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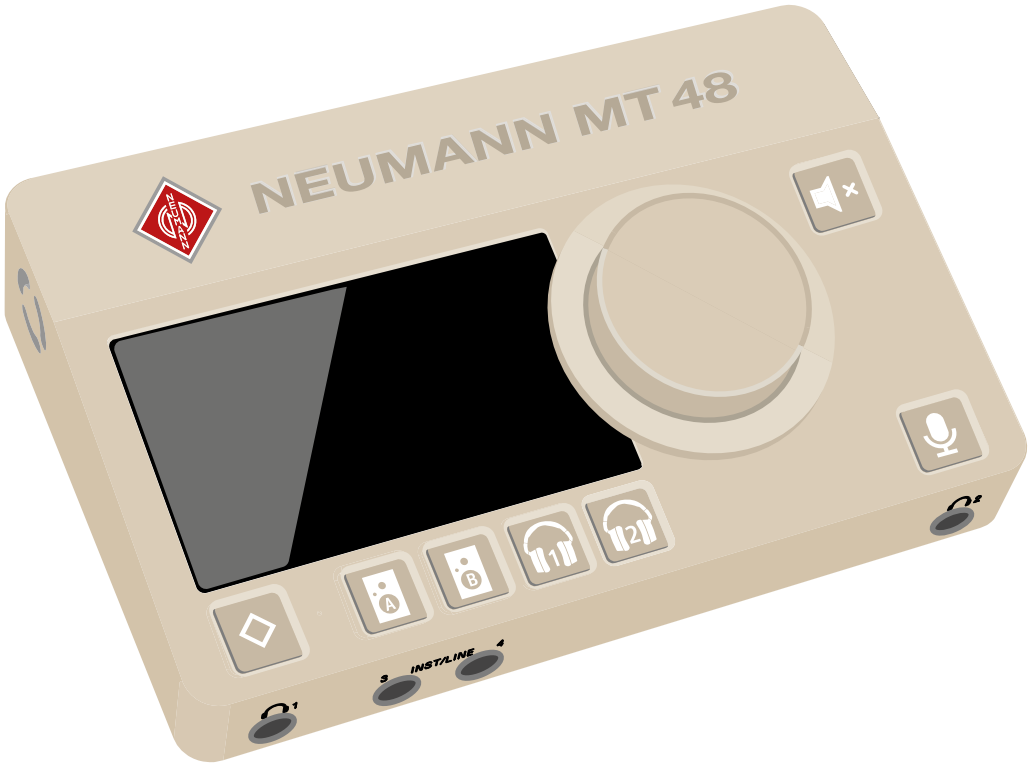
MT 48

AUDIO INTERFACE

A brief introduction



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Preface: Mission Statement

The MT 48 brings reference-class sound to your desktop. It is the first audio interface to unlock the full potential of Neumann microphones, studio monitors, and headphones as well as high class gear from other manufacturers. The MT 48 builds upon the knowledge and experience of our friends, partners and now colleagues from Merging Technologies, famous among recording professionals for state-of-the-art AD/DA converters and networked audio solutions. The MT 48 is the perfect marriage of Neumann sound quality and Swiss precision.

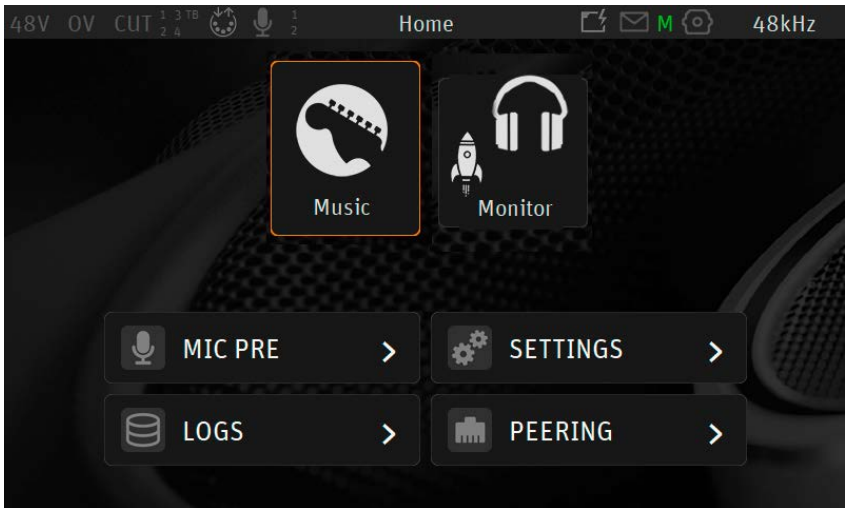
Analog-to-digital conversion in the MT 48 sets a new benchmark. With an enormous dynamic range of 136 dB, the MT 48 has four times the resolution of competing devices. Analog circuitry is on the highest level, too. The microphone preamps offer up to 78 dB of clean gain for capturing every detail from any microphone, be it high sensitivity condensers or low output ribbons. The same philosophy applies to the built-in headphone amplifiers, which are extremely powerful and ultra-low impedance to drive any headphones to their absolute best.

Just as important as sound quality is monitoring! Not only for the engineer but also for the artist. Musicians play better, singers sing better with inspiring sound in their headphones. That's why the MT 48 is also a fully-fledged mixing console with sophisticated EQ and dynamics in every channel plus a nice sounding reverb processor. Despite this complexity, the MT 48 remains easy to use, thanks to an intuitive touchscreen user interface: Adjust levels and panning at the touch of a finger. No more shuttling between the audio interface and a controller app on your computer: All parameters can be set on the device itself.

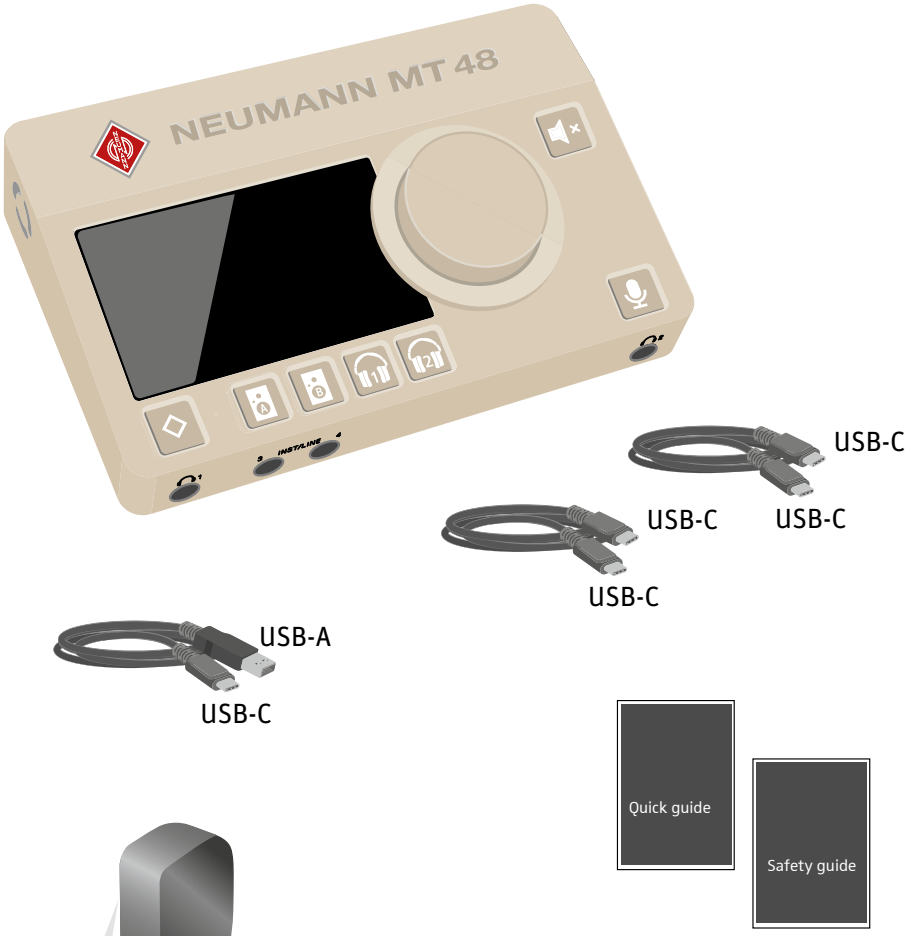
The MT 48 is thus the ideal companion for self-recording musicians. But it can handle large setups just as well, as it offers two expansion options, ADAT and AES67. While ADAT has long been a standard in project studios for adding up to eight I/O channels, AES67 is a modern open standard for networked audio with enormous capabilities. From the proverbial bedroom studio though location work and broadcast applications to professional recording, the MT 48 always offers outstanding performance. Whatever you're up to, the MT 48 is ready for it.

The MT 48 supports two Missions. The Music Mission (default ex-factory) and the Monitor Mission (released in February 2024).

The Music Mission oriented towards a music, recording, bands and studio projects, while the Monitor Mission supports multichannel and is the perfect Monitor Controller for mixing and mastering. Refer to the Monitor Mission Manual for specific details on the activation and operations.



What‘s in the box?



The MT 48 comes with two USB-C to USB-C cables and one USB-C to USB-A cable as well as an external power supply including adapters for different countries.

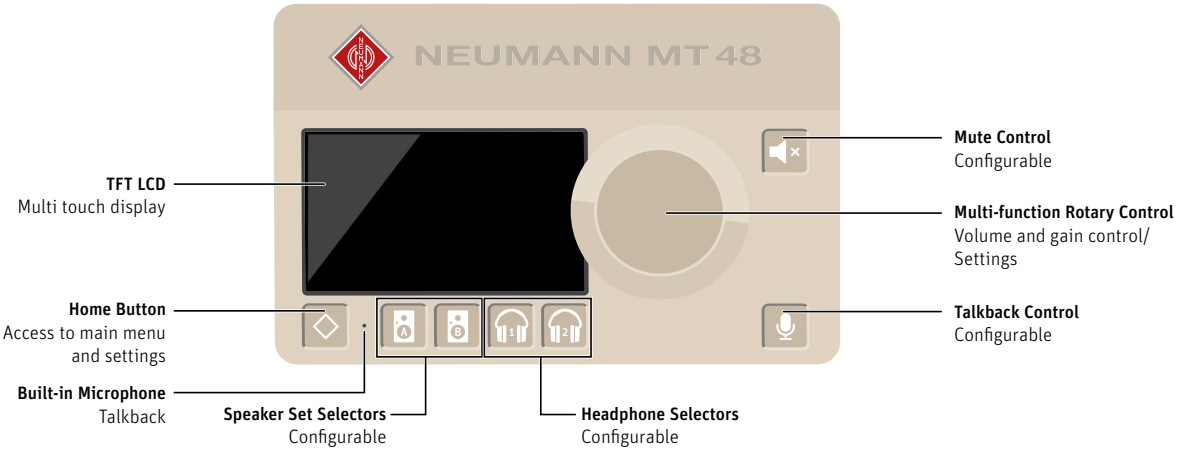
Please read the Quick Guide and Safety Guide before using the MT 48.

TRS to DIN5 MIDI adapters are available as an optional accessory.

System requirements

Windows	Mac
Windows 10 64-bit or Windows 11 64-bit	macOS 10.15 to 15.x (Catalina – Big Sur – Monterey – Ventura – Sonoma – Sequoia)
Software Requirement: MT 48 Toolkit must be installed.	Intel and Apple Silicon platforms supported
	Driverless: Plug and Play supported on macOS.
	Installation of MT 48 Toolkit required for full functionality.

Overview



- **TFT LCD**: The MT 48 features a capacitive touch display.
- **Home button** with the Neumann diamond for navigation purposes.
- **Multi-function Rotary Control** for parameter value changes.
- **Selector Buttons**: Pressing the Speakers A, Speakers B, Headphone 1 and Headphone 2 buttons selects one of four separate Mixers for each output pair and assigns the large Rotary Control to volume control for the selected stereo output.
- **Mute Control**: Pressing the mute button silences the selected stereo output, pressing the mute button for more than 2 seconds mutes all outputs.
- **Talkback Control**: A Talkback function is integrated. Pressing the Talkback button activates the built-in Talkback Mic, which is below the pinhole* located between the Home and Speakers A buttons.

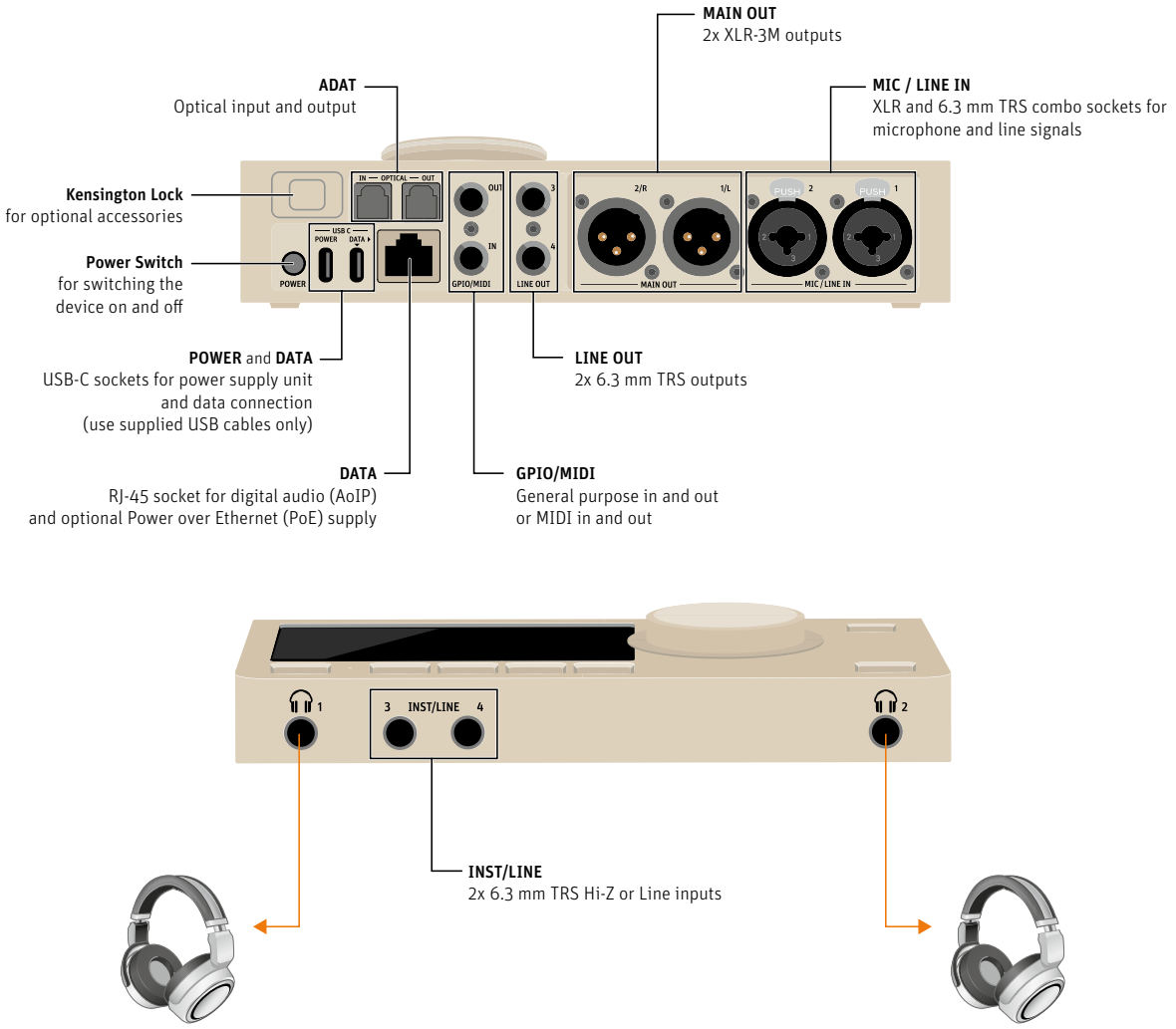
CAUTION
Damage to the product

Avoid touching the built-in microphone or applying pressure while engaged.

▶ Do not insert any object or liquid into the hole, or use a vacuum over the mic hole as it could permanently damage the microphone.

- **Bottom Panel**
  3/8” thread for mounting the MT 48 on a mic stand

Inputs and Outputs



Analog I/O

The MT 48 has four analog inputs and eight analog outputs. The latter are configured as stereo pairs.

- **Mic/Line Inputs 1+2** are located on the back of the unit as combo jacks. Use the XLR part for microphones and use the TRS part for line sources.
- **Instrument/Line Inputs 3+4** are located on the front of the unit as a matter of convenience, when you plug in a guitar or bass. These inputs can also be used for line signals.
- **Main Outs 1+2** with XLR connectors are located on the back. Usually, these will be connected to your studio monitor loudspeakers.
- **Line Outs 3+4** with TRS connectors are located on the back of the unit. These can be used for a secondary pair of monitor loudspeakers or other purposes, such as sending a line signal to an external headphone amplifier in the live room or vocal booth.
- **Two stereo headphone outputs** are located on the front. Headphone output 1 and 2 are independent of each other and can be addressed separately.

Digital I/O

In addition, the MT 48 may be expanded via ADAT / S/PDIF optical and AES67

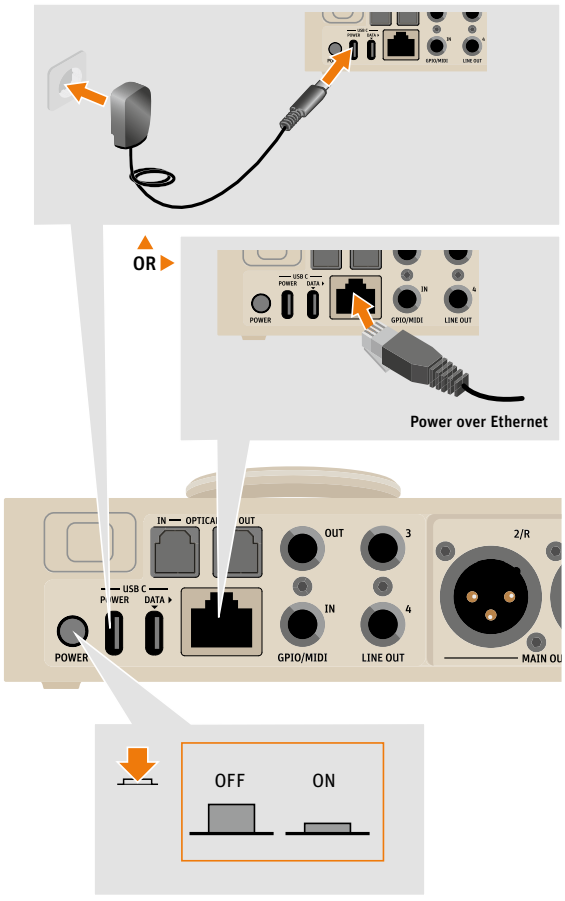
- **ADAT** allows for adding up to eight input and output channels via dedicated converters. ADAT channel count depends on the sampling rate used: 8 channels up to 48 kHz, 4 channels up to 96 kHz and 2 channels up to 192 kHz. The optical connectors can also be used in S/PDIF mode for stereo I/O up to 96 kHz.
- **AES67** is an open standard in networked audio, which has been embraced by the broadcast market and is now gaining traction in the world of recording as well. AES67 supports high channel counts even at double and quad sampling rates. The AES67 port is bidirectional, so input and output devices can be connected to the MT 48 at the same time using a standard ethernet switch.

GPIO/MIDI

- These two TRS connectors can be used in GPIO mode for switching, e.g. connecting a foot-switch or remote switching a recording light indicator. Or you can use them in MIDI mode as MIDI in and out, using TRS to DIN5 adapter cables (sold separately).

Powering

- Please use the included power supply and connect it to the USB-C port labeled POWER using one of the cables provided.
- Alternatively, the MT 48 also works via Power over Ethernet with switches that support the PoE+ standard.
- Switch the device on/off using the power button.



Register your product

To register your MT 48, please go to **mt48.neumann.com**

Driver installation

Although the MT 48 works on Mac systems without driver installation, you need to install the MT 48 Toolkit for full access to all of its features. On Windows systems, installing the MT 48 driver is mandatory.

- Go to mt48.neumann.com and download the installer package to your computer.
- Double click the installer for your computer operating system and follow the on-screen instructions.
- **Important note for Mac users**: macOS requires the drivers to be approved. During the installation, you will have a “System extension blocked” message, go in the Security preferences to unlock the driver.
- After restart, you will notice the Neumann logo of the MT 48 AGENT in the taskbar/menu bar of your computer.
- If your MT 48 requires a firmware update, see Appendix “Updating the firmware” on page 40

Setup

- ▶ Place the MT 48 on a hard surface. Make sure it stands on its rubber feet to allow airflow to the MT 48’s bottom plate.
- ▶ Connect the power supply to the USB-C port labelled “Power”
- ▶ Connect the MT 48’s USB-C port labelled “Data” to your computer using the supplied cables (USB-C to USB-C or USB-C to USB-A).
  The MT 48 uses the High Speed protocol supported by USB 2.0 or higher ports. Should the MT 48 not work properly on a USB 3.0 (or higher) port, please use a USB 2.0 port. Some non-Intel USB 3.x chipsets are not fully backward compatible for use with audio devices.
- ▶ Press the Power button on the MT 48. Booting takes about 60 seconds.
  After a few seconds, the Neumann symbol of the MT 48 Agent in the taskbar/menu bar of your computer becomes solid to show that the MT 48 is connected and can be used.

Connecting your gear

The MT 48 includes a sophisticated mixer for each of its four output pairs, so you do not need a monitor controller.

- ▶ Connect your main speakers directly to the Main Outs (XLR, balanced).
- ▶ If you have a secondary set of speakers, connect them to Line Outs 3+4 (TRS, balanced)
- ▶ Connect your headphones to the headphone outs 1 and 2 on the front. Both outputs can be addressed separately and have their own mixers.
- ▶ Connect microphones or line devices to Mic/Line Inputs 1+2 (XLR/TRS combo, balanced) For microphones, use XLR cables only. For line sources use the TRS input.
- ▶ Connect your guitar or bass to Line/Instrument Inputs 3+4 (TRS) on the front. These inputs have a very high impedance (1 MΩ unbalanced, 2 MΩ balanced) to capture the tone of magnetic pickups without losses. But they can also be used for low impedance sources such as electronic keyboards, drum machines or line level studio gear.
- ▶ GPIO/MIDI: Per default these ports are set to MIDI mode. Connect Synths, drum machines and other MIDI gear using TRS to DIN5 adapters (sold separately). Alternatively, you can use these ports for various switching functions, e.g. to connect a footswitch for activating talkback.

Adding I/O via ADAT / S/PDIF and AES67

The MT 48 offers three ways for adding more I/O channels: I/O via ADAT / S/PDIF, AES67 and Dante

ADAT

- ADAT is an established standard used predominantly in project studios. ADAT allows for up to 8 channels via optical connectors at sampling rates up to 48 kHz. At higher sampling rates the channel count reduces to 4 channels at double sampling rates up to 96 kHz and 2 channels at quad rates up to 192 kHz.
- Alternatively, the optical I/O can be used in S/PDIF mode, allowing for 2 channels at all sampling rates up to 96 kHz.

AES67

- AES67 is an open standard for audio-over-IP networks which ensures interoperability between various audio network protocols such as DANTE, RAVENNA. AES67 was developed by the Audio Engineering Society (AES) and has been embraced by the broadcast market but is gaining more traction in recording environments as well. AES67 uses ethernet connections and can handle high channel counts even at high sampling rates. The AES67 port on the MT 48 is bidirectional, it can send and receive audio data simultaneously. Using a standard ethernet switch you could connect a Neumann AES67 loudspeaker and Merging converters at the same time. Note: The MT 48 supports the RAVENNA protocol, which is fully compliant with AES67 and SMPTE ST 2110.
- Peering is a particularly convenient form of AES67/RAVENNA. Currently this only works with devices by Merging Technologies (who are now part of Neumann), such as Hapi, Horus, and Anubis. These high-end converters can be seamlessly integrated via Peering for I/O expansion and remote controlling their preamps and other functions from the MT 48.

Dante®

- Dante is a network protocols that delivers uncompressed, multi-channel, low-latency digital audio over a standard Ethernet network using Layer 3 IP packets. Developed in 2006 by the Sydney-based Audinate, Dante builds on previous audio over Ethernet and audio over IP technologies.
- Latest generation of produced MT 48 serial numbers 10 000 and above are Dante Ready™. A Dante Ready device allows you to purchase and add audio channels to a device after initial purchase using the Dante Activator feature in Dante Controller.
- Ex factory, the MT 48 continues to be delivered with the RAVENNA/AES67 firmware preinstalled. For Dante operation, Neumann provides an alternative Dante firmware free of charge. In addition, an 8x8 or 64x64 I/O Dante channel license is required, which must be purchased from Audinate. This ensures there are no additional costs for users who do not require Dante.
- For more information, please follow our Dante Appendix guide or visit our Dante FAQs.
- Owners of an MT 48 with a serial number below 10000 who require Dante compatibility can have their device made Dante Ready™ as a retrofit. This requires a hardware modification, which Neumann Service will perform for a fee.

Configuring the MT 48 to your needs

USB I/O

The MT 48 driver supports up to 32 input and 16 output channels at sampling rates up to 48 kHz.

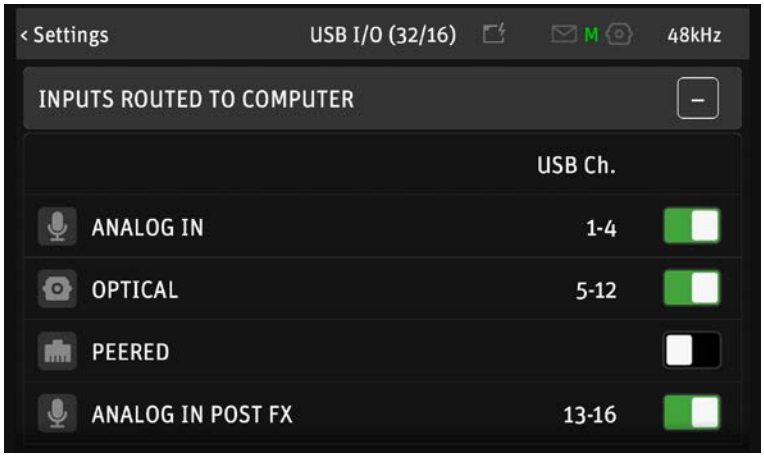
At higher sampling rates up to 96 kHz, the channel count reduces to 16 input and 16 output channels

At highest rates up to 192 kHz, the MT 48 driver can handle up to 8 input and 16 output channels.

Under **SETTINGS → USB I/O** you can select which channels are sent to and received and from the computer via USB. Channel numbers are assigned dynamically.

This menu has three sections:

Section 1: Inputs Routed to Computer



- **Analog In**: these are the two Mic/Line Inputs and the two Line/Instrument Inputs.
- **Optical**: the ADAT or S/PDIF input. Activate this only if you have ADAT or S/PDIF devices and want to record their output. If you use optical devices only for monitoring (e.g. a CD player for reference tracks), there's no need to activate it as a recording input.
- **Peered**: This only applies if you use AES67/RAVENNA gear that is Peering enabled, e.g. Hapi, Horus, or Anubis converters from Merging Technologies.

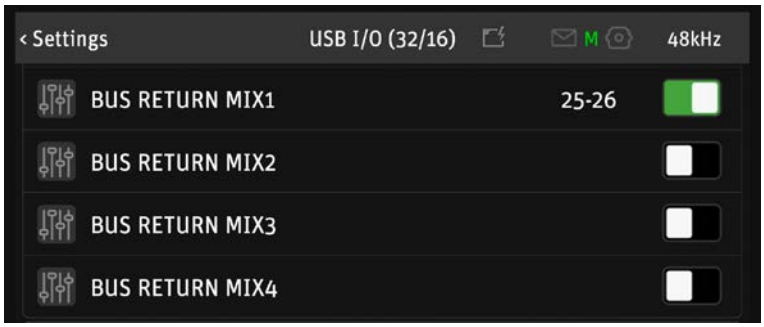
Analog, Optical and Peered Inputs can also be recorded Post FX, i.e. including the EQ and Dynamics processing in their respective channels:

- **Analog In Post FX**
- **Optical Post FX**
- **Peered Post FX**

Having this option on separate driver channels allows you to record the same source with and without processing at the same time, if you wish. That way you can use the on-board processing for creative sound shaping and record a raw backup on a separate track, in case you decide otherwise.

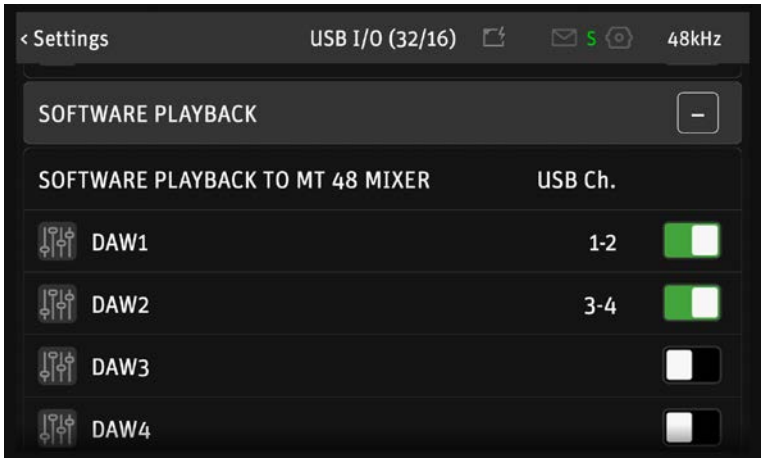
If you don't intend to record sources with processing included, you can simply uncheck the Post-FX entries. Conversely, if you like to record processed signals only, you can uncheck the unprocessed inputs and only select their Post FX counterparts.

- **Built-in Mic** – activate this only if you intend to record the Talkback Mic in your session. Deactivating this entry does not deactivate the Talkback function as such. Please note: Input 4 is shared with the Built-in Talkback. Connecting an instrument or line source to Input 4 will deactivate the built-in Talkback (channel 5).
- **Bus Return Mix 1/2/3/4** – This allows you to record the output of one (or more) of the internal mixers, including signal processing in the input channels and even the reverb effect. A typical use case would be streaming. For instance, you could plug in a vocal microphone and a guitar for a musical performance, apply EQ, Dynamics and Reverb and plug in another microphone for the show host, without Reverb. Routing Bus Return Mix 1/2/3/4 to your streaming application, your listeners will hear the entire mix, including effects.



Please note: It is a good idea to uncheck any channels you are not going to use. This gives you a cleaner workflow in your recording software.

Section 2: Software playback
Software playback to MT 48 mixer



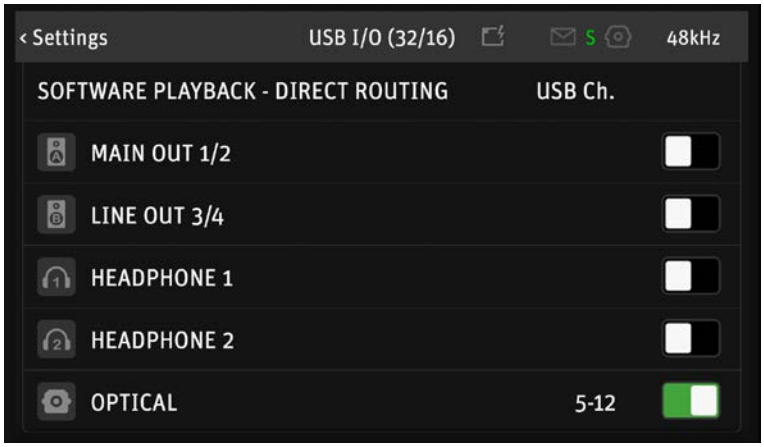
DAW1, **DAW2**, **DAW3** and **DAW4** are stereo channels from your computer software to the MT 48. Per default, DAW1 and DAW2 are activated. These channels appear as stereo channel strips in your MT 48 Mixer. This allows you to create monitoring mixes with the live inputs balanced against the DAW signal. DAW1 is also where you receive system sounds and the audio output from consumer software such as media players.

Having more than one DAW channel in your mixer is handy for many applications:

▶ Put a click track on a separate DAW channel

- Use DAW1 for system sounds and consumer software only and use DAW2 for pro audio applications
- Create stems on up to four DAW channels for the artist to create his/her own mix from melodic instruments/vocals/bass/drums
- Mix reference: Compare your DAW Mix with tracks played out from streaming/media player software.

Section 3: Software Playback - Direct Routing



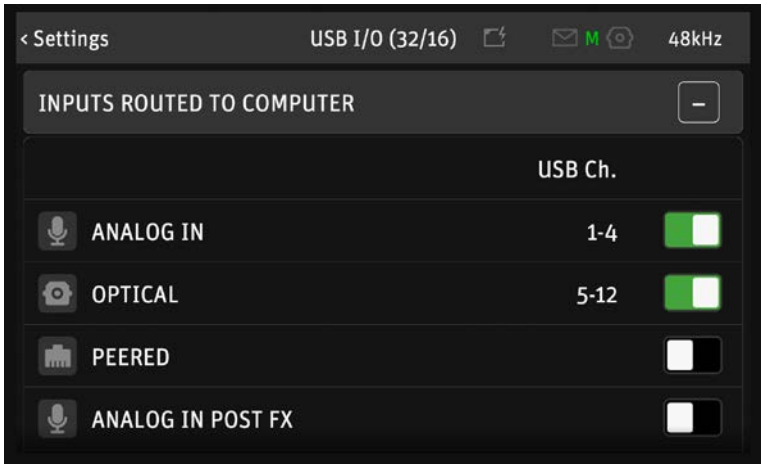
Channels activated in this section bypass the MT 48 Mixer. Usually, that's NOT what you want. But it is a useful option in some scenarios, for instance:

- If you want to manage your monitoring in the DAW instead of the MT 48. Direct Routing allows you to address Headphones and speakers without going through the MT 48 mixer. Keep in mind, though, that the MT 48 Mixer offers much lower latency compared to DAW monitoring.
- If you want to use external outboard as hardware plug-ins in your DAW. In this case, bypassing the MT 48 mixer via Direct Routing makes a lot of sense.

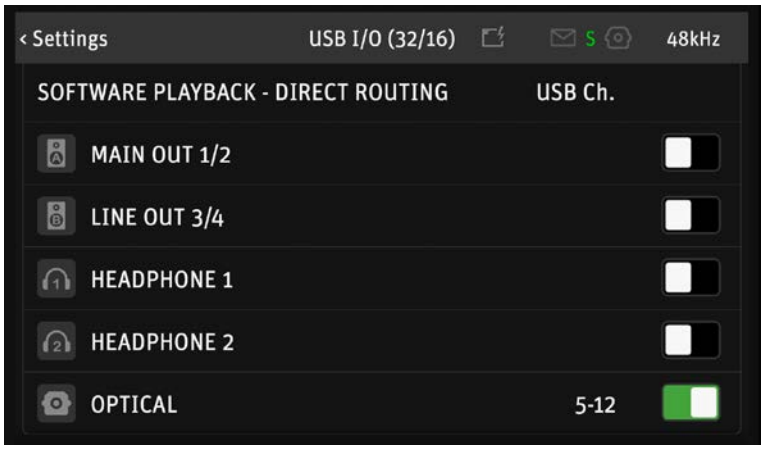
Routing example: Using external outboard as hardware plug-ins in your DAW

Let's say you have an ADAT AD/DA converter that you want to use to integrate analog outboard into your workflow as" hardware plug-ins" in your DAW. (You could do the same with analog inputs/outputs 3+4 or Peered inputs/outputs).

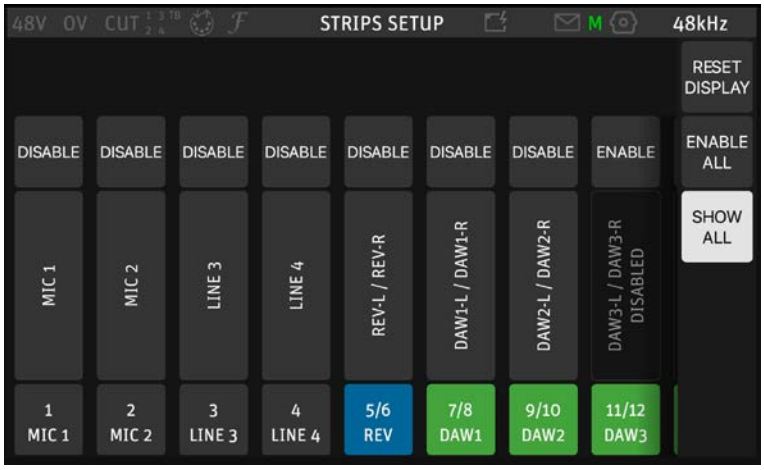
1. Connect the outputs and inputs of your ADAT converter to the inputs and outputs of your outboard gear.
2. Go to SETTINGS → USB I/O → INPUTS ROUTED TO COMPUTER and activate OPTICAL. Your ADAT Inputs can now be addressed via the channel numbers displayed.



3. Go to SETTINGS → USB I/O → SOFTWARE PLAYBACK – Direct Routing and activate OPTICAL. The ADAT Outs can now be addressed directly in your DAW via the channel numbers displayed.



4. In the MT 48 Mixer, remove the ADAT Input channels. Go to MENU → STRIPS SETUP and disable all ADAT channels and clear the ADAT Group.



Now you can address outboard gear connected to your ADAT converter via driver I/O channels.

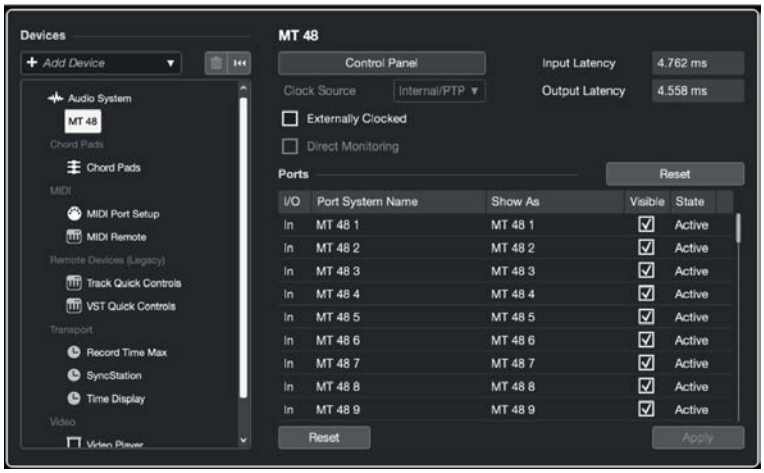
DAW setup

When you start your recording software or DAW, as it is usually called (DAW = Digital Audio Workstation), it will usually detect your new audio interface and ask which driver to use. Depending on your computer system and recording software, there may be more than one choice. Select the MT 48 driver. On Windows systems, most DAWs rely on the ASIO protocol. ASIO was developed by Steinberg specifically for audio processing with low latency – which is crucial for using virtual instruments such as softsynths and realtime audio processing (e.g. amp modeling). ASIO offers much better performance than Windows system drivers, so almost all manufacturers have adopted Steinberg's ASIO driver protocol.

On the Macintosh, system drivers (CoreAudio) offer excellent low latency performance. Some manufacturers, however, still use ASIO. In practice, this makes little difference. Depending on your DAW software, you may have two driver options: "MT 48" and "MT 48 High Precision". Either one should work fine. The High Precision driver works with 32 Bit resolution, which isn't necessary since the normal driver's 24 Bit resolution covers a dynamic range of up to 144 dB – enough for the MT 48.

DAW buffer size / latency settings

To adjust the Latency settings in your DAW, go to Preferences or Studio Setup – please refer to your DAW manual. There will be button to open the Control Panel for the MT 48 that allows you to specify the Buffer Size.



Selecting a small buffer size will give you lower latency, but it will also increase the CPU load. Selecting a larger buffer size will reduce the CPU load but results in higher latency. A good starting point is usually 64 samples. This gives you low enough latency for a direct response when you play softsynths while maintaining good CPU performance.

Keep in mind that low latency is only needed for realtime processing, e.g. playing virtual instruments. There is input latency and output latency. Output latency is the delay between hitting a key and the resulting sound. Input latency is irrelevant when you play virtual instru-

ments as the input is MIDI data. Input latency only comes into play when you process an audio input signal in real time. For instance, when you plug a guitar into the MT 48's instrument input and process its audio signal with amp modeling software.

Please note that the latency/buffer size settings only relate to your DAW. **The monitor mixers inside the MT 48 always work with ultra-low latency, regardless of the DAW settings.** So, you do not need a small buffer size for audio recording or mixing. You only need low latency in when playing virtual instruments or realtime processing an input signal with plug-ins in your DAW.

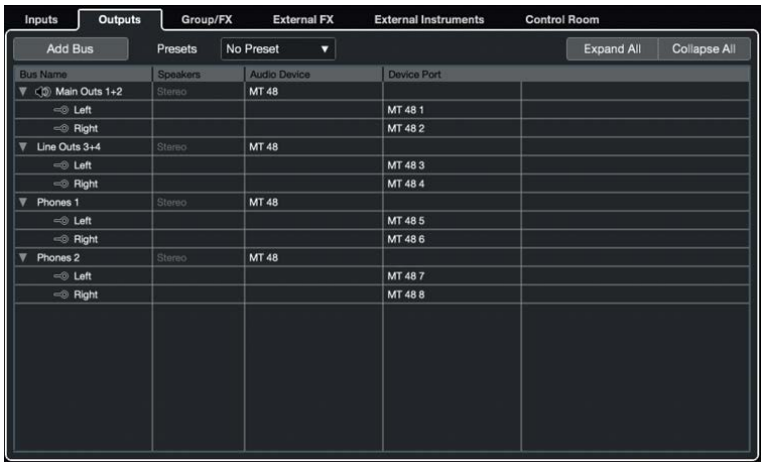
Setting up inputs and outputs in your DAW

In most DAWs inputs and outputs must be activated before you can use them. This is usually done in Preferences or I/O Setup or Audio Connections – please refer to your DAW manual.

Usually, it is best to setup inputs as stereo pairs as you can use stereo inputs to record mono tracks (by specifying the left or right channel), but you cannot use mono inputs to record stereo tracks.

Create stereo inputs for all hardware inputs you previously activated in MT 48 USB I/O SETUP. Channel naming is not transmitted to your computer. To avoid confusion, it's a good idea to name the channels in your DAW accordingly. This is what it looks like in Steinberg Cubase/ Nuendo 12; other DAWs work similarly:

Do the same for the Output channels:



Now the correct names will be displayed whenever you select an input or output in your DAW.

Working with the MT 48

Understanding the mixer structure

The MT 48 has a separate mixer for each of its four output pairs:

- Mix1 → Speakers A
- Mix2 → Speakers B
- Mix3 → Headphone 1
- Mix4 → Headphone 2

Pressing the hardware button for SPEAKERS A, SPEAKERS B, HEADPHONE 1, or HEADPHONE 2 will automatically display the respective Mix on the touchscreen and assign the Rotary control to the output level (volume) of the selected output.

If you do not want or need separate mixes for each output pair, you can assign MIX2, MIX3 and/or MIX4 to follow MIX1.

▶ Go to MENU → SETTINGS → MONITORING and scroll down to the bottom of the page:

Set like this, SPEAKERS B and HEADPHONE 1 get the same mix as SPEAKERS A while HEADPHONE 2 can have a separate monitoring mix for a musician or singer.

Mixes following MIX1 still have their own output level setting. So when you press the hardware buttons for SPEAKERS A, SPEAKERS B, HEADPHONES 1, or HEADPHONES 2 you can adjust their output volume individually (including MUTE).

Pressing the Home Button opens the alternate mixer view.

Each input channel has inserts for EQ and Dynamics processing. EQ and DYN in color indicate that processing is activated. The Analog Inputs as well as Peered Inputs also have a PREAMP page.

Please note:

- PREAMP, EQ, and DYN are set for all channels at the same time.
- The same goes for the REV SEND level, as there is only one reverb processor for all four mixers. However, you can adjust the overall reverb level in each mixer using the REV CHANNEL FADER.
- FADER LEVELS and PAN positions can be adjusted individually for each Mixer

Preamp

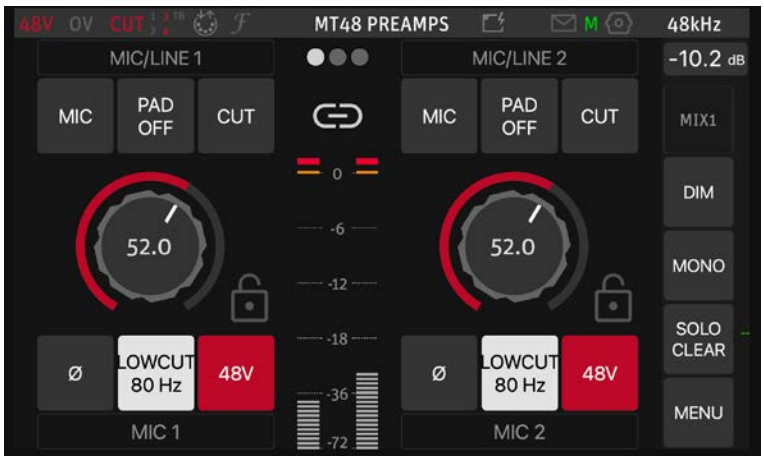
Pressing the PREAMP button in the alternate mixer view opens the PREAMP page.

Alternatively, the preamp (as well as many other functions) can be accessed by pressing on the channel name. This opens a menu; select PREAMP.

Each preamp has a gain knob as well as several additional functions.

- MIC/LINE (Inputs 1+2) switches between Microphone and Line mode. INST/LINE (Inputs 3+4) switches between Instrument and Line mode.
- PAD (Mic mode only) allows to attenuate the input signal by 12 or 24 dB to accommodate high input levels.
- CUT mutes the input.
- Ø reverses the polarity of the input signal (often called “phase reverse”)
- LOW CUT activates a filter to combat rumble. The cut-off frequency is selectable: 40/60/80 Hz. Please note that the low cut filter always affects the recorded signal, even on the Pre-FX driver channels.
- 48V (Mic mode only) activates 48 Volts phantom power for condenser microphones (and other microphones with active electronics)
- GAIN is adjusted by selecting the gain knob and then turning the Rotary Controller. It’s good practice to leave at least 10 dB headroom. The Meter in the middle has a peak hold function, which can be reset by tapping the Meter section.

For stereo operation, two channels can be linked by pressing on the chain symbol above the level display. Any offset in the gain setting is maintained, which allows you to compensate differences in sensitivity.



Settings can be locked by clicking on the lock symbol next to the gain knob.

Tip: You can go to the neighboring channels’ preamp page by swiping to the left or right.

EQ

The MT 48 mixer features a sophisticated 4-band equalizer on every channel. This EQ is based on the acclaimed EQ-X of Merging's award winning Pyramix digital audio workstation. Like EQ-X, the MT 48 equalizer works without audible artifacts.

Each of its four bands can has four controls:

- TYPE: low cut, low shelf, peak, high shelf, high cut
- FREQUENCY: 20 Hz … 20 kHz
- GAIN: ±24 dB
- Q: 1 (wide) …100 (very narrow)

Adjust values by selecting a parameter on the touchscreen and turning the Rotary Control.

The resulting curve is shown in the display.

Please note: The EQ Gain and Frequency can be adjusted with touch movement on TFT. Make sure you update to the latest Firmware to benefit from such feature.

Selectors link to Physical Outputs

Option to unbind the Physical Buttons to the Physical Analog Outputs, so by example you can use the MT 48 Speakers or Headphones selector buttons to control a Mixer's that is routed to ADAT or peered outputs.

USB directing routed playbacks require a long physical button push for volume control.

Dynamics

The MT 48 mixer features complex Dynamics processor on every channel. It has three sections: Gate, Compressor, and Limiter, each with their own bypass switch. So you can use one or two or even all three sections at the same time. In addition, there is a global bypass switch for the entire Dynamics processor.

Each band can be bypassed individually. A global EQ bypass is to the right of the frequency display.

Gate

A gate is useful to suppress unwanted noise in (short) pauses of the wanted signal.

Gate has the following parameters:

THRESHOLD	adjusts the level when the gate opens
ATTACK 0.1 … 200 ms	adjusts how fast the gate opens
RELEASE 12 ms … 2 s	adjusts how fast the gate closes
MAKEUP GAIN	to compensate for gain loss (usually not needed for gate), can be switched to auto gain for automatic gain compensation

Compressor

A compressor is often used to make levels more even. Loud signal become softer, soft signals become louder (via makeup gain).

The compressor section has the following parameters:

THRESHOLD	adjusts the level when compression sets in
RATIO 1:1 (no compression) … 32:1 (hard compression)	adjusts the compression ratio, i.e. the factor by which a gain increase is reduced
ATTACK 0.1 … 200 ms	adjusts how fast gain reduction sets in after the signal exceeds the threshold
RELEASE 12 ms … 2 s	adjusts how fast gain reduction fades after the signal falls below the threshold
MAKEUP GAIN	to compensate for gain loss This can be switched to AUTO GAIN for automatic gain compensation

Limiter

A limiter is basically the same as a compressor but with a very high compression ratio. Limiters are used to make sure a signal does not exceed a certain level, e.g. as overload protection.

The limiter section has the following parameters:

THRESHOLD	adjusts level when limiting sets in
ATTACK 0.1 … 200 ms	adjusts how fast gain reduction sets in after the signal exceeds the threshold
RELEASE 12 ms … 2 s	adjusts how fast gain reduction fades after the signal is below the threshold
MAKEUP GAIN	to compensate for gain loss This can be switched to AUTO GAIN for automatic gain compensation.

Sidechain

Instead of reacting to the input signal, the Dynamics processor can also be triggered by an external signal. You can use any channel that has Dynamics processing activated(!).

A typical application would be “ducking”. This is a technique you have often heard applied on the radio: Music plays, and as soon as the announcer speaks, the music is attenuated so the voice can be clearly heard without interrupting the music completely. This is done by feeding the music signal through a compressor and sending the voice signal to its sidechain input. The compressor thus does not react to the music, gain (volume) is controlled by the voice signal. As soon as the announcer speaks, the voice signal “ducks” the music.

Please note: Sidechaining affects all three sections of the Dynamics processor.

Reverb

Every channel has a send to the built-in reverb. Please note that there is only one reverb processor for all four mixers. So, the send level in the individual channels is the same in all four mixers. However, the Reverb has its own output fader. Using the REV fader, the total amount of reverb can be adjusted in each mixer individually.

Please note that reverb is for monitoring purposes only, i.e. to give singers and musicians a more inspiring sound in their headphones. Reverb is not recorded, not even in the PostFX driver channels.

The Reverb parameters can be edited. Press on the REV channel name and select Edit REV:

The controls are similar to what you would find in a plug-in or hardware unit:

LOW CUT	10 Hz … 20 kHz
ROOM SIZE	small … large
HIGH CUT	10 Hz … 20 kHz
PREDELAY	0 … 300 ms
DECAY	0 … 100%
DIFFUSION	0 … 100%

Snapshots

EQ, Dynamics, and Reverb each have their own snapshots (presets), which, of course, is particularly useful for the Reverb:

Monitor controller functions

Just like a professional mixing console, the MT 48 includes monitor controller functions.

- DIM reduces the level of the selected output. The default attenuation is 20 dB, but the amount can be adjusted in SETTINGS → MONITORING
- MONO sums both stereo channels to mono output. Checking mono compatibility is still good practice as many Bluetooth boxes, smartphones, and kitchen radios effectively produce mono sound.
- CLEAR SOLO is useful to disengage the solo function on all channels at once.
- Every channel has a SOLO button. The ones on the DAW channels are particularly helpful so you can listen to your DAW mix without the live inputs.
- SOLO PFL and SOLO EXCLUSIVE options are available under the SETTINGS → MONITORING

MT 48 mixer details

Basic operation of the MT 48 mixer is pretty much self-explanatory, but there are some useful functions that may not be obvious.

Pressing on the channel name opens a menu:

The entries PREAMP, EQ, and DYNAMICS allow you to access the respective pages without switching to the Alternate Mix View.

- MUTE is handy for silencing individual channels.
- NAME lets you rename the channel.
- COLOR lets you choose a color for the channel name lettering and an indicator strip below the channel name.
- LINK WITH RIGHT/LEFT allows you to link two neighboring channels for stereo operation.
- GROUP allows you to assign the selected channel to one of eight groups. This group then has a common fader for all its channels. As default, this is the case with the ADAT input channels.

Pressing on a Group channel name opens a different menu:

- MUTE silences the entire group with all contained channels.
- EXPAND/COLLAPSE displays or hides all channels contained in this group.
- NAME lets you rename the group.
- CLEAR GROUP dissolves the group. This does not remove the channels contained in the group, they will be displayed as individual channels in the mixer.

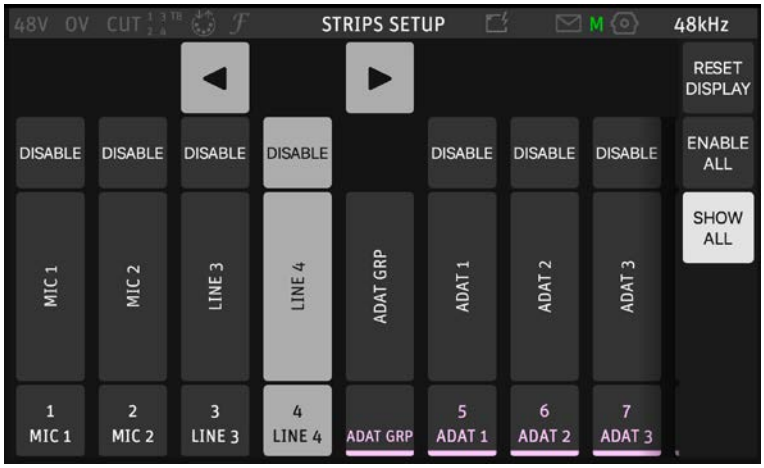
Strips setup

STRIPS SETUP allows you to configure the MT 48 mixer to your needs. Please note that changes to the Strips Setup apply to all four mixers.

▶ Go to MENU → STRIPS SETUP

Here you can disable channel strips or enable channel strips that are currently hidden (disabled). If you enable more strips than the screen can display (eight), you can reach the additional channels by swiping.

You can also re-arrange channel strips in a new order. When you press on a strip, arrows appear that allow you to move the respective strip to the left or right.



There are three buttons on the right-hand side:

- RESET DISPLAY restores the default order (but does not enable or disable strips).
- ENABLE ALL enables all channel strips.
- SHOW ALL STRIPS displays all strips, including those which are currently hidden (disabled).

MT 48 status bar

The Status Bar gives an overview over the current state of the MT 48:

48V	48V indicates if phantom power is active on one or more channels.
OV	OV indicates overload in the MT 48 mixer.
CUT 1 3 TB 2 4	CUT numbers + TB (Talkback) in red indicate inputs that are cut (muted).
	DIN5 symbol indicates MIDI activity.
	Fine precision for value changes enabled/disabled.
	Network port with flash indicates Power over Ethernet.
	Letter symbol lights up when a message or error has been received.
M	Clock status: A green M indicates the MT 48 is Master; a green S indicates the MT 48 is slave. Yellow color indicates locking is in progress, red color indicates improper synchronization (check your setup!).
	Optical Input as Clock source: If the optical input is Master, the icon appears green. Yellow color indicates locking is in progress, red color indicates improper synchronization (check your setup!). If the optical input is not used as Clock source, the icon is colorless.
48kHz	44.1/48/88.2/96/176.4/192 kHz is the current sampling rate.

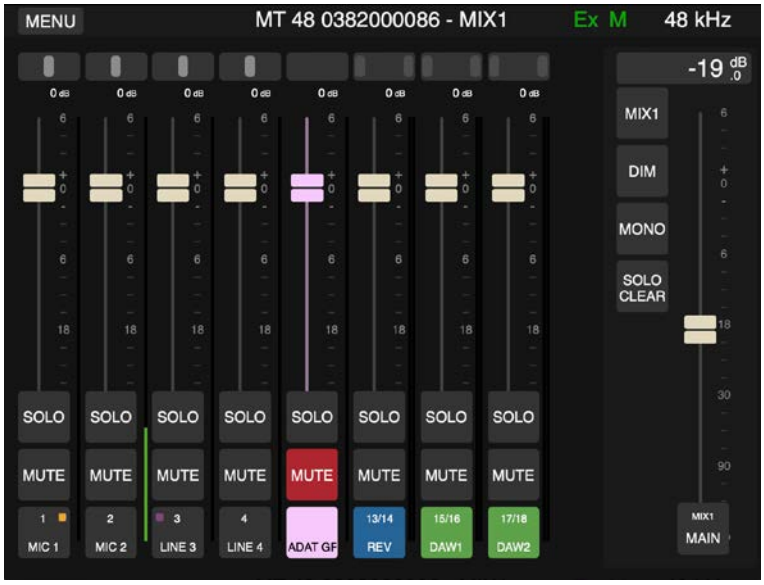
Remote controlling the MT 48

The MT 48 can be remote controlled from the connected computer using the MT 48 Remote Control App.

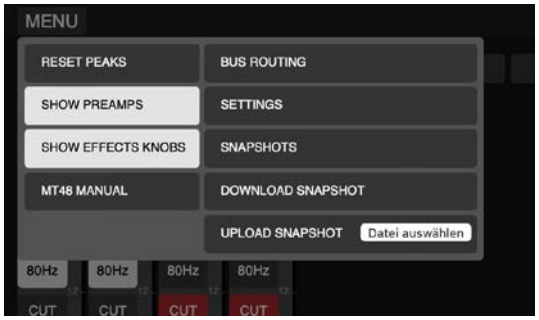
On your computer, go to the taskbar (Windows) or menu bar (Mac) and click on the Neumann logo of the MT 48 AGENT. Select LAUNCH REMOTE CONTROL APP. The Remote Control App looks and behaves just like the MT 48 screen and includes software counterparts for the MT 48's hardware buttons.

Web control

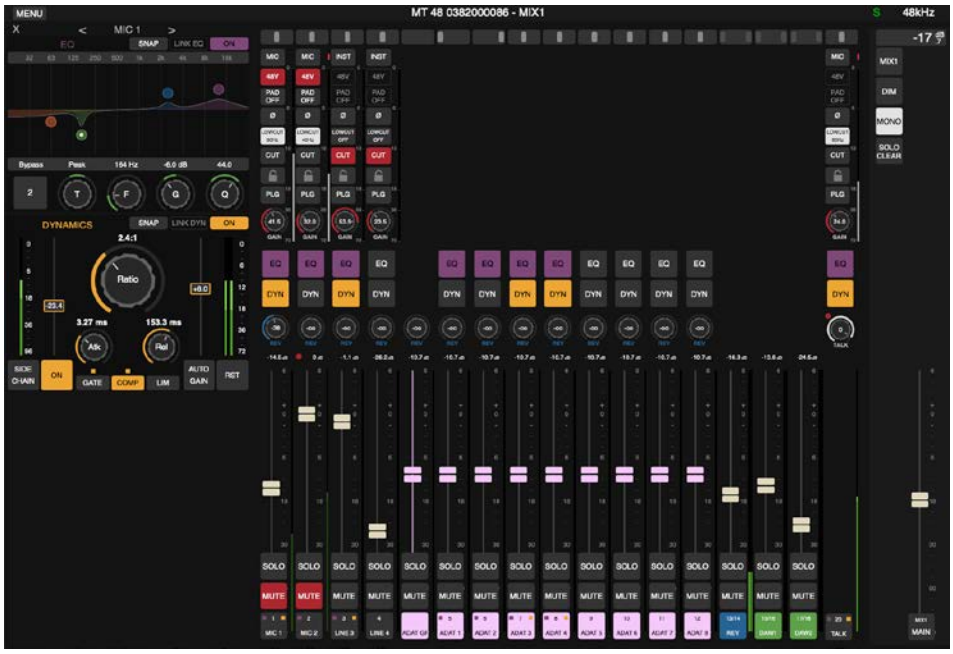
In a similar fashion, the MT 48 can also be controlled via a web interface in your browser. Click on the Neumann logo in your computer's taskbar/menu bar and select OPEN WEB CONTROL.



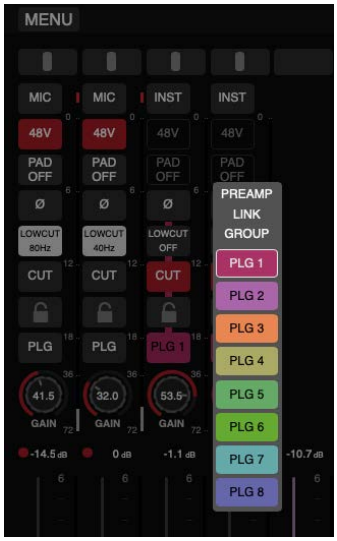
Per default, the Web Control browser page looks similar to the Remote Control App. However, the Web Control surface is optimized for touchscreen, i.e. for use with a tablet computer (see below), and it can be configured in various ways. The Web Control browser window has no size constraints so it can display more than eight channel strips.



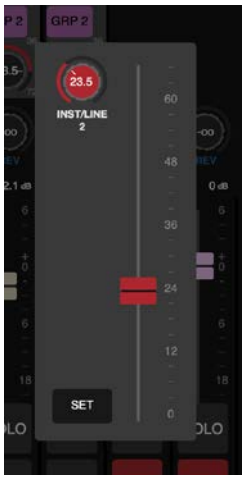
Pressing the MENU button in the upper left-hand corner, you can choose to display the preamp controls in all preamp-equipped input channels and/or the Effects knobs (EQ and DYN buttons and REV level). The latter will also give you edit panels in the left-hand side, if you click on an EQ or DYN button or open REV edit.



The Preamps section has additional functions not found on the hardware unit or the Remote Control app. These additional functions are especially useful in larger setups with many preamp channels. Instead of the PREAMP LINK feature, the Web Control features up to eight PREAMP LINK GROUPS (PLG) which the preamp channels can be assigned to. This allows you to control gain, polarity, low cut and cut (mute) on two or more channels simultaneously.



Please note that when you press on a rotary control (e.g. preamp gain or REV level) to change its value, a panel opens with a slider. This is to facilitate using the Web Control from a touch-screen device, e.g. a tablet computer.



Controlling the MT 48 from your tablet

The Web Control can be used to remote control the MT 48 from another computer. However, this requires the MT 48 to be part of your home network.

A typical use case would be to remote control mixes from one or more tablet computers so the artist(s) can adjust their own monitor mix.

To accomplish this, connect the MT 48's AES67 port to your WiFi router via a CAT5 or CAT6 ethernet cable. If that's not possible, you can also connect the MT 48 to a WiFi extender that has an ethernet port.

To find out the web address of your MT 48, go to the MT 48 AGENT, i.e. the Neumann logo in the menu bar/taskbar of your computer and select OPEN WEB CONTROL. Please note that you must deactivate WiFi on your computer as you are already connected to your home network via the MT 48. The web address in your browser window will work on any web device within your network.

A custom image can be displayed when starting the MT 48. The image can be uploaded from the MT 48 Web App browser page under Settings›Info by selecting "Upload Startup Image" entry line.

Image Specifications:

- Supports PNG files only

- Image dimension should be 800x480 (the image will be scaled if needed)
- Image size must be smaller than < 2 MB

The upload of the image can take up to 15 secs to load. If the upload fails, the browser will display page 404 or 403 error. Please make sure you've respected the ratio and file format mentioned above.

▶ Restart the MT 48 once the upload is completed.

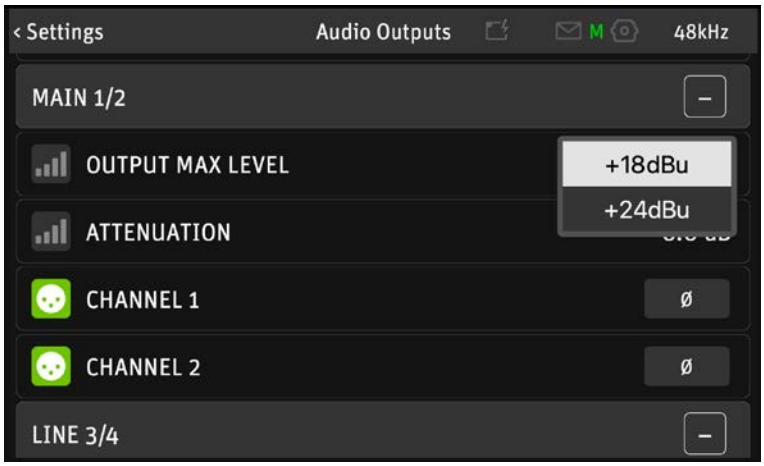
Users can revert to the original image by selecting "Reset Startup Image".

Advanced settings

Output levels

The analog outputs can be switched between high and low level with +24 dBu or +18 dBu @ 0 dBFS. The signal level can be switched for output pairs 1+2 and 3+4 individually.

▶ Go to MENU → SETTINGS → AUDIO OUTPUTS:



+24 dBu results in slightly better signal-to-noise but may be too hot for some equipment. When directly connecting your (active) studio loudspeakers, the lower signal level +18 dBu may be useful to prevent ear blasting SPLs. If needed, you can dial in additional signal attenuation.

Speakers A/B latch mode

When using two sets of speakers, you may want to avoid both speaker pairs playing at the same.

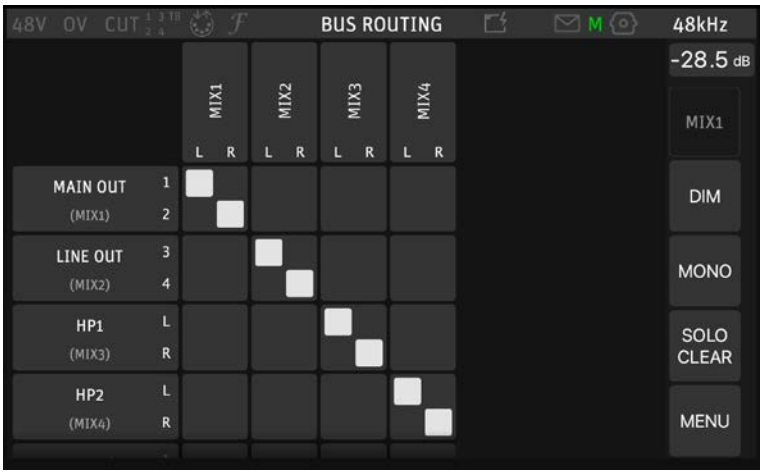
▶ Go to MENU → SETTINGS → MONITORING and scroll down to the bottom of the page:

▶ Activate "MIX 1-2 IN A/B MODE".
Now you can easily toggle between both speaker pairs. Pressing the SPEAKERS A button automatically mutes SPEAKERS B and vice versa.

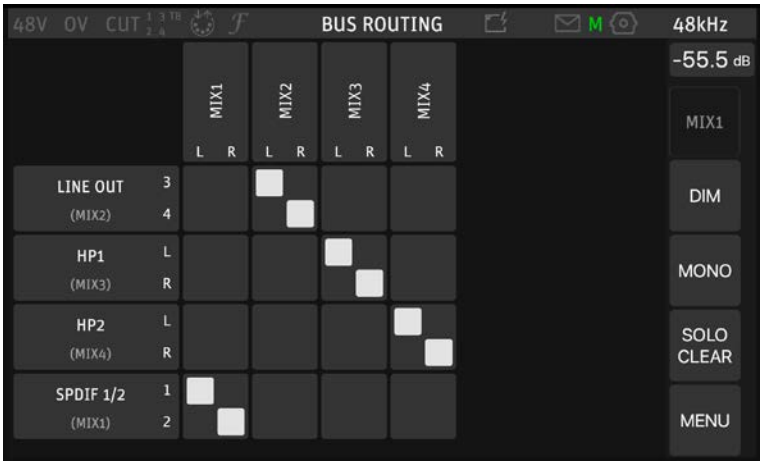
Output routing

Per default, the MT 48's four mixes are routed to Speakers A, Speakers B, Headphones 1 and Headphones 2. If needed, the mixes can be re-routed to different outputs. If, for instance you have monitor speakers with an S/PDIF input (such as the Neumann KH 150) you could re-route Mix 1 or Mix 2 to the optical output in S/PDIF mode (for the KH 150 you'd need an S/PDIF optical to coaxial converter).

▶ Go to MENU → BUS ROUTING



▶ Now scroll down to see the S/PDIF output and route Mix 1 (or whatever Mix you prefer) to the S/PDIF output (stereo).

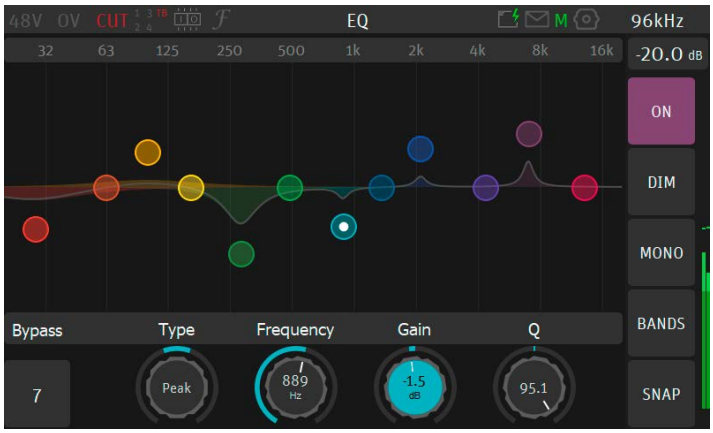


Output EQ

A 4-band parametric equalizer is available for each mixer output. This allows for basic room adjustment on the loudspeaker outputs. For Neumann DSP loudspeakers, we of course recommend MA 1, which is a much more sophisticated room alignment solution. But if you are using other speakers, the Bus EQ can be of great help to compensate for resonances in your listening environment or nonlinearities in the loudspeakers themselves. Bus EQ can also be useful on the headphones outputs, if you have to work with less than linear headphones or headphones you are not accustomed to.

▶ Go to MIX 1/2/3/4 (on the upper right-hand side) → BUS EQ

The MT 48 Music Mission Bus EQ now supports up to 12 bands of EQ on the 4 x Mixers output buses. Use the “BANDS” button to add or remove EQ Bands”.



Output dynamics

Similarly, mixer outputs can also be processed with Dynamics, if needed. The functions are the same as in the input channels Dynamics module: Gate, Compressor and Limiter.

Configuring the headphone outputs

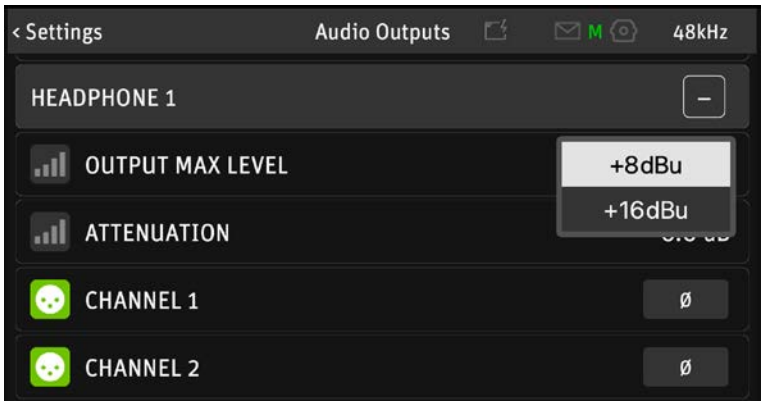
A huge number of different headphones are offered today. Two key parameters vary greatly: sensitivity and impedance. Low sensitivity and/or high impedance headphones need a very powerful headphone amplifier to produce high enough volume. Low impedance headphones require a headphone amplifier with super low output impedance for fast transient response and an optimal sound balance.

The MT 48 features excellent headphone amplifiers offering both high power and ultra-low output impedance (0.035 Ω!). So it can drive any headphones to their absolute best.

High/Low output setting

The maximum output level of the headphone amplifiers can be switched between +8 dBu or +16 dBu. The +16 dBu setting provides loud output for high impedance/low sensitivity headphones. For low impedance/high sensitivity headphones, the +8 dBu setting is advisable to avoid ear blasting levels. The output level setting can be set for both headphone outputs individually.

▶ Go to SETTINGS → AUDIO OUTPUTS and scroll down to OUTPUT MAX LEVEL under HEADPHONE 1 and HEADPHONE 2.



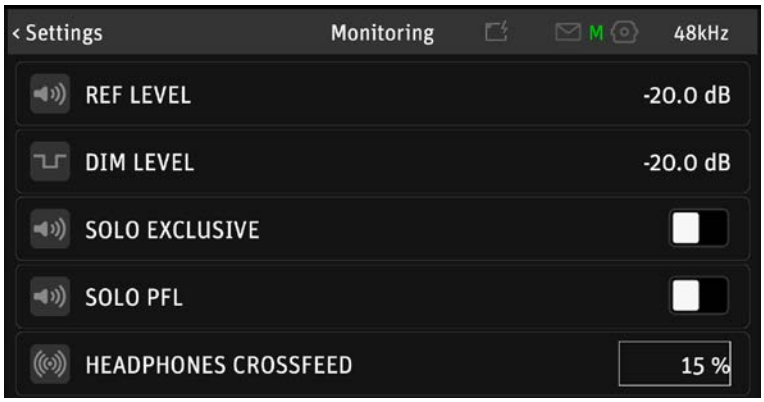
You can even reverse the signal polarity for each ear cup on each headphone output. Reversing one earcup (temporarily) can be useful to check for wiring mistakes. Reversing both cups is a useful option for some older headphones that came with reversed polarity. Correct polarity is crucial for singers in front of a microphone, so the electric path is in phase with acoustic path (air, skull bones).

Crossfeed

When listening on headphones, the signal on the left channel will only reach the left ear, and the signal on the right channel will only reach the right ear. With loudspeakers this is not the case: The signal on the left speaker will also reach the right ear and the signal on the right speaker will also reach the left ear.

This can be simulated via crossfeed: Part of the left channel is also fed to the right ear cup, and part of the right channel is fed to the left earcup. Listening on headphones thus becomes more like listening on loudspeakers. The amount of crossfeed can be adjusted from 0% (no crossfeed) to 100% The default setting is 0%.

▶ Go to SETTINGS → MONITORING and scroll down to HEADPHONES CROSSFEED.



Please note: The Crossfeed setting affects both headphone outputs.

Talkback

The MT 48 has a built-in talkback mic, which is located below the pinhole between the HOME and SPEAKERS A buttons.

To configure Talkback, go to MENU → SETTINGS → MONITORING and scroll down:

Under the Monitoring Settings find the “Link Talk Button to Preamps Cut” entry, this option is useful for users that route the Talkback within a DAW channel and need full Talkback control using the MT 48 Talkback soft button.

Note: this new setting is available as of Firmware 1.8.x and above.

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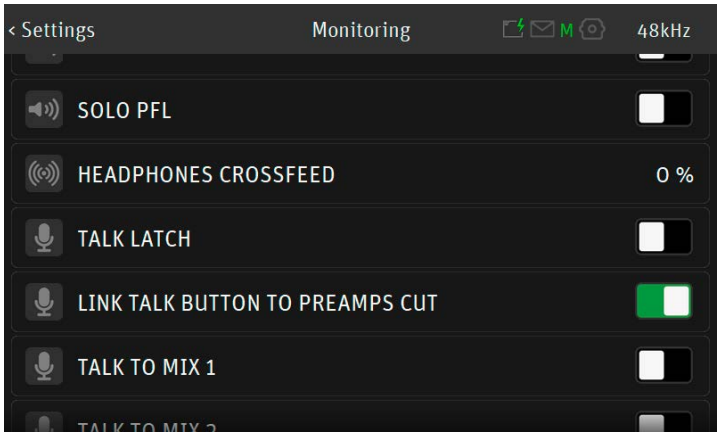
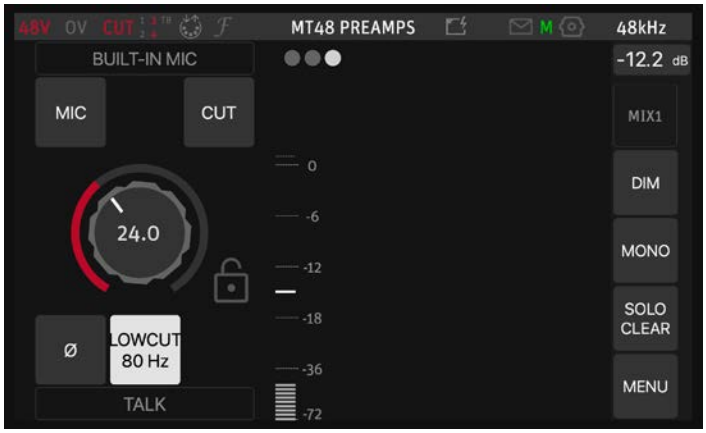


With TALK LATCH activated, the talkback mic is only open as long as you press the talkback button.

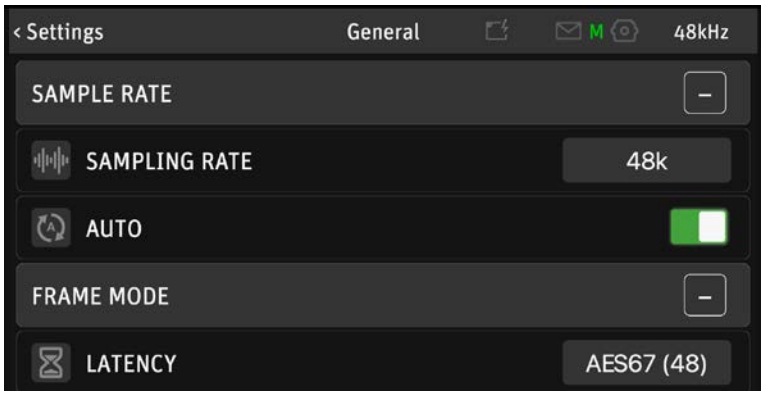
You can select which mixers Talkback talks to: Mix 1 (Speakers A), Mix 2 (Speakers B), Mix 3 (Headphones 1) and Mix 4 (Headphones 2).

To adjust the built-in mic gain, go to MENU → TALK PREAMP



Changing the sampling rate/clocking

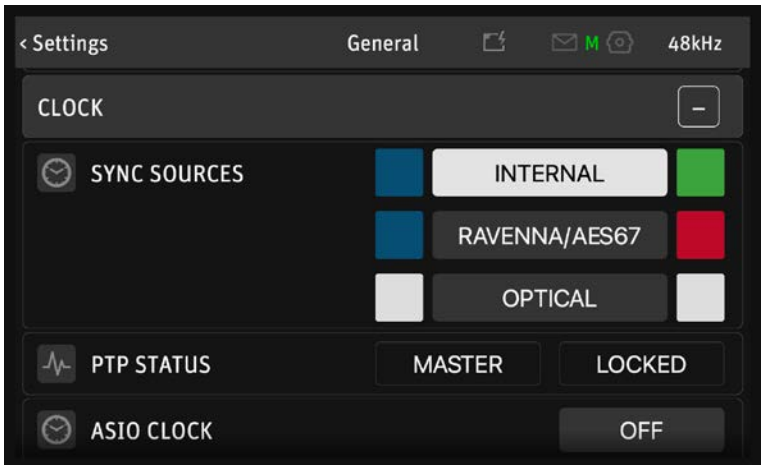
To set the sampling rate for the MT 48, go to MENU → GENERAL → SAMPLING RATE



With AUTO setting activated, the MT 48 follows the sampling rate setting in your recording software. Or if the MT 48 is in slave mode, it will automatically follow the master clock.

FRAME MODE | LATENCY sets the buffer size in a AES67/RAVENNA setup. Ignore this point if you're not using any AES67 devices with your MT 48.

Per default, the MT 48's sync source is its internal clock (INTERNAL). To change the clock settings, go to MENU → GENERAL and scroll down to CLOCK.



When set to **RAVENNA/AES67** the MT 48 follows an external PTP Master.

Note: The Precision Time Protocol (PTP) is a protocol used to synchronize clocks throughout a computer network. Also known as IEEE 1588 or IEC 61588, it is a protocol designed to synchronize real-time clocks in the nodes of a distributed system. PTP Clocks allow for time resolution to the nanosecond.

When set to **OPTICAL**, the MT 48 follows the clock embedded in an ADAT or S/PDIF input signal.

Color indicators

BLUE indicates a sync source is available; WHITE means this source is unavailable.

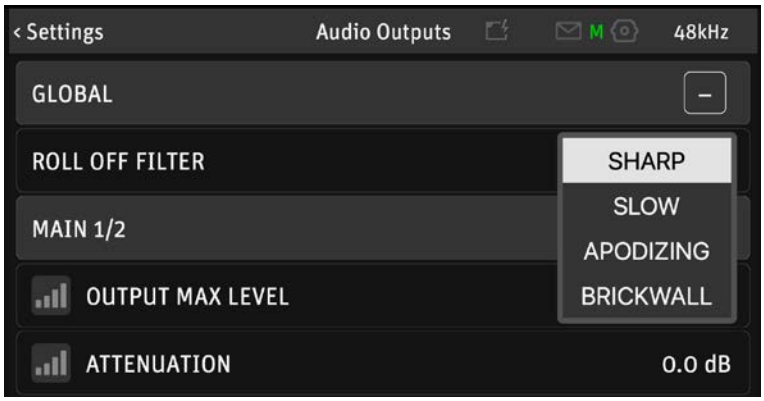
GREEN indicates Sync is working; RED means Sync is impossible.

ASIO CLOCK is deactivated by default – this is not a mistake! This setting is only relevant to AES67 setups, where the PTP master has to generate its clock according to the ASIO clock. In setups without AES67 network devices, leave ASIO CLOCK deactivated.

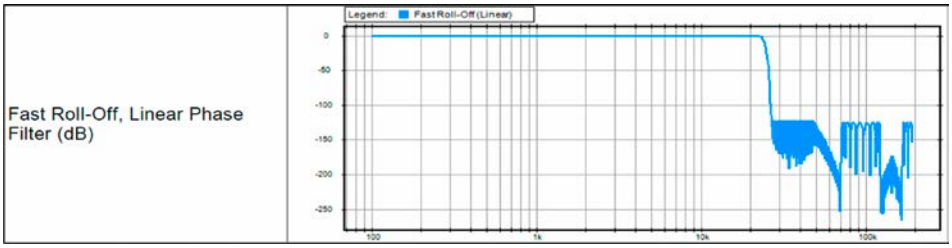
DAC filter

The DA converters offer four output filter modes:

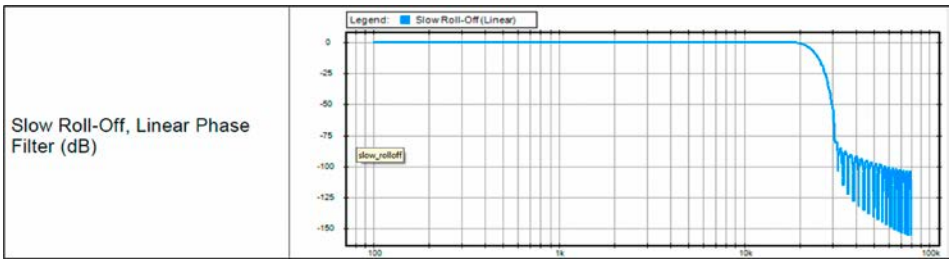
▶ Go to MENU → SETTINGS → AUDIO OUTPUTS → ROLL OFF FILTER:



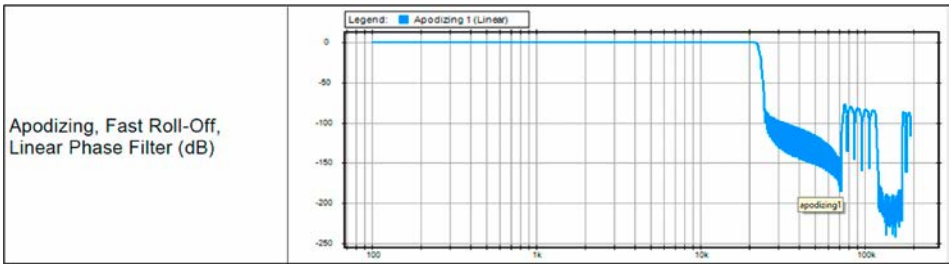
Sharp (default setting): Offers a flat frequency response with an attenuation of 3 dB at 0.484 x FS (23.2 kHz @48k), which has the tradeoff of 35 samples latency.



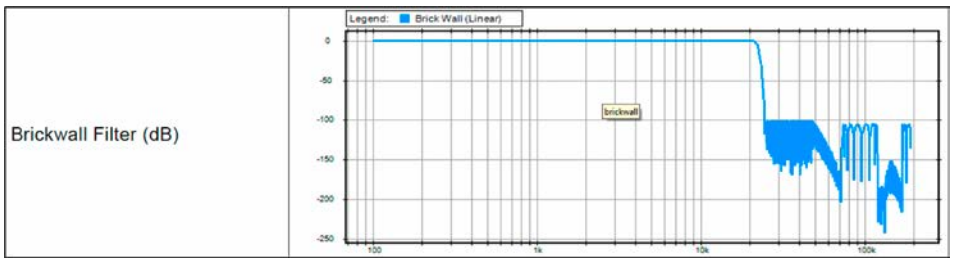
Slow: Offers the lowest latency of 9 samples, with the tradeoff of a gentle frequency response attenuation reaching -3dB at 0.45 x FS (21.6 kHz @48k).



Apodizing: Fast roll-off, linear phase filter. Avoids pre-ringing for super accurate transient response. Latency of 35 samples.



Brickwall: Ensures rejection of more than -100dB at Nyquist (0.50 x FS, 24 kHz @48k). Latency of 35 samples



Tip: For a super-direct feel use SLOW mode during recording; this is especially great for singers. For mixing and mastering, use one of the other three options, depending on your preferences.

OSC - OPEN SOUND CONTROL SUPPORT

(Supported as of firmware 1.8.x and above)

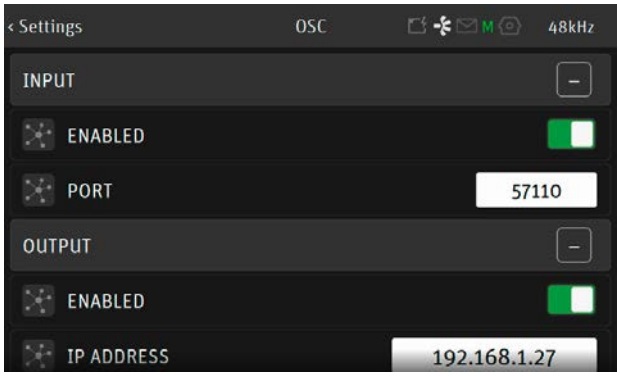
Open Sound Control (OSC) is a protocol for communication among computers, sound synthesizers, and other multimedia devices such as Remote Controllers. OSC is optimized for modern networking technology. Compared to protocols such as MIDI, OSC's advantages include interoperability, accuracy, flexibility, and enhanced deployment.

Our initial deployment is for the MT 48 Music Mission.

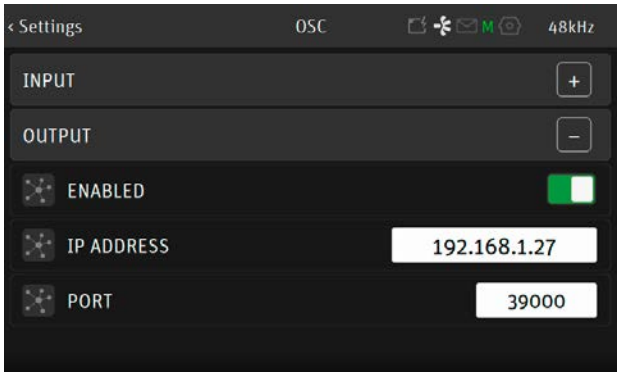
Settings configuration

Enable both input and output port, set the target IP address and adapt the Port to match the remote controller network device.

Input



Output



OSC Dictionary

General functionalities

OSC Address	Data	Comments
/music/get_all event	void	Send back all controls value

Strips by Buses

BUS_ID : Is a 1 based number to control a specific bus/mix

BUS_ID	Anubis
1	MIX
2	MIX-ALT
3	CUE 1
4	CUE 2
5	CUE 3
6	CUE 4
7	CUE 5

STRIP_IDX: 1 based number defining the strip order

OSC Address	Data	Comments
/music/bus/BUS_ID/strip/STRIP_IDX/fader	float, min -144.5, max 6.0	
/music/bus/BUS_ID/strip/STRIP_IDX/pan	float, min -1.0, max 1.0	
/music/bus/BUS_ID/strip/STRIP_IDX/solo	bool	
/music/bus/BUS_ID/strip/STRIP_IDX/mute	bool	
/music/bus/BUS_ID/nb_of_strips	int	

Note: Linked strips are described as a single Strip when collapsed and two separate strip when un-collapsed. When collapse or un-collapsed a Strip, it updates all the STRIP_IDX affected.

OSC Address	Data	Comments
/music/bus/BUS_ID/master_gain	float, min -144.5, max 6.0	
/music/bus/BUS_ID/mute	bool	
/music/bus/BUS_ID/dim	bool	
/music/bus/BUS_ID/mono	bool	

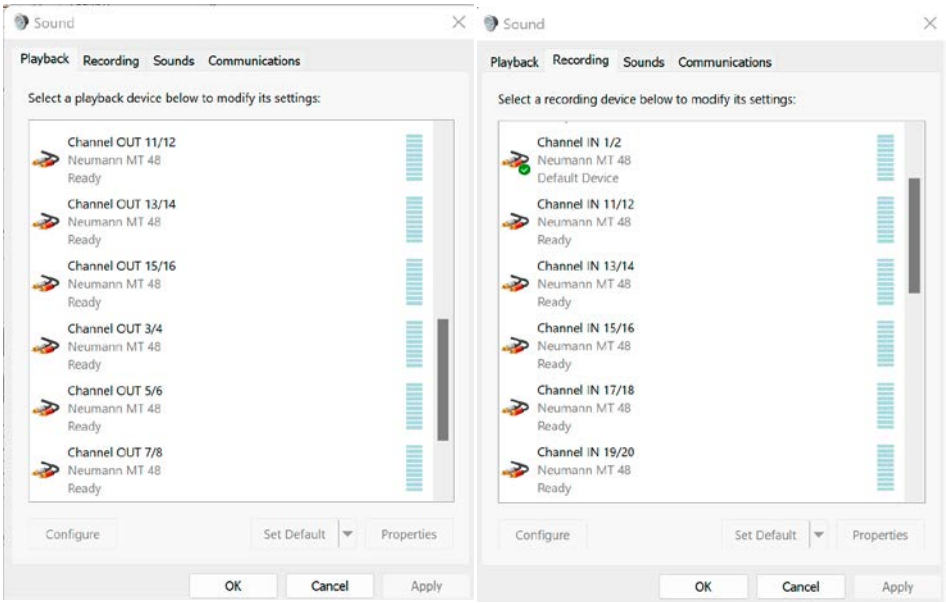
MakePro Support

- You can import it in Glue (Download Glue | MakePro X Downloads)
- Then create your Control Device layout and assign the target definition to the layout.

Windows WDM Routing flexibility

WDM channels can be individually selected per banks for IO routing. This allows flexible routing for application such as Chrome Browser, or system sound running applications.

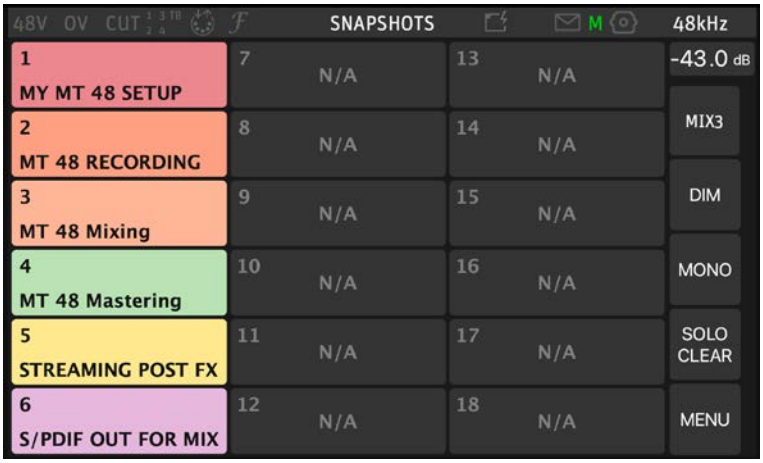
Note: Available as of MT 48 Toolkit 1.8.x and above.



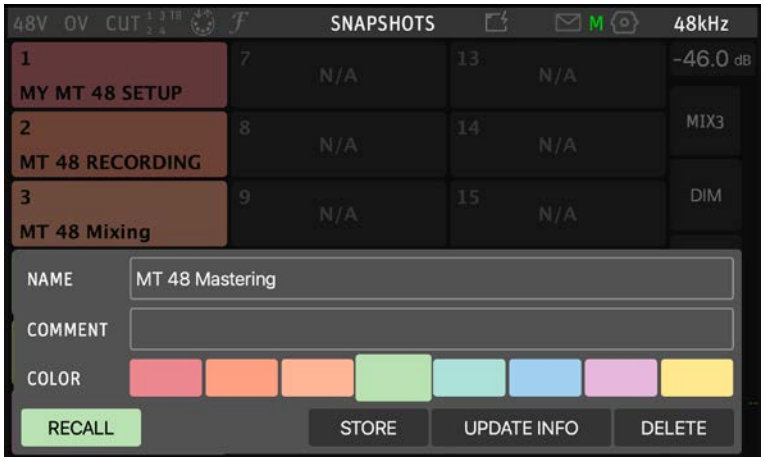
Device snapshots

All settings can be stored and recalled using device snapshots.

▶ Go to MENU → SNAPSHOTS



Once you've configured the MT 48 to your needs, it's a good idea to store your settings in a device snapshot. You can also use snapshots to switch between several configurations for various tasks, e.g. recording, mixing, and mastering.



The STORE dialog allows you to type in a name (and a comment if you like) using a virtual keyboard and choose a color.

Recalling Snapshots is just as easy. The RECALL dialog even lets you select and deselect various functions that you may want to include or exclude from the Snapshot you're about to recall. Or simply press TOTAL RECALL for taking over all settings from the Snapshot you are recalling.

Appendix

Updating the firmware

We recommend updating the firmware from the MT 48 Agent by selecting "Show Available Updates" refer to the Toolkit guide for all details. For Manual Firmware update, follow the procedure below (that is the current one).

To find out if your MT 48 is on the newest firmware, go to MENU → SETTINGS → INFO

Updating requires the MT 48 driver/agent to be installed on your computer system.

The firmware update procedure is easy:

1. Download the newest MT 48 toolkit software, unzip and locate the firmware file.
2. With the MT 48 connected to your computer, go to the Neumann logo in the menu bar (Mac)/taskbar(Windows) to open the drop-down menu.
3. In the menu, select OPEN MAINTENANCE PAGE
4. A browser window opens. Click SELECT FILE and locate the firmware file you downloaded.
5. Click UPDATE and wait for the procedure to end. Make sure your computer does not enter sleep mode during the update process. The updating procedure takes about 10 minutes. Don't panic if the process appears to get stuck. Do not abort, just wait.
6. Once the update is completed, the message REBOOT DEVICE appears. Confirm.
7. After rebooting the device, your MT 48 is on the latest firmware.

Using the MT 48 with the Neumann MA 1 Automatic Monitor Alignment

MA 1 is a hardware/software solution that allows easy and precise alignment for Neumann DSP speakers. Acoustic data of your room is gathered using the included measurement microphone in several spots around your listening position. The resulting correction curve is then stored inside the DSP speakers.

This process requires the speakers' network ports to be connected to your computer for data transfer while audio signals are sent to the speakers via the analog or digital outputs of your audio interface, in this case the MT 48.

The MT 48 contains a converter that translates between USB and AES67/Ravenna. It is important to know that a computer should not be part of two networks at the same time. Therefore, **the network connection to your speakers should not be made from your computer's network**

port but via the MT 48's AES67 network port (regardless of whether your Neumann speakers are AES67 variants or the standard version). This avoids address conflicts. Connect your loudspeakers using a standard IPv6 capable ethernet switch. Please note that the MT 48 has a gigabit port that is not directly compatible with Non-AES67 Neumann speakers' 100Mbit ports. So a gigabit network switch is required to translate between the two, even if you only connect one Non-AES67 Neumann loudspeaker such as the KH 750 DSP subwoofer.

FAQ

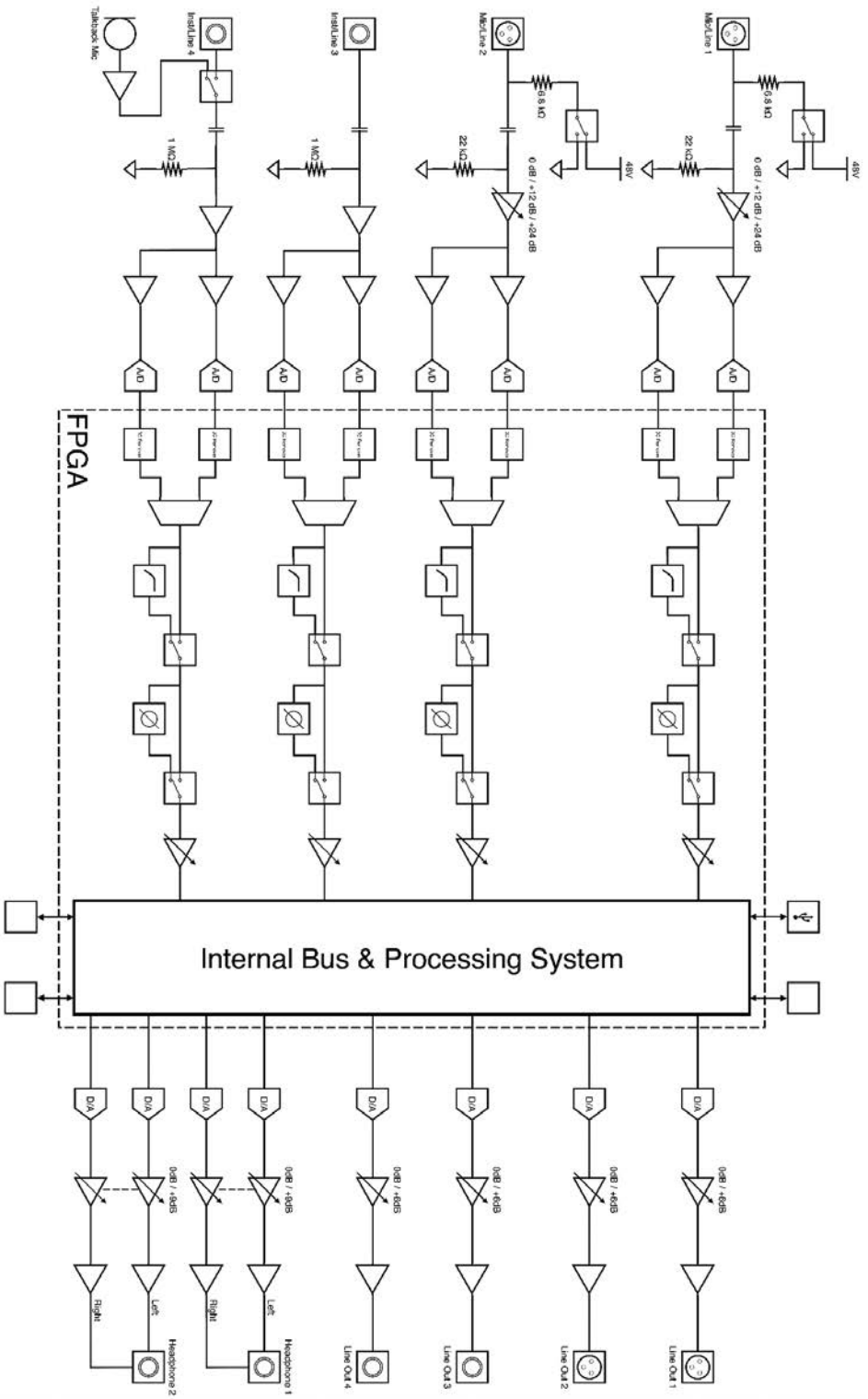
The MT 48 runs rather warm. Is my device defective?

- Don't worry, that's normal. The MT 48 contains a high performance DSP, and the aluminum enclosure serves as a heat sink.

The fan makes noise when I set Fan Mode to Mid or High. What can I do?

- Just set FAN MODE to Low, which is the default. In this setting the fan is inaudible except in very hot environments. The fan is merely a safety measure to keep the electronics from overheating, if the room temperature is unusually high.

Can the MT 48 be used with Merging Anubis Missions?

- Not a launch, but additional missions may be offered at a later point.

MT 48 signal flow