innosonix

MAXX SERIES

MA16 HP²

MA24 HP²

MA32 HP²

MANUAL

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INDEX

Chapter 1. SAFETY INSTRUCTIONS

1.1. GENERAL

Before using the product, please read this manual and follow all Safety Instructions. They are used to protect you, help to avoid equipment defects and damages resulting from improper use. Keep this manual in a safe place.

CAUTION: THE POWER SUPPLY CORD IS USED AS THE MAIN DISCONNECT DEVICE, ENSURE THAT THE SOCKET-OUTLET IS LOCATED/INSTALLED NEAR THE EQUIPMENT AND IS EASILY ACCESSIBLE

ATTENTION: LE CORDON D'ALIMENTATION EST UTILISÉ COMME INTERRUPTEUR PRINCIPAL. LA PRISE DE COURANT DOIT ÊTRE SITUÉE OU INSTALLÉE À PROXIMITÉ DE L'ÉQUIPEMENT ET ÊTRE FACILE D'ACCÉS

CAUTION - DANGER OF EXPLOSION IF BATTERY IS INCORRECTLY REPLACED. REPLACE ONLY WITH THE SAME OR EQUIVALENT TYPE.

ATTENTION - DANGER D'EXPLOSION LORSQUE LA BATTERIE N'EST PAS REMPLACÉE CORRECTEMENT. REMPLACER UNIQUEMENT AVEC DES BATTERIES IDENTIQUES OU D'UN TYPE ÉQUIVALENT

CAUTION - THESE SERVICING INSTRUCTIONS ARE FOR USE BY QUALIFIED SERVICE PERSONNEL ONLY. TO REDUCE THE RISK OF ELECTRIC SHOCK DO NOT PERFORM ANY SERVICING OTHER THAN THAT CONTAINED IN THE OPERATING INSTRUCTIONS UNLESS YOU ARE QUALIFIED TO DO SO.

ATTENTION - CES CONSIGNES D'ENTRETIEN DOIVENT ETRE UNIQUEMENT EMPLOYES PAR LE PERSONNEL DE SERVICE QUALIFIÉ. POUR RÉDUIRE LE RISQUE DE CHOC ÉLECTRIQUE NE PAS EFFECTUER DES REPARATIONS AUTRES QUE CEUX CONTENUS DANS LES INSTRUCTIONS D'UTILISATION A MOINS QUE VOUS SOYEZ QUALIFIE POUR LE FAIRE

The amplifier is a device of protection class 1. Make sure that the protective conductor (earth) is connected properly. A missing earth can lead to dangerous voltages at the enclousure!

Keep the device away from dust, moisture, water and other liquids! In such case, the further usage is prohibited!

The amplifier has a relatively high output power and possibly can be a hazard for people and speakers. Pay particular attention to any defective set volume.

Do not touch the housing of the device, during operation. The surfaces can be hot. After switching off the device, wait 30 minutes till touching the device.

In the following cases it is necessary to return the amplifier for examination to the manufacturer. Contact details can be found on our website: www.innosonix.de

- The unit has been dropped, mechanically damaged or treated improperly.
- The power cord or plug has been damaged.
- Objects have fallen into the unit.
- Liquid has been spilled into the unit.
- The unit is not operating normally.
- The device displays errors.

1.2. VENTILATION AND COOLING

Built-in and 19-inch racks must be ventilated adequately.

The active cooling system inside the device creates front to back ventilation.

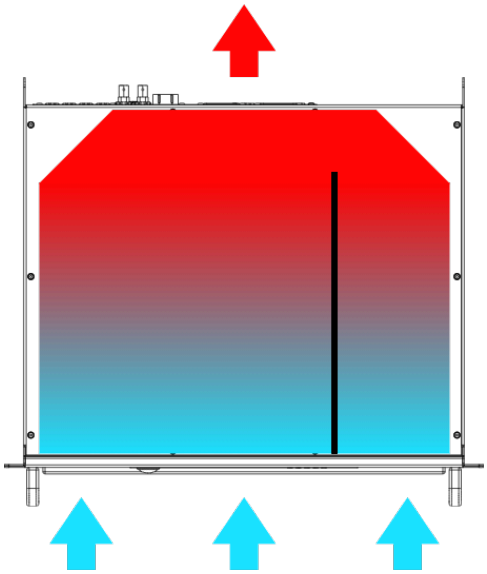


Figure 1. MAXX/HP² Ventilation

1.3. OPERATING CONDITIONS

Enviromental operating temperature	0 - 40°C			
Thermal dissipation	Fan, variable speed, temperature controlled front to rear airflow			
	32 CH			
	@230V		@110V	
amps power off	31 kcal/h	123 BTU/h	31 kcal/h	123 BTU/h
idle	84 kcal/h	333 BTU/h	88 kcal/h	349 BTU/h
1/8 power @ 4 Ohm	269 kcal/h	1066 BTU/h	270 kcal/h	1071 BTU/h
	24 CH			
	@230V		@110V	
amps power off	31 kcal/h	123 BTU/h	31 kcal/h	123 BTU/h
idle	71 kcal/h	282 BTU/h	72 kcal/h	286 BTU/h
1/8 power @ 4 Ohm	210 kcal/h	833 BTU/h	212 kcal/h	840 BTU/h

Chapter 2. TECHNICAL SPECIFICATION

2.1. MAxx/HP² front view

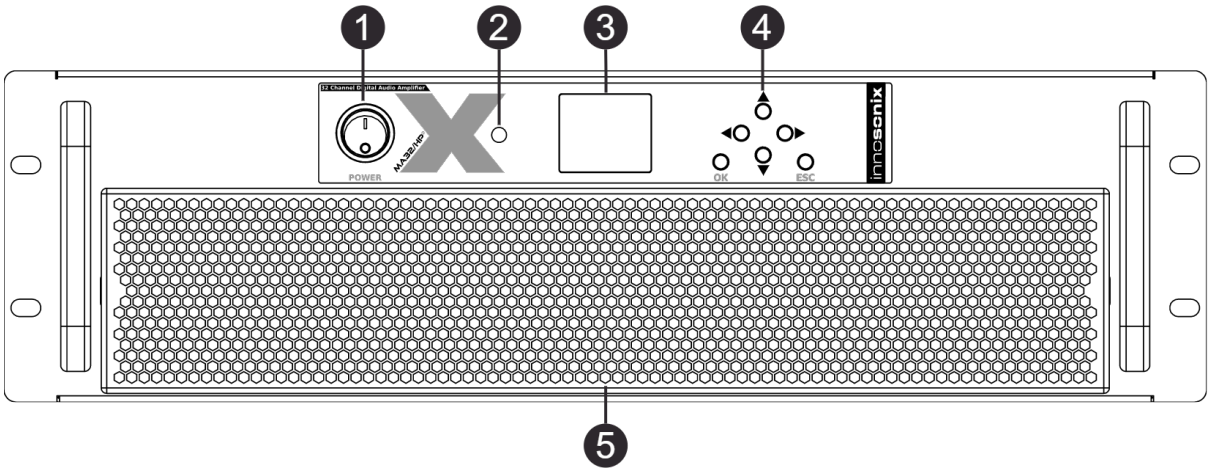


Figure 2. MA32/HP² front view

Table 1. DEVICE ELEMENTS FRONT

NR	DESRIPTION	NOTE
1	POWER SWITCH	Hard disconnect Mains
2	POWER LED	POWER LED
3	DISPLAY	DISPLAY
4	BUTTONS	DISPLAY
5	VENTILATION GRILLS	Magnetically attached see: SERVICE

2.2. MAxx/HP² back view

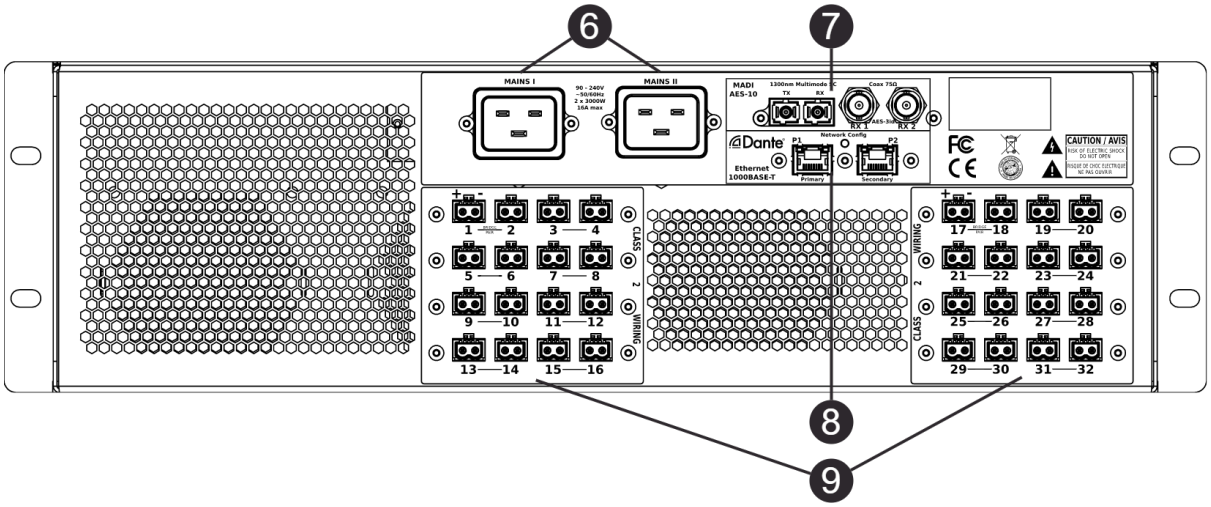


Figure 3. MA32/HP² back view

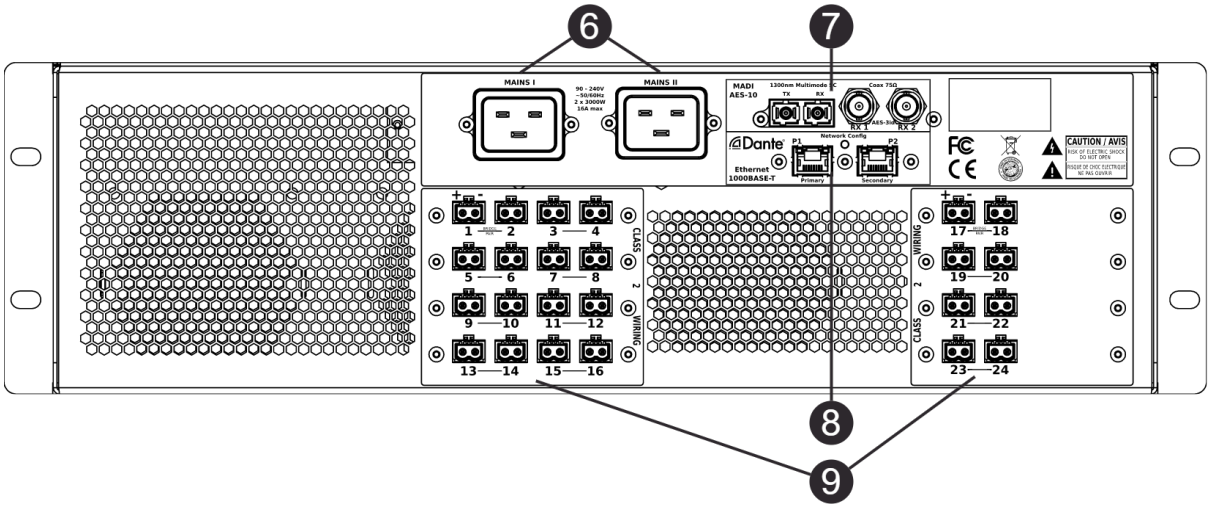


Figure 4. MA24/HP² back view

Table 2. DEVICE ELEMENTS BACK

NR	DESRIPTION	NOTE
6	MAINS SUPPLY	MAINS SUPPLY
7	MADI	MADI (AES10) / AES3
8	ETHERNET/DANTE	ETHERNET / DANTE
9	AMP OUTPUTS	AMP OUTPUT

2.3. DIMENSIONS & WEIGHT

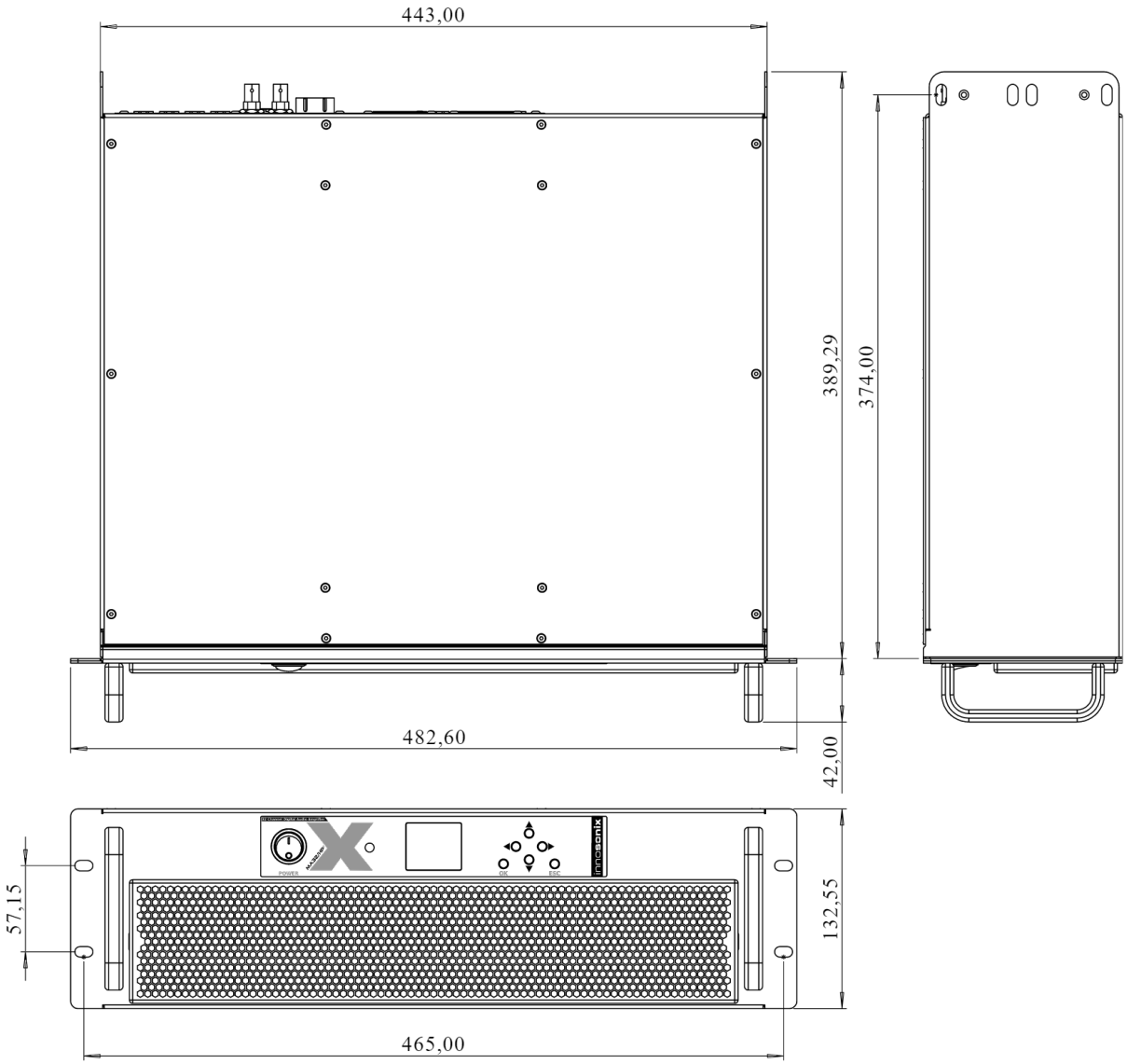


Figure 5. MAXX/HP² dimensions

Dimensions	W 482.60mm (19") H132.55mm (3 RU), D 389.29mm	
	32CH	24CH
Weight	19.6 kg	18.3 kg
Dimensions Boxed	60 x 52 x 25 cm	
Weight Boxed	22.6 kg	21.3 kg

2.4. CONNECTIONS & CABLE

Control input connectors	RJ45 (1Gbit/s Ethernet)
Audio signal input connectors	RJ45 (DANTE / AES67), BNC 75R (MADI Coax, AES3id), SC Optic (MADI Fibre)
Speaker connector	Wuerth Elektronik 691352710002
	Phoenix Contact MSTB 2,5/ 2-ST - 1754449
AC mains	2x IEC C20 inlet / IEC C19 plug

2.4.1. MAINS SUPPLY

Power supply	Two Universal, regulated switch mode with PFC (Power Factor Correction)
Operating Voltage	90 - 264VAC 50/60Hz
AC Current typ.	14.3A per PSU
Inrush Current	60A max.
Suggested circuit breaker	2x B16
Earth Leakage Current	<0.75mA / 240V

32 CH Version		
Power Factor	@230V	@110V
amps power off	0.46	0.85
idle	0.64	0.95
1/8 power @ 4 Ohm	0.96	0.98
Consumption / current draw	@230V	@110V
amps power off	36W	36W
idle	98W	102W
1/8 power @ 4 Ohm	1514W	1532W

24 CH Version		
Power Factor	@230V	@110V
amps power off	0.44	0.83
idle	0.58	0.93
1/8 power @ 4 Ohm	0.94	0.99

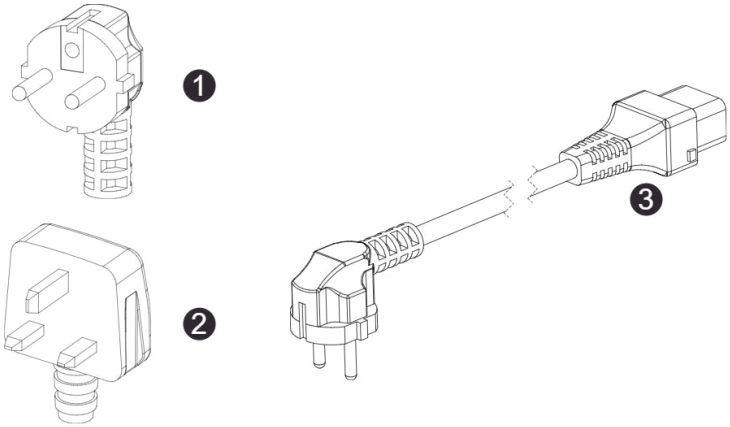
Consumption / current draw	@230V	@110V
amps power off	36W	36W
idle	83W	84W
1/8 power @ 4 Ohm	1145W	1150W

The devices contains an internal fuse see: FUSES

The Amplifier comes with a power cord according to the planned place of use.

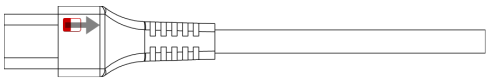
There are two separate power supplies in the amplifier. They can be used individually or together to increase power output. If only half power is needed, they can be seen as completely redundant.

2.4.2. AVAILABLE POWER CORDS



NR	DESRIPTION
1	3-pin Schuko CEE 7/7
2	3-pin GB BS 1363A
3	IEC-LOCK C19

To release the cable from the amplifier, the red Button has to be pulled to the back.



2.4.3. ETHERNET / DANTE

There are two differnet network devices inside the amplifier, the Control Module and the Dante® Module. There are three different network modes that determine which device can be reached at which network port.

Do not connect both Port to the same Switch if no different VLANs are configured.

NETWORK MODES

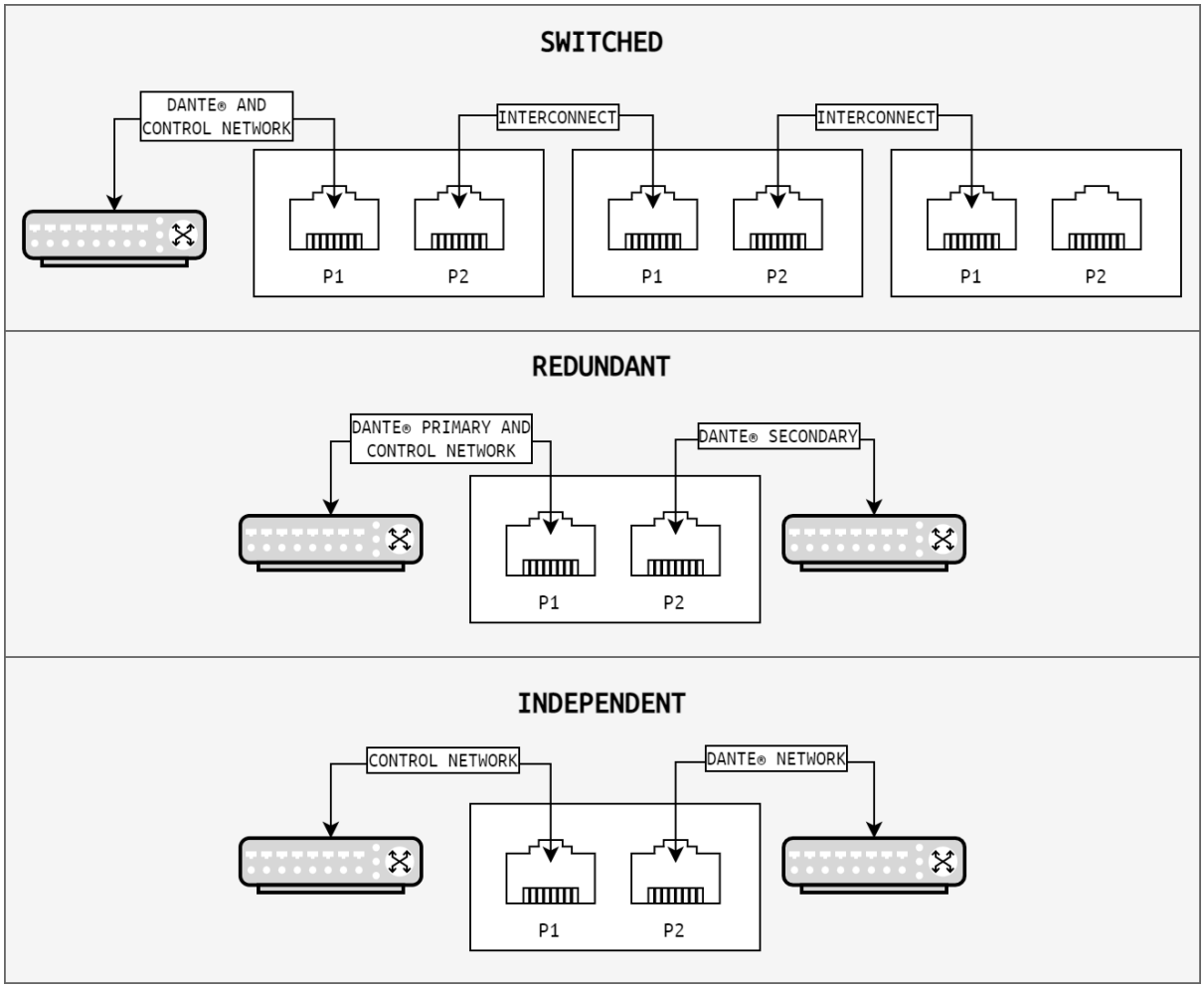


Figure 6. NETWORK MODES

To **change** the network modes, use the **DEVICE** page DANTE SETTINGS.

Alternatively, the **Dante Contoller** Software can be used. (DOWNLOAD HERE)

Open the **Dante Controller** and go to Device View:

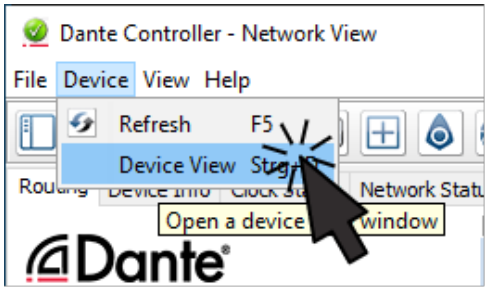


Figure 7. Dante Controller

The Device View Popup appears:

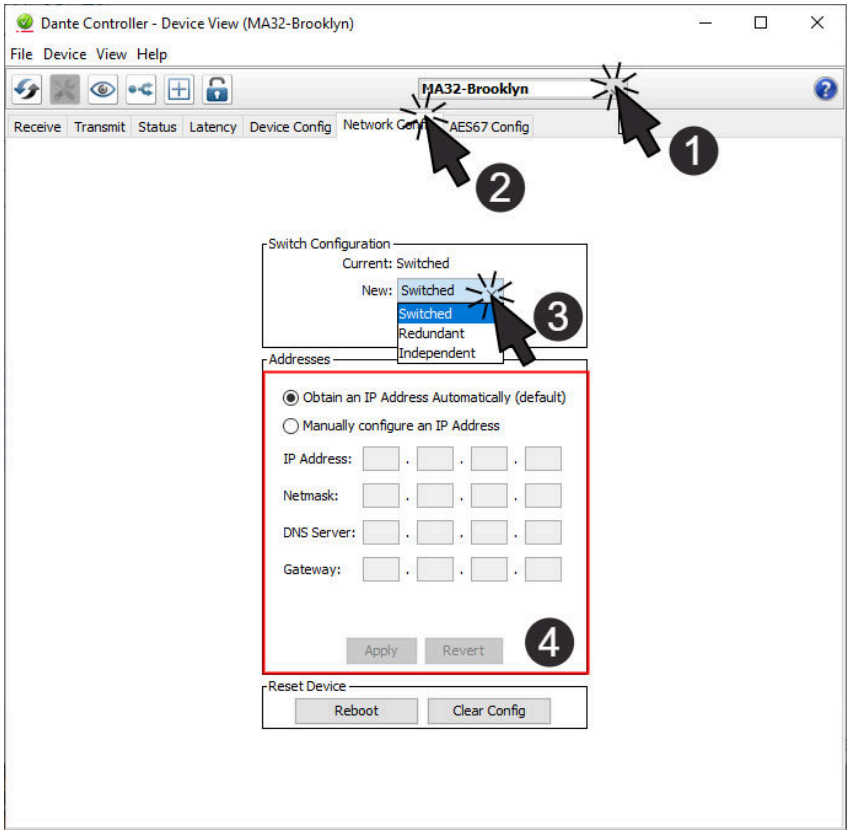


Figure 8. Dante Device View

Select the device in the Dropdown ❶ and change to tab **Network Config** ❷. The currently selected network mode appears in the **Switch Configuration** box. By selecting the new Mode ❸, a Popup opens that asked, whether you're really sure to do that. After clicking **YES**, the Brooklyn Module inside the Amplifier restart, so it can take a few seconds till the Dante Device is back

online.

DANTE IP SETTINGS

To control and change IP Settings of DANTE, use the **DEVICE** page DANTE SETTINGS, or use ❹ in Dante Device View.

CONTROL IP SETTINGS

There are several ways to change the **IP** of an Amplifier. See DISPLAY MENU to change on Display Menu, DEVICE to change on Website or use the external software **IDFM** (see IP SETTINGS).

There are three different IP Types available:

Table 3. IP TYPES

TYPE	DESCRIPTION
static	set IP, SUBNET and GATEWAY manually
dhcp	system tries to get a DHCP release, there is also an auto ip fallback, if no lease available
auto-ip	force zeroconf IP, device will get an address with a 169.254/16 prefix (that is, 169.254.xxx.xxx)

HOSTNAME

With mDNS the device is also available with its hostname. With hostname **AMP1** the local name is **AMP1.local**. The Name can be used to call every network service, like a webbrowser, **http://AMP1.local**.

There are several ways to change the **Hostname** of an Amplifier. See DEVICE to change on Website or use the external software **IDFM**.

NETWORK SERVICES

- Full remote control via the website hosted on the device (see WEBSITE)
- REST-API, JSON based web service for integration in media control systems (see RESTful API)
- mDNS name resolution and servicediscovery (INFO HERE)
- syslog integration to send notifications to external syslog server (INFO HERE)

2.4.4. MADI (AES10) / AES3

MADI (**M**ultichannel **A**udio **D**igital **I**nterface) or **AES10** is a standard that defines electrical characteristics and the data format of an interface that carries multiple channels of digital audio. There are two Coaxial Inputs available, which also can be used as AES3 Inputs, one optical input and an optical output. On every Input the device supports **44.1kHz / 48kHz** with **56 / 64** channels and **88.2 kHz / 96kHz** with **28 / 32** channels.

MADI Optical

To use the Optical MADI interface, a **1300nm multimode** cable with **SC** connectors is required.

MADI Coaxial / AES3

The two BNC (75 Ohm) jacks are multifunctional inputs and can be used as AES10 MADI or AES3 interface.

To use the BNC input for AES3, a 110 to 75 Ohm impedance transformer like (NADITBNC-F) or (NADITBNC-FX) is required.

Every AES3 input has an asynchronous samplerate converter enabled which can hanlde samplerates from 32kHz - 192kHz.

2.5. AMP OUTPUT

To connect speakers, use 2-Pol Terminal Blocks.

Output Power (EIAJ Test Standard 1kHz 1% THD)	4Ω		8Ω	8Ω Bridge-Mode
	280W		140W	500W
Max output Voltage	52 V_{peak}			104 V_{peak}
Max output Current Limited	20 A_{peak}			
Emergency Shutdown Current	31 A_{peak}			
DC Offset	<25mV			
Frequency response	10Hz-20kHz / 4-8Ω: +0.5 -0.5dB			
S/N typ	105dBA			
Analog Gain	Software Adjustable, 0dBFS on any Input Interface: $20V_p$ - $60V_p$ (default: $60V_p$)			
THD+N @ 4Ω	1W	10W	280W	300W
	< 0.05%	< 0.05%	1%	2%
SMPTE IMD	< 0.1% @ 1/8 Power 4Ω			
CCIF IMD	< 0.1% @ 1/8 Power 4Ω			
Output impedance	typ 10 mΩ			
Crosstalk	channel enabled		channel disabled	
	typ < 75dB	typ 90dB (distant channels)	typ < 100dB	
Protection	Overtemperature, DC and Overcurrent			

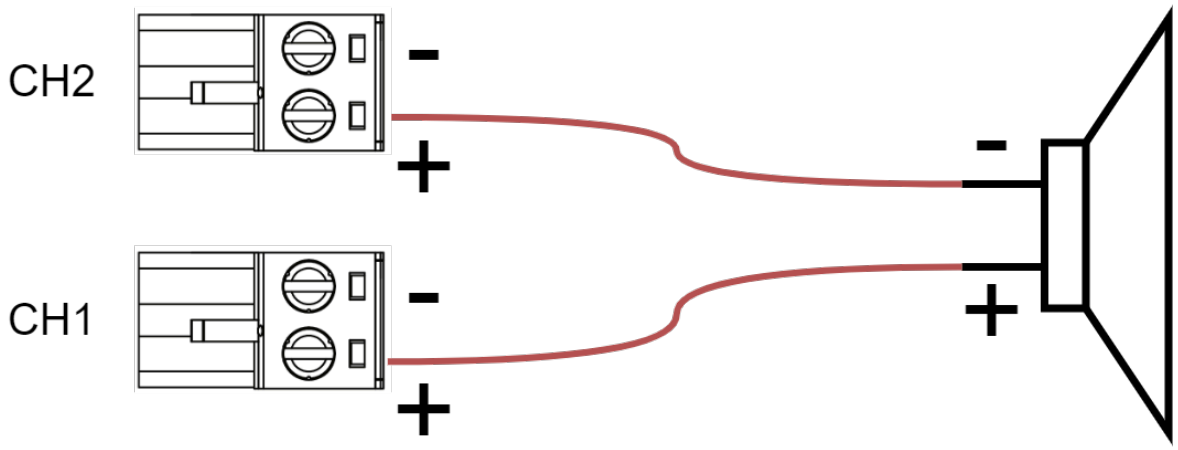
2.5.1. Bridge Mode

In bridge mode, only adjacent channels can be used together, like 1/2, 3/4, 5/6, ... 31/32. The **+** Pin of both channels has to be connected to the speaker, the **-** stays unused in this mode.

CH1 + @ "speaker +"

CH2 + @ "speaker -"

Speaker Wiring in Bridge Mode



2.5.2. Overcurrent behaviour

1. The integrated software **CURRENT** Limiter (LIMIT) will always try to limit the output current to it's Max output Current Limited as specified in the datasheet. This will also protect the amplifier output stage when a hard short between the output terminals is present. The amp will drive ~0V and its maximum current specified.
2. If the software limiting will fail, an additional hardware comparator will shut down the amplifier output stage if Emergency Shutdown Current is exceeded. The channel has to be manually power cycled to recover from that state.

2.6. POWER DISTRIBUTION

- The power supply can deliver 5000W continuous power. To ensure a stable operation in overload situations, the MAXX/HP² involves an overall power limiter. With an attack time of 100ms and a release of 3s the limiter softly reduces the gain of all channels simultaneously not to exceed the maximum available power. At 110V lines, a derating of 20% must be considered.
- The pulse power of 32 * 240W is buffered through 65.000uF of capacitors and is certainly available as a short burst only.
- An amplifier efficency of 85% can be expected.
- All amplifiers are sourced by one powersupply.
- Assuming a evenly distribution of load between all channels, the MAXX/HP² is able to source 140W continous sine power per channel. At 110V line it is after all 120W.

Chapter 3. IDFM (FIRMWARE UPDATE AND IP CONTROL)

The IDFM (**I**nnosonix **D**iscovery and **F**irmware **M**anager) is available for Windows 10, MAC OSX and Linux. Have a look at our Download Area

It is desinged to discover MAXX Devices across subnets and across network modes. It also handles Firmware updates of MAXX Devices.

3.1. DISCOVERY

The Discovery process starts after opening the IDFM Tool. All available Devices will appear in the list view.

! Be sure the firewall allows TCP and UDP connections.

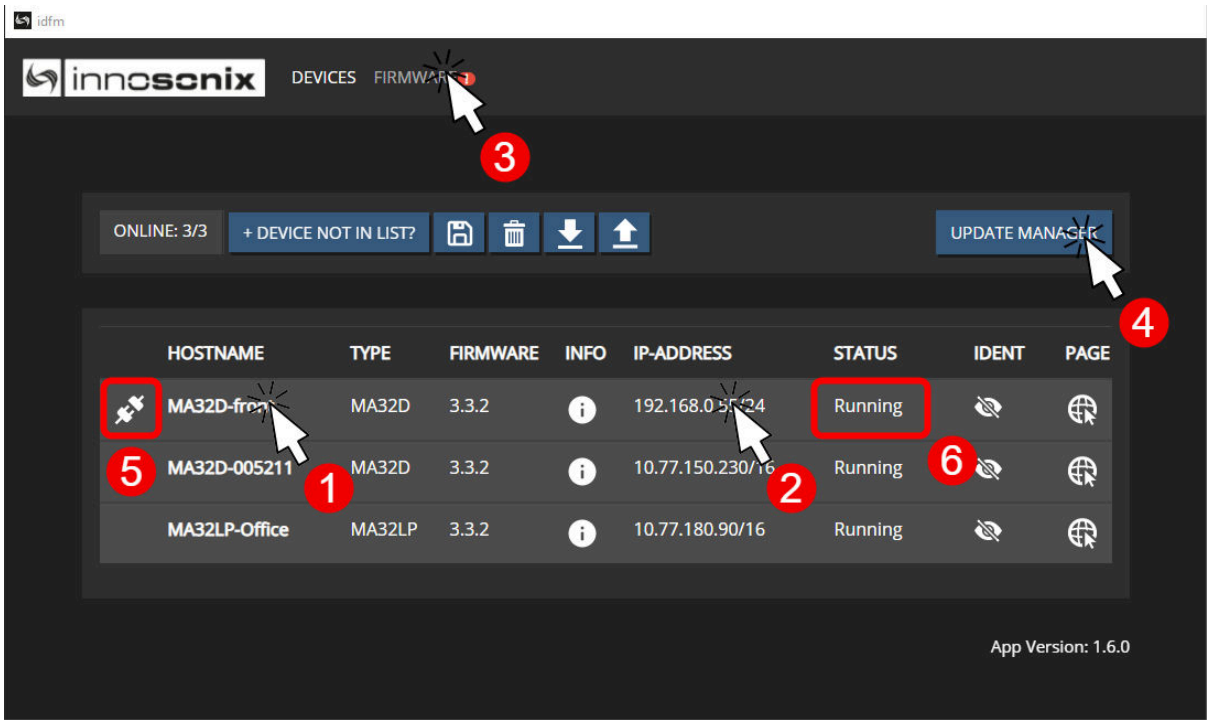


Figure 9. IDFM Discovery

NR	DESCRIPTION	REFERENCE

1	Change Hostname	
2	Change IP Settings	IP SETTINGS
3	Download / Import Firmware Files	see FIRMWARE
4	Update Devices	UPDATE
5	Device not in same Subnet cannot be updated	
6	Actual Device Status (Update Status)	

3.2. IP SETTINGS

After clicking on the IP Address in IDFM Discovery view, following popup appears to change IP Settings.

IP Settings are described here Control IP

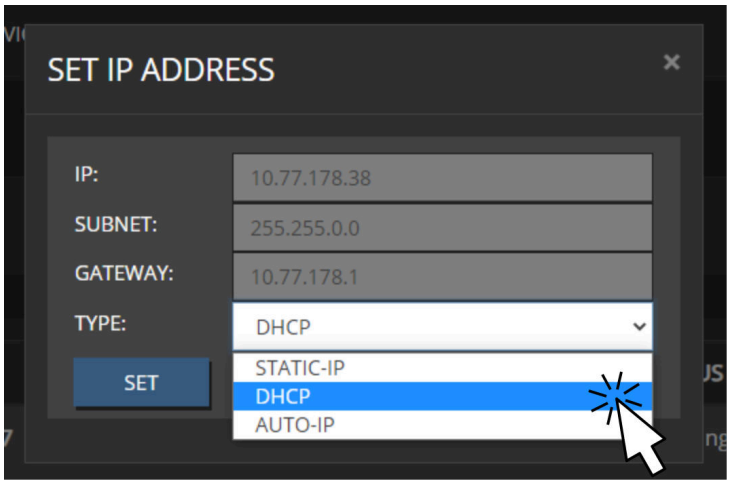


Figure 10. IDFM IP Settings

3.3. FIRMWARE STORAGE

To update the firmware of a MAXX Device, the correct Firmware must be available in the firmware storage.

If there is no Internet connection available, the newest firmware cannot be loaded from our server ❷. With ❶ a firmware image files can be uploaded manually.

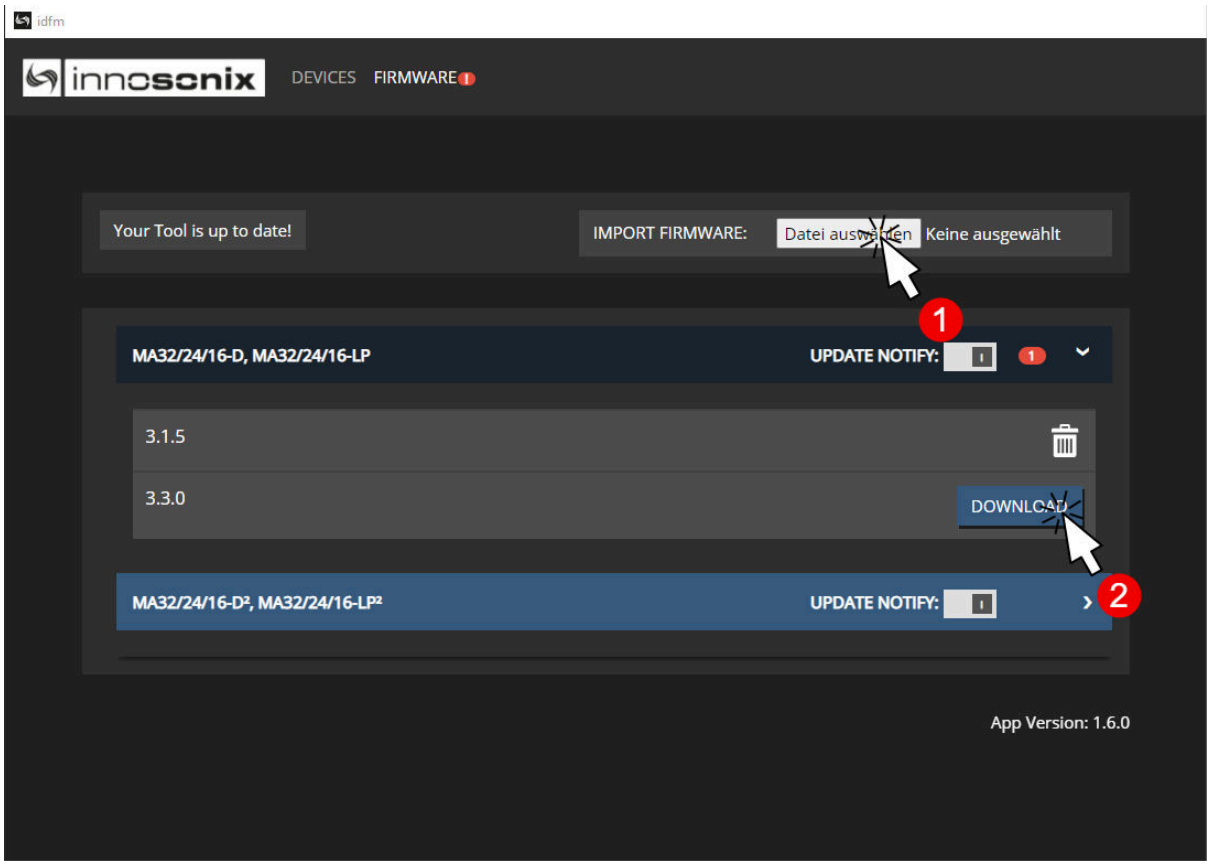


Figure 11. IDFM Firmware storage

3.4. FIRMWARE UPDATE

After loading a correct firmware file to the FIRMWARE STORAGE, the firmware can be selected in the firmware update popup. If no Firmware is selected, the device will be ignored. After confirming the update, the update status can be seen at ❻ on IDFM Discovery.

! After firmware Update completed, the Device restarts autimatically.

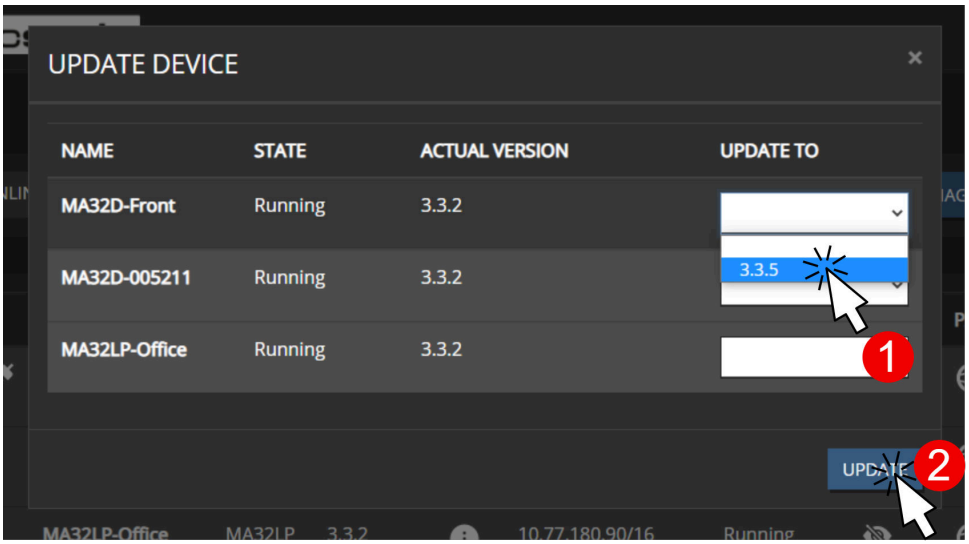


Figure 12. IDFM Firmware update

Chapter 4. DSP (internal)

A DSP is a digital signal processing chain inside the FPGA that calculates the volume control, filtering and limiting parameters on the selected Input Source. There are as many DSP channels as amplifier outputs on the MAXX device. DSPs are "hardwired" to the corresponding amplifier, e.g. DSP channel 1 supplies an amplifier that is wired to CH1 Jack on the rear panel.

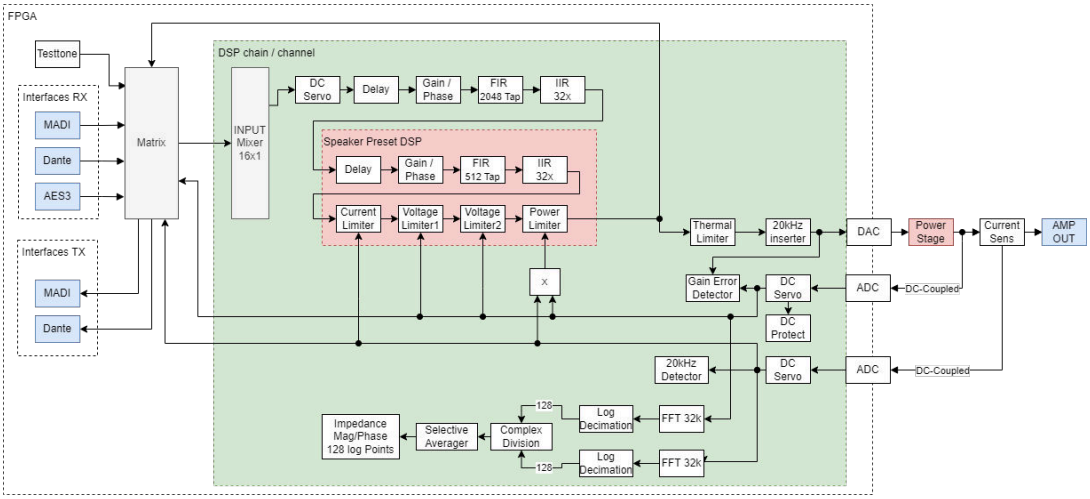


Figure 13. DSP Block Image

DSP Features

Architecture	FPGA based 32-bit fixed point
Inputs	16 x input matrix per channel (DANTE / AES67 via Dante Module / AES3 / MADI)
	sine, white- pink- brown-noise
Level Control	Mute, Volume, Phase
Filter per channel	32 x EQ / Highpass / Lowpass
Filter types	bell, notch, highshelf, lowshelf, allpass 1th / 2nd order
High- Lowpass types	6 - 48dB/Oct, Bessel, Butterworth, Linkwitz/Riley, Variable Q
FIR Filter	2048 Tabs, ASCII file import
Fraction Delay	48000 Samples / 330m / 1000ms (resolution 0.001 units) per channel
CurrentLimiter	Threshold [Ap]
Voltagelimiter	2 x Threshold [Vp], Attack, Release
Powerlimiter	Threshold [W], Attack, Release
Speakerdetection	20kHz Pilot Tone generating with Volume, Threshold, Debounce

	Latency @ 44k1 / 48kHz to Analog
digital input / without ASRC on AES3	41 samples, (24 samples DSP + 17 samples DAC)
Input	**Latency @ 44.1kHz to Analog**
DANTE / MADI	0.930 ms
AES3@44.1kHz	2.028 ms
AES3@48kHz	1.996 ms
AES3@88.2kHz	1.811 ms
AES3@96kHz	1.797 ms
Input	**Latency @ 48kHz to Analog**
DANTE / MADI	0.854 ms
AES3@44.1kHz	2.021 ms
AES3@48kHz	1.854 ms
AES3@88.2kHz	1.690 ms
AES3@96kHz	1.6875 ms
Input	**Latency @ 88.2kHz to Analog**
DANTE / MADI to Analog	0.896 ms
AES3@44.1kHz	2.008 ms
AES3@48kHz	1.916 ms
AES3@88.2kHz	1.432 ms
AES3@96kHz	1.416 ms
Input	**Latency @ 96kHz to Analog**
DANTE / MADI	0.823 ms
AES3@44.1kHz	1.952 ms
AES3@48kHz	1.856 ms
AES3@88.2kHz	1.365 ms
AES3@96kHz	1.330 ms

Chapter 5. Frontpannel

5.1. DISPLAY

5.1.1. OVERVIEW

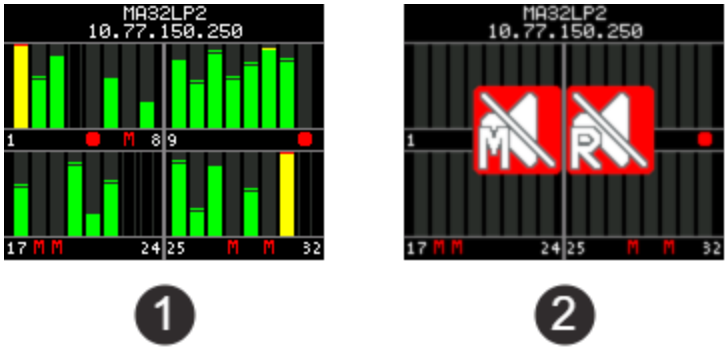


Figure 14. OVERVIEW EXAMPLES

The Overview Page appears at startup, and after a 30 seconds without any user interaction. Every channel has its own Levelmeter ranging from **-60dBFS** to **0dBFS** with **PEAK** as a bar and **HOLD** as a horizontal line. The Overview ❶ shows **CHANNEL MUTE** State (CH 7, 18, 19, 18, 30), if the amplifier channel is **disabled** (CH 4, 23, 24, 32) or the amp channel has an **error** (CH 5, 16).

If **Master Mute** or **Remote Mute** is active, the crossed loudspeaker with the letter **M** or **R** appears ❷.

5.1.2. DISPLAY DEVICE LOCK

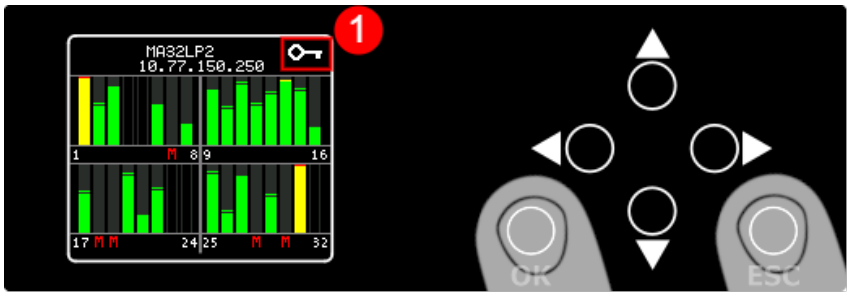


Figure 15. DEVICE LOCK

To enable and disable Display Device Lock, hold **OK** and **ESC** for about 2 seconds. The Device Lock prevents setting changes, like IP. The small Key ❶ shows activated Device Lock on every Page.

5.1.3. DISPLAY MENU

Multiple menu pages can be accessed by pressing the **LEFT** or **RIGHT** button and cycling through the menu selection. Use the **UP**, **DOWN**, **ENTER**, **ESC** button to modify settings.

Mute

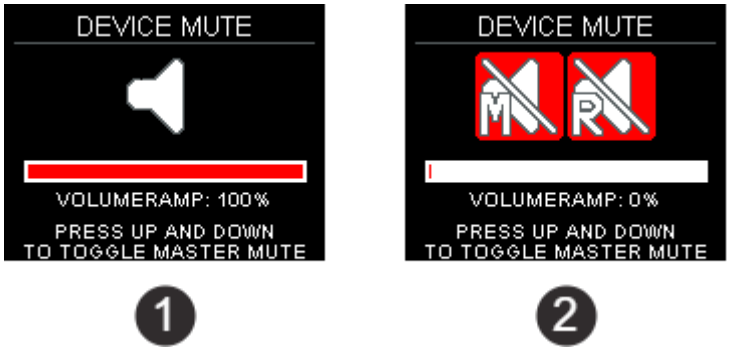
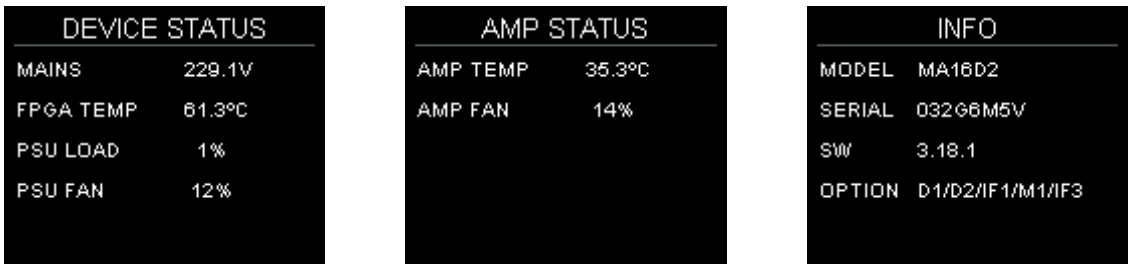


Figure 16. MUTE PAGE EXAMPLES

The Mute Page appears, if Master or Remote Mute changed to Active. If **Master Mute** or **Remote Mute** is active, the crossed loudspeaker with **M** or **R** appears ❷. To toggle **Master Mute**, press **UP** and **DOWN** simultaneously.

The **Volume Ramp Bar** ❶ only appears if the Volume Ramp is activated in Device Settings. It shows the actual state during ramp-up. To cancel ramping, press **UP** and **DOWN**. This will activate Master Mute.

General

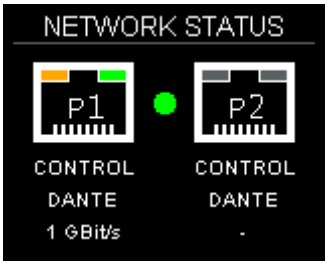


Network

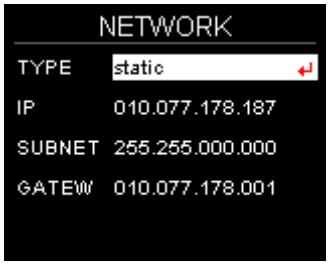
The **NETWORK STATUS** page represents the two Ethernet jacks on the backside, including the VLAN configuration LED. (see DANTE SETTINGS)

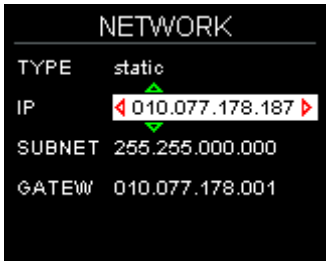
Each Port functionality is listed below the jack symbols based on the currently active VLAN configuration.

The two network jack LEDs indicate link, speed as well as traffic.



On the **NETWORK** page, the current IP address information can be shown/modified.

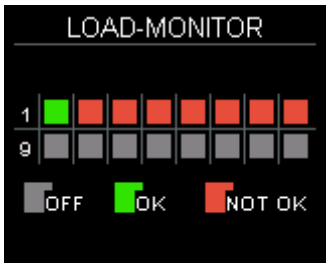




Channel

Those pages are dedicated to showing information for each channel. The LED, next to each channel number, will indicate an **OK** or **ERROR** state. (see AMP STATUS)

On the **LOAD-MONITOR** page, the output of the configured 20kHz pilot tone detection when enabled.



5.2. POWER LED

Table 4. POWER LED states

COLOR	DESRIPTION
GREEN	everything is ok
ORANGE	system is booting up
RED	one or more channels are in error state
BLUE	Mains dropout, by pulling the mains cable or press the power switch

Chapter 6. WEBSITE

To open the control page, use a regular web browser like Chrome, Firefox, Safari and enter the IP address or hostname into the address line.

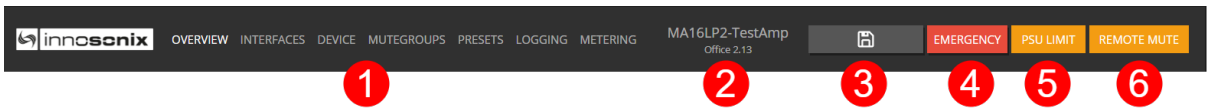
Like http://192.168.0.100 or by using the hostname http://AMP1.local

The website is the main User Interface to control every setting and get status informations of the amplifier.

Some DSP function and inputs are optional and depend on the software and hardware options of the device.

In single edit, value fields and buttons with blue background indicate the value is changed but not currently set to the device. In multi-edit it also indicates different values on the selected channel.

6.1. HEADER



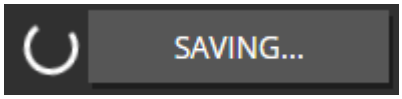
NR	DESCRIPTION	REFERENCE
1	navigate through pages	PAGES
2	click to change hostname or location	
3	save button / autosave status	SAVE INTERNAL STORAGE
4	overall amp status	AMP STATUS
5	PSU Limit indication	PSU LIMIT
6	Remote Mute active indication	DEVICE

6.1.1. PAGES

Table 5. PAGES

IDENTIFIER	DESCRIPTION	REFERENCE
OVERVIEW	status and settings of amp channel	OVERVIEW
INTERFACES	device interface status and config	INTERFACES
DEVICE	device specific settings	DEVICE
MUTEGROUPS	mutegroup settings	MUTEGROUPS
PRESETS	device/channel preset edit/save/call/store	PRESETS
LOGGING	syslog with syslog server settings	LOGGING
METERING	show input / output level and measured voltage / current / power	METERING

6.1.2. SAVE INTERNAL STORAGE



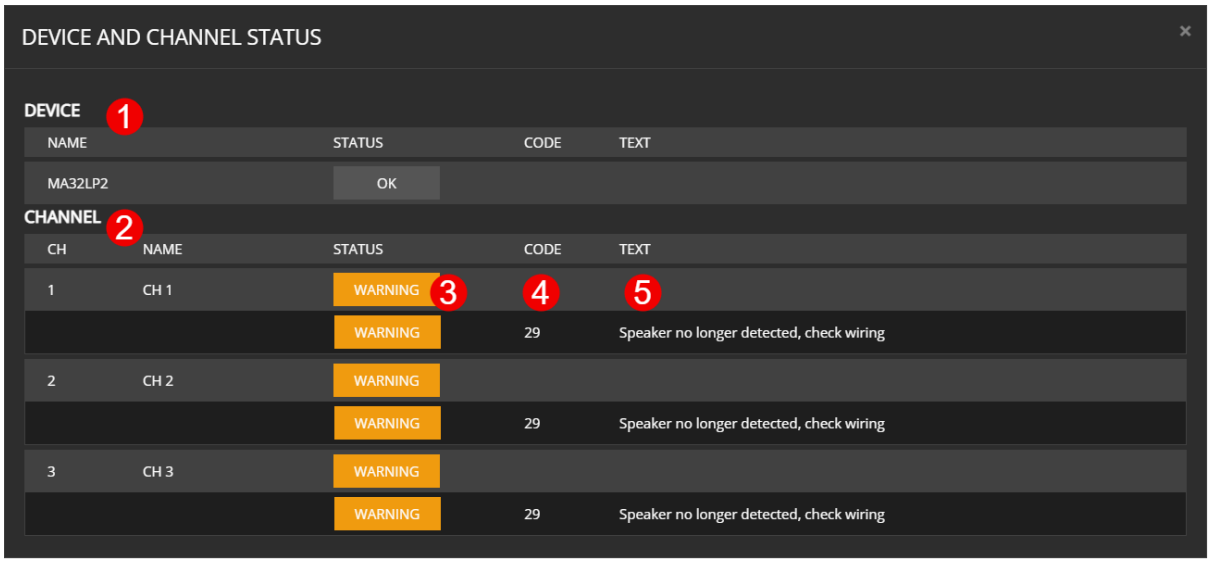
Settings changes will be automatically saved after 10 seconds. An immediately save of changed settings can be triggered by clicking on the **AUTOSAVE IN:** button.

6.1.3. PSU LIMIT

If maximum power of the PSU is reached, the amplifier reducts the output with an extra limiter, to avoid shutting down the amplifier. The indicator LED (WEBPAGE HEADER ❺) starts blinking, if reduction is active. To see the actual reduction value and load, see DEVICE ❹

6.1.4. AMP STATUS

The Amp Status shows all currently applicable errors. To see the chronological sequence of errors see LOGGING.



NR	DESCRIPTION	REFERENCE
1	device errors	
2	channel errors	
3	severity level	SEVERITY LEVEL
4	error code	ERROR CODES
5	error text	

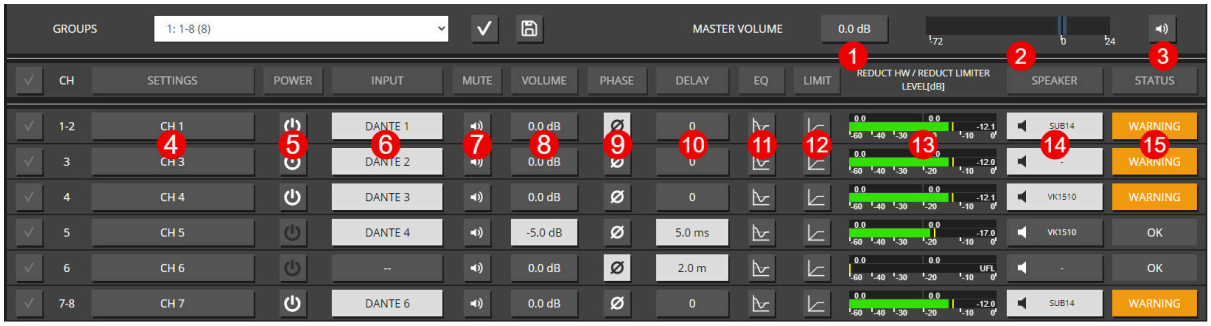
6.2. FOOTER

On the footer, a temperature overview can be seen, which shows the FPGA temperature as well as the maximum temperature on all amp modules.

If the **AMP** temperature exceeds 80°C, a showdown of all amplifier modules is performed to prevent damage. The amplifier automatically enables all amplifiers again, if the temperature decreases.

6.3. OVERVIEW

All channel setting can be done to single and multi-channel (see SELECTION AND GROUPING for multi-channel selection details).



NR	DESCRIPTION	REFERENCE
1	click to set master volume	
2	change master volume by moving the slider	
3	click to toggle master mute	
4	click opens channel settings modal	CHANNEL SETTINGS
5	click to toggle amplifier power	POWER
6	click opens input patching modal (∑ indicates, that multiple inputs are set)	INPUT
7	click to toggle channel mute (blinks if muted by mutegroup MUTEGROUPS)	MUTE
8	click opens channel volume modal	CHANNEL VOLUME
9	click to toggle channel phase (reverses phase)	PHASE
10	click opens channel delay modal	DELAY
11	click opens channel eq modal	PEQ

12	click opens channel limiter modal	LIMIT
13	sample synchonous channel level after processing, hw limiter (sum of PSU Limit and Thermo Limit) and biggest limiter reduction	
14	click opens channel speaker settings modal	SPEAKER SETTINGS
15	if error occurs, click opens channel status modal	error shown like AMP STATUS

If the ANALYZER has detected a valid speaker impedance, the **minimum** impedance in ohms of the whole spectrum range is indicted in the small speaker icon.

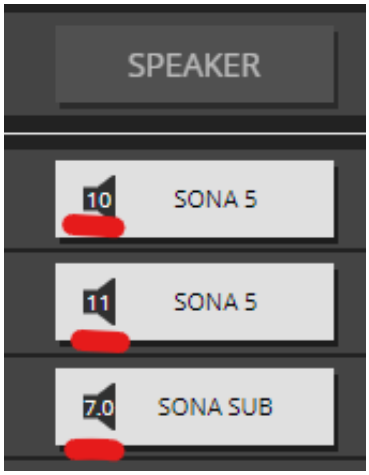


Figure 17. Speaker minimum impedance

6.4. SELECTION AND GROUPING

Multiple channels can be selected by clicking on ✓. This feature enables the "multi-channel edit" functionality indicated by the active headline buttons (SETTINGS, POWER, ...). The headline buttons open the corresponding modal.

> The saved selection groups will be used as mute groups MUTEGROUPS and can be selected in the channel edit modal header MODAL HEADER.

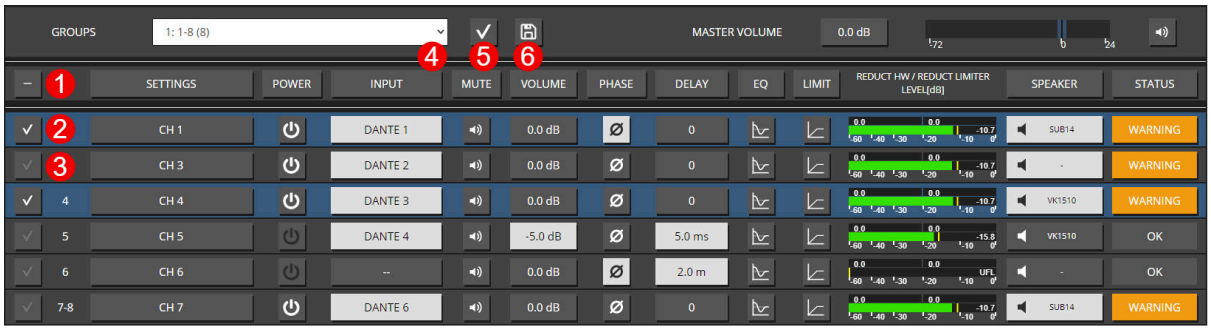


Figure 18. WEBPAGE GROUPING

NR	DESCRIPTION
1	Select/deselect all channels.
2	Channel is selected. Indicators are the white hook and the blue background of the channel line.
3	Channel is not selected.
4	Channel Groups drop-down list.
5	Loads the selected group to the selection.
6	Saves active selection to the selected group.

6.5. MODAL HEADER

The MODAL HEADER allows switching through a channel selection, controlling the volume/mute, showing the OUTPUT LEVEL meter for the selected channel and navigating through all settings.

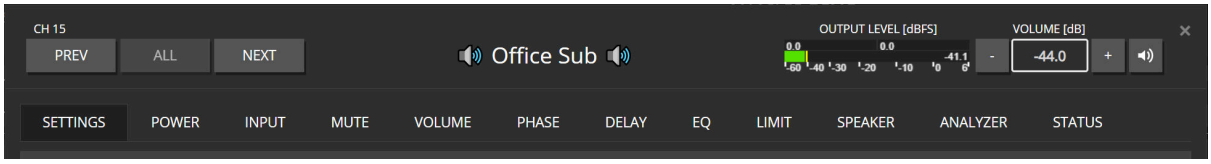


Figure 19. MODAL HEADER single-channel

When opening the modal with multiple channel selected, the "multi-edit" functionality is enabled and the header is reduced to only show common settings. The "PREV / ALL / NEXT" can be used to cycling through the selected channels and edit each channel independently, resulting in the MODAL HEADER single-channel view again.

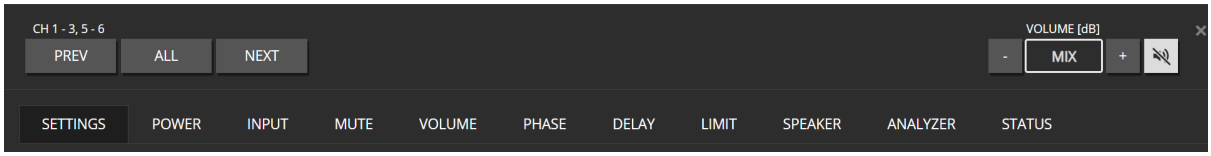
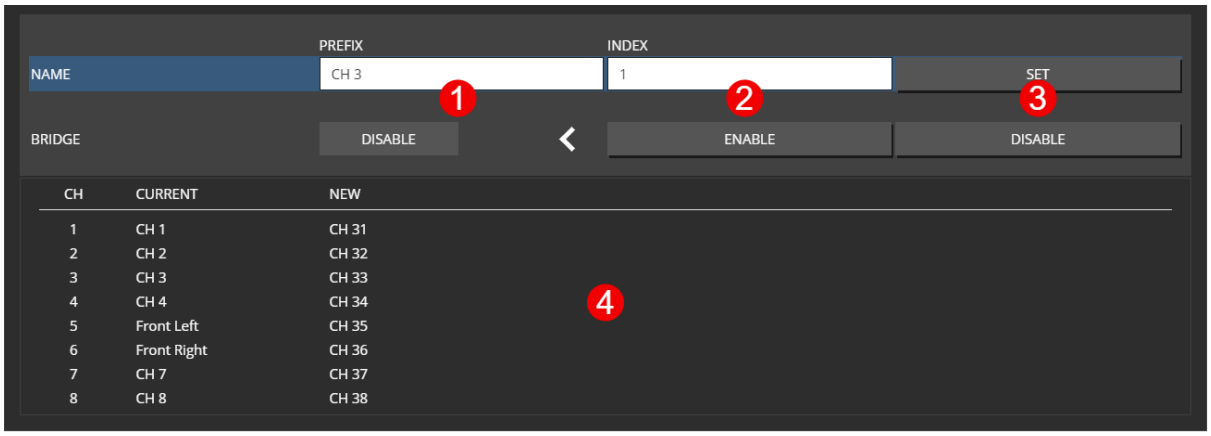


Figure 20. MODAL HEADER multi-channel

The OUTPUT LEVEL meter and VOLUME / MUTE give you better control over your system while browsing the settings tabs, since you can always modify the volume and mute in case anything is unexpected.

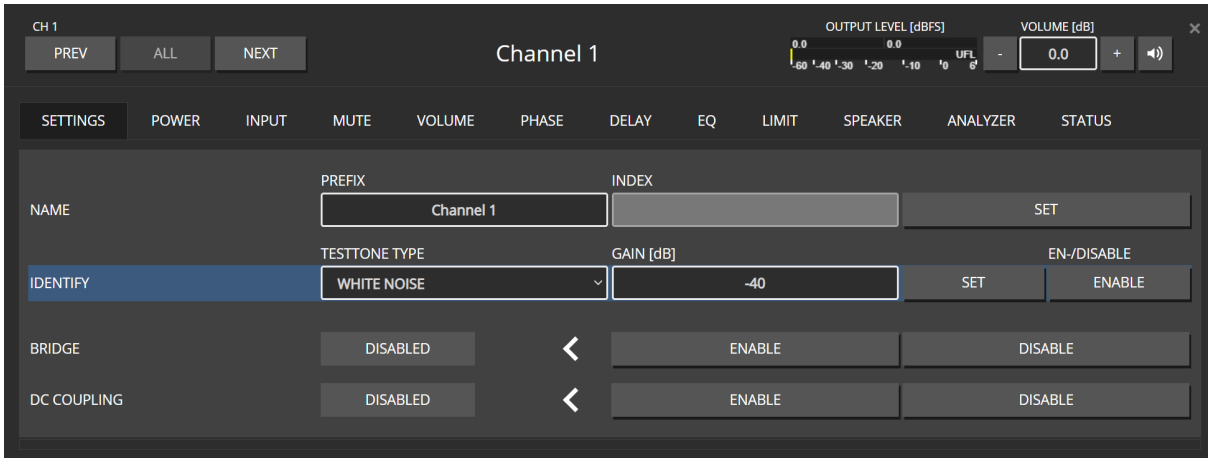
6.6. CHANNEL SETTINGS

6.6.1. NAME



NR	DESCRIPTION
1	Set channel prefix, which will be concatenated with the "INDEX" as final channel name.
2	Set an optional index which is incremented for each selected channel. (only available in multi-edit)
3	Execute changes.
4	Preview of channel names.

6.6.2. IDENTIFY

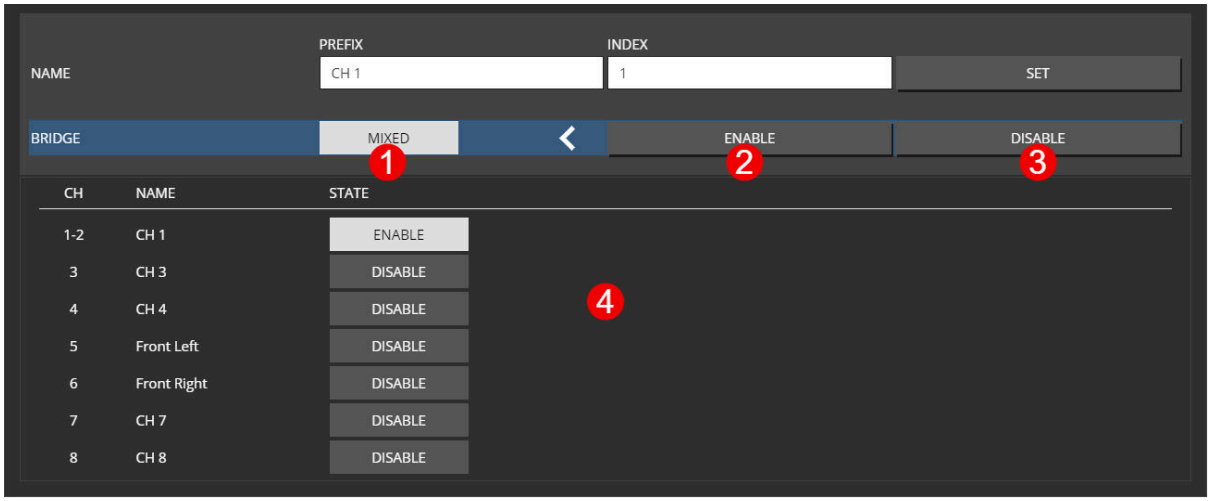


The IDENTIFY can be used to identify the connected speaker by applying a TESTTONE to the amplifiier output. Once the IDENTIFY is enabled the PREV/NEXT button can be used to cycle through a slection or all channels to identify and name multiple speakers.

As a safety mechanism, the tone will only play as long as the channel pop-up stays open. Closing the pop-up or a connection loss will set the channel to its previous state.

The channel must be powered up to hear the IDENTIFIER tone.

6.6.3. BRIDGE MODE



NR	DESCRIPTION
1	Indicates summarized state of selected channels.
2	Enable bridge mode for selected channels.
3	Disable bridge mode for selected channels.
4	States for all selected channels.

Only adjacent channel pairs can be set to bridge mode, channel 1/2 or 3/4 ...

Enabling the **BRIDGE MODE** for a channel pair will clear all settings of the EVEN channel.

6.6.4. DC COUPLING

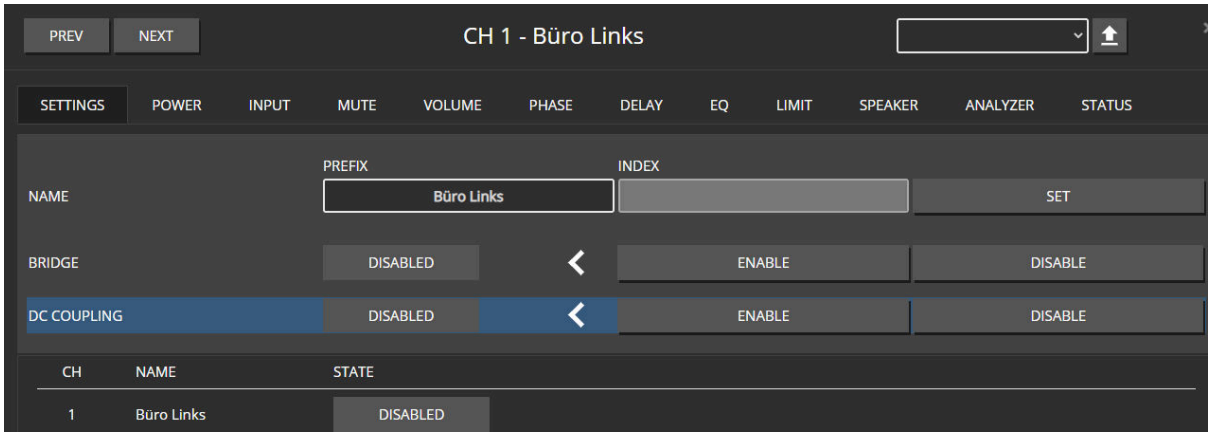


Figure 21. DC COUPLING

Some scientific measurements require the amplifier to output a common DC voltage to bias the speaker coil. By enabling **DC COUPLING** in internal DC, servo filters are frozen and allow passing DC in the input signal to the amplifier output stage.

Use this setting with caution, since it can easily burn your speaker chassis if not used correctly.

The integrated DC protection is still enabled, which will shut down the amplifier if DC reaches ~-3dB of the amplifier rail voltage. This is used as a safety mechanism in case of transistor damage.

6.6.5. POWER

Power-off a channel which will stop the class-d amp from switching to save power.

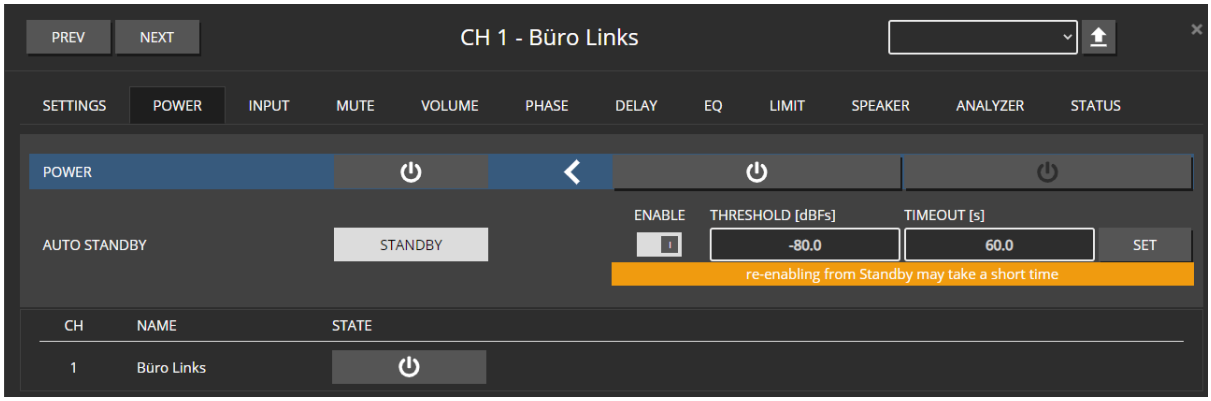


Figure 22. POWER

6.6.6. AUTO STANDBY

This feature allows additional power savings by automatically powering down individual channels when no more input signal is present for a configurable amount of time.

> Power up, after detecting an input level, will require some milliseconds, so only use it if your application can tolerate this.

Table 6. AUTO STANDBY STATES

STATE	DESCRIPTION
PLAYING	Normal operation, when the channel is powered up.
TIMEOUT	The input Level is below the threshold. The timeout counter is counting up until it reaches the configured value.
STANDBY	Channel is powered down and waiting for reactivation by an input signal.

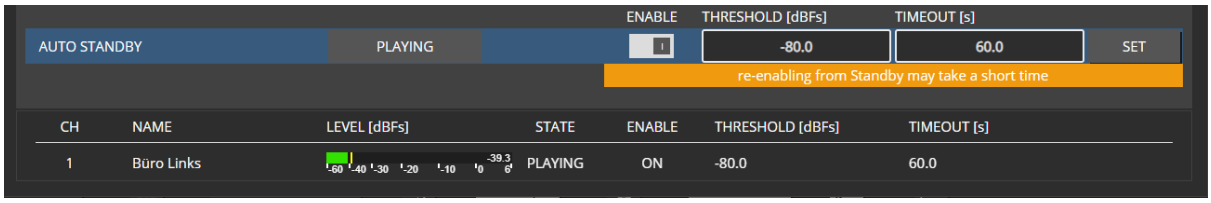


Figure 23. STANDBY PLAYING

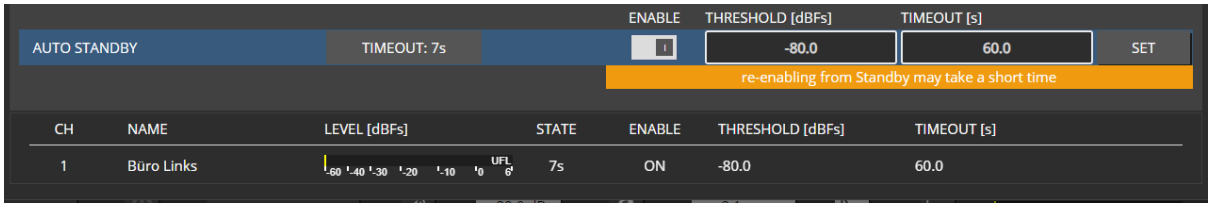


Figure 24. STANDBY TIMEOUT

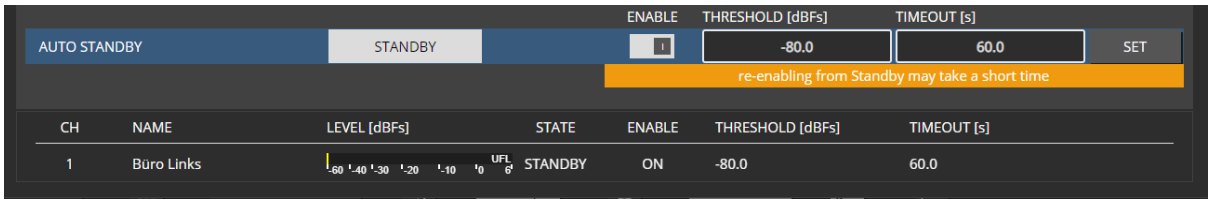


Figure 25. STANDBY

If any channel is in the **STANDBY** state, the **POWER** button on the overview page is fading back and forth.

Figure 26. STANDBY on OVERVIEW page

6.6.7. INPUT

Each DSP channel has its own 16x1 input mixer which allows a summation of up to 16 different sources with individual gains.

The INPUT view differs slightly in functionality and status views, depending on editing a single or multiple channels.

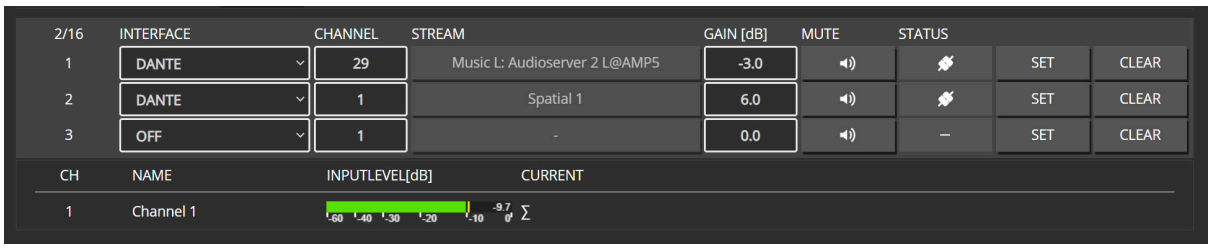


Figure 27. SINGLE CHANNEL

CONTROL	DESCRIPTION
INTERFACE	List of available interface to assign to this DSP channel. (depends on your hard- and software-options, so can slightly variating between devices)
CHANNEL	Different interface providing a different amount of channels (MADI 64, DANTE 32 / 64, AES3 2, etc)
STREAM	Additional information of the selected Interface and channel, as well as selection of STREAMs see STREAM
GAIN	Gain of this INPUT patch
MUTE	Mute this INPUT patch
STATUS	Either a indication of the interface lock (MADI, AES3, etc) or information about the stream status when using DANTE or STREAM as INTERFACE option

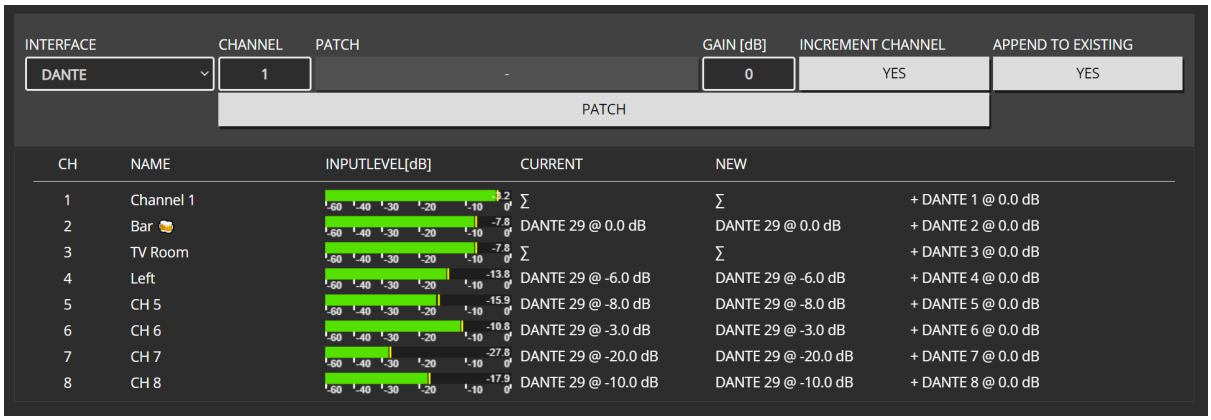


Figure 28. MULTI CHANNEL

When selecting multiple channels, it's possible to assign them channels from one INTERFACE

simultaneously.

CONTROL	DESCRIPTION
INCREMENT CHANNEL	If enabled, the INTERFACE CHANNELs is incremented per selected DSP channel. Like DANTE 1, 2, 3, 4, etc. Otherwise, only the selected INTERFACE CHANNEL is patched to all selected DSP channels
APPEND TO EXISTING	Use this to append the INTERFACE CHANNEL to existing patches, otherwise they will be removed

A preview of the selected combination of **INCREMENT CHANNEL** and **APPEND TO EXISTING** can be seen in the channel list below. **CURRENT** indicates the current patch, **NEW** the upcomming result.

∑ indicates, that multiple inputs are set.

STREAM

STREAMs are like all other INTERFACEs which could be used as INPUT to a DSP channel, except that they are based on Dante stream labels and not on interface numbers.

Like if you use DANTE 1 as input for a DSP channel, the corresponding Dante Stream has to be manually assigned via the Dante Controller.

With STREAM as INPUT, a corresponding Dante Stream in the Network could be assigned directly via its name. In that case, the amplifier automatically subscribes to that Dante stream on an unused Dante channel and uses that as INPUT to the channel.

Receiving a Dante Stream requires a stream label, which is the combination of "Transmit Channel" @ "Hostname".

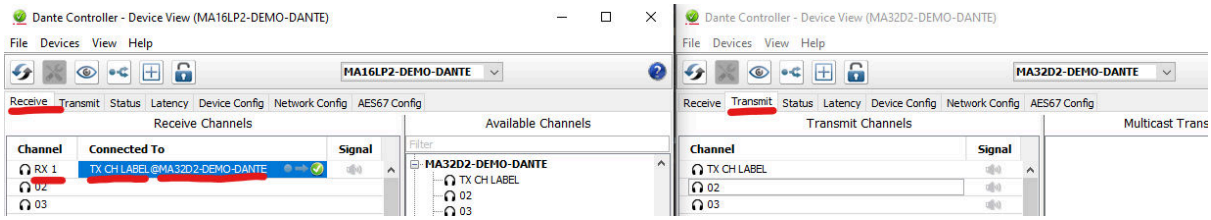
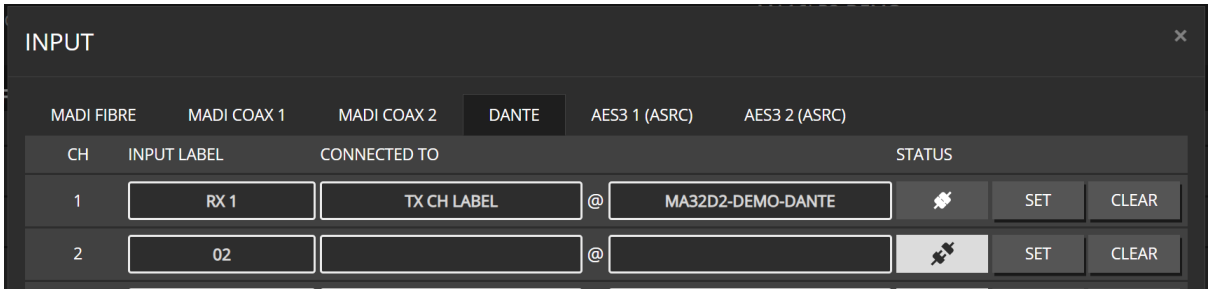


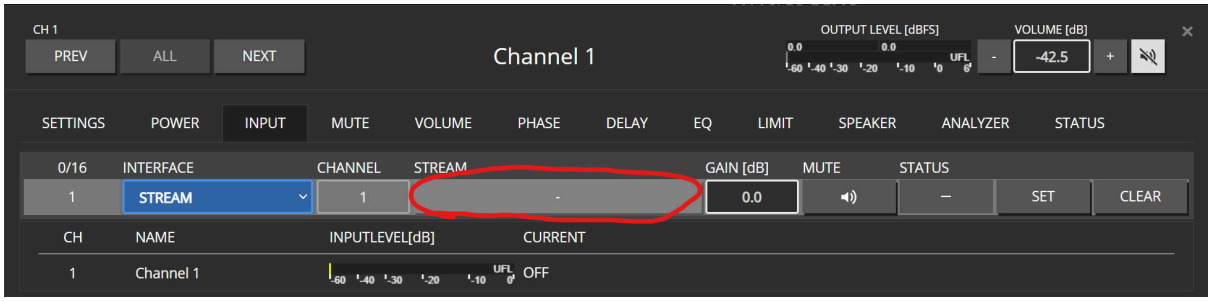
Figure 29. Dante Controller Streams

In the example above, **TX CH LABEL** is the name, and **MA32D2-DEMO-DANTE** is the hostname of the actaul Dante device. An optional name to the Recieve Channel could be assigned as well, in this case **RX 1**.

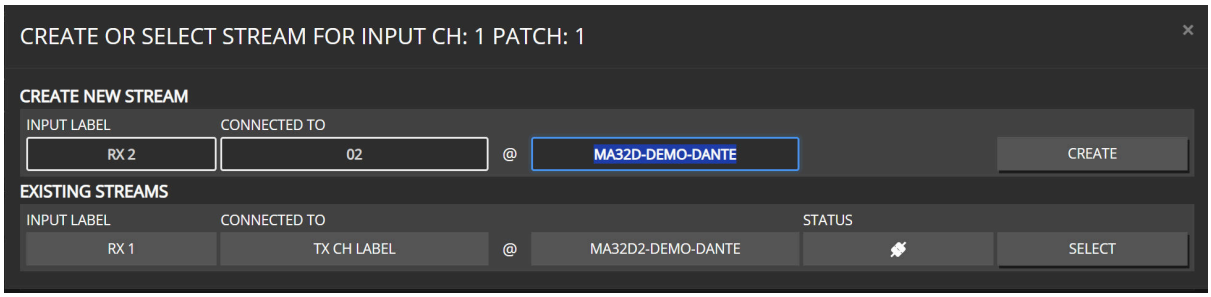
On the amplifier's control page, the same information can be shown in the INPUT section.



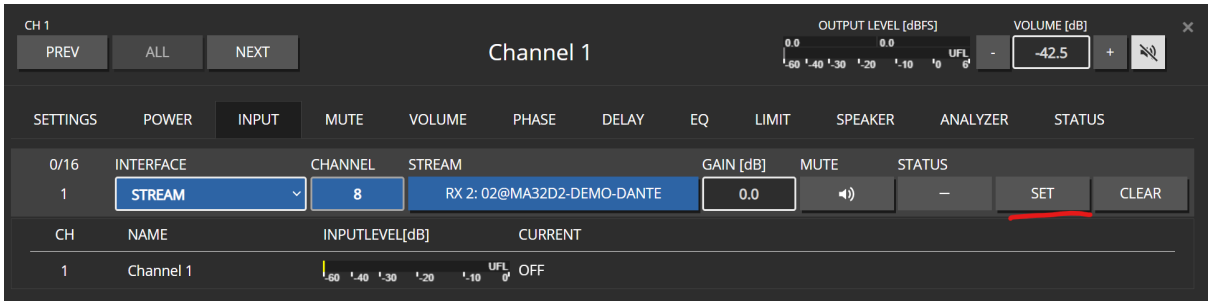
To assign a new STREAM as INPUT to a channel, click on the gray **STREAM** box marked in read below.



A new popup appears, where either a NEWS STREAM can be created, or any existing STREAM already patched via the Dante Controller can be selected.



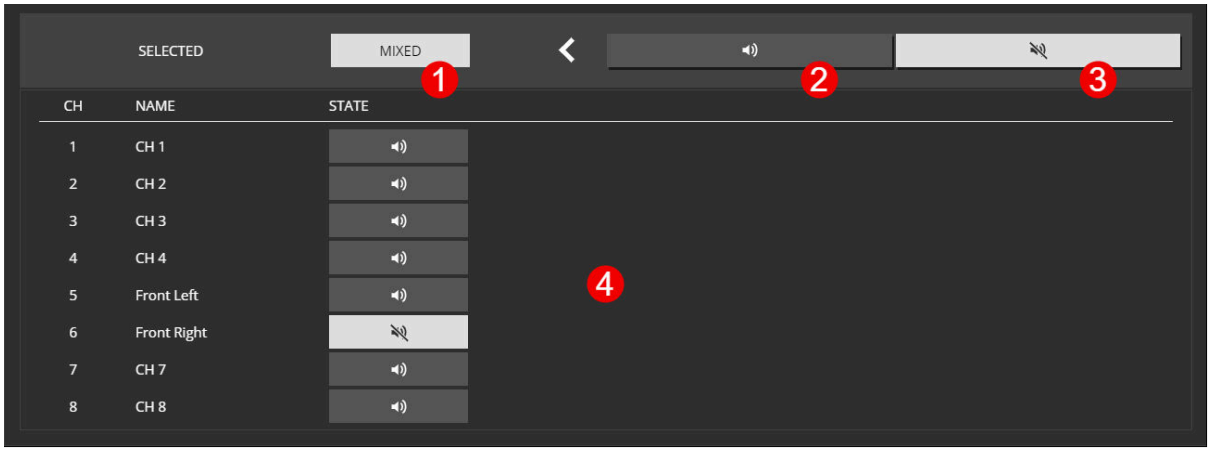
Once a STREAM is selected, the INTERFACE patch has to be finally confirmed via SET.



Once the Stream is set, a reload of the CHANNEL data is required to see the final STREAM status and assigned Dante Slot in the INPUT section. This can either be

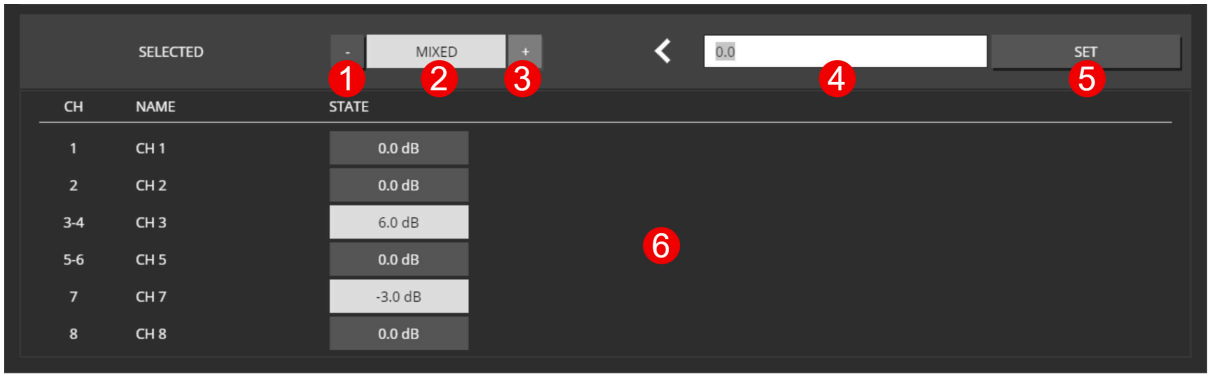
done, by closing the POPUP or press the NEXT then PREV channel button.

6.6.8. MUTE



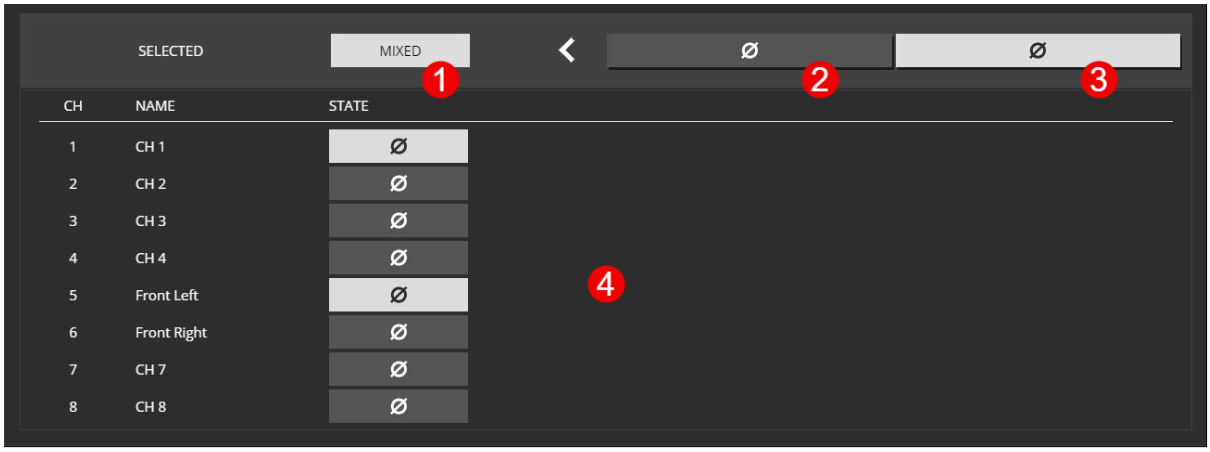
NR	DESCRIPTION
1	Indicates summarized state of selected channels.
2	Unmutes all selected channels.
3	Mutes all selected channels.
4	Shows states of all selected channels.

6.6.9. CHANNEL VOLUME



NR	DESCRIPTION
1	Decreases volume of selected channels by 1 dB.
2	Indicates summarized the state of selected channels.
3	Increases volume of selected channels by 1 dB.
4	Channel volume to set.
5	Apply Settings.
6	Shows vales of selected channels.

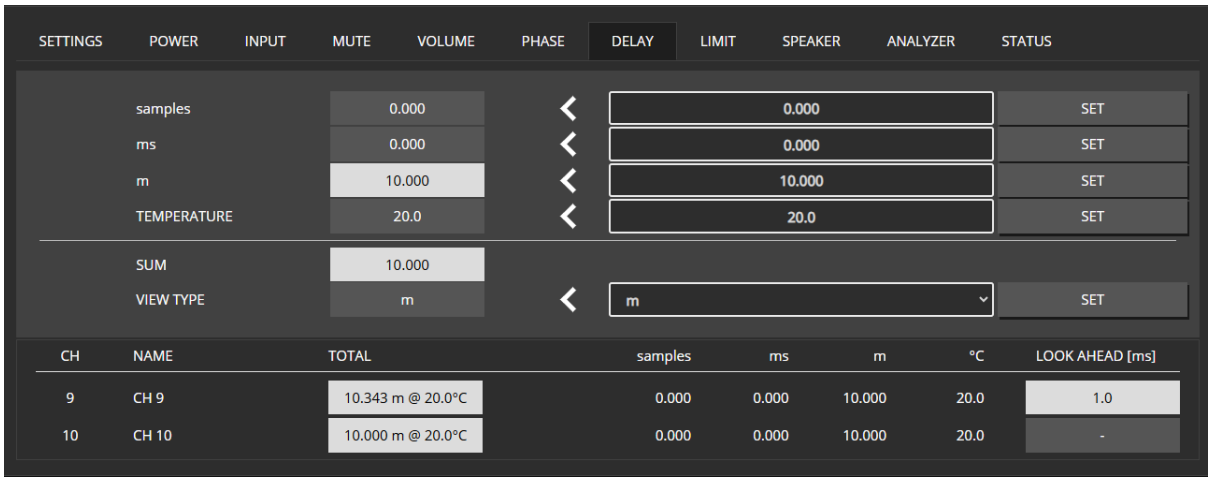
6.6.10. PHASE



NR	DESCRIPTION
1	Indicates summarized state of selected channels.
2	Set normal phase.
3	Set reverse phase.
4	Shows vales of selected channels.

6.6.11. DELAY

The delay now allows you to simultaneously set a value in all three available formats: samples, milliseconds (ms), and meters (m). To calculate the delay in meters, a room/air temperature is required to calculate the correct speed of sound.



The **SUM** field shows the summation of all delay values set above, while the **VIEW TYPE** determines the unit type for the **SUM**, **TOTAL** and the delay field on the **OVERVIEW** page.

An additional Look Ahead Delay indicated in the right column is added to the **TOTAL** column, which represents the final delay value the channel is affected by.

It is also possible to enter "negtiave" values into the delay fields but the **SUM** of all delays, can't become negative.

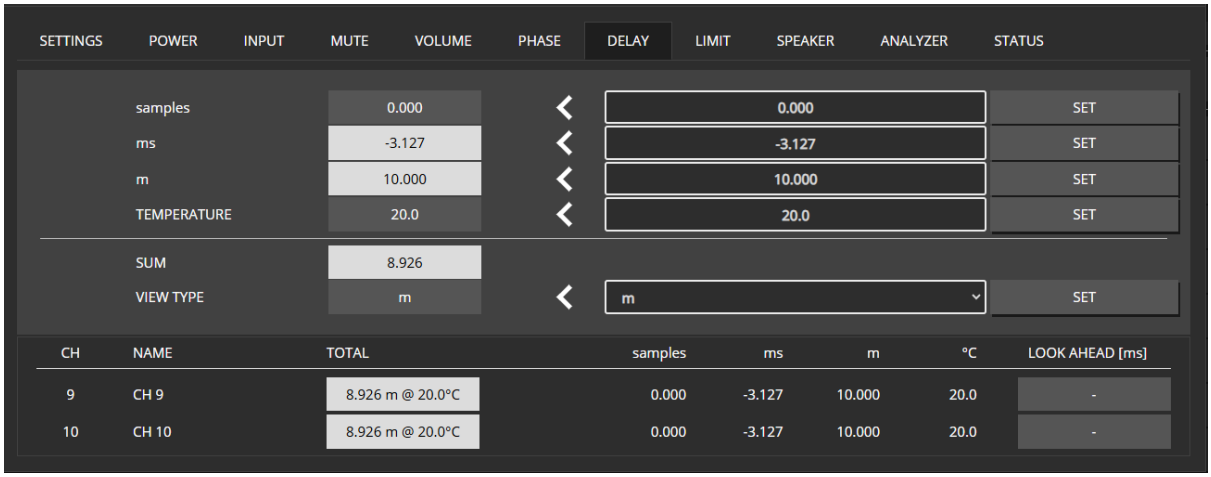


Figure 30. Negative Delay

6.6.12. PEQ

There are 32 EQ slots that can be set with several EQ types. Some EQ types need more than one EQ slot. 18dB/24dB Low/High passes require two, while 48dB Low/High require four slots.

When values are changed but not set to the device, the EQ is in preview mode which is indicated by **PREVIEW** in the HEADLINE, and the PEQ plot only shows the theoretical EQ curve. The current enabled EQs are plotted in the output curve.

Figure 31. PEQ Overview

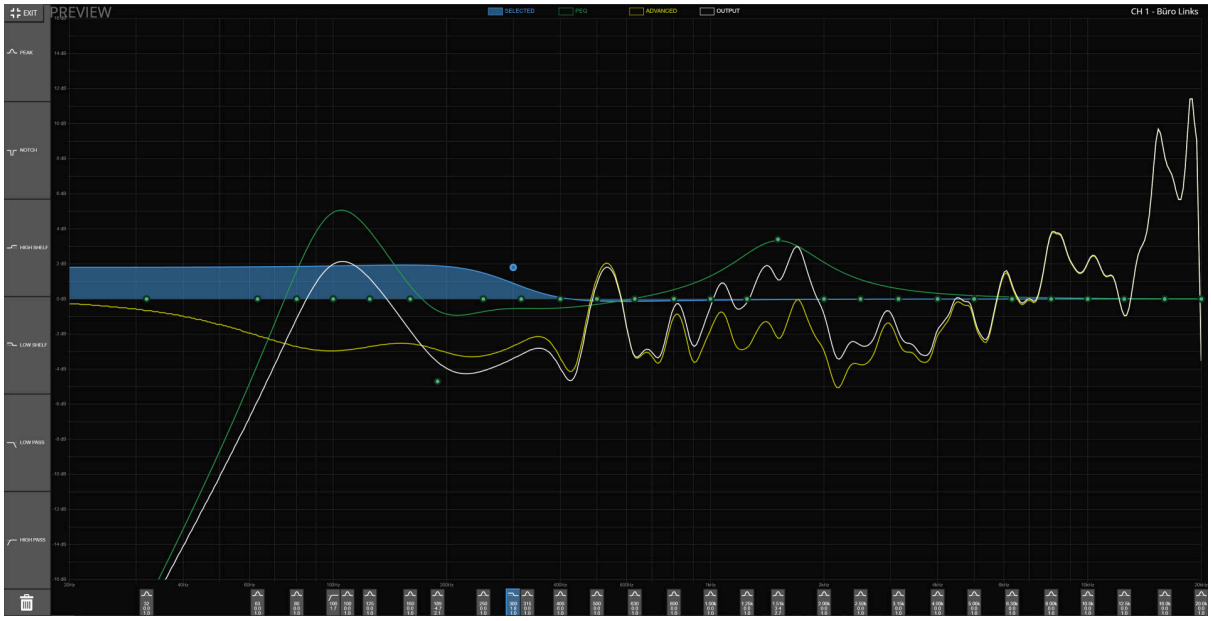


Figure 32. PEQ Fullscreen

The Plot shows the overall EQ for this channel, indicated by the white **OUTPUT** curve. It's the result of **PEQs** and **ADVANCED** EQs.

All EQs parameters are shown in the list below the plot, where each parameter can be manually adjusted. Don't forget to press ENTER or SET to confirm your changes.

Each EQ (or ALL) can be ENABLED/DISABLED via the toggle button on the left of each row (in the headline for ALL) to make a simple A/B comparison.

When changing the EQ TYPE, it's not automatically applied to prevent damage when accidentally choosing the wrong type. It has to be confirmed by pressing the **SET** button.

PEQ Add / Remove

New PEQs can be added by dragging the desired EQ type from the left selection into the curve plot, or by pressing the **ADD** Button below.

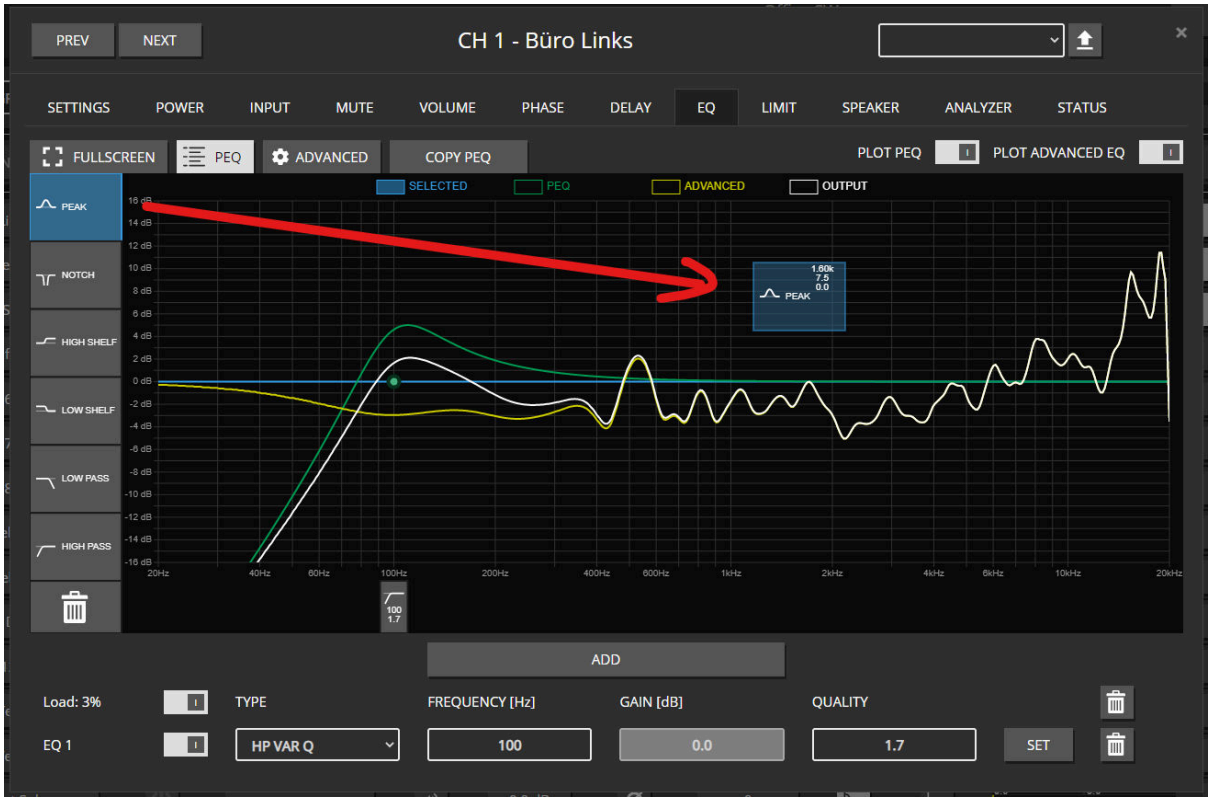


Figure 33. Drag & Drop

Removing a PEQ by dragging its ribbon below the plot on the bin icon or pressing the icon next to the EQ in the overall list below.

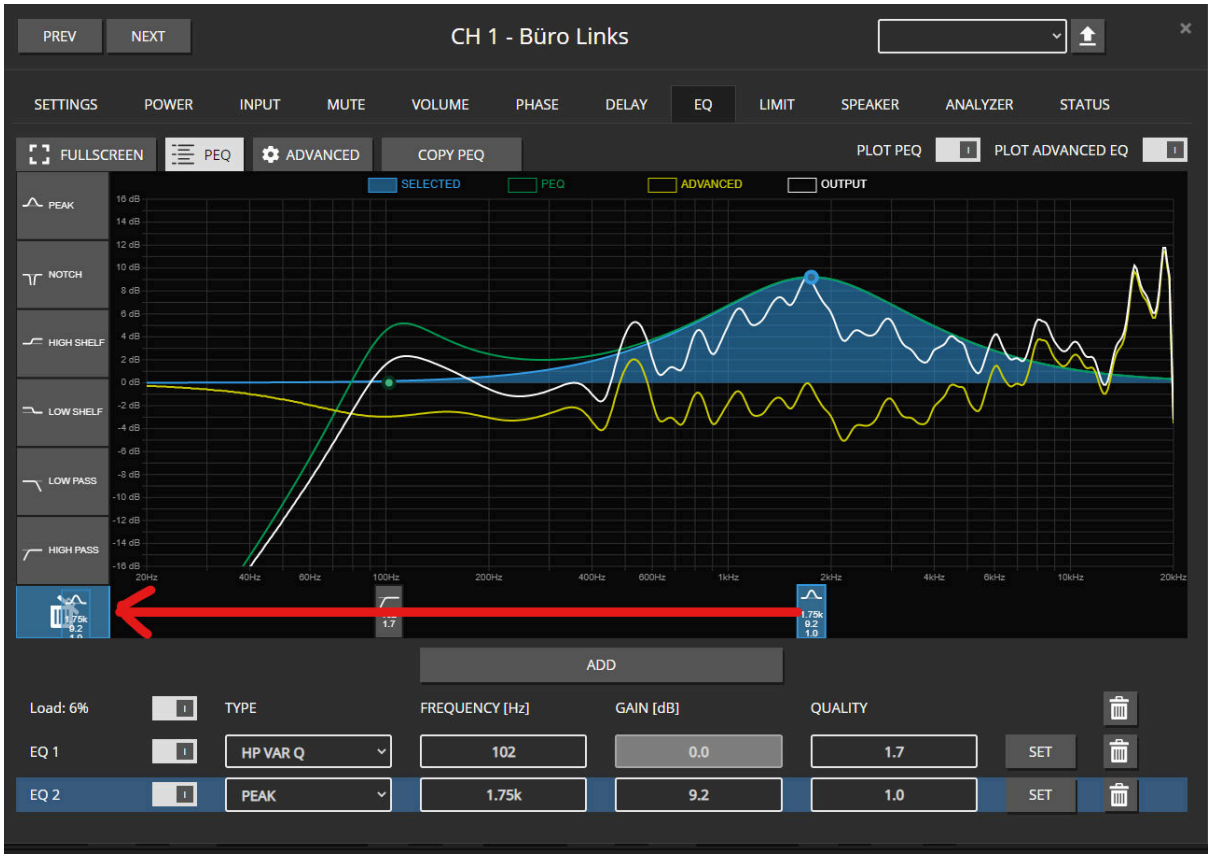


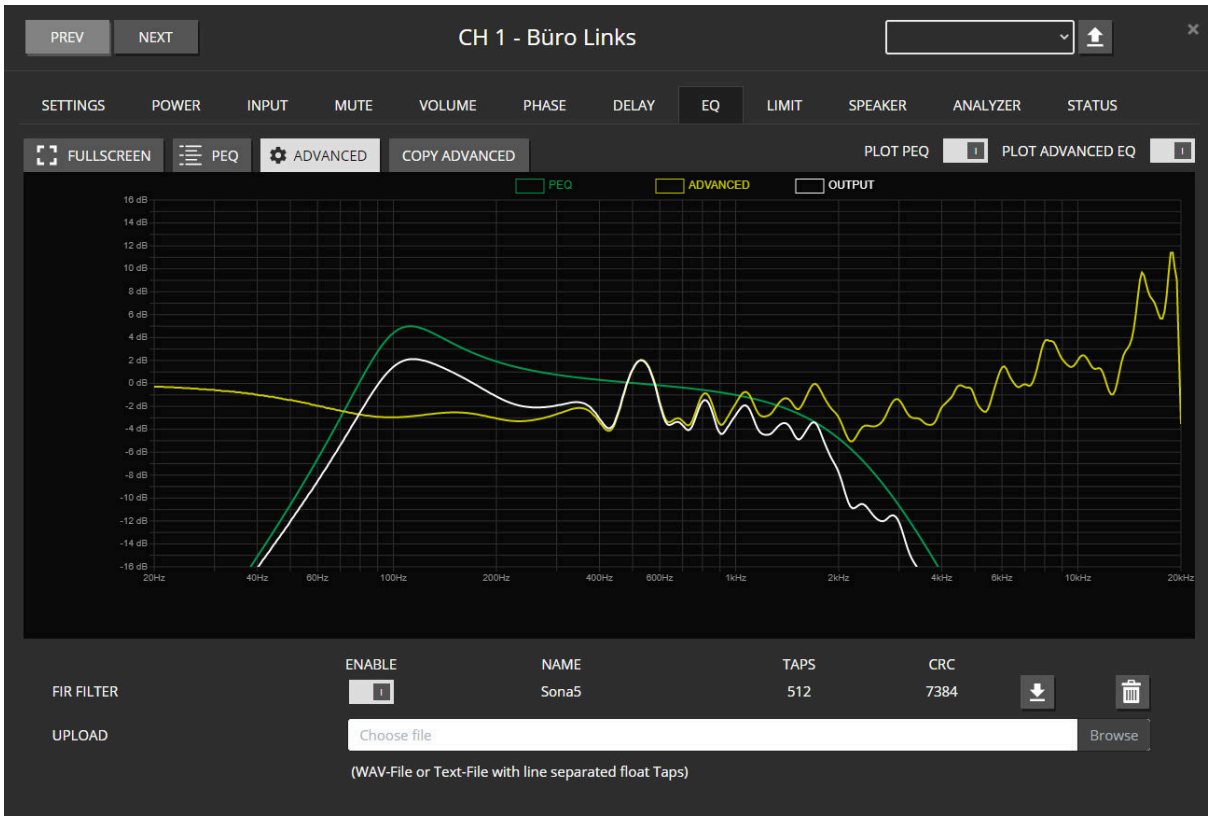
Figure 34. Remove an EQ

ADVANCED EQ (FIR)

The **ADVANCED** EQ currently supports loading FIR impulse responses to be convoluted over the output signal. (see: Supported File Formats)

Once a file is loaded, it the FIR filter engine can be **ENABLED/DISABLED** via the toggle button. The **TAPS** indicates the length of the filter, while the **CRC** value is a simple checksum over all coefficients, which easily allows the user to compare different revisions of files. The same CRC on different channels guarantees that exactly the same FIR filter is loaded.

> The FIR engine always runs on 1xFS of the selected input sync interface. (e.g. 32, 44.1, 48kHz) Thus, the FIR coefficients have to export for the desired sample rate and can not be dynamically adapted.



Supported File Formats

The Range of each coefficient is limited -4.0 to 3.999 due to the internal fix point representation. The maximum number of Taps is 2048.

WAV The WAV File has to be 32Bit Float 48kHz

Coefficient File

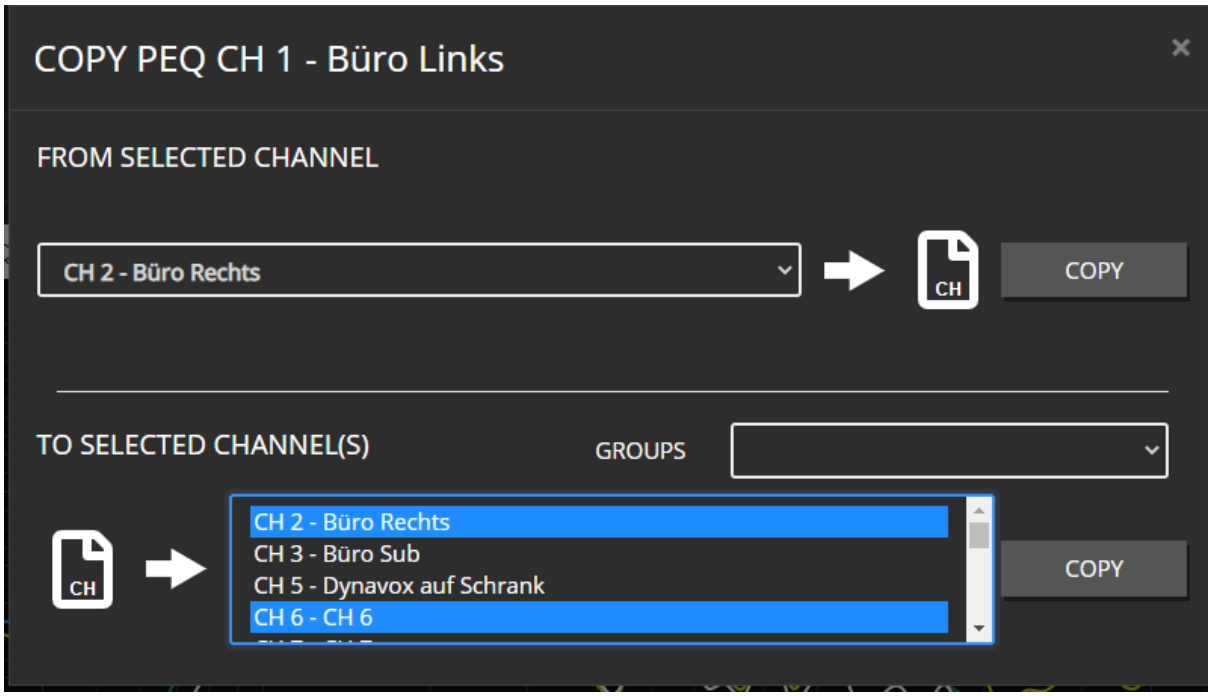
line in file	coefficient	multiplied with sample
1	-0.1648560000	* n
2	0.0737233000	* (n-1)
3	-0.0973907000	* (n-2)
4	0.0139486000	* (n-3)
5	-0.0406976000	* (n-4)
6	-0.0222878000	* (n-5)
7	0.0227421000	* (n-6)
...		
2048	-0.0066785500	* (n-2047)

line in file	coefficient	multiplied with sample
1	-2.03711E-0003	* n
2	-2.03711E-0003	* (n-1)
3	-7.42133E-0004	* (n-2)
4	-2.41038E-0003	* (n-3)
5	1.85561E-0004	* (n-4)
6	-3.39548E-0003	* (n-5)
7	5.96577E-0004	* (n-6)
8	-1.78038E-0003	* (n-7)

COPY TO / COPY FROM

Both **PEQ** and **ADVANCED** can be copied from different channels into the currently selected one, or can be copied to others channels.

The Copy to / Copy From pop-up can be opened by pressing the **COPY PEQ** or **COPY ADVANCED**.



COPY FROM allows just a single channel to be selected, while **COPY TO** allows one or multiple. Selection is performed by CTRL + click / Shift + click, or by using pre-defined Selection GROUPS from the overview page.

6.6.13. LIMIT

Each channel offers four limiters, 1x **CURRENT**, 2x **VOLTAGE**, 1x **POWER**.

> All thresholds are configured in **peak** values, for simple sine wave signals, the corresponding **RMS** value can be calculated by **peak** / sqrt(2).

On a single-channel edit, all level meters are shown simultaneously. While selecting multiple channels, only level meters of the same limiter type can be seen in the list below.

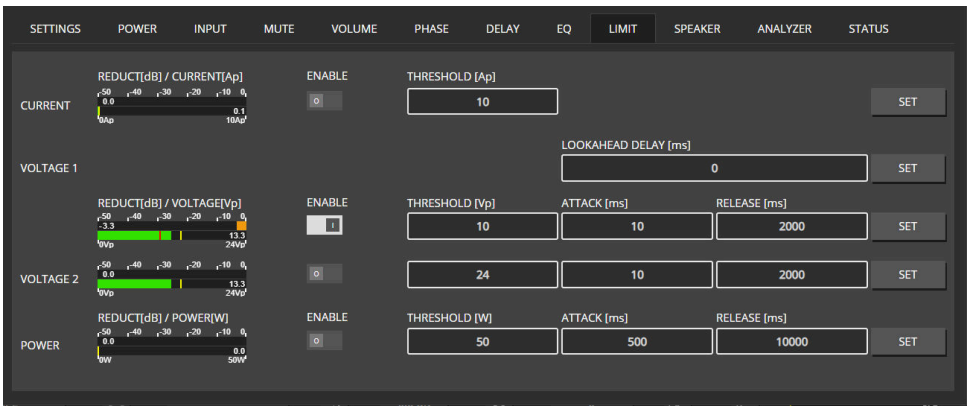


Figure 35. Single Channel Edit

Figure 36. Multi Channel Edit

CURRENT limiter is used as simple short circuit protection, or can be used to limit the power on certain impedance minimums where otherwise the voltage had to be reduced so much, that overall power would be too low.

The **POWER** limiter is for a long time, RMS limiting. The active power is calculated by multiplying "voltage * current", sample by sample. Normally just active for high-power subs since, for regular speakers, the actual RMS power is really low with music.

Disabled limiters are set to the maximum threshold internally. Due to the internal headroom, it is still possible to see some reduction if the maximum thresholds are reached.

Look Ahead Delay

The first voltage limiter offers an additional look-ahead delay. This delay is added to the signal chain, while the side chain input to the limiter is used without delay.

Example Settings

In most cases, the Voltage Limiter 1 is suitable enough to achieve simple output limiting.

The Look-Ahead delay mainly helps to prevent overshoots. This is mainly required if you want it as a hard clip limiter or to prevent the slightly "plopping" noise when hard clipping occurs due to reaching rail voltage maximums.

So the recommendation is setting THD ~ 50-53V, Attack 1-5 ms, Release 100-300ms, and setting the look-ahead delay as high as it can be tolerated up to a maximum of attack time + 1-2ms.

After that, sometimes setting the Voltage Limiter 2, with a little lower threshold like the Limiter 1 (like 1-5 volts) and attack 200 - 500 ms, release 1000 - 5000 ms. This will just make it sound a little bit better if constantly running in the limiting area since the second limiter will overtake the reduction from the first one. Since it has much more release time, the signal will not be deformed as much as limiter 1 is doing with its 1-5ms attack.

6.6.14. SPEAKER SETTINGS

SINGLE EDIT

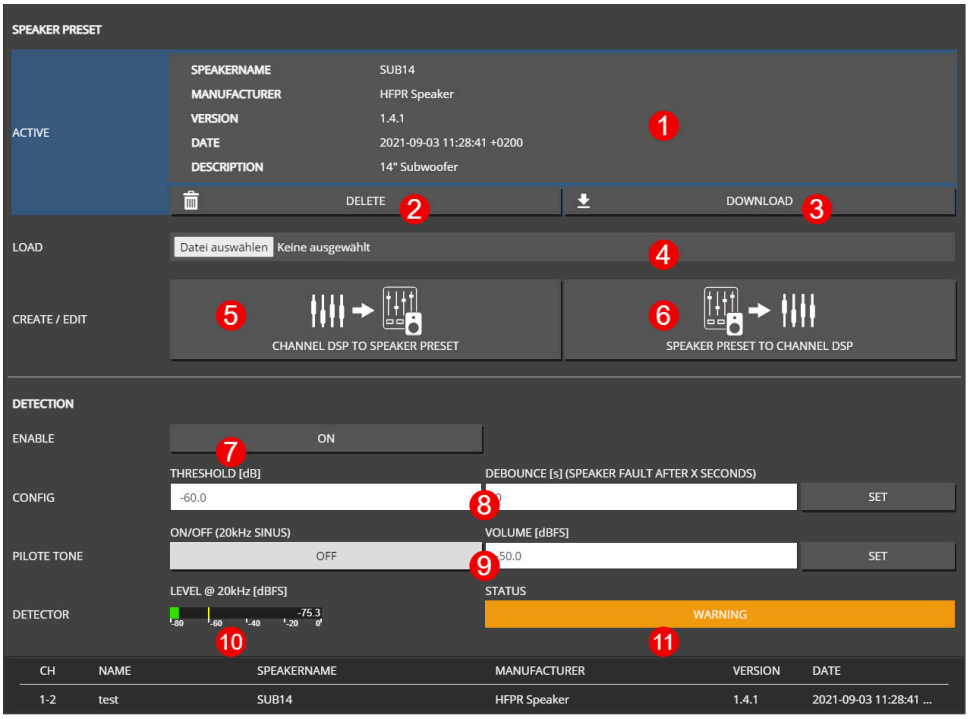


Figure 37. SPEAKER SINGLE CHANNEL EDIT VIEW

NR	DESCRIPTION
1	Metadata of the active speaker preset.
2	Remove the speaker preset.
3	Download the speaker preset file to share it or apply to others channels.
4	Load a speaker preset file from your computer.
5	Create a speaker preset from channel dsp data WEBPAGE SPEAKER PRESET CREATE.
6	Load the speaker preset dsp data to the channel dsp WEBPAGE SPEAKER PRESET LOAD.
7	Enable/Disable speaker detection.

8	Set detection threshold and debounce. The 20 kHz current value has to be lower than the threshold for "debounce" seconds to trigger an error.
9	Set 20 kHz pilot tone generator level in dBFS which will be added to the actual output signal of the amplifier.
10	Shows measured current at 20 kHz (yellow line indicates threshold)
11	Shows actual speaker detection status.

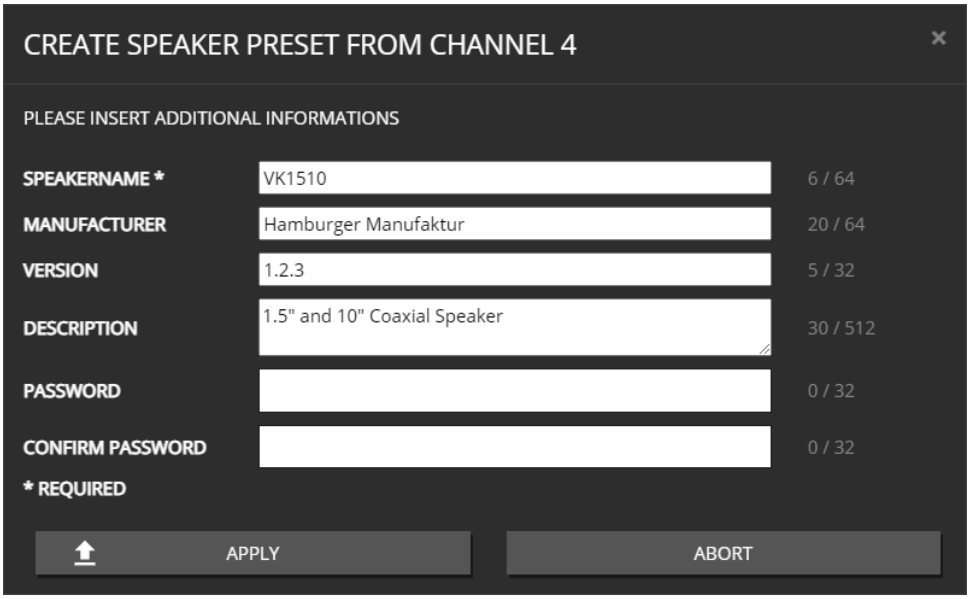


Figure 38. WEBPAGE SPEAKER PRESET CREATE

To create a speaker preset, tune your speaker with the channel DSP settings to your needs. The parameter which can be used inside the speaker preset are: VOLUME, PHASE, DELAY, 32x PEQ, ADVANCED EQ (FIR Filters with 512 Taps), LIMIT.

Once happy with your parameter work, create the speaker preset by clicking on the "CHANNEL DSP TO SPEAKER PRESET" button. This will copy all parameters listed above to a fully separated "SPEAKER DSP" and free up the "CHANNEL DSP".

Information like a speaker name is mandatory, all other parameters are optional. If the data have to be secure, please insert a password. This password is only required to load the speaker preset to the channel DSP (for further editing).

The FIR Filter in the speaker preset can only have 512 taps, instead of the full 2048 taps which are possible in the channel DSP.

The LIMIT section only exists once. If a limiter is used inside a speaker preset, it will just block that specific limiter from being used by the user.

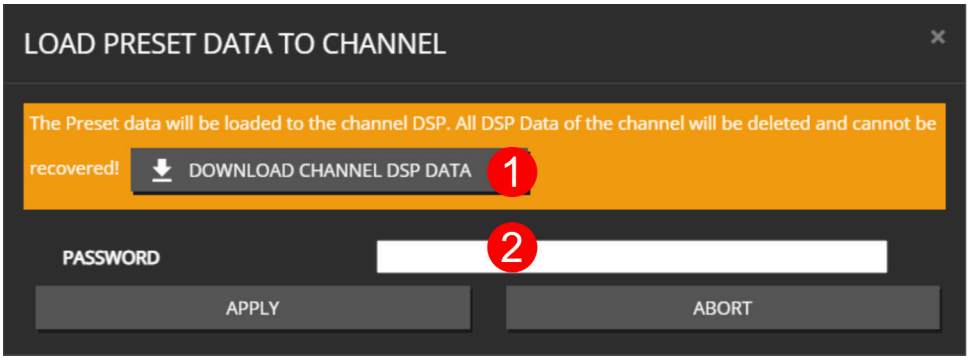


Figure 39. WEBPAGE SPEAKER PRESET LOAD

NR	DESCRIPTION
1	Since the channel DSP will be overwritten by the speaker preset values, a backup of the currently loaded settings can be downloaded as channel preset.
2	If a speaker preset is created with a password, the password is required to load the data to the channel

MULTI EDIT

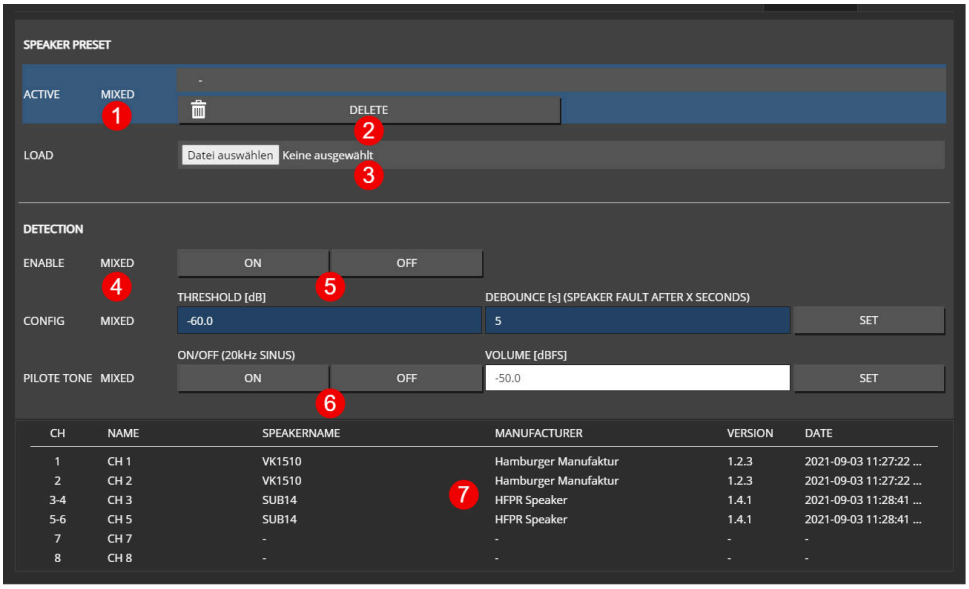


Figure 40. SPEAKER MULTI CHANNEL EDIT VIEW

NR	DESCRIPTION
1	Mixed speaker preset indicator (different speakers presets are loaded on the selected channel)
2	Delete the currently loaded speaker preset from all selected channels.
3	Load a speaker preset file from your computer to all selected channels.
4	Mixed value indicator of the speaker detection section.
5	Enable/Disable speaker detection.
6	Set 20 kHz pilot tone generator level in dBFS which will be added to the actual output signal of the amplifier.
7	Shows actual speaker detection status.

6.6.15. ANALYZER

The integrated impedance analyzer performs a continuous measurement of the connected speaker impedance over the full frequency spectrum based on the supplied music signal.

It's enabled when the POWER of the corresponding channel is active.

The analyzer measures up to 32 channels simultaneously.

This is performed by using two 32k FFTs per channel for voltage and current, which are divided through each other to calculate the complex impedance. Additional log decimation and smoothing is applied afterwards to achieve a higher signal-to-noise ratio even to estimate the spectrum with a relatively low output signal. Depending on the output level, the measurement process may take some time. The higher the signal level, the less time it takes to get valid results.

Since the music signal does not always contain enough information on certain frequencies, a **CONFIDENCE** level indicates the level of trust that can be taken to certain frequency bins. When the **CONFIDENCE** of a certain bin reaches **1.0**, its indicated by **GREEN** curve plot. The momentary output of the impedance analyzer is indicated by **GRAY** colour and is present FYI.

Figure 41. Real Time Impedance

On the **DETAIL** view, the output of the two complexes FFTs can be seen, which are just for information.

The spectrum plots are not intended as a reference for the audio performance of

> the actual output stage, which is much better than the simple measurements ADCs used to collect the data for the FFTs.

MEASUREMENT

To perform a reference sweep, the **MEASUREMENT** tab can be used. Multiple channels can be started simultaneously with different levels, depending on the connected speaker impedance it may require some gain adjustments to measure the full impedance spectrum.

> In the reference mode, all channel settings are temporarily bypassed, which means a full frequency sweep with the configured **GAIN** is performed.

Figure 42. SWEEP in progress

Figure 43. SWEEP done

After the measurement is finished (or any time when the spectrum is completely built up by music), the **GREEN** impedance curve could be saved as a **REFERENCE** curve which is persistent during reboots.

This can be helpful to perform A/B comparisons over time to detect the ageing or failures of speakers.

More automatic analysis functions based on the reference are coming in further releases.

Figure 44. REFERENCE saved

6.7. INTERFACES

6.7.1. INTERFACE STATUS

Select the interface to derive the audio clock source. In most applications, only one interface like MADI or DANTE is used to supply audio data AND the audio clock.

In the **SYNC SELECT** column, the preferred interface is selected. If the interface is not locked or supplies an invalid or not supported audio rate, the next interface with valid settings is automatically selected.

The auto sync select priority is as ordered in the list from **MADI FIBRE** down to **AES3 2(ASRC)**

GREEN colour indicates the currently selected sync interface.

> AES3 receivers can also be selected as clock sources. Despite the fact that the clocks are synchronous, the audio data are routed through the sample rate converter.

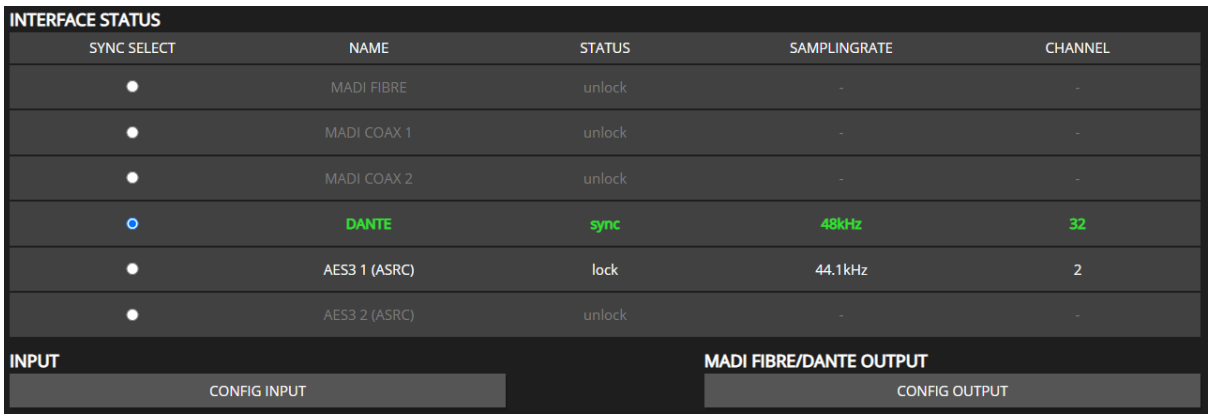


Figure 45. WEBPAGE INTERFACES

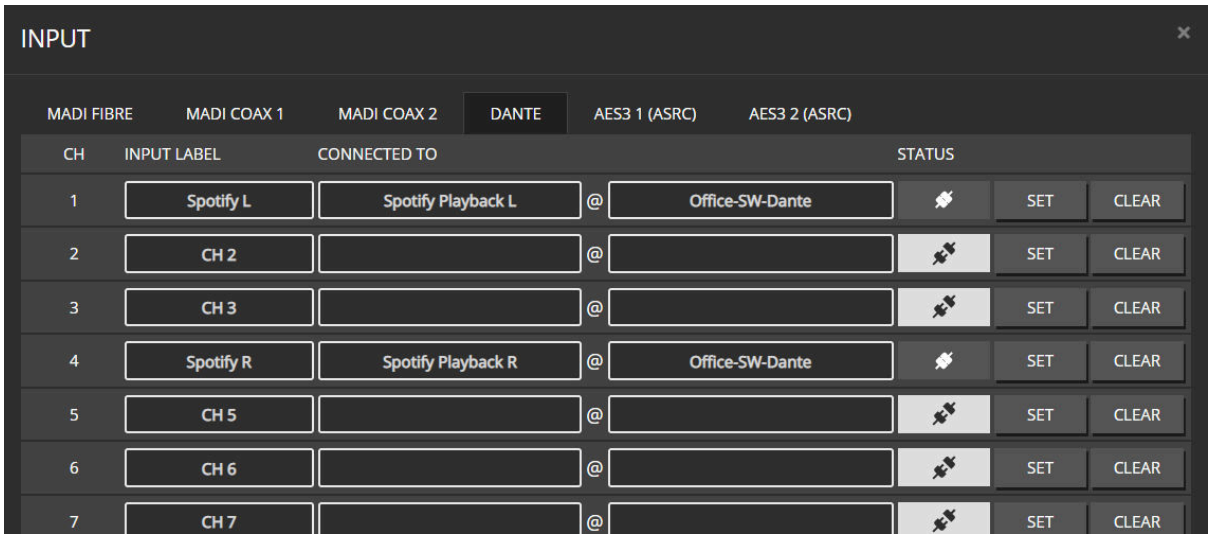
Table 7. SYNC STATUS values

STATUS	DESCRIPTION
unlock	No valid carrier or word-clock was detected on that interface.
lock	Valid carrier and word-clock but not in phase with the internal audio clock.
sync	Valid carrier and word-clock AND in phase with the internal audio clock.
error	Unsupported sample rate.

6.7.2. INPUT

The **INPUT** pop-up allows the labelling of all channels of each input interface and shows individual stream status.

The most interesting part is the **DANTE** tab, which allows assigning streams to the Dante module. The **STATUS** is also used on the global **OVERVIEW** page when this Input Stream uses an **INPUT** to a channel but does not supply any valid stream or data.



6.7.3. MADI FIBRE/DANTE OUTPUT

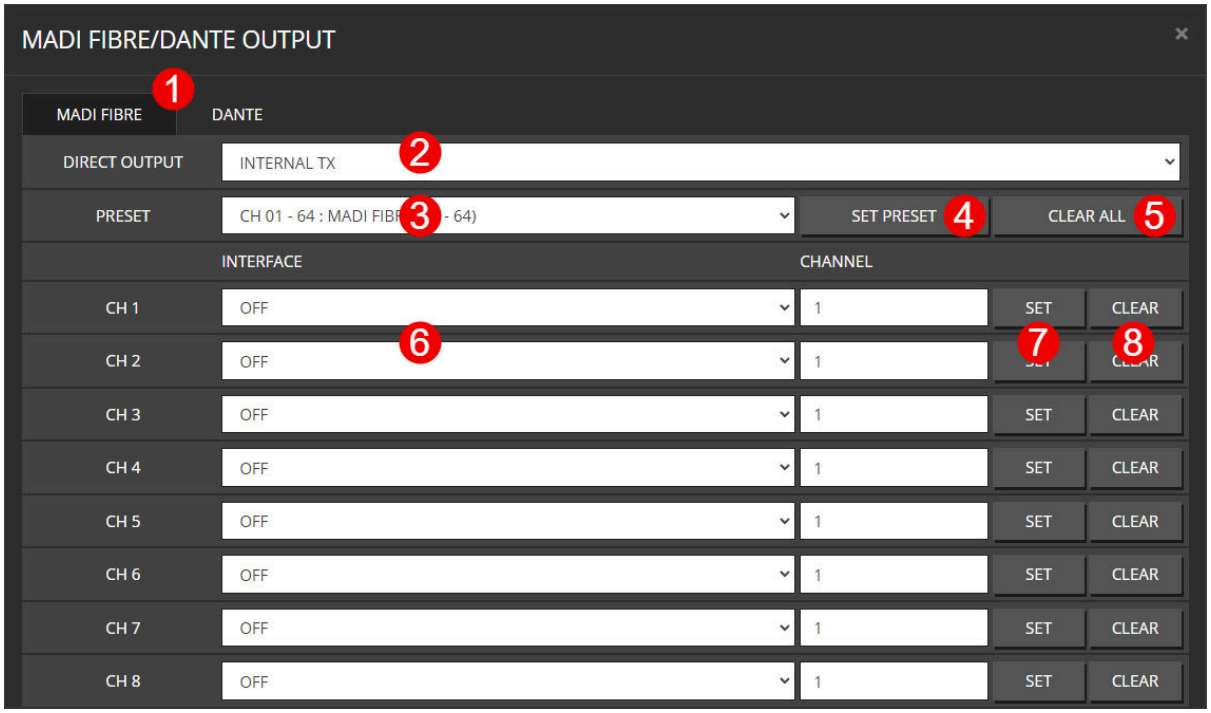


Figure 46. WEBPAGE INTERFACES OUTPUT

NR	DESCRIPTION
1	Select Output to Config
2	Select the source of the MADI FIBRE TX jack. This can either be "INTERNAL TX" which uses the internal MADI transmitter, or any of the other MADI RX jacks. In the case of an RX jack, the connection is direct routed through the FPGA with almost zero latency, but a jitter build-up has to be in mind. The direct through connection can be used to daisy-chain two devices to utilize all 64 channels on a MADI signal. Device 1 consumes CH1-32, device 2 CH33-64.
3	Select a Preset to Set
4	Set Selected Preset
5	Clear all Output Patchings
6	Select a Interface/Source with Channel
8	Set selected Source and Channel to interface output channel

9	Clear Patching from interface output channel

6.8. DEVICE

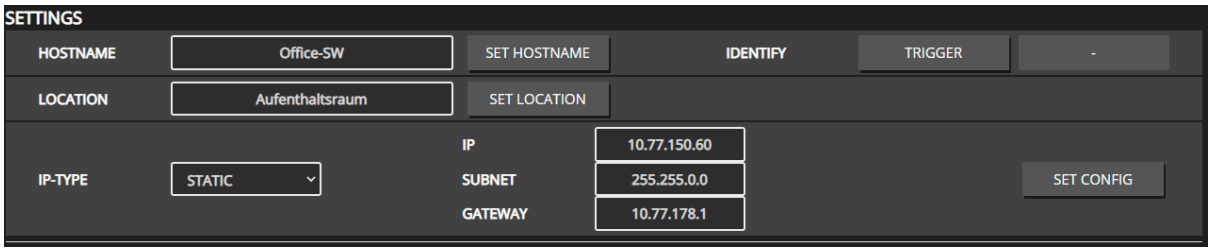
The **DEVICE** page offers different global settings that affect the amplifier device.

6.8.1. SETTINGS

The **HOSTNAME** is used in DNS for IP resolving. **LOCATION** is just a string to add some additional information to the device, like where is it located.

When **IDENTIFY** is active, the device will visually identify itself by blinking leds. But will automatically stop after a few seconds.

Changing the **IP-TYPE** requires confirmation by pressing **SET CONFIG**. The web interface has to be reconnected to the new IP address.



6.8.2. DANTE SETTINGS

IDENTITY is the hostname of the integrated Dante module, as seen in the Dante Controller.

VLAN-CONFIG determine which physical Ethernet port is connected to the amplifier controller and Dante module board. This is also indicated by the LED between the Ethernet ports on the device's backside. see NETWORK MODES

MODE	LED COLOUR
Switched	GREEN
Redundant	YELLOW
Independent	BLUE

Do not use the same hostname for the amplifier and the Dante Module if using **Switched** or **Redundant** mode since the mDNS hostname resolve protocol will not be able to determine the IP address of the devices.

It's pretty handy to use the **SYNC WITH HOSTNAME** option, which will use the

amplifier's **HOSTNAME** and append **-DANTE** to it, which is used as Dante **IDENTITY**.

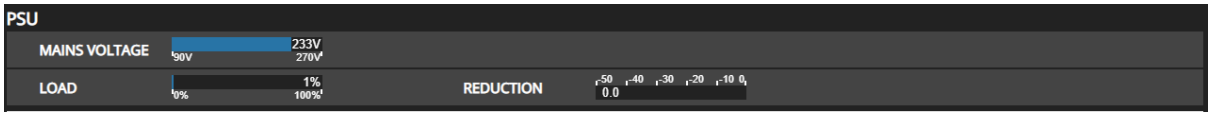
6.8.3. TIME

Set system time and time zone. If the device is connected to the internet, it will try to synchronize its RTC (real time clock) to an NTP time-server. The system time is mainly used in Syslog entries to tag events.

6.8.4. PSU

Measures the mains voltage, which is used for derating the PSU in 100 / 110V cases. If the device has more than one PSU, it will indicate which one is plugged in. (HP² only)

When **LOAD** reaches 100%, the overall power limiter will reduce the output level of all channels simultaneously, not to overload the PSU. The **REDUCTION** meter is also shown on the **OVERVIEW** page as **REDUCT HW** level meter.



6.8.5. HOUSING

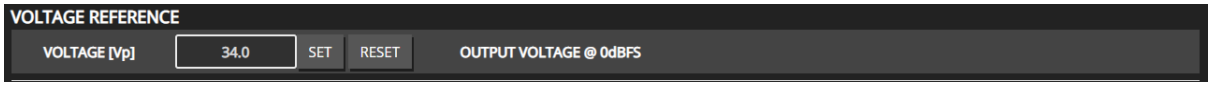
FAN MODE **NORMAL** is the recommended mode to keep all components as cool as possible to improve lifetime. When not much output power is required, the FAN MODE could be changed to **SILENT** or **PASSIVE** (LP² only) which will use different fan speeds to reduce noise.

MODE	CPU FAN	HOUSING FAN	PSU FAN
NORMAL	100%	100%	Temperature & Load controlled
SILENT	Temperature controlled	Temperature controlled	Temperature & Load controlled
PASSIVE (LP² only)	OFF, but Temperature controlled if exceeds	OFF, but Temperature controlled if exceeds	not present

6.8.6. VOLTAGE REFERENCE

VOLTAGE REFERENCE defines the maximal output peak-voltage when feeding an 0dBFS signal on any input interface. Due to different maximum rail voltages based on the hardware device, a clipping could occur.

This is intended to match the gain of different amplifiers to output the same voltage with a given digital input signal.



6.8.7. DEVICE MUTE

En-/disable mute on startup, if enabled, the **MASTER MUTE** will be set on every start up. An additional **VOLUME RAMP** can be enabled, which will linearly increase the dB value until it is reached its desired value after EVERY **MASTER MUTE** unmute.

6.8.8. REMOTE MUTE

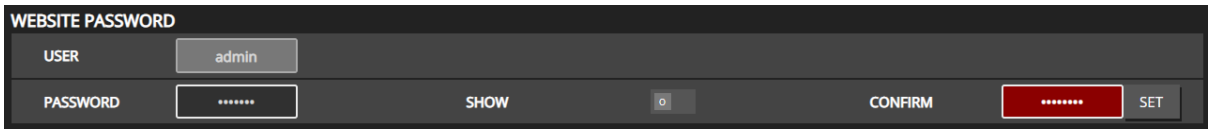
Enable remote mute, this will provide a GPI interface to mute the entire device, also known as dead man switch. Which require an external Innosonix Remote Mute Server, multiple devices can share one server.

6.8.9. WEBSITE PASSWORD

USE with caution, to not lock you out of your device.

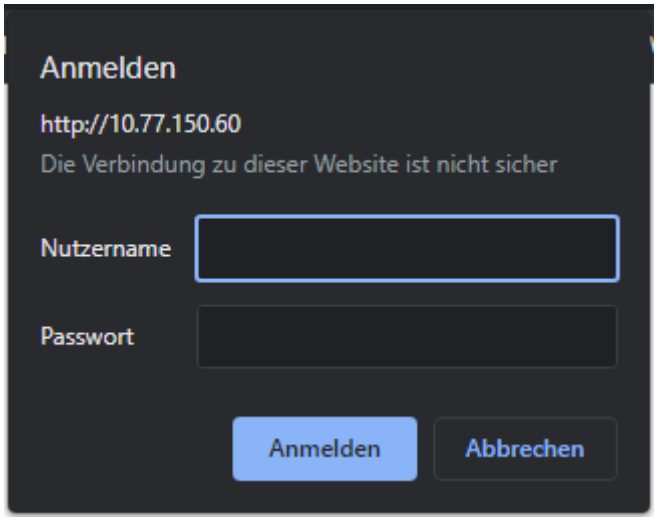
Set an HTTP password which restricts access to this device configuration page. The password will only restrict web page access and NOT any API transactions to allow existing media controls still work but prevent prevents unauthorized access by users.

Username is fixed to **admin**, while the password can be entered secretly or shown by enabling the SHOW switch.



DO NOT LOOSE YOUR PASSWORD!

Once a password is set, accessing the webpage on a **new** web browser instance will ask for the username and password. Existing sessions will be stored in your local browser cache; you don't need to enter the password again.



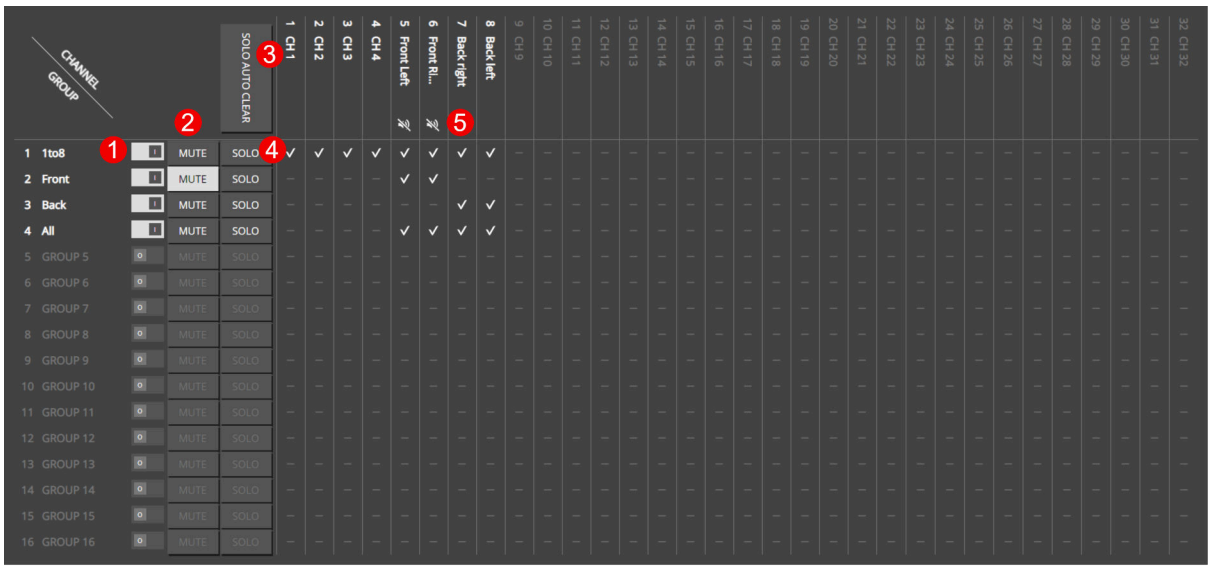
On an logged in session, the website password can be removed by clicking on **CLEAR PASSWORD**

6.8.10. DEVICE REBOOT

Simply performs a complete reboot of the device.

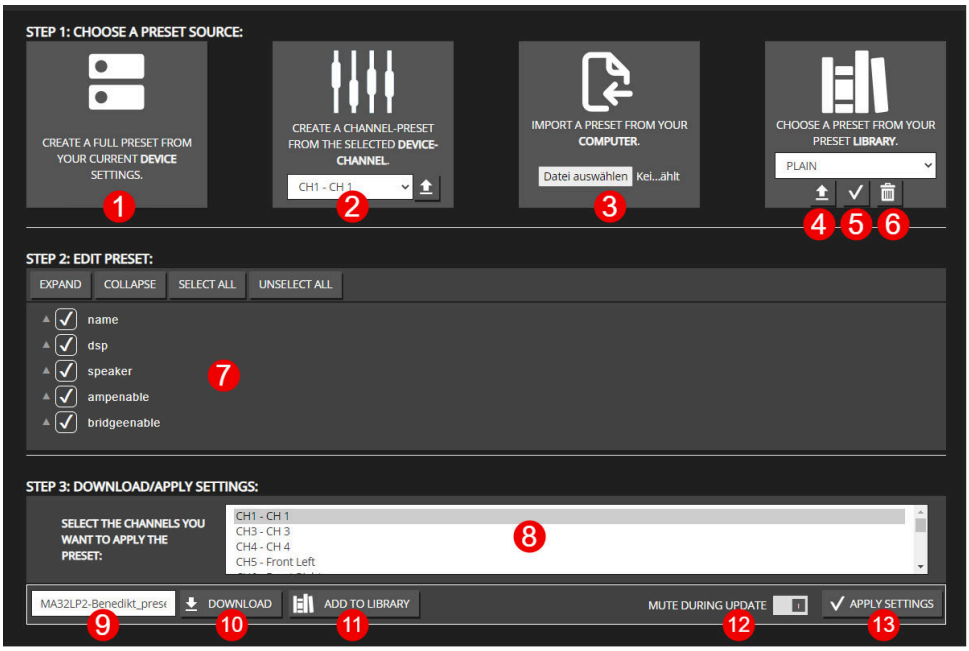
6.9. MUTEGROUPS

Mute groups assignment are derived from selection groups (SELECTION AND GROUPING).



NR	DESCRIPTION
1	Only enabled mute groups are taken into account when the final mute result is calculated.
2	Mute / unmute mute group.
3	If solo auto clear is activated, only one solo can be active.
4	Active SOLO for the corresponding group. All other channels in an active mute group will be muted.
5	A speaker symbol and blinking MUTE button on the OVERVIEW indicates that the channel is muted due to a mute group assignment.

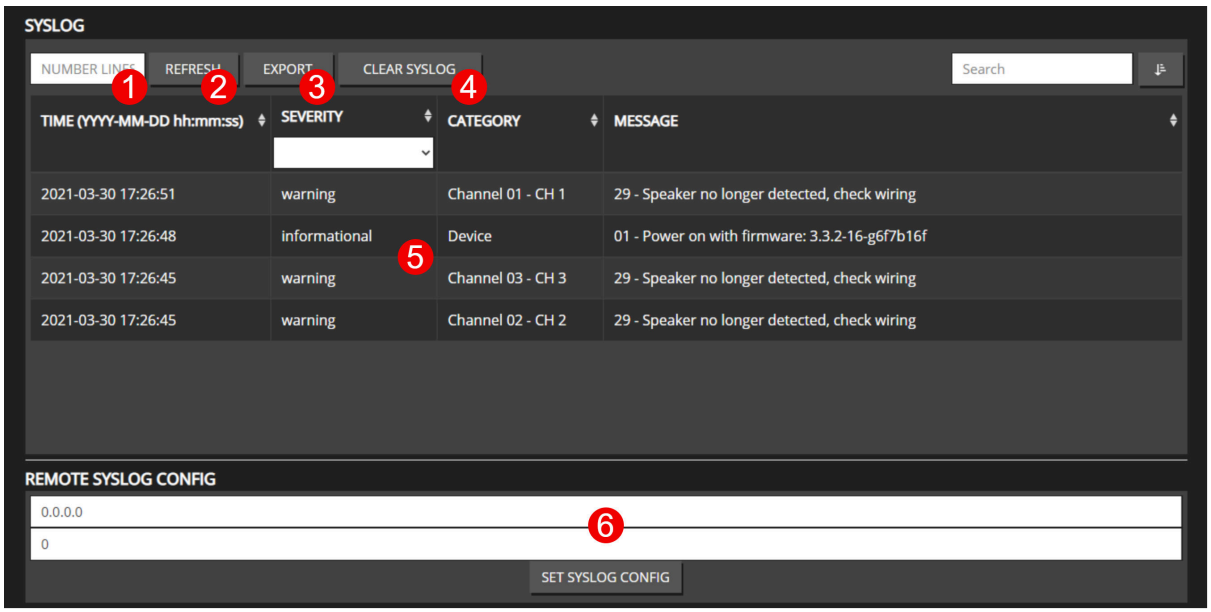
6.10. PRESETS



NR	DESCRIPTION
1	Load all device settings to the preset editor (7). Device presets do include fixed mapping of parameters to specific channels.
2	Load setting from one specific channel to preset editor (7).
3	Upload a file from your computer to the editor (7) . It can either be a channel or device preset.
4	Load preset from preset library in preset editor (7).
5	Recall the selected preset from the library to the device. This functionality can also easily be triggered via the RESTful-API to do a simple scene switch.
6	Delete selected preset from the library.
7	edit settings tree
8	select (multiple) channel to load preset to (only available if channel preset is loaded into the preset editor)
9	preset name to save in library or download

10	save selected settings as preset to library
11	download selected settings as preset file
12	mute device / channel during update settings from preset
13	apply selected settings to device

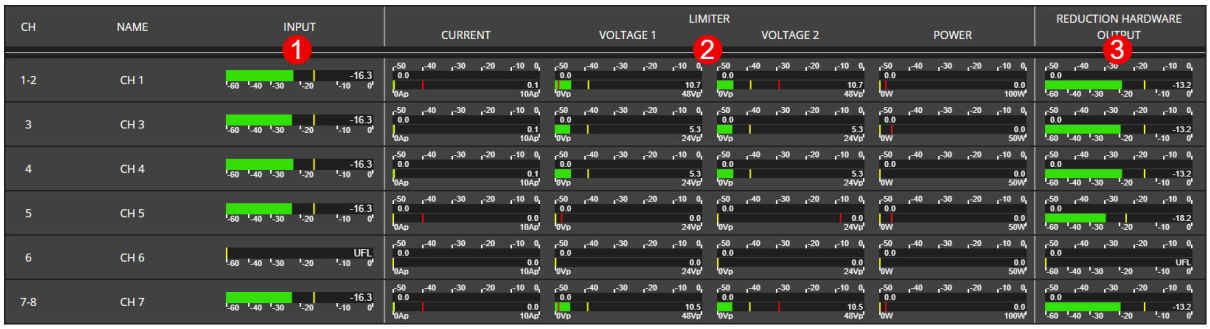
6.11. LOGGING



NR	DESCRIPTION
1	select syslog lines to load
2	refresh syslog table
3	export complete syslog from device
4	delete syslog files on the device
5	chronological errors

6 config connection to external syslog server

6.12. METERING



NR	DESCRIPTION
1	input level after input mixer
2	measured current, voltage and power with limiter reductions
3	output level with hardware reduction (sum of PSU Limit and Thermo Limit)

Chapter 7. ERROR CODES

Table 8. SEVERITY LEVEL

TYPE	DESCRIPTION
EMERGENCY	system is unusable
ALERT	action must be taken immediately
CRITICAL	critical conditions
ERROR	error conditions
WARNING	warning conditions
NOTICE	normal but significant condition
INFO	informational

Table 9. ERROR CODES

NR	SEVERITY	DESCRIPTION
1	INFO	Power on
2	INFO	IP mode set to DHCP
3	INFO	IP mode set to AUTO IP
4	INFO	IP mode set to STATIC IP
5	ALERT	UDP Discovery error, device no longer available, please try to restart the device
6	INFO	device reboots for software update
7	ALERT	Interfaces monitoring and control no longer available, please try to restart the device
8	ALERT	Speaker monitoring no longer available, please try to restart the device
9	ERROR	Display Interface no longer available, please try to restart the device
10	INFO	Samplingrate changed, EQs, Limiter, FIR Filter will be recalced
11	ALERT	DSP monitoring/control no longer available, please try to restart the device
12	EMERGENCY,	Hardware verification failed, no Audio available
13	ERROR	Metering no longer available, please try to restart the device
14	ERROR	Amplifier overcurrent error
15	ALERT	Amplifier overcurrent Shutdown
16	ALERT	Amplifier recurring overcurrent error, check wiring and powercycle channel to try again

NR	SEVERITY	DESCRIPTION
17	EMERGENCY	Amplifier communication error, please try to restart the device
18	WARNING	Amplifier overtemp
19	ALERT	Controller monitoring no longer available, please try to restart the device
20	ALERT	FAN controller no longer available, please try to restart the device
21	CRITICAL	Overtemp emergency shutdown init, all Fans will turn up, till temperature out of critical range
22	ALERT	PSU monitoring no longer available, please try to restart the device
23	ERROR	No settings file available using default settings
24	ERROR	Settings file corrupted, file will be deleted
25	ALERT	All Settings files corrupted, start with default settings
26	ALERT	User Settings cannot be saved anymore, please try to restart the device
27	ALERT	User Settings cannot be changed anymore, please try to restart the device
28	ALERT	User Settings cannot be restored correctly, please try to restart the device
29	WARNING	Speaker no longer detected, check wiring
30	EMERGENCY	Wrong PD Type installed
31	ERROR	No Calibration File available, Amp using default values
32	CRITICAL	Power distribution overcurrent, try to restart
33	EMERGENCY	Amp Module Hardware Error
34	EMERGENCY	Amplifier Shutdown caused by PSU Overcurrent
35	ALERT	Remote Mute no longer available, please try to restart the device
36	EMERGENCY	Start without initing all Amps
37	ALERT	DC not OK
38	EMERGENCY	Amplifier Shutdown caused by overtemp emergency shutdown
39	EMERGENCY	Power Distribution cannot be load, please try to restart the device
40	WARNING	Link unlock
41	WARNING	CRC errors
42	WARNING	Negative Rail Converter ready timeout
43	WARNING	Fan dirty or stuck, check logging for further informations
44	WARNING	Fan dirty, please clean Fan
45	ALERT	Fan stuck, please check Fan
46	EMERGENCY	Amplifier Shutdown caused by dc protection

NR	SEVERITY	DESCRIPTION
47	CRITICAL	Amplifier Shutdown caused by overtemperature
48	WARNING	Switching Frequency Error (Channel will be restarted)
49	EMERGENCY	PSU Shutdown caused by dc protection
50	ERROR	DC Detection not ok, syslog no longer prevented
51	ERROR	Mains Dropout Detection not ok, syslog no longer prevented

Chapter 8. RESTful API

There is a RESTful API with JSON data implemented on the device. Every Parameter can be set, and every status can be read over this Interface. All available commands are documented at REST API DOC on the webpage.

The Base URL is: **${HOST_IP}/rest-api/**.

Table 10. REST API HTTP REQUEST TYPES

TYPE	DESCRIPTION
GET	Get settings or status data
PUT	Set device/channel settings
OPTIONS	Get settings value range and unit
DELETE	Delete resource from device

For **PUT** and **DELETE** HTTP requests, an authentification TOKE in the HTTP header is required:

token:f4005bf8507999192162d989d5a60823

The command line tool **curl** can be used to execute a rest api request which allows some easy evaluation and debugging mechanism.

See some examples below.

8.1. GET DEVICE INFOMRATIONS

COMMAND	info/device
TYPE	GET
CURL-COMMAND	curl ${HOST_IP}/rest-api/info/device
RESPONSE	

```
{
    "model_name": "MA32LP2",
    "channel": 32,
    "options": ["D1","D2","IF1","M1","IF3"],
    "psu_fan": true,
    "housing_fan": true,
    "sd_card": true,
    "rtc": true,
    "sw_revision": "3.3.0",
    "fpga_revision": "2.9.1",
    "loader_revision": "2.1.4",
    "image_id": 1,
    "serial": "140619000221"
}
```

8.2. SET CHANNEL MUTE

COMMAND	settings/channel/{channel_id}/dsp/mute
TYPE	PUT
CURL-COMMAND	curl -X PUT -H 'token: f4005bf8507999192162d989d5a60823' -d "{\"value\":true}" ${HOST_IP}/rest-api/settings/channel/1/dsp/mute

On Success, the server responded with a **200** response.

On Error, the server returns a error message with a **400** response.

8.3. GET CHANNEL VOLUME OPTIONS

COMMAND	settings/channel/{channel_id}/dsp/volume
TYPE	OPTIONS
CURL-COMMAND	curl -X OPTIONS ${HOST_IP}/rest-api/settings/channel/1/dsp/volume
RESPONSE	

```
{"value": [-72.0, 24.0, 0.1 , "dB"]}
{"value": [MIN  , MAX , STEP, UNIT]}
```

8.4. REMOVE PRESET WITH NAME TEST

COMMAND	preset/storage/{preset_name}
TYPE	DELETE
CURL-COMMAND	curl -X OPTIONS ${HOST_IP}/rest-api/preset/storage/test

On Success, the server responded with a **200** response.

On Error, the server returns a error message with a **400** response. .DELETE error example

```
{
    "error": "preset not available: test"
}
```

Chapter 9. SERVICE

CAUTION - THESE SERVICING INSTRUCTIONS ARE FOR USE BY QUALIFIED SERVICE PERSONNEL ONLY. TO REDUCE THE RISK OF ELECTRIC SHOCK DO NOT PERFORM ANY SERVICING OTHER THAN THAT CONTAINED IN THE OPERATING INSTRUCTIONS UNLESS YOU ARE QUALIFIED TO DO SO.

ATTENTION - CES CONSIGNES D'ENTRETIEN DOIVENT ETRE UNIQUEMENT EMPLOYES PAR LE PERSONNEL DE SERVICE QUALIFIÉ. POUR RÉDUIRE LE RISQUE DE CHOC ÉLECTRIQUE NE PAS EFFECTUER DES REPARATIONS AUTRES QUE CEUX CONTENUS DANS LES INSTRUCTIONS D'UTILISATION A MOINS QUE VOUS SOYEZ QUALIFIE POUR LE FAIRE

9.1. FUSES

The devices contains internal fuses which are inaccessible to ordinary and instructed persons.

9.2. FIRMWARE UPDATE

It is recommended to update the software to the latest version. To keep the software up to date, see FIRMWARE UPDATE.

9.3. FILTER CLEANING

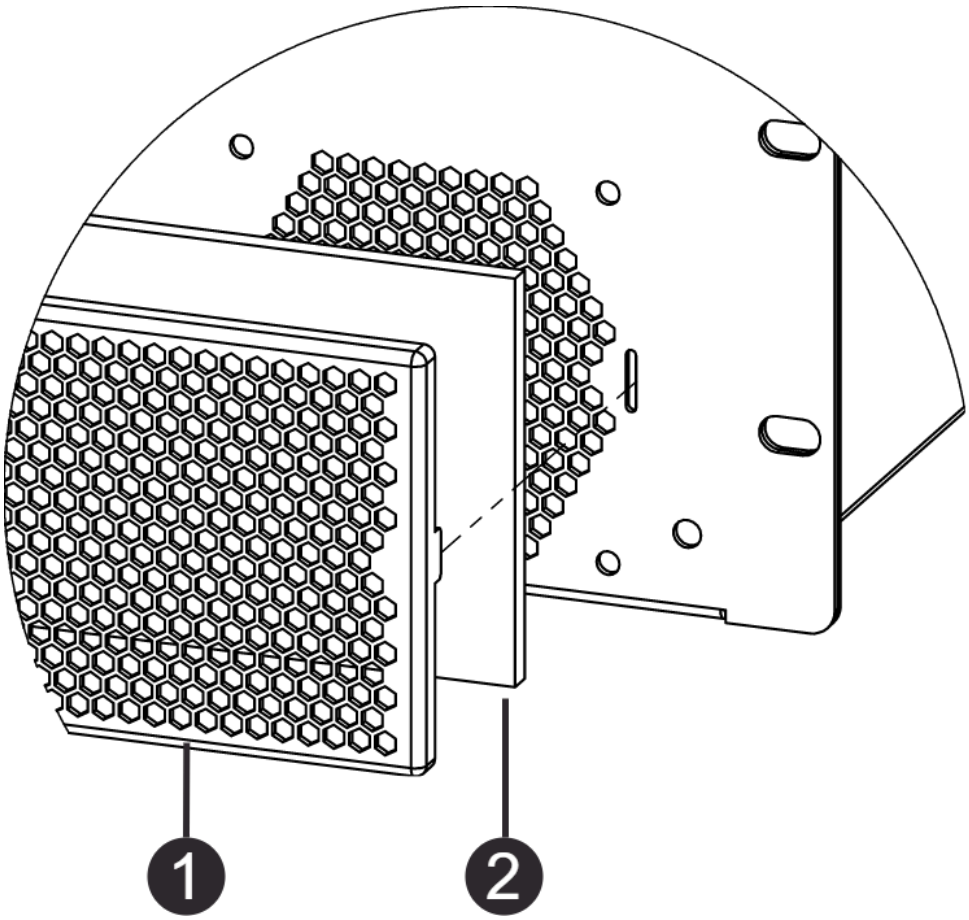


Figure 47. MAXX/HP² filter cleaning / replacement

Please clean the filter when dirty. Depending on the installation environment, a regular check is highly recommended.

Remove the grill ❶ by gently pulling on it, it is attached with magnets, no tools required. Clean the filter ❷ with compressed air and put it back together.

9.4. SPARE PARTS

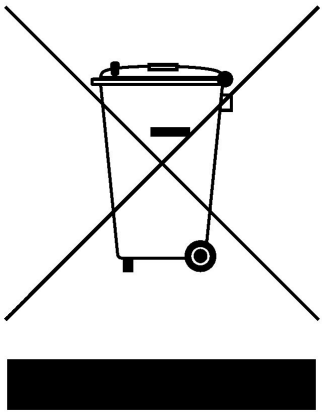
Table 11. SPARE PARTS

INNOSONIX PART NUMBER	DESRIPTION	REFERENCE
12578	2-Pol Speaker Connector	CONNECTIONS & CABLE
13403	air filter foam	FILTER CLEANING
13115	fan grill	FILTER CLEANING
13356	power cord C19 Typ E/F 2m (IEC-LOCK C13 to 3-pin Schuko CEE 7/7)	AVAILABLE POWER CORDS

INNOSONIX PART NUMBER	DESRIPTION	REFERENCE
13358	power cord C19 Typ G 2m (IEC-LOCK C13 to 3-pin GB BS 1363A)	AVAILABLE POWER CORDS

Chapter 10. DISPOSING

Electrical and electronic equipment must be disposed of separately from normal waste at the end of its operational lifetime. Please dispose of this product according to the respective national regulations or contractual agreements. If there are any further questions concerning the disposal of this product, please contact the manufacturer.



Chapter 11. EU Declaration of Conformity

The company Innosonix GmbH declares under sole responsibility that the products **MA24/HP²** and **MA32/HP²** complies with the following directives and standards

- EMC Directive **2014/30/EU**
- Low Voltage Directive **2014/35/EU**
- RoHS Directive **2011/65/EU**

11.1. EN 55032:2012

Electromagnetic compatibility of multimedia equipment - **Emission requirements:**
Radiated, Conducted: Class A Limits

11.2. EN 55103-2

EMC Compatibility – Product Family Standard for Audio, Video, Audio-Visual and Entertainment Lighting Control Apparatus for Professional Use, Part 2: Immunity

- **EN 61000-4-2:2008 Ed 2.0**
  Testing and measurement techniques - Electrostatic discharge immunity test
- **EN 61000-4-3:2010 Ed 3.2**
  Radiated, Radio-Frequency, Electromagnetic Immunity (Environment E3, criteria B)
- **EN 61000-4-4:2007**
  Radiated, Radio-Frequency, EMC Immunity (Environment E3, Criteria B)
- **EN 61000-4-5:2006**
  Surge Immunity (Criteria B)
- **EN 61000-4-6:2006**
  Immunity to Conducted Disturbances Induced by Radio-Frequency Fields (Criteria A)
- **EN 61000-4-11:2004**
  Voltage Dips, Short Interruptions and Voltage Variation

11.3. EN 62368-1:2014/AC:2015

Audio/video, information and communication technology equipment **Part 1: Safety requirements**

11.4. MANUFACTURER

Innosonix GmbH
Hauptstraße 35
D - 96482 Ahorn

Chapter 12. FCC Declaration of Conformity

This equipment has been tested and found to comply with the limits for a Class A digital device, pursuant to Part 15 of the FCC Rules. These limits are designed to provide reasonable protection against harmful interference when the equipment is operated in a commercial environment. This equipment generates, uses, and can radiate radio frequency energy and, if not installed and used in accordance with the instruction manual, may cause harmful interference to radio communications.

Operation of this equipment in a residential area is likely to cause harmful interference in which case the user will be required to correct the interference at his own expense.

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