

D*AP4

Digital Audio Processor

D*AP4 FLX

D*AP4 LM Edition

Manual





Hardware Features

- **1RU** compact 19" processing device with front side info display
- **Dual power supply** second power supply for redundancy
- **Front panel info display** for signal activity, IP address, status alert
- **Two hidden touch buttons** to change the content of the info display
- **Remote Panel** optional X*AP RM₁ panel
- **Audio inputs** balance/unbalanced AES – manual selection
- **Audio outputs** balance/unbalanced AES
- **One interface slot** I/O expansion slot for one option board at a time
- **RJ45 network connector** 100BaseT full duplex Ethernet interface
- **USB B connector** built in USB < > serial adapter to access the device service port
- **8 GPI/Os** 8 balanced inputs, 8 relay closure combined on a 25pin D-Sub
- **Aux power supply** isolated 5V supply for external wiring
- **External sync IN** 75Ohm input (Word Clock, AES, Black Burst, Tri-Level)
- **Sync OUT** 75Ohm Word Clock output

Software Features in general

The **D*AP4** may be purchased as a **Level Magic Edition** and will appear as **D*AP4 LM** or based on the software licenses of the **FLX** concept. In this case it appears as **D*AP4 FLX**. Pls. contact your local dealer for details.

- **LevelMagic** loudness management according to ITU BS.1770-1/-2/-3
EBU R128, ATSC A/85, ARIB TR-B32, Free TV OP-59, Portaria 354
- **Dynamic filter** **optional** SPECTRAL SIGNATURE™ dynamic EQ
- **EQ** **optional** 5 band parametric
- **Dynamics** **optional** compressor, expander / gate
- **Fail over** **optional** automatic switch over with signal loss detection
- **Voice over** **optional** stereo or mono voice over extra program input, pan
- **FM Conditioner** **optional** MPX limiter and Pre-Emphasis
- **Loudness measurement** in reference to the selected standard
- **SNMP agent** SNMP v1, see D*AP4-MIB
- **Remote control** EmBER plus protocol or X*AP RM1 remote panel, mobile UI and legacy GPI/Os

Content

	<u>page</u>
Introduction	3
D*AP4 front panel view	4
D*AP4 rear view	4
Block Diagram	5
Audio Processing Blocks	6
Control, Operating & Event Concept	6
Getting Started – quick start guide	7
Getting Started – IP setup in general	8
Getting Started – IP setup of the D*AP4 – via console interface	8
Getting Started – IP setup of the D*AP4 – via web browser	9
Operating – menu structure of the X*AP RM1 remote panel	10
Operating – menu structure of the X*AP RM1 remote panel – operating displays	10
Operating – menu structure of the X*AP RM1 remote panel – menu tree	13
Setup GUI – connecting with the D*AP4	14
Setup GUI – SYSTEM – System Status	15
Setup GUI – SYSTEM – Overview	16
Setup GUI – SYSTEM – Admin	16
Setup GUI – SYSTEM – Setup	19
Setup GUI – SYSTEM – Remote Access – X*AP Remote	20
Setup GUI – SYSTEM – Remote Access – Mobile UI	21
Setup GUI – SYSTEM – the preset concept in detail	22
Setup GUI – SYSTEM – Preset Cleanup	22
Setup GUI – SYSTEM – SNMP	23
Setup GUI – SYSTEM – Backup / Restore	24
Setup GUI – SYSTEM – Firmware Update	25
Setup GUI – SYSTEM – Reboot	25
Setup GUI – INTERFACES – AES I/O	26
Setup GUI – INTERFACES – SDI I/O Interface – Overview	28
Setup GUI – INTERFACES – SDI I/O Interface – Local Routing	29
Setup GUI – INTERFACES – SDI I/O Interface – Setup	30
Setup GUI – INTERFACES – SDI I/O Interface – De-Embedder	30
Setup GUI – INTERFACES – SDI I/O Interface – Embedder	31
Setup GUI – INTERFACES – MADI Interface – Status / Setup	32
Setup GUI – INTERFACES – MADI Interface – Local Routing	33
Setup GUI – INTERFACES – Dante I/O Interface – Status	34
Setup GUI – INTERFACES – Dante I/O Interface – Inputs	36
Setup GUI – INTERFACES – Dante I/O Interface – Ouputs	37
Setup GUI – INTERFACES – Dante I/O Interface – Network	37
Setup GUI – INTERFACES – 4 Ch Analog I/O Interface	39
Setup GUI – INTERFACES – AES Interface – Status / Setup	40
Setup GUI – ROUTING	41
Setup GUI – AUDIO PROCESSOR – Overview	42
Setup GUI – AUDIO PROCESSOR – Setup	43
Setup GUI – AUDIO PROCESSOR – Input – Inputs	44
Setup GUI – AUDIO PROCESSOR – Input – Phase Rotator	44
Setup GUI – AUDIO PROCESSOR – Fail Over	46
Setup GUI – AUDIO PROCESSOR – De-Esser	46
Setup GUI – AUDIO PROCESSOR – Filter – Spectral Signature	47
Setup GUI – AUDIO PROCESSOR – Filter – Equalizer	50
Setup GUI – AUDIO PROCESSOR – Dynamics	52
Setup GUI – AUDIO PROCESSOR – Voice Over	54
Setup GUI – AUDIO PROCESSOR – Level Magic	55
Setup GUI – AUDIO PROCESSOR – FM Conditioner	56
Setup GUI – AUDIO PROCESSOR – Output	61
Setup GUI – AUDIO PROCESSOR – Delay	62
Setup GUI – AUDIO PROCESSOR – Mobile UI	62
Setup GUI – MESUREMENT – Setup	64
Setup GUI – MESUREMENT – Loudness	65

Content

	<u>page</u>
Setup GUI – EVENTS – Overview	66
Setup GUI – EVENTS – Triggers – Sources – Remote Hotkeys	67
Setup GUI – EVENTS – Triggers – Sources – Network	68
Setup GUI – EVENTS – Triggers – Sources – Parameters	68
Setup GUI – EVENTS – Triggers – Configuration – Trigger Equation	69
Setup GUI – EVENTS – Events – Preset Events	70
Setup GUI – EVENTS – Events – Parameter Events	71
Setup GUI – EVENTS – Events – Measurement Events	72
Setup GUI – EVENTS – Events – I/O Events	72
Setup GUI – EVENTS – Events – Bypass Events	73
Setup GUI – EVENTS – Actions – Event Actions	73
Setup GUI – EVENTS – Actions – Bypass Actions	74
Technical Data – 4 Channel Audio Processor [D*AP4 LM EDITION, D*AP4 FLX]	75
Technical Data – Option Board SDI I/O (3G/HD/SD) [O_DAP_SDI_a]	76
Technical Data – Option Board Analog Out [O_DAP_8DA_a]	77
Technical Data – Option Board Analog Out [O_DAP_ADDA_a]	78
Technical Data – Option Board AES/EBU I/O [O_DAP_AES_a]	79
Technical Data – Option Board MADI I/O, BNC [O_DAP_MB_a]	79
Technical Data – Option Board MADI I/O, Optical [O_DAP_MO_MM_a]	80
Technical Data – Option Board MADI I/O, Optical [O_DAP_MO_SM_a]	80
Technical Data – Option Board Audio-over-IP DANTE™ [O_DAP_DANTE_a]	81
Technical Data – Rear Connectors – pin assignment	82
Technical Data – Optional Interface Modules – pin assignment	83
Technical Data – GPI wiring	84
Safety Information	85
Warranty	85

Introduction

The **D*AP4** is a processing platform that may be bought in different versions as a 2 channel or 4 channel processor. The **D*AP4 LM** is a fully featured loudness control device **except** EQs, Spectral Signature and FM conditioner while the **D*AP4 FLX** will be tailor-made to your specification. Both devices may be upgraded later on locally.

This manual refers to a fully featured **D*AP4 LM**.

The **D*AP4** focuses on automatic and adaptive loudness management compliant with all current broadcast audio loudness recommendations including ITU.1770 standards (revisions 1, 2 and 3) as well as recommended practices ATSC A/85 (2011/2013), ARIB TR-B32, Free TV OP-59, Portaria 354 and EBU R128. The **D*AP4** features loudness normalization and dynamic range processing for up to two stereo programs of audio. Sophisticated fail-over and voice-over round off the **D*AP4** feature set. Dynamic range control and Level Magic™ are based on a unique multi-loop control principle.

LEVEL MAGICII™

The algorithm offers adaptive wideband control with exceptionally high audio quality uncompromised loudness management without any coloration, pumping, distortion or modulation effects by combining three major gain changing elements:

- Transient Processor
- Adaptive AGC
- Distortion-free true peak limiter

Spectral Signature™

Jünger Audio's Spectral Signature™ dynamic equalizer gives you a powerful creative tool to control the spectral balance of your program. Spectral Signature™ analyzes incoming audio and compares its spectrum with a predetermined reference curve. This allows dynamic EQ corrections to be applied, if necessary, to give a consistent sound impression. Spectral Signature™ is an optional feature for the D*AP4.

FM Conditioner

A powerful option for FM broadcast and other FM applications like analog cable head ends to pre-control the MPX power and peak deviation of a stereo FM signal, including true peak and pre-emphasis limiting.

System Integration

All system parameters are remotely accessible, allowing the unit to be integrated and remotely controlled by broadcast control systems. This helps users to apply individual processing to their programs, which is a key feature for well-managed loudness control.

Loudness measurement

To check compliance of programs with your local loudness regulations, the unit analyzes loudness and true peak levels from input signals and transfers the measurement data via Ethernet to an optional measurement and logging software anywhere in your network. These measurements can be triggered by automation systems via GPIs, via network or even manually on the **X*AP RM1** remote panel. The **D*AP4** can also generate SNMP or GPI/O alarms in case pre-determined limits are exceeded.

Web configuration

A web interface also allows easy and intuitive setup and configuration anywhere in your network.

Interfaces and system security

Audio I/Os range from onboard AES I/O to optional 3G/HD/SD-SDI I/O including video delay and analog I/O. All combined I/O interfaces support power fail bypass relays as standard. With optional redundant PSU and SNMP integration the unit ensures maximum operational safety.

D*AP4 front panel view



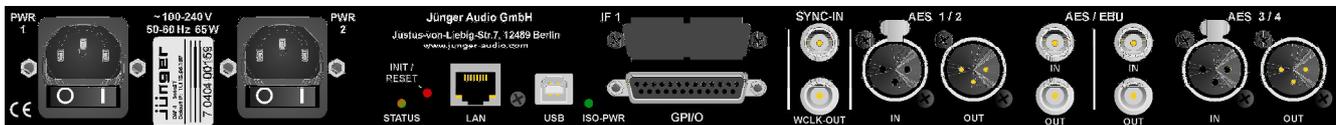
The front panel of the **D*AP4** has a 3 line status display and two hidden touch buttons ~ 2.5cm left of the display. **Button 1** = Home will switch back to the power up display no matter which display level you are in. **Button 2** controls the multi level display:

- Level 1** Power up display [Device type, firmware version]
- Level 2** Status [OK / Error] / Device Name / IP address
- Level 3** IN / OUT peak meter
- Level 4** Program 1 Out - short term loudness
- Level 5** Program 2 Out - short term loudness
- Level 6** Program 1 Out - integrated loudness and integration time
- Level 7** Program 2 Out - integrated loudness and integration time

The measures of the loudness displays depend on the setup of the respective loudness mode (see AUDIO PROCESSOR > SETUP > Loudness Mode).

Display background color
 Green = device status OK
 Red = device status ERROR

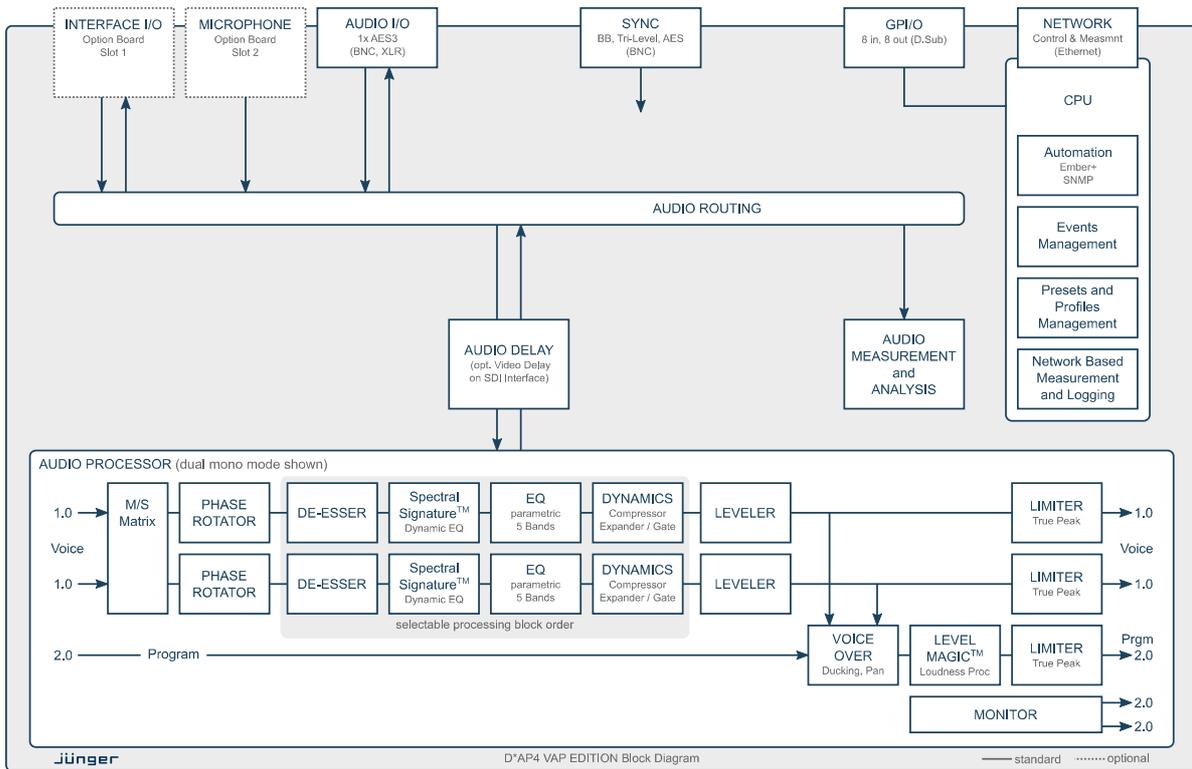
D*AP rear view



For fail safe operation, the **D*AP4** provides two independent power supplies. These power supplies operate in load balance. The status of both PSUs are combined with other status information and displayed as back light color the front panel display.

- STATUS** shows the status of the device controller.
- INIT / RESET** pressing the INIT button briefly will warm start the device controller. Holding down the button until the **STATUS** LED flashes 5 times will initialize the **D*AP4** to factory default
- LAN** RJ45 socket for Ethernet connection to a LAN
- USB** USB 2.0 type B socket to connect the built in **USB >> serial** converter with an external PC
- ISO-PWR** lights up if the isolated 5V power supply for GPI /O application is turned on
- Interface 1** slot to mount one of the optional interface boards (SDI, MADI, DANTE, AES, analog)
- GPI/O** 25pin Sub-D female connector to interface with the 8 optical isolated general purpose inputs and 8 solid state relay closure outputs
- SYNC IN** 75Ohm BNC connector to connect with external sync sources
- WCLK-OUT** 75Ohm BNC connector to synchronize external devices to the **D*AP4** internal word clock
- AES 1/2 IN / OUT** AES3 (XLR) and AES3id (BNC) input (selectable via GUI) / output (parallel)
- AES 3/4 IN / OUT** AES3 (XLR) and AES3id (BNC) input (selectable via GUI) / output (parallel)

Block Diagram



The above schematic shows the principal blocks of the **D*AP4**.

The core of the unit is the audio processor with 4 inputs and 4 outputs.

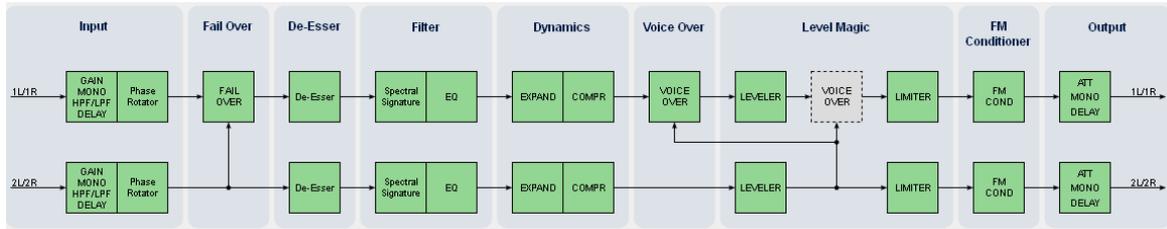
An AES I/O on the motherboard is provided for digital line operation. The respective connectors have relay bypass for power fail operation. The bypass circuit may be disabled by an internal jumper. For the 2 channel version only one AES I/O is fitted.

An interface slot is provided to carry optional 3G / HD / SD-SDI, AES I/O, MADI, DANTE or even analog expansion modules. It allows for extremely flexible interfacing of the **D*AP4** in TV or radio installations.

The sync. circuit can deal with all formats to integrate the **D*AP4** into digital facilities with a sample rate from 44.1 or 48kHz. Other devices may be synchronized by the word clock output of the **D*AP4**.

The **D*AP4** has 8 balanced **GPIs** and 8 relay closure **GPO** contacts. This enables the user to simply recall presets or call events, change device configurations and report general status information.

Audio Processing Blocks



Above you see the various function blocks of the audio processor rendered by the **DSP** engine. Each function block has its representation in the GUI by individual tab sheets. You may simply click on the respective graphical area as an alternative way to navigate through the GUI.

It is important to understand that the physical input interfaces of the device (SDI DE-EMBEDDER, AES IN) must be routed to the **DSP** inputs in order to process it. Similarly the **DSP** outputs must be routed to output interfaces (SDI EMBEDDER, AES OUT). You will find those settings by clicking on the **ROUTING** tab.

Control Concept

The communication between external applications or the **X*AP RM1** remote panel, is based on **TCP/IP over Ethernet**.

The setup GUI utilizes web technology. At the time of editing this manual the functionality of the web GUI is optimized for Firefox 30.x and higher.

The setup GUI can be complemented by other application programs running on MS Windows® XP, W7, W8 like the Junger Application Manager **J*AM**. Operator access will also be available for mobile devices running an appropriate browser on iOS or Android.

An **SNMP** agent may be activated to incorporate the device into a station monitoring system.

For 3rd party remote applications, **Junger Audio** highly recommends using the **Ember+** protocol which is widely distributed in the European broadcast industry. The user community is also increasing rapidly world wide. By default, the **X*AP RM1** remote panel and the **D*AP4** "talk" Ember natively.

Operating Concept

Further below you will see that the setup GUI for the device is grouped into several parameter areas. One can reach the parameters via a 3 tier navigation by tabs which may have sub tabs, and the sub tabs may have pages embedded or extra soft buttons for groups of parameters.

Each function block (parameter area) has dedicated presets. The presets can be recalled at any time during operation, either by manual intervention via the embedded web server (browser based GUI), automatically by the internal event manager or by external applications.

For all relevant settings an **ON AIR** and a **PRESET** part exists. I.e. you may either edit the parameters **ON AIR** or **offline** for the respective part of the **D*AP4**.

The presets of the **D*AP4** are persistent by nature. You are working directly on the preset memory. I.e. you need not to worry about storing such presets, the **D*AP4** does it for you.

Event Concept

The **D*AP4** incorporates a sophisticated event management system.

Events may be combined to perform actions. The **D*AP4** offers these event types:

- * **Preset Events** for System set up, Interfaces, Routing, Audio Processing etc.
- * **I/O Events** to control GPOs
- * **Bypass Events** for pre-configured bypass scenarios
- * **Measurement Events** to control loudness measurements per program

These events may be combined to form **Actions** which are fired by **Triggers**.

Triggers are defined by a logical combination (AND, OR, XOR) of two random trigger sources.

Trigger source may be GPIs, hotkeys of the **X*AP RM1** remote panel, network commands, parameters, other active events, other active triggers (nested trigger), or device status information (e.g. sync lost).

Getting Started – quick start guide

Before the **D*AP4** can be used, there are some basic configuration steps which must be followed in the order set out below. This example assumes you will process one stereo program that is embedded into SDI group1 Ch1/2.

- * Connect the SDI signal (from a source like the station router or video server) to the SDI IN.
- * Connect the SDI OUT connector to your destination device (station router or monitor box).
- * Connect the BNC SYNC IN to the Black Burst reference of your TV station.
- * Hook up the device to the station PC network
 - Consult your IT administrator for assistance if you are not sure about this procedure
 - Connect it to a switch or hub or directly to a PC / LapTop by an Ethernet cable (some PCs need a cross over [1:1] cable when connected with the D*AP4 directly)
 - Find an unused IP address - ask your administrator!
 - Assign it that IP address and set the network mask accordingly, a gateway is optional (see next page for details)
- * Open a browser (FireFox recommended) and connect with the device
 - Type in the IP address as an URL
- * Set the **sync source**
 - SYSTEM > Setup > Sync Source Priority > **Choice 1=Sync-In BB**
leave all other **Choices x=OFF** (for the beginning)
 - SYSTEM > Setup > System Clock > **Sample Rate=Follow Source**
- * Set the routing to the Audio Processor (DSP)
 - ROUTING > SDI DE-EMBEDDER > **DM1=DSP 1**
 - ROUTING > SDI DE-EMBEDDER > **DM2=DSP 2**
- * Set the routing from the Audio Processor (DSP)
 - ROUTING > DSP > **DSP 1=SDI EMBEDDER > EMB 1**
 - ROUTING > DSP > **DSP 2=SDI EMBEDDER > EMB 2**
- * Enable the SDI embdder
 - INTERFACES > SDI I/O Interface > Embedder > **SDI OUT Gr1=ON** (check box)
 - Check if the routing radio buttons will connect "**From Routing EMB 1/2**" to "**SDI Out Gr1 1/2**"

Now you should hear your source stereo program signal at the destination and you may start experimenting with the various parameters of the audio processing blocks.

Getting Started – IP setup in general

The process of installing a **D*AP4** into an **IP network** is as follows:

1. Ask the system administrator for a unique IP addresses of the network, the netmask and gateway address
2. Assign the **D*AP4** an IP address

You have 2 choices to assign the **D*AP4** an **IP address**:

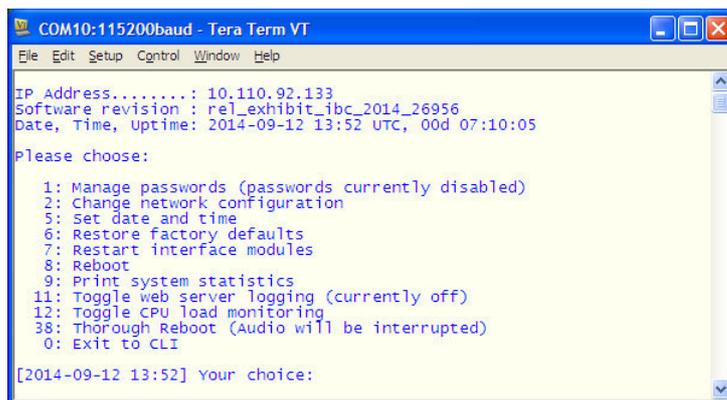
- * From the serial console interface
- * Via Web browser

Important Note! If you are not familiar with setting up devices for IP communication, we highly recommend you consult your system service or IT department to assist you.

Getting Started – IP setup of the **D*AP4** – **via console interface**

The tool to change the IP configuration of the **D*AP4** can be selected via the console interface. You must connect it with the PC via an **USB A to B** cable. This will install the driver for the built-in **USB to serial converter**. Now you can open a terminal program. Here you must select the virtual **COM port** assigned by the OS. The communication parameters are:

115200kBaud, 8, N, 1 no hand shake. Pressing **<ENTER>** will open the console menu:



[2014-08-22 12:01] Your choice:

Select item "2": **<ENTER>**

Current network configuration

IP Address: 10.110.24.128
Netmask ...: 255.255.0.0
Gateway ...: 10.110.0.1

Enter new IP address, press ENTER to cancel:

You must enter the new IP address (e.g.): "192.168.178.78" **<Enter>**

Enter new netmask, press ENTER to cancel:

You must enter the new netmask (e.g.): "255.255.255.0" **<Enter>**

Enter new gateway address, press ENTER to configure without gateway:

You may press **<Enter>** to skip this point or

You may enter the new gateway address (e.g.): "192.168.178.1" **<Enter>**

Important Note! The gateway entry is optional but you must take care that the gateway address matches the network mask related to the device IP address! If you are not sure simply enter **0.0.0.0** or leave it without an entry.

Changing Network configuration

Network configuration has been changed. Please reboot the device to activate the new settings.

Select item "8: Reboot" <ENTER>

Do you want to reboot the device ?

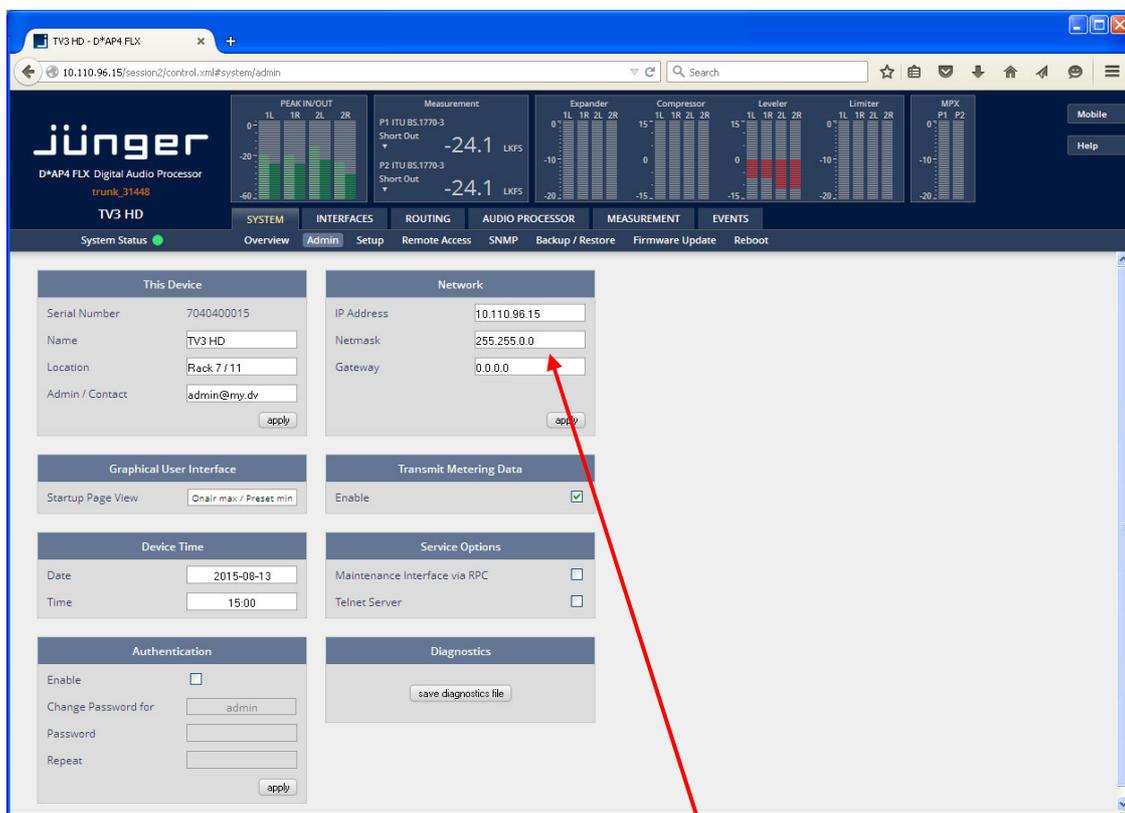
Press small "y" <ENTER>

Rebooting the device

After reboot has finished, the new IP configuration is active and will be displayed at the top of the configuration menu.

Getting Started – IP setup of the D*AP4 – via web browser

- * Read the **default IP address** printed on the label at the rear of the device.
- * Set up network parameters of your PC to fit the default IP address of the **D*AP4** (e.g. default IP +1 and net mask = 255.255.0.0).
- * Connect the **D*AP4** with the PC either by a Ethernet patch or a cross over cable (if the PC does not support Auto MDI-X) or via a switch.
- * Open a browser and type the default IP address of the **D*AP4** into the URL field and press <ENTER>. This will open the **AUDIO PROCESSOR** tab sheet of the GUI.
- * Click on <SYSTEM> and afterwards the "Admin" tab:



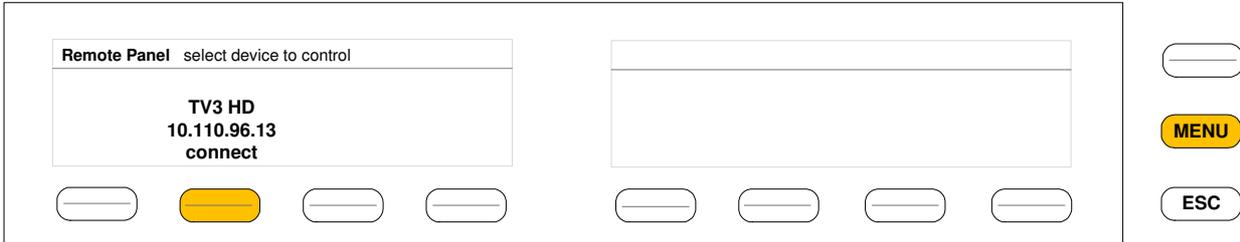
Enter the desired network configuration and press <apply>

Afterwards you must reboot the **D*AP4** in order to activate the new IP configuration.

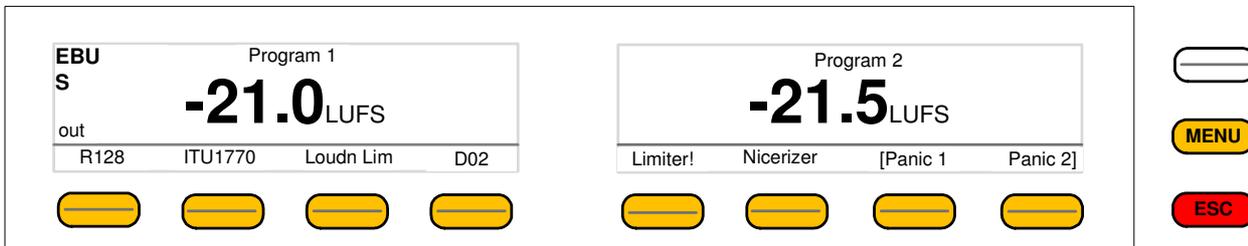
Important Note! After reboot neither the **web browser** nor the **X*AP RM1** remote panel may be able to communicate with the **D*AP4**. You must change back the IP configuration of the PC and fill in the **new IP** address in the URL field. You must change the **X*AP RM1** remote panel settings as well to attach this device again.

Operating - menu structure of the X*AP RM1 remote panel

Power up display – may show up to four D*APx enabled for remote control for this X*AP RM1 remote panel. The example below has just one D*AP4 unit [given name "TV3 HD"] attached for remote control. The status is "connect" (i.e. you may connect with that device). See X*AP RM1 manual for details.



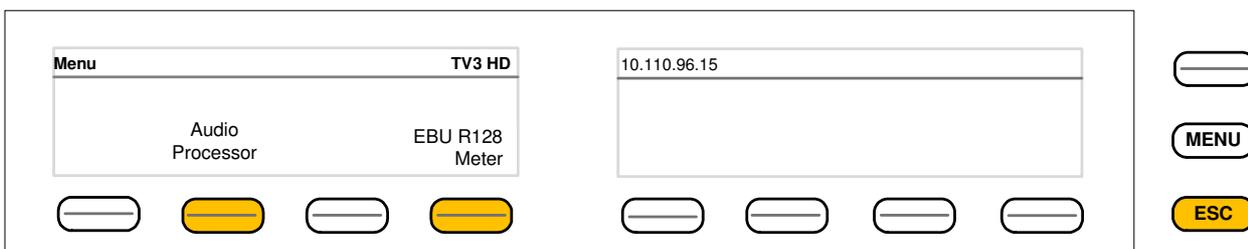
Pressing that button will connect with that D*AP4. Now the X*AP RM1 remote panel will gather all necessary information from that D*AP4 unit (this may take a few seconds). When finished the main operating display opens up:



From here you may fire pre-defined hotkeys and observe the loudness of the processed programs. Because this is the main operating display, the <ESC> button lights red to indicate that the power up display is below the main operating display. Pressing <ESC> sends you back to the power up display (device selection).

Operating – menu structure of the X*AP RM1 remote panel – operating displays

When you press the <MENU> button, the upper operating display opens up:



When pressing the <ESC> button you will return to the main operating display.

Operating display – EBU R128 Meter

The meter style (ITU BS.1770-x / ATSC / EBU etc.) is defined by the settings of:

AUDIO PROCESOR > Level Magic > Loudness Mode (example is for EBU R12()).

The above menu serves as a display of measurement values and offers the metering control buttons (reset & pause / continue).

<ESC> returns to the **main operating display**.

Operating display – Audio Processor

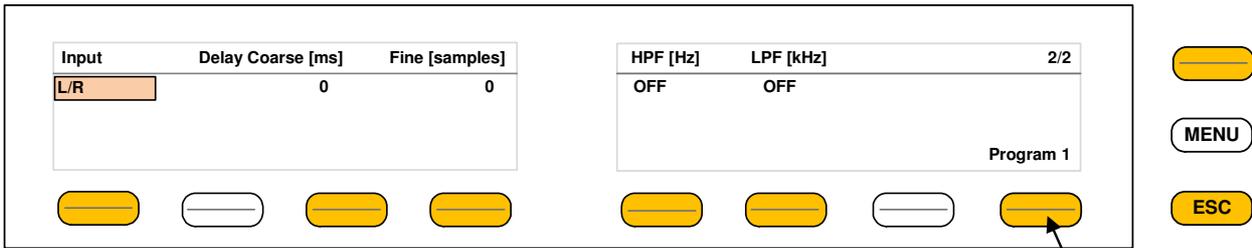
<ESC> returns to the **operating display**.

This menu gives access to tweak the various function blocks. The active <Shift> ● button indicates that there is another page (2/2):

The example below explains how to set parameters via the **X*AP**. E.g. if you press <Input> all **p**arameters for the input function block will be accessible:

Here you are at the input ● section of the first program. The <Shift> ● button again toggles between two pages and gains access to the remaining parameters:

Here is another example for **page 2/2** after pressing the **<Shift>** button:



The **D*AP4** knows two independent programs:

See **SYSTEM > Setup > Program Configuration = 2 x 2**. You can select between both ● here.

You are able to **link / unlink** the respective processing blocks (see **AUDIO PROCESSOR > Input**):

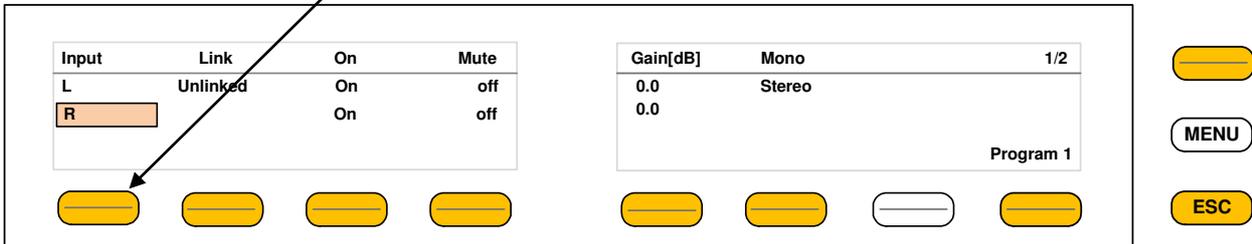


The above example shows both channels of "Program 1" (default name) in **Linked** mode.

When you press **hotkey #2** you are able to unlink both channels. Now you must simply push the rotary encoder (or turn it counter clockwise / clockwise) to toggle between **Linked** and **Unlinked** condition.

In case of **Unlinked**, the display shows two independent parameter sets.

By pressing **hotkey #1** ● you can toggle the between the channel that is under control:



The examples above demonstrates the general way how to setup parameters of the **AUDIO PROCESSOR** of the **D*AP4 VAP**:

- * Select a parameter
- * Change it by using of the **Rotary Encoder**.
 - Push it to toggle states
 - Turn it to increment / decrement values.

Important Note! Not all processing blocks can be linked / unlinked. Carefully compare the settings via the web GUI if you are not certain about individual settings. In general the **X*AP RM1** menus are a duplication of the GUI settings. To access all parameters of a function block you must sometimes use the **<Shift>** button. E.g. the equalizer has 5 pages for one program channel!

Operating – menu structure of the **X*AP RM1** remote panel – **menu tree**

Power Up Display – select a remote device

<MENU> opens **X*AP RM1** remote panel IP setup menu.

Hotkey #

- 1 <Address> setup
- 2 <Netmask> setup
- 3 <Gateway> setup
- 4 < empty >
- 5 Device 1 setup IP & ON / OFF
- 6 Device 2 setup IP & ON / OFF
- 7 Device 3 setup IP & ON / OFF
- 8 Device 4 setup IP & ON / OFF

<ESC> back to **power up** display

<connect> will connect with that particular **D*AP4** and opens the **main operating** display:

Hotkey #

- 1 <R128>
- 2 <ITU1770>
- 3 <Loudn Lim>
- 4 <D02>
- 5 <Limiter!>
- 6 <Nicerizer>
- 7 <[Panic 1>
- 8 <Panic 2]>

<ESC> will jump back to **power up** display

<MENU> opens **upper operating** display:

Hotkey #

1 <empty>

2 **<Audio Processor>** opens up the function block selection

- | <u>Hotkey #</u> | Hotkey # (after pressing <Shift>) |
|------------------------|-----------------------------------|
| 1 <Input> | <Level Magic> |
| 2 <Phase Rotator> | <FM Conditioner> |
| 3 <Spectral Signature> | <empty> |
| 4 <Fail Over> | <empty> |
| 5 <De-Esser> | <empty> |
| 6 <Dynamics> | <empty> |
| 7 <Level Magic> | <empty> |
| 8 <Voice Over> | <Output> |

<ESC> back to operating display

3 <empty>

4 **<EBU R 128>** opens the loudness meter display

Hotkey #

- 1 <Input>
- 2 <Fail Over>
- 3 <Spectral Signature>
- 4 <Equalizer>
- 5 <Dynamics>
- 6 <Voice Over>
- 7 <Level Magic>
- 8 <Output>

<ESC> returns to the **upper operating display**

5 <empty>

6 <empty>

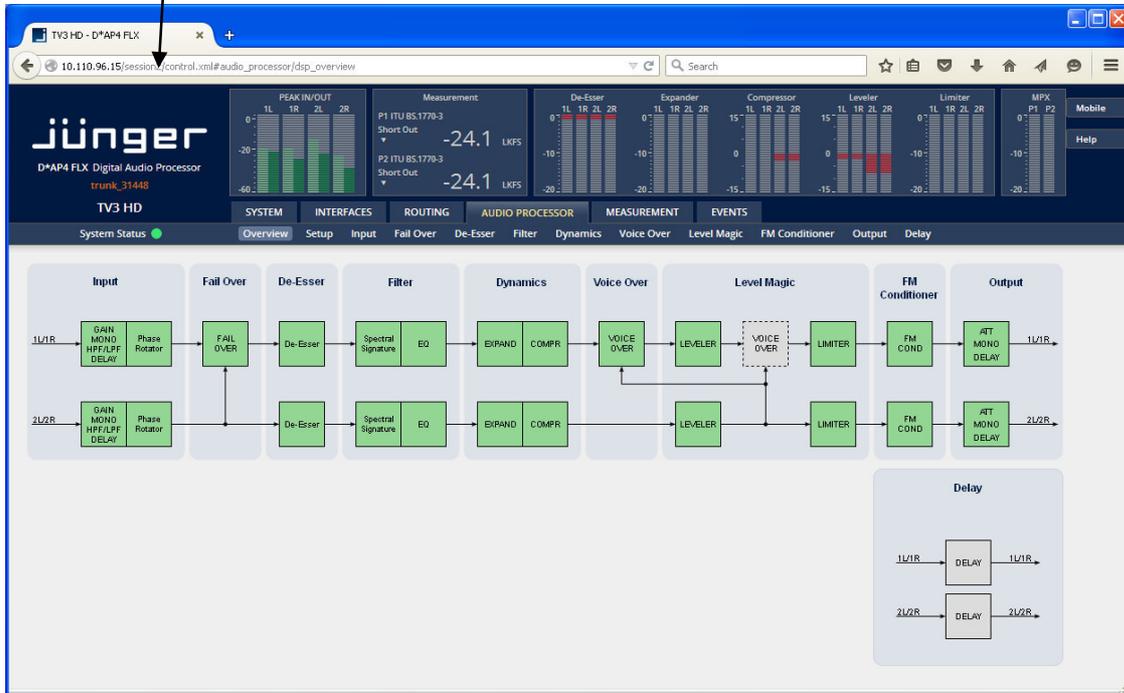
7 <empty>

8 <empty>

<ESC> returns to the **main operating display**

Setup GUI – connecting with the D*AP4

You must open a browser and enter the **IP address** of the **D*AP4** into the **URL** field and press **<Enter>**. The browser will retrieve the necessary information and open up the entrance page:



The entrance page is the **AUDIO PROCESSOR** pane with its sub pane **Overview**. If you are returning from other pages or if you reload your browser content (by pressing **<F5>**) it may show a different page due to caching of the browser.

In the top section you see several bar graph displays for signal levels as well as for gain reduction display of several function blocks.

On the following pages we will go through the various panes to perform the basic setup of the device.

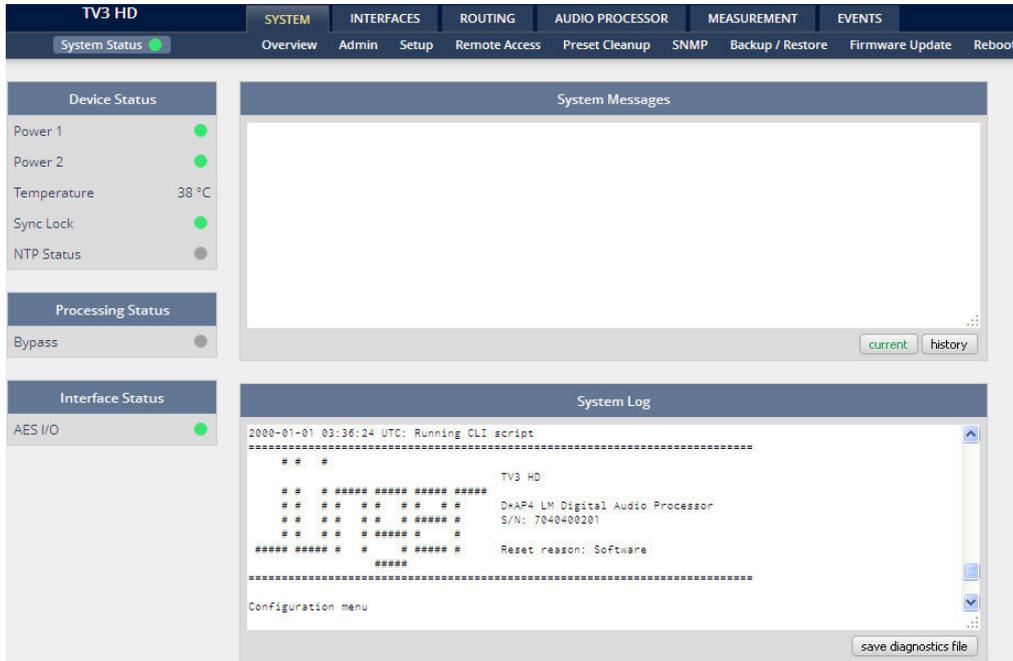
You must set up the synchronization source. You may also give the device a name, tell it its location and define an administrative contact which may be used by the monitoring system of your house (e.g. via SNMP).

You must set up the installed interface module and finally set the signal routing.

You will find those settings under the **SYSTEM** link.

Setup GUI – SYSTEM – System Status

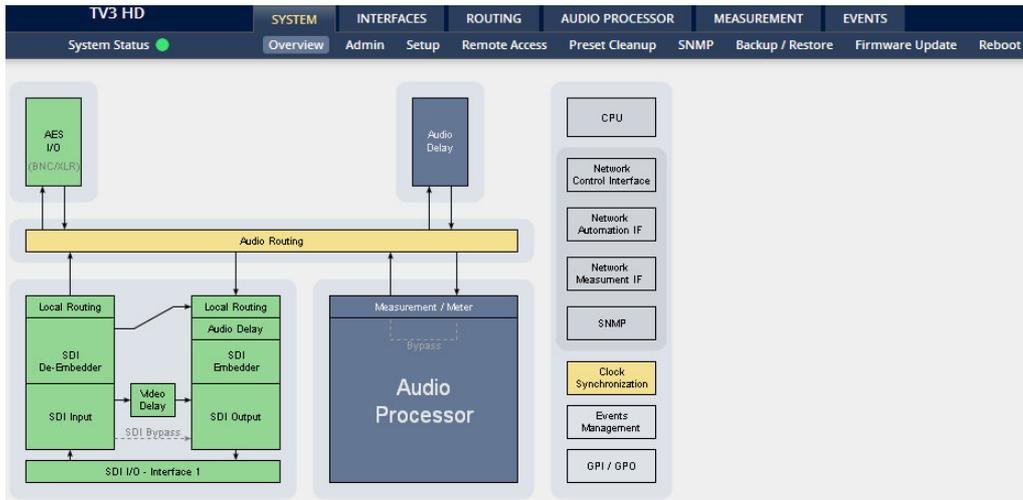
The system status is a special link you can reach independently from where you are:



The **System Status** page provides a top level view of the various status information available for the device.

Device Status	Provides the top level status of the D*AP4 .
Power 1	Status of the first power supply (left hand side of rear).
Power 2	Status of second power supply (to the right of the first power supply)
Temperature	Measured on the surface of the main PCB.
Sync Lock	Turns red if the external sync source is lost or unstable.
NTP Server Status	Is grey if the NTP server synchronization is turned off. It is green if the clock is synchronized. Turns red if the clock can not be synchronized by one of the NTP servers.
Processing Status	
Bypass	Turns red if Bypass is activated.
Interface Status	
AES I/O	Turns red if an AES input that is internally in use (i.e you have routed it to an input of a function block) has detected an error.
System Messages	[current / history] Displays a list of messages produced by the system controller.
System Log	The system controller activities will be logged. This log information may be downloaded from the device and sent to Junger Audio. In case of a problem you can press: <save diagnostics file> from here or from: SYSTEM > Admin > Diagnostics.

Setup GUI – SYSTEM – Overview



The graphical overview shows the main building blocks of the device including the options installed, in this example SDI interface placed into the interface 1 location (see rear view).

You may click on the boxes and the respective setup page will open. The navigation is based on URLs so you may use the <Back> navigation button of the browser to return to this page.

Setup GUI – SYSTEM – Admin

TV3 HD		SYSTEM	INTERFACES	ROUTING	AUDIO PROCESSOR	MEASUREMENT	EVENTS			
System Status ●		Overview	Admin	Setup	Remote Access	Preset Cleanup	SNMP	Backup / Restore	Firmware Update	Reboot
This Device Serial Number: 7040400010 Name: TV3 HD Location: Rack 14 Admin / Contact: you@youtv.com <input type="button" value="apply"/>		Network IP Address: 10.110.96.10 Netmask: 255.255.0.0 Gateway: 10.110.0.1 <input type="button" value="apply"/>		Device Time Date (Local): 2016-06-16 Time (Local): 12:32 Date (UTC): 2016-06-16 Time (UTC): 10:32 Get Time from: NTP Server Primary NTP Server: 5.9.110.236 Secondary NTP Server: 10.110.2.7 Update Rate (min): 30 The NTP server must be an IP address and cannot be a domain name. For external NTP server the Network Gateway needs to be configured accordingly.						
Graphical User Interface Startup Page View: Onair max / Preset max		Transmit Metering Data Enable: <input checked="" type="checkbox"/>		Authentication Enable: <input type="checkbox"/> Change Password for: admin Password: <input type="text"/> Repeat: <input type="text"/> <input type="button" value="apply"/>						
		Service Options Maintenance Interface via RPC: <input type="checkbox"/> Telnet Server: <input type="checkbox"/>		Diagnostics <input type="button" value="save diagnostics file"/>						

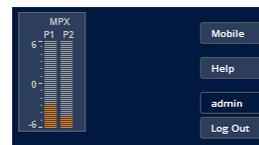
This Device	Input fields for information utilized by higher level services.
Serial Number	The electronic serial number. It is printed on a label at the rear of the device.
Name	Give the device a meaningful name that may be used by name services and SNMP management.
Location	The place where the D*AP4 is located.
Admin / Contact	E-mail address of a person in charge.
Graphical User Interface	[Onair max / Preset max, Onair max / Preset min, Onair min / Preset max, Last Used] Defines the appearance of the parameter panes in the ON AIR vs. the PRESETS area (which one will be visible).
Authentication	To prevent non-authorized people from changing D*AP4 settings the administrator may assign passwords for either the admin and/or an operator. While the admin is allowed to set everything, an operator is just allowed to load presets. Parameters will be reset if the operator attempted to change it.
Enable	[ON / OFF] The administrator may turn authentication OFF.
Change Password for	[admin / operator] Select which password you will set / change
Password	Type in a password Default passwords are: admin (for admin) and operator (for operator).
Repeat	Repeat that password

Important Note! The authentication may be enabled / disabled from the **console** interface as well (see page 8 "1: Manage Password") via USB connection but also via Telnet! If you have higher security demands you should turn the Telnet server off. Authentication will be turned off and passwords will be reset if one initializes the device to factory default (see Reboot - page 19, INIT/RESET rear button - page 4).

If there was an authentication failure, the **admin** will be notified at the next proper login about such conditions. The pop up appears for each login that has failed. It shows the IP address of the device that caused the authentication failure.



After a correct login the status "who" (e.g. admin) and a **<Log Out>** button are available from the GUI in the upper right corner:



Network	IP address setup, see above: getting started – IP setup of the D*AP4 – via web browser
IP Address	A proper address for your network – default [10.110.xxx.yyy]
Netmask	The net mask of your network – default [255.255.0.0]
Gateway	The optional gateway address – default [0.0.0.0]

Transmit Metering Data [OFF / ON]
metering data will be streamed via UDP protocol. In order not to receive such data by external applications you may disable it.

Service Options

- Maintenance Interface via RPC** [OFF / ON]
For administrative use to enable communication with factory tools.
- Telnet Server** [ON / OFF]
Enables a telnet server to connect to the consol interface via Telnet (IP port 21).

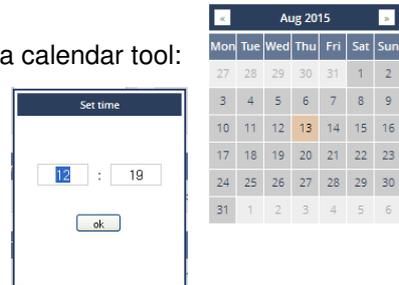
Diagnostics

- <get diagnostics file>** Pressing this soft button will start the assembly of a diagnostics file. The file will be presented in XML format for download. If you experience unexpected behavior of the device you may be asked by the Junger service team to send such file by e-mail for analysis.

Device Time

Allows you to set the device clock. At the factory it will be set to UTC (Coordinated Universal Time).

- Date (Local)** If you click into the **Date (local)** input field, a calendar tool: appears to select month and year.
- Time (Local)** If you click into the **Time (local)** input field, you will be able to set the device time.
- Date (UTC)** Similar as above for local date setting.
- Time (UTC)** Similar as above for local time setting.
- Get Time from** [Manual Setting / Browser / NTP Server]
If set to **NTP Server** the D*AP4 will look for the below servers to synchronize the internal clock.

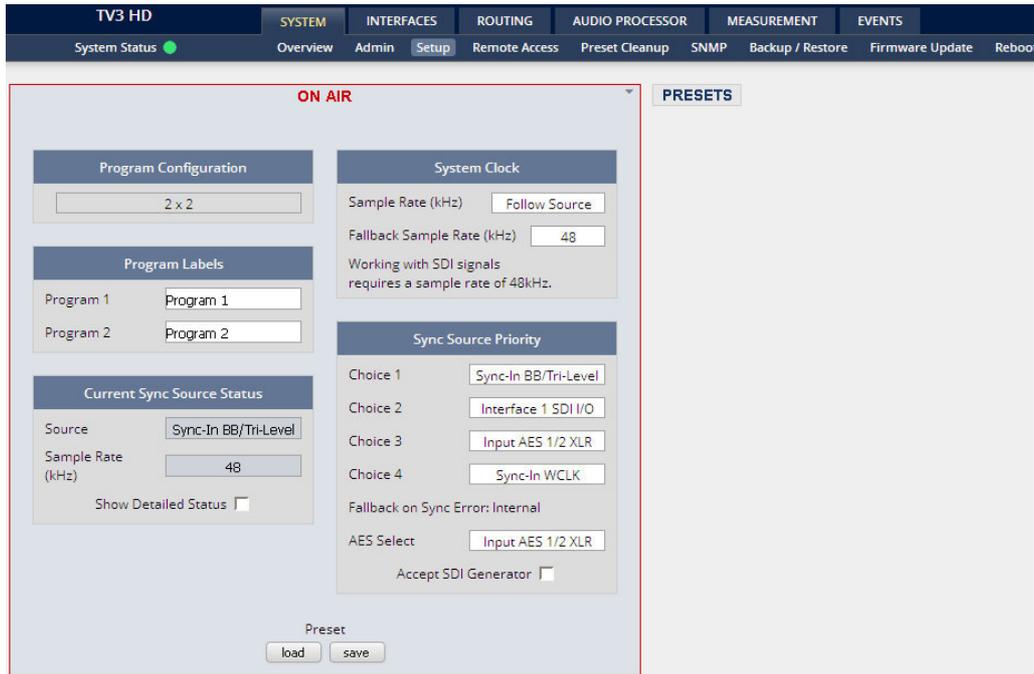


- Primary NTP Server** [5.9.110.236] default set to a publicly accessible NTP server via internet. This is used for device testing an may be overwritten at any time.
- Secondary NTP Server** [10.110.2.7] default set to an internal NTP server from Junger Audio. This is used for device testing and may be overwritten at any time.

Important Note! If it is impossible to synchronize the internal clock to one of the two NTP servers an SNMP "ntpStatusTrap" will be issued by the SNMP agent (if enabled SYSTEM > SNMP > Enable = ON).

- Update Rate (min)** [1 ... 1440]
Interval of synchronizing the internal clock of the D*AP4.

Setup GUI – SYSTEM – Setup



Program Configuration

[2 x 2]

Shows the program configuration (2 times 2 channel). This is also the default configuration of the audio processing blocks.

Program Labels

Program 1

Each of the two possible programs has a name that will be used as a reference for the display of parameters and their setup.

Program 2

You may edit the default names.

Current Sync Source Status

Source

shows the status of the 5 tier sync priority circuit

Sample Rate

active sync source

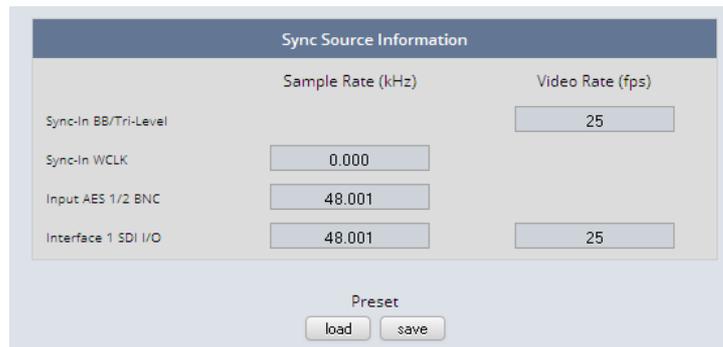
measured sample rate

Show detailed status

[ON / OFF]

If you enable the checkbox you will get this information:

Sync Source Information



You will get detailed information about the measured rates of possible sync sources

System Clock

- Sample Rate** [Follow Input / 44.1 / 48 / 88.2 / 96]
- Fallback Sample Rate** [44.1 / 48 / 88.2 / 96]
- Fallback Video rate** [25 / 29,97 / 30]

Sync Source Priority

- Choice 1 – 4** [OFF / Internal / Sync-In WCLK / Sync-In AES {or Input AES 1/2 XLR or Input AES 1/2 BNC} / Interface 1 {SDI I/O or AES I/O or non} / Sync-In Black Burst/Tri-Level]
- Fallback on Sync Error:** [Internal]
If the selected sync source is not available the next source will be selected. If none of the pre selected sync sources is not available, the source will fall back to the internal clock oscillator.
- AES Select** [Sync-In AES / Input AES 1/2 XLR / Input AES 1/2 BNC]
Select from which physical input the AES sync must be taken.
- Accept SDI Generator** [ON / OFF]
For rare application you may use the SDI generator (if an SDI I/O interface is installed) as the sync source. In this case downstream equipment must be synchronized to the **D*AP4**. See **INTERFACES > SDI I/O interface > Setup** for details.

Important note! It is not possible to gen lock the SDI generator. The generator will run on its own internal 27MHz crystal clock.

Setup GUI – SYSTEM – Remote Access – X*AP Remote

The **X*AP** can control multiple **D*AP4s** one by one and a single **D*AP4** may be controlled from multiple **X*APs**. This requires a flexible remote concept that allows you to recall pre-set configurations from the **X*AP** panel or from the Mobile UI. You can control pre-settings of the **EVENTS** system via remote access from the **X*AP** remote panel or from a mobile **UI** on a tablet, a smart phone or even via a browser session from any PC in the network.

To better understand the possibilities of these settings it is recommended to study the comprehensive **EVENTS** system of the **D*AP4**.

Here you can define feature sets for an **X*AP**. At the moment it allows you to limit the access of a specific **X*AP** to load profiles only or to gain full access.

X*AP Remote		Mobile UI	
X*AP Remote	X*AP Remote Feature Set		
IP Address			
Default / Not listed	Standard Set		
10.110.68.129	Standard Set		
10.110.68.128	Standard Set		
	Standard Set		

Multiple **X*AP** remote panels may gain access to the **D*AP4**. For each **X*AP** you may pre-set general functions:

This function is not implemented yet, so you see: "Full Access" instead.

*Profiles will be set up in the **EVENTS > Preset Events (Profiles)** area. They combine a number of presets of several processing blocks.*

IP Address

In the first line you define the access policy for an "unknown" **X*AP** that connects with this **D*AP4** for the first time. The other lines are used to pre-define features for known **X*APs**. When enabling an unknown **X*AP** to connect with this **D*AP4**, the respective **IP address** will be inserted automatically into the next empty line.

Setup GUI – SYSTEM – Remote Access – **Mobile UI**

For **mobile UIs** on tablets, smart phones or PCs you can assign the features via its **IP address**:

X*AP Remote		Mobile UI	
Mobile UI Device	Mobile UI Features		
IP Address	Hotkeys	Actions	
Default / Not listed	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	
	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	
	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	
	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	
	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	
	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	
	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	
	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	
	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	

IP Address

When connecting from a mobile device that is not pre-set, the respective **IP address** will be automatically inserted.

Hotkeys

[ON / OFF]
Turn it on if the **UI** should display the hotkeys of the **X*AP** that is connected with this **D*AP4**.

Actions

[ON / OFF]
Turn it on if the **UI** should show actions to trigger from the **UI**.

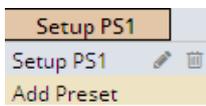
Setup GUI – SYSTEM - the **preset concept** in detail

The example above shows the **preset concept** of the **D*AP4**. It is a general feature of the device and you will come across it in almost every area. For all relevant settings one set of **ON AIR** parameters and a practically unlimited number of **PRESETS** are available. The count depends on the NV memory space left. If you want to load parameters from a preset to the **ON AIR** area or save parameters from the **ON AIR** area to a preset, you must press **<load>** or **<save>**:



A dialog opens to select the desired preset. When you press **<ok>** the selected action will be executed. When you press the little pencil icon the preset name turns *italic* and you may edit it.

To generate a new preset offline, you must click into the preset name field below the **PRESET** headline:



The pull down offers **"Add Preset"**. If you select this a new entry to the list will be generated. Clicking on the small trash bin symbol will delete that preset. You may change the default name "Preset x" by clicking the small pencil icon. Now the default name becomes *italic* and you may edit that name.

If you have selected the new preset or one of the existing presets indicated by the name displayed at the top, you may edit the parameter values.

Important Note! The presets of the **D*AP4** are persistent by nature. You are working directly on the preset memory, i.e. you need not worry about storing such presets. The **D*AP4** does it for you.

On the other hand you must be aware that you are **overwriting the actual preset settings!** If you want to keep original values (e.g. from a factory preset) you must simply **copy** the content of an existing one to the clip board, add a new preset, name it differently and **paste** the clip board to it.

At the bottom of the **PRESET** part you find the soft buttons to **<copy>** the content of that preset to the clip board or to **<paste>** the content of the clip board into an other preset which you have selected before pasting.

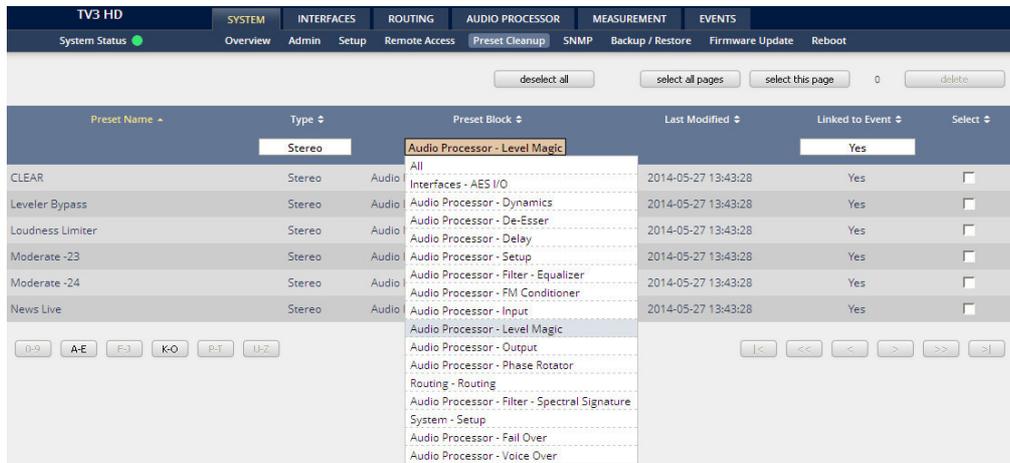
You may also **<export>** or **<import>** the preset content to / from a file.

Setup GUI – SYSTEM – **Preset Cleanup**

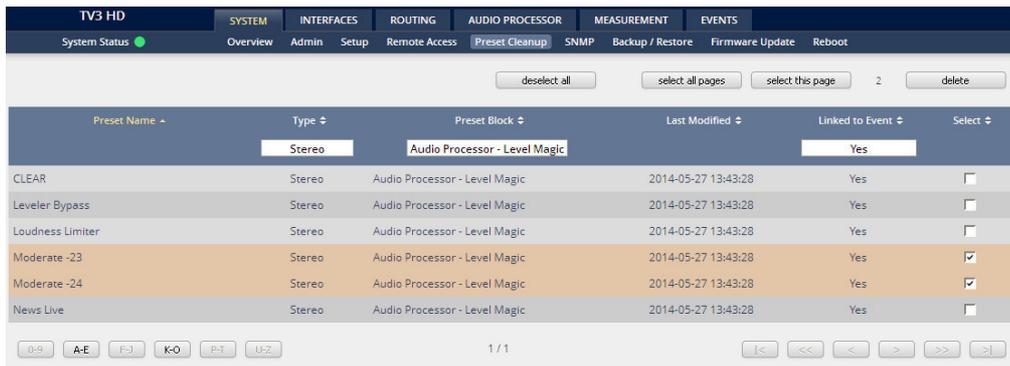
It is sometimes desirable to delete presets which are used by multiple events without stepping through all processing blocks and deleting the respective presets one by one. This pane offers you a tool to remove presets via a central access point:

Preset Name	Type	Preset Block	Last Modified	Linked to Event	Select
50 Hz Hum Remover	Stereo	Audio Processor - Filter - Equalizer	2014-05-22 14:52:34	No	<input type="checkbox"/>
60 Hz Hum Remover	Stereo	Audio Processor - Filter - Equalizer	2014-05-22 14:52:34	No	<input type="checkbox"/>
Anti-Sizzle	Stereo	Audio Processor - De-Esser	2015-04-02 10:41:21	No	<input type="checkbox"/>
Auto	Stereo	Audio Processor - Phase Rotator	2015-04-02 10:39:29	No	<input type="checkbox"/>
B42 Female	Stereo	Audio Processor - De-Esser	2015-04-02 11:37:00	No	<input type="checkbox"/>
B42 Male	Stereo	Audio Processor - De-Esser	2015-04-02 11:30:21	No	<input type="checkbox"/>
Band Pass	Stereo	Audio Processor - Input	2014-05-26 11:47:21	No	<input type="checkbox"/>
BNC SRC OFF		Interfaces - AES I/O	2015-04-02 11:58:26	Yes	<input type="checkbox"/>
BNC SRC ON		Interfaces - AES I/O	2015-04-02 11:57:59	No	<input type="checkbox"/>
BNC SRC ON		Interfaces - AES I/O	2015-04-02 11:59:29	Yes	<input type="checkbox"/>
Bus Compression	Stereo	Audio Processor - Dynamics	2014-05-26 13:48:27	No	<input type="checkbox"/>
Bypass		Routing - Routing	2014-05-22 14:55:07	No	<input type="checkbox"/>

You can sort the table by pressing on one of the column headlines. You can qualify your selection by the "Type" selector and / or the "Preset Block", "Linked to Event", "Last Modified" column headlines. The pull down lists allow to reduce the number of presets displayed:

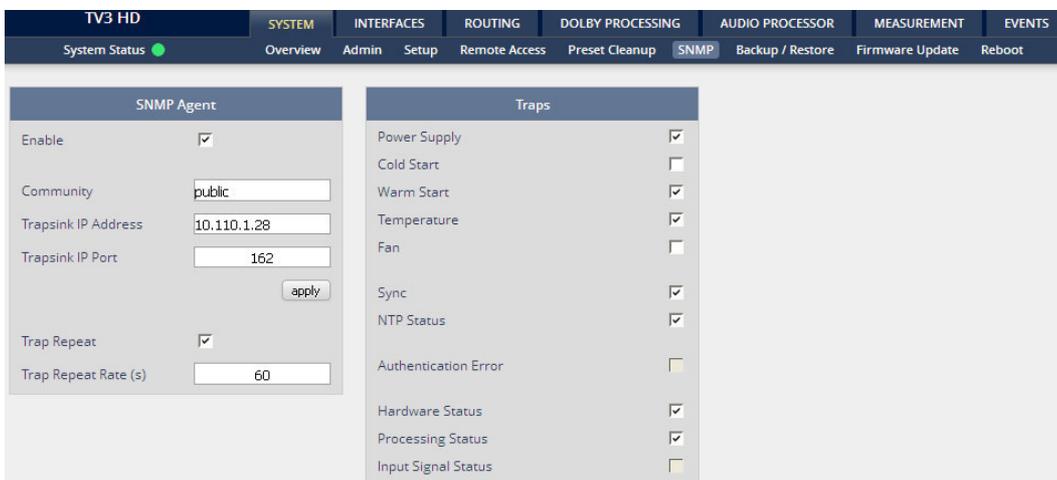


The soft buttons at the bottom left hand side may also be used to search through the table by sorting it by the first letter or leading number. The arrow buttons at the bottom right hand side can be used to scroll through the table if the selection is too big for one page:



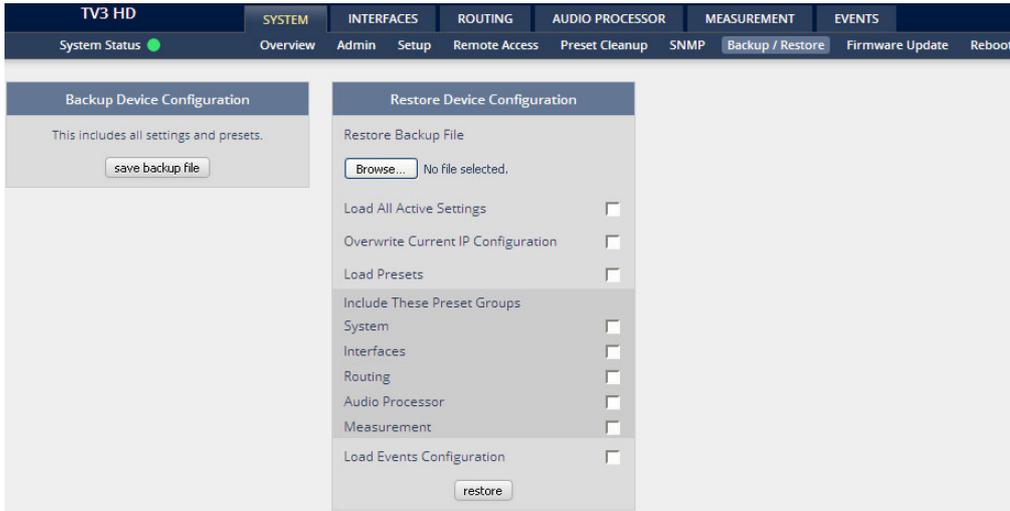
A selection is made by clicking on a line to activate the check box. Once you have made your selection (highlighted lines) you can press the **<delete>** soft button to execute the process. This will remove the selected presets permanently from the device.

Setup GUI – SYSTEM – SNMP

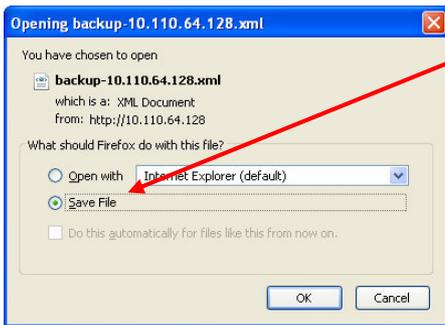


This pane is meant for basic settings of the **SNMP Agent** of the device. If you are not familiar with the use of SNMP protocol for system monitoring you should not enable the SNMP agent.

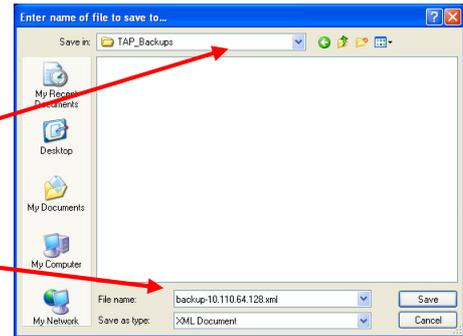
Setup GUI – SYSTEM – Backup / Restore



Here you can **backup** the complete **device** and **restore** parts or all of it .If you press **<backup>** the device controller will collect all necessary data and assemble it to an XML file. Finally you will get a pop up message:



You must select: **<Save File>**.
After pressing **<OK>**, the system file dialog opens:



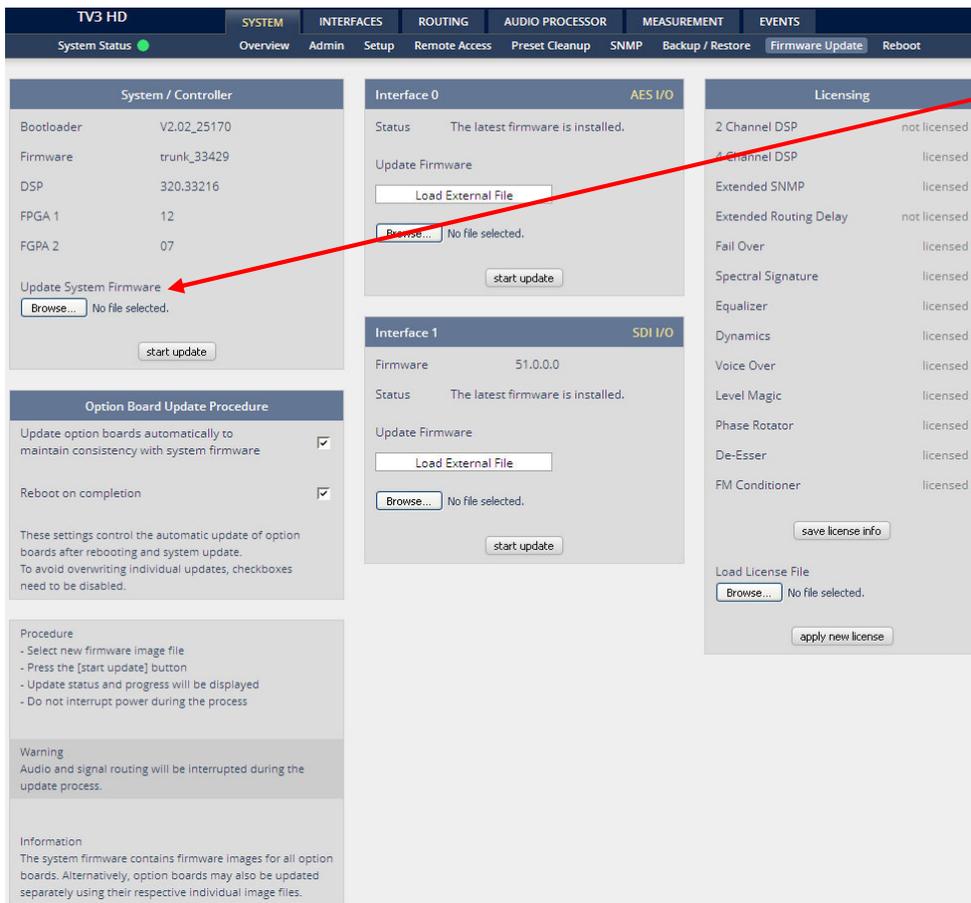
Select a folder and alter that default file name if needed.

Similar applies to the restore process. You must select the desired backup file which you want to restore and check the necessary option(s) under "Restore Device Configuration".

Setup GUI – SYSTEM – Firmware Update

The files to update the **D*AP4** will be available in **ZIP** format. You must unpack them to your PC in order to use them for the update procedure. Here an example path name from the ZIP file:
 junger_dap4_mei_firmware/base_unit_image

The folder /base_unit_image contains an image file for the **D*AP4** core system in the format (example): "rel_dap4-me_i_3_0_2-25852.img". The other folders contain update files for components, like the optional interface boards in the format: "rsdi150_v51.sdi" or for the **X*AP RM1** remote panel in the format: xap_125105.img.



To update the **D*AP4**, you must

<Browse ...> to find the respective firmware file (which you have unzipped before) and press **<start update>**.

After finishing the procedure the device will automatically reboot.

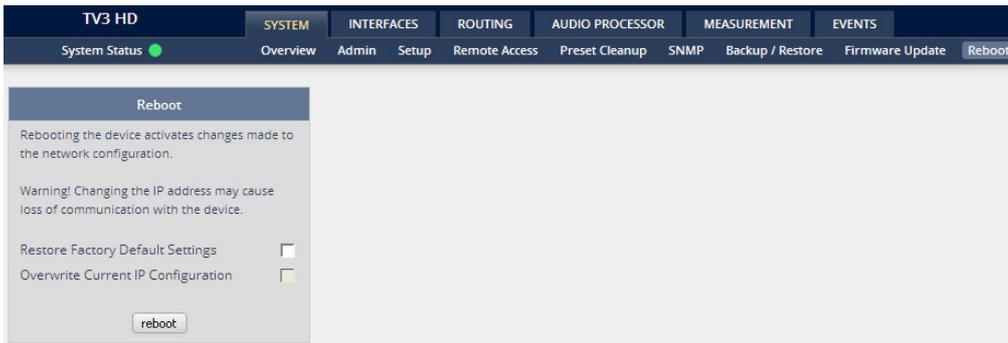
You may also update the firmware of an installed SDI (Interface 1) or AES I/O (Interface 0).

Finally you can see the options of your device which you have bought a license for. When you buy a license you must provide the **license info** file and in return you will get a **new license** file which you must apply to the device here.

Important Note! After the update of the latest firmware image you must observe the **Status** messages displayed in the middle below the firmware version of Interfaces x. If it indicates that you don't have the latest firmware installed you should select the respective file(s) via the drop down box and press the **<start update>** soft button afterwards. But you can also upload an external file in case you need a specialized version for any reason that is not contained in the uploaded firmware image. Same applies to all interface boards.

You must secure the power connection during the update procedure. Especially if you have turned on automatic update of option boards. There is a potential risk to crash the **Dante** board firmware when you lose power during the module update (see interface description how to recover).

Setup GUI – SYSTEM – Reboot



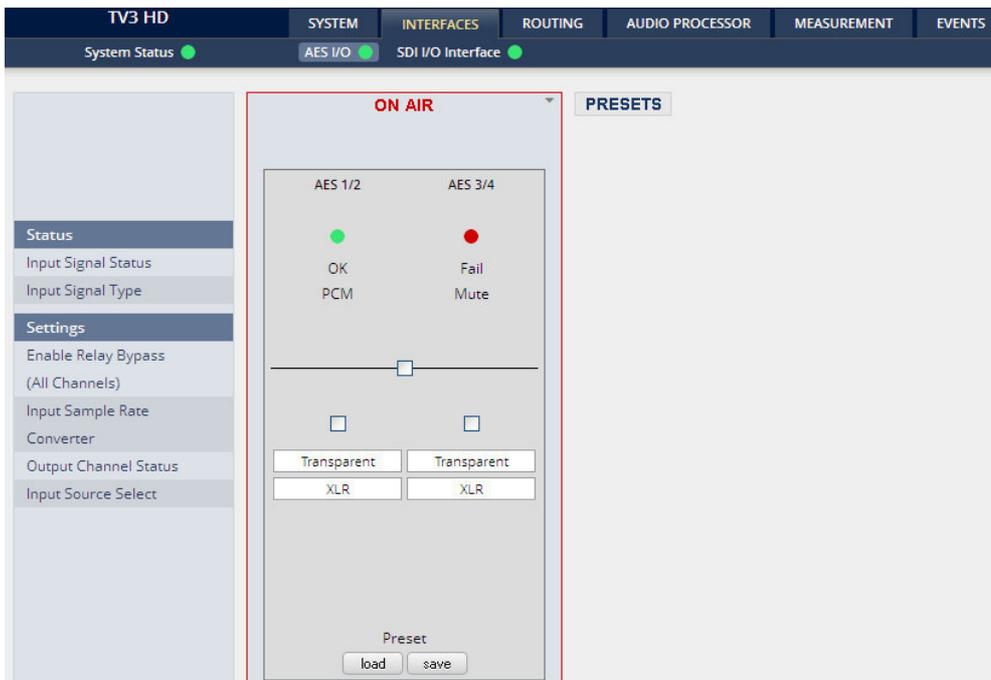
Restore Factory defaults

Will clean up the parameter and preset memory and will initialize all parameters to their factory default values and will reset passwords and turn authentication off.

Overwrite Current IP IP Configuration

You may exclude the current IP settings from this process to keep your existing settings.

Setup GUI – INTERFACES – AES I/O



Status

[green / red / yellow] The soft LED represents the status

Input Signal Status

[OK / Fail]

Fail = no carrier, unlock, cranky [too much jitter]

Input Signal Type

[Mute / PCM / Non PCM]

The **Non PCM** (e.g. Dolby encoded signal) status will be retrieved from a logical combination of the Validity flag and the channel status.

If one of the inputs is not assigned by the ROUTING section, its status will not be incorporated into the System Status (see upper left hand side above).

Settings

Enable Relay Bypass

[ON / OFF]

For fail save operation bypass relays are provided to connect AES IN / OUT in case of a power fail. One may enable such relay manually here.

Input Sample Rate Converter

[ON / OFF]

For asynchronous sources it is possible to turn a **SRC** on. If an **SRC** is turned on and the input status becomes **Non-PCM**, the **SCR** will be turned OFF automatically in order to maintain the original data structure of the encoded bit stream (e.g. Dolby E).

Output Channel Status

[Transparent / Prof PCM / Prof Non-PCM / Cons PCM / Cons Non-PCM]

The channel status can either be transparent from the input source of the **D*AP4** or may be overwritten.

Transparent
Prof PCM
Prof Non-PCM
Cons PCM
Cons Non-PCM
Transparent

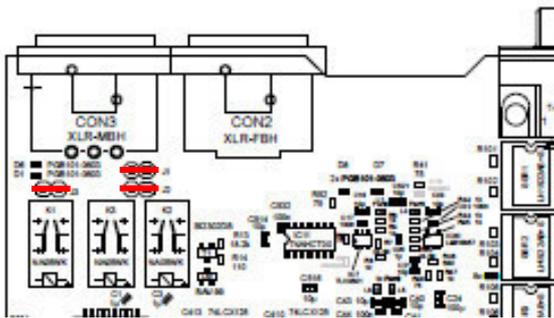
Input Source Select

[BNC / XLR]

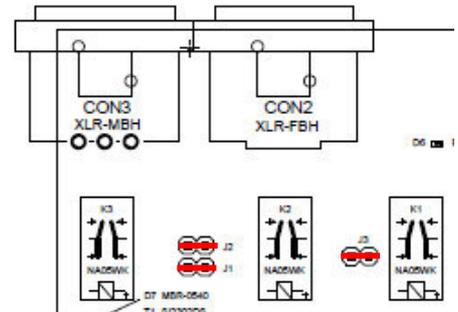
You must select here which input is in use. (AES3id = BNC or AES3 = XLR).

Important note! The AES relay bypass circuit of the AES I/Os may be deactivated inside the **D*AP4**. You must open the cover plate from the **D*AP4** unit and locate the red jumpers shown in the schematic below:

AES 1/2 on the main PCB



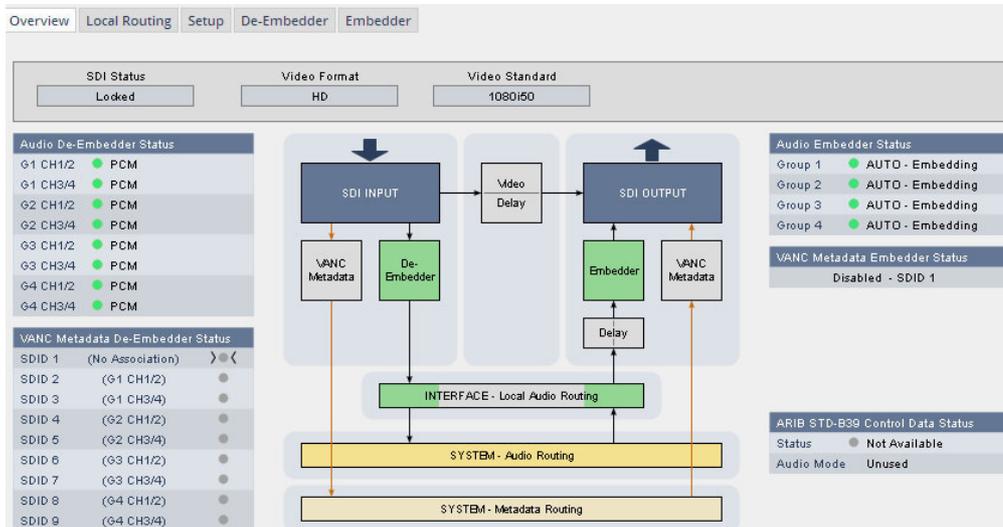
AES 3/4 on interface slot 0



You must remove the jumpers to de-activate the AES I/O relay power fail circuit.

Set up GUI – INTERFACES – SDI I/O interface – **Overview**

If the D*AP4 is equipped with an optional **SDI** interface the following settings will be available. This pane has five sub panes imbedded:

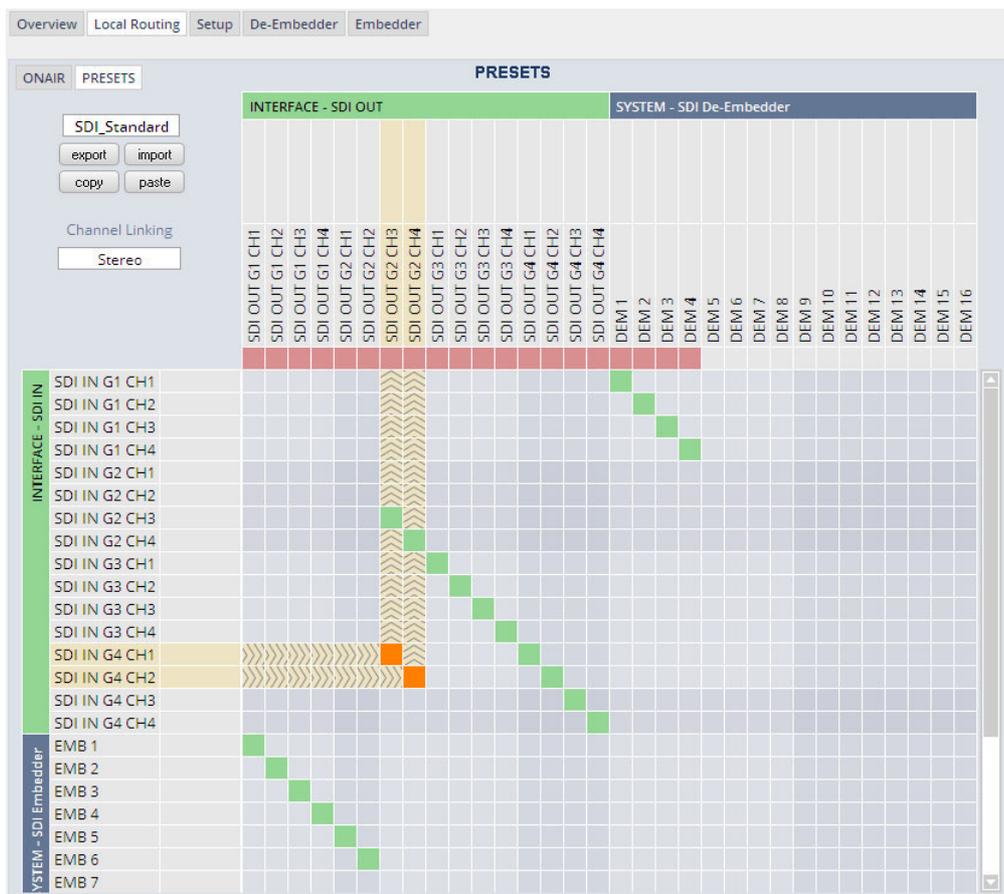


The overview pane shows all relevant information of that interface:

- SDI Status** [Locked / Unlocked]
- Video Format** [SD / HD /3G / N/A]
- Video Standard** [actual decoded standard (e.g. 1080i50) / No SDI Lock]
- Audio De-Embedder Status** [PCM / Dolby E / Dolby Digital / Dolby Digital Plus / MPEG-4 HE AAC / MPEG-4 AAC / N/A]
- VANC Metadata De-Embedder Status** The respective soft LED will turn green to indicate the SDID found in the stream while the angle brackets indicate the SDID selected in the de-embedder set-up as a pre-selected stream.
- Audio Embedder Status** [AUTO – Embedding / AUTO – Replace Audio / OFF / Delete]
 - Group 1 – 4** The embedding process distinguishes between 4 different modes for each group independently:
 - Embedding** – a new group will be built
 - Replace** – the structure of the group from the input is kept and the audio content is simply replaced
 - Delete** – the group from the input is deleted
 - OFF** – the embedder from that group is turned off
- VANC Metadata Embedder Status** [Enabled / Disabled & selected SDID#]
For details see **SMPTE 2020-2** standard.
- ARIB STD-B39 Control Data Status** Meta information standard
 - Status** [Available / Not Available]
 - Audio Mode** See **ARIB** Japanese standard "Structure of Inter-Stationary Control Data Conveyed by Ancillary Data Packets"
http://www.arib.or.jp/english/html/overview/doc/2-STD-B39v1_2.pdf

Set up GUI – INTERFACES – SDI I/O interface – Local Routing

The SDI interface comes with a local routing matrix to shuffle audio signals from and to the system (device) (i.e. to and from the central device router) and from and to the physical de-embedders / embedders. The example below shows an example routing that sends signals 1:1 from the physical de-embedders [INTERFACE – SDI IN G2 CH3 ... SDI IN G4 CH4] to the physical embedders [SDI OUT G2 CH3 ... SDI OUT G4 CH4]. The signals from the physical de-embedders [SDI IN G1 CH1 ... SDI IN G1 CH4] are sent to the device router [DEM 1 ... DEM4] while the device router outputs [SYSTEM – SDI Embedder EMB 1 ... EMB 6] are routed to the first 6 SDI channels [SDI OUT G1 CH1 ... SDI OUT G2 CH2]:



You must use the scroll bar to navigate through the matrix. In the upper left hand corner you can select between the **ONAIR** and the **PRESETS** view of the matrix. On the **ON AIR** page you will also see the device signal labels (see ROUTING section further below for details).

Channel Linking

[mono / stereo]

You can decide if the routing must be performed in mono or stereo mode (where adjacent odd/even channels are routed at once).

You may select cross points by hovering with the mouse over the little squares and select / deselect cross points with a left mouse button click.

Mouse over

Color codes of cross points:

dark blue

Possible new cross point.

orange

You are about to reconnect a cross point.

grey

Cross point is not allowed (i.e. routing will cause a loop and will not therefore be performed) or dedicated input is not activated.

red

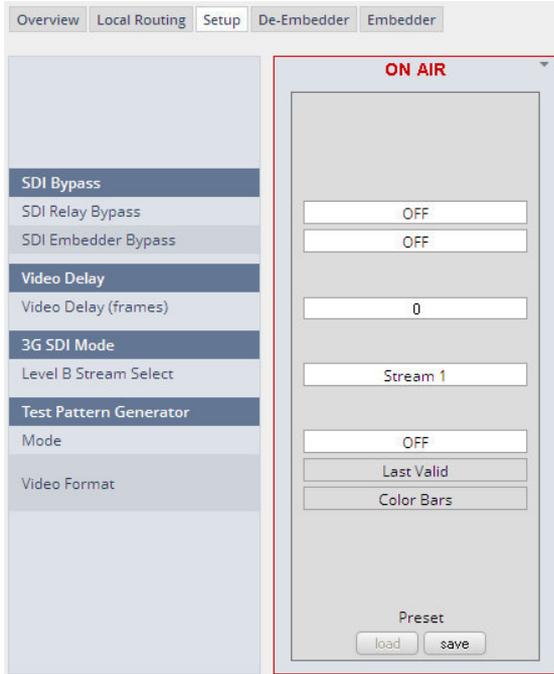
You are about to disable a cross point.

An animated signal flow



will help you when navigating through the matrix.

Set up GUI – INTERFACES – SDI I/O interface – Setup



SDI Bypass

SDI Relay Bypass Will deactivate the **Bypass Relay**. It provides a shortcut from **SDI-IN** to **SDI-OUT1** and disconnects the de-embedder from the SDI input. This relay also serves as a **fail bypass** if the power is off. This feature maintains the SDI signal for downstream equipment.

SDI Embedder Bypass Will pass the embedded audio data from the de-embedder to the embedder 1:1. This function preserves the original ancillary data structure.

Video Delay

Video Delay (frames) [0 ... 15]
For compensation of any kind of audio processing delay within the chain of devices you may use a **Video Delay**. Position “0” turns off the delay function.

3G SDI Mode

Level B Stream Select

A 3G-SDI signal may have two HD sub streams (e.g. for 3-D TV), AKN as 3G-B standard select between stream 1 or 2 for embedded audio. See SMPTE 425M for details.

Test Pattern Generator

The interface offers a test generator to either check downstream connections during installation or for use in case of an input fail but you may also use it to move 16 independent audio channels over a single coax cable from point to point.

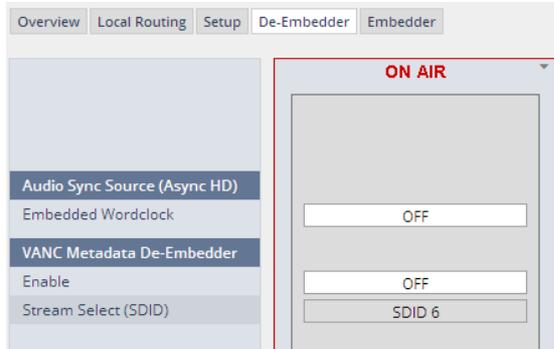
Mode

[OFF / AUTO (Input Loss) / Always ON]

Video Format

[Last valid / one of the defined SD / HD 3G formats (see specs)]
[Color Bars / Black Frame]

Set up GUI – INTERFACES – SDI I/O interface – De-Embedder



Audio Sync Source (Async HD)

The HD SDI standard allows for asynchronous audio. This is critical if you have decided to synchronize the device on such signal. Here you find a solution. You may either use the embedded word clock or the SDI carrier itself as a reference.

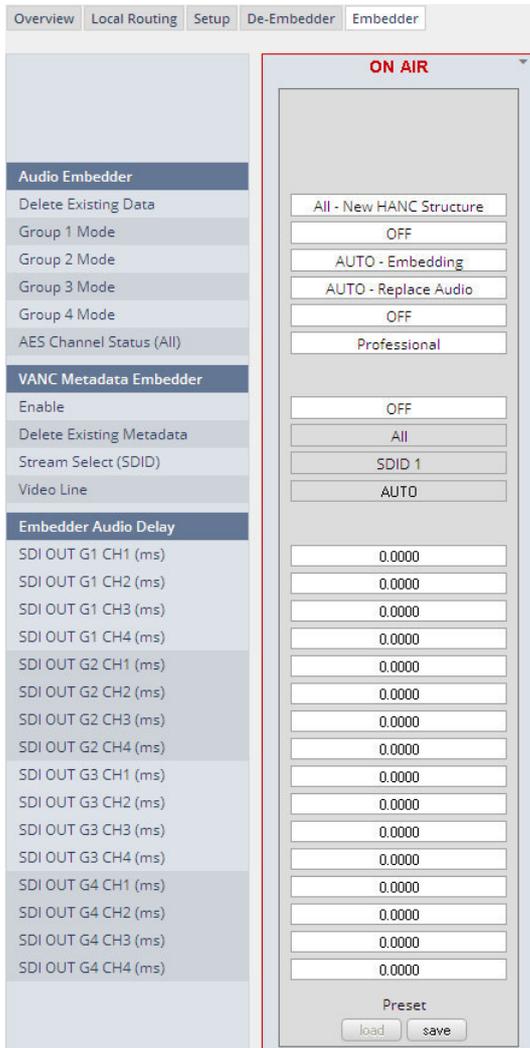
Embedded Word Clock

[Auto / De-Embedder CH1 (DEM 1) / OFF]
OFF = synchronized to the SDI carrier.

Auto = In case of asynchronous audio it is synchronized automatically to the SDI carrier.

DEM1= From de-embedder group 1 channel 1.

Set up GUI – INTERFACES – SDI I/O interface – Embedder



Audio Embedder

Here you set the general functions of the embedder

Delete Existing Data [ALL – New HANC Structure / OFF]

Group 1 – 4 Mode [OFF / AUTO – Embedding / AUTO – Replace Audio / Delete]

See SDI I/O Interface > Overview for details.

AES Channel Status

[Transparent / Professional]
If Professional these values are used:

Format: Professional
Audio Mode: [Audio / Non Audio]
Emphasis: None
Freq. Mode: Locked
Sample Freq.: 48kHz
Channel Mode: Not Indicated
User Bits: None
Auxiliary Bits: 24Bit
Audio Word Length: Not indicated

Important note! If you generate a new AES channel status the **Audio Mode** will be automatically set to **Non Audio** (AKA "other") for both channels, if for example an adjacent pair (1/2, 3/4) carries a Dolby E stream.

VANC Metadata Embedder

The embedder can insert one Dolby metadata stream into the Vertical Ancillary Data

Enable

[ON / OFF]

Delete Existing Metadata

[All / OFF]

Stream Select (SDID)

[SDID 1 ... SDID 9]

Video Line

[Auto / 9 ... 44]

The line number depends on the actual video standard how many VANC lines are available for data insertion.

Embedder Audio Delay

Each embedder signal may be delayed independently. This may be useful for Lips Sync alignment if a video delay is used.

Important Note! You must take care that for Dolby encoded signals the adjacent pairs are set to the same delay values in order not to destroy the data structure.

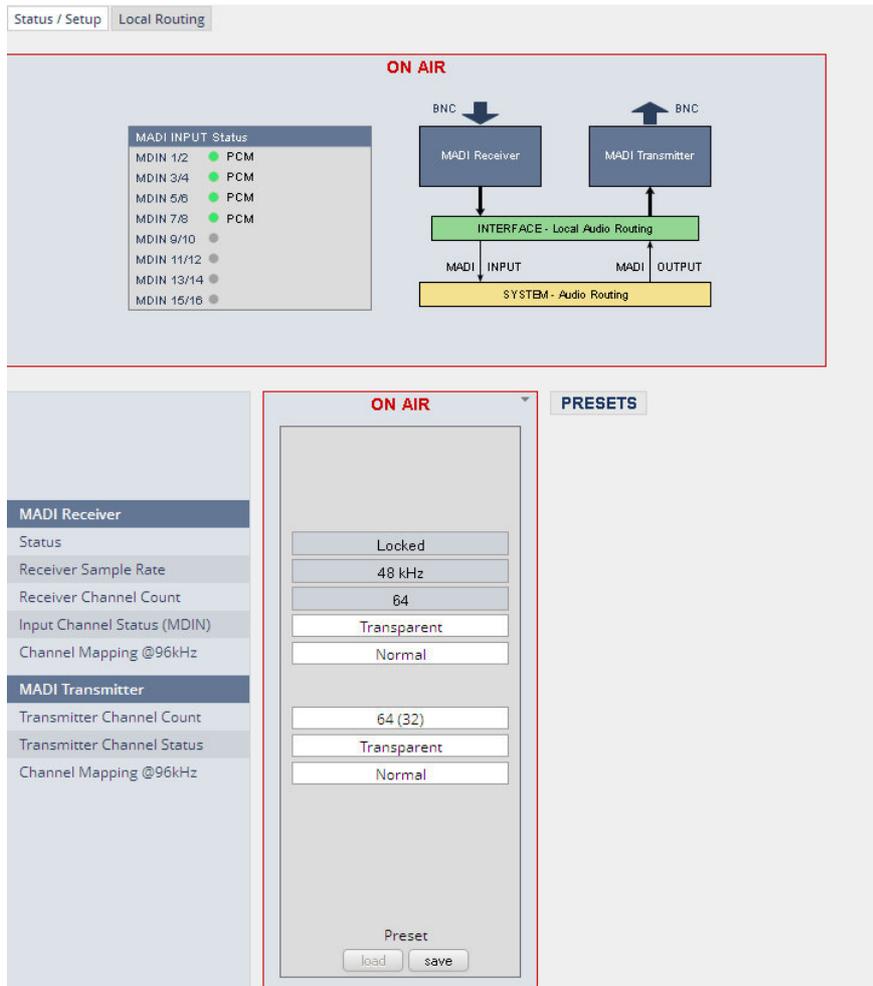
SDI OUT G1 CH1 (ms) [0.0000 ... 340.000]

to

SDI OUT G4 CH16 (ms) [0.0000 ... 340.000]

Set up GUI – INTERFACES – MADI Interface – **Status / Setup**

The implementation of MADI for the **D*AP4** is based on the option module O_DAP_MB (BNC) or O_DAP_MO_MM (MADI optical multi mode fiber) or O_DAP_MO_SM (MADI optical single mode fiber). Since the V*AP is a 4 channel processing device not all 64 MADI channels are available for device I/O. The first 8 channels are available via the MADI local router to the device router. They appear at the device router pane as MDIN 1 .. 8 and MDOUT 1 ... 8. These channels can be routed to and from any of the local routing sources MADIRX 1 ... 64 and MADITX 1 ... 64 respectively.



MADI Receiver

- Status** [Locked / Locked-Async / Error]
The timing of the audio decoding is locked to the MADI clock. If the internal timing of the D*AP4 is different "Locked-Async" is displayed.
- Receiver Sample Rate** [44.1 / 32 / 48 / 88.2 / 96kHz / Unknown]
The measured sample rate from the received MADI stream.
- Receiver Channel Count** [32 / 56 / 64]
Depends on the upstream MADI transmitter settings.
- Input Channel Status (MDIN)** [Transparent / Professional]
One may overwrite the input channel status by a set of professional ones.
- Channel Mapping @ 96 kHz** [Normal]

MADI Transmitter

Transmitter Channel Count [64 (32) / 56 (28)]
 Depends on the internal sample rate and the desired number of MADI channels. The numbers in brackets are valid for 96kHz.

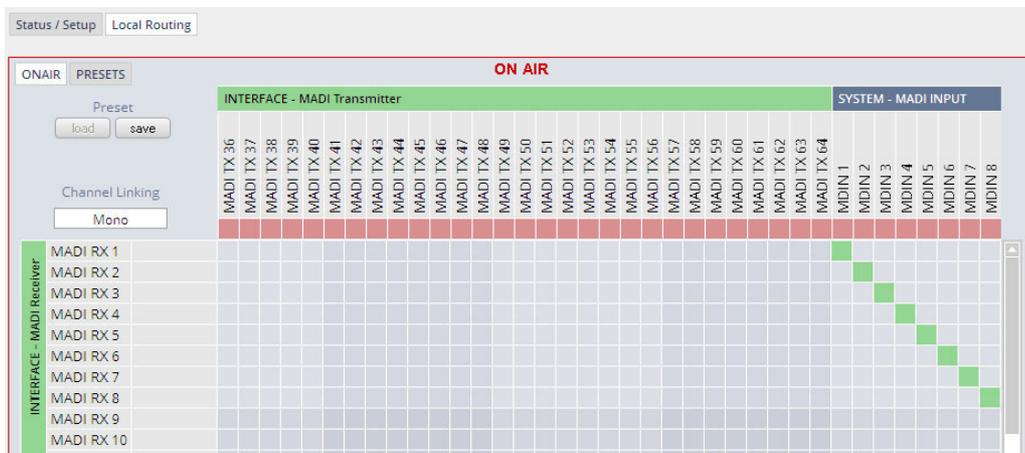
Transmitter Channel Status [Transparent / Professional]

Channel Mapping @ 96 kHz [Normal]

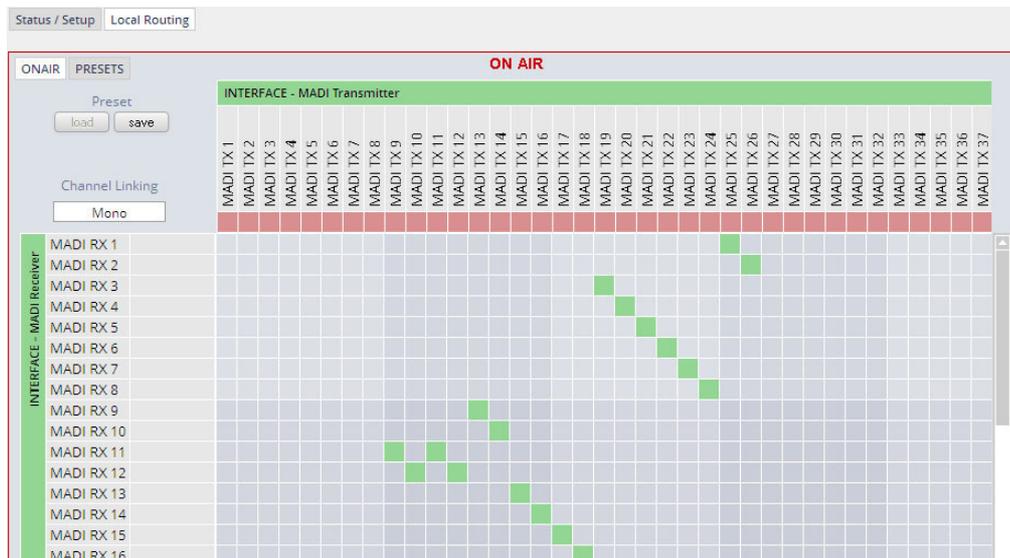
Set up GUI – INTERFACES – MADI Interface – Local Routing

Below are some excerpts from the local routing pane. Single channels from or to the **D*AP** may be connected with the MADI transmitter or MADI receiver respectively.

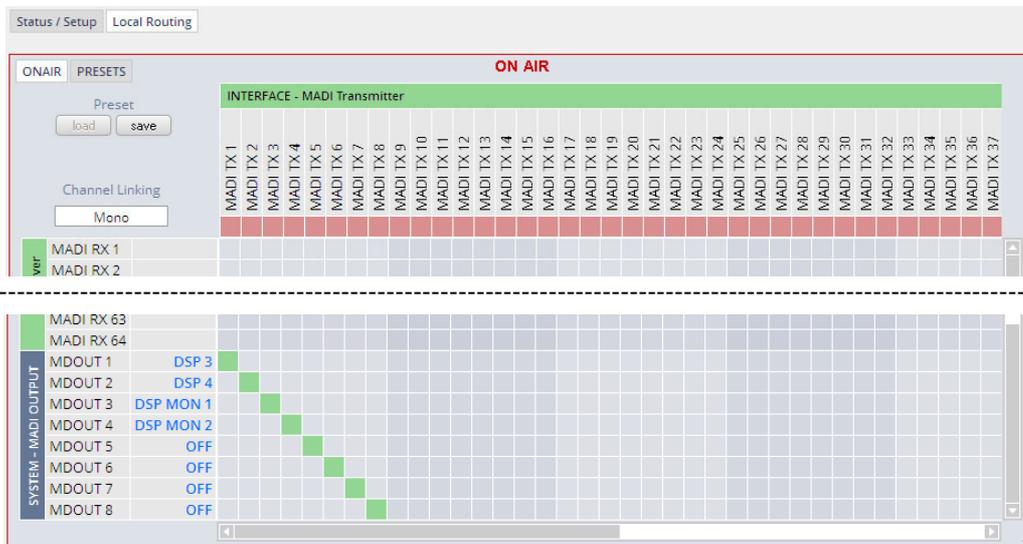
The example below shows the first 8 MADI channels from the receiver (MADI RX 1 ... MADI RX 8) connected with the device inputs SYSTEM MADI INPUT (MDIN 1 ... MDIN 8):



The **Local Routing** pane can also be used to route MADI signals from the receiver directly to the transmitter and vice versa:



You can also assign device outputs (MAOUT 1 ... MDOUT 8) to MADI transmitter channels (MADITX 1 ... MADITX 8). For better visibility the matrix has been divided by cutting off the middle part:



Set up GUI – INTERFACES – Dante I/O Interface – **Status**

The **Dante** interface connects a **D*AP8** to an audio over IP (AoIP) network. Junger Audio has committed itself to the quasi industry standard **Dante** developed by the company **Audinate**:



"Based on industry standards, Audinate created **Dante**, an uncompressed, multi-channel digital media networking technology, with near-zero latency and synchronization ... One cable does it all.

Dante does away with heavy, expensive analog or multicore cabling, replacing it with low-cost, easily-available CAT5e, CAT6, or fiber optic cable for a simple, lightweight, and economical solution. **Dante** integrates media and control for your entire system over a single, standard IP network."

The network infrastructure for **Audio over IP** must be able to handle the **IP multicast**. So it needs a bit of care when it comes to network gear. The recommendation is to separate the control network from the audio network.

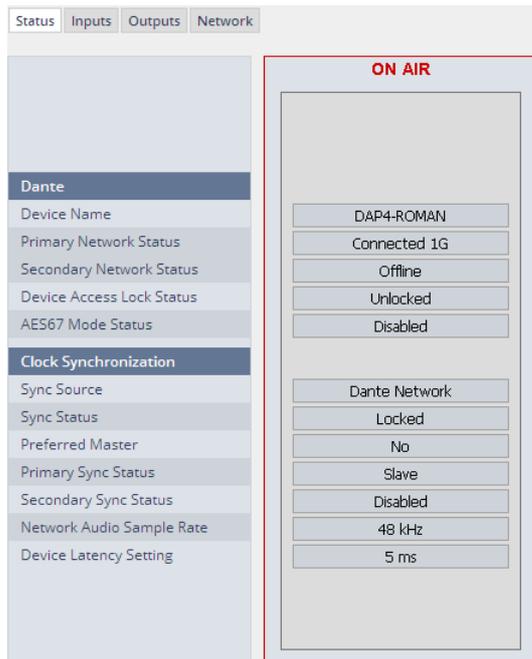
For details pls. refer to the Audinate web-site: <https://www.audinate.com>. Here you will find many useful application videos and FAQs.

To configure such an audio network you need the **DanteController** software. You can download it from the **Audinate** web site. People who want to interface a PC or MAC to such an audio network can use the **VirtualSoundcard** or even more sophisticated the **Via**, an applications software from **Audinate**. The **Virtual Sound Card** provides audio drivers to connect with common audio tools while **Via** allows you to connect network audio resources with PC audio resources like analog line / Mic / USB-Audio / even applications (Skype, youtube you name it) directly.

We highly recommend to read the **Audinate** documents to understand how to set-up and operate a real-time **AoIP** network.

Looking at the rear panel the RJ45 connector on the left is the primary port while the second connector acts either as a redundant or as a switch port. Both RJ45s have built in LEDs. The left one shows network activities (flashing green) while the right one indicates the interface speed, with **green=1Gbit/s** and **off=100MBit/s**.

Below is the Status page of the **DANTE** interface board:



Dante

Device Name The name you gave the interface board via the **DanteController**:
Device > Device View > Device Config

Primary Network Status [Offline / Connected + bandwidth]

Secondary Network Status [Offline / Connected + bandwidth]

Device Access Lock Status [Unlocked / Locked]
See Dante Controller

AES67 Mode Status [Disabled / Enabled]
See Dante Controller to enable it.

Clock Synchronization

Sync Source [Dante Network / DA*P is Master]
Here you define the reference clock for this **Dante** module.

Important Note! If this parameter is set to "Dante Network", the **D*AP4** must be synchronized to the same clock as the network clock master (whoever it is). It **must** be set to "Dante Network" if this module is to become the "Preferred Master" of the network.

Sync Status [Unlocked / Locked / Locked-Async]

The sync source for the **Dante** interface is the **Dante** network. If no network cable is connected the interface is "Unlocked". If it is connected to a network it will be "Locked". If the **D*AP4** is set to synchronize to other than the **Dante** interface it will show "Locked-Async".

Preferred Master [No / Yes]

The **Dante** algorithm automatically looks for the best clock master inside the network but one may force a **Dante** module to become the clock master.

Primary Sync Status [Slave / Master]

Network Audio Sample Rate [44.1 kHz / 48 kHz / 88.2 kHz / 96 kHz]

Depending on the A*P device type the sample rate is limited to the device specification.

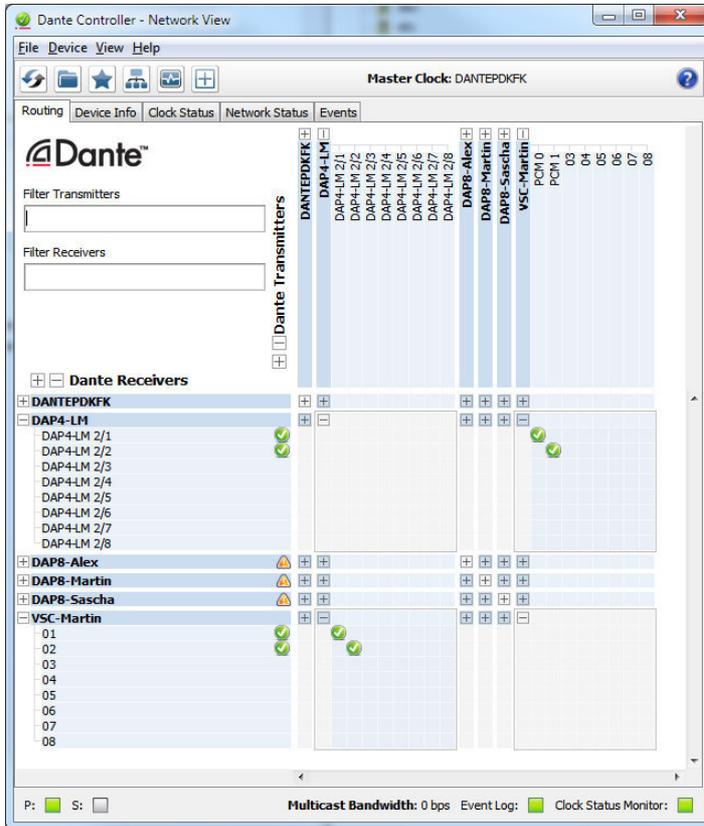
Device Latency Setting [5ms]

You can allow for a certain transmission latency if you face network problems of any kind.

Set up GUI – INTERFACES – Dante I/O Interface – Inputs

The **DanteController** software gives you an overview of all members of such a **DANTE** network. You can assign channel labels for the inputs (from the network to the device interface). Those labels will automatically appear in the **D*AP4** and will be displayed there.

Here is a glimpse on the GUI of the **DanteController**:



As an example you see here a "DAP4-LM" (name given by the Dante Controller) that has assigned the labels DAP-4 2/1 ... 2/8 for both the inputs and the outputs.

Beside a few more devices on that network, we see the unfolded outputs of a **DanteVirtualSoundcard** (VSC) named "VSC-MARTIN" on the upper right hand side.

The top horizontal area shows the transmitters while the receivers are shown vertically on the left hand side.

The outputs PCM 0 and PCM 1 from the VCS are assigned to the **D*AP4** inputs DAP4-LM 2/1 and 2/2 while two outputs from the "DAP4-LM" are assigned to the VSC inputs "01" and "02".

We see the labels assigned by the DanteController software in the "Channel" column:

Status Inputs Outputs Network			
Inputs	Channel	Connected	Status
DTIN 1 ● PCM	DAP4-LM 2/1	PCM 0 @ VSC-Martin	Connected (Unicast)
DTIN 2 ● PCM	DAP4-LM 2/2	PCM 1 @ VSC-Martin	Connected (Unicast)
DTIN 3 ● PCM	DAP4-LM 2/3	no subscription	No Subscription
DTIN 4 ● PCM	DAP4-LM 2/4	no subscription	No Subscription
DTIN 5 ● PCM	DAP4-LM 2/5	no subscription	No Subscription
DTIN 6 ● PCM	DAP4-LM 2/6	no subscription	No Subscription
DTIN 7 ● PCM	DAP4-LM 2/7	no subscription	No Subscription
DTIN 8 ● PCM	DAP4-LM 2/8	no subscription	No Subscription

Inputs

Eight inputs are pre-defined for the **DANTE** interface installed in a **D*AP4**. They are organized in pairs and the input status is shown by soft LEDs (green = PCM audio / yellow = non audio/ grey no audio).

Channel The labels assigned to that channel by the **DanteController**

Connected The source of the audio signal.

Status [No Subscription / Subscription Unresolved / Wait / Naming Problem / Loopback / Idle / Subscription in Progress / Connected (Unicast) / Connected (Multicast) / Manual Config / Format Problem / QoS Problem / Latency Problem / Clock Domain Problem / Link Down / Fail / Unknown]
The DANTE module provides very detailed status information. In regular operation one will not see much of it.

Set up GUI – INTERFACES – Dante I/O Interface – **Outputs**

Outputs	Channel	Channel Label
DTOUT 1	01	DAP4-LM 2/1
DTOUT 2	02	DAP4-LM 2/2
DTOUT 3	03	DAP4-LM 2/3
DTOUT 4	04	DAP4-LM 2/4
DTOUT 5	05	DAP4-LM 2/5
DTOUT 6	06	DAP4-LM 2/6
DTOUT 7	07	DAP4-LM 2/7
DTOUT 8	08	DAP4-LM 2/8

Outputs The signals from the **DANTE** board to the network. They will also appear in the device **ROUTING** section.

Channel Numeric count of the channels.

Channel Label Up to 16 labels can be configured for each stream from the interface to the network. This allows configuring multi layer routing.

Set up GUI – INTERFACES – Dante I/O Interface – **Network**

Dante Redundancy The DANTE interface allows redundant network operation. Pls. refer to manufacturer's documentations of your Ethernet equipment on supported switching configuration and redundant operation.

Mode	[Switched / Redundant]
Redundant	– The interface will duplicate the audio traffic to both Ethernet ports.
Switched	– The second port behaves like a standard switch port allowing daisy-chaining through the interface. I.e. IP configuration is only available for Redundant mode.

Important Note! When set to switched mode, do **not** connect both ports to the same network (same Ethernet switch) if it does not support STP (Spanning Tree Protocol). This is the case for most of the off-the-shelf (office) switches. Doing so will cause a race condition where IP packets are circling around from the external switch to the second **Dante** (switch) port and back via the first port. This will tear down your network and may create a bunch of new "friends" in your facility.

Primary Address Setup	Setup of the primary network interface
Network Status	[Offline / Connected + bandwidth]
DHCP – Automatic IP Config.	[OFF / ON]
IP-Address	
Netmask	
DNS Server	
Gateway	
MAC Address	
Secondary Address Setup	Setup of the secondary network interface
Network Status	[Offline / Connected + bandwidth]
DHCP – Automatic IP Config.	[OFF / ON]
IP-Address	
Netmask	
DNS Server	
Gateway	
MAC Address	[unknown / address]

Important Note! It may happen by accident that the update of the Dante module fails. E.g. if the firmware update option: SYSTEM > Firmware Update > Option Board Update is set to "Update option boards automatically" and the device loses power during this process, the Dante module will be in the fail-save state. This is indicated in the Dante Controller software.

In this case you must repair it by aid of a Dante tool. You can download it from the website:

<https://www.audinate.com/content/dante-firmware-update-manager-v31009-windows>

Pls. keep in mind that the PC, that runs the Dante update manager must be in the Dante network (if you have separated the networks as recommended) and not in the device control network.

The update manager performs two tasks, the recovery from the fail-safe state and the update of an valid Junger basic firmware for the Dante module.

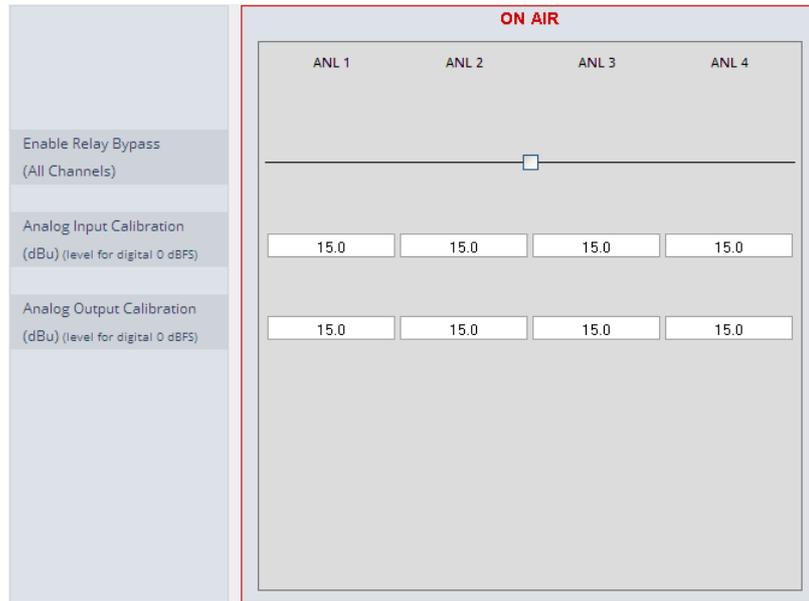
After you have managed to recover from fail-safe you must power cycle the **D*AP8** and update the module manually to the latest Junger firmware using the Dante update manager. The file is part of the zip file that you can download from the Junger web-site.

You will find the Junger recovery firmware here (version numbers are examples only):

rel_dap4_mei_4_0_1.zip > junger_dap4_mei_firmware > Dante_recovery_image > DT-100-v1.0.3-7.dnt

Set up GUI – INTERFACES – 4 Ch ANALOG I/O Interface

An additional analog interface can be installed in the **Interface** slot.
It provides 4 additional analog line inputs and outputs on a 25pin D-Sub connector:



Enable Relay Bypass
(All Channels)

[ON / OFF]

Power fail bypass relay that may be activated from the GUI

Analog Input Calibration
(dBu) (level for digital 0dBFS)

[0 ... 15.0 ... 24.0]

A/D conversion parameter. It defines the analog input level in dBu to reach a digital full scale signal.

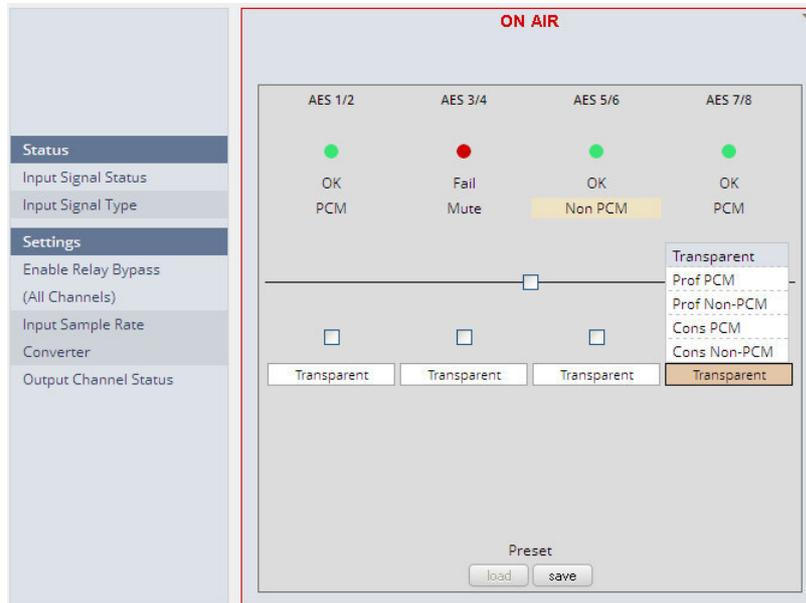
Analog Output Calibration
(dBu) (level for digital 0dBFS)

[0 ... 15.0 ... 24]

D/A conversion parameter. It defines the analog output level in dBu for a digital full scale signal.

Set up GUI – INTERFACES – AES Interface – **Status / Setup**

An additional AES3 interface can be installed in the **Interface** slot. It provides 4 additional AES3 inputs and outputs on a 25pin D-Sub connector:



Status

Input Signal Status green [OK] / red [Fail]

Input Signal Type [Mute / PCM / Non PCM]

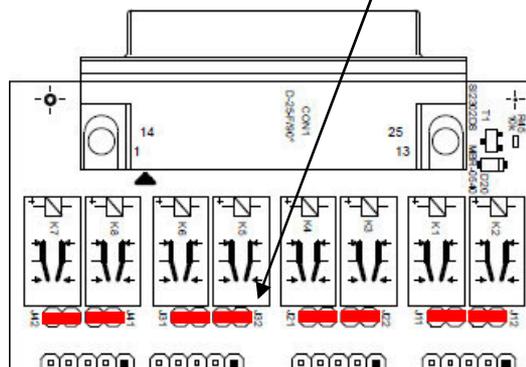
Settings

Enable Relay Bypass (All Channels) [ON / OFF]
Power fail bypass relay that may be activated from the GUI

Input Sample Rate Converter [ON / OFF]

Output Channel Status [Transparent / Prof PCM / Prof Non-PCM / Cons PCM / Cons Non-PCM]
Controls the channel status for the AES output. It provides a set of useful channel status information (e.g. to prevent non audio signals to be fed to speakers).

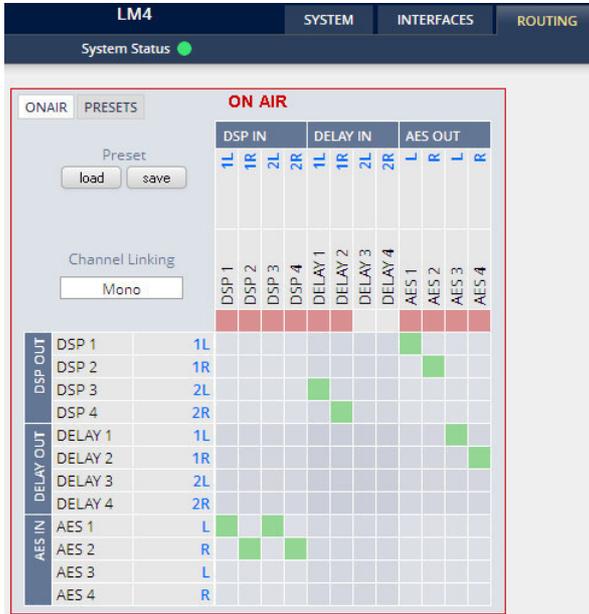
Important note! The AES relay bypass circuit of the I/Os is activated on the option board. It is possible to deactivate it if necessary. You must open the cover plate from the **D*AP4** unit and locate the jumper shown in the schematic below. You must remove the jumpers ● to de-activate the AES I/O relay power fail circuit.



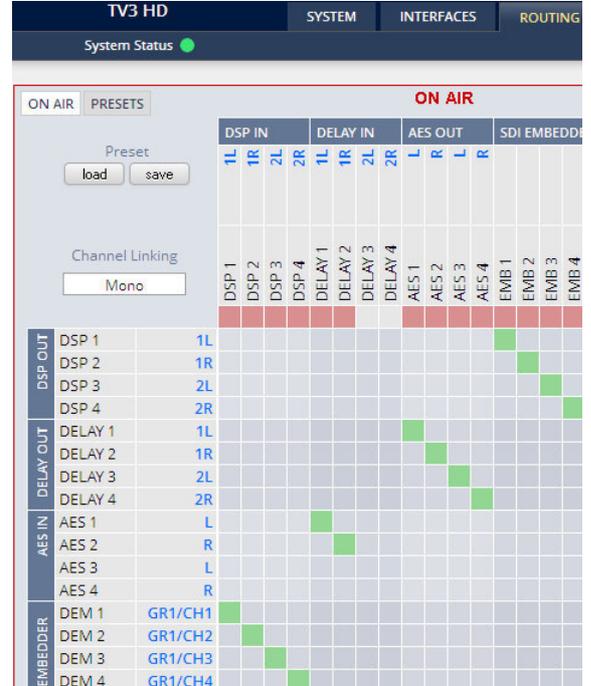
Setup GUI – ROUTING

This is the core of the **D*AP4** as it defines the audio signal flow inside the device. It appears differently depending on the type of optional interface boards installed.

Example without an interface board:



example for an SDI interface board:



Each functional block of the device has a source label and a destination label. Additional **blue** signal labels give an indication of the type of signal that is expected or issued by the respective function block or I/O interface (e.g. **1L/1R** for AES or **2L/2R** for DSP 3/4 and so forth).

Top / horizontal (column headlines) = signal destinations

- DSP** [DSP 1 ... DSP 4]
The DSP inputs carrying the signal type labels **1L, 1R, 2L, 2R** of the two programs 1/2 (see AUDIO PROCESSOR > Overview).
- DELAY** [DELAY 1 ... 4]
The four inputs of the extra delay lines of the D*AP4
- AES** [AES 1 ... 4]
The AES outputs of the device.

Left hand / vertical (line headlines) = signal sources

- DSP** [DSP 1 ... DSP4]
The DSP outputs carrying the signal type labels **1L, 1R, 2L, 2R**
- DELAY** [DELAY 1 ... 4]
The four outputs of the extra delay lines of the **D*AP4**
- AES** [AES 1 / AES2]
The AES inputs of the **D*AP4**.

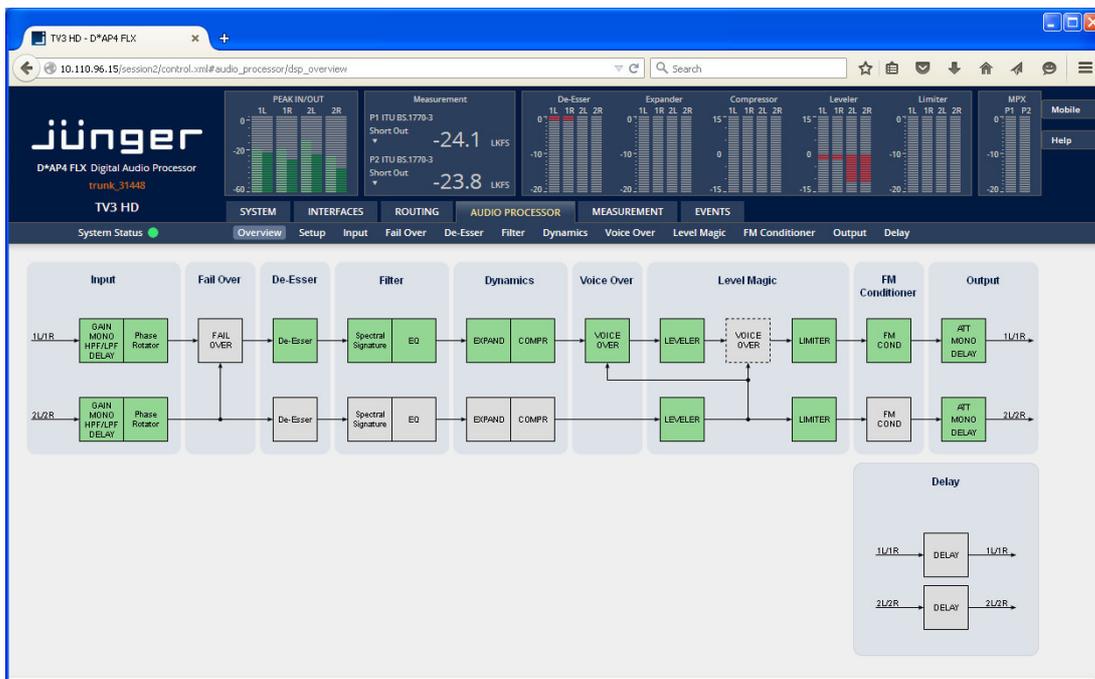
The routing example on the left hand side shows that the AES inputs 1,2 are connected to the inputs DSP 1,2 (label **1L/1R**) of the DSP. The outputs DSP 1,2 (label **1L/1R**) are connected with the inputs DELAY 1,2 and with the outputs AES 1,2 (label **L, R**). The outputs DELAY 1,2 finally are connected with the outputs AES 3,4 (label **L,R**).

Important Note! If a different optional interface board is installed the matrix will be expanded by the pre-defined number of I/Os with their labels:

SDI	[O_DAP_SDI_a]	DEM 1 ... DEM 16 and EMB 1... EMB 16
MADI	[O_DAP_MB_a / O_MO_MM_a / _MS_a]	MDIN 1 ... MDIN 8 and MDOUT 1 ... MDOUT 8
DANTE	[O_DAP_DANTE_a]	DTIN 1 ... DTIN 8 and DTOUT 1 ... DTOUT 8
4 Ch ANALOG I/O	[O_DAP_ADDA_a]	ANL 1 ... ANL 4 and ANL 1 ... ANL 4
AES	[O_DAP_AES_a]	AES 1 ... AES 8 and AES 1 ... AES 8

Setup GUI – AUDIO PROCESSOR - **Overview**

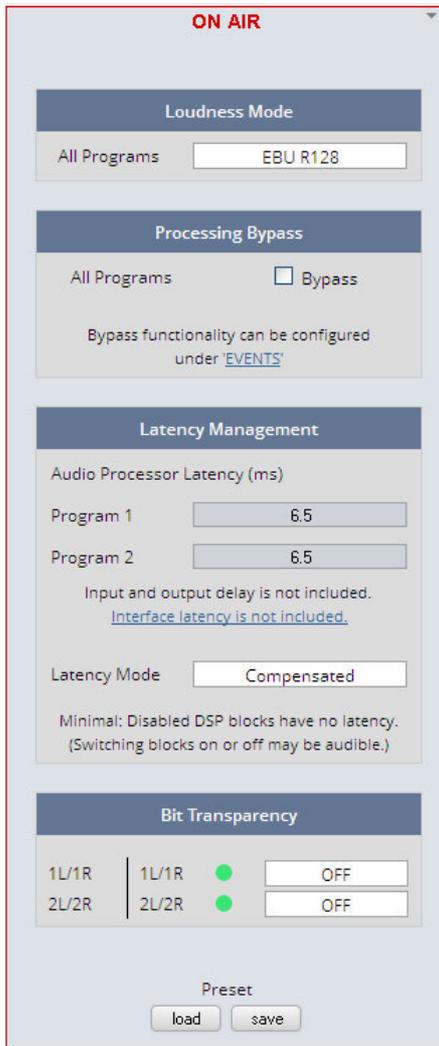
The overview shows the actual signal flow and the audio processor blocks, rendered by the DSPs.



The processing blocks in use, which may be activated from their individual setup panes, will be indicated in green. I.e. blocks shown in grey are not activated by the user.

To navigate through the various processing blocks you may either click on the graphical block or use the tabs provided in the navigation bars below the bar graph displays.

Setup GUI – AUDIO PROCESSOR - Setup



Loudness Mode

EBU R128
Level
ITU BS.1770-1
ITU BS.1770-2
ITU BS.1770-3
EBU R128
ARIB TR-B32
ATSC A/85 (2011)
ATSC A/85 (2013)
Free TV OP-59
Portaria 354

In order to meet the regulations of regions or countries you must select the loudness control mode here. Beside of the weighing curves several measurement duration and loudness ranges have been defined. Some regulations are based on the same measurement (e.g. ITU BS.1770-2) but defined in a different regional norm. You must check with your local authority for correct settings if you must comply with regulations.

Processing Bypass

[ON / OFF]

You may turn the bypass ON/OFF from here by activating the check box. The bypass functionality may be configured at the **EVENTS > Actions** pane where the link will direct you to.

Latency Management

In a latency critical environment it might be desirable to have the lowest possible latency. So it is useful to actually bypass a process that is not in use. In normal operation, switching audio processing modules on and off does not result in a change of latency and thus does not cause audible glitches or clicks.

Program 1

Display of the actual latency

Program 2

Display of the actual latency

Latency Mode

[Minimal / Compensated]

"horizontal" compensation for one program. Disabled audio processing blocks are taken out of the processing

chain and are no longer causing a delay. However switching blocks on or off can cause clicks and glitches, even in unaffected channels, as the latency compensation is recalculated.

Bit Transparency

1L/1R | 1L/1R
2L/2R | 2L/2R

For non audio signals which may appear at the input of a program chain permanently or time by time you can select the behavior here.

[OFF / ON / AUTO]

[OFF / ON / AUTO]

You may force the DSP to pass through the audio stream untouched in case there is encoded audio present. The AUTO mode is triggered by the AES channel status.

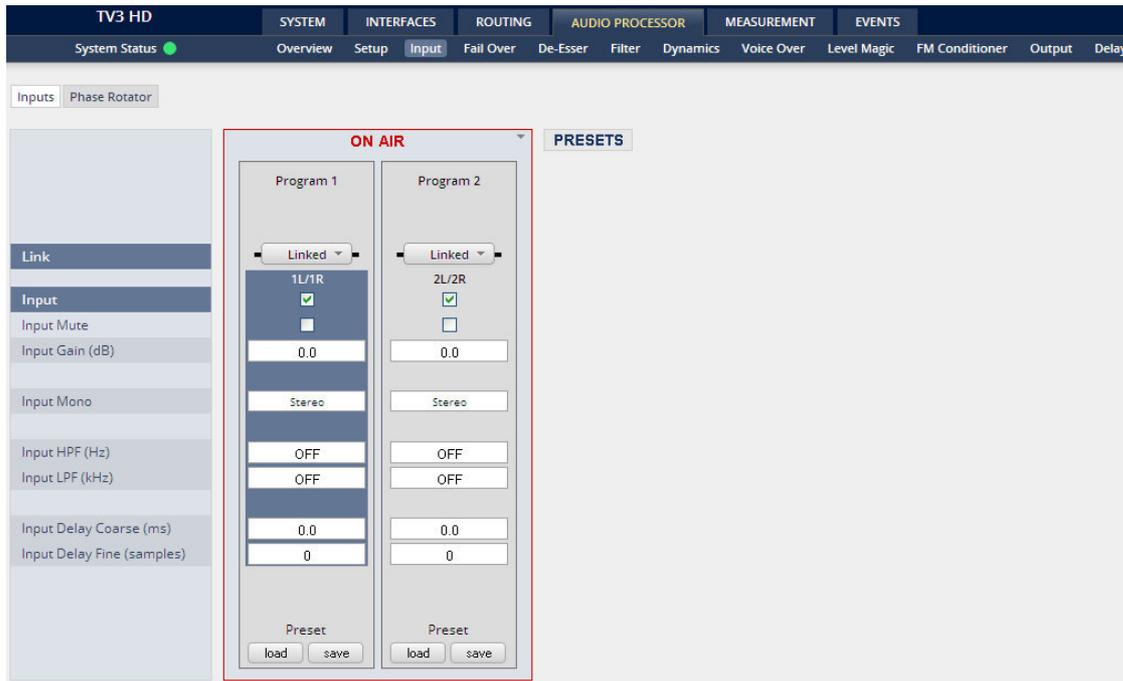
System Latency [ms]:

	44.100kHz	48.000kHz	88.200kHz	96.000kHz
Base Latency				
AES IN to AES OUT	4.26	4.08	3.13	3.05
Additional Latencies				
Spectral Signature	2,9	2,66	2,9	2,66
Sample Rate Converter	0,21	0,21	0,21	0,21

Base latency consists of 1 ms Limiter Look Ahead Time and all system inherent processing and input/output delays.

Activating sample rate converters will add additional latency (< 1ms).

Setup GUI – AUDIO PROCESSOR – Input – Inputs

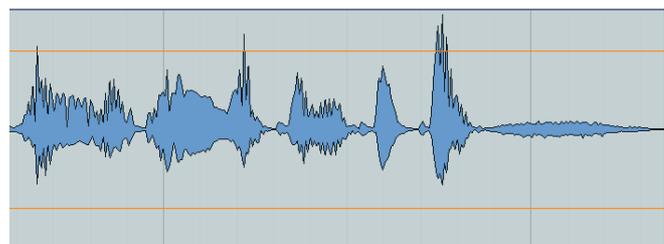


You may set the input conditions for both program channels (1L/1R) and (2L/2R) here:

Link	[Unlinked, Linked] For stereo operation you may link the setup parameters
Input	[Enable / Disable] Enables or disables the input section
Mute	[ON / OFF]
Input Gain (dB)	[-80.0 ... 0.0 ... 20.0]
Mono	[L/R Stereo / L+R Mono / L/L Mono / R/R Mono]
Input HPF (Hz)	[OFF / 20 / 40 / 80 / 120]
Input LPF (kHz)	[OFF / 15 / 20 / 22]
Input Delay Coarse(ms)	[0.0 ... 2000.0]
Input Delay Fine (samples)	[0 ... 2000]

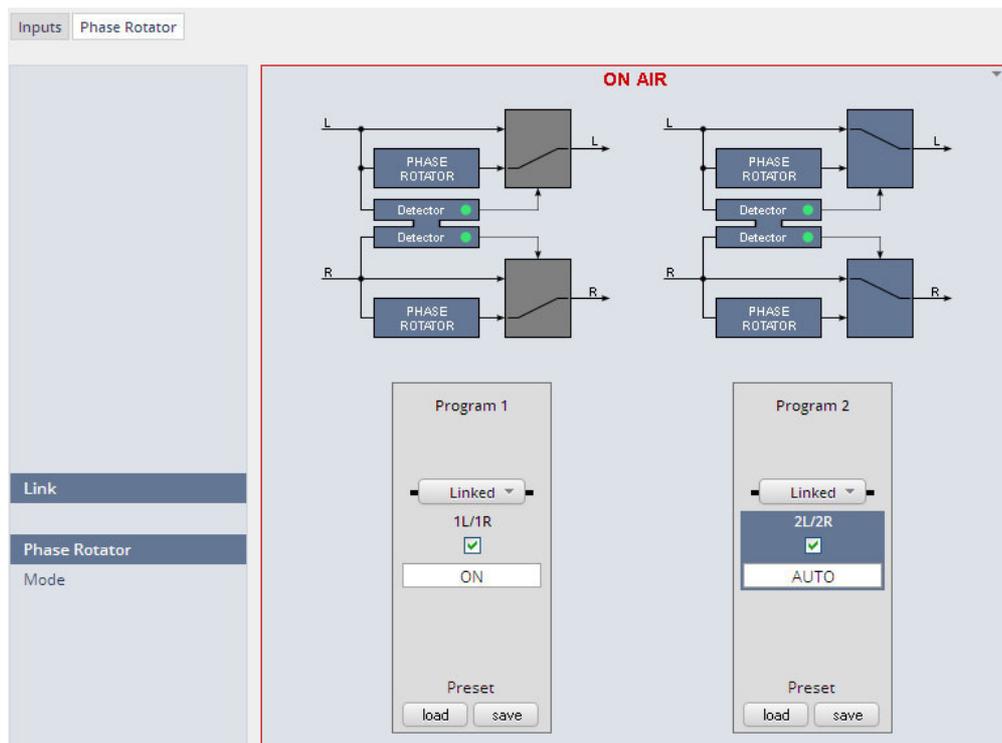
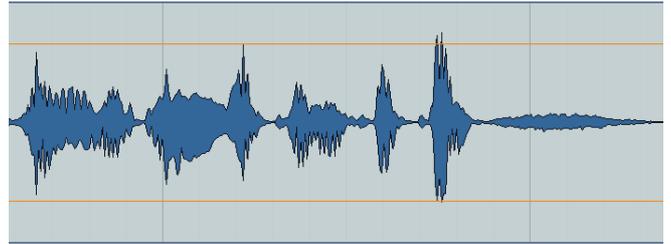
Setup GUI – AUDIO PROCESSOR – Input – Phase Rotator

When working with human voice, one regularly experiences issues with imbalanced waveforms. Imbalanced in this context means that the positive or negative half of the alternating signal carries more power than the other. The problematic result of this type of imbalance is unnecessarily applied dynamics processing (e.g. signal limiting) or loss of headroom.



The Phase Rotator detects this type of imbalance and automatically applies a complex phase wrapping filter to restore symmetry.

Please keep in mind that this system is not effective against DC offset. In this case a high pass filter should be applied instead.



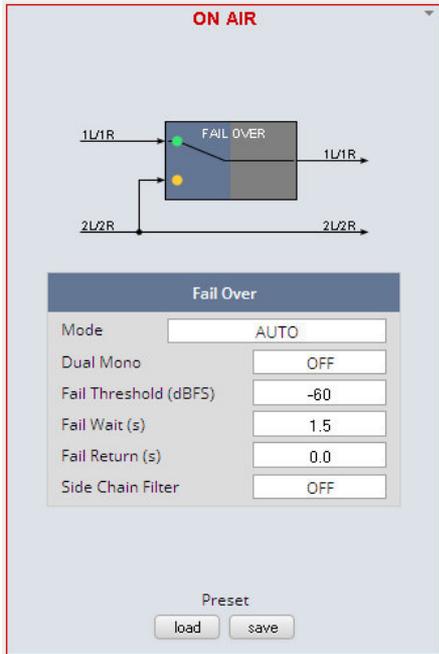
Each program has an independent Phase Rotator.

Link [Unlinked / Linked]
For stereo you may link the setup parameters of both voice channels.

Phase Rotator [ON / OFF]
Mode [OFF / ON / AUTO]

- OFF
System is inactive
- ON
System always applies phase wrapping
- AUTO
Unbalanced waveforms are automatically detected and phase wrapping is applied only if necessary.

Setup GUI – AUDIO PROCESSOR – Fail Over



Fail Over

The D*AP4 offers a fail over circuit for automatic operation. It will switch to 2L/R in case 1L/1R fails.

MODE

[FIX 1L/1R / FIX 2L/2R / AUTO]
In AUTO mode the switch over happens in case of an input failure.

Dual Mono

[OFF / AUTO]
A detector looks after the input signal. If it is a left [L] or right [R] only it converts that signal either to [L/L] or [R/R].

Fail Threshold (dBFS)

[-80 ... -60 ... -40]
RMS weighted input level for fail detection.

Fail Wait (s)

[1.5 ... 10.0]
Elapsed time after fail detection until the switch over will happen.

Fail Return (s)

[0.0 ... 10.0]
Elapsed time after detection of a proper input signal until the switch back to the program input.

Side Chain Filter

[OFF / ON]
A high pass filter (300Hz) and a low pass filter (3000Hz) is applied to the detector side chain (not the audio path) to prevent hum and noise from blocking fail over switching.

Setup GUI – AUDIO PROCESSOR – De-Esser



Link

[Unlinked, Linked]
For stereo operation you may link the setup parameters for both voice channels.

De-Esser

[ON / OFF]

Frequency (Hz)

[1000 ... 3000 ... 16000]

Range (dB)

[-20.0 ... 0.0]

Filter Type

[Peak / High Shelf]

Q

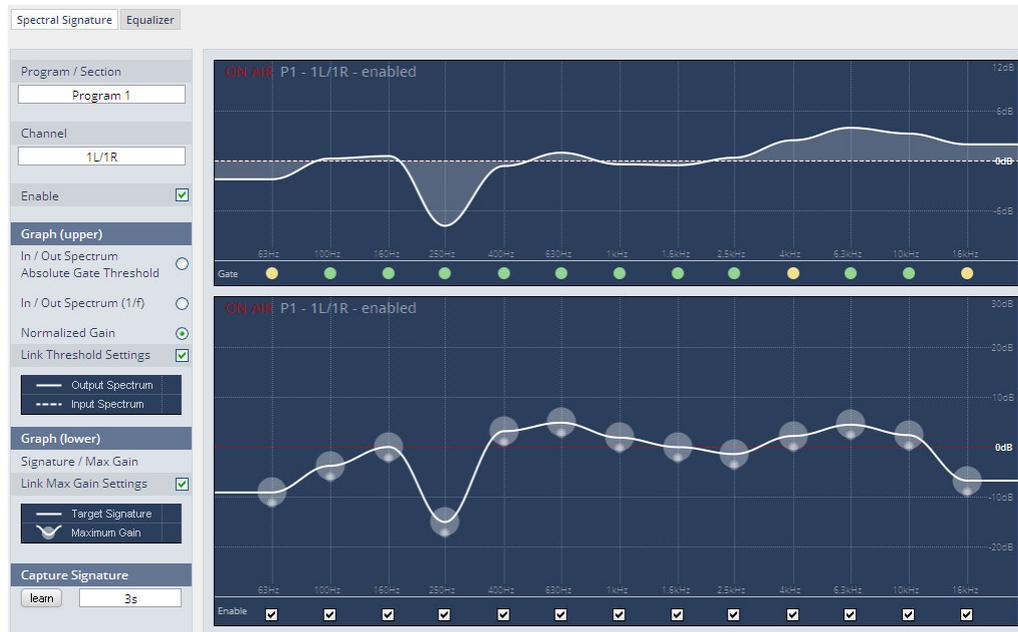
[0.4 ... 1.0 ... 8.0]

Side Chain Listen

[OFF / active]

Setup GUI – AUDIO PROCESSOR – Filter – **Spectral Signature**

Spectral Signature is a highly sophisticated dynamic multiband filter to boost (or reduce) spectral parts of the processed audio signal dynamically. It punches through a reference spectrum to the processed audio signal.



Program / Section

[Program1 / Program2 / Preset]
 Selects the program for which Spectral Signature will be displayed. Since this view does not allow the display of a preset page side by side as usual one must select "Preset" to get to the preset editor.

Channel

[1L/1R, 2L/2R, 1L, 1R, 2L, 2R]
 Depending on the program selected and the link status (see below lower graph) the channel under control will be displayed here.

Enable

[ON / OFF]
 Enables / disables Spectral Signature for the selected program. Please note: For convenient operation, this function is also available (in the Expert section, see below) within the web interface.

Graph (upper)

The upper graph is a metering window, illustrating the difference between the input (dotted line) and the output (solid line) signal. This window can be used in two different ways:

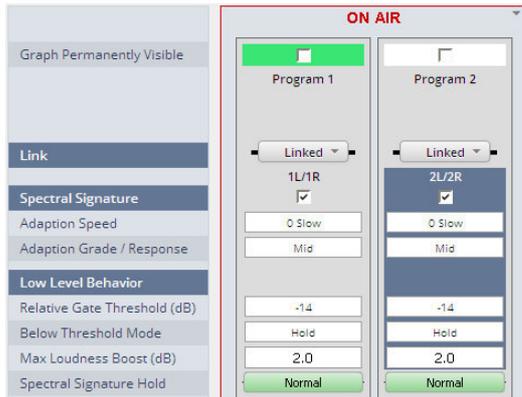
Input / Output Spectrum
Absolute Gate Threshold

[alternative selection]
 The spectrum is shown in absolute values (related to digital full scale). This is very helpful to get an impression of the frequency response of the signal. Also, in this mode the absolute gate threshold can be set within the graph by grabbing and dragging the lower transparent white area. The gate LED row at the bottom indicates whether the absolute or relative gate of the band is closed (yellow) or open (green). A gray LED indicates that the band is switched out.

Normalized Gain

[alternative selection]
 This is very useful to see the actual amount of amplification or attenuation within each band. In this setting the Absolute Gate Threshold cannot be set.

Link Threshold settings	<p>[ON / OFF]</p> <p>The absolute gate threshold can be set individually for every single band. However, in most cases this is not necessary. Checking this box links all gate thresholds together. This connection is absolute, differences between bands will be overwritten. Please note: For convenient operation, this function is also available (in the Expert section, see below) within the web interface.</p>
Graph (lower)	<p>It may show the reference curve for all programs by a color code. Above we see an additional pink line that represents Program 2. It may be disabled by the "Graph Permanently Visible" switch below the graph display.</p>
Signature / Max Gain (dB)	<p>[0 ... 12]</p> <p>Spectral Signature does not work with an absolute level reference. Its frequency response is based on level differences between bands only. Thus a signature is only represented on a relative graph showing the level positions related to the neighboring bands. In consequence, having a straight line does not mean Spectral Signature is not doing anything or is in a 'neutral' status. A straight line would cause Spectral Signature to modify the input signal towards the frequency response of white noise which is, in most cases, not desirable.</p> <p>On mouse over you can read the actual setting of a particular band (BAND 5 above).</p> <p>To change a band, just grab and drag the corresponding sphere. It is recommended to use the 'Learn' function first (see below). Every single band can have an individual max gain value that limits the maximum amplification and attenuation. To set this value, grab and drag the smaller sphere on the bottom of the main sphere. The max gain setting is indicated by the size of the main sphere. The lowest and highest values are indicated by a flashing edge.</p>
Link Max Gain Settings	<p>[ON / OFF]</p> <p>Instead of dialing in all max gain settings individually per band, this link function is a handy tool for basic setup. This connection is absolute, differences between bands will be overwritten.</p>
Enable	<p>[ON / OFF]</p> <p>Checkboxes on the bottom of the lower graph can be used to bypass single bands from processing.</p>
Capture Signature	<p>Spectral Signature is a dynamic filter tool to even out differences between signals of different source or condition. It does not have an absolute reference. Only if the incoming signals frequency response equals the reference response (signature), will Spectral Signature operate in a neutral manner. To create a reference spectrum, which is called 'Signature', start your reference signal and hit the 'Learn' button. After a couple of seconds (see below), the Signature is updated. If the input signal does not change, the upper graph shows that the input and output curves are alike. If the incoming signal spectrum changes, Spectral Signature starts to even out the tonal differences, without destroying the original structure.</p>
<learn>	<p>[manual / 1s ... 30s / 1min]</p> <p>Determines the time over which the input frequency response is integrated to create the signature. A shorter time is sufficient for single channel signals, where the content remains stable over time (for example a presenter microphone). Longer time settings are appropriate for mixed content or buses (for example a studio output).</p>



Graph Permanently Visible

The color code of the column headers will change depending on the program selected for gain change (see upper display). White color represents the selected program. If you select program 2 for example it becomes white while Program 1 is colored light green.

Link

[Unlinked, Linked]
For stereo operation you may link the setup parameters

Spectral Signature

Adaption Speed

[ON / OFF]

[0 Slow / 1 / 2 Mid / 3 / 4 Fast]

This parameter affects the time taken for the bands to reach their target values. Fast settings even out differences between sources, but can lead to audible transitions. They are well suited for single channel signals, for example to even out sound differences due to movement in front of a microphone. Slower settings remain unobtrusive, but cannot bring down differences very quickly. They are suitable for mixed content or buses with varying content. The overall spectrum remains well balanced without drastic sonic changes.

Adaption Grade / Response

[Soft / Mid / Hard]

In order to achieve a stable and natural behavior, the intensity of the gain change needs to process according to a response curve. This curve is defined by a ratio. A high ratio means that a difference of 5 dB results in a gain change of almost the same amount. A low ratio means that the actual gain applied is lower. A ratio of 2:1 would bring the amplification up to 2.5dB in this example. The max gain value is applied after the ratio calculation. As these ratios are not static, they have been combined into three preset responses. The average ratio increases from 'soft' to 'hard'.

Low Level Behavior

Relative Gate Threshold (dB)

[-10 -14 ... -20 / OFF]

To prevent a band from amplifying noise (especially hum), a relative gate can be set. If the energy within one band is lower than this gate, no amplification will take place. This is especially useful, when mixed content with highly varying frequency response is processed (for example a radio station output with alternating presenter voice and music).

Max Loudness Boost (dB)

[0.0 ... 1.5 ... 12.0]

The human hearing is not a linear system. When levels get low, humans perceive less bass within the signal and the sound becomes subjectively thin and tiny. This phenomenon is well known and documented as the 'equal loudness contours'. By setting up Max Loudness Boost the system compensates for this difficulty of the human perception and raises the bass bands as levels decrease. Our intelligent system compensates the frequency response independently from the absolute playback level. Max Loudness Boost is the amount of gain that the system is allowed to build up, not a static gain value. We suggest to experiment with a start setting of 4.

Spectral Signature Hold

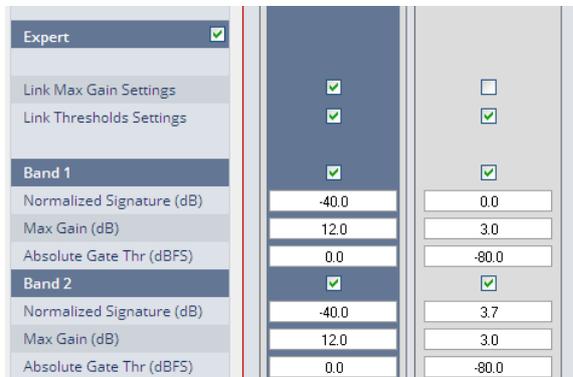
<Normal / <Hold>

Hold freezes the dynamic adaption and preserves the current frequency response curve, for as long as the function is activated. This can be useful for example to prevent adaption to an inserted audio signal (intermission, advertising...).

Expert

[ON / OFF]

All parameters within the Expert section are duplicated in the Signature and Spectrum graphs. They can be used to enter numerical values directly. Changes are reflected in the graphs and likewise in reverse.



Link Max Gain Settings [ON / OFF]

Link Threshold Settings [ON / OFF]

Band 1 [ON / OFF]

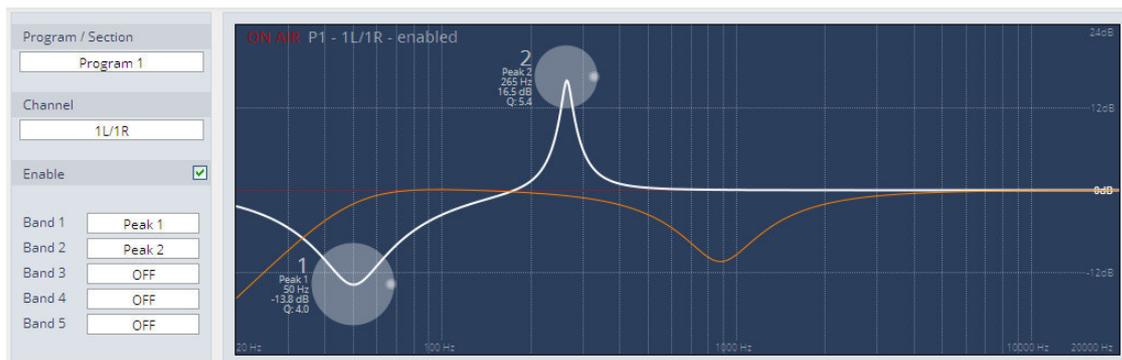
Normalized Signature level [-40.0 ... 0 ... 40.0]

Max Gain [0.0 ... 3.0 ... 12.0]

Absolute Gate Threshold [-84.0 ... -80.0 ... 0.0]

Band 2 ... 16 similar parameters as Band 1

Setup GUI – AUDIO PROCESSOR – Filter – Equalizer



The graphical EQ offers 5 bands. The characteristic of each band can be setup either left hand side of the graph or alternatively for each band further below.

Program

[Program1 / Program2 / Preset]

Selects the source for which the curve will be displayed. This selection depends on the Voice Channel Mode (see SYSTEM > Setup) and whether or not the channels are linked for stereo operation.

Channel

[1L/1R, 2L/2R, 1L, 1R, 2L, 2R]

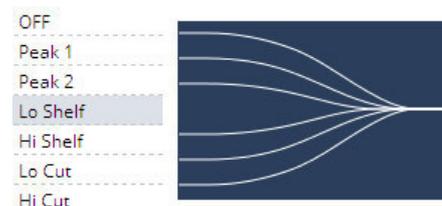
Depending on the program selected and the link status (see below graph) the channel under control will be displayed here.

Enable

[On / OFF]

Band 1 ... 5

[OFF / Peak 1 / Peak 2 / Lo Shelf / Hi Shelf / Lo Cut / Hi Cut]
Filter characteristic will be selected by this pop-up:





Graph Permanently Visible

[ON / OFF]
The color code of the column headers will change depending on the program selected for display (see upper) display. White color represents the selected program. If you select program 1 for example it becomes white while Program 1 is colored orange.

Link

[Unlinked, Linked]
For stereo operation you may link the setup parameters

Equalizer

[ON / OFF]

Band 1

Filter Type

[OFF / Peak 1 / Peak 2 / Lo Shelf / Hi Shelf / Lo Cut / Hi Cut]

Frequency (Hz)

[20 ... 20000]

Gain (dB)

[-20.0 ... 20.0]

Q

[0.4 ... 4.0]

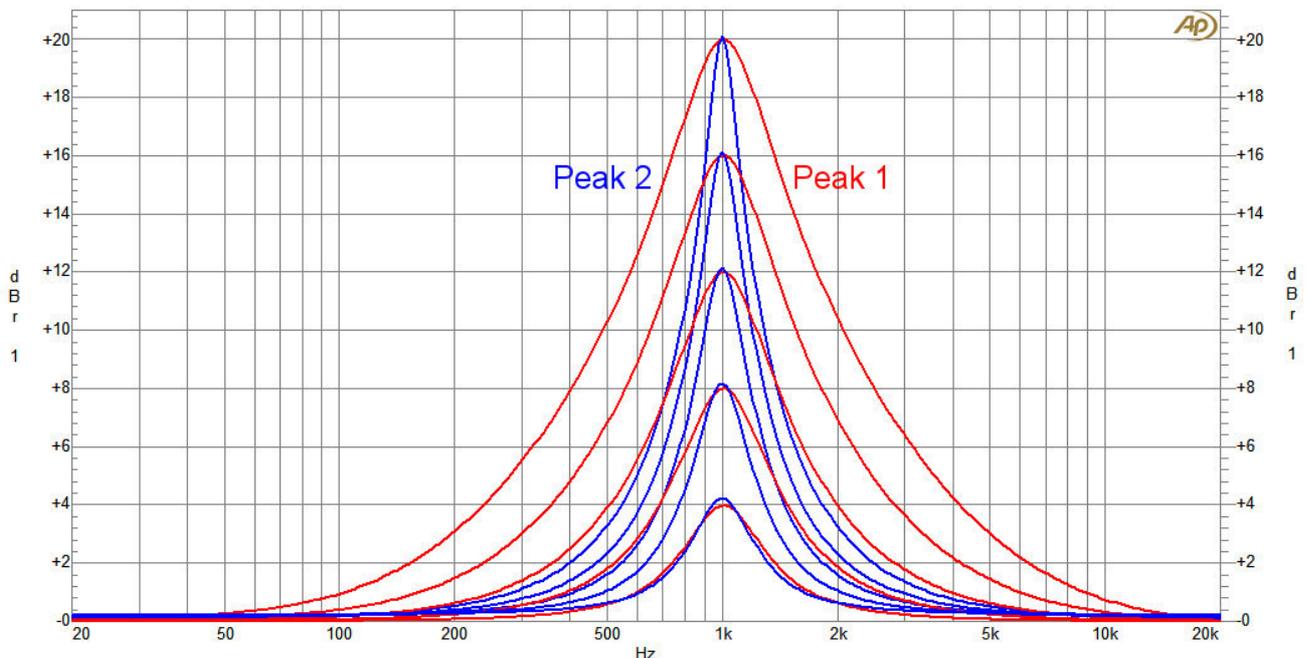
Band 2 ... 5

similar parameters as Band 1

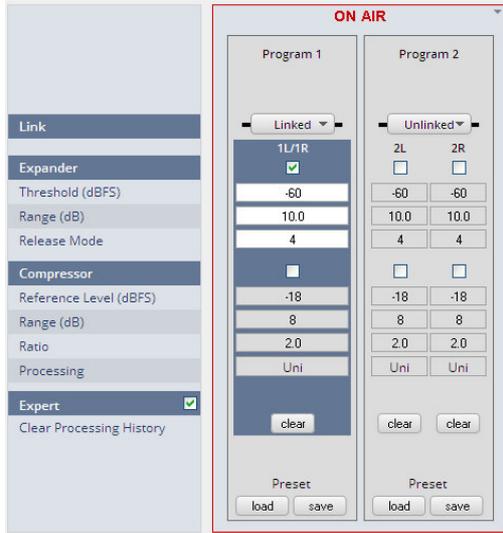
The **EQs** offer two different peak modes:

Peak 1: The bell curves of the **Peak 1** filter features constant quality (Q) over gain. Q is defined at -3dB below peak. It does not change when altering gain.

Peak2: The bell curves of the **Peak 2** filter also features constant quality (Q) over gain. But Q is defined at 50 % of gain. Subjectively the bell curve becomes sharper when increasing gain, but this is only true for the lower 6-8dB of gain.

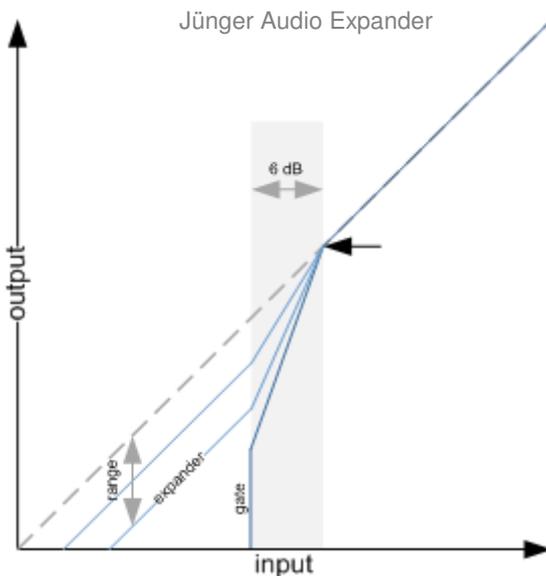


Setup GUI – AUDIO PROCESSOR – Dynamics

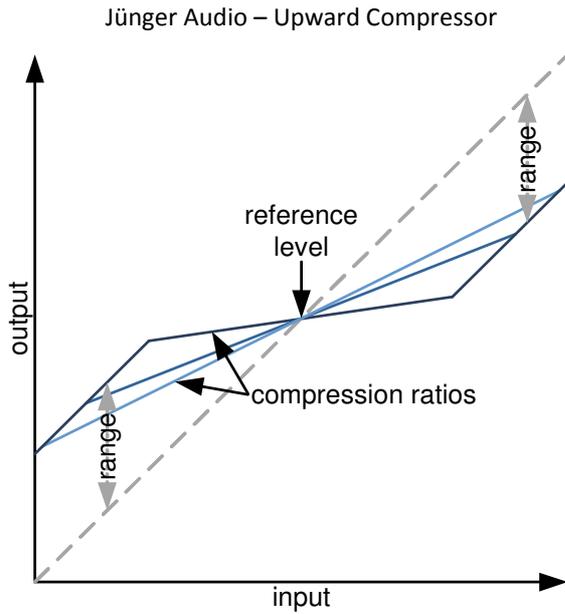


Link	[Unlinked, Linked] For mono operation you may unlink the setup parameters (see Program 2 for example).
Expander	[ON / OFF]
Threshold (dB)	[-60.0 ... -20.0]
Range (dB)	[0.0 ... 10.0 ... 20.0 / Gate]
Release Mode	[0 ... 4 ... 9]
Compressor	[ON / OFF]
Reference Level (dBFS)	[-40 ... -18 ... 0]
Range (dB)	[0 ... 8 ... 20]
Ratio	[1.1 ... 2.0 ... 4.0]
Processing	[Live / Speech / Pop / Uni / Classic]
Expert	[ON / OFF]
Clear Processing History	<clear> pressing the soft button will clear the processing history of the dynamics control loops.

The parameters of the dynamic section are explained below in reference to the curves:



Threshold	Signals below threshold are processed, signals above pass unaffected. Please be aware that this is only true in Gate mode, as the Expander mode features soft knee characteristics.
Range	Maximum reduction range. Its value determines the maximum reduction of the input signal. This parameter is sometimes called 'floor', but differs in terminology. A floor level is defined as an absolute value in dBFS, no matter where the threshold is set. Range defines the relative range of reduction in dB below threshold and is thus independent from absolute values. When it is set to 'Gate' the input signal is muted.
Release Mode	The release mode controls the timing of the closing of the Gate/Expander. Release mode 0 is very fast and even short gaps or signal intermissions lead to gain reduction. On the other end of the scale, 9 is a very slow mode with a relaxed handling of gaps and low level periods. All modes feature the same super fast opening when the signal returns above threshold.



Reference Level (dBFS) Not to be confused with threshold, this parameter defines the turning point of the response curve from upward to downward compression (see picture). When set to 0dBFS, the signal is amplified according to the ratio and range settings.

Range (dB) This defines the range over which dynamic compression is applied as defined by the ratio setting. Signals outside of this range are still reduced or amplified but not altered in their dynamic structure.

Ratio Determines the amount of gain reduction by a selectable ratio. Although the same in mathematical terms, understanding is easier when differentiating between upward and downward compression:

Processing

The timing characteristics of the compressor are generated adaptively according to the incoming signal structure. The overall timing can be set up from fast and responsive settings (lower numbers) to relaxed settings (higher numbers) without detailed access to the actual micro timings. The names behind some of the numbers may help to easily find adequate values to your content.

Expert

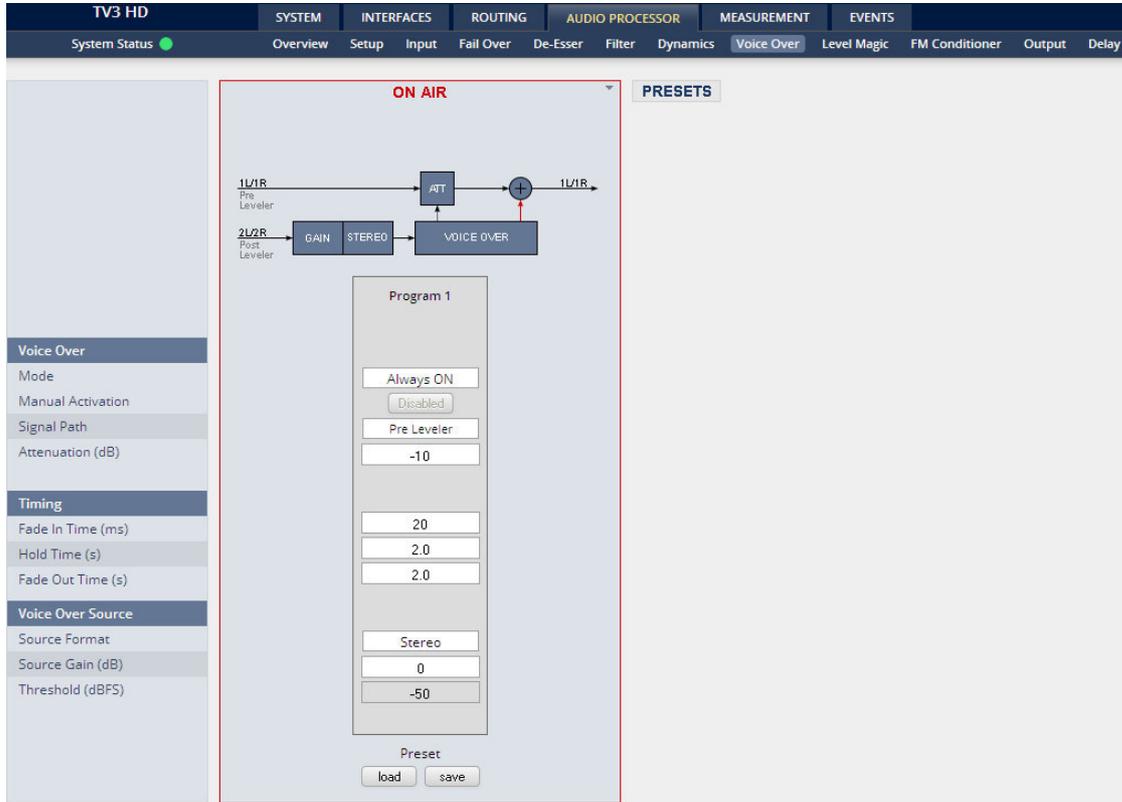
Clear Processing History

[ON / OFF]

This is a triggered action that resets the dynamic processing without any release time. Imagine it as a short circuit to the timing circuits of an analog dynamic processor which discharges the whole system and immediately returns the dynamic gain to its neutral state. This function is useful to reset the process when switching programs (e.g. from movie to commercial breaks).

Setup GUI – AUDIO PROCESSOR – Voice Over

The voice over section allows for manual (mixing) / automatic (ducking) of a voice channel over the program feed. The dynamic schematic in the top of the pane shows the actual signal flow:



Voice Over Mode

[OFF / Always ON / AUTO]

Defines the operating mode of the voice over block. AUTO will detect the signal in the voice channel and will automatically perform the voice over (ducking).

Manual Activation

[Enable / Disable]

Mostly used for testing

Signal Path

[Pre Leveler / Post Leveler]

See AUDIO PROCESSOR > Overview for the actual location of the circuit in the signal path.

Attenuation (dB)

[-30 ... -10 ... 0]

Timing

Fade In Time (ms)

[10 / 20 ... 1000]

Hold Time (ms)

[0.0 ... 2.0 ... 10.0]

Fade Out Time (ms)

[0.0 ... 2.0 ... 10.0]

Voice Over Source

Source Format

[Stereo / Mono LL / Mono RR / Mono L+R]

Source Gin (dB)

[-20 ... 0 ... 20]

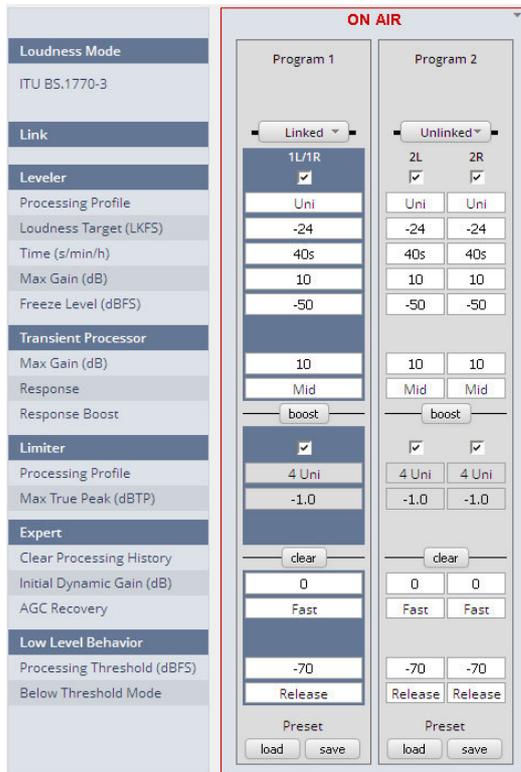
Threshold (dBFS)

[-60 ... -50 ... -40]

Sets the threshold for detection in AUTO mode.

Setup GUI – AUDIO PROCESSOR – Level Magic

This function block is used for loudness control of the program paths.



Loudness Control Mode [display of the setting from AUDIO PROCESSOR > Setup > Loudness Mode]

Link [unlinked / linked]
defines the coupling of the control circuits

Leveler [ON / OFF]

Processing Profile [Live / Speech / Pop / Uni / Classic]

Loudness Target for different modes
 Level [0 ... -50dBFS]
 ITU [0 ... -50LKFS]
 EBU [0 ... -50LUFS]

Time (s/min/h) [10, 20, 40 / 1, 2, 5, 10, 20, 40 / 1, 2]

Max Gain (dB) [0 ... 10 ... 40]

Freeze Level (dBFS) [-60 ... -50 ... -20]

Transient Processor

Max Gain (dB) [0 ... 10 ... 15]

Response [Soft, Mid, Hard]

Response Boost <boost>

Limiter [OFF / ON]

Processing Profile [Live / Speech / Pop / Uni / Classic]

Max True Peak (dBTP) [-20 ... -9.0 ... 0.0]

Expert [ON / OFF]

Clear Processing History <clear>

Initial Dynamic Gain (dB) [-40 ... 0 ... 15]

AGC Recovery [Fast / Normal]

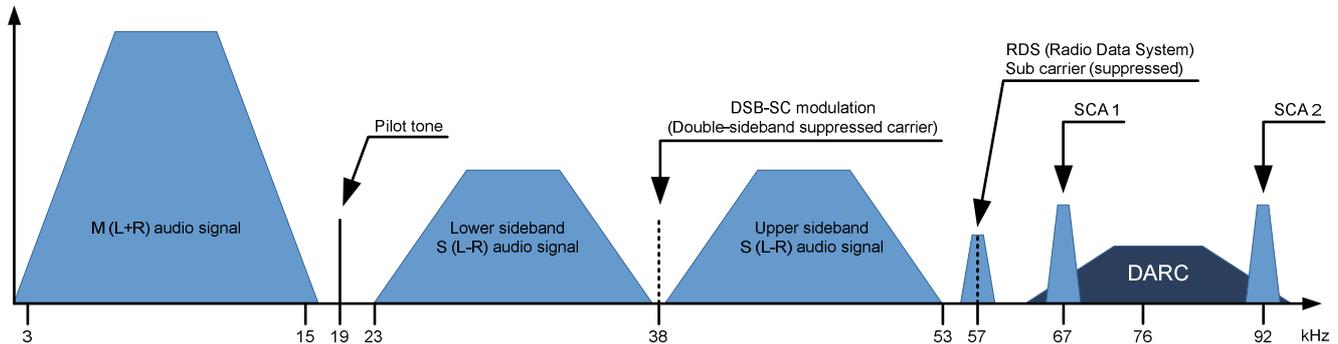
Low Level Behavior

Processing Threshold (dBFS) [-80 ... -70 ... -20]

Below Threshold Mode [Hold / Release]

For details regarding LevelMagic parameters see the bulletin:
 "Junger processing parameter description" on the Junger web site:
<http://junger-audio.com/downloads>.

Setup GUI – AUDIO PROCESSOR – FM Conditioner



FM radio broadcast is not just frequency modulated audio. It consists of different signals and services that share the 'space' available on the FM carrier. A typical stereo radio signal spectrum may look like this:

Mono audio signal (M=L+R)	30Hz to 15kHz base band
Stereo pilot tone at 19kHz	approximately 9 % of 75kHz deviation
Stereo audio signal (S=L-R)	30Hz to 15kHz base band
DSB-SC carrier	Double-sideband suppressed carrier
RDS signal	Radio Data Signal at 1187,5bit/s
DARC signal	Data Radio Channel at about 16 000bit/s
SCA signal	14kHz (narrow) or 26kHz (wide) bandwidth for auxiliary audio services

To calculate the overall MPX power the spectrum power of all consisting signals needs to be considered.

Please note that within the **FM Conditioner** Web UI only RDS and SCA Deviation can be set as additional services. As SCA and DARC normally cannot be used simultaneously due to their overlapping frequency bands, the SCA Deviation parameter can be used for DARC also. To calculate the overall deviation, all of the services in use must be taken into account in order to not exceed the modulation limits defined by the ITU (see below). After setup this happens internally and is not of concern for the **FM Conditioner** user.

When dealing with the audio processing side of FM broadcast, four main parameters come into focus:

- * **Deviation** Δf_c of the transmission frequency (carrier) f_c
- * **MPX Power** of the modulating signal (modulator)
- * **Pre-Emphasis** to enhance the signal-to-noise ratio of FM transmission
- * **Baseband bandwidth** of all involved services (audio signals and auxiliary data)

ITU-R BS.412 has standardized the maximum values for these parameters. Broadcasters must comply with these limits to not exceed the planned coverage or interfere with adjacent programs. They are:

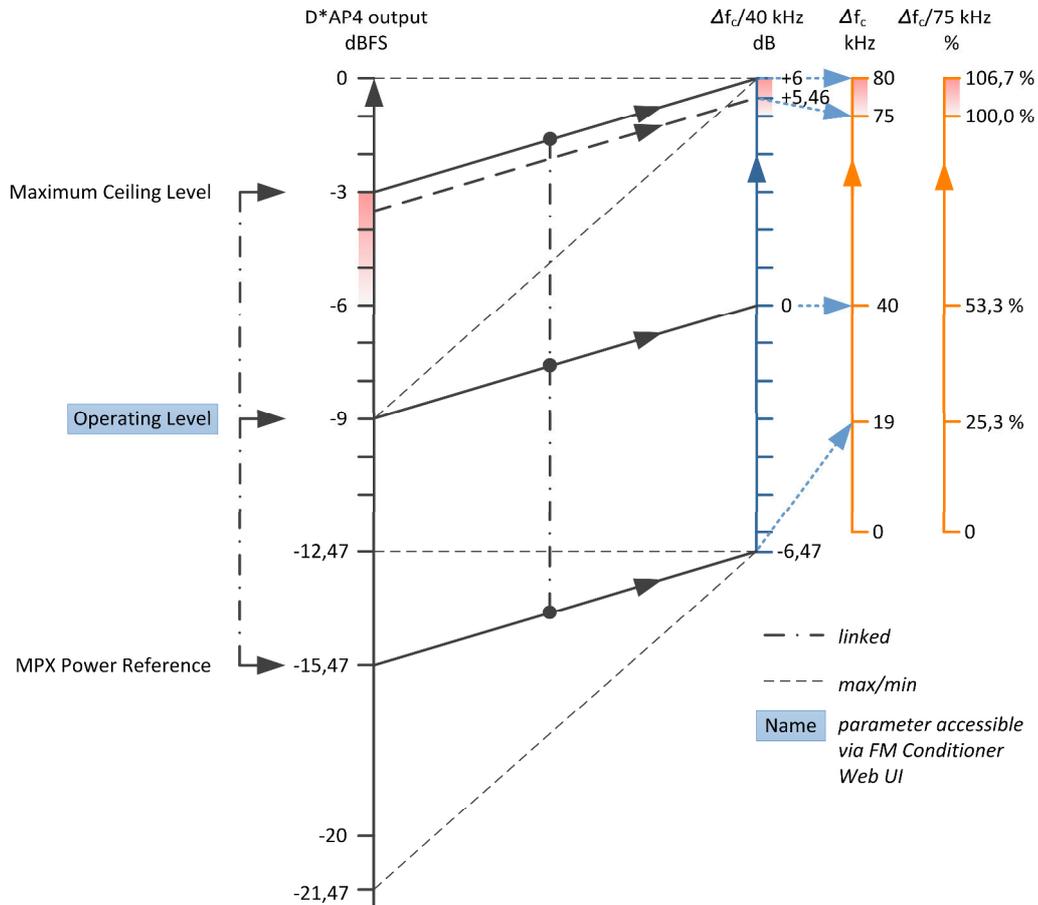
- * Maximum peak deviation of ± 75 kHz
- * Maximum MPX power of 0dB_r
- * A typical audio baseband cut-off at 15kHz to ensure undisturbed transmission of the 19kHz stereo pilot tone
- * For mono operation a typical audio baseband bandwidth of 17.5kHz is utilized (no pilot tone necessary)

Calibration

MPX power is measured in a random interval of 60 s. MPX power of 0dB_r should be equal to the modulation power of a stationary sine signal that causes a deviation of ± 19 kHz. A stimulus frequency of 500Hz is recommended.

The necessary tasks to comply to this rule 'are simple': take your pocket power measurement instrument, hook up your always at fingertip reference antenna, tune into your transmitter and measure... Now adjust the relevant audio parameters if necessary. Since this is not feasible for studio equipment we must calculate MPX power prior to modulation and translate them for the studio output. For a precise calculation all technical equipment needs to be gain matched and calibrated.

For Calibration



The important step in calibration is to set up the **Operating Level**. A stationary sine signal at this level must cause a FM carrier deviation of ± 40 kHz. If the input level (at FM HPA or uplink line) for this reference modulation is known, just set the Operating Level in the **FM Conditioner** accordingly. In most installations this will be the case.

In a lot of stations +6 dBu (analog) or -9 dBFS (digital) for a 500 Hz tone is reference level. It may be designated as the operating level and defined at 0 dB relative (displayed on a peak level meter). Please be careful with this type of reference level scale, as this analog operating level of 0 dB is not the same as 0 dB MPX power!

If the reference modulation is unknown, a sine test tone needs to be applied and the frequency deviation of the FM carrier needs to be measured over the air. Start with a generator level of -9 dBFS and change this value until ± 40 kHz deviation is attained. Please note that any Junger Audio unit in the chain between generator and FM HPA needs to be bypassed first. Calibration is performed without any processing, additional service or pilot signal considered!

If the Reference Level of your setup differs from -9 dBFS the Setup Gain of the FM Conditioner can be used for level matching.

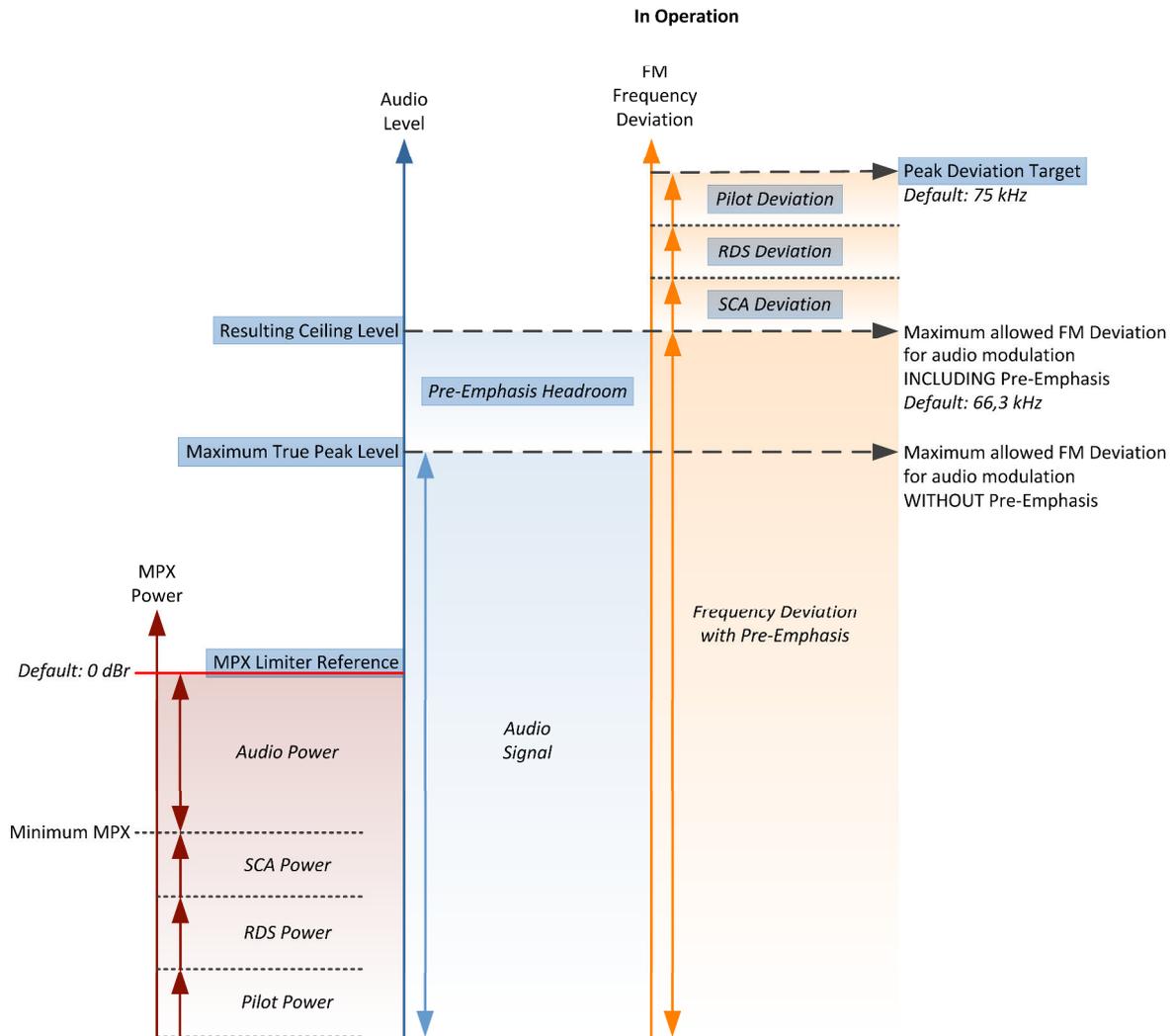
The second step of calibration is to set the values for **Pilot Tone, RDS, SCA(DARC) Deviation**. The necessary values depend on the setup of the respective encoders. Please refer to the respective manuals.

After calibration the **FM Conditioner** will now display the available audio headroom. Here is an example with an assumed deviation of ~ 12 % of 75kHz for the extra services:

$$20 \cdot \log (75\text{kHz} - 8.8\text{kHz}) / 40\text{kHz} = 4,4\text{dB}$$

or -4.6dBFS

This calculation is performed internally and updates automatically when any of the involved parameters change. The resulting value is called the Ceiling. It is important to know that the **Resulting Ceiling** is calculated with the **Pre-Emphasis** filtering of the FM transmitter included. Thus the wideband true peak level of the audio signal before **Pre-Emphasis** needs to be lower. A look at the level relation diagram may help understanding this:



Pre-Emphasis is a filter system where the higher frequencies are raised by a shelving filter at transmission stage and equivalently reduced at the receiver end. The **Pre-Emphasis** filter utilizes a time constant of 50 μs (or 75 μs in the USA) which results in a gain of 10dB at 10kHz. This procedure creates a significantly improved signal-to-noise ratio.

But as the increased high frequency energy adds to the MPX power, it needs to be considered within the **FM Conditioner**. There are two facilities to deal with **Pre-Emphasis**. First a **Pre-Emphasis Headroom** parameter reduces the maximum wide band level by lowering the true peak limiter threshold. This results in lower overall audio levels, but increased high frequency transparency. Second a process called **Pre-Emphasis Limiter** reduces the high frequency amount of the audio signal dynamically and thus creates 'space' for the additional **Pre-Emphasis** shelving. When activated the **Pre-Emphasis Limiter** prevents high frequency over-modulation. To reduce its effect the **Pre-Emphasis Headroom** needs to be increased. If the signal structure requires permanent Pre-Emphasis Limiter action, the system automatically reduces the input gain of the True Peak Limiter to improve sound quality. Please note that very short transients may not be 'caught' by the **Pre-Emphasis Limiter**. This however happens by principle and has no practical relevance for the FM transmission.

When activating the **FM Conditioner** it takes over control of the True Peak Limiter algorithm.

Important Note! The TP Limiter parameters on the Level Magic page in the Web UI are automatically overridden. The **True Peak Max** value cannot be set by the user, as it is automatically calculated and set to the Ceiling level minus the **Pre-Emphasis Headroom**. With no **Pre-Emphasis Headroom** Maximum True Peak equals Ceiling.

The MPX Limiter algorithm

The most important part of the FM Conditioner processor certainly is the **MPX Limiter**. As MPX power is a value that is calculated with one minute of integration time, limiting is a very complicated task. In theory 60 seconds look ahead time seems appropriate but of course not practically applicable for a real time processor. Thus the Junger Audio **MPX Limiter** works with a complex prediction algorithm that adapts to the incoming signal structure. Still the limiter reference level is a brickwall threshold and considered sacrosanct. In case of 'emergency' the **MPX Limiter** will reduce the signal level drastically to prevent any threshold violation. Chances are that the **MPX Limiter** of the **FM Conditioner** is the best MPX brickwall available today.

Please note that the **MPX Limiter Reference** can of course be violated when the incoming levels are high and the **MPX Limiter** was just switched on. By measurement principle it may take up to one minute for the **MPX Limiter** to settle.

The **MPX Limiter Profile** influences the speed and range of the process and in consequence the neutrality to the incoming sound quality. With softer settings the system needs to apply a buffer zone between the **MPX Limiter Reference** and the measured MPX power of the audio signal. Though this buffer zone is always very small, with harder settings it becomes even smaller and a higher MPX power can be transmitted. The optimal setting depends on the type and style of program that is broadcast.

A very handy feature of the **FM conditioner** is the visualization of the development of MPX power over time:



The screenshot above shows an example curve of the calculated MPX power [MPX Power 60 s] (orange). For convenient analysis and set up the graph allows to display a second MPX curve [MPX Power User] (green) with a user defined integration time. This is especially useful to see why the **MPX Limiter** starts working although the MPX power has not reached threshold. The purple curve shows the gain reduction action of the MPX Limiter. The center white line represents the selected MPX Limiter reference.

Program [Program 1 / Program 2]
Selects which of the two programs will be displayed on the graph

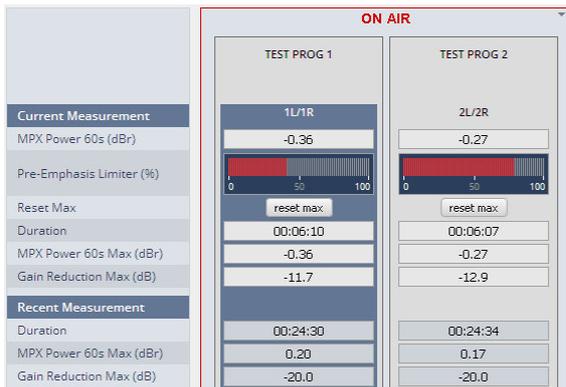
Graph

Enable MPX Power User [OFF / ON]
Enables the display of the user-defined curve

User Integration Time (s) [1 ... 40]
This is only a measurement parameter that has no influence on the MPX Limiter processing

Time Range (min) [1 / 2 / 5 / 10 / 20 / 30]
Sets the time scale for the MPX power display. The Current Measurement displays the numerical values of the MPX power measurement:

Current Measurement [1L/1R and 2L/2R]



MPX Power 60s (dBr) current MPX power
Pre-Emphasis Limiter (%) bar graph to show the percentage of high frequency filtering applied

Reset Max <reset max>
soft button resets Duration, MPX Power (60s) Max and Gain Reduction Max

Duration elapsed time since <reset max> was depressed

MPX Power 60s Max (dBr) current maximum MPX power

GAIN Reduction Max (dB) current maximum gain reduction

Recent Measurement

Duration elapsed time of recent measurement since <reset mpx max> was previously depressed

MPX Power 60s Max (dBr) maximum MPX power value detected

GAIN Reduction Max (dB) maximum gain reduction allied by the MPX limiter

In the next section the operator can set the audio relevant parameters of the FM conditioning process:

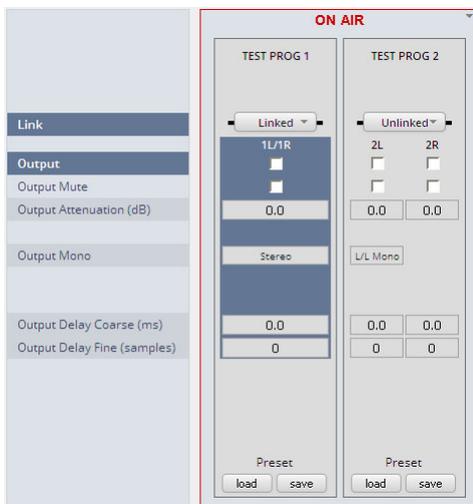


FM Conditioner	[ON / OFF]
Setup Gain (dB)	[-4.0 ... 0.0 ... 10.0] can be used to adapt loudness processed signals to MPX criteria or level matching
Pre-Emphasis Headroom (dB)	[0.0 ... 2.0 ... 15.0]
MPX Limiter Profile	[Soft / Mid / Hard]
True Peak Limiter Profile	[0 / 1 Live / 2 Speech / 3 Pop / 4 Uni / 5 / 6 Classic / 7 / 8 / 9]
True Peak Max (dBTP)	[read only]

The expert mode allows the administrator to set up the parameters of the transmitter and to turn the **MPX Limiter** on or off.

Expert	[ON / OFF]
Pre-Emphasis	[OFF / 50µs / 75µs]
Operating Level (dBFS)	[-15.0 ... -9.0 ... -6.0]
Peak Deviation Target (kHz)	[35.0 ... 75.0 ... 80.0]
Pilot Deviation (kHz)	[0.0 ... 6.0 ... 15.0]
RDS Deviation (kHz)	[0.0 ... 2.0 ... 4.0]
SCA Deviation (kHz)	[0.0 ... 15.0]
Resulting Ceiling (dBFS)	calculated from the operating level, the sub-carrier (pilot-tone) and the respective RDS and SCA deviation
MPX Limiter	[OFF / ON]
Reference Level (dBr)	[-4.0 ... 0.0 ... 4.0]
Low Pass Filter (15kHz)	[OFF / ON]

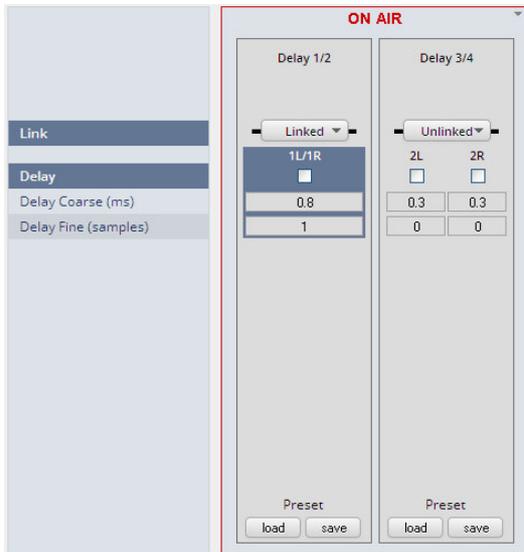
Setup GUI – AUDIO PROCESSOR – **Output**



Link	[unlinked / linked] defines the coupling of the control circuits
Output	[ON / OFF]
Mute	[ON / OFF]
Attenuation (dB)	[-80.0 ... 0.0]
Mono	[L+R Mono / LL Mono / RR Mono / Stereo]
Output Delay Coarse (ms)	[0.0 ... 2000.0]
Output Delay Fine (samples)	[0 ... 2000]

Setup GUI – AUDIO PROCESSOR – Delay

The D*AP4 has an independent audio delay that may be routed to any signal path inside the device.



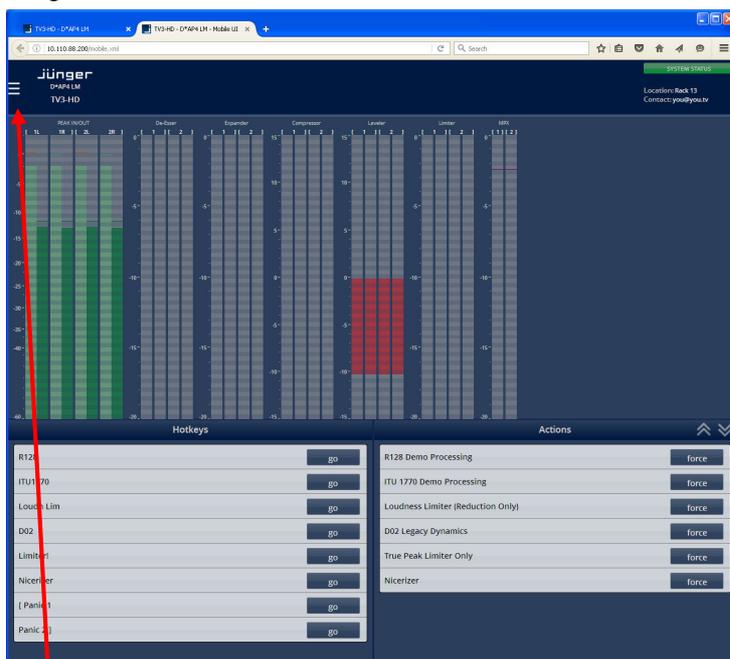
Link	[unlinked / linked] defines the coupling of the control circuits
Delay	[ON / OFF]
Output Delay Coarse (ms)	[0.0 ... 2000.0]
Output Delay Fine (samples)	[0 ... 2000]

Important Note! If the audio delay is routed into an internal signal path via the device routing matrix one must add an initial delay of 27 samples.

Setup GUI – AUDIO PROCESSOR – Mobile UI

The D*AP4 provides an extra UI for live applications that may be used on tablets or mobile phones but may also be displayed on a PC's web browser. When you click on the <Mobile> soft button in the upper right area of the GUI above the <Help> button, a new tab in your current browser will open up.

But you can also use the URL: "<IP-address>/mobile" to open the operator UI elsewhere (in a browser of a different PC or a mobile phone or a tablet). For mobile devices it requires network integration of the D*AP4 via a WLAN.



At the bottom left you have a representation of the Hotkey settings (see EVENTS > Triggers > Remote Hotkeys).

By pressing the respective <go> soft button you will trigger an action the same way a remote Hotkey would do.

On the bottom right you have a selected number of actions available to trigger (see EVENTS > Actions > Event Actions).

By pressing the respective <force> soft button you will trigger an action the same way a trigger would do.

In the ● upper left hand corner you can open the Options settings:



You can enable / disable full screen display.

Here you can decide between three arrangements of the soft buttons:

Buttons A shows the rectangle buttons with assigned dark colors, active ones are highlighted.

Buttons B shows the greyish rectangle buttons, active ones are highlighted.

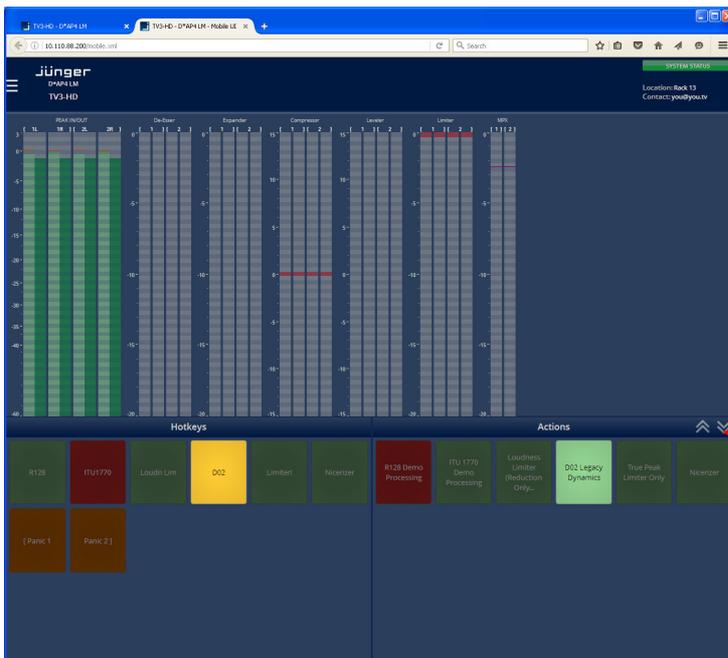
List shows the initial button list display

Desktop Version

But you can also open another tab where the GUI will be loaded.

The color scheme will be defined on the EVENTS > Triggers > Remote Hotkey page and / or EVENTS > Actions > Event Actions. The brightness of the buttons depends on the momentary status of the represented function. Dark color means inactive, medium bright color means that the function is triggered and bright color means that the D*AP4 has fully finished the operation.

As stated above, the active status of a hotkey or the result of a specific action will be high lighted in the bright version of the assigned color. Below is an example where the action "D02 Legacy Dynamics" was triggered by the hotkey "D02", so both are shown as active:

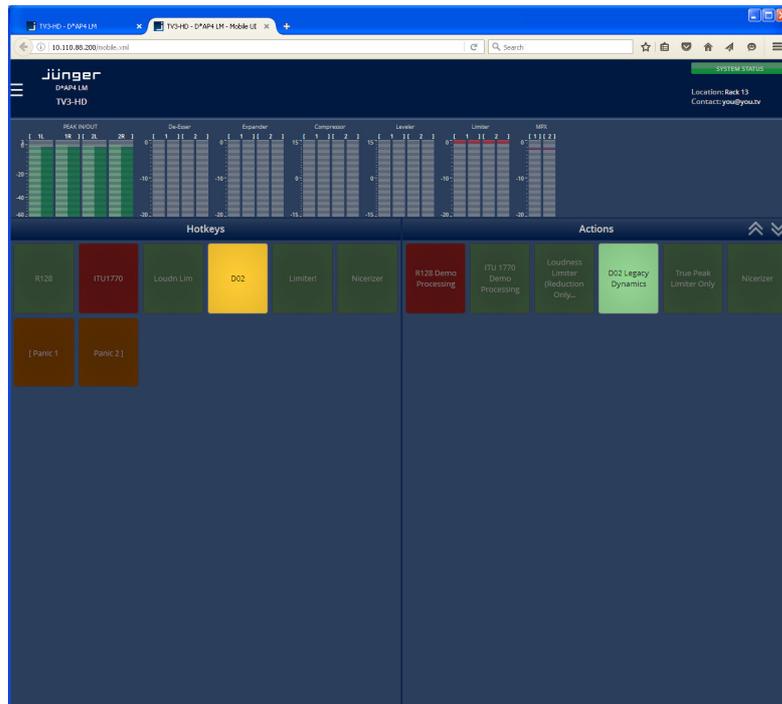


If you press a hotkey the color becomes medium bright immediately. Since many actions (e.g. reconfiguration of the MAP) may take a few moments the status of a button finally turns bright to acknowledge that the action has been performed. If the color turns back to the dark version the action was not successfully finished.

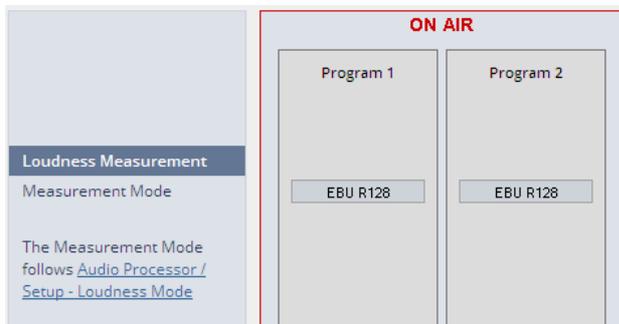
The up and down arrows here will change the size of the bar graph display to allow for more room for buttons to be displayed. This is a four stage feature:

Small > medium size > large > no bar graph display.

Below is an example for the small size version:



Setup GUI – MEASUREMENT – Setup



As stated at the left side the measurement mode must be set-up here:
 AUDIO PROCESSOR > Setup > Loudness Mode

Setup GUI – MEASUREMENT – Loudness

The **D*AP4 LM** offers a sophisticated loudness measurement tool for the input and output of the program path of the device. The three control buttons **<pause>**, **<reset>**, **<reset max>** may be used to manually control the actual measurement. The pane shows the two measurement blocks for both programs:



Loudness Mode [EBU R 128]
setting from AUDIO PROCESSOR > Setup > Loudness Mode

Current Measurement [hh:mm:ss]
Time elapsed since measurement started (excluding pauses)

Integrated Loudness (LUFS)

Loudness Range (LU)

Short-Term Loudness (LUFS) numeric and convenient bar graph display

Momentary Loudness (LUFS) numeric and convenient bar graph display

Short Term Max (LUFS)

Momentary Max (LUFS)

True Peak Max (dBTP)

Recent Measurement [hh:mm:ss]
Total time of the recent measurement

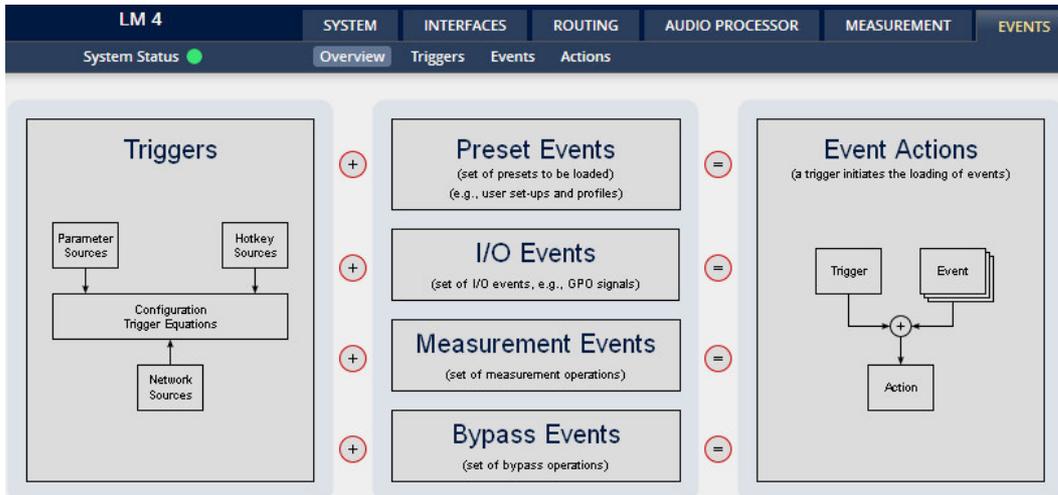
Important Note! The measures of the parameters above depend on the loudness mode selected at AUDIO PROCESSOR > Setup pane.

The measurement data may also be streamed to the **J*AM** (Junger Application Manager) to feed the external loudness measurement and loudness logging tool. The **J*AM** is a PC software package that one can download from the Jungeraudio.com web site. To perform loudness measurement and loudness logging one must buy a hardware (USB) dongle.

Setup GUI – EVENTS – Overview

As mentioned previously, D*AP4 includes a sophisticated **event management** system. The event system performs **Actions**. These **actions** are built from **Events**. **Actions** may be triggered manually (via the X*AP RM1 remote panel **Hotkeys**), semi-automatically (triggered by network commands or GPIs) and automatically (triggered by changes of parameters and/or the internal status) or as a combination of all three.

The overview shows the building blocks of the action management of the D*AP4:



A **trigger** is subdivided into a trigger **type** and a trigger **source**. E.g. a GPI is a trigger type while its number (the physical input) represents the trigger source. Other trigger types have sources which must be configured, like the names of **X*AP Hotkeys** or network triggers.

- Hotkey Sources You may assign hotkeys of the **X*AP** remote and / or the **mobile UI** to become a trigger source.
- Network Sources Received via the I-s-b EmBER+ protocol.
- Parameter Sources Device parameters / status information grouped into system and Interfaces.

The triggers will be defined by its trigger equation that may be the logical combination of 2 trigger sources.

The **D*AP4** knows five different **event types**:

- Preset Events System / Interfaces / Routing / Audio Processor / Program / Delay
- Parameter Events Measurement
- Measurement Events Pause / Continue / Reset / Reset Max / Start / Pause / Stop
- I/O Events GPOs / Loudness Measurement
- Bypass Events Program 1 / Program 2

The **D*AP4** has two different **action types**:

- Event Actions executes the predefined events
- Bypass Actions executes pre-defined bypass scenarios, independent on the bypass events

An action runs like a flip-book inside the **D*AP4**. This powerful technology spans from simply recalling a certain parameter over specific parameter combinations (household name: "Preset") to the complete reconfiguration of the **D*AP4** including all signal routing, processing parameters and so forth. It allows you to create your own **snap shots** where you decide what is part of it and what is not! But it also enables **fail over** scenarios where the **D*AP4** will automatically react to the system and/or parameter status.

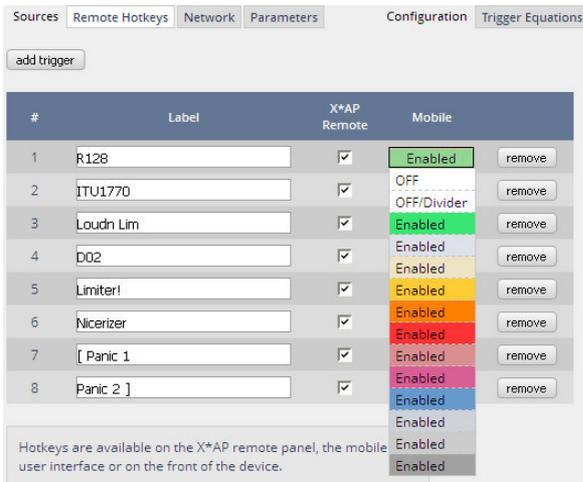
The steps to set up the **EVENTS** system are as follows:

1. **Define** - trigger sources
2. **Configure** - triggers by logical combination of pre-defined trigger sources
3. **Set up events** - by selecting presets or GPOs or bypass blocks or measurement controls
4. **Create actions** - what should happen? Which trigger will launch which event?

The following explanations refer to the factory default settings of the EVENTS system.

Setup GUI – EVENTS – Triggers – Sources – Remote Hotkeys

Hotkeys are the 8 buttons of an **X*AP RM1** remote panel. You may give them names and enable them to show up as active on the **X*AP RM1** remote panels main operating menu.



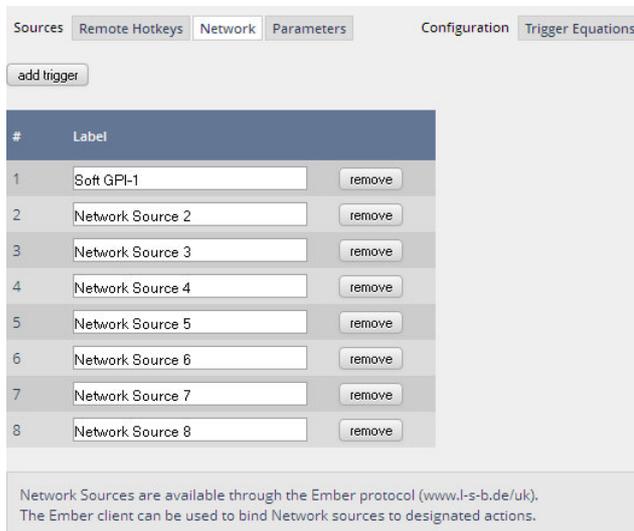
#	Label	X*AP Remote	Mobile	
1	R128	<input checked="" type="checkbox"/>	Enabled	remove
2	ITU1770	<input checked="" type="checkbox"/>	OFF	remove
3	Loudn Lim	<input checked="" type="checkbox"/>	Enabled	remove
4	D02	<input checked="" type="checkbox"/>	Enabled	remove
5	Limiter1	<input checked="" type="checkbox"/>	Enabled	remove
6	Nicerizer	<input checked="" type="checkbox"/>	Enabled	remove
7	[Panic 1	<input checked="" type="checkbox"/>	Enabled	remove
8	Panic 2]	<input checked="" type="checkbox"/>	Enabled	remove

- #** The number of the Hotkey on the **X*AP RM1** remote panel, counting from left to right.
- Label** Each Hotkey may have a label that appears in the display of the **X*AP RM1** remote panel above that button.
- X*AP Remote** [ON / OFF]
If you turn it off the respective Hotkey on the **X*AP RM1** remote panel becomes inactive - no label is displayed and the button background light turns off.
- <remove>** will remove a line from the list. This will automatically disable the respective front panel button.

The number of hotkey triggers is not limited. You may also add virtual hotkeys which can be used by a Mobile UI for example, that may have more than 8 compared to the **X*AP RM1**.

Setup GUI – EVENTS – Triggers – Sources – Network

Network triggers are based on the **EMBER+** protocol. See code.google.com/p/ember-plus/ for details. The **D*AP4** receives such triggers over the TCP/IP network. The triggers are issued by a remote device or a broadcast automation system. You may assign these triggers to virtual panels, physical buttons or play list events, of Ember+ enabled control instances.



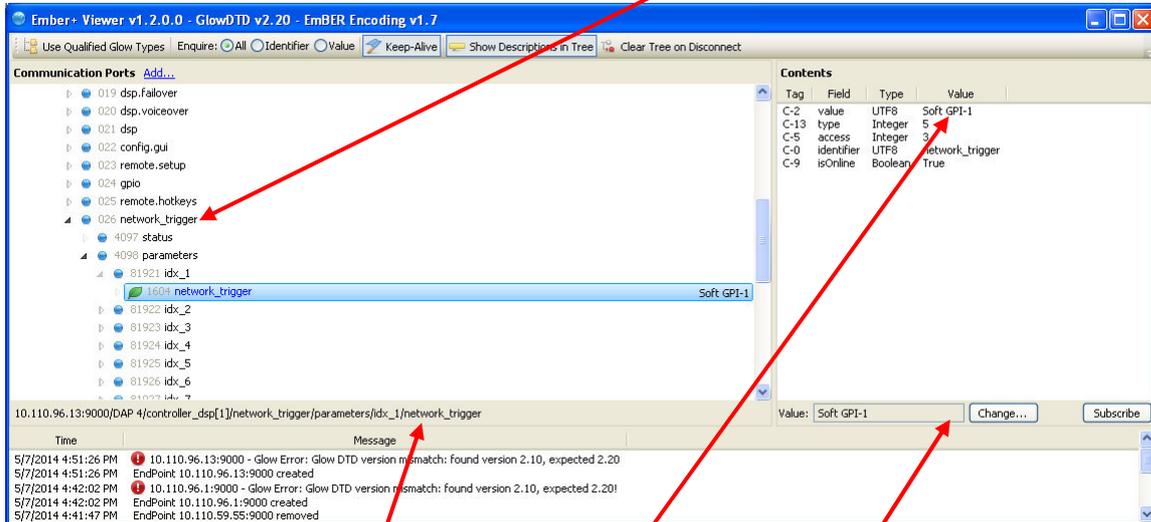
#	Label	
1	Soft GPI-1	remove
2	Network Source 2	remove
3	Network Source 3	remove
4	Network Source 4	remove
5	Network Source 5	remove
6	Network Source 6	remove
7	Network Source 7	remove
8	Network Source 8	remove

- #** The number of the network trigger.
- Label** Label of that network trigger. It appears on the **Configuration** pane as well as in the **EMBER+** tree of the setup interface of a control instance.

We have given the first Network trigger the name **"Soft GPI-1"** to explain how it works. It will be activated from the automation system if the pre-defined event has been reached in the play list.

Lets have a look at the "Soft GPI-1" trigger.

If you open the **EmBER+** viewer tool you must drill down to **network_trigger**:



The path is also shown at the bottom of the main window:

As a value you will receive the trigger name that you have given to the first network trigger **"Soft GPI-1"**.

The automation may now fire that trigger by simply returning the string.

For demonstration how it works press **<Change ... >** In the popup press **<OK>**. This works as sending a pulse (behaves like a push button).

But the automation may also give it a value of **"1"** to set the trigger and later on a **"0"** to reset it. What best to do depends on the functionality you need for a specific application.

Setup GUI – EVENTS – Triggers – Sources – **Parameters**

Label	Category	Subcategory	Parameter	Expression 1	Expression 2		
Sample Rate 48kHz Trigg...	System	Setup	Current Sample Rate	=	48	-	remove
AES Input Fail	Interfaces	AES I/O	Input Status All	=	Fail	-	remove
Fail Over Alarm	Audio Processor	Fail Over	Failover Status	=	<multiple values>	-	remove

Above is an example of possible parameter trigger sources.

Setup GUI – EVENTS – Triggers – Configuration – **Trigger Equation**

To form a **Trigger** you may logically combine two trigger sources below.

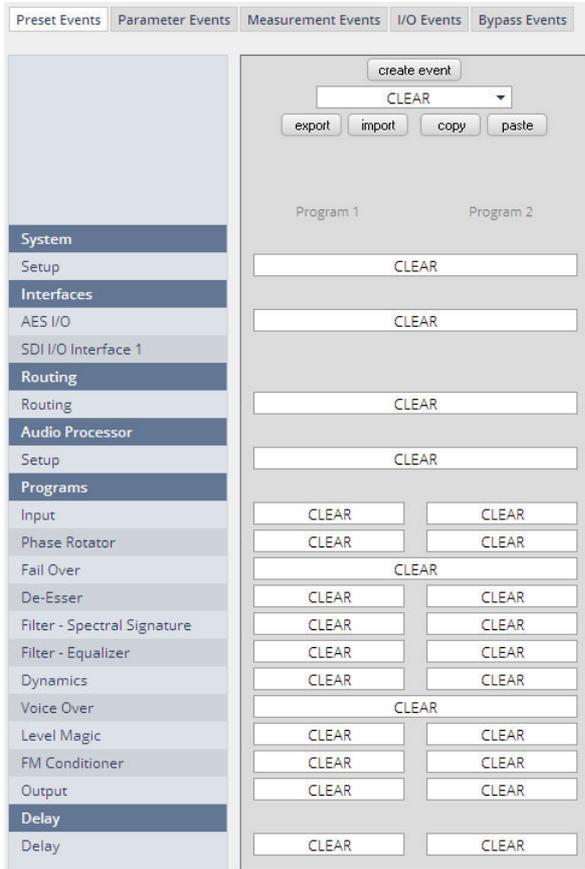
Trigger	Invert	Type	Source 1	Source	Logic	Invert	Type	Source 2	Source
Trigger 1	<input type="checkbox"/>	Hotkey	1 R128		or	<input type="checkbox"/>	GPI	1	
Trigger 2	<input type="checkbox"/>	Hotkey	2 ITU1770		or	<input type="checkbox"/>	GPI	2	
Trigger 3	<input type="checkbox"/>	Hotkey	3 Loudn Lim		or	<input type="checkbox"/>	GPI	3	
Trigger 4	<input type="checkbox"/>	Hotkey	4 D02		or	<input type="checkbox"/>	GPI	4	
Trigger 5	<input type="checkbox"/>	Hotkey	5 Limiter!		or	<input type="checkbox"/>	GPI	5	
Trigger 6	<input type="checkbox"/>	Hotkey	6 Nicerizer		or	<input type="checkbox"/>	GPI	6	
Trigger 7	<input type="checkbox"/>	Hotkey	7 [Panic 1		or	<input type="checkbox"/>	GPI	7	
Trigger 8	<input type="checkbox"/>	Hotkey	8 Panic 2]		or	<input type="checkbox"/>	GPI	8	
48 kHz Fail	<input checked="" type="checkbox"/>	Parameter	1 Sample Rate 48kHz Trigger		or	<input type="checkbox"/>	-		
48 kHz OK	<input type="checkbox"/>	Parameter	1 Sample Rate 48kHz Trigger		or	<input type="checkbox"/>	-		
AES Input Fail	<input type="checkbox"/>	Parameter	2 AES Input Fail		or	<input type="checkbox"/>	-		
Fail Over Active	<input type="checkbox"/>	Parameter	3 Fail Over Alarm		or	<input type="checkbox"/>	-		
Sync Lost	<input checked="" type="checkbox"/>	Sync Lock	System Sync Lock		or	<input type="checkbox"/>	-		
Bypass Active	<input type="checkbox"/>	Bypass	Processing Bypass		or	<input type="checkbox"/>	Bypass	AES Bypass	
Panic	<input type="checkbox"/>	Hotkey	7 [Panic 1		and	<input type="checkbox"/>	Hotkey	8 Panic 2]	

- Trigger** Here you define a label for the trigger (e.g. "48kHz fail").
- Source 1** The first source of a logical combination of two trigger sources.
- Invert** [ON / OFF]
If the type of trigger allows an inverted operation it can be defined here.
- Type** [GPI / Hot Key, Network / Parameter / Event active / Trigger effective/ Bypass / Sync Lock]
- Source** [e.g. for GPIs it will be 1 ... 8]
It acts like an index for the respective trigger type (in case of GPIs it is the physical GPI number or in case of X*AP Hotkeys the key number (1 of 8)).
- Logic** Kind of logical operation [and, or, xor].
- Source 2** Second source for the logical combination of two trigger sources.
If only one source exists, you may leave it unassigned [-].

Now since we have prepared the ignition, we must load the bombs ...
I.e. we define what will happen if ... so we must define the events.

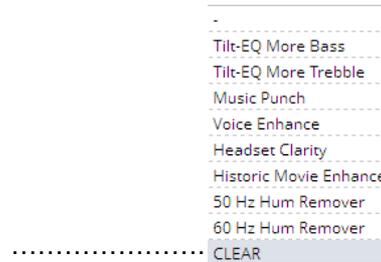
Setup GUI – EVENTS – Events – **Preset Events**

A preset event is a group of presets you may load at one occasion to the On Air parameters of function blocks. When executing such an event you may for example change the Equalizer, compressor and EQ settings, by simply assigning the individual preset of your choice to the processing block or the system, to an interface, to the routing, to the program path or even to the monitor output:



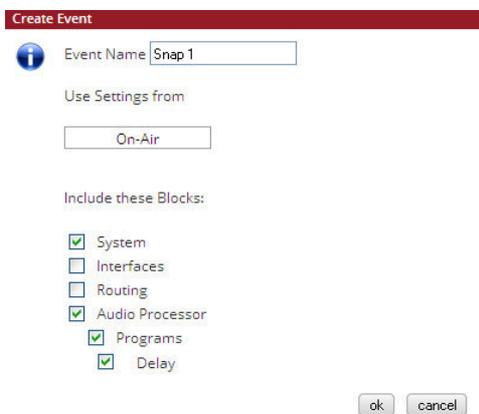
Some blocks have empty selection drop downs because nothing is selected while others don't have such drop down at all because no presets exists there.

As an example you see here the pull down list of all existing presets from the Filter – Spectral Signature section:



Here you have the possibility to reconfigure the **D*AP4** completely, partially or to change a few audio parameters marginally. Reconfigure also means you can use this part to create Events which act like **snap shots!**

To create a new preset event you must press **<create event>**:



Event name

[Snap 1]
A unique name to address this preset event later in the action manager.

Use Settings from On Air

[ON Air / Existing Event / Empty]

The events manager will **copy** all **On Air parameters** to **new** presets in **all** function blocks (that have been selected via the "Include" check boxes).

Existing Event

you will be asked from which event it originated. The presets of the selected event will be copied to the new event and may be marginally tuned afterwards to form a slightly different event.

Empty

Creates a set of empty fields where you may select the preset of your choice for the respective function block or leave it empty if no changes are needed ...

Important Note! This is the way to create your own **snapshot**. The new presets will be automatically given the name of this event! So be careful to select meaningful names. You will find them later on in all your function blocks!

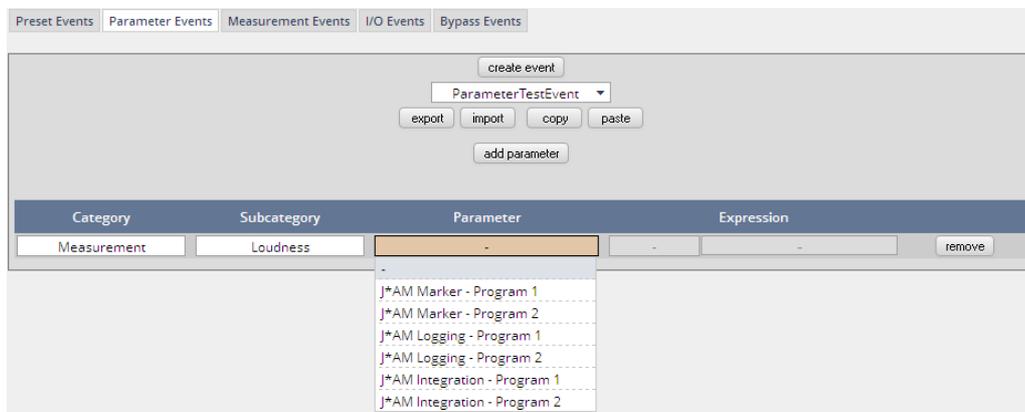
Include these Blocks

[System / Interface / Routing / Audio Processor / Voice 1 / Voice 2 / Program / Monitor]

You can tell the event manager which function blocks must be included in this event (or not).

Setup GUI – EVENTS – Events – **Parameter Events**

Parameter events are designed to change parameters when executing an action. You define the parameter here:



This example shows the possible parameters to control the loudness measurement of the loudness logger of the Junger Audio application manager **J*AM**.

Category [System / Audio Processor / Measurement]

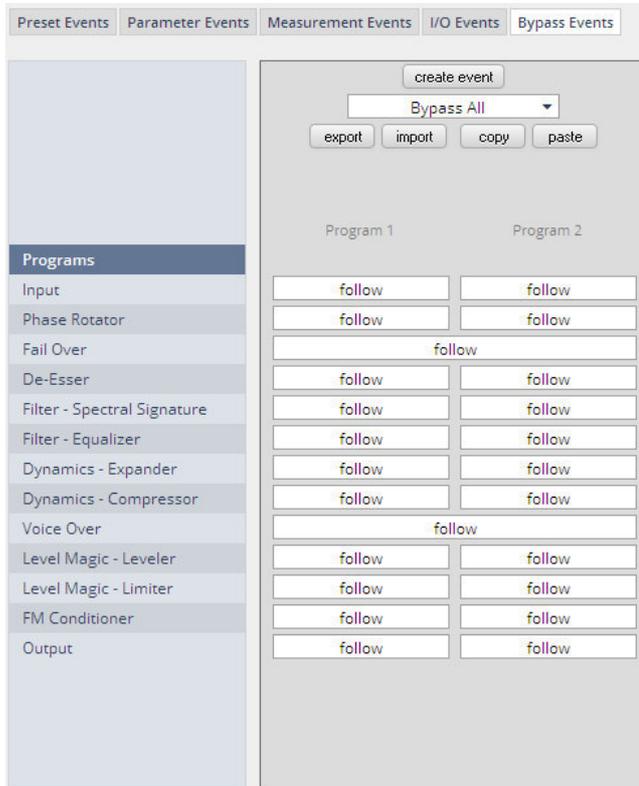
Subcategory [in case of Category = System >> Setup]
 [in case of Category = Audio Processor >> Voice Over]
 [in case of Category = Measurement >> Loudness]

Parameter depending on the pre-selected Subcategory you may define the relevant parameter which you want to set (e.g. to control the integration measurement of the J*AM).

Expression If applicable, the value of the parameter that will be set if the parameter event is triggered during an event action (e.g. set a marker for the J*AM live plot)

Setup GUI – EVENTS – Events – **Bypass Events**

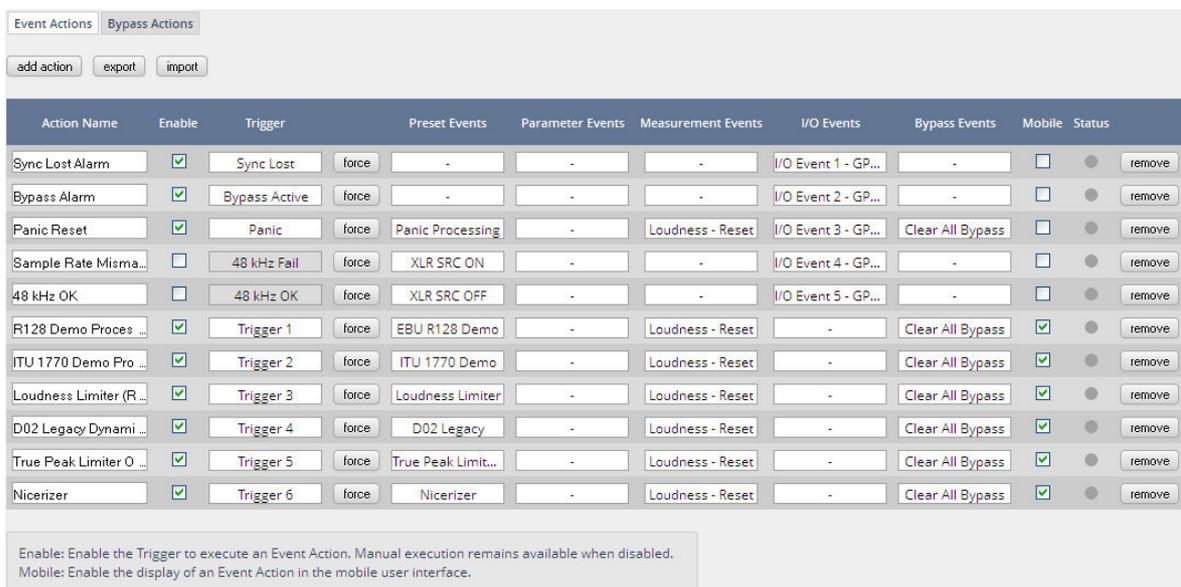
The **D*AP4** allows you to bypass some or all of the function blocks. This can be used for A/B comparison for all or for a subset of function blocks:



Now we have loaded the bombs, let's ignite them ...

Setup GUI – EVENTS – Actions – **Event Actions**

This is the point where all previously set sub functions will be combined:



Here you create the action!

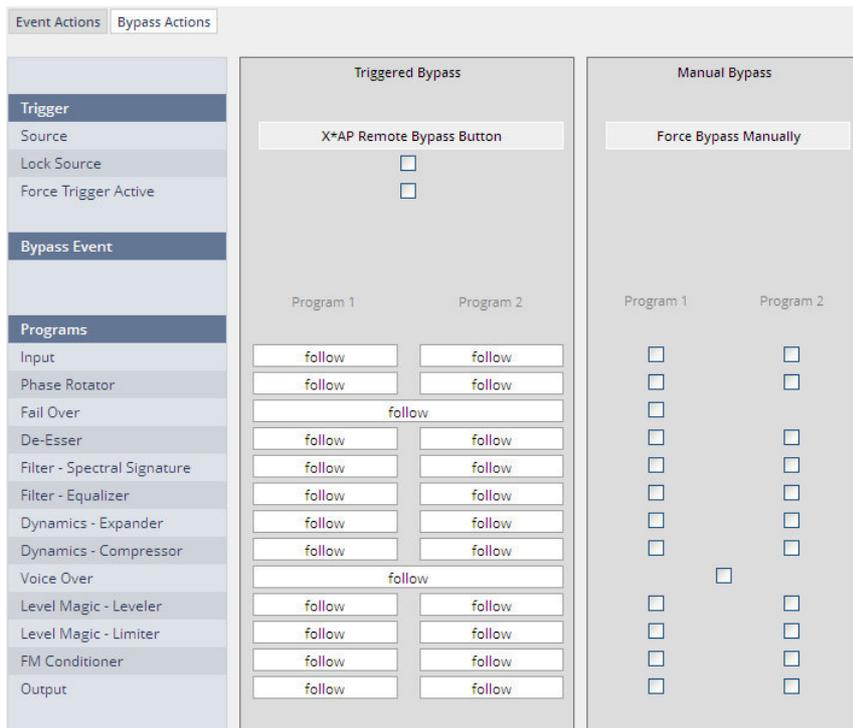
You must give the action a name, select a trigger (from the Trigger Equation) and select the respective Preset - Measurement - I/O and/or Bypass event.

Done ...

Setup GUI – EVENTS – Actions – **Bypass Actions**

The bypass action is bound to the **<BYPASS>** button of the **X*AP RM1** remote panel. You must simply select "follow" or "-" for the setup field.

But it also allows you to turn the bypass of one, some or all function blocks on. You must simply tick the check boxes in the right hand panel:



Trigger

Source

The X*AP and / or manual selection

Lock Source

[ON / OFF]

The **X*AP RM1** remote panel **<BYPASS>** button may be disabled / enabled here.

Force Trigger Active

[ON / OFF]

Force the bypass function from the GUI instead of the **X*AP RM1** remote panel **<BYPASS>** button.

Bypass Event

Programs

Audio processing blocks of the respective program path.

Input ... Output

[follow / -]

Technical Data - 4 Channel Audio Processor [D*AP4 LM EDITION, D*AP4 FLX]

General	<ul style="list-style-type: none"> • 4 channel audio processor (2 stereo programs) • Expandable by hard and software options 		
Audio Sample Rate	44.1, 48, 88.2, 96kHz, (32 ... 196kHz @ input with SRC) ±150ppm sync input capture, ±25ppm master-sync stability		
AES/EBU Inputs	Relevant specifications comply with AES3-X-2009, IEC 60985 and AES11-2009		
	4 channels (2 stereo inputs), 2 XLR-3 connectors and 2 BNC connectors, alternative inputs - user settable input selector		
	24bits, transparent forwarding of PCM and compressed audio (w/o SRC) 24bits, PCM, sample rate converter (SRC) activated		
	Impedance	110Ohm differential (XLR-3) 75Ohm single-ended (BNC)	
	Input level	0.3 ... 5Vpp @ 110Ohm differential (XLR-3) 0.3 ... 5Vpp @ 75Ohm single-ended (BNC)	
	Sample Rate Converter (SRC)	THD+N -120dB @ 0dBFS, 1kHz Latency < 0.3ms	
AES/EBU Outputs	Relevant specifications comply with AES3-X-2009, IEC 60985 and AES11-2009		
	4 channels (2 stereo outputs), 2 XLR-3 connectors and 2 BNC connectors, both connector types carry the same signal		
	24bits, transparent forwarding of PCM and compressed audio		
	Impedance	110Ohm differential (XLR-3) 75Ohm single-ended (BNC)	
	Output voltage	3Vpp (typ.) @ 110Ohm differential (XLR-3) 1Vpp (typ.) @ 75Ohm single-ended (BNC)	
	Power fail relay bypass between AES/EBU inputs and outputs (can be deactivated by jumper)		
Sync Input	Multi-standard synchronization interface for AES/EBU, wordclock or video-sync (black burst, tri level), complies with AES11-2009 and relevant audio or video standards		
	Connector type	BNC	
	AES/EBU input	0.3 ... 5Vpp @ 75Ohm single-ended	
	Wordclock input	1 ... 5Vpp @ 75Ohm single-ended	
	Video-sync input	1Vpp (nom.) @ 75Ohm single-ended	
		Rates supported: 23.975, 24, 24.975, 25, 29.97, 30, 49.95, 50, 59.94, 60fps (SD and HD)	
	On-board audio ports and master-sync capable option boards may also be selectable as sync source.		
Sync Output	Word clock output, complies with AES11-2009		
	Connector type	BNC	

	Wordclock output	2.4V (typ.) @ 75Ohm single-ended
Network Interface	RJ45 connector, 10/100Mbit Ethernet auto sense, full duplex, auto MDI/X	
USB Interface	USB 2.0 connector to internal console interface	
GPI Signals	8 general purpose inputs (GPI), divided into 2 groups with separate common signal, isolated	
	Connector type	D-Sub25 connector female, same for GPO
	Input conditions	3 ... 24Vdc, < 5mA
	Auxiliary supply	5V (nom.), 200mA (max.), isolated
GPO Signals	8 general purpose outputs (GPO), SPST, divided into 2 groups with separate common signal, isolated	
	Connector type	D-Sub25 connector female, same for GPI
	Output conditions	24Vac/dc (max.), 120mA (max.)
Expansion Slot	1 general purpose expansion slot for option boards	
Power Supply	Dual power supply, automatic fail over, 85 ... 264Vac, 50 ... 60Hz, 58W (max.)	
Environmental	Operating temperature 0 ... 50°C, fan cooled, Non-operating -20 ... 70°C, Humidity < 90%, non-condensing	
Physical	19", 1 RU, 27 cm depth, net weight ca. 5 kg, shipping weight ca. 7.5 kg	

Technical Data – Option Board SDI I/O (3G/HD/SD) [O_DAP_SDI_a]

Standards	Video complies with SMPTE 424/425M (3G, Level A and B), SMPTE 292M (HD) or SMPTE 259M (SD). Automatic format detection. Audio embedding and de-embedding complies with SMPTE 299M (3G, HD) or SMPTE 272M-AC (SD). Metadata embedding and de-embedding complies with SMPTE 2020-2.	
Video Data Rate	2970/2967Mbps (3G), 1485/1483.5Mbps (HD), 270Mbps (SD)	
Video Formats	1080p23.975, 24, 25, 29.97, 30, 50, 59.94, 60 1080i50, 59.94, 60 720p23.975, 24, 25, 29.97, 30, 50, 59.94, 60 625i50, 525i59.94, ...	
Video Delay	User selectable 0 ... 15frames, can be disabled	
Audio	24bits, transparent forwarding of PCM and compressed audio	
Audio Channels	16 inputs and 16 outputs (4 groups with 4 channels each)	
Audio Sample Rate	48kHz (SDI compliant)	
Audio Delay	Embedder audio delay selectable 0 ... 320 ms per channel	
Metadata (RDD6)	1 channel input and 1 channel output, SDID selectable	
BNC Input	Impedance	75Ohm
	Return loss	> 15dB, 5 ... 1485MHz > 10dB, 1485 ... 2970MHz
	Cable length (max.)	250m @ SD for Belden 1694A cable 230m @ HD for Belden 1694A cable 140m @ 3G for Belden 1694A cable

	Jitter tolerance	> 0.7UI (Alignment)
BNC Output	Impedance	75Ohm
	Output voltage	0.8Vpp (typ.)
	Return loss	> 15dB, 5 ... 1485MHz > 10dB, 1485 ... 2970MHz
	Output jitter	< 0.2UI (Alignment), < 0.5UI (Timing)
Audio Latency	Input to Output	Embedder and de-embedder combined HD, 3G < 0.6ms SD typ. 1.5ms (< 2 ms)
General Features	<ul style="list-style-type: none"> • Power fail relay bypass (may be activated via GUI) • Lip-Sync compensation for processed and non-processed audio signals • Dedicated routing for non-processed channels, all channels (max. 16) can be routed to/from the device or looped through • Test pattern generator • Master-sync capable • ITU-R BT.1685 / ARIB STD-B39 metadata support 	

Technical Data – Option Board Analog Out [**O_DAP_8DA_a**]

Audio	24bit D/A-converter	
Audio Channels	8 output channels (e.g. for speakers)	
Audio Sample Rate	44.1, 48, 88.2, 96kHz	
Analog Outputs	8 channels	
	Connector type	D-Sub25 connector female
	Output Level (max.) (0dBFS equiv.)	0 ... 24dBu, adjustable in 0.5dB steps
	Impedance	50Ohm (typ.), differential
	THD+N	-91dB @ 0dBFS = 15dBu, 1kHz
	Dynamic range	> 103dB (RMS)
	Crosstalk attenuation	> 103dB @ 0dBFS = 15dBu, 1kHz
	Frequency response	20Hz ... 22kHz (< ±0.3dB) @ 48kHz 20Hz ... 43kHz (< ±0.3dB) @ 96kHz
General Features	<ul style="list-style-type: none"> • Power fail glitch prevention • Balanced analog outputs • Electrical isolation between outputs and device 	

Technical Data – Option Board Analog I/O [O_DAP_ADDA_a]

Audio	24bit sigma-delta A/D-converter, 24bit D/A-converter	
Audio Channels	4 input channels, 4 output channels	
Audio Sample Rate	44.1, 48kHz	
Analog Inputs	4 channels	
	Connector type	D-Sub25 connector female, same for outputs
	Input Level (max.) (0dBFS equiv.)	0 ... 24dBu, adjustable in 0.5dB steps
	Impedance	20kOhm (typ.), differential
	THD+N	-93dB @ 0dBFS = 15dBu, 1kHz
	Dynamic range	> 110dB (RMS)
	Crosstalk attenuation	> 93dB @ 0dBFS = 15dBu, 1kHz
	CMRR	> 71dB @ 0dBFS = 15dBu, 1kHz
	Frequency response	20Hz ... 22kHz (< ±0.1dB) @ 48kHz 20Hz ... 43kHz (< ±0.1dB) @ 96kHz
Analog Outputs	4 channels	
	Connector type	D-Sub25 connector female, same for inputs
	Output Level (max.) (0dBFS equiv.)	0 ... 24dBu, adjustable in 0.5dB steps
	Impedance	50Ohm (typ.), differential
	THD+N	-91dB @ 0dBFS = 15dBu, 1kHz
	Dynamic range	> 103dB (RMS)
	Crosstalk attenuation	> 103dB @ 0dBFS = 15dBu, 1kHz
	Frequency response	20Hz ... 22kHz (< ±0.3dB) @ 48kHz 20Hz ... 43kHz (< ±0.3dB) @ 96kHz
General Features	<ul style="list-style-type: none"> • Power fail relay bypass between inputs and outputs • Balanced analog inputs and outputs • Electrical isolation between inputs, outputs and device 	

Technical Data – Option Board AES/EBU I/O [O_DAP_AES_a]

Standards	Relevant specifications comply with AES3-X-2009, IEC 60985 and AES11-2009	
Audio	24bits, transparent forwarding of PCM and compressed audio (w/o SRC) 24bits, PCM, sample rate converter (SRC) activated	
Audio Sample Rate	44.1, 48, 88.2, 96kHz, (32 ... 196kHz @ inputs with SRC)	
Inputs	8 channels (4 stereo inputs)	
	Connector type	D-Sub25 connector female, same for outputs
	Impedance	110Ohm or 75Ohm, jumper selectable (110Ohm default)
	Input level	0.3 ... 5Vpp @ 110Ohm differential 0.3 ... 5Vpp @ 75Ohm single-ended
	Sample Rate Converter (SRC)	THD+N -120dB @ 0dBFS, 1kHz Latency < 0.3 ms
Outputs	8 channels (4 stereo outputs)	
	Connector type	D-Sub25 connector female, same for inputs
	Impedance	110Ohm or 75Ohm, jumper selectable (110Ohm default)
	Output voltage	3Vpp (typ.) @ 110Ohm differential 1Vpp (typ.) @ 75Ohm single-ended
General Features	<ul style="list-style-type: none"> • Power fail relay bypass (can be deactivated by jumper) • Input sample rate converters (SRC) • Electrical isolation between inputs, outputs and device (if configured for differential mode, 110Ohm) • AES3 channel status management, non-audio detection • Master-sync capable 	

Technical Data – Option Board MADI I/O, BNC [O_DAP_MB_a]

Standards	Relevant specifications comply with AES10-2008 and AES11-2009.	
Audio	24bits, transparent forwarding of PCM and compressed audio	
Audio Sample Rate	44.1, 48, 88.2, 96kHz, (88.2, 96kHz short framing)	
BNC Input	64/56 channels @ 44.1 and 48kHz, 32/28 @ 88.2 and 96kHz Processable by D*AP8: 16 channels @ 44.1, 48kHz Processable by D*AP4: 8 channels @ 44.1, 48, 88.2, 96kHz	
	Impedance	75Ohm
	Input level	0.15 ... 0.8Vpp @ 75Ohm
	Cable length (max.)	150m (Belden 1694A)
BNC Output	64/56 channels @ 44.1 and 48kHz, 32/28 @ 88.2 and 96kHz Processable by D*AP8: 16 channels @ 44.1, 48kHz Processable by D*AP4: 8 channels @ 44.1, 48, 88.2, 96kHz	
	Impedance	75Ohm

	Output voltage	0.6Vpp (typ.) @ 75Ohm
General Features	<ul style="list-style-type: none"> • Input cable equalizer for extended range and robustness • Reference grade word clock recovery, master-sync capable • Dedicated routing for non-processed channels, all channels (max. 64) can be routed to/from the device or looped through • AES3 channel status management, non-audio detection 	

Technical Data – Option Board MADI I/O, Optical [O_DAP_MO_MM_a, O_DAP_MO_SM_a]

Standards	Relevant specifications comply with AES10-2008 and AES11-2009.	
Audio	24bits, transparent forwarding of PCM and compressed audio	
Audio Sample Rate	44.1, 48, 88.2, 96kHz, (88.2, 96kHz short framing)	
Optical Input, LC	64/56 channels @ 44.1 and 48kHz, 32/28 @ 88.2 and 96kHz Processable by D*AP8: 16 channels @ 44.1, 48kHz Processable by D*AP4: 8 channels @ 44.1, 48, 88.2, 96kHz	
	Connector type	LC (IEC 61754-20)
	Center wavelength	1310nm (typ.), 1270 ... 1360nm
	Input optical power	[O_DAP_MO_MM_a]: -31 ... -8dBm, OM2 multimode (50/125µm) [O_DAP_MO_SM_a]: -23 ... -8dBm, singlemode (9/125µm) (standard values, others on request)
	Cable length (max.)	[O_DAP_MO_MM_a]: 1.5km, OM2 multimode [O_DAP_MO_SM_a]: 2km, singlemode (standard values, others on request)
Optical Output, LC	64/56 channels @ 44.1 and 48kHz, 32/28 @ 88.2 and 96kHz Processable by D*AP8: 16 channels @ 44.1, 48kHz Processable by D*AP4: 8 channels @ 44.1, 48, 88.2, 96kHz	
	Connector type	LC (IEC 61754-20)
	Center wavelength	1310nm (typ.), 1270 ... 1360nm
	Output optical power	[O_DAP_MO_MM_a]: -23 ... -14dBm, OM2 multimode (50/125µm) [O_DAP_MO_SM_a]: -15 ... -8dBm, singlemode (9/125µm) (standard values, others on request)
BNC Output	Optical and BNC output carry the same signal.	
	Impedance	75Ohm
	Output voltage	0.6Vpp (typ.) @ 75Ohm
General Features	<ul style="list-style-type: none"> • Field-replaceable optical module (SFP) • Reference grade word clock recovery, master-sync capable • Dedicated routing for non-processed channels, all channels (max. 64) can be routed to/from the device or looped through • AES3 channel status management, non-audio detection • Parallel outputs (BNC/LC) for media conversion 	

Technical Data – Option Board Audio-over-IP DANTE™ I/O [O_DAP_DANTE_a]

Standards	Audio-over-IP by Dante™ Digital Audio Networking Standard
Audio	24bits, transparent forwarding of PCM and compressed audio
Audio Sample Rate	44.1, 48, 88.2, 96kHz
Inputs and Outputs	2 x Gigabit Ethernet RJ45 connectors (100M/1Gbit), primary and secondary port
Inputs	Processable by D*AP8: 16 channels @ 44.1, 48kHz Processable by D*AP4: 8 channels @ 44.1, 48, 88.2, 96kHz
Outputs	Processable by D*AP8: 16 channels @ 44.1, 48kHz Processable by D*AP4: 8 channels @ 44.1, 48, 88.2, 96kHz
General Features	<ul style="list-style-type: none"> • AES67 compliant • Network master-sync can be provided by D*AP device • Master-sync capable (for D*AP device) • Non-audio detection for input channels • Glitch-free Dante™ audio redundancy using dual Ethernet networks

Technical Data - Rear Connectors - **pin assignment**

8x GPI/O

Mic / Line IN

connector:	GPI/O
female	25-pin D-Sub
1	GPI_1, 2, 3, 4 common
2	GPI_1
3	GPI_2
4	GPI_3
5	GPI_4
6	GPI_5, 6, 7, 8 common
7	GPI_5
8	GPI_6
9	GPI_7
10	GPI_8
11	
12	
13	Isolated 5V +
14	GPO_1, 2, 3, 4 common
15	GPO_1
16	GPO_2
17	GPO_3
18	GPO_4
19	GPO_5, 6, 7, 8 common
20	GPO_5
21	GPO_6
22	GPO_7
23	GPO_8
24	Isolated 5V -
25	Isolated 5V -

connector:	Mic / Line input
female	XLR
1	GND
2	IN +
3	IN -
Shield	Virtual GND

Technical Data - Optional Interface Modules – pin assignment

4x analog I/O [O_DAP_ADDA_a]

4x AES I/O [O_DAP_AES_a]

8x analog out [O_DAP_8DA_a]

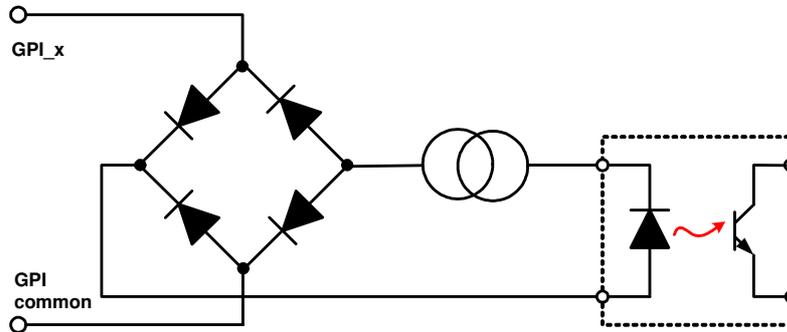
connector:	4 x analog I/O
female	25-pin D-Sub
1	OUT-4 +
2	GND
3	OUT-3 -
4	OUT-2 +
5	GND
6	OUT-1 -
7	IN-4 +
8	GND
9	IN-3 -
10	IN-2 +
11	GND
12	IN-1 -
13	
14	OUT-4 -
15	OUT-3 +
16	GND
17	OUT-2 -
18	Out-1 +
19	GND
20	IN-4 -
21	IN-3 +
22	GND
23	IN-2 -
24	IN-1 +
25	GND

connector:	4x AES I/O
female	25-pin D-Sub
1	OUT-4 +
2	GND
3	OUT-3 -
4	OUT-2 +
5	GND
6	OUT-1 -
7	IN-4 +
8	GND
9	IN-3 -
10	IN-2 +
11	GND
12	IN-1 -
13	
14	OUT-4 -
15	OUT-3 +
16	GND
17	OUT-2 -
18	OUT-1 +
19	GND
20	IN-4 -
21	IN-3 +
22	GND
23	IN-2 -
24	IN-1 +
25	GND

connector:	8 x analog out
female	25-pin D-Sub
1	OUT-8 +
2	GND
3	OUT-7 -
4	OUT-6 +
5	GND
6	OUT-5 -
7	OUT-4 +
8	GND
9	OUT-3 -
10	OUT-2 +
11	GND
12	OUT-1 -
13	
14	OUT-8 -
15	OUT-7 +
16	GND
17	OUT-6 -
18	OUT-5 +
19	GND
20	OUT-4 -
21	OUT-3 +
22	GND
23	OUT-2 -
24	OUT-1 +
25	GND

Technical Data – GPI wiring

The device offers a unique circuitry to save GPI setups from hum and noise influence in complex installations. Here the principle circuit of one of the 8 GPI inputs:

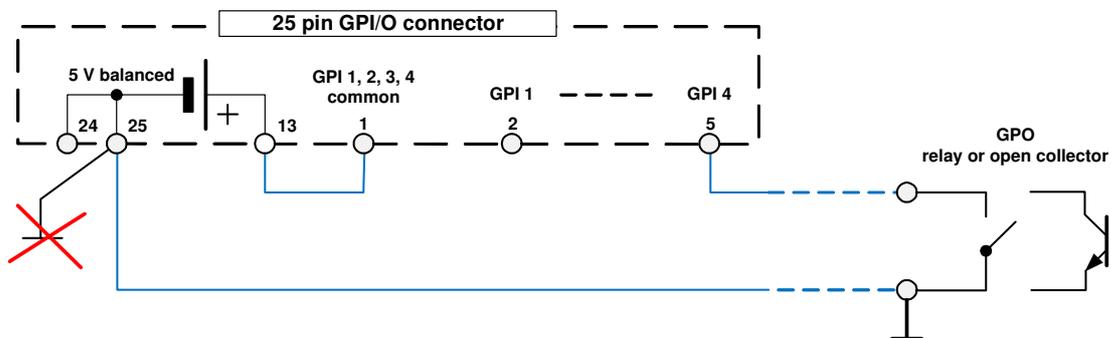


At the GPI input is a **bridge rectifier** i.e. you do **not** need to care about the polarity of the input voltage. A **constant current source** in line with the **optical coupler** limits the current. You must simply provide a voltage in the range from 5 V to 30 V to activate a GPI.

If you have open collector outputs or simple relay closures as the driving GPOs (this technique is commonly known as "low active" and will be found in most legacy equipment), you must wire up an auxiliary voltage supply.

The device provides such auxiliary power supply. It offers a balanced 5 V source that you can imagine as a battery.

Here an example how to wire up GPI #4:



We strongly recommend to spend a wire for ground connection instead of using the chassis common grounds of an installation.

Safety Information

Electrical

- Safety classification:** Class 1 – grounded product / Schutzklasse 1
Corresponding to EN 60065:2002
- Power connection:** The device must be connected to a power socket that provides a protective earthing conductor.
- Power switch:** The power switch is a toggle switch placed at the rear of the device. The ON / OFF position is indicated by engravings [I] / [o] on the lever. It must be reached without difficulty.
The devices may be equipped with dual power supply, in this case it will have two power cords and switches. You must inform yourself about the location and assignment of the switches.
- Water protection:** The device must not be exposed to splash or dripping water. It is permitted to place a container filled with liquids (e.g. vases) on top of the device.

Service safety

- Only qualified personnel should perform service procedures.
- Do not service alone:** Do not perform internal service or adjustments of the device unless another person capable of rendering first aid and resuscitation is present.
- Disconnect power:** To avoid electrical shock, switch off the device power, then disconnect the power cord from the mains power. Do not block the power cord; it must remain accessible to the user at all times

To avoid fire or personal injury

- Mounting:** It must be placed on a flat surface or must be mounted into an 19" rack. It is recommended to use metal brackets (sheet steel angle) to support the device.
- Provide proper Ventilation** this case and if the device has a built in fan, a gap of at least 1cm must be left between the device edge and the steel angle. It is highly recommended to leave a gap of at least 1RU above and below the device.
- Use proper power cord** Use only the power cord specified for this product and certified for the country of use.
- Do not operate without covers** Do not operate this product with covers or panels removed.
- Do not operate with suspected failures** If you suspect that there is damage to this product, have it inspected by qualified service personnel.
- Risk of explosion:** The device contains a lithium battery. If replaced incorrectly or by a different or inadequate type an explosion may occur.

Warranty

Standard Junger Audio one-year warranty on parts and labor.

Specifications are subject to change without notice

Headquarters

Jünger Audio GmbH
Sales · Service & Support
Justus-von-Liebig-Str. 7
12489 Berlin, Germany
phone +49 30 677 721 0
fax +49 30 677 721 46
support@jungeraudio.com
sales@jungeraudio.com
www.jungeraudio.com

Jünger Audio Asia

Junger Audio Pte. Ltd.
Sales · Service & Support
33, Ubi Ave 3, no. 06-62 Vertex
408868 Singapore
phone +65 6509 8171
fax +65 6748 1740
asia@jungeraudio.com

Jünger Audio (China Office)

Junger (Beijing) Technology Ltd.
Sales · Service & Support
Unit 068, 2nd Floor, No. 38 GuangQu Road,
ChaoYang District, 100021 Beijing, China
phone +86 137 0102 6664
china@jungeraudio.com

Jünger Audio USA

Service & Support
2100 B2 Walsh Ave
Santa Clara, CA 95050-2590, USA
phone +1 408 330 9215
usa@jungeraudio.com



 www.facebook.com/jungeraudio

 www.twitter.com/jungeraudio