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REVIEW: NEUMANN MT48

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HIGH-END FOR YOUR POCKET

COMPACT USB AND AOIP AUDIO INTERFACE NEUMANN MT48

The consolidation in our industry is noticeable on all fronts, so we were hardly surprised when the news broke that Swiss manufacturer of professional DAW and converter systems, Merging Technologies, joined the Sennheiser Group. Shortly after the acquisition of Merging, Neumann, the Berlin-based microphone specialist that has also been part of Sennheiser for many decades, introduced its first audio interface, the MT48. Visually, the MT48 closely resembles Merging's Anubis Audio-over-IP audio interface. Does this mean the manufacturer is offering us old wine in new bottles? Not at all, because in reality, the order of events was the opposite ‚behind the scenes.' The two companies decided to jointly develop an audio interface and release it under the Neumann brand. Through this collaboration, they eventually decided to continue their journey together on a business level as well. The first result of this joint effort is now the MT48, which we will explore in this review.

Externally, the resemblance between the Anubis and the MT48 is undeniable. Both feature the same precision-machined aluminum housing, with only slight adjustments to their corporate identity. Visually, the MT48 is an absolute eye-catcher and seamlessly integrates into Neumann's high-quality portfolio. However, the first differences become apparent when looking at the rear panel. The MT48 is essentially a modified version of the Merging Anubis, tailored to the market of demanding musicians and producers. As a result, a large part of Anubis's capabilities has been inherited and augmented with additional features, while some parts of the software have been, at least temporarily, omitted.

Overview

The MT48 is a multi-channel audio interface with built-in functionality as a monitor controller, as well as DSP resources for effects and an extensive audio mixer. Its computer interface relies on the Audio-over-IP network Ravenna/AES67 protocol, with only one network port available compared to the Anubis (in its current SPS version). The most significant unique feature of the MT48 compared to its older sibling is that Neumann has integrated a USB interface, allowing direct connection to the computer. The physical interface uses USB-C, and the underlying protocol is USB 2.0. In fact, there are two USB-C ports on the back of the device, with one serving solely as a compact power supply connection. Neumann provides a powerful external power supply, as the power provided by the computer alone is insufficient to satisfy the MT48's hunger for power. While the decision for a dual USB-C port may seem unusual at first glance, it proves to be quite practical. The interface and cables are widely available, the port is compact and reliable, and in a pinch, a power bank can even be used while traveling. With the elimination of the second RJ45 port, the MT48 had enough space for an optical interface, which can be used for ADAT (up to eight channels) and TOSLink (two channels). All other input and output options are the same as those on the Anubis. There are two combined microphone/line inputs, two instrument/line inputs, four line outputs, and two headphone outputs. Two additional quarter-inch jacks provide either a simple GPI/O signal or MIDI connectivity.

The device is operated using the large touch display, a highly responsive rotary encoder, and seven backlit rubber buttons. Due to its high feature density and resulting power consumption of up to 25 watts, the compact housing sometimes requires active cooling. This is achieved by a small fan that sucks warm air out of the device, which we will discuss further in the Practical Use and Listening section. Audio-wise, the MT48 supports sample rates up to 192 kHz. No premium version is currently available for even higher sample rates or DSD, like with Anubis. The converters and DSP resources are the same as those found in the standard version of Anubis.

USB and Control

Of course, the touch display is not the only way to operate the device. When installing the driver package on Windows 10 or macOS (from 10.15 Catalina), besides audio drivers, you also gain access via a remote control app or a browser interface. The browser interface can be utilized with Firefox, for example. Both control concepts largely correspond in functionality to operating the device on the front panel. The browser control is also possible in network operation, eliminating the need to install drivers. All three control concepts can be used simultaneously, allowing you, for instance, to keep the monitoring controls accessible on the device while working with DSP effects via the remote control app. The USB audio driver supports ASIO under Windows and can transmit up to 32 inputs into the PC and up to 16 channels back at a sample rate of 48 kHz. Thus, mixes and other signals can also be transferred besides physical connections. Furthermore, the device is partially class-compliant and can be used without driver installation on certain devices. For example, it is compatible with iPads, but we have had no luck with Android devices and our Linux PC so far.

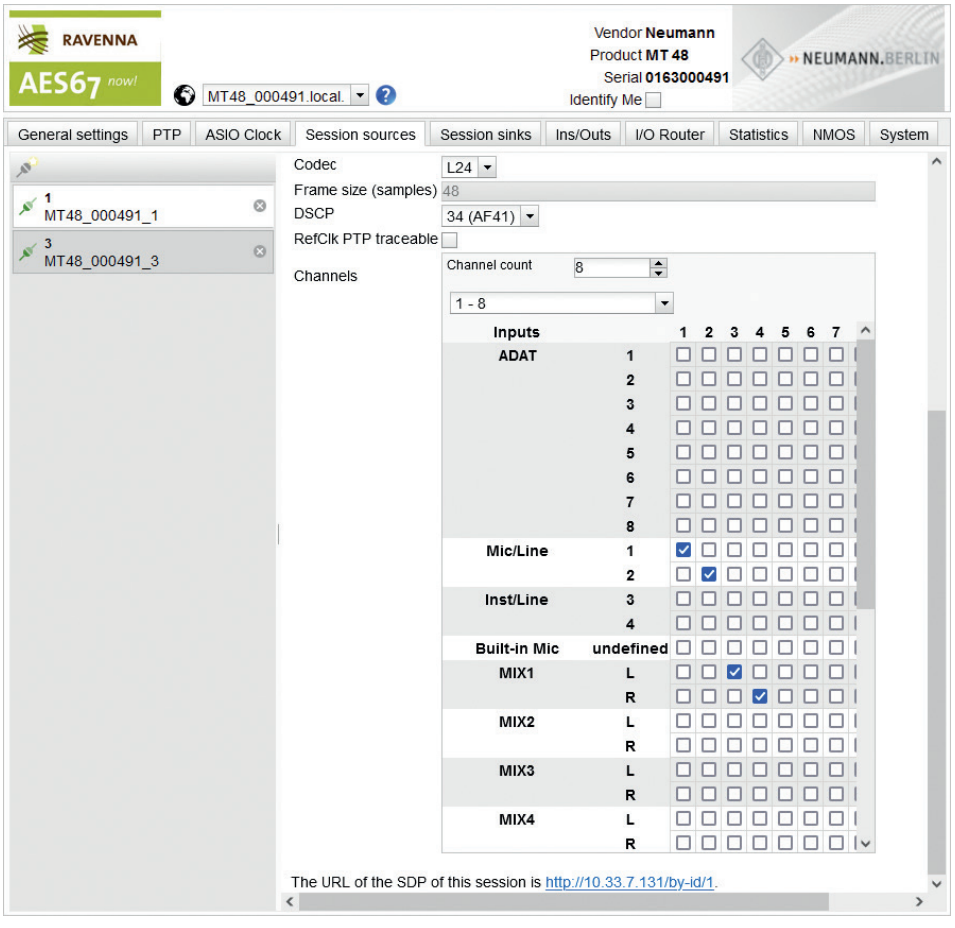
Routing and Mixer

The audio routing provides the option to either directly link the physical inputs and outputs with the USB connection or route them through four separate mixer instances. These four mixers operate independently and can be assigned to individual stereo outputs. However, a mix cannot be sent to multiple outputs simultaneously, which can be rather inconvenient if you want to use the same mix for both headphone outputs, for example. The mix bus can be sent back to the DAW for recording purposes. Each of the four mixers has an equalizer and a dynamics section, consisting of a gate, compressor, and limiter, on its mix bus. All four mixers share a common send effect, which can be accessed via a post-fader send. Consequently, the send levels cannot be different on the same channel but rather are set equally in all four mixers. The effect return, on the other hand, is available as an independent source everywhere. Currently, the only effect provided is a reverb. The quality of the effects is evaluated in the „Practical Use and Listening“ section. With Anubis, there is an option to purchase a license for the superb-sounding Eventide Blackhole Reverb. Perhaps this option will also be made available for the MT48 in the future.

Measurements

The Merging Hapi has been our reference converter in our Berlin editorial studio for many years. Therefore, we were eager to see if the MT48 could match or even surpass the quality offered. By the way, regarding the quality of the analog inputs and outputs, the MT48 and the Anubis do not differ. As always, we use our Audio Precision APx555 to measure the technical performance. Let‘s start with the line inputs. The maximum input level is +24.36 dBu for 0 dBFS. The

noise level is a sensational -136.5 dBu RMS unweighted (20 Hz to 20 kHz). The quasi-peak comparison value according to ITU-R BS.468-4 reaches -125.4 dBu. As expected, the noise spectrum, depicted in Diagram 1, is perfectly free from disturbances. Such high dynamics can only be achieved with multi-stage converters, and the handover point can usually be recognized from the THD ratio curve as a function of input level. Diagram 2 shows its progression and illustrates the step at around -26 dBFS. At full scale, the THD ratio reaches 0.0005%, and its lowest value of 0.00018% is achieved below -9 dBFS. The THD ratio curve over frequency response is a somewhat mixed bag. Diagram 3 shows the results at -1 dBFS and -9 dBFS. The best and worst values



The Advanced page allows for configuring network streams and creating direct routings for Audio-over-IP.

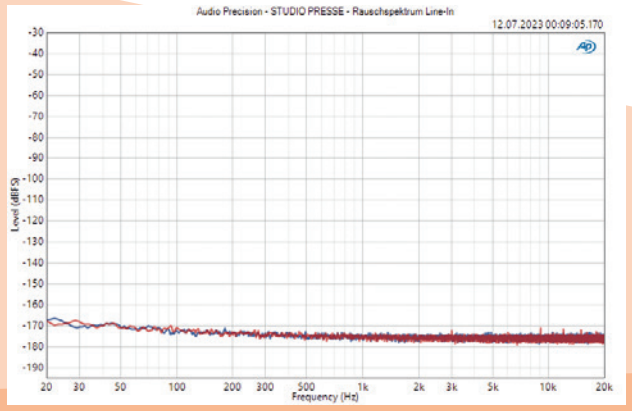


Diagram 1: Noise spectrum - line input

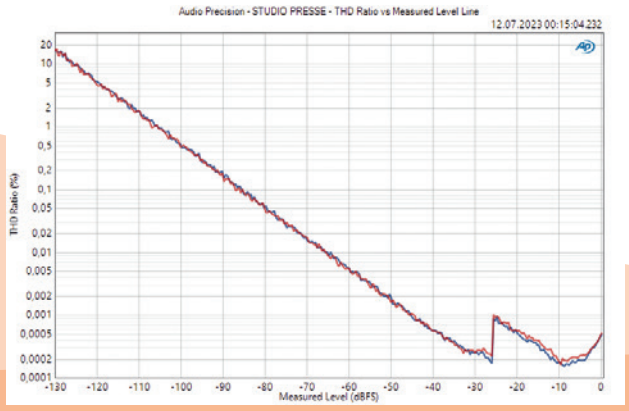


Diagram 2: THD ratio response of the line input over input level. The point of switching between the two converter stages is clearly visible

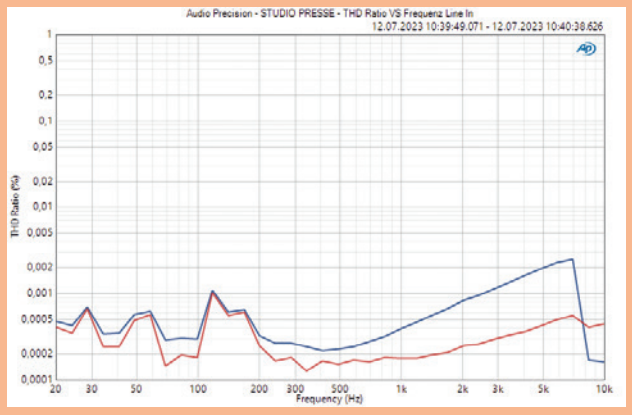


Diagram 3: THD ratio response of the line input over frequency at -1 dBFS (blue) and -9 dBFS (red)

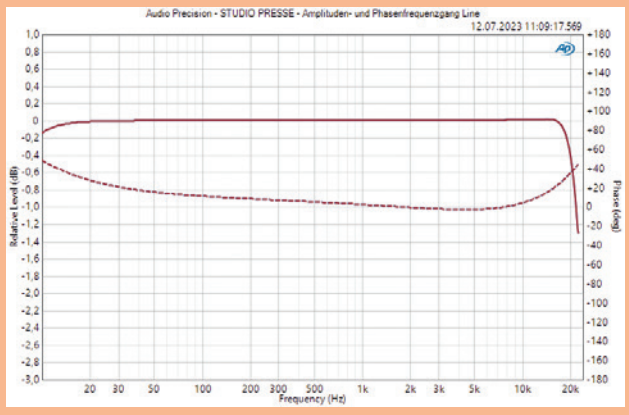


Diagram 4: Amplitude (solid line) and phase frequency response (dashed line) of the line inputs at 48 kHz sample rate

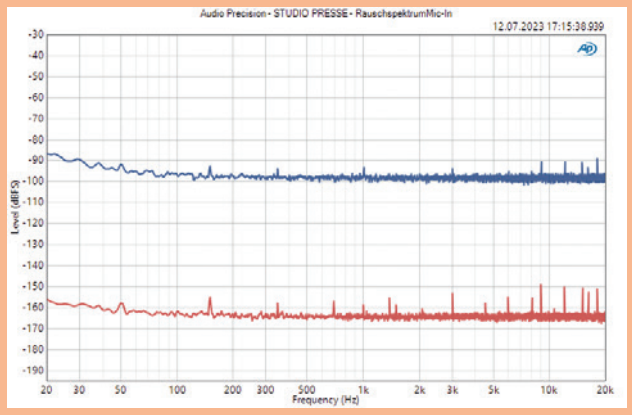


Diagram 5: Noise spectrum of the microphone input at minimum (red) and maximum (blue) gain

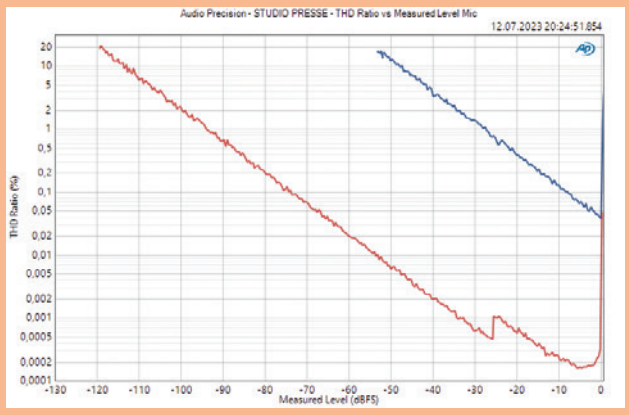


Diagram 6: THD ratio response of the microphone input over input level at minimum (red) and maximum (blue) gain

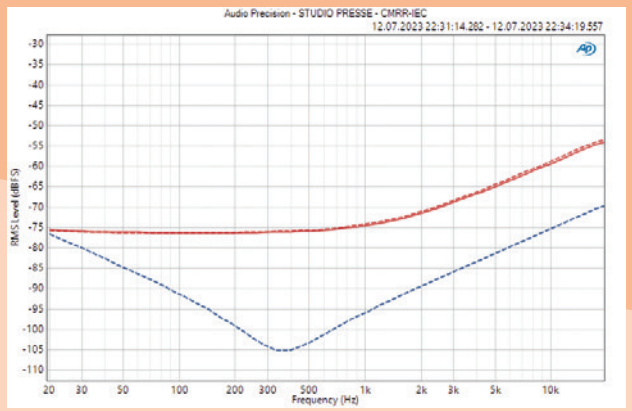


Diagram 7: Excellent result in the Common Mode Rejection Ratio (CMRR) measurement according to IEC for microphone inputs. The top curve represents the result

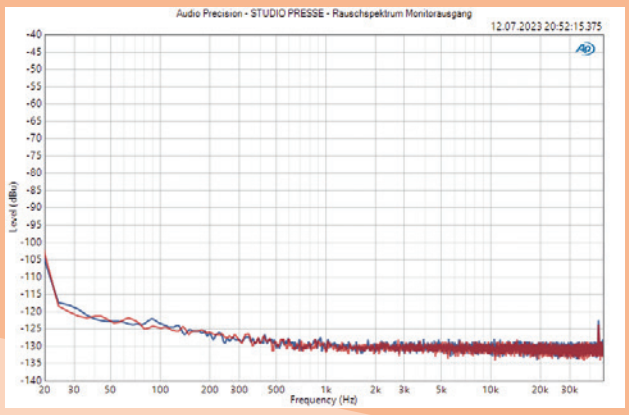


Diagram 8: Noise spectrum of the line output used as the monitor output

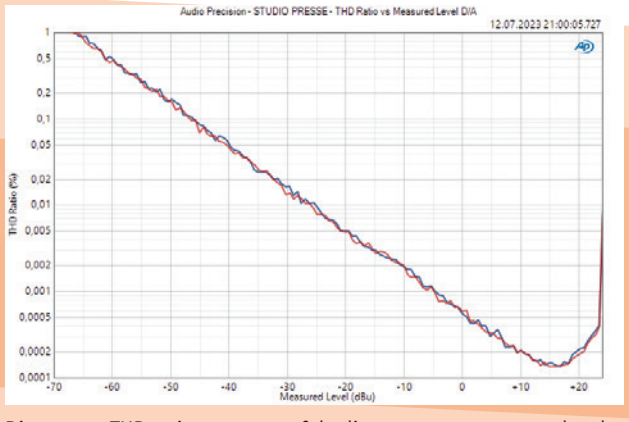


Diagram 9: THD ratio response of the line output over output level

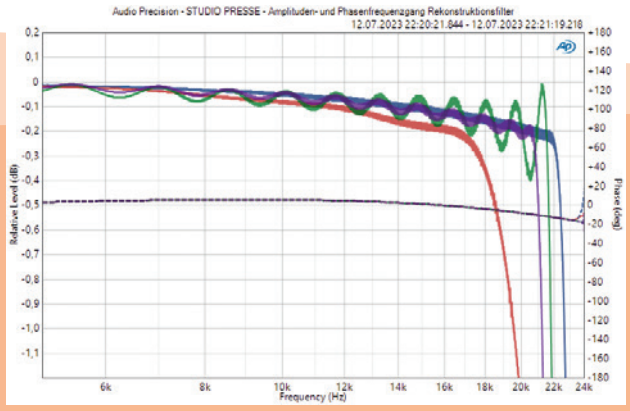


Diagram 10: Comparison of different reconstruction filters: Sharp (blue), Slow (red), Apodizing (green), and Brickwall (purple)

are almost an order of magnitude apart, with the rise being most relevant in the lower frequencies. So, the remarkable dynamic range comes with slight drawbacks in distortion behavior, although to be honest, this is nitpicking at a relatively high level. The amplitude and phase frequency responses in Diagram 4 are absolutely flat. The anti-aliasing filter cannot be switched for the inputs. When switched to microphone input, the dynamic range is slightly reduced but still remains a sensational 125.4 dB RMS unweighted (20 Hz to 20 kHz) with a minimal gain of 12 dB. The input sensitivity can be lowered in two stages using a pad. In the -24 dB operating mode, the input achieves the same values as in line mode, suggesting that the line switch probably performs the same function (activating the pad), and the signal path is identical in both cases. With maximum gain, the usable dynamic range drops to 59.3 dB. Mathematically, the equivalent input noise is 125.4 dB. However, this value plays a minor role since the crucial factor in practice is solely the digital dynamic range at the converter‘s output. The 100 dB dynamic range limit is reached at a gain of approximately 38 dB. Adding digital gain in the DAW would yield half a decibel of improved signal-to-noise at best. Therefore, it is better to leave some headroom and adjust the level afterward if in doubt. There is no sonic disadvantage in doing so. The noise spectrum in Diagram 5 shows small spike disturbances that remain very low in level, even at maximum gain. The harmonic distortion behavior is similar to line mode, with slightly higher values as expected. The lowest value of the THD ratio is achieved at -5 dB and is at 0.0002%. Diagram 6 displays the progression of the THD ratio over the input level at minimum and maximum gain. The amplitude and phase frequency responses do not differ from line mode. A look at the common-mode rejection ratio (CMRR) according to IEC provides an above-average result, even with inserted unbalancing resistors. The measurement result is documented in Diagram 7. Now, let‘s focus on the outputs. At the XLR outputs, exactly 24 dBu is present at 0 dBFS when the monitor controller is set to 0. The noise level is -97.1 dBu RMS unweighted (20 Hz to 20 kHz). This results in a dynamic range of 121.1 dB. The corresponding quasi-peak comparison value is -86.3 dBu, indicating that there are no disturbances to be expected here either. Diagram 8 confirms this assumption. Inspired by the slight increase at the lower end of the spectrum, we took a look at the output DC offset. With about 6 mV, it is well below our threshold of 10 mV and is, therefore, flawless. The THD ratio drops significantly below 0.0002% in the upper operating range. As a result, the converter does not reach high-end performance levels but operates within a top class. The THD curve over the output level can be seen in Diagram 9. As expected, the frequency responses are completely smooth, so instead of a broad overview, let‘s focus on the upper end of the spectrum. The device allows switching between the four available reconstruction filters of the D/A converter. Diagram 10 shows the results of the various filter settings in comparison. There is little reason to deviate from the default ‚Sharp‘ setting. Outputs 1&2 and 3&4 do not differ measurably. Now, let‘s conclude the measurements with the two headphone amplifiers. The maximum output power of the headphone amplifier is 184 mW at 32.5 ohms when the output is set to +16 dBu. However, the manufacturer warns with an information panel about high volumes when using low-impedance (<200 ohms) headphones. Switching to +8 dBu will result in a maximum output power of 110 mW for those who want to play it safe. The following measurements were still performed at the higher setting. The noise level of the headphone output is -104 dBu RMS unweighted (20 Hz to 20 kHz), resulting in a total dynamic range of precisely 114 dB. The noise spectrum in Diagram 11 is again completely free from disturbances, and the DC offset level is at 2 mV, as indicated by the quasi-peak value of -93 dBu. The THD

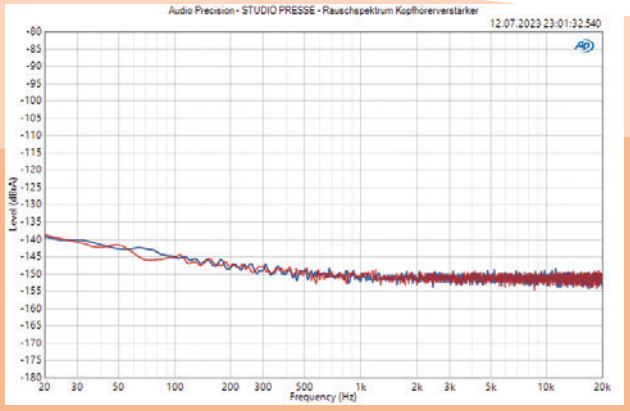


Diagram 11: Noise spectrum of the headphone amplifier

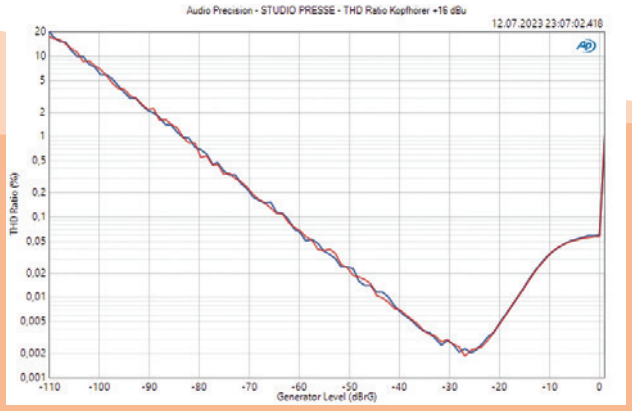


Diagram 12: THD ratio response of the headphone amplifier over output level

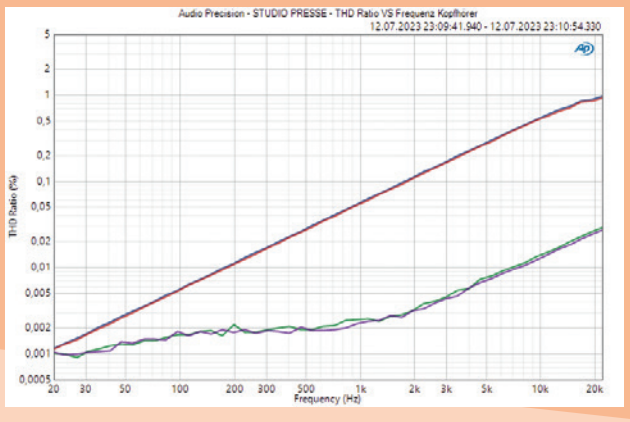


Diagram 13: THD ratio response of the headphone amplifier over frequency at maximum output level 0 dBr (blue and red) and -27 dBr (purple and green)

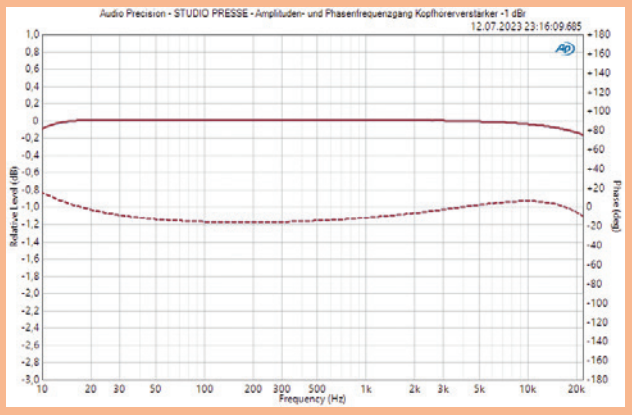


Diagram 14: Amplitude (solid line) and phase frequency response (dashed line) of the headphone outputs at 48 kHz sample rate

ratio in the upper operating range is 0.056%, slightly above our threshold of 0.05%. Its best value is achieved at around -27 dBr, which is in the critical range for ‚normal' monitoring levels. An interesting observation is also the THD ratio's frequency response. Diagram 13 displays the comparison between full-scale output and -27 dBr. Finally, let's look at the amplitude and phase frequency responses in Diagram 14. The overall verdict of the measurements is good, leaning toward very good. The offered dynamics of the inputs are beyond any doubt. However, the distortion values cannot quite keep up. Therefore, compared to a stationary device, there are still some minor limitations, but customers who choose the MT48 opt for very high-class technology.

Hands-on and Listening Test

Operating the device is quick and intuitive. Most functions are where you would expect them to be. However, some menus are quite nested, so you may occasionally get a bit ‚lost.' By pressing the large diamond-shaped button, you can always return to the default screen or, in some situations, simply go back to the previous page. The level metering is well executed; the switch states and configurations, in particular, are clearly displayed. The only slightly unusual aspect is the display of level reduction in the dynamics section, which changes depending on the output level. Channels and buses can be assigned names and colors, significantly enhancing clarity.

The current settings can be saved using snapshots and recalled separately for individual sections. Unlike Anubis, there are currently no application-specific ‚missions.' This means that the configuration and operation of the device cannot be switched between different application modes. Instead, the workflow is similar to the ‚Music Mission' and is exclusively limited to stereo operation. For most users, this is not a limitation. Sonically, the MT48 absolutely lives up to its promises. Recording with the microphone inputs is simply enjoyable. In a direct comparison to our Hapi converter, we could hardly detect any tonal differences; the two perfectly complement each other. And, of course, that's also a seal of approval for the compact interface, as keeping up with a Hapi is quite a feat. The sound is immaculately clean, well-defined,

and completely noise-free. Due to the extremely low noise floor, gain-staging doesn‘t need to be perfect. It’s ok to leave some headroom, which can be a lifesaver, especially in live recording situations. The stereo image is crystal clear, allowing for perfect signal localization, and one gets the impression that the input stages are genuinely bringing out the best in the connected microphones. It‘s a lot of fun and gives beginners the peace of mind to worry about one less ‚problem.‘ The same applies to the front instrument inputs, by the way. Our SG guitar sounds punchy and rounded, with clean transients on the attack. We had the impression that the lower mids are slightly restrained, which only adds to the clarity. Other DI boxes or instrument inputs may have a bit more body, which can sound pleasant but, as always, depends on the instrument used. In any case, the DI functionality meets the device‘s demands. The outputs are also beyond reproach. In the test, we not only compared the monitor output with our Hapi but also inserted our mastering chain through the remaining outputs. The result was an excellent little mastering setup that manages to showcase the advantages of analog processing without obvious drawbacks from additional conversion. That‘s precisely why we chose the Hapi back then, and we would do the same with the small Neumann interface today. Listening to the golden ‚box‘ is truly a pleasure. The headphone amplifier sounds good and is low in noise; however, we did notice some slight limitations compared to our reference device. Our Funk MTX offers even more clarity and transient accuracy, especially at higher volumes with our Dan Clark Audio Aeon headphones.

The reverb effect cannot be recorded. This is not a significant issue, as it primarily serves for monitoring purposes. It’s also evident in its sound - even though it is adequate for creating a more pleasant acoustic environment for a singer, it does not match the quality of high-end plug-ins. The reverb tail sounds metallic and audible modulations limit its aesthetic appeal. Today, the quality of even budget-friendly effects in DAWs is so high that such ‚detours‘ via a DSP only seem worth it if something truly exceptional is offered. Sonically, the equalizer and compressor are also ‚standard fare,‘ useful as problem-solvers during recording, but they don‘t provide substantial added value beyond that. Removing some troublesome resonances with the EQ and tightening the monitor signal with the compressor is no problem and is quick to set up. However, recording with these effects should generally be avoided, as there are few technical advantages, and you become tied to the intrinsic sound of both processes. Nevertheless, these minor criticisms are of little concern, as the MT48 shares these characteristics with almost all other audio interfaces on the market. For monitoring purposes, the offered effects are usually more than sufficient.

One question that naturally arises: Can you hear the fan? The answer is a definite ‚Yes.‘ The system offers three cooling levels to choose from. At the lowest level, the fan remains off for a long time and only turns on in extreme situations. However, the enclosure becomes quite warm in this mode. The manufacturer‘s manual assures that there is no need to worry about this since the case essentially acts as a heatsink and is allowed to get warm. However, when the fan does eventually turn on, it is clearly audible. With such a small fan, it‘s unavoidable and can be quite annoying, as is often the case with many laptops. There is an option for the fan to turn off automatically when the talkback button is engaged so it doesn‘t bleed into the headphones, which is a helpful feature. The Remote Control app has proven to be immensely useful for operation. It mirrors the touchscreen menus, which is great because all functions can be found in the same place. But there are also slight drawbacks since the operation was initially designed for touch input and not

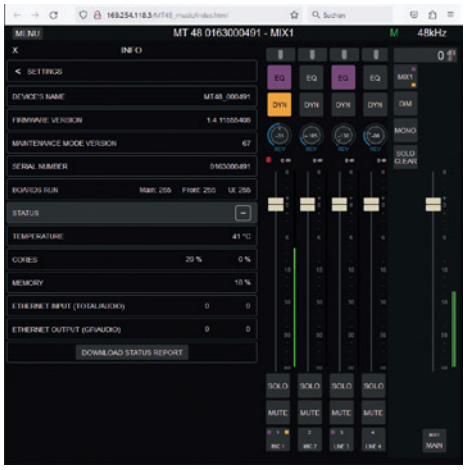
The Remote Control-App window closely resembles the device‘s touchscreen display

Here, EQ settings can be conveniently adjusted...

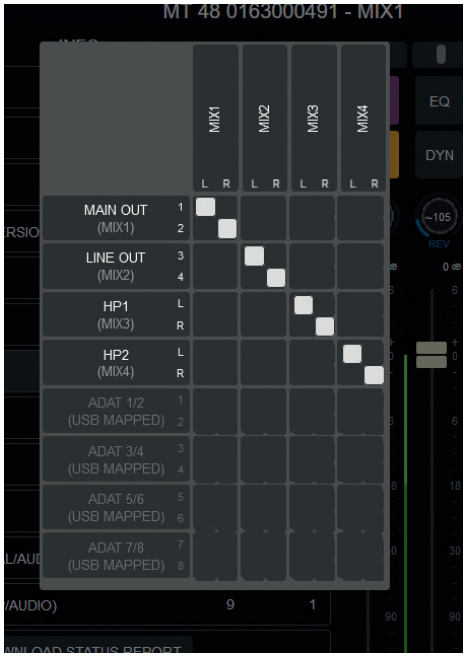
...and dynamics can be easily controlled

Snapshots can be saved and recalled through the Snap menu. During recall, a filter function (shown in the image) allows the selection of which parts should be loaded

The web browser view offers a better overview



Additionally, there are expanded system information details



An example view of the Mix Bus routing

the mouse. This leads to the intuitive use of the mouse wheel for scrolling in the app, which, unfortunately, doesn't work as intended. Instead, it inadvertently adjusts the monitoring level, as the mouse wheel is always linked to the same parameter that the hardware encoder on the device controls. To scroll, you have to ‚swipe' with the mouse instead. One quickly gets used to the operation and gets along well with it. Operating through a web browser is very practical since it can be done from any device on the network. This means you can use your tablet, phone, or laptop as a control device on the side. As a self-producing musician, you can adjust while standing in front of the microphone away from the DAW. But in general, there is value to having a browser link, as many more parameters can fit simultaneously on a large computer screen compared to the touch display.

In the broadcasting world, Audio-over-IP takes center stage today, so naturally, we wanted to see how the MT48 performs in this regard. The network streams utilize the Ravenna protocol but can easily be configured to meet the requirements of AES67, making them compatible with other audio network participants supporting this standard. As we have an extensive AES67 network in place, we were able to test this functionality effortlessly. Inputs and outputs are configured through a dedicated website accessed via a web browser. This interface is consistent across all Merging devices, enabling users to find their way around immediately. As a result, the setup is straightforward. All physical inputs are presented in a matrix and can be easily assigned to network streams. Four stereo paths are available for transmission to the DAW, allowing the network streams to be recorded. However, sources from the DAW coming through USB cannot currently be routed to the network right away. For this purpose, the detour via the mixers is necessary, which unfortunately restricts the functionality as a network front-end for the DAW. However, Neumann has already informed us that future firmware updates will expand this aspect. The streams can be manually configured, connected via SDP, or discovered on the network using the SAP protocol. In the simplest case, the interface locates the desired streams in a dropdown menu populated with sources made known via SAP. The MT48 also supports NMOS IS-04 and IS-05, enabling connections to be configured through a central controller, such as Nvidia's free NMOS controller, or more comprehensive solutions like VideoIPath. All of this works seamlessly, making the MT48 a pleasantly uncomplicated network participant.

The integration benefits from Merging's experience with network connectivity, as the manufacturer was among the very first to offer Ravenna. By the way, those planning to use the MT48 with an audio network should consider downloading Merging's helpful software tools. For example, Aneman and MTDiscovery make configuration remarkably easy and swift, allowing users to work from a central instance instead of managing each participant individually.

Conclusions

The decision for Neumann to partner with Merging for their first standalone audio interface was a smart move. With Anubis, Merging had an arrow in their quiver that perfectly met the quality demands of the renowned Berlin-based manufacturer. Neumann's modifications precisely target the intended market and provide genuine added value for music production customers. The MT48 sounds excellent, performs reliably, offers flexible operation, and holds significant potential for future expansions. Oh, and it looks fantastic too. What more could one ask for? In our book, they could hardly have laid a better foundation for the collaborative future of the colleagues from Puidoux and Berlin!