D*AP8

Digital Audio Processor

D*AP8 FLX
D*AP8 TAP Edition
D*AP8 CODEC Edition

Manual

Hardware features

• **D*AP8**	1RU / 19" generic compact 8 channel processing unit
• **X*AP RM1**	**optional** 1RU remote panel
• **Dolby decoder**	**optional** built in Dolby D/D+/E decoder incl. metadata emulation
• **Dolby encoder**	**optional** built in Dolby D/D+/AAC/HE-AAC **or** Dolby E encoder
• **Dolby metadata I/O**	two 9-pin D-Sub connectors (RS485)
• **4x AES (BNC) I/O + SRC**	on board AES I/Os with relay bypass and (selectable) SRC per input
• **Two interface slots**	expansion slots for **optional** I/O boards: 3-G/HD/SD-SDI, MADI, Dante, 4x AES I/O, 4Ch Analog I/O, 8 Ch Analog Out
• **RJ45 network connector**	100BaseT full duplex Ethernet interface
• **USB connector**	built in USB < > serial adapter to access the service port
• **8x GPI/O**	balanced inputs and SSR contacts on a 25pin Sub-D
• **Aux power supply**	isolated 5V supply for external GPI/O wiring
• **External sync IN**	BNC input (Word Clock, AES, Black Burst, Tri-Level)
• **Sync OUT**	BNC Word Clock output

Software features of D*AP8

• **TP limiter**	Junger Audio true peak limiter control algorithm
• **LevelMagic II™**	**optional** loudness management according to ITU BS.1770-1/-2/-3/-4 EBU R128, ATSC A/85, ARIB TR-B32, Free TV OP-59, Portaria 354
• **Dynamic filter / EQ**	**optional** SPECTRAL SIGNATURE™ dynamic filter and 5 band parametric EQ
• **Dynamics**	**optional** compressor, expander / gate
• **Fail over Upmix**	**optional** automatic fail over / optional 5.1 Upmix
• **Voice over**	**optional** 5.1 or stereo voice over from program input
• **Dolby metadata emulation**	**optional** Dolby metadata emulation
• **Dolby metadata generator**	generates RDD6 complied metadata
• **Loudness measurement**	in reference to the respective standard
• **SNMP agent**	SNMP v1, see D*AP8-MIB
• **Remote control**	**X*AP RM1** remote panel, EmBER plus protocol and legacy GPI/Os

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Introduction

The **D*AP8** is a generic platform with several software options that can be accomplished by an optional **X*AP RM1** remote panel. The bundle is designed to allow the operator a direct access to major functions and important parameters.

It is available as **D*AP8 TAP EDITION** that replaces the former **T*AP** (Television Audio Processor).
While the **D*AP8 FLX EDITION** can be combined to the customer needs at the moment of ordering but can later be upgraded in the field with more options. Similar applies to the **D*AP8 CODEC** edition that is focused on **Dolby** encoding / decoding. All three editions are covered by this manual and the MEI (Multi Edition Image) firmware image. Depending on the licensed features you may not have all functions available.
A special version, the **D*AP8 MAP EDITION** is also available to suite the special needs of monitoring. It is the successor of the **Dolby DP570** Multi Channel Audio Tool and has its own manual.

For level and loudness measurement and logging applications the **D*AP8** may be used as a measurement box that sits close to the signal sources while measurement data will be streamed over the network to a PC for live display and/or storing of such data.

The heard of the **D*AP8** is a sophisticated audio processor. It renders all functions of the audio blocks, as well as level and loudness measurements.

A comprehensive **Dolby** subsystem including a stand alone metadata generator is provided for optional decoding, emulation and encoding. The influence of metadata on PCM audio signals can be monitored either directly from the monitoring section of a mixing console or from a decoded Dolby E stream. It allows you to hear how the metadata will influence the listening experience on the customers side without insertion of a consumer format encoder / decoder. The metadata emulation part incorporates a **Dolby** stream decoder. An optional **Dolby Digital**/**Digital plus** or a **Dolby-E** encoder can be added to the device.

The four **AES3id I/Os** on the motherboard may be complemented by a variety of interface modules that can be installed as an option into the **D*AP8** interface slots.

Comprehensive routing set-ups allow almost every signal flow from hardware inputs, from and to optional **Dolby** decoder / encoder, from the audio processor itself to hardware outputs as well as the metadata I/Os, the metadata generator and the metadata emulator.

Routing paths, the enabling and disabling of audio processing blocks and the setting of processing parameters can be pre-configured by individual **presets** dedicated to each function block. The content of the **presets** can be displayed and edited off-line while the device is on duty. These **presets** may either be recalled on demand by the operator via the GUI, the **X*AP RM1** remote panel hot keys or external systems, but may also be part of complex scenarios defined by the administrator and automatically executed by the event manager of the device or by operator intervention.

The **D*AP8** provides a web based setup GUI and can be controlled by an **X*AP RM1** remote panel that displays status and metering information and allows user intervention.

Junger Audios application manager **J*AM** is available as an add on and can be attached with a few simple clicks to the **D*AP8** so that users can log loudness data as well as display it as a live plot on a PC screen in real time or simply display level bar graphs and numeric loudness values. For production / post-production needs a built-in LTC reader will be available in the near future. So loudness logging may then be performed in regard to relative time as well as to time of day.

Completing the feature set of the **D*AP8** is the availability of an **SNMP** agent, which provides traps and status polling.

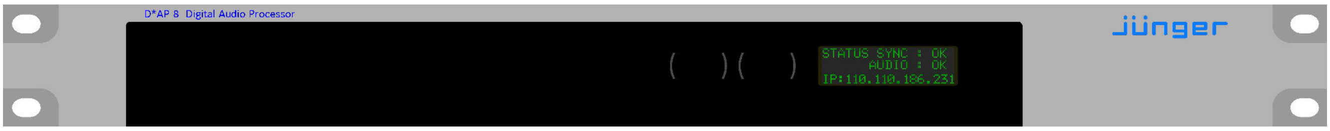
As with most advanced tools, the **D*AP8** can be driven in a variety of ways, depending on requirements and ideas of the user. These can range from simple and straightforward to quite complex set ups.
Although this manual explains the functions and general operation of the **D*AP8**, it does not give detailed scenarios because the operational needs of today's broadcasters vary so widely between organizations and their work flows and cover so many different parameters – from ingest to studio operation, from master control rooms to play-out, or even rebroadcast applications.

Junger Audio is more than happy to discuss your particular requirements with you and to convey your ideas and solutions to other users of the **Junger Audio Processors** community.

Hardware concept

The **D*AP8 editions** are based on the **D*AP8 device** that carries all relevant connectors.
An optional **X*AP RM1** remote panel can be used to control the **D*AP8**.

D*AP8 front panel view



The front panel of the **D*AP8** 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 peak meter (10x)
Level 4	OUT peak meter (10x)

The total number of display levels depends on the number of programs. For 5.1 + 2 mode (2 programs) we will have 4 more levels while for 4 x 2 (4 programs) we will have 8 more levels:

Level 5 - 8	Program 1 - 4 Out - short term loudness
Level 9 - 12	Program 1 - 4 Out - integrated loudness and integration time

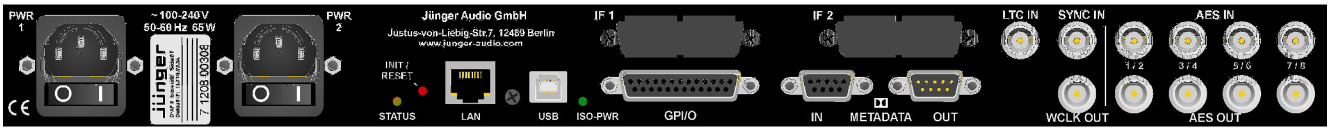
The measures of the loudness displays depend on the setup of the respective loudness mode
(see AUDIO PROCESSOR > SETUP > Loudness Mode).

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

X*AP RM1 front panel view

The **X*AP RM1** remote panel is powered by POE (Power Over Ethernet) or an external wall plug PS and designed to control multiple **D*AP8 unit**s one at a time. For details of operation see extra manual "XAP_manual_EN_140328.pdf" or later.

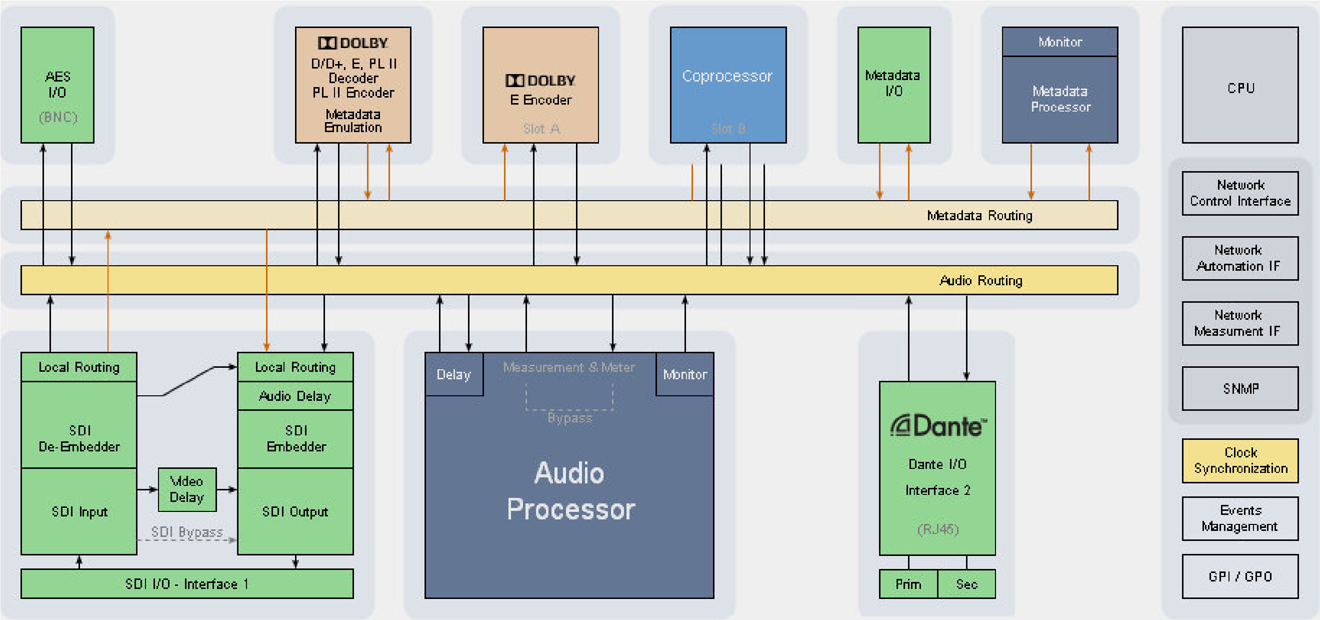
D*AP8 rear view



For fail safe operation the **D*AP8** 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	LED indicates the status of the device controller. It becomes green at the end of a successful boot process
INIT / **RESET**	pressing the **INIT** / **RESET** button briefly will warm start the device controller. Holding down the button until the **STATUS** LED flashes 5 times will initialize the **D*AP8** 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 to reach the console interface of the system controller
ISO-PWR	lights up if the isolated 5V power supply for GPI /O application is turned on
GPI/O	25pin Sub-D female connector to interface with the 8 optical isolated general purpose inputs and 8 solid state relay closure outputs
Interface 1	slot to mount one of the optional interface boards (SDI, AES, analog)
Interface 2	slot to mount one of the optional interface boards (SDI, AES, analog)
METADATA IN	9pin Sub-D female connector to receive and send Dolby® serial metadata
METADATA OUT	9pin Sub-D male connector to send Dolby® serial metadata
LTC IN	*LTC timecode input not activated jet*
SYNC IN	75Ohm BNC connector to connect with external sync sources
WCKL-OUT	75Ohm BNC connector to synchronize external devices to the **D*AP8** internal word clock
AES IN 1/2 – 7/8	AES3id inputs
AES OUT 1/2 – 7/8	AES3id outputs

Device block diagram:



The above schematic shows the principal blocks of a fully loaded **D*AP8.**

The core of the is the audio processor with 10 inputs, 8 outputs and a 2ch monitor output.

Dolby Decoding/Emulation is based on a hardware decoder option.

It also provides a **Dolby E encoder** that can be licensed

An optoional **Dolby encoder** may be fitted to provide an encoded output either in **Dolby E (D-E)** or one of the consumer formats **Dolby Digitial (D-D)** AKA **AC3**, **Dolby Digital plus (D-D+)** or **AAC** and its derivates. This will save rack space and installation cost and offers a fully integrated solution.

Four AES I/Os on the motherboard are provided for digital line operation. The respective connectors have relay bypass for power fail operation. The bypass ciruit may be disabled by internal jumpers.

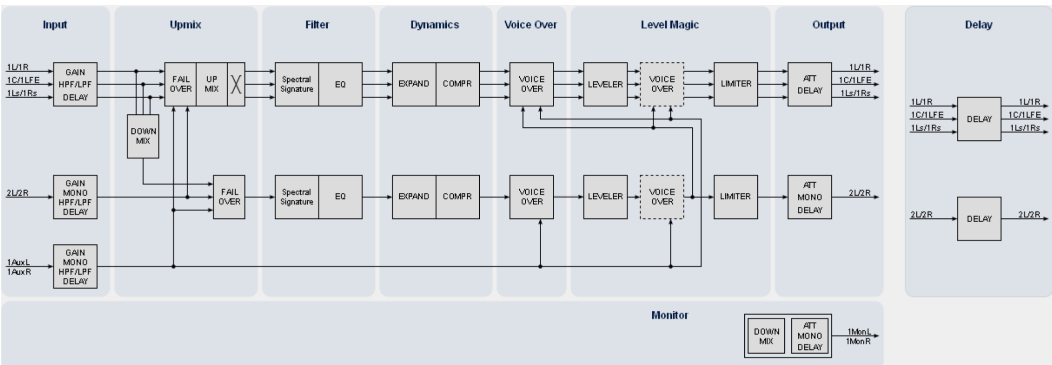
Two interface slots are provided to carry optional 3G / HD / SD-SDI, MADI, Dante, AES I/O or even analog expansion modules. It allows for extremely flexible interfacing of the **D*AP8** in TV installations.

For comprehensive metadata processing the **D*AP8** has 9-pin serial metadata I/O connectors. All metadata functions are centralized in a metadata generator. Furthermore you will have the possibility to emulate the influence of **Dolby** metadata on the audio signals for stereo or surround configurations and surround down mixes, without the need to involve an encoder and a decoder.

The sync circuit can deal with all practical formats to integrate the D*AP8 into digital facilities. Other devices may be synchronized by the word clock output of the **D*AP8 unit**. The frame reference for D-E encoding, can be shifted to align the D-E guard band.

The **D*AP8** has 8 balanced GPIs and 8 SSR closure GPOs. 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.

For additional functions like FM-processing or watermarking, one may buy a co-processor module JDSPA.
It must be routed into the signal paths as well.

Control concept

The communication between the **X*AP RM1** remote panel, the **D*AP8 unit**, setup and operating tools, 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 35.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 is also available on the device and may be explored via a SNMP monitoring system.

For **3rd party** remote control Junger highly recommends using the l-s-b **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*AP8** "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 via tabs which may have sub tabs and 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 function block of the **D*AP8**.

The presets of the **D*AP8** are persistent by nature. You are working directly on the preset memory.
I.e. you must not worry about storing such presets. The **D*AP8** does it for you.

Event concept

The **D*AP8** incorporates a sophisticated event management system.
Events may be combined to perform actions. The **D*AP8** offers these event types:

* **Preset Events** for System set-up, Interfaces, Routing, Audio Processing, Dolby related settings etc
* **Parameter Events**
* **Measurement Events** for pre-configured measurement scenarios
* **I/O Events** for GPOs
* **Bypass Events**

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

Triggers are defined by a logical combination (AND, OR, XOR) of two random trigger sources.
A 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 – **IP setup in general**

The process of installing a **D*AP8** into an **IP network** is as follows:

1. Ask the system service IT people for two unique IP addresses of the network, for the netmask and if a gateway address is necessary
2. Assign the **D*AP8** an unique IP address

   You have two choices to assign the **D*AP8 VAP** an **IP address**:
   * From the serial console interface
   * Via Web browser

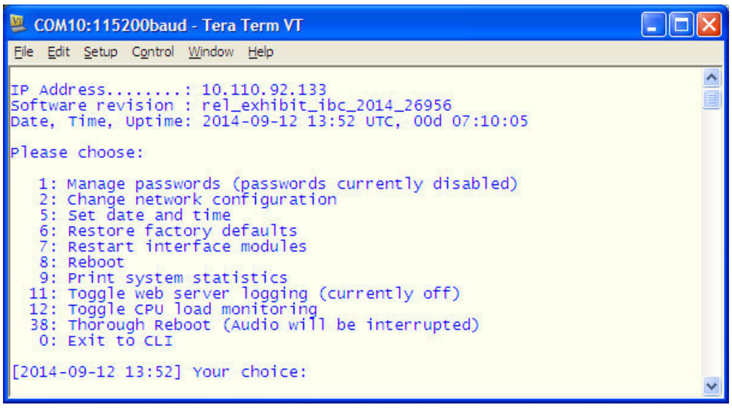
3. Assign the **X*AP RM1** remote panel a unique IP address configuration
4. Attach the **D*AP8** to the **X*AP RM1** remote panel

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 – **via console interface**

The tool to change the IP configuration of the **D*AP8** 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 ?

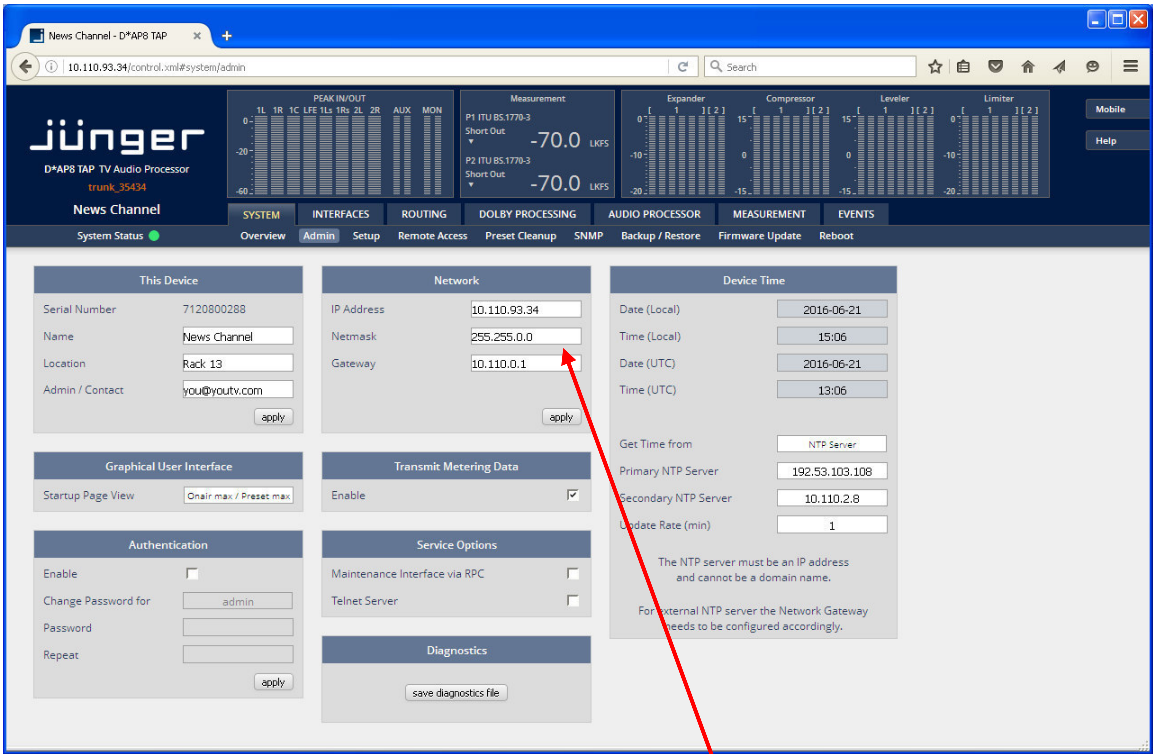
Press small **"y" <ENTER>**

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*AP8** – **via web browser**

* Read the **default IP address** printed on a label at the rear of the device.
* Set up network parameters of your PC to fit the default IP address of the **D*AP8 unit** (e.g. default IP + 1 and net mask = 255.255.0.0).
* Connect the **D*AP8** with the PC either via an Ethernet patch cable (if the PC supports Auto-MDI(X), an Ethernet cross over cable or via patch cables with a switch.
* Open a browser and type the IP address of the **D*AP8** 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:



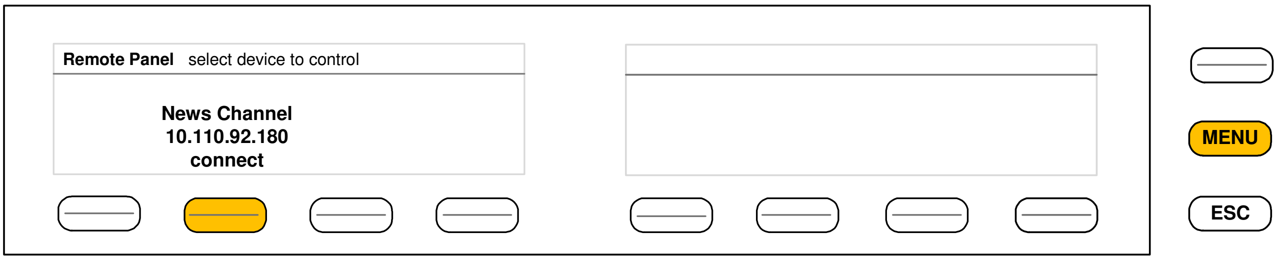
Enter the desired network configuration and press **<apply>**

Afterwards you must reboot the **D*AP8** in order to activate the new IP configuration.

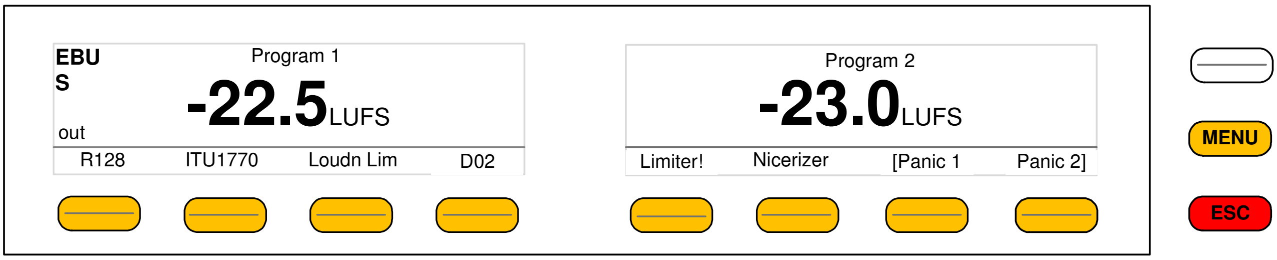
Important Note! After reboot neither the **web browser** nor the **X*AP RM1** remote panel will be able to communicate with the **D*AP8 unit**. You must fill in the new IP address in the URL field and change the **X*AP RM1** remote panel settings to attach this device with its new IP address.

Getting started – basic **X*AP RM1** remote panel operation

Power up display – may show up to four **D*AP8s** enabled for remote control for this **X*AP RM1** remote panel**.** This example has just one **D*AP8** named "NEWS Channel" attached for remote control while the status is "connect" (i.e. you may connect with that device). See **X*AP RM1** manual for details.



Pressing one of these buttons will connect with the respective **D*AP8.**
Now the **X*AP RM1** remote panel will gather all necessary information from that **D*AP8** (may take a few seconds) and open up the **main operating display**:



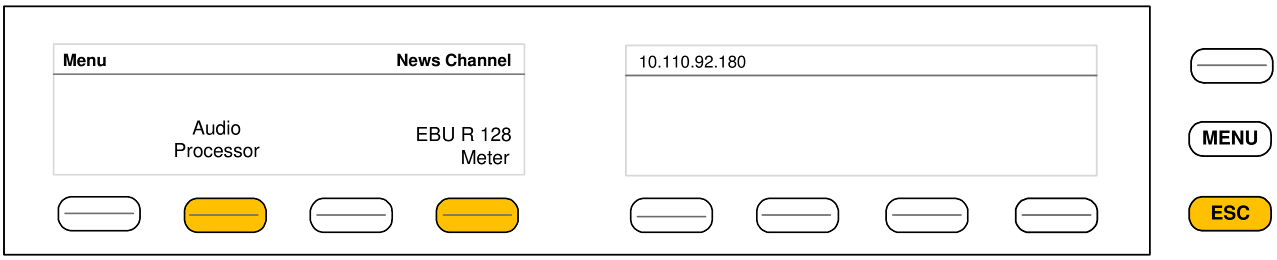
From here you may fire pre-defined hotkeys and observe the status of the volume setting.
Because this is the main operating display, the **escape** button will light up **red** to indicate that the **power up display** is below the **main operating display**. Pressing **<ESC>** returns you back to the **power up display** (device selection).

The hot keys may be programmed by the administrator of the device to recall global settings (see EVENT management for details) and therefore may have dedicated names.

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

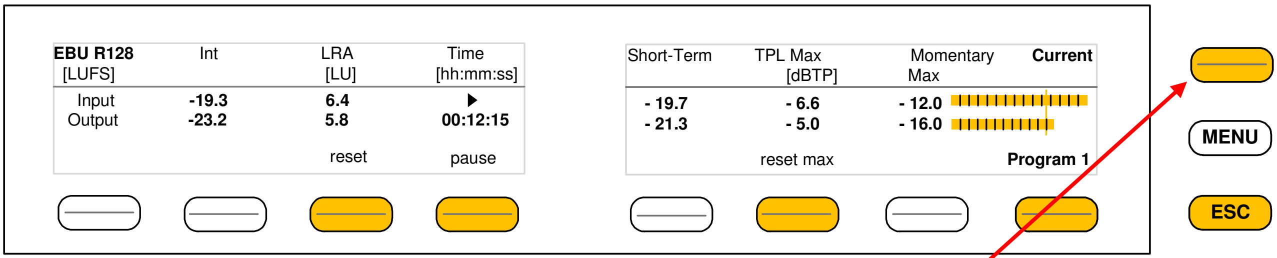
Important Note! The functions described below expect a proper routing of the signal from hardware interfaces to the audio processor and back (see ROUTING pane).

When pressing the **<MENU>** button, the main page of the operating menu opens up:



This menu allows for high level selections like the control of the audio processor or showing the meter display.

The fourth key <**EBU R128 Meter**> opens the loudness measurement display:

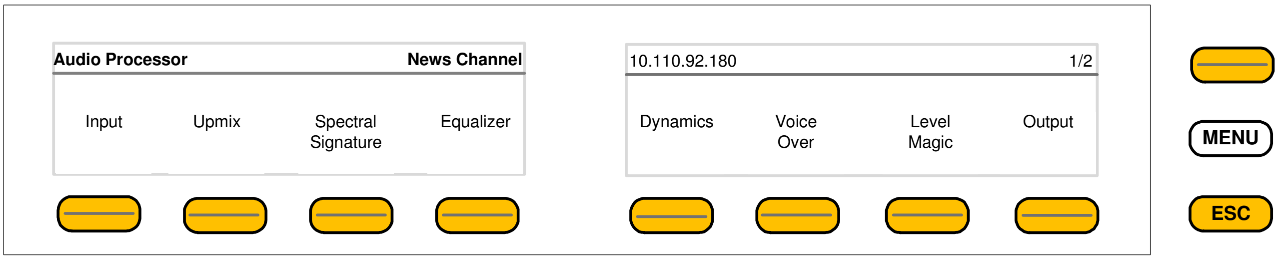


The highlighted keys will control the measurement process. While the **<Shift>** key toggles the display between **Current** and **Recent** measurement.
The display represents the measurements of **Integrated-** / **Short Term**- and **Momentary-Loudness** as well as **LRA** [LU] - the loudness range and **Max TPL** [dBTP] - the maximum true peak level.

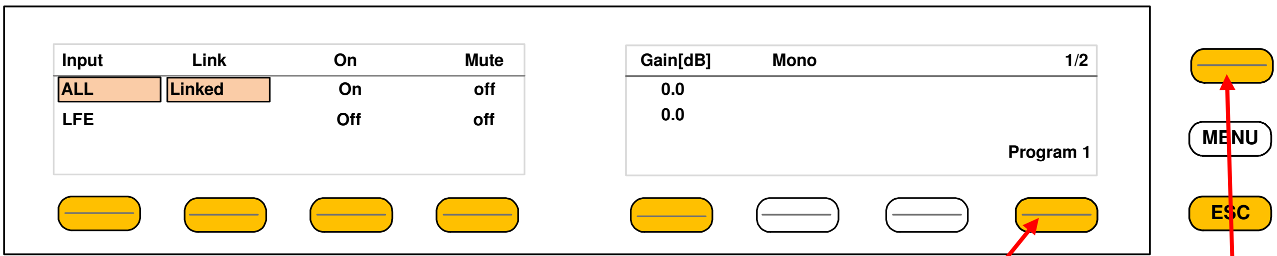
The measure for the EBU meter display is **[LUFS]** (Loudness Units Full Scale) as long as not defined differently. For details pls. refer to the EBU-Tech 3341 document.
You may leave this display by pressing <ESC>. This will bring you back to the first page of the operating display.

The key #8 switches between the programs of the D*AP8 **Program x**
(see block diagram AUDIO PROCESSOR > Overview). The other keys will do what is written above tem.

The second key of the operating display opens the Navigation for the Audio Processor (DSP) function blocks:



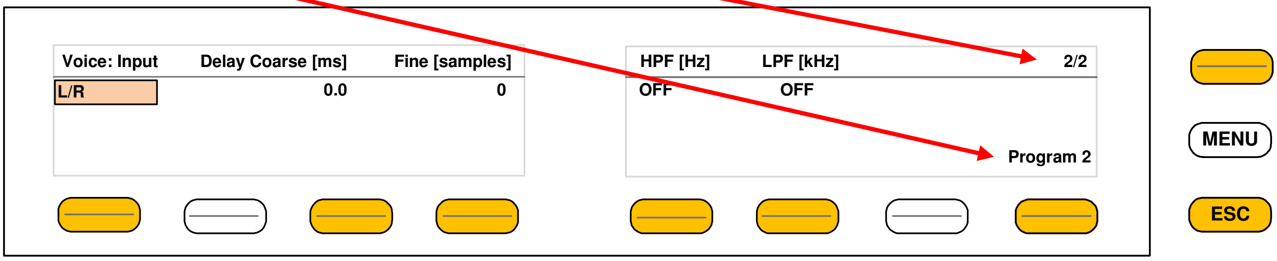
The highlighted key will open the parameter settings for the respective function block. The **<Shift>** key opens a second page where you will reach **Aux Input** and **Monitor** control.
You will find the parameter description further down in this manual explained step by step in relation to the GUI tab sheets, we will show here the principle how to control parameters via the **X*AP RM1 by** using the example of the **AUDIO PROCESSOR > Input** function block. After pressing key #1 the following display appears:



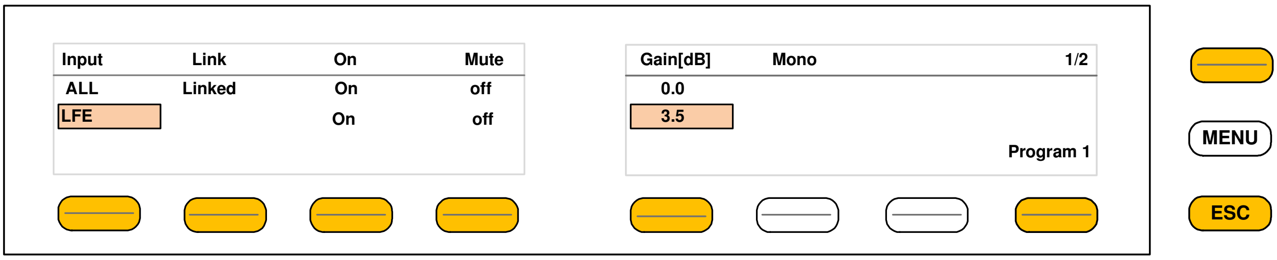
The device can process up to 4 programs (4 x 2). You must select the program that you will control via key #8. Above you see the display of program 1 of a **5.1 + 2** set-up.

This display has two pages. Page **1/2** (of two) is displayed at the moment. When you press the **<Shift>** key the next page **2/2** appears.

Below you see a display for **page 2** of **program 2**:



Back to page **1/2**:
The display is divided into columns above the function keys and horizontal lines representing audio channels [LFE] or groups of channels [ALL = L/R/C/Ls/Rs].
Function key #1 selects one of the available lines and selects between [ALL] and [LFE].
If you want to change the **Gain** for the **LFE** you must press key #1 to select the second line.
If you press the Gain key #5 the second gain parameter in the Gain column will be highlighted and you can change that gain by simply turning the rotary encoder:



Similar applies for all parameter setting via the **X*AP RM1:**

1. Select the program [program 1]
2. select the page where the desired parameter is in [1/2]
2. select the channel / group of channels [LFE]
3. select the parameter [Gain]
4. set the parameter by the rotary encoder [3.5]

ON/OFF parameters can be changed either by pressing the rotary encoder or turning it **counter clockwise > off** or **clockwise > on**.

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

Power Up Display

<MENU> opens **X*AP RM1** remote panel IP setup menu. See extra manual for details.

<Address> Setup
<Netmask> Setup
<Gateway> Setup
< empty >
Device 1 Setup IP & ON / OFF
Device 2 Setup IP & ON / OFF
Device 3 Setup IP & ON / OFF
Device 4 Setup IP & ON / OFF

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

<connect> will connect with that particular **D*AP8** and opens the **main operating** display:

Hotkey #
1 user defined
2 user defined
3 user defined
4 user defined
5 user defined
6 user defined
7 user defined
8 user defined

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

<MENU> opens o**perating** displays:

Hotkey #
1 <empty>
2 <Audio Processor page 1/2> <Audio Porcessor page 2/2>

<Input>	<Aux Input>
<Upmix>	<monitor>
<Spectral Signature>	<empty>
<Equalizer>	<empty>
<Dynamics>	<empty>
<Voice Over>	<empty>
<Level Magic>	<empty>
<Otput>	<empty>

3 <empty>
4 <EBU R128 Meter>
5 <empty>
6 <empty>
7 <empty>
8 <empty>

<ESC> back to **main operating** display

Setup GUI – connecting with the **D*AP8** – AUDIO PROCESSOR > **Overview**

You must open a browser and enter the **IP address** of the **D*AP8** unit into the **URL** field and press **<Enter>**. The browser will fetch the necessary information and open 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 area you have several bar graph displays for IN/OUT peak level of the audio processor, the AUX input and the Monitor output. A numeric display of loudness measurement and other bar graphs to display the gain changes of the function blocks: expander, compressor leveler and limiter complement upper display.

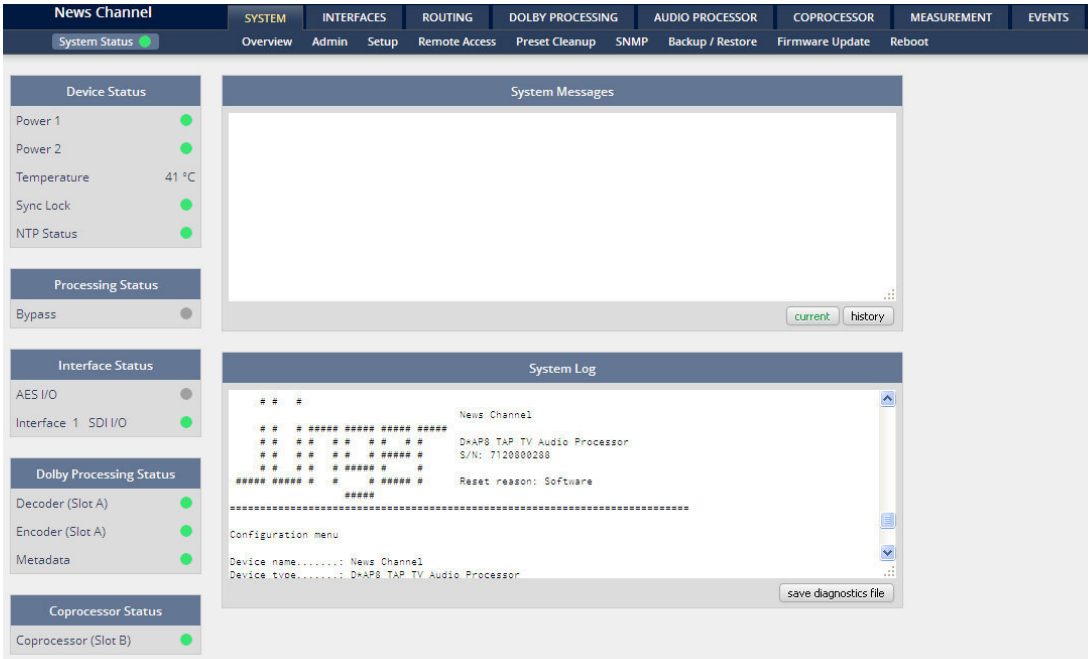
On the following pages we will go through the various panes to perform the basic setup of the device.

You must setup 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 monitoring systems of your company (e.g. via SNMP).

You must setup the installed interface modules and finally set the signal routing. Those settings you will find 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 hardware status of the **D*AP8 unit.**

Power 1 status of the first power supply (left hand side from rear).

Power 2 status of second power supply (right hand side from rear).

Temperature measured on the surface of the main PCB.

Sync Lock turns red if the external sync source is removed or unstable.

NTP Server Status Is green if the NTP server synchronization is turned off or 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.

Interface 1 SDI I/O turns red if the SDI input is not locked (not present or bad SDI signal).

Dolby Processing Status

Decoder (Slot A) turns orange if the input signal is **not** Dolby encoded (PCM).

Encoder (Slot A) status of the first D-E encoder (if license is installed).

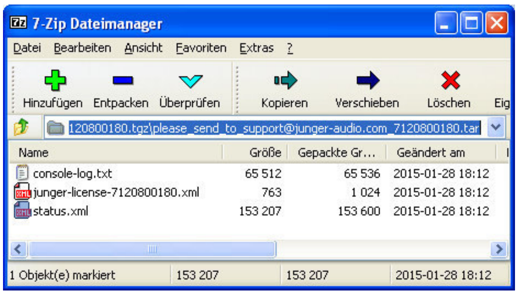
Metadata status of the metadata.

Coprocessor Status turns red if the co-processor encounters a problem.

System Messages <current> / <history>
Displays a list of messages produced by the system controller.

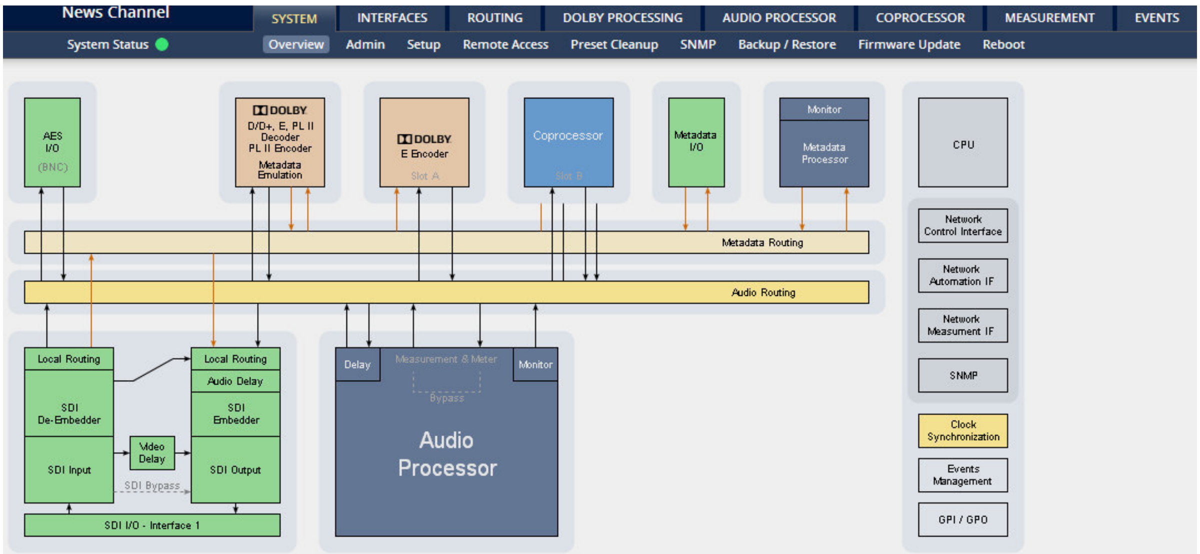
System Log The system controller activities will be logged. If there is a suspicious behavior we recommend to warm-start the D*AP8 by pressing the rear **<INIT** / **RESET>** button briefly. This will keep the log information for later investigation. If you do a power cycle instead the previous log information get lost.

<save diagnostics file> Pressing this soft button will start the assembly of files to help with diagnostics. The packed **.tar** archive contains 3 files:



The console log from the System Status pane, the license file and the status XML 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.

Setup GUI – SYSTEM – **Overview**

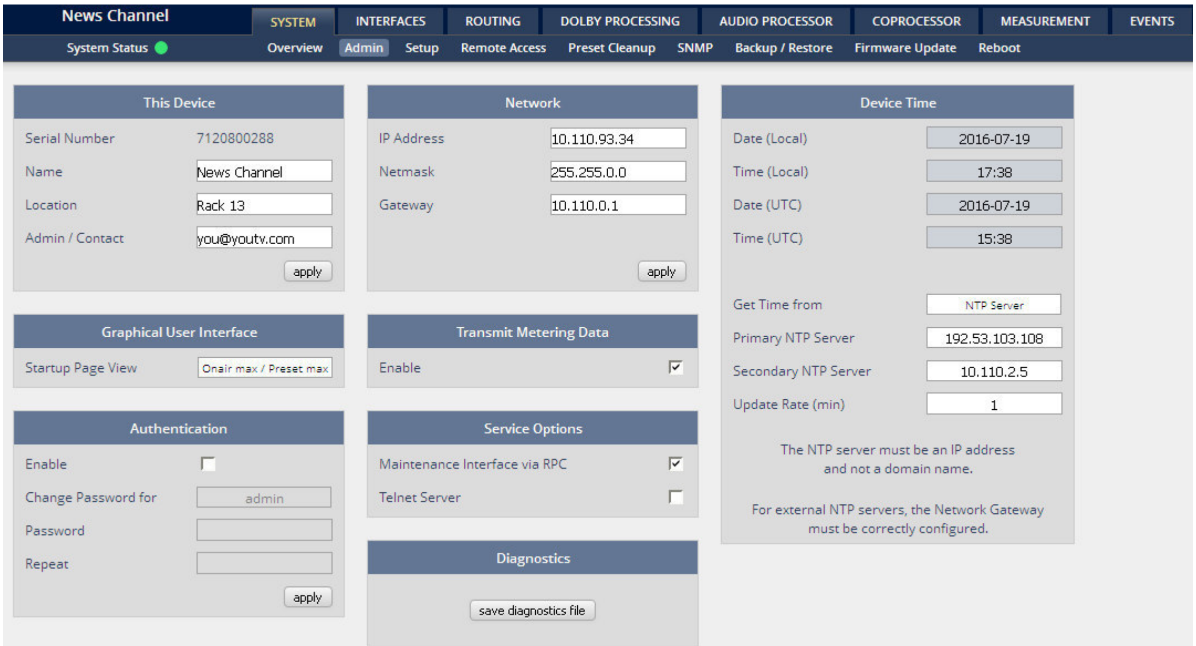


The graphical overview shows the main building blocks of the device including options installed, in this example a **SDI** interface placed into the interface 1 location and a **JDSPA** co-processor as well as the Dolby **CAT1100** OEM module, that runs the Dolby decoder / metadata emulation and a Dolby E encoder.

A fully loaded unit can have a Dolby (D / D+ / E / PL II) decoder / metadata emulator including a PL II encoder and up to two independent Dolby E encoders or one Dolby E and two consumer (Dolby D / D+ / AAC) format encoders installed.

You may click into the boxes and the respective 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**



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*AP8** 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 when you open a page).

Authentication To prevent non-authorized people from changing **D*AP8** 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.

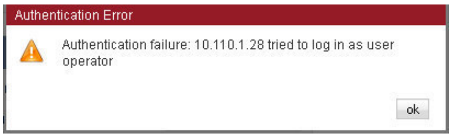
Change Password for [admin / operator]
Select which password you will set / change

Password type in a password
Default passwords are: admin (for admin) and operator (for operator).

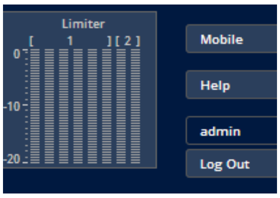
Repeat repeat that password

Important Note! The authentication may be enabled / disabled form 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 :



Network IP address setup, see above:
getting started – IP setup of the **D*AP8** – **via web browser**
– **via console interface**

IP Address A proper address for your network – default [10.110.xxx.yyy]

Netmask The net mask of your network – default [2555.255.0.0]

Gateway The optional gateway address – default [0.0.0.0]

Transmit Metering Data [ON / OFF]
Metering data will be streamed via UDP protocol. In order to receive such data by external applications and the GUI, you must enable it.

Service Options

Maintenance Interface via RPC [OFF / ON]
For administrative use to enable communication with factory tools**.**

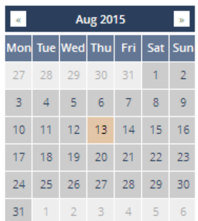
Telnet Server [ON / OFF]
Enables a telnet server to connect to the consol interface via TCP (port 23). You must be aware about the security risks if you do that over the internet!

Diagnostics

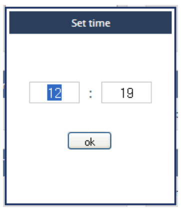
<save diagnostics file> Pressing this soft button will start the assembly of a diagnostics file.
The file will be presented in XML format for download.
If you experience unexpected behavior of the device you may be asked by the Junger service team to send such file by e-mail for analysis.

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.

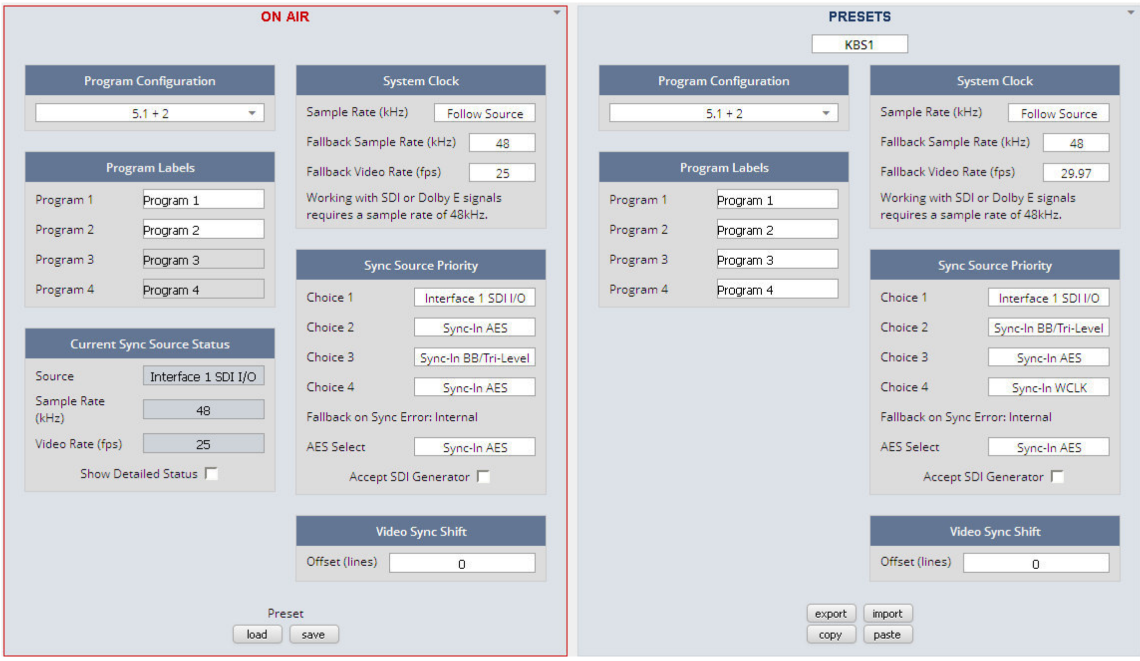
Primary NTP Server [5.9.110.236] default set to a publicly accessible NTP server via internet. This is used for device testing an may be overwritten at any time.

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. If no secondary NTP server is available set the address to 0:0:0:0 to avoid an error message regarding duplicated NTP server address setting.

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**

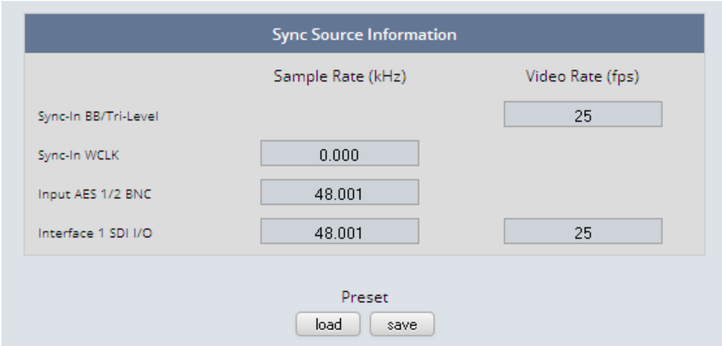


Program Configuration [5.1 + 2 / 4 x 2]
Shows the program configuration (2 times 2 channel). This is also the default configuration of the audio processing blocks.

Program Labels Each of the two possible programs has a name that will be used
Program 1 - 4 as a reference for the display of parameters and their setup.

Current Sync Source Status shows the status of the 5 tier sync priority circuit
Source active sync source
Sample Rate measured sample rate
Show detailed status [ON / OFF]
If you enable the checkbox you will get detailed information about the measured rates of possible sync sources.

Sync Source Information



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

System Clock

Sample Rate [Follow Source / 44.1 / 48]

Fallback Sample Rate [44.1 / 48]

Fallback Video rate [25 / 29,97 / 30]

Sync Source Priority

Choice 1 – 4 [OFF / Internal / Sync-In WCLK / Sync-In AES / Interface 1 SDI I/O (if fitted) / 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 BNC … AES 7/8 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*AP8**.
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.

Video Sync Shift For applications like Dolby E encoding it might be necessary to move the timing reference point.

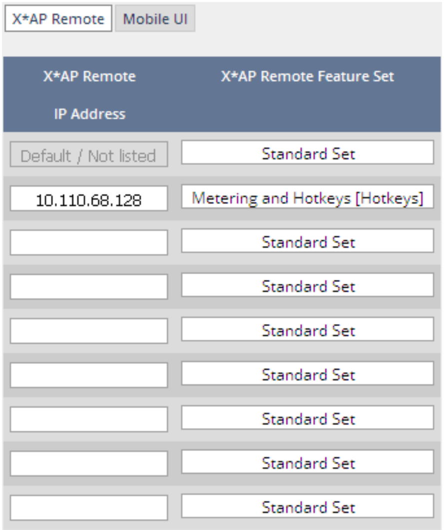
Offset (lines) [-1023 … 0 … 1023]
The number of lines which the reference point can be moved in either direction.

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

The **X*AP** can control multiple **D*AP8s** one by one and a single **D*AP8** may be controlled from multiple **X*AP**s. This requires a flexible remote concept that allows you to recall pre-set configurations via the **X*AP** panel or via 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 you study the comprehensive **EVENTS** system of the **D*AP8**.

At the moment of connecting a particular **X*AP** with a **D*AP8** the **X*AP** configuration will be transferred to that **X*AP**. I.e. configuration must take place at the **D*AP8.** You will decide here which feature set a particular **X*AP** is allowed to control:



For each **X*AP** you will be able to pre-set a **Feature Set**:

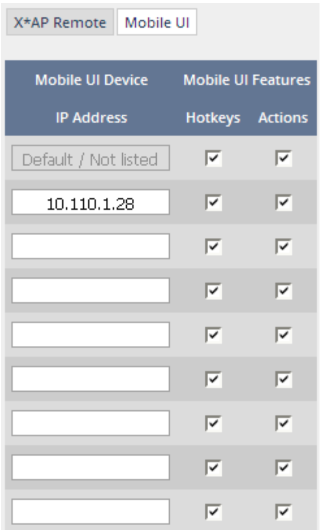
X*AP Remote

IP Address — In the first line: [Default / Not listed] you define the access policy for an "unknown" **X*AP** that connects with this **D*AP8** for the first time. The other lines are used to pre-define features for known **X*APs**. When connecting from an unknown **X*AP**, the respective **IP address** will be inserted automatically into the next empty line.

X*AP Remote Feature Set — You can select between a "Standard Set" that is full access for now and the access to "Metering and Hotkeys". I.e. any user who connects from an **X*AP** with this **D*AP8** will have limited access to metering and pre-defined hot-keys (see EVENTS > Triggers > Remote Hotkeys).

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

For the moment the **Mobile UI** implementation is limited to emulating hotkeys and to triggering actions. Profiles will be coming soon. They will then represent the combination of presets.



For each **Mobile UI** you will be able to pre-set if the UI is allowed to fire Hotkeys or trigger Actions.

Mobile UI Device

IP Address — In the first line: [Default / Not listed] you define the access policy for an "unknown" device that connects with this **D*AP8** for the first time. The other lines are used to pre-define features for known **UI**s. When connecting from an unknown **UI**, the respective **IP address** will be inserted automatically into the next empty line.

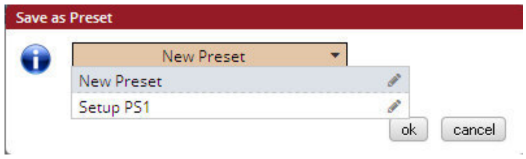
Mobile UI Features

Hotkeys — Pre-defined Hotkeys from **EVENTS > Triggers > Remote Hotkeys**

Actions — Pre-defined Actions from EVENTS > Actions.

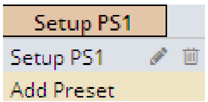
Setup GUI – SYSTEM - the **preset concept** in detail

The example above shows the **preset concept** of the **D*AP8.** 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 on 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 on 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.

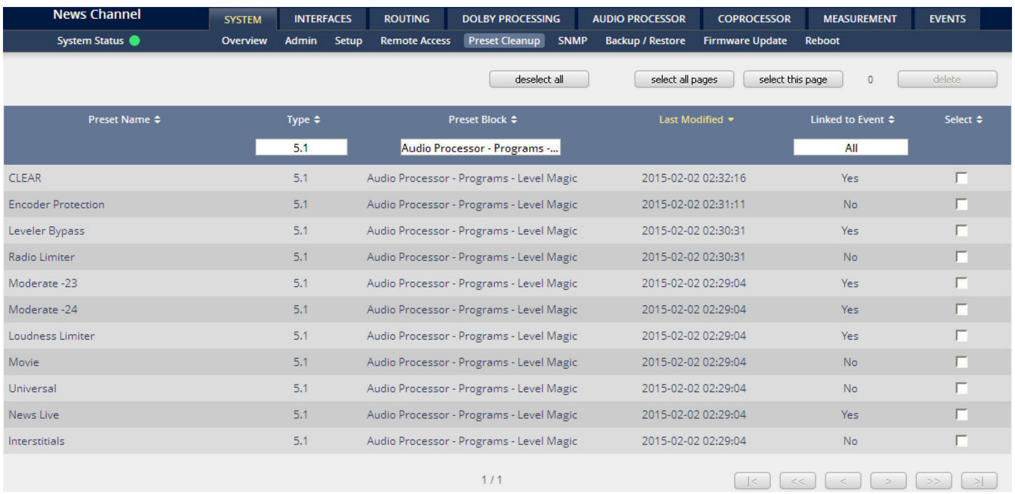
Important Note! The presets of the **D*AP8** are persistent by nature. You are working directly on the preset memory, i.e. you need not worry about storing such presets. The **D*AP8** does it for you.
On the other hand you must be aware that you will **overwrite the actual preset settings!** If you want 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 will 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.

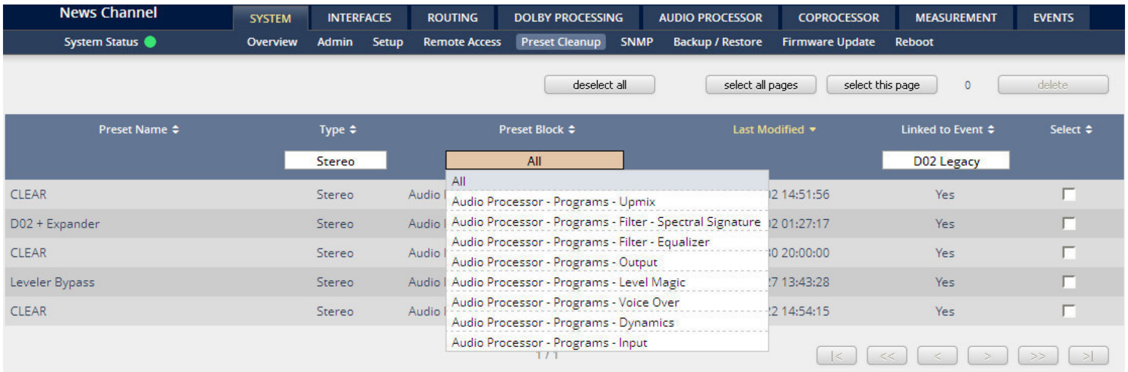
You may also **<export>** or **<import>** the preset content to / from a file.

Setup GUI – SYSTEM – **Preset Cleanup**

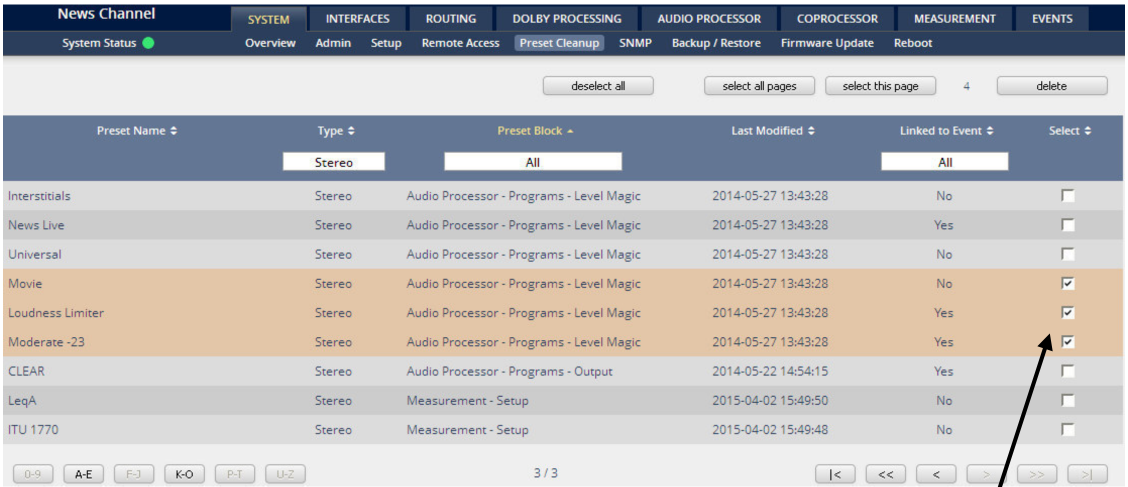
It is sometimes desirable to delete presets which are used by multiple events without stepping through all processing blocks and deleting the respective presets one by one. This pane offers you a tool to delete presets from a central access point:



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:

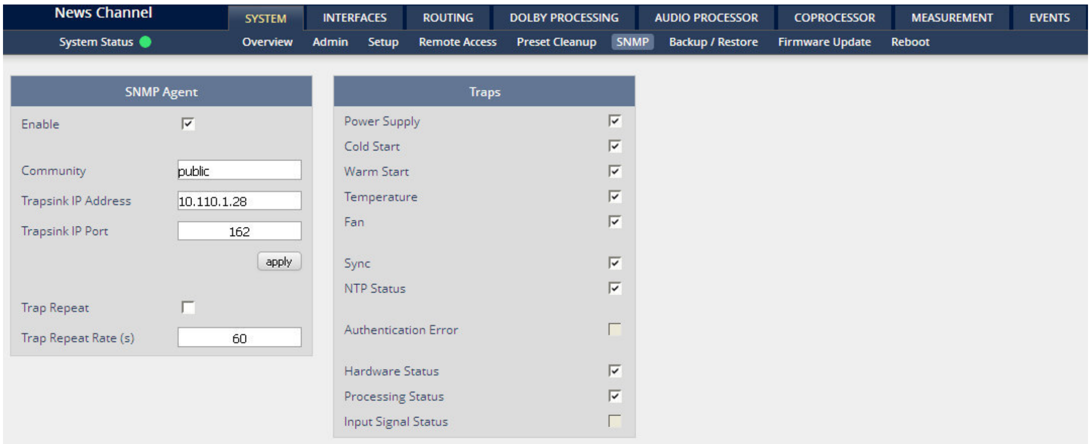


The soft buttons at the bottom left hand side may also be used to search through the table by sorting it by the first letter or leading number. The arrow buttons at the bottom right hand side can be used to scroll through the table if the selection is too big for one page:



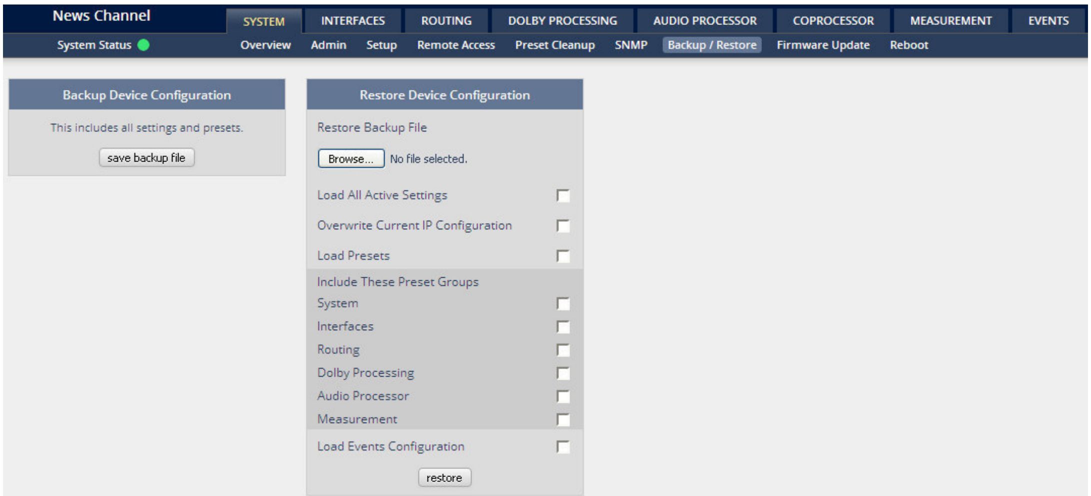
A selection is made by clicking on a line to activate the check box. Once you have made your selection (highlighted lines), you can press the **<delete>** soft button to execute the process. This will remove the selected presets permanently from the device.

Setup GUI – SYSTEM – **SNMP**

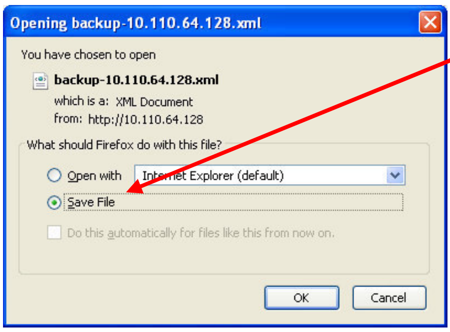


This pane is meant for basic settings of the **SNMP Agent** of the device. If you are not familiar with the use of SNMP protocol for system monitoring you should not enable the SNMP agent.

Setup GUI – SYSTEM – **Backup** / **Restore**

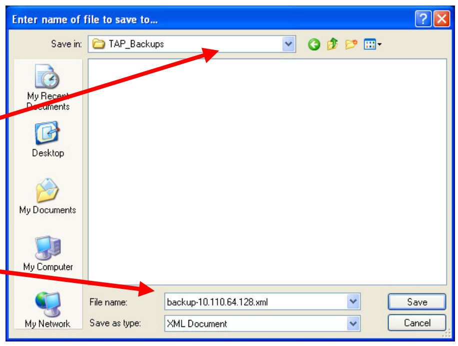


Here you can **backup** the complete **device** and **restore** parts or all of it .If you press **<backup>** the device controller will collect all necessary data and assemble it to an XML file. Finally you will get a pop up message:



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

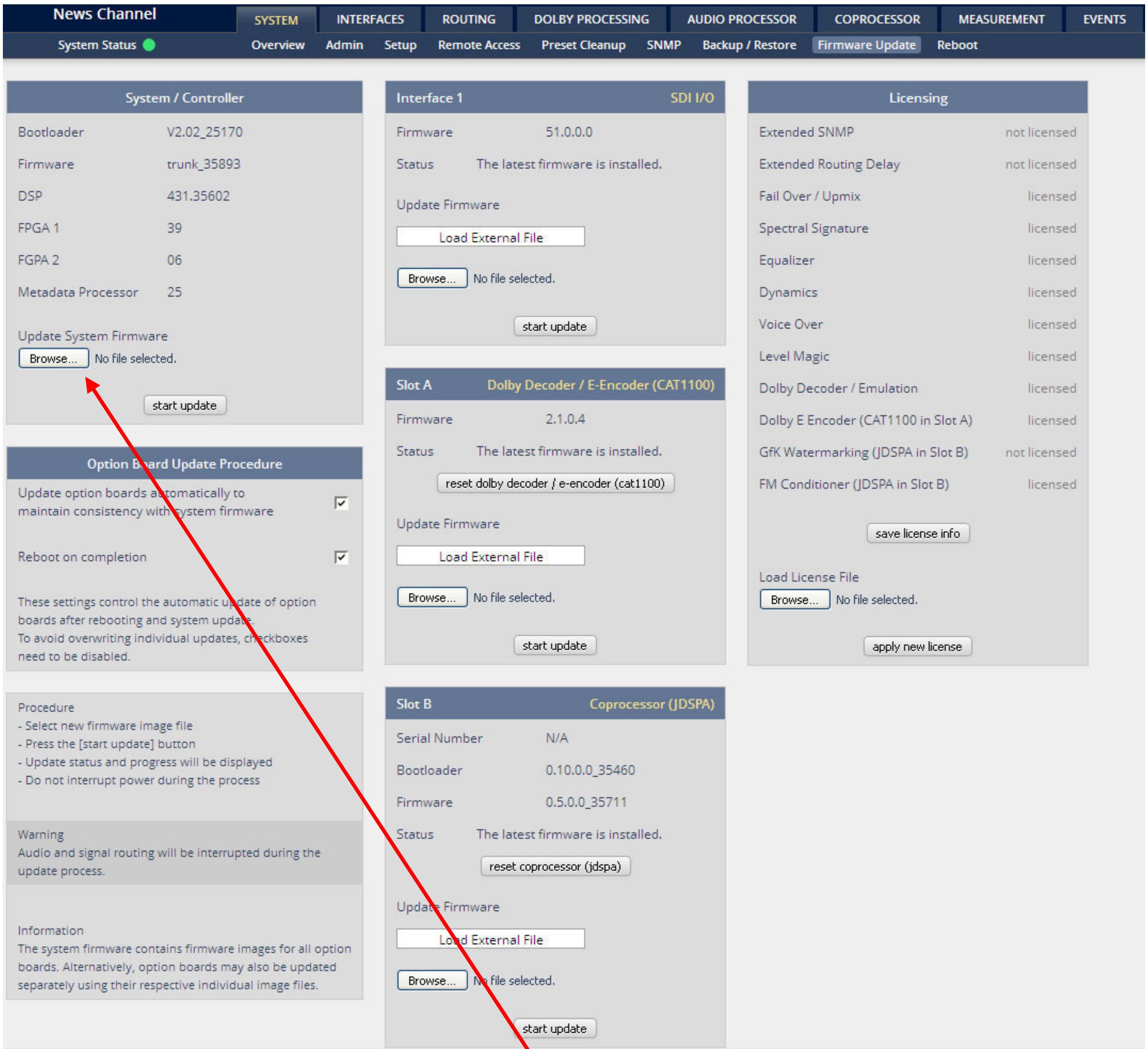
Select a folder and alter that default file name if needed.



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 file to update the **D*AP8** comes in **ZIP** format. You must unpack it to your PC's hard drive.
It contains also the manual a quick start guide the version history and a folder with the firmware for the **X*AP** remote panel. The folder /base_unit_image contains the so called "image" file for the **D*AP8**. Here an example: "rel_dap8_mei_1_2_3-34186.img". It is a bundle that brings the latest firmware versions for all interfaces and Dolby modules with it.



To update the **D*AP8**, you must **<Browse …>** to find the respective firmware file (which you have unzipped before) and press **<start update>**. If you do not want to upload all individual module firmware files for any reason, you may take the "rel_dap8_mei_4_3_4-basic-34186.img" file.
After finishing the update the device will automatically reboot.

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 or modules in Slot A / B. 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 and the Dolby OEM boards. See Interface 1 below as an example.

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).

Interface 1 — **SDI/O**

You may also update the firmware of an optionally installed SDI board or other interface boards.

Firmware — Display of actual installed firmware.

Status — [The latest firmware is installed. / A firmware update is available]

Update Firmware — [Load External File / x.y.z.]
You can decide if you want to upload it manually or take the latest module firmware "x.y.z" that came with the release image (recommended). You may **<Browse…>** the file system and select a file of your choice.

Interface 2 — *If you have two interface boards installed, similar applies to the second one.*

Slot A — **Dolby Decoder / E-Encoder (CAT1100)**

For the example above we have the optional Dolby decoder installed.
It is based on the Dolby OEM board CAT1100.
The status says: "The latest firmware is installed".

<reset dolby decoder / e-encoder (cat1100)> — Pressing this soft button will warm start that module.

Slot B — **Coprocessor (JDSPA)**

For the example above we have the optional **Coprocessor** installed.
It is based on the **JDSPA** module.

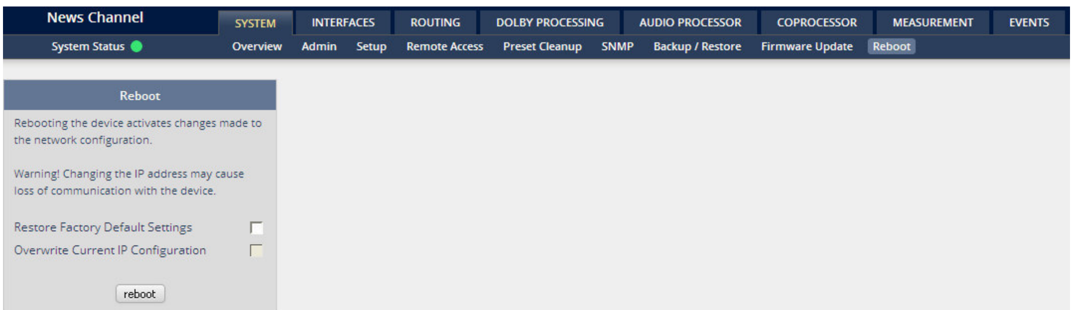
<reset coprocessor jdspa)> — Pressing this soft button will warm start that module.

Licensing — Here you can see a list of the licensed options of the **D*AP** family.

<save license info> — When you buy a license you must provide the "**license info"** file which you may obtain here.

Load License File — In return you will get a "**license"** file which you must apply to the device here. You must **<Browse …>** to find the respective license file (which you have unzipped before) and press **<apply new license>**.

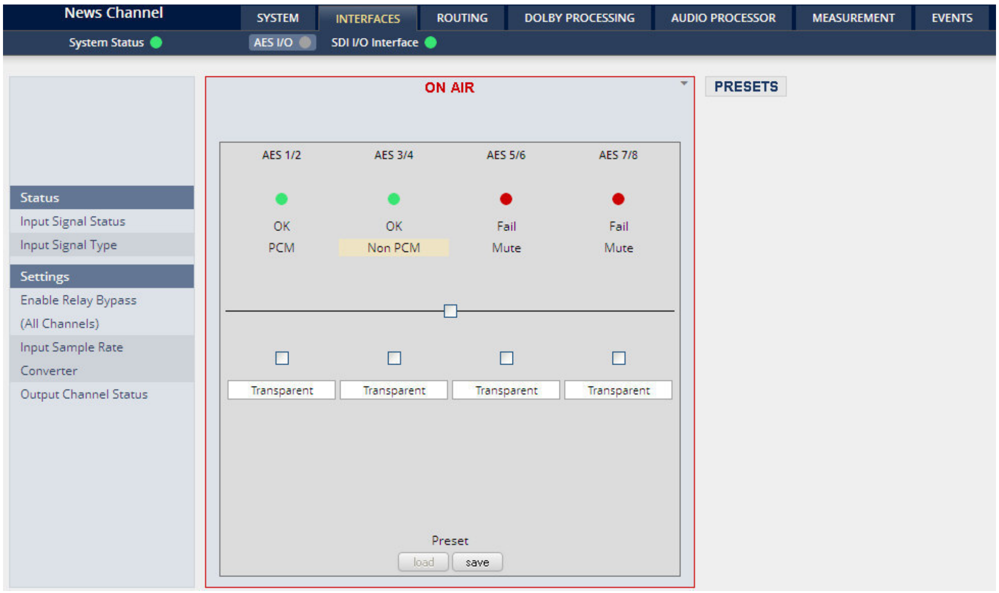
Setup GUI – SYSTEM – **Reboot**



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.

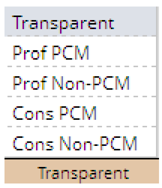
Important Note! The input signal status is logically combined and represented as part of the System Status. If one of the inputs is not assigned by the ROUTING matrix, its status will not be incorporated into the System Status. If non of the inputs is routed the Interface Status > AES I/O status soft LED becomes grey.

Settings

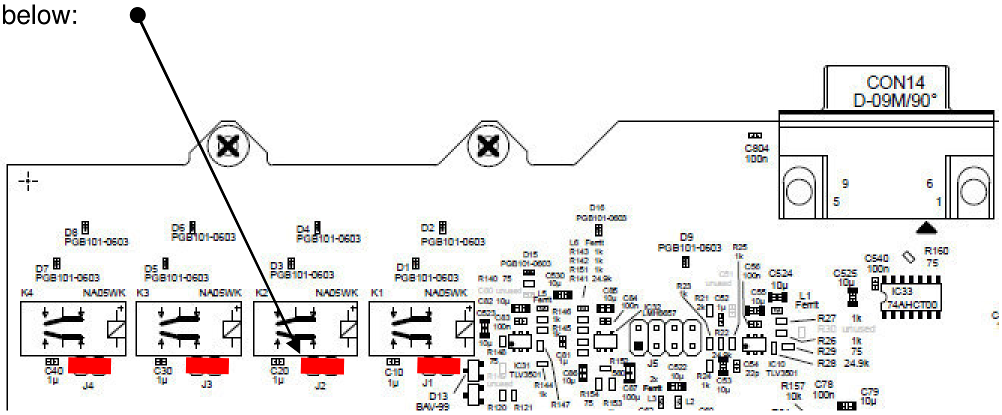
Enable Relay Bypass — [ON / OFF]
For fail save operation bypass relays are provided to connect AES IN / OUT in case of a power fail. One may enable such relays manually here.

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*AP8** or may be overwritten.



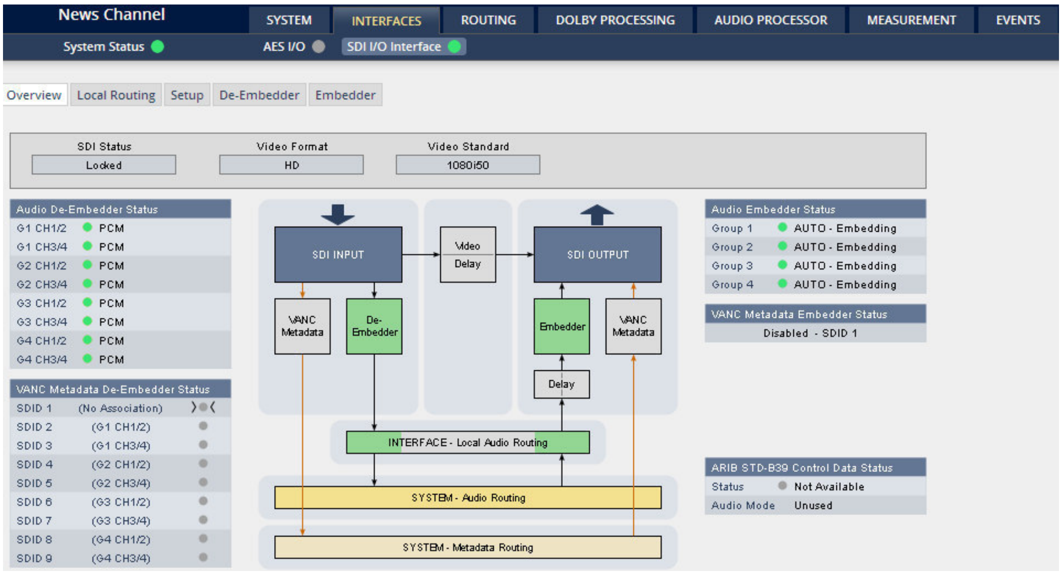
Important note! The AES relay bypass circuit of the AES I/Os may be deactivated inside the **D*AP8.** You must open the cover plate from the **D*AP8** unit and locate the red jumpers shown in the schematic below:



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

Setup GUI – INTERFACES – SDI I/O interface – **Overview**

If the **D*AP8** is equipped with an optional **SDI** interface the following settings will be available.
This pane has five sub panes embedded:



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 one has 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 simple replaced

Delete – the group from the input is deleted

OFF – the embedder for that group is turned off

VANC Metadata Embedder Status [Enabled - / Disabled – (selected SDID#)]
For details see **SMPTE 2020-2** standard.

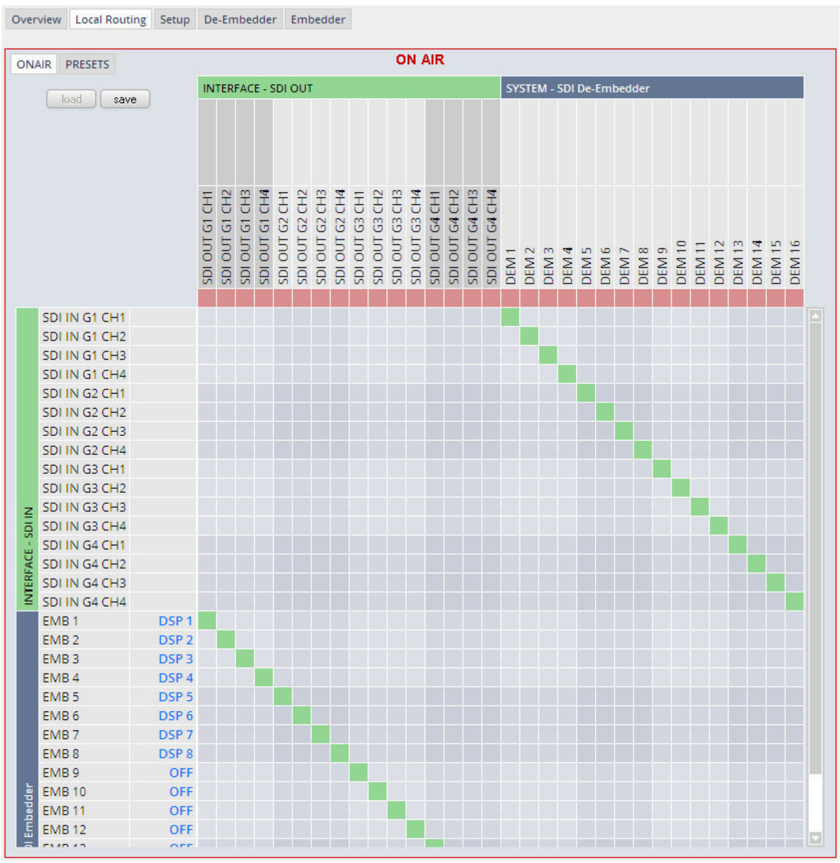
ARIB STD-B39 Control Data Status Meta information standard.

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

Setup 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 the default routing that sends all signals 1:1 from the physical de-embedders [INTERFACE – SDI IN G1 CH1 … SDI IN G4 CH4] to the internal device matrix
[SYSTEM – SDI De-Embedder DEM 1 … DEM 16].
The signals from the device router [SYSTEM – SDI Embedder EMB 1 … EMB 16] are routed by default 1:1 to the physical embedders [INTERFACE – SDI OUT G1 CH1 … G4 CH4].

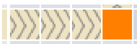


You must use the scroll bar to navigate through the matrix. In the upper left corner you can select between the **ONAIR** and the **PRESETS** view of the matrix. On the **ON AIR** page you will also see the device signal labels (see ROUTING section further below for details).

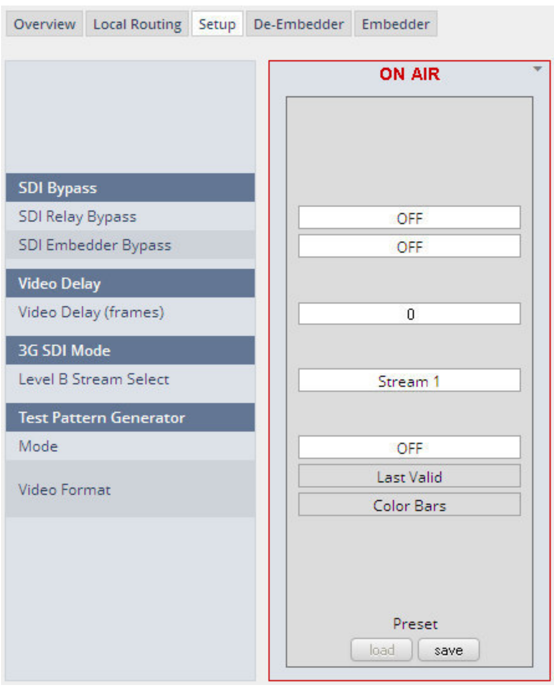
Channel Linking [mono / stereo]
You can decide if the routing must be performed in mono or stereo mode (where adjacent odd/even channels are routed at once).

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. The color of the respective squares changes:

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.

Setup 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

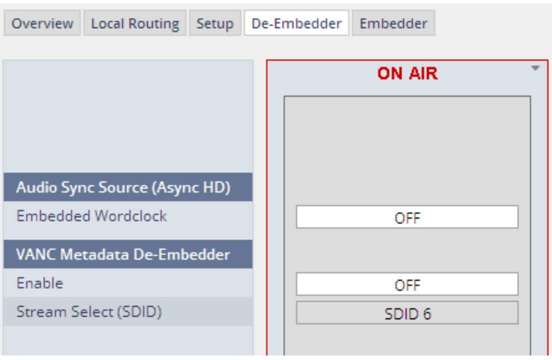
Level B Stream Select — A 3G-SDI signal may have two HD sub streams (e.g. for 3-D TV), AKN as 3G-B standard select between stream 1 or 2 for embedded audio. See SMPTE 425M for details.

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]

Setup GUI – INTERFACES – SDI I/O interface – **De-Embedder**

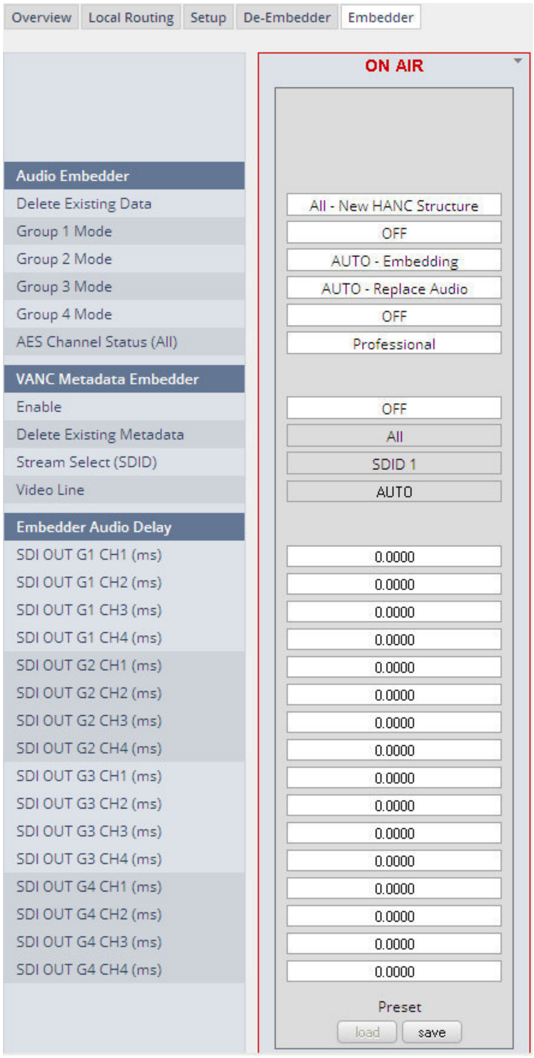


Audio Sync Source (Async HD) — The HD SDI standard allows for asynchronous audio. This critical if you have decided to synchronize the device on such signal. Here you find a solution. You may either use the embedded word clock

Embedded Word Clock — [Auto / De-Embedder CH1 (DEM 1) / OFF
OFF = synchronized to the SDI carrier

Auto = In case of a-sync audio it is synchronized automatically to the SDI carrier
DEM1= from de-embedder channel 1

Setup 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]
In case of 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 an adjacent pair (1/2, 3/4 …..) carries a Dolby E stream for example.

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 VAN 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 must be set to the same delay values not to destroy the data structure.

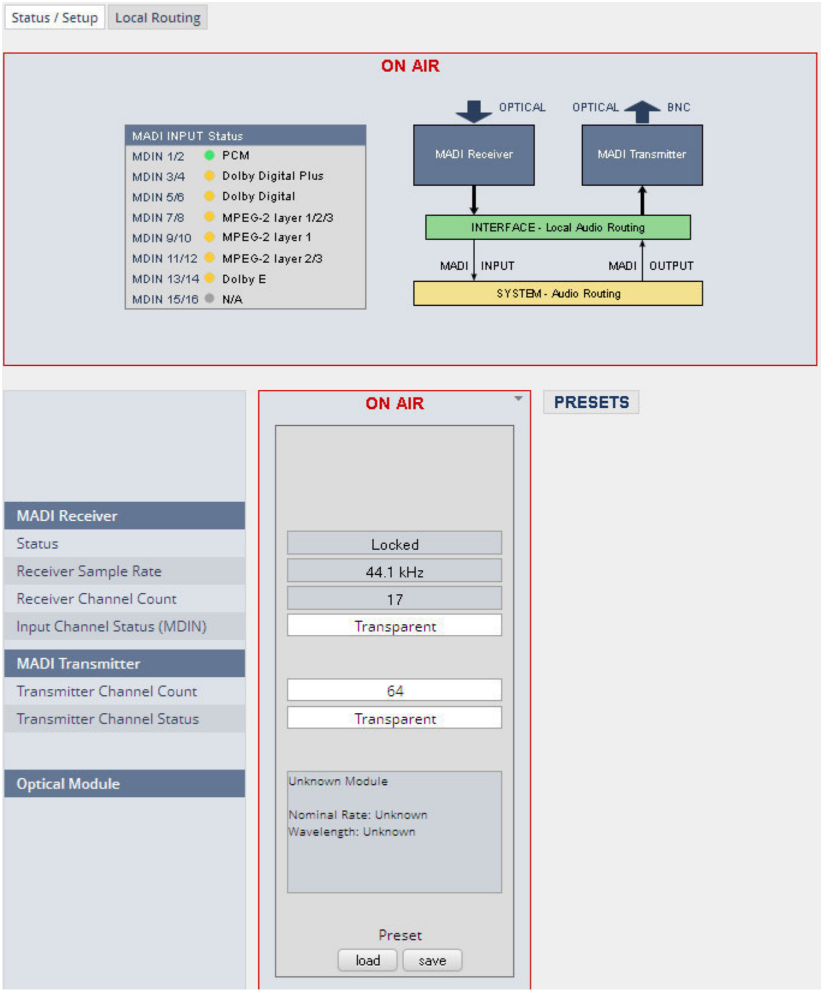
SDI OUT G1 CH1 (ms) [0.0000 … 340.000]

to

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

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

The implementation of MADI for the **D*AP8** 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 **D*AP8** is an eight channel processing device not all 64 MADI channels are available for device I/O. The first 16 channels are available via the MADI local router to the device router. They appear at the device router pane as MDIN 1 .. 16 and MDOUT 1 … 16.
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*AP8** 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]

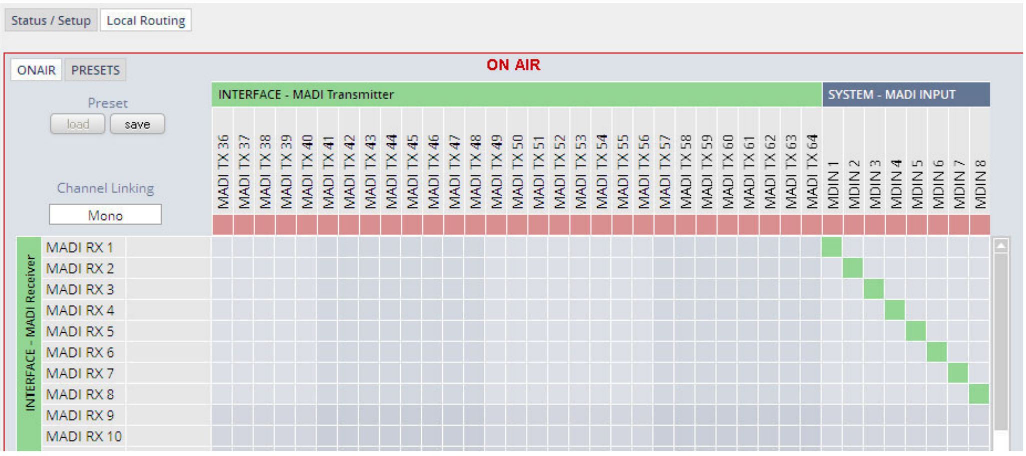
MADI Transmitter

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]

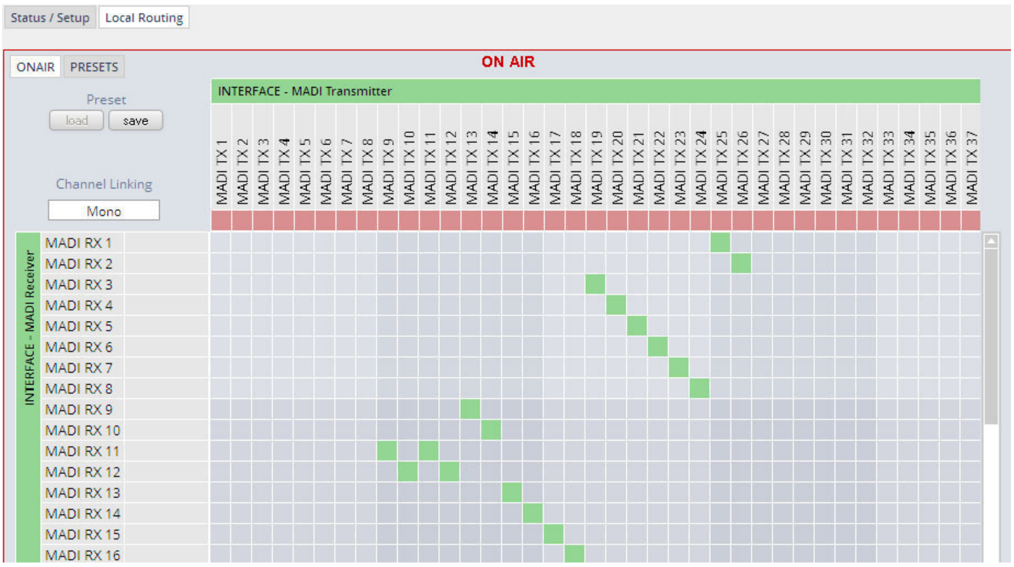
The connection for fiber cable is made by a LC connector. Looking at the rear panel the transmitter is the left one and the receiver the right one.

Setup GUI – INTERFACES – MADI Interface – **Local Routing**

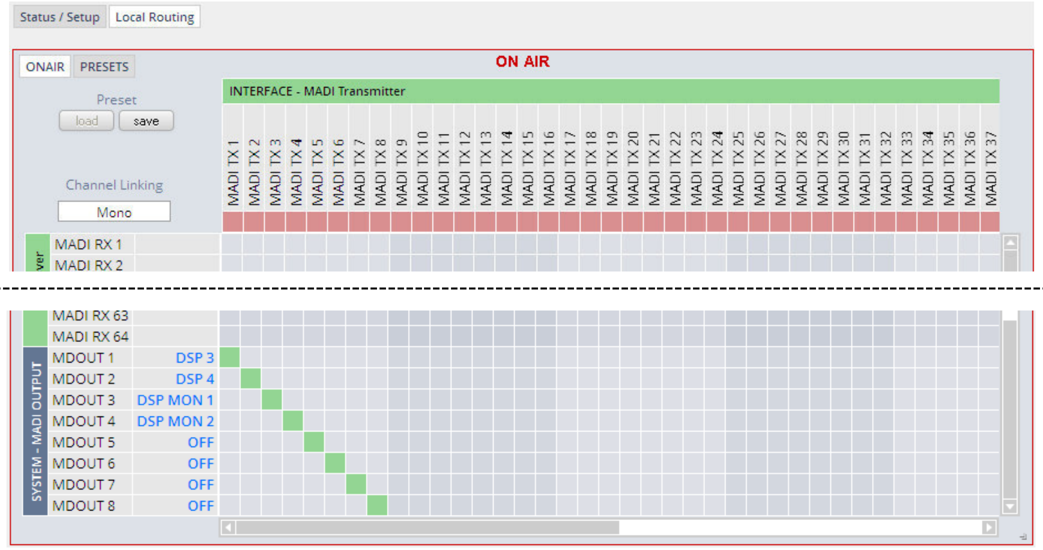
Below are some excerpts from the local routing pane. Single channels from or to the **D*AP8** may be connected with the MADI transmitter or MADI receiver respectively.
The example below shows the first eight MADI channels from the receiver (MADI RX 1 … MADI RX 8) connected with the device inputs **SYSTEM - 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 16) to MADI transmitter channels
For better visibility the matrix has been divided by cutting off the middle part:



You must use the scroll bars to navigate through the huge matrix.

Setup 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 AoIP must be able to handle the IP multicast. The recommendation is to separate the control network from the audio network.

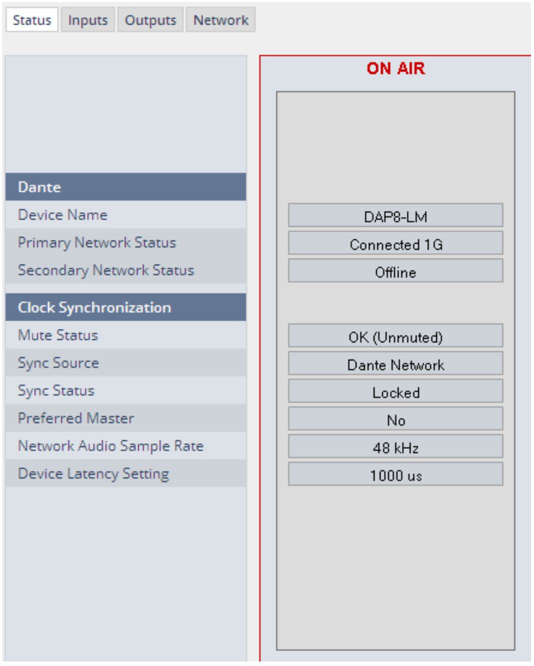
For details pls. refer to the Audinate web-site: https://www.audinate.com. Here you will find many useful application videos and FAQs.

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** software from **Audinate**. It provides standard audio drivers to connect with common sound tools.

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:



The parameters you see here must be set via the **DanteController** software.

Dante

Device Name — The name you gave the interface board via the **DanteController**.

Primary Network Status — [Offline / Connected + bandwidth]

Secondary Network Status — [Offline / Connected + bandwidth]

Clock Synchronization

Mute Status — [OK (Unmuted) / Muted]

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*AP8** 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*AP8** 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.

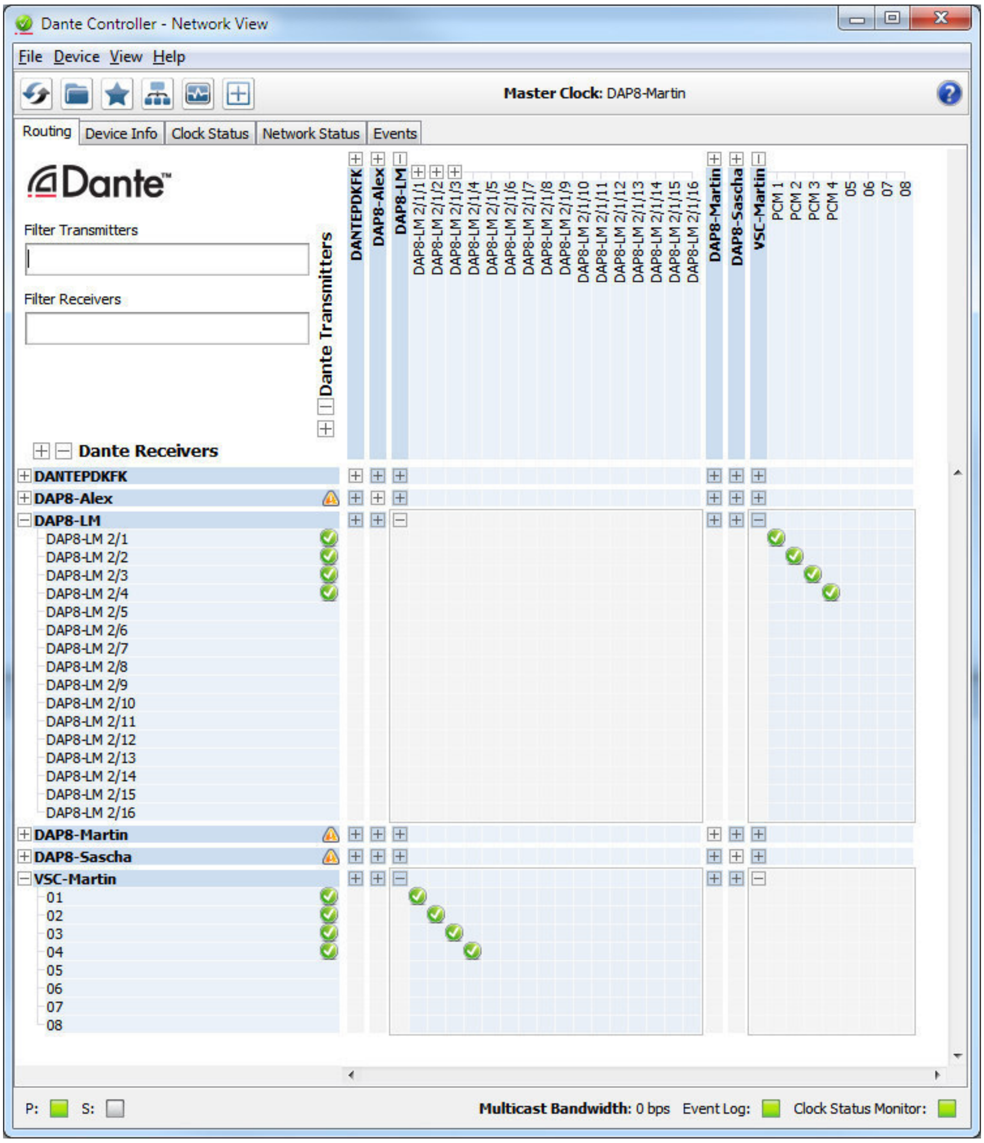
Network Audio Sample Rate — [44.1 kHz / 48 kHz / 88.2 kHz / 96 kHz]
Depending on the A*P device type the sample rate is limited to the device specification.

Device Latency Setting — [1000 µs]
You can allow for a certain transmission latency if you face network problems of any kind.

Setup 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*AP8** and will be displayed there.

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



As an example you see here a "DAP8-LM" (name given by the DanteController) that has assigned the labels DAP8-LM 2/1 … 2/16 for the inputs and DAP8-LM 2/1/1 … 2/1/16 for the outputs.
For the outputs you can assign up to 16 different labels used for multi layer routing.

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 1 … PCM 4 from the VCS are assigned to the **D*AP8** inputs DAP8-LM 2/1 … 2/4 while four outputs DAP-8 LM 2/1/1 … 2/1/4 are assigned to the VSC inputs 01 … 04.

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

Status | Inputs | Outputs | Network

Inputs		Channel	Connected	Status
DTIN 1	PCM	DAP8-LM 2/1	PCM 1 @ VSC-Martin	Connected (Unicast)
DTIN 2		DAP8-LM 2/2	PCM 2 @ VSC-Martin	Connected (Unicast)
DTIN 3	PCM	DAP8-LM 2/3	PCM 3 @ VSC-Martin	Connected (Unicast)
DTIN 4		DAP8-LM 2/4	PCM 4 @ VSC-Martin	Connected (Unicast)
DTIN 5	PCM	DAP8-LM 2/5	no subscription	No Subscription
DTIN 6		DAP8-LM 2/6	no subscription	No Subscription
DTIN 7	PCM	DAP8-LM 2/7	no subscription	No Subscription
DTIN 8		DAP8-LM 2/8	no subscription	No Subscription
DTIN 9	PCM	DAP8-LM 2/9	no subscription	No Subscription
DTIN 10		DAP8-LM 2/10	no subscription	No Subscription
DTIN 11	PCM	DAP8-LM 2/11	no subscription	No Subscription
DTIN 12		DAP8-LM 2/12	no subscription	No Subscription
DTIN 13	PCM	DAP8-LM 2/13	no subscription	No Subscription
DTIN 14		DAP8-LM 2/14	no subscription	No Subscription
DTIN 15	PCM	DAP8-LM 2/15	no subscription	No Subscription
DTIN 16		DAP8-LM 2/16	no subscription	No Subscription

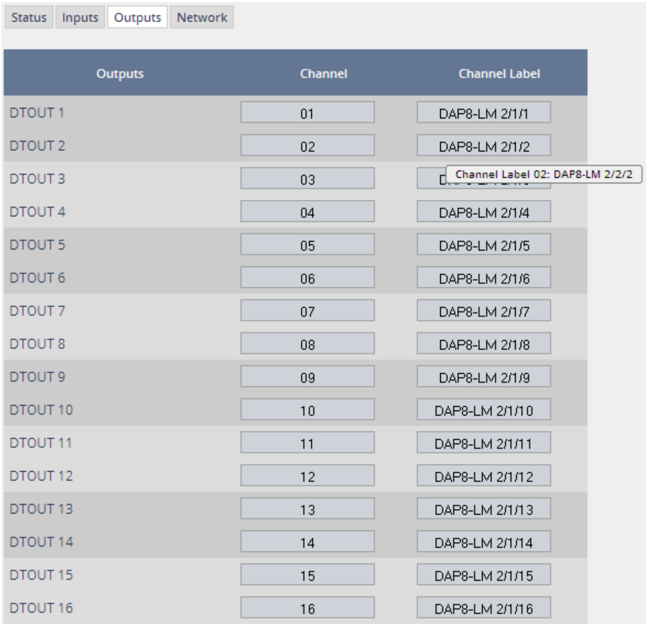
Inputs 16 inputs are pre-defined for the **Dante** interface installed in a **D*AP8**. They are organized in pairs and the input status is shown by soft LEDs (green = PCM audio / yellow = non audio/ grey no audio).

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

Connected The source of the audio signal.

Status [No Subscription / Subcription 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.

Setup GUI – INTERFACES – Dante I/O Interface – **Outputs**



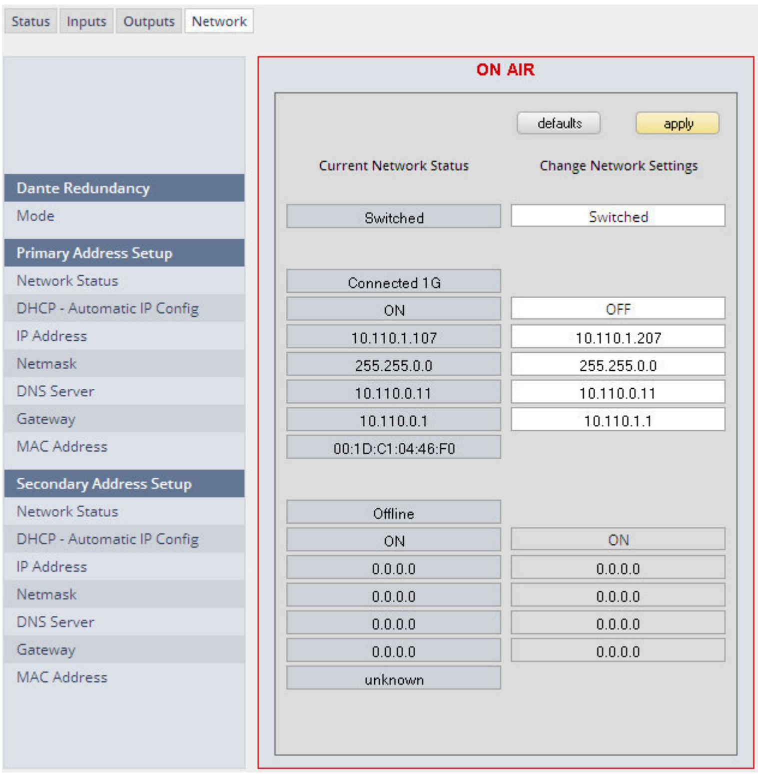
Outputs The signals from the **Dante** board to the network. They will also appear in the device **ROUTING** section.

Channel Numeric count of the channels.

Channel Label Up to 16 labels can be assigned for each stream from the interface to the network.

When you hover with the mouse over the channel labels, you will get a tool tip that shows the other (if any) labels assigned to the same outputs assigned fro multi layer routing.

Setup 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 redundant operation.

Mode	[Switched / Redundant]	
	Redundant –	The interface will duplicate the audio traffic to both Ethernet ports. Both ports must have different IP addresses
	Switched –	The second port behaves like an Ethernet switch port allowing daisy-chaining through the interface. I.e. IP configuration of the second port 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]

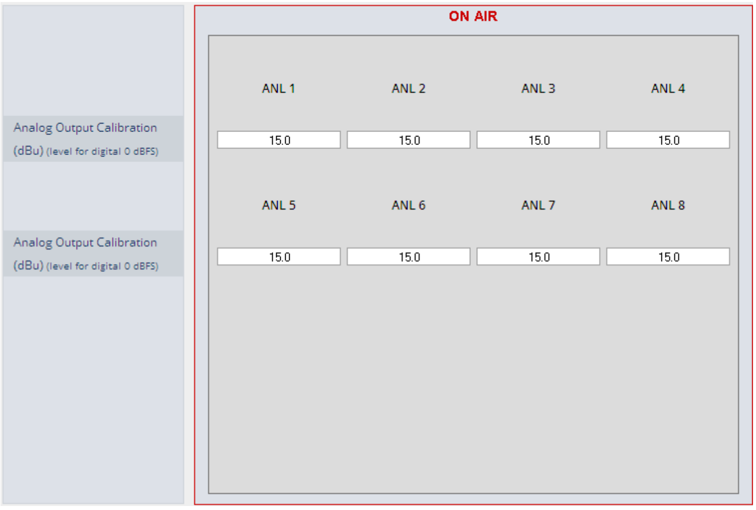
Important Note! It may happen by accident that the update of the Dante module fails. E.g. if the firmware update option: SYSTEM > Firmware Update > Option Board Update is set to "Update option boards automatically …." and the device loses power during this process, the Dante module will be in the fail-save 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_dap8_mei_4_3_4.zip > junger_dap8_mei_firmware > Dante_recovery_image > DT-100-v1.0.3-7.dnt

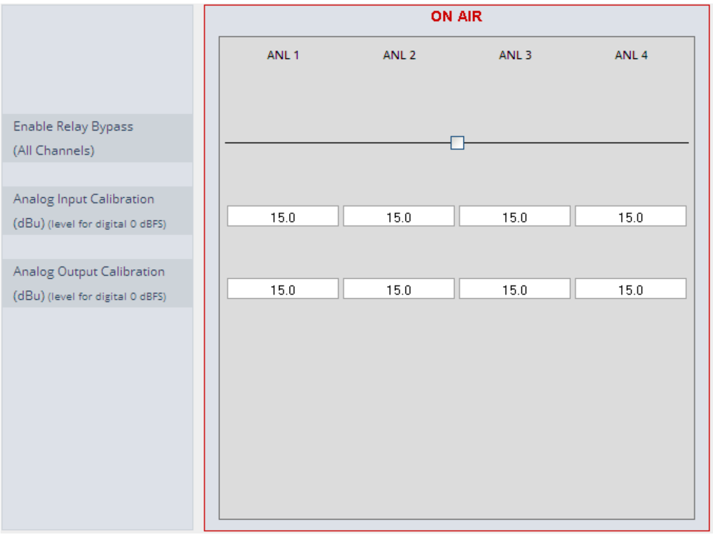
Setup GUI – INTERFACES – **8 Ch Analog Out Interface**



Analog Output Calibration (dBu) (level for digital 0 dBFS)	sets the factor for D/A conversion
ANLx (dBu)	[0.0 … 15.0 … 24.0] output level for output "x" at 0dBFS. The default setting of 15.0dBu correlates to the 6dBu = -9dBFS conversion.

Setup GUI – INTERFACES – **4 Ch Analog I/O Interface**

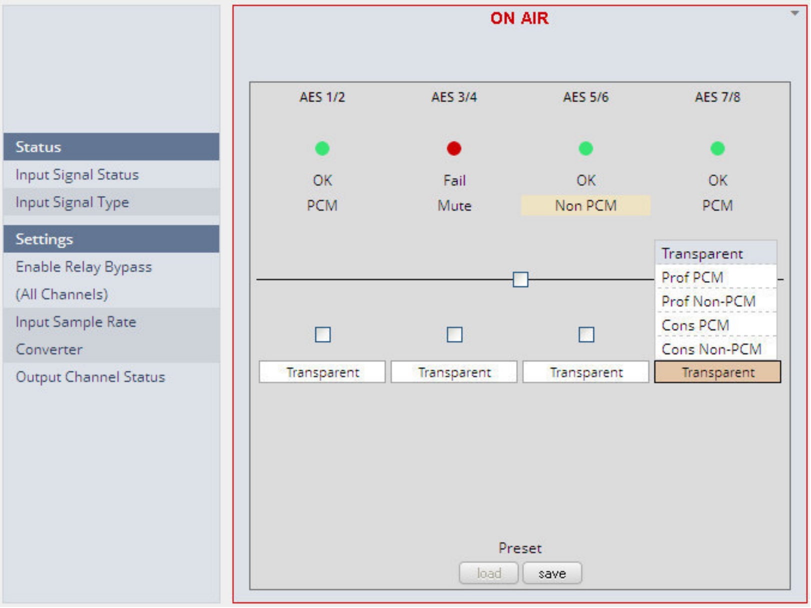
An additional analog interface can be installed in the **Interface** slot.
It provides 4 additional analog line inputs and outputs on a 25pin D-Sub connector:



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 0 dBFS)	[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 0 dBFS)	[0 … 15.0 … 24] D/A conversion parameter. It defines the analog output level in dBu for a digital full scale signal.

Setup 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:



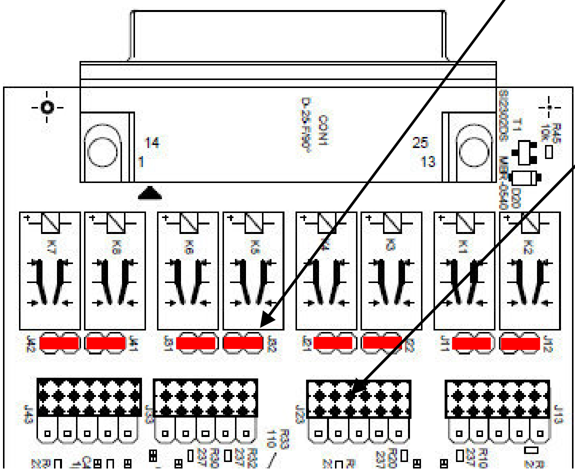
Status

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*AP8** 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.



The bulk jumpers J13, 23, 33, 43 ● at the bottom of the picture are meant for setting the I/Os to unbalanced operation.

Putting them into the lower position will turn to unbalanced.
Factory default setting is balanced.

Setup GUI – **ROUTING**

This is the core of the **D*AP8** because it defines the audio signal flow inside the device:

Each functional block of the device has a source- and a destination-label.
Vertically at the left hand side you will find the outputs of function blocks / hardware interfaces.
The labels are organized hierarchically. I.e. we have source group names like DSP OUTPUT, AES INPUT, DECODER EMULATION OUTPUT etc. And single channel (AKA mono) signal labels like **DTIN x** [x=1 … 16] for the **Dante** interface, **AESx** [x=1 … 8] for the AES inputs or **DEC x** [x=1 … 10] for the Dolby interface. CAT1100**.**

If applicable the labels have bluish dynamical signal descriptors [e.g. **1L / 1R / 1C** and so forth].

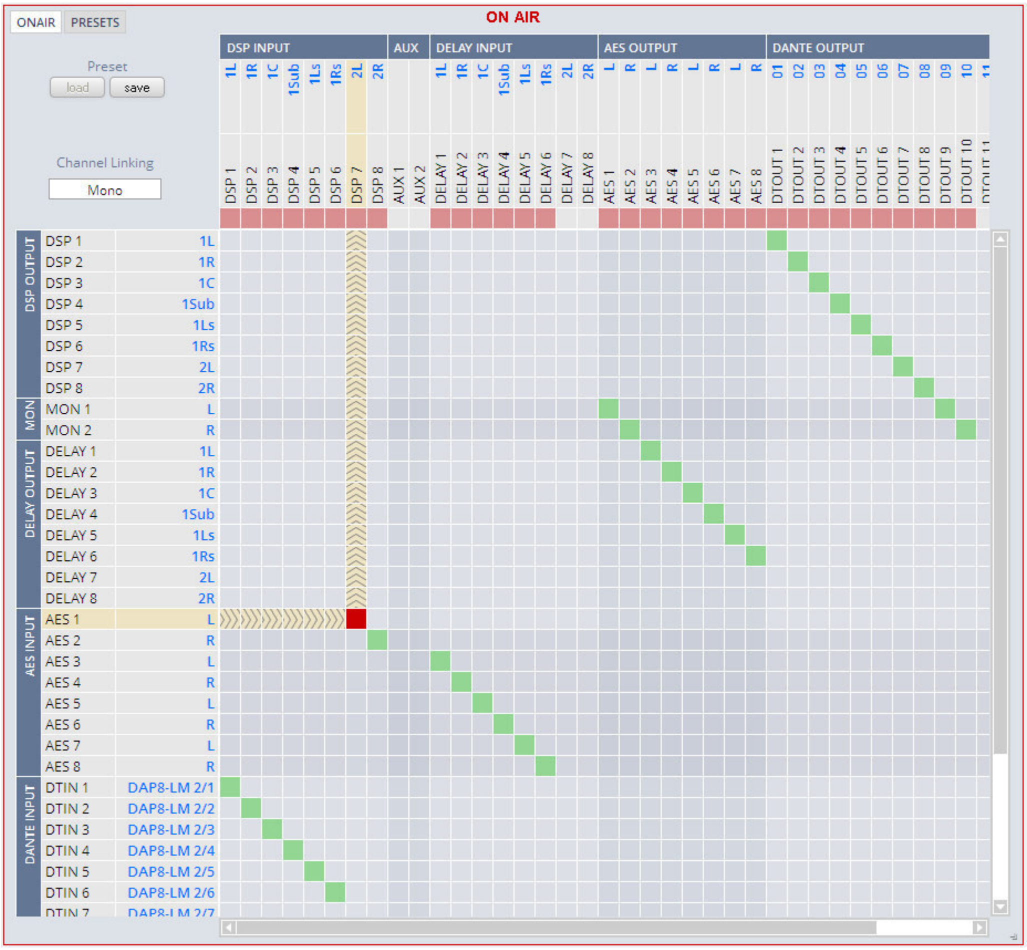
Horizontally at the top of the ROUTING pane you will find the group names for destinations like, DSP INPUT, AES OUTPUT, ANALOG OUTPUT, Dante OUTPUT, DECODER/EMULATION etc. **a**nd their respective single channel labels like **DSP x** [x=1 … 8] **AUX 1,2** for the 10 audio processor outputs or feeds to the hardware interfaces, like **AESx** [x=1 … 8] for the AES outputs, **MADOUTx** [x=1 … 16] for MADI outputs or ENCx [x=1 … 8] for the Dolby encoder inputs.

If applicable the labels have bluish dynamical signal descriptors [e.g. **1L / 1R / 1C** and so forth].

Green quads show active cross points. Due to the number of I/Os in total one must scroll through the matrix to set or disable cross points. To give you an indication while scrolling which outputs have an active connection, red quads are shown in the top of the matrix beneath the output labels.

The matrix is organized for single channel (mono) routing but it may also be controlled in 2-channel (stereo) mode:

Channel Linking [mono / stereo]
You may set cross points either in mono mode or pair wise for stereo routing.



Due to the size of the graphic you must select between **<ONAIR>** and **<PRESET>** view in the upper left corner.

Important Note! If a different optional interface board is installed the matrix will be expanded by the pre-defined number of **Inputs and Outputs** for the **D*AP8** platform with their labels:

Signal:	Option board:	Input label:	Output label:
SDI	[O_DAP_SDI_a]	**DEM 1 … DEM 16**	**EMB 1… EMB 16**
MADI	[O_DAP_MB_a / O_MO_MM_a / _MS_a]	**MDIN 1 … MDIN 16**	**MDOUT 1 … MDOUT 16**
Dante	[O_DAP_Dante_a]	**DTIN 1 ... DTIN 16**	**DTOUT 1 ... DTOUT 16**
4 Ch ANALOG I/O	[O_DAP_ADDA_a]	**ANL 1 … ANL 4**	**ANL 1 … ANL 4**
8 Ch ANALOG out	[O_DAP_8DA_a]		**ANL 1 ... ANL 8**
AES	[O_DAP_AES_a]	**AES 1 … AES 8**	**AES 1 … AES 8**
Dolby Decoder	[O_DAP_Dolby_DEC_b]	**DEC 1 … DEC 10**	**DEC 1 … DEC 8**
Dolby E Encoder (A)	[O_DAP_Dolby_EENC_b]	**ENC 1 … ENC 8**	**ENC 1/ENC 2**
Dolby D Encoder (B)	[O_DAP_Dolby_DENC_a]	**ENC 1 … ENC 8**	**ENC 1 … ENC 4**
Dolby E Encoder (B)	[O_DAP_Dolby_EENC_a]	**ENC 1 … ENC 8**	**ENC 1/ENC 2**

Source label

DSP x	Outputs of the audio processor (DSP)
MON x	Monitor outputs of the audio processor (DSP)
DELAY x	Outputs of the extra delay lines (independent from the audio DSP)
AES x	Outputs from the hardware AES receiver on the motherboard
DEM x	Outputs of the SDI local routing matrix
MDIN x	Outputs of the MADI local routing matrix
DTIN x	Outputs of the Dante Interface
DEC x	Output of the optional Dolby decoder / emulation board
ENC x	Output of the Dolby encoders

Destination label

DSP x	Inputs of the audio processor (DSP)
AUX x	Aux inputs of the audio processor (DSP)
DELAY x	Inputs of the extra delay lines (independent from the audio DSP)
AES x	Inputs of the AES transmitters on the motherboard
EMB x	Inputs of the SDI Local Routing matrix
MDOUT x	Inputs of the MADI local routing matrix
DTOUT x	Inputs of the Dante Interface
DEC x	Input of the optional Dolby decoder / emulation board
ENC x	Inputs of the optional Dolby encoders

Mouse over Pls. see "Setup GUI – INTERFACES – SDI I/O interface – **Local Routing"** for details.

Setup GUI – **DOLBY PROCESSING** in general

The Dolby metadata system is quite complex to describe in detail in a product manual such as this. If you are not familiar with it, we recommend you study the many publications from **Dolby Inc.** Especially the **Dolby Metadata Guide** is essential for understanding the parameters. For details please visit the Dolby web site:

http://www.dolby.com/gb/en/professional/technology/landing.html

We cannot guarantee that the link is active forever so you may browse other Dolby resources as well. Specifically concerning metadata we also recommend the SMPTE document RDD6-2008.

So we must assume that you are familiar with this topic.

Metadata emulation means that Dolby metadata will be applied to listen to the effect of it without the need for encoding / decoding that may become a costly setup and introduces a lot of latency.

The aim is to check the influence of the **Dialnorm** (dialog normalization) value and the **DRC** (dynamic range control) settings.

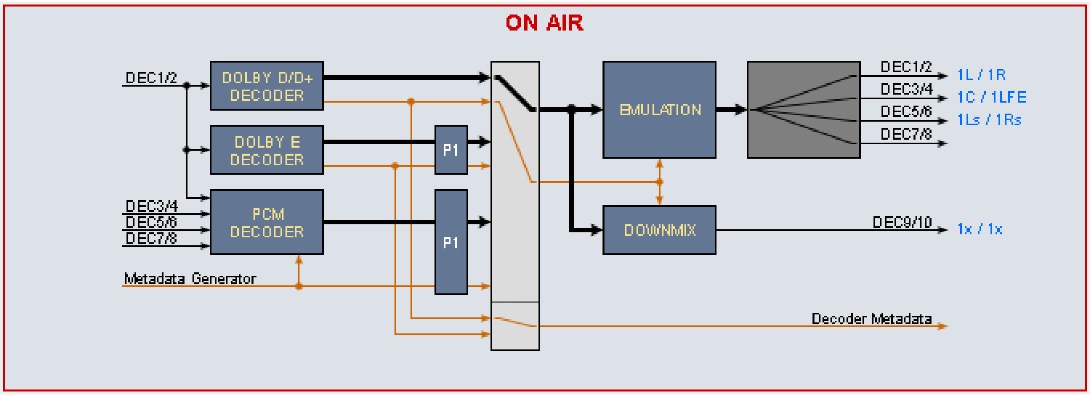
Important Note! The **D*AP8** platform is designed to operate an "all Dolby format" decoder and two independent encoders **A** and **B**. Encoder **B** can be consumer format (D-D, D-D+, AAC) **or** Dolby E professional while encoder **A** can be a second Dolby E. All solutions are based on the **D*AP8** options model and require extra hardware and/or licenses.

Setup GUI – DOLBY PROCESSING – **Decoder/Emulation**

The Decoder/Emulation functions are built from the Dolby OEM board **CAT1100.** The graphic below illustrates the signal flow through it.

Important Note! The module **must** be routed into both the audio- **and** the metadata-signal paths. In order to decode a Dolby stream you **must** feed it to input **DEC1/2**. The metadata must be routed by the metadata router: **DOLBY PROCESSING > Metadata > Routing**.

The page embedded graphic shows the building blocks of the CAT1100 module. On the left hand side you have the decoding blocks, a signal router in the middle, and on the right hand side you have the downmix and the emulation part. You also can see the actual signal flow and their labels depending on the input signal status.



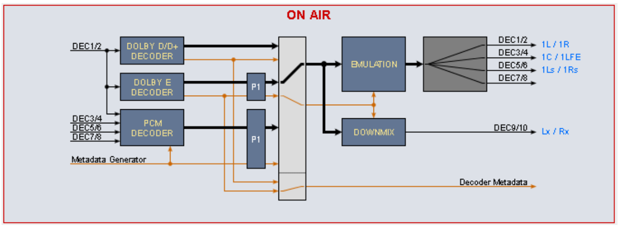
The emulation of the influence of metadata can be performed only on one program at a time. In the above case program 1 **"P1"** is pre-selected for emulation. But the signal is actually coming from the D/D+ decoder because a **D+** signal is present at DEC 1/2 input and will be decoded automatically. The metadata set of the **D+** stream has a channel mode of **3/2.** Therefore the output labels show a surround signal **1L/1R**, **1C/1LFE**, **1Ls/1Rs**, while the downmix output label is **Lx/Rx**.

If you feed PCM signals you have the setup mostly used for live or post pro mixing. The **D*AP8** may be connected to a monitoring insert of the mixing desk. The sound engineer can now switch between his mix and the emulated version of his surround mix or the downmix of it. He may now change DRC and downmix metadata by the generator to see how it would sound at home.

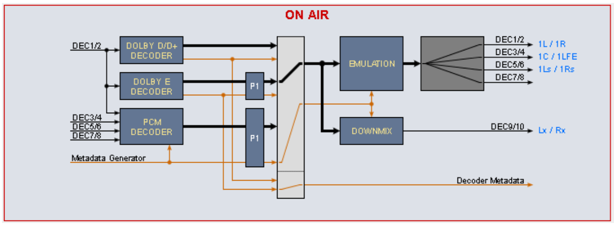
But he can also use external metadata from 9-pin input or from a SDI VANC stream which are routed to the metadata generator.
(see DOLBY PROCESSING > Metadata > Routing > Metadata Destination = D.Sub In).

Similar applies if one wants to listen to the influence of metadata from encoded streams.
A professional decoder would normally not apply metadata to the decoded audio as a TV set or a STB implementation would do. With emulation you can listen to it. This example shows a **Dolby E** decoding situation with **metadata** for **emulation** coming from

the decoder:



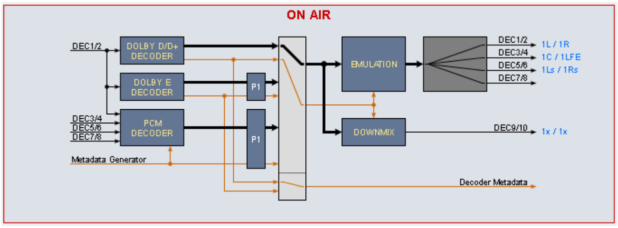
and alternatively from the generator:



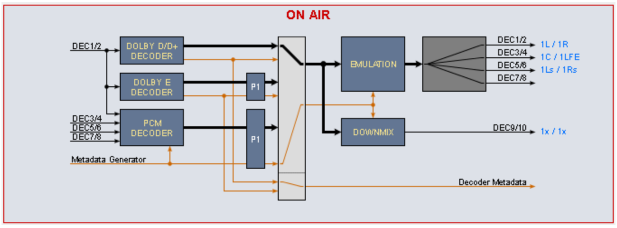
The right hand scenario allows for partially or fully overwriting the encoded metadata
(see DOLBY PROCESSING > DECODER/ EMULATION > Emulation > MD Generator overwrites encoded Metadata = ON)

Same applies to Dolby D / D+ decoding.

Metadata from decoder:

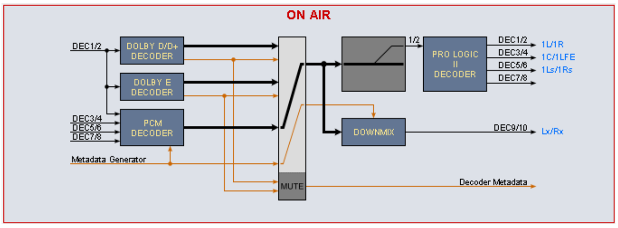


metadata from generator:

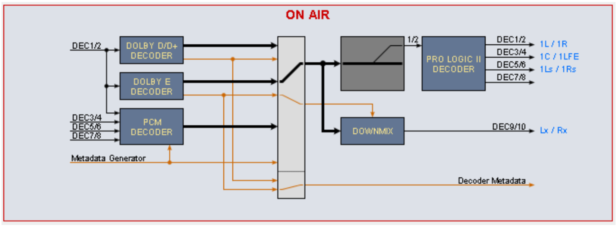


A special application is **Dolby Pro Logic** decoding. The **Pro Logic** technology does not have metadata like its younger digital family members. So in case a **Dolby Pro Logic** signal must be evaluated it will be passed straight through to the **Pro Logic** decoder.
But you may also listen to the Lt/Rt downmix (the Pro Logic format) by decoding it.

Decoding of **Pro Logic** from PCM input:



Pro Logic decoding from a D-E stream:

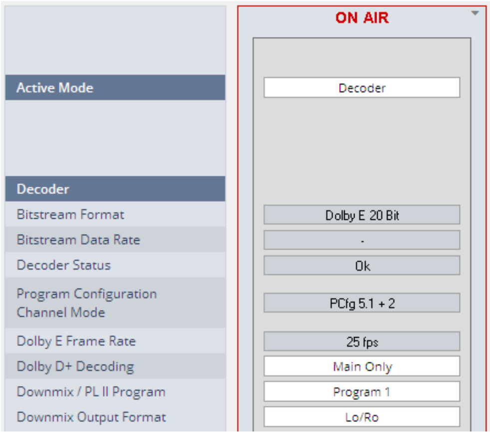


The configuration at the right hand side will only work if the channel mode of the selected program is 2/0. It will be used if the **Dolby Surround Mode** is set to **"Dolby surround encoded"** and one wants to listen to the decoded surround signals.

The **D*AP8** distinguishes between two major modes: **Decoding** (only) and **Decoding/Emulation**.
For the decoding part we have pre-settings for each decoding type. The format detection is automatic so the desired general settings like DRC modes must be set manually prior to decoding.

Important Note! For parameter consistency reasons the preset editor can only be used for the respective active mode of the ON AIR area. If the preset active mode does not match the one from ON AIR, preset set-up is disabled and you will get the message **"Setup not available for this mode"**.

Setup GUI – DOLBY PROCESSING – Decoder/Emulation - **Decoder**



Active Mode	**= Decoder**

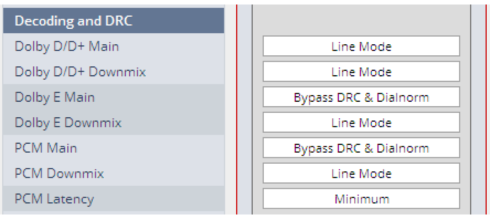
Decoder

Bitstream Format	[PCM / Dolby E 16/20/24 Bit Dolby Digital / Dolby Digital plus (I0, I0D0, I0I1, I0D0I1)] where Ix and Dx stands for independent and dependent sub stream IDs
Bitstream Datarate	[of a D-D or D-D+ stream]
Decoder Status	[OK / Fail]
Program Configuration **Channel Mode**	[in case of D-E] [in case of D-D / D-D plus]
Dolby E Frame Rate	[detected by the D-E decoder]
Dolby D+ Decoding	[Main Only, Mixed Main & AD, AD Only] Dolby Digital plus supports associated services like the provision of extra **dialog** or sending an audio descriptive (AD) track for **visually impaired** people or allows for separate **commentary** etc. that may be mixed automatically or by user intervention (depending on the consumer decoder implementation). This selection allows you to listen to the main program only, the main and the associated audio description (AD) signals mixed together or the associated audio descriptive (AD) signal only. It works only for streams where two **Dolby Digital plus** elementary streams are multiplexed (AKA single PID operation). For dual PID streams you may listen to the main and the associated signals independently only, because the Dolby OEM module has only one decoder input.
Downmix / PL II Program	[Program 1 / Program 2] Selects the program for downmix or PL II decoding. The drop down field becomes red colored if there is no second program available (e.g. PL II decoding from a D-D / D-D+ stream).
Downmix Output Format	[AUTO / Lt/Rt / Lo/Ro / Pro Logic II] AUTO=from Metadata, Lt/Rt (Pro Logic encoded), Lo/Ro (Stereo), Pro Logic II encoded.

The decoding functions of the **D*AP8** are implemented to meet all possible applications in the field. Besides monitoring for QA, broadcasters use decoded consumer format (D-D/D+) streams for turn around or backup applications. On the one hand they receive it from suppliers to add content to their bouquet and on the other hand they must maintain older distribution systems (cable head ends) which are based on AC3 encoding but (e.g.) are fed by D-D+. So often they can not / will not rely on the received Dialnorm / DRC settings because they prefer to add automatic levelling and standard DRC settings to all signals to have seamless loudness across their bouquet. That's why we offer to skip DRC & Dialnorm if it makes sense for the application.

Important Note! Metadata will be applied to the downmix output at any time. Either from the decoder or from the MD Generator (if input format is PCM). The selection is only regarding the DRC profile which will be used.

General settings are available for each of the possible input signal types (Dolby D/D+ / Dolby E / PCM):



Decoding and DRC

Dolby D/D+ Main [Bypass DRC & Dialnorm, Apply Dialnorm Only Line Mode, RF Mode, Mute Dolby D/D+]
This is a common setting for both D-D or D-D+.

D/D+ Downmix [Line Mode, RF Mode]

Dolby E Main [Bypass DRC & Dialnorm / Mute Dolby E]

Dolby E Downmix [Line Mode / RF Mode]

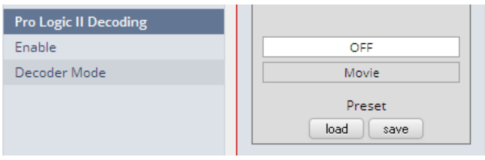
PCM Main [Mute PCM / Bypass DRC & Dialnorm]
Mute PCM is useful if one expects corrupted Dolby E blocks (if one runs a VTR or a switching device upstream is expected not to switch within the Dolby E guard band). In this case other than decoded Dolby E will not be audible.
Bypass DRC & Dialnorm must be used as an alternative setting (Mute PCM=OFF).

PCM Downmix [Line Mode / RF Mode]

PCM Latency [Matched, Minimum]

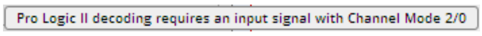
ProLogic II Decoding

There are a lot of **Pro Logic** / **Pro Logic II** consumer decoders installed and a lot of archived footage still has this sound track format. If you either must check such existing tracks or eventually produce such a sound track using the **Dolby DP563** (Pro Logic II encoder), you may also listen to the decoded signal via the **D*AP8**.



Pro Logic II Decoding

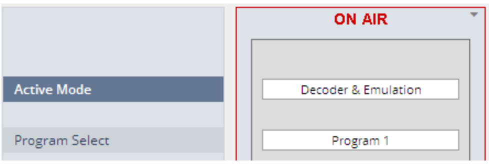
Enable [OFF / ON]
When you hover with the mouse over that pull down, a hint will be displayed:



Decoder Mode [Movie / ProLogic Emulation]

Setup GUI – DOLBY PROCESSING – Decoder/Emulation – **Decoder & Emulation**

For emulation five more parameters are available:



Active Mode **= Decoder & Emulation**

Program Select [Program 1 … Program 8]
SMPTE RDD6 standard defines up to 8 independent programs. For the emulation process you must select one program at a time.

Pls. refer to the **Decoder** > **Program Configuration** to see how many programs belong to an actual Dolby E stream.

Emulation

Unity Gain Mode [OFF / ON]
For applications like live mixing or others where the level must not be changed but listening to the influence of DRC is desired.

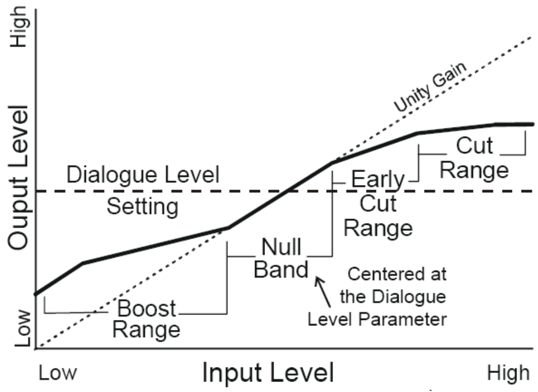
MD Generator overwrites encoded Metadata [OFF / ON]
If you want to see how different metadata will "sound" for already encoded signals you may decode it and apply different ones to it.

Decoding and DRC

Custom Mode Boost Factor [0 … 0.5 … 1]
If you want to check out different DRC behaviour from the defined profiles you may set the lower level boost factor here.

Custom Mode Cut Factor [0 … 0.5 … 1]
If you want to check out different DRC behaviour from the defined profiles you may set the higher level cut factor here.

Here a simplified DRC characteristic curve, published by Dolby®:



Important Note! Dolby Digital and Digital plus encoded streams do **not** contain metadata for DRC but pre-calculated gain words which may be applied to the decoded audio to decrease dynamic range for home reproduction. That's why you will **not** get a display of such metadata from the **Input** if consumer format streams are decoded. Similar applies to the professional metadata which is used to setup consumer format encoders (e.g. filters) and which is not present in the metadata stream as well.

Status display of **Decoder/Emulation** / **Encoder A** / **Encoder B** / **Metadata** (soft LEDs)

Green
* Dolby encoded stream at the input
* Metadata valid from the generator

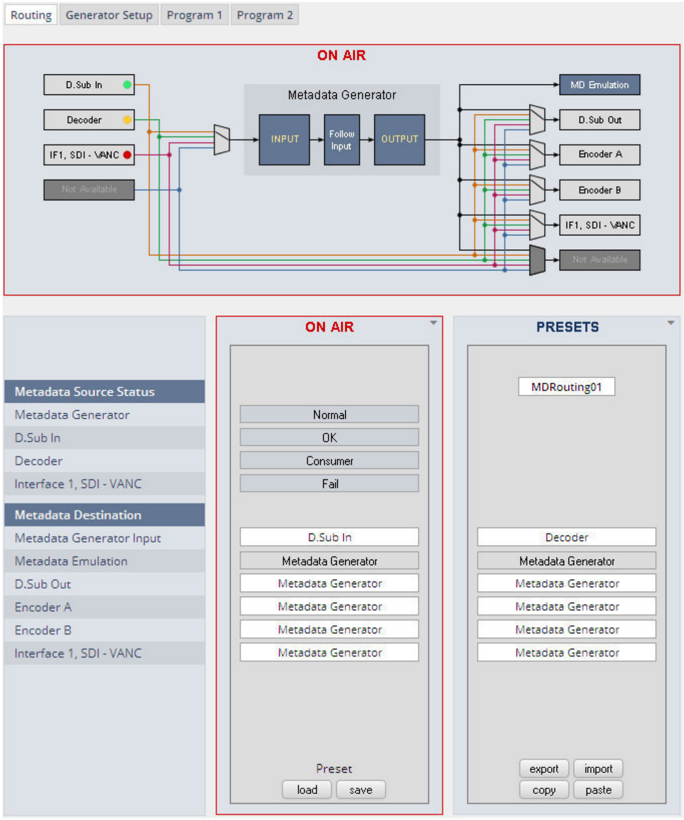
Orange
* Dolby E frame rate mismatch
* MD generator has entered the reversion mode
* Dolby E encoder has entered the reversion mode

Red
* If the decoder receives corrupted (e.g. asynchronous) or no metadata
* Internal error

Important Note! If no input metadata is available for PCM emulation and you tick a **<Follow Input>** checkbox, the generator enters the reversion mode as well.

Setup GUI – DOLBY PROCESSING – Metadata – **Routing**

The center of the **D*AP8** Dolby processing is the **Metadata Processor**. It can be the point of origin of metadata but it may also modify existing metadata from available sources:



The metadata processor of the **D*AP8** has a maximum of seven metadata destinations and four sources which can be routed individually.

The **Metadata Generator** in the middle can run independently but may take metadata from an available source at the **"Input"**, may select some or all of it in the "**Follow Input"** section and present a complete set of metadata at the "**Output"**.

Metadata Source Status - colors
The respective soft LED turns **red** if **no** metadata is present or the metadata are corrupted.
It turns **green** if a **RDD 6** compliant metadata stream is detected.
It turns **yellow** if an AC3 or similar (D-D+) signal is decoded.

Metadata Source Status
[OK / Consumer / Fail / Not Available]
The word "**CONSUMER"** will be displayed to indicate that only a metadata subset is provided.

Metadata Destination	[OFF / D.Sub In / SDIx - VANC (if present) / DECODER (if present)] The destinations can have any of the system sources assigned except of the emulation engine [MD Emulation].

Setup GUI – DOLBY PROCESSING – Metadata – **Generator Setup**

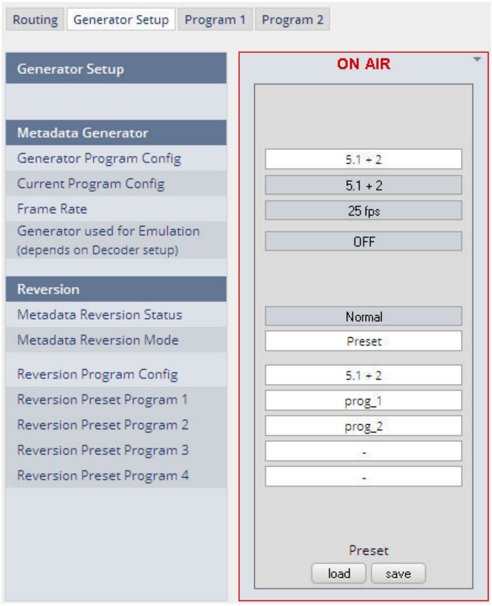
The metadata processor generates **SMPTE RDD 6** standard compliant metadata. It supports the most relevant program configurations for broadcast applications (5.1 / 5.1+2 / 3x2 / 4x2) used with Dolby E 16 or 20Bit bit depth. Since the number of programs from an external RDD 6 stream may differ from the generator setup, "off-size" program configurations will be handled this way:

If the input program configuration has more programs (e.g. 4x2) than the generator setup (e.g. 5.1+2) and you click on a "surplus" program (Program 3 or Program 4), only an Input table will be displayed while for the other programs an input and an output table is shown.

If the input program configuration has less programs (e.g. 3x2) than the generator setup (e.g. 4 x 2) and you click on a "surplus" program (e.g. Program 4), an empty input table will be shown.

If the metadata generator is set up for "Follow Input" and the input program configuration does not match the possible ones of the metadata generator it enters the reversion mode.

The output from the **metadata generator** is the source for the **emulator engine** but may also be selected for optional built-in encoders and for metadata transport interfaces like **9-pin** (RS485) or **VANC** (SMPTE 2020).



Metadata Generator

Generator Program Config. [Follow Input / 5.1+2 / 4 x 2 / 5.1 / 3 x 2]

Current Program Config. displays the actual program configuration used by the generator.

Frame Rate display of the frame rate SYSTEM > Setup > Video Rate (fps).

Generator used for Emulation (depends on Decoder Setup) [OFF / ON] shows if the generator is used for emulation or not.

Reversion

Metadata Reversion Status [Normal / Reversion] Display of the reversion mode status.

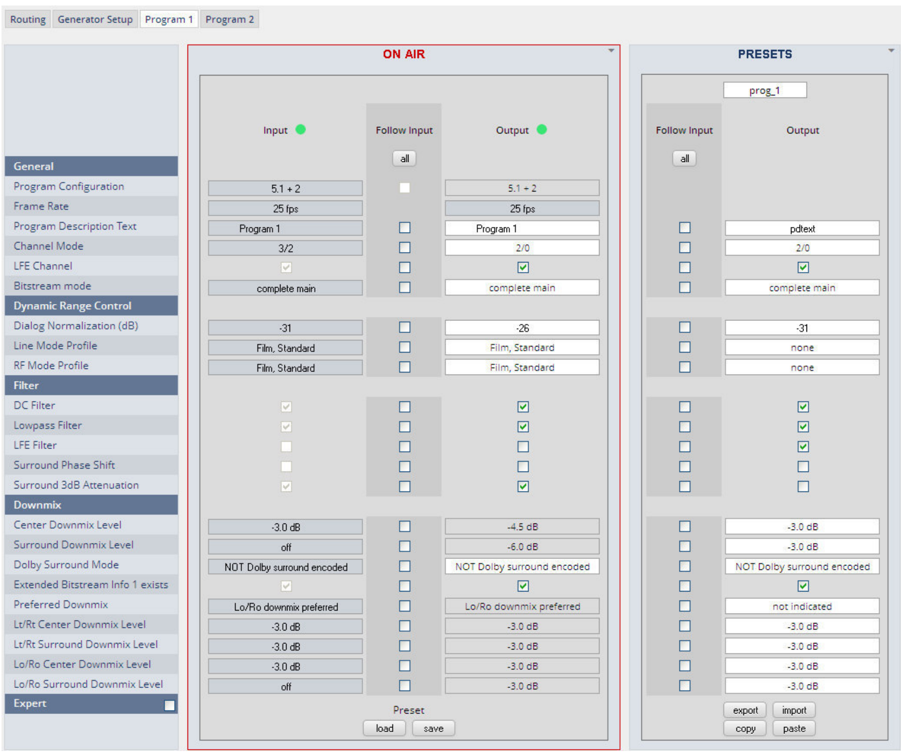
Metadata Reversion Mode [Last Valid / Preset] Selection of what happens in case of input metadata failure.

Reversion Program Program Config. [5.1+2, 4 x 2, 3 x 2] Pre-selection of the program configuration for reversion mode.

Reversion Preset Program x You can select a preset for **Program x** to become the Reversion preset for that program.

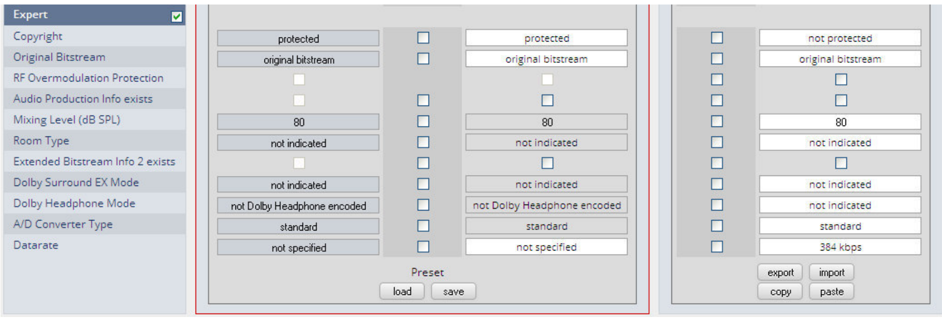
Important Note! There is only one set of reversion presets for all programs. You must be careful when you assign reversion presets to programs. It may be a good idea to name the presets used for reversion mode after the program number it is meant for.

Setup GUI – DOLBY PROCESSING **–** Metadata – **Program x**



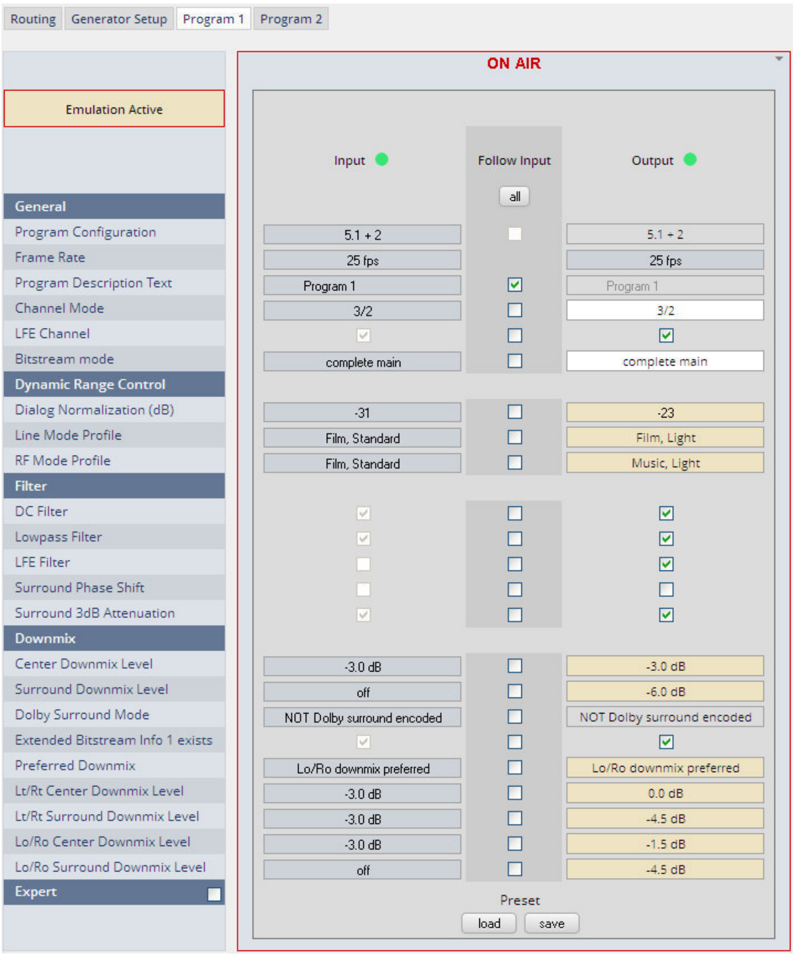
Above you can see the input metadata of the processor and you can decide about the metadata output. You may set it to follow the input or you may overwrite it. The table shows the most relevant metadata.

The Expert checkbox gives you access to more specific metadata:



Important Note! Dolby advises that the **RF Overmodulation Protection must be off**. Therefore Junger automatically turns it off. You are not able to set this parameter and no <Follow Input> check box exists, except for the preset parameters which will be ignored when loading it.

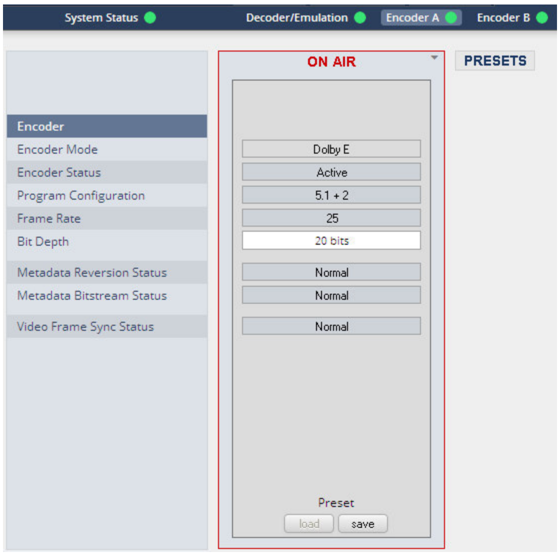
If **Emulation** is active and the option **"MD Generator overwrites encoded Metadata"** is turned on, the metadata are used for emulation are highlighted by a yellowish background:



This example shows the metadata from **Program 1** of a Dolby E encoded stream.

Setup GUI – DOLBY PROCESSING – optional Dolby E encoder – **Encoder A**

If the optional Dolby E **encoder** is **licensed** (see SYSTEM > Firmware Update > Licensing) the UI shows it as Encoder A:

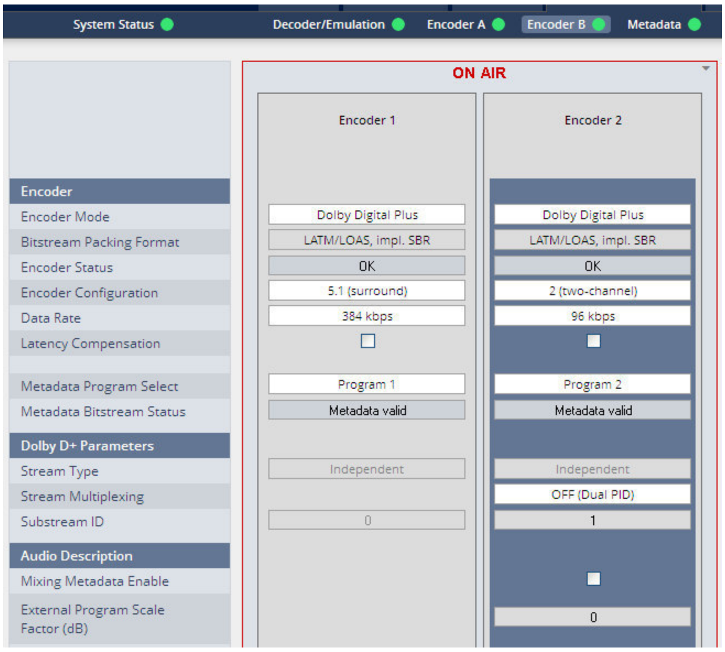


Encoder

Encoder Mode	[Dolby E]
Encoder Status	[Active / Metadata Reversion / Fail]
Program Configuration	[3x2 / 4x2 / 5.1 / 5.1 +2] Set by the generator
Frame Rate	[25 / 30 / 29,97 / Unknown]
Bit Depth	[20 bits / 16 bits]
Metadata Reversion Status	[Normal / Reversion]
Metadata Bitstream Status	[Normal / Fail]
Video Frame Sync Status	[present at Dolby E frame rate]

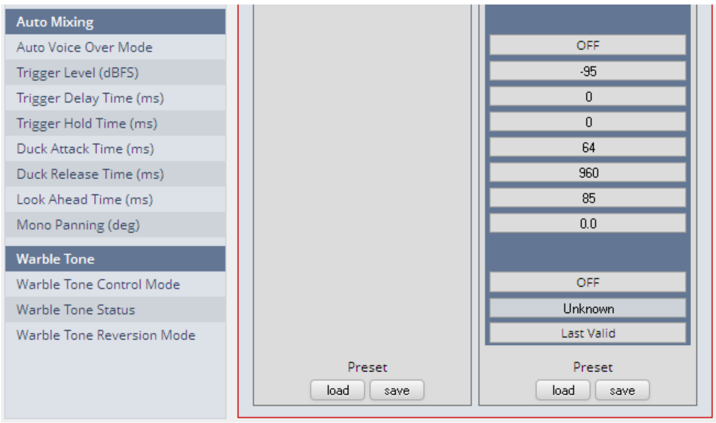
Setup GUI – DOLBY PROCESSING – optional consumer format encoder – **Encoder B**

The **D*AP8** offers the option to install a consumer format (Dolby Digital / Dolby Digital plus / HE-AAC (v1/v2) / AAC) or another optional Dolby E encoder. If an encoder is installed it shows up under **DOLBY PROCESSING**. This example has a consumer format encoder installed:



The **OEM** module from Dolby called **CAT561**. The implementation for the **D*AP8** platform provides two encoded outputs. Both outputs may have independent consumer formats. If both encoders are set for **Dolby Digital plus** encoding special features like providing associated services (e.g. an extra audio track for visually impaired people, AKA audio descriptive service - AD) are available.

Encoder	**Encoder 1** (similar applies to **Encoder 2** accept from setup where both encoders are used for associated services).
Encoder Mode	[Dolby Digital plus, Dolby Digital, Dolby Digital Puls HE-AAC v1, Dolby Digital Pulse HE-AAV v2, Dolby Digital pulse AAC] Here you may select the encoding format for the respective encoder
Bitstream Packing Format	AAC encoded bit-streams may be packed in different container formats. This parameter allows you to select one from the many possible formats.
Encoder Status	[OK, Fail]
Encoder Configuration	[2 (two-channel), 5.1 (surround)]
Data Rate	The data rate that is used for encoding
Latency compensation	[ON / OFF] For parallel encoding of different formats the same latency may be desirable. In this case both encoders will have the same latency of 305ms. If you turn latency compensation OFF, latency will be reduced to 135ms for Dolby Digital.
Metadata Program Select	[Program 1 … Program 8] Here you can select a program number of the RDD6 metadata set that shall be used for consumer encoding. If you are about to encode a 5.1 program that comes with a Dolby E stream as program 1, you must select Program 1 here.
Metadata Bitstream Status	[Metadata valid, Metadata not present]
Dolby D+ Parameters	
Stream Type	[Independent, Dependent] The streams which are encoded by both encoders can either be independent (i.e. there is no signal relationship of the audio signals) or dependent (if you use both encoders to encode 8 audio channels for 7.1 encoding).
Stream Multiplexing	[OFF (Dual PID) / ON /Single PID]]
Substream ID	[1, 2, 3] Since the encoded streams can be multiplexed by an on-board multiplexer they must have individual (sub-) stream IDs, so a de-multiplexer "knows" which data belong to which stream. If there is no intention to multiplex them together, the D*AP8 sets both IDs to "0".
Audio Description	Audio description service employs both encoders to allow for the so called **receiver mix**. I.e. the mix between the program sound and the narrator who performs the audible scene description can be done in the decoder. This saves a lot of audio bandwidth compared to the so called broadcast mix, where two independent audio mixes are transmitted to the receiver at home.
Mixing Metadata Enable	[ON / OFF]
External Program Scale Factor	[-50 … 0 … 12] To remote control the mixing of associated services you can change the level of the main program with this parameter.



Audio Mixing

Auto Voice Over Mode
[OFF / ON]
In case of ON, the ducking parameter below will be used by the receiver to perform the mixing.

Trigger Level (dBFS)
[-96 … 0]
Level of the associated audio channel that will turn on the ducking.

Trigger Delay Time (ms)
[0 … 4992]
Time that must elapse before ducking becomes active after the trigger detects a signal that is above the trigger level.

Trigger Hold Time (ms)
[0 … 4992]
Time the ducker stays open after trigger becomes active.

Duck Attack Time (ms)
[0 … 4992]
Time the ducker needs to fully open up.

Duck Release Time (ms)
[0 … 4992]
Time the ducker needs to fully close.

Look Ahead Time (ms)
[0 … 85]
Time to look in advance for the level in the associated channel.

Warble Tone

Warble tone is a BBC invention to encode the volume and PAN values into one audio track while the other track carries the narrators voice signal.

Warble Ton Control Mode
[OFF / ON]

Warble Tone Status
[Unknown / Not Available / Not Valid / Valid]

Warble Tone Reversion Mode
[Last Valid / Internal / Automatic]

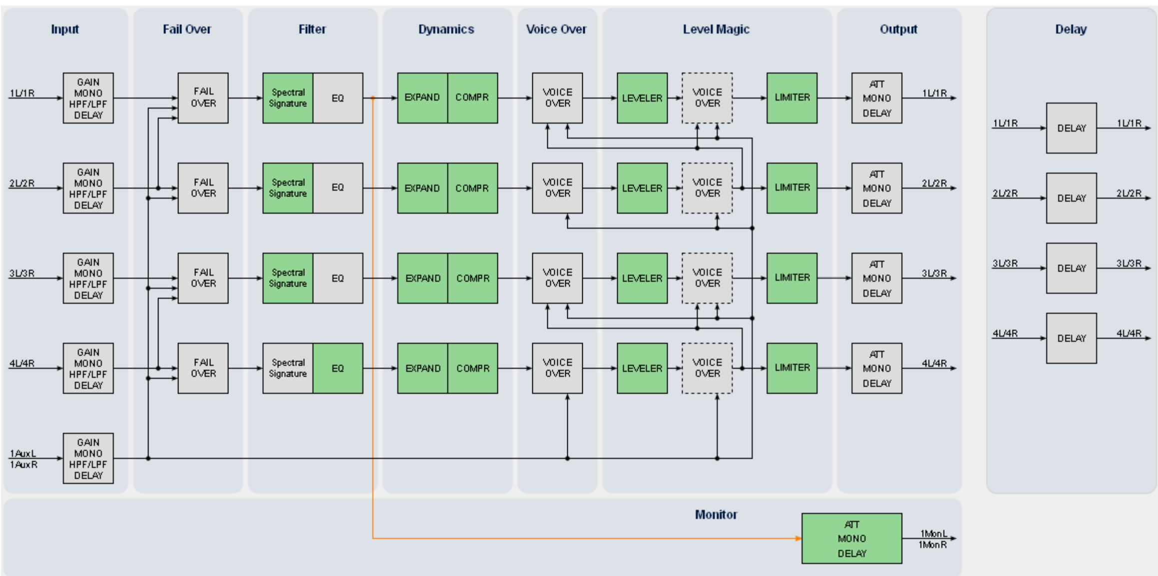
Setup GUI – AUDIO PROCESSOR - **Overview**

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.

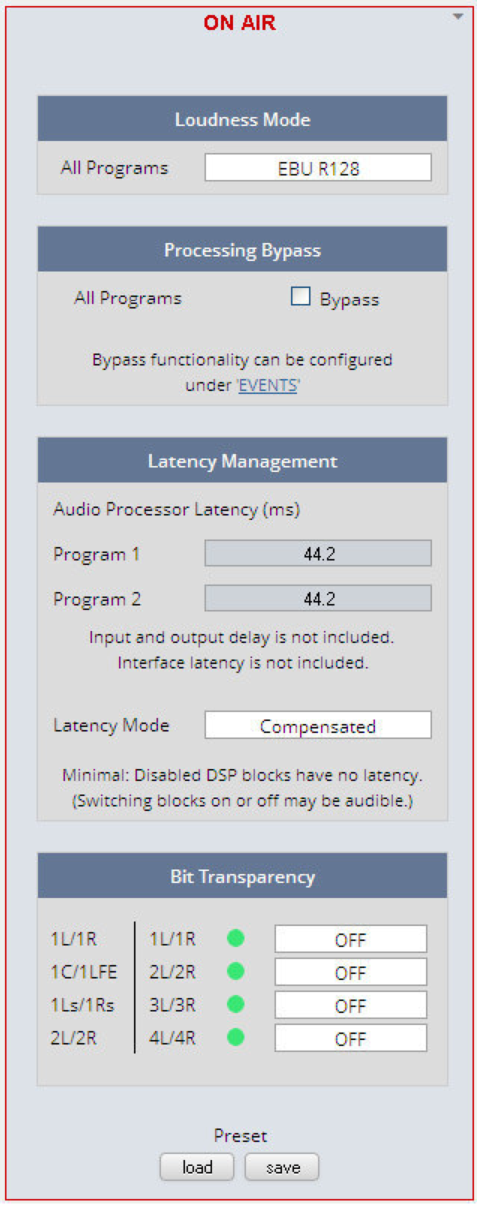
To navigate through the various processing blocks you may either click on the graphical block above or use the tabs provided in the navigation bars below the bar graph displays.

The **D*AP8** knows two major setups fro the number of programs which may be processed.
See SYSTE > Setup > Program Configuration
This is a general setting and has influence on all following GUI pages of the audio processor. Above we see the appearance for 5.1 +2. If the device is set for 4 x 2 it will look this way:



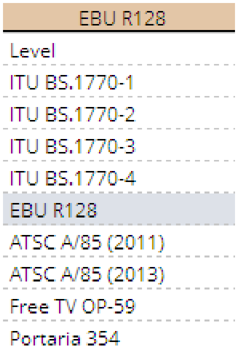
Important Note! The explanation of the Audio Processor functions and their respective parameters will be done for the 5.1 + 2 program configuration. The 4 x2 values are a subset and may contain less functions (e.g. no upmix is available in that mode) but if necessary both configurations will be explained.

Setup GUI – AUDIO PROCESSOR - **Setup**



Loudness Mode

All Programs



In order to meet the regulations of regions or countries you must select the loudness control mode here. Beside of the weighting curves several measurement duration and loudness ranges have bee 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 / 2

Display of the actual latency. This example has turned upmix on for program 1. I.e. program 2 is compensated so both have 40+ ms.

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.

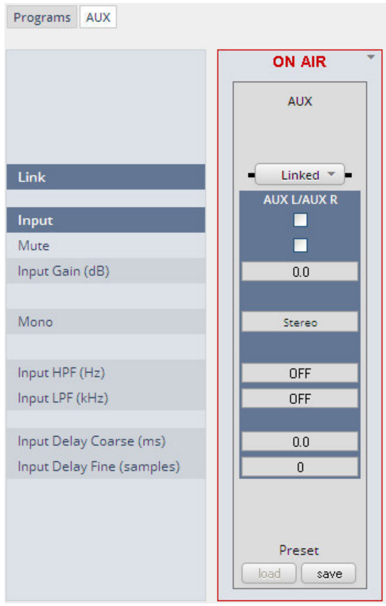
Bit Transparency

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

1L/1R | 1L/1R	[OFF / ON / AUTO]
1C/1LFE | 1L/2R	[OFF / ON / AUTO]
1Ls/1Rs | 1Ls/Rs	[OFF / ON / AUTO]
2L/2R | 2L/2R	[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.

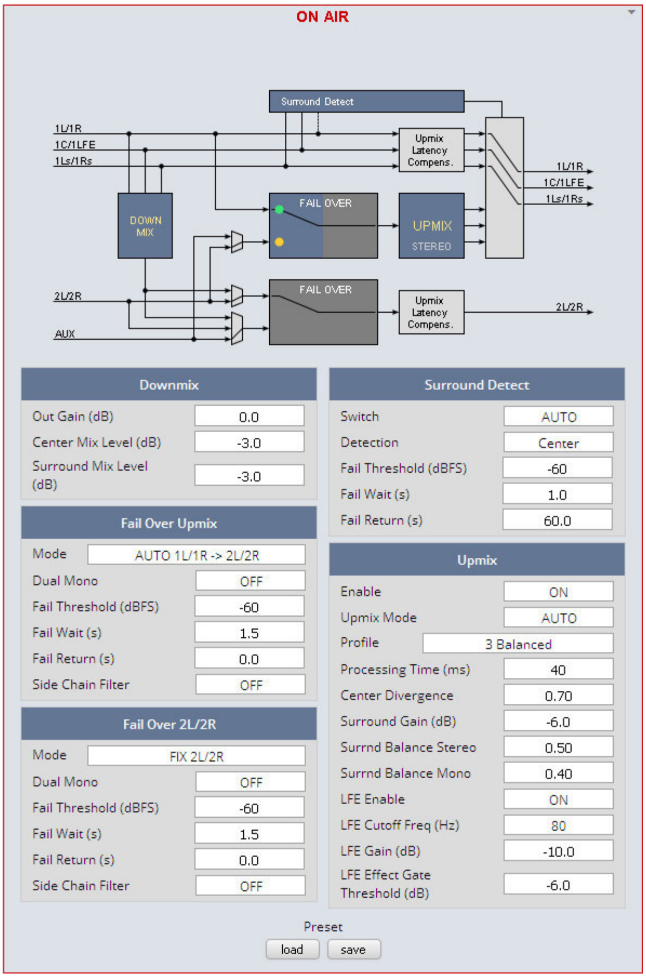
Setup GUI – AUDIO PROCESSOR – **Input**



You may set the input conditions for both programs (Program 1 and 2) and the AUX path here:

Link (Program 1)	[Quad / Movie / Live / Linked / Linked & LFE] You may select one of the possible multichannel modes to enable gang setting of the parameter values
Link (Progarm 2)	[Linked / Unlinked] For stereo operation you may link the setup parameters Similar applies fro the AUX path.
Input	[Enable / Disable] Enables or disables the input section
Mute	[ON / OFF]
Input Gain (dB)	[-80.0 … 0.0 … 20.0]
Mono	[Stereo / L+R Mono / L/L Mono / R/R Mono] Not applicable for multichannel program 1
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 – **Upmix (5.1 + 2) & 2ch Fail Over**



Junger Audio provides a **new** 5.1 upmix algorithm for upmixing stereo or even mono sources to multichannel surround sound while remaining acoustically downmix compatible. This is a real-time process which does a frequency analysis of the input signal. As known from the mathematical theory, the longer the time for such an analysis the better the result. But this will introduce more delay for the audio path, compared to the video. This delay, if acceptable in general, may be compensated by the video delay of the SDI embedder.

Please note that presets created with earlier firmware version are **not compatible** with the new upmix algorithm!

You may take the upmix source signal from either the surround Left/Right input (in case it provides stereo PCM instead of surround L/R) or from pre-selectable inputs (2L/2R or AUX).

The **Surround Detect** circuit monitors the input channels to decide if the surround signal has disappeared in order to do an automatic upmix if desired. But the upmix may also be forced by an event of the system that loads a preset configuration, that turns the upmix permanently on.

Downmix

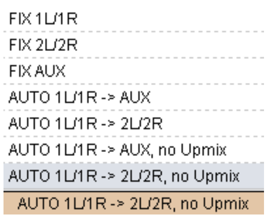
Out Gain (dB) [-20.0 … 20.0]
output gain of the downmix signal

Center Mix Level (dB) [0.0 … -12.0]

Surround Mix Level (dB) [0.0 … -12.0]

Fail Over Upmix switch that provides the upmix block with an input signal for upmix or pass through if the source is not intended do be used for upmixing.

Mode



The switch may be permanently [FIX] connected with either the 1L/1R, 2L/2R or AUX input but it may also perform an [AUTO] switch over from 1L/1R to AUX or 1L/1R to 2L/2R if the first signal fails.
Both options may also turn the upmix off [no Upmix]. I.e. the fail over signal will not be upmixed.

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

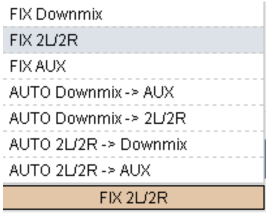
Fail Wait (s) [1.5 … 10.0]
Elapsed time after fail detection until the switch over will happen.

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.

Fail Over 2L/2R Switch that provides an independent stereo fail over circuit.

Mode



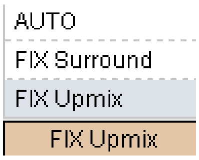
the switch may be permanently [FIX] connected with either the Downmix, 2L/2R or the AUX input but may also perform an [AUTO] switch over from the first input to the alternative input.

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)	[-60 … -40]	see description for **Fail Over Upmix** block above.
Fail Wait (s)	[1.5 … 10.0]	see description for **Fail Over Upmix** block above.
Fail Return (s)	[0.0 … 10.0]	see description for **Fail Over Upmix** block above.
Side Chain Filter	[OFF / ON]	see description for **Fail Over Upmix** block above.

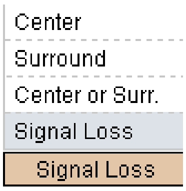
Surround Detect To perform an automatic upmix in case the main surround signal fails.

Switch



The surround switch may be permanently [FIX] connected with the surround input or the upmix output but it may also perform an [AUTO] switch over in case the surround input fails.

Detection



Here you can decide which channels must be observed for signal loss to operate the surround switch. This switch is independent from the upmix state! You are able to feed the 1L/1R output even if the upmix is not activated either by "**Upmix Enable**=Off" or by "**Fail Over Upmix**=AUTO no upmix" setting of that switch.
Signal Loss=All channels are gone.

Fail Threshold (dBFS)	[-80 … -40]	see description for **Fail Over Upmix** block above.
Fail Wait (s)	[0.0 … 10.0]	see description for **Fail Over Upmix** block above.
Fail Return (s)	[0.0 … 120.0]	see description for **Fail Over Upmix** block above.

Upmix

Enable [OFF / ON]

Upmix Mode [Mono / Stereo / Auto]

Profile [1 Front Projection, 2 Emphasize Front, 3 Balanced, 4 Emphasize Surround, 5 Wrap Surround]

1 Front Projection – Optimized for a stable surround image, independent from correlation of the input signal. Opens a stage-like presentation over the front speakers and uses the rear channels for ambience creation.

2 Emphasize Front – Based on setting 1 with a less strict front projection.

3 Balanced – A balanced distribution of the signal between the front and rear channels. Without overemphasizing the rear channels.

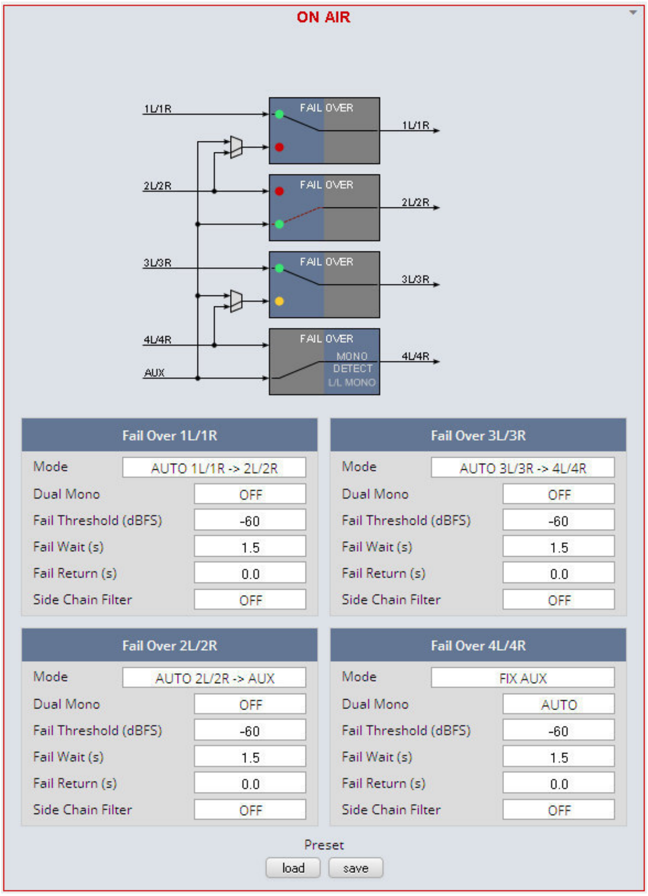
4 Emphasize Surround – The distribution between the front and rear channels is highly dependent on the correlation of the input signal. Highly uncorrelated signals may create emphasized surround channels.
5 Wrap Surround – Even distribution of the signal between all channels, to create a feeling of being 'wrapped in sound' for creating spectacular effects.

Processing Time (ms)	[3 … 100] the processing time has great influence on the quality of the upmix process but of course alters the latency of the audio signal. It is highly recommended to allow as much processing time as possible. One can e.g. rise the processing time instead of adding audio delay to compensate for a delayed video line. Depending on the system latency requirements (ingest vs. live broadcast) you may change the processing time accordingly.
Center Divergence	[0.0 … 1.0] the upmix process assembles a center signal from the input stereo. It may either be fed to the center channel only (0.0) or spread between L/R (1.0). The effect will be a wider presentation of center signals in a surround sound image. Please note that the signal does not completely disappear from one source (L/R or C) depending on the selected profile.
Surround Gain (dB)	[0 … -24.0] sets the level of Ls/Rs channels.
Surround Balance Stereo	[0.0 … 1.0] defines the amount of direct sound mixed into the surround channels. 0.0 provides pure ambient sound while 0.1 to 1.0 will increase the amount of direct sound. Works only, when upmix mode is set to Stereo or switched to Stereo in Auto mode.
Surround Balance Mono	[0.0 … 1.0] defines the amount of direct sound mixed into the surround channels. 0.0 provides pure ambient sound while 0.1 to 1.0 will increase the amount of direct sound. Works only, when upmix mode is set to Mono or switched to Mono in Auto mode. For Auto mode lower values (0.2 – 0.4) are recommended to prevent unwanted effects when auto switching between Mono and Stereo.
LFE Enable	[OFF / ON / Effect Gate] you may turn this option on if the upmix process shall generate a subwoofer signal that will appear in the LFE channel. When using the Effect Gate function the system interactively processes the subwoofer signal and generates a signal that comes very close to a real LFE signal, without creating permanent rumble and bass excitation.
LFE Cutoff Freq (Hz)	[60, 80, 100, 120] set the cutoff frequency for the generated LFE signal.
LFE Gain (dB)	[-20.0 … 20.0] you can set the LFE level here
LFE Effect Gate Threshold (dB)	[0.0 … -20.0] set the relative threshold of the Effect Gate processor.

Important Note! If you encode the surround signals from the upmix to Dolby format we recommend to set the center and the surround downmix level to -3dB for best downmix compatibility.

setup GUI – AUDIO PROCESSOR – **Fail Over** (4 x 2 program configuration)

For the **4x2 Program Configuration** (SYSTEM > Setup > Program Configuration) the **D*AP8** offers **four** independent **Fail Over** circuits (see Overview sketch).



The source for the Fail Over circuit can be either the adjacent program input (e.g. input 2L/R for the program input 1L/1R) or the **AUX** input. The **Mode** switch will select the respective signal path.

See the example above for the four program outputs :

program 1 (1L/1R) has a valid input signal and is prepared for auto switch over to the second program input **2L/2R.**

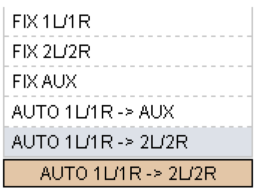
program 2 (2L/2R) has no valid input and has automatically switched over to the **AUX** input.

program 3 (3L/3R) has a valid input and is prepared for auto switch over to input **4L/4R,** input **4L/4R** has valid input. This is indicated by the **yellow** soft LED.

program 4 (4L/4R) is fix connected to **AUX.** Signal input is mono L.

Fail Over 1L/1R Example description of the fail over function blocks

MODE



The Fail Over output can be permanently connected to:

* its program input **1L/1R**

* its adjacent program input **2L/2R**

* or to the **AUX** input.

Automatic switch over in case of an input failure may be configured for the **AUX** or the adjacent **2L/2R** input.

Dual Mono
[OFF / AUTO]
A detector looks for 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 … -40]
RMS weighted input level for fail detection

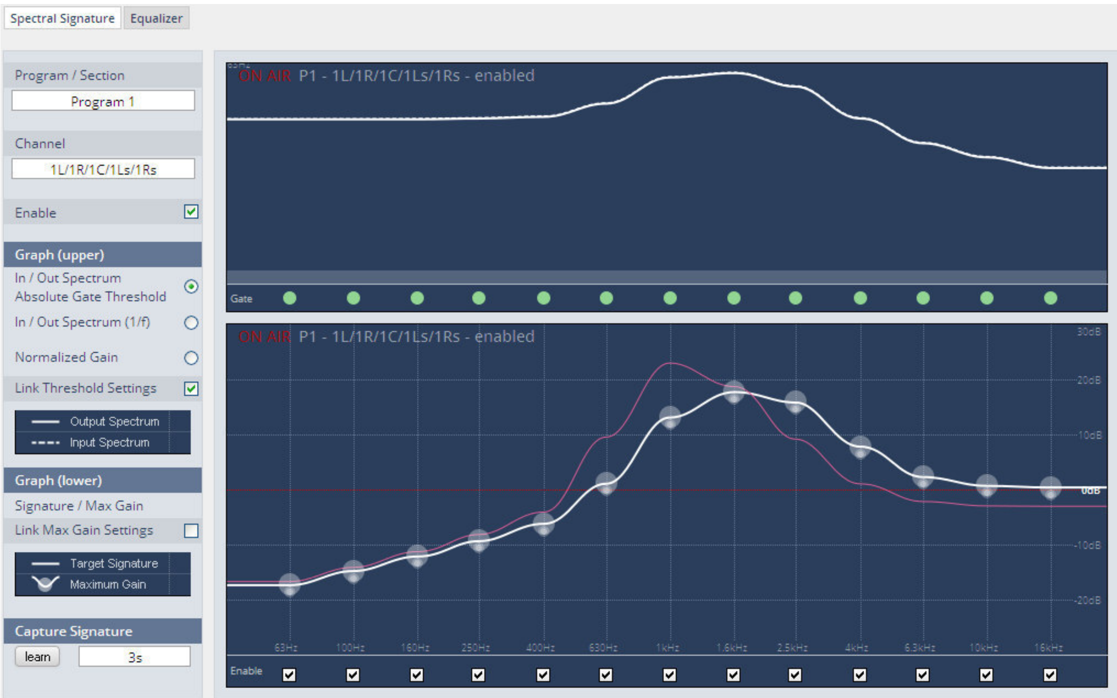
Fail Wait (s)
[1.5 … 10.0]
elapsed time after fail detection until the switch over will happen

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 – 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

[ON / OFF]
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.

Graph (lower)

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.

Signature / Max Gain (dB)

[0 … 12]
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.
On mouse over you can read the actual setting of a particular band (BAND 5 above).
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.

Link Max Gain Settings

[ON / OFF]
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.

Enable

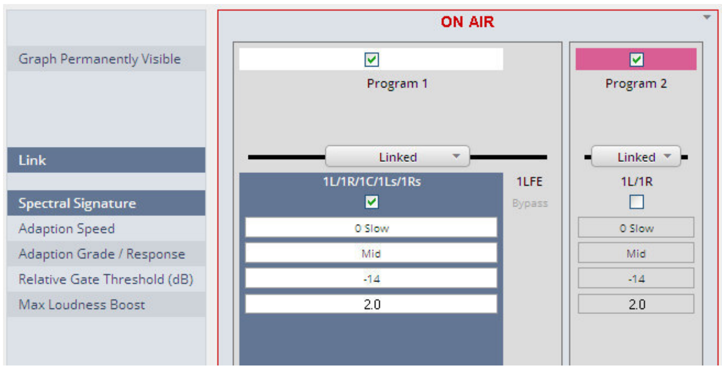
[ON / OFF]
Checkboxes on the bottom of the lower graph can be used to bypass single bands from processing.

Capture Signature

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.

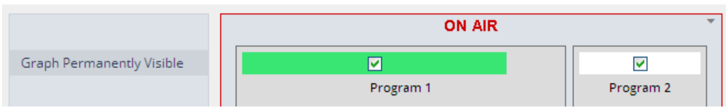
<learn>

[manual / 1s … 30s / 1min]
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).



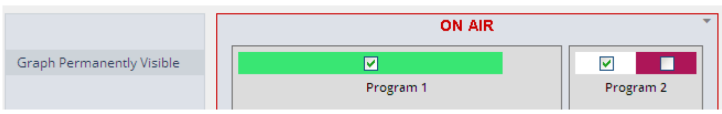
Graph Permanently Visible

[ON / OFF]
The color code of the column headers will change depending on the program selected for gain change (upper) display.
White color represents the selected program while pink represents the second program. If you select program 2 for example it becomes white while Program 1 becomes green:



Link

[Linked / Unlinked] for program 2
I.e. for non stereo operation you may unlink the function and a 3rd color is introduced because we have now 3 independent Spectral Signature processes running:



Spectral Signature [ON / OFF]

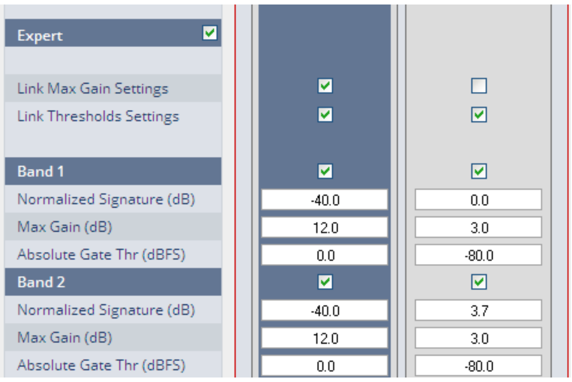
Adaption Speed [0 / 2 Mid / 3 / 4 Fast]
This parameter affects the time taken for the bands to reach their target values. Fast settings even out differences between sources, but can lead to audible transitions. They are well suited for single channel signals, for example to even out sound differences 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 5dB 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'.

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.

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]

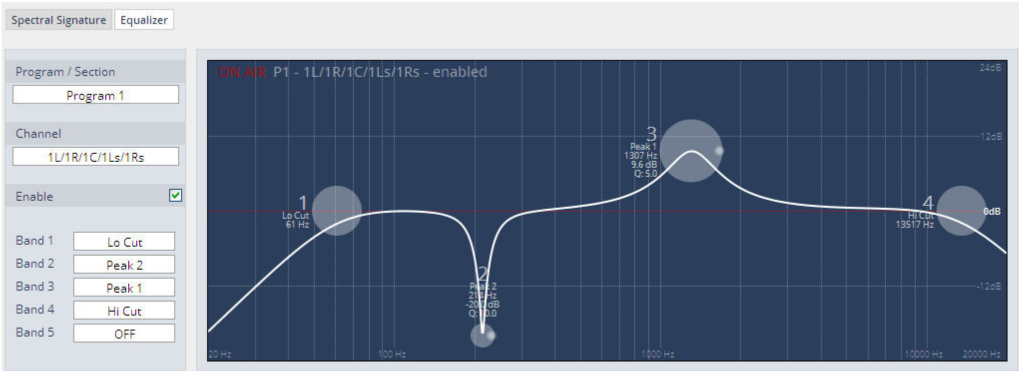
Normalized Signature level [-40.0 … 0 … 40.0]

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**



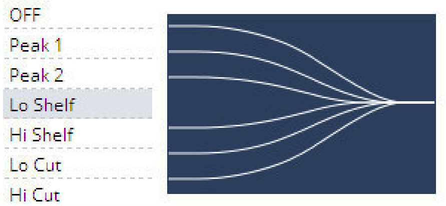
The graphical EQ offers 5 bands. The characteristic of each band can be setup either left hand side of the graph or alternatively for each band further below.

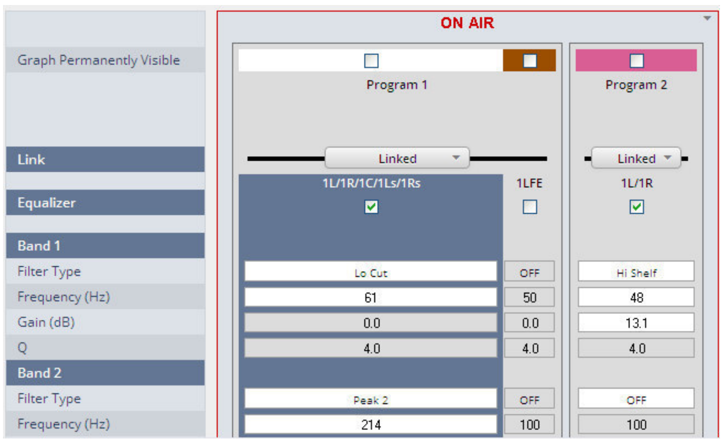
Program [Program1 / Program2 / Preset]
Selects the source for which the curve will be displayed.
This selection depends on the Voice Channel Mode
(see SYSTEM > Setup) and whether or not the channels are linked for stereo operation.

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 :

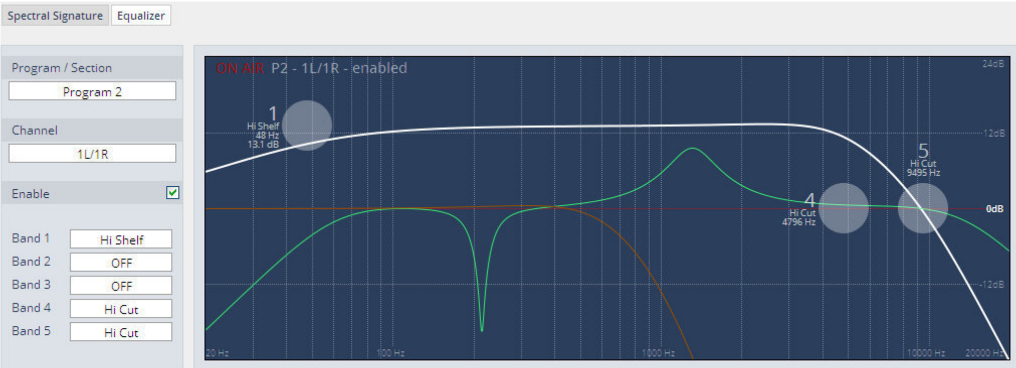




Graph Permanently Visible [ON / OFF]
The color code of the column headers in the lower display will change depending on the selected program.
White color represents the actual selected program while brown represents the LFE channel in the example above and pink represents the second program.

Important Note! For numeric input double click into the parameter field. You must use the period as a decimal separator. For graphical input use the left mouse button and drag it horizontally to change frequency and vertical to change gain while the mouse wheel will change the Q value.

If one selects another program for EQ setting the color code changes accordingly. If all 3 columns in the ON AIR section are activated for processing (Equalizer = ON) you can see three graphs in the display:



Link (Program 1) [Quad / Movie / Live / Linked / Linked & LFE]
You may select one of the possible multichannel modes to enable gang setting of the EQ parameter values.

Link (Program 2) [Linked / Unlinked]
For stereo operation you may link the setup parameters.

Equalizer [ON / OFF]

Band 1 The parametric EQ function block offers 5 bands for each program / channel (if unlinked).

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

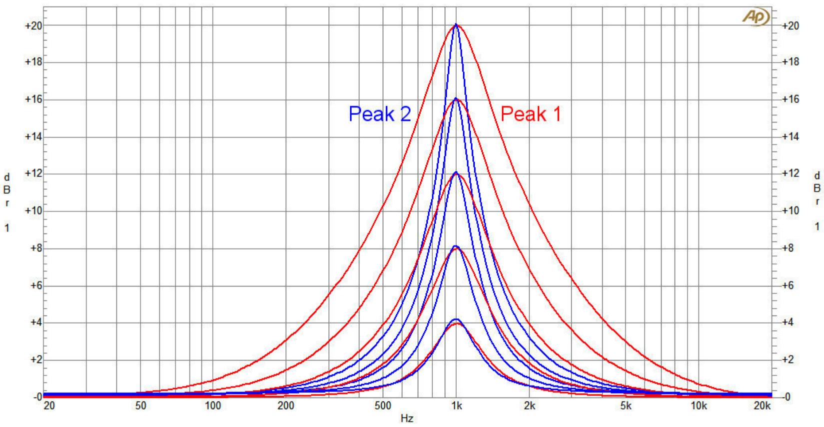
Frequency (Hz) [20 … 20000]

Gain (dB) [-20.0 … 20.0]

Q [0.4 … 4.0]

Band 2 … 5 similar parameters as Band 1

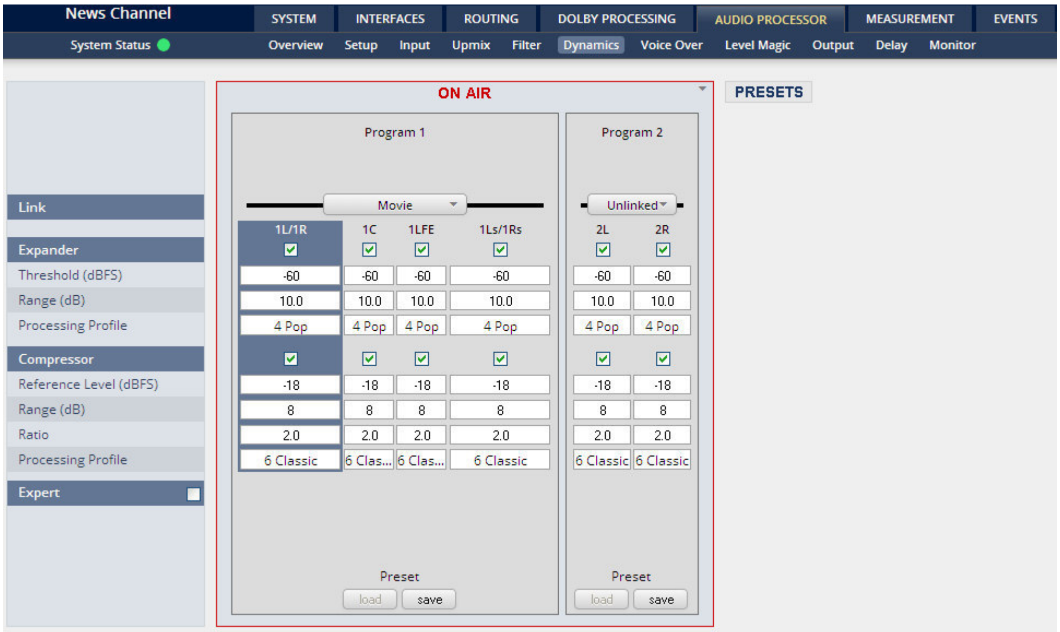
The **EQs** offer two different peak modes :



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.

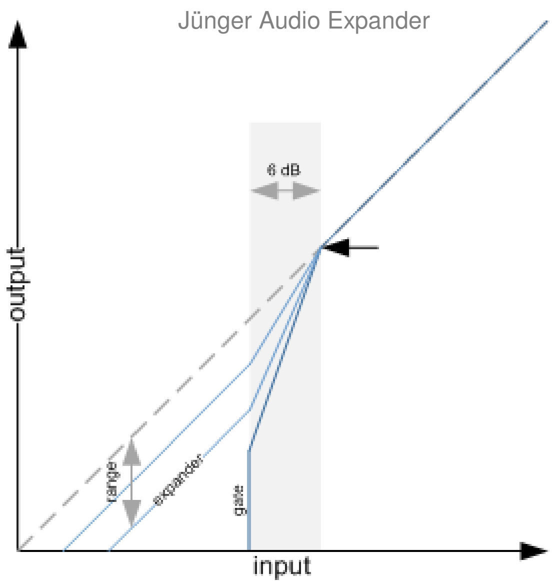
Peak2:
The bell curves of the **Peak 2** filter also features constant quality (Q) over gain.
But Q is defined at 50% of gain. Subjectively the bell curve becomes sharper when increasing gain, but this is only true for the lower 6-8dB of gain.

Setup GUI – AUDIO PROCESSOR – **Dynamics**



Link (Program 1)	[Quad / Movie / Live / Linked / Linked & LFE] You may select one of the possible multichannel modes to enable gang setting of the EQ parameter values.
Link (Progarm 2)	[Linked / Unlinked] For stereo operation you may link the setup parameters.
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.

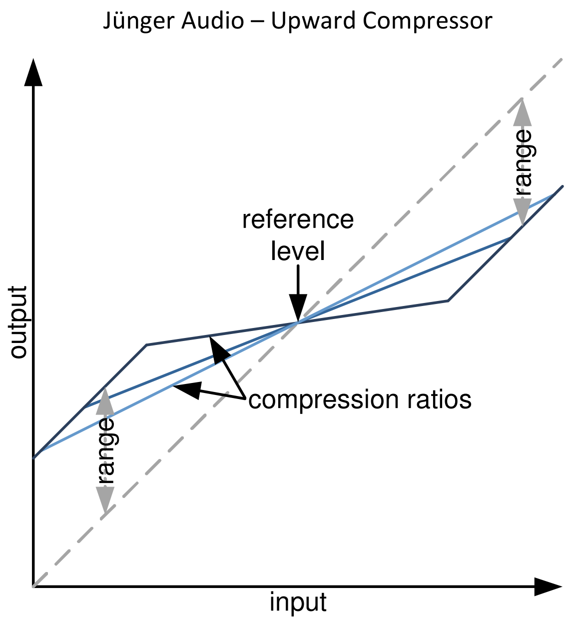
The parameters of the dynamic section are explained below in reference to the curves :



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:

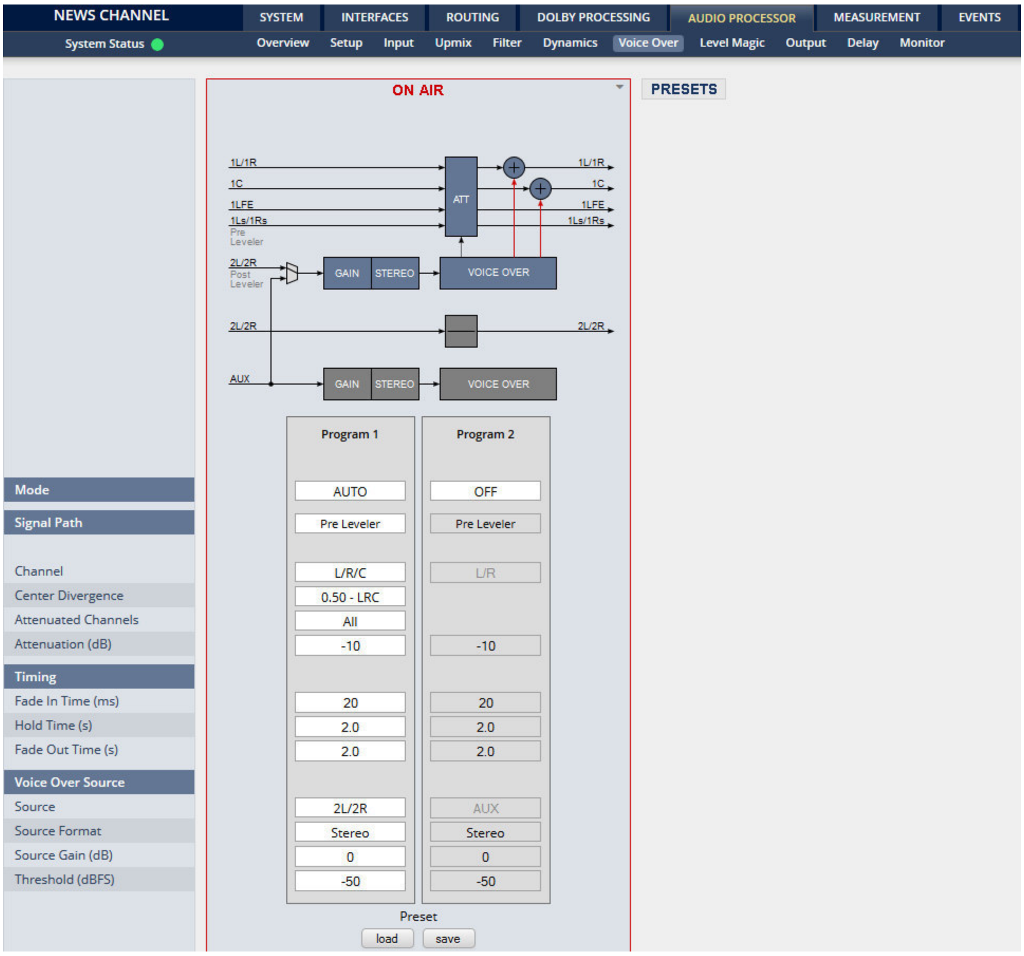
Processing The timing characteristics of the compressor are generated adaptively according to the incoming signal structure. The overall timing can be set up from fast and responsive settings (lower numbers) to relaxed settings (higher numbers) without detailed access to the actual micro timings. The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

Expert [ON / OFF]

Clear Processing History 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 :



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).

Signal Path [Pre Leveler / Post Leveler]
See AUDIO PROCESSOR > Overview for the actual location of the circuit in the signal path.

Channel	[C / L/R / L/R/C] Here you define the channels where the voice signal must be mixed to.
Center Divergence	[0.0 – C only ... 0.5 – LRC ... 1.0 LR only] For the mix you can define the width of the voice signal
Attenuated Channels	[All / Selected] Which channels must be attenuated when voice over is active.
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**

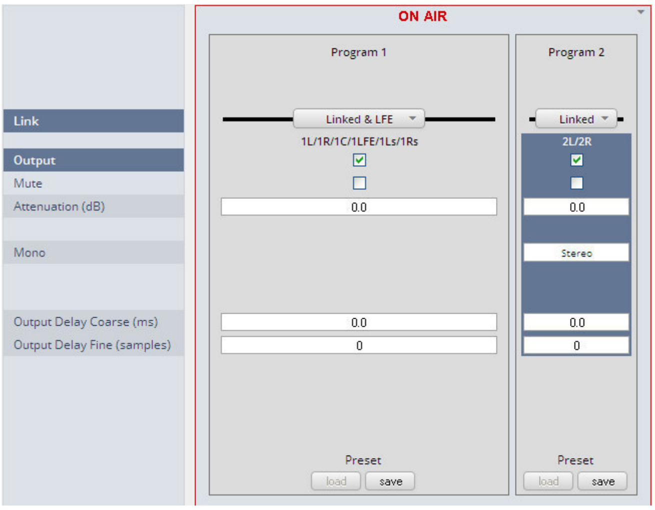
This function block is used for loudness control of the program path.

Loudness 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: "Jünger Processing Parameter Description" on the Junger web site: http://junger-audio.com/en/downloads section White Papaers.

Setup GUI – AUDIO PROCESSOR – **Output**



Link (Program 1) [Quad / Movie / Live / Linked / Linked & LFE]
You may select one of the possible multichannel modes to enable gang setting of the EQ parameter values.

Link (Progarm 2) [Linked / Unlinked]
For stereo operation you may link the setup parameters.

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*AP8** 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

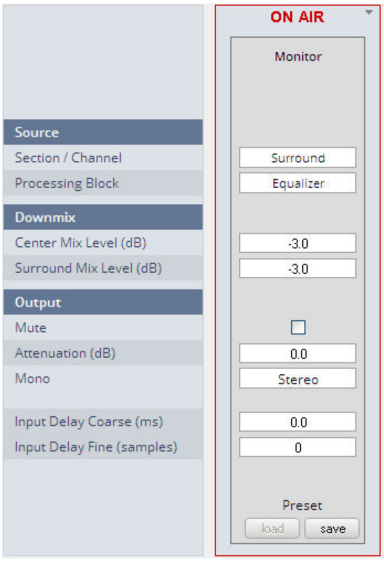
Delay [ON / OFF]

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 – **Monitor**



The monitor output of the DSP can be connected to the outputs of individual function blocks of one of the programs.

Source

Section / Channel	[Surround / 2L / 2R]
Processing Block	[OFF (Mute) / Input / Input Conditioner / Equalizer / Level Magic / Output Conditioner]

Downmix

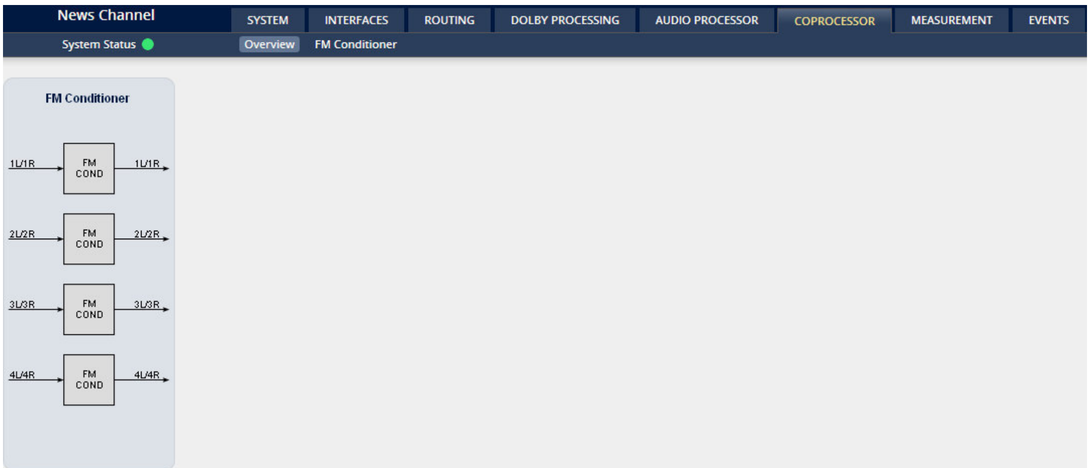
Center Mix Level (dB)	[-12.0 … -3.0 … 0.0]
Surround Mix Level (dB)	[-12.0 … -3.0 … 0.0]

Output

Mute	[OFF / ON]
Attenuation (dB)	[-80.0 … 0.0]
Mono	[Stereo / L+R Mono / L/L Mono / R/R Mono]
Delay Coarse (ms)	[0.0 ... 2000.0]
Delay Fine (samples)	[0 ... 2000]

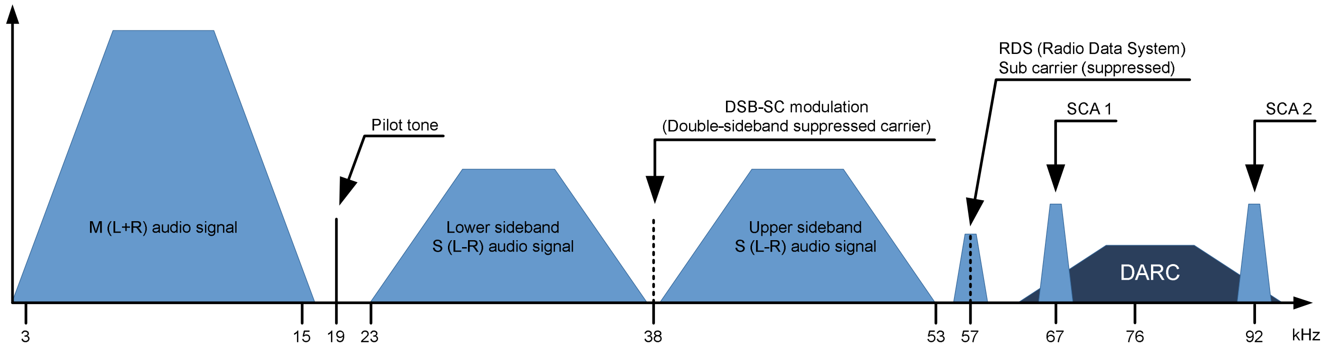
Setup GUI – COPROCESSOR – **Overview**

If you have installed the optional co-processor JDSPA it is possible to extend the functionality of the **D*AP8** by extra FM conditioners. E.g. it allows you to use the **D*AP8** for multiple FM processing channels in one device. In this case an additional tab **COPROCESSOR** appears in the top navigation line:



Above you see four FM conditioner function blocks with their labels as they appear in the routing matrix (1L … 4R) of the **D*AP8** under COPROCESSOR OUTPUT and COPROCESSOR INPUT COPR1 … COPR8.

Setup GUI – COROCESSOR – **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:

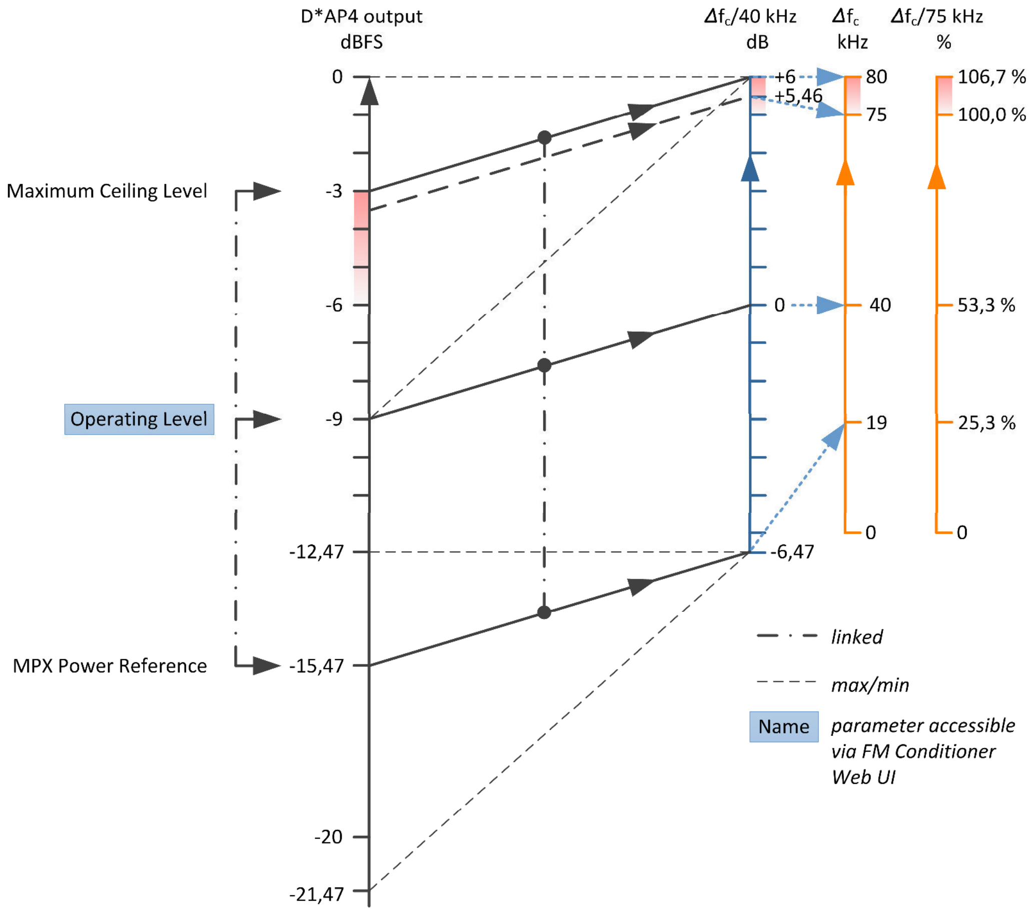
* Maximum peak deviation of $\pm$75kHz
* Maximum MPX power of 0dBr
* A typical audio baseband cut-off at 15kHz to ensure undisturbed transmission of the 19kHz stereo pilot tone
* For mono operation a typical audio baseband bandwidth of 17.5kHz is utilized (no pilot tone necessary)

Calibration

MPX power is measured in a random interval of 60 s. MPX power of 0dBr should be equal to the modulation power of a stationary sine signal that causes a deviation of $\pm$19kHz. 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 ±40kHz. 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 +6dBu (analog) or -9dBFS (digital) for a 500Hz tone is reference level. It may be designated as the operating level and defined at 0dB relative (displayed on a peak level meter). Please be careful with this type of reference level scale, as this analog operating level of 0dBr is not the same as 0dBr 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 -9dBFS and change this value until ±40kHz 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!

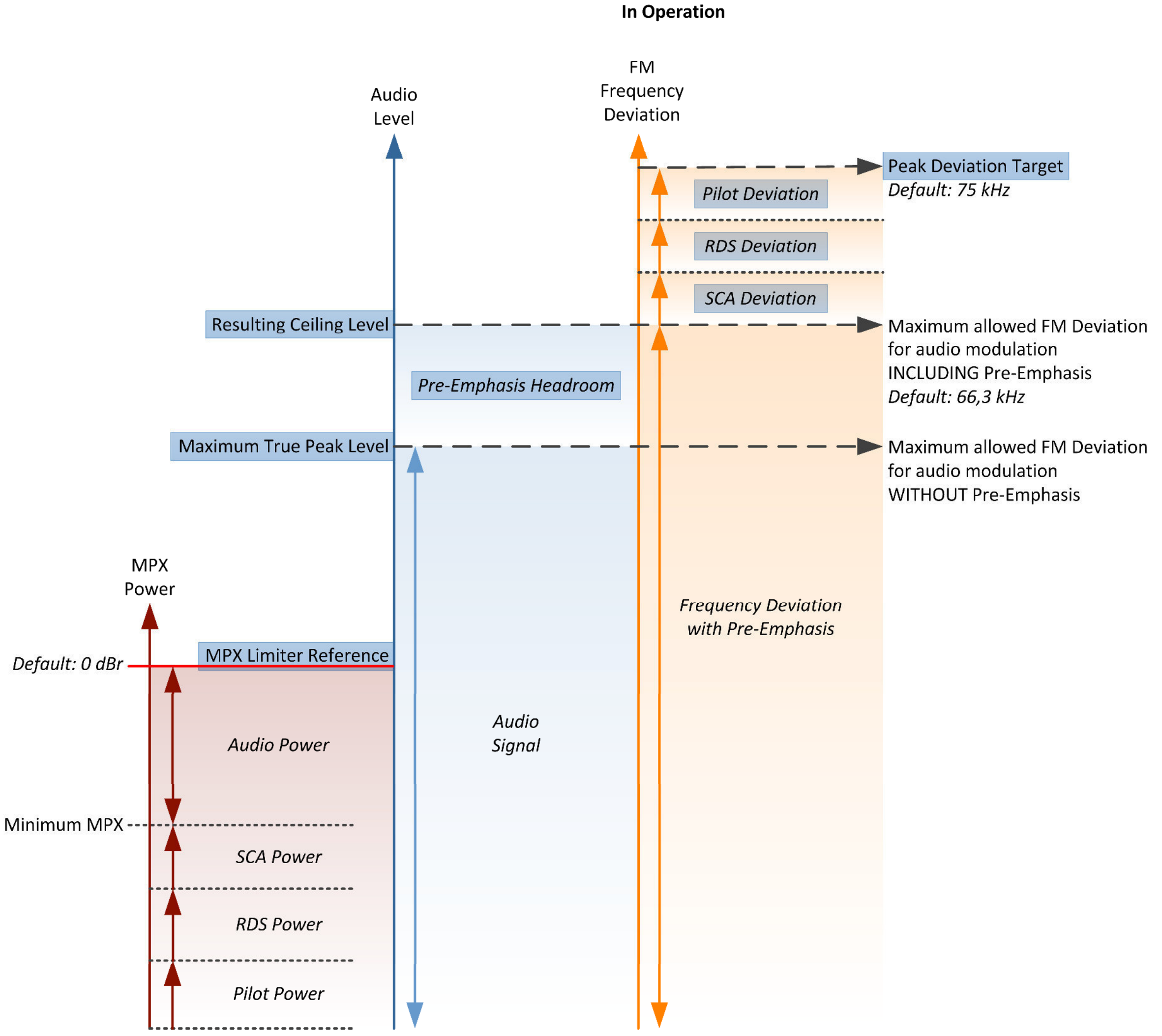
If the Reference Level of your setup differs from -9dBFS the Setup Gain of the FM Conditioner can be used for level matching.

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*log (75kHz – 8.8kHz) / 40kHz = 4,4dB
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 'catched' 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 **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** **True Peak Max** 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 / Program 3 / Program 4]] Selects which of the programs will be displayed on the graph. The available number of programs depends on SYSTEM > Setup > Program Configuration

Graph

Display 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, 2L/2R, 3L/3R and 4L/4R]
MPX Power 60s (dBr)	current MPX power
Pre-Emphasis Limiter (%)	bar graph to show the percentage of high frequency filtering applied
True Peak Limiter Gain Reduction (dB)	bar graph (duplicates the GUI top display)
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 max>** was previously depressed
MPX Power 60s Max (dBr)	maximum MPX power value detected
Gain Reduction Max (dB)	maximum gain reduction applied 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]

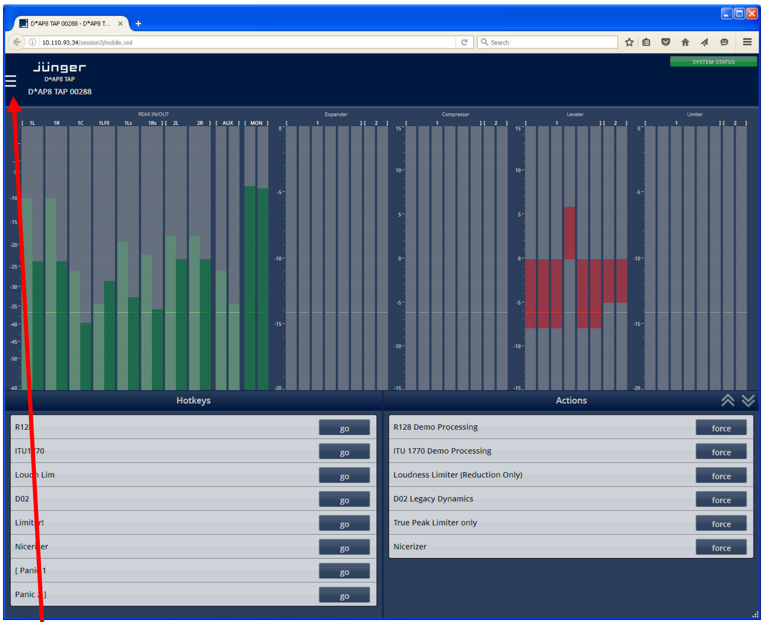
The expert mode allows the administrator to set up the parameters of the transmitter and to turn the **MPX Limiter** on or off.

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 (dBr)	[-4.0 … 0.0 … 4.0]
Low Pass Filter (15kHz)	[OFF / ON]

Setup GUI – AUDIO PROCESSOR – **Mobile UI**

The **D*AP8** 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*AP8** via a WLAN.



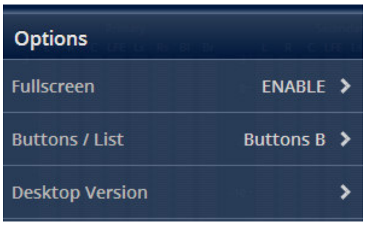
At the bottom left you have a representation of the Hotkey settings (see EVENTS > Triggers > Remote Hotkeys).

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.

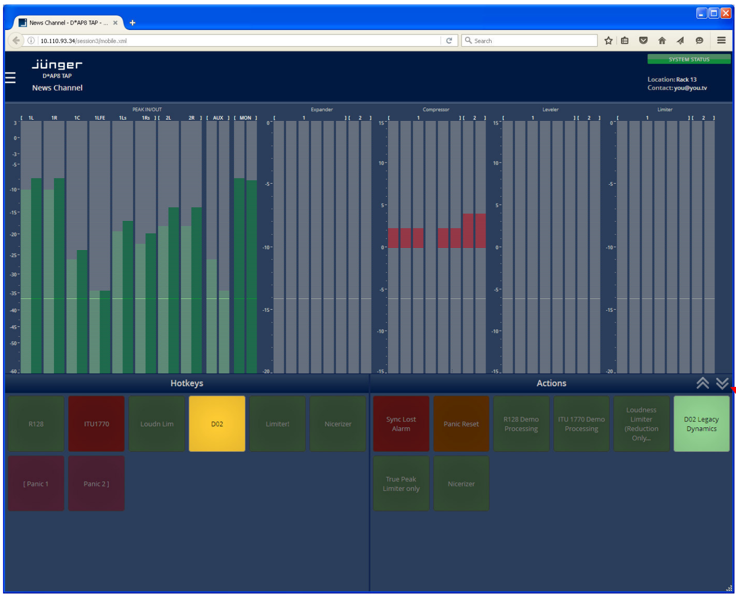
Buttons B shows the greyish rectangle buttons, active ones are highlighted.

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*AP8 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:

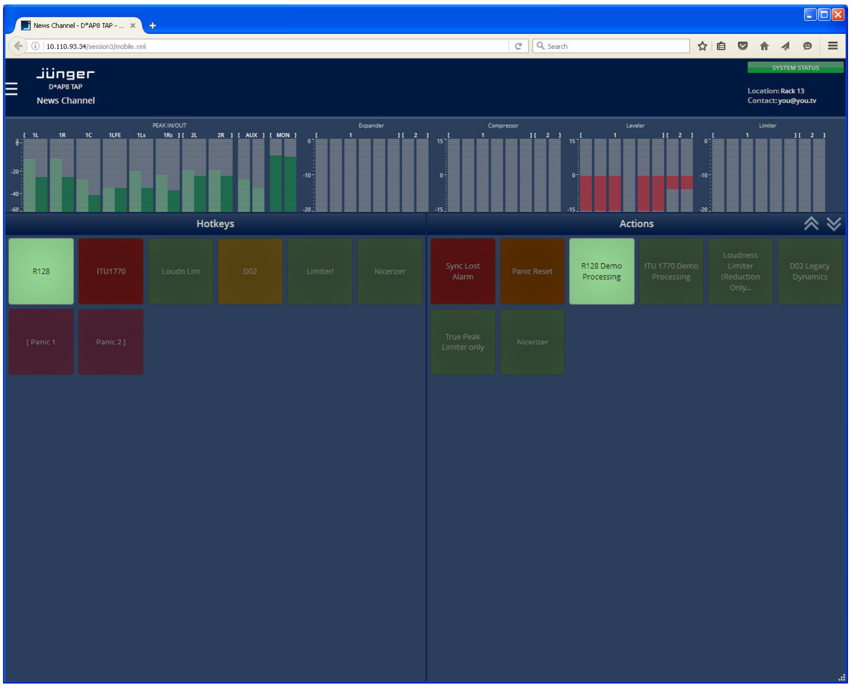


If you press a hotkey the color becomes medium bright immediately.
Since many actions (e.g. reconfiguration of the MAP) may take a few moments the status of a button finally turns bright to acknowledge that the action has been performed. If the color turns back to the dark version the action was not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

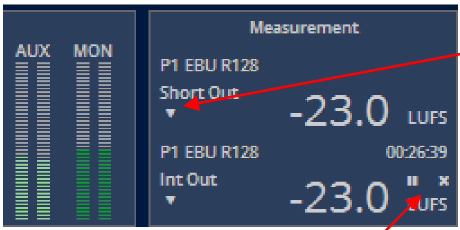
Small > medium size > large > no bar graph display.

Below is an example for the small size version:

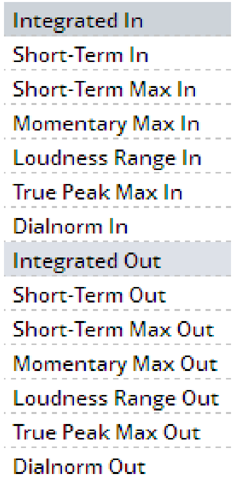


Setup GUI – **MEASUREMENT**

In the top of the GUI you can read these metering data of a preselected source:



When you click on the little triangle over here you will get a selection of the measurement formats available:



This display also shows the duration of the measurement. If the **Speech Gate** is active for the **Dialogue Intelligence™** algorithm, the numbers become yellowish when the measurement has paused because there is no speech detected for the moment.

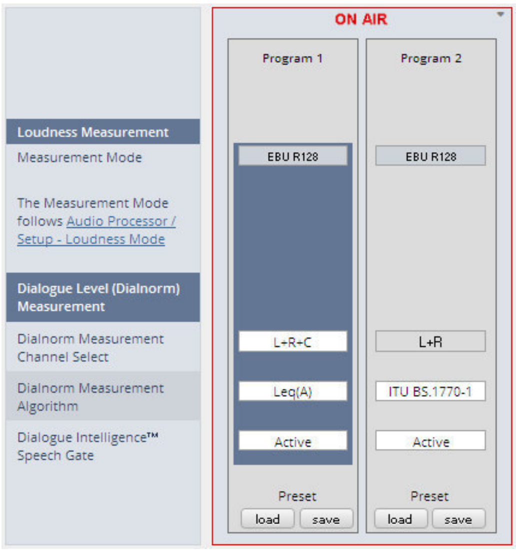
The other two buttons will control the measurement:
|| **<start>** / **<pause>** / **<continue>**
x <reset>.

Setup GUI – MEASUREMENT – **Setup**

Dialog Level (Dialnorm) Measurement:

Beside the ability to measure loudness by above standards, the **D*AP8** offers the feature to measure the long-term A-weighted average level of dialogue within a presentation. A Dolby Digital / Digital plus consumer decoder (e.g. a Set Top Box) will normalize the output level to -31dBFS by applying a shift based on the Dialog Level parameter setting. The rule is: -31 - (dialog level value) = shift applied.

Example (dialog level measured = -23dB): -31 - (-23) = -8dB shift applied.



Loudness Measurement

Measurement Mode	Follows AUDIO PROCESSOR > Setup > Loudness Mode

Dialogue Level (Dialnorm) Measurement

Dialnorm Measurement Channel Select	[L / R / C / L+R / L+R+C]
Dialnorm Measurement Algorithm	[ITU-BS.1770-1 / Leq(A)]

Dialog Intelligence™ Speech Gate	[OFF / Active] The **Dialog Intelligence™** algorithm developed by **Dolby® Inc.** searches for portions of the audio content where speech is present. Such portions may trigger the loudness measurement. If it is activated and no speech is detected, the number display becomes yellowish.

Setup GUI – MEASUREMENT – **Loudness**

The **D*AP8 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.

Loudness Mode — 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)**
- **Dialnorm** — -70.0 indicates that no speech has been detected. If it is activated in the setup but no speech is recognized by the algorithm in this case, the background of the display box turns yellowish.
- **Short-Term Loudness (LUFS)** — numeric and convenient bar graph display
- **Momentary Loudness (LUFS)** — convenient bar graph display
- **Short Term Max (LUFS)**
- **Momentary Max (LUFS)**
- **True Peak Max (dBTP)**

Recent Measurement — same parameters as current measurement

- **Integration time (hh:mm:ss)** — Total time of the recent measurement

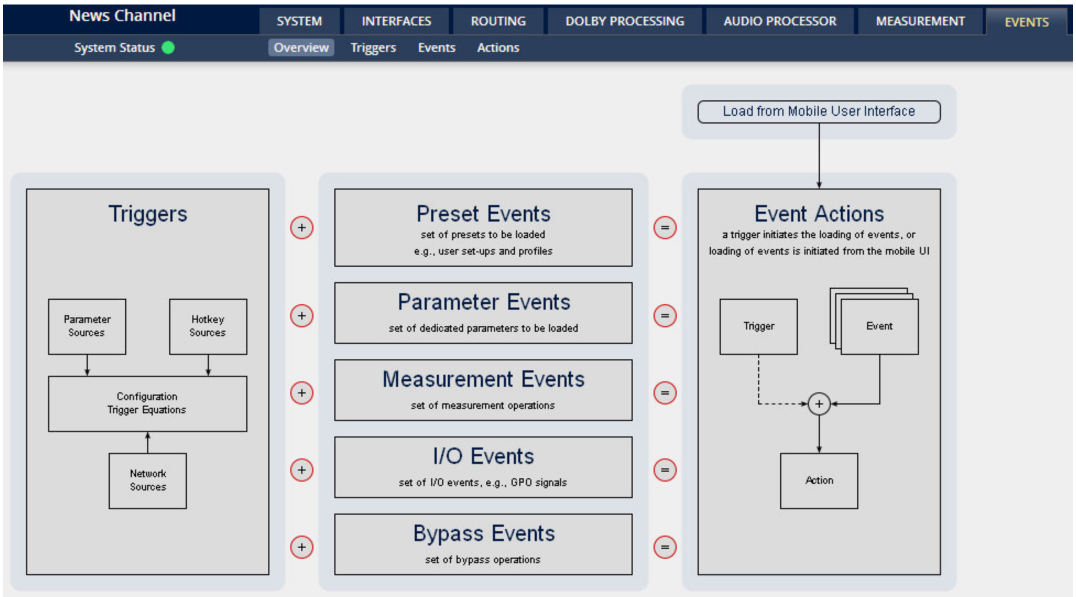
Important Note! The measures of the parameters above depend on the loudness mode selected at AUDIO PROCESSOR > Setup pane.

The measurement data may also be streamed to the PC based **J*AM** (Junger Application Manager). The **J*AM** is a graph display and logging tool 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**

The **D*AP8** offers a sophisticated **event management** system.
The event management system performs A**ctions.** 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 **EVENTS** system of the **D*AP8.** The examples further below are taken from the actual set of factory pre-set events based on a number of useful presets:



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 systems and Interfaces.

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

The **D*AP8** knows five different **event types**:

Preset Events (Profiles)	System / Interfaces / Routing / Dolby Processing / Audio Processor / Programs / AUX / Delay / Monitor / Measurement
Parameter Events	System / Measurement
Measurement Events	Loudness 4 x 2 / Loudness 5.1 + 2
I/O Events	GPOs
Bypass Events	Programs / AUX

The **D*AP8** has two different **action types:**

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

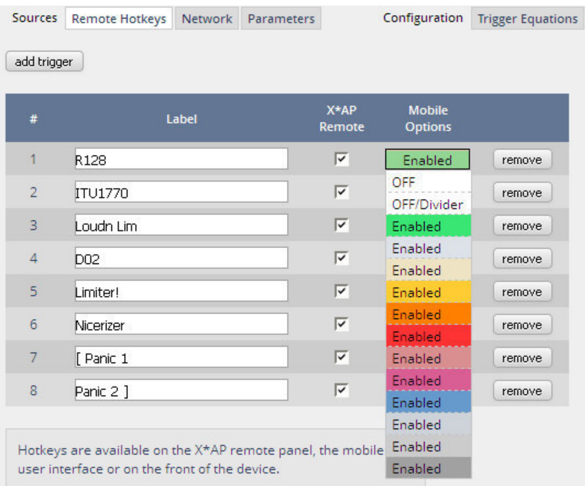
An action runs like a flip-book inside the **D*AP8**. This powerful technology spans from simply recalling a certain system parameter over speaker or Dolby specific parameter combinations (household name: "Preset") to the complete reconfiguration of the **D*AP8** including all signal routing, processing 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 several **fail over** scenarios where the **D*AP8** 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 the pre-defined trigger sources
3. Set up events - by selecting presets for function blocks
4. Create actions - what will happen - which trigger will launch which event? Or what will happen in case someone presses the **<BYPASS>** button at the **X*AP RM1** or engages the **<Force Trigger Active>** check box (see EVENTS > Actions > Bypass Action).

Setup GUI – EVENTS – Triggers – Sources – **Remote Hotkeys**

Hotkeys are the eight buttons of an **X*AP RM1** remote panel. You may give them names and enable them to become active on the **X*AP RM1** remote panels main operating menu:

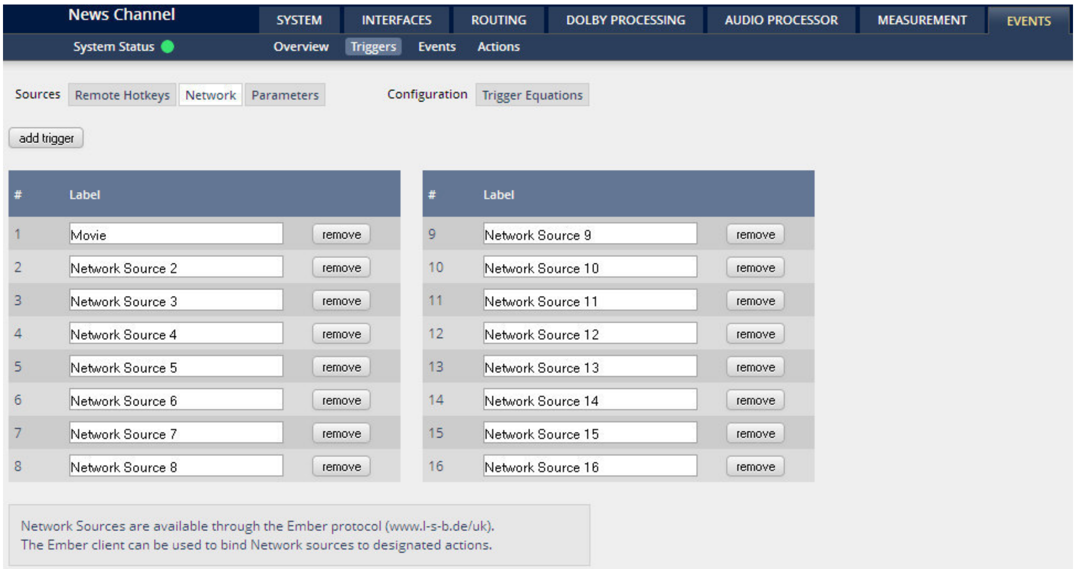


<add trigger>	You can add lines here.
#	The number of the Hotkey on the **X*AP RM1** remote panel, counting from left to right.
Label	Each Hotkey may have a label that appears in the display of the **X*AP RM1** remote panel above that button.
X*AP Remote	[ON / OFF] If you un-check, the respective Hotkey on the **X*AP RM1** remote panel becomes inactive - no label is displayed and the button background light turns off.
Mobile Options	The buttons of the **Mobile UI** can be assigned an individual color. You will see the bright color if the button is active and the dark color if it is inactive. [OFF/Divider] acts as a place holder for a button that is temporarily set to 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 graphical UI for example that may have more than 8 compared to the **X*AP RM**$_1$.

Setup GUI – EVENTS – Triggers – Sources – **Network**

Network triggers are based on the **EmBER+** protocol from Co. l-s-b http://www.l-s-b.de/en.
The **D*AP8** receives such triggers over the TCP/IP network. The triggers are issued by a device that has implemented the **EmBER+** protocol (e.g. VSM server, broadcast automation system). You may assign these triggers to virtual panels as well as physical (e.g. LBP) buttons of a VSM installation. But also a broadcast automation system may have an **EmBER+** server running that will trigger events in the **D*AP8**.



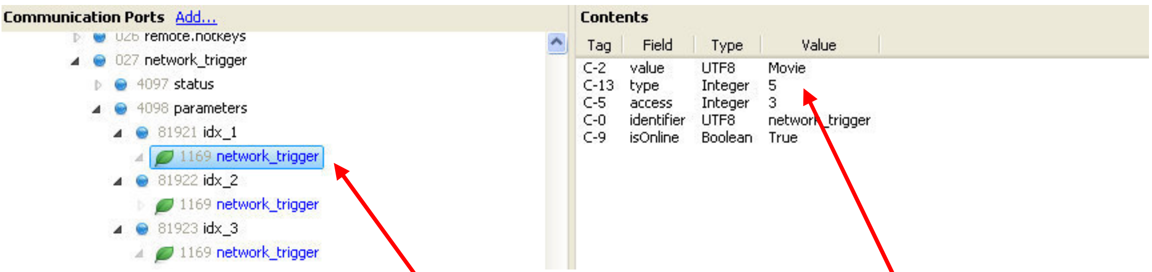
The number of a network trigger.

Label Label of that network trigger. It will be used on the **Configuration** pane and serves as a reference for 3rd party software implementation (e.g. broadcast automation systems). As an example you see the name of the first Trigger **"Movie".**

<remove> will remove a line from the list.

The name of the respective trigger may be selected via an **EmBER+** enabled device to fire that trigger. By means of a setup tool you must configure such network triggers in order to remote control the **D*AP8**. You will find the Ember+ protocol details, the implementation guidelines as well as an example here: code.google.com/p/**ember**-plus/

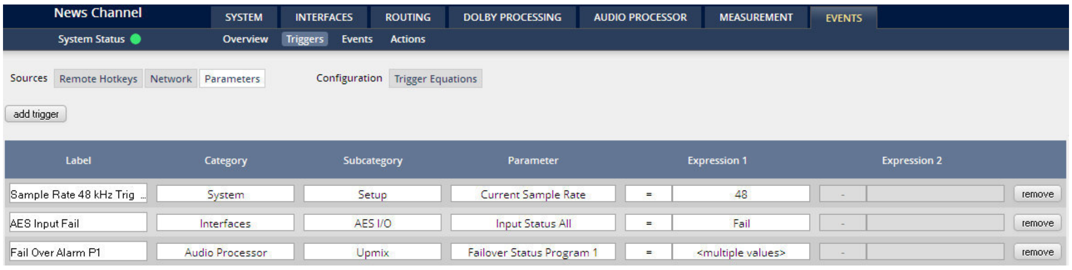
Below is a screen shot of the **EmBER+** viewer tool:



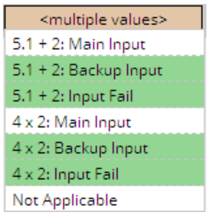
In the EmBER tree you go to:
"Device" > controller_dsp > network_trigger > parameters > e.g. "idx_1"

As a value you will receive the trigger name from the **D*AP8**.
In this example it is the trigger named: **"Movie"** – the 1st trigger from the example above.

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

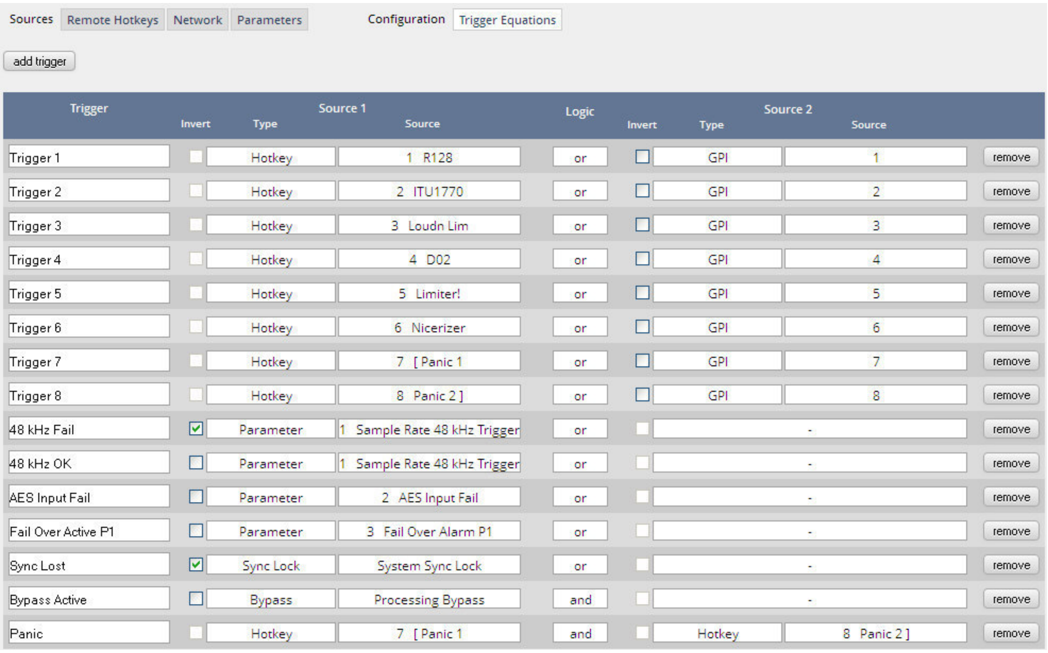


Above is an example of parameter trigger sources. The phrase **<multiple values>** indicates that more than one value of the parameter "Status" is bound to that trigger source:
If you click into the "Expression 1" box you see two greenish marked entries.
I.e. if one of these values is true, "Expression 1" is true.



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

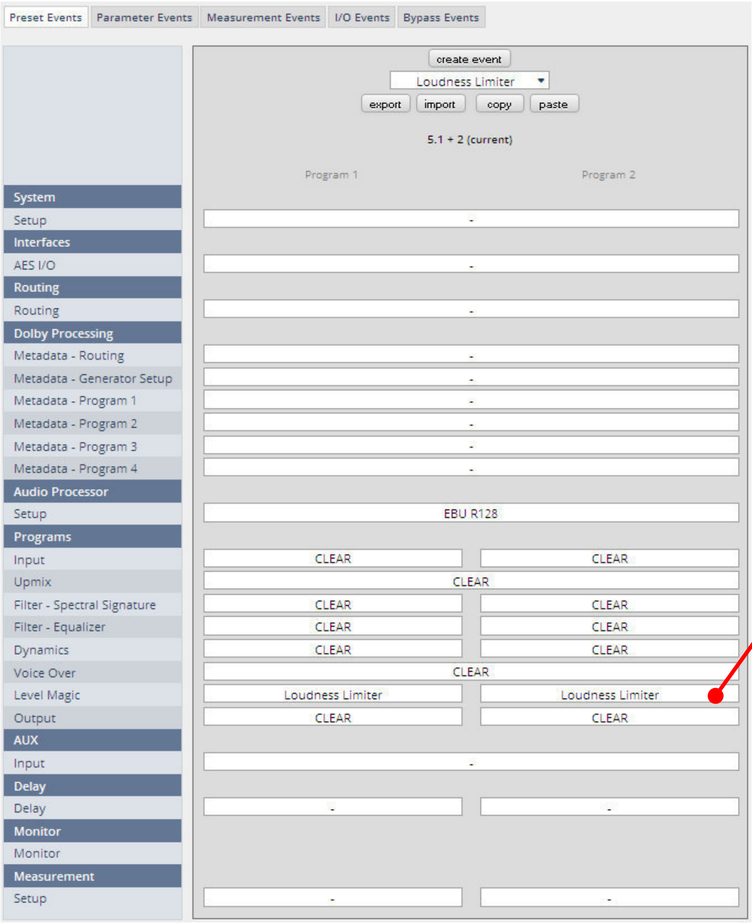
To form a trigger you may logically combine two trigger sources:



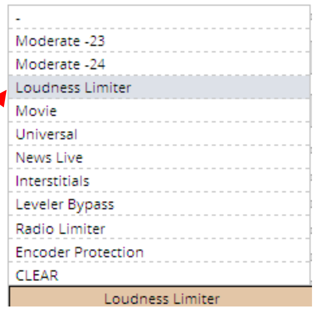
Trigger	Here you define a name for the trigger ("Trigger 1").
Source 1	The first source of a logical combination of two trigger sources.
Invert	[ON / OFF] If the type of trigger allows an inverted operation it can be defined here.
Type	[GPI / Hot Key, Network / Parameter / Event active / Trigger effective/ Bypass / Sync Lock]
Source	[e.g. for GPIs it will be 1 … 8] It acts like an index for the trigger type (In case of GPI it is the physical GPI number or in case of X*AP RM1 Hotkeys it is the key number ….
Logic	[and / or / xor] The kind of logical operation.
Source 2	Second source for the logical combination of two trigger sources. If only one source exists, you may leave it unassigned [-].

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

A **Preset Event** is a group of presets you may load on one occasion to the On Air parameters of function blocks. When executing such an event you may for example change the 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:



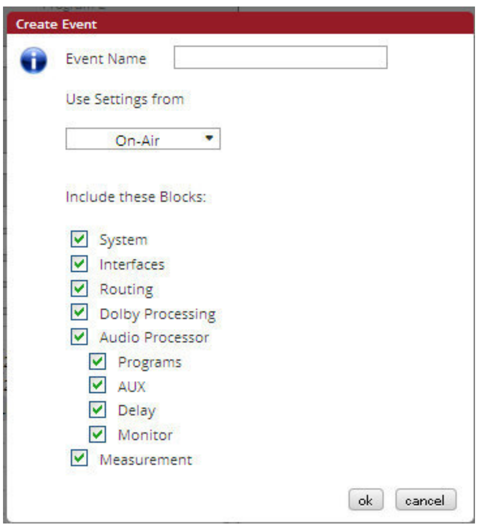
This picture shows an excerpt from the Preset Events pane where a few presets are pre-selected for the event: "Loundess Limiter".
If no preset is selected you have a dash in the drop down field. Some function blocks (e.g. Monitor) even have no preset assigned at all at the moment so there is no drop down box.



Pull down list of all factory default presets of the Level Magic, Loudness Limiter being one of them.

The **Preset Events** allow you to reconfigure the **D*AP8** completely, partially or to change a few audio parameters marginally.

You are also able to create a new preset event semi-automatically by pressing **<create event>**:



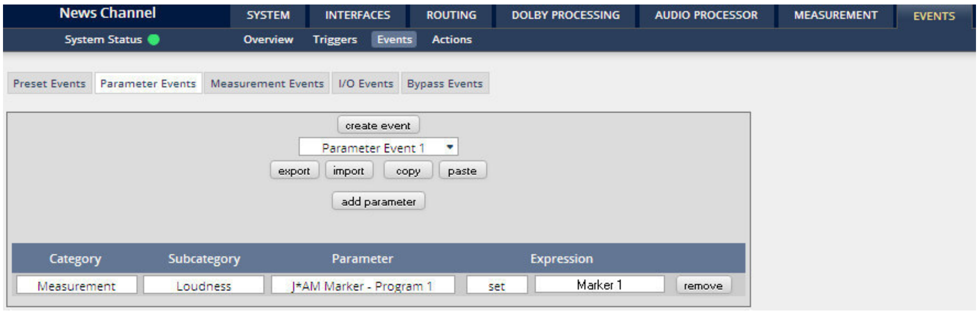
Event name	[New Event] default A unique name to address this preset event later in the action manager.
Use Settings from	[On Air / Existing Event / Empty]
"On Air"	The events manager will copy all On Air parameters to new presets in all function blocks, (that have been selected via the **"Include these Blocks"** check boxes).
"Existing Event"	The presets of the selected event will be copied to the new event and may be tuned afterwards to form a slightly different event.

"Empty"	Creates a set of empty boxes where you may select the preset of your choice for the respective function block or leave it empty if no changes are needed …
Include these Blocks:	[System / Interface / Routing / Dolby Processing /Audio Processor **>**Programs / >AUX / >Delay / >Monitor / Measurement]. You can tell the event manager which function blocks must be included in this event (or not).

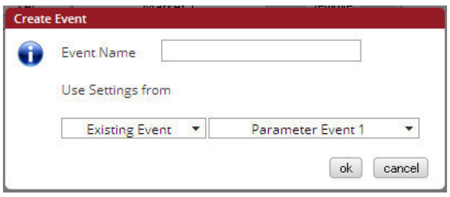
Important Note! This is the way to create your own **snap shot**. The new presets will be automatically given the name of this event! So be careful to select meaningful names. You will find them later on in your function blocks!

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

Right now the **D*AP8** supports parameter events to remote control the measurement / logging related features of the **J*AM**:



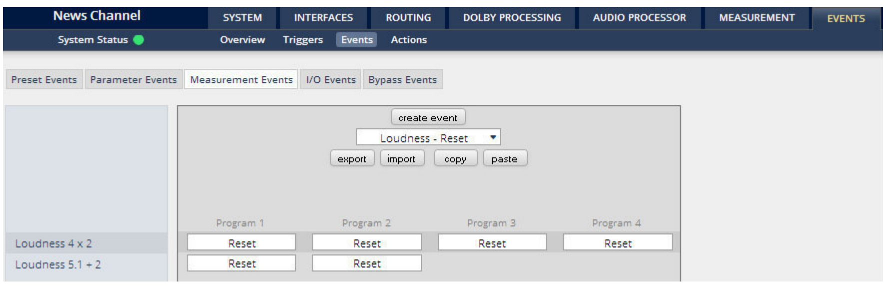
The above example selects the category **"Measurement"** with its sub category **"Loudness"**. From the list of possible parameters, the setting of a marker **"J*AM Marker – Program 1"** has been selected. This marker will appear in the log file if that preset event is executed. When you press **<create event>** these choices are provided:



Event Name	your choice
Use settings from	[Existing Event / Defaults / Empty]

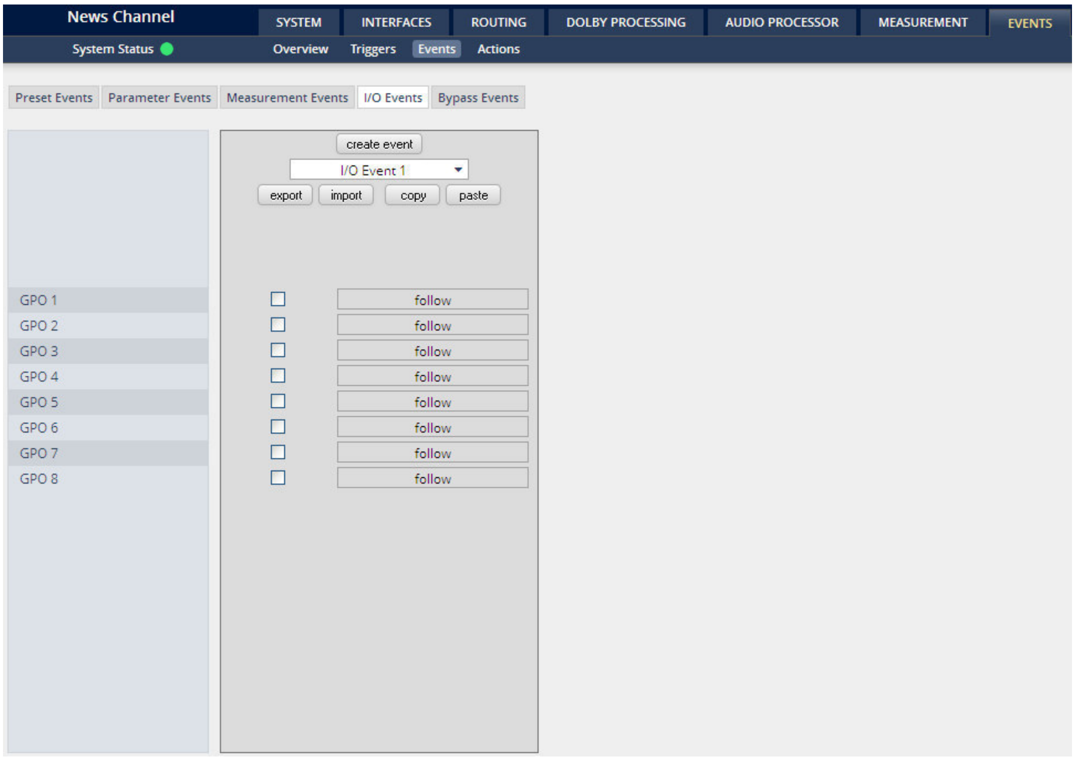
Setup GUI – EVENTS – Events – **Measurement Events**

A measurement event is used to control the **D*AP8** internal loudness meter. (See MEASUREMENT > Loudness). For the example below **"Reset"** has been pre-selected for all possible program configurations and their respective programs:



Setup GUI – EVENTS – Events – **I/O Events**

I/O Events at the moment control the **GPO**s of the **D*AP8**:

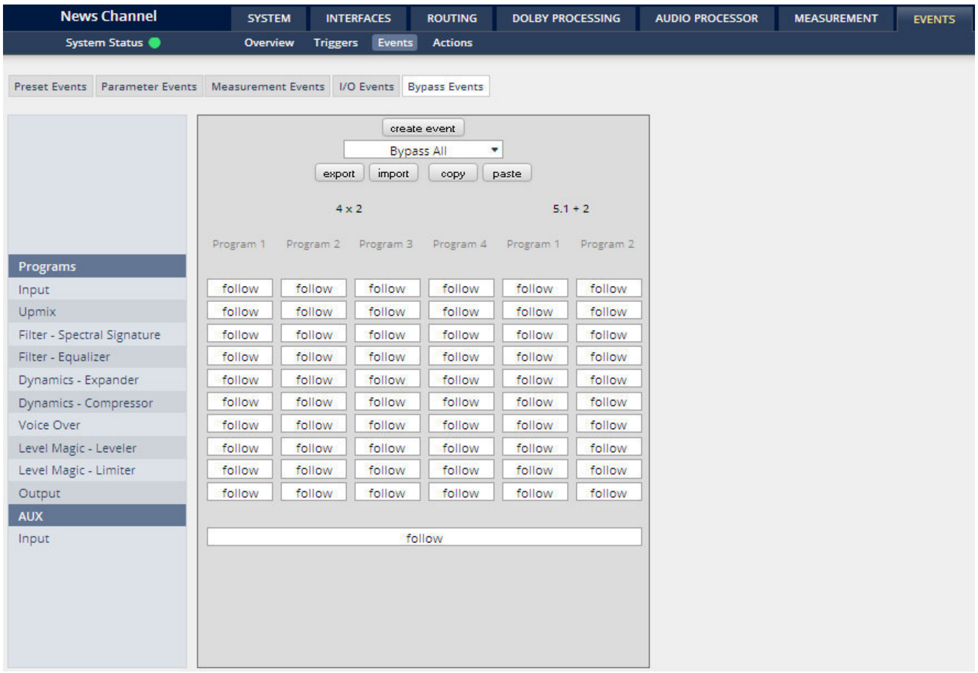


Each GPO (when enabled for that I/O event) can be set to one of these actions:

Clear	Turns a GPO off that was previously turned on.
Set	Turns a GPO on.
Follow	The GPO follows the state of the trigger.
Toggle	The trigger will toggle that GPO.

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

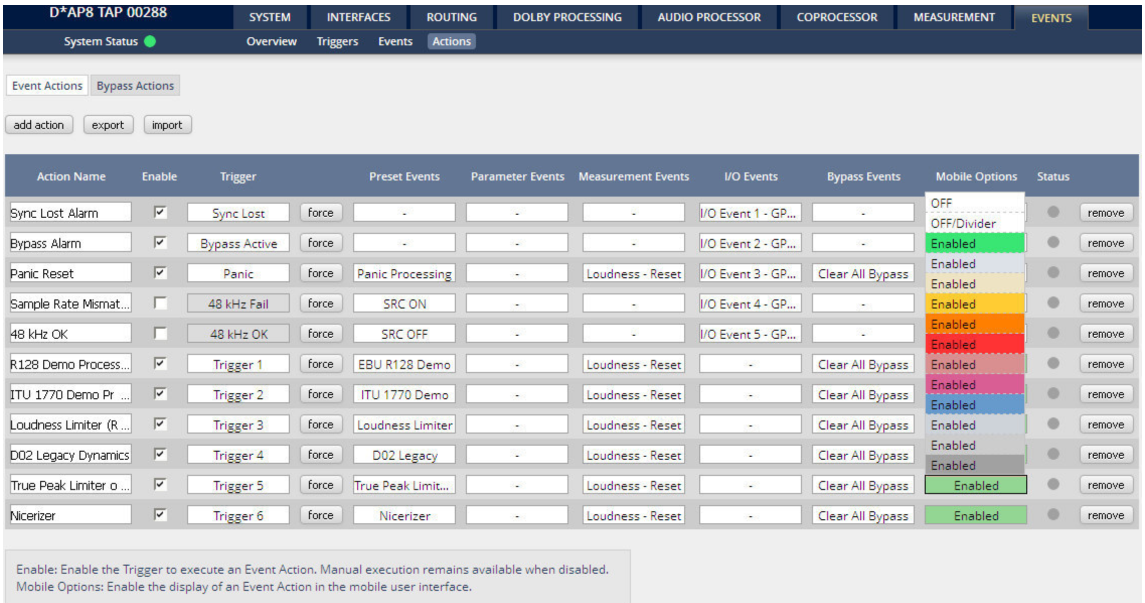
Here you can configure complex scenarios to bypass function blocks of the programs:



The options for controlling the bypass are: [clear / set / follow / toggle]

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

This is the point where all previously created sub-functions are combined:



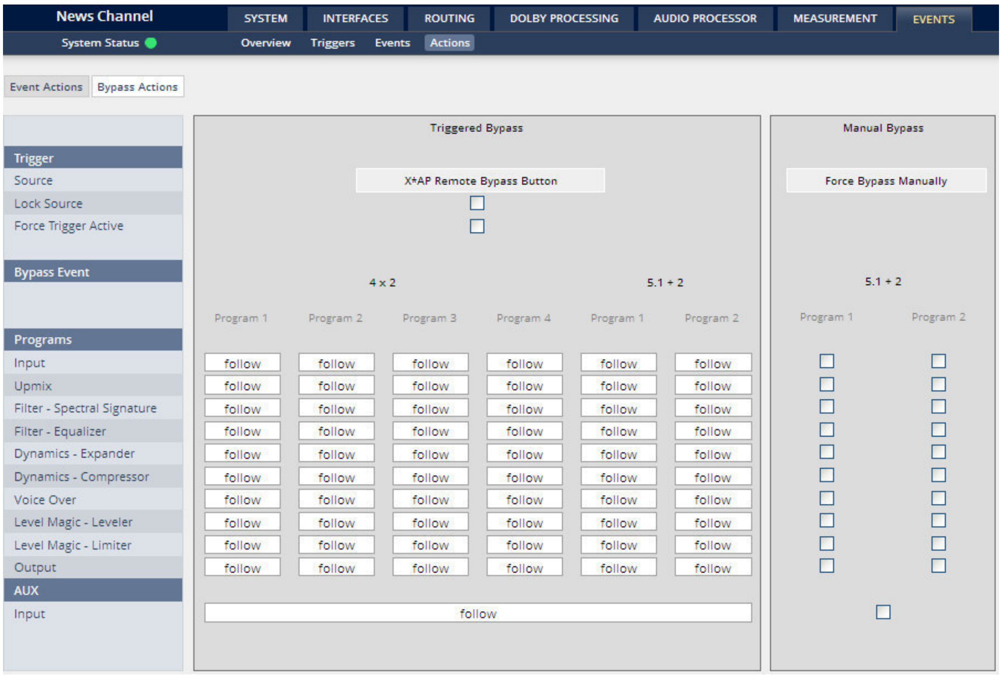
Here you create the action!

You should give the action a meaningful name, select a trigger (from one of the trigger equations) and select the respective event(s) you need to perform the desired action. In the "**Mobile Options"** column you can define whether the action may be turned on via a soft button of the **Mobile UI**. You can assign a specific color for that button. [OFF/Divider] acts as a place holder for a button that is temporarily set to OFF.

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 "-" in the setup field.

But it also allows you to turn the bypass on for, some or all function blocks by simply enabling the check boxes in the right hand panel:



Triggered Bypass

Trigger

Source — The **X*AP RM1 <BYPASS>** button is the trigger source

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 — [4 x 2 / 5.1 + 2]

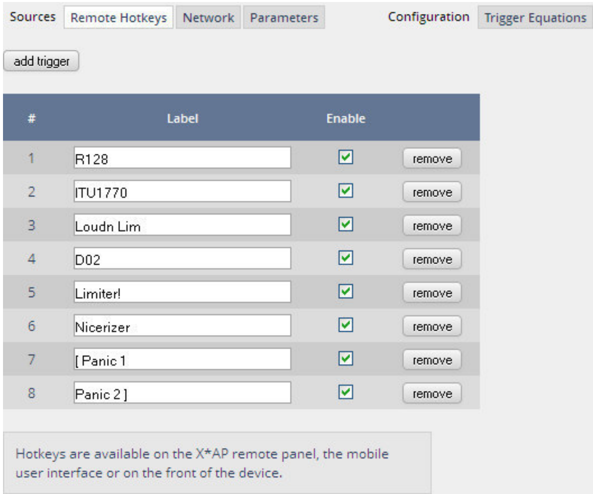
Programs — [Input / Upmix / Filter – Spectral Signature / Filter – Equalizer / Dynamics – Compressor / Voice Over / Level Magic – Leveler / Level Magic –Limiter / Output]

AUX — [Input]

Manual Bypass — You can use the check boxes for manually bypassing the respective function block of that program.

Setup GUI – EVENTS – Actions – Event Actions – **Factory Defaults**

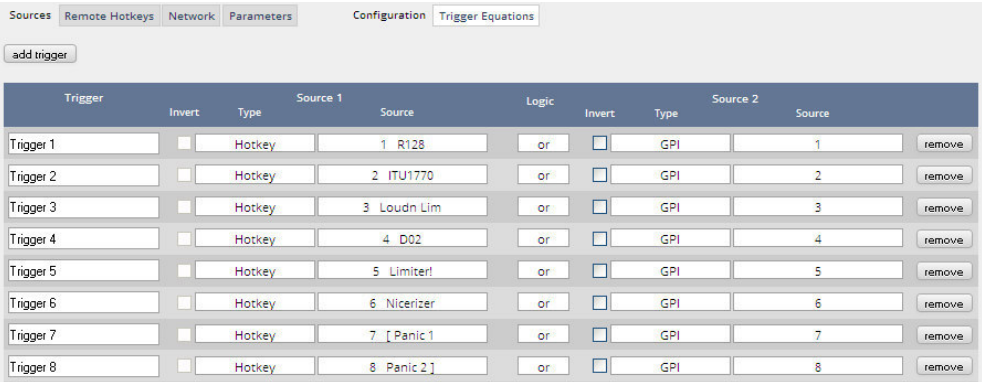
The **D*AP8** comes with a set of factory default presets and event settings which you can use as they are or modify them to your needs. The examples below explain the relationship of the **EVENTS** system.



Above you see the factory default **EVENTS > Event Actions** that come with the **D*AP8**.

They are prepared to ease the operation of some functionalities of the device. Eight of these actions may be triggered manually from the **X*AP RM1** remote panel.
See the **"Remote Hotkeys"** settings.

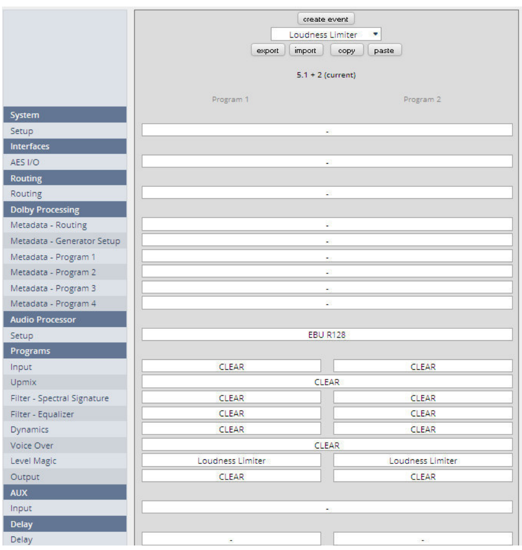
The remote hotkeys are used by the following **"Trigger "Equations"**:



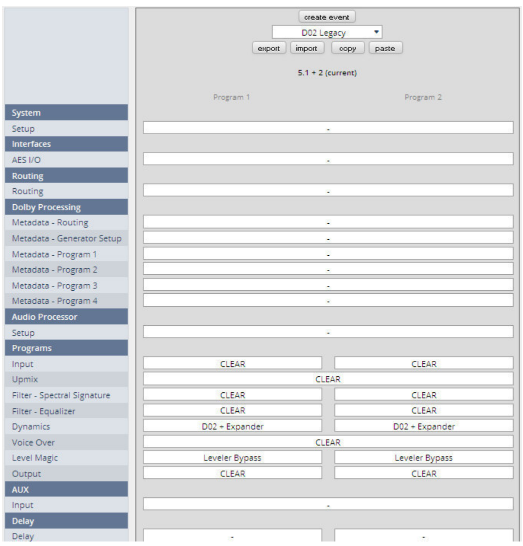
I.e. the trigger named **"Trigger 1"** will be fired if one depresses the **hotkey # 1** that is named **"R128"**. That name appears above the first hotkey of the **X*AP RM1.**

On the page **EVENTS > Events > Preset Events** you see the combinations of individual presets for each of the factory default **"Preset Events"**. Below are two examples of such preset events:

"Loudness Limiter"



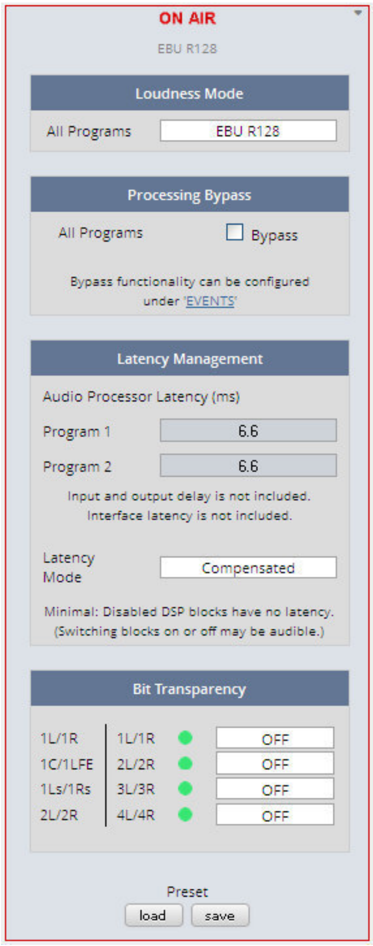
"D02 Legacy"



In the examples above you can see the differences of the respective Setup / Level Magic / Dynamics and Clear preset.

Below are two of the five parameter sets that will be loaded by their presets. The respective preset name is displayed in grayish above the active mode display:

AUDIO PROCESSOR > Setup: "EBU R128"



AUDIO PROCESSOR > Dynamics: "D02 Original"

Technical Data - 8 Channel Surround Audio Processor
[D*AP8 TAP EDITION, D*AP8 CODEC EDITION, D*AP8 FLX]

General	• 8 channel audio processor (4 stereo programs or 1 surround and 1 stereo program) • 2 channel (1 stereo) auxiliary input • 2 channel (1 stereo) monitor output • Expandable by hard and software options	
Audio Sample Rate	44.1, 48kHz, (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	
	8 channels (4 stereo inputs), 4 BNC connectors	
	24bits, transparent forwarding of PCM and compressed audio (w/o SRC) 24bits, PCM, sample rate converter (SRC) activated	
	Impedance	75Ohm single-ended
	Input level	0.3 … 5Vpp @ 75Ohm single-ended
	Sample Rate Converter (SRC)	THD+N -120dB @ 0 BFS, 1kHz Latency < 0.3ms
AES/EBU Outputs	Relevant specifications comply with AES3-X-2009, IEC 60985 and AES11-2009	
	8 channels (4 stereo outputs), 4 BNC connectors	
	24bits, transparent forwarding of PCM and compressed audio	
	Impedance	75Ohm single-ended
	Output voltage	1Vpp (typ.) @ 75Ohm single-ended
	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
Metadata Input	Relevant specifications comply with SMPTE RDD6-2008 (Dolby Metadata).	
	Connector type	D-Sub9 connector female

	Input conditions	110Ohm RS485, 0.2 … 5Vpp differential
Metadata Output	Relevant specifications comply with SMPTE RDD6-2008 (Dolby Metadata).	
	Connector type	D-Sub9 connector female, same conn. as input, D-Sub9 connector male, output only Both connectors carry the same signal.
	Output conditions	3Vpp (typ.) @ 110Ohm differential, RS485
Timecode Input	LTC timecode input, BNC, currently not supported (TBD)	
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@5V nom.
	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 Slots	2 general purpose expansion slots for option boards, 2 internal expansion slots for Dolby encoding, decoding and emulation	
Power Supply	Dual power supply, automatic fail over, 85 … 264Vac, 50 … 60Hz, 58W (max.)	
Environmental	Operating temperature 0 … 50ºC, fan cooled (dual fan), Non-operating -20 … 70ºC, Humidity < 90%, non-condensing	
Physical	19", 1RU, 27cm depth, net weight ca. 5kg, shipping weight ca. 7.5kg	

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	2970/296Mbps (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 … 320ms 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)
Video Latency	Input to Output	120 … 200pixel, depends on video standard
Audio Latency	Input to Output	Embedder and de-embedder combined HD, 3G < 0.6ms SD typ. 1.5ms (< 2ms)
General Features	• 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 8 Ch 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	• Power fail glitch prevention • Balanced analog outputs • Electrical isolation between outputs and device	

Technical Data – Option Board 4 Ch Analog I/O **[O_DAP_ADDA_a]**

Audio	24bit sigma-delta A/D-converter, 24 bit 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	• 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 < 0.3ms
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	• 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	• 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	• 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	• AES67 compliant (when available) • 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**

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:	**Metadata IN**
female	9-pin D-Sub
1	GND
2	Tx (-)
3	Rx (+)
4	GND
5	
6	GND
7	Tx (+)
8	Rx (-)
9	GND

connector:	**Metadata OUT**
male	9-pin D-Sub
1	GND
2	
3	Tx (+)
4	GND
5	
6	GND
7	
8	Tx (-)
9	GND

Technical Data - Optional Interface Modules – **pin assignment**

4x analog I/O [O_DAP_ADDA_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

4x AES I/O [O_DAP_AES_a]

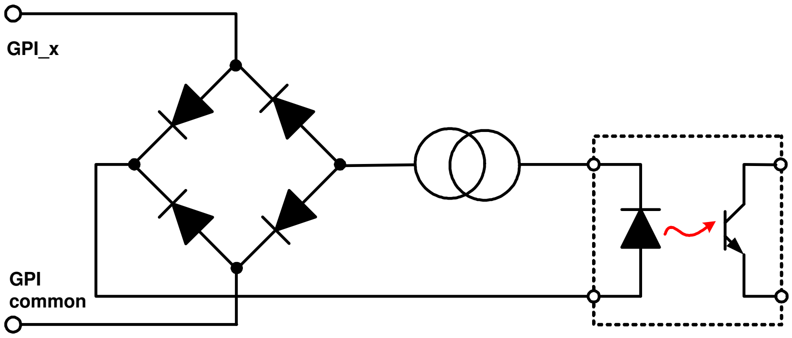
connector:	4x AES I/O
female	25-pin D-Sub
1	OUT-4 +
2	GND
3	OUT-3 -
4	OUT-2 +
5	GND
6	OUT-1 -
7	IN-4 +
8	GND
9	IN-3 -
10	IN-2 +
11	GND
12	IN-1 -
13	
14	OUT-4 -
15	OUT-3 +
16	GND
17	OUT-2 -
18	OUT-1 +
19	GND
20	IN-4 -
21	IN-3 +
22	GND
23	IN-2 -
24	IN-1 +
25	GND

8x analog out [O_DAP_8DA_a]

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 **eight GPI** inputs:

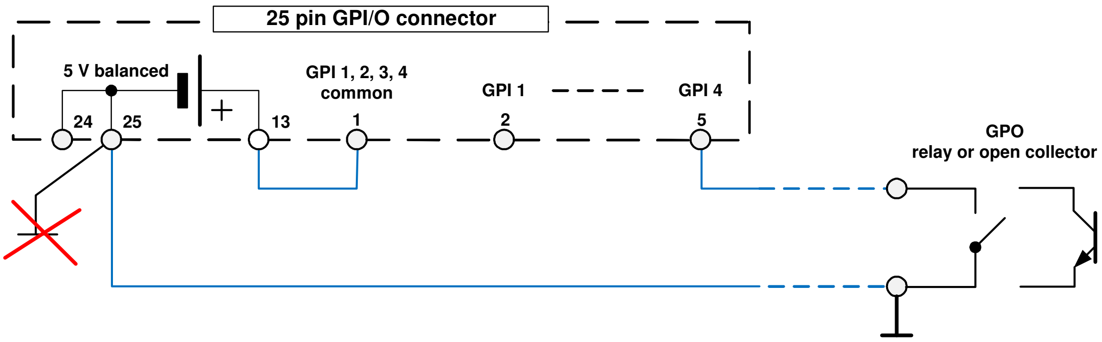


At the GPI input is a **bridge rectifier** I.e. you do **not** need to care about the polarity of the input voltage. A **constant current source** in line with the **optical coupler** limits the current.
You must simply provide a voltage in the range from 5V to 30V to activate a **GPI**.

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 5V source that you can imaging as a battery.

Here an example how to wire up GPI #4:



We strongly recommend to spent 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

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