Audio Vertriebs
Entwicklungsgesellschaft
mbH
STUTTGART

Andante 8

Digital Mixer
with 8 mic/line inputs and 4 outputs

User Guide

Table of contents

Introduction

Thank you for choosing the 16-channel digital mixer “Andante 8” from AVE.
No matter what your requirements are, due to its advanced technical design, it ensures trouble-free use with sound exposure in a variety of facilities such as churches, courtrooms, government offices, conference rooms, schools, universities, etc.

The following information relates to Andante 8. It is also applicable for Andante 8 because only the number of inputs and outputs are different.

The Andante 8 is equipped with its own software. This makes the programming via laptop or PC easily and conveniently.

Safety instructions

Located inside the unit are hazardous voltages. Do not remove the cover. Internal modifications or service work should only be conducted by qualified service personnel.

The Andante 8 is supplied with a proper power supply cable. At one end of this cable is a three-pronged AC power connector (IEC plug) and at the other end of a CE-standard-compliant Schuko-plug for connection to a 230 V / 50 Hz AC voltage source. Please ensure that this power supply cable is not damaged. Do not use defective or damaged power cables!

Scope of delivery

Please check immediately upon receipt the package integrity, the contents for completeness and proper delivery of the unit.

The delivery scope of the unit includes:

- The “Andante 8”
- USB - cable
- Power supply cable

Please keep the operating instructions in a safe place and make them available to qualified personnel for making necessary changes to the device.

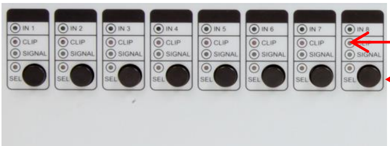
This **user guide** is divided into **3 sections**:

Section 1

The **first section** explains how the **user** can easily and comfortably change the overall volume and retrieve the presets (up to 20) directly on the “Andante 16” device without using the laptop or PC.
The volume control in "user mode" is restricted in order to prevent operating errors. If the protection is activated, the volume is adjusted by pressing and turning the v/v button and to select a preset by pressing and turning the p/p button.

Control of the entire volume
By turning the v/v button the total volume will be increased or decreased (master volume).
Select preset
With the p/p button a desired preset is selected.
Input level 1-8
Press the select (Sel) button and turn v/v key, to increase or to decrease the volume.
Output level 1-4
Press the select (Sel) button and turn v/v key, to increase or to decrease the volume.
Press the v/v button to return to the main menu.

8 inputs
Each input has the same functions



LED Display:
Input, Clip, Signal, Select
Select button
LED display Select

Matrix

Matrix Select buttons for assigning inputs to the outputs

4 outputs
Each input has the same functions

LED Display:
Output, Clip, Signal, Select
Select button

Display
Volume/Value

Power switch
Preset/Parameter USB connection

Power socket RSC 232 Aux Out Outputs 1-4 CD / Aux Input Inputs 1 – 8

Section 2

The **second section** is intended for **qualified personnel**, which adjusts the settings via PC or laptop and the programming of the parameters, the automatic, the matrix and the delays, etc. **The following modifications must to be carried out by qualified personnel.**

Installing the software

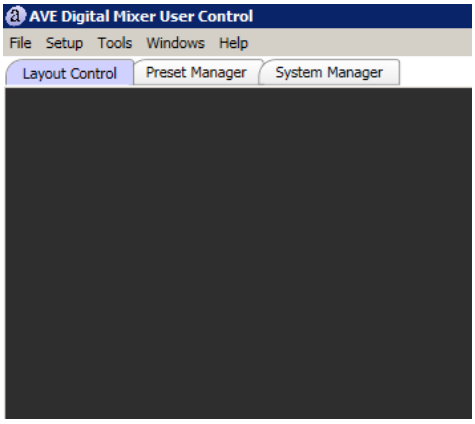
Start the installation program by clicking on the appearing desktop folder and then clicking on **setup.exe.**

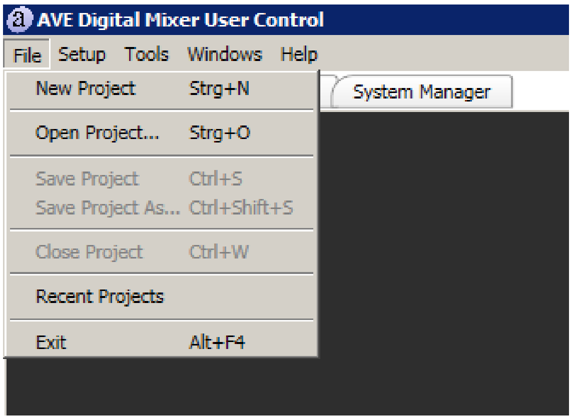
Then follow the installation instructions.
Enter your user name, your organization name and the license number that was included.

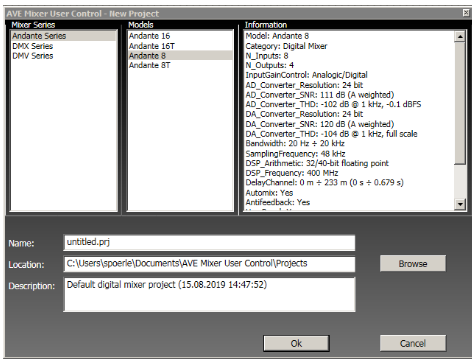
After installation is complete you will have the following shortcut on the desktop

Programming of the Andante 8

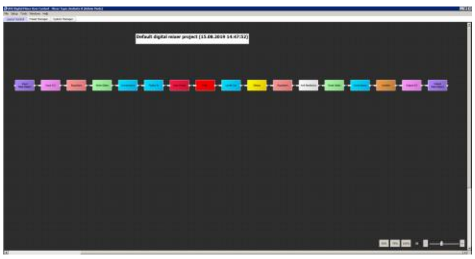
Restart your PC or your laptop, so the program can be executed properly. Use the supplied connector cable between your laptop and Andante 16. Start the program and click with the left mouse bottom on **File** and then on **New Project.**



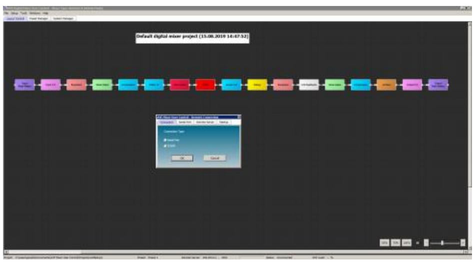




Specify the type of device and confirm by pressing **OK**.



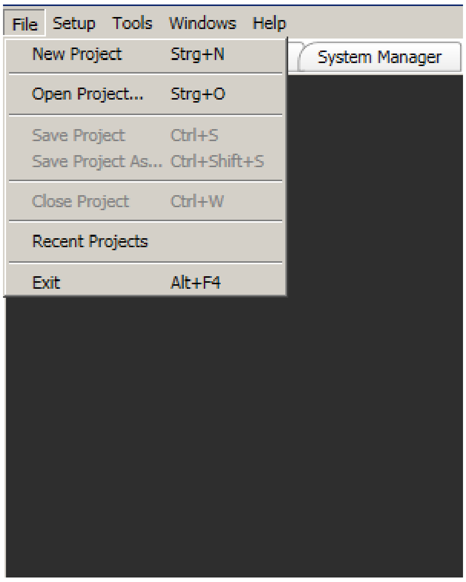
This window shows all the functions of the mixer. With the zoom on the bottom right you can adjust all commands on the monitor



Click with the left mouse button on **Setup** and then on **Remote Connection,** confirm with **OK**. After a short time.the display shows **Remote Connected.**

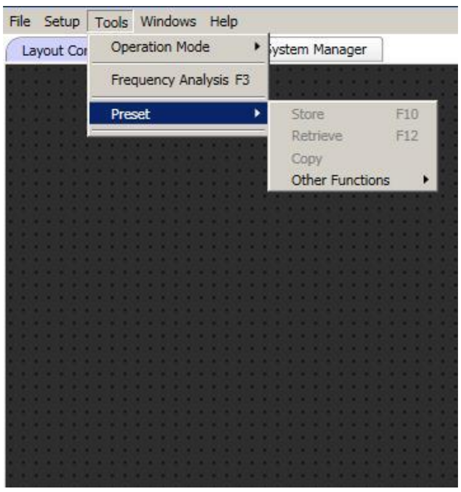
Now you can use all the functions you want to program.

If you use a DHPC router connection make sure that the IP number of the router is correct. For this purpose press the preset bottom twice. Then turn the knob until the IP number appears. Press with the left mouse bottom **Setup** and then **Remote Connection** afterwards **Remote Server.** Enter the IP adress and confirm with OK. On the display appears **Remote Connected**.



When you click **File** in the menu bar**,** a pull-down menu with following selection options is opened:

New Project	A new project is created
Open Project	An existig project is opened
Save Projekt	The project is saved
Save Project As	The project can be saved under a different name
Close Project	The project will be closed
Recent Projects	Saved projects can be considered



When you click **Tools** in the menu bar**,** a pull-down menu with following selection options is opened:

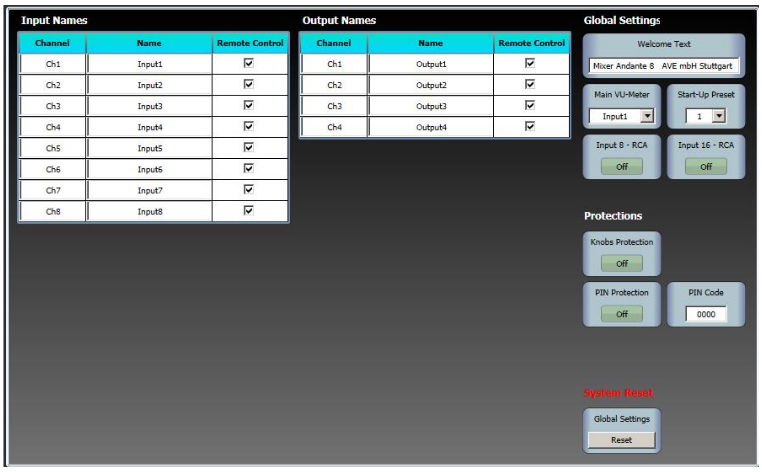
Operation Mode	Choice between Online and Offline
Preset Store	Storing presets
Preset Retrieve	Retrieve presets
Preset Copy	Copying of presets
Other Functions	**Store Without Sending** allows corrections during a live transmission without switching in mute mode.

When you click **Windows** you can select between **Layout control**, **Preset Manager** and **System Manager.**

Der Preset Manager

Before you save a configuration you have to give the preset a name to retrieve it again via Preset Retrieve.

Der System Manager



Start-Up Preset
Choice of the start preset

Main VU-Meter
Choice of the VU meters

Knobs Protection
By pressing and turning the volume control you can adjust the volume on the device.

PIN Protection
Activation of the PIN of the Admin Mode

PIN Code
Enter of an own PIN Code (Admin Mode)

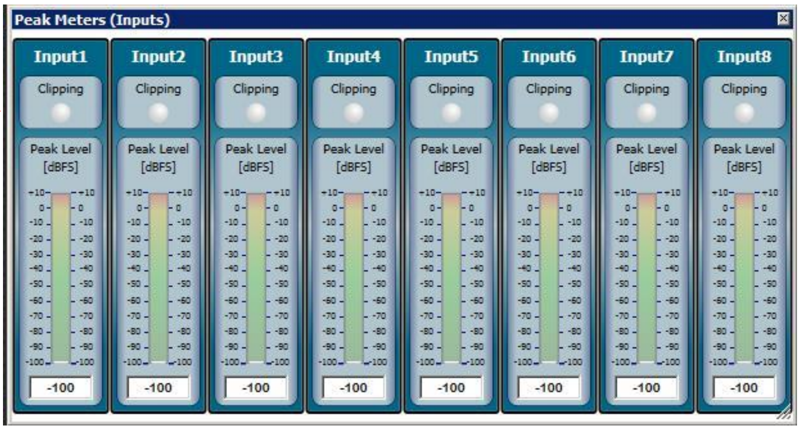
System Reset

Layout Control

Peak meters (Input)

Clipping indicator

Peak level indicator

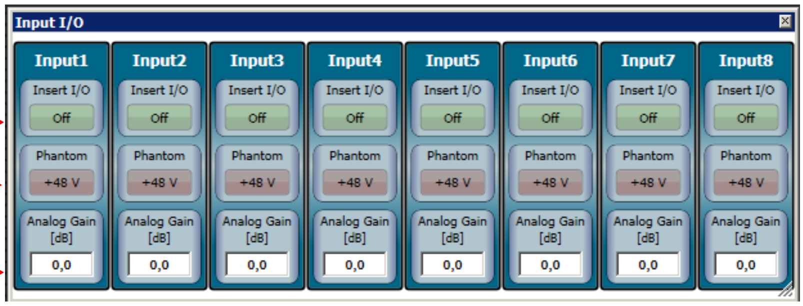


Input On/Out

Input on/out

Phantom power

Analog Gain [dB]

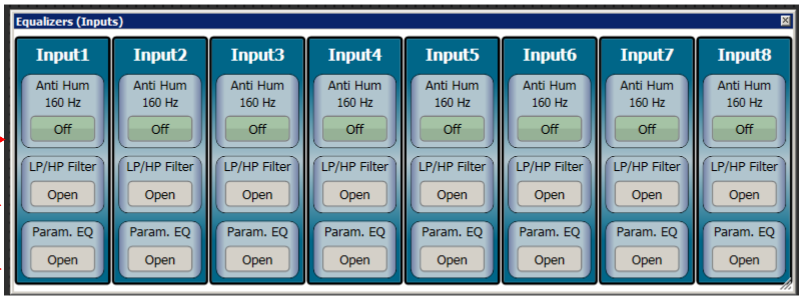


Parametric Equalizer (Input)

Hum filter 160 Hz

Lowpass/highpass filter

8 band parametric equalizer



8 parameters are available for each input and each output. The settings can be copied to each other input and output. The same applies to the highpass/lowpass filters.

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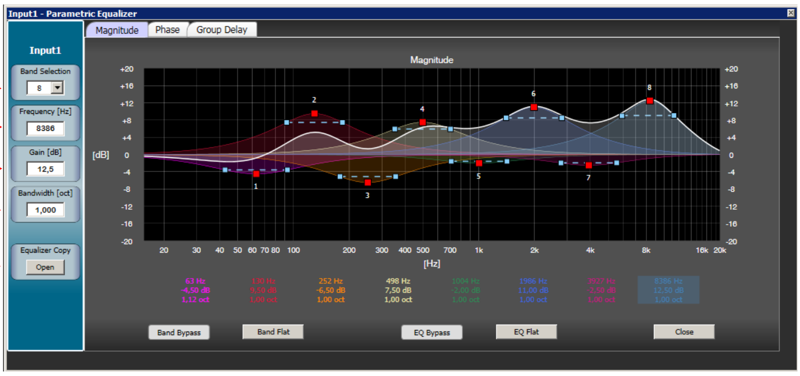
Band Selection

Frequency [Hz]

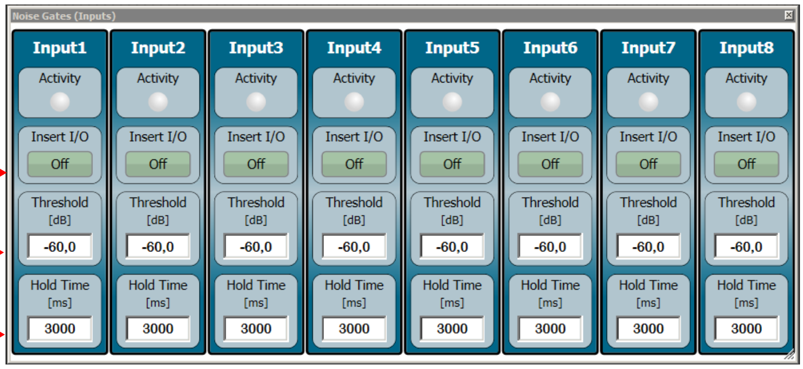
Gain [dB]

Bandwidth [Oct]

Equalizer Copy



Noise Gates (Input)



Noise Gate On/Out

Threshold [dB]

Hold time [ms]

Compressor (Input)

On/Out

Threshold [dB]

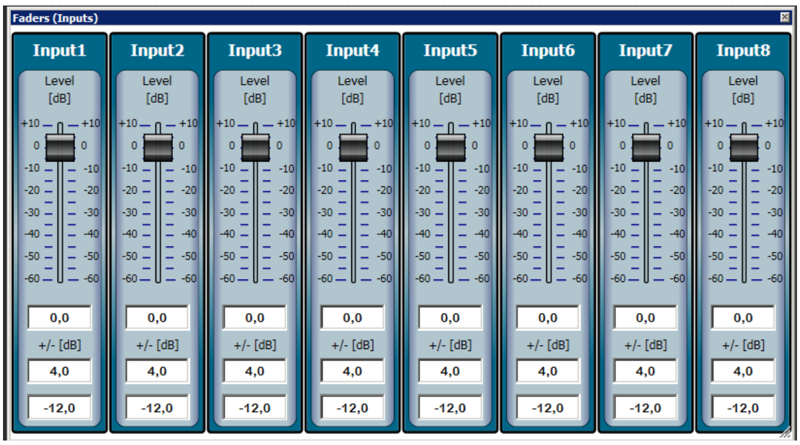
Ratio

Post Gain [dB]

Attack time [ms]

Release time [ms]

Faders (Input)



Level Setting
Input option for programmer
to change the volume at the input

Automixer (Eingang)

Select

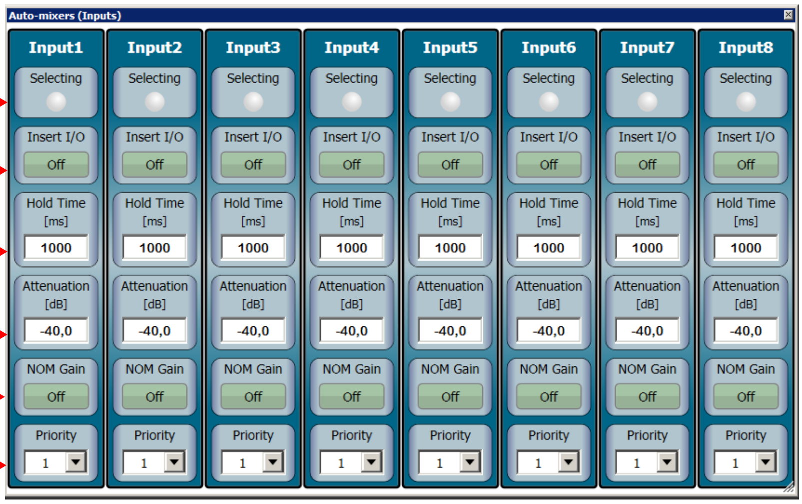
Automixer on/out

Hold time [ms]

Attenuation [dB]

NOM Gain on/out
Reducing the level when microphones are activated simultaneously

Priority



Detailed description of the automixer on page 16

Matrix

Matrix on/out

Level Setting [dB]

After pressing the right mouse bottom a pull-down menu opens. It allows the activation of all matrix nodes or different rows or columns of nodes.
Routing of all inputs and outputs with additional level setting.
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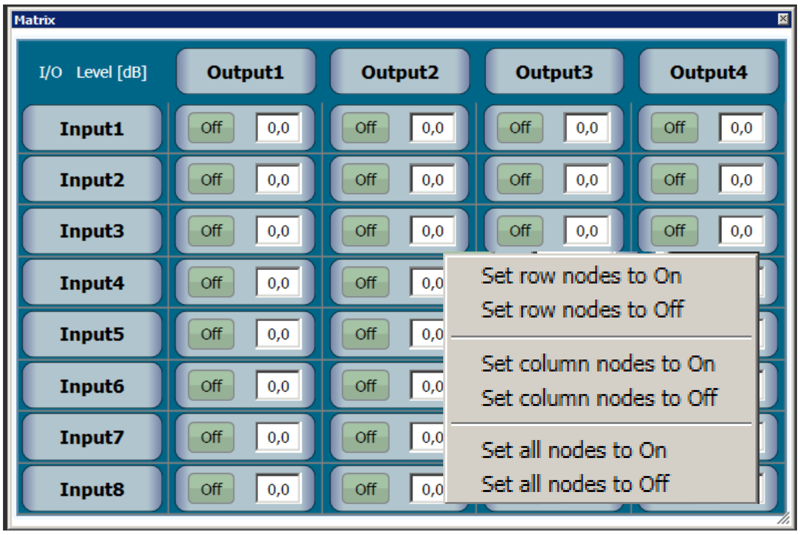
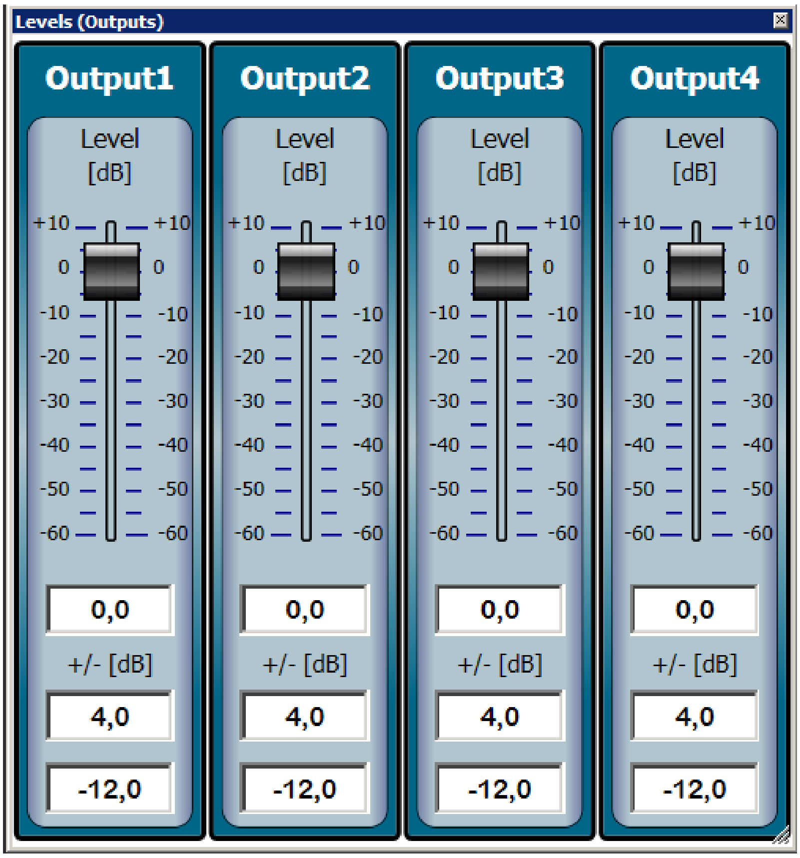


Level (Output)

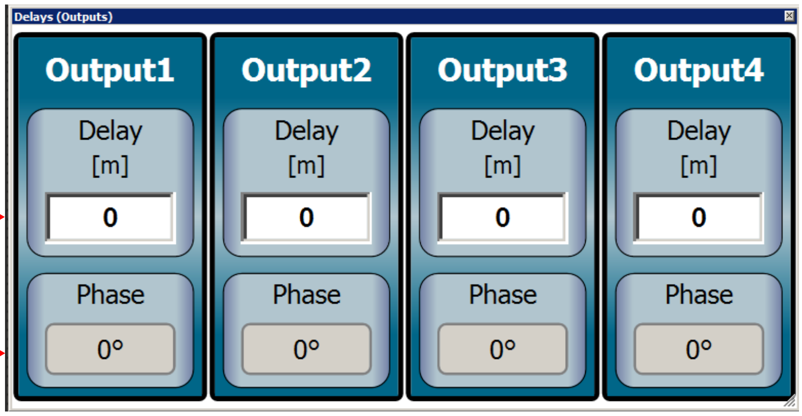
Level setting
Input option for programmer to change the volume at the output.



Delay (Output)

Delay [m]

Phase

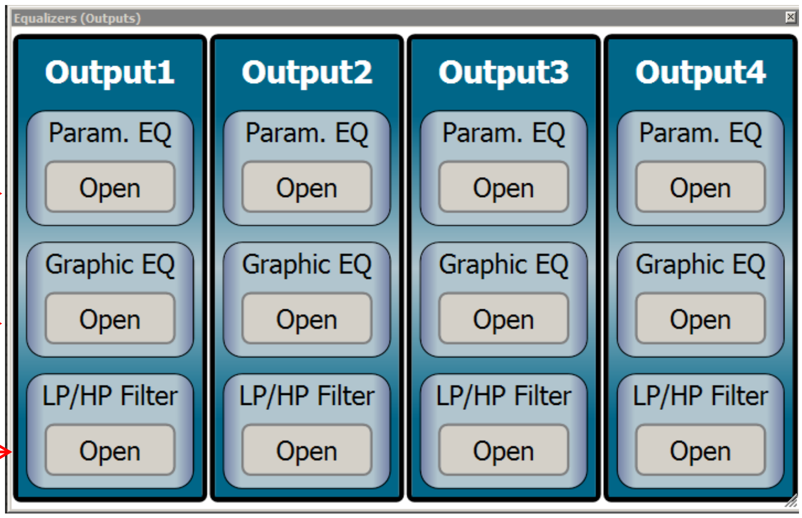


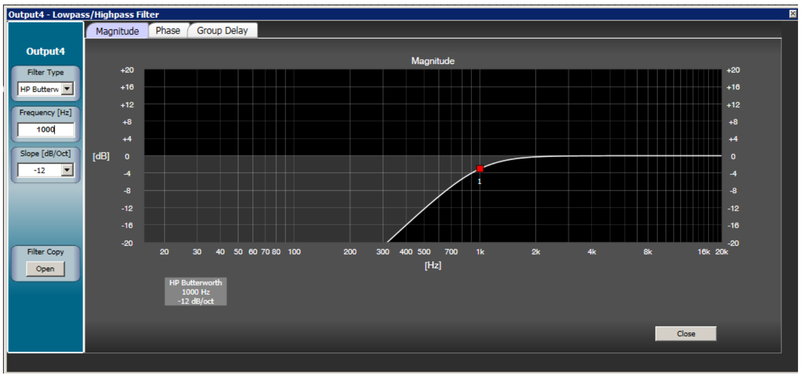
Equalizer (Output)

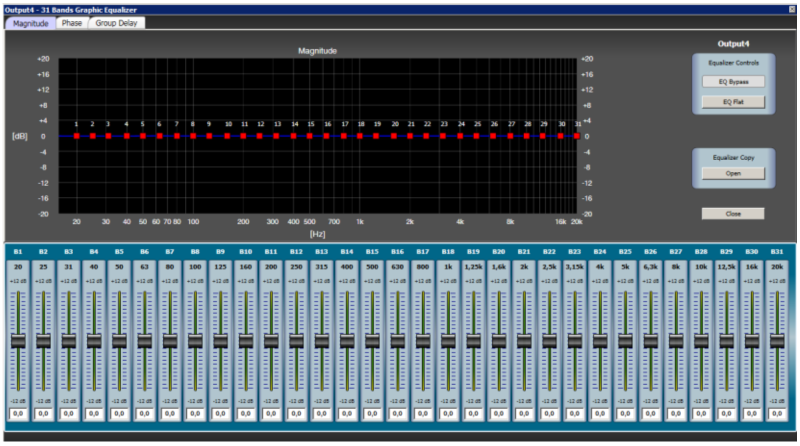
Parametric EQ

GraphicEQ

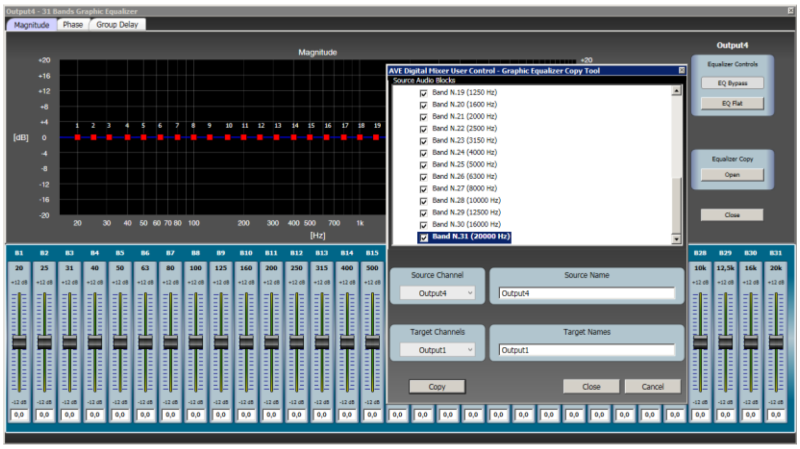
Lowpass-/Highpass filter





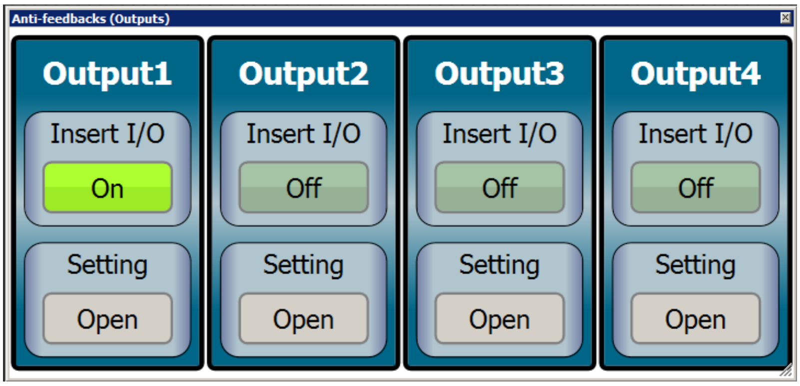


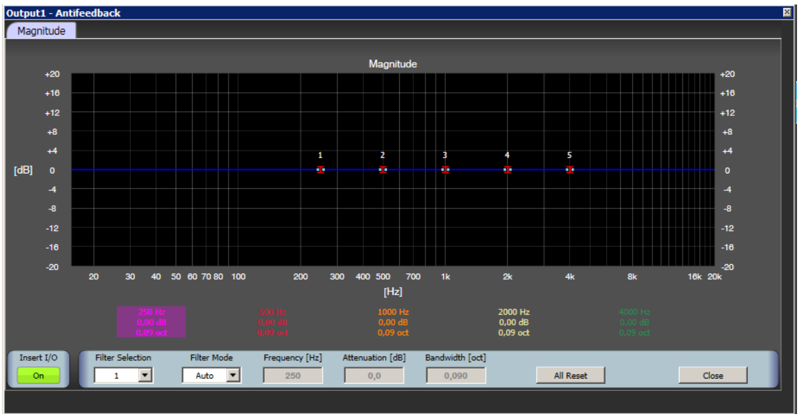
Each output has a 31 band graphic equalizer.
All parameters can be copied to the other EQs or individually adjusted for each output.



Anti-Feedbacks (Outputs)

Detailed description of the antifeedback functions on page 16



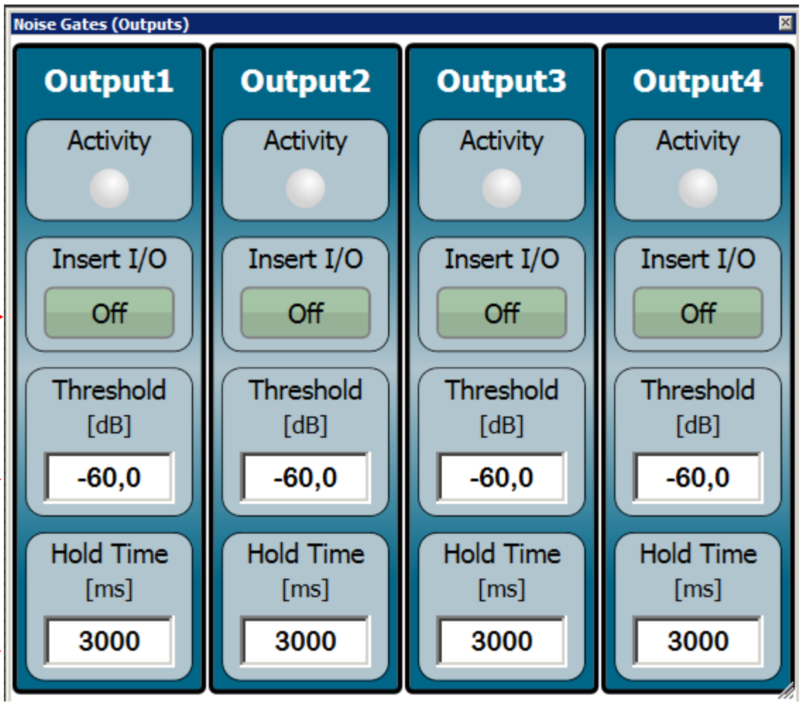


Noise Gates (Output)

On/Off

Threshold [dB]

Hold time [ms]



Compressor (Output)

On/Off

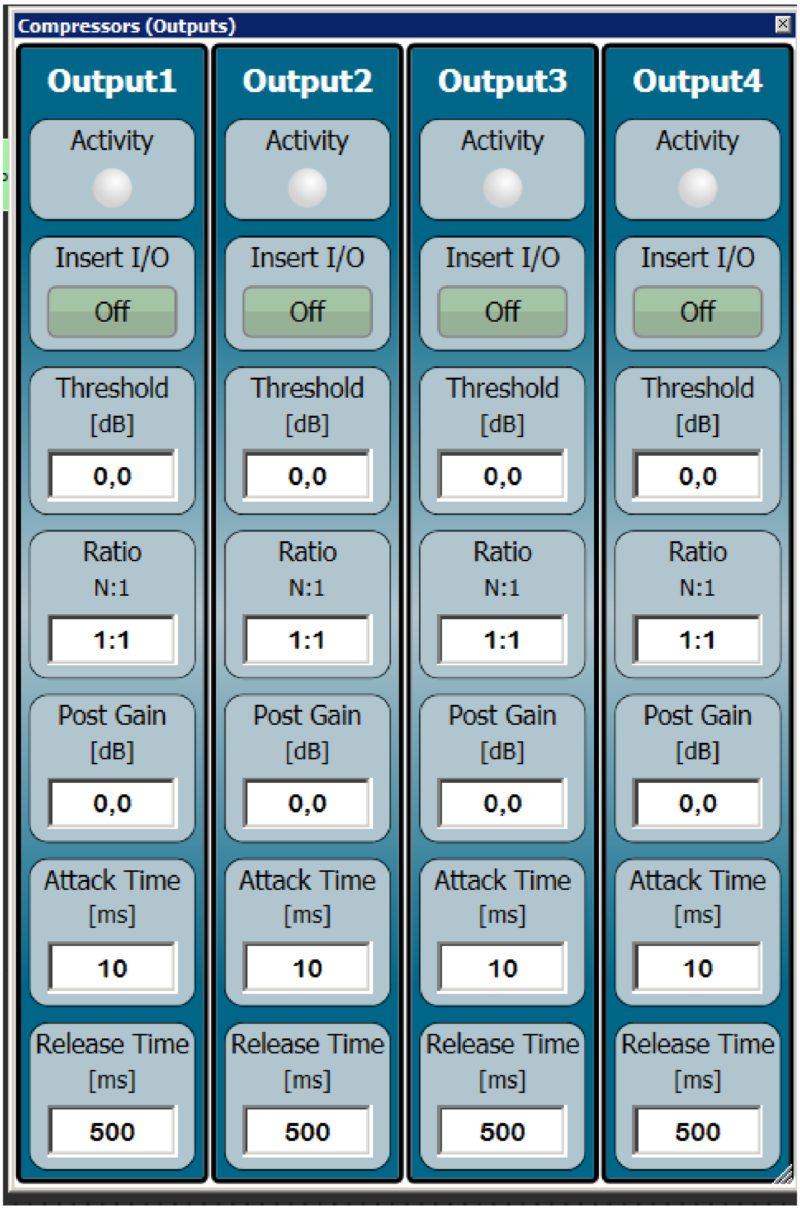
Schwelle [dB]

Ratio

Post Gain [dB]

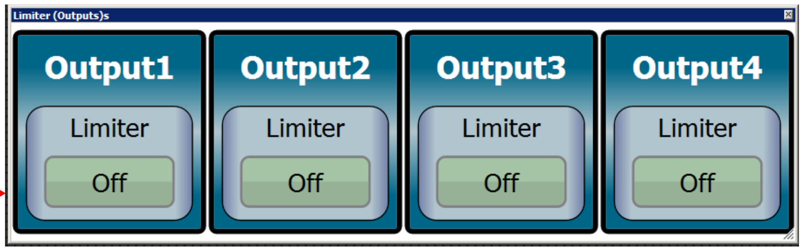
Attack time [ms]

Release time [ms]



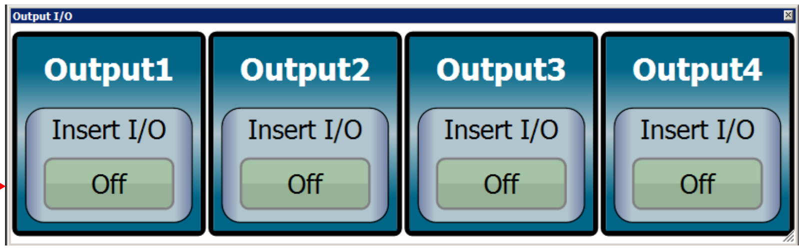
Limiter (Output)

Limiter On/Off



Output On/Off

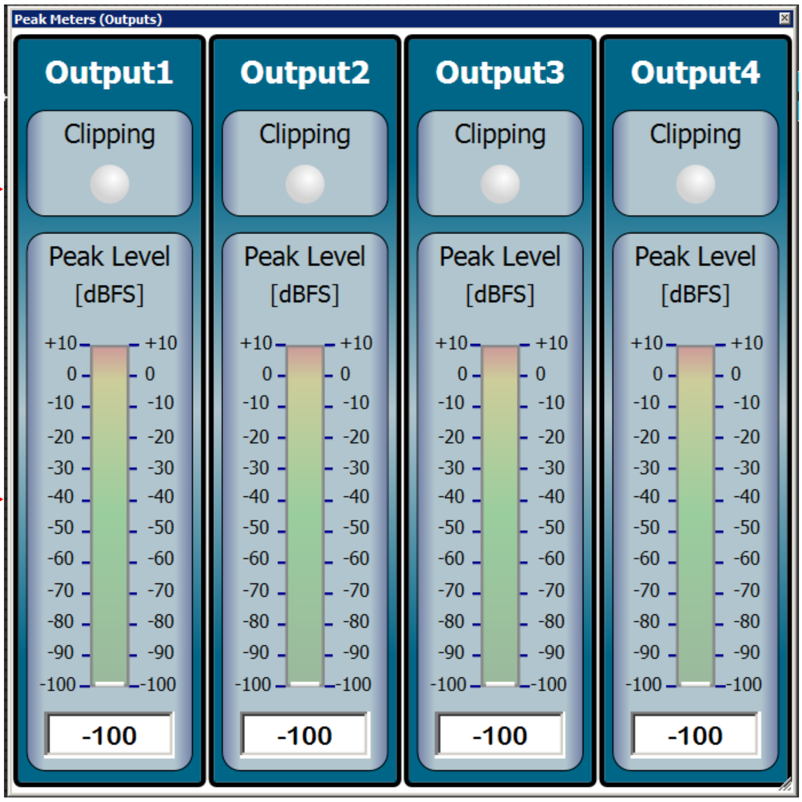
Output On/Off



Peakmeter (Output)

Clipping

Peak Level



Overview adaptive automixer

Automixer processing is useful in situations where multiple microphones are used, such as in churches, board rooms, conference rooms or courtrooms. Automixer solves the typical problems that occur when multiple mics are used.
In fact, the voice signal picked up by multiple microphones suffers from multi-path interference, also called comb filtering. This multi-path interference affects the quality of the sound source by emphasizing sound at some frequencies and canceling the sound at others
Moreover, multiple opened microphones could easily create multiple potential feedback loops and consequently input gain must be limited.
Automixer is a real-time sound algorithm that automatically reduces the level of a microphone's audio signal when it is not being used and maintain a steady limit on the overall signal level of the opened microphones.

Automixer has the following benefits:

- Lowering hiss, rumble, reverberation and other extraneous noise that occur when several microphones operate simultaneously.
- Improving gain before feedback.
- Reducing the comb-filtering effect.

Automixer is ideally suited for applications such as churches, board rooms, conference rooms or courtrooms.

Gating

The automixer unit controls the microphone's state as following:

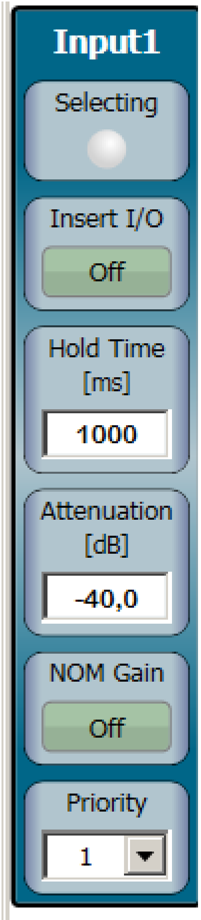
- If the RMS level is above the current adaptive threshold, the microphone's state is open i.e. the block applies no attenuation to the signal.
- If the RMS level is below the current adaptive threshold, the microphone's state is attenuated i.e. the block applies an off attenuation to the signal.

Adaptive Threshold

The internal algorithm dynamically sets the gate threshold level in accordance with the ambient noise. For example, in a noisy room the automixer would increase the threshold level to prevent any of the microphone channels from being triggered on by the noise. On the other hand, in a quiet environment, the threshold level must be lowered.

Last Microphone Hold

Die Funktion „Last Microphone Hold“ lässt das zuletzt aktivierte Mikrofon offen, auch wenn der Schallpegel unter den eingestellten Schwellenwert fällt.



Selecting
This LED is on when the microphone channel is gated on.
For those input channels with automixer mode disabled, this LED is always off.

Insert I/O
Turns on/off channel gating. When disabled, the microphone state for this channel remains bypassed (no automixer mode) at all times and its level does not contribute to adaptive threshold computing or NOM attenuation.

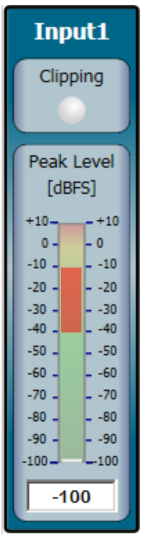
Hold Time
Sets the minimum time an microphone stays open after the user stops talking (i.e. RMS input level drops below the adaptive threshold level).
Use this control to prevent the gate from opening and closing due to momentary pauses in speech.
Typical values are comprised between 1 second and 2 seconds.

Attenuation
The off attenuation parameter allows attenuation of inactive inputs from 0 dB to -60 dB.
When hold time is elapsed, the off attenuation is smoothly inserted.
Practical experience has shown that a 15 dB – 20 dB attenuation is a good compromise.

NOM GAIN
When NOM GAIN is active for a channel, the automixer will reduce the gain for the given channel by 3 dB
for every doubling of open channels (-10log(NOM) rule).
The NOM GAIN, if enabled,
is singularly applied on each channel.

NOM	NOM GAIN
1	0 dB
2	-3 dB
3	-3 dB
4	-6 dB
5	-6 dB
6	-6 dB

Priority
has relevance to assign preference on opening channels.
Compatibly with the actual number of open microphones, the channels with higher priority will be opened before of the lowest ones.
It does not mean strictly that microphones with higher priority can override ones with lower priority.
Generally, each time two or more channels must be opened, priority level is used to give a precedence order.
Priority increases from 1 (lowest) to 5 (highest).
A special case of priority behavior regards the highest priority that serve to disable all other channels.
With PRIORITY equal to 5, use the additional parameter **RESET TIME** to determines how long to wait for channel restoring.
Typical values of RESET TIME are between 1 second and 5 seconds.



Application example:
The highest priority reveals useful inside churches during organ playing, where a separate microphone with PRIORITY equal to 5, placed close to the organ pipes, gates off all microphones in presbyter.

NOTE:
When you set up your mixing system, make sure you get the right gain on the analog input gain control. Automixer works better when the gain has been properly set to give the algorithm adequate signal.

For automixer proper working, best microphone level range is between -40 dB and -10 dB on Peak Level Meter.

Overview Antifeedback

Audio feedback (also known as acoustic feedback, simply as feedback, or the Larsen effect) is a special kind of positive feedback which occurs when a sound loop exists between an audio input (for example, a microphone or guitar pickup) and an audio output (for example, a loudspeaker).

When acoustical feedback occurs in a sound system, it is because the gain of the system is too high. Feedback will occur at specific frequencies before others; these are the frequencies with the most gain. If the gain at only these specific frequencies is reduced, then the system can operate with more overall gain before it feeds back, without a perceptible difference in tonal quality.

The antifeedback algorithm inside the mixer unit discriminates accurately between feedback and non-feedback sounds (speech and music). When this algorithm detects feedback, it inserts a -3 dB, 1/11 octave (Q = 16) notch filter into the audio path to reduce the gain at the frequency which is feeding back. If the feedback does not stop, the filter depth is increased in -0.5 dB steps (up to -20 dB) until the feedback stops.

The antifeedback algorithm could assign up to 5 notch filters for each one of first four output channel.

The antifeedback filtering chain is composed by two typologies of notch filters: **AUTO** and **MANUAL.**

AUTO Filters

AUTO filters are managed automatically by the identification feedback algorithm, which inserts them at detected feedback frequencies. AUTO filters have fixed bandwidth equal to 1/11 octave, narrow enough to remove feedback effectively and preserve audio quality.

In case of all AUTO notch filters are set and a new feedback frequency is detected, the antifeedback will remove the oldest AUTO filter to reassign it at the new critical frequency.

AUTO filters can be locked switching their filter mode from AUTO to MANUAL. In such case, their setup is saved with preset storing operation.

AUTO filters would be appropriate in a system that has several non-stationary wired/wireless microphones.

MANUAL Filters

MANUAL filters are settable by user, typically at the end of “Ring-Out” setup procedure (see below), and their configuration is saved with preset storing operation.

Notch attenuation is choosable in the range from 0 dB to -20 dB while notch bandwidth is selectable between 6,67 and 0,014 octaves (Q = 0,1 and Q = 100 respectively).

MANUAL filters would be suitable in a sound system which has fixed microphone and loudspeaker locations. In this type of system, the most feedback frequencies are defined by the room size and the microphone and loudspeaker placement, and will not change appreciably.

Setup Procedures for Feedback Control

There are two basic ways in which to setup the antifeedback: the “Ring Out” and the "Be Alert" procedures.

Ring Out” Procedure

The “Ring Out” method is a precautionary measure in which the preset volume is raised beyond the normal setting to deliberately make the system feed back. The antifeedback will then set its notch filters. Then, the preset volume is reduced slightly to restore sound system stability.

This setup procedure is commonly used for sound systems which are operated near the feedback point where a perceptive annoying “metallic” sound is noticed as well as an altered tone quality.

Follow these steps:

1. Place speakers and microphones in performance positions.
2. Adjust the analog gain for each microphone channel. Try to set each microphone as close as possible to the desired performance level while remaining under the point of feedback. If feedback occurs, slightly reduce the analog gain or the channel fader until the feedback disappears.
3. Adjust for the desired tonal qualities using the parametric or graphical equalizer panels (eventually after tonal compensations correct the Fader Level or Output Level channel).
4. Reduce the mixer's preset volume (about -20 dB) respect the level at which the system will be operated and mute all other mixer channels which you do not intend to ring out.
5. Active antifeedback pushing INSERT I/O button (green light) and leave all filters in AUTO mode.
6. Slowly raise the mixer's preset volume until the antifeedback eliminates the first few feedback tones (all AUTO filters are engaged in magnitude panel). You should be a few dB above the nominal preset volume.
7. Restore preset volume to the level at which the system will be operated.
8. Finally, for sound system with:
   - stationary microphones: set each notch filter to MANUAL mode
   - wireless microphones: leave each notch filter in AUTO mode.
   - hybrid microphones or to improve protection against unexpected feedbacks: arrange the first three or four

Be Alert" Procedure

For the "Be Alert" method, the antifeedback is simply activated in the sound system and all filters are left in AUTO mode. If the feedback occurs, the antifeedback is ready to delete it.
This setup procedure is used for those systems, which already have sufficient gain-before-feedback margin but need protection from sudden feedbacks, due mainly to non-stationary microphones or erroneous preset volume adjusting by user during performance.

Follow these steps:

1. Place speakers and microphones in performance positions.

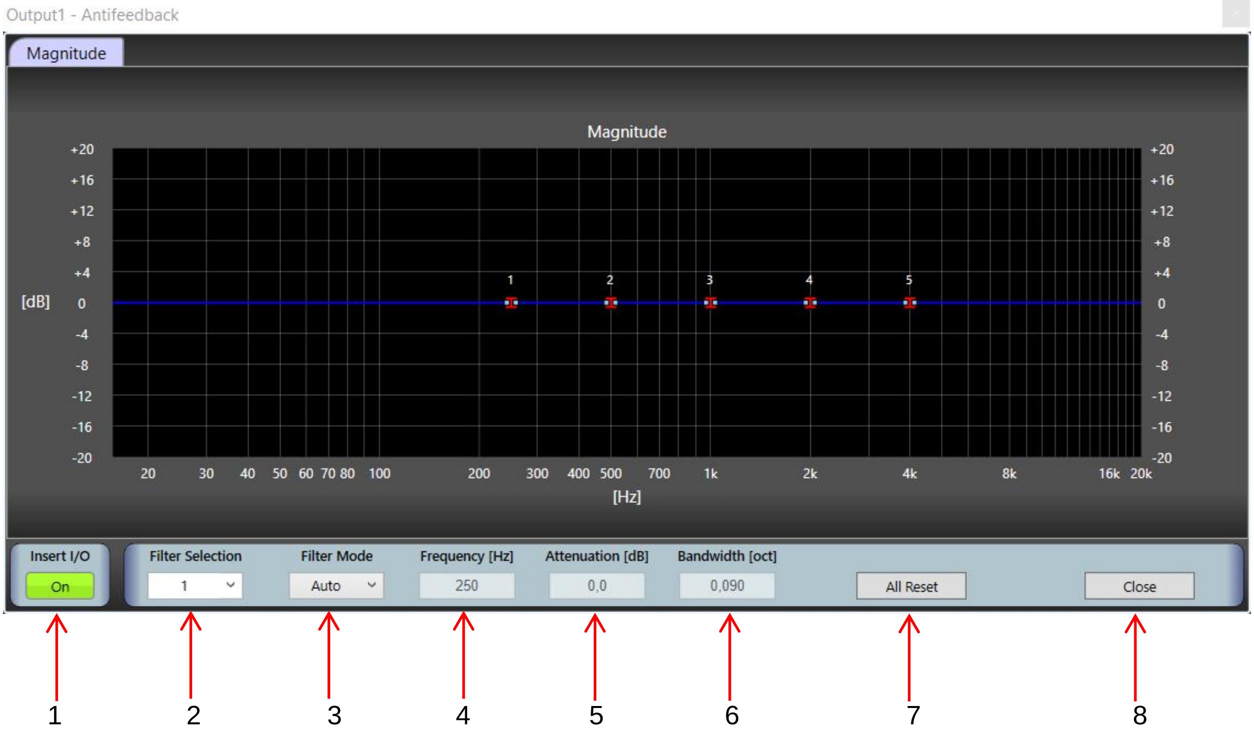
2. Adjust the analog gain for each microphone channel. Set each microphone to the desired performance level while remaining under the point of feedback.

3. Adjust for the desired tonal qualities using the parametric or graphical equalizer panels (eventually after tonal compensations correct the Fader Level or Output Level channel).

4. Active antifeedback pushing INSERT I/O button (green light) and leave all filters in AUTO mode.

Antifeedback Bedienfeld

Dies ist die Haupt-Bedienoberfläche des Antifeedbacks. Sie stellt die eingesetzten Sperrfilter grafisch dar. Die automatische Aktualisierung ermöglicht eine Echtzeitkontrolle über die gerade ablaufende Antifeedback Anwendung.



1	**INSERT I/O**	switches antifeedback on or out. If it is disabled this channel remains completely unaffected. (no antifeedback mode) This button shows a green light if the antifeedback channel is on. **WARNING!** Make sure that the adjusted volume is decreased before the antifeedback application is switched off.
2	**FILTER SELECTION**	Select the blocking filter within the antifeedback filter chain.
3	**FILTER MODE**	**Auto:** If the system detects a feedback frequency the blocking filter is automatically deployed. If in rare particular cases the autofilter is used in some feedback frequencies this frequencies are reduced in steps of 0.5 dB. In auto mode the bandwidth of the notch filter is 0.09 octaves (Q=16) This bandwidth is a good combination of response time and fidelity. **Manual:** The notch filter is set by the user with fixed frequency, attenuation and bandwidth independent of a feedback detection. **Important remarks:** Filter in auto mode are not saved in the adjusted configuration. When the system was switched off and switched on again all auto filters are placed in their standard frequencies with 0 dB attenuation. Filter in manual mode are saved in the adjusted configuration. When the system was switched off and switched on again all manual filters are reset to the previous settings
4	**FREQUENCY**	AUTO Filter Mode: shows the current notchfilter frequeny. MANUAL Filter Mode: sets a fixed notchfilter frequeny.
5	**ATTENUATION**	In AUTO-Filter mode the current notchfilter attenuation is displayed. In Manual Filter mode a fixed notchfilter attenuation is set.
6	**BANDWIDTH**	In AUTO-Filter mode the current notchfilter bandwidth is displayed. In Manual Filter mode a fixed notchfilter bandwidth is set. A higher bandwidth offers a better protection. A lower bandwidth uses a narrow frequency range and changes less signal tones.
7	**RESET**	deletes all antifeedback settings of the channel.,
8	**CLOSE**	closes the antifeedback panel

Section 3

Admin Mode

The third section must be executed by qualified personnel..

Press and hold the p/p button and turn the power on until the following appears on the display:
Admin Mode - Preset 1

With **Preset 1 - 20** the following functions can be performed:
Input 1:
Press the Select button
w/w button input on/off
p/p input gain - w/w dB Level + -
p/p Volume Control - w/w dB Level + -
p/p phantom power
w/w on/off p/p automix function
w/w on/off
p/p RMS VU meter-level input 1.
To return to the main menu press the w/w button briefly.
The same procedure applies for Input 2 – 16.
The same procedure applies to Preset 2 – 20.

Output 1-8
Press the Select button
w/w button output on/off
p/p button level - w/w dB level setting + -
p/p button RMS VU-meter level output.
The same procedure applies to Output 2 – 8.
To return to the main menu press the w/w button briefly.

Matrix
Select button 1 or 2
You will see the following:
Matrix Status Input 1 to Output 1
p/p button matrix level w/w button dB level setting + -
The same procedure applies for input 2 – 16 at output 1.
The same procedure applies for input 1-16 at output 2 – 8.
To return to the main menu press the w/w button briefly.

Preset Setup
By briefly pressing the p/p button you will be asked to enter a pin – a four digit number.
Insert pin (default = 0000):

- ➔ Turn the w/w button for letters, special characters and numbers.
- ➔ If you press the p/p button briefly, the cursor jumps to the next position.

After entering 4 numbers, press the p/p button to reach the next menu. This inscription appears on the display:
Save Changes
Exit Ok
Turn the p/p button until **Preset Setup** appears on the display, press briefly to go to the menu.
By pressing the button again **Preset Selection** appears. With w/w button **Choose Preset** occurs.
Turn the p/p button for **Preset Status** - w/w button on/off
Turn the p/p button for Master Volume - w/w for volume regulation + -
Turn the p/p button for number of open microphones - w/w choice of the number of open microphones.

By turning the p/p button the following appears Preset 1
Name (enter preset name)

- ➔ Turn the w/w button for letters, special characters and numbers.
- ➔ If you press the p/p button briefly, the cursor jumps to the next position.

By turning the p/p button the following appears:
Preset 1: In EQs
Preset 1: Copy
Turning of the w/w button: Selects which copy from Preset 1 (in EQs) to Preset 2-20
Briefly press the p/p button: copy starts.
By turning the p/p button the following appears:
Preset 1: Out EQs
Preset 1: Copy
Turn w/w button: Selects which copy from Preset 1 (Out EQs) to Preset 2-20
Briefly press the p/p button: copy starts.
Turn p/p button
Preset 1: Setup
Preset 1: Copy
Turn w/w button: Selects which copy from Preset 1 (Setup) to Preset 2-20
Briefly press the p/p button: copy starts.
Turn p/p button
Preset 1 Setup
Reset
Briefly press p/p button to delete the displayed preset
Briefly press w/w button - turn p/p button (go back)
Save Changes
-> Exit OK
Press w/w button: Exits the menu without saving
Press p/p button 2 times: Save
By turning the p/p button you leave the **Preset Setup** and enter

System Setup Briefly press to enter the menu.

Startup preset

Turn w/w button:	Selection of presets, with which the device starts at switching on.
Turn p/p button:	Main VU-meter
Turn w/w button:	Selects the VU meter in the display input 1-16 or output 1-8
Turn p/p button:	Panel protection
Turn w/w button:	Panel protection on/off in user mode
Turn p /p button:	PIN code request
Turn w/w button:	PIN code request on/off in admin mode
Turn p/p button:	PIN code change
Press p/p button:	confirm new PIN code

Turn p/p button: RCA Input 8 Enable (turn on input 8 to RCA - CD, cassette)
Turn w/w button: on/off
Turn p/p button: RCA Input 16 Enable (turn on input 16 to RCA - CD, cassette)
Turn p/p button: welcome text (any entry)
Procedure as text or numerical entry (see above)
Turn p/p button: LCD contrast level
Turn w/w button: sensitivity adjustment
Turn p/p button: global settings - reset
Press p/p button: confirmation No / Yes

Attention: If you press yes, you will delete all previous settings and reset the unit back to factory condition!

Technical Specifications

Analog Input Section

Number of balanced inputs	7 + 1 (XLR type connector)
Number of unbalanced inputs	1 (RCA)
Dynamic range	122 dB (“A” weighted)
Analog gain (digitally adjustable)	0 dB ÷ 70 dB with 0.5 dB steps
Nominal sensitivity (balanced input)	-84 dBu (38,8 μV_{rms})
Phantom power (digitally activated)	+48 V stabilized, very low noise
Balanced input impedance (XLR)	5,8 kΩ @ 1 kHz
Unbalanced input impedance (RCA)	14,7 kΩ @ 1 kHz
Frequency response (20 Hz ÷ 20 kHz @ +4dBu)	-0.5 dB ÷ 0 dB
Maximum balanced input level	20.2 dBu (7.92 V_{rms})
Input protections	radio frequency interference (RFI) transient voltage spikes external overvoltage

Analog Output Section

Number of balanced outputs	6 + 2 (XLR type connector)
Number of unbalanced outputs	2 (RCA type connector)
Dynamic range	121 dB (“A” weighted)
Residual noise of output driver	-101 dBu (20 Hz ÷ 20 kHz)
Nominal level (balanced output)	4 dBu (1,23 V_{rms})
Maximum level (balanced output)	20.2 dBu (7,92 V_{rms})
Output impedance	50 Ω typical
Output protections	short circuits radio frequency interference (RFI) transient voltage spikes external overvoltage

Analog to Digital Converter

Bit resolution	24-bit
Converter type	sigma delta
Sampling frequency (Fs)	48 kHz
Signal to noise ratio (SNR)	111 dB (“A” weighted @ 48 kHz)
Dynamic range	111 dB (-60 dB_{FS})
Total harmonic distortion (THD)	-102 dB (1 kHz, -0,1 dB_{FS})
Oversampling factor	128 Fs

Digital Signal Processor

DSP	32-bit / 40-bit, Floating-Point 400 MHz - 2,5 ns instruction cycle Super Harvard Architecture 2,4 GFLOPS, 2Mbits SRAM

Digital to Analog Converter

Bit resolution	24-bit
Converter type	sigma delta
Sampling frequency (Fs)	48 kHz
Signal to noise ratio (SNR)	117 dB (“A” weighted @ 48 kHz)
Dynamic range	117 dB (-60 dB_{FS})
Total harmonic distortion (THD)	-104 dB (1 kHz, -0,1 dB_{FS})
Delay time	0.66 ms
Oversampling factor	256 Fs

Digital Processing

Inputs Blocks:

Highpass filter (anti hum, anti rumble, ect)	Butterworth filter type with cutting frequency at 160 Hz and slope12 dB/Octave
5-PEQs equalizer	Frequency: 20 Hz ÷ 20 kHz Gain: -15 dB ÷ 15 dB Bandwidth: 0,014 ÷ 6,672 oct
Noise gate	Threshold: -60 dB_{FS} ÷ 0 dB_{FS} Hold Time: 100 ms ÷ 10 s
Dynamic compressor range	Threshold: [-90 dB_{FS} ÷ 20 dB_{FS} Ratio: R=1:1 ÷ R=20:1 Post Gain: -20 dB ÷ 20 dB Attack Time: 1 ms ÷ 250 ms Release Time: 10 ms ÷ 2500 ms
Automix function	Hold Time: 100 ms ÷ 5000 ms Attenuation: -60 dB ÷ 0 dB NOM Gain: increase post gain of -3 dB for each doubling of opened automix channels)
Priority:	1 (lowest) ÷ 5 (highest) Max opened channels: 1 ÷ 16
Fader level	[-60 dB ÷ 10 dB]

Input / Output Routing Matrix:

Matrix size	8 In / 4 Out
Matrix cross point level adjusting	[-60 dB ÷ 10 dB]

Output Blocks:

5-PEQs equalizer	Frequency: 20 Hz ÷ 20 kHz Gain: -15 dB ÷ 15 dB Bandwidth: 0,014 ÷ 6,672 oct
31-Bands graphic equalizer	Gain: -12 dB ÷ 12 dB
Noise gate	Threshold: -60 dB_{FS} ÷ 0 dB_{FS} Hold Time: 100 ms ÷ 10]
Dynamic compressor range	Threshold: -90 dB_{FS} ÷ 20 dB_{FS} Ratio: R=1:1 ÷ R=20:1 Post Gain: -20 dB ÷ 20 dB Attack Time:1 ms ÷ 250 ms Release Time:10 ms ÷ 2500 ms

Limiter	Threshold fixed at 0 dB_{FS}
Delay	0 m ÷ 233 m, 0 ms ÷ 679 ms
Phase control	0°, 180°
Output level	-60 dB ÷ 10 dB
Master level	-60 dB ÷ 10 dB

Data Connections

Front panel	USB 2.0
Rear panel	RS232 @ 38400 kbit/s

Display

LCD	20 characters x 2 lines

PSU Module

AC range	230 VAC ± 10%
Input frequency	47 Hz to 67 Hz
Power consumption	max 33 W
Analog voltages	+48 VDC, ±15 VDC, +5 VDC
Digital voltages	+3.3 VDC, +1,2 VDC
Voltage regulators	linear type (no switching noise)

Dimensions and Weight

Height	88 mm
Width	484 mm
Depth	260 mm + 60 mm connector
Weight	6 kg

Temperature Range

Indoor	0°C to 40°C (32°F to 102°F)

Compliances

AES48-2005 grounding scheme
2002/95/EC
CE

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