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FORMAT CONVERTER, MATRIX MIXER, DSP

Prodigy MP

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Prodigy MP

Prodigy MP from DirectOut Technologies is a modularly equipable mainframe suitable for a wide range of audio engineering tasks. With audio conversion and FPGA-based DSP functionality, it covers live applications, fixed installations and broadcast.

Copy and measurements: Anselm Goertz | Images: Anselm Goertz

In recent years, the MADI format, which is mainly used in broadcast, has been joined by new IP-based formats such as Dante, Ravenna or SoundGrid, which - thanks to their rather simple handling – have quickly spread and established themselves in fixed installations and in mobile applications. At the same time, it is not uncommon for several formats to be used in one project. In such cases, it is a matter of bringing everything together, synchronizing it and, above all, avoiding interference, losses and dropouts. Generally, a distinction has to be made between pure audio data protocols – such as MADI, ADAT or even AES/EBU – and IP-based audio networks. Dante, Ravenna and AES67 operate on OSI layer 3 and rely on IP addresses. SoundGrid operates on OSI layer 2 and relies on MAC addresses. AVB actually operates on layer 3, but uses layer 2 protocols for synchronization, it is therefore a mix of layer 2 and 3. One challenge is to bring these mixed systems' quite different formats together. And, additionally, also integrate analogue signals that also want to find their way into the audio system.

A typical example for such applications such as those described above are festivals where bands bring their own consoles, resulting in a situation where Dante, SoundGrid and perhaps a playout via MADI for the OB van have to be managed at the FOH location. Large stages additionally require a lot of playouts for the main system, delay lines, side and front fills, subwoofers, and more. This is where the Prodigy MP comes in, as it is born out of these very diverse requirements. In the MP version, in addition to its function as an audio converter, the Prodigy is also equipped with powerful FPGA-based DSP functionality. This review will take a closer look at how far the device lives up to its promising Prodigy name.

The Prodigy's manufacturer DirectOut GmbH is a com-

pany founded in Mittweida, Germany, in 2008 by several audio engineers with the goal of developing and marketing high-quality and high-performance audio format converters, interfaces and signal distributors for broadcast, studio, live and installation applications. Specifically, the company's portfolio includes a whole range of 19" devices that process, convert or forward audio signals for or from MADI systems.

Modular design

In order to be as flexible as possible and not to have to sell unnecessary hardware to the user, the Prodigy is designed as a mainframe with slots for individual configuration. The software also applies this concept, offering different bundles with DSP functions, special tools and services. As is often the case with audio devices, a look at the Prodigy's rear is more revealing than a look at the front. Here one finds four slots for analogue inputs and outputs or digital AES3 format signals, two slots for audio network interfaces and two for MADI connections. Three network ports are permanently integrated in the mainframe as are a MIDI interface, two word clock inputs and outputs as well as a GPIO connector and two power supplies that operate redundantly. The latter is an important aspect when it comes to operational reliability, as is the option of operating two devices in mirror mode or configuring the inputs with automatic failover.

The two network slots can be mixed as desired or even equipped identically, as can the four slots for analogue or AES3 cards. The MADI slots 1 and 2, which are smaller in terms of dimensions, can be equipped with connectors for BNC cables, with an SC multi-mode or single-mode card for optical connections or with a connector for an SFP transceiver.

Analogue and AES3 module

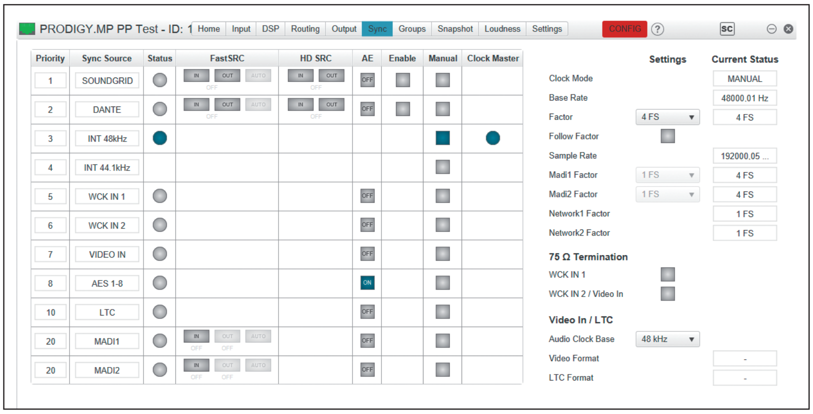
Four versions and three combination modules are available for the analogue modules. Equipped with eight analogue inputs each, the AN8.I module is available for line level while the MIC8.LINE.I and MIC8.HD.I modules with preamp inputs are available for microphone and line level signals. The HD version offers a 10 dB better signal-to-noise ratio at higher gain and, with an equivalent input noise (EIN) of -128 dBu, corresponds to the standard for high-quality mic preamps. The AN8.O is available for eight analogue outputs with levels that can also be set by jumper, in this case of 15, 18 or 24 dBu. The three combi cards correspond to the three analogue input modules extended by the eight-channel output module. All connections feature 25-pin D-sub connectors and comply with the AES59 Tascam standard.

For signals in digital AES3 format, two cards are available for the Prodigy, each featuring four two-channel inputs and outputs: the AES4.IO and the AES4.SRC.IO, which is additionally equipped with sample rate converters. Here again, the connection is also made via 25-pin D-sub in line with the AES59 Tascam standard. Breakout boxes in a 19" format and with 1 RU are available for users that want to connect analogue or digital signals individually via XLR connectors. The five available options offer either 16 analogue inputs or outputs, a combination of eight inputs and outputs each, or – for digital signals – eight AES3 inputs and outputs each with XLR connectors for 110 Ω cabling or 16 inputs and outputs each with 75 Ω BNC sockets in AES-iD format. The latter enables connections over distances of up to 1000 m provided suitable coax cables are in place.

Networks and synchronisation

When it comes to networks, DirectOut offers cards for Dante, Ravenna and SoundGrid. All three versions are each

Prodigy MP mainframe Dual power supply, two MADI interfaces, three Ethernet ports, MIDI and clock connections, analogue and AES3 IO cards with D-Sub25 connectors as well as network IOs for Dante and SoundGrid



SRC Overview of the available clock sources and sample rate converters (Fig. 1)

equipped with three network ports (2 × RJ45 and 1 × SFP) and an internal switch, enabling redundant networks for audio signal transmission. The remaining third port can then also be used for the device management port's Ethernet connection if required.

Normally, all devices in an audio network work synchronously. In this network, one of the devices or a special clock generator sets the clock as the leader and all other devices follow this clock (followers). This always works well when the entire audio transmission is based on one network format. If, however, several formats are used, for example if one console at the FOH operates with Dante and another with Ravenna, then the leader/follower principle no longer works. The data streams, even if they use the same sample rate, must then be adjusted with sample rate converters (SRC).

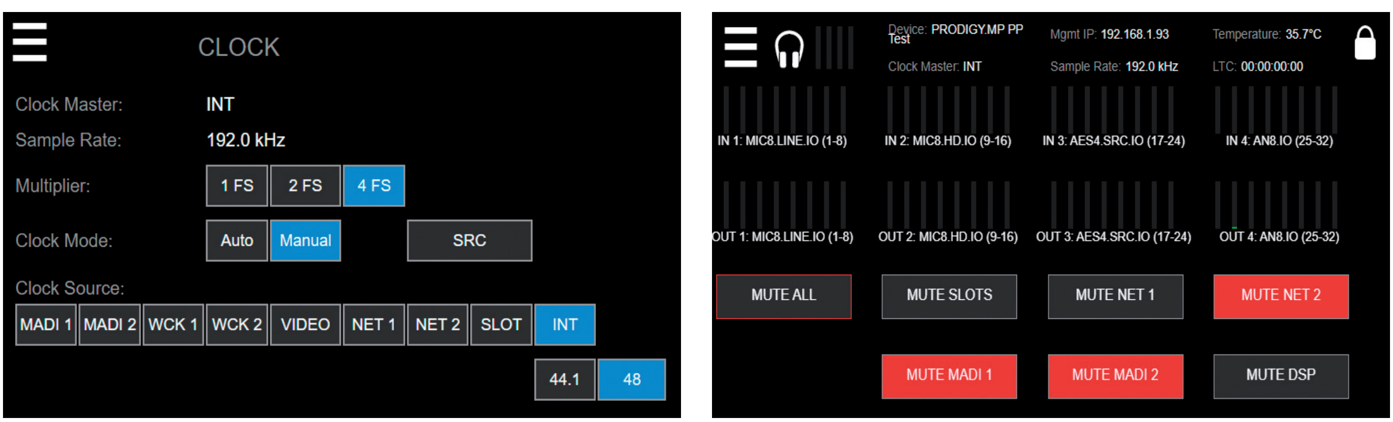
For this purpose, DirectOut offers two SRC versions: the FastSRC with 0.15 ms latency and the HD SRC with 1 ms latency. The FastSRC is an integral part of the Prodigy MP for all signal inputs and outputs via the network slots and the MADI interface. The HD SRC, on the other hand, requires considerably more computing power and is only optionally available on network cards with the suffix "SRC" in the type designation. There is no HD SRC available for MADI signals. In addition to the latency, the difference between the SRCs can also be identified in terms of the audio quality: with FastSRC, small compromises regarding the audio quality have to be made in favour of short latency and low computing power requirements. For the AES3 card, the SRC is only available in the inputs and works independently for the four inputs, so that four signals that are non-synchronous and have different sample rates can be adjusted simultaneously to the Prodigy's internal clock.

Fig. 1 shows an overview of the Prodigy's possible synchronisations to external or internal clock sources with priority and availability. When it comes to the two Dante and SoundGrid network cards, one can see that the HD SRC option could also be selected, in other words, the cards installed there are equipped with the SRC option.

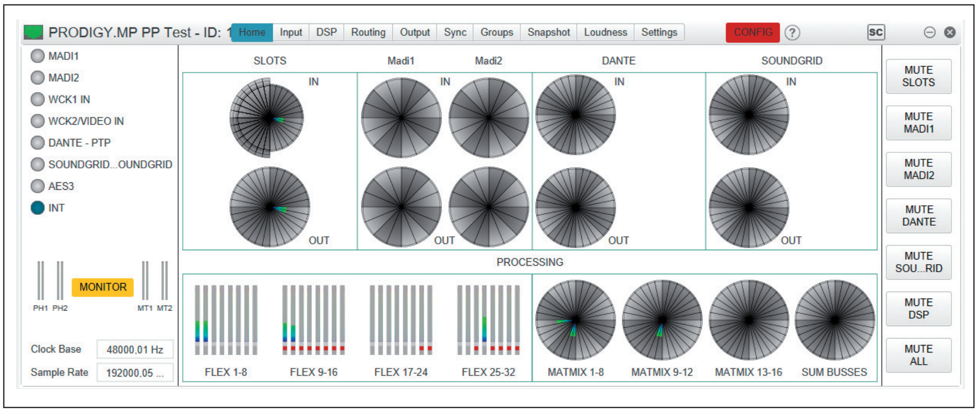
Operation

As the Prodigy's range of functions and the number of possible inputs and outputs (416 and 420 respectively) are enormous, good operating software for configuration and a user interface are of considerable importance. Three options are currently available for this purpose.

Simple functions such as level monitoring, mute, gain or a clock overview as well as the information on the device's status can be carried out directly via the large, high-res colour



Display on the device or in the browser of the pages for the clock setting (left) and an overview of all of the IO slots' levels and mutes (Fig. 2)



Ingenious overview of the level ratios of all (!) inputs and outputs, the flex channels and the matrix mixers in the Prodigy (Fig. 3)

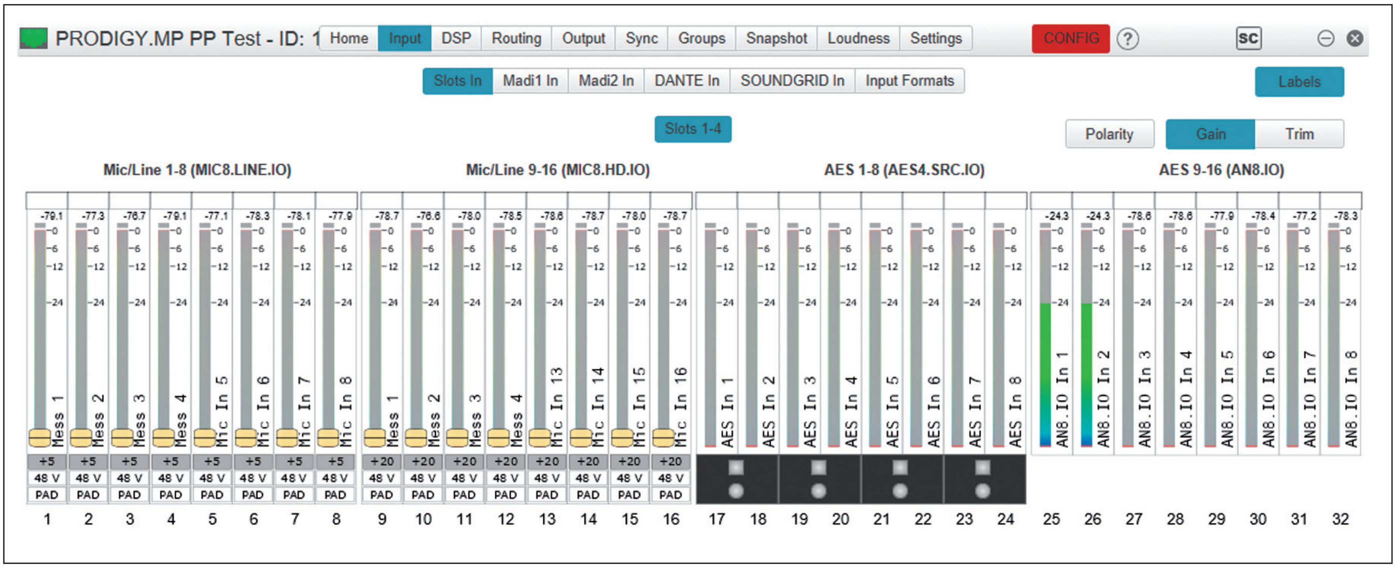
touch display (see Fig. 2). All possible operations can also be accessed on a website via a standard browser. All users have to do is to be on the same network and enter the Prodigy's IP address. Any tablet or smartphone can be used as a remote control for simple functions and monitoring.

Software

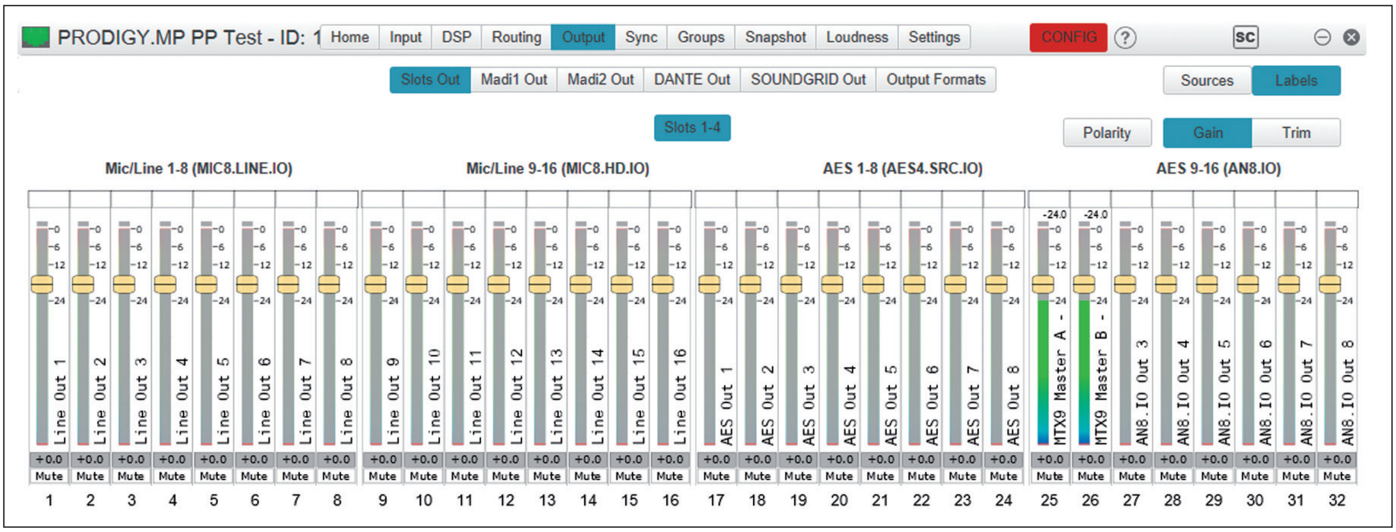
Complete operation and configuration of the Prodigy, however, requires a PC and the Globcon software, which is available for Windows, MacOS and soon also for Linux. The software by LGSF Engineering GmbH from Hanover, Germany, was developed in close cooperation with DirectOut, but is not limited exclusively to their devices. Rather, it is understood as a universal tool for operating media technology devices. The Globcon project is financed by the participating manufacturers and is available to users free of charge.

The Globcon software distinguishes between a Show and Config mode, whereby the latter only allows users to define function blocks or inserts in the channels. When starting the software, the Prodigy first appears with a small overview screen showing either the levels of all inputs and outputs or all DSP channels. If one then opens the device in the software, the home screen in Fig. 3 appears, presenting a more comprehensive display of all important states. The levels are very cleverly displayed using circular graphics, where all of a slot's channels are respectively summarised. In Fig. 3, the Prodigy was operated with a sample rate of 192 kHz, thereby reducing the number of channels in the network inputs and outputs. In addition to the inputs and outputs, the screen also displays the level ratios of the flex channels, the matrix mixers and the summing buses. On the far right of the

Breakout connected via solid analogue or digital patch cables



Inputs The 32 inputs of slots 1 to 4 with level meters and gain settings for the inputs with preamps (Fig. 4)



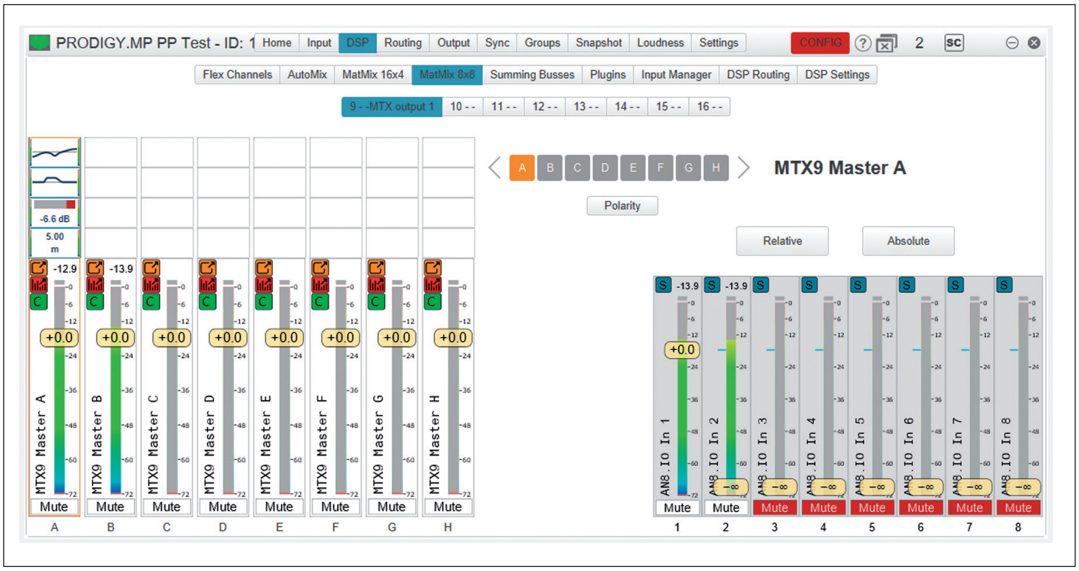
Outputs The 32 outputs of slots 1 to 4. The three analogue modules are equipped with the output option AN8.O (Fig. 5)

image, one can also identify the mute switches, which can be used to mute entire slots or everything together.

Inputs and outputs

Windows with 32 channels each are available to operate the inputs and outputs available in the hardware's slots or networks. In total, 64 inputs and outputs each for Madi M1 and M2 and 128 each for networks N1 and N2 are available. In addition, there are eight inputs and outputs each in the S1 to S4 slots. This brings the total to 416 inputs and 416 outputs. As for the outputs, one must add the two stereo headphone outputs PH1 and PH2 on the front of the unit. Polarity, gain, trim and mute can be set for all inputs and outputs. The channel labels show either the internal assignment (source or sink) or a name assigned by the user. For the analogue cards with preamps, gain, PAD and 48 V phantom power are also set here.

All inputs and outputs as well as the internal channels of the summing buses, matrix mixers and flex channels can be linked via an all-encompassing matrix. The matrix is structured in such a way that users can select the sources and sinks according to affiliation first – for example the matrix mixers' outputs as sources and network 1 as physical outputs – and then set the links in the now clear frame. If all of the matrix's ways were always visible, the overview would suffer as a result.



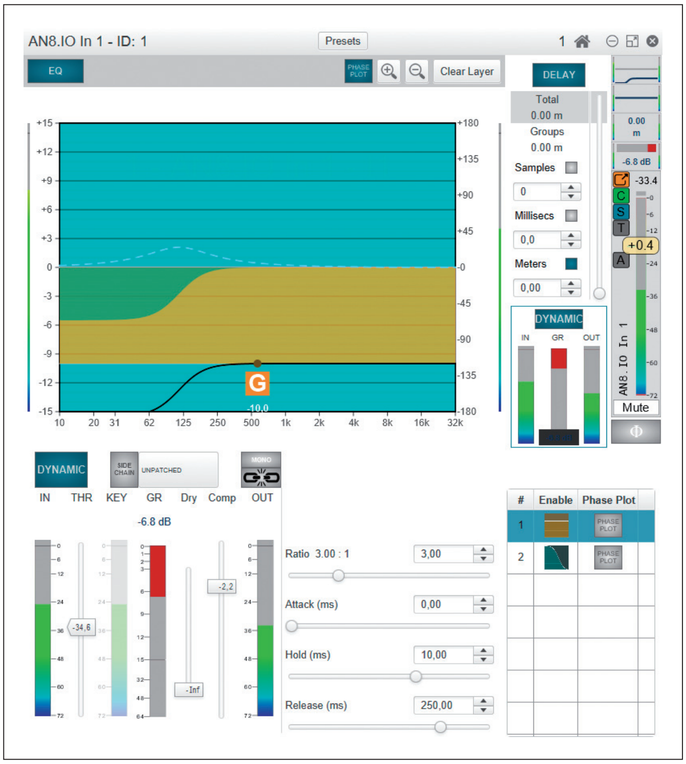
One of 16 matrix mixers Four processing modules can be used in each of the outputs (left) (Fig. 6)

Matrix mixer and flex channels

The Prodigy's matrix mixers allow users to mix 16 or eight inputs to four or eight outputs respectively, or internally to additional DSP channels such as a flex channel. In total, there are eight 16 × 4 and 8 × 8 matrix mixers each. Fig. 7 shows an example of an 8 × 8 mixer where an individual mix can be created from inputs 1 to 8 for each of the eight outputs A to H. Each output features four slots with a selection of signal processing functions. This can be a simple delay, an IIR filter bank, an FIR filter bank or even a dynamics processor. All control elements can also be removed or copied individually and arranged again on a separate surface to suit the application in question.

A specialty of the Prodigy's signal processing are the 32 flex channels, which can be individually defined as an input or output channel strip. Every flex channel also includes four slots with a selection of signal processing functions. Users must note the direction of the signal flow, which is top-down for input channel strips, in other words, processing first with the fader at the end. It is exactly the other way round when it comes to the output channel strips: down-top with the processing located behind the fader.

Individual channel strips with all details of signal processing can also be displayed completely using the C (channel strip) button. Fig. 7 shows an example of this, where filter curves, level displays and faders are clearly under control. The parameters can be set via the faders, graphically or directly with numerical values.



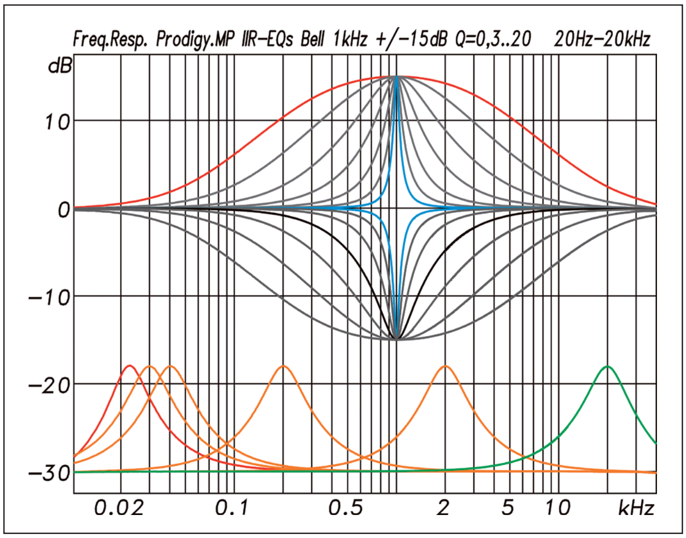
Detailed view of a flex channel with all parameters for the filters, dynamics and delay (Fig. 7)

Filters

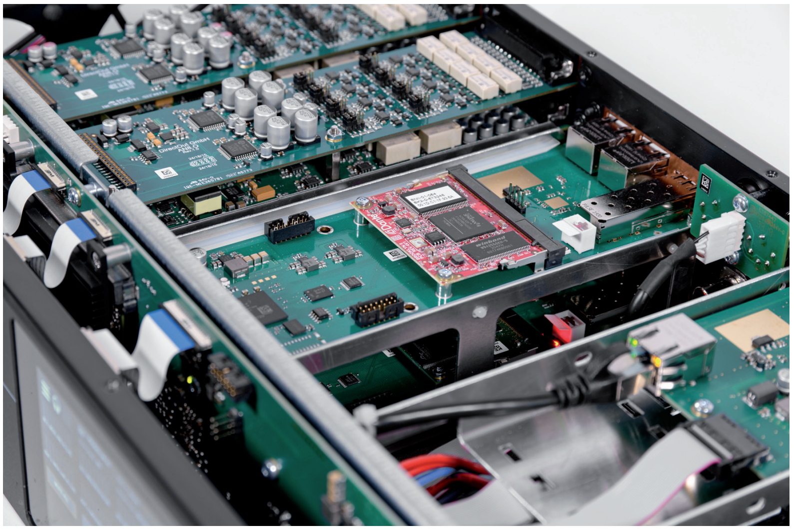
Due to the large number of possible filter functions, it is, of course, not possible to display them all. Filters can be adjusted via two processing modules, the IIR and the FIR equaliser. The IIR filters are classic digital filters with a so-called BiQuad structure, which simply put behave like analogue filters. The FIR filters are used here in a somewhat modified and extended form. The structure, which we will not discuss in detail here, is somewhat more complex than that of conventional FIR filters and allows a filter curve with interpolation points on a logarithmic frequency axis. This means that these filters can be used much more effectively at low frequencies than conventional FIR filters, whose interpolation points lie above the linear frequency axis. However, one has to accept a small disadvantage: this type of FIR filter does not offer the possibility of adjusting amplitude and phase independently of each other. It is therefore not possible to create linear-phase filters or to equalise loudspeakers in phase.

The IIR EQ module offers the usual range of functions with 1st and 2nd order high and low pass filters, Bell filters, all-pass filters as well as low and high shelf filters with variable quality. The curves shown in Fig. 8 are an example of the Bell filters. The centre frequency can be varied from 20 Hz to 20 kHz and the quality from 0.3 to 20 with a maximum gain of ±15 dB. The lower part of the graph shows Bell filters with a constant quality of 2.0 and a fixed gain of +12 dB for centre frequencies from 20 Hz to 20 kHz.

Thanks to the high sampling rate, the Bell curve's usual compression at high frequencies does not occur. The green curve at 20 kHz has exactly the same shape as the other curves at lower frequencies. There is only a slight deviation at low frequencies below 30 Hz, where the 20 Hz Bell filter's



Some exemplary curves of the IIR-EQ for frequencies from 20 Hz to 20 kHz, a quality between 0.3 and 20 as well as gain values of a maximum of ±15 dB. The filter curves are not compressed even at high frequencies and behave exactly like an analogue filter (Fig. 8)

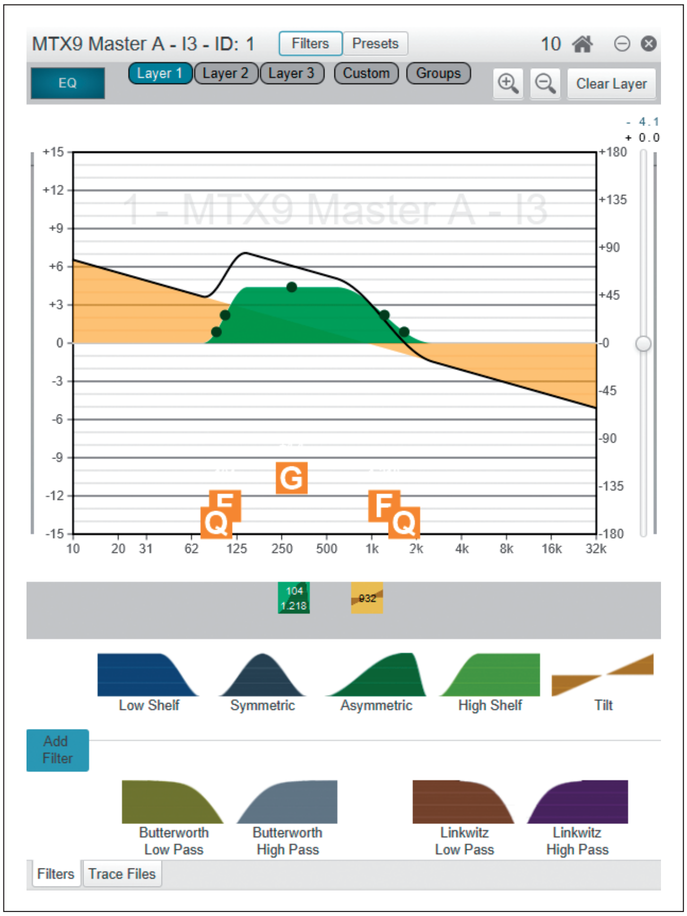


Dante IO card with the Brooklyn II module in the front of the picture, behind it two analogue IOs with optional AN8.O output board

centre frequency has shifted slightly upwards. Additionally, there are some more unusual filter types on the second page of the filter selection for the IIR EQ. Here, users will find a bandpass and a notch filter as well as RIAA production and playback filters. The latter are used to lower the low frequencies before cutting a pressing for vinyl records and to raise them again afterwards during playback. The level difference between 20 Hz and 20 kHz is 40 dB.

The FIR EQ module, which enables asymmetrical filters, shelf filters with extreme quality values as well as Linkwitz-Riley high and low passes with up to 48 dB/Oct slopes, is even more interesting. However, as already mentioned, these are not linear-phase due to the special FIR structure. Another rather unusual filter type is the tilt filter, which generates a linearly rising or falling straight line as a filter function above the logarithmically scaled frequency axis. The slope is set via a quality value and ranges from -1 dB/Oct to +1 dB/Oct. Occasionally referred to as "sound scales", they are well suited for adjusting recordings that sound too sharp or too thin or for tuning sound systems that are "too linear".

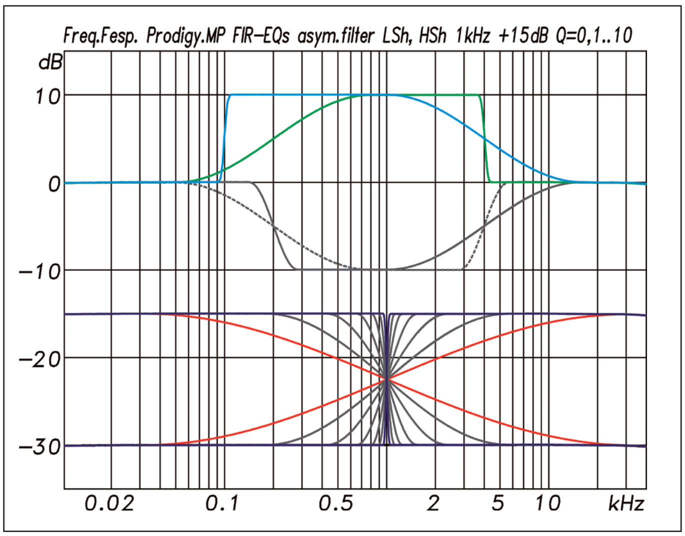
For the FIR EQs, three independent layers with the filter types mentioned above can be set and saved as pre-sets and transferred using copy and paste. The three layers are well suited for comparing filter settings or also for making them available to different users independently of each other. This way, the three layers in the PA's playout way can be used for the calibration of the system, for an adjustment to the room acoustics and for the band technician, without one having to interfere with the other's filters.



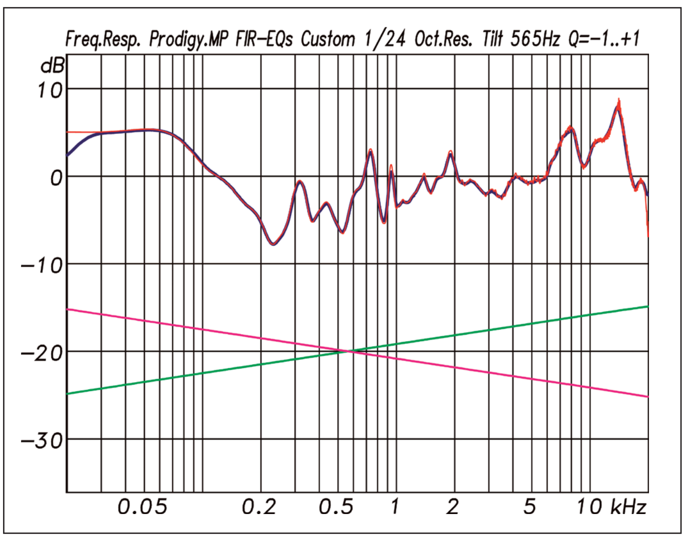
FIR-EQ module with asymmetric filters and special tilt filters, among others (Fig. 9)

The Prodigy has another special function for calibrating a loudspeaker system. With the fourth so-called "custom" layer, filters defined via simple table values can be implemented or derived directly from a measurement. Fig. 11 shows a filter curve which was imported from a measuring programme using a CSV table and which is displayed correctly in all details. Here again, only the amplitude can be specified and no linear-phase or filters that operate independently are possible. In this example, the filter function was imported via a CSV table with 244 interpolation points with 1/24 Oct spacing from 20 Hz to 20 kHz.

In combination with a Smaart measuring system and a plug-in, users can go one step further by taking the measurements and using these to directly create an equalising filter via the invert function. If necessary, the measured curve can be smoothed and its dynamics can also be compressed. The lower and upper cut-off frequencies for the inversion can also be freely defined so as not to create extreme compensations. The measuring microphones required for Smaart can be connected via the Prodigy's microphone inputs, just as the measuring signal from Smaart can be sent to the speakers via the Prodigy's output paths. For this, the measuring computer with the software is connected to the Prodigy via Dante using a DVS (Dante Virtual Soundcard).



Some exemplary filter options with asymmetric functions (top) and with shelf filters (bottom), whose quality can be defined from 0.1 to 10 (Fig. 10)

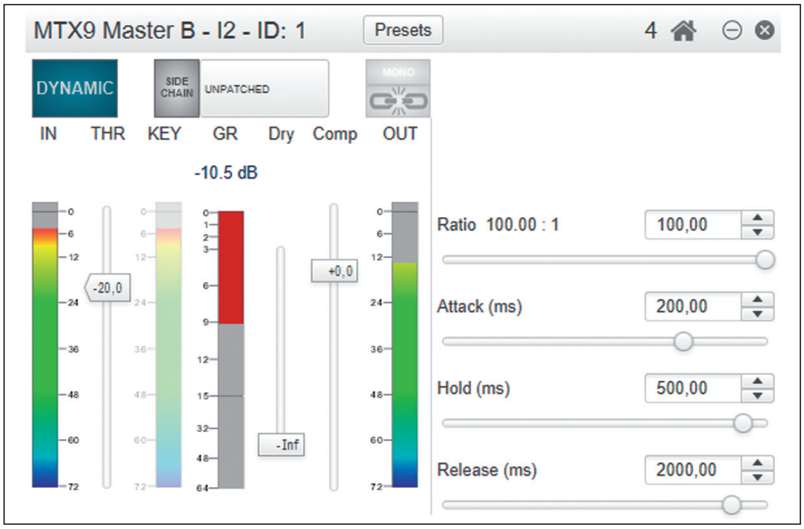


Custom FIR filters (top), which can be created using a simple table in CSV format or also directly using measurement data. In the lower part of the graph, a measurement of the tilt filters, whose slope can be adjusted via a quality value between -1 dB/Oct and +1 dB/Oct (Fig. 11)

Dynamic processors

Another processing module contains the dynamic function, which can be used in stereo or mono mode. A side-chain is available for controlling the compressor, which, for example, can be supplied with a desired pre-filtering from a flex channel. For the compressor itself, the usual threshold and ratio parameters as well as the three time constants for attack, hold and release can be set in very wide ranges (Fig. 12). A fader labelled "dry" also allows the compressor's signal to be mixed back with the original signal.

Fig. 13 shows an exemplary measurement of the compressor function for the parameters set in Fig. 12. A 1 kHz sine signal with a level jump of +20 dB between 1 s and 3 s was used as the input signal (blue curve). The compressor's red output signal shows the function where all set values can be easily understood.



Parameterisation of the dynamic functions (Fig. 12)

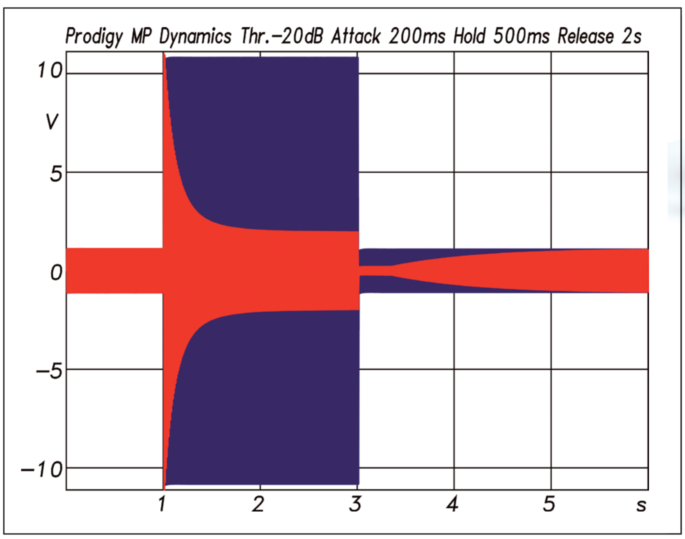
Further functions: loudness measurement, auto mix

The Prodigy's wide range of functions does not allow us to explain or examine everything in detail here. But to name a few, the device also features further functions such as automatic mixer, loudness measurement, the possibility to integrate external plug-ins as well as many other small specialities that can be important for certain tasks. For users who want to get a good overview of the device, we recommend the freely available Globcon software, where a virtual Prodigy with all hardware and software options is fully mapped and can also be configured. A series of well-made videos, including webinars and tutorials, on DirectOut's homepage also provides an overview of various applications and explains the Prodigy's functions.

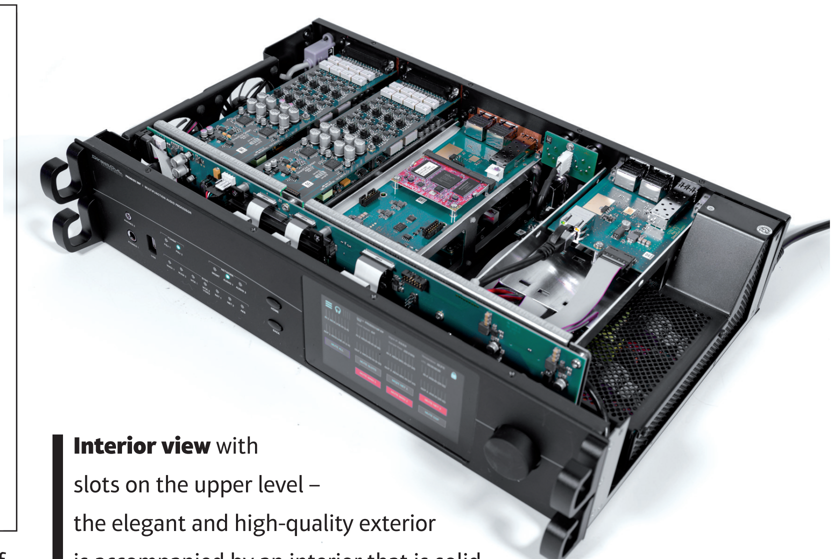
Costs

The price structure is as complex as the device itself, with a lot of hardware options and various software packages as well as individual software licences.

The basic Prodigy MP without network modules or slot cards with the Advanced software package can be found in



Reaction of the compressor to a sinus burst with a level jump of +20 dB between 1 s and 3 s (Fig. 13)



Interior view with slots on the upper level – the elegant and high-quality exterior is accompanied by an interior that is solid and tidy despite the large amount of technology

Remote Operation can also be carried out via a browser on any device in the same way as it can be on the device itself

Slot cards		
AN8.I	8 × Line In	350.00
AN8.O	8 × Line Out	350.00
AN8.IO	8 × Line In/Out	600.,00
MIC8.LINE.I	8 × Mic/Line In	700.00
MIC8.LINE.IO	8 × Mic/Line In/Out	925.00
MIC8.HD.I	8 × Mic/Line In	1,150,00
MIC8.HD.IO	8 × Mic/Line In/Out	1,375.00
AES4.IO	4 × AES3 In/Out	350.00
AES4.SRC.IO	4 × AES3 In/Out	700.00
Network cards		
RAV.IO	Ravenna	1,375.00
RAV.SRC.IO	Ravenna with HD-SRC	2,125.00
DANTE.IO	Dante	1,200.00
DANTE.SRC.IO	Dante with HD-SRC	1,950.00
SG.IO	SoundGrid	1,000.00
SG.SRC.IO	SoundGrid with HD-SRC	1,750.00
Madi modules		
BNC.IO		225.00
SC.IO MM	Multi mode	225.00
SC.IO SM	Single mode	400.00
SFP.IO		175.00

The reviewed device without breakout boxes and without patch cables with the Advanced package is available for a total price of 14,300 € net.

the price list for €6,500 net. With the simpler Essential package, which does not include DSP functionality, the Prodigy is already available for as little as €4,500. At €8,100, users will get the ultimate Unlimited package, which includes all current and future DSP resources and tools.

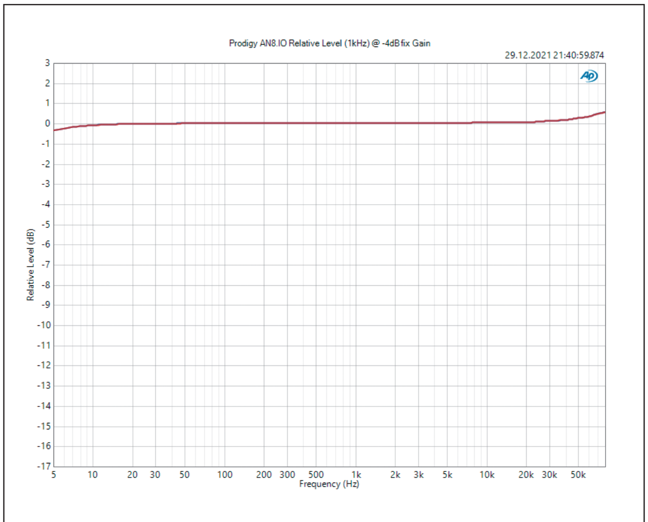
Summary: Powerful!

Anyone who has taken the time to read this review in its entirety up to this point will inevitably understand the conclusion: DirectOut's Prodigy MP lives up to its name and is capable of virtually every signal processing function in modern audio technology. Like a spider in an audio system's web, the Prodigy can record all kinds of analogue and digital signals as well as all popular audio network formats with up to 416 channels, synchronise them to each other, process them and distribute them again via just as many channels and formats. The extremely powerful FPGAs's available DSP functions leave nothing to be desired and predestine the Prodigy for use in large PA systems, in complex fixed installations and also in studios. Despite the large range of functions, however, the Globcon software succeeds in presenting the unit clearly and making it fully operable and usable after a certain period of familiarisation.

The Prodigy's hardware meets the highest demands and is absolutely top class in terms of workmanship and material quality – which is then also appropriately reflected in the price. All in all, the Prodigy is probably one of the most powerful 19" units on the audio market. Thanks to its flexible hardware and software, it can be used for numerous tasks and thanks to its modular design, it is also futureproof.

Prodigy MP: measurements of analogue IOs

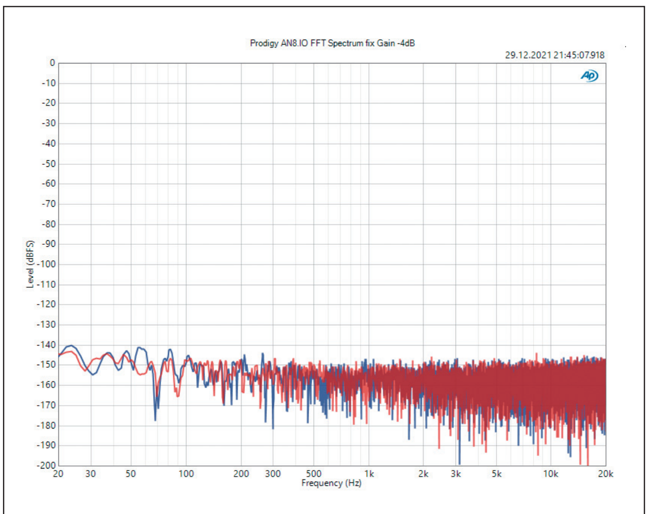
All acoustic signals that are to be digitally processed, stored or transmitted inevitably come into contact with ADCs and DACs. They are therefore of decisive importance for the audio signal chain's final quality. Technically, the ADC side is particularly demanding, as the analogue signals present there can be very different in level. These range from microphone signals in the mV range to a line level with several V. In order to adapt these to the ADCs, a preamp is required that can process the signals appropriately with adjustable amplification. The gain range covered is usually 50 dB or more. Noise, distortion as well as the frequency and phase response are therefore important characteristics of the analogue input and output modules. Keeping these aspects in mind, the Prodigy's three analogue input modules and the output module were examined using the Audio Precision APx555 measuring system. In order to keep the scope of the measurements somewhat in check, not all results are listed here, but rather only those with the most important key values for assessing the modules.



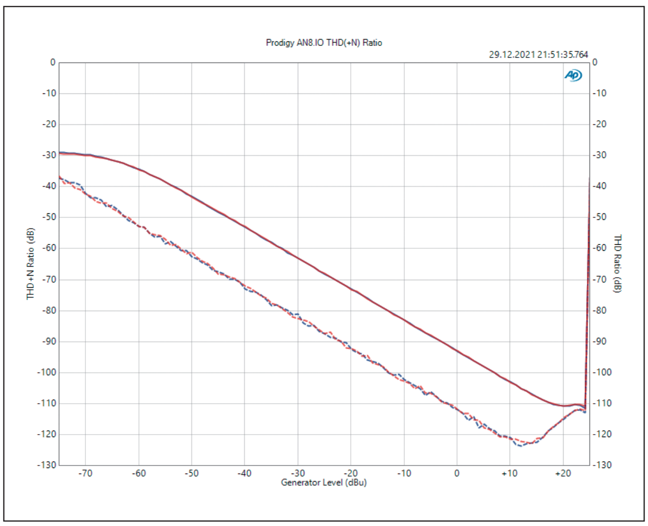
The AN8.I's frequency response with a fixed gain of -4 dB referred to +20 dBu (measurement at 192 kHz sample rate with a plot relative to the value at 1 kHz) (Fig. 14)

AN8.I

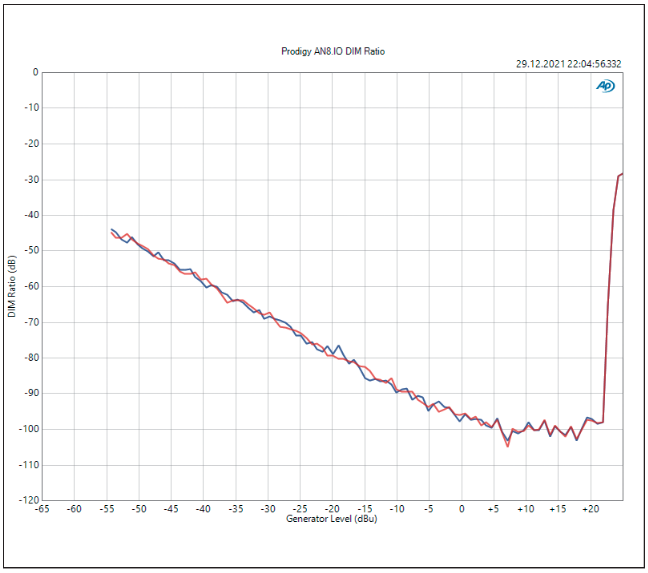
The simplest input module is the AN8.I with eight analogue inputs exclusively for line-level signals. Input sensitivity related to 0 dBfs full level can be set via jumpers on the module: +15, +18, +24dBu. Full scale (0 dBfs) on the digital side is achieved at +24.3 dBu input level, allowing users to connect all common sources without the risk of overload. As expected, the frequency response shown in Fig. 14 is perfect and ranges from 5 Hz to above 50 kHz without significant deviations. The noise level on the digital side without a signal at the input is -117 dBfs unweighted and -120 dBfs with A-weighting. Monof-



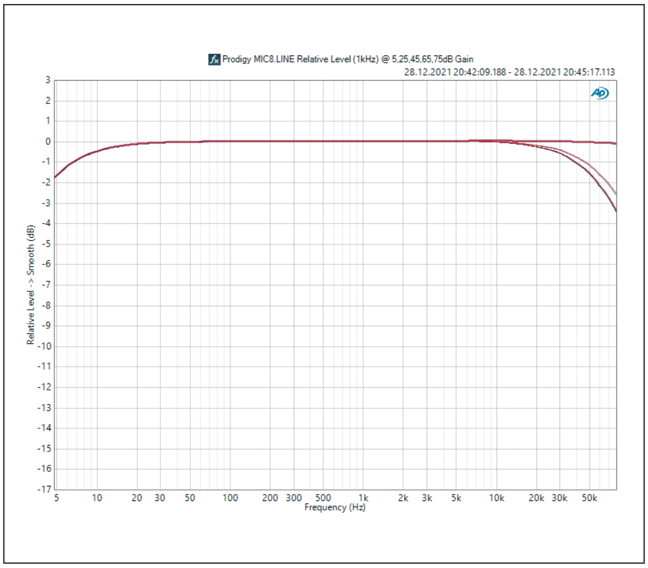
The AN8.I's noise level with a sum level (20-20k) of 117(120) dBfs(A) (Fig. 15)



The AN8.I's THD () and THD+N (- - -) as a function of the level up to a maximum of +25 dBu (x-axis in dBu, Fig. 16)



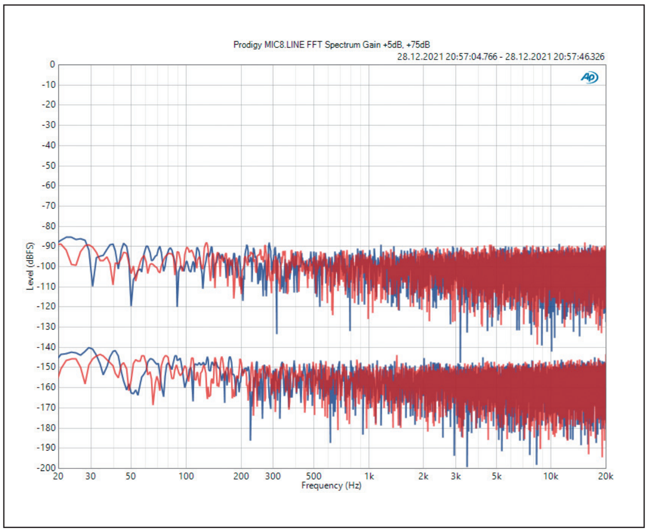
The AN8.I's transient intermodulation distortions (DIM100) for input levels from -65 dBu to +25 dBu (Fig. 17)



The MIC8.LINE.I's frequency response for gain values from +5 to +75 dB referred to +20 dBu (measurement at sample rate 192 kHz with a plot relative to the value at 1 kHz) (Fig. 18)

requency components are not visible in the interference spectrum (Fig. 15). The THD and THD+N distortion measurements measured at 1 kHz are also excellent, with a THD minimum of less than -120 dB and 111 dB directly before the clip limit.

Another criterion on the subject of distortion is the measurement of transient intermodulation distortion (DIM100), which is said to be of particular relevance for the sound characteristics. Here, instead of a steady sine wave, a mixed signal of a steep-edged square wave of 3.15 kHz and a sine wave of 15 kHz is measured. The square's steep curves are a challenge for the circuit and can lead to a short-term overload and thus to distortions, which are detected with this measurement. Fig. 17 shows the measurement result for the AN8.I's two channels with outstandingly good values of -100 dB.



The MIC8.LINE's noise level at minimum gain of +5 dB and at maximum gain of +75 dB with a sum level (20-20k) of 117(120) dBfs(A) and -61(63) dBfs(A). This results in an EIN value of -116 dBu (Fig. 19)

MIC8.LINE.I

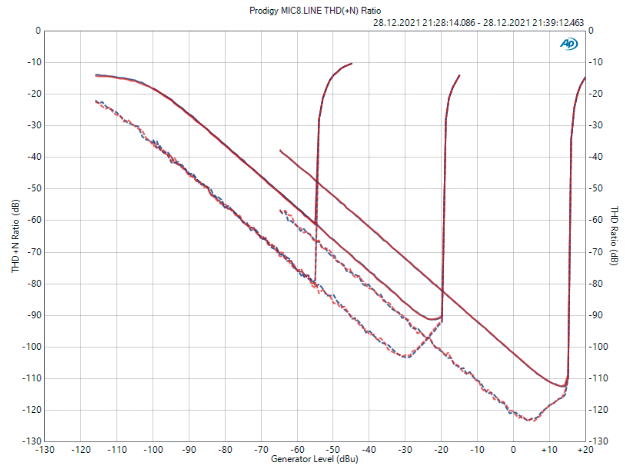
The MIC8.LINE.I card with preamps is somewhat more complex in design, the preamps' gain can be adjusted between +5 and +75 dB using the software. In addition, there is a PAD that attenuates the level by 9 dB. The reference value here is also +20 dBu, in other words, without PAD, full scale at the input is achieved at +15 dBu with a minimum gain of +5 dB. With PAD, the value is +24 dBu. The gain setting on this card is simplified in four analogue steps for the gain ranges from 5...16 dB, from 17...24 dB, from 25...34 dB and from 35...75 dB. Within these ranges and above +35 dB, the gain change takes place on the digital level, so that the available S/N no longer improves.

If we first have a look at the frequency response in Fig. 18, the curve in the relevant frequency range is again perfectly straight. Only at higher amplifications do the curves begin to drop slightly far beyond 20 kHz.

Gain [dB]	S/N (lin) [dB]	S/N (A) [dB]
+5	117	120
+15	107	110
+25	110	112
+35	101	103
+45	91	93
+55	81	83
+65	71	73
+75	61	63

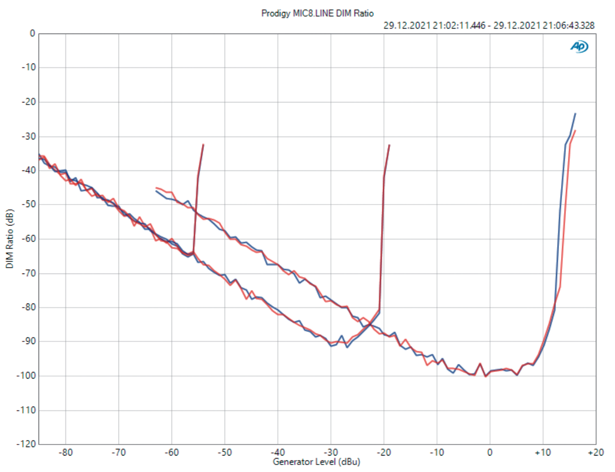
The MIC8.LINe.I's S/N values depending on the set gain value. Above +35 dB, the S/N decreases synchronously with the gain's increase, which, here, is only converted on the digital level (Tab. 01)

For a minimum gain of +5 dB, the measurement of the noise level and the corresponding FFT spectrum delivers values of -117(-120)dBfs(A). These are comparable to the AN8.I card's results. For the gain range from +5 to +35 dB, one can see the influence of the analogue gain steps. Above +35 dB, the S/N then decreases synchronously with the increase in gain, since here amplification only takes place on the digital level. This is exactly what can be seen in the distortion measurements for gain values of +5, +40 and +75 dB in Fig. 20 and Fig. 21. The



The MIC8.LINE's THD () and THD+N (- - -) curve for gain values of +5, +40 and +75 dB (f.r.t.l.) depending on the level (x-axis in dBu) (Fig. 20)

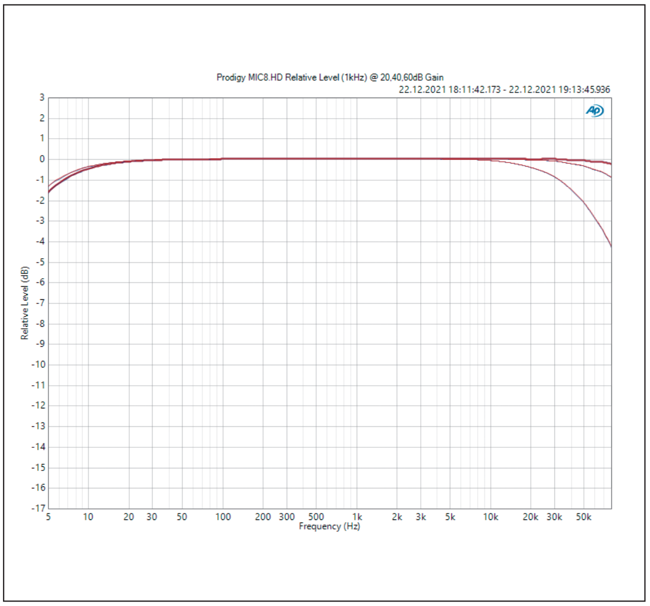
curves at +40 dB and +75 dB are initially congruent and differ only by the digital level shift of 35 dB, so that the clip limit is once at -20 dBu and once at -55 dBu. Regardless of this, the distortion values at +5 dB gain correspond to the AN8.I's very good values. At +40 dB gain on the highest analogue gain stage, values of below -100 dB and -91 dB are also still achieved at the clip limit. The same applies to the measured values for the transient intermodulation distortions in Fig. 21.



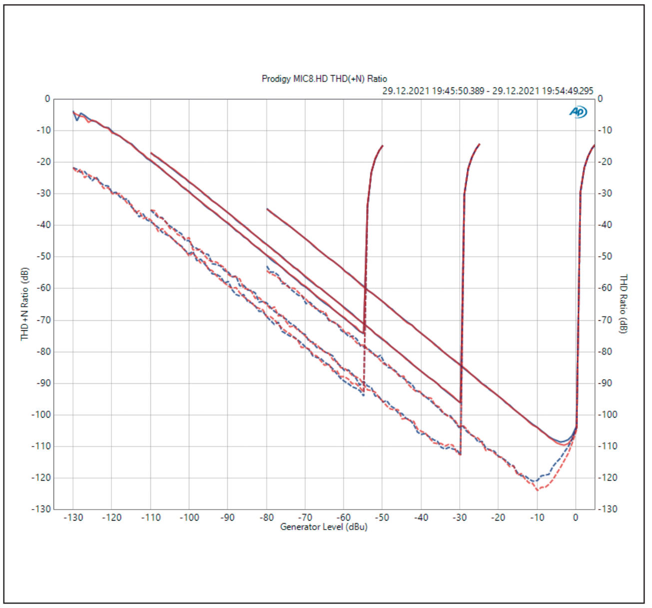
The MIC8.LINE's transient intermodulation distortions (DIM100) for gain values of +5, +40 and +75 dB (f.r.t.l.) (Fig. 21)

MIC8.HD.I

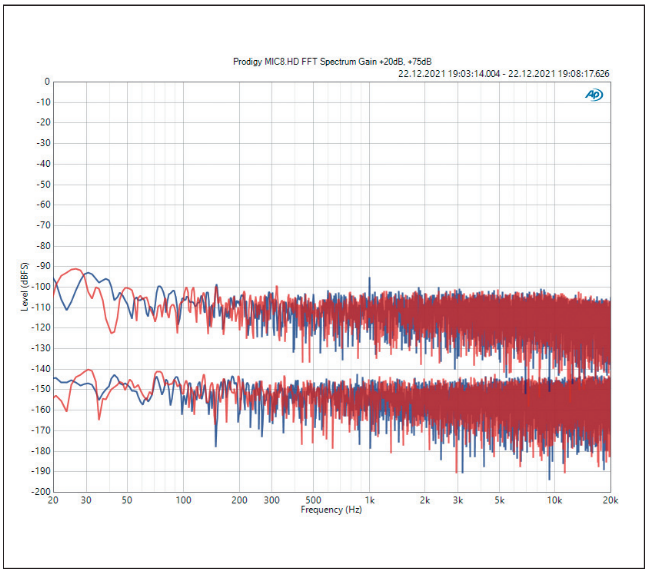
In comparison to the MIC8.LINE.I, the MIC8.HD.I card works with a completely different circuit for the preamp. Here, an integrated preamp with digital gain adjustment is used, whose gain can be set between +20 and +75 dB using the software. In addition, PAD is included that attenuates the level by 30 dB. The reference value is again +20 dBu, in other words, without PAD, full scale is achieved at +0 dBu at the input with a minimum gain of +20 dB. With PAD, the value is +30 dBu, meaning that all sources from a quiet microphone to a high-level line output with +27 dBu can be connected. The HD card's frequency responses display the already familiar behaviour with a slight level drop above 20 kHz at high gain values.



The MIC8.HD.I's frequency response for gain values from +20 to +60 dB referred to +20 dBu (measurement at 192 kHz sample rate with a plot relative to the value at 1 kHz) (Fig. 22)



The MIC8.HD's THD () and THD+N (- - -) curve for gain values of +20, +50 and +75 dB (f.r.t.l.) depending on the level (x-axis in dBfs) (Fig. 24)



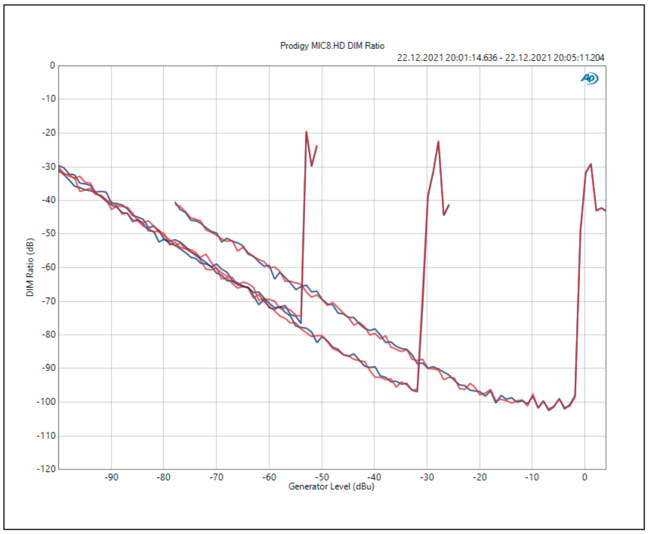
The MIC8.HD's noise level at minimum gain of +20 dB and at maximum gain of +75 dB with a sum level (20-20k) of 114(117) dBfs(A) and -73(75) dBfs(A). This results in an EIN value of -128 dBu (Fig. 23)

Gain [dB]	S/N (lin) [dB]	S/N (A) [dB]
20	114	-117
30	110	-112
40	103	-106
50	95	-98
60	86	-89
70	77	-80
75	73	-75

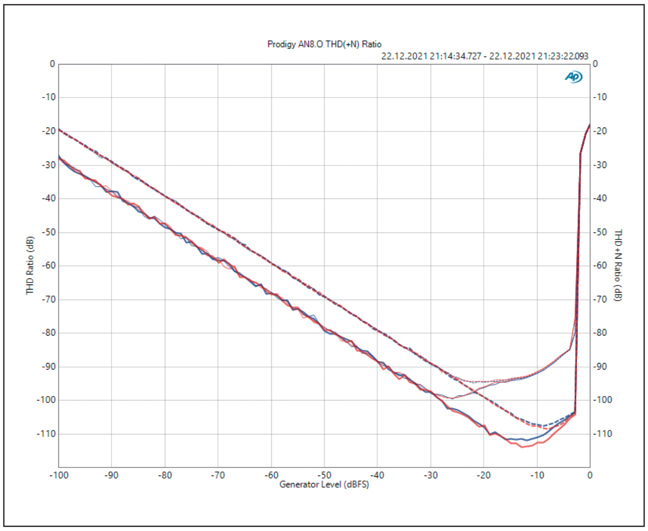
The MIC8.HD.I's S/N values depending on the set gain value (Tab. 02)

In comparison to the MIC8.LINE, the S/N behaves differently due to the different circuit concept with a purely analogue gain setting over the entire range. The result is that better S/N values are achieved at high gains. At a gain of 75 dB, the MIC8. HD achieves an S/N of 73(75) dB(A), which corresponds to an EIN value of -128 dBu and is thus 12 dB better than the MIC8. LINE's result.

Accordingly, the distortion value curves in Fig. 24 and Fig. 25 also run somewhat differently depending on the input level. A higher gain here not only means reaching the clip limit earlier, but also less distortion and less noise at the same level. Regardless of this, this card's THD and DIM values are also very



The MIC8.HD's transient intermodulation distortions (DIM100) for gain values of +20, +50 and +75 dB (f.r.t.l.) (Fig. 25)



The AN8.O output module's THD () and THD+N (- - -) curve as a function of the modulation (x-axis in dBfs) with a load of 200 kΩ and of 300 Ω. The measurement was made with a digital gain of 3 dB to make the clip limit more visible (Fig. 26)

good and meet the expectations one would have of a high-quality microphone input. For the sake of completeness, it should also be mentioned that the MIC8.HD.I card also features individually switchable phantom power.

AN8.O

Last but not least, let us take a look at the AN8.O analogue output card, which is available either as a separate module or as an add-on for the input modules, allowing users to implement up to 32 analogue inputs and outputs with the Prodigy's four slots. On the card, the maximum output level can be set individually to +15, +18 or +24 dBu for all eight outputs via a jumper, which facilitates matching to subsequent devices. In the +24 dBu setting, the measured noise level at the output was -92.2 dBu and 94.5 dBu(A), respectively, resulting in a very good S/N of 118.5 dB(A). The frequency response (no Fig.) was of course perfectly straight with a corner frequency of -1 dB at the upper end of 60 kHz.

The AN8.O's distortion values (Fig. 26) were also measured with the setting for an output level of +24 dBu. For this purpose, The THD measurement was carried out for two load conditions: once with the measuring system's usual input resistance of 200 kΩ and once with a load of 300 Ω. However, loads of this size are likely to be encountered rather rarely in practice. Only if a large number of receivers are fed in parallel from one output can it happen that the load to be driven for the line level output drops to values below 1 kΩ. With the 300 Ω load, the distortion values increase just before the clip limit from a very good 104 dB to a still good -84 dB. The output's internal resistance is 100 Ω, so that with a load of 300 Ω, the level at the receiver drops by 2.5 dB.

Conclusion of the measurements

All of the Prodigy's analogue modules fulfil their tasks and deliver good to very good measurement results. Decisive for successful use is the choice of the right module: if line level is required at the input as well as at the output, then the matter is clear. Both the AN8.I and the AN8.O card are an unqualified recommendation in all situations.

Users dealing with low-level signals have the choice between the MIC8.LINE.I and the MIC8.HD.I. For microphones, the HD version is the first choice. For the MIC8.HD.I, all measured values – including an EIN value of -128 dBu – turn out perfectly. For the last 40 dB of its adjustable gain range, the MIC8.LINE.I uses only a digital gain and therefore cannot match the HD version for low level signals. However, it is probably also intended that the MIC8.LINE.I is used for line level and somewhat weaker signals and, if necessary, also for microphones, but in principle, when it comes to connecting microphones, it is better to use the HD version. ■