



innosonix



MAXX-SERIES

DIGITAL POWER AMPLIFIERS

MA32/D

USER MANUAL

INDEX

1. SAFETY INSTRUCTIONS	1
1.1. GENERAL	1
1.2. INSTALLATION INSTRUCTIONS	2
1.3. VENTILATION AND COOLING	2
2. HARDWARE	3
2.1. CONTROLS	3
2.2. ETHERNET	4
2.3. AMPLIFIER	4
2.3.1. VOLTAGE, CURRENT AND POWER	4
2.4. GAIN	5
2.5. PROTECTION CIRCUITS	6
2.5.1. AMP CHIP / CHANNEL MAPPING	6
2.6. SPEAKER CONNECTION	7
3. HEAT GENERATION AND DISSIPATION	8
4. AUDIO OVERVIEW	10
4.1. INPUTS	10
4.2. OUTPUTS	11
4.3. DSP	11
4.3.1. INPUT SECTION	12
4.3.2. SIGNAL PROCESSING	13
4.3.3. SPEAKER MONITORING	13
4.4. CLOCKING	13
5. QUICKSTART	14
6. FRONT PANNEL	15
6.1. MENU STRUCTURE	15
6.2. CHANGING SETTINGS	17
6.3. DEVICE LOCK	18
7. OVERVIEW	20
8. DSP CONFIG	22
8.1. GENERAL	23
8.2. PATCH HELPER	24
8.3. FILTER	26
8.3.1. IIR	26
8.3.2. FIR	28
8.4. LIMITER	31
8.5. SPEAKER	32
8.6. COPY TO	33
8.7. PRESET	33
9. DEVICE CONFIG	35

9.1. INPUT.....	35
9.2. MIXER	36
9.3. OUTPUT.....	37
9.3.1. MADI	37
9.3.2. DANTE.....	38
9.3.3. PATCH HELPER.....	38
9.4. MUTE GROUPS.....	39
9.5. AMPLIFIER.....	41
9.6. PSU	42
9.7. FAN	42
9.8. REMOTE MUTE	42
9.9. MUTE BEHAVIOUR.....	43
9.10. HOSTNAME.....	45
10. LOCK	47
11. PRESETS.....	48
12. SOFTWARE UPDATE	50
13. SNMP.....	52
14. AES70 (OCA)	53
15. UDP DISCOVERY.....	55
15.1. DISCOVER MESSAGE	55
15.2. STATIC IP MESSAGE	56
15.3. DHCP MESSAGE	57
15.4. AUTO IP MESSAGE.....	59
16. DIMENSIONS AND WEIGHT	60
16.1. DIMENSIONS	60
16.2. WEIGHT.....	60
17. DISPOSING OF OLD APPLIANCES	61
18. EU DECLARATION OF CONFORMITY	62
18.1. EN 55032:2012.....	62
18.2. EN 55103-2	62
18.3. EN 62368-1:2014/AC:2015	63
18.4. MANUFACTURER	63

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.



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 enclosure!



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



Do not open the housing, there is no service work inside! Leave such work to a specially qualified technician.



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.

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. INSTALLATION INSTRUCTIONS

All devices can be installed in a 19-inch rack. Screw the devices at each of the two Mounting holes of the mounting bracket on the front. Use Screws with a sufficiently large head diameter and lock washers.

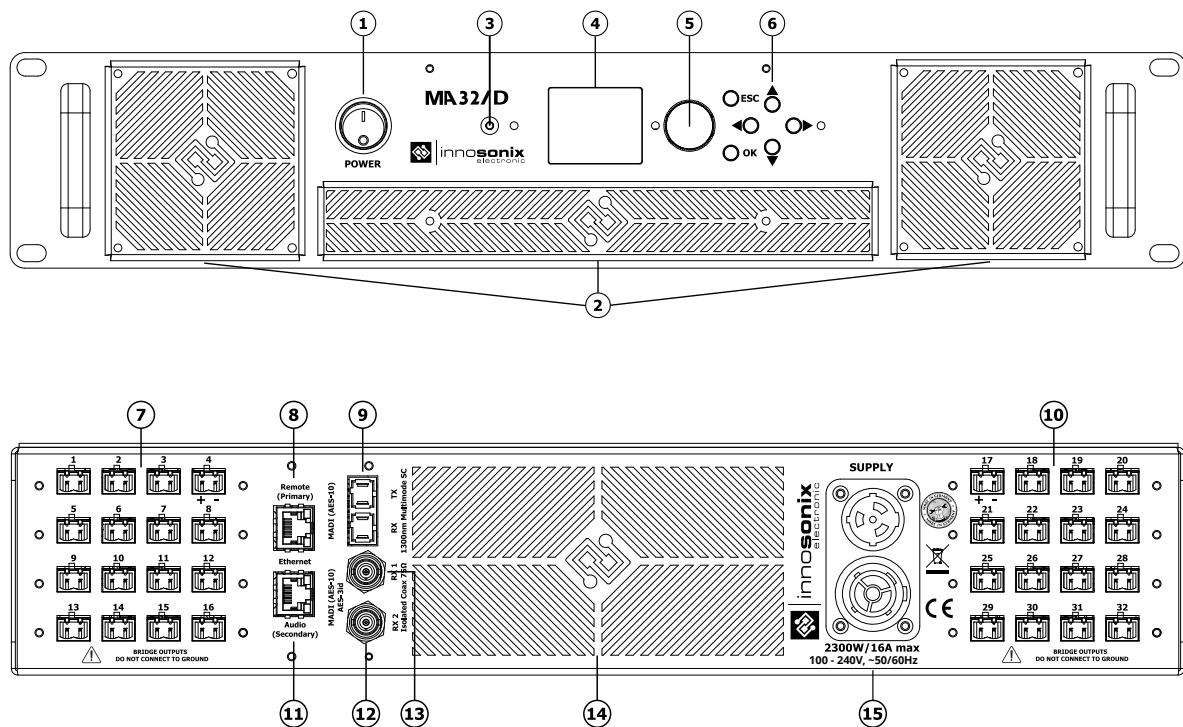
We recommend mounting the unit cantilevered. The use of guide rails and bearing surfaces can affect the convection.

1.3. VENTILATION AND COOLING

Built-in and 19-inch racks must be ventilated adequately. The passively cooled devices must not be placed directly over each other since the housing is cooled by convection. There must be at least 1U space under and above the unit.

Chapter 2. HARDWARE

2.1. CONTROLS



Element Nr.	Function
1	Power switch disconnects the unit from the mains
2	Ventilation grille. Please clean to ensure an unhindered airflow through the device
3	Power LED. Red, Yellow or Green
4	TFT Display
5	Encoder knob for adjusting values
6	Keys to enter values and menu navigation
7	Speaker terminals channel 1-16
8	Network connection for remote control and web server. 10 / 100BaseT Ethernet (DANTE Primary 1000BaseT)

Element Nr.	Function
9	MADI (AES10) Optical. SC Connector for Multimode Optics 1310nm
10	Speaker terminals channel 17 - 32
11	Network connection for audio transport (Optional). 1000BaseT Gigabit Ethernet (DANTE Secondary 1000BaseT)
12	MADI (AES10) BNC Secondary. Connection for 75Ohm coaxial cable, e.g. RG59
13	MADI (AES10) BNC Primary. Connection for 75Ohm coaxial cable, e.g. RG59
14	Ventilation grills (front to back ventilation)
15	Mains Connection. Neutrik Powercon True 1 with Loop-through facility. 90 - 240V AC 2300W or 16A max

2.2. ETHERNET

- IEEE802.3 10/100Base-T Ethernet
- Static IP-Address / Dynamic IP-Address via DHCP
- Full remote control via Web
- JSON web service for integration in media control systems
- Across subnets working UDP broadcast discovery protocol which can be used to set IP-Address
- FTP access for FIR Filter- and Presetfiles

2.3. AMPLIFIER

2.3.1. VOLTAGE, CURRENT AND POWER

This Unit uses full bridge Class-D amplifiers. Only by the extremely high efficiency of a "switched" amplifier, it is possible to accommodate 32 channels in 2U.



Due to the H-Bridge concept used, all outputs are biased by half VCC which is approximately 24V. So no common ground for outputs exist.



All outputs are internal bridged pairs, no external bridging is possible

- **Internal Operating Voltage** 48V
- **Maximum voltage at the speaker terminal** 94Vpp 33VRms
- **Maximum output current** 15A peak
- **Power** 16 Ohm → 68W RMS (1% THD+N), 140W Peak 8 Ohm → 130W RMS (1% THD+N), 280W Peak 4 Ohm → 270W RMS (1% THD+N), 530W Peak

POWER DISTRIBUTION



- The power supply can deliver 2000W continuous power. To ensure a stable operation in overload situations, the MA32 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*530W is buffered through 65.000uF of capacitors and is certainly available as a short burst only.
- An amplifier efficiency of 85% can be expected.
- All amplifiers are sourced by one powersupply.
- Assuming a evenly distribution of load between all channels. The MA32/D is able to give 50W continous sine power per channel. At 110V line it is after all 40W.
- Maximum continuous power output per channel is limited to 200W at 4 Ohms because of thermal reasons. (Continous power longer than 10 seconds). There is no limitation at 8 Ohm loads.

2.4. GAIN

32dB

Input in dBFS + 32dB = Output in dBV

Example: -12dBFS → 20dBV

2.5. PROTECTION CIRCUITS

The amplifier contains the following protection circuits to ensure safe operation under all conditions.

Protection Function	Description
Temperature	An assembly consisting of 8 channels shuts down at 85°C heat sink temperature. The remaining 24 stay active. With a Hysteresis of 25 ° C, i.e., at 60 ° C, the affected group turns on again. Each MOSFET bridge on a module has a shutdown threshold of 120 ° C. In this case, only two channels are affected, and the other 30 remains active.
Direct Current	If the polarity of the signal changes not for more than a second, the affected channel switches off and restarts again after a few seconds.
Short Circuit	The affected channel is switched off when the maximum current of 15A is exceeded. Recovery time is about two seconds.
Low Impedance	The amplifier works unconditionally stable at low impedance as long as the maximum current is not exceeded.

2.5.1. AMP CHIP / CHANNEL MAPPING

Every Amp Chip has two output channels. The assignment of Amp to Output channels is in the table below.

Internal Amplifier	Output Channel	Amplifier Module
Amp 1	24 / 29	1
Amp 2	23 / 30	1
Amp 3	22 / 31	1

Internal Amplifier	Output Channel	Amplifier Module
Amp 4	21 / 32	1
Amp 5	17 / 28	2
Amp 6	18 / 27	2
Amp 7	19 / 26	2
Amp 8	20 / 25	2
Amp 9	5 / 16	3
Amp 10	6 / 15	3
Amp 11	7 / 14	3
Amp 12	8 / 13	3
Amp 13	4 / 9	4
Amp 14	3 / 10	4
Amp 15	2 / 11	4
Amp 16	1 / 12	4

If an error occurs, the power led (element no. 3) changes from green to yellow, and the display (Item no. 4) shows detailed information. An example error message for a static error in an Amplifier channel:



2.6. SPEAKER CONNECTION

The speaker terminal is a connector from Wuerth Elektronik. It is a 2-pin connector with 5mm grid and item number 691352710002. Phoenix contact MSTB 2,5/ 2-ST - 1754449 can be used, too. In procurement problems, please feel free to contact us.

We recommend a cable with a cross-section of at least 1.0 mm² for wiring. It is not critical to install several couples of different amplifier channels in parallel.

Chapter 3. HEAT GENERATION AND DISSIPATION

The idle power dissipation of the amplifier is approx 165W.

The maximum arise of heat is about 640 W. This has to be considered in the planning of ventilation and air conditioning as "worst case".

Table 1. Dissipation loss at different sine output power levels at 230V 50Hz

Power per channel	Total output power	Active power at line	Total loss
0 W	0 W	165 W	165 W
1 W	32 W	210 W	178 W
2,5 W	80 W	270 W	190 W
5 W	160 W	370 W	210 W
7,5 W	240 W	470 W	230 W
10 W	320 W	570 W	250 W
12,5 W	400 W	670 W	270 W
15 W	480 W	770 W	290 W
17,5 W	560 W	870 W	310 W
20 W	640 W	970 W	330 W
22,5 W	720 W	1070 W	350 W
25 W	800 W	1170 W	370 W
27,5 W	880 W	1270 W	390 W
30 W	960 W	1370 W	410 W
32,5 W	1040 W	1470 W	430 W
35 W	1120 W	1570 W	450 W
37,5 W	1200 W	1670 W	470 W
40 W	1280 W	1770 W	490 W
42,5 W	1360 W	1880 W	520 W
45 W	1440 W	1980 W	540 W
47,5 W	1520 W	2080 W	560 W
50 W	1600 W	2190 W	590 W

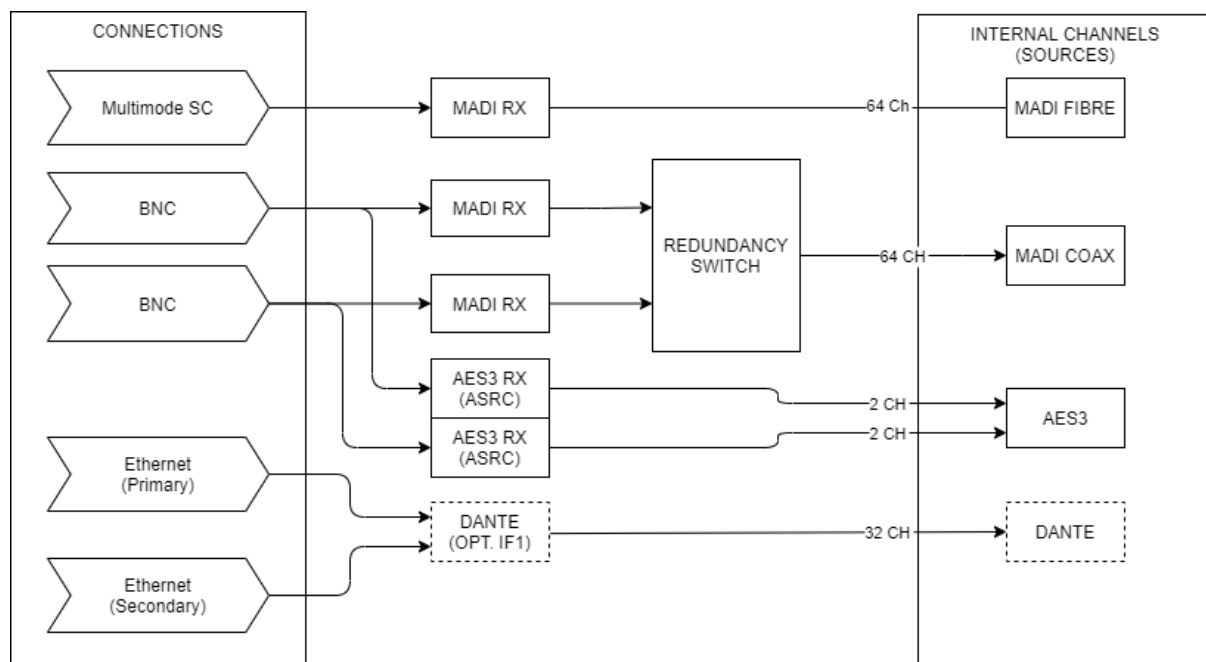
Power per channel	Total output power	Active power at line	Total loss
52,5 W	1680 W	2290 W	610 W
55 W	1760 W	2400 W	640 W

Chapter 4. AUDIO OVERVIEW

This chapter is intended to give an overview of all the available audio parts inside an MAXX-Device.

4.1. INPUTS

The MAXX devices offer various audio interfaces, like MADI and AES3, over different connectors. Both BNC connectors can receive either a MADI or AES3i signal without any user configuration. Those MADI ports also feature an integrated redundancy switch which can be used for seamless switchover if one connection fails. Redundancy mode is always enabled and appears internal like only one MADI COAX interface, the switch automatically picks the first locked port and uses it until it unlocks. Optional an Audinate Dante hardware option is available, which allows the MAXX Device to seamless integrate into Dante Networks.



Each audio input channel from the various interfaces is now called a **Source**, which means a MADI Input supplies up to 64 **Sources**, an AES3 just two.

4.2. OUTPUTS

Besides the amplifier outputs, some Audio Input Interfaces offers an Output Interface as well, e.g. MADI over Fibre or the optional DANTE interface. Due to the flexible FPGA patchbay, the user got the possibility to assign individual Sources to any possible output interface.



Each audio output channel that can transmit audio data is now called a **Destination**. A **Destination** can only be sourced by one **Source**, but one **Source** can be patched into multiple **Destination**.

4.3. DSP

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.

The internal signal path of a MAXX Device depends on the following software and hardware options:

- **Opt. M1** Speaker Monitoring

Monitoring of the connected speakers using 20kHz pilot tone to detect defective speakers or failure in the supply cable to the speaker.

- **Opt. D1** DSP

DSP functions for each channel: (EQ, High-/Lowpass, Compressor, Limiter, Volume, Delay)

- **Opt. D2** FIR Filter

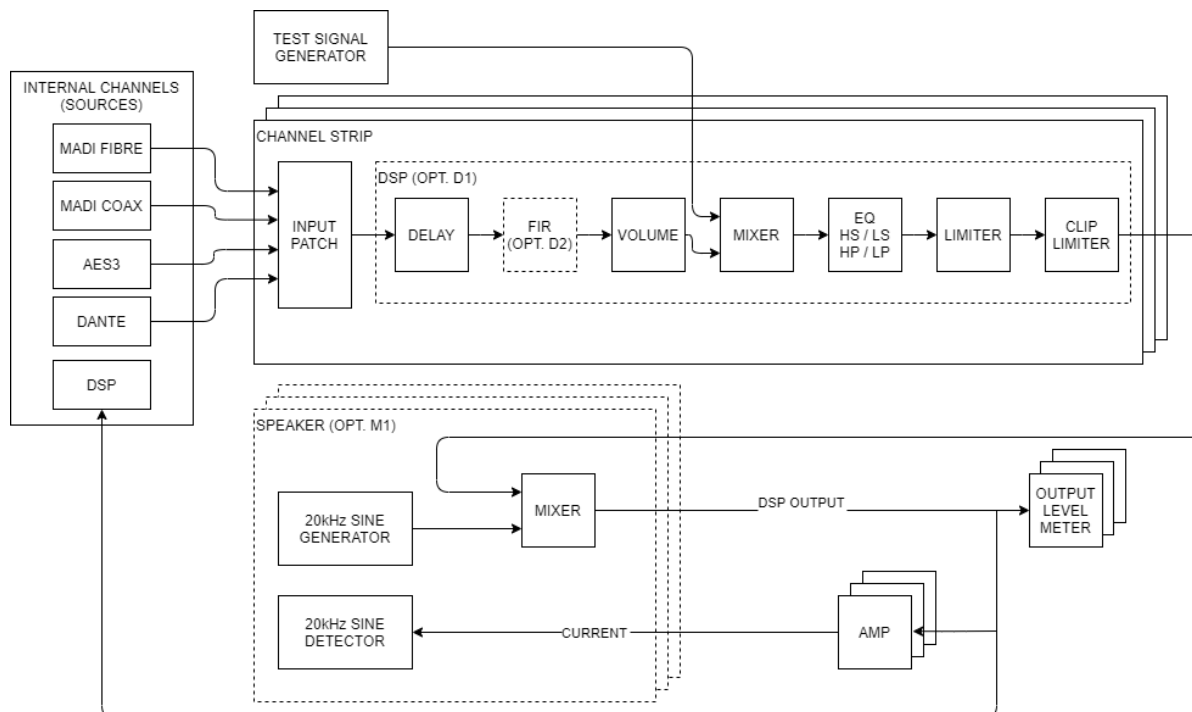
2048 tap filter / channel

- **Opt. IF1** Audinate Dante Interface

32 input channels over Dante IP-Network.

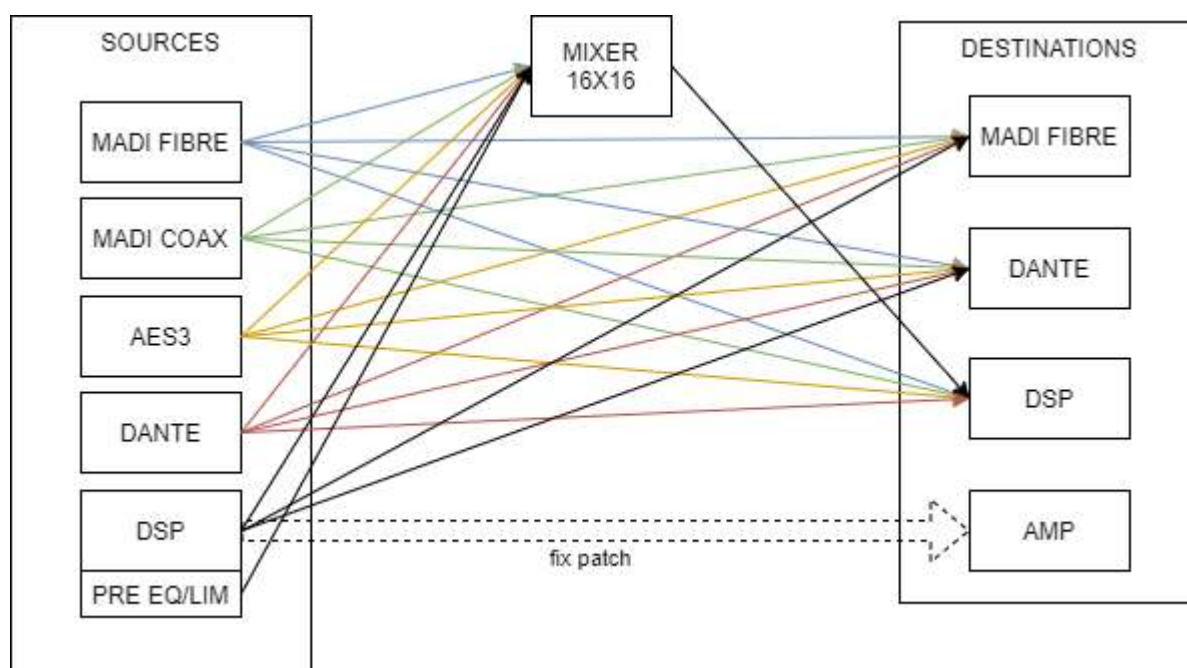


Option **D1** and **D2** can be used independently



4.3.1. INPUT SECTION

The user has the great possibility to select any of the available **Sources** as Input of a DSP channel. Let's call this a **Patch** between a **Source** and a **Destination**.



4.3.2. SIGNAL PROCESSING

The actual signal processing happens in the DSP which has the following features:

- Delay
- FIR filter
- Volume / Gain
- Test Signal Generator
- EQs / High- Lowpass / High- Lowshelf
- RMS Limiter (1:inf)
- PEAK Limiter

4.3.3. SPEAKER MONITORING

The optional Speaker Monitoring option will insert an additional 20 kHz pilot signal on the end of the DSP signal chain, which is then used to monitor your speakers' connection and voice coil.

4.4. CLOCKING

Because there is no asynchronous sample rate converter used in any of the input interfaces to keep a MAXX Device's overall latency as low as possible, a suitable audio clock source has to be selected. Every available input interface which fulfils the requirements of sample rate can synchronise the internal signal processing flow.

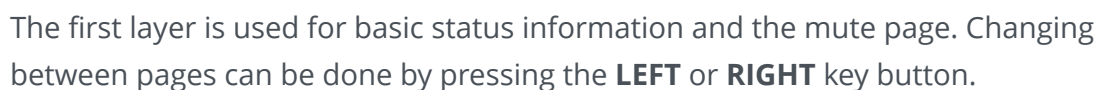
The internal clock is generated using an ultra low jitter digital DDS generator, which keeps running even if the input sync source is removed.

Chapter 5. QUICKSTART

Let's start with a short quickstart tutorial that should explain the basic workflow of working with MAXX Devices. Details of used menu pages are described in the following sections. For example, we want to patch MADI Fibre Input 1-32 to all amplifier channels 1-32.

1. connect the power cord, a MADI connection and a RJ45 ethernet cable to your remote pc or local network
2. power on the amplifier and wait for boot up is completed
3. check / set ip setting on the display: "**CONFIG** → **ETHERNET**" (see [FRONTPANNEL](#))
4. connect with a webbrowser to the amplifiers IP-Address
5. check if audio input is locked : "**DEVICE CONFIG** → **INPUT**" and set **SYNC TO** to **FIBRE** (see [INPUT](#))
6. now let's connect the audio inputs to the DSPs: **DSP CONFIG** → **PATCH HELPER**, select **MADI FIBRE** as **SOURCE** and set **INCREMENT** to **ON** because we want an incremental patch (MADI 1 → DSP1, MADI 2 → DSP2, ...)
7. set **START DSP CH.** to **1** and **END DSP CH.** to **32** and apply the patch with **SUBMIT** (see [PATCH HELPER](#))
8. check the **INPUT** column on the **OVERVIEW** page for correct channel assignment. The **LEVEL** column should indicate outgoing signals from the incoming MADI stream if present. (see [OVERVIEW](#))
9. that's it, enjoy your music

6.1. MENU STRUCTURE



Shows all DSP output levels within a range of -50dBFS to 0dBFS. A yellow colour indicates a level between -3dBFS and -0.1dBFS, whereas red indicates a full-scale 0dBFS sample. The hostname of this device is also displayed in the first line. (see [HOSTNAME](#))

Each LED indicates the corresponding speaker status depending on the settings on the **SPEAKER** page. (see [SPEAKER](#))



This page is only present if the optional **M1** feature is included.

MUTE

On this page, the master mute can be toggled by pressing the **OK** button. If **RAMP UP VOLUME AFTER MUTE** on [MUTE BEHAVIOUR](#) page is enabled, the current volume ramp state is indicated as well.

While being on the **OUTPUT LEVEL** screen, the sub-menu can be entered by pressing the **OK** button. Switching between the sub menu pages can be done by pressing the **LEFT** or **RIGHT** key.

STATUS

Lists all amplifier channel with its corresponding temperature and a status indicator

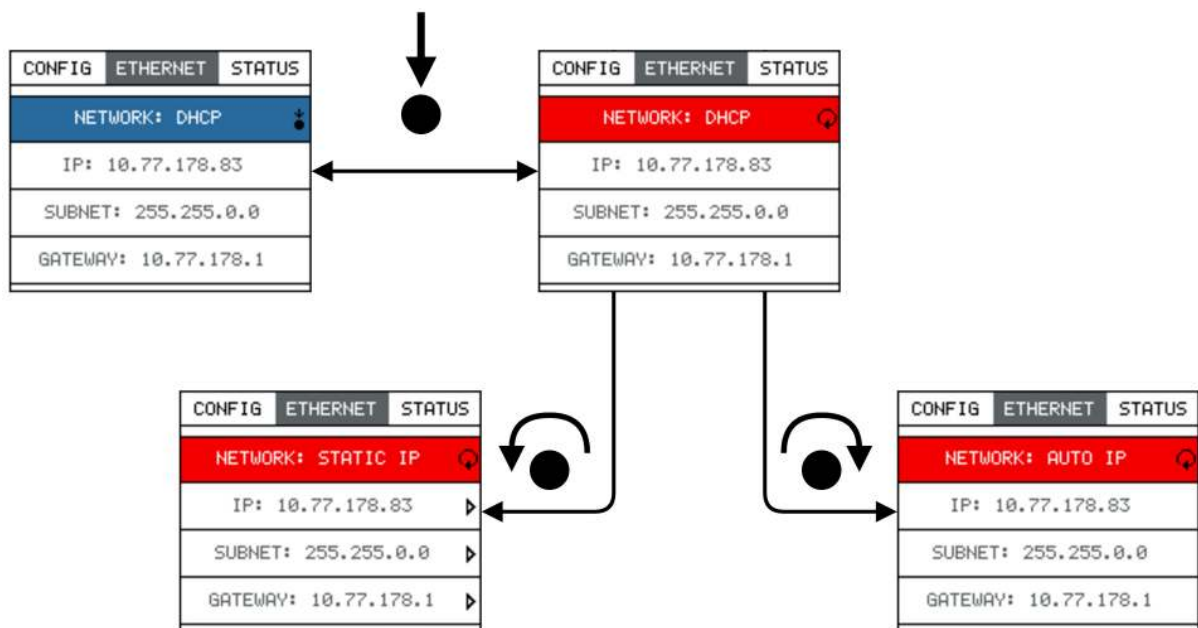
- **green checkmark** Normal operating condition.
- **red x** A protection circuit of the amplifier has triggered. That can be caused by an over current or over temperature error.
- **exclamation mark** If the temperature of an amplifier reaches its maximum operating temperature a temperature warning is triggered.

CONFIG

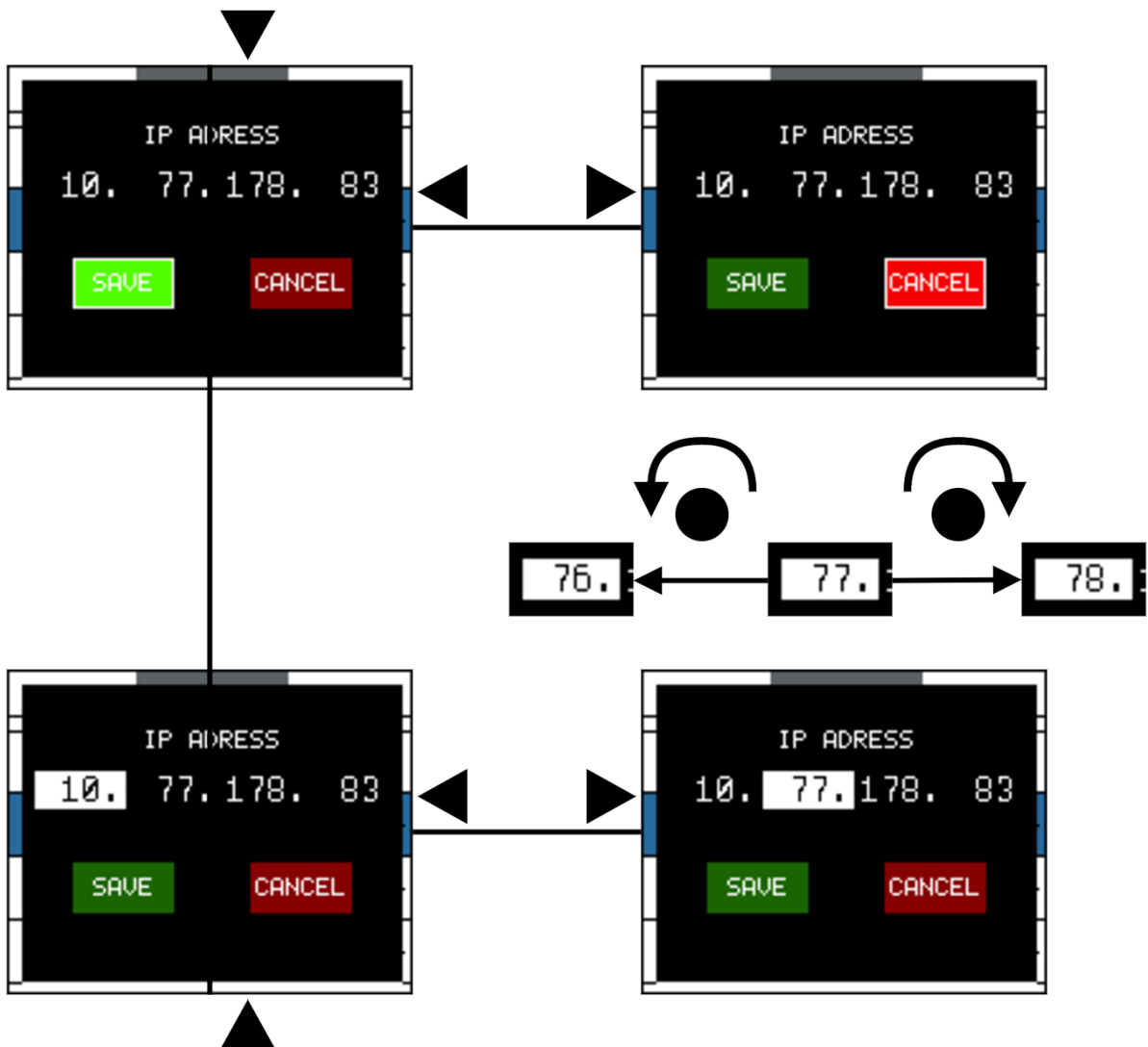
- **INPUT STATUS** Shows information about all audio input interfaces.
- **DISPLAY** The display brightness can be modified to suit individual needs. It's also possible to set a timeout, after which the brightness is set to a minimum.
- **FIRMWARE** Shows the current firmware version of this MAXX-Device and the serial number as well.
- **Ethernet**
 - **STATIC IP** user configured IP address
 - **DHCP** The MAXX-Devices looking for an available DHCP server and acquires its IP from it. If a timeout is reached, an AUTO IP fallback is implemented.
 - **AUTO IP** Auto IP acquisition as defined in RFC 3927.

6.2. CHANGING SETTINGS

To modify any of the sub menu settings, the desired entry must be selected by pressing the rotary encoder. After that, the value can be changed by rotating the encoder.

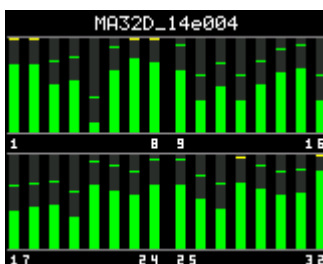


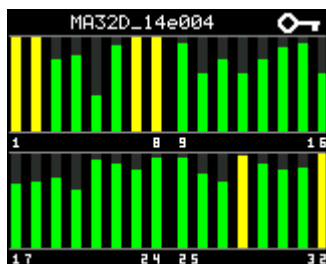
On the IP Settings screen, the currently selected section of the IP address can be changed by rotating the encoder while navigating between the entries is done by the key cross.



6.3. DEVICE LOCK

DEVICE LOCK is a simple mechanism to restrict operation on the front panel of an MAXX-Device, like a "key lock", which is indicated by a small key symbol on the upper right of the level meter page.





To toggle the **DEVICE LOCK**, hold down the **ESC** button followed by the **OK** for at least 3 sec.

Chapter 7. OVERVIEW

To keep an overview of all DSP channels and states on the MAXX-Device, the OVERVIEW page is the point to start. It shows you all the important settings and states which are useful for your daily work. Links to the corresponding DSP channels and EQ / Limiter settings are included as well, which allows quick access to the desired DSP parameter.

MASTERMUTE

REMOTE MUTE

SYNC

AMP STATUS

PSU LIMIT

MA32D_14e004

innosonix

BUILD V2.3.0-3448

SERIAL 170815000020

OVERVIEW

DSP CONFIG

DEVICE CONFIG

PRESETS

SOFTWARE UPDATE

CH	NAME	INPUT	VOLUME	LEVEL	FIR	EQ	LIMIT	GR	DELAY	AMP	SP
1	Front L	FIBRE 1	0dB	-1.3dB	Bypass	E	L	0dB	0	24.5°C	
2	Front R	FIBRE 2	0dB	-1.3dB	Bypass	E	L	0dB	0	24.5°C	
3	REAR L	FIBRE 3	-8dB	MUTE	C12	E	L	-10.6dB	0	25°C	
4	REAR R	FIBRE 4	-9.7dB	MUTE	C12	E	L	-8.6dB	0	25°C	
5	SUB	FIBRE 5	4dB	MUTE	Bypass	E	L	0dB	40	24.5°C	
6	CENTER	FIBRE 6	-17.3dB	MUTE	OMNI	E	L	0dB	8877	24.5°C	
7	STUDIO 1	FIBRE 7	0.3dB	MUTE	Bypass	E	L	0dB	0	24.5°C	
8	STUDIO 2	FIBRE 8	0.3dB	MUTE	Bypass	E	L	0dB	0	24.5°C	
9	Ch 9	FIBRE 1	-3.9dB	-4.8dB	Bypass	E	L	0dB	0	25°C	
10	Ch 10	FIBRE 2	-20dB	-20.8dB	Bypass	E	L	0dB	0	25°C	
11	Ch 11	FIBRE 3	-12.2dB	-13dB	Bypass	E	L	0dB	0	24.5°C	
12	Ch 12	FIBRE 4	-20dB	-20.8dB	Bypass	E	L	0dB	0	24.5°C	
13	Ch 13	FIBRE 5	-11.5dB	-21.8dB	Bypass	E	L	-10.9dB	2000	24.5°C	
14	Ch 14	FIBRE 6	-5.5dB	-21.3dB	Bypass	E	L	-16.4dB	2000	24.5°C	
15	Ch 15	FIBRE 7	-3.5dB	-4.4dB	Bypass	E	L	0dB	0	24.5°C	
16	Ch 16	FIBRE 8	-20dB	-20.8dB	Bypass	E	L	0dB	0	24.5°C	
17	Ch 17	DANTE 1	-16.3dB	UFL	Bypass	E	L	0dB	0	24.5°C	
18	Ch 18	DANTE 2	-14.3dB	UFL	Bypass	E	L	0dB	0	24.5°C	
19	Surround 1	COAX 19	-13.2dB	MUTE	Bypass	E	L	0dB	0	24°C	
20	Surround 2	COAX 20	-20dB	MUTE	Bypass	E	L	0dB	0	24°C	
21	Surround 3	COAX 21	-3.2dB	MUTE	Bypass	E	L	0dB	0	24.5°C	
22	Surround 4	COAX 22	-5.9dB	MUTE	Bypass	E	L	0dB	0	24.5°C	
23	FX 1	COAX 23	-8.6dB	-19.4dB	C12	E	L	-10dB	0	25°C	
24	FX 2	COAX 24	-2.3dB	-15dB	C12	E	L	-15.5dB	0	25°C	
25	Ch 25	DANTE 3	-2.3dB	UFL	OMNI	E	L	0dB	8877	24°C	
26	Ch 26	DANTE 4	-10.3dB	UFL	OMNI	E	L	0dB	8877	24°C	
27	Ch 27	COAX 1	-14.9dB	-26.7dB	OMNI	E	L	0dB	8877	24.5°C	
28	Ch 28	COAX 1	-1.4dB	-14.6dB	OMNI	E	L	-1.7dB	8877	24.5°C	
29	Ch 29	COAX 1	-5.6dB	-20.4dB	Bypass	E	L	0dB	40	25°C	
30	Ch 30	COAX 1	-8.5dB	-23.4dB	Bypass	E	L	0dB	40	25°C	
31	Ch 31	COAX 1	-7.4dB	-21dB	C12	E	L	-6.8dB	0	24.5°C	
32	Ch 32	COAX 1	5.2dB	-20.2dB	C12	E	L	-18.3dB	0	24.5°C	

FPGA: 47°C

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API REFERENCE

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Phone: +49 9561 71599-00

Innosonix GmbH

Hauptstr. 35

96102 Nümb.

Starting with overall device flags on the upper left corner:

MASTERMUTE

Indicator for the MASTER MUTE flag, RED = active. (see [MASTERMUTE](#))

REMOTE MUTE

Indicator for the REMOTE MUTE flag, RED = active. (see [REMOTE MUTE](#))

SYNC

Indicates if the selected **SYNC SOURCE** is valid concerning **SAMPLERATE** and **LOCK** state. (see [INPUT](#))

- **RED** selected sync source is **not** valid
- **GREEN** selected sync source is valid

AMP STATUS

Indicates any amplifier failures.

- **RED** failure
- **GREEN** no failure

PSU LIMIT

Indicates different states of the PSU Limiter gain reduction (see [PSU](#)):

- **RED** > -0.1dB
- **YELLOW** between -0.1 and -3dB
- **GREEN** < -3dB

HOSTNAME

The name, centred on the upper part of the overview pages is the hostname of the MAXX-Device. (see [HOSTNAME](#))

BUILD

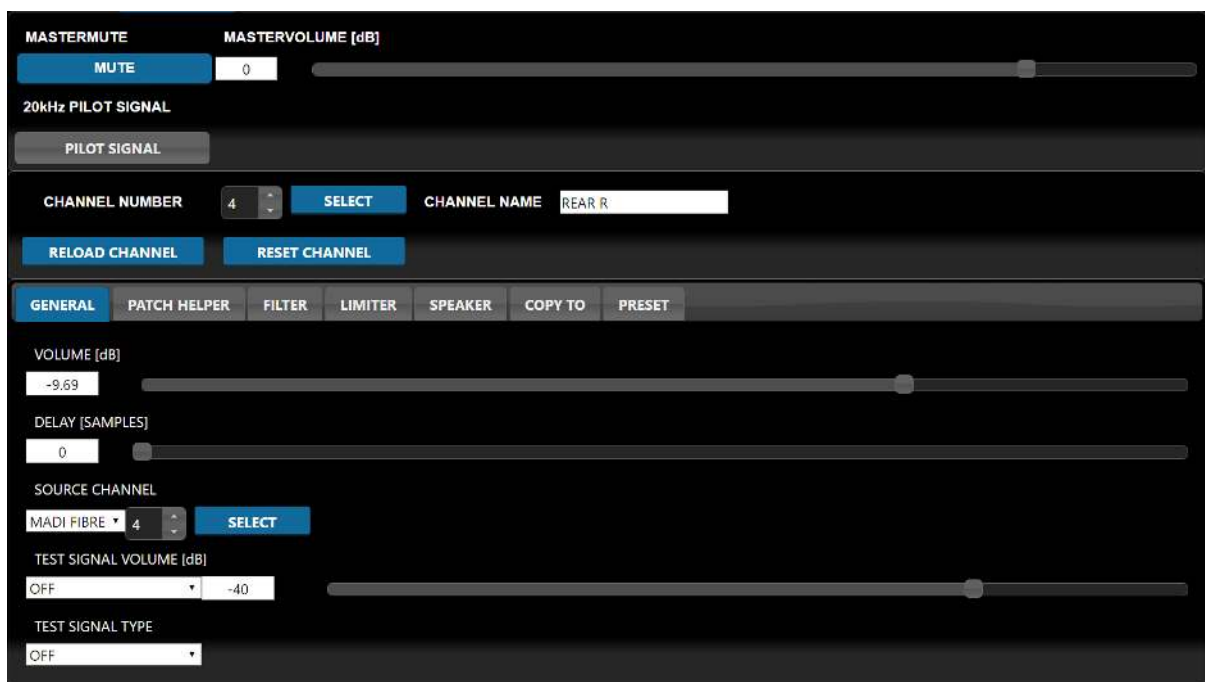
Shows the current firmware version of this MAXX-Device

SERIAL

Unique serial number of this device.

Chapter 8. DSP CONFIG

The **DSP CONFIG** tab gives access to all parameters which deals with the actual audio path inside the MAXX Device, including Patching, Volume, Filter, Limiter, Speaker Monitoring as well as Channel Preset management. **MASTERMUTE**, **MASTERVOLUME**, **PILOT SIGNAL** and **TEST SIGNAL TYPE** switch are global parameters for all DSP channels. All other settings refer to the individual selected channel.



The screenshot shows the DSP CONFIG interface with the following settings:

- MASTERMUTE:** MUTE button.
- MASTERVOLUME [dB]:** Slider set to 0.
- 20kHz PILOT SIGNAL:** PILOT SIGNAL button.
- CHANNEL NUMBER:** 4, with a SELECT button.
- CHANNEL NAME:** REAR R.
- RELOAD CHANNEL** and **RESET CHANNEL** buttons.
- Tabs:** GENERAL, PATCH HELPER, FILTER, LIMITER, SPEAKER, COPY TO, PRESET.
- VOLUME [dB]:** Slider set to -9.69.
- DELAY [SAMPLES]:** Slider set to 0.
- SOURCE CHANNEL:** MADI FIBRE, 4, with a SELECT button.
- TEST SIGNAL VOLUME [dB]:** Slider set to -40.
- TEST SIGNAL TYPE:** OFF.

MASTERMUTE

A global mute switch for all DSP channels

MASTERVOLUME

A global offset volume which affects all DSP channels.

20 kHz PILOT SIGNAL

Enables the 20 kHz pilot signal which is configured in each channel separately.



This button is only present if the optional **M1** feature is included.

CHANNEL NUMBER

Select current channel number to edit its specific settings. Selection has to be applied by pressing the **SELECT** button.

CHANNEL NAME

Enables you to enter or edit the **CHANNEL NAME** (maximum length 30 characters). The Name has to be applied by pressing the **SET NAME** button.

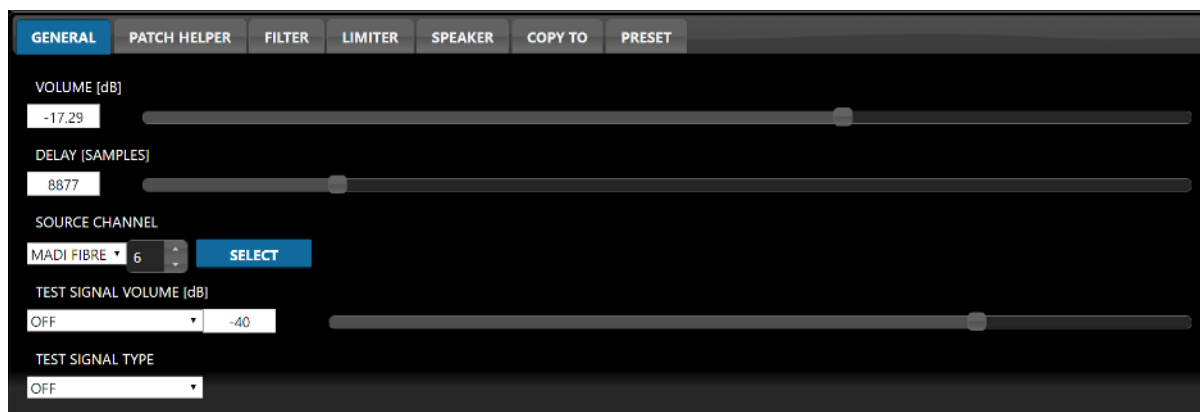
RELOAD CHANNEL

Request all information out of the amplifier again.

RESET CHANNEL

Resets all settings of the corresponding DSP channel.

8.1. GENERAL



PHASE

Reverse the Phase of each channel.

VOLUME

Volume of each channel, adjustable from -100dB to +24dB.



The total output Volume is a combination of MASTER VOLUME and the VOLUME of the corresponding channel.

MASTER VOLUME = -6dB, **VOLUME** = +2, total output volume = -4dB



Total amplification cannot exceed +24 dB. If the combination of **MASTER VOLUME** and **VOLUME** is greater than +24dB, the gain factor will be clamped.

DELAY

An independent signal delay is available for each channel to allow delay settings of up to 48000 samples.



The actual delay in ms depends on the supplied audio sample rate.

SOURCE CHANNEL

Select the desired Source channel from any available audio interface. The selection has to be confirmed by pressing the **SELECT** button.



There also can be selected **MIXER** as input source. (For further details see [MIXER](#))



Be aware that the selected **SOURCE CHANNEL** is synchronous to the selected **SYNC SOURCE** on [INPUT](#) page, otherwise audio artifacts are possible!

TEST SIGNAL VOLUME

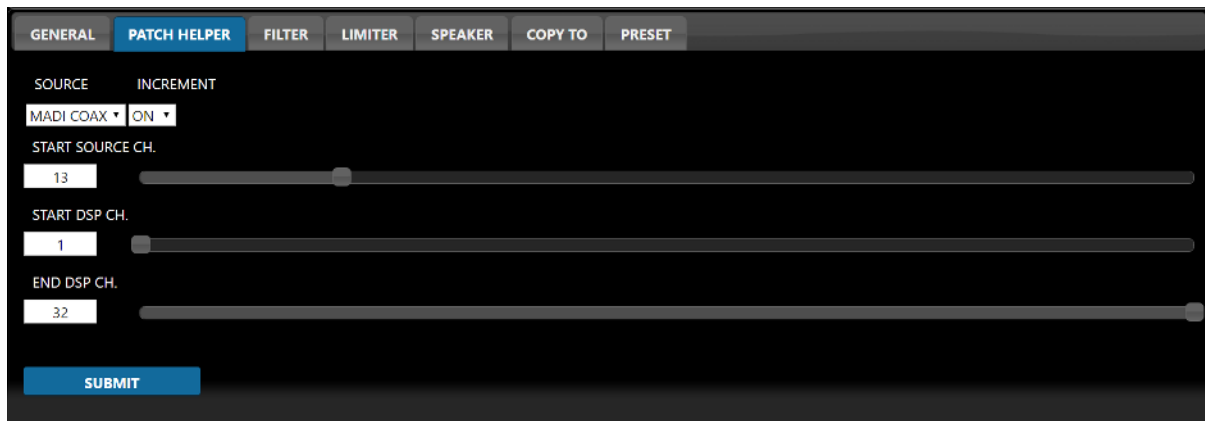
A global test signal can be enabled on each channel with individual output volume.

TEST SIGNAL TYPE

Only one test signal generator is present inside the FPGA, so the signal type shared between all DSP channels.

8.2. PATCH HELPER

This **PATCH HELPER** feature allows the user to assign incremental channel ranges to the DSP inputs quickly.



SOURCE

In combination with **START INPUT CH.**, choose from which interface the channels should be supplied.

INCREMENT.

If the option is **ON**, the Source channel will be incremented during patching. If deactivated the same source channel will be patched to all selected DSP channels.

Table 2. Example: Increment Ch. active

Source Ch. →	DSP CH.
MADI COAX CH. 13	1
MADI COAX CH. 14	2
.....
MADI COAX CH. 44	32

Table 3. Example: Increment Ch. not active

Source Ch. →	DSP CH.
MADI COAX CH. 13	1
MADI COAX CH. 13	2
.....
MADI COAX CH. 13	32

START DSP CH. / END DSP CH.

Selects the range of DSP channels to be patched.

SUBMIT

To apply the selected configuration you have to press **SUBMIT**



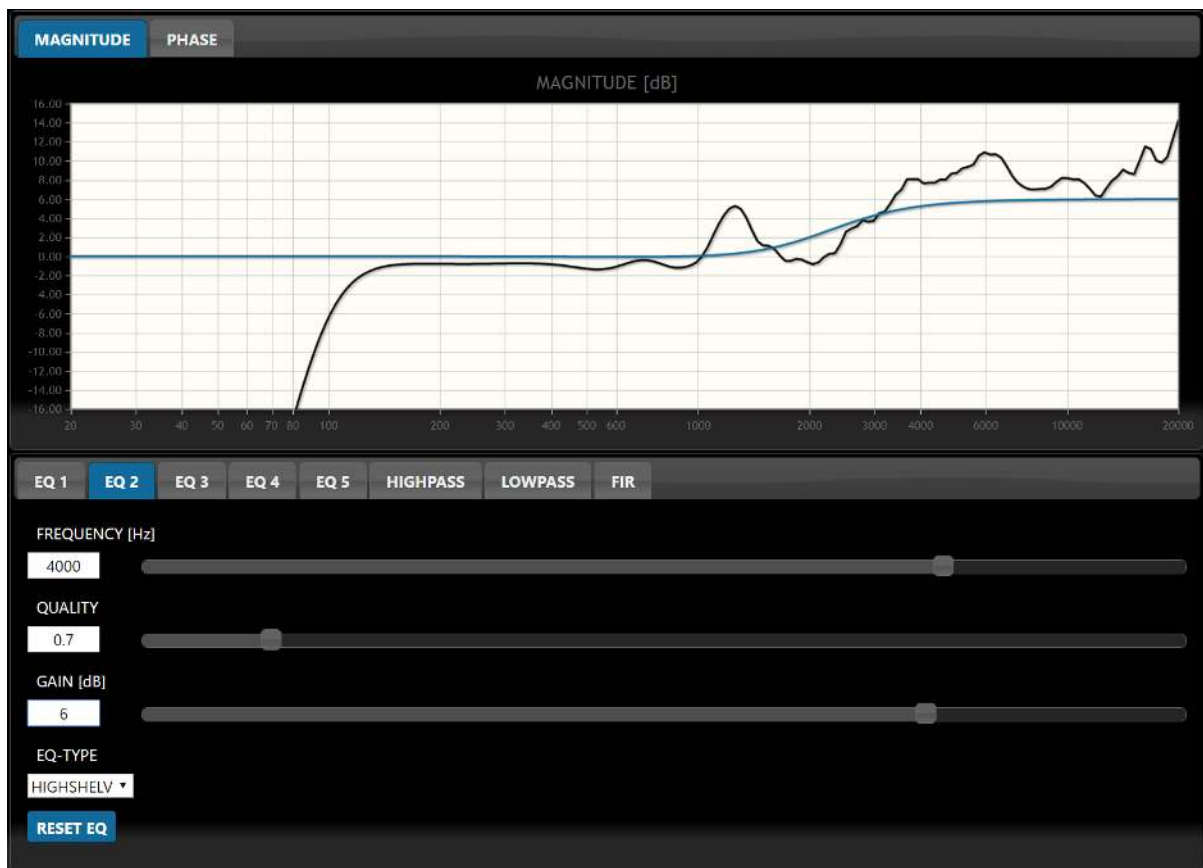
The patch will be applied **immediately**. For this reason, it can lead to immediate level changes at the speaker outputs depending on the level of the newly selected Source channels.

8.3. FILTER

The **FILTER** tab gives an overview of all enabled filters and allows modifying their parameter. The resultant magnitude is shown in the plot window in black. The currently selected filter is printed blue.

8.3.1. IIR

This section provides control over the included minimum phase biquad IIR filters, which can be configured to different filter types, including high and low pass types.



EQ-TYPE

Choose a type, additional parameters will show up depending on the type. EQ-Types can be:

- OFF
- BELL (peak filter)
- NOTCH
- HIGHSHELV
- LOWSHELV

HIGHPASS-TYPE / LOWPASS-TYPE

Choose a type, additional parameters will show up depending on the type. HIGHPASS-/LOWPASS-Types can be:

- OFF
- 6 dB/Oct
- 12 dB/Oct Bessel / Butterworth / Linkwitz-Riley
- 18 dB/Oct Bessel / Butterworth
- 24 dB/Oct Bessel / Butterworth / Linkwitz-Riley
- 48 dB/Oct Bessel / Butterworth / Linkwitz-Riley
- Variable-Q

FREQUENCY

Filter frequency (center / corner frequency), adjustable from 20Hz to 20kHz.

QUALITY

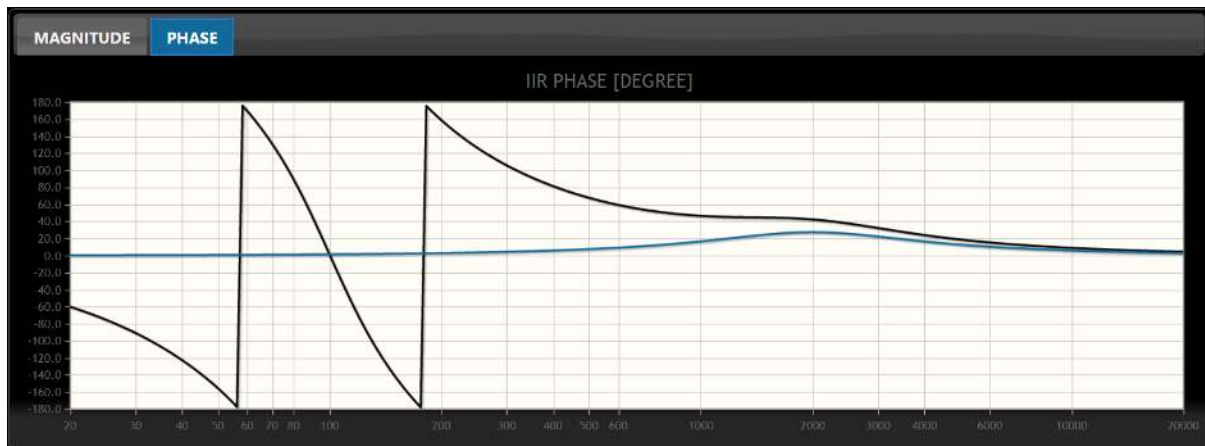
Q of the filter, adjustable from 0.1 to 5.0 in 0.1 steps.

GAIN

Gain, adjustable from -12dB to +12dB in 0.1dB steps

RESET EQ

Resets all settings of the selected filter right away.



The phase plot only shows the IIR phase but not the FIR phase. It's due to the possibility of mixing minimum phase IIR and FIR filters which can have a linear phase; the resulting phase plot wouldn't give any useful information.

8.3.2. FIR



This tab is only present if the optional **D2** feature is included.

Any FIR filter that is uploaded to the system can be assigned to any DSP channel. Filters are based on coefficient files supplied by the user with a maximum length of 2048 taps. The coefficients inside the file represent the desired filter's impulse response and are convolved with the input sample data.



One coefficient per line, no separation chars.



Coefficient representation is float.

File extension has to be .fir



The filter file is automatically padded with zeros or truncated to fit into the internal FIR unit.

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

Table 4. Coefficient Example1

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)

Table 5. Coefficient Example2

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)



The actual coefficient file only has to include the values out of the coefficient column from the two examples above.



ACTIVE FILTER

Indicates the currently assigned FIR filter name to this channel.



The filter's name will be saved to the channel, so there must be an FIR Filter in the list below, with the exact Name. If the File is deleted, the FIR Filter will be bypassed at the next startup.

LOAD SELECTED

After selecting the desired filter in the list, it gets finally loaded to hardware with this button.

BYPASS FILTER

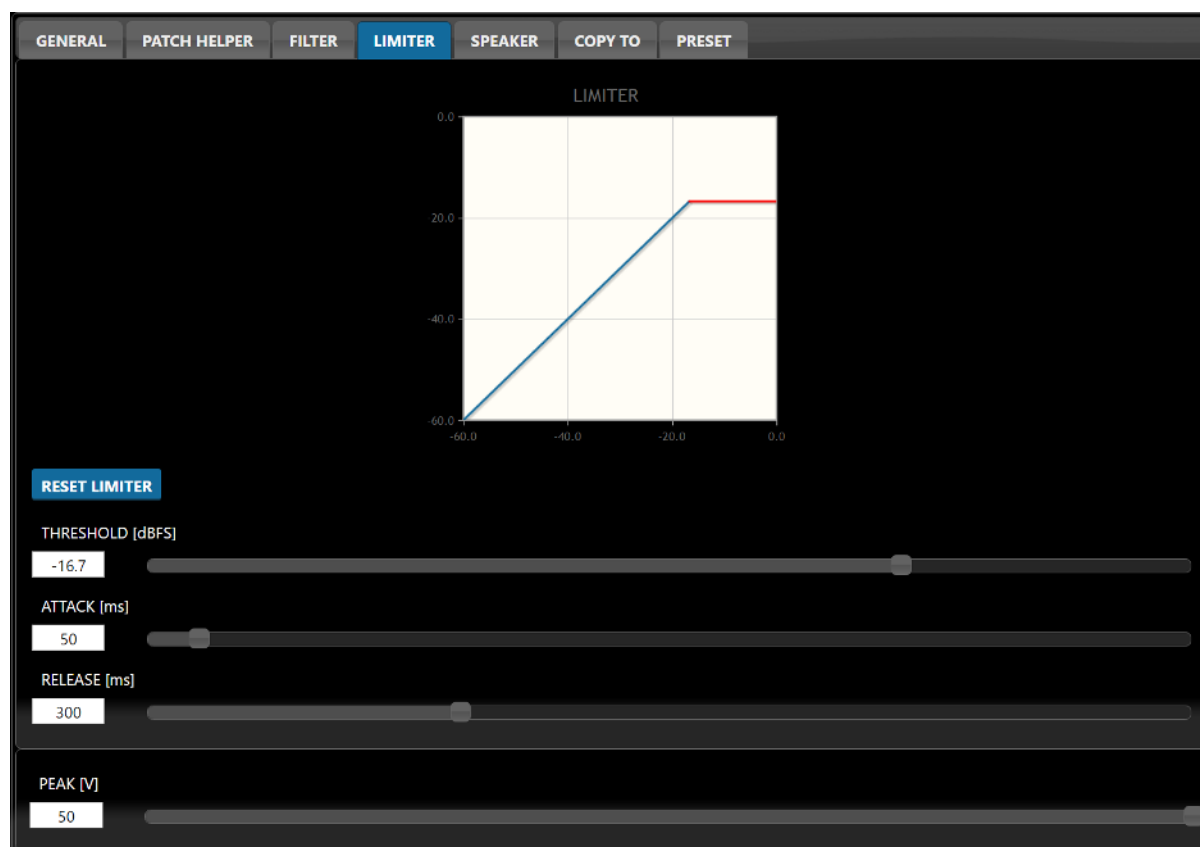
Loads a simple bypass filter in the FIR unit, to deactivate any filter behaviour. All coefficients are set to zero except the first one, which is set to 1.0.

DELETE SELECTED

Removes the filter definition from the file system.

8.4. LIMITER

In the **LIMITER** tab, parameters for the RMS-Limiter and Peak-Limiter can be changed.



THRESHOLD

Threshold in dbFS, which the RMS limiter tries to reach.

ATTACK

Time in ms in which the Limiter damps the DSP output signal to the set **THRESHOLD** if it exceeds it.

RELEASE

Time in ms in which the Limiter releases the gain damping factor to 0dB if the input level drops below **THRESHOLD**

PEAK

This parameter belongs to the PEAK-Limiter, which performs hard truncation of the output signal if it would exceed the set output level in Volts.



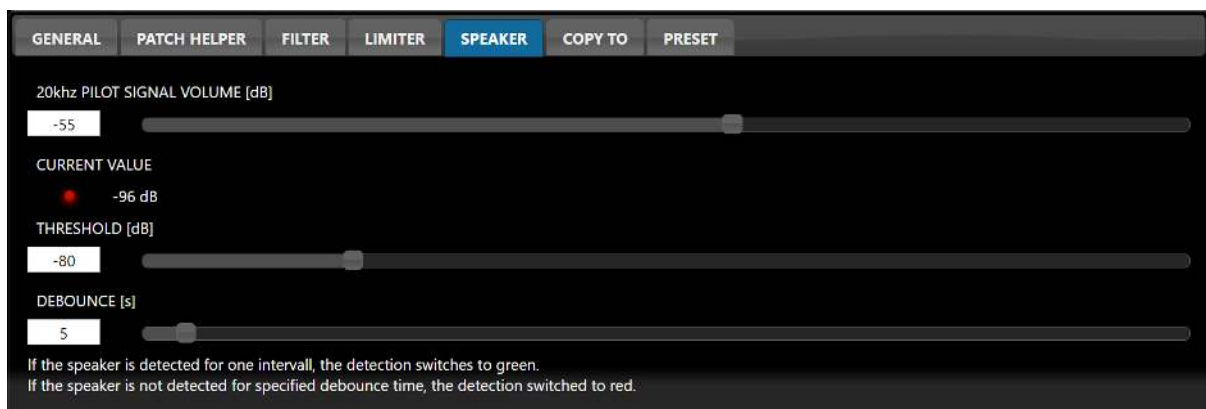
Due to hard intervention into the output waveform, this type of limiting produces many harmonics that may become audible. So this parameter is only intended as a final speaker protection mechanism to prevent damage resulting from overcurrent.

8.5. SPEAKER

The output current of all channels is measured and fed to a 20kHz detector to monitor your speakers' connection and voice coil.



The tab is only present if the optional **M1** feature is included.



PILOT SIGNAL VOLUME

Volume of the 20kHz pilot signal which can be inserted at the end of the DSP chain

CURRENT VALUE

This value indicates the actual output current on 20kHz. A LED indicator next to it shows the speaker connection state:

- **RED** not connected or broken
- **GREEN** OK

THRESHOLD

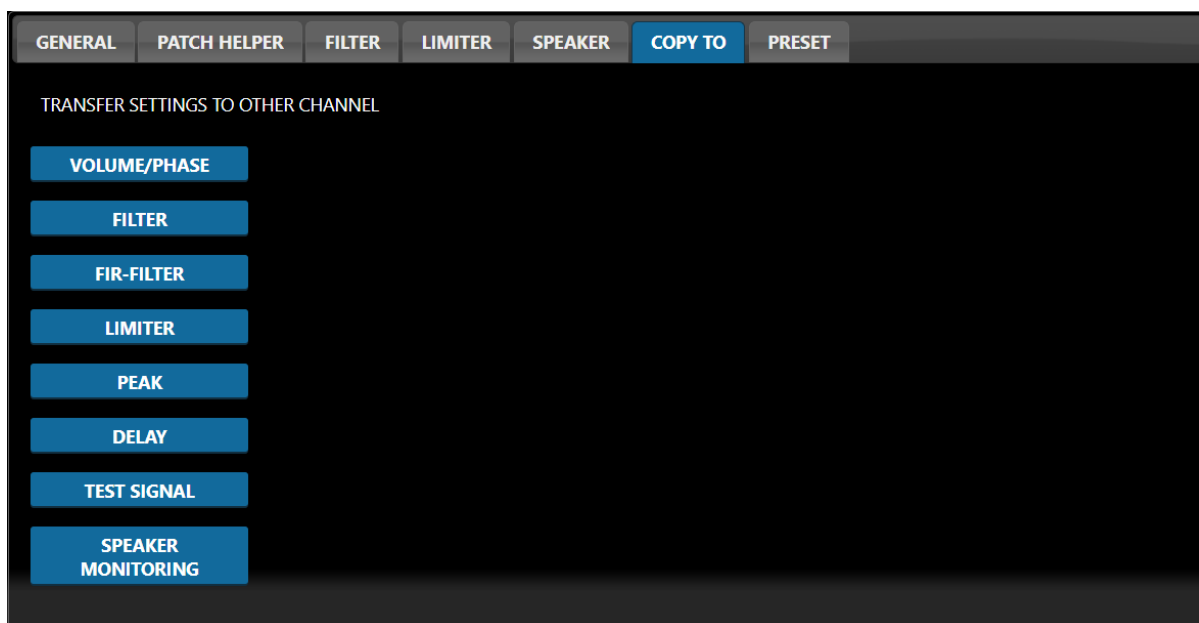
Threshold of the output current to trigger the speaker state to change to OK

DEBOUNCE

The **CURRENT VALUE** has to fall below the **THRESHOLD** for at least **DEBOUNCE** period to assume a connected speaker is in bad condition.

8.6. COPY TO

The **COPY TO** tab offers a convenient way to quickly transfer DSP settings from one channel to ALL others. It's sometimes useful for resetting or set an individual parameter on all channels at the same time. So, configure the desired parameters on the currently selected channel and copy them to all. A Popup appears where you have to confirm to prevent accidentally overriding wanted settings.

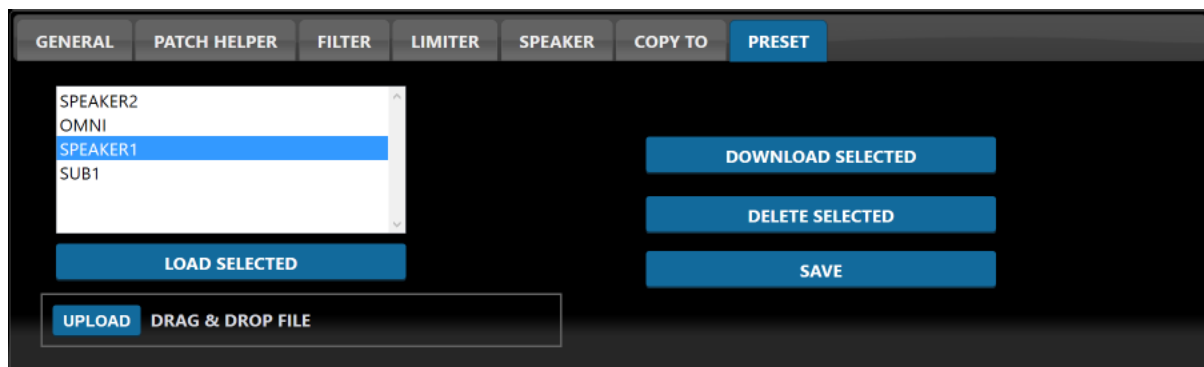


8.7. PRESET

This **PRESET** tab allows the user to save a complete channel config including EQ, Delay, Limiter settings, loaded FIR-Filter and all other parameters in the **DSP CONFIG** tab except the **INPUT CHANNEL** setting.



Channel **PRESET** are intended to store information about specific speaker types which can be loaded into each channel such a speaker is connected.



Chapter 9. DEVICE CONFIG

9.1. INPUT

The **INPUT** section gives a status overview of all audio interfaces.

INPUT	MIXER	OUTPUT	MUTE GROUPS	AMPLIFIER	PSU	FAN	REMOTE MUTE	MUTE BEHAVIOUR	HOSTNAME	LOCK
SYNC TO DANTE	TYPE		STATUS	SAMPLINGRATE		CHANNEL				
	MADI COAX 1		UNLOCK							
	MADI COAX 2		UNLOCK							
	MADI FIBRE		LOCK	48kHz		64				
	AES3 1 (ASRC)		UNLOCK							
	AES3 2 (ASRC)		UNLOCK							
	DANTE		SYNC	48kHz		32				

SYNC TO

Select the interface which is used to synchronise the internal audio clock.

TYPE

Due to the redundancy feature of the **MADI COAX** input (see [INPUTS](#)), only one connection can supply audio channels for internal use. If both **MADI COAX** ports are connected, the currently active port is indicated by white colour while the passive port is marked in grey.

STATUS

Shows the current state of this interface.

- **SYNC** Valid carrier and internal clock synced to interface
- **LOCK** Valid carrier and frame detected.
- **UNLOCK** No carrier or frame detected.

SAMPLINGRATE

Shows the actual samplingrate of the corresponding interface.

CHANNEL

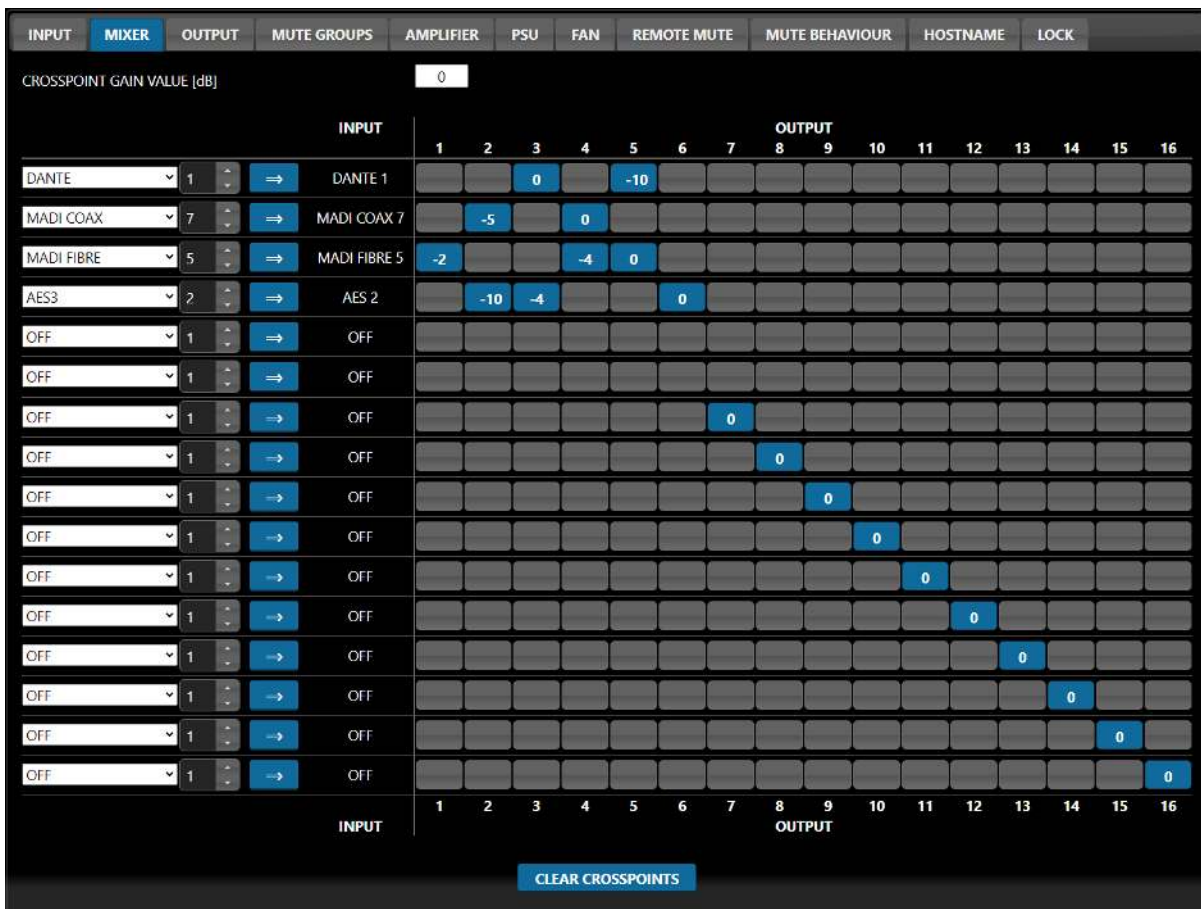
Shows the number of available channel of the corresponding interface:

- **MADI COAX/FIBRE** indicates the actual frame length 56CH or 64CH
- **AES3 1/2** 2CH

- **DANTE** 32CH or 64CH (only with **IF1** or **IF2**, see [DSP](#))

9.2. MIXER

The **MIXER** section is used to mix several inputs to Mixer channels, which can be used as input on channels. (see [INPUT CHANNEL](#))



CROSSPOINT GAIN VALUE [dB] 0

INPUT		OUTPUT															
		1	2	3	4	5	6	7	8	9	10	11	12	13	14	15	16
DANTE	1			0		-10											
MADI COAX	7		-5		0												
MADI FIBRE	5	-2			-4	0											
AES3	2		-10	-4			0										
OFF	1							0									
OFF	1								0								
OFF	1									0							
OFF	1										0						
OFF	1											0					
OFF	1												0				
OFF	1													0			
OFF	1														0		
OFF	1															0	
OFF	1																0

CLEAR CROSSPOINTS

INPUT

There are 16 **INPUTS**, which can be assigned to the Mixer. Input devices depend on your hardware and software options. To assign an **INPUT**, select a Device and Channel and assign it with the ⇒ Button.

OUTPUT / CROSSPOINTS

There are 16 **OUTPUTS**, which depends on set Crosspoints. To set a crosspoint, type in a **CROSSPOINT GAIN VALUE** and press on a grey, not activated, crosspoint button. A click on an activated Crosspoint deactivates it. To change the volume, deactivate the Crosspoint and activate it again as described above.

The **CLEAR CROSSPOINTS** Button deactivates all crosspoints and clears the gain value.

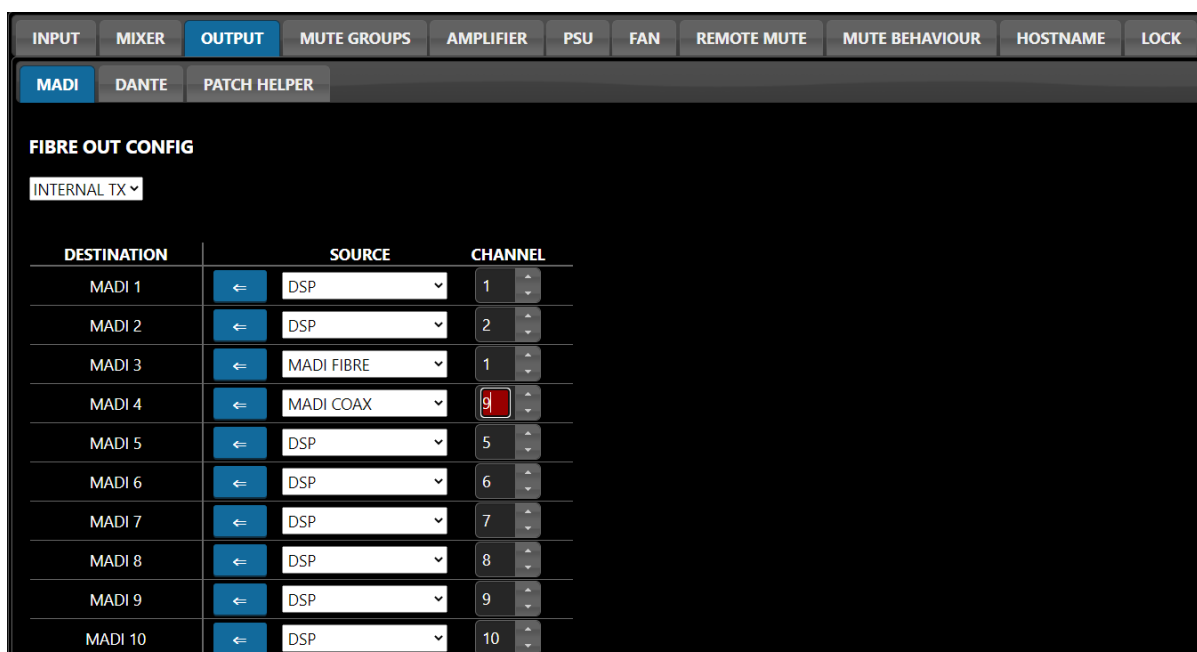
9.3. OUTPUT

The **OUTPUT** section is used to assign specific channels to the audio transmitter interfaces.



This is NOT the amplifier channel assignment!

9.3.1. MADI



DESTINATION		SOURCE	CHANNEL
MADI 1	←	DSP	1
MADI 2	←	DSP	2
MADI 3	←	MADI FIBRE	1
MADI 4	←	MADI COAX	9
MADI 5	←	DSP	5
MADI 6	←	DSP	6
MADI 7	←	DSP	7
MADI 8	←	DSP	8
MADI 9	←	DSP	9
MADI 10	←	DSP	10

In the MADI tab, the channel assignment for the optical MADI transmitter port can be made. Each output channel can be assigned individually from all available Sources inside the device.

FIBRE OUT CONFIG

image::/builds/maxx/manual_v2/source/includes/chapters/web/pictures/device_config/madi_tx_routing.svg[]

The **FIBRE OUT CONFIG** allows the user to use the MAXX-Device as a simple media converter without any need for additional hardware. Where the MADI FIBRE TX interface can either be driven by the internal MADI transmitter or bypass any of the available MADI RX interfaces electrically.



This feature allows daisy-chain multiple devices together to utilise all 64 channel from a single MADI stream.



FIBRE OUT CONFIG does not influence the available input interface.

PATCH (⇐)

Allows the individual channel assignment for the internal MADI transmitter, only available if **FIBRE OUT CONFIG** is set to **INTERNAL TX**. Each patch has to be applied by pressing the ⇐ button. To easily patch an incremental channel range, the **PATCH HELPER** feature is also available.



Unapplied patches are marked with a RED channel selector.

9.3.2. DANTE

INPUT

MIXER

OUTPUT

MUTE GROUPS

AMPLIFIER

PSU

FAN

REMOTE MUTE

MUTE BEHAVIOUR

HOSTNAME

LOCK

MADI

DANTE

PATCH HELPER

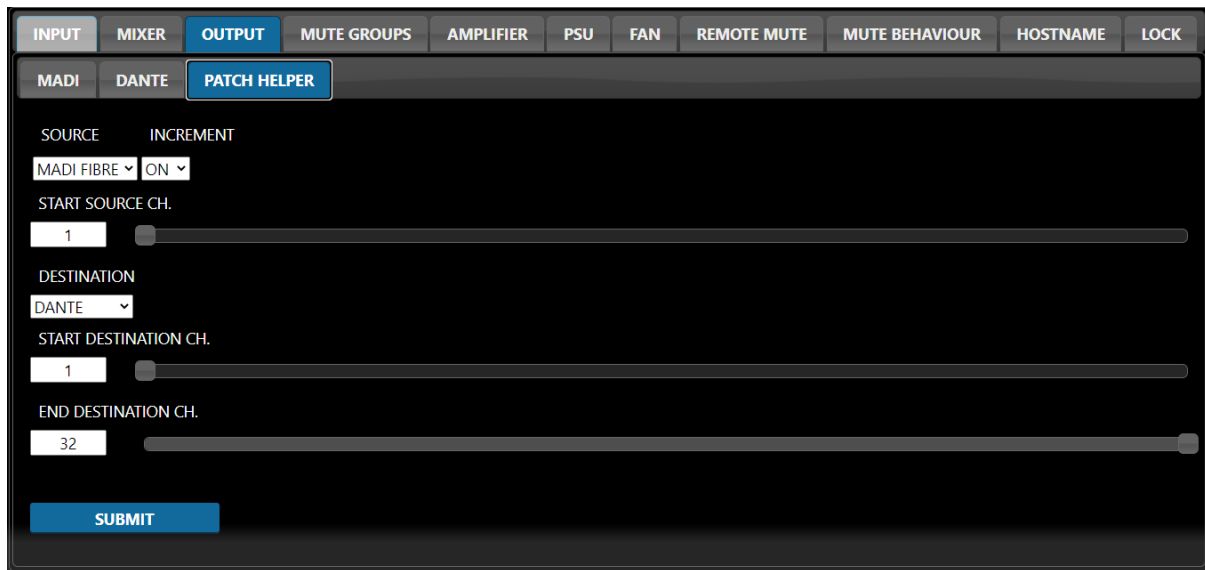
DESTINATION		SOURCE	CHANNEL
DANTE 1	⇐	MADI FIBRE	1
DANTE 2	⇐	MADI FIBRE	2
DANTE 3	⇐	MADI FIBRE	3
DANTE 4	⇐	MADI FIBRE	4
DANTE 5	⇐	MADI FIBRE	5
DANTE 6	⇐	MADI FIBRE	6
DANTE 7	⇐	MADI FIBRE	7
DANTE 8	⇐	MADI FIBRE	8
DANTE 9	⇐	MADI FIBRE	9
DANTE 10	⇐	MADI FIBRE	10



Only available if the optional **IF1** feature is included.

Offers the output channel assignment for the **DANTE** interface which is very similar to the [MADI](#) page.

9.3.3. PATCH HELPER



The screenshot shows the **PATCH HELPER** configuration page. It features a top navigation bar with tabs: INPUT, MIXER, OUTPUT, MUTE GROUPS, AMPLIFIER, PSU, FAN, REMOTE MUTE, MUTE BEHAVIOUR, HOSTNAME, and LOCK. Below this, there are sub-tabs: MADI, DANTE, and PATCH HELPER. The PATCH HELPER section includes the following controls:

- SOURCE**: A dropdown menu set to **MADI FIBRE**.
- INCREMENT**: A dropdown menu set to **ON**.
- START SOURCE CH.**: A slider control with a value of **1**.
- DESTINATION**: A dropdown menu set to **DANTE**.
- START DESTINATION CH.**: A slider control with a value of **1**.
- END DESTINATION CH.**: A slider control with a value of **32**.
- SUBMIT**: A blue button at the bottom.

This section is very similar to the [PATCH HELPER](#) feature in the **DSP CONFIG** tab, except the **DESTINATION** can be chosen as well.

9.4. MUTE GROUPS



The screenshot shows the **MUTE GROUPS** configuration page. It features a top navigation bar with tabs: INPUT, MIXER, OUTPUT, MUTE GROUPS, AMPLIFIER, PSU, FAN, REMOTE MUTE, MUTE BEHAVIOUR, HOSTNAME, and LOCK. Below this, there are sub-tabs: MADI, DANTE, and MUTE GROUPS. The MUTE GROUPS section includes the following controls:

- AUTO CLEAR**: A blue button.
- Mute Group Headers**: A row of buttons for groups 1 through 8, each with a label (FRONT, REAR, SUB, ALL, CENTER, STUDIO, SURROUND, Group 8) and a state indicator (SOLO or MUTE).
- Channel Matrix**: A table with 10 rows (CH. 1 to 10) and 8 columns (1 to 8). Each cell contains a crosspoint indicator (a red dot) and a toggle button (blue for SOLO, red for MUTE).

On the **MUTE GROUPS** page, the assignment for eight mute groups can be made. An indicator of the current mute state of each group is present as well. The channel assignment to a mute group can be made by toggling the crosspoint in the matrix view. Once the channel assignment is complete, each group's state can be changed by the buttons on the web page or via the control protocol, which is, for example, controlled by

an external media control device.

NR. / NAME

Name and number of this DSP channel for an easy assignment.

GROUP NR / NAME

Mute Groups can have individual names as well. Change the name in the text box and confirm with Enter key.

MUTE INDICATOR

The little-led indicator next to the channel name shows the actual mute state of this DSP channel. It's the result of various combinations of **MUTE** and **SOLO** states of the assigned groups.

- **RED** muted
- **GRAY** not muted

MUTE

The MUTE button itself is used to toggle the state of its groups as well as an indicator of its current state.

- **RED** muted
- **BLUE** not muted



MUTE features the highest priority and cannot be toggled by any other event.

SOLO

The **SOLO** button temporarily enables only the assigned DSP channels in this group and mutes all other groups. Only DSP channels assigned to a **MUTE GROUP** are affected by an active **SOLO** button.



SOLO features a lower priority than **MUTE**, which means that DSP channels with an active **MUTE** stay muted even if another group is assigned to features a **SOLO** state.

AUTO CLEAR

The **SOLO** state can be active in multiple groups simultaneously. The **AUTO CLEAR** functionality changes this behaviour and automatically deactivates all remaining **SOLO** buttons if a new one is activated.

9.5. AMPLIFIER

INPUT

MIXER

OUTPUT

MUTE GROUPS

AMPLIFIER

PSU

FAN

REMOTE MUTE

MUTE BEHAVIOUR

HOSTNAME

LOCK

CH.	NAME	STATE
1	Front L	ON
2	Front R	ON
3	Rear L	OFF
4	Rear R	OFF
5	Sub	ON
6	Center	ON
7	Studio 1	ON
8	Studio 2	ON
9	Ch 9	OFF
10	Ch 10	OFF
11	Ch 11	OFF
12	Ch 12	OFF
13	Ch 13	OFF
14	Ch 14	OFF
15	Ch 15	OFF
16	Ch 16	OFF

CH.	NAME	STATE
17	Ch 17	OFF
18	Ch 18	OFF
19	Surround 1	OFF
20	Surround 2	ON
21	Surround 3	ON
22	Surround 4	OFF
23	FX 1	OFF
24	FX 2	OFF
25	Ch 25	OFF
26	Ch 26	OFF
27	Ch 27	OFF
28	Ch 28	OFF
29	Ch 29	OFF
30	Ch 30	OFF
31	Ch 31	OFF
32	Ch 32	OFF

ENABLE ALL

DISABLE ALL

To reduce the idle power and the idle noise, each unused amplifier channel can be disabled separately.

STATE

The button can power down the amplifier channel and used as an indicator of the actual state.

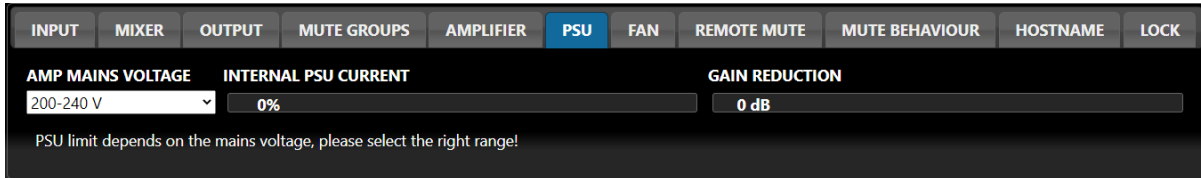
ENABLE ALL

Powers on all amplifier channel.

DISABLE ALL

Powers off all amplifier channel.

9.6. PSU



A compressor with slow release time is responsible for limiting the maximum power drawn from the power line. The limitation smoothly reduces the gain at all channels simultaneously so that no acoustic picture distortion will occur in overload situations.

AMP MAINS VOLTAGE

Please select your current mains voltage range, current-limiting depends on it.

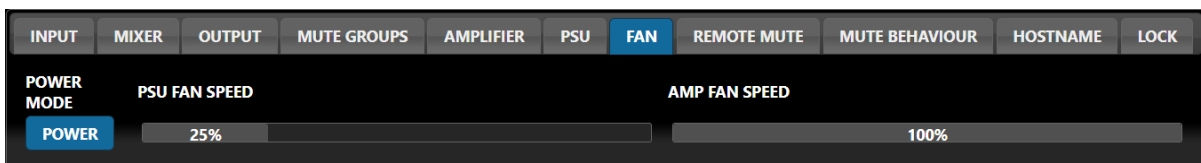
INTERNAL PSU CURRENT

Slider which indicates the internal **PSU** current on the secondary side.

GAIN REDUCTION

Indicates the **GAIN REDUCTION** of the output limiter in dB.

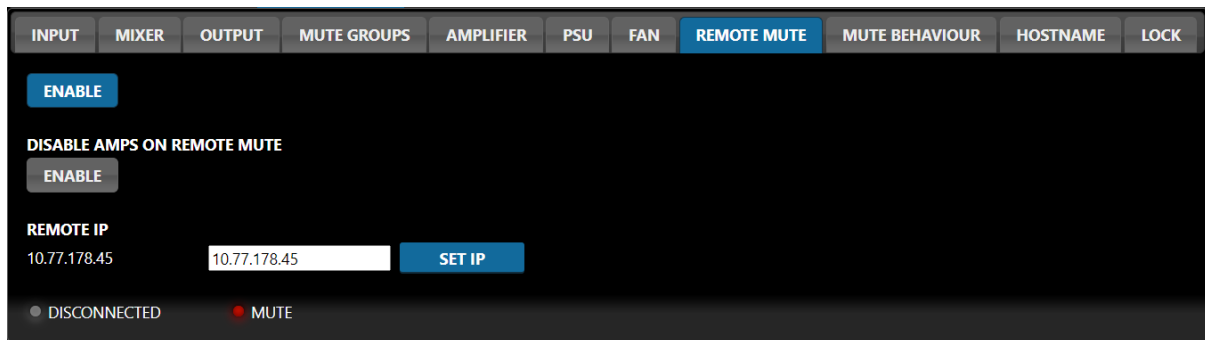
9.7. FAN




Only present on the MA32/D device.

The **FAN** power mode is intended for a high current intensive usage of the amplifier in industrial or scientific applications where fan noise doesn't matter, and a continuous output signal is required. If enabled, the idle, as well as the maximum fan speed, is increased. The two sliders indicate the current speed within a range of 0-100%.

9.8. REMOTE MUTE



A convenient way to mute all amplifier outputs with a simple GPI, an optional remote mute device is offered, which is connected over IP. Multiple MAXX-Devices can connect to the same remote mute device what offers the possibility to mute an unlimited amount of amplifiers by only a single hardware button.

ENABLE

The main enable switch enables the remote mute feature.

DISABLE AMPS ON REMOTE MUTE

If enabled, channels also power off if remote mute triggered.

REMOTE IP

Shows the IP of the remote mute device. It can be changed in the text box next to it.

CONNECTION STATE

Indicates the connection state between the MAXX-Device and the Remote Mute Device. If the connection is established, the indicator switches to **CONNECTED**, otherwise it shows **DISCONNECTED**. The connection is also observed by periodical ping messages to detect connection failures.

MUTE STATE

Indicates the resulting remote mute state, which can be **MUTE** or **RUN**. **MUTE** is set either by the **DISCONNECTED** state or by GPI state on the remote mute device.

9.9. MUTE BEHAVIOUR

The **MUTE BEHAVIOUR** controls the volume and mute state after powering on the device or toggling the master mute switch.



With the two settings **MUTE ON START** and **RAMP UP VOLUME AFTER MUTE**, the following combinations are possible:

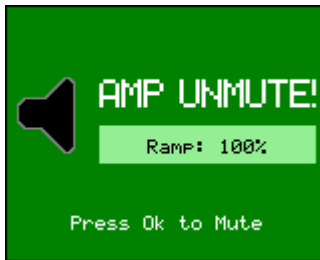
MUTE = ON, RAMP = OFF

In this case, the amplifier starts with muted outputs. By pressing the **OK** button on the LCD user panel or deselect the master mute on the web page, the volume is set to the values stored per channel, and the amplifier continues in normal operation.



MUTE = OFF, RAMP = ON

After turning the unit on or unset **MASTER MUTE** on the web page, it will automatically ramp up the volume on each channel to the desired final values.



MUTE = ON, RAMP = ON

After turning the unit on, all outputs are muted. The Ramp-Up-Process has to be started manually by pressing the **OK**-Button or toggle the Master Mute on the web page. Pressing **OK**-Button again, the Ramp-up process is cancelled, and the amplifier is muted again.



Values for the **RAMP UP TIME** in the range of about 2-5 seconds gives the amplifier a nice "soft-start" behaviour. Larger values of about 10-20 seconds combined with **MUTE ON START** are a useful tool to assist in the commissioning of new speaker systems to approach the desired nominal level and possibly initiate an abort if a problem occurs.

9.10. HOSTNAME

The **HOSTNAME** is a unique name of the MAXX Device inside a network environment. It's also easy to identify multiple devices at different locations in the network.

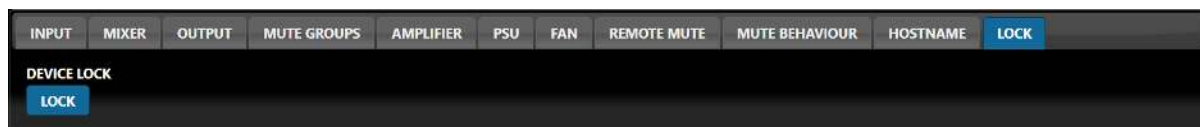
By default, the **HOSTNAME** is a combination of the MAXX Device type and its Ethernet MAC Address's last six digits to make the name unique.



The user can change the name freely, but it's advisable to keep the naming unique between multiple MAXX-Devices.

INPUT	MIXER	OUTPUT	MUTE GROUPS	AMPLIFIER	PSU	FAN	REMOTE MUTE	MUTE BEHAVIOUR	HOSTNAME	LOCK
<div>MA32D-14e0cc</div> <p>Set empty string to get default hostname.</p>										

Chapter 10. LOCK



DEVICE LOCK

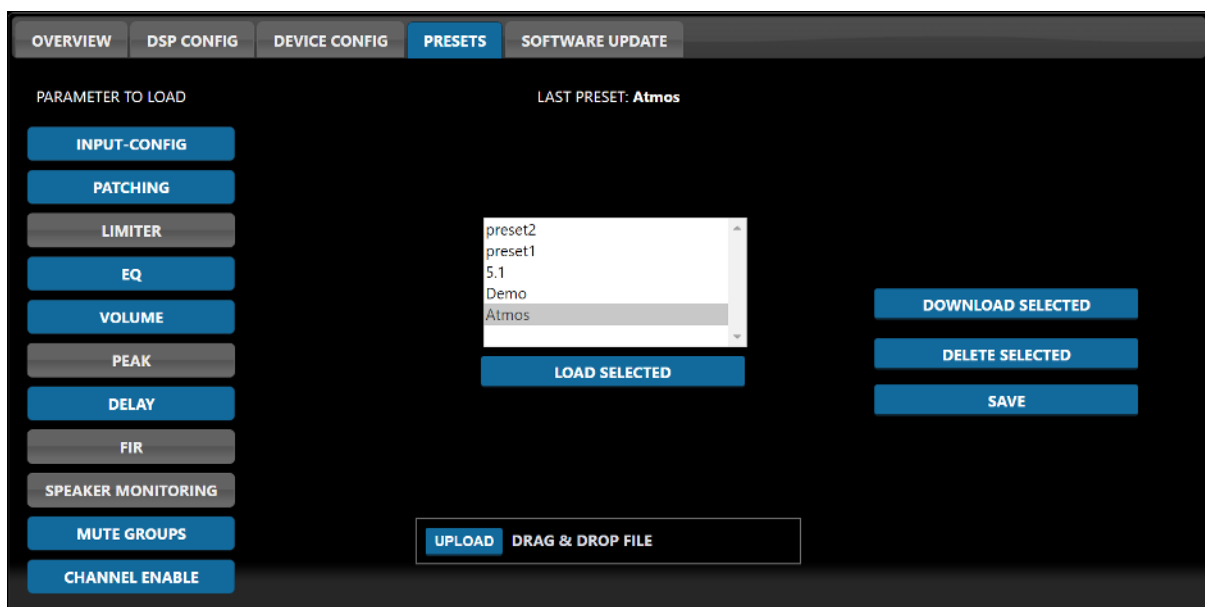
Toggles the **DEVICE LOCK** state on the front panel (see [DEVICE LOCK](#))

Chapter 11. PRESETS

In contrast to [CHANNEL PRESETS](#), global **PRESETS** always stores the entire configuration of the amplifier. Therefore, global **PRESETS** are intended to switch between different settings of complete loudspeaker systems or just different audio supply sources.



When saving the presets, the currently active settings are stored, but only certain parameters can be loaded during a preset recall.



SAVE

Stores the currently active configuration of the MAXX-Device into a preset, which can be named by the user. The saved preset is added to the selection list from where it can be loaded, deleted, or downloaded to the host PC.

LAST PRESET

Indicates the last loaded preset.

PARAMETER TO LOAD

- **INPUT-CONFIG** loads the SYNC TO from [INPUT](#)
- **PATCHING** loads the DSP channel assignment (see [INPUT CHANNEL](#))
- **LIMITER** loads the RMS [LIMITER](#) settings
- **EQ** loads the [IIR](#) section of the DSP chain

- **VOLUME** loads the master volume as well as the volume for each DSP channel (see [VOLUME](#))
- **PEAK** loads only the [PEAK](#) value from the LIMITER tab
- **DELAY** loads the [DELAY](#) parameter
- **FIR** loads the [FIR](#) filter name only
- **SPEAKER MONITORING** loads the [SPEAKER](#) settings
- **MUTE GROUPS** loads the [MUTE GROUPS](#) channel assignment, its **MUTE** / **SOLO** state and its name
- **CHANNEL ENABLE** loads the **CHANNEL ENABLE** for each channel (see [AMPLIFIER](#))



The user have to ensure that all required FIR filter files are present on the device, if not the filter will be bypassed.

Chapter 12. SOFTWARE UPDATE

OVERVIEW
DSP CONFIG
DEVICE CONFIG
PRESETS
STATUS
SOFTWARE UPDATE

1. DOWNLOAD USER DATA (PRESETS, CHANNEL PRESETS, FIR FILTER)

DOWNLOAD
2. UPLOAD & UPDATE

UPLOAD

DRAG & DROP FILE

!Do not turn off amp during update is in progress!
3. Wait until update is completed. Amp will be restarted after update. Reload Webpage after reboot.
4. UPLOAD USER DATA

UPLOAD USER DATA

DRAG & DROP FILE

Step	Explanation
1	DOWNLOAD a complete backup of the device. It includes presets, channelpresets and FIR filters
2	Upload a firmware file (*.inx). This process takes several minutes. After the upload is complete, the update file will be automatically extracted and copied to the flash.
3	After the Update has been completed, the amp will restart automatically.
4	With the UPLOAD USER DATA function the config backup created in step 1 can be uploaded again.



The update process is completely "fail-safe". It always writes an unused flash memory while the currently active flash memory remains unchanged. After the update is complete, the MAXX Device will boot from the recently updated memory. If the update image was corrupted by a power loss or any failures during the update process, the old firmware version is booted after the Device's restart.



Steps 1 and 4 can be used to create a complete backup of the amplifier or transfer it to other devices.



All amplifier outputs are muted during the update process.

Chapter 13. SNMP

At Version V2.3.8 and above, the amplifier supports SNMP requests. SNMP V1 and V2c is supported. All values are read-only.



We offer CDS, CMI or MIB files.

Please contact us to get the corresponding files.

The following Values are supported:

- **MASTER MUTE STATUS** indicates master mute status. (Off = 1 / On = 2)
- **REMOTE MUTE STATUS** indicates remote mute status. (Off = 1 / On = 2) (see [REMOTE MUTE](#))
- **SYNC STATUS** indicates audio sync is ok. (OK = 1 / Fault = 2)
- **PSU LIMIT STATUS** indicates if the power supply is in limit and the PSU limiter is working. (Off = 1 / PSU limiter active = 2) (see [PSU](#))
- **TEMPERATURE SENSORS** gives max Amp temperature and FPGA temperature. (in °C)
- **CHANNEL STATUS (1-32)** indicates whether amplifier on given channel is ok. (OK = 1 / Fault = 2)
- **SPEAKER STATUS (1-32)** indicates whether speaker is detected or not. Every speaker has its own value. (OK = 1 / Fault = 2)

Chapter 14. AES70 (OCA)

The AES70 standard protocol is implemented to control the parameters listed below.

The server can be reached via TCP port **49523** by the client application.

Table 6. Available OCA objects

OcaONo	description	type	range
0x1	standard device manager	OcaDeviceManager	
0x4	standard description manager	OcaSubscriptionManager	
0x1000	standard network manager	OcaNetwork	
0x2000	master mute	OcaMute	
0x2001	master gain	OcaGain	
0x2002	sync select	OcaUint32Actuator	MADI COAX = 0 MADI FIBRE = 1 AES3_1 = 2 AES3_2 = 3 DANTE = 4
0x2010-0x2030	channel gain (1-32)	OcaGain	

OcaONo	description	type	range
0x2100-0x2120	channel input patch (1-32)	OcaUint32Actuator	OFF = 0 MADI_COAX = 1 - 64 MADI_FIBRE = 101 - 164 AES3 = 201 - 204 DANTE = 301 - 332
0x2200-0x2220	speaker detector (1-32)	OcaBooleanSensor	true = Speaker OK false = no Speaker or Speaker NOT OK
0x2300	amp status	OcaBooleanSensor	true = Amp OK false = Amp NOT OK
0x2301	PSU limit active	OcaBooleanSensor	true = PSU Limit active false = PSU Limit NOT active
0x2302	get audio sync status	OcaBooleanSensor	true = Sync OK false = Sync NOT OK

Chapter 15. UDP DISCOVERY

In every IP configuration, it is possible to discover the device over ethernet. If the device is on the same network, you can discover it via UDP broadcast. The communication uses the UDP port: **9453**. Beside discover it is possible to change IP settings. The configurations are described below:

Table 7. Network config types

config	type (discovery response)	description
Static IP	0	To use in fixed installations. The device always starts with the same, user configured, IP address.
DHCP	1	To use with a DHCP Server. The Device tries to get an IP address from DHCP Server. If it is not possible to get a DHCP address, the device automatically jumps to config type four.
AUTO IP (Zeroconf)	2	To use with a direct connection between device and PC. The device uses a random IP between 169.254.0.1 - 169.254.254.254 to communicate with the PC.
AUTO IP (Fallback)	3	If the device gets no IP via DHCP it automatically jumps to AUTO IP Mode.

The structure of used UDP Messages is described below:

15.1. DISCOVER MESSAGE

The package for discovering devices is **INX-DISCOVER**. The response message of discovered devices is as followed:

Table 8. Discovery response message

byte(s)	content
1-10	INX-DEVICE
11	–
12-29	IP:XXX.XXX.XXX.XXX
30	–
31-49	SUB:XXX.XXX.XXX.XXX
50	–
51-68	GW:XXX.XXX.XXX.XXX
69	–
70-85	MAC:XXXXXXXXXXXX
86	–
87-93	TYPE:XX
94	–
95-130	HOST:HOSTNAME

15.2. STATIC IP MESSAGE

To bring the device from **DHCP** or **AUTO-IP** to **STATIC IP** mode or to change the IP address followed package can be sent to the device:

Table 9. Set static IP requests message

byte(s)	content
1-10	INX-SET-IP
11	–
12-29	IP:XXX.XXX.XXX.XXX
30	–
31-49	SUB:XXX.XXX.XXX.XXX
50	–
51-68	GW:XXX.XXX.XXX.XXX
69	–

byte(s)	content
70-85	MAC:XXXXXXXXXXXX

If IP is successfully set, the response is as followed:

Table 10. Set static IP response message

byte(s)	content
1-10	INX-IP-SET
11	–
12-29	IP:XXX.XXX.XXX.XXX
30	–
31-49	SUB:XXX.XXX.XXX.XXX
50	–
51-68	GW:XXX.XXX.XXX.XXX
69	–
70-85	MAC:XXXXXXXXXXXX

If an error during IP set occurs the response is as followed:

Table 11. Set static IP error response message

byte(s)	content
1-10	INX-IP-SET
11	–
12-19	IP-ERROR
20	–
21-36	MAC:XXXXXXXXXXXX

15.3. DHCP MESSAGE

To bring the device from **STATIC-IP** or **AUTO-IP** to **DHCP**, followed package can be sent to the device:

Table 12. Set DHCP request message

byte(s)	content
1-12	INX-SET-DHCP
13	–
14-29	MAC:XXXXXXXXXXXX

If the device successfully set the IP mode to **DHCP**, the response is as followed:

Table 13. Set DHCP response message

byte(s)	content
1-12	INX-DHCP-SET
13	–
14-29	MAC:XXXXXXXXXXXX

If the device is in **AUTO-IP (Fallback)** mode, you can trigger new tries for DHCP request. The device tries to communicate with the DHCP server on the network. If the device gets no IP via DHCP it automatically jumps to **AUTO IP** Mode, again. The request package is as followed:

Table 14. DHCP release request message

byte(s)	content
1-16	INX-RELEASE-DHCP
17	–
18-33	MAC:XXXXXXXXXXXX

If the device successfully starts a DHCP release, the response is as followed:

Table 15. DHCP release response message

byte(s)	content
1-16	INX-DHCP-RELEASE
17	–
18-33	MAC:XXXXXXXXXXXX

15.4. AUTO IP MESSAGE

To bring the device from **STATIC-IP** or **DHCP** to **AUTO-IP**, followed package can be sent to the device:

Table 16. Set AUTO IP request message

byte(s)	content
1-15	INX-SET-AUTO-IP
16	–
17-32	MAC:XXXXXXXXXXXX

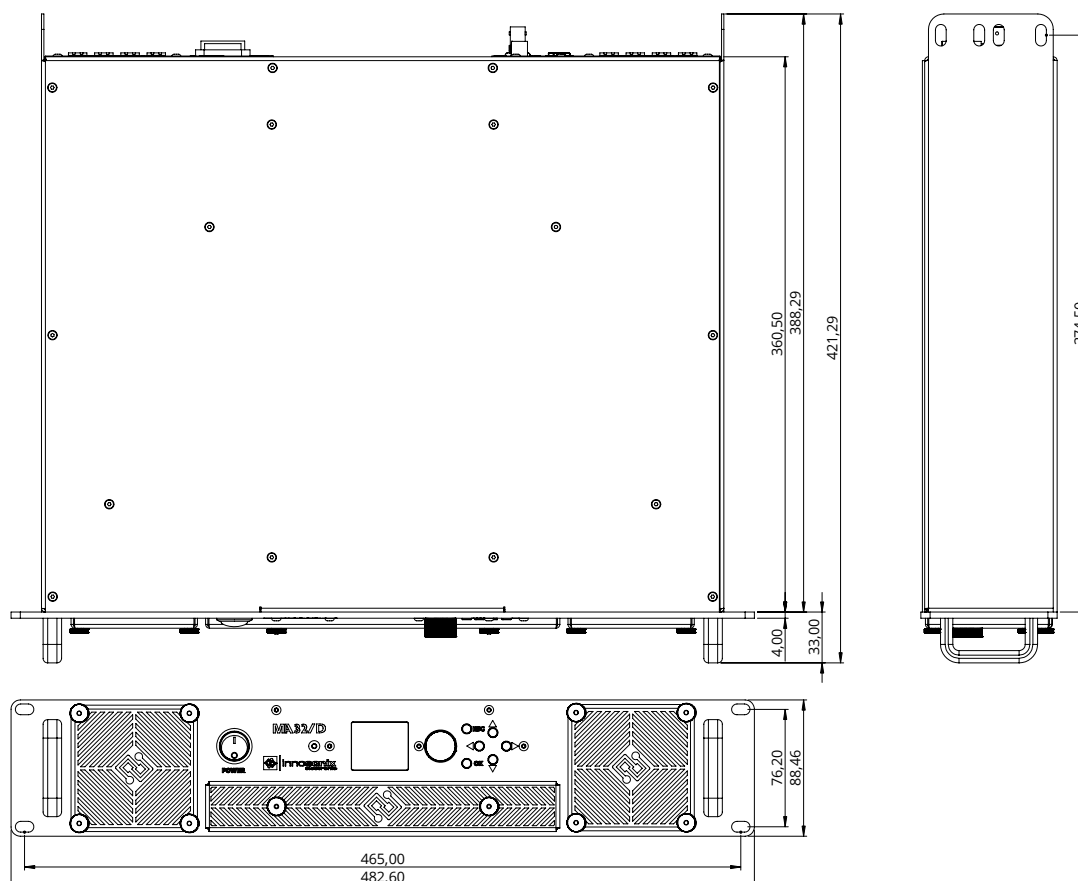
If the device successfully set the IP mode to **AUTO-IP**, the response is as followed:

Table 17. Set AUTO IP response message

byte(s)	content
1-15	INX-AUTO-IP-SET
16	–
17-32	MAC:XXXXXXXXXXXX

Chapter 16. DIMENSIONS AND WEIGHT

16.1. DIMENSIONS

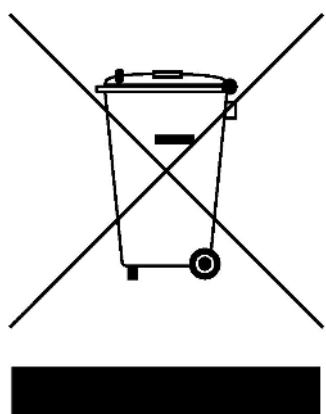


16.2. WEIGHT

15.0 kg

Chapter 17. DISPOSING OF OLD APPLIANCES

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 18. EU DECLARATION OF CONFORMITY

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

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

18.1. EN 55032:2012

Electromagnetic compatibility of multimedia equipment - **Emission requirements:**

Radiated, Conducted: Class A Limits

18.2. 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**

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

- **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

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

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

18.4. MANUFACTURER

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