Audio Vertriebs
Entwicklungsgesellschaft
mbH
STUTTGART

DMV-64

Digital automatic mixing amplifier
with 6 mic/line inputs and 4 outputs

User guide

Table of contents

Introduction

Thank you for choosing the 6-channel digital mixer DMV-64 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 DMV-64 is equipped with its own software and offers users a wide range of practical functions.

Streaming Media Server:
DMV-64 is a live audio streamer..

It records audio from an input/output channel, encodes it and sends it to a streaming server.

DMV-64 can do live streaming, so people can hear your video while it is being recorded.

It can be used to create an internet radio station or a privately running jukebox and many things in between.

It is very versatile in that new formats can be added relatively easily.

DMV-64 supports open standards for communication and interaction.

DMV-64 supports TCP/IP, UDP, RTSP and RTMP streaming protocols.

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 DMV-64 comes with an approved power 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 DMV-64
- Power supply cable
- Cat 5 cable

Der DMV-64 is delivered with three presets set at the factory.
The presets are located in 11 and 12. .

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

Section 1

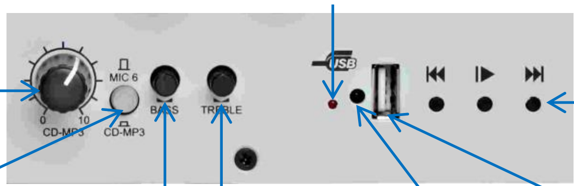
The **first section** explains how the **user** can easily and comfortably change the master volume and retrieve the presets (up to 20) directly on the DMV-64 device without using the laptop or PC. (User Modus)

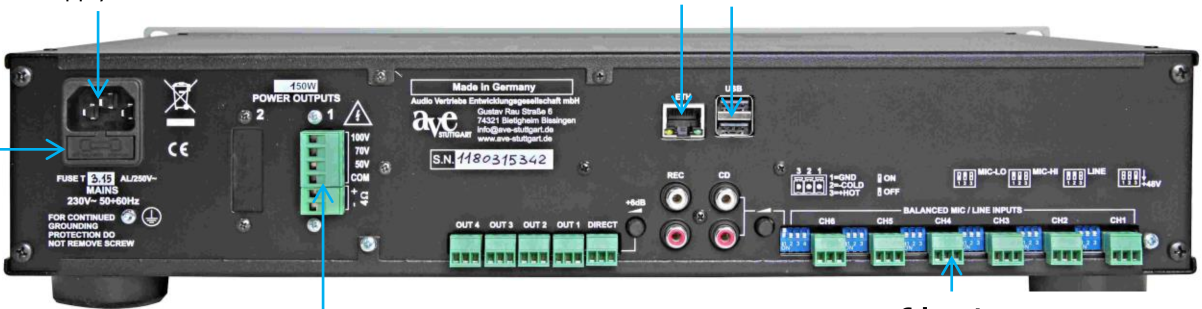
Input Level 1 – 6
By tuning the volume control of the 6 digital inputs the volume will be increased or decreased. kö
Input 6 can be switched between CD/USB and Micro/Line.

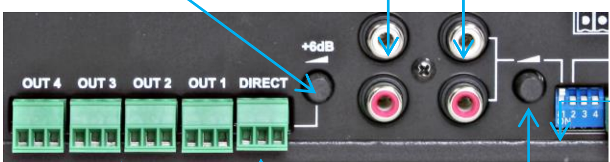
Control of the master volume
By turning the volume/value control te master volume will be increased or decreased.

Select preset
By turning the preset control the desired preset is selected.

Note: If the protection is activated, the volume and preset selection is adjusted by pressing and then turning.



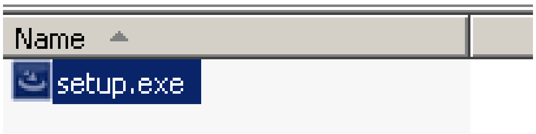




Section 2

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 only be carried out by technical specialists.**

The parameter menue/ Programming of the DMV-64



You will 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 this link on the desktop

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Before starting programming set the manual functions of the inputs (Advice: volume control approx. 12 o'clock) and the DIP-switches on the backside. .

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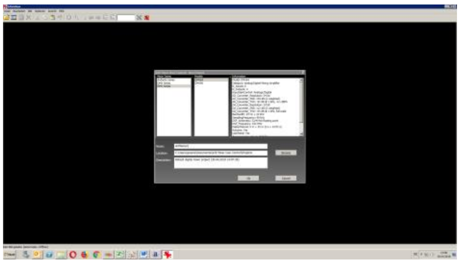
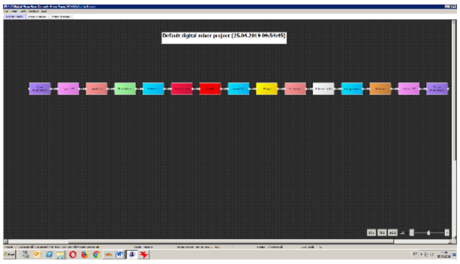
Programming of the DMV-64

Restart your PC or your laptop, so the program can be executed properly. Use the supplied connector cable between your laptop and DMV-64. After starting the program klick with the left mouse button **File** and then **New Projekt.**

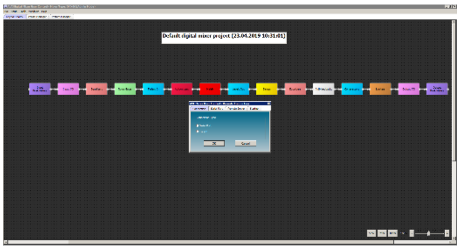
The following picture appears:



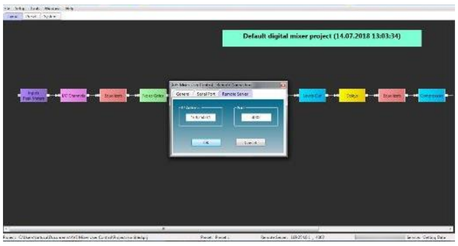
In the menu bar you find under **File** and **Setup,** details for handling the program. Specify the type of the device and confirm by pressing OK.



This window shows all functions of the mixer. Select in the upper bar a project name. You can adjust all commands to your monitor with the zoom on the bottom right.



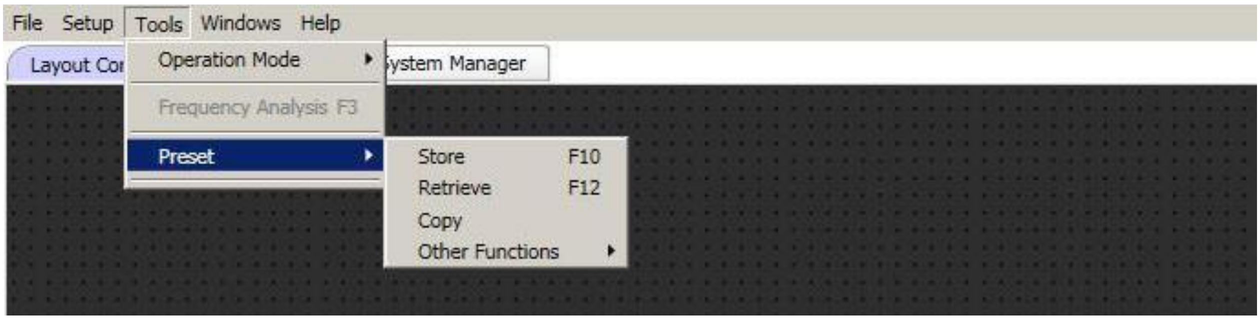
Press with the left mouse bottom **Setup** and then **Remote Connection.** Confirm with OK. After a short time on the display appears **Remote Connected.** Now you have access to all functions you want to program.



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

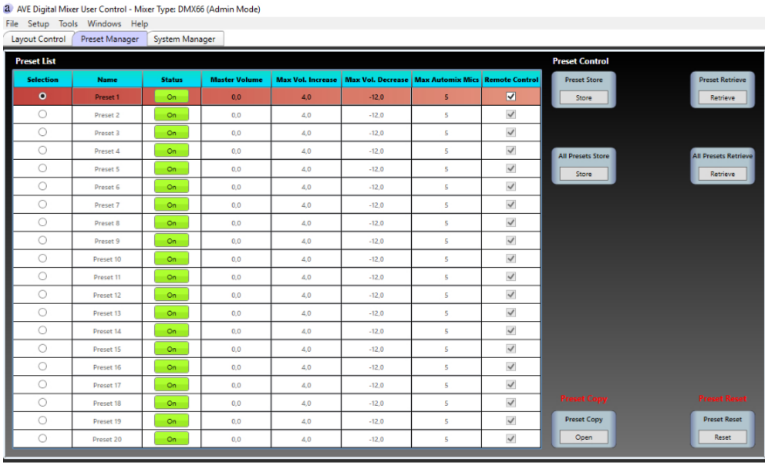


When you click **Tools** in the menu bar**,** a pull-down menu with following selection options is opened:

Operation Mode		Choice betwee Online und Offline	
Preset	**Store**	Storing presets	
	Retrieve	Retrieve presets	
	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.

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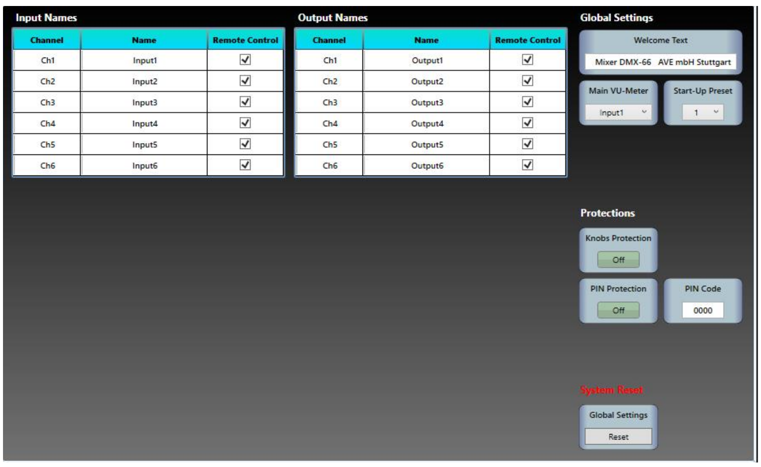
Der System Manager



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.

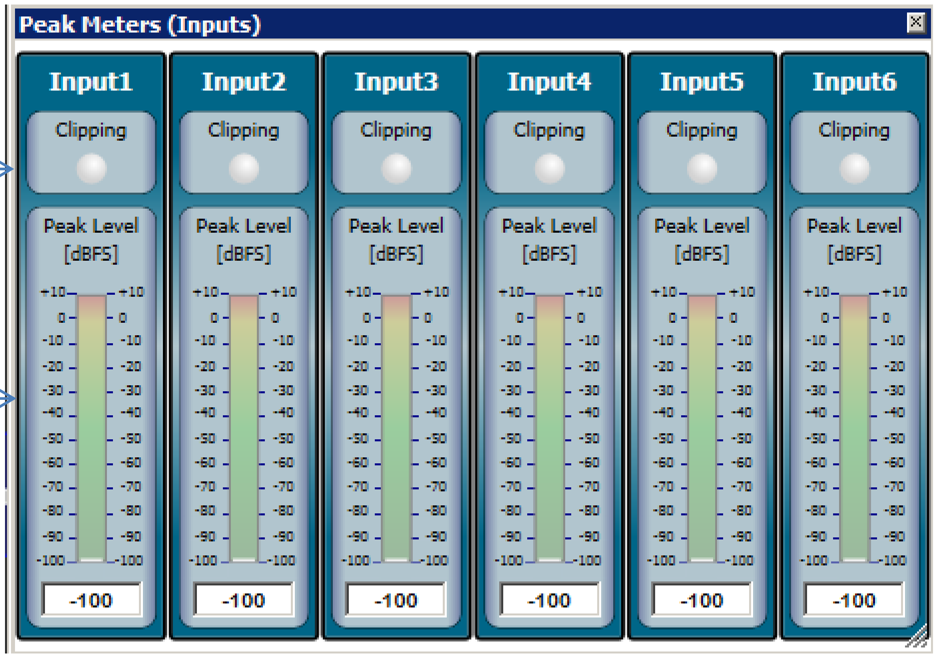
PIN Protection
Activation of the PIN of the Admin Mode

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

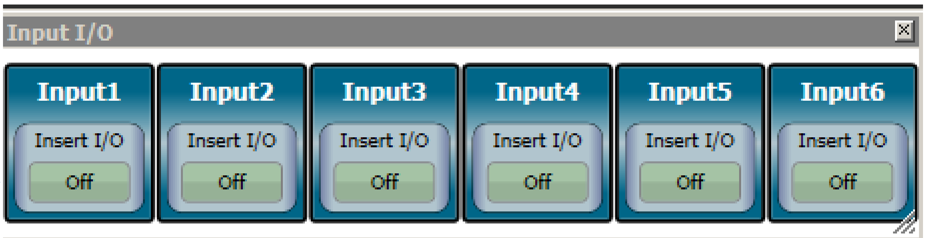
System Reset

Clipping indicator

Peak level indicator



Inputs on/out



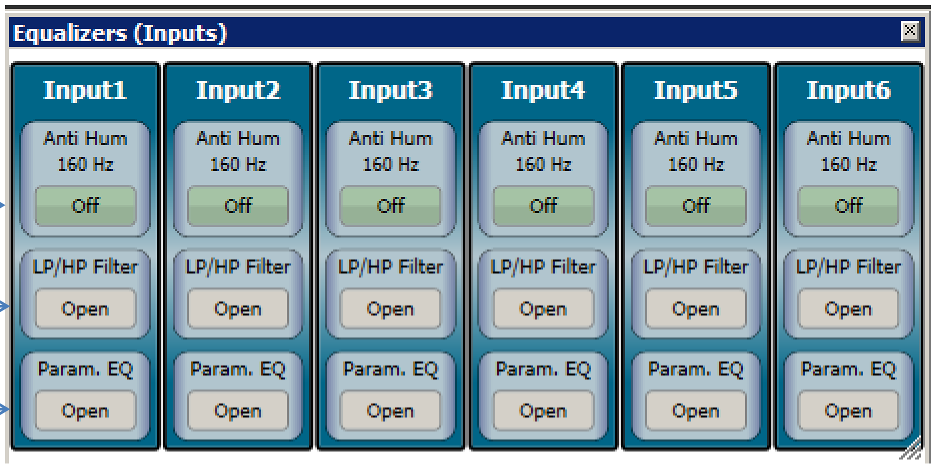
Parametric Equalizer:
5 parameters are available for each input and each output. The settings can be copied to each other input and output.
The same applies to the highpass/lowpass filters.

At the output there is a 5 band parametric equalizer as well.

Hum filter 160 Hz

Lowpass/highpass filter

5 band parametric equalizer



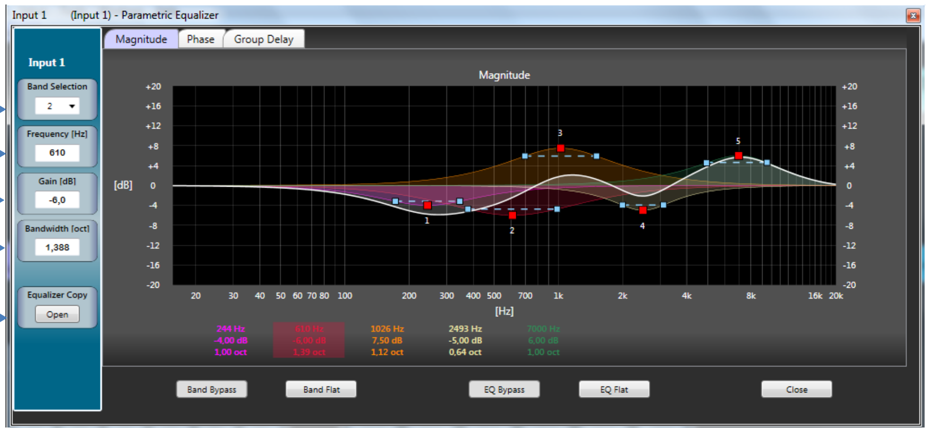
Band Selection

Frequency [Hz]

Gain [dB]

Bandwidth [Oct]

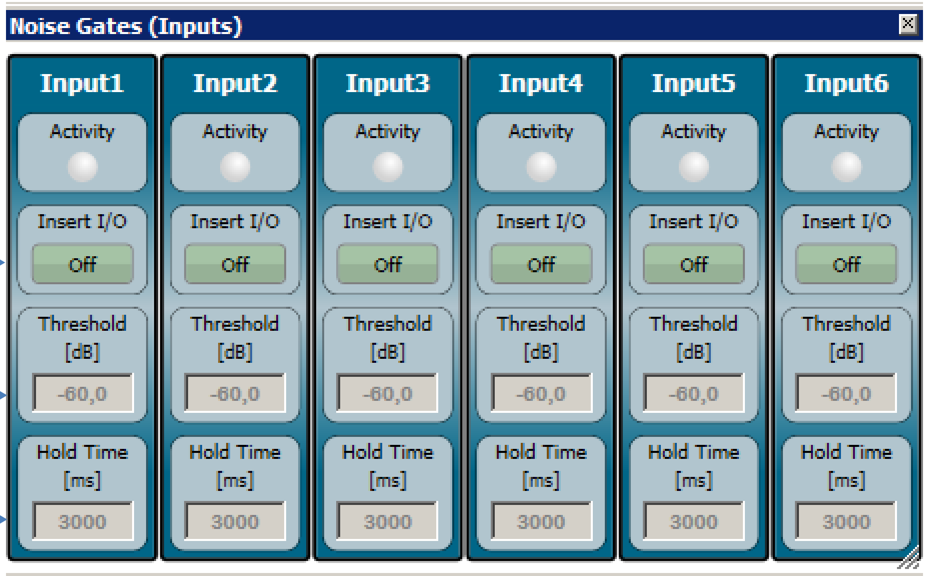
Equalizer Copy



Noise Gate On/Out

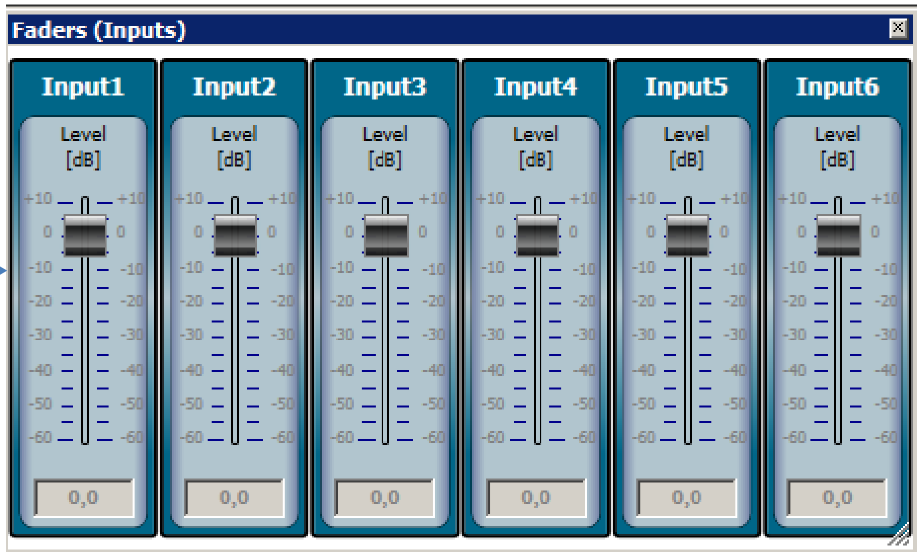
Threshold [dB]

Hold time[ms]



Fader Level Setting

Input option for programmer
to change the volume at the input



Selecting

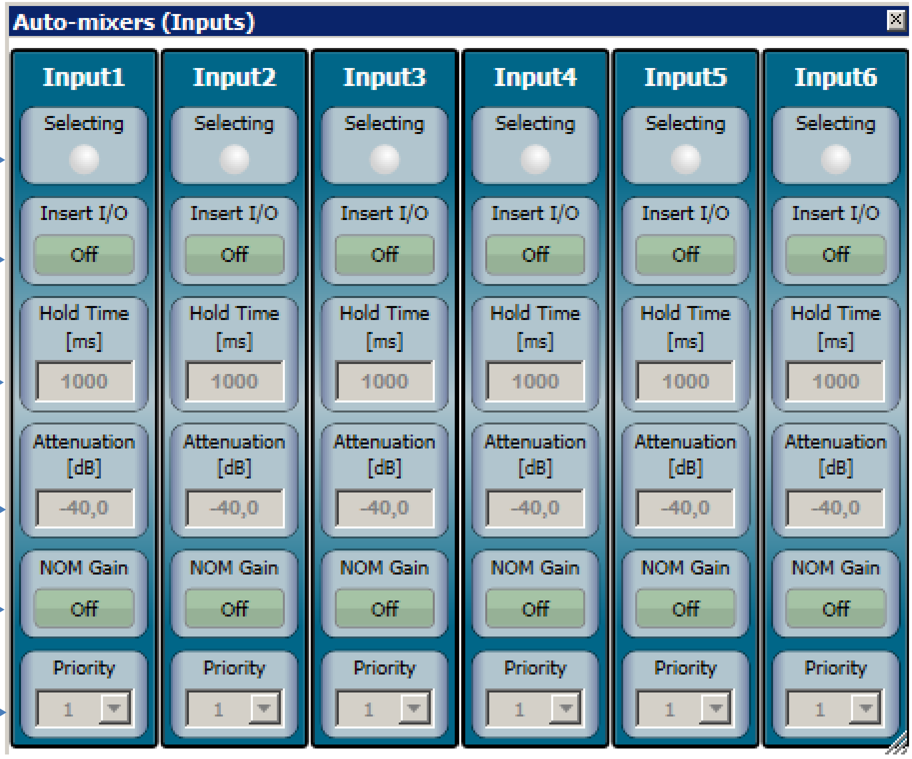
Automixer On/Out

Hold time [ms]

Attenuation [dB]

NOM Gain On/Out
Reducing the level when microphones are activated simultaneously

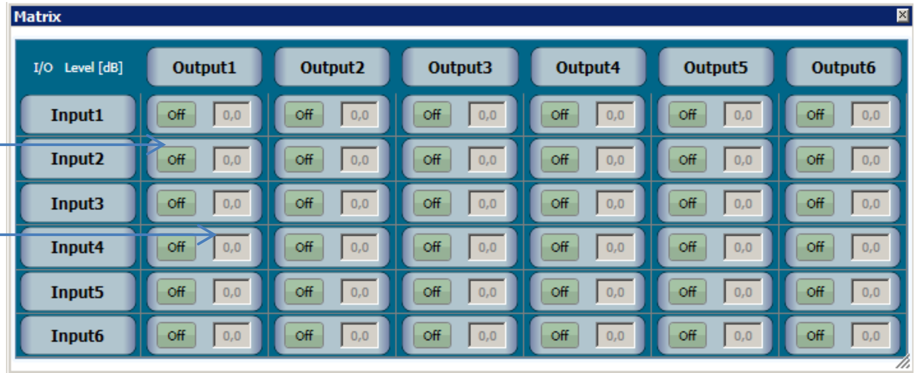
Priority



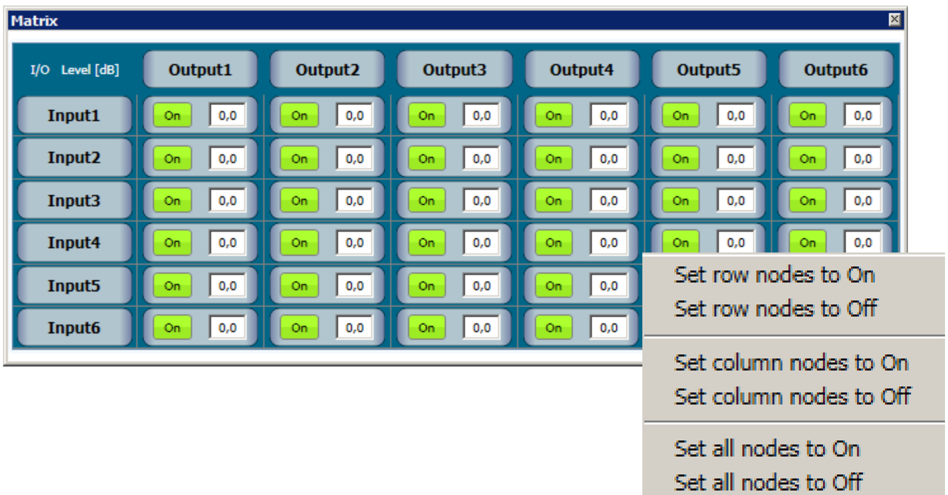
Detailed description of the automixer on page 15

Matrix On/Out

Level setting [dB]

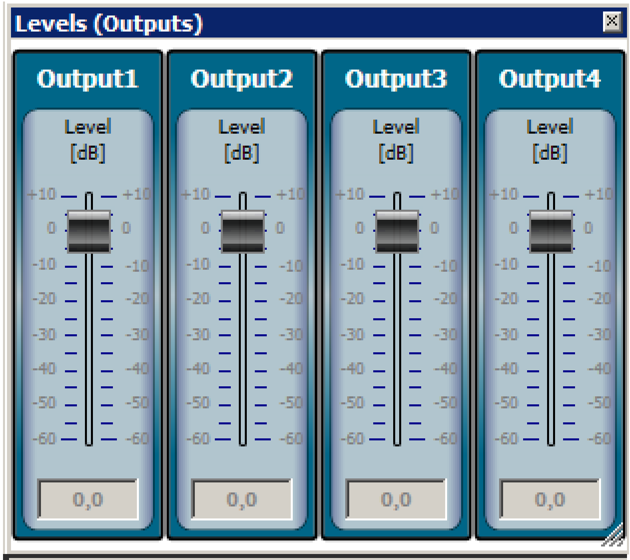


After pressing the right mouse bottom a pull-down menu opens. It allows the activation of all matrix nodes or different rows or columns of nodes. Routing of all inputs and outputs with additional level setting.



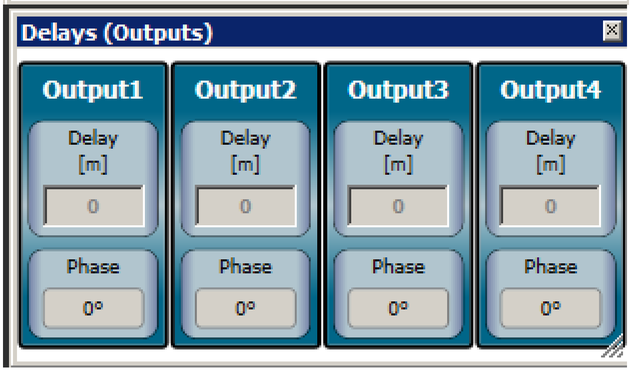
Output

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



Delay [m]

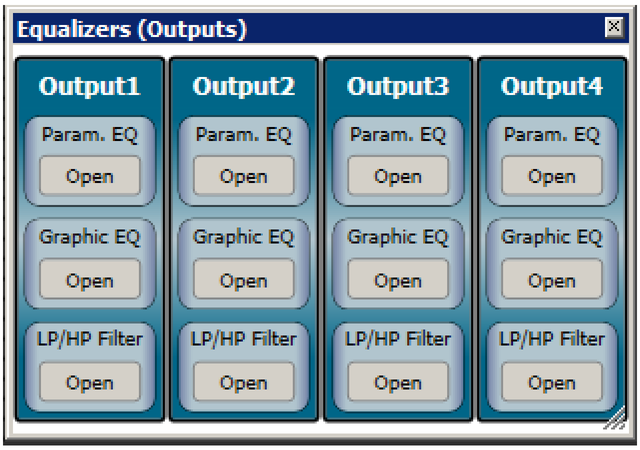
Phase

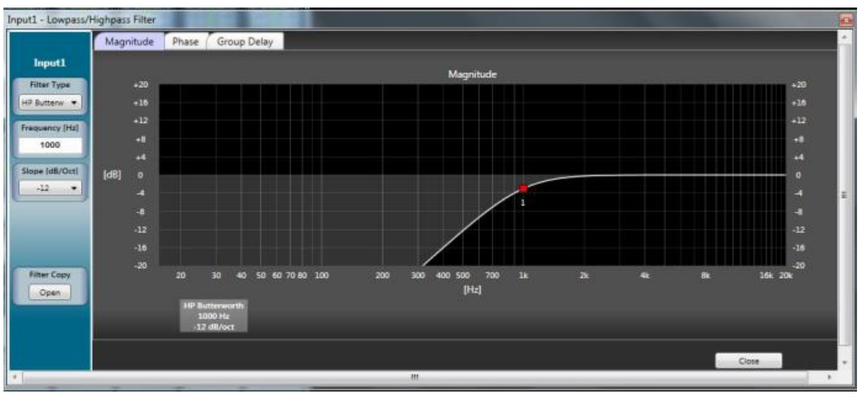


Parametric EQ

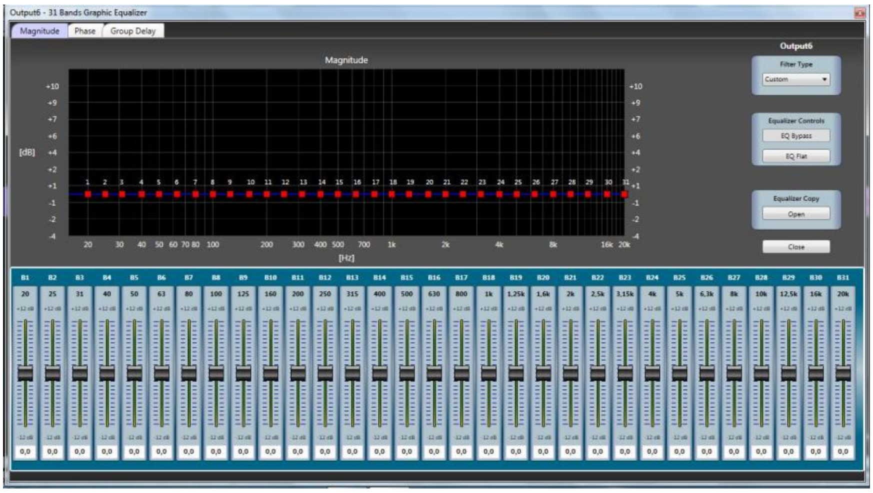
Graphic EQ

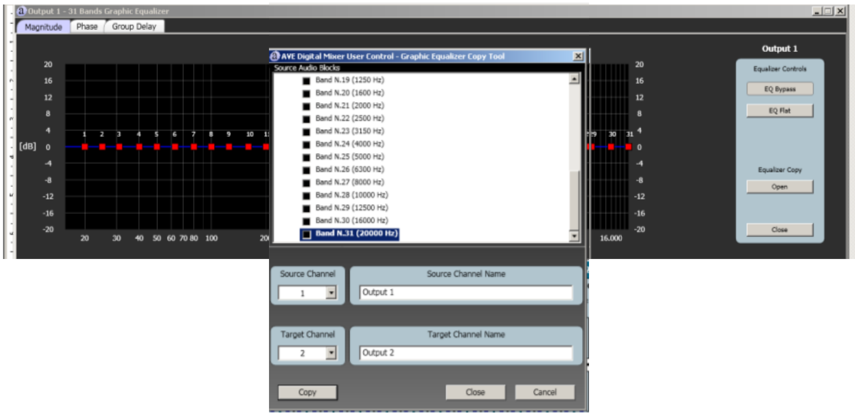
Lowpass/Highpass filter



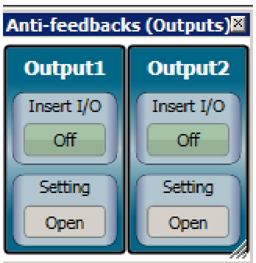


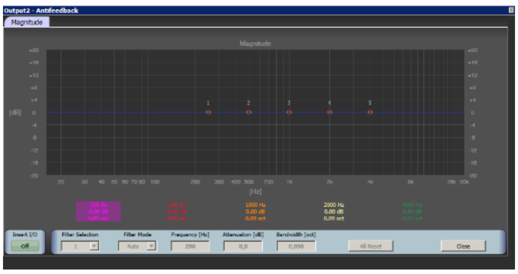
Each output has a 31 band graphic equalizer. All parameters can be copied to the other EQs or individually adjusted for each output.



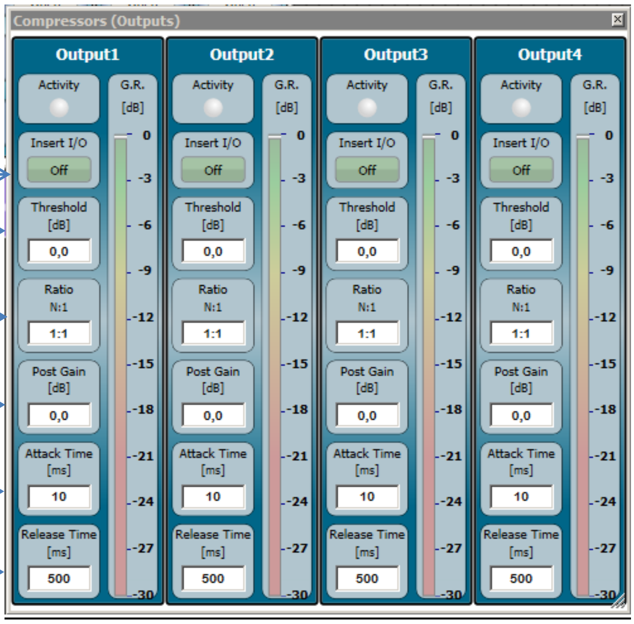


Detailed description of the antifeedback functions on page 17

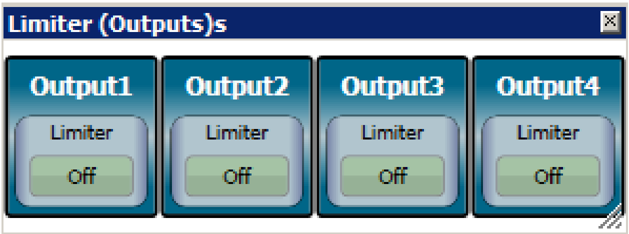




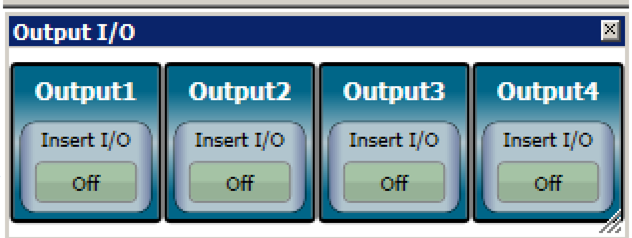
On/off
Threshold [dB]
Ratio
Post Gain [dB]
Attack Time [ms]
Release Time [ms]



Limiter On/Out

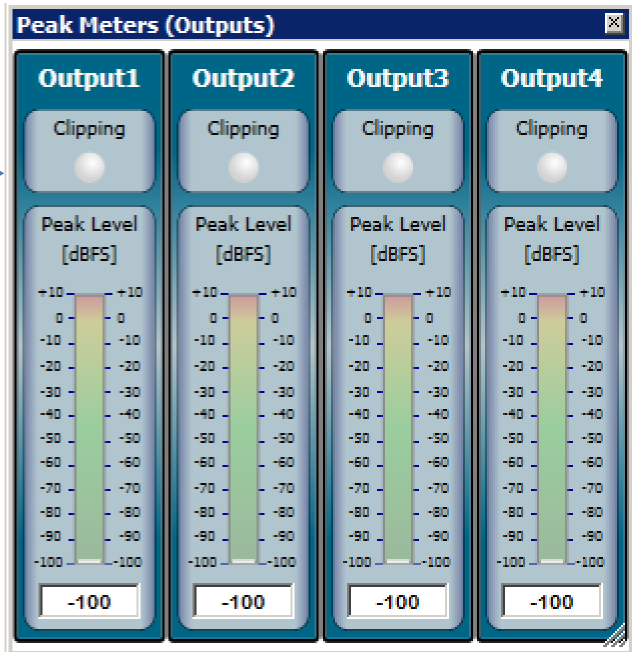


Outputs On/Out



Clipping

Peak level [dBFS]



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** feature keeps the most recently activated microphone even if the sound level drops below the threshold setting.

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.

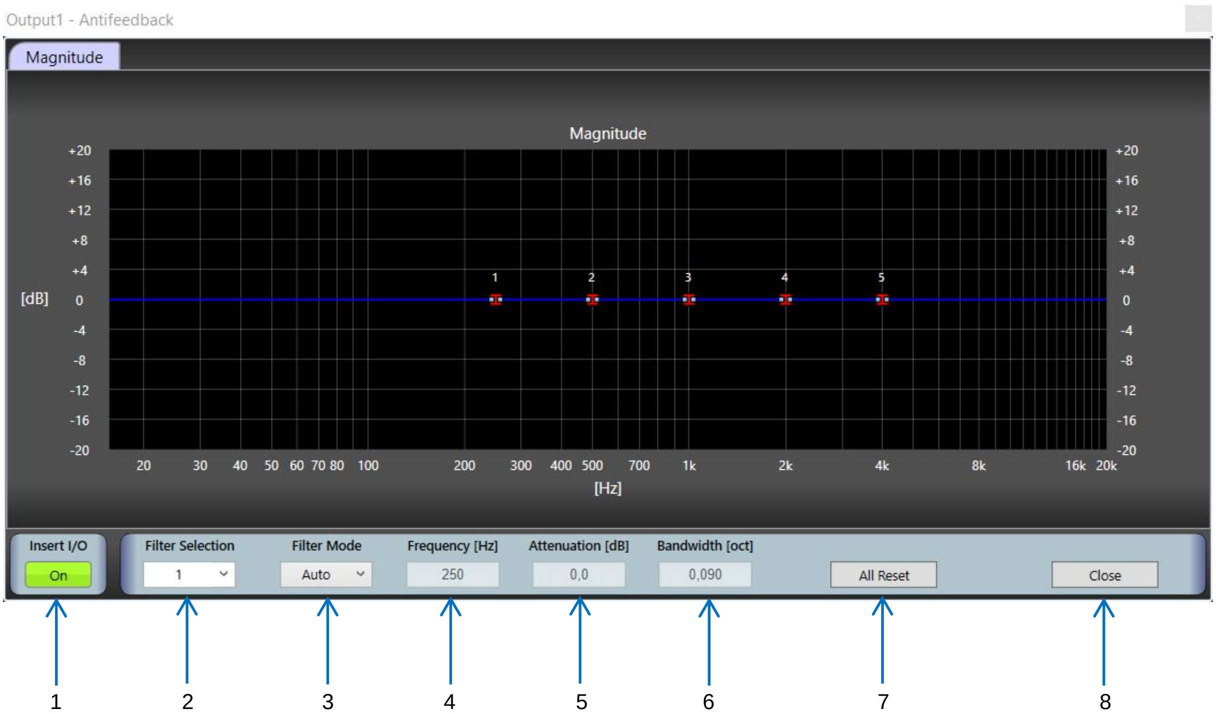
Application example:
The highest priority reveals useful inside churches during organ playing, where a separate microphone with PRIORITY equal to 5, placed close to the organ pipes, gates off all microphones in presbyter.

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.

Antifeedback panel

This is the main antifeedback user interface. It allows viewing of the graphical response of the notch filters engaged. The auto refresh gets a real-time view of what is currently in the Anti Feedback setup.



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 applification 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

Technical specifications

Analog inputs

• Number of balanced inputs	6 (Phoenix 3,5 mm connector)
• Number of unbalanced inputs	1 (RCA connector)
• Analog gain (manually adjustable)	0 dB - 30 dB
• Nominal sensitivity (balanced input) MIV-HI	-58 dBu (1mV$_{rms}$)
• Nominal sensitivity (balanced input)MIV-LO	-34 dBu (15mV$_{rms}$)
• Nominal sensitivity (balanced input)MIV-HI	-19 dBu (87mV$_{rms}$)
• Phantom power (Activating at DIP-Switch	+48 Volt stabilized, very low noise
• Balanced input impedance (Phoenix)	5 kΩ @ 1 kHz
• Unbalanced input impedance (RCA)	33 kΩ @ 1 kHz
• Balanced input CMRR	<60 dB @ 1 kHz
• On Mic (20 - 20 kHz weighted) Rs =150 Ohm	-126 dBV
• Frequency response MIC (-3dB)	160 Hz bis 20 kHz
• Frequency response LINE (-3dB)	40 Hz bis 20 kHz
• Input protections	radio frequency interference (RFI) transient voltage spikes external overvoltage

Analog outputs

• Number of balanced outputs	6 (Phoenix 3,5 mm connector)
• Number of unbalanced outputs	1 (RCA connector)
• Dynamik range	120 dB (“A” weighted)
• Residual noise of output driver	-100 dBu (20 Hz ÷ 20 kHz)
• Nominal level (balanced output)	0 dBu (7,75 V$_{rms}$)
• Maximum level (balanced output)	20 dBu (7,75 V$_{rms}$)
• Balanced output impedance	140 Ω
• Unbalanced output impedance	70 Ω
• Ausgangsschutz	short circuits Overvoltage protection

Analog to digital converter

• Bit resolution	24-bit
• Converter type	sigma delta
• sampling frequency (Fs)	48 kHz
• Signal to noise ratio (SNR)	104 dB (“A” weighted @ 48 kHz)
• Dynamic range	104 dB (-60 dB$_{FS}$)
• Total harmonic distortion (THD)	-93 dB (1 kHz, -1 dB$_{FS}$)
• Oversampling factor	512 Fs

Digital Signal Processor

• DSP	32-bit / 40-bit, Floating point 150 MHz - 6,6 ns cycling rate Super Harvard Architecture 900 MFLOPS 1Mbits SRAM, two-channel

Digital to analog converter

• Bit resolution	24-bit
• Converter type	sigma delta
• Sampling frequency (Fs)	48 kHz
• Signal to noise ratio (SNR)	112 dB (“A” weighted @ 48 kHz)
• Dynamic range	112 dB (-60 dB_{FS})
• Total harmonic distortion (THD)	-94 dB (1 kHz, 0 dB_{FS})
• Delay time	0,58 ms
• Oversampling factor	512 Fs

Digital Processing

Input

• Highpass/lowpass filter (anti hum, anti rumble and more)	Butterworth filter type with adjustable cutting frequency and selectable slope 12/24/48 dB/Octave
• 5 parametric equalizuer PEQ	Frequency [20 Hz ÷ 20 kHz] Gain [-15 dB ÷ 15 dB] Bandwidth [0,01 ÷ 6 oct]
• Noise Gate	Threshold [-80 dB_{FS} ÷ 0 dB_{FS}] Hold time [100 ms ÷ 10 s]
• Automix function	Hold time [100 ms ÷ 5000 ms] Attenuation [-60 dB ÷ 0 dB] NOM Gain (increase post gain of -3dB for each doubling of opened automix channels) Max. opened channels [1 ÷ 6]
• Volume control	[-100 dB ÷ 10 dB]

Routing Matrix:

• Matix size	6 inputs / 6 outputs
• Matix crosspoint level adjusting	[-60 dB ÷ 10 dB]

Audio output

• Easy adjustment of the sound columns	AT-N series Live SM series
• 31 bands graphic equalizer	Gain [-12 dB ÷ 12 dB]
• 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 ÷ 35 m], [0 ms ÷ 100 ms]
• Phase control	[0°, 180°]
• Output level	[-100 dB ÷ 10 dB]
• Master level	[-100 dB ÷ 10 dB]

Data connections

• Rear panel	ETHERNET 802.3 Wi-Fi 802.11 USB 2.0

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	84 mm
• Width	484 mm
• Depth	340 mm + 60 mm connector
• Weight	5.6 kg (1x480W) 6.8 kg (2x320W and 2x480W) 12 kg (1x150 W and 1x240 W) 13.5 kg (2x150 W)

Temperature range

• Indoor	0°C to 40°C (32°F bis 102°F)

Zertifizierungen

- AES48-2005 grounding scheme
- 2002/95/EC
- CE

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