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FACHMAGAZIN FÜR VERANSTALTUNGSTECHNIK

TEST FROM ISSUE **2 I 2021**

MATRIX MIXER

Dynacord MXE5

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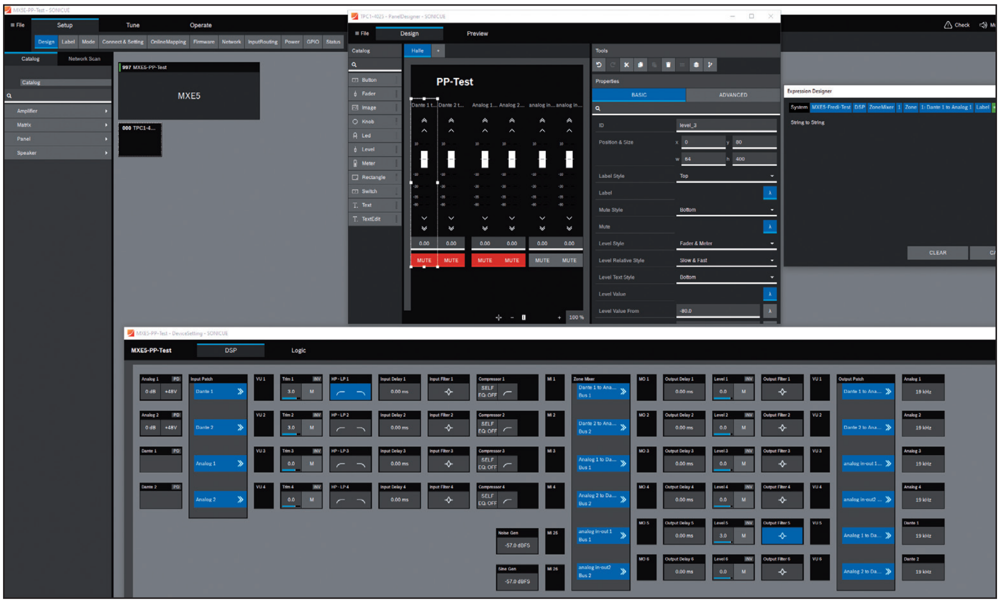
24 × 24 mixer with powerful DSP hardware, Dante interface, and integration with the new Dynacord SONICUE sound system software.

Text and measurements: Anselm Goertz | Photos: Anselm Goertz

With the MXE5 model, Dynacord launches a new series of devices. The designation "MXE" for "Matrix Mix Engine" already reflects the basic function of the device in its name. The core of the MXE5 is a freely configurable and divisible audio matrix with 24 × 24 crossover points, flanked by input and output channels with many signal processing functions as well as complete patch panels on the input and output side. On the hardware side, the MXE5 features 12 analog inputs with full-featured preamps, eight analog outputs, and a Dante interface with 24 in- and outputs each. The technical specifications are all hallmarks of an audio quality of the highest standards – a sample rate of 96 kHz and a signal-to-noise ratio of 115 dB, a THD+N of 94 dB and a latency of <0.45 ms.

The MXE5 was developed for use in live sound and fixed installations. In live sound, the typical application is signal mixing and routing, e.g. as an interface between the mixing console and a large PA with multiple outputs. Applications in fixed installations are also diverse, when it comes to supplying zones from the main stage to the lounge in clubs or distributing signals to diverse areas in a stadium. Moreover, the MXE5 can be used as a system manager or controller in audio networks that comply with the OCA standard. OCA stands for Open Control Architecture and is defined by the AES70 standard. The signal transmission to the other devices in the network is done via Dante. Both together, the OCA protocol for media control and the signal transport with Dante, are found in OMNEO media networking architecture, developed by Dynacord parent company Bosch. In addition to the new MXE series, the TGX and IPX

Dynacord MXE5 with the individually programmable panel TPC-1



▌**Sonicue software** with MX5E and Touch Panel Controller TPC-1 (Fig. 1)

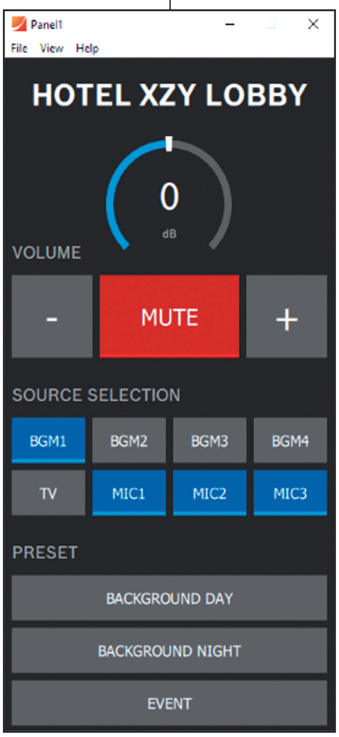
amplifiers also support OMNEO IP architecture. Older devices, such as the P64 DSP often found in large installations or the DSA amplifiers, can be integrated via MXE Logic. OMNEO-based networks can be set up with single or multiple redundancy, which is particularly important for safety-relevant systems. Redundancy in the first layer can be achieved by a ring-shaped network architecture. This is done using the "Rapid Spanning Tree Protocol" (RSTP), which deactivates redundant paths in the network during normal operation in order to activate them should an interruption occur elsewhere. If a switch in a ring fails or a cable is interrupted, the remaining signal path is automatically used. Without RSTP, this would not be possible, as otherwise network conflicts would occur during normal operation. Working with two rings for the primary and secondary Dante network adds another fallback layer.

Sonicue Software

The MXE5 is configured using Dynacord's new SONICUE sound system software, designed to configure, control and monitor complete systems.

Sonicue has three basic modes: Setup, Tune and Operate. The designations are self-explanatory. Once the program is started, you can either search for devices that are already available in the network, or you can first put together a complete system offline. The catalog of available devices includes speakers, amplifiers, matrix mixers and devices for operation. The latter can be a PC, a tablet or even a wall panel like the TPC-1. A TPC-1 has also been supplied for testing. Equipped with a 5.7" HD touch display, the compact device can be inserted into a standard wall-mounting box and assigned selected functions for operating the system. Power is supplied either with PoE via the network connection or via a separate power supply unit.

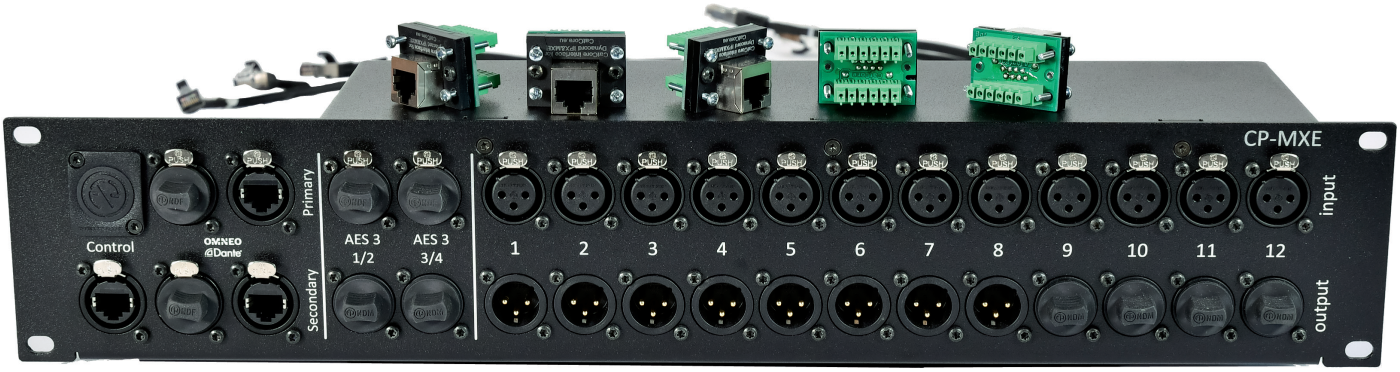
The definition of the interface including individual background images is created in the Sonicue software.



▌**Panel Designer** in the Sonicue-Software (Fig. 2)

The amplifiers and matrix mixers currently available in the catalog are the in-house models of the IPX, TGX, H/TG (via RCM-28)), L and C series, as well as the DSP systems from the MXE series. In addition to the MXE5 presented here, an MXE10 with three times the DSP processing power of the MXE5 and additional connection options in AES/EBU format is planned for the future. A software update for the MXE5 has also been announced, which will then also allow free configuration without a fixed structure, as was previously known from the P 64 / N8000 DSP system.

After the system is assembled and "wired" in the Sonicue interface, "tuning" takes place with the selection of setups, filters, delays and gain settings, as well as possible system tests using the built-in signal generators. The final step is the operation of the system in "Operate" mode with level displays, error detection, impedance monitoring and much more. All de-



Connection panel for the MXE5 (and also the future MXE10) with AES3 inputs and outputs and 12 analog outputs. The panel is connected to the MXE5 via twisted pair network cables with RJ45 connectors and small adapters to the Euroblock connectors on the device

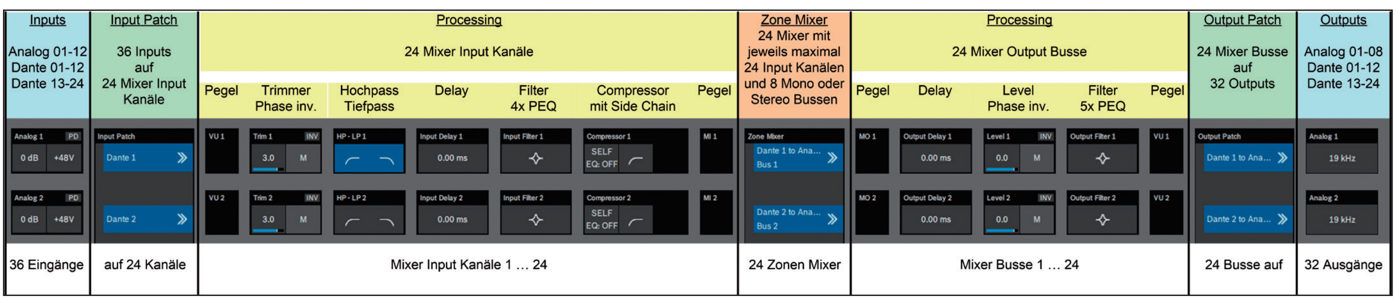
tails of the software cannot be described here. To explore Sonicue in detail, we therefore recommend downloading the free software available from the Dynacord homepage (www.sonicue.com). The software includes a thematic help and a dynamic help, which displays information about the function currently being processed. When using Sonicue software, especially together with the help pages, it is advisable to have at least one large screen, preferably supplemented by a second one for the help, otherwise the necessary overview can easily get lost.

Also worth mentioning are the possibilities to program your own functions via Logic Designer and to create API communication.
However, detailing this would go beyond the scope of this test. For interaction with Crestron media controls or a QSC Q-SYS system, corresponding software plug-ins are available for the MXE5.

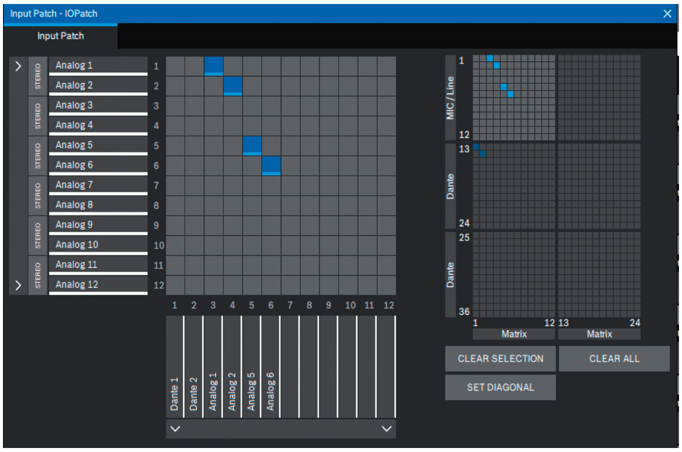
MXE5 Features

To better understand the function of the MXE5, it is useful to have a look at the block diagram in the software. Fig. 3 shows an excerpt with additional explanations. The leftmost column lists all inputs present in the hardware and actively routed. Inputs that are not routed in the input patch do not appear here. For the MXE5, that's a maximum of 12 analog inputs and another 24 inputs from the Dante network. This is where the preamp gain and phantom power can be set for the analog inputs. All inputs, including those from the Dante network, also feature pilot tone detection and notch filters with adjustable frequencies. The next column is the Input Patch, where the total of 36 inputs can be freely assigned to the 24 input paths (or channel lines). The hardware input that is assigned to the input channel to the right of it is indicated by the name in the Input Patch. Typically, this will not be Dante 1 … etc., but the actual name of the source. The Processing column contains the signal processing for the respective channel with various filters, Delay, Gain and dynamic functions. Upstream and downstream of the processing chain there is a level meter each.

The zone mixers are arranged in the center. Up to 24 independent mixers can be defined here, which map a mix of a maximum of 24 inputs to a maximum of eight mono or stereo busses. All mixers can use any of the signals from



Audio signal processing in the MXE5 in 96 kHz mode. At 48 kHz sample rate, speaker processing is added on the output side (Fig. 3).



Input Patch for 12 analog and 24 Dante inputs to be assigned to the 24 mixer input channels (Fig.4)

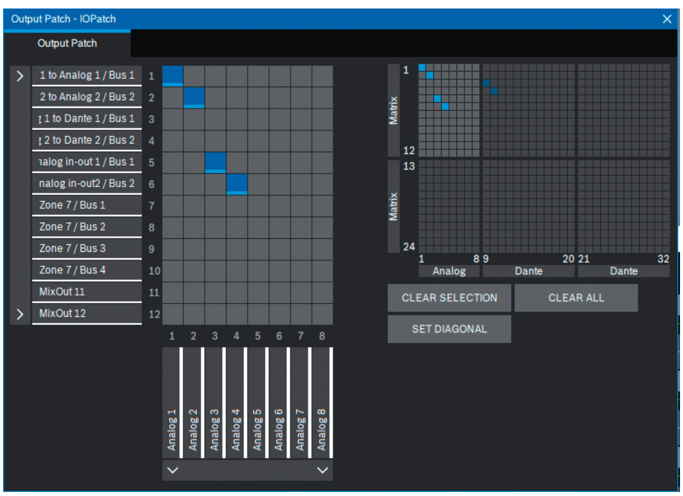
Zone mixer example with six inputs and four output buses, the buses all receive the same mix and are only used to play out this mix at different levels if required (Fig. 5)

the 24 input paths. The maximum number of buses from the total of all mixers is also 24 and is specified by the 24 mixer output paths. However, the term "output bus" is a bit misleading here, since not each bus of a zone mixer receives its own mix, as commonly used in a mixing console. All busses of a mixer receive the same mix and can only be adjusted differently in the output level.

The subsequent processing in the outputs is similar to that of the inputs, except for the compressor, which is not present here. The 24 buses can then be freely routed to the 32 outputs available in the hardware in the output patch. All outputs including those in the Dante network can provide a pilot tone adjustable in level and frequency.

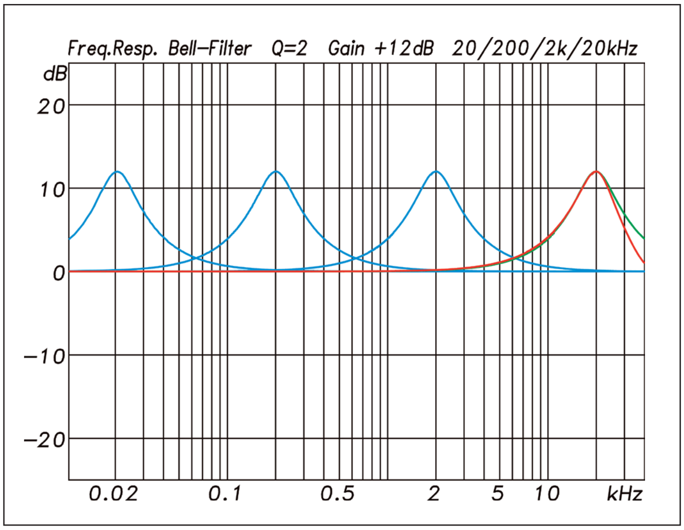
Filters

The filter block in the input and output paths includes four and five fully parametric EQs, respectively, which can be defined as Bell filters, high-pass or low-pass, or high- or low-shelv. Only the inputs include the option of a notch filter and only the outputs include the option of an allpass filter. The notch filter in the inputs can be used to avoid feedback at critical frequencies and the allpass in the outputs can be useful for matching subwoofers and tops. All filter types are very flexible in their parameters with a gain from -18 to +12 dB and grades from 0.4 to 40. Shelving filters and all-passes can be set in the quality from 0.4 to 2.0. A practical feature of the filter blocks is the possibility to store four setups in



Output Patch for the 24 mixer outputs, which can be assigned to the 32 outputs of the hardware here (Fig. 6).

Filter adjustment in the outputs with five parametric EQs. Four settings can be stored in the Memory, between which the user can switch quickly for listening tests (Fig. 7)



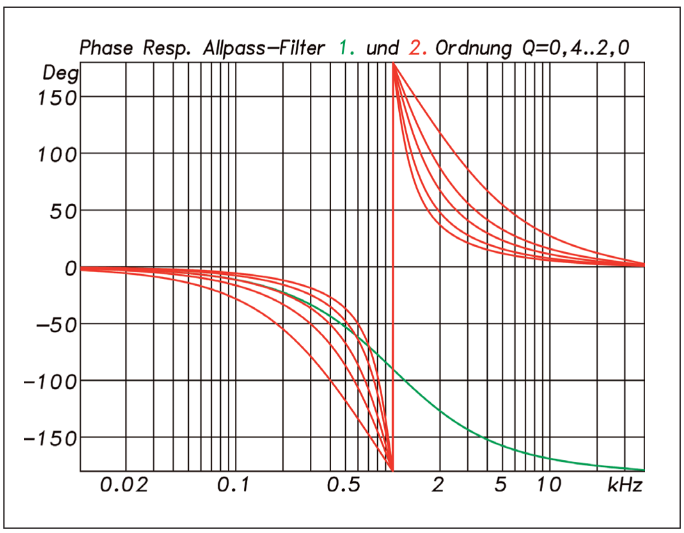
Bell-Filter with a grade Q of 2 and 12 dB gain at 20, 200, 2k and 20 kHz (red). The underlying green curve shows the calculated course of the 20 kHz filter (Fig. 8)

the memory and to recall and switch them directly for a quick comparison.

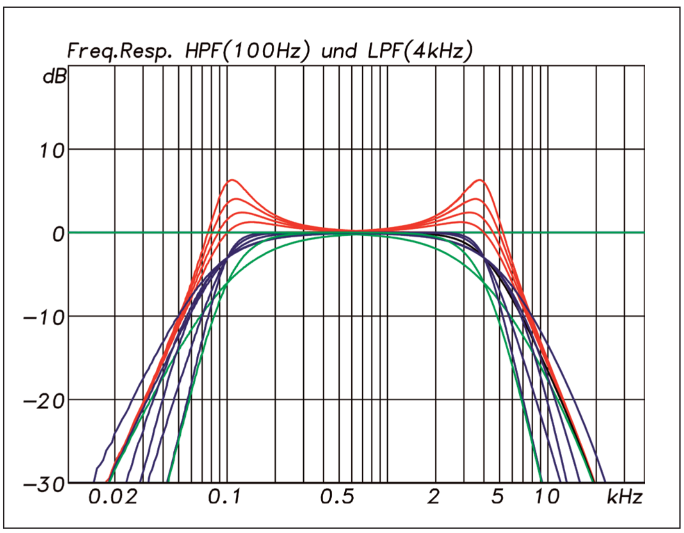
In terms of measurement, only a few details of the filters will be discussed in more detail. Fig. 8 shows a Bell filter for frequencies from 20 Hz to 20 kHz. A critical point here is the distortion of the filter curve as soon as you approach half the sample rates. The term "distortion" is not meant in the sense of distortion of the audio signal, but as distortion of the curve. Basically, the latter is not critical at first. However, the deviation from the calculated ideal curve, as known from analog filters, is often perceived as unpleasant. This can be avoided by either increasing the sample rate until there are no or only minor effects in the audible frequency range, or by compensating the filter curve in advance by calculation, which of course is only possible up to half the sample rate. The MXE5 can work with a sample rate of 96 kHz at almost full functionality, thus circumventing the problem in a simple way. Fig. 8 shows the calculated ideal curve and the measured curve, where there is no noticeable difference up to above 20 kHz.

Another metrological aspect to check for filters is possible distortion of the audio signal with high-grade filters for low frequencies. Here, rounding errors can lead to considerable signal distortions in some cases. Primarily affected are DSP systems with fixed-point arithmetic and inadequate bit depth. In the MXE5, signal processing is done with 32/40-bit floating-point arithmetic of the Sharc DSPs, where this issue does not occur. A measurement with a Bell filter of grade 40 at 50 Hz with 2 dB gain thus caused the THD value to rise only from 98 to -94 dB.

Among the many possible filter types, let's take a quick look at the all-pass filters that are found in the output paths. As the name suggests, these filters have no effect on the amplitude response, which is always equal to 1 (0 dB) regardless of the frequency. However, the phase response is different: A 1st order all-pass rotates the phase over the frequency by 180° and a 2nd order all-pass by 360°. At the set center frequency, the phase rotation is then 90° or 180°.



Phases of the 1st (green) and 2nd order (red) all-pass filters. The 2nd order filters can also be adjusted in grade, with the amplitude response not being affected by these filters (Fig. 9)



High-pass and low-pass filters in the input paths, here exemplified at 100 Hz and at 4 kHz (Fig. 10)

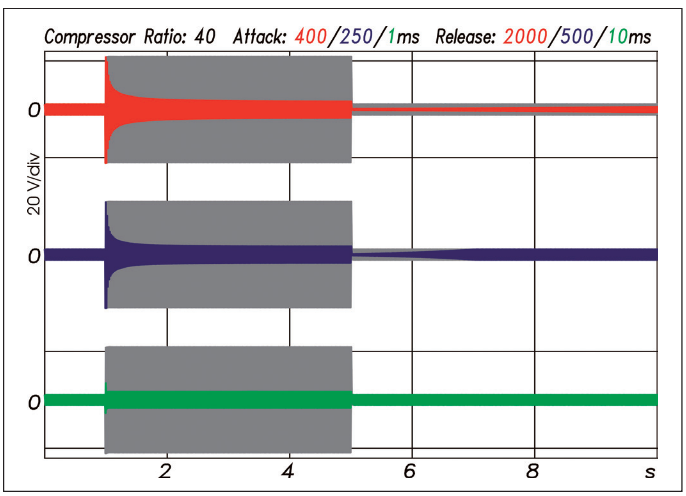
Gate, ducker and compressor with parameters and characteristic curves, in 96 kHz mode only the compressor is available (Fig. 11)

Fig. 9 shows some examples for a frequency of 1 kHz. With 2nd order all-pass filters, the slope of the phase rotation can also be adjusted via the filter grade. All-pass filters are mainly used when the individual paths of a loudspeaker have to be matched to each other or when subwoofers and tops have to be brought into phase alignment. A simple phase inverse usually does not yield a satisfactory result.

In addition to the filter block described above, only the mixer inputs feature explicit high-pass and low-pass filters that can be configured with slopes of 12, 18 or 24 dB/oct. Fig. 10 shows the high-pass and low-pass curves for 12 dB filters with adjustable grade (red), for Bessel and Butterworth filters of 2nd, 3rd and 4th order (blue) and for Linkwitz-Riley filters with 12 and 24 dB/Oct. (green).

Dynamics Functions

Apart from the filters, the functions for controlling the signal dynamics rank among the most frequently used ones. Especially when microphones are involved, it is important to get a grip on the level differences, which can be considerable. The compressor is ideally suited for this purpose, as it begins to reduce the gain at a certain level (threshold), thus limiting the signal peaks. In the MXE5, compressors are found in the inputs. Fig. 11 shows the configuration window with all parameters and the resulting characteristic curve. The threshold setting in dBu is somewhat irritating, as it only fits the analog inputs and then only for 0 dB preamp gain with +22 dBu = 0 dBFS. It would be more appropriate to specify this directly in dBFS with reference to full modulation. A quick measurement of the compressor functions (Fig. 12) confirms that all values are maintained as set. This may sound obvious, but unfortunately it often is not. Switching the MXE5 from 96 kHz mode to 48 kHz adds the functions of a gate and a ducker to the input paths.



Compressor with different attack and release time constants (Fig. 12)

Speaker Processing

The speaker processing functions are also available in 48 kHz mode only and for a maximum of eight outputs. In addition to complete setups for Dynacord's and Electro-Voice's proprietary products, there are also manufacturer-verified setups for Fulcrum Acoustics speakers, as well as for some other well-known brands, though these are then not manufacturer verified. When using the predefined setups, the settings behind them remain hidden. Thus, the user can only specify the gain of the amplifier used for this channel. Switching to Custom mode unlocks almost all of the speaker processing functions. The only exception are the FIR filters, which can only be used with predefined setups.

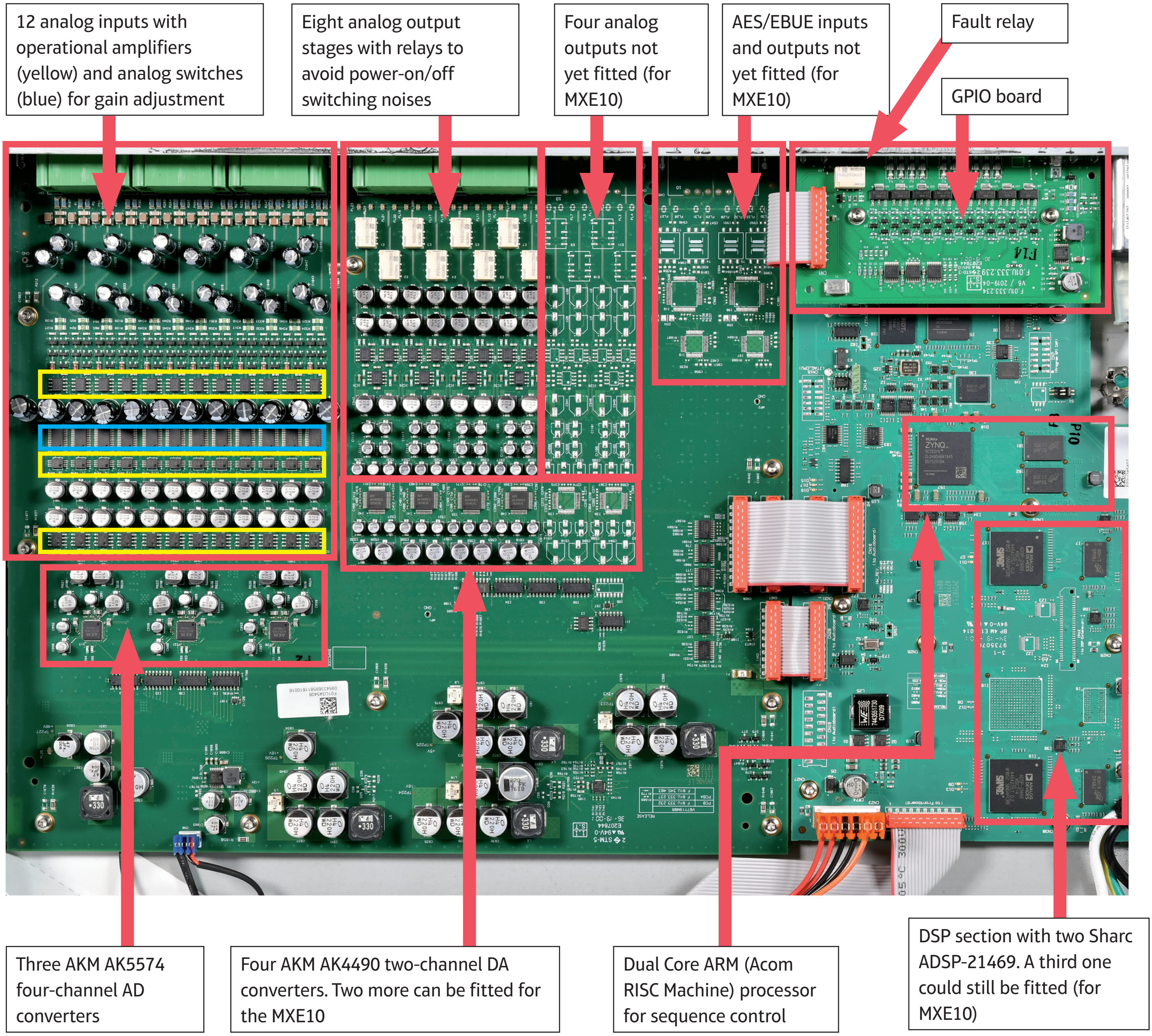
Speaker processing includes the following functions in Custom Mode:

Inside view of the MXE5: rear left the large PCB with all analog inputs and outputs, right of it the DSP board and the power supply unit front right

- PEQ with six parametric filters including allpass, notch and shelving filters
- XOVER with high and low pass filters 1st to 4th order with the usual characteristics as well as settings for level, delay and phase inverse
- LIMITER with RMS and peak limiters

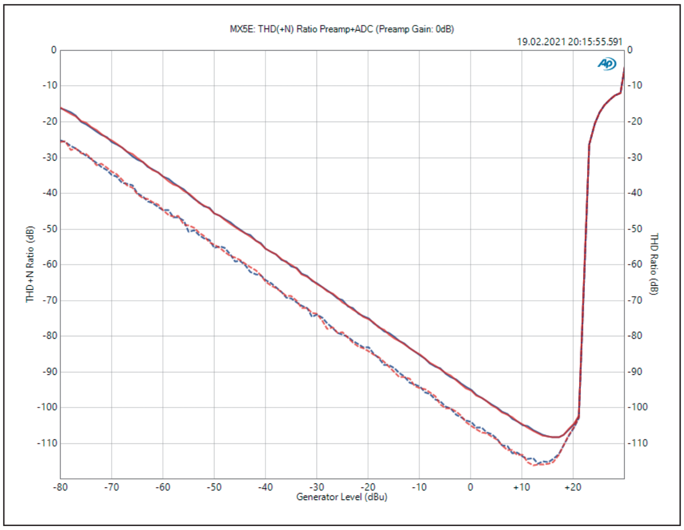
The limiters are particularly interesting here, with separate settings and sensible time constants that allow good adaptation to the speakers used. For this purpose, the user specifies a permissible voltage value in V_{pk} for the peak limiter and in V_{rms} for the RMS limiter. The values for the two limiters in the MXE5's output are calculated from this together with the likewise adjustable amp gain in dB. At first glance, the displayed value for the threshold DSP might be a bit confusing. In the example in Fig. 13, a correct maximum output level of the MXE5 of 5.78 dBu is calculated for a maximum output voltage at the amplifier of 60 V_{rms} at 32 dB gain. Now, setting the peak limiter to double the value of 120 V_{pk} results in an output level of the MXE5 that is only 3 dB higher, i.e. 8.78 dBu. At first glance, one would have now expected 6 dB more. However, since a level value in dBu always refers to 0.775 V_{rms}, the calculation was not based on the 120 V_{pk}, but on an RMS value that is 3 dB lower. Strictly speaking, however, the latter only applies to a sinusoidal signal. It would therefore have been more consistent to specify only V_{pk} values for the AMP threshold as well as for the DSP for the peak limiter and only V_{rms} values for the RMS limiter.

Irrespective of this, the question arises as to how to best set the limiters now. A data sheet of the connected driver usually specifies a performance value according to AES for a 2-hour endurance test. The value given here is the thermal continuous load capacity of the driver, from which the voltage value for the RMS limiter is to be calculated together with the impedance of the driver. But careful, this value is not calculated with the nominal impedance, but with the impedance minimum. Thus, if an AES (2 h) power value of 100 W is specified for a high-frequency tweeter with 16 Ω nominal impedance, this would correspond to a voltage value of 40 V_{rms} at 16 Ω. However, if this high-frequency tweeter has an impedance minimum of 12 Ω in its working range, then it is only 34.6 V_{rms}. For the value of the peak limiter, the AES test signal can be used as a guide, which features a crest factor of 2 (6 dB). The peak value then corresponds to the double RMS value. In practice, however, the crest factor of a music signal is significantly greater than 6 dB. Therefore, the peak limiter actually always takes effect first. Destruction of the driver due to a thermal overload is therefore rather rare. An exception, however, is the catastrophic case for the driver in the form of feedback, where a sinusoidal signal is initially limited only in the peak value by the peak limiter. The RMS value for a sinusoidal signal is then already exceeded by at least a factor of two, where a fast and safe intervention of the RMS limiter is the only remedy. The value for the RMS limiter should therefore always be chosen very conservatively and with a short attack time constant. Normally, this limiter will hardly or not at all be-

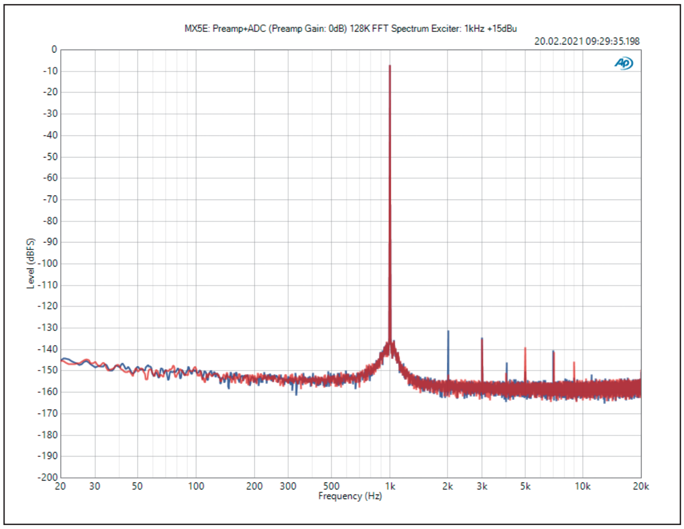


come active, so that no loss of level has to be feared. However, if an emergency occurs, immediate and, above all, safe reaction is required.

However, all of what has been said so far about the limiter settings only works reliably if, on the one hand, the set gain value of the power amplifier is correct and if the power amplifier is also capable of converting the set values. If the power amplifier starts clipping before the set value for the peak limiter is reached or the power supply collapses before the RMS value is reached, the limiters are powerless. In such case, the values would have to be adjusted to the power amplifier. As the time constants set for attack and release in Fig. 13 show, very short times are set for the peak limiter, which do not allow any overshoot with an attack of 0 ms. The situation is quite different with the RMS or thermal limiter, where the time constants are based on how fast the voice coil of a driver heats up. A large 5" coil with thick wire and a large heat capacity takes much longer to do this than the small, lightweight coil of a high-frequency tweeter. However, exact values for this are hard to find, as they also depend on the ambient temperature, the heat capacity of the magnet and various



THD+N and THD (solid and dashed, respectively) as a function of input level in dBu at 0 dB preamp gain, clip limit is +22 dBu (Fig. 14).

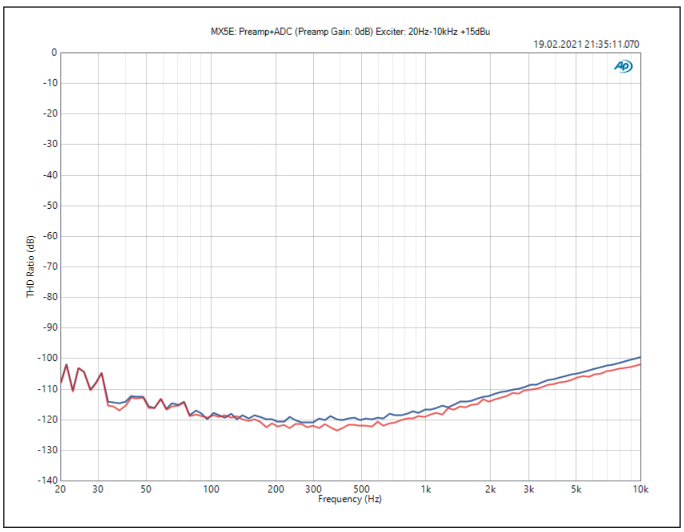


FFT spectrum at 0 dB gain for a 1 kHz signal with +15 dBu, the distortion components are more than 120 dB below the fundamental (fig. 15)

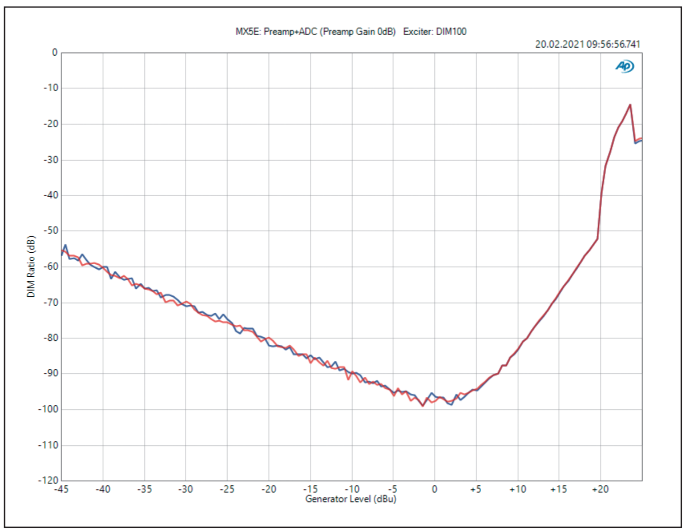
other factors. For a high-frequency tweeter with coil diameters of 1-2", the attack time constant of 1 s for the RMS limiter should be a safe choice. Low-frequency woofers and other large drivers with 4" or even larger coils have much longer time constants, where the maximum value of 5 s can always be set with confidence.

Audio Quality

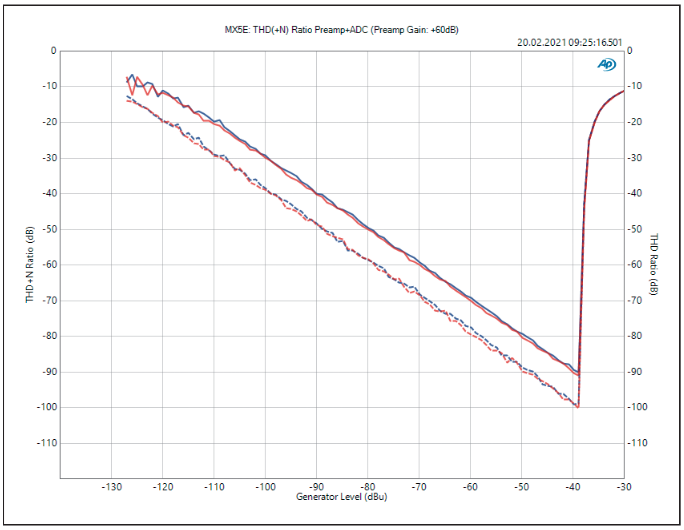
In the presentation of the MXE5, Dynacord clearly emphasizes the audio quality and also explicitly recommends the matrix mixer for challenging applications. Now, what are challenging applications, one might ask. Sophisticated demands usually arise when it is necessary to transmit a high signal dynamic range without noise or other interference being heard in quiet passages, and on the other hand loud signal peaks are not distorted or limited. In extreme cases, this can result in the need to map a signal dynamic range of 100 dB or more with reserves at the upper and lower ends. Considering that a signal chain can quickly lose a few dB due to possible mismatches and necessary headroom, the individual devices in the chain should offer a S/N of at least 110 dB or more. However,



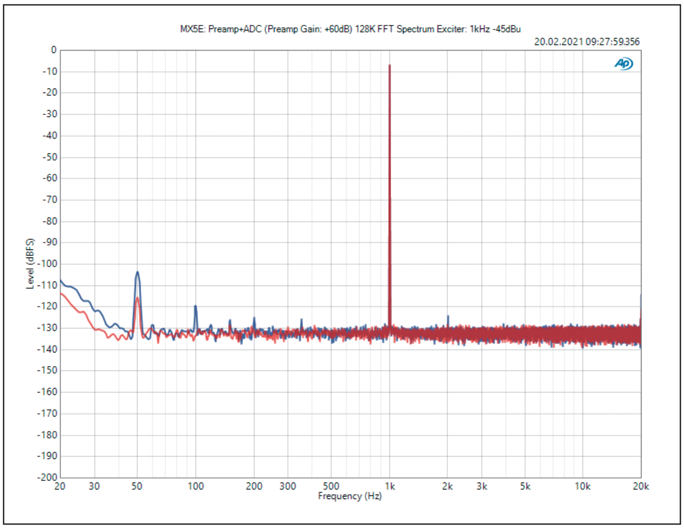
THD(f) as a function of frequency at 0 dB gain and +15 dBu input level (Fig. 16)



Transient intermodulation distortion (DIM100) as a function of input level at 0 dB gain (Fig. 17)



THD+N and THD (solid and dashed, respectively) as a function of input level in dBu at +60 dB preamp gain, clip limit is -38 dBu (Fig. 18)



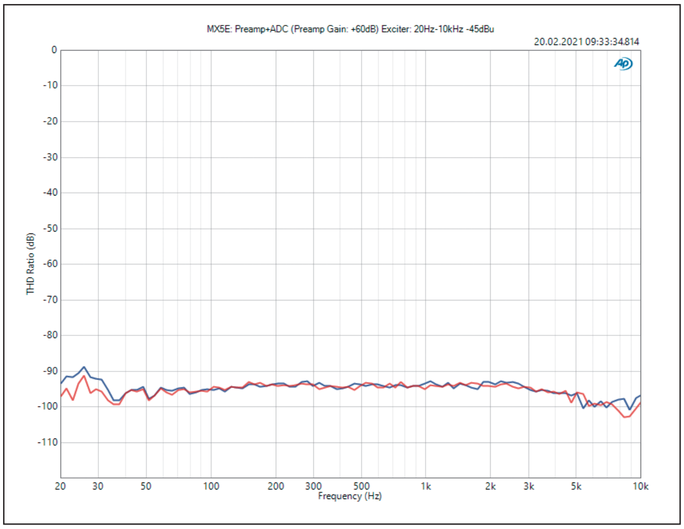
FFT Spectrum at +60 dB gain for a 1 kHz signal at -45 dBu, the distortion components remain low but now disappear in the noise floor (Fig. 19)

the standard, be it for power amplifiers, controllers, breakout boxes or mixers, is rather a few dB below that, which is more than sufficient for most applications. Yet when it comes to theaters, concert halls or sophisticated clubs, this may no longer be enough. Especially for matrix mixers like the MXE5, the situation is aggravated by the fact that they are located directly before the power amplifiers in the signal chain, where there is no downstream fader.

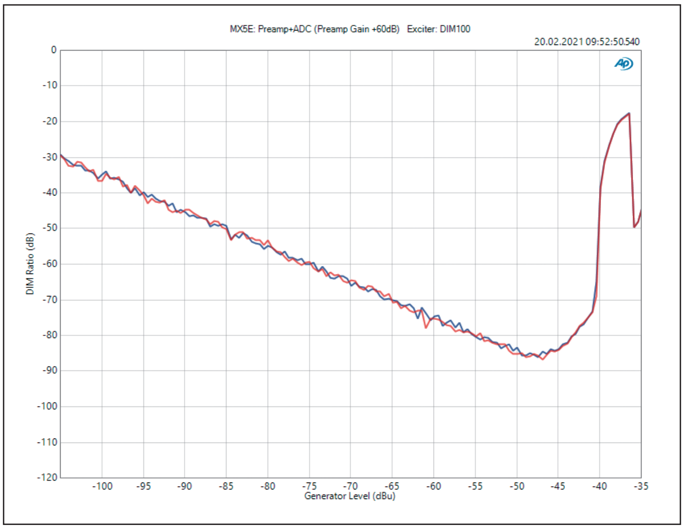
To illustrate this, here is a small calculation example: the MXE5 delivers a maximum of +22 dBu at the output with a S/N of 118 dB. This is followed by a hi-quality power amplifier with a S/N of 118 dB as well, which is, however, already fully modulated at +10 dBu at the input. The gain of the power amplifier is 32 dB. This produces an interference signal with a 0.5 mV voltage at the output of the power amplifier. If this power amplifier drives a tweeter path with a sensitivity of 110 dB at 2.83 V, this will result in an interference level of 35 dB at 1 m distance. Even though no one will be sitting within 1 m of the speaker, this would still be perceptible a few meters further away in a quiet environment. If there is more than one loudspeaker in a theater hall, maybe ten or even more, each dB more in the signal-to-noise ratio will be appreciated. If the MXE5 had a S/N at the output of only 105 dB in the calculation example above, it would already create 48 dB noise at a distance of 1 m in front of the speaker.

MXE5 Measurements

We conducted the following measurements for the MXE5 in our test: frequency response, interference level, THD and DIM for the analog inputs with preamp at minimum and at maximum gain and for the analog outputs. In each case, the measurements were carried out with the measurement signal being fed in or tapped via the Dante network. All measurements were performed with a sample rate of 96 kHz. The frequency response runs from 20 Hz to 40 kHz within a tolerance range of ±0.1 dB at the preamps' minimum gain of 0 dB. At maximum gain in the preamp of +60 dB, the ripple remains consistently low, but a high-pass response with a -3 dB cutoff frequency of about 20 Hz is added. On the output side, the frequency response is also completely smooth, as expected, with corner frequencies at from 15 Hz at the low end and 40 kHz at the high end with a level drop of only 0.5 dB. The measured value for the S/N at the analog outputs is 117 dB unweighted and 119.3 dB with A-weighting. For the analog inputs, the S/N value depends on the gain setting of the preamp. Likewise, at 0 dB, 117 dB unweighted and 119.7 dB with A-weighting are achieved. At maximum gain, the A-weighted S/N is 91 dB with an input sensitivity of 38 dBu at full modulation, from which an equivalent input noise (EIN) of a very good 129 dBu (20-20 k, A-weighted) can be calculated. In terms of S/N, the MXE5 clearly meets the high expectations.



THD(f) as a function of frequency at +60 dB gain and -45 dBu input level (Fig. 20)



Transient intermodulation distortion (DIM100) as a function of input level at +60 dB gain (Fig. 21)

Distortion Measurements

As for analog input distortion, THD and THD+N measurements were performed at minimum and at maximum gain. The measurement signal from the Audio Precision APx555 was fed into the analog inputs and output via the Dante outputs without using any internal signal processing. The signal then returns to the measurement system via the DVS (Dante Virtual Soundcard) on the PC associated with the APx. The results of the THD curves from Fig. 14 at 0 dB gain speak for themselves without lengthy explanations. The THD value drops to 116 dB at the minimum, 5 dB below the clip limit. Expressed as a percentage, this corresponds to 0.00015%. The corresponding FFT spectrum in Fig. 15 confirms the value and also shows a favorable distribution of the distortion components with falling values towards the higher order, if this even needs to be evaluated at all with values in this quasi vanishingly small order of magnitude. If we look at the THD values as a function of frequency (Fig. 16), they increase slightly toward higher frequencies as usual at 6 dB/oct. but this still means a value of -100 dB at 10 kHz. So everything is fine here. Only when measuring the transient distortions with a mixed square-sine signal, the preamp seems to become a bit stressed. The achieved minimum of 98 dB at 0 dBu signal level is a very good value, however, beyond that the distortion values quickly rise to-52dB up to the clip limit. The second DIM100 measurement at +60 dB gain from Fig. 21 does not reveal this behavior and delivers a value of -72 dB at the clip boundary, which is 20 dB better. This suggests that the preamp's first input stage does not handle steep signal edges at high levels very well.

Gain	Noise 20 Hz – 20 kHz		Max. input
dB	dBFS	dBFS(A)	dBu
0	-117	-119,7	+22
10	-113,7	-116,2	+12
20	-115,3	-118,0	+2
30	-110,7	-113,2	-8
40	-108,0	-111,0	-18
50	-99,0	-102,0	-28
60	-89,2	-91,0	-38

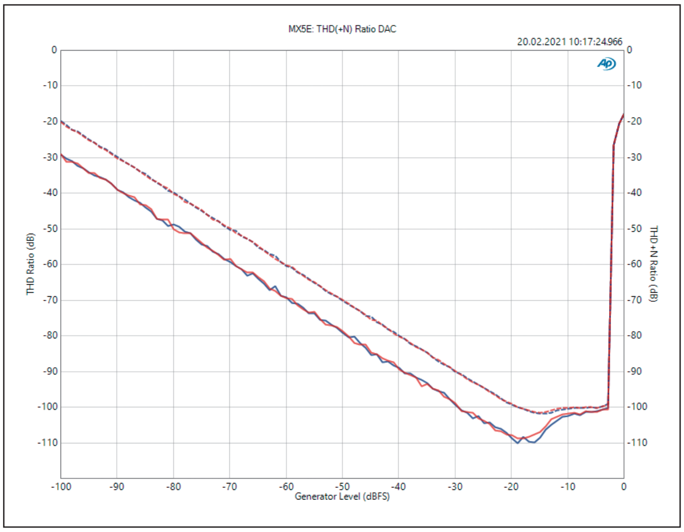
S/N values of the input paths with preamp and ADC depending on the set gain. At maximum gain, the equivalent input noise (EIN) is -129 dBu (20-20 k, A-weighted). (Tab. 1)

All other distortion measurements at maximum gain also deliver top values. The THD in Fig. 18 reaches an extremely good value of 100 dB even at a maximum gain of +60 dB directly upstream of the clip limit. The fact that the good values are not only achieved at 1 kHz is shown by the THD(f) measurement as a function of frequency at maximum gain in Fig. 20.

Three other THD measurements pertain to the MXE5's analog outputs. The signal was fed in via Dante and tapped again at the analog outputs. The THD curves in Fig. 22 drop to -110 dB at the minimum and rise only slightly to 100 dB up to the clip limit. The distribution of harmonics is favora-

Euroblock connectors for the analog inputs and outputs. On the left, the network control port and the OMNEO interface (Dante, OCA) with primary and secondary network connection

ble with rapidly decreasing values toward higher orders. Measured as a function of frequency, the good values of the 1 kHz measurement prove to be valid for the entire frequency range. There is no illustration of the last two measurements.



THD+N and THD (solid and dashed, respectively) of the analog outputs as a function of the input level in dBFS with +3 dB digital gain (Fig. 22)

Conclusion

With the MXE5 matrix mixer, Dynacord has a universally applicable DSP system in its portfolio that can be used for a wide variety of applications. In small installations, the MXE5 can assume the role of organizer of all sources and output paths. In large networks, the MXE5 can act as a central signal distributor, processor, communications center, and watchdog over the system. Owing to its outstanding audio quality and the abundant DSP functions available, it can be used in large theaters, sports stadiums and at festivals without any obstacles.

However, the MXE5 matrix mixer is only half the story, on the hardware side. In terms of software for the configuration and individualized operation of the Dynacord devices, the new Sonicue software is emerging as the successor to IRIS-Net, which has been widely used for many years. After a short period of familiarization, Sonicue makes it possible to assemble, configure and operate entire systems quickly and clearly. The MXE5 Matrix Mix Engine 24 × 24 is priced at (net) €2,890, the touch-panel controller (5.7") TPC-1 at €990. ■