DirectOut TECHNOLOGIES

PRODIGY.MP

User's Manual

Version 2.6

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About This Manual

How to Use This Manual

This manual guides you through the installation and operation of the device. Use the Table of Contents at the beginning of the manual or Index Directory at the end of the document to locate help on a particular topic. You can access more information and latest news by visiting on the DirectOut website at www.directout.eu.

Conventions

The following symbols are used to draw your attention to:

TIPS

indicate useful hints and shortcuts.

NOTES

are used for important points of clarification or cross references.

WARNINGS!

alert you when an action should always be observed.

CHAPTER 1: Overview

Introduction

Welcome to PRODIGY.MP, DirectOut's multifunction audio processor supporting multiple formats, flexible I/O, dual network audio, DSP functions, sample rate conversion and powerful hard- and software.
PRODIGY.MP has been designed to address numerous applications in live sound, broadcast, installation and recording, and it's hardware can be configured to specific requirements.

Feature Summary

Audio Network Modules *	2 slots (1 x SFP, 2 x RJ45 Socket): - Dante (64 ch) - RAVENNA (128 ch) - SoundGrid (128 ch) - AVB / MILAN (128 ch) - MADI2 (128 ch) - BNC or SFP
MADI Modules *	2 slots : - SFP (empty cage without module) - SC-Socket multi/single-mode - coaxial BNC, 75 Ω
Converter Modules*	4 slots - individually configurable - 8 channels each: - Analog I/O - Line In / Out - Mic I/O - Mic In / Line Out - Digital I/O - AES3 In / Out
Headphones	1 x 6.3 mm jack & 1 x 3.5 mm jack
Word Clock (I/O):	4 x coaxial BNC (WCK2 switchable to Video sync) (75 Ω termination switchable)
Supported Audio Network Standards (depending on module)	Dante / AES67 RAVENNA / AES67, ST 2110-30 /-31, ST 2022-7 SoundGrid AVB / MILAN
Sample Rate Conversion	FastSRC™ on MADI I/Os, Audio Network I/Os HD SRC on Audio Network I/Os with SRC option AES4.SRC.IO modules for AES3 inputs
DSP Functions	Flex Channels with EQ, Dynamics, Delay Matrix Mixers, Summing Busses, External Insert-Points, DSP Routing
MADI Formats	56/64 channel, 48k/96k Frame, S/MUX
Sample Rates	44.1, 48, 88.2, 96, 176.4, 192 kHz +/-12.5%
Management Port	1 x SFP, 2 x RJ45 Socket (Gigabit-Ethernet)
Device Control	remote: via globcon or browser, Ember+, OSC local: via touch-display
MIDI	2 x DIN socket
GPIO	1 x DSUB-9 (2 x GPI, 2 x GPO, MOSFET switches)
USB Port	2 x USB 2.0 port for legacy control of ANDIAMO devices
Power Supply	This device is equipped with two wide range power supplies (84 V to 264 V AC / 47 Hz to 63 Hz / safety class 1)

* see „Slots- Connecting Audio“ on page 31 for a comprehensive description of the different modules available.

How it works

All modules inserted into the mainframe offer a variety of inputs and outputs that are managed by an internal routing matrix.
There are four converter slots equipped with eight channels per module and interface direction for conversion of up to 32 audio channels. Combined with the two audio network options, two MADI options and the headphones outputs the maximum channel capacity of the device increases to 416 inputs and 420 outputs.
Signal processing is managed internally by a DSP routing matrix which connects hardware I/Os with the DSP processing blocks - such as Flex Channels, Matrix Mixer, Summing Busses and Plug-Ins (EQ, Delay, Dynamics).

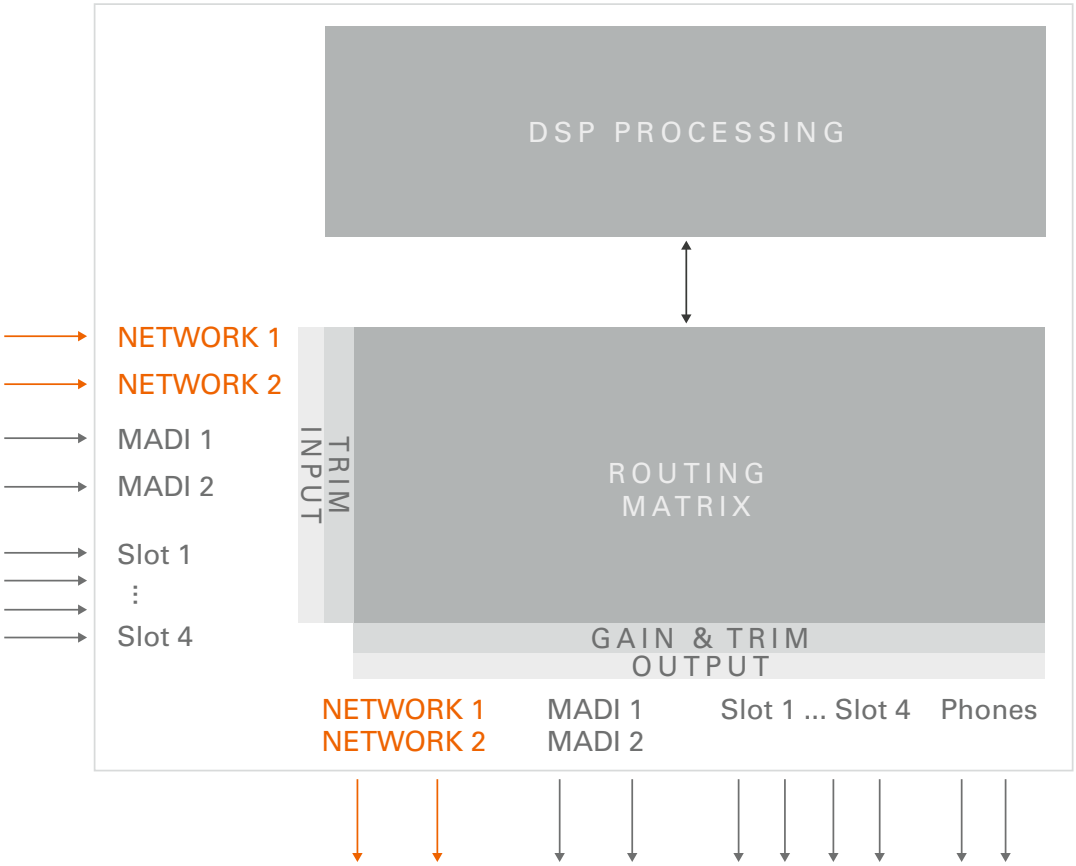
Applications

PRODIGY.MP can be used for bidirectional conversion between analog and digital signals, signal processing, signal distribution, conversion between network audio formats, streaming audio, conversion between audio networks and MADI environments, conversion of different MADI signals.

Typical applications include:

- Live PA measurements, tuning and control
- Centralised virtual sound-check-station (collecting feeds from stage racks for multi-track recording, distribution of playback to different consoles with different formats and sample rates)
- Enhanced playback-station (with EARS and LTC reader)
- Signal collection and processing in broadcast and fixed installation (MicPres, EQs, Limiters, Delay, Summing Busses, Matrixes, Mixers, …)
- Format and sample rate conversions of network audio streams
- Signal distribution (routing matrix)
- Clock extractor/distributor with automatic redundancy and selectable priorities
- Enhancing the level of redundancy of complex live and realtime audio environments
- Stream monitoring (headphones / hardware outputs)

Scheme


DSP PROCESSING
NETWORK 1
NETWORK 2
MADI 1
MADI 2
Slot 1
Slot 4
TRIM
INPUT
ROUTING
MATRIX
GAIN & TRIM
OUTPUT
NETWORK 1
NETWORK 2
MADI 1
MADI 2
Slot 1 ... Slot 4
Phones

CHAPTER 2: Legal issues & facts

Before Installing This Device

WARNING!

Please read and observe all of the following notes before installing this product:

- Check the hardware device for transport damage.
- Any devices showing signs of mechanical damage or damage from the spillage of liquids must not be connected to the mains supply, or disconnected from the mains immediately by pulling out the power lead.
- All devices must be grounded. The device is grounded through its IEC power connections.
- All devices must be connected to the mains using the three-cord power leads supplied with the system. Only supply electrical interfaces with the voltages and signals described in these instructions.
- Do not use the device at extreme temperatures. Proper operation can only be guaranteed between temperatures of 5° C and 45° C and a maximum relative humidity of 80 %, non-condensing.
- The cabinet of the device will heat up. Do not place the device close to heating sources (e.g. heaters). Observe the environmental conditions.

Defective Parts/Modules

WARNING!

This device contains no user-serviceable parts. Therefore do not open the device. In the event of a hardware defect, please send the device to your DirectOut representative together with a detailed description of the fault.
We would like to remind you to please check carefully whether the failure is caused by erroneous configuration, operation or connection before sending parts for repair.

First Aid (in case of electric shock)

WARNING!

- Do not touch the person or his/her clothing before power is turned off, otherwise you risk sustaining an electric shock yourself.
- Separate the person as quickly as possible from the electric power source as follows:
  - Switch off the equipment.
  - Unplug or disconnect the mains cable.
- Move the person away from the power source by using dry insulating material (such as wood or plastic).
- If the person is unconscious:
  - Check their pulse and reanimate if their respiration is poor.
  - Lay the body down and turn it to one side. Call for a doctor immediately.
- Having sustained an electric shock, always consult a doctor.

Updates

DirectOut products are continually in development, and therefore the information in this manual may be superseded by new releases. To access the latest documentation, please visit the DirectOut website: www.directout.eu.

This guide refers to System Build 13.

Intended Operation

PRODIGY.MP is designed for signal processing / conversion / routing between analog, digital and audio network signals. Digital audio refers to AES3 and AES10, audio network refers to Dante, SoundGrid and RAVENNA (including AES67 and SMPTE ST 2110-30 /-31).

WARNING!

No compensation can be claimed for damages caused by operation of this unit other than for the intended use described above. Consecutive damages are also excluded explicitly. The general terms and conditions of business of DirectOut GmbH are applied.

Conditions of Warranty

This unit has been designed and examined carefully by the manufacturer and complies with actual norms and directives.
Warranty is granted by DirectOut GmbH over the period of 36 months for all components that are essential for proper and intended operation of the device. The date of purchase is applied for this period.
Consumable parts (e.g. battery) are excluded from warranty claims.

WARNING!

All claims of warranty will expire once the device has been opened or modified, or if instructions and warnings were ignored.
For warranty claims please contact the dealer where your device was acquired.

Conformity & Certificates

CE

This device complies with the basic requests of applicable EU guidelines. The appropriate procedure for approval has been carried out.

RoHS

(Restriction of the use of certain Hazardous Substances)
This device was constructed fulfilling the directive on the restriction of the use of certain hazardous substances in electrical and electronic equipment 2011/65/EU and 2015/863.

WEEE

(Directive on Waste Electrical and Electronic Equipment)
Due to the directive 2002/96/EC for waste disposal this device must be recycled.
For correct recycling please dispatch the device to:
DirectOut GmbH,
Hainichener Str. 66a
09648 Mittweida
Germany
Only stamped parcels will be accepted!
WEEE-Reg.-No. DE 64879540

Contact

DirectOut GmbH
Hainichener Str. 66a, 09648 Mittweida, Germany
Phone: +49 (0)3727 5650-00
Mail: sales@directout.eu
www.directout.eu

Contents

The contents of your PRODIGY.MP package should include:

- 1 x PRODIGY.MP (19'', 2 RU)
- 2 x power chord with self locking mechanism
- 1 x Hardware Guide

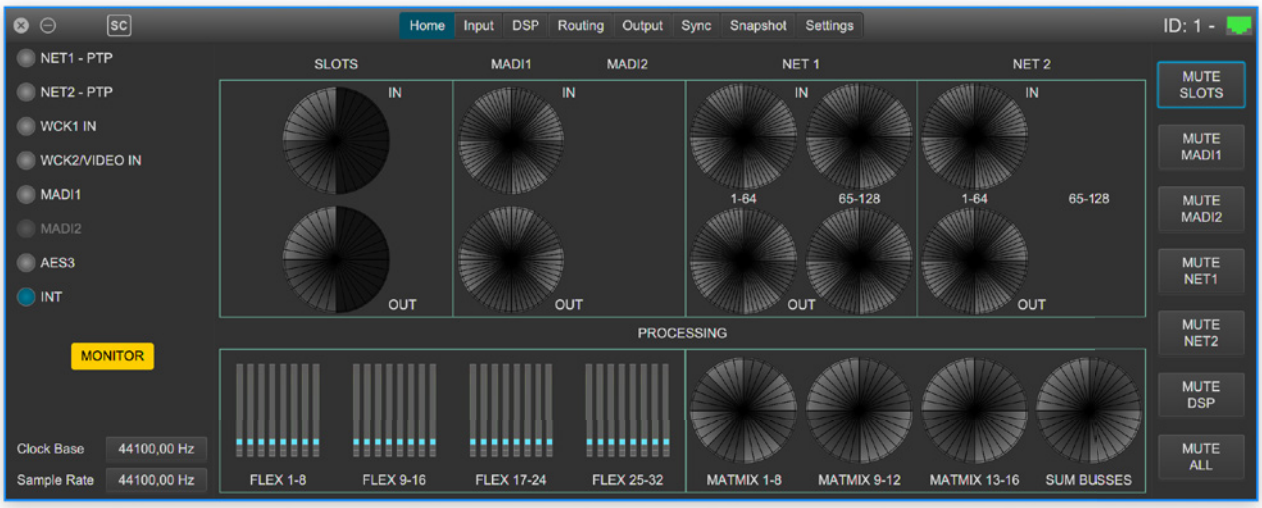
The device provides slots for audio network modules, MADI modules and converter modules. The modules may be delivered separately and require installation first to complete the delivery.

TIP

Keep any packaging in order to protect the device should it need to be dispatched for service.

Remote Control

To access all functions of the device it is required to install the globcon remote control.



globcon is a free, global control software platform for the management of professional audio equipment. Almost all products of the DirectOut product portfolio are supported by globcon.

Link: www.globcon.pro

Accessories

BREAKOUT

The BREAKOUT series is a range of adaptor boxes available in different variants to extend the coverage of the ANDIAMO and PRODIGY series. They are equipped with XLR or BNC connectors on the front panel and DSUB-25 connectors on the rear panel. Audio signals are carried passively between the front and rear panels.

The small form factor and angle brackets also allow for mounting the devices on the back of an ANDIAMO or PRODIGY unit.

BREAKOUT.AN8 - analog input / output, 8 channels
Article code: DOBOB0889

BREAKOUT.AN16I - analog input, 16 channels
Article code: DOBOB0890

BREAKOUT.AN16O - analog output, 16 channels
Article code: DOBOB0891

BREAKOUT.AES - digital input / output, 8 AES3 ports (16 channels)
Article Code: DOBOB0887

BREAKOUT.AESID - digital input / output, 16 AESid ports (32 channels)
Article Code: DOBOB0888

Patch Chords

Cabling from Cordial provides appropriate connection of the BREAKOUT with your device to ensure proper transmission of the audio signals.

Name	Description	Article code
DSUB25.AN50	Analog patch cable for connection with BREAKOUT.AN16I, AN16O, AN8, transferring 8 audio channels, length 0.5 m	DOCAA0334
DSUB25.AN100	Analog patch cable for connection with BREAKOUT.AN16I, AN16O, AN8, transferring 8 audio channels, length 1.0 m	DOCAA0335
DSUB25.AES50	Digital patch cable for connection with BREAKOUT.AES or AESid transferring 8 audio channels, length 0.5 m	DOCAA0332
DSUB25.AES100	Digital patch cable for connection with BREAKOUT.AES or AESid transferring 8 audio channels, length 0.5 m	DOCAA0333

SFP Transceiver - MADI

Two different optical SFP transceiver for MADI transmission are available from DirectOut GmbH:

- Multimode SFP transceiver with LC connectors (No: DOICT0129)
- Singlemode SFP transceiver with LC connectors (No: DOICT0130)

Specification of the optical SFP transceivers:

SFP	Multimode	Singlemode
Wavelength TX nominal	1310 nm	1310 nm
Wavelenght RX nominal	1310 nm	1310 nm
Distance	2 km	10 km
Powerbudget (dB)	> 11 dB	> 12 dB
Protocols	Fast Ethernet OC3/STM1	Gigabit Ethernet, Gigabit Fibre Channel
Bandwidth from	100 Mbit/s	1.050 Gbit/s
Bandwidth	155 Mbit/s	1.250 Gbit/s
Laser	FP	FP
Receiver Type	PIN	PIN
Connector	LC	LC
Wavelength TX min	1260 nm	1260 nm
Wavelength TX max	1360 nm	1360 nm
Wavelength RX min	1260 nm	1260 nm
Wavelength RX max	1620 nm	1600 nm
Transmit min	- 19.00 dBm	- 9.00 dBm
Transmit max	- 14.00 dBm	- 3.00 dBm
Receive min	- 30 dBm	- 21.00 dBm
Receive max (Receiver overload)	- 5.00 dBm	- 3.00 dBm
Temperature (min)	0° Celsius	0° Celsius
Temperature (max)	70° Celsius	70° Celsius
Type of DDM/DOM	internal	internal
Extinction Ratio	8.20 dB	9 dB

SFP Transceiver - Network

Two different optical SFP transceiver for 1Gbit/s ethernet transmission are available from DirectOut GmbH:

- SFP Copper Transceiver, RJ-45 (No: DOICT0132)
- SFP Optical Transceiver, optical LC Single Mode (No: DOICT0133)

Specification

SFP	Copper - RJ45	Optical - LC single mode
Distance	100 m via CATe cable	2000 m
Wavelength	-	1310 nm
Datarate	12 Mbit/s to 1.25 Gbit/s	2.67 GBit/s

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CHAPTER 3: Installation

Installing the Device

1. Open the packaging and check that the contents have been delivered complete and undamaged.
2. Fix the device in a 19'' frame with four screws, or place it on a non-slip horizontal surface.

WARNING!

Avoid damage from condensation by waiting for the device to adapt to the environmental temperature. Proper operation can only be guaranteed between temperatures of 5° C and 45° C and a maximum relative humidity of 80%, non-condensing.
Ensure that the unit has sufficient air circulation for cooling.
Do not cover the fan outlets and the slots at the sides of the device!
Do not block the fans by putting objects through the protective grid!

3. For installation of separately delivered modules follow the instructions in the manual enclosed with the packaging of the modules.

1	Audio Network Slots - see page 44
2	MADI Slots - see page 49
3	Converter Slots - see page 55

4. Connect the signal cables with the installed modules. For more details about connecting audio network, MADI, AES3 and analog signals see„Slots- Connecting Audio“ on page 31.

5. Connect a network cable to PORT 1,2 or 3 of the MANAGEMENT sockets to control the device via network.

6. Using the power cord provided connect the PSUs to a matching power supply.

The delivered power cords provide a self-locking mechanism to prevent an accidental disconnect. To unlock the connection press the red release mechanism.

WARNING!

This device must be connected to the mains using the three-cord power leads supplied with the system. Only supply the voltages and signals indicated (84 V – 264 V).

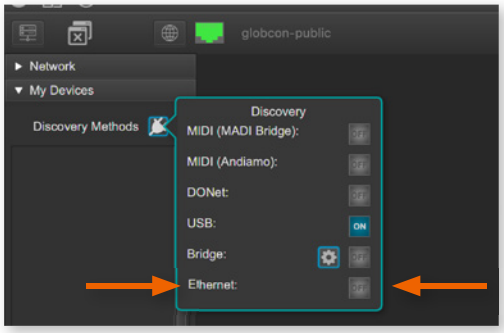
7. Turn on the power switches:

Check the display on the front panel for warnings.

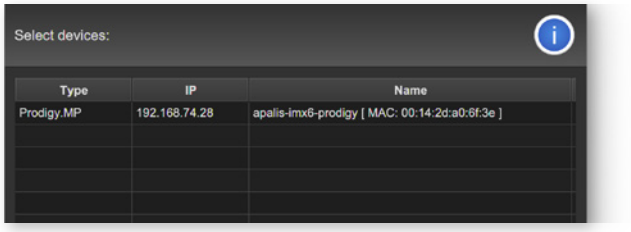
8. Enter http://<IP Address> (default IP: DHCP) in the navigation bar of your browser to open the control website. The IP address is assigned by the DHCP server of your network and can be checked or altered on the display at the front panel.

9. Install globcon on your computer
To access all functions the use of globcon is required - see page 16.

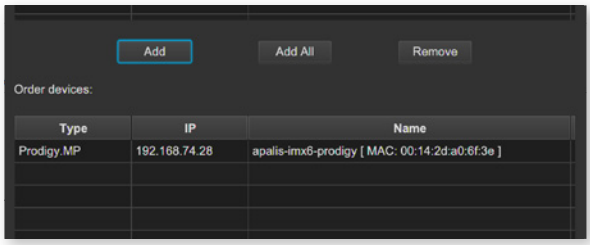
10. Launch globcon control
Select the port of the network where your device is connected to
Sidebar - My Devices - Click Discovery Methods
and set Ethernet Discovery to 'ON'



Select the device from the list and click 'Add'

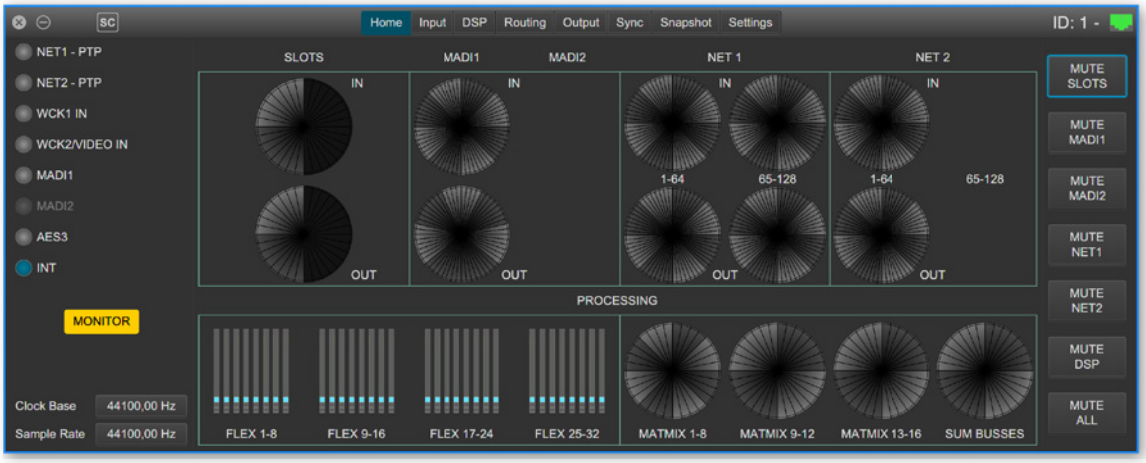


Confirm your selection with ‘OK’



The device is displayed in the overview of globcon
Double-Click the tiny display of the device to open the control.

More information and video tutorials about globcon are available at www.globcon.pro.



NOTE

For globcon control of the device the network infastructure is required to have ports 5002, 5003 and 5004 unfiltered for TCP traffic.
Since the device discovery uses multicast-DNS (mDNS), the device may be discovered but cannot be controlled if the ports 5002, 5003 and 5004 are filtered.

CHAPTER 4: Operation

Introduction

This chapter describes the basic operation of the device.
Note that throughout this manual, the abbreviation FS refers to sample rate or sample frequency. So, when dealing with scaling factors, the following sample rates can be written as:

- 44.1 kHz or 48 kHz = 1 FS
- 88.2 kHz or 96 kHz = 2 FS
- 176.4 kHz or 192 kHz = 4 FS

Global Control

The display on the front panel indicates the power supply. The power switches are on the back panel:

PSU 1 & PSU 2	**2 Switches** Enable / disable power supply.
PSU 1 & PSU 2	**2 C13 sockets** Connect the power supply here (84- 264 V AC).

WARNING

Before switching off the power supply, the system must be shut down first - see “Shutdown” on page 75.

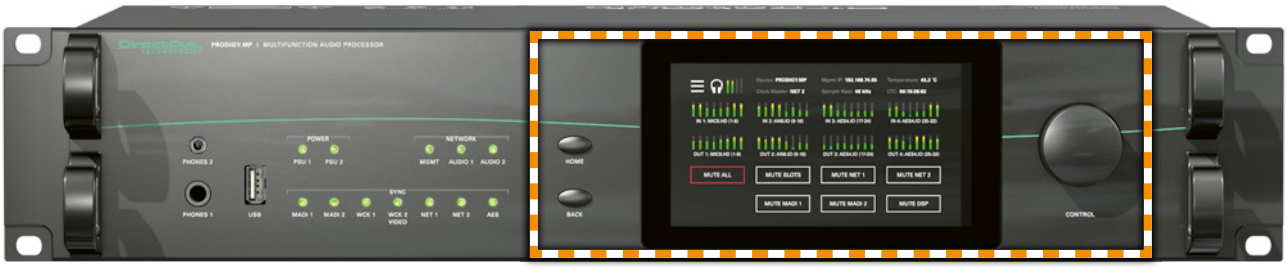
POWER PSU 1 & PSU 2	**LED RGB - indicates state of power supply** (OFF) = power supply not working (ON, green) = power supply working (blinking, red) = power supply was active and is now inactive.

NOTE

The green LEDs (PSU 1 & PSU 2) indicate that a working power supply is connected to the power supply unit. Note that an unlit LED does not guarantee that the device is free of voltage. To ensure that the device is completely disconnected from mains voltage, the power chords must be disconnected.

Managing Device

The device is equipped with a touch-display, an encoder and two push buttons for local control. For remote operation a network socket is provided at the rear panel to operate the device via a browser based GUI or via globcon.



HOME	**Push-button to access the HOME screen.** Press shortly to call the top menu.
BACK	**Push-button to return to previous page.** Press shortly to return to the previous page.
Display	**Touch-Display to navigate the menu, adjust settings and for monitoring.** tap, swipe- have fun!
CONTROL	**Encoder to adjust values and confirmation.** Rotate to modify values. Press to confirm a setting.

Networking

PRODIGY.MP uses separate network links for managing the device and network-audio transmission. LEDs on the front panel indicate the state of all network links.

MANAGEMENT **1**	**2 x RJ 45 socket & SFP slot** Connect here for network control and firmware updates.
MANAGEMENT PORT 1 & 2	**LED orange - indicates the link state of the network connection.** (ON) = device link active (OFF) = device link not active
MANAGEMENT PORT 1 & 2	**LED green - indicates the activity state of the network connection.** (ON) = data sent or received (OFF) = no data transmission
AUDIO NETWORK 1 AUDIO NETWORK 2 **2**	**Option Slots for audio network modules** See „Slots- Audio Network Modules“ on page 44.

NETWORK MGMT	**LED - indicates the state of the controller for management.** (OFF) = controller not available (ON) = controller ready (ON, blue) = controller shut down, device is ready to be switched off**
NETWORK AUDIO 1 & AUDIO 2	**LED - indicates the state of the network audio modules.** (OFF) = network module not ready (ON) = network module ready (blinking, red) = network module in panic mode*

* please contact support@directout.eu

** to ensure proper file operation a shut down must be initiated before switching the device off - see “Shutdown” on page 75.

Slots - Connecting Audio

PRODIGY.MP offers three different slot-types that can be assembled individually according the particular requirements.

Audio Network Slots **1**	**2 x Option slot for Audio Network Modules** See „Slots- Audio Network Modules“ on page 44
MADI IO Slots **2**	**2 x Option slot for MADI Modules** See „Slots- MADI Modules“ on page 49
Converter IO Slots **3**	**4 x Option slot for Converter Modules** See „Slots- Converter Modules“ on page 55

WARNING!

All module slots must be fitted with a module or a blind plate each. Otherwise live parts become accessible which may cause serious harm to your health. An open housing may also cause inappropriate operation conditions due to an insufficient electromagnetic shielding.

Slots - Audio Network Modules

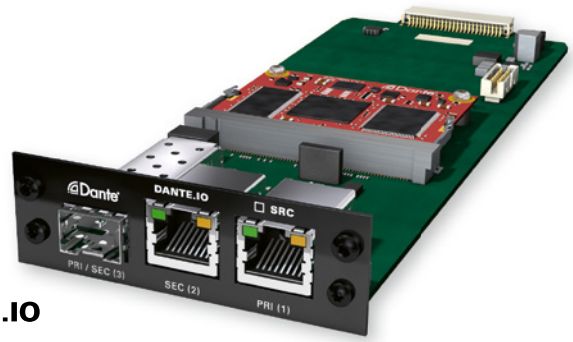
One audio network slot offers the choice between four audio network protocols. Each module is equipped with 1 x SFP and 2 x RJ45-sockets (Gigabit-Ethernet).

Module	Protocol	Capacity	SRC*
DANTE.IO	Dante / AES67	64 ch in / out	no ☐ SRC
DANTE.SRC.IO	Dante / AES67	64 ch in / out	yes ◉ SRC
RAV.IO	RAVENNA / AES67 / SMPTE ST 2110-30 /-31	128 ch in / out	no ☐ SRC
RAV.SRC.IO	RAVENNA / AES67 / SMPTE ST 2110-30 /-31	128 ch in / out	yes ◉ SRC
SG.IO	SoundGrid	128 ch in / out	no ☐ SRC
SG.SRC.IO	SoundGrid	128 ch in / out	yes ◉ SRC
MILAN.IO	AVB / MILAN	128 ch in / out	no ☐ SRC
MILAN.SRC.IO	AVB / MILAN	128 ch in / out	yes ◉ SRC

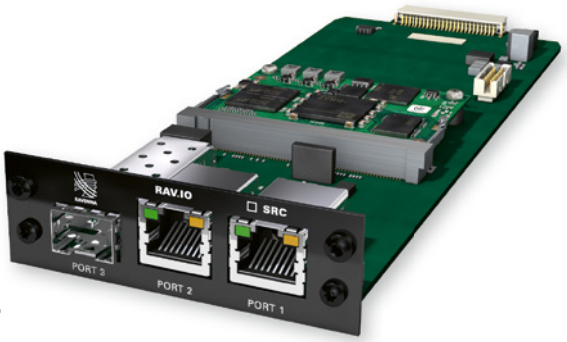
* The SRC versions are equipped with a bidirectional sample rate converter (HD SRC).
They are marked with a dot on the module plane.

NOTE

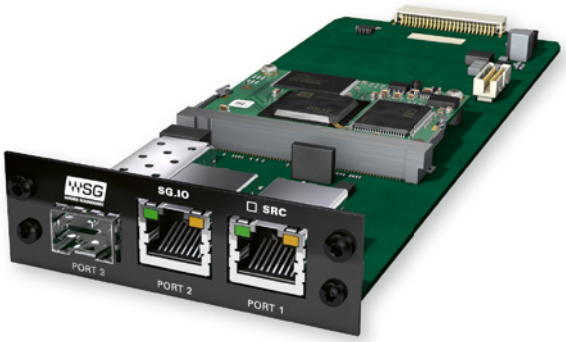
Multi-port MADI modules for extended MADI connectivity also fit into the audio network slots. Please refer to MADI modules on page 39.



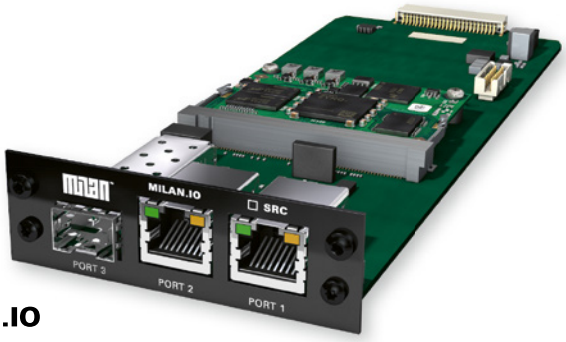
DANTE.IO
DANTE.SRC.IO



RAV.IO
RAV.SRC.IO



SG.IO
SG.SRC.IO

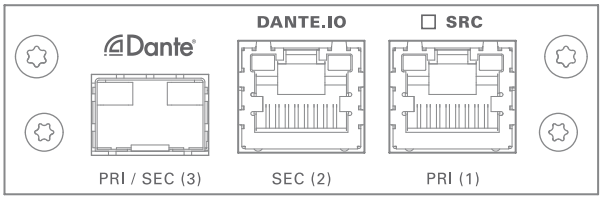


MILAN.IO
MILAN.SRC.IO

DANTE.IO / DANTE.SRC.IO

Three network ports are available for transmission of audio signals (Dante / AES67). The built-in network switch can be operated in three modes:

- Switched (all ports in the same network)
- Redundant (1 = Primary, 2 = Secondary, 3 = Primary)
- Red_Sec (1 = Primary, 2 & 3 = Secondary)



The switch will be configured automatically by selecting the respective operation mode in Dante Controller.

PRI (1)	**RJ45 socket (1 Gbit/s)** Network interface - connect here for network transmission. Primary Port when used in redundant operation.
SEC (2)	**RJ45 socket (1 Gbit/s)** Network interface - connect here for network transmission. Secondary Port when used in redundant operation.
PRI (1) SEC (2)	**LED orange - indicates the link state of the network connection.** (ON) = device link active (OFF) = device link not active
PRI (1) SEC (2)	**LED green - indicates the activity state of the network connection.** (ON) = data sent or received (OFF) = no data transmission
PRI / SEC (3)	**SFP cage** Insert SFP transceiver here and connect to the network.* Primary or secondary (selectable) port when used in redundant operation.

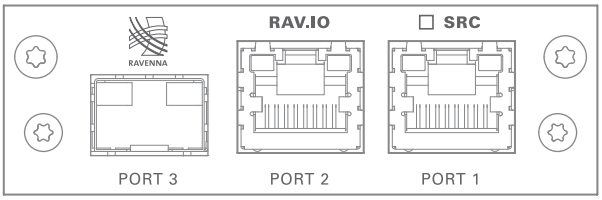
* See p 20 for SFP-transceiver that are available by DirectOut.

NOTE

If redundancy is being used, secondary interfaces should be connected to a second separate network. Secondary interfaces cannot communicate with primary interfaces.

RAV.IO / RAV.SRC.IO

Three network ports are available for transmission of audio signals (RAVENNA / AES 67 / SMPTE 2110-30 / -31). The module supports Seamless Protection Switching according to SMPTE 2022-7.



PORT 1 PORT 2	**RJ45 socket (1 Gbit/s)** Network interface - connect here for network transmission.
PORT 1 PORT 2	**LED orange - indicates the link state of the network connection.** (ON) = device link active (OFF) = device link not active
PORT 1 PORT 2	**LED green - indicates the activity state of the network connection.** (ON) = data sent or received (OFF) = no data transmission
PORT 3	**SFP cage** Insert SFP transceiver here and connect to the network.*

* See p 20 for SFP-transceiver that are available by DirectOut.

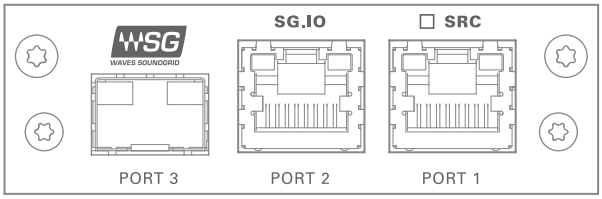
The configuration of the built-in network switch is available via the browser interface of the RAV.IO-module.

NOTE

Further devices that require PTP synchronisation should be connected to an external PTP-aware switch.

SG.IO / SG.SRC.IO

Three network ports are available for transmission of audio signals (SoundGrid).

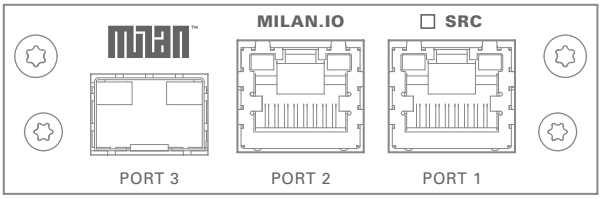


PORT 1 PORT 2	**RJ45 socket (1 Gbit/s)** Network interface - connect here for network transmission.
PORT 1 PORT 2	**LED orange - indicates the link state of the network connection.** (ON) = device link active (OFF) = device link not active
PORT 1 PORT 2	**LED green - indicates the activity state of the network connection.** (ON) = data sent or received (OFF) = no data transmission
PORT 3	**SFP cage** Insert SFP transceiver here and connect to the network.*

* See p 18 for SFP-transceiver that are available by DirectOut.

MILAN.IO / MILAN.SRC.IO

Three network ports are available for transmission of audio signals (AVB / MILAN).



PORT 1 PORT 2	**RJ45 socket (1 Gbit/s)** Network interface - connect here for network transmission.
PORT 1 PORT 2	**LED orange - indicates the link state of the network connection.** (ON) = device link active (OFF) = device link not active
PORT 1 PORT 2	**LED green - indicates the activity state of the network connection.** (ON) = data sent or received (OFF) = no data transmission
PORT 3	**SFP cage** Insert SFP transceiver here and connect to the network.*

* See p 18 for SFP-transceiver that are available by DirectOut.

The configuration of the built-in network switch is available via the browser interface of the RAV.IO-module.

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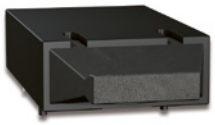
Slots - MADI Modules

Two MADI slots offer the choice between three connection standards for transmission of 64 channels @ 1 FS. For extended MADI connectivity there are multi-port MADI modules that fit into the audio network slots.

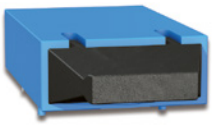
Module	Connection
BNC.IO	coaxial BNC input / output 75 Ω
SFP.IO	SFP cage *
SC.IO	SC-socket duplex multi-mode **
MADI2.SRC.IO	2 x SFP cage *
MADI2.BNC.SRC.IO	4 x coaxial BNC input / output 75 Ω
MADI4.IO / MADI4.SRC.IO	4 x SFP cage * (for PRODIGY.MX only)

* matching SFP transceiver are available from DirectOut - see page 17.

** The SC ports are multi-mode as default, single-mode SC ports are available on request. The housing of single-mode ports is colored blue.



multi-mode



single-mode

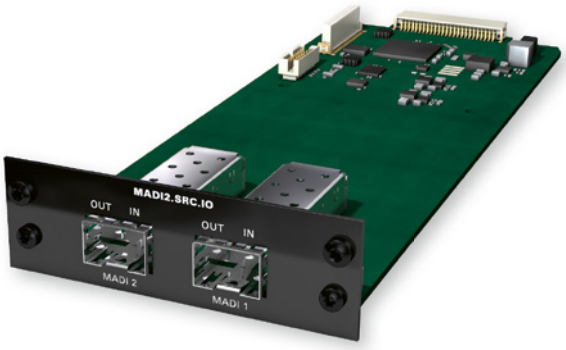
Single-port MADI Modules

BNC.IO

SC.IO

SFP.IO

Multi-port MADI Modules

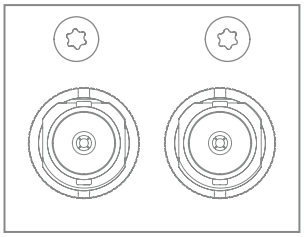


MADI2.SRC.IO

MADI2.BNC.SRC.IO

MADI4.IO
MADI4.SRC.IO
(PRODIGY.MX only)

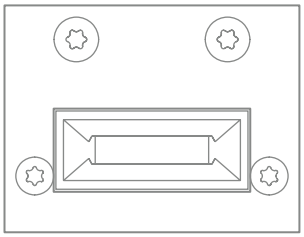
BNC.IO



BNC	**2 x BNC socket (coaxial)*** OUT: MADI output, connect for MADI output signal here. IN: MADI input, connect MADI input signal here

* The labelling of the MADI ports is printed on the Mainframe.

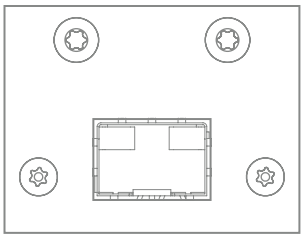
SC.IO



SC	**2 x SC socket (optical)*** OUT: MADI output, connect for MADI output signal here. IN: MADI input, connect MADI input signal here

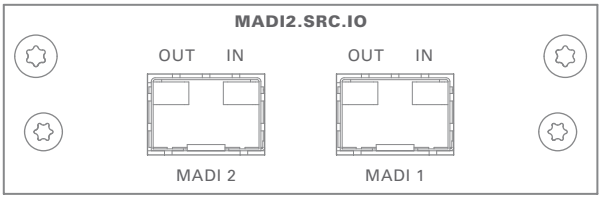
* The labelling of the MADI ports is printed on the Mainframe.

SFP.IO



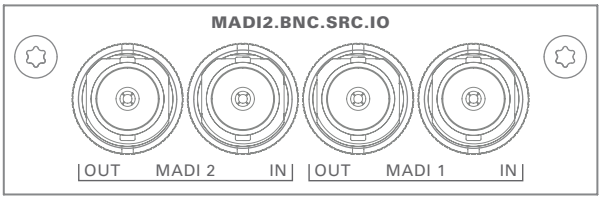
SFP	**SFP cage** Insert SFP transceiver here and connect MADI input/output.

MADI2.SRC.IO



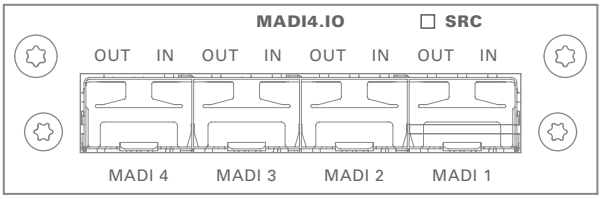
SFP	**2 x SFP cage** Insert SFP transceiver(s) here and connect MADI input/output.

MADI2.BNC.SRC.IO



BNC	**2 x BNC socket (coaxial)*** OUT: MADI output, connect for MADI output signal here. IN: MADI input, connect MADI input signal here

MADI4.IO / MADI4.SRC.IO (PRODIGY.MX only)



SFP	**4 x SFP cage** Insert SFP transceiver(s) here and connect MADI input/output.

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Slots - Converter Modules

The converter slots can be equipped with analog or digital modules. The pin-out of the DSUB-25 sockets is according to AES59 ('Tascam').

Analog Modules

Module	Input	Output
AN8.O		8 ch line out
AN8.IO	8 ch line in	8 ch line out
AN8.I	8 ch line in	
MIC8.HD.I	8 ch mic input (HD)	
MIC8.HD.IO	8 ch mic input (HD)	8 ch line out
MIC8.LINE.IO	8 ch mic input	8 ch line out
MIC8.LINE.I	8 ch mic input	

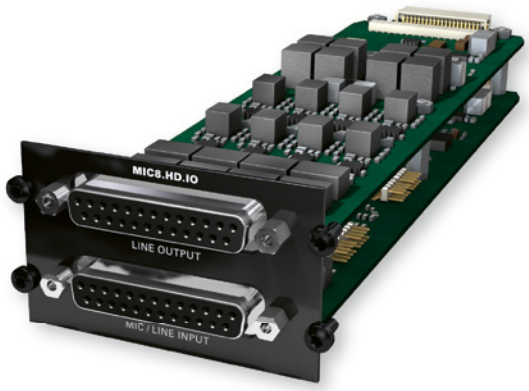
AN8.O

AN8.IO

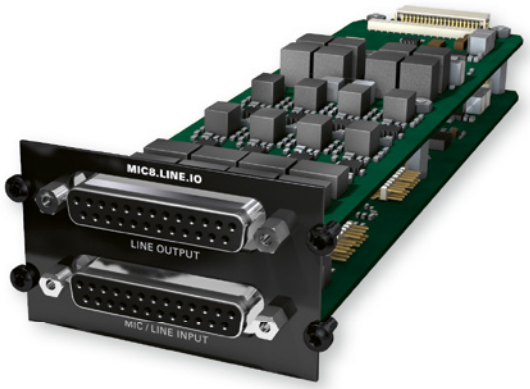
AN8.I

MIC8.HD.I

MIC8.HD.IO



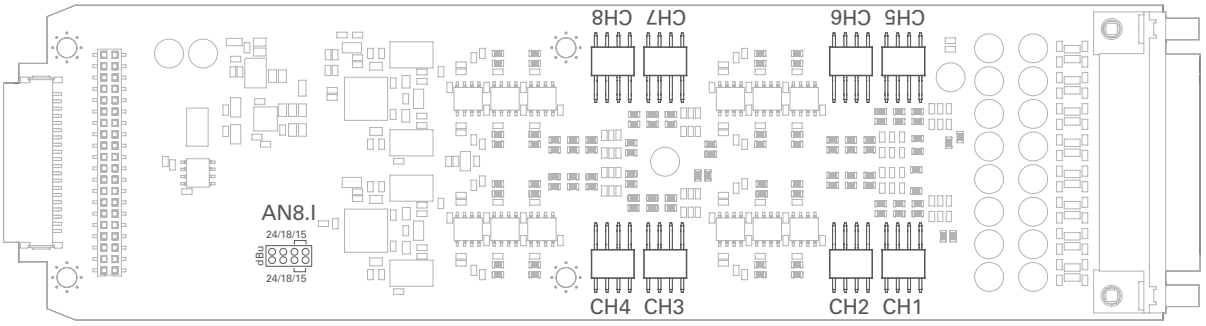
MIC8.LINE.IO



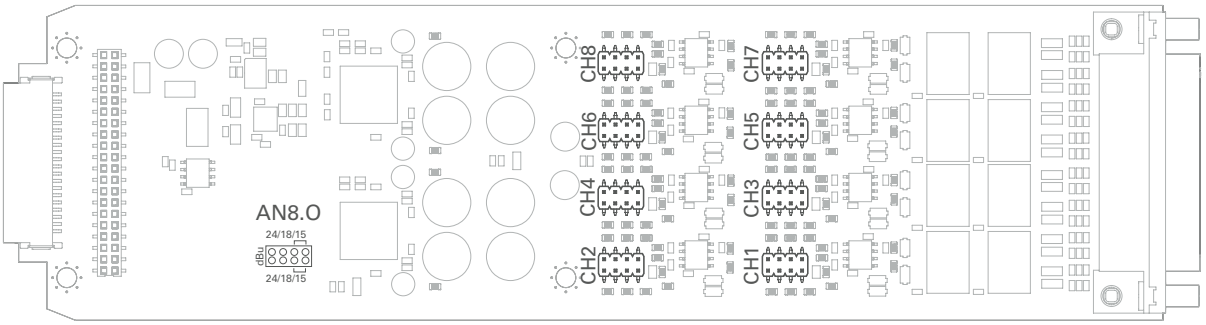
MIC8.LINE.I

Reference Level

The analog reference level of line inputs and outputs can be adjusted via jumper - for each channel individually. Check the label on the particular board. At modules with two boards the input is at the bottom and the pins are accessed from the side, where as the output is at the top and the pins are accessed from the top.

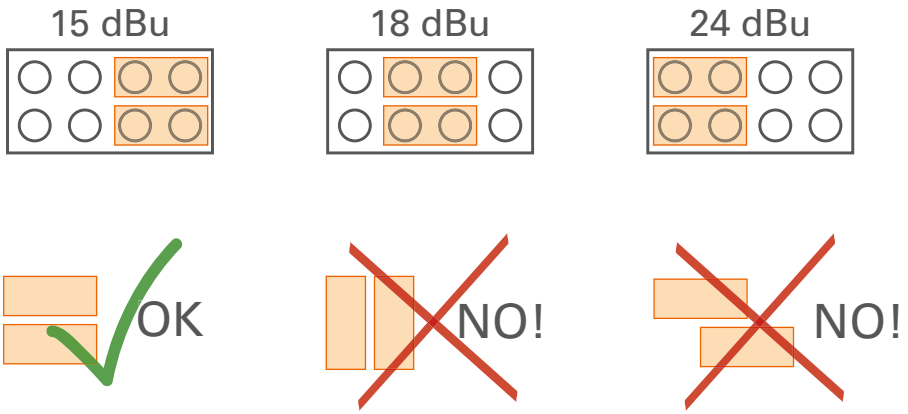


AN8.I - access from sideways



AN8.O - access from top

Level Settings



WARNING

To prevent damage from the board only set the jumpers parallel in horizontal direction - as illustrated above.

NOTE

The pinout of the digital and analog I/O is different. Check for appropriate cabling to ensure proper operation and to avoid damages caused by improper connections.

The analog outputs are fed by the D/A converters and not a split-out of the analog inputs.

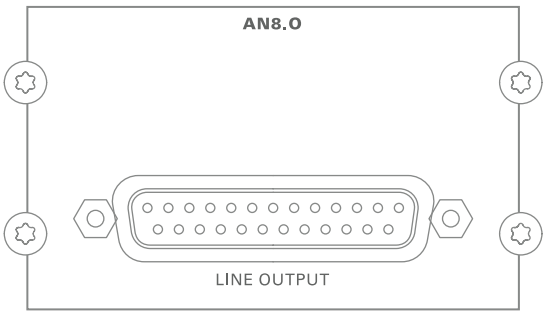
WARNING

Do not connect voltage sources to the analog outputs. This may cause damage at the output stages. Observe the technical specifications listed in this document.

WARNING

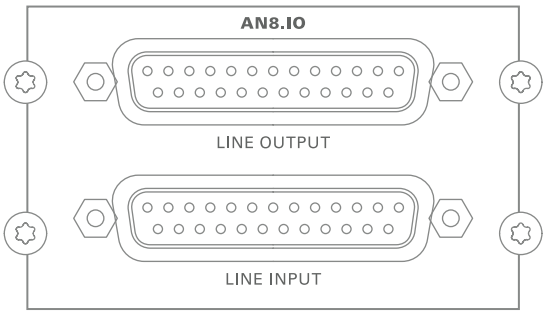
The line output is not servo balanced. Do not connect the negative lead to ground. This may cause damage at the output stage. Observe the technical specifications listed in this document.

AN8.O



LINE OUTPUT	**DSUB-25 Port (analog pinout)** Analog audio output (balanced) - connect for line level audio here

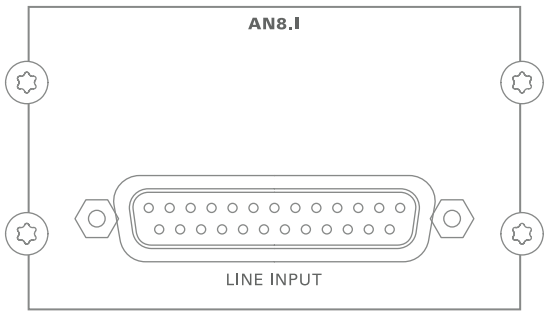
AN8.IO



LINE OUTPUT	**DSUB-25 Port (analog pinout)** Analog audio output (balanced) - connect for line level audio here
LINE INPUT	**DSUB-25 Port (analog pinout)** Analog audio input (balanced) - connect line level audio source here

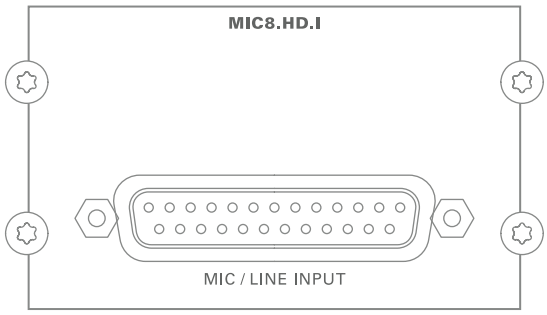
The pinout complies with AES59 ('TASCAM pinout') - see „Appendix A- DSUB-25 Pin assignment“ on page 148.

AN8.I



LINE INPUT	**DSUB-25 Port (analog pinout)** Analog audio input (balanced) - connect line level audio source here

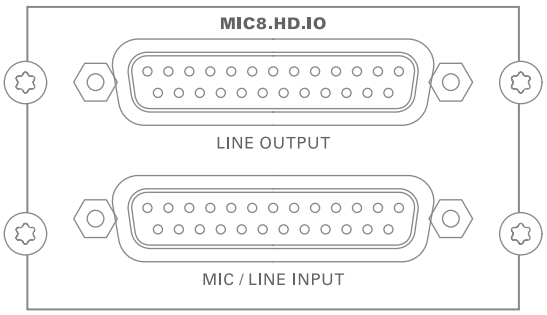
MIC8.HD.I



MIC / LINE INPUT	**DSUB-25 Port (analog pinout)** Analog audio input (balanced) - connect mic / line level audio source here

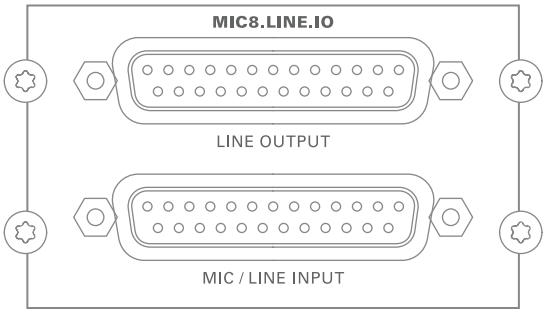
The pinout complies with AES59 ('TASCAM pinout') - see „Appendix A- DSUB-25 Pin assignment“ on page 148.

MIC8.HD.IO



LINE OUTPUT	**DSUB-25 Port (analog pinout)** Analog audio output (balanced) - connect for line level audio here
MIC / LINE INPUT	**DSUB-25 Port (analog pinout)** Analog audio input (balanced) - connect mic / line level audio source here

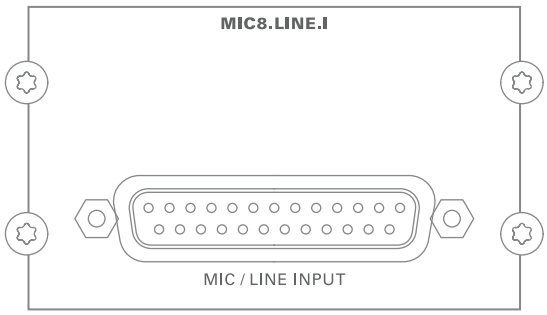
MIC8.LINE.IO



LINE OUTPUT	**DSUB-25 Port (analog pinout)** Analog audio output (balanced) - connect for line level audio here
MIC / LINE INPUT	**DSUB-25 Port (analog pinout)** Analog audio input (balanced) - connect mic / line level audio source here

The pinout complies with AES59 ('TASCAM pinout') - see „Appendix A- DSUB-25 Pin assignment“ on page 148.

MIC8.LINE.I



MIC / LINE INPUT	**DSUB-25 Port (analog pinout)** Analog audio input (balanced) - connect mic / line level audio source here

The pinout complies with AES59 ('TASCAM pinout') - see „Appendix A- DSUB-25 Pin assignment“ on page 148.

Digital Modules

Module	Input	Output
AES4.IO	4 port AES3 in	4 port AES3 out
AES4.SRC.IO	4 port AES3 in with SRC	4 port AES3 out

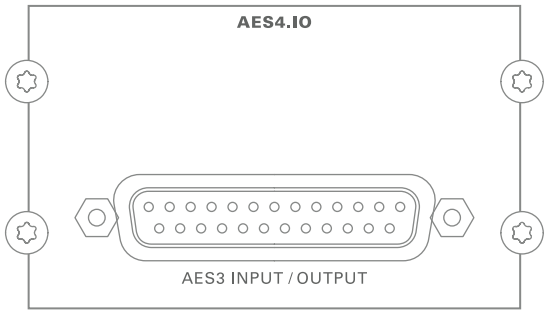
AES4.IO

AES4.SRC.IO

NOTE

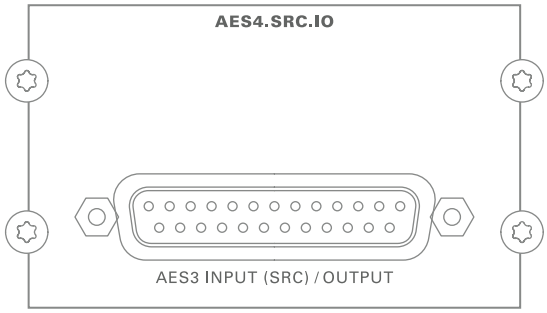
An AES4.IO transports eight audio channels - two audio channels per port.

AES4.IO



AES3 INPUT / OUTPUT	**DSUB-25 Port (digital pinout)** Digital audio input and output (AES3) - connect for digital audio here

AES4.SRC.IO



AES3 INPUT (SRC) / OUTPUT	**DSUB-25 Port (digital pinout)** Digital audio input and output (AES3) - connect for digital audio here. Input offers switchable Sample Rate Conversion.

The pinout complies with AES59 ('TASCAM pinout') - see „Appendix A- DSUB-25 Pin assignment“ on page 148.

Word clock

Two word clocks inputs and outputs are available. The word clock output provides the system clock that is either derived from PTP / Network Audio input, AES input, word clock input, MADI input or internal clock generator. WCK 2 input may be switched to video sync.

WCK 1 OUT	**BNC socket (coaxial), 75 Ω** System clock output - connect for word clock output signal here.
WCK 1 IN	**BNC socket (coaxial), 75 Ω** Connect word clock or AES3 DARS (Digital Audio Reference Signal) here.
WCK 2 OUT	**BNC socket (coaxial), 75 Ω** Word clock output - connect for secondary word clock output signal here.
WCK 2 IN	**BNC socket (coaxial), 75 Ω** Connect secondary word clock or AES3 DARS (Digital Audio Reference Signal) or video signal here.

Both word clock inputs also accept a AES3 frame (AES11).

When switched to video sync, the WCK 2 input accepts black burst or tri-level sync.

Termination (75 Ω) for the word clock inputs is switchable locally or via remote control.

Clocking

The device offers several options for clocking.

- PTP (Precision Time Protocol) - IEEE1588-2008 / PTPv 1 & 2 *
- SoundGrid (SoE) *
- AVB *
- Word Clock / Video
- MADI input *
- AES3 input *
- internal clock generator

* depending on the modules installed.

When PTP (network) is selected the device can either act as slave or as network grandmaster.

The front panel informs about selected clock sources and their lock / sync state.

SYNC	**LED RGB - indicates the lock / sync state of MADI input, Network-clock, word clock or AES3 input.**
MADI 1	(OFF) = no signal lock
MADI 2	(ON, green) = signal lock, in sync
WCK 1	(ON, blue) = signal lock, in sync, selected clock source
WCK 2 / VIDEO	(ON, yellow) = signal lock, sample rate conversion active
NET 1	(blinking, green) = signal lock, not in sync with selected clock source
NET 2	(blinking, red) = input selected as clock source and no signal lock.
AES	

USB

Two USB ports at the front and rear panel are used for legacy control of ANDIAMO devices via globcon bridge.

USB (front / rear)	**USB 2.0 socket (Type B)** Connect here for legacy control.

MIDI

PRODIGY.MP offers MIDI control to either trigger device functions (input) or to control external equipment via globcon (output).

MIDI OUT / IN	**2 x DIN 5-pin connector** OUT: MIDI output, connect for MIDI output signal here. IN: MIDI input, connect MIDI input signal here.

GPIO

General Purpose Input and Output
Two MOSFET switches (2 x GPO) can be triggered. A power supply (12 V, max. 200 mA) is also provided. This allows to remote control external devices; e.g. a recording light.

Two GPIs can be triggered by connecting the input pin with ground (GND) or by a voltage source between input pin and ground. The high level of the voltage may range between 2 V and 30 V due to a safety limiter in the input.

GPIO	**DSUB-9 socket (female)** Connect for GPIO application here.

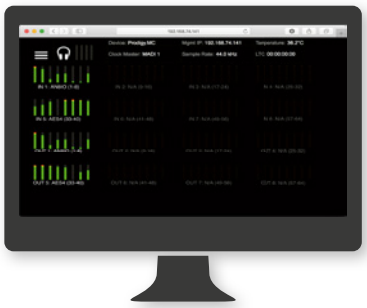
Observe the pin out - see „Appendix B- DSUB-9 Pin assignment“ on page 149.

CHAPTER 5: Managing Device

Introduction

PRODIGY.MP can be managed via multiple user interfaces:

- local touch display on the front panel



- browser based control / website

- globcon control

Further a JSON API (JavaScript Object Notation) is available for management.

This documentation explains the management via touch-display and browser control. For the use of globcon control there are video tutorials available that can be found at www.globcon.pro.

NOTE

The network settings of the audio network modules are not managed via globcon, refer to the particular chapters RAV.IO (p 82) or DANTE.IO (p 128).

Getting started

1. Check the hardware - the housing must be closed completely.
   Empty slots require covering with a blank plate.

2. Check network connection with one of the three MGMT ports on the rear panel.

3. Check the power cabling and switch on the device.

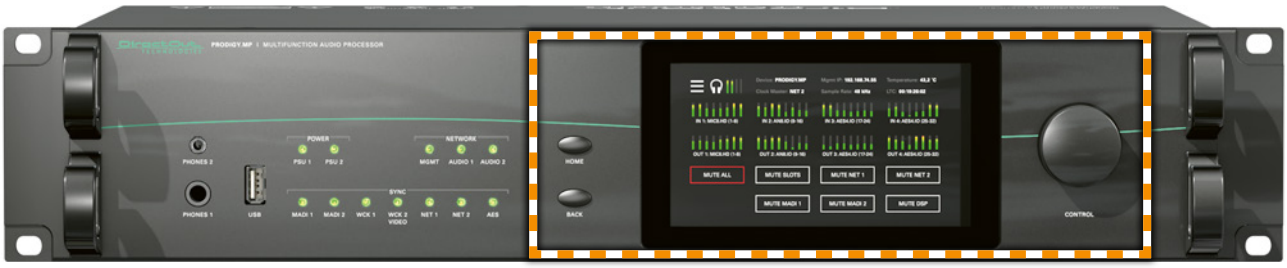
4. After the boot process of the device is complete. Check the display on the front panel for the IP address.

NOTE

The device will look for a DHCP server in the network to obtain its IP address. However the IP address can be changed manually via the touch display - see “NETWORK SETTINGS” on page 74.

Local operation

The device is equipped with a touch-display, an encoder and two push buttons for local control. For remote operation a network socket is provided at the rear panel to operate the device via a browser based GUI or via globcon control.



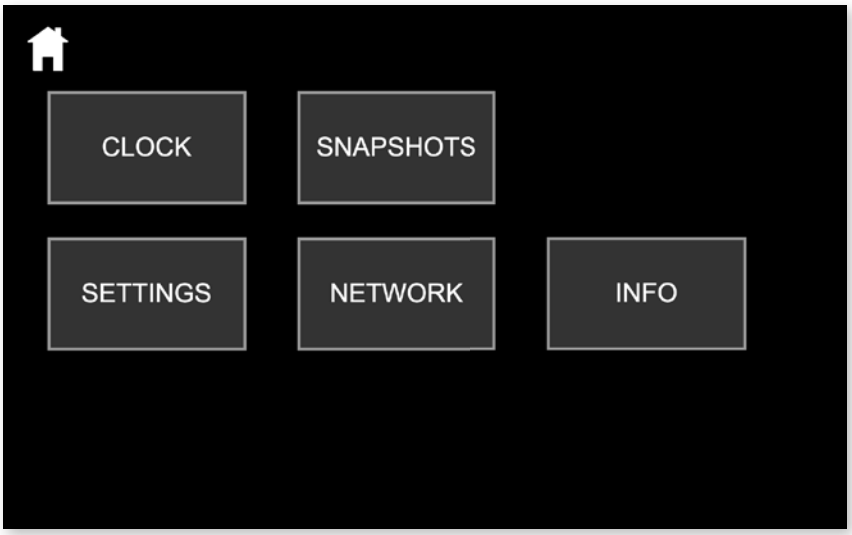
HOME	**Push-button to access the HOME screen.** Press shortly to call the top menu.
BACK	**Push-button to return to previous page.** Press shortly to return to the previous page.
Display	**Touch-Display to navigate the menu, adjust settings and for monitoring.** tap, swipe- have fun!
CONTROL	**Encoder to adjust values and confirmation and for volume control of the phones outputs.** Rotate to modify values. Push to confirm a setting or to zero a value. Phones volume control - active in HOME view: push to enter volume control or to toggle the selection between PHONES 1 and PHONES 2. Rotate to modify values. Automatic timeout of the control display is about 5 seconds.

NOTE

For remote operation via browser, enter the IP-Address in the navigation bar of your browser. Local and remote display are identical.

For remote operation via globcon control, you need to install globcon on your computer first - see page 16. Then follow the steps described in "CHAPTER 3: Installation"on page 24.

MAIN MENU

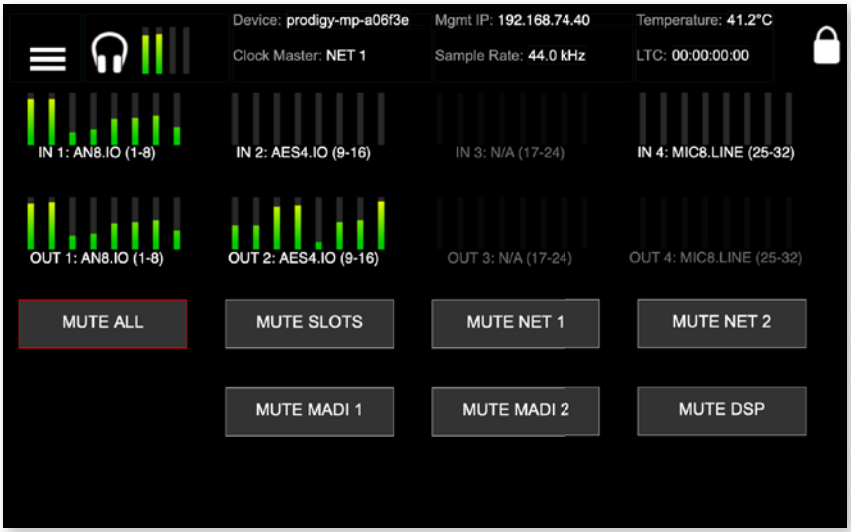


The MAIN MENU displays:

- Icons to access the HOME menu and further settings

Tap / click	**to open**
Home icon	HOME Menu
Clock	Clock Settings
Snapshots	Snapshot list for recall
Settings	Network Settings for the Management Ports
Info	Display of Device Name, System Build, FPGA Version, cored version, Frontpanel version, License information Download Support Archive

HOME



The HOME screen displays:

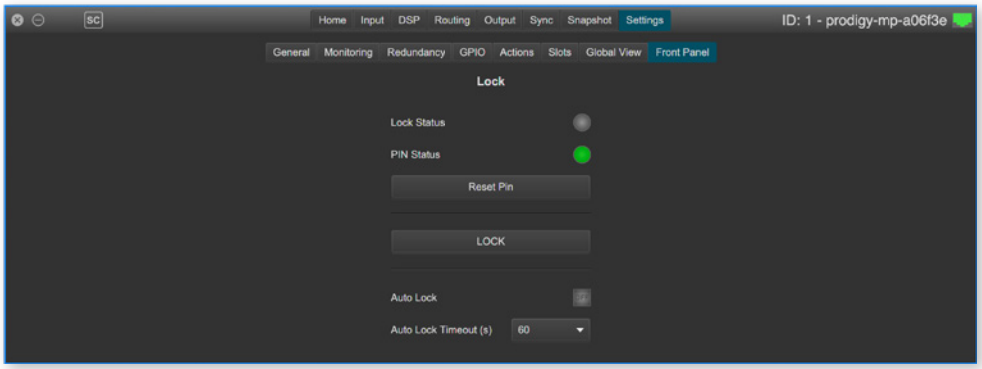
- Level metering (input / output) of all eight converter slots
- Level metering of headphones outputs
- Device name (Device)
- IP address for device management (Mgmt IP)
- Temperature
- Clock Master
- Sample Rate
- LTC Reader display (LTC)
- MUTE buttons to mute all or individual outputs or signal paths
- Lock icon to lock / unlock access

Tap / click	**to open / <function>**
Burger icon	Main Menu
Clock Master	Clock Settings
Mgmt IP	Network Settings
Level Meter	zoomed display of eight channels
Encoder (CONTROL)	Phones volume control or to toggle the selection between PHONES 1 and PHONES 2. Rotate to modify values. Automatic timeout of the control display is about 5 seconds. *

* When controlling via browser the phones volume control is accessed by pressing the <ENTER> key on the keyboard. Level is adjusted via arrow keys (left and right).

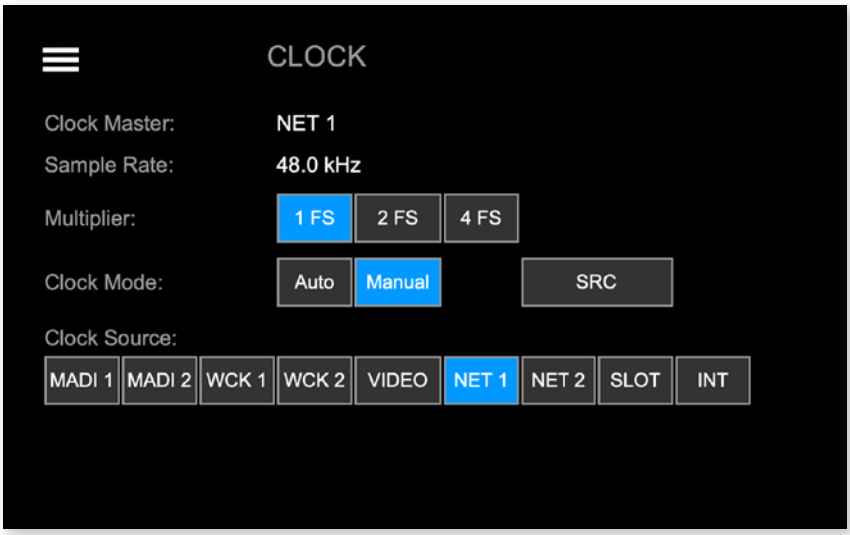
Tap / click	**to open / <function>**
Lock icon	Lock or unlock the device control with a pin code. *
MUTE ALL	Mute / Un-mute all physical outputs and signal paths
MUTE SLOTS	Mute / Un-mute all outputs of converter slots
MUTE NET 1	Mute / Un-Mute output of network slot NET 1
MUTE NET 2	Mute / Un-Mute output of network slot NET 2
MUTE MADI 1	Mute / Un-Mute output of MADI slot MADI 1
MUTE MADI 2	Mute / Un-Mute output of MADI slot MADI 2
MUTE DSP	Mute / Un-Mute of DSP blocks (not bypassing) **

* A four-digit pin code needs to be defined when locking for the first time. To unlock device the pin code is required. The pin code can be reset in the PRODIGY.MP plugin of globcon (Settings / Front Panel) when the device is locked or in the device settings (see "DEVICE SETTINGS" on page 75) when the device is unlocked.



** MUTE DSP will interrupt the signal chain at the DSP block section, but not mute a physical output.

CLOCK



The CLOCK Settings displays:

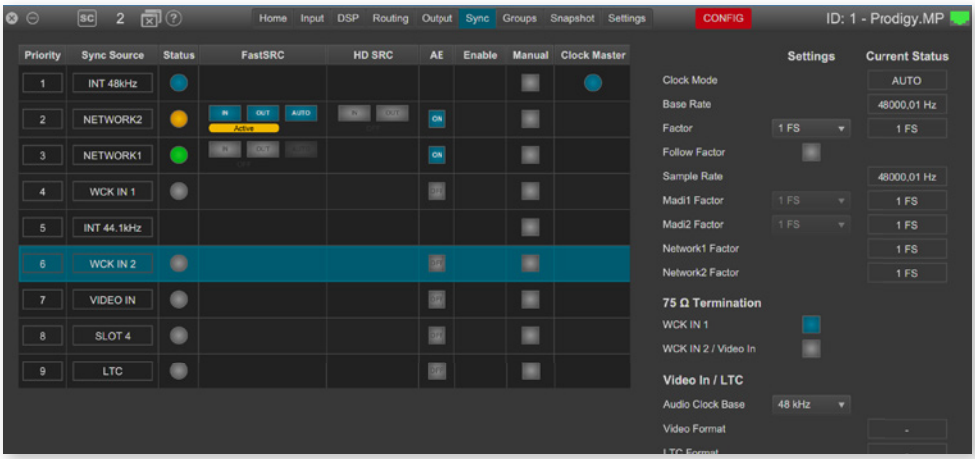
- Clock Master: Current master for system clock
- Sample Rate: measured sample rate
- Icons to adjust the scaling factor of the base rate
- Icons to select Clock Mode (Auto or Manual) and to open SRC settings
- Icons to select the clock master

Tap / click	**to open / set***
Burger icon	Main Menu
1 FS	Scaling factor to 1 FS (usually 44.1 or 48 kHz)
2 FS	Scaling factor to 2 FS (usually 88.2 or 96 kHz)
4 FS	Scaling factor to 4 FS (usually 176.4 or 192 kHz)
Auto	Clock source is set automatically following the priority settings in globcon.**
Manual	Clock source is set manually according the selection below (Clock Source:).
SRC	Submenu to activate / deactivate SRC - see p 68
MADI 1	MADI 1 input as Clock Master
MADI 2	MADI 2 input as Clock Master
WCK 1	Word Clock input 1 as Clock Master
WCK 2	Word Clock input 2 as Clock Master

VIDEO	Video signal @ Word Clock input 2 as Clock Master***
NET 1	Audio Network Slot 1 as Clock Master
NET 2	Audio Network Slot 2 as Clock Master
SLOT	AES3 input as Clock Master ****
INT	Internal Generator as Clock Master
44.1	Base Rate to 44.1 kHz [INT]
48	Base Rate to 48 kHz [INT]

* the value set is marked by the particular icon with a blue background

** The priority setting allows to define several fallbacks if a clock source fails. It also offers reverting to a higher prioritized clock source (Auto Enable / Enable) when its signal returns.

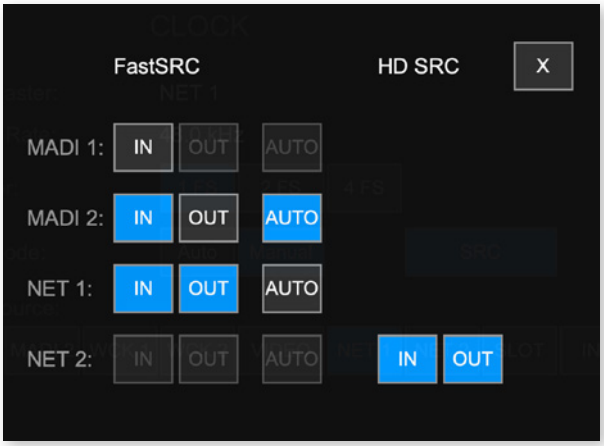


*** Word Clock input 2 also accepts analog video signal (black burst, tri-level)

**** requires AES4.IO or AES4.SRC.IO installed - the first AES3 port receiving a valid signal will be used as clock source.

FastSRC™ / HD SRC

The FastSRC™ is a low latency sample rate converter that is available for the MADI I/Os and the audio network I/Os. HD SRC is a state-of-the-art sample rate converter which is offered with the audio network modules RAV.SRC.IO, DANTE. SRC.IO and SG.SRC.IO.



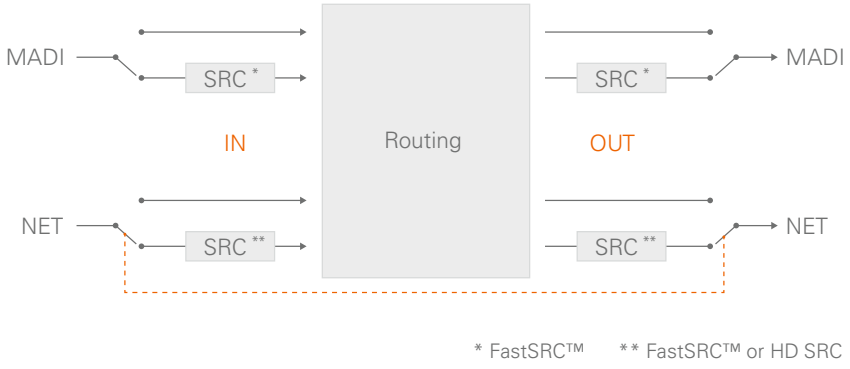
The SRC screen displays:

- buttons to activate or deactivate FastSRC™ or HD SRC at individual I/Os

Tap / click	**to open / set***
X	Clock page
MADI 1/2 - IN	Activate / Deactivate FastSRC™ at MADI 1/2 input
MADI 1/2 - OUT	Activate / Deactivate FastSRC™ at MADI 1/2 output
MADI 1/2 - AUTO	Activate / Deactivate Automatic Bypass for FastSRC™ at MADI 1/2
NET 1/2 - IN or OUT	Activate / Deactivate FastSRC™ at NET 1/2 input & output*
NET 1/2 - AUTO	Activate / Deactivate Automatic Bypass for FastSRC™ at NET 1/2 input & output
NET 1/2 IN or OUT (HD SRC)	Activate / Deactivate HD SRC at NET 1/2 input & output**

* For the audio network I/Os the FastSRC™ may be activated for input and output together only.

** With HD SRC active, the FastSRC™ and the automatic bypass become inactive.



NOTE

If an input is used as clock source for the device it is neither necessary nor recommended to activate the FastSRC™ or HD SRC.

Automatic Bypass for FastSRC™

With ‚AUTO' activated sample rate conversion is applied depending from the sync state of the input signal:

- in sync with the system clock
- not in sync with the system clock

AUTO	FastSRC™	Signal in Sync	Sample Rate Conversion	Latency added
n.a.	OFF	yes / no	inactive	no
OFF	ON	yes / no	active	< 0.15 ms
ON	ON	yes	inactive	< 0.15 ms
		no	active	
		yes again*	automatic bypass after 60 seconds	

* 60 seconds after the signal is in sync with the system clock again, sample rate conversion is bypassed.

The automatic switch-over will not change the latency (less than 0.15 ms) of the signal path.

About FastSRC™

DirectOut's FastSRC™ is a low latency sample rate converter for when two digital interfaces of a device must work in different clock domains.

FastSRC™ combines good sound quality with very low latency of less than 0.15 msecs and is invaluable in live sound applications and a "life-saver" in critical situations.

High-end sample rate conversion requires noticeable processing power, so for the very best audio quality we recommend DirectOut's dedicated SRC products such as the SRC version of the audio network modules or the multi-port MADI modules or the MADI.SRC- see About HD-SRC.

About HD SRC

DirectOut's HD SRC is a state-of-the-art sample rate converter with a latency of about 1 msec.

HD SRC is available for the audio network modules:

- RAV.SRC.IO
- DANTE.SRC.IO
- SG.SRC.IO
- MILAN.SRC.IO

or the multi-port MADI modules:

- MADI2.SRC.IO
- MADI2.BNC.SRC.IO
- MADI4.SRC.IO.

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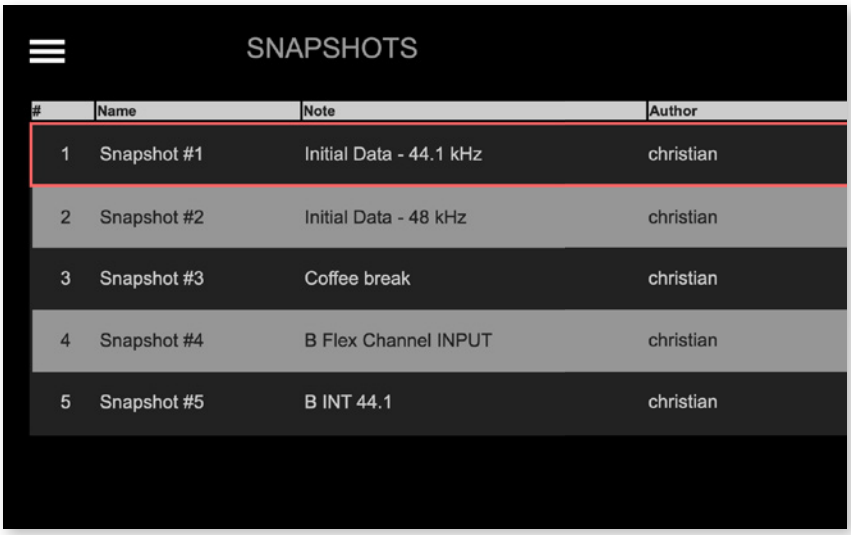
SNAPSHOTS



The SNAPSHOT screen displays:

- list of snapshots that are stored in the device (up to 99 snapshots)

Tap / click	**to recall**
snapshot entry	snapshot - after confirming a prompt

Each snapshot which is created in globcon will be stored into the device.

As a snapshot can be stored within a globcon project, it is possible to reconfigure a device including snapshots via 'Push' function from globcon.

The selected snapshot is surrounded by a red frame.

NOTE

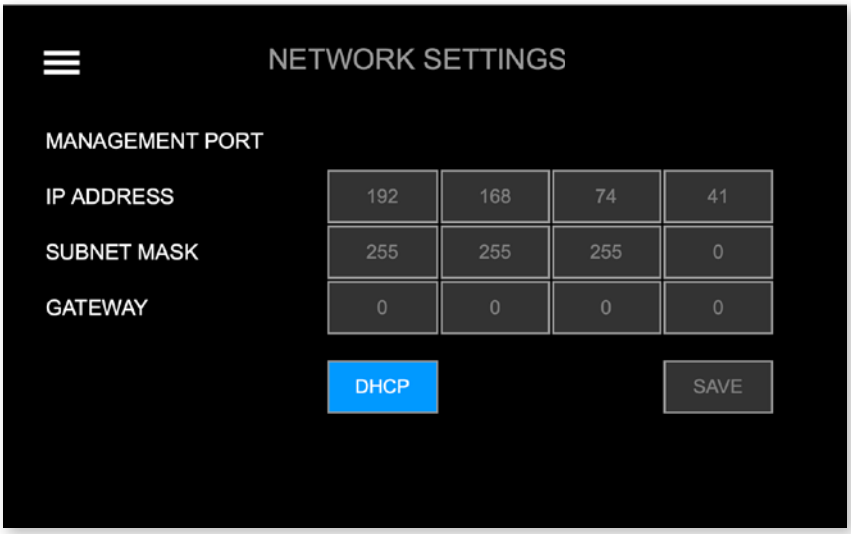
Only compliant snapshots are listed and available for recall.

Snapshots - Compliance

compliant	Snapshot Data Set matches the Configuration Settings of the device.
non-compliant	Snapshot Data Set does not match the Configuration Settings of the device. Conform procedure required for recall in Show Mode or recall with full Scope Set in Configuration Mode, thus changing the Configuration Settings.

See document 'Info PRODIGY.MP Supplemental' for more information about the Operating Modes and Snapshots.

NETWORK SETTINGS



The NETWORK Settings displays:

- IP Address for the management ports (MGMT)
- Subnet Mask for the management ports (MGMT)
- Gateway for the management ports (MGMT)

Tap / click	**Function**
Burger icon	Link to Main Menu
DHCP	Enable / Disable DHCP * grey = DHCP OFF, manual input required blue = DHCP ON, network settings are assigned automatically
SAVE	Stores changes to the device
Input field	Input of values **

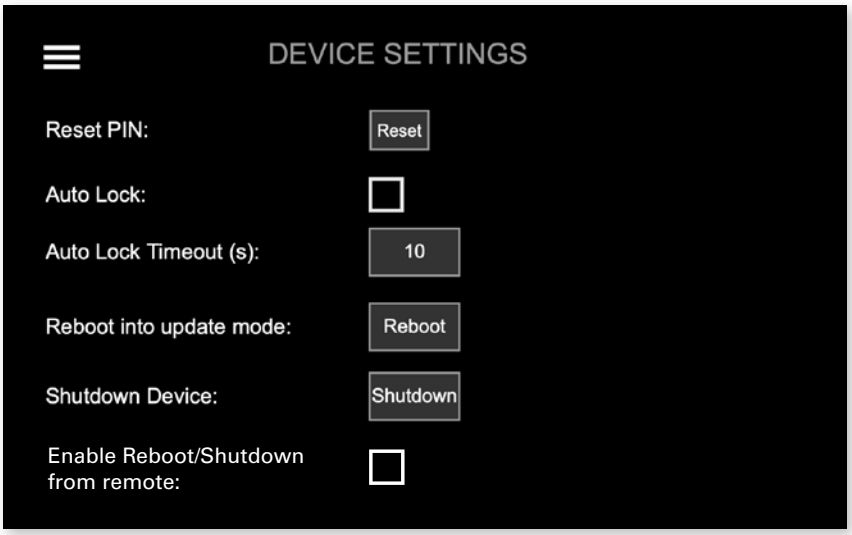
* DHCP (Dynamic Host Control Protocol) = device's IP Address is assigned automatically by a DHCP server in the network

** the values are entered via the Encoder if DHCP is OFF

Enter values:

- select field
- turn the Encoder to adjust the value or use the arrow keys (left and right)
- de-select field
- tap SAVE to store the changes

DEVICE SETTINGS



The DEVICE Settings allows to:

- Reset the four-digit PIN code used for device lock
- Enable / disable Auto Lock
- Define the timeout for Auto Lock function
- Reboot the device into update mode for system updates or license installation
- Shut down the device to ensure proper file operation before switching the device off.
- Manage security setting for shutdown or reboot from remote - see p 76.

Tap / click	**Function**
Reset	Resets the PIN code.
Auto Lock	Enable ■ Disable □
Auto Lock Timeout (s)	Timeout in seconds to lock the device automatically when Auto Lock is enabled. Values: 10 to 180 seconds
Reboot	Reboots the device in update mode
Shutdown	Terminates all services, fans may run at full speed, leds and display are switched off, LED MGMT = blue
Enable Reboot/ Shutdown from remote*	Enable ■ Disable □

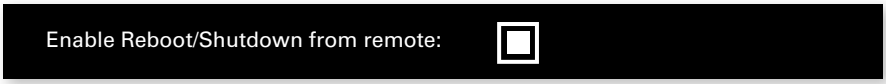
* at front panel only

Enter values:

- select field
- turn the Encoder to adjust the value or use the arrow keys (left and right)
- de-select field

Shutdown / Reboot into update mode

Both commands can also be used remotely via globcon or the web UI, once the feature has been activated on the front panel:



This setting is persistent and will be restored after a power-cycle or reboot.

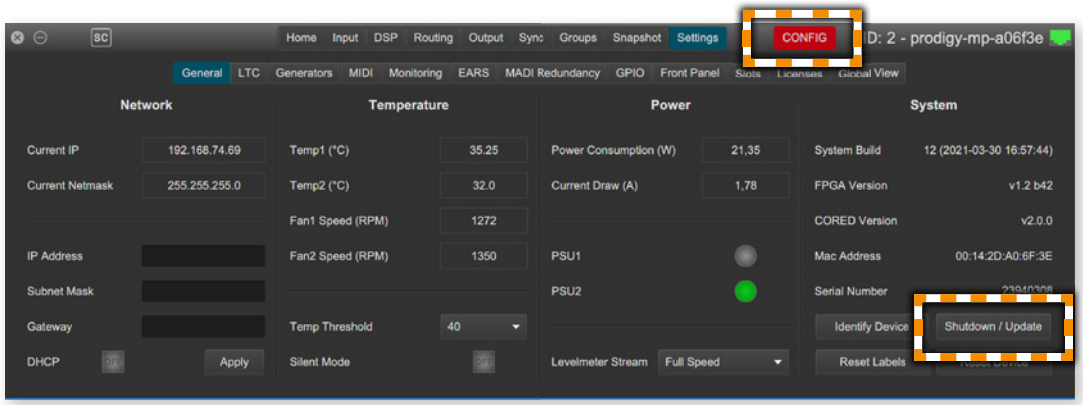
WARNING

Enabling 'Reboot/Shutdown from remote' can be a security risk, if the management network can be accessed freely. It is disabled by default.

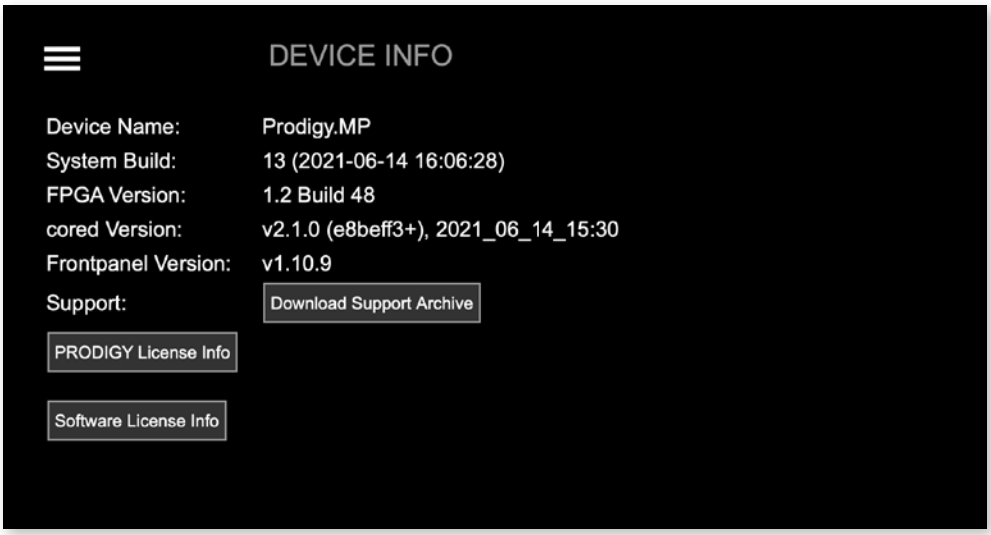
To use the new commands in globcon:

- activate the Configuration Mode
- go to Settings - General

You will be prompted to type 'shutdown' or 'update' to use either command.



DEVICE INFO



DEVICE INFO displays:

- Device Name
- System Build: Version and date
- FPGA Version: audio processing
- cored Version: device management
- Frontpanel Version: local operation
- Support: Support Archive
- License information about installed licenses and their service state
- License information for open source software that is used by this device

Tap / click	**to open / generate**
Burger icon	Main Menu
Download Support Archive *	a support archive (*.pdgs) Example: prodigy_support_2019-11-19_12-34-33.pdgs
PRODIGY License Info	subpage with information about installed licenses and service state
Software License Info	subpage with license information about open source software

* the support archive is generated and downloaded into the downloads folder of your browser. You may send it to support@directout.eu for troubleshooting.

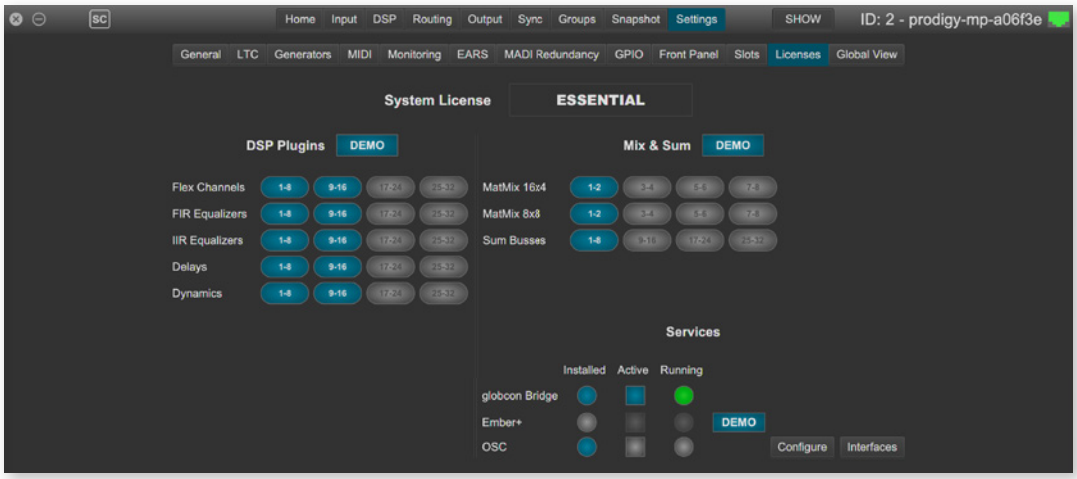
PRODIGY License Information

For enhanced scalability the PRODIGY series can be ordered with different system licenses - Essential, Advanced, Unlimited. Additional single licenses can be acquired on demand.

The PRODIGY License Information displays:

- system license (Essential, Advanced or Unlimited)
- services - license availability and its operating status
- single licenses - availability

For testing purposes a demo session can be initiated via globcon after startup to activate the feature for 30 minutes.

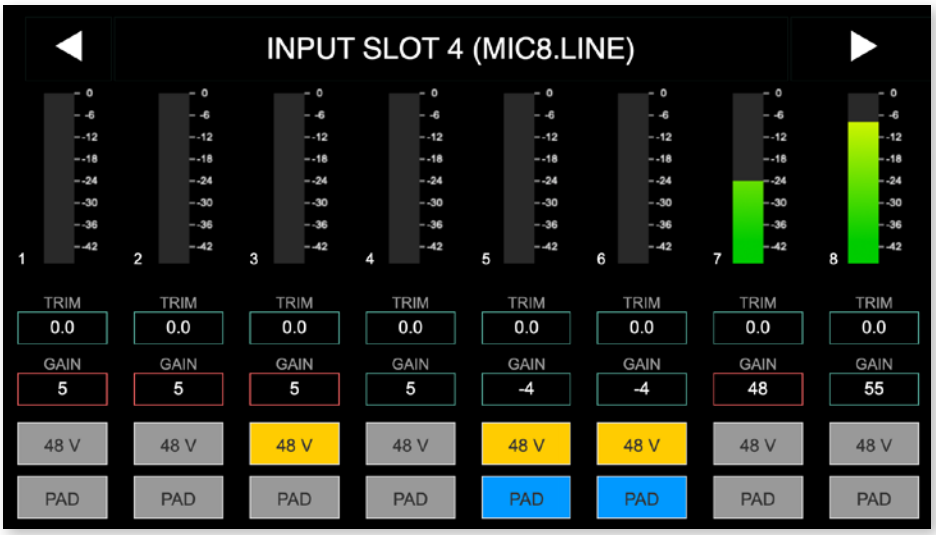


To acquire a license please contact your local DirectOut representative.

NOTE

Licenses are installed via license files which are coupled with the device's serial number. To install a license file the device must be operated in update mode - see "Appendix D- System Update & License Installation" on page 151.

LEVEL METER - INPUT (MIC8.LINE.IO)



The LEVEL METER - INPUT (MIC8.LINE.IO) displays:

- Level meters for a single converter slot- 8 channels
- Slot number and module type inserted
- control for input level, mute and phantom power (depending on the module)

Tap / click	**to**
Arrow left	navigate to previous module*
Arrow right	navigate to next module*
INPUT SLOT	return to HOME screen
Trim** / Gain	select field(s) for modifying values of trim and gain. selected unselected
48 V	phantom power (48 V) 48 V enabled 48 V disabled
PAD	enable / disable PAD (-9 dB MIC8 / - 30 dB MIC8.HD) PAD enabled PAD disabled

* the arrows are used for circular navigation through slots 1 to 4 input, followed by slot 1 to 4 output.

** displayed with a yellow background, when the value is not zero

To adjust the level:

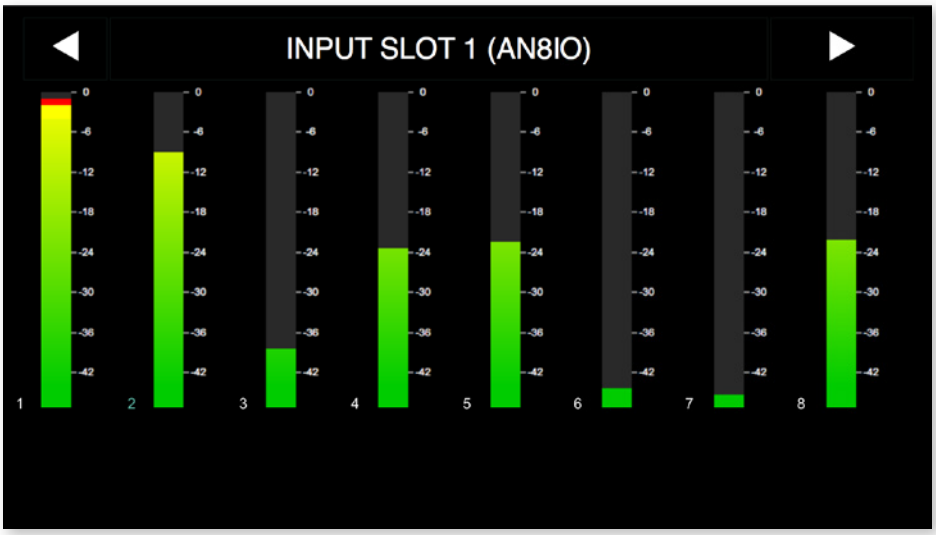
- select field(s)
- turn the Encoder or use the arrow keys (left and right)
- un-select field(s) when finished
- to zero the value, press the Encoder

Values:

Gain: module dependent

Trim: from - 24 dB to + 24 dB

LEVEL METER - INPUT (AN8.IO)



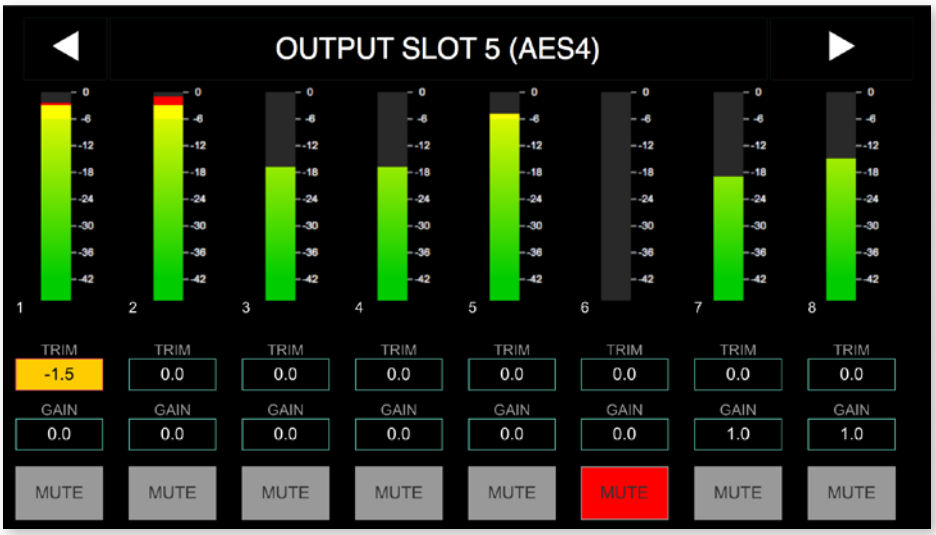
The LEVEL METER - INPUT displays:

- Level meters for a single converter slot- 8 channels
- Slot number and module type inserted

Tap / click	**to**
Arrow left	navigate to previous module*
Arrow right	navigate to next module*
INPUT SLOT	return to HOME screen

* the arrows are used for circular navigation through slots 1 to 8 input, followed by slot 1 to 8 output.

LEVEL METER - OUTPUT



The LEVEL METER - OUTPUT displays:

- Level meters for a single converter slot - 8 channels
- Slot number and module type inserted
- control for level, mute and phantom power (depending on the module)

Tap / click	**to**
Arrow left	navigate to previous module*
Arrow right	navigate to next module*
OUTPUT SLOT	return to HOME screen
Trim** / Gain	select field(s) for modifying values of trim and gain. selected unselected
MUTE	enable / disable mute MUTE enabled MUTE disabled

* the arrows are used for circular navigation through slots 1 to 4 input, followed by slot 1 to 4 output.

** displayed with a yellow background, when the value is not zero

To adjust the level:

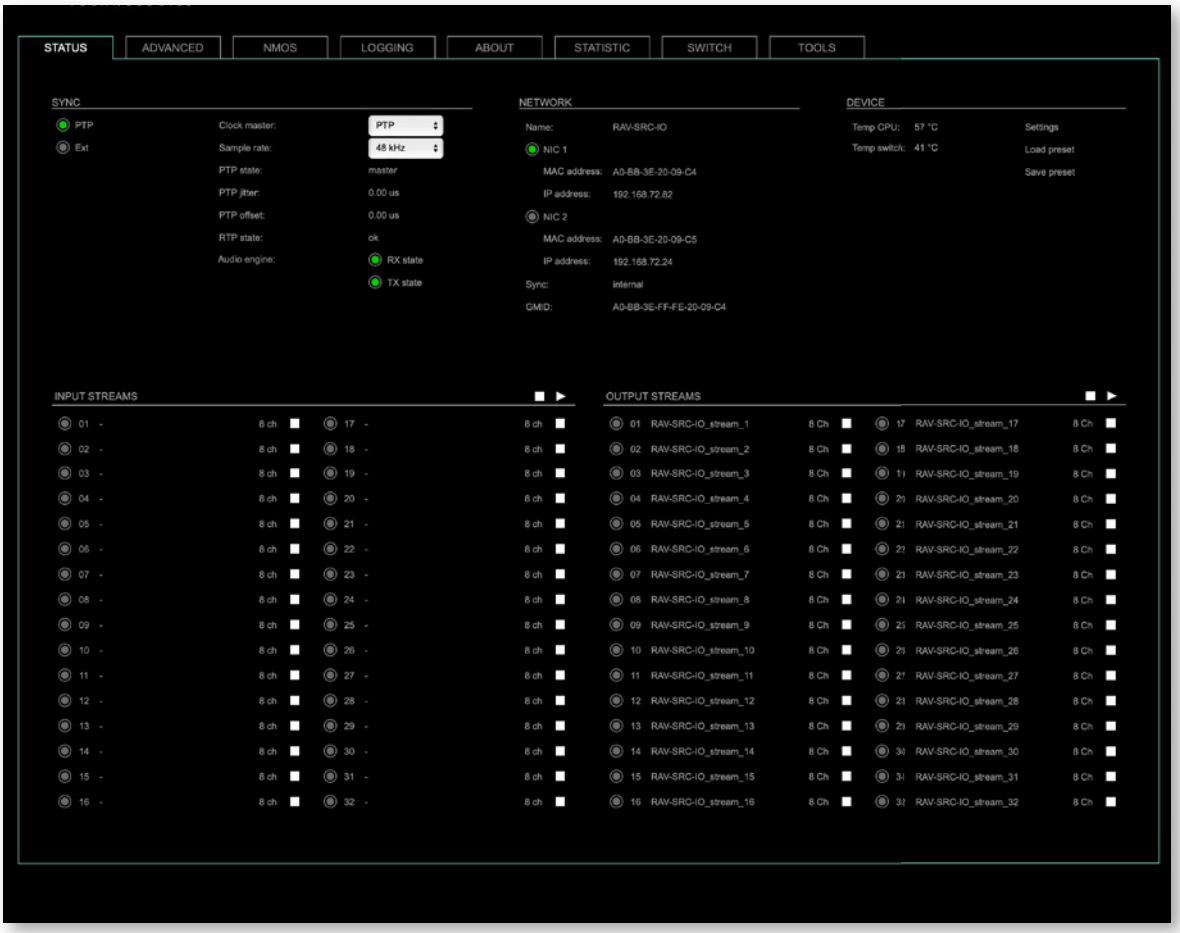
- select field(s)
- turn the Encoder or use the arrow keys (left and right)
- un-select field when finished
- to zero the value, press the Encoder

Values:
Gain: module dependent
Trim: from - 24 dB to + 24 dB

CHAPTER 6: RAV.IO

Introduction

RAV.IO is an audio network module for RAVENNA / AES67.
All functions of the device are accessible through a browser based interface (hmtl5 / javascript). The size of the window and the zoom level can be varied. The page is organized in tabs, pulldown menus or hyperlinks offer access to the values of a parameter. Some values use an input field (e.g. IP address).



NOTE

Refers to module software: SW 0.94 / HW 0.29

Connecting Audio Network

To access the control page:

- connect the network with one port
- enter http://<IP Address> (default IP @ PORT 1: 192.168.0.1) in the navigation bar of your browser

The three physical network ports (Port 1 to 3) are managed by two independent internal network interfaces (NIC 1 / NIC 2).

Port 1 is fixed assigned to NIC 1.
Port 2 and 3 can be assigned to either NIC 1 or NIC 2 on the SWITCH tab - see p 121.

NOTE

If NIC 1 and NIC 2 are connected to the same switch, they must be configured to different subnets - see “Network Settings” on page 87.

Status - Overview

The tab ‘STATUS’ is divided into several sections:

- SYNC - monitoring sync state, clock selection, links to I/O settings
- NETWORK - display network info, link to network settings
- DEVICE - monitoring device info, link to device settings, phones level control
- INPUT STREAMS - monitoring and control input streams, link to input stream settings
- OUTPUT STREAMS - monitoring and control output streams, link to output stream settings

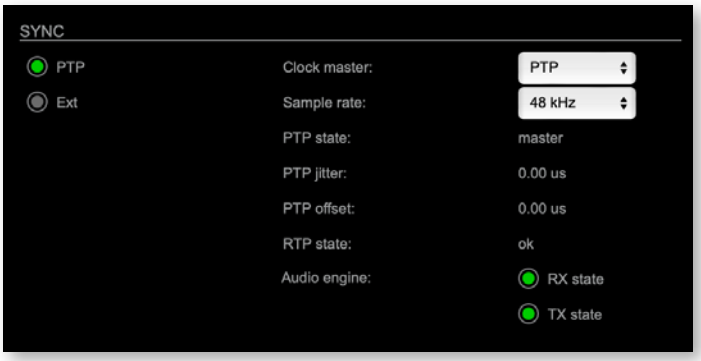
Hyperlinks open a popup window to adjust related settings. Most settings are updated immediately without further notification. To exit a popup window click the button in the top right corner.

Mouse overs are used to display further information (e.g. connection speed of network link).

NOTE

The web user interface updates itself when changes are applied by other instances (other browsers, external control commands).

Status - Sync



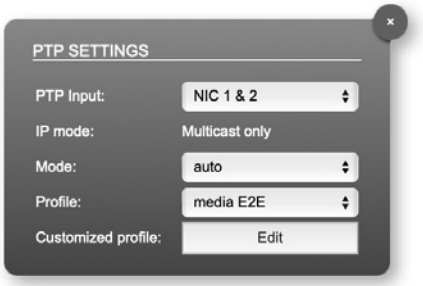
PTP, Ext	Displays clock source and state for the main frame: (OFF) = not locked (ON) = locked and in sync with clock master (blinking) = locked but not in sync with clock master
Clock master	Pulldown menu to select clock source of the main frame (PTP, extern)
Sample rate	Pulldown menu to adjust sample rate of the main frame (44.1 / 48 / 88.2 / 96 / 176.4 / 192 kHz).
PTP state	State of PTP (Master / Slave).
PTP jitter	PTP-clock jitter per second
PTP offset	Offet relative to PTP-clock master
RTP state	Status of packet processing (OK, Error*)
Audio engine RX state	State of module's audio engine- receiving (ON) = ok, receiving data (blinking) = not all received packets can be processed
Audio engine TX state	State of module's audio engine- sending (ON) = ok, sending data (blinking) = not all packets can be sent to the network

* Error: packet time stamps are out of bounds.
Possible reasons: stream offset may be too small or transmitter or receiver are not synced properly to the Grandmaster.

Hyperlinks:

- PTP / PTP state (p 85)

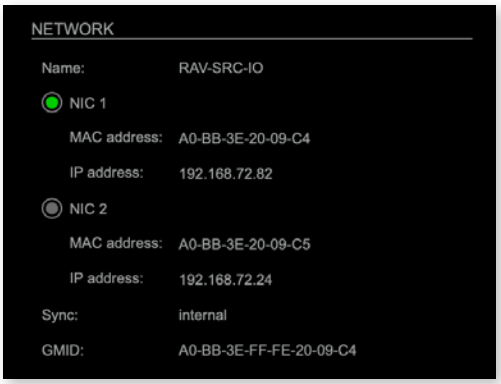
PTP Settings



PTP Input	NIC selection for PTP clock input. 'NIC 1 & 2' means input redundancy.
IP Mode	PTP via multicast, unicast or in hybrid mode. *
Mode	PTP-clock master / slave configuration is auto negotiated between devices in the network. Module's master / slave state may change automatically.
Profile	PTP profile selection (default E2E, default P2P, media E2E, media P2P, customized)
Customized profile	Edit opens the tab 'ADVANCED' to adjust the custom profile.

See „Advanced- PTP Clock Setting“ on page 111 for more details.

Status - Network



Name	Module's name in the network. Used e.g. for mDNS service. The name needs to be unique throughout the network.
NIC 1 / NIC 2	Monitoring state of network interface controller (OFF) = not connected (ON) = connected with the network
MAC address	Hardware identification of network interface controller.
IP address	IP address of device
Sync	Selected NIC for PTP sync
GMID	Grand Master ID (PTP)

Hyperlinks

- Name / IP address (p 87)

Mouse over:

- LED NIC 1- indicating link state and connection speed
- LED NIC 2- indicating link state and connection speed

NOTE

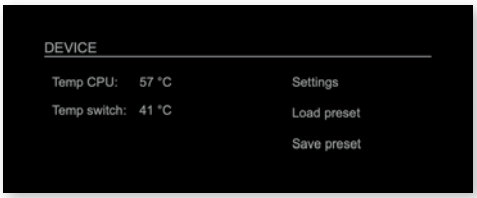
If NIC 1 and NIC 2 are connected to the same switch, they must be configured to different subnets - see "Network Settings" on page 87.

Network Settings

The two network interface controllers (NIC 1 / NIC 2) are configured individually.

Device name	Input field - Module's name in the network. Used e.g. for mDNS service. The name needs to be unique throughout the network.
Dynamic IP address (IPv4)	Switch to enable the device's DHCP client. IP address is assigned by DHCP server. If no DHCP is available the IP address is determined via Zeroconf.
Static IP address (IPv4)	Switch to disable the device's DHCP client. Manual configuration of network parameters.
IP address (IPv4)	Module´s IP Address
Subnet mask (IPv4)	Module's subnet mask
Gateway (IPv4)	IP address of gateway
DNS server (IPv4)	IP address of DNS server
Apply	Button to confirm changes. Another popup window will appear to confirm a reboot of the module.
Direct routing	IP addresses of devices outside the subnet, to enable multicast traffic; e.g. Grandmaster or IGMP querier. Mark checkbox to activate.

Status - Device

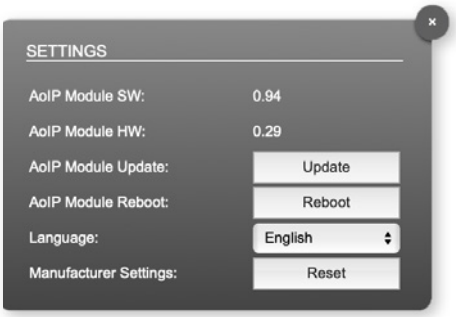


Temp CPU	Display temperature of CPU core in degree Celsius. It may reach 95 °C without effecting the performance of the device.
Temp switch	Display temperature of network switch in degree Celsius
Settings	Opens a popup window to configure the device.
Load preset	Opens a dialog to store the device settings to a file. Filetype: .rps
Save preset	Opens a dialog to restore the device settings from a file. Filetype: .rps

Hyperlinks:

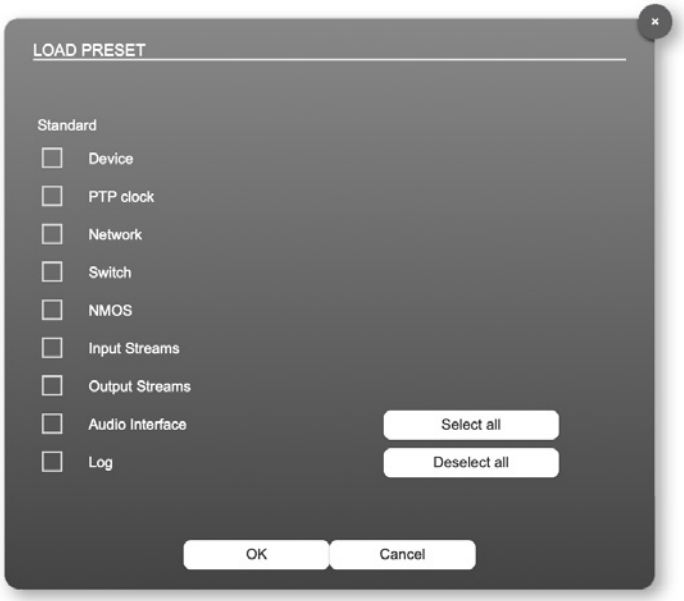
- Settings (p 88)
- Load preset (p 89)
- Save preset

Settings



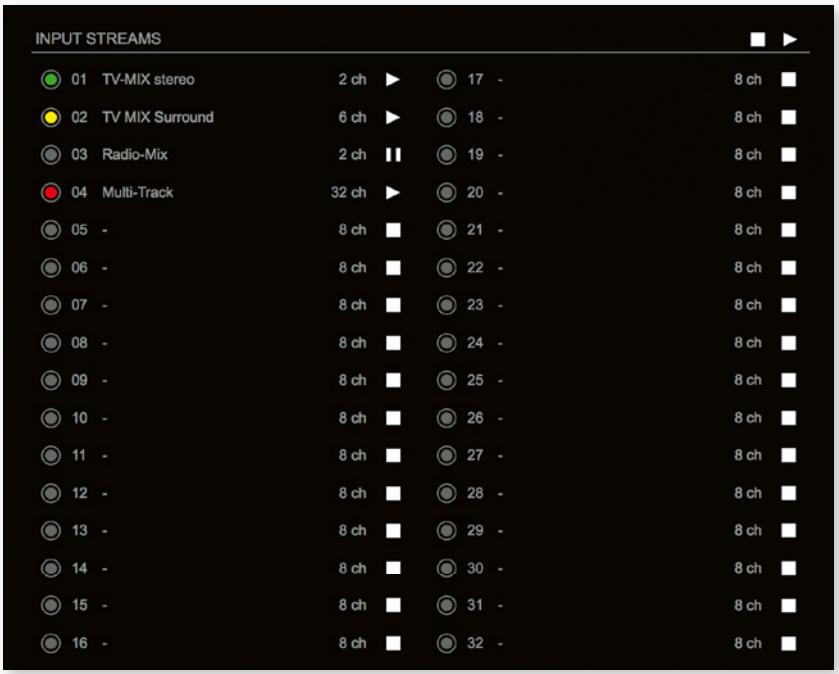
AoIP Module SW	Module´s software version. It is updated together with hardware version via network.
AoIP Module HW	Module´s bitstream version. It is updated together with software version via network.
AoIP Module Update	Opens a dialog for selection of the update file - see “RAV.IO- Firmware Update” on page 127.
AoIP Module Reboot	Restart of the AoIP module. Confirmation required. Audio transmission will be interrupted.
Language	Menu language (english, german).
Manufacturer Settings Reset	Restore device settings to factory defaults. Confirmation required.

Load Preset



The device configuration can be stored to a single file (.rps).
Restoring the configuration a dialog prompts for selection of individual settings. This enhances flexibility at setup changes when a particular adjustment shall be preserved or just a single adjustment shall be restored.

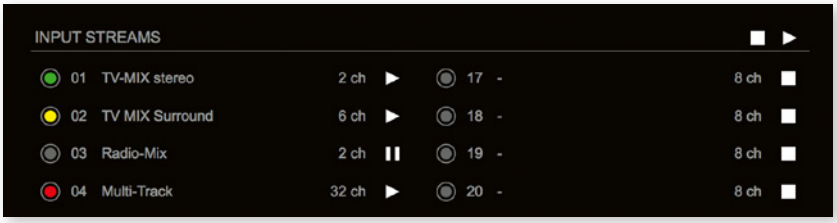
Status - Input Streams



The module can subscribe up to 32 streams. The overview displays the basic information of each stream. The input stream name can be set manually (discovery protocol: manually, see page p 99) overriding the SDP's stream name information.

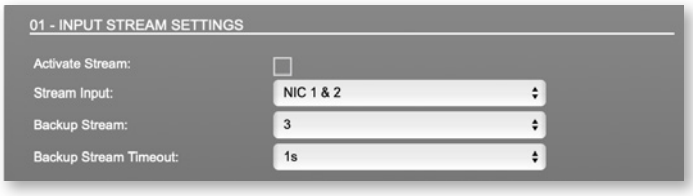
A backup stream can be defined as source after an adjustable timeout. A central active / inactive switch allows to toggle the stream state of all input streams at once.

01 to 32	State of incoming streams (OFF) = stream not activated (ON) = stream activated, receiving data (ON) = stream activated, receiving data via one NIC only (input redundancy) (blinking) = stream activated, not receiving data (unicast, connection not established)
01 to 32 Name	Name of stream gathered from SDP or set manually in the stream settings dialog.
01 to 32 xx ch	Number of audio channels transported by the stream



01 to 32 ❚❚ ■ ▶	Click to activate or deactivate single stream. ▶ = stream activated ■ = stream deactivated ❚❚ = stream not active, defined as backup-stream
INPUT STREAMS ■ ▶	Click to activate or deactivate all streams. ▶ = activate all streams ■ = deactivate all streams (requires confirmation)

Backup Streams

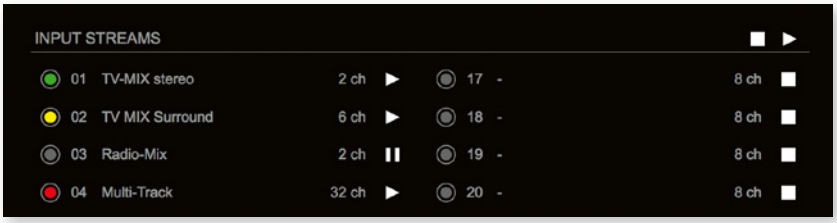


Example:
Backup stream (input 3) that will act as source in the audio matrix if the current session (input 1) fails. Switch-over occurs after the defined timeout (1s).
Stream 3 is marked accordingly in the status view

Input 1 failed and Input 3 becomes active after the timeout.

NOTE

In case the main input fails the main stream is stopped (IGMP LEAVE) before the backup stream is being activated. This behaviour ensures that the required network bandwidth does not increase in case of a failure.



Hyperlinks:

- Name (p 94)

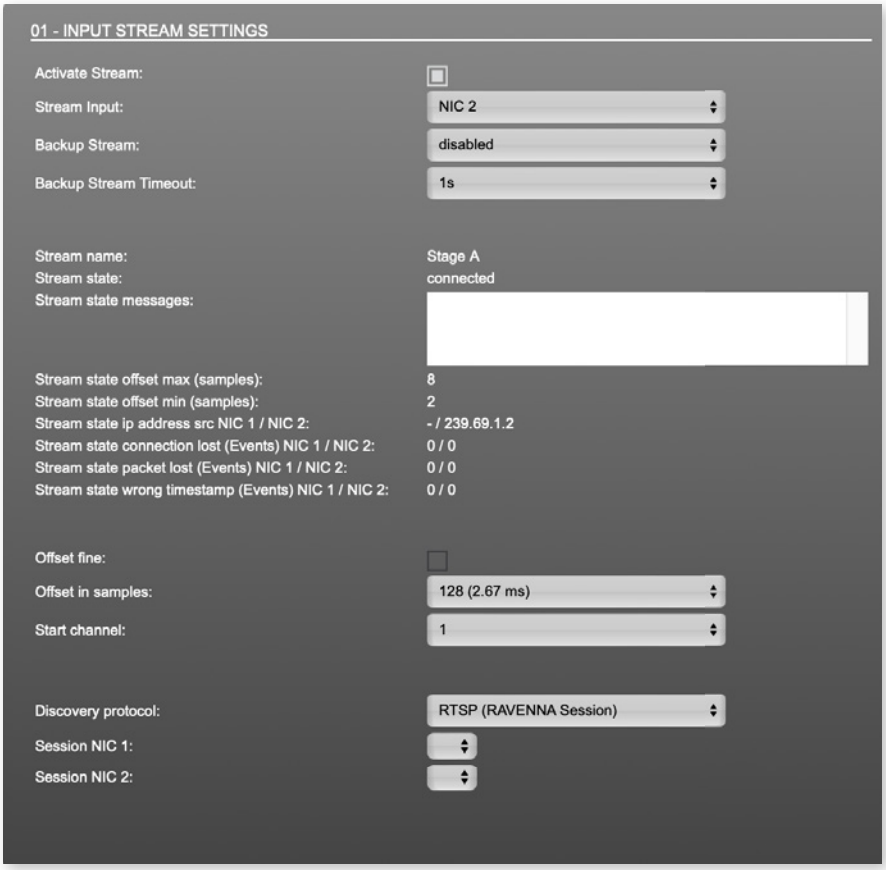
Mouse over:

- LED- indicating stream state

NOTE

Source-Specific Multicast (SSM) support for IGMP v3, v2 and v1 (SSM via protocol only in IGMP v3, SSM via internal filtering is applied for IGMP v2 and v1) - see “Source Specific Multicast” on page 99.

Input Stream Settings



Up to 32 input streams can be subscribed. Each stream is organized in a 'RAVENNA session' (SDP = Session Description Protocol) that describes the stream parameters (audio channels, audio format, etc.).

The stream settings allow to adjust the processing of the received audio data (offset, signal routing). The receiving of stream data starts once the stream has been enabled.

The settings displayed vary depending on the selected discovery protocol.

TIP

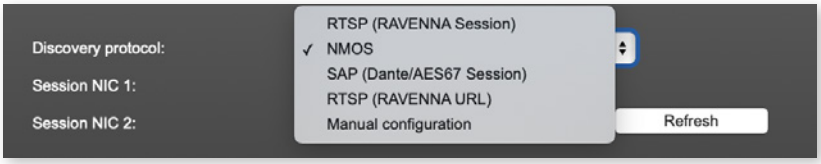
A sample offset of at least doubled packet time (samples per frame) is recommended
Example: Samples per frame = 16 (0.333 ms) ➪ Offset ≥ 32 (0.667 ms)

It may be helpful to alter the stream discovery protocol if an expected stream can't be discovered by the device.

Activate stream	Stores parameters and activates or deactivates the receiving of audio data. (Unicast: additionally the negotiation of the connection)
Stream input	Selects one or both NICs used for stream input. Both NICs means input redundancy.
Backup Stream	Selects a backup stream that will act as source in the audio matrix if the current session fails. Switch-over occurs after the defined timeout.
Backup Stream Timeout	Defines timeout [1 s to 120 s] before switch-over to backup stream.
Stream name	Name of stream gathered from SDP
Stream state	Information about stream state: connected not connected receiving data read successfull error
Stream state message	Status info related to stream state.
Stream state offset max	Measured value (maximum). A high value indicates that the media offset of the source might not match the adjusted media offset of the device.
Stream state offset min	Measured value (minimum). The offset should not become negative.
Stream state ip address src NIC 1 / NIC 2	Multicast address of input stream subscribed at NIC 1 / NIC 2. Unicast transmission: IP address of sender.
Stream state connection lost NIC 1 / NIC 2	counter indicates the number of incidents where the network connection was lost (link down)
Stream state packet lost (Events) NIC 1 / NIC 2	counter indicates the number of lost RTP packets
Stream state wrong timestamp (Events) NIC 1 / NIC 2	counter indicates the number of packets with invalid timestamp
Offset fine	Enables adjustment of offset in increments of one sample.
Offset in samples	Modules output delay of received audio data (input buffer).

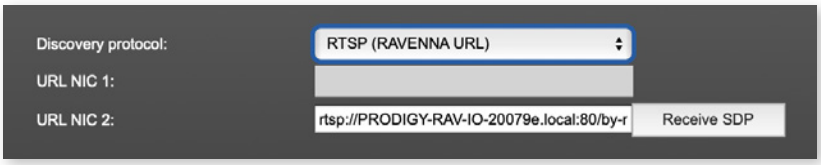
Start channel	Assignment of first stream channel in the audio matrix. E.g. stream with two channels, starting at channel 3 is available at channel 3 & 4 of the routing matrix.
Discovery protocol	Connection protocol or manual setup. RTSP = Real Time Streaming Protocol SAP = Session Announcement Protocol
Session NIC 1	Selection of discovered streams at NIC 1
Session NIC 2	Selection of discovered streams at NIC 2

Stream Discovery in AoIP environments is a colorful mixture of different mechanisms. To serve a successful stream management RAV.IO provides a bunch of options, not making operation easier but effective.



Discovery RTSP (Session)

Discovery RTSP (URL)

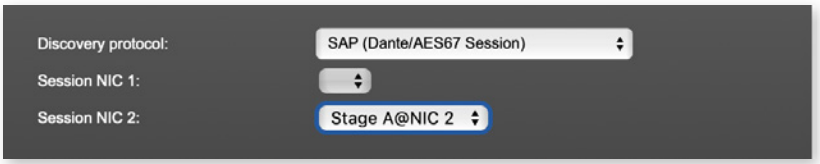


URL	URL (Uniform Ressource Locator) of the session of the device that is serving streams. Examples: rtsp://192.168.74.44/by-id/1 or rtsp://PRODIGY-RAV-IO.local:80/by-name/Stage_A
Receive SDP	Recalls the stream configuration of the defined session(s).

NOTE

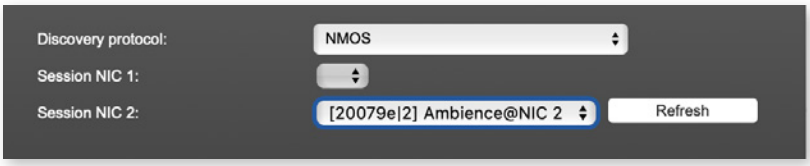
In case the automatic stream announcement and discovery of RAVENNA streams fails or cannot be used in a given network, the stream's SDP file can also be obtained via an RTSP URL.

Discovery SAP



SAP is used in Dante environments.

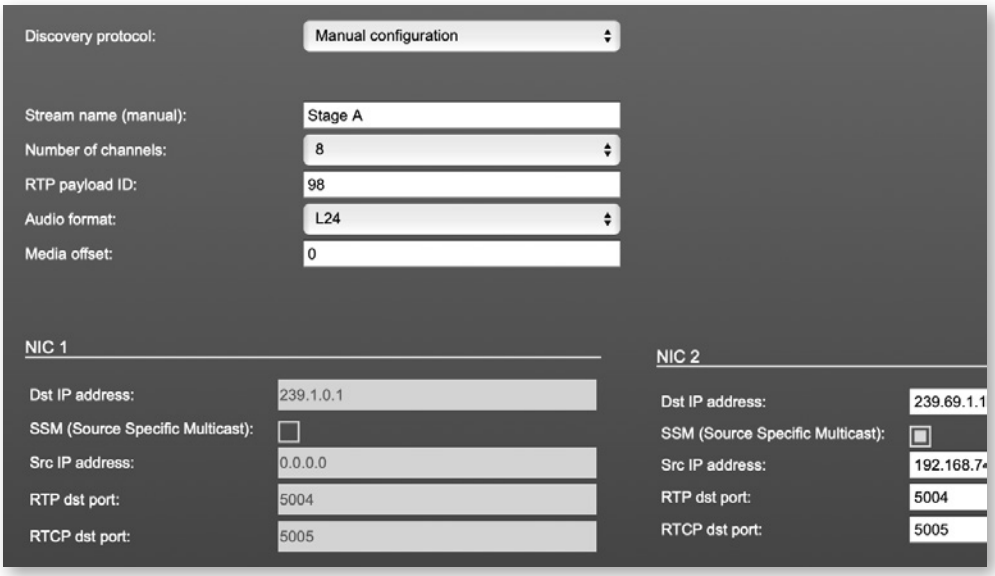
Discovery NMOS



Session	[MAC Address of sender] stream name @NIC
Refresh	Initiates a scan for available streams.

NMOS is suited for use in SMPTE ST 2110 environments.

Manual Setup



Stream name (manual)	Stream name for display in status view and matrix. Can be specified individually, different than the name gathered from the SDP.
Number of channels	Number of audio channels in the stream
RTP-Payload-ID	RTP-Payload-ID of the audio stream (Real-Time Transport Protocol). Describes the format of the transported content.
Audio Format	Stream's audio format (L16 / L24 / L32 / AM824)
Media Offset	Offset between stream's timestamp and PTP-clock
Dst IP address	Multicast IP address of audio stream
SSM	Activate Source Specific Multicast filter for this stream.*
Src IP address	IP address of sending device.*
RTP dst port	Stream's destination port for RTP
RTCP dst port	Stream's destination port for RTCP (Real-Time Control Protocol)

* An RTP packet contains the IP address of the sender (source IP) and the stream's multicast address (destination IP). With SSM activated the receiver only accepts RTP packets of a certain destination IP that are originated by a sender with the specified source IP.

NOTE

RTP Payload ID must match between sender and receiver.

Status - Output Streams

The device can send up to 32 streams. The overview displays the basic information of each stream.

01 to 32	State of outgoing streams (OFF) = stream not activated (ON) = stream activated, sending data (ON) = stream activated, stream output via both NICs selected, but one NIC is not linked to the network.
01 to 32 Name	Name of stream defined in the settings
01 to 32 xx ch	Number of audio channels transported by the stream
01 to 32 ■ ▶	Activate or deactivate stream. ▶ = stream activated ■ = stream deactivated
OUTPUT STREAMS ■ ▶	Click to activate or deactivate all streams. ▶ = activate all streams ■ = deactivate all streams (requires confirmation)

Hyperlinks:

- Name (p 102)

Mouse over:

- LED- indicating stream state

TIP

AES67 Streams
To create output streams for interoperability in AES67 environments please consult the information document Info- AES67 Streams.

TIP

SMPTE 2110-30 /-31 Streams
To create output streams for interoperability in SMPTE ST 2110 environments please consult the information document Info- ST2110-30 Streams.

Both documents are available at http://academy.directout.eu.

Output Stream Settings

Up to 32 output streams can be sent to the network. Each stream is organized in a session (SDP = Session Description Protocol) that describes the stream parameters (audio channels, audio format, etc.).

Each stream may be labelled with an individual stream name (ASCII) which is useful for enhanced comfort at organizing the setup.

The stream settings allow to adjust the processing of the sent audio data (blocks per frame, format, signal routing, ...). The sending of stream data starts once the stream has been enabled.

Once the stream is active, the SDP data is displayed and may be copied from the window or downloaded via http://<rav-io>/sdp.html?ID=<stream no.>.

Activate stream	Stores parameters and activates or deactivates the receiving of audio data. (Unicast: additionally the negotiation of the connection)
Stream Output	Selects one or both NICs used for stream output. Both NICs means output redundancy.
Stream name (ASCII)	Individually defined name of an output stream. It is used in the URL which is indicated in different ways below.*
RTSP URL (HTTP tunnel) (by-name) / (by id)	Current used RTSP-URL of stream with HTTP port used for RTSP, stream name or stream id.
RTSP URL (by-name) / (by id)	Current used RTSP-URL of stream with stream name or stream id.
SDP	SDP data of the active stream.
Unicast	If activated, the stream is sent in unicast mode.**
RTP payload ID	Stream's payload id
Samples per Frame	Number of blocks containing payload (audio) per ethernet frame - see packet time on p 94.
Audio format	Stream's audio format (L16 / L24 / L32 / AM824) ***
Start channel	Assignment of first stream channel from the audio matrix. E.g. stream with eight channels, starting at channel 3 is fed from channel 3 to 10 of the routing matrix.
Number of channels	Number of audio channels in the stream.
RTP dst port	Stream's destination port for RTP
RTCP dst port	Stream's destination port for RTCP (Real-Time Control Protocol)
Dst IP address (IPv4)	Stream's IP address for multicast (should be unique for each stream).

* Only ASCII characters are allowed.

** A unicast stream can only be received by one device. If a device is already receiving the stream, further connection calls by other clients are answered with ‚service unavailable' (503). The release time after disconnect or interruption of the client's connection amounts to about 2 minutes.

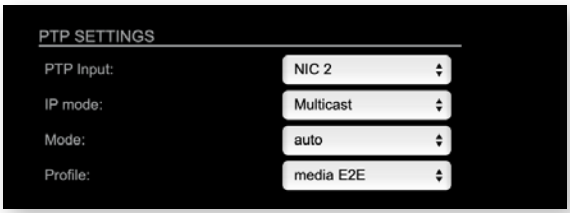
*** L16 = 16 bit audio / L24 = 24 bit audio / L32 = 32 bit audio / AM824 = standardized according to IEC 61883, allows AES3 transparent transmission (SMPTE ST 2110-31).

Advanced - Overview

The tab 'ADVANCED' is divided into several sections:

- PTP SETTINGS - definition of PTP source, mode and profile
- PTP PROFILE CURRENT SETTINGS - definition of a customized PTP profile
- CURRENT PTP MASTER - monitoring PTP characteristics
- PTP STATISTIC - monitoring device's PTP state, jitter and delay
- PTP CLOCK SETTINGS - definition of adaption algorithms to reduce jitter
- NETWORK ADVANCED SETTINGS - definition of network and QoS characteristics
- PTP JITTER - graphical display of measured PTP jitter

Advanced - PTP Settings

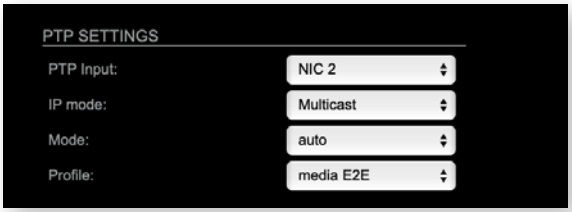


PTP Input	Selects one or both network ports used for PTP input. Both ports means input redundancy. *
IP Mode	Multicast = Sync messages and delay request are sent as multicast message to every node within the network. Hybrid = Sync messages are sent as multicast, delay requests are sent as unicast messages directly to the Grandmaster or Boundary Clock.** Unicast = Sync messages are sent as unicast, delay requests are sent as unicast messages directly to the Grandmaster or Boundary Clock.***

* Using redundant PTP-operation a switch-over is triggered not only at signal loss of the Grandmaster but depends on the quality of the PTP clock. Changes (e.g. clock class) are observed permanently and the algorithm decides for the best signal present.

** Hybrid Mode reduces the workload for all nodes in the network as they do not receive the (unnecessary) delay requests from other devices anymore.

*** Unicast Mode may help when multicast routing is not possible within the network. As an opposite to the Hybrid Mode it increases the workload of the grandmaster since sync messages must be sent to each single slave individually.



Mode	auto	= PTP-clock master / slave configuration is auto negotiated between devices in the network. Module's master / slave state may change automatically.
	slave only	= PTP-clock slave configuration is preferred. Module clocks to another device in the network
	preferred master	= PTP-clock master configuration is preferred. Module acts as network grandmaster. Priority values are adjusted automatically to ensure Grandmaster status. *
	master only	= PTP-clock master is forced. **
Profile	Selects predefined PTP profile (default E2E, default P2P, media E2E, media P2P) or activates customized PTP profile.	

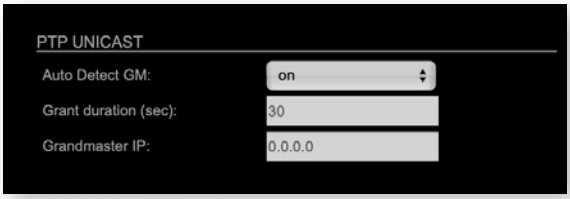
* If more than one device announces as PTP-clock master the network Grandmaster is determined following the Best Master Clock Algorithm (BMCA).

** 'Master only' configures the device to act as Unicast Grandmaster. This setting is available only with PTP Mode set to 'unicast'

NOTE

PTP profile ‚customized' allows for individual adjustment of the PTP parameters. If profile is set to ‚media' or ‚default' the PTP parameters cannot be altered and are displayed only. Factory default setting is PTP Media Profile E2E.

Advanced - PTP Unicast



Auto Detect GM	on = enables the automatic detection of the grandmaster * off = IP address of grandmaster needs to be defined manually
Grant duration (sec)	Time period during which the slave receives sync messages from the grandmaster.**
Grandmaster IP	IP address of the grandmaster. ***

* 'Auto Detect GM' is a proprietary function and might not be supported by 3rd party GMs.

** Depending on the temporary workload of the grandmaster the negotiation may fail.

*** This value is used only with 'Auto Detect GM' set to <off>.

About PTP Unicast

Since the BMCA is not available with PTP unicast, the PTP properties of the devices require some extra configuration.

Example:

Grandmaster	IP Mode Unicast, Mode Master only
Slave(s)	IP Mode Unicast, Mode Slave Only, Auto Detect GM ON, Grant Duration 30 sec

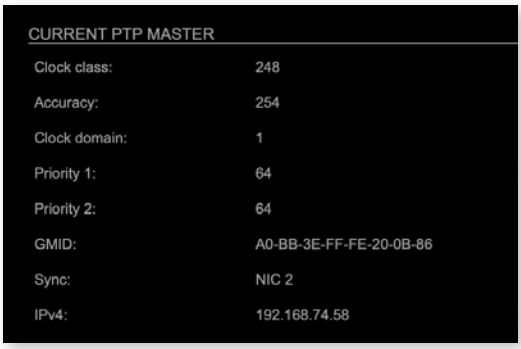
Advanced - PTP Profile Customized Settings

The settings become available with PTP profile set to 'customized'.

Clock class	PTP-clock's class according to IEEE 1588 [read only]
Accuracy	PTP-clock's accuracy according to IEEE 1588 [read only]
Clock domain NIC 1	PTP-clock's domain at NIC 1
Clock domain NIC 2	PTP-clock's domain at NIC 2
Priority 1	Priority setting for master announcement (the smaller the value the higher the priority)
Priority 2	If value 'Priority1' (and other PTP-clock parameters) of more than one device in the network match: Priority setting for master announcement (the smaller the value the higher the priority)
Announce	Intervall of sending announce-packets for auto-negotiation.
Sync	Intervall of sending sync-packets to the PTP-clock slaves in the network.
Min delay request	Intervall of sending End-To-End packets of PTP-clock slave to PTP-clock master. To determine the offset slave-to-master.
Min pdelay request	Intervall of sending Peer-To-Peer packets between two PTP-clocks. To determine the offset master-to-slave and slave-to-master.
Announce receipt timeout	Number of missed announce-packets (threshold) to reinitialize the negotiation of PTP-clock master.

One step clock	Timestamp of PTP-clock is integrated in PTP-sync-packets. No follow-up packets are sent. No = Two step clock is used
Slave only	Yes = PTP-clock is always slave.
Delay mechanism	E2E - Offset slave-to-master is determined by End-To-End packets. P2P - Offset master-to-slave and slave-to-master is determined by Peer-To-Peer packets.

Advanced - Current PTP Master



Monitoring display only.

Clock class	PTP-clock's class according to IEEE 1588
Accuracy	PTP-clock's accuracy according to IEEE 1588
Clock domain	PTP-clock's domain at selected NIC
Priority 1	Priority setting for master announcement (the smaller the value the higher the priority)
Priority 2	If value 'Priority1' (and other PTP-clock parameters) of more than one device in the network match: Priority setting for master announcement (the smaller the value the higher the priority)
GMID	ID of current Grandmaster
Sync	Selected NIC for PTP clock
IPv4	IP address of Grandmaster

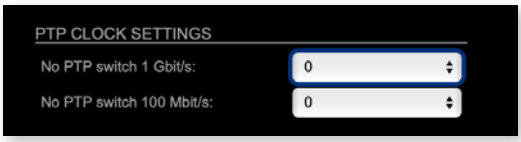
Advanced - PTP Statistic

Monitoring display only.

PTP state	Information about current PTP-clock state: intialize error deactivated receiving data pre master master passive not calibrated slave
PTP jitter	PTP-clock jitter in microseconds (µs)
PTP offset	Offset relative to PTP-clock master
PTP master to slave	Absolute offset master-to-slave in nanoseconds
PTP slave to master	Absolute offset slave-to-master in nanoseconds
Current PTP time (TAI):	Date and time information from GPS source*
Current PTP time (TAI) (RAW):	RAW TAI from GPS source*

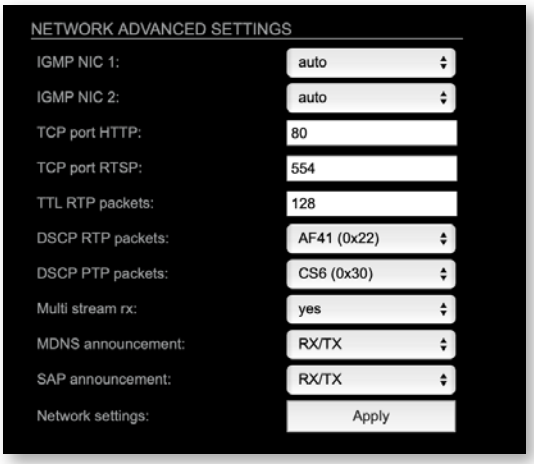
* Temps Atomique International- if no GPS source is available for PTP time-stamping,the date / time display starts at 1970-01-01 / 00:00:00 after every reboot of the device.

Advanced - PTP Clock Setting



No PTP Switch 1 Gbit/s	Adapted PTP-clock algorithm to reduce clock jitter using 1 GB network switches without PTP support. Max. number of 1 Gbit/s switches: less than 10
No PTP Switch 100 Mbit/s	Adapted PTP-clock algorithm to reduce clock jitter using 100 MB network switches without PTP support. Max. number of 100 Mbit/s switches: 1

Advanced - Network Advanced Settings



IGMP NIC 1	Definition or auto-select of IGMP version used to connect to a multicast router at NIC 1.
IGMP NIC 2	Definition or auto-select of IGMP version used to connect to a multicast router at NIC 2
TCP port HTTP	TCP port for HTTP
TCP port RTSP	TCP port for RTSP
TTL RTP packets	Time-To-Live of RTP packets - default: 128
DSCP RTP packets	DSCP marking of QoS of RTP packets - default: AF41
DSCP PTP packets	DSCP marking for QoS of PTP packets - default: CS6*
Multi stream rx	If activated, the device allows to subscribe to the same multicast stream more than one time - default: off
MDNS announcement	Announcement of streams via MDNS can be controlled to optimize network traffic or CPU load. Values: Off, RX, TX or RX/TX **
SAP announcement	Announcement of streams via SAP can be controlled to optimize network traffic or CPU load. Values: Off, RX , TX or RX/TX **
Network settings Apply	Confirms and saves changes being made. Reboot required.

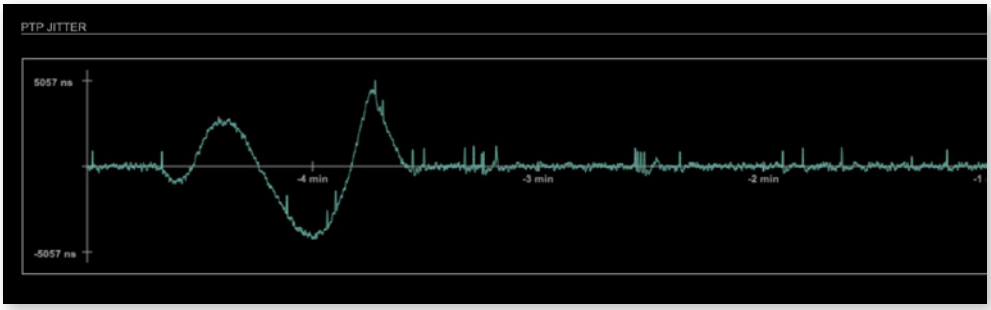
* AES67 specifies EF, but some implementations use EF for Audio streaming. To avoid overlapping of RTP and PTP packets in the same queue CS6 has been chosen as default.

** RX = receive, TX = transmit, RX/TX = receive and transmit

NOTE

Source-Specific Multicast (SSM) support for IGMP v3, v2 and v1 (SSM via protocol only in IGMP v3, SSM via internal filtering is applied for IGMP v2 and v1) - see “Source Specific Multicast” on page 99.

Advanced - PTP Jitter

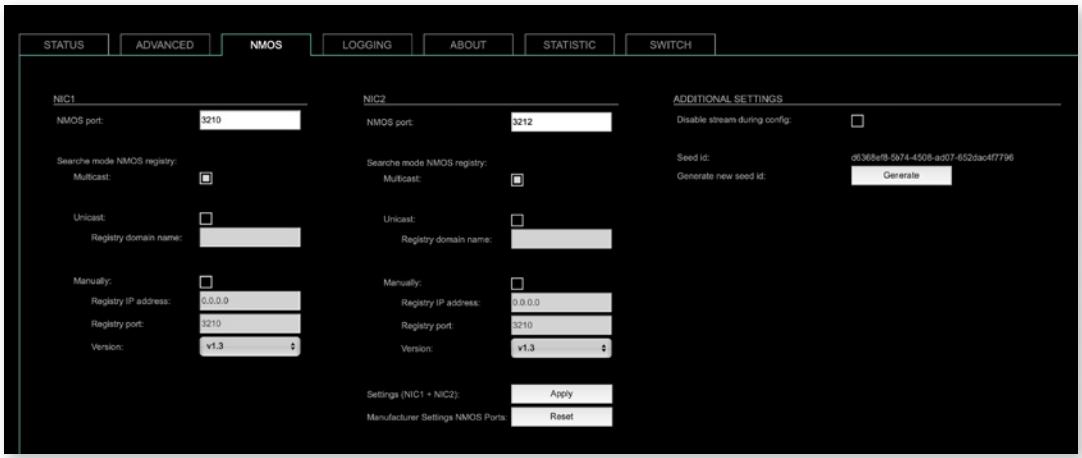


Graphical display of measured PTP jitter.

NOTE

An error message next to Jitter measurement is displayed if delay requests are not being answered by Grandmaster.

NMOS - Overview



NMOS provides a family of specifications related to networked media for professional applications. It is produced by the Advanced Media Workflow Association (AMWA).

Support for NMOS is introduced with the AoIP Module version SW 0.17 / HW 0.46 according to the specifications:

- IS-04 Discovery & Registration
- IS-05 Device Connection Management

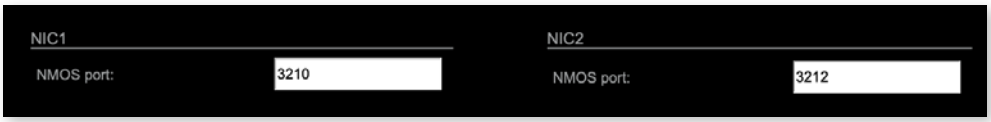
IS-04 allows control and monitoring applications to find the resources on a network. Resources include Nodes, Devices, Senders, Receivers, Sources, Flows…

IS-05 provides a transport-independent way of connecting Media Nodes.

More information: https://specs.amwa.tv/nmos/

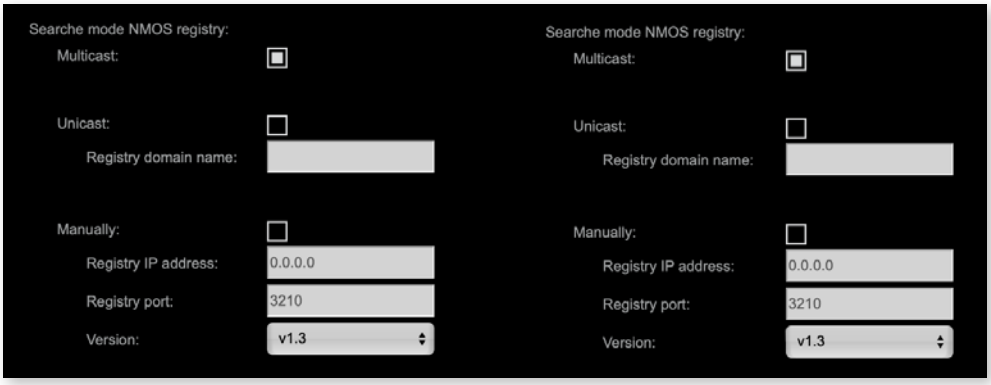
NMOS port - NIC1 & NIC2

The port entries for NIC1 and NIC2 are pre-configured by default.
Modifications are possible but not necessary.



NMOS port (NIC1 + NIC2)	Port address. Reboot required after modification.

Search mode NMOS registry



Multicast	use mDNS to determine and connect to the registry server
Unicast	use DNS-SD to connect to the registry server
Registry domain name	DNS resolvable domain name of the registry server
Manually	
Registry IP address	
Registry port	
Version	Support of NMOS API version

NMOS - Additional Settings

Disable stream during config	Automatically disable and re-enable streams when settings are changed via NMOS (recommended)
Seed id	Unique identifier, subordered entities are derived from the seed id.
Generate new seed id Generate	Generates a new unique identifier. Reboot required.

NMOS uses a logical data model based on the JT-NM Reference Architecture to add identity, relationships and time-based information to content and broadcast equipment. Hierarchical relationships group related entities, with each entity having its own identifier.

The identifiers are persistent across restarts of the device in order to make them useful over a period of time longer than a single production deployment.

New identifiers may be generated manually if required.

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Logging

The tab 'LOGGING' displays logging depending on the 'Log Settings'. The logging can be enabled individually for different protocols, each of with an adjustable filter. An adjustable log level specifies the information detail of each entry.

To save a log the content of the view can be copied and pasted to a text document.

Log Level

0	log data
1	level and log data
2	protocol, level and log data
3	protocol, process-id of requesting process, process-id of running process, level and log data
4	protocol, process-id of requesting process, process-id of running process, level, processor time in ticks and log data
5	protocol, process-id of requesting process, process-id of running process, level, processor time in ticks, file name and line and log data

Protocol Types

ARP	Address Resolution Protocol
BASE	Basic operation of module
DHCP	Dynamic Host Configuration Protocol
DNS	Domain Name System
FLASH	Process for updating the module
IGMP	Internet Group Management Protocol
MDNS	Multicast Domain Name System
NMOS	Network Media Open Specification
PTP	Precision Time Protocol
RS232	Serial Protocol
RTCP	Real Time Control Protocol
SAP	Session Announcement Protocol
TCP	Transmission Control Protocol
Zeroconf	Zero Configuration Protocol

Log Filter

NONE	logging disabled
ERROR	error occurred
WARNING	warnings- condition that may lead to unwanted behavior or an error
INFO 1	log info* + warning + error
INFO 2	log info* + warning + error
INFO 3	log info* + warning + error
INFO 4	log info* + warning + error

* increasing amount of log info starting from ‚INFO 1‘

Log Operation

Save log	Downloads the current log entries to a text-file (log.txt).
Clear log	Deletes all log entries without further prompt.
Scroll lock	Interrupts automatic scrolling of the list view to allow copying the content to a text file via copy & paste. If scrolling is stopped for a longer period of time the display may not list all entries.

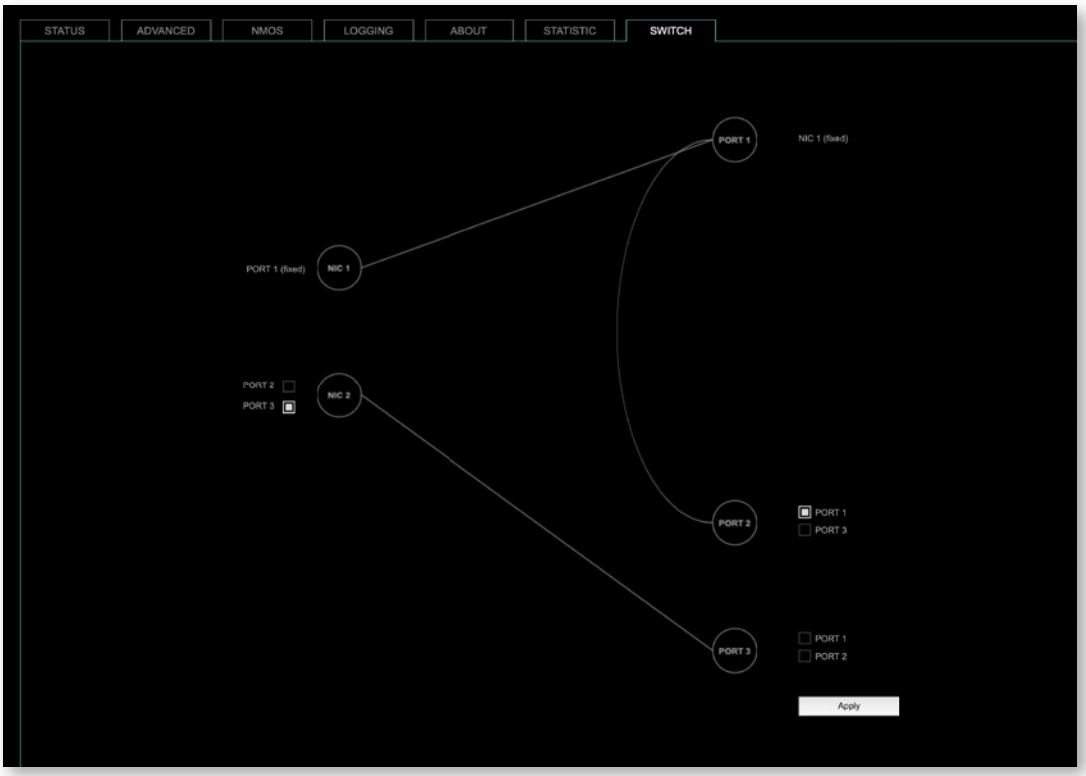
Statistic

The tab 'STATISTIC' displays an overview of the CPU load of the particular processes, an error counter and a monitor display to indicate the incoming (RX) and outgoing (TX) network traffic on both network ports individually.

Details	Displays a list of input streams and related events (connection lost, packet lost, wrong timestamp) of received audio packets.
Reset	Resets the packet statistic

See "Protocol Types" on page 119.

Switch



The module features two RJ45 sockets and an SFP cage managed by two independent network interfaces (NIC 1 / NIC 2).

- Port 1 is fixed assigned to NIC 1.
- Port 2 and 3 can be assigned to either NIC 1 or NIC 2 on the SWITCH tab

NOTE

If you want to use a port that is not assigned to a NIC e.g. to patch the device's management port (MGMT) into the audio network, you can link it to one of the audio ports.

NOTE

To access the module's control page it is required to connect the management network to one of the ports that is directly attached to a NIC - see next page.

To give the very best PTP synchronisation performance, the switch incorporates advanced timestamping between the external PORTS and the internal NICs. As a consequence, the on-board switch cannot be used to connect other PTP devices via a single shared connection to the wider network.

Please connect all other PTP devices directly to your system's network switch.

Tools

The tab 'TOOLS' offers a generator to ping any IP address (IPv4) from either NIC 1 or NIC 2. The result is displayed at the 'Output'.

IP address (IPv4)	Enter IP address (IPv4) to be pinged
Interface	Select NIC 1 or NIC 2
Start	Sends ping to the specified IP address from selected NIC.

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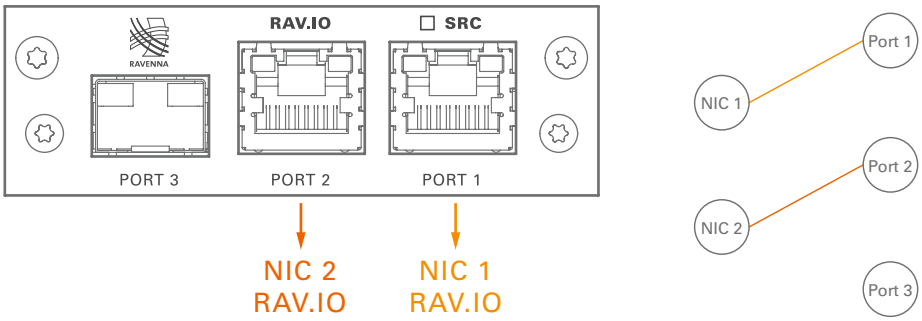
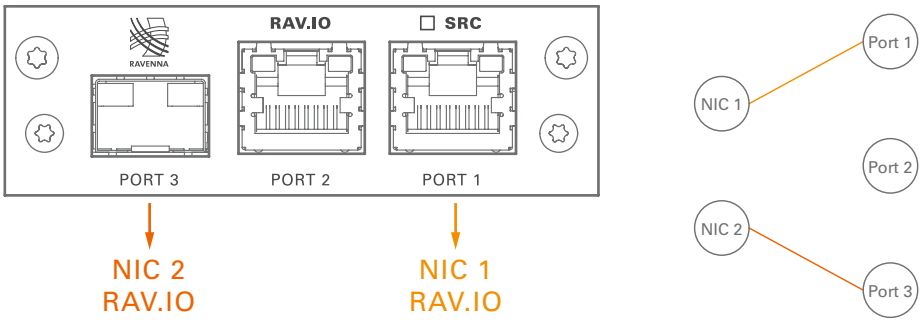
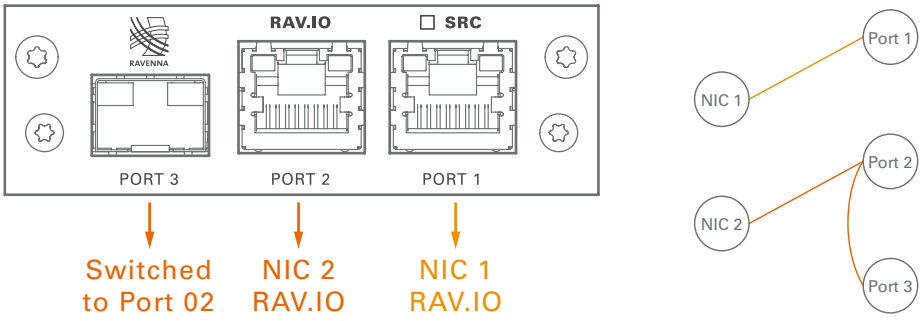
SWITCH - Configurations



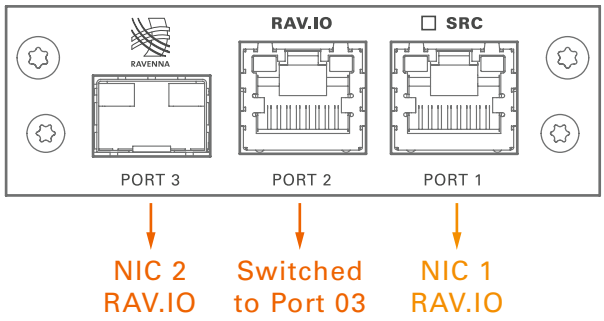
RAV.IO Audio & Control via Port 1 and Port 2

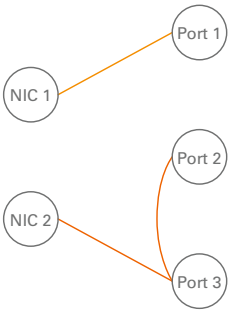


RAV.IO Audio & Control via Port 1 and Port 3

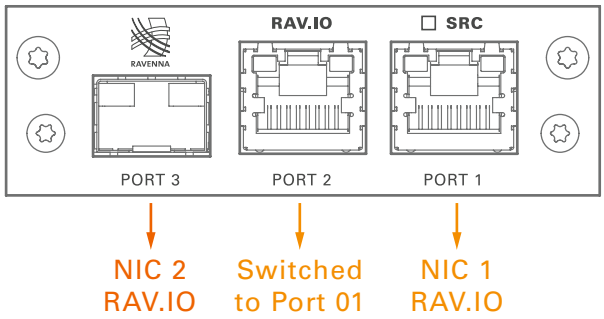


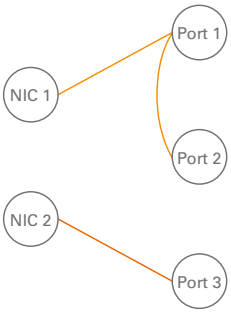
RAV.IO Audio & Control via Port 1 and Port 2
Extra ethernet traffic via Port 3 (NIC 2)



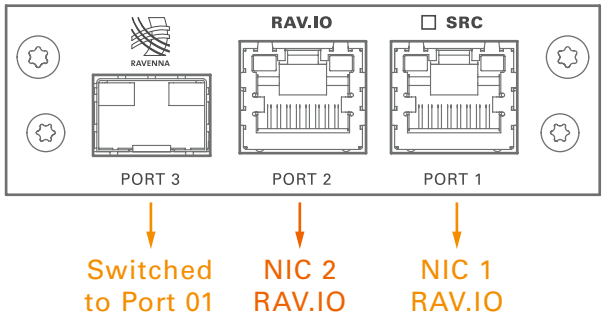


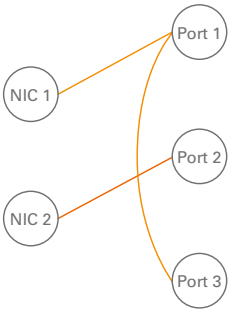
RAV.IO Audio & Control via Port 1 and Port 3
Extra ethernet traffic via Port 2 (NIC 2)





RAV.IO Audio & Control via Port 1 and Port 3
Extra ethernet traffic via Port 2 (NIC 1)*



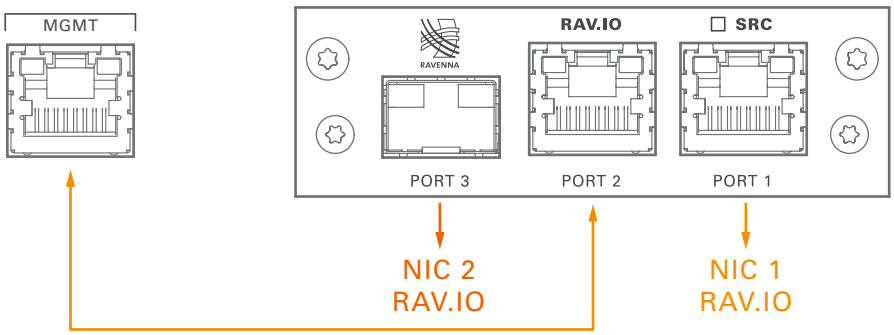


RAV.IO Audio & Control via Port 1 and Port 2
Extra ethernet traffic via Port 3 (NIC 1)

* see example on the following page.

Device Management over the audio network

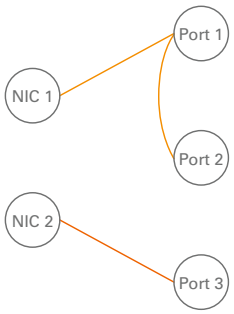
This configuration is exemplary to illustrate the possibility to integrate the management data that is used to control the device into the audio network.



- connect the MGMT port of the device with Port 2 of the RAV.IO
- connect the RAVENNA network to Port 1 of the RAV.IO
- connect the RAVENNA network to Port 3 of the RAV.IO (if needed)

Switch configuration

RAV.IO Audio & Control via Port 1 and Port 3
Extra ethernet traffic via Port 2 (NIC 1)



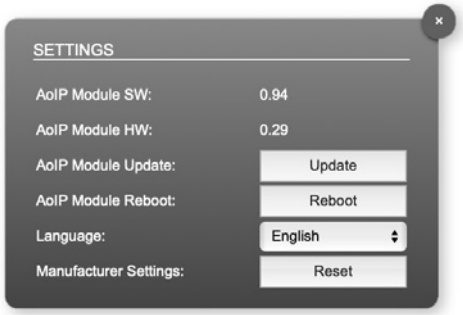
RAV.IO - Firmware Update

The RAV.IO module is updated via network.

Visit www.directout.eu and navigate to a PRODIGY product page.
Download:

- PRODIGY RAV.IO Firmware

Open the control page of the module and navigate to the tab STATUS and click SETTINGS in the top right corner (p 88).



Click 'Update' and browse to the update file after unzipping first.
Example: rav_io_hw_0_29_sw_0_94.update

Folllow the instructions displayed.

WARNING!

It is strongly recommended to backup the device configuration (Save Preset) before running any update.

CHAPTER 7: DANTE.IO

Introduction

DANTE.IO is an audio network module for Dante / AES67.
All functions of the device are managed via the Dante Controller application.

To control the functions of the host device connect your computer with the management network port (MGMT) and use the globcon application.

Dante Controller

To control a Dante network the application 'Dante Controller' needs to run on a computer that is connected to the audio network.

The software is available from the Audinate website (requires a free registration):
https://www.audinate.com/products/software/dante-controller

A detailed documentation about using Dante Controller is available here:
https://dev.audinate.com/GA/dante-controller/userguide/webhelp/

At startup Dante Controller scans the network for connected Dante devices a shows them automatically in the 'Network View'.

The IP address of a Dante device must fit into the network environment for proper operation of a Layer 3 based network. However Dante Controller will mark a device entry red if the IP Address is not matching the network environment and helps resolving the issue.

NOTE

DANTE.IO supports control via Dante Domain Manager.

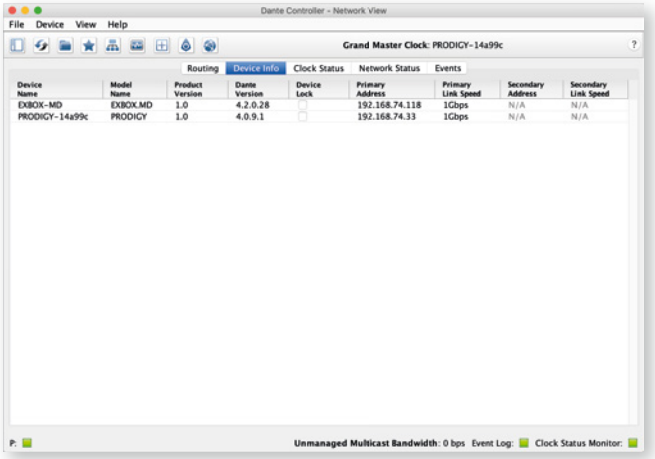
When a device is enrolled in a domain, it can be viewed and configured in Dante Controller only by DDM users that are members of the domain.

To use an enrolled device outside of the domain it is required to un-enroll it from DDM first or to reset the device via Dante Controller.

User Guide ‘Dante Domain Manager’ (Chapter: Enrolling Devices in Domains)
https://www.audinate.com/learning/technical-documentation

Network View

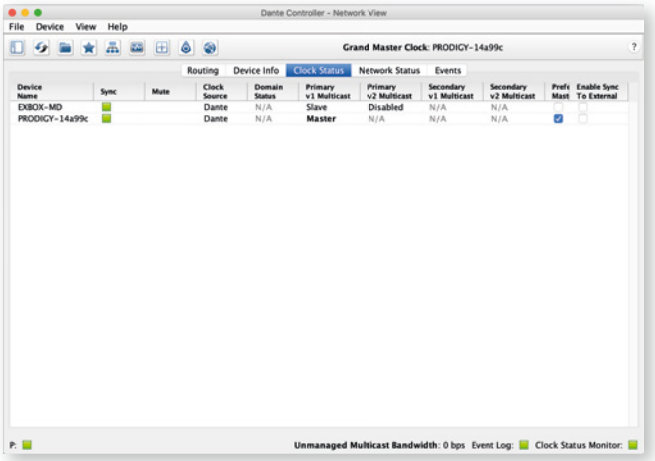
The 'Network View' is organised in several tabs. All detected Dante devices are shown on a list.



The DANTE.IO is delivered with the network interface set to DHCP as a default. Dante Controller can be used to discover the device on the network and change the IP configuration if necessary - see “Network Config” on page 132.

Clock Status

The tab ‘Clock Status’ informs about the clock settings of each connected device and allows to modify them.

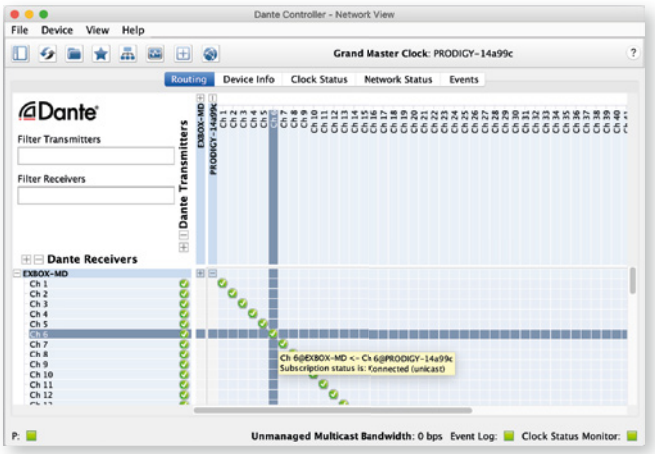


‘Preferred Master’ sets the device as PTP Grandmaster in the network

‘Enable Sync to External’ allows to clock the device from an external source that is defined in the clock settings of PRODIGY- e.g. AES, MADI or internal. The device shall become Grandmaster of the network then.

Routing

The audio signal routing is accessible in the tab 'Routing'



'Dante Receivers' in the vertical column displays the receiving devices (= destinations).

'Dante Transmitters' in the horizontal row displays the transmitting devices (= sources).

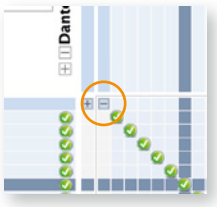
The channel list can be expanded or collapsed for each device.

Connections are made by clicking into the matrix.

TIP

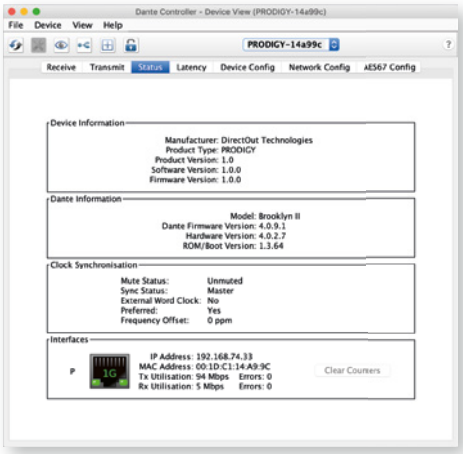
To patch a 1:1 connection:
Hold CTRL + Click the minus-symbol

To un-patch a 1:1 connection:
Hold CTRL + SHIFT + Click the minus-symbol



Device View

The 'Device View' is also organised in to several tabs.
It can be opened by double-clicking on to a device name in the 'Network View'.



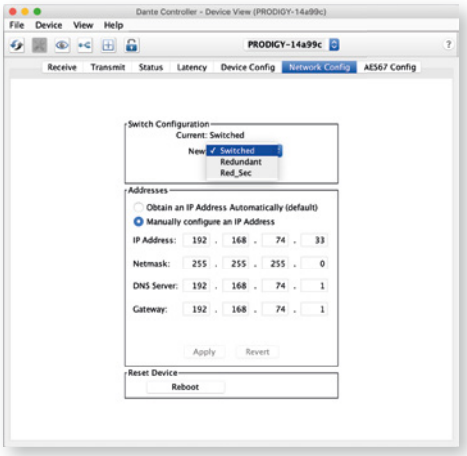
The status tab informs about the current firmware and software versions.

TIP

See “DANTE.IO- Firmware Update” on page 134.

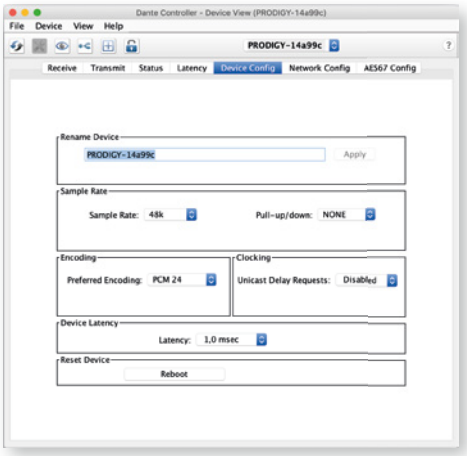
Network Config

The tab 'Network Config' of the device view provides access to the operating mode of the built-in network switch and the network settings of the device.



Device Config

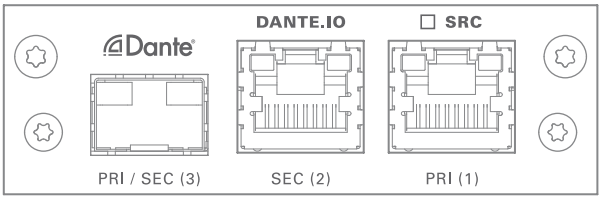
The tab ‘Device Config’ to adjust the device name, sample rate, encoding modes and more.



Switch - Configuration

Three network ports are available for transmission of audio signals and remote control. The built-in network switch can be operated in three modes:

- Switched (all ports in the same network)
- Redundant (1 = Primary, 2 = Secondary, 3 = Primary)
- Red_Sec (1 = Primary, 2 & 3 = Secondary)



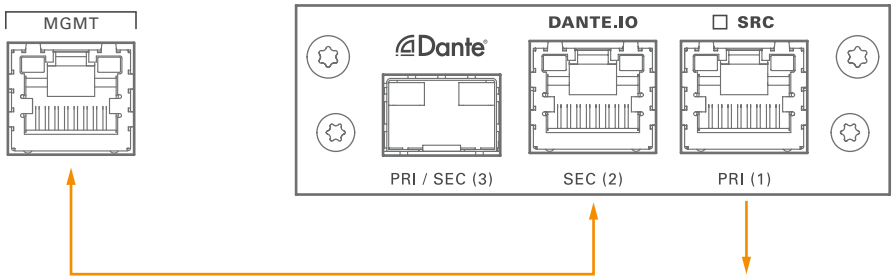
The switch will be configured automatically by selecting the respective operation mode in Dante Controller.

NOTE

If redundancy is being used, secondary interfaces should be connected to a second separate network. Secondary interfaces cannot communicate with primary interfaces.

Device Management over the audio network

This configuration is exemplary to illustrate the possibility to integrate the management data that is used to control the host device into the audio network.



- set the switch configuration to 'Switched'
- connect the MGMT port of the device with Port 2 of the DANTE.IO
- connect the DANTE network to Port 1 of the DANTE.IO
- connect the DANTE network to Port 3 of the DANTE.IO (if needed)

DANTE.IO - Firmware Update

The device can be updated either via the:

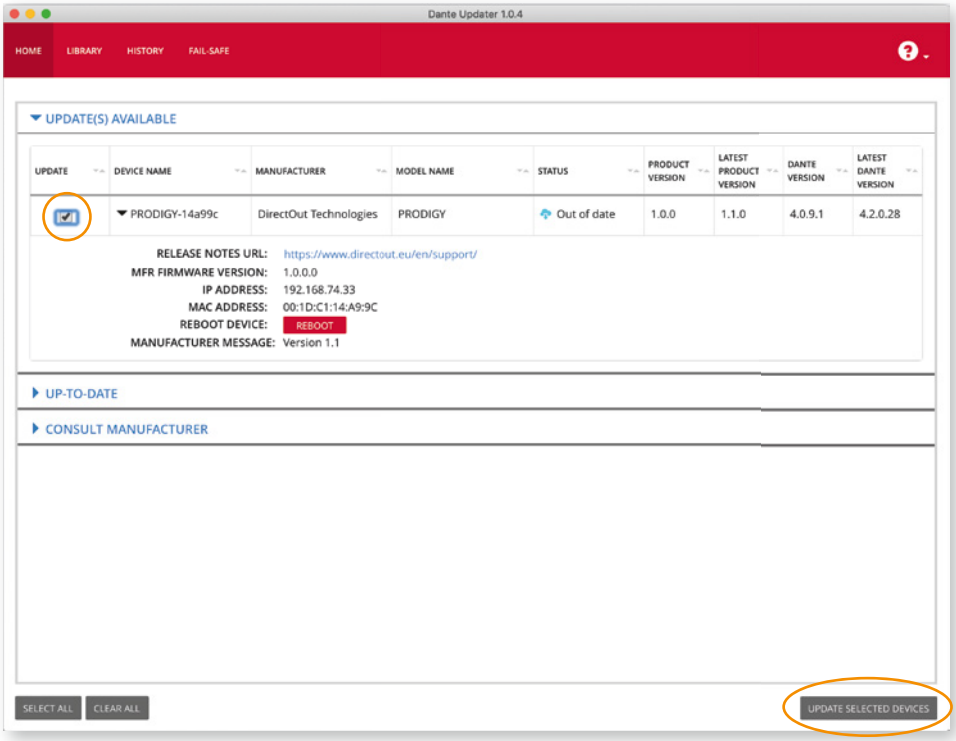
- online procedure using the 'Dante Updater' which is integrated in the latest version of 'Dante Controller'.
- offline procedure using an update file and ‚Dante Firmware Update Manager'

WARNING

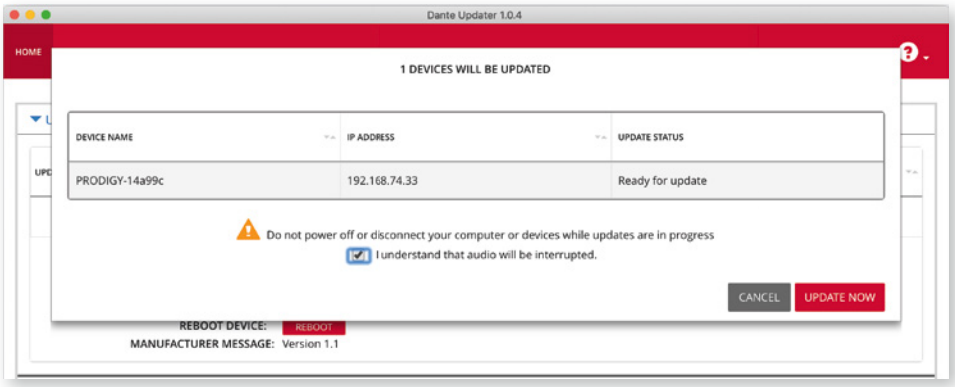
It is strongly recommended to backup the device configuration before running any update.

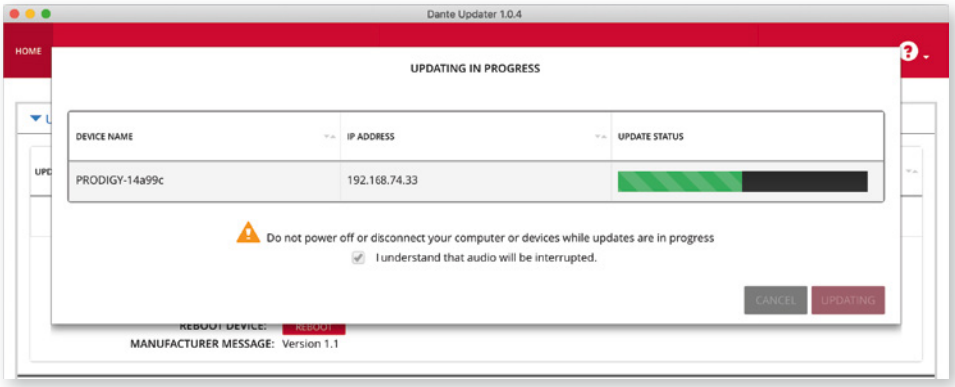
Online Procedure

1. Open Dante Controller
2. Menu: View- Dante Updater (CMD-U)
3. Select device to update and click 'Update Selected Devices'

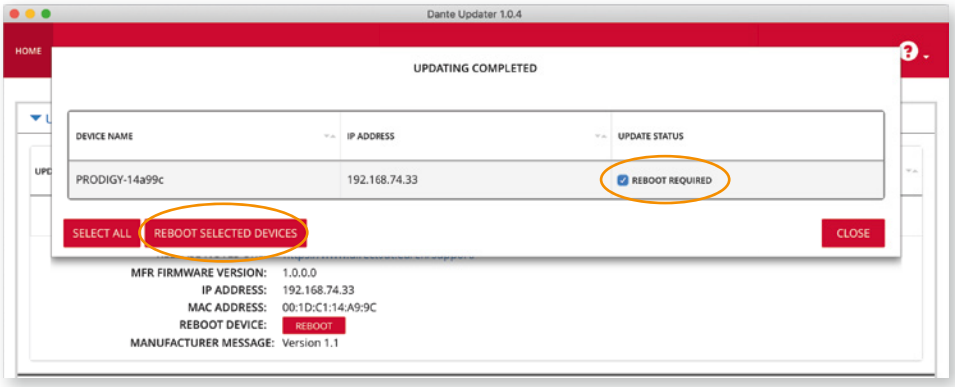


4. Confirm and take your time until the update procedure has finished.

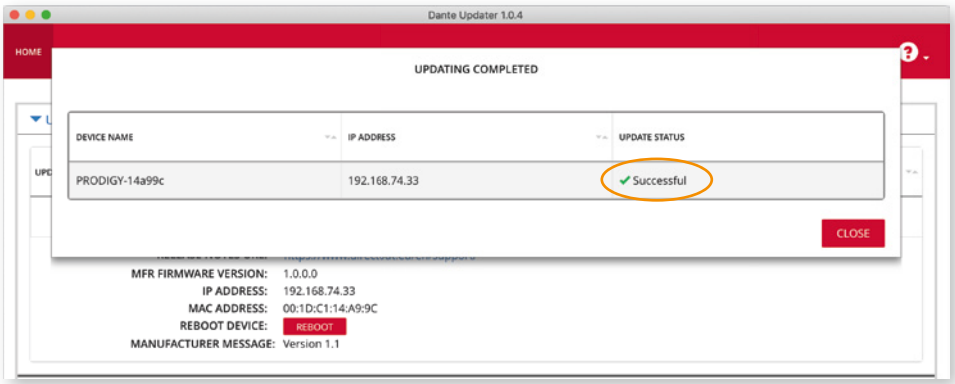




5. Mark the updated device for reboot and click 'Reboot Selected Devices'.



6. After Reboot the Updater will report the Update Status.



Offline Procedure

1. Download the update file from the product page at www.directout.eu.

2. Open ‚Dante Firmware Update Manager' and follow the instructions.
https://www.audinate.com/products/firmware-update-manager

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CHAPTER 8: Troubleshooting and Maintenance

Troubleshooting

To identify a possible defect with the device please consult the following table. If the fault cannot be resolved using these instructions, please contact your local DirectOut representative or visit support.directout.eu.

Issue	Possible reason	Solution
Device doesn't work.	Power supply is broken.	Check that the power supply switch is on, that the device is connected to the power supply and that the socket is working. Defective fuses must be exchanged by qualified service personal only.
Optical port does not work.	Optic is dirty.	Use an air supply to carefully remove any dust. Never use objects for cleaning.
No signal at the output port.	Connections (input / output) are mixed up.	Check the connections and change the cables if necessary. Check the routing matrix.
No signal at the output port.	Signal cable defective.	Exchange the signal cable.
MADI signal at the input is not stable.	Signal source is defective or bad signal condition (Jitter > 1 ns)- e.g. due to exceeded length or bad screening attenuation of signal cable.	Change the source or use appropriate cables.
Clicks in the audiosignal.	Input source is not in sync with clock master of the box.	Check the status of input LED and check clock setting of the connected device.

Maintenance

To clean the device, use a soft, dry cloth. To protect the surface, avoid using cleaning agents.

NOTE!

The device should be disconnected from the power supply during the cleaning process.

CHAPTER 9: Technical Data

Dimensions (sketch on page 150)

- Width 19'' (483 mm)
- Height 2 RU (89 mm)
- Depth 10'' (254 mm)
- Weight about 10 kg

Power Consumption

- 20 W to 110 W, module dependent

Power Supply

- 2 x 84 V- 264 V AC / 47 Hz- 63 Hz / Safety class 1

Fuses

- Fuse 250 V- 4 A (slow-blow) – 2 fuses per power supply

Environmental Conditions

- Operating temperature +5°C up to +45°C
- Relative humidity: 10%- 80%, non condensing

Display

- 5.0" Backlight LED LCD, IPS Transmissive
- Resolution: 480 x 854 px

Control

- 1 x Encoder knob with push functionality
- 2 x Push-Buttons

Network

- 2 x RJ45 Gigabit Ethernet, 1 x SFP cage
- for managing the device (remote control)

Remote Control

- globcon software control
- integrated web server with UI (HTML, JavaScript)
- Ember+ (license) - MGMT <IP Address>:9000
- OSC (license)

Word Clock

- 4 x BNC socket (input / output), WCK 2 switchable to Video Sync
- Video sync: analog, black burst or tri-level
- Impedance: 75 Ω (termination switchable)
- AES11 (DARS also supported)

Sample Rate

- 30 - 50 kHz @ 1 FS
- 60 - 100 kHz @ 2 FS
- 120- 200 kHz @ 4 FS

Phones Out 1

- 1 x 6.3 mm TRS jack, mono / stereo
- Output level: max. +18 dBu
- SNR:-115 dB RMS (20 Hz- 20 kHz) /-118 dB(A)
- THD+N @ 0 dBFS:-105 dB @ 600 Ω

Phones Out 2

- 1 x 3.5 mm TRS jack, mono / stereo
- Output level: max. +12 dBu
- SNR:-115 dB RMS (20 Hz- 20 kHz) /-118 dB(A)
- THD+N @ 0 dBFS:-105 dB @ 600 Ω

USB

- 2 x USB socket (Type B)
- for legacy control via globcon bridge

GPIO

- 2 x GPI (MOSFET switch),
- 2 x GPO (MOSFET switch)

MADI Port SC optical

- 1 x SC socket FDDI (input / output)
- ISO/IEC 9314-3
- Wave length 1310 nm
- Multi-Mode 62.5/125 or 50/125

MADI Port BNC coaxial

- 2 x BNC socket (input / output)
- Impedance: 75 Ω
- 0.3 V up to 0.6 V (peak to peak)

MADI Port SFP

- 1 x SFP (empty cage without module)

MADI Format (I/O)

- 48k Frame, 96k Frame
- 56 channel, 64 channel

Analog I/O level line

- +15 / +18 / +24 dBu changeable via jumper on module

MIC8.HD.I

- Input sensitivity:-55 dBu to +30 dBu
- EIN:-128 dBu
- THD @-1 dBFS:-113 dB
- Frequency response:-0.15 dB (10 Hz) /-0.15 dB (20 kHz)
- 30 dB PAD (switchable)
- +48 V phantom power (switchable)

MIC8.LINE.IO

- Input sensitivity:-55 dBu to +24 dBu
- EIN:-118 dBu
- SNR:-115 dBFS (20 Hz- 20 kHz) /-118 dB(A) @ 0dB Gain
- THD @-1 dBFS:-113 dB
- Frequency response:-0.5 dB (10 Hz to FS/2)
- 9 dB PAD (switchable)
- +48 V phantom power (switchable)

A/D

- SNR:-117.6 dB RMS (20 Hz- 20 kHz) /-119.9 dB(A)
- THD @-1 dBFS: <-119 dB
- Frequency response: <-0,15 dB (10 Hz) /-0,15 dB (20 kHz)

D/A

- SNR:-116.8 dB RMS (20 Hz- 20 kHz) /-119.5 dB(A)
- THD @-1 dBFS:-109 dB
- Frequency response:-0,5 dB (10 Hz) /-0,15 dB (20 kHz)

AN8.IO
- 8 ch line input / output, 2 x DSUB-25, balanced

AN8.O
- 8 ch line output, 1 x DSUB-25, balanced

AN8.I
- 8 ch line input, 1 x DSUB-25, balanced

MIC8.HD.I
- 8 ch mic high definition input, 1 x DSUB-25, balanced

MIC8.HD.IO
- 8 ch mic high definition input / line output, 2 x DSUB-25, balanced

MIC8.LINE.IO
- 8 ch mic/line input / line output, 2 x DSUB-25, balanced

MIC8.LINE.I
- 8 ch mic/line input, 1 x DSUB-25, balanced

AES4.IO
- 4 port AES3 input / output, 1 x DSUB-25 (8 audio channels)

AES4.SRC.IO
- 4 port AES3 input with SRC / output, 1 x DSUB-25 (8 audio channels)

BNC.IO

- MADI, 64 ch @ 1 FS, 32 ch @ 2 FS, 16 ch @ 4 FS
- 1 x coaxial BNC input, 1 x coaxial BNC output, 75 Ω

SC.IO

- MADI, 64 ch @ 1 FS, 32 ch @ 2 FS, 16 ch @ 4 FS
- 1 x SC-Socket duplex multi-mode (single mode on request)

SFP.IO

- MADI, 64 ch @ 1 FS, 32 ch @ 2 FS, 16 ch @ 4 FS
- 1 x SFP cage (matching SFP transceiver available from DirectOut)

MADI2.SRC.IO

- MADI, 128 ch @ 1 FS, 64 ch @ 2 FS, 32 ch @ 4 FS
- 2 x SFP cage (matching SFP transceiver available from DirectOut)

MADI2.BNC.SRC.IO

- MADI, 128 ch @ 1 FS, 64 ch @ 2 FS, 32 ch @ 4 FS
- 2 x coaxial BNC input, 2 x coaxial BNC output, 75 Ω

RAV.IO / RAV.SRC.IO

- Network Audio RAVENNA / AES67, 128 ch @ 1 FS, 64 ch @ 2 FS, 32ch @ 4 FS
- 2 x RJ45, 1x SFP

DANTE.IO / DANTE.SRC.IO

- Network Audio Dante, 64 ch @ 1 FS, 32 ch @ 2 FS, 16ch @ 4 FS
- 2 x RJ45, 1x SFP

SG.IO / SG.SRC.IO

- Network Audio Waves SoundGrid, 128 ch @ 1 FS, 64 ch @ 2 FS
- 2 x RJ45, 1x SFP

MILAN.IO / MILAN.SRC.IO

- Network Audio AVB / MILAN, 128 ch @ 1 FS, 64 ch @ 2 FS, 32 ch @ 4 FS
- 2 x RJ45, 1x SFP

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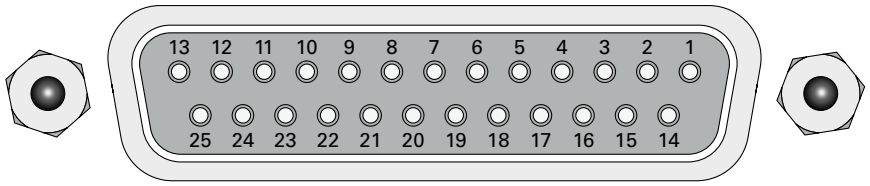
Appendix A - DSUB-25 Pin assignment

The pinout of the DSUB-25 connectors for the transmission of analog and AES3 audio signals follows the AES59 specification.

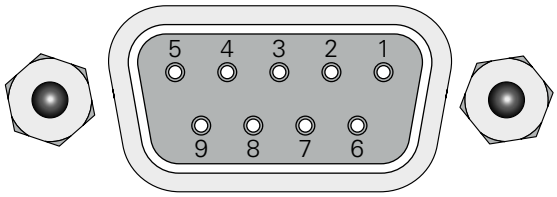


jack- female

PIN	Signal analog	Signal digital
1	CH 8 +	CH 4 OUT +
2	GND	GND
3	CH 7 –	CH 3 OUT –
4	CH 6 +	CH 2 OUT +
5	GND	GND
6	CH 5 –	CH 1 OUT –
7	CH 4 +	CH 4 IN +
8	GND	GND
9	CH 3 –	CH 3 IN –
10	CH 2 +	CH 2 IN +
11	GND	GND
12	CH 1 –	CH 1 IN –
13		
14	CH 8 –	CH 4 OUT –
15	CH 7 +	CH 3 OUT +
16	GND	GND
17	CH 6 –	CH 2 OUT –
18	CH 5 +	CH 1 OUT +
19	GND	GND
20	CH 4 –	CH 4 IN –
21	CH 3 +	CH 3 IN +
22	GND	GND
23	CH 2 –	CH 2 IN –
24	CH 1 +	CH 1 IN +
25	GND	GND

Appendix B - DSUB-9 Pin assignment

The pinout of the DSUB-9 connector for the GPI (General Purpose Input) and GPO (General Purpose Output) connection.



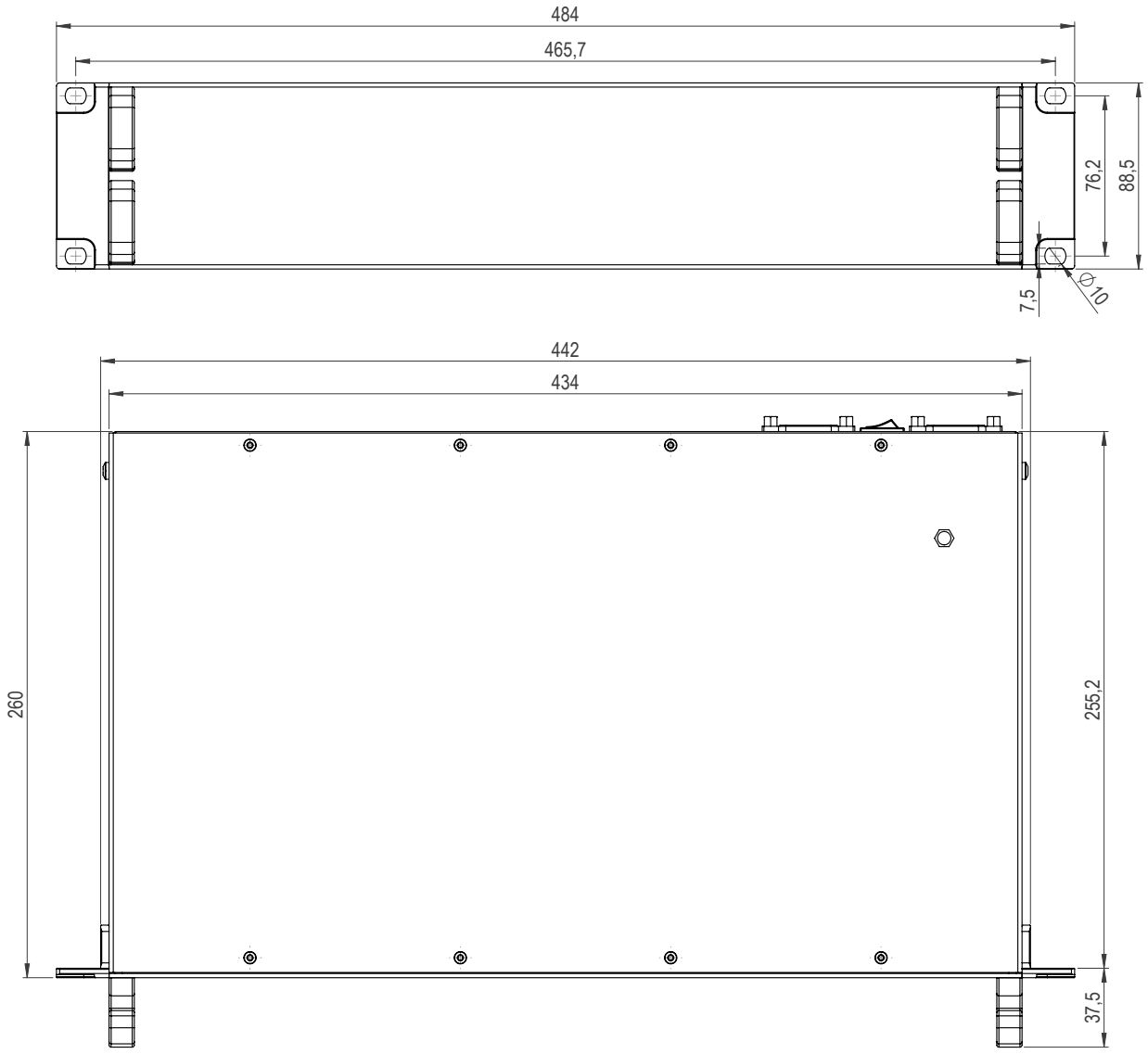
jack- female

PIN	Signal
1	GND
2	GND
3	GND
4	+ 12 V
5	+ 12 V
6	GPI 2
7	GPI 1
8	GPO 2
9	GPO 1

GPI - 3.3 V CMOS compatible, low active
Can be triggered by connecting the input pin with ground (GND) or by a voltage source between input pin and ground.
Input voltage: max 30 V DC

GPO - Open drain MOSFET switch - max. 30 V, max. 200 mA.
Power supply: + 12 V, max. 200 mA (in total)
The pinout (3, 4, 5, 8, 9) complies with the GPO of PRODUCER.COM

Appendix C - Dimensions



Appendix D - System Update & License Installation

To update the system of PRODIGY or to install a license the device must be rebooted in Update Mode.

WARNING!

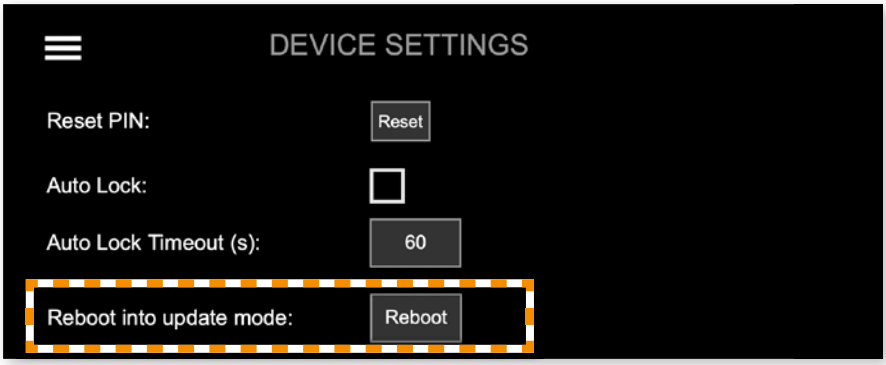
It is strongly recommended to backup the device configuration (Save Preset) before running any update.

1. Download Image Archive from the product page at www.directout.eu

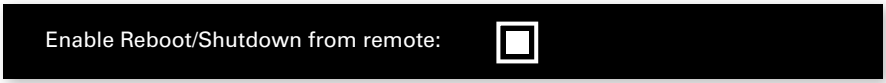
2. Unzip the Image file => prodigy_<xx>_system_update_<build>_<date>.pdgy

3. Start the device in Update Mode:
   Method A
   - Press and hold the button HOME on the front panel and switch the device on.
   - Once the DirectOut logo disappears you may release the button.

   Method B (introduced with system build 26 [MC] or build 12 [MP])
   - front panel: open SETTINGS and click Reboot



This command can also be used remotely via globcon or the web UI, once the feature has been activated on the front panel:



This setting is persistent and will be restored after a power-cycle or reboot.

WARNING

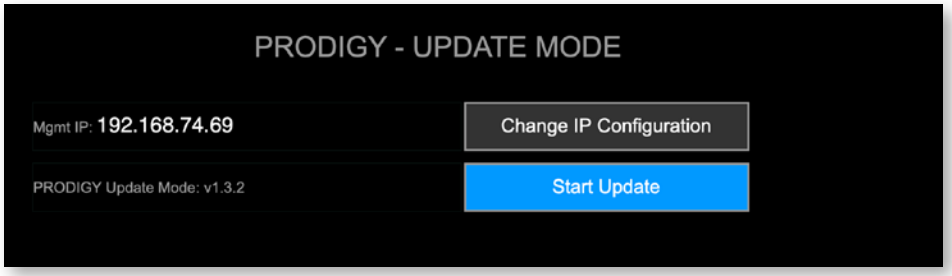
Enabling 'Reboot/Shutdown from remote' can be a security risk, if the management network can be accessed freely. It is disabled by default.

4. Enter the device's IP Address in your browser (Mozilla Firefox or Google Chrome). It is displayed in the front panel display. The IP address may vary in update mode when the network settings are set to DHCP.

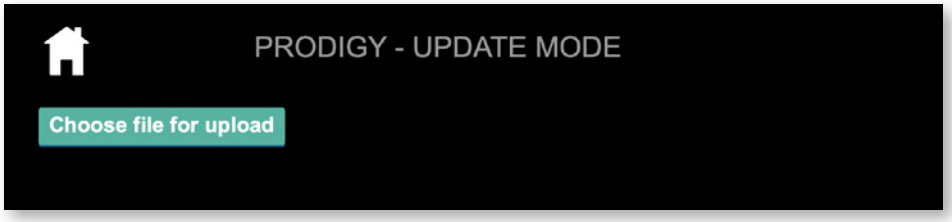
TIP

To keep the identical IP address in both operating modes, configure the network settings manually before rebooting into update mode.

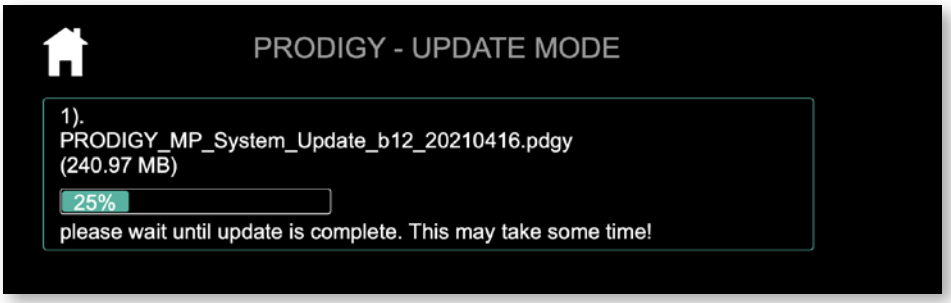
5. Click Start Update



6. Choose the downloaded file for upload (*.pdgy)

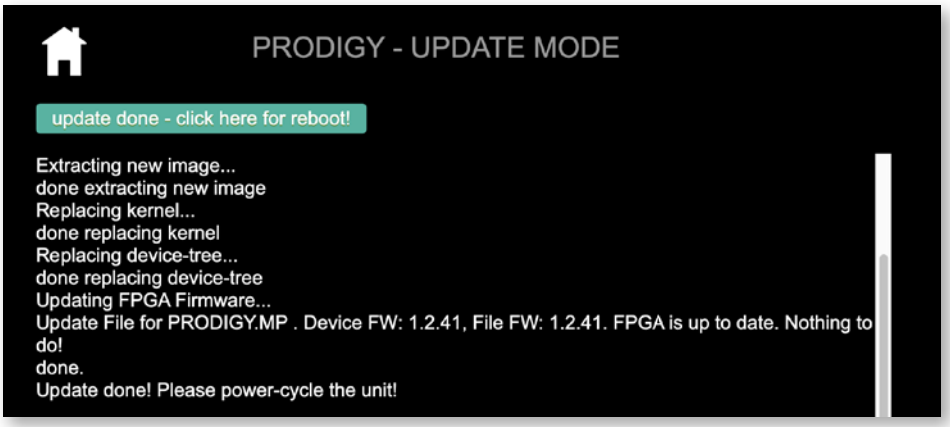


The update process will start after confirming your selection.



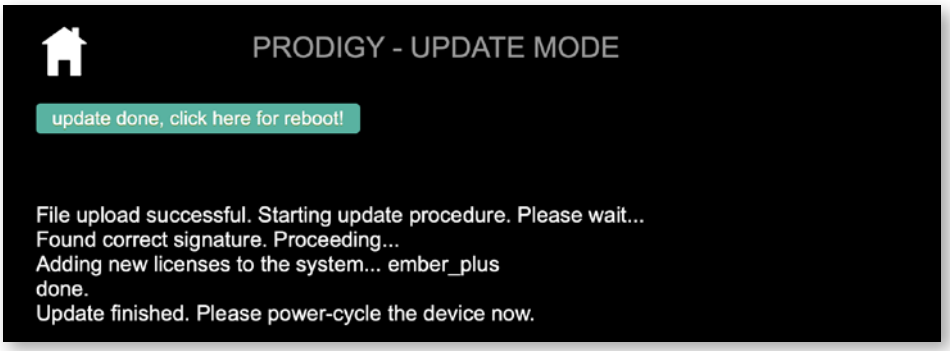
The entire update may take a couple of minutes.
Once the update is stored into the device a reboot message will appear.
The update progress messages are displayed below the upload box (green rectangle).

7. Once the reboot message appears you can power cycle the unit.



TIP

Installation of multiple licenses (not a system update!):
To install multiple licenses without rebooting the device in between, click on the HOME icon after the installation of a license has been completed successfully.



8. Check the new build version in the SETTINGS DEVICE INFO.

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