USER MANUAL

MZ 8060

- AUDIO MATRIX WITH 8 AUDIO
INPUTS AND 6 AUDIO OUTPUTS

RCF

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WARNING ⚠

IMPORTANT
Before connecting and using this product, please read this instruction manual carefully and keep it on hand for future reference.
The manual is to be considered an integral part of this product and must accompany it when it changes ownership as a reference for correct installation and use as well as for the safety precautions.
RCF S.p.A. will not assume any responsibility for the incorrect installation and / or use of this product.

WARNING: To prevent the risk of fire or electric shock, never expose this product to rain or humidity (indoor use only).

SAFETY PRECAUTIONS
1. All the precautions, in particular the safety ones, must be read with special attention, as they provide important information.

2. POWER SUPPLY FROM MAINS
- The mains voltage is sufficiently high to involve a risk of electrocution: never install or connect this product when its power cord is plugged in.
- Before powering up, make sure that all the connections have been made correctly and the voltage of your mains corresponds to the voltage shown on the rating plate on the unit, if not, please contact your RCF dealer.
- The metallic parts of the unit are earthed by means of the power cord.
- An apparatus with CLASS I construction shall be connected to a mains socket outlet with a protective earthing connection.
- Protect the power cord from damage. Make sure it is positioned in a way that it cannot be stepped on or crushed by objects.
- To prevent the risk of electric shock, never open this product: there are no parts inside that the user needs to access.
- The mains plug is used as the disconnect device and it shall remain readily operable.

3. Make sure that no objects or liquids can get into this product, as this may cause a short circuit. This apparatus shall not be exposed to dripping or splashing.
No objects filled with liquid (such as vases) and no naked sources (such as lighted candles) shall be placed on this apparatus.

4. Never attempt to carry out any operations, modifications or repairs that are not expressly described in this manual.
Contact your authorized service centre or qualified personnel should any of the following occur:
- The product does not function (or functions in an anomalous way).
- The power supply cord has been damaged.
- Objects or liquids have got in the unit.
- The product has been subject to a heavy impact.

5. If this product is not used for a long period, disconnect the power cord.

6. If this product begins emitting any strange odours or smoke, switch it off immediately and disconnect the power cord.

7. Do not connect this product to any equipment or accessories not foreseen.
For suspended installation, only use the dedicated anchoring points and do not try to hang this product by using elements that are unsuitable or not specific for this purpose.
Also check the suitability of the support surface to which the product is anchored (wall, ceiling, structure, etc.), and the components used for attachment (screw anchors, screws, brackets not supplied by RCF etc.), which must guarantee the security of the system /

installation over time, also considering, for example, the mechanical vibrations normally generated by loudspeakers.

To prevent the risk of falling equipment, do not stack multiple units of this product unless this possibility is specified in the user manual.

8. RCF S.p.A. strongly recommends this product is only installed by professional qualified installers (or specialised firms) who can ensure correct installation and certify it according to the regulations in force.

The entire audio system must comply with the current standards and regulations regarding electrical systems.

9. Supports and trolleys

The equipment should be only used on trolleys or supports, where necessary, that are recommended by the manufacturer. The equipment / support / trolley assembly must be moved with extreme caution. Sudden stops, excessive pushing force and uneven floors may cause the assembly to overturn.

10. Mechanical and electrical factors need to be considered when installing a professional audio system (in addition to those which are strictly acoustic, such as sound pressure, angles of coverage, frequency response, etc.).

11. Hearing loss

Exposure to high sound levels can cause permanent hearing loss. The acoustic pressure level that leads to hearing loss is different from person to person and depends on the duration of exposure. To prevent potentially dangerous exposure to high levels of acoustic pressure, anyone who is exposed to these levels should use adequate protection devices. When loudspeakers capable of producing high sound levels are used, it is necessary to wear ear plugs or protective earphones.

See the technical specifications in speaker user manuals to know their maximum sound pressure levels.

12. Install this product far from any heat source.

13. Never force the control elements (keys, knobs, etc.).

14. Do not use solvents, alcohol, benzene or other volatile substances for cleaning the external parts of this product. Use a dry cloth.

NOTES ABOUT AUDIO SIGNAL CABLES

To prevent the occurrence of noise on microphone / line signal cables, use screened cables only and avoid putting them close to:
- **Equipment that produces high-intensity electromagnetic fields**
- **Mains cables**
- **Loudspeaker lines.**

RCF S.P.A. THANKS YOU FOR PURCHASING THIS PRODUCT, WHICH HAS BEEN DESIGNED TO GUARANTEE RELIABILITY AND HIGH PERFORMANCES.

INSTALLATION

This device can be installed into a 19" rack cabinet (1 unit) thanks to its rack ears.

DESCRIPTION

MZ 8060 is a highly versatile digital audio matrix (and automixer) based on a powerful DSP and dedicated to installed sound systems and other audio applications.

It has 8 mic / line inputs on removable connectors with selectable 48 V phantom power (each input from 5 to 8 has a single RCA socket) and six outputs, of which four (A to D) on removable screw terminals, two (E – F) on RCA sockets.

Sound processing includes parametric equalizers, noise-gates, compressors, limiters, delays, ducking, automatic level and gain controls, priority functions, levels and gains on all matrix nodes.

All parameters can be edited through a dedicated RDNET software module.

Each MZ 8060 can work as a dual automixer with 2 independent settings.

MZ 8060 has 10 logic inputs (GPIN), 4 logic outputs (GPOUT) plus 2 (RL1, RL2) with relays. All logic inputs and outputs have removable screw terminals and are fully configurable.

Two MZ 8060 (MASTER and SLAVE) can be linked each other to get a 16-input 12-output matrix through RJ 45 digital ports on their rear panels.

A slot on rear panel allows to integrate the optional RCF AC RDNET IN/OUT PLUG board to get remote control through RDNET network.

Up to twelve BM 3003 paging microphones (linked in daisy-chain on the same line) can be connected to the dedicated RJ 45 port, which can be used as alternative to input 1.
Each of the three BM 3003 buttons can be configured to page one or more zones (audio outputs).
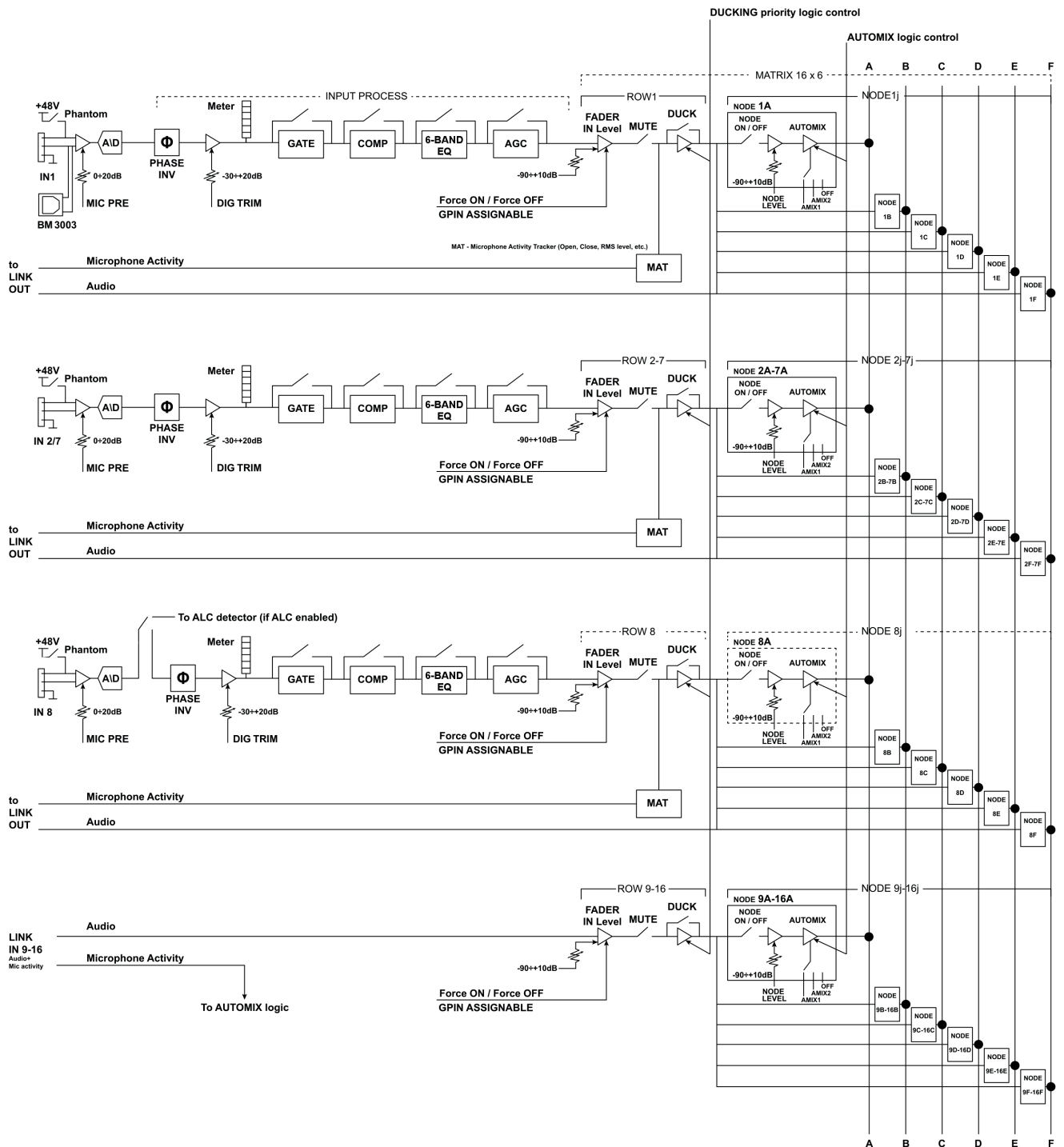All paging microphones are interlocked one another and report in-use and busy indications.
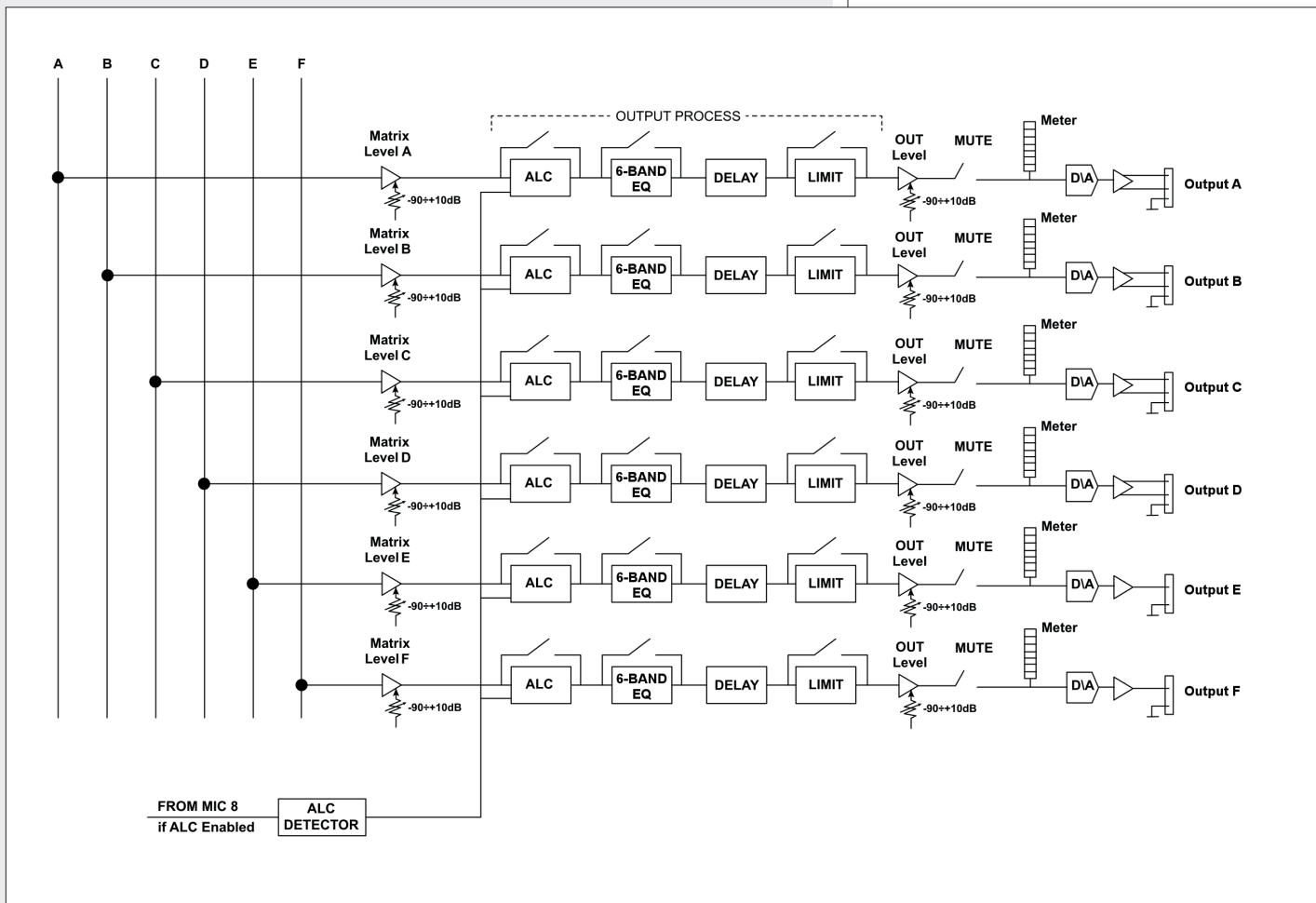
MAIN FEATURES

- 8-input 6-output digital audio matrix and automixer.

- 40-bit floating point DSP, sampling frequency: 48 kHz, 24-bit a/d d/a converters.

- 8 mic / line balanced audio inputs with 48 V 'Phantom' power, with noise gates, compressors, parametric equalizers, automatic gain controls, ducking function.

- 4 balanced and 2 unbalanced LINE audio outputs, all with level controls, parametric equalizers, delays, limiters.

- 10 programmable logic inputs (GPIN).

- 4 programmable logic outputs (GPOUT) plus 2 with relay (RL1, RL2).

- Dual automixer with 2 independent settings.

- 16-input 12-output audio matrix and 4 automixers (two for MASTER unit outputs and two for SLAVE unit outputs) by linking together two MZ 8060.

- Configurable ALC ('Automatic Level Control') when a noise detection microphone is connected to the input no.8 .

- Either USB link (standard) or RDNET link (through the optional AC RDNET IN/OUT PLUG board).

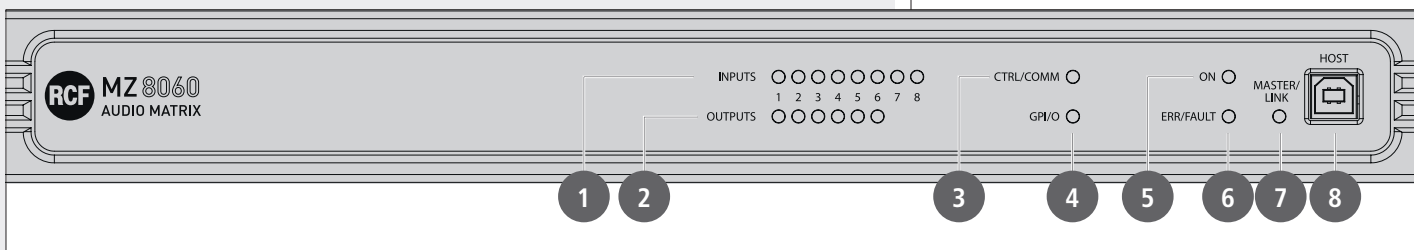- RS 485 serial port for remote control via third-party software.

AUDIO INPUT BLOCK DIAGRAM

AUDIO OUTPUT BLOCK DIAGRAM

OUTPUT PROCESS

Matrix Level A — ALC — 6-BAND EQ — DELAY — LIMIT — OUT Level — MUTE — Meter — D\A — **Output A**
-90÷+10dB / -90÷+10dB

Matrix Level B — ALC — 6-BAND EQ — DELAY — LIMIT — OUT Level — MUTE — Meter — D\A — **Output B**
-90÷+10dB / -90÷+10dB

Matrix Level C — ALC — 6-BAND EQ — DELAY — LIMIT — OUT Level — MUTE — Meter — D\A — **Output C**
-90÷+10dB / -90÷+10dB

Matrix Level D — ALC — 6-BAND EQ — DELAY — LIMIT — OUT Level — MUTE — Meter — D\A — **Output D**
-90÷+10dB / -90÷+10dB

Matrix Level E — ALC — 6-BAND EQ — DELAY — LIMIT — OUT Level — MUTE — Meter — D\A — **Output E**
-90÷+10dB / -90÷+10dB

Matrix Level F — ALC — 6-BAND EQ — DELAY — LIMIT — OUT Level — MUTE — Meter — D\A — **Output F**
-90÷+10dB / -90÷+10dB

A B C D E F

FROM MIC 8 if ALC Enabled — ALC DETECTOR

FRONT PANEL

RCF **MZ 8060** AUDIO MATRIX

INPUTS ○○○○○○○○
        1 2 3 4 5 6 7 8
OUTPUTS ○○○○○○

CTRL/COMM ○
GPI/O ○

ON ○
ERR/FAULT ○

MASTER/ LINK ○

HOST

1 **2** **3** **4** **5** **6** **7** **8**

1 **INPUTS** LEDs: each LED indicates the signal level of its respective audio input.

**2** **OUTPUTS** LEDs: each LED indicates the signal level of its respective audio output.

NOTE: EACH LED (**1** AND **2**) IS OFF WHEN ITS AUDIO SIGNAL LEVEL IS BELOW −60 dBFS, GREEN = −60 dBFS, ORANGE = −12 dBFS, RED = IN THE RANGE −3 TO 0 dBFS (PEAK).

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3 **CTRL/COMM** LED: it blinks when receiving control data.

④ **GPI/O** LED: when lit, it indicates the activation of at least a logic input / output.

⑤ **ON** LED: when lit, it indicates the MZ 8060 audio matrix is properly powered and operating.

⑥ **ERR/FAULT** LED: when lit, it indicates a fault or data transmission / reception errors.

⑦ **MASTER/LINK** LED: when lit, the MZ 8060 audio matrix is the MASTER unit. When off, the MZ 8060 is the SLAVE unit. When blinking, the MASTER unit is not linked to the SLAVE (or vice-versa) or due to a LINK fault.

⑧ **HOST:** USB (type B) port to link a computer (PC) locally.

REAR PANEL ➡️
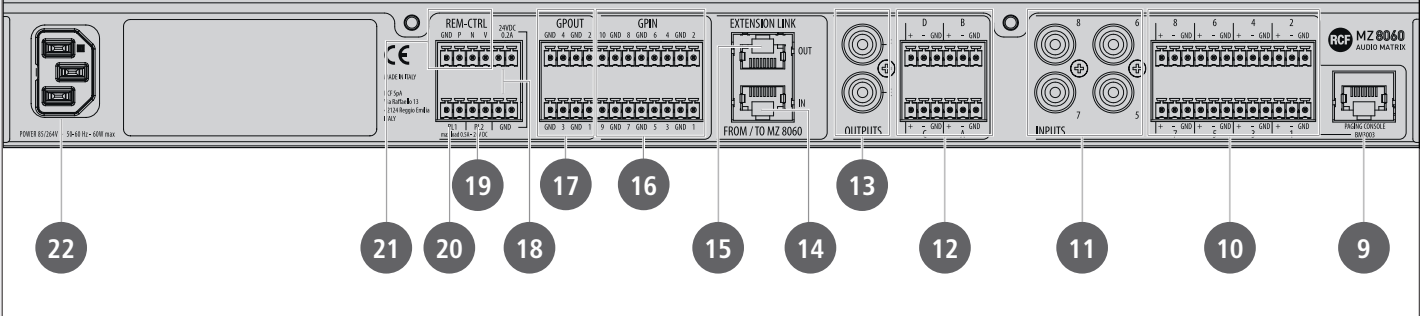
⑨ **PAGING CONSOLE (input 1):** audio input with RJ 45 socket (alternative to the input 1 with removable connector ⑩) for a line made of max. 12 RCF BM 3003 paging microphones linked in 'daisy-chain' (read the BM 3003 user manual).

⑩ **AUDIO INPUTS 1 to 8:** balanced line level inputs for removable connectors. Pins:

+	Audio signal, hot
–	Audio signal, cold
GND	Ground

(11) **AUDIO INPUTS 5 to 8:** (alternative to those for removable connectors (10)), unbalanced line level inputs with RCA sockets.

(12) **AUDIO OUTPUTS A to D:** balanced line level outputs for removable connectors. Pins:

+	Audio signal, hot
–	Audio signal, cold
GND	Ground

(13) **AUDIO OUTPUTS E and F:** unbalanced line level outputs with RCA sockets.

(14) **EXTENSION LINK IN:** input to link the EXTENSION LINK OUT of the second MZ 8060 (optional). Read the manual section 'How to link two MZ 8060 each other'.

(15) **EXTENSION LINK OUT:** output to link the EXTENSION LINK IN of the second MZ 8060 (optional). Read the manual section 'How to link two MZ 8060 each other'.

(16) **GPIN 1 ÷ 10** 'Generic Purpose Inputs' (removable connectors)
Logic inputs that can be activated by other device commands.
Each logic input can be set via software. It can be activated when its two pins (for example, '1' and its respective ground GND) are either shorted or open.

(17) **GPOUT 1 ÷ 4** 'Generic Purpose Outputs' (removable connectors)
Logic outputs that can be set via software.
NOTE: LOGIC OUTPUTS CANNOT BE USED TO POWER OTHER DEVICES (I.E. RELAYS).
OUTPUT VOLTAGE CAN BE EITHER 5 V (HIGH) OR 0 (LOW).

(18) **24 VDC** output (two positive 24 V dc pins and two ground pins)
24 V dc output (for removable connectors) available to power other devices (max. current: 0.2 A).

(19) **RL2** logic output with relay (removable connector) that can be set via software.
Max. current on contacts: 0.5A. Max. applicable voltage: 24 V dc.

(20) **RL1** logic output with relay (removable connector) normally assigned to the device fault indication. To enable its user software control, it is necessary to set the JMP1 internal jumper (see the respective manual section).
Max. current on contacts: 0.5A. Max. applicable voltage: 24 V dc.

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21 **REM-CTRL** REM-CTRL RS 485 serial port for remote control via third-party software (removable connector).

GND	Ground
P	Non-inverting pin (RS 485 A)
N	Inverting pin (RS 485 B)
V	24 V dc

22 Input for the power cord (to be connected to mains through an earthed socket only).

HOW TO LINK TWO MZ 8060 EACH OTHER

Two MZ 8060 can be linked each other to get a 16-input 12-output matrix through the EXTENSION LINK ports (RJ 45 sockets) on their rear panels.

Connect the **EXTENSION LINK OUT** **15** output of the first MZ 8060 to the **EXTENSION LINK IN** **14** input of the second MZ 8060 and vice-versa.

THE TWO CABLES MUST BE UNSHIELDED **CAT 6** AND NOT LONGER THAN **20** CM (ABOUT **8** INCHES).

WHEN USING THE **PC** SOFTWARE (READ THE RESPECTIVE MANUAL SECTION 'SOFTWARE CONFIGURATION > UNIT SETTINGS'), SET THE FIRST **MZ 8060** TO **MASTER** AND THE SECOND **MZ 8060** TO **SLAVE**.

EXTENSION LINK

OUT

MZ 8060 (1)

IN

FROM / TO MZ 8060

EXTENSION LINK

OUT

MZ 8060 (2)

IN

FROM / TO MZ 8060
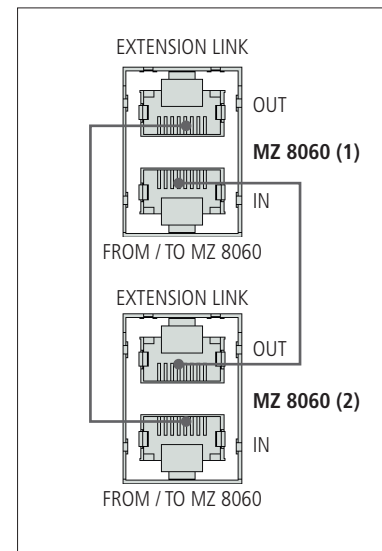
NOTES ABOUT THE RDNET SOFTWARE

The **RDNET** software is protected by international copyright laws and is to be used to configure the **RCF RDNET** system only.
It is not allowed to modify or change or try to decompile this software.
In no event shall RCF S.p.A. be liable to end-users for any damage whatsoever, including but not limited to financial damages for loss of business profits or business information due to the software use or inability to use this product.
The foregoing provision is effective even if RCF S.p.A. has been advised of the possibility of such damages.
Even if the SOFTWARE has any material, verifiable and reproducible program errors, RCF S.p.A. shall have no obligation to modify such errors.

Minimum requirement: a PC with either Microsoft 'Windows® Vista' or '7' (or later) operating system, having an available USB port. The operating system shall also have the drivers for the emulation of the serial RS 232 interface via the USB port.

Before installing a new software release, it is necessary to remove the previous version (if installed). Verify also that the new release is the right one for the PC operating system: either 32-bit or 64-bit.
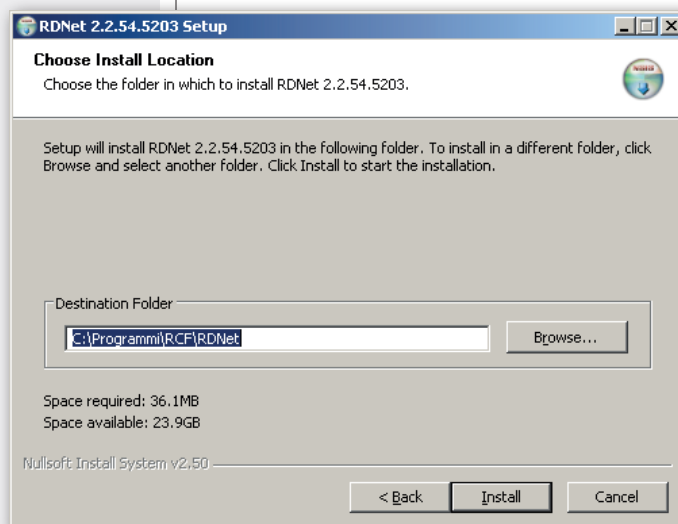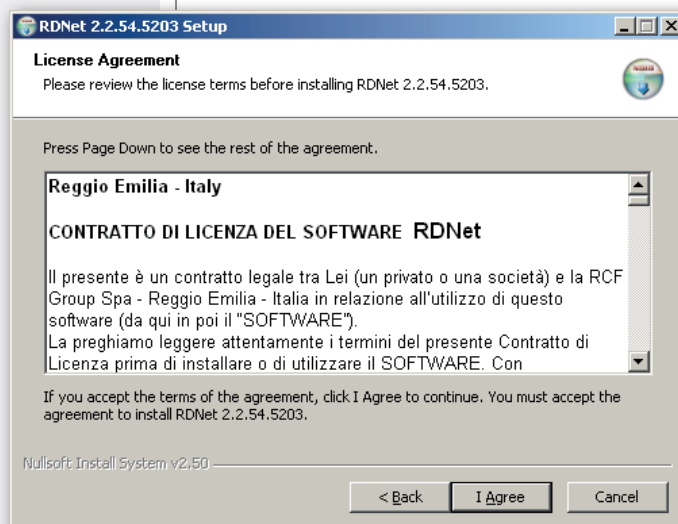
Run rcf_rdnetsetup_(release).exe to start the setup wizard.

Click NEXT > to proceed.

Read the license agreement.
Click 'I Agree' to accept and proceed with the software installation.

It is now possible to change the installation folder (directory) of the RDNET software (or keep the default path).

Click INSTALL to proceed.

As soon as the installation ends, it will be necessary to reboot your computer. Choose either '**Reboot Now**' to reboot immediately or 'I **want to reboot later**" to reboot later manually.

Click FINISH to quit.
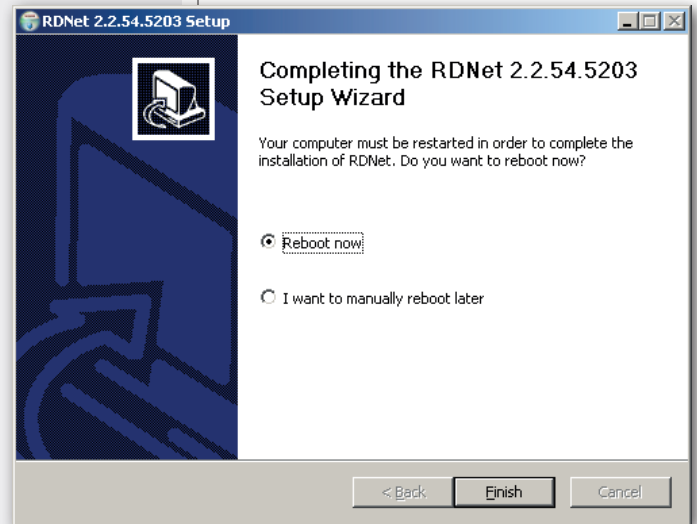
START THE RDNET SOFTWARE (REL. 2.2.54.5203) VIA USB

NOTE: THE PC OPERATING SYSTEM SHALL ALSO HAVE THE DRIVERS FOR THE EMULATION OF THE SERIAL RS 232 INTERFACE VIA THE USB PORT.
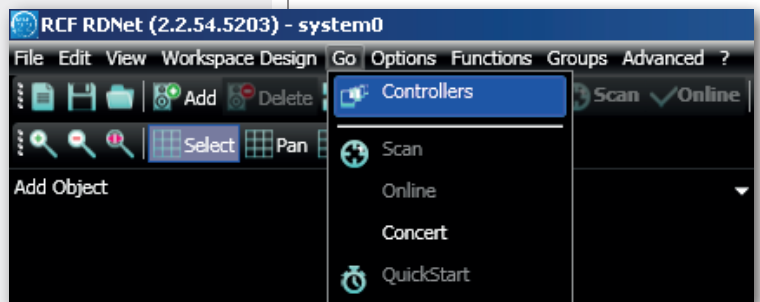
Link (through a suitable cable) the computer USB port to the one of the MZ 8060 audio matrix (labelled HOST **8**).

Run the RDNET software (in Windows, click):
Start > Programs > RCF > RDNet > RDNet
(or double-click **RDNet** icon on the desktop).

First, in the main menu, click '**Go**' and then '**Controllers**'.

In the '**Controllers**' section (bottom left in the main window), click '**Match All**' (or click each single 'Match' button or simply drag and drop) to add all linked USB devices (COM) to the '**Workspace Controllers**' list (bottom right in the main window).
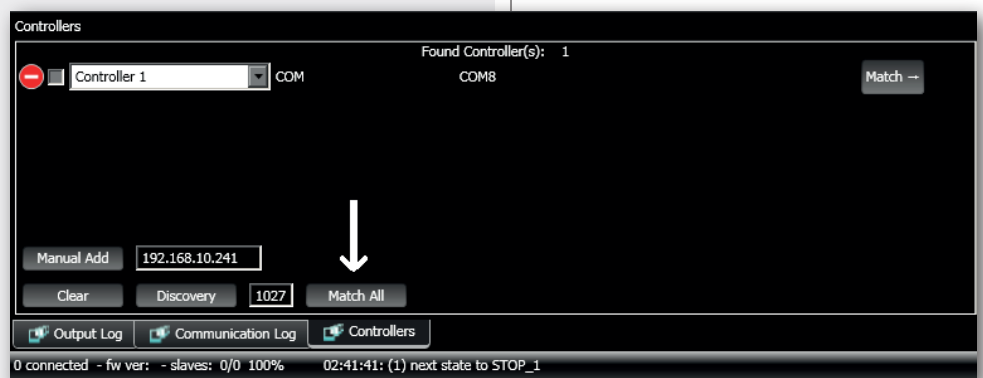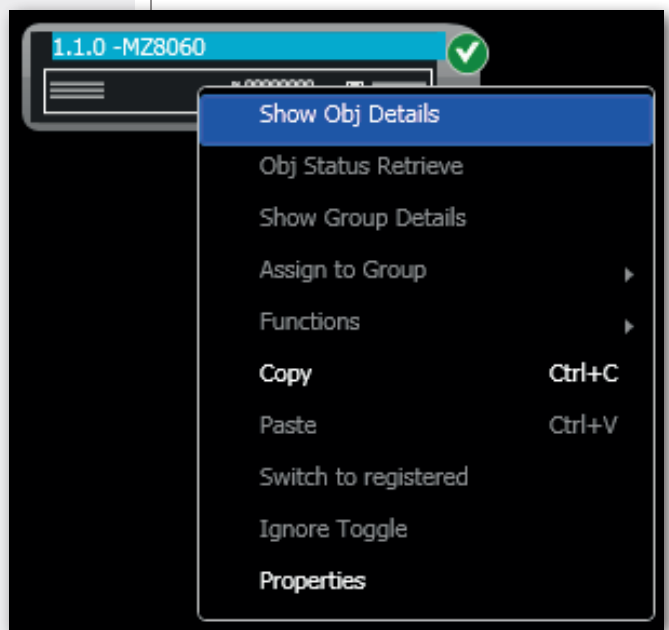
In the main menu, now click '**Quickstart**': in the bottom right-hand corner of the main window (in the bottom bar) the word '**OFFLINE**' becomes '**ONLINE**'.
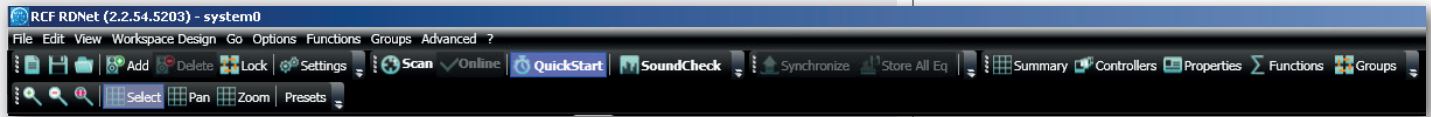
The MZ 8060 module now appears.
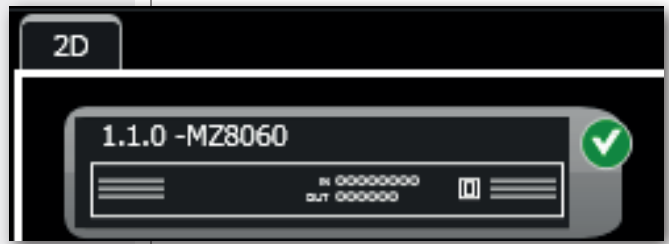
Point the PC mouse on the MZ 8060 module and double-click (or right-click and select '**Show Obj Details**') to open the audio matrix setting window.

To upgrade the MZ 8060 firmware (main menu: "**Advanced > Firmware Upgrade...**"), refer to the RDNET user manual.

SOFTWARE CONFIGURATION

The main window includes the 8 MASTER INPUTS and the 6 MASTER OUTPUTS of the MASTER unit (the first or the only MZ 8060); the SLAVE INPUTS and SLAVE buttons (to manage the SLAVE unit, which is the second optional MZ 8060) are initially disabled.
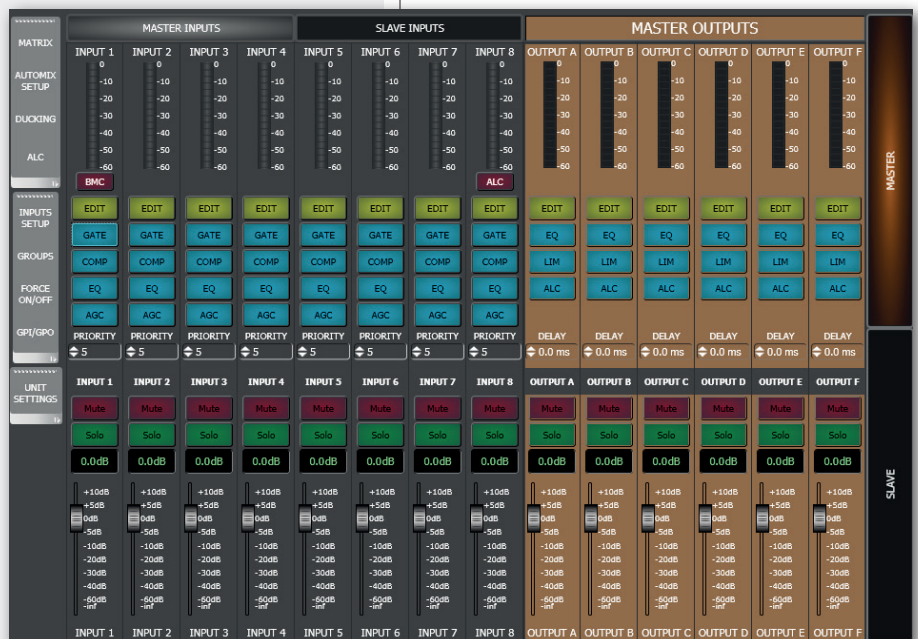
First, click UNIT SETTINGS (in the main window) to open the 'Global settings' window.

UNIT SETTINGS
Select either **MASTER** for the MASTER unit or **SLAVE** for the SLAVE unit.

LINK CONFIG
Select either **MASTER ONLY** if there is a single MZ 8060 or **MASTER + SLAVE** for two MZ 8060 linked each other.
The MASTER + SLAVE setting enables all SLAVE unit software controls.

SYSTEM CONFIG CONTROL
Click **SAVE to save the current software configuration to MZ 8060**.
The configuration includes the MASTER / SLAVE settings, the initial PRESET, audio input settings (e.g. PHANTOM power, phase, MIC/LINE settings, digital trimmers), logic inputs and outputs, all labels, remote controls and groups.
Click **LOAD** to load (into the software) the configuration previously saved to MZ 8060.

UNITS RESET
Click **RESET** to reset all MZ 8060.

PRESET MANAGER

It is possible to store and recall up to 16 different presets.

Click **UNIT SETTINGS** (in the main window) to open the 'Global settings' window, which includes the preset manager.

SAVE: saves the current settings to the selected preset (chosen from 1 to 16 in the **PRESET** list).
A **NAME** (text box) can be given to each preset.

Note: TO SAVE ALL PRESET NAMES TOO, IT IS NECESSARY TO SAVE THE ENTIRE CONFIGURATION. CLICK **UNIT SETTINGS** IN THE MAIN WINDOW, THEN **SAVE** IN **SYSTEM CONFIG CONTROL**.

RECALL: recalls the selected preset (previously saved).
WARNING: SAVE CURRENT SETTINGS TO A PRESET BEFORE RECALLING ANOTHER ONE (OTHERWISE ALL CURRENT SETTINGS WILL BE LOST)!
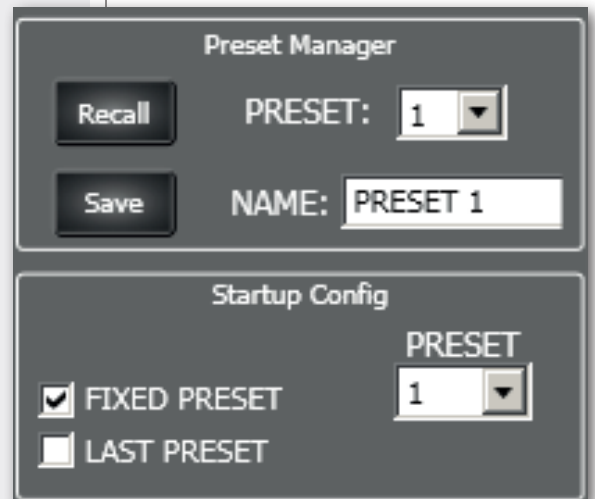
STARTUP CONFIG
It allows to choose which preset will be loaded as first after MZ 8060 start-up:
- **FIXED PRESET**, the one chosen in the **PRESET** list.
- **LAST PRESET**, the last saved or recalled preset (in chronological order).

INPUT CHANNEL SETTINGS

Each audio input has a level bar **1**, from –60 to 0 dB.
0 dB is the peak level before distortion.

All input controls are here listed:

EDIT 2 opens or closes the setting window of the equalizer, the compressor, the automatic gain control (AGC), the noise gate and the digital trimmer.
The INPUT 1 also includes all settings for BM 3003 paging microphone.

GATE 3 toggles the noise gate (automatic channel opening only if its signal exceeds the predetermined threshold).
To access its parameters, click **EDIT 2** and select the **DSP** window.

COMP 4 toggles the compressor.
To access its parameters, click **EDIT 2** and select the **DSP** window.

EQ 5 toggles the parametric equalizer.
To access its parameters, click **EDIT 2** and select the **EQ** window.

AGC 6 toggles the automatic gain control.
To access its parameters, click **EDIT 2** and select the **DSP** window.

PRIORITY 7: priority level setting from 0 (lowest) to 10 (highest) for the 'ducking' function (automatic level attenuation of one or more audio inputs when a signal is detected at the input with the highest priority level).

INPUT (n) **8**: audio input label that can be edited after a double-click on the text.

MUTE 9: (when lit) mutes the audio input.

SOLO 10: (when lit) opens the selected input only (and mutes all the others).

11 Audio input level control. The level can be adjusted by moving the fader (by keeping it with the mouse left button) or directly changing its value (above the fader: click the number and then insert the new value with the numeric keyboard).

Click **INPUTS SETUP** (in the main window) to edit other input options.
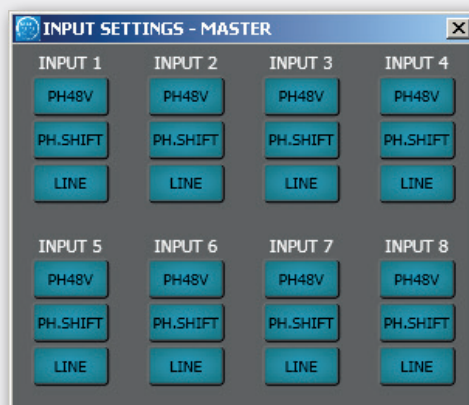
PH48V: toggles the Phantom power (48 V) in its respective input.
Phantom is usually needed by electret microphones.

Warning: never turn Phantom power on when not necessary (i.e. for LINE level signal from CD / MP3 players, dynamic microphones) or the used cable is unbalanced.
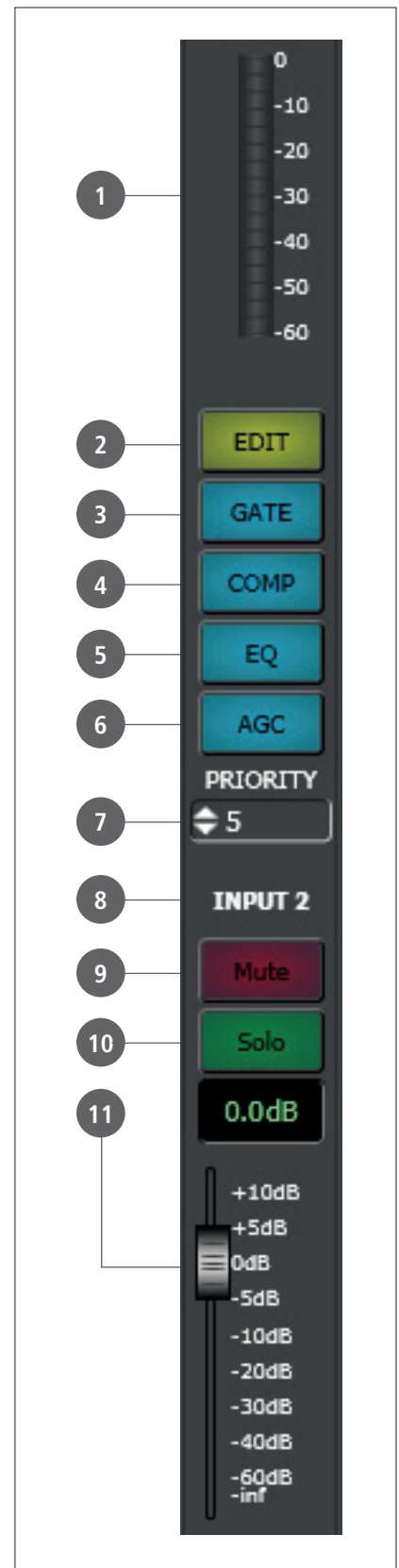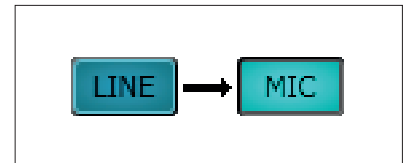
PH.SHIFT:
(when lit) inverts the signal phase.
Do not turn this option on (but particular needs).

LINE: changes the input sensitivity from **LINE** (for signals coming from other mixers, CD / MP3 players, etc.) to **MIC** (microphones that are not pre-amplified).
Click MIC to turn back to LINE.

EQUALIZER
Click **EDIT** to open the **EQ** setting window (of its respective audio input).

It is possible to set up to six independent filters (Filter 1, 2, 3, 4, 5, 6).

Enabled: click the checkbox to toggle each filter.
Move the mouse on its white background (making it green) to highlight a filter (if present) on the EQ graph.

Type:
- **DISABLED**: the filter is disabled (by-pass).
- **PEQ**: parametric equalizer that allows to adjust the level at the settable centre frequency and specify the Q factor (the adjusted level can be widened or narrowed).
- **SHELVING HIGH**: increase or decrease the level of all frequencies above the selected frequency by the specified amount.
- **SHELVING LOW**: increase or decrease the level of all frequencies below the selected frequency by the specified amount.
- The **BUTTERWORTH, LINKWITZ-RILEY, BESSEL** options are only available for the filters no.1 ('high-pass') and no.6 ('low-pass').

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Slope: filter slope setting (dB / oct).

Q: Q factor setting.

Gain (dB): filter gain setting.

Freq (Hz): either filter frequency selection or PEQ central frequency setting.

File Commands:
-   **SAVE**: it saves to PC the current equalization as .rde file.
-   **LOAD**: it loads from PC an equalization previously saved as .rde file.

Eq. options:
-   **FLATTEN**: it disables all filters (flat frequency response).
-   **BYPASS EQ.**: it disables the equalization, but without changing filter settings.
-   **PHASE PLOT**: if selected, the phase plot is shown (green line).

Devices commands:
-   **STORE**: it sends and stores the equalization to MZ 8060.
-   **SEND**: it sends (without storing) the equalization to MZ 8060.

For each filter, frequency and gain can be adjusted either graphically (through the mouse) by dragging the little coloured square or in an analytical way (by inserting the values in the cells or rotating the control).
The overall equalization is shown as a red line, the intervention of the selected filter as a green line, the intervention of a filter that is not selected as a white line.

For example, to apply a 12 dB attenuation for frequencies below 80 Hz: enable the filter 1 by clicking its 'Enabled' checkbox, select the SHELVING LOW filter type, set the gain to –12 dB and frequency to 80 Hz.
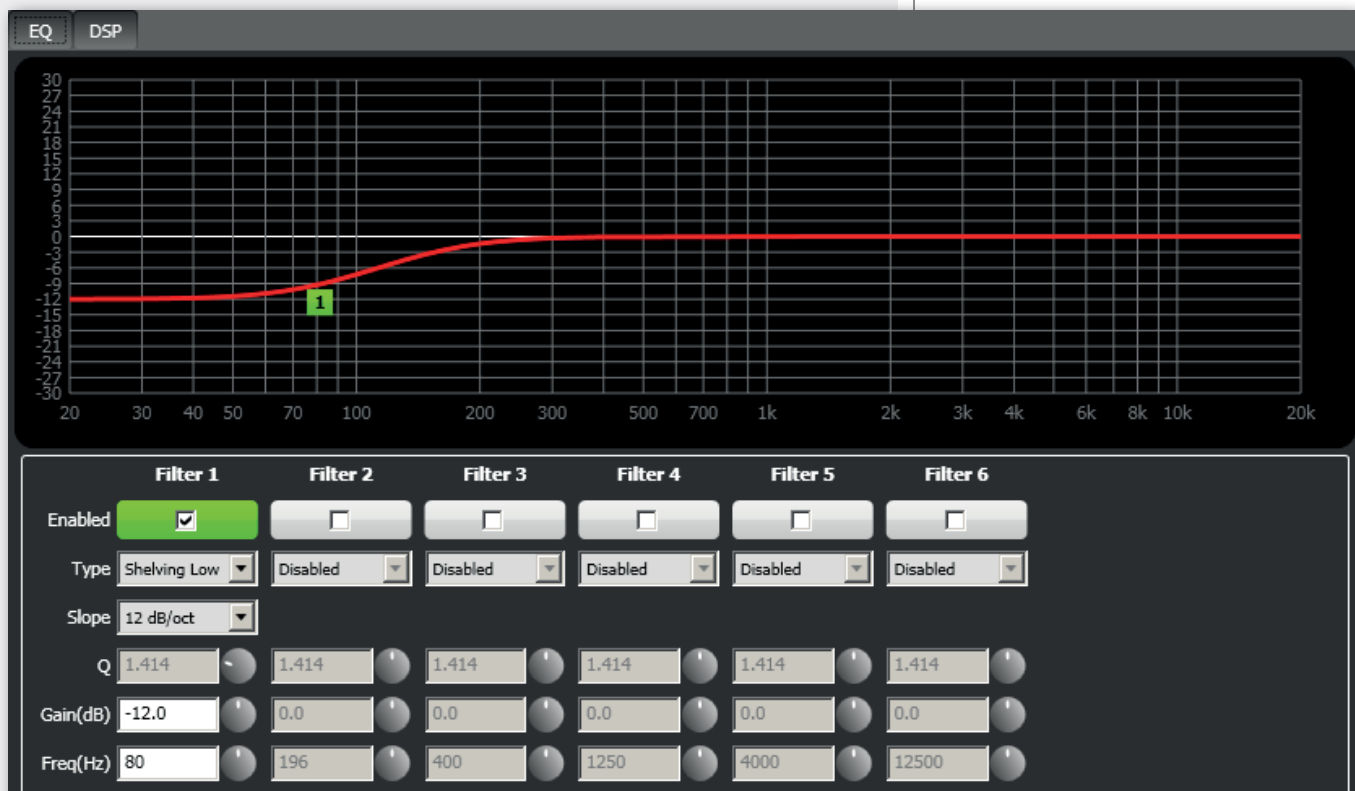
To add a second filter (i.e. PEQ): enable the filter 2, select the PEQ filter type, its gain and central frequency, then set its Q factor.
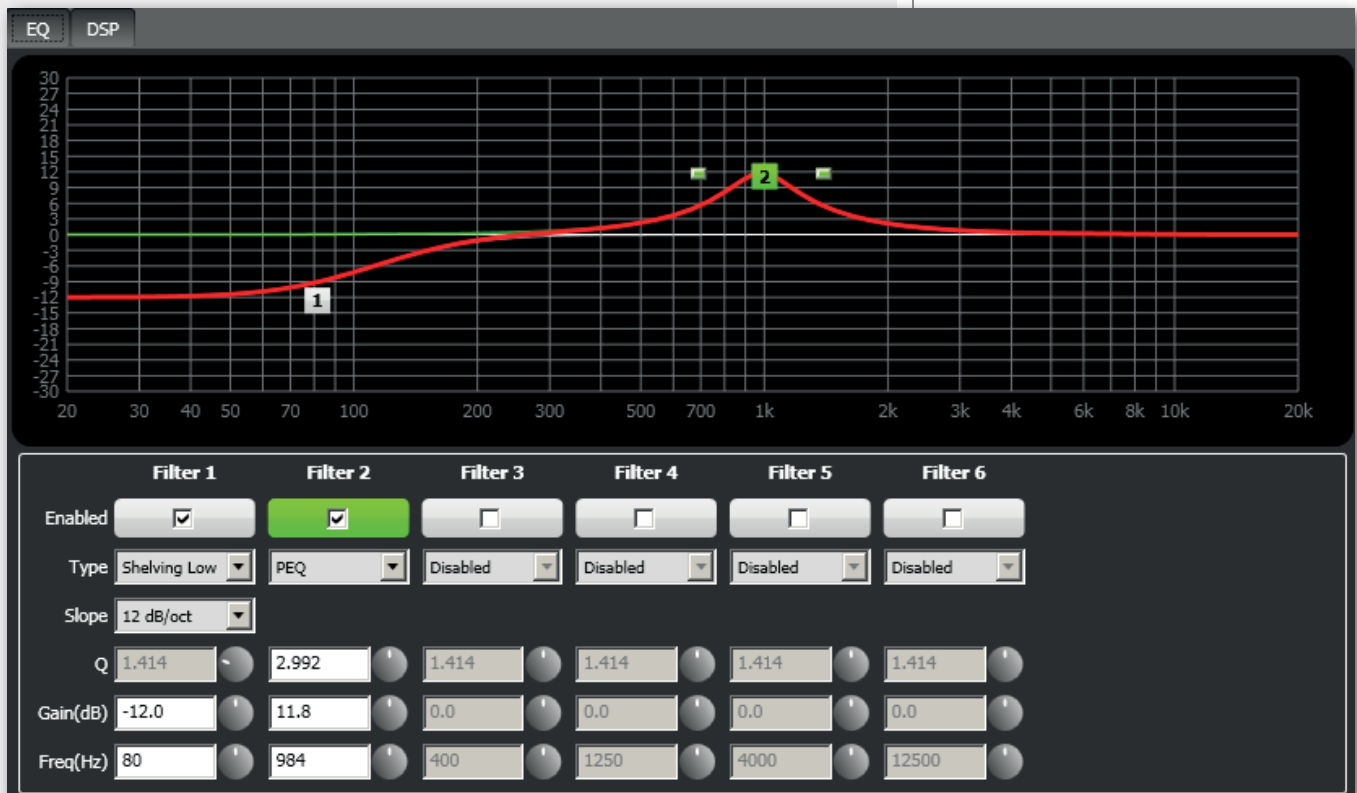
Finally, click either STORE (send and store) or SEND (send without storing) all EQ settings to MZ 8060.

COMPRESSOR

The compressor does not modify a signal having a level lower than the predetermined threshold and compresses a signal with higher level.
To access its parameters (of a single input), click **EDIT** on the respective channel and select the **DSP** window.
THRESHOLD and RATIO values can be adjusted either by moving the respective faders or dragging (with the mouse) the red points in the graph.

THRESHOLD: adjusts the level above which the compressor compresses the signal.

OUT GAIN: adjusts the output gain (useful to minimize the signal attenuation due to compression).

Attack: adjusts the compressor attack time.

Release: adjusts the compressor release time.

Ratio: sets the ratio between the original signal (above the threshold) and the compressor output.
For example: if ratio is 2:1, a signal that exceeds the threshold of 2 dB will be compressed to 1 dB (above the threshold).
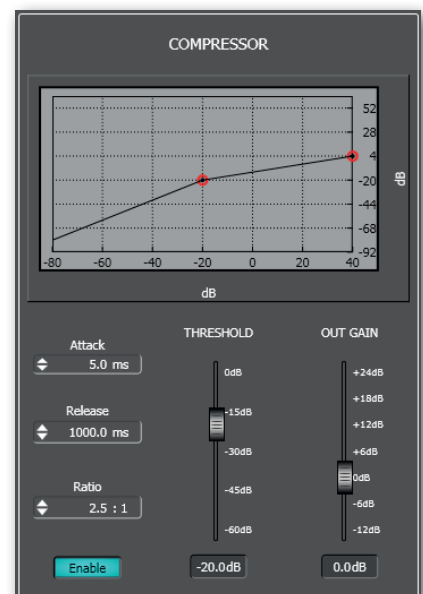Indicatively: 2:1 ratio is suitable for voice, from 2:1 to 4:1 for musical instruments.
A compressor having a ratio higher than 10:1 operates as a limiter.
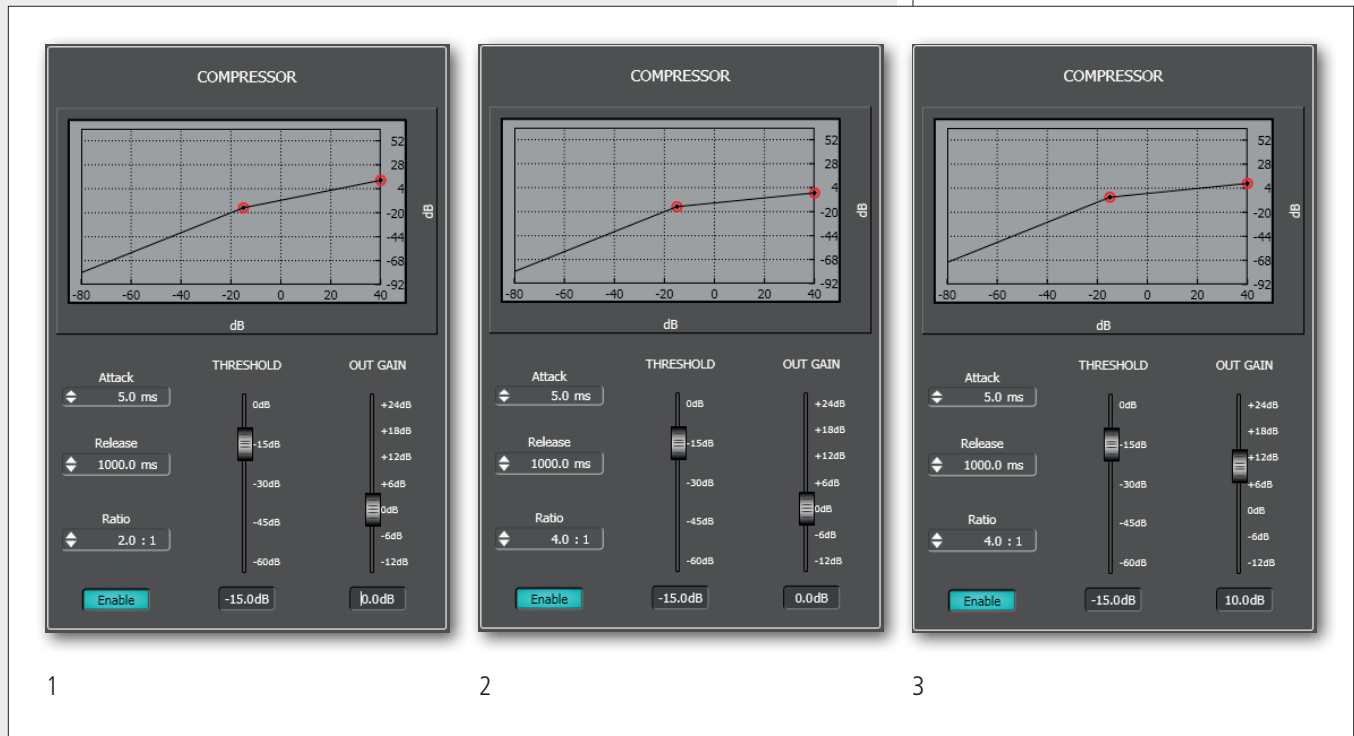A 1:1 ratio is an invalid setting, as it does not compress at all.

Enable: toggles the compressor (keep it on when using).

USE EXAMPLE:

1. Set **THRESHOLD** to a level lower than 0 dB (e.g. −15 dB).
2. Set **RATIO** (i.e. 4:1).
3. Adjust **OUT GAIN** in order to increase the signal output gain (to minimize the compressor attenuation, i.e. +10 dB).

1   2   3

AUTOMATIC GAIN CONTROL (AGC)

The automatic gain control does not modify a signal having a level lower than the predetermined threshold, but increases or reduces a signal above the threshold to get the desired average level ('target'), within the set time and according to the compression/expansion ratio.

To access AGC parameters (of a single input), click **EDIT** on the respective channel and select the **DSP** window.

THRESHOLD: adjusts the level above which the AGC operates.

TARGET: adjusts the optimal average signal level to be obtained.

Time: adjusts the AGC intervention time.

Ratio: sets the ratio between the original signal (above the threshold) and the AGC output.
Use a ratio in the 1.5 ÷ 3 range, in order to avoid excessive compression / expansion, which may lead to too low or distorted output signals.
A 1:1 ratio is an invalid setting, as the AGC does not operates at all.

Enable: toggles the AGC (keep it on when using).

USE EXAMPLE:

1. Set **THRESHOLD** to –45 dB.

2. Adjust the **TARGET** fader to fix the optimal average level of the signal (e.g. –20 dB).

3. Set the intervention **TIME** to 60 seconds.

4. Set **RATIO** (e.g. 2:1).

NOISE GATE

The noise gate automatically reduces an audio input level when its signal is below the defined threshold (or absent).
It allows to either mute a channel when no signal is detected at its input or open it when necessary.

To access the noise gate parameters (of a single input), click EDIT on the respective channel and select the **DSP** window.

THRESHOLD: adjusts the level below which the noise gate attenuates a signal.

GATE RANGE: adjusts the signal reduction when its level is below the threshold.

Attack: adjusts the noise gate attack time.

Release: adjusts the noise gate release time.

Enable: toggles the noise gate (keep it on when using).

USE EXAMPLE:

1. Set a low **THRESHOLD** (e.g. –50 dB) to ensure the immediate channel opening even with signals having a low amplitude (but higher than the background noise).

2. Adjust the **GATE RANGE** value according to your needs
(for instance, a –80 dB setting entails the audio input muting when its signal does not reach the chosen threshold).

3. Set both the ATTACK time (i.e. 1 ms) and the RELEASE time (i.e. 100 ms).
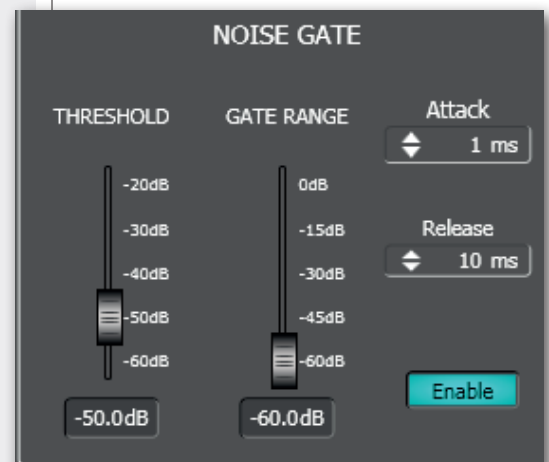
DIGITAL TRIMMER

The digital trimmer adjusts the audio input sensitivity.
It can help optimize a too low or too high audio signal.

To access the digital trimmer (of a single input), click **EDIT** on the respective channel and select the **DSP** window.

Set the **DIG. TRIM** fader to either a positive value to increase the input sensitivity or a negative value to attenuate (0 dB: no change).

Use this fader with caution, in order to avoid signal distortion (too high) or an insufficient level.

BM 3003 SETUP

To access the (common) BM 3003 paging microphone settings, click **EDIT of the INPUT 1** and select the **DSP** window.

Each of the three BM 3003 paging microphone zone buttons corresponds to a checkbox group: **A ZONE**, **B ZONE** and **C ZONE**.

Each BM 3003 zone button can be assigned to one or more buses (that correspond to MZ 8060 audio outputs A to F) of both MASTER unit and SLAVE unit (if available).

Click ENABLE when using BM 3003 paging microphones. In the main windows, the INPUT 1 (MASTER) red BMC lights up.
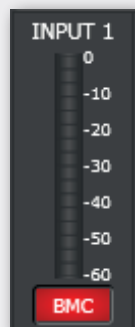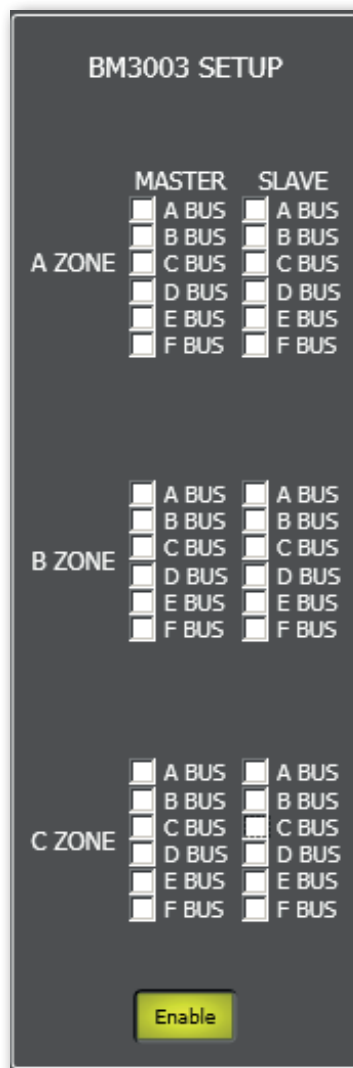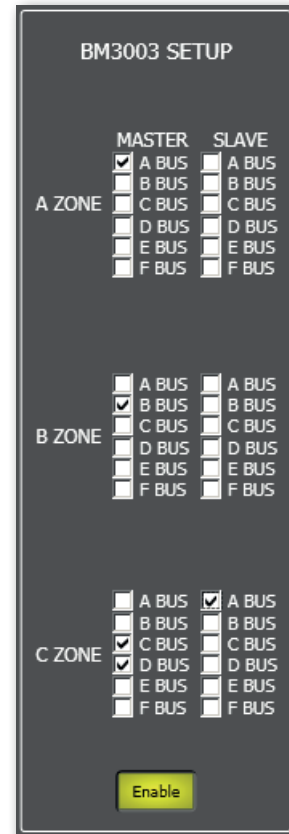
EXAMPLE

The first zone button (A ZONE) of all BM 3003 paging microphones is assigned to the MASTER A output only.

The second zone button (B ZONE) of all BM 3003 paging microphones is assigned to the MASTER B output only.

The third zone button (C ZONE) of all BM 3003 paging microphones is assigned to the MASTER C, D and SLAVE A outputs.

AUTOMATIC LEVEL CONTROL (ALC)

This function automatically adjusts the level of one or more audio outputs according to the detected ambient noise.
If the SLAVE unit is present, two **ALC** will be available: the first for MASTER unit audio outputs (with ALC enabled) only, the second for the SLAVE unit audio outputs (with ALC enabled) only. This selection needs to be made in advance by clicking either MASTER or SLAVE on the far right of the main window.

If ALC is enabled, the audio input no.8 (of the selected unit) is reserved for the microphone used as noise detector and its red ALC indication lights up (below the signal level bar).
Note: set the input 8 to MIC (INPUTS SETUP).

When the ambient noise (detected by the microphone linked to the input 8) is above the threshold, only audio outputs (of the same unit, MASTER or SLAVE) with ALC enabled are processed.

Click **ALC** in the main window.

THRESHOLD: adjusts the ambient noise level threshold above which the ALC operates.

GAIN / ATTEN: sets either the maximum (if positive) or the minimum (if negative) guaranteed level of audio outputs with ALC enabled.

Attack: adjusts the ALC attack time.

Release: adjusts the ALC release time.

Ratio: gain / attenuation ratio relative to each single decibel of detected noise above the threshold. For example, if the threshold is set to –40 dB and the ambient noise is 5 dB higher (–35 dB), a RATIO value 1.00:1 will entail either a 5 dB gain (if the GAIN / ATTEN fader is positive) or a 5 dB attenuation (if the GAIN / ATTEN fader is negative), a RATIO value 2.00:1 will entail a 10 dB gain / attenuation, etc... In any case, the maximum gain or attenuation will never exceed the limit level set by the GAIN / ATTEN fader.

Enable: toggles the ALC (keep it on when using).

AUTOMATIC LEVEL CONTROL (ALC)

23
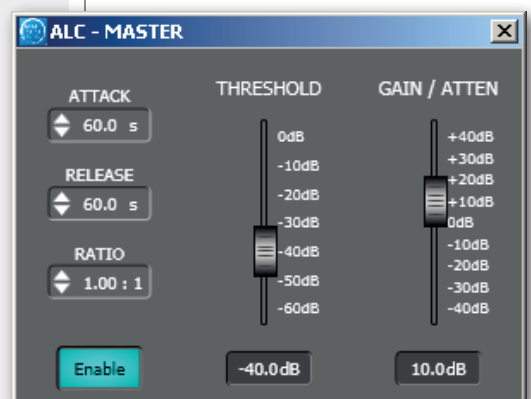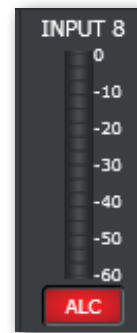
AUTOMIXER

The automixer function automatically controls and manages the maximum number of audio inputs (or channels) that are completely open at the same time, reducing the level of those in excess and compensating the volume of the final mix.
This is useful, for instance, when using multiple microphones in the same room.

Two independent automixers are available: **AMIX1** and **AMIX 2**.

Important: in the MIXER MATRIX section, it is necessary to assign all matrix nodes (input / output combinations) to one of the two automixers (see the respective manual section).
The assignment of each matrix node to one of the two is exclusive.

If the SLAVE unit is present, four automixers will be available: two for the MASTER unit BUS / audio outputs only, two for the SLAVE unit BUS / audio outputs only.
The selection needs to be made in advance by clicking either MASTER or SLAVE on the far right of the main window.

Click **AUTOMIX SETUP** in the main window.

Select the automixer to set by clicking either **AMIX1** or **AMIX 2**.

MAX NOM: setting of the maximum number of audio inputs (or channels) that can be fully opened at the same time.

NOM ATTENUATION: mixed signal attenuation setting.
The attenuation is increased gradually (the set value in decibels) in function of the doubling of the number of mixed inputs.
For example, if the value is set to –3 dB, with two open inputs the mixed signal will be reduced by 3 dB, with four open inputs by 6 dB, with eight open inputs by 9 dB, etc...

SENSITIVITY: setting of the needed audio signal (relative level expressed in decibels) above the ambient noise level (detected by the same input) needed to fully open a channel in the mixed signal.
For example, if the value is set to 6 dB, when using a microphone, a channel will be fully opened as soon as the detected signal is 6 dB above the detected ambient noise.

OFF GAIN: setting of the attenuation (in decibels) applied to signals of all channels detected as 'off', in which the signal does not reach the needed level (set by the SENSITIVITY parameter) above the ambient noise.

HOLD TIME: time expressed in seconds, in which a channel still stays open from when its detected signal level is below the threshold necessary for full activation.

DEFAULT CHANNEL: setting of a preferential channel that is always kept open.
It is possible to select (only) one of the following three options:
- **DISABLED:** no default channel, only channels having a signal level above the activation threshold get opened (and within the maximum allowed number of open channels).
- **STATIC:** the default channel is the one selected through the below **STATIC CHANNEL** parameter.
- **LAST ON:** the default channel is the last activated (in chronological order).

PRIORITY FUNCTION ('DUCKING')

This function allows to get automatic level attenuation of one or more audio inputs (if selected with **DUCK** buttons) when a signal is detected at the input(s) with the highest priority level.
If there are more inputs all having the highest priority level, the function can be activated by each of these.
If the SLAVE unit is present, two 'ducking' functions will be available: the first for MASTER unit audio outputs only, the second for the SLAVE unit audio outputs only. This selection needs to be made in advance by clicking either MASTER or SLAVE on the far right of the main window.

The priority level, from 0 (lowest) to 10 (highest), is set in each input channel by PRIORITY parameters.

Click **DUCKING** in the main window.

The **THRESHOLD** fader only refers to audio input(s) having the highest priority level and allows to set the threshold (signal level expressed in decibels) above which the 'ducking' function gets activated and all the other inputs (with lower priority) get attenuated.
The reference audio source is / are the only input / inputs all having the absolute highest priority level.

RANGE: allows to adjust the attenuation (dB) to apply to all inputs with lower priority levels when the 'ducking' function is in progress.

Attack: adjusts the priority function attack time.

Release: adjusts the priority function release time.

DUCK: each button toggles the 'ducking' function for the respective input.
If the SLAVE unit is present, it will be possible to select either the MASTER unit inputs (by clicking **MASTER INPUTS**) or the SLAVE unit inputs (by clicking **SLAVE INPUTS**).

AUDIO OUTPUTS AND SLAVE UNIT CONTROLS

The 6 audio outputs (A-F) of the MASTER unit (single or first MZ 8060) are on the right of the software main window.

NOTE: ALL CONTROLS OF THE 6 AUDIO OUTPUTS OF THE SLAVE UNIT (OPTIONAL SECOND MZ 8060, IF PRESENT) CAN BE ENABLED BY CLICKING UNIT SETTINGS IN THE MAIN WINDOW AND THEN SELECTING **MASTER+SLAVE** AS LINK CONFIG CHOICE.

On the far right of the software main window, there are two buttons to choose which MZ 8060 needs to be edited: click either **MASTER** to access all MASTER unit parameters or **SLAVE** (if enabled) for all SLAVE unit parameters.

MASTER AND SLAVE OUTPUTS HAVE DIFFERENT BACKGROUND COLOURS.

NOW, IT IS POSSIBLE TO SELECT THE SLAVE UNIT AUDIO INPUTS TOO.

NOTE:
AFTER SELECTING THE SLAVE OUTPUTS, THE LOGIC INPUT/OUTPUT (GPI/GPO) SETTINGS ARE NOT AVAILABLE.

AUDIO OUTPUT SETTINGS

Each audio output has a level bar **1** , from –60 to 0 dB.
0 dB is the peak level before distortion.

All audio output controls are now listed.

EDIT **2** opens or closes the setting window of both the equalizer and the limiter.

EQ **3** toggles the parametric equalizer.
Click **EDIT** **2** to access its parameters.

LIM **4** toggles the limiter.
Click **EDIT** **2** to access its parameters.

ALC **5** toggles the automatic level control that operates according to the ambient noise detected by a microphone (connected to the input no.8).
Read the 'Automatic Level Control (ALC)' manual section.

DELAY **6** delay time setting (in milliseconds, up to max. 5 seconds) applied to the output signal.
Except in specific cases (e.g. second line loudspeakers to be delayed), leave this parameter to 0.

**OUTPUT (n)** **7** : audio output label that can be edited after a double-click on the text.

MUTE **8** (when lit) mutes the audio output.

SOLO **9** (when lit) open the selected output only (and mutes all the others).

10 Audio output level control.
The level can be adjusted by moving the fader (by keeping it with the mouse left button) or directly changing its value (above the fader: click the number and then insert the new value with the numeric keyboard).
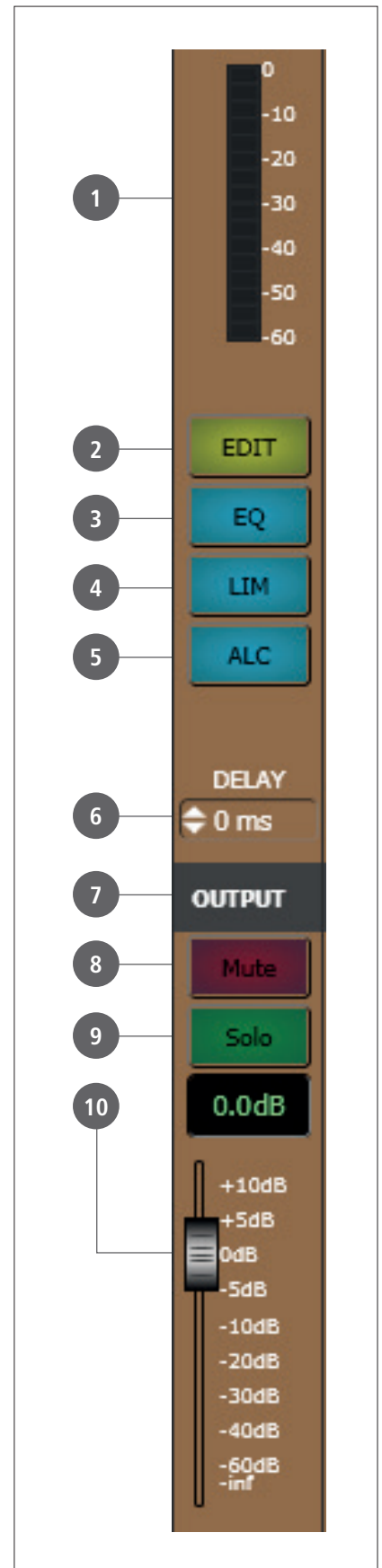
EQUALIZER

Click **EDIT** to open the EQ/LIMITER setting window (of the respective audio output).

It is possible to set up to six independent filters (Filter 1, 2, 3, 4, 5, 6).

Enabled: click the checkbox to toggle each filter.
Move the mouse on its white background (making it green) to highlight a filter (if present) on the EQ graph.

Type:
- **DISABLED**: the filter is disabled (by-pass).
- **PEQ**: parametric equalizer that allows to adjust the level at the settable centre frequency and specify the Q factor (the adjusted level can be widened or narrowed).
- **SHELVING HIGH**: increase or decrease the level of all frequencies above the selected frequency by the specified amount.
- **SHELVING LOW**: increase or decrease the level of all frequencies below the selected frequency by the specified amount.
- The **BUTTERWORTH, LINKWITZ-RILEY, BESSEL** options are only available for the filters no.1 ('high-pass') and no.6 ('low-pass').

Slope: filter slope setting (dB / oct).

Q: Q factor setting.

Gain (dB): filter gain setting.

Freq (Hz): either filter frequency selection or PEQ central frequency setting.

File Commands:
- **SAVE**: it saves to PC the current equalization as .rde file.
- **LOAD**: it loads from PC an equalization previously saved as .rde file.

Eq. options:
- **FLATTEN**: it disables all filters (flat frequency response).
- **BYPASS EQ.**: it disables the equalization, but without changing filter settings.
- **PHASE PLOT**: if selected, the phase plot is shown (green line).

Devices commands:
- **STORE**: it sends and stores the equalization to MZ 8060.
- **SEND**: it sends (without storing) the equalization to MZ 8060.

For each filter, frequency and gain can be adjusted either graphically (through the mouse) by dragging the little coloured square or in an analytical way (by inserting the values in the cells or rotating the control).
The overall equalization is shown as a red line, the intervention of the selected filter as a green line, the intervention of a filter that is not selected as a white line.

For example, to apply a 6 dB attenuation at 800 Hz through a parametric equalizer: enable the filter 1 by clicking its 'Enabled' checkbox, select the PEQ filter type, set its Q to 1, its gain to –6 dB and frequency to 800 Hz.
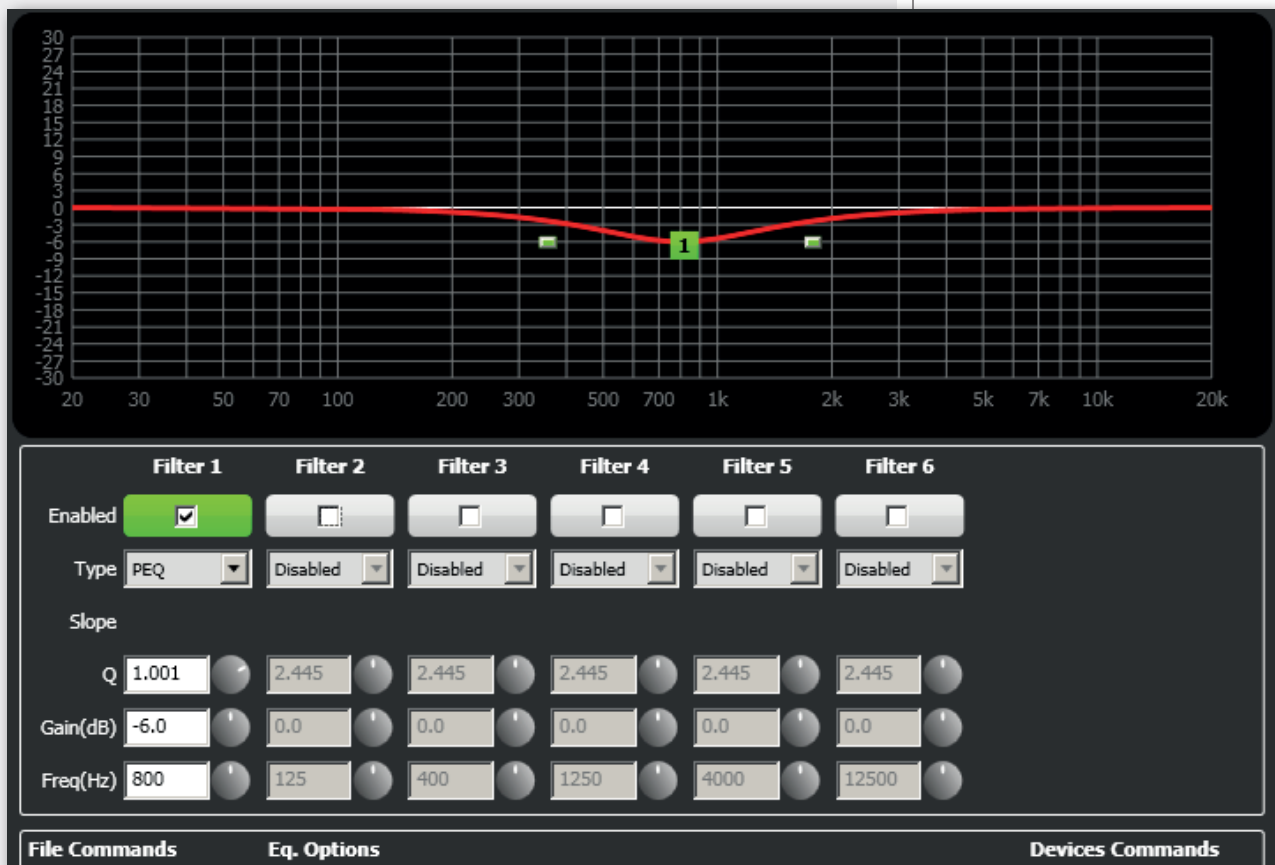
Finally, click either STORE (send and store) or SEND (send without storing) to send all EQ settings to MZ 8060.

LIMITER

The limiter does not modify a signal having a level lower than the predetermined threshold and limits a signal with higher level.
A limiter is in fact a compressor having a high compression ratio.
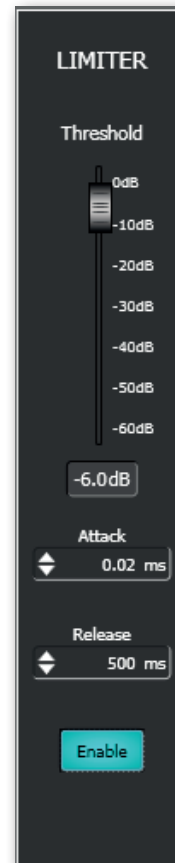It can be really useful to avoid signal distortion due to too high levels.

THRESHOLD: adjusts the level above which the limiter compresses the signal (suggested value: –6 dB).

Attack: adjusts the limiter attack time.

Release: adjusts the limiter release time.

Enable: toggles limiter on (suggested setting) / off.

MIXER MATRIX

Click MATRIX (in the software main window) to access the audio matrix window to match the 8 MASTER inputs (plus the 8 SLAVE inputs, if available) and either the 6 MASTER outputs (A-F buses) or the 6 SLAVE outputs (A-F buses, if available).

IMPORTANT:
- In the **MIXER MATRIX** window, click either **MASTER INPUTS** (to select the 8 MASTER inputs) or **SLAVE INPUT** (to select the 8 SLAVE INPUTS).

- MASTER or SLAVE output / bus selection is made by clicking either the **MASTER** or the **SLAVE** button on the far right of the software main window.

- There can be up to 4 matrix tables:

    1.   MASTER inputs to MASTER outputs / buses.
    2.   SLAVE inputs to MASTER outputs / buses.
    3.   SLAVE inputs to SLAVE outputs / buses.
    4.   MASTER inputs to SLAVE outputs / buses.

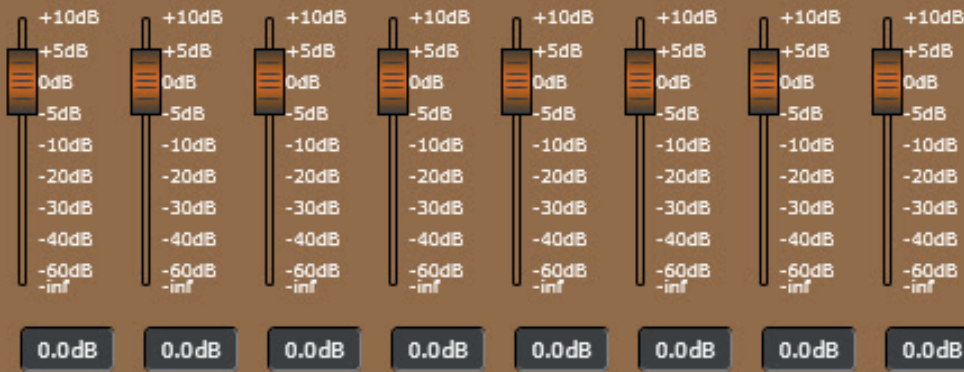## MIXER MATRIX - MASTER

		INPUT 1	INPUT 2	INPUT 3	INPUT 4	INPUT 5	INPUT 6	INPUT 7	INPUT 8	
0.0 dB	A BUS	ON	ON	ON	ON	ON	ON	ON	ON	MASTER INPUT
		A-MIX1	A-MIX1	A-MIX1	A-MIX1	A-MIX1	A-MIX1	A-MIX1	A-MIX1	
		A-MIX2	A-MIX2	A-MIX2	A-MIX2	A-MIX2	A-MIX2	A-MIX2	A-MIX2	
0.0 dB	B BUS	ON	ON	ON	ON	ON	ON	ON	ON	
		A-MIX1	A-MIX1	A-MIX1	A-MIX1	A-MIX1	A-MIX1	A-MIX1	A-MIX1	
		A-MIX2	A-MIX2	A-MIX2	A-MIX2	A-MIX2	A-MIX2	A-MIX2	A-MIX2	
0.0 dB	C BUS	ON	ON	ON	ON	ON	ON	ON	ON	
		A-MIX1	A-MIX1	A-MIX1	A-MIX1	A-MIX1	A-MIX1	A-MIX1	A-MIX1	
		A-MIX2	A-MIX2	A-MIX2	A-MIX2	A-MIX2	A-MIX2	A-MIX2	A-MIX2	
0.0 dB	D BUS	ON	ON	ON	ON	ON	ON	ON	ON	
		A-MIX1	A-MIX1	A-MIX1	A-MIX1	A-MIX1	A-MIX1	A-MIX1	A-MIX1	
		A-MIX2	A-MIX2	A-MIX2	A-MIX2	A-MIX2	A-MIX2	A-MIX2	A-MIX2	
0.0 dB	E BUS	ON	ON	ON	ON	ON	ON	ON	ON	
		A-MIX1	A-MIX1	A-MIX1	A-MIX1	A-MIX1	A-MIX1	A-MIX1	A-MIX1	
		A-MIX2	A-MIX2	A-MIX2	A-MIX2	A-MIX2	A-MIX2	A-MIX2	A-MIX2	
0.0 dB	F BUS	ON	ON	ON	ON	ON	ON	ON	ON	SLAVE INPUT
		A-MIX1	A-MIX1	A-MIX1	A-MIX1	A-MIX1	A-MIX1	A-MIX1	A-MIX1	
		A-MIX2	A-MIX2	A-MIX2	A-MIX2	A-MIX2	A-MIX2	A-MIX2	A-MIX2	

ROW LEVELS

+10dB
+5dB
0dB
-5dB
-10dB
-20dB
-30dB
-40dB
-60dB
-inf

0.0dB	0.0dB	0.0dB	0.0dB	0.0dB	0.0dB	0.0dB	0.0dB

MATRIX NODES

For each bus, there is:
- A rotary control (−90 to +10 dB) for any general settings of the signal level in the respective BUS (the default setting is 0 dB: no adjustment).
- A button (in the picture above **A BUS**) for the choice of the bus to which ROW LEVEL sliders (matrix node levels, see below) are assigned, sliders that change their colour according to the selected BUS.

Every audio input has the following controls:

ON: toggles the audio input in the correspondent bus. It is the main switch of the node.

A-MIX1: assigns the node to the control of the automixer no.1, which automatically adjusts the audio input level according to the number of open channels.

A-MIX2: assigns the node to the control of the automixer no.2 (alternative to the automixer no.1).

IMPORTANT: automixers need to be set in the AUTOMIX SETUP window, see the 'Automixer' manual section.

ROW LEVELS: each matrix node has a slider to adjust its level (from −40 a +20 dB, default setting is 0 dB: no adjustment). The BUS selection is made by clicking one of the buttons from **A BUS** to **F BUS**. Sliders change their colour according to the selected BUS.
With the MASTER unit only, slider total number is 48 (8 inputs x 6 BUS / outputs). If the SLAVE unit is linked, there will be 192 sliders (16 inputs x 6 BUS / outputs x 2 matrices).
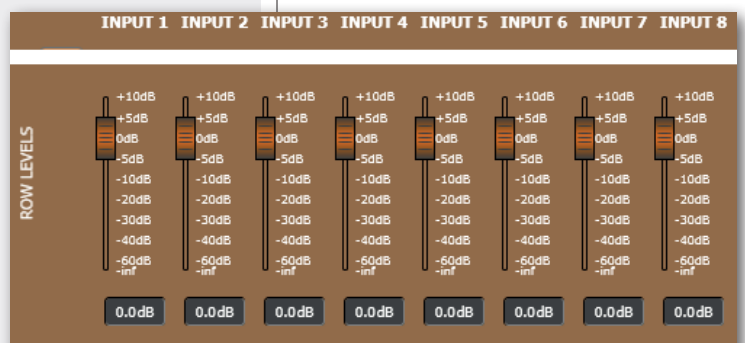
A 'group' is a setting that includes some audio inputs and outputs. Any event (e.g. the activation of a GPI logic input) that refers to a group simultaneously acts on all audio inputs and outputs assigned to that group.

For each MZ 8060 (MASTER or SLAVE), it is possible to set up to 16 **CONTROL** groups and 16 **MUTE** groups.
Before editing group settings, always check (on the far right of the software main window) which unit (MASTER or SLAVE) is selected.
Click **GROUPS** in the main window.

CONTROL GROUP

CTRL GROUP (n): group name that can be edited after a double-click on the text.

EDIT: opens the window of audio input / output selection.
The green light beside (when lit) means that at least one input / output has been selected (if the light is off, the group has not been configured yet).

FADER: group level control (default setting: 0 dB). The level can be adjusted by moving the fader (by dragging it with the mouse left button) or directly changing its value (click the number and then insert the new value with the numeric keyboard).

ENABLE: toggles the group (keep it on when using).

MUTE GROUP

MUTE GROUP (n): group name that can be edited after a double-click on the text.

EDIT: opens the window of audio input / output selection.
The green light beside (when lit) means that at least one input / output has been selected (if the light is off, the group has not been configured yet).

ENABLE: toggles the group (keep it on when using).
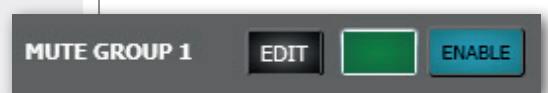
ASSIGNMENT OF AUDIO INPUTS AND OUTPUTS TO A GROUP

Click **EDIT** of either a CONTROL or MUTE group to open the window of audio input / output selection.

MASTER INPUTS: click checkboxes to select the MASTER unit audio inputs to be included in the group.

SLAVE INPUTS: click checkboxes to select the SLAVE unit audio inputs (if available) to be included in the group.

OUTPUT: click checkboxes to select the audio outputs (of either the MASTER or SLAVE unit currently selected) to be included in the group.

DESELECT: clear all checkboxes.

After selecting audio inputs and outputs, close the window (by clicking the X in the top right-hand corner).

FORCE ON/OFF (SET BY THE MASTER UNIT CONFIGURATION ONLY)

This function allows to either increase (FORCE ON) or reduce (FORCE OFF) an input level forcedly to both MASTER and SLAVE units.

Its activation can be made by an external command linked to a MASTER UNIT logic input (GPIN).

Click **FORCE ON/OFF** in the main window.
(M) INPUT n: MASTER unit input.
(S) INPUT n: SLAVE unit input.
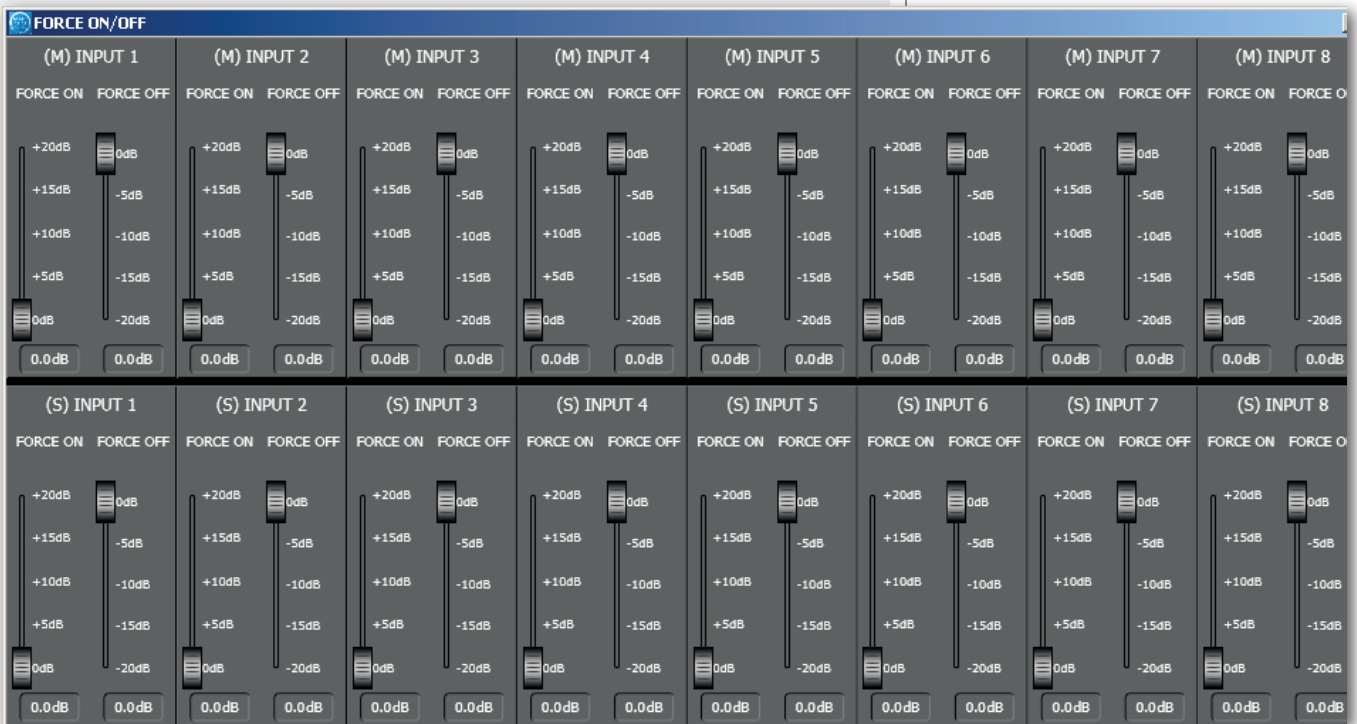
AUDIO INPUT SETTINGS

FORCE ON: SETS HOW MUCH (UP TO +20 dB) THE AUDIO SIGNAL LEVEL WILL BE INCREASED DURING A FORCE ON EVENT.

FORCE OFF: SETS HOW MUCH (UP TO −20 dB) THE AUDIO SIGNAL LEVEL WILL BE REDUCED DURING A FORCE OFF EVENT.

NOTE: THE USE OF FORCE ON / FORCE OFF FADERS IS EXCLUSIVE (THE ADJUSTMENT OF ONE CLEARS THE SETTING OF THE OTHER).

AUDIO INPUT SETTINGS

35
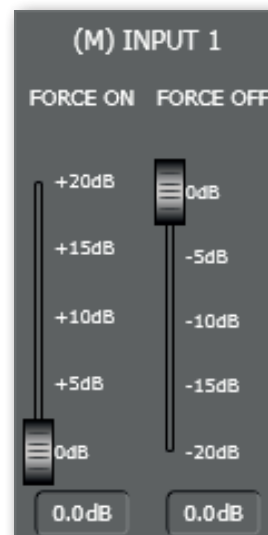
GPI/GPO (LOGIC INPUTS AND OUTPUTS, MASTER UNIT ONLY) ➡ •↗

In the MASTER unit only, it is possible to set up to 10 logic inputs (GPIN), 4 logic outputs (GPOUT 1 – 4) plus 2 (GPOUT5 and GPOUT6, marked as RL1, RL2 on the rear panel) with relay.

Click **GPI/GPO** in the main window.

LOGIC INPUTS (GPIN)

The GPIN screen page allows to set up to 10 logic inputs.
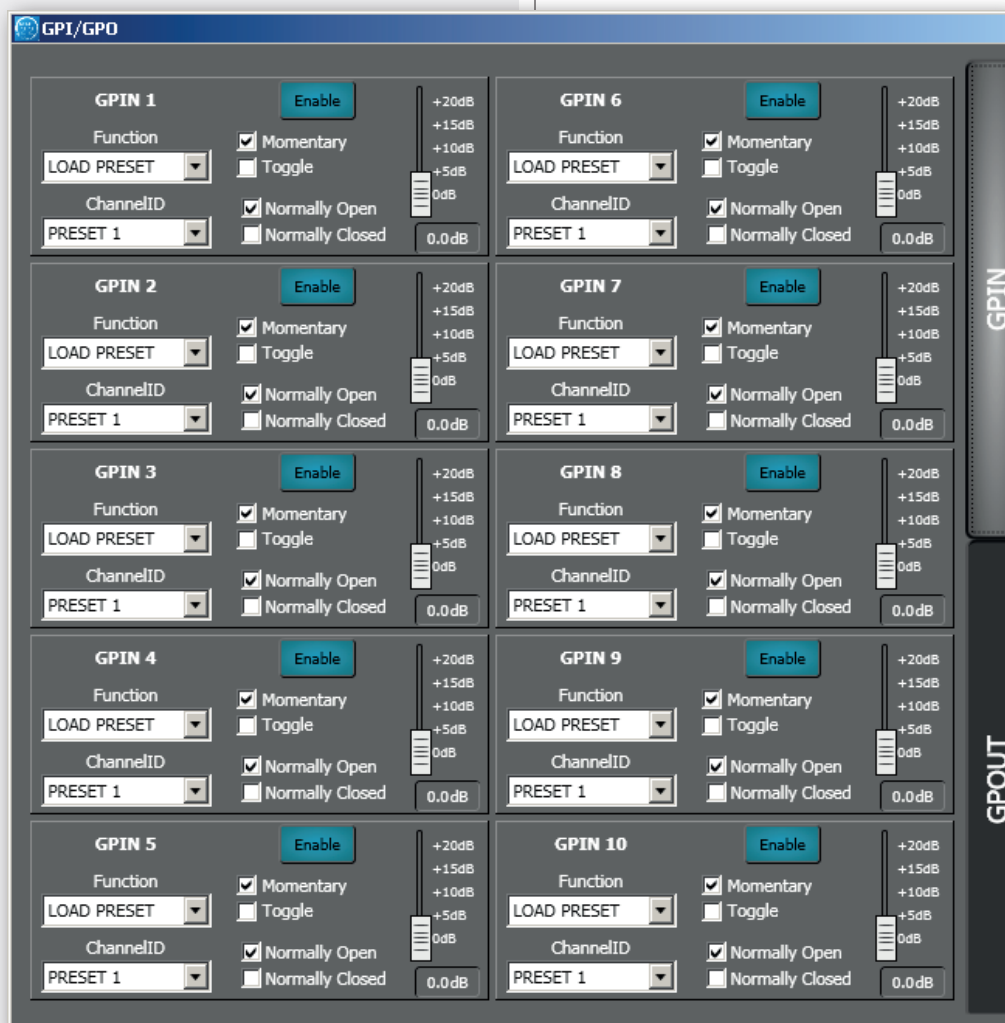
Each logic input includes the following parameters:

Function: selection of the function assigned to the logic input (read the next paragraph).

ChannelID: parameter depending on the selected FUNCTION, it can be either a preset or an audio input / output or a group.

Momentary: the selected function is applied only when the logic input is kept activated.

Toggle: every logic input activation toggles the selected function.

Normally Open: the logic input is activated when its contacts are shorted (or off when its contacts are open).

Normally Closed: the logic input is activated when its contacts are open (or off when its contacts are shorted).

SLIDER: allows to adjust the signal level.

Enable: toggles the logic input (keep it on when used).

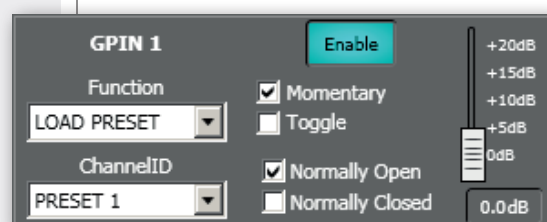| FUNCTION	CHANNEL ID (parameter)	DESCRIPTION
LOAD PRESET	PRESET	It loads the selected preset.
FORCE ON/OFF	INPUT	Forced level gain / attenuation of the selected audio input, setting that needs to be made in the FORCE ON/OFF section.
MUTE ALL OUTS	---	It mutes all audio outputs.
MUTE MASTER INPUT	INPUT	It mutes the selected audio input to the MASTER unit audio outputs.
MUTE MASTER OUTPUT	OUTPUT	It mutes the selected audio output of the MASTER unit.
MASTER MUTE GROUP	MUTE GROUP	It turns on the selected MUTE group of the MASTER unit.
MASTER IN LEV INCR	INPUT	It increases the selected audio input level to the MASTER unit outputs (use the slider to adjust).
MASTER IN LEV DECR	INPUT	It reduces the selected audio input level to the MASTER unit outputs (use the slider to adjust).
MASTER OUT LEV INCR	OUTPUT	It increases the selected MASTER unit audio output level (use the fader to adjust).
MASTER OUT LEV DECR	OUTPUT	It reduces the selected MASTER unit audio output level (use the slider to adjust).
MASTER CTRL GRP INCR	CTRL GROUP	It increases the selected MASTER unit CTRL GROUP signal level (use the slider to adjust).
MASTER CTRL GRP DECR	CTRL GROUP	It reduces the selected MASTER unit CTRL GROUP signal level (use the slider to adjust).
MUTE SLAVE INPUT	INPUT	It mutes the selected audio input to the SLAVE unit audio outputs.
MUTE SLAVE OUTPUT	OUTPUT	It mutes the selected audio output of the SLAVE unit.
SLAVE MUTE GROUP	MUTE GROUP	It turns on the selected MUTE group of the SLAVE unit.
SLAVE IN LEV INCR	INPUT	It increases the selected audio input level to the SLAVE unit outputs (use the slider to adjust).
SLAVE IN LEV DECR	INPUT	It reduces the selected audio input level to the SLAVE unit outputs (use the slider to adjust).
SLAVE OUT LEV INCR	OUTPUT	It increases the selected SLAVE unit audio output level (use the fader to adjust).
SLAVE OUT LEV DECR	OUTPUT	It reduces the selected SLAVE unit audio output level (use the slider to adjust).
SLAVE CTRL GRP INCR	CTRL GROUP	It increases the selected SLAVE unit CTRL GROUP signal level (use the slider to adjust).
SLAVE CTRL GRP DECR	CTRL GROUP	It reduces the selected SLAVE unit CTRL GROUP signal level (use the slider to adjust).

LOGIC OUTPUT (GPOUT)

Click **GPI/GPO** in the main window, then select **GPOUT** in the GPI/GPO window.

Six logic outputs are available, of which both the fifth **GPOUT 5 (RL1)** and the sixth **GPOUT6 (RL2)** have relays.

NOTE: AS FACTORY DEFAULT, THE **GPOUT 5 (RL1)** LOGIC OUTPUT WITH RELAY IS ASSIGNED TO THE DEVICE FAULT INDICATION. TO ENABLE ITS USER SOFTWARE CONTROL, IT IS NECESSARY TO SET THE **JMP1** INTERNAL JUMPER (SEE THE RESPECTIVE MANUAL SECTION).

Each logic output includes the following parameters:

Function: selection of the function assigned to the logic output (read the next paragraph).

ChannelID: parameter depending on the selected FUNCTION, it can be either a preset, an audio input / output or a MUTE group.

GPOUT 1, 2, 3, 4

Normally High: the logic output is normally high (5 V) and will switch to low (0) when activated.

Normally Low: the logic output is normally low (0) and will switch to high (5 V) when activated.

GPOUT 5, 6 (RL1, RL2)

Normally Open: the internal relay contacts are normally open and will get shorted as soon as the logic output is activated.

Normally Closed: the internal relay contacts are normally closed (shorted) and will get open as soon as the logic output is activated.

Enable: toggles the logic output (keep it on when using).
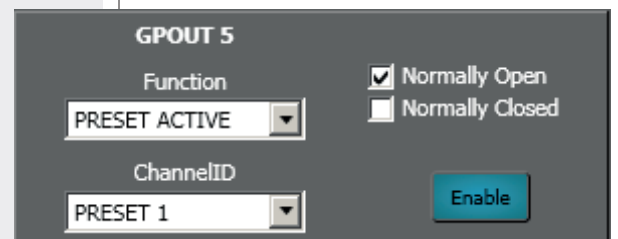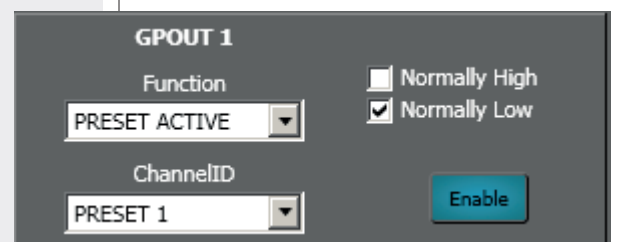
FUNCTION	CHANNEL ID (parameter)	DESCRIPTION
PRESET ACTIVE	PRESET	The logic output will get activated when the selected preset is recalled.
MUTE ALL OUTS	---	The logic output will get activated when all audio outputs are muted due to the activation of a logic input set to MUTE ALL OUTS.
MUTE MASTER INPUT	INPUT	The logic output will get activated when the selected audio input is muted to the MASTER unit audio outputs.
MUTE MASTER OUTPUT	OUTPUT	The logic output will get activated when the selected MASTER unit audio output is muted.
MASTER MUTE GROUP	MUTE GROUP	The logic output will get activated when a MASTER unit MUTE group is turned on.
MUTE SLAVE INPUT	INPUT	The logic output will get activated when the selected audio input is muted to the SLAVE unit audio outputs.
MUTE SLAVE OUTPUT	OUTPUT	The logic output will get activated when the selected SLAVE unit audio output is muted.
SLAVE MUTE GROUP	MUTE GROUP	The logic output will get activated when a SLAVE unit MUTE group is turned on.

AC RDNET IN/OUT PLUG BOARD INSTALLATION

The **AC RDNET IN/OUT PLUG** optional board allows the MZ 8060 remote automixer control via RDNET.
Refer to the RDNET CONTROL 8 manual about the complete RDNET software use and further information.

1. Unplug the power cord.

2. Remove the blank panel on the side of the input for the power cord 22 (on the MZ 8060 rear panel) by unscrewing the 4 screws with a TORX T10 screwdriver.

3. Connect the flat cable inside the MZ 8060 automixer to the AC RDNET IN/OUT PLUG board.

4. Insert the AC RDNET IN/OUT PLUG board into the MZ 8060 automixer slot and fasten it with the 4 screws.
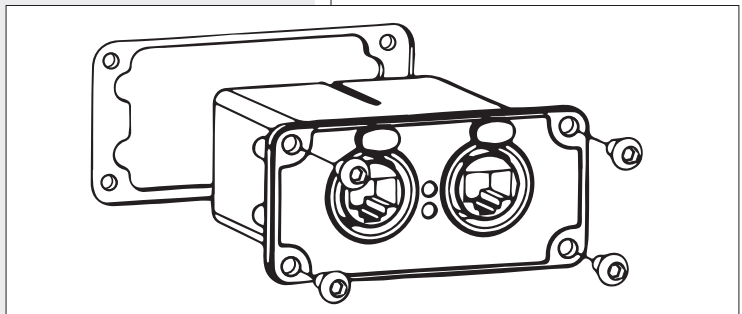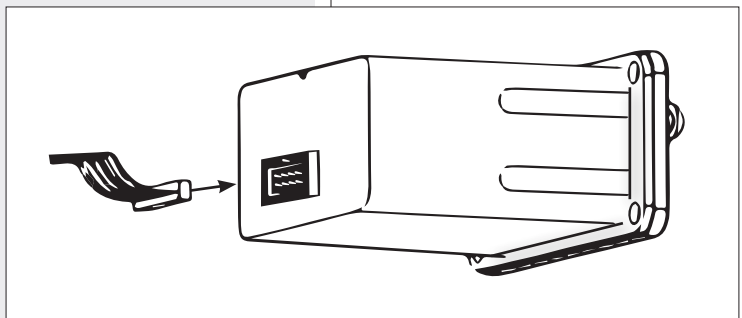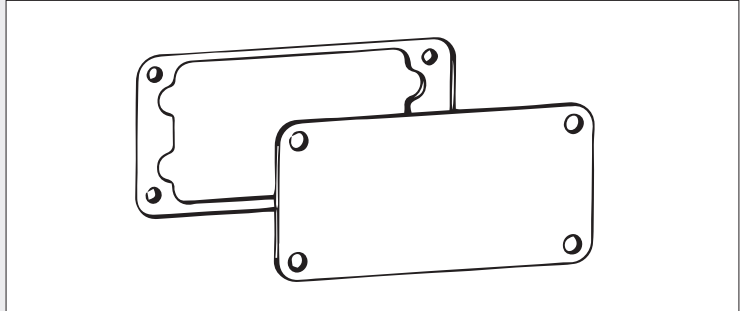
Each of the eight RDNET CONTROL 8 ports can be connected to max. 32 (daisy-chained) compatible audio devices.
Each **RDNET CONTROL 8** port shall be connected to the **DATA INPUT** of a RDNET device and the **DATA LINK** parallel output to the **DATA INPUT** of the next device.
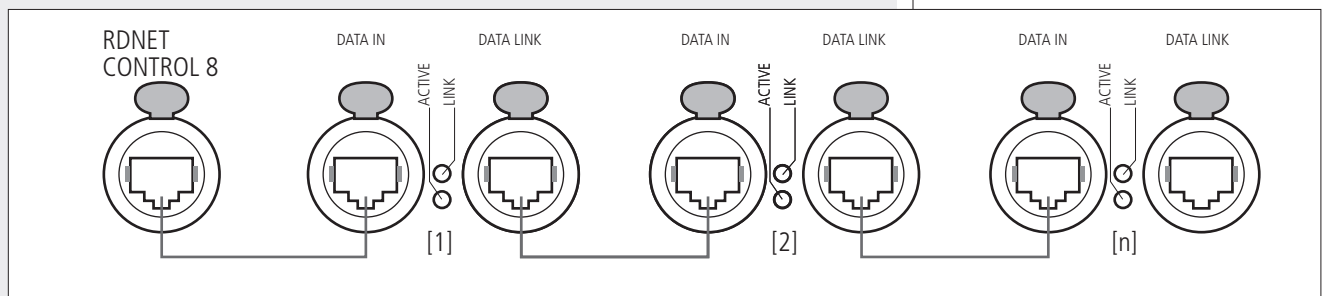
In the example above, [n] is a number from 3 to 32 (max. number of audio devices connected to a subnet).

The total CAT5 cable length of a single subnet cannot exceed 900 metres (c. 2950 feet).

IMPORTANT: internal jumper settings shall be carried out only by either RCF or an authorised service centre.

As default, the **RL1** logic output with relay is assigned to the device fault indication. Set the **JMP1** internal jumper (on the digital board) to the position **1-2**, in order to enable the user software control.
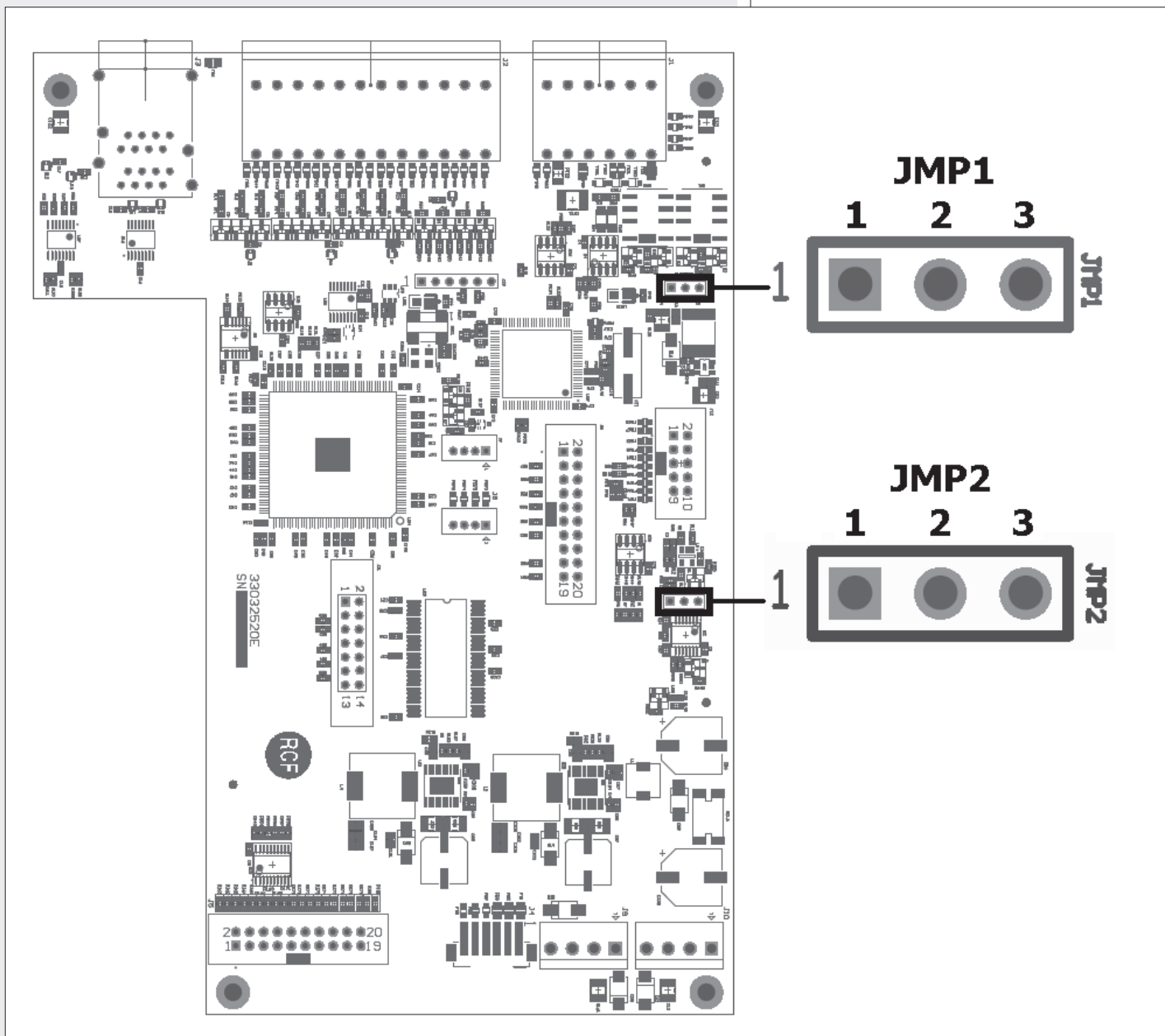
JUMPER	POSITION	FUNCTION
JMP1	1-2	The **RL1** logic output is software controlled.
	2-3	The **RL1** logic output is assigned to the device fault indication (factory default).
JMP2	1-2	A device fault mutes all audio outputs (factory default).
	2-3	A device fault does NOT mute all audio outputs.

SPECIFICATIONS

AUDIO PERFORMANCE DATA

A/D – D/A converters:	24 bits
Sampling rate:	48 kHz
Max. audio input level:	+24 dBu (balanced)
Max. MIC input gain:	+50 dB
Max. audio output level:	+24 dBu
Output impedance:	50 Ω
Frequency response:	20 Hz ÷ 20 kHz (±0.5 dB)
Dynamic range:	> 114 dB
Distortion (THD+N, 1kHz@ +22 dBu):	< 0.01% (analogue input to analogue output)

ELECTRICAL DATA

Power:	100 ÷ 240 V ac (50 / 60 Hz)
'Phantom' power:	48 V

MECHANICAL DATA

Dimensions (l, h, p):	482 mm, 44 mm, 320 mm (1 unit - 19″ rack) (18.98″, 1.73″, 12.60″)
Net weight:	3.475 kg (7.66 lbs)

GENERAL NOTES

The RS-485 protocol for third-party software control is text-based (ASCII string). Commands are sent as simple character strings with terms separated by spaces and completed with a carriage return character <CR> (ASCII code decimal 13 or hex 0x0D).

The general command form is:

```
<COMMAND> <PARAMETER> … <PARAMETER><CR>
```

A single white space character must be included between a command name and each parameter. However, it is allowed to add an arbitrary number of extra white spaces in-between each field if desired (to get a better readability).
Furthermore, both command and string parameters parsing is case-insensitive.

When the device accepts a given command, it will respond with a string as the following:

```
ACK <COMMAND> <PARAMETER> … <PARAMETER> <DATA><CR>
```

On the contrary, when a command is not accepted or if its execution fails for any reason (e.g. the specified controller type does not exist or a given parameter value is out of range), the device will respond with a string as the following:

```
NACK <COMMAND> <PARAMETER> … <PARAMETER><CR>
```

In general, reply strings will be always formed by an acknowledge/not-acknowledge string followed by an echo of the original command (comprising command parameters too) plus an ending (optional) data field. This last field used to return some information to the caller (e.g. the mute state of a channel) will commonly appear in reaction to "get" commands while will not in reaction to "set" commands.

RS 485 CONTROL PORT SETUP

RS-485 control is generally made by using 3rd-party controllers such as certain AMX or CRESTRON models or any RS-485 equipped remote controller device that can be adapted to this protocol.
For experimental purposes, a terminal emulation software such as PUTTY or TERATERM can be run on a PC instead.

In any case, set up the remote controller for:

Baud	38400
Data Bits	8
Stop Bits	1
Parity	None

THIRD-PARTY SOFTWARE PROTOCOL COMMANDS

GET_MODEL
Returns the model name and firmware version of the connected device.
Command syntax is the following:

GET_MODEL<CR>

If command execution succeeds, the device will respond to the above command as follows:

ACK GET_MODEL <MODEL NAME><CR>

with **<MODEL NAME>** field containing a string description of the attached device.

GET_IO_SIZE
Returns the total number of audio input / output channels available.
Command syntax is the following:

GET_IO_SIZE<CR>

If command execution succeeds, the device will respond to the above command as follows:

ACK GET_IO_SIZE <INPUT SIZE> <OUTPUT SIZE><CR>

<INPUT SIZE> is the number of available audio input channels and **<OUTPUT SIZE>** is the available audio output channels.

GET_LABEL
Retrieves the label of either an audio input / output channel or a preset or a mute / control group or even a GPI logic input / GPO logic output.
Generic command syntax is the following:

GET_LABEL <CONTROLLER TYPE> <PARAMETER><CR>

<CONTROLLER TYPE> can be one of the following values:

M_IN	to retrieve the label of a master input channel
S_IN	to retrieve the label of a slave input channel
M_OUT	to retrieve the label of a master output channel
S_OUT	to retrieve the label of a slave output channel
PRST	to retrieve the label of a preset
M_MGRP	to retrieve the label of a master device mute group
S_MGRP	to retrieve the label of a slave device mute group
M_CGRP	to retrieve the label of a master device control group
S_CGRP	to retrieve the label of a slave device control group
GPIN	to retrieve the label of a master device logic input (GPI)
GPOUT	to retrieve the label of a master device logic output (GPO)

<PARAMETER> is the channel number for all "controller types" except for **PRST** and must be in the range: 1 to 8 for master/slave audio inputs, 1 to 6 for master/slave audio output, 1 to 16 for master/slave mute and control groups, 1 to 10 for logic inputs (GPI) and 1 to 6 for logic outputs (GPO).

For **PRST** controller type, **<PARAMETER>** is instead the preset number and must be in the range 1 to 16.

If command execution succeeds, the device will respond to the above command as follows:

ACK GET_LABEL <CONTROLLER TYPE> <PARAMETER> <LABEL><CR>
---

with **<LABEL>** field containing the requested label string (max. 20 characters long).

EXAMPLE 1

GET_LABEL M_MGRP 15<CR>

Returns mute group 15 of master device label string.

The device will respond to the above command with a string similar to the following:

ACK GET_LABEL M_MGRP 15 FOO BAR<CR>

where **FOO BAR** is the current label of 15th master mute group.

EXAMPLE 2

GET_LABEL PRST 3<CR>

Returns third preset label string.

EXAMPLE 3

GET_LABEL S_IN 6<CR>

Returns slave device input channel 6 label string.

GET_PRESET
Returns active preset ID.
Command syntax is the following:

GET_PRESET<CR>

If command execution succeeds, the device will respond to the above command as follows:

ACK GET_PRESET <PRESET NUMBER><CR>
---

with **<PRESET NUMBER>** field equal to active preset ID in the range 1 to 16.

SET_PRESET
Recalls and activates the preset corresponding to given ID number.
Command syntax is the following:

SET_PRESET <PARAMETER><CR>
---

in which **<PARAMETER>** is the ID number of the preset to load and must be in the range 1 to 16.

If command execution succeeds, the device will respond to the above command as follows:

ACK SET_PRESET <PARAMETER><CR>

GET_MUTE

Retrieves the current mute state of either an input / output channel or a mute group or an audio matrix node.

Generic command syntax is the following:

```
GET_MUTE <CONTROLLER TYPE> <PARAMETER 1> <PARAMETER 2><CR>
```

<CONTROLLER TYPE> can be one of the following values:

M_IN	to retrieve the mute state of a master device input channel
S_IN	to retrieve the mute state of a slave device input channel
M_OUT	to retrieve the mute state of a master device output channel
S_OUT	to retrieve the mute state of a slave device output channel
M_MGRP	to retrieve the state of a master device mute group
S_MGRP	to retrieve the state of a slave device mute group
M_MTX	to retrieve the mute state of a master device audio matrix node
S_MTX	to retrieve the mute state of a slave device audio matrix node

<PARAMETER 1> is the channel number for all controllers types except for **M_MTX** and **S_MTX** and must be in the range: 1 to 16 for master/slave audio inputs, 1 to 6 for master/slave audio outputs, 1 to 16 for master/slave mute groups.

Input channels 1 to 8 belong to the master unit, input channels 9 to 16 belong to the slave unit.

For **M_MTX** and **S_MTX** controller types, **<PARAMETER 1>** is the matrix input channel and must be in the range 1 to16 (for both master/slave audio matrices), while **<PARAMETER 2>** is the matrix output channel and must be in the range 1 to 6 (for both master/slave audio matrices).

If command execution succeeds, the device will respond to the above command as follows:

```
ACK GET_MUTE <CONTROLLER TYPE> <PARAMETER 1> <PARAMETER 2>
<MUTE STATE><CR>
```

with **<MUTE STATE>** field equal to 0 if the addressed channel, mute group or node is in OFF state, otherwise equal to 1 if in ON state.

EXAMPLE 1:   **GET_MUTE M_MGRP 3<CR>**

   Returns "0" if master device mute group 3 is in OFF state (unmuted), otherwise returns "1" if it is in ON state (muted).

   The device will respond to the above command with a string similar to the following:

   ACK GET_MUTE M_MGRP 3 1<CR>

EXAMPLE 2:   **GET_MUTE M_IN 1<CR>**

   Returns "0" if master input 1 on master device is in OFF state (unmuted), otherwise returns "1" if it is in ON state (muted).

EXAMPLE 3:   **GET_MUTE S_MTX 3 6<CR>**

   Returns "1" if master input 3 to slave output 6 matrix node is in OFF state (disabled), otherwise returns "0" if it is in ON state (enabled).

SET_MUTE
Updates the current mute state of either an input / output channel or a mute group or an audio matrix node.
Generic command syntax is the following:

SET_MUTE <CONTROLLER TYPE> <PARAMETER 1> <PARAMETER 2>
<PARAMETER 3><CR>

<CONTROLLER TYPE> can be one of the following values:

M_IN	to change the mute state of a master device input channel
S_IN	to change the mute state of a slave device input channel
M_OUT	to change the mute state of a master device output channel
S_OUT	to change the mute state of a slave device output channel
ALL_OUT	to change the mute state of both master and slave devices output channels altogether
M_CGRP	to change the state of a master device mute group
S_CGRP	to change the state of a slave device mute group
M_MTX	to change the mute state of a master device audio matrix node
S_MTX	to change the mute state of a slave device audio matrix node

<PARAMETER 1> is the channel number for all controllers types except for **M_MTX**, **S_MTX** and **ALL_OUT**. It must be in the range: 1 to 16 for master/slave audio inputs, 1 to 6 for master/slave audio outputs, 1 to 16 for master/slave mute groups.

Input channels 1 to 8 belong to the master unit, input channels 9 to 16 belong to the slave unit.

<PARAMETER 2> is the target mute state of the addressed input channel, output channel or mute group for all controllers types except for **M_MTX**, **S_MTX** and **ALL_OUT**. It must be in the range 0 to 255. A "0" value will be interpreted as "turn mute state to OFF", while any value in the range 1 to 255 will be interpreted as "turn mute state to ON".

For **M_MTX** and **S_MTX** controller types, **<PARAMETER 1>** is the matrix input channel (in the range 1 to 16), **<PARAMETER 2>** is the matrix output channel (in the range 1 to 6) and **<PARAMETER 3>** is the target state (0 for disabled, 1:255 for enabled) of the addressed matrix node controller.

For **ALL_OUT** controller type, **<PARAMETER 1>** is the only needed parameter and represents the target state (0 for disabled, 1:255 for enabled) of both master/slave audio output channels.

If command execution succeeds, the device will respond to the above command as follows:

ACK SET_MUTE <CONTROLLER TYPE> <PARAMETER 1> <PARAMETER 2>
<PARAMETER 3><CR>

EXAMPLE 1:   **SET_MUTE M_IN 5 1<CR>**
  Mutes master input 5 on the master unit.

  The device will respond to the above command with the following string:
  ACK SET_MUTE M_IN 5 1<CR>

EXAMPLE 2:   **SET_MUTE S_IN 15 27<CR>**
  Mutes slave input 7 on the slave unit.

EXAMPLE 3:   **SET_MUTE M_MTX 5 5 0<CR>**
  Activates master input 5 to master output 5 matrix node.

EXAMPLE 4:   **SET_MUTE ALL_OUT 0<CR>**
  Unmutes all outputs of both master and slave units.

GET_LEVEL

Retrieves current level of either an input / output channel or a control group or an audio matrix node fader.

Generic command syntax is the following:

```
GET_LEVEL <CONTROLLER TYPE> <PARAMETER 1> <PARAMETER 2><CR>
```

<CONTROLLER TYPE> can be one of the following values:

M_IN	to retrieve a master device input fader level
S_IN	to retrieve a slave device input fader level
M_OUT	to retrieve a master device output fader level
S_OUT	to retrieve a slave device output fader level
M_CGRP	to retrieve a master device control group fader level
S_CGRP	to retrieve a slave device control group fader level
M_MTX	to retrieve a master device audio matrix node level
S_MTX	to retrieve a slave device audio matrix node level

<PARAMETER 1> is the channel number for all controllers types except for **M_MTX** and **S_MTX**.

It must be in the range: 1 to16 for master/slave input fader levels, 1 to 6 for master/slave output fader levels, 1 to 16 for master/slave control group fader levels.

Input channels 1 to 8 belong to the master unit, input channels 9 to 16 belong to the slave unit.

For **M_MTX** and **S_MTX** controller types, **<PARAMETER 1>** is the matrix input channel and must be in the range 1 to 16 (for both master/slave audio matrices), while **<PARAMETER 2>** is the matrix output channel and must be in the range 1 to 6 (for both master/slave audio matrices).

If command execution succeeds, the device will respond to the above command as follows:

```
ACK GET_LEVEL <CONTROLLER TYPE> <PARAMETER 1> <PARAMETER 2>
<LEVEL><CR>
```

with **<LEVEL>** field indicating the current level in the range 0 to 255.

Some faders have a range of −90 dB to +10 dB, while the **GET_LEVEL** command return value has a range of 0 to 255.

Consequently, the following formula can be used to convert from returned value to dB:

$$dB\ Level = -90 + 100 \cdot \frac{Returned\ Value}{255}$$

Notice that if a fader has a different range than −90 dB to +10 dB, the formula that can be used to convert from returned value to dB will be the following:

$$dB\ Level = Min\ Level + (Max\ Level - Min\ Level) \cdot \frac{Returned\ Value}{255}$$

where *Min Level* and *Max Level* are the fader lower and upper bounds in dB, respectively.

EXAMPLE 1: **GET_LEVEL M_IN 3<CR>**
Returns master input 3 fader level as set on the master unit.

The device will respond to the above command with a string similar to the following:
ACK GET_LEVEL M_IN 3 239<CR>

EXAMPLE 2: **GET_LEVEL M_IN 9<CR>**
Returns slave input 1 fader level as set on the master unit.

EXAMPLE 3: **GET_LEVEL S_MTX 3 6<CR>**
Returns master input 3 to slave output 6 matrix node level.

SET_LEVEL
Updates current level of either an input / output channel or a control group or an audio matrix node fader.
Generic command syntax is the following:

> **SET_LEVEL <CONTROLLER TYPE> <PARAMETER 1> <PARAMETER 2>**
> **<PARAMETER 3><CR>**

<CONTROLLER TYPE> can be one of the following values:

M_IN	to update a master device input fader level
S_IN	to update a slave device input fader level
M_OUT	to update a master device output fader level
S_OUT	to update a slave device output fader level
M_CGRP	to update a master device control group fader level
S_CGRP	to update a slave device control group fader level
M_MTX	to update a master device audio matrix node level
S_MTX	to update a slave device audio matrix node level

<PARAMETER 1> is the channel number for all controllers types except for **M_MTX** and **S_MTX**.
It must be in the range: 1 to16 for master/slave input fader levels, 1 to 6 for master/slave output fader levels, 1 to 16 for master/slave control group fader levels.
Input channels 1 to 8 belong to the master unit, input channels 9 to 16 belong to the slave unit.

<PARAMETER 2> is the target level of the addressed fader controller for all controller types except for **M_MTX** and **S_MTX**. It must be in range 0 to 255.
While most faders have a range of –90 dB to +10 dB, the following formula can be used to calculate the **<PARAMETER 2>** value corresponding to a given target fader level in dB:

$$Parameter2\ Value = 255 \cdot \frac{(dB\ Level + 90)}{100}$$

Notice that if a fader has a different range than –90 dB to +10 dB, the formula that can be used to convert from target fader level in dB to the corresponding **<PARAMETER 2>** value will be the following:

$$Parameter2\ Value = 255 \cdot \frac{dB\ Level - Min\ Level}{Max\ Level - Min\ Level}$$

where *Min Level* and *Max Level* are the fader lower and upper bounds in dB, respectively.

For **M_MTX** and **S_MTX** controller types, **<PARAMETER 1>** is the matrix input channel number in the range 1 to16, **<PARAMETER 2>** is the matrix output channel number in the range 1 to 6, **<PARAMETER 3>** is the target level of the addressed matrix node controller and must be in range 0 to 255. As for other controller types, proper values of **<PARAMETER 3>** can be computed by applying the same formulas reported above (used to calculate second parameters of other level controllers types).

If command execution succeeds, the device will respond to the above command as follows:

ACK SET_LEVEL <CONTROLLER TYPE> <PARAMETER 1> <PARAMETER 2> <PARAMETER 3><CR>

EXAMPLE 1: **SET_LEVEL M_IN 2 229<CR>**
Sets master input 2 fader level to 0 dB on the master unit.

The device will respond to the above command with the following string:

ACK SET_LEVEL M_IN 2 229<CR>

EXAMPLE 2: **SET_LEVEL S_IN 10 245<CR>**
Sets slave input 2 fader level to +6 dB on the slave unit.

EXAMPLE 3: **SET_LEVEL M_MTX 2 4 199<CR>**
Sets master input 2 to master output 4 matrix node level to –12 dB.

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