# D*AP4 VAP Edition

## Voice Audio Processor

**Manual**


Hardware Features

- **1RU** compact 19" processing device with front side info display
- **Dual power supply** second power supply for redundancy
- **Front panel info display** for signal activity, IP address, status alert
- **Remote Panel** optional X*AP $RM_1$ panel
- **Optional mic inputs** optional dual high end mic preamp module with phantom power
- **Optional AES42 input** optional module for digital mic / line input
- **Balanced AES I/O** AES line input / output for desk inserts or program input
- **One interface slot** I/O expansion slot for one option board
- **3G / HD / SD SDI module** option board with SDI de-embedder / embedder and relay bypass
- **4x AES I/O module** option board with 4x AES3id I/Os and relay bypass
- **4Ch analog I/O module** option board with 4 analog line I/Os and relay bypass
- **RJ45 network connector** 100BaseT full duplex Ethernet interface
- **USB B connector** built in USB < > serial adapter to access the device service port
- **8 GPI/Os** 8 balanced inputs, 8 relay closure on 25pin Sub-D
- **Aux power supply** isolated 5V supply for external wiring
- **External sync IN** 75Ohm input (Word Clock, AES, Black Burst, Tri-Level)
- **Sync OUT** 75Ohm Word Clock output

Software Features

- **2 main processing channels** chain of processing blocks, mono / stereo operation
- **Program Path** 2Ch input for a program signal including EQ, Dynamics, LevelMagic
- **Input stage** mute, gain, polarity, HPF, LPF
- **M/S matrix** encode, stereo width, decode
- **Phase Rotator** corrects imbalanced waveforms
- **De-esser** frequency, range, type, Q
- **Filter** spectral signature & 5x full parametric EQs
- **Dynamics** expander, upward compressor, downward compressor, soft limiter
- **Leveler** automatic level control for the voice channel
- **Voice over** stereo or mono voice over extra program input, pan
- **LevelMagic™** processor for the program (AUX) path
- **Output stage** true peak limiter, mute, attenuation
- **Monitor outputs** extra feeds from the DSP to monitor DSP processing blocks
- **SNMP agent** SNMP v1, see D*AP4 VAP-MIB
- **Remote control** EmBER plus protocol (e.g. for VSM integration), 3rd party API
- **Mobile user interface** graphical operator UI optimized for use on mobile devices

# Content

Content

# Introduction

Primarily designed to apply individual processing to voice based applications the 2 channel **D*AP4 VAP** toolset includes HP/LP filtering, dynamic section, full parametric EQ, and de-essing. A dedicated voice leveler aids the integration of voice programs into loudness based broadcasting. Targeted at radio stations and TV production voice-over applications, this unit will make your daily life easier and let you focus on content.

With Spectral Signature™ dynamic EQ, you will have a tool with automatic and dynamic EQ control to balance spectral differences to one specific voice only when necessary. **Spectral Signature™** analyzes incoming audio and compares the spectral structure with individual predetermined voice “footprints”. On this basis, dynamic EQ corrections will be applied only when necessary to achieve consistent results. Spectral Signature™ is a **D*AP4 VAP** standard feature while a separate program I/O path allows for voice over either in manual controlled or automatic (ducking) mode. The program path has a separate LevelMagic processor for conditioning the program output after voice over.

The **D*AP4 VAP** offers interfaces to allow integration in existing environments via AES insert or with an optional analog board adding 2 high quality mic-preamps. Preset management can be controlled via network integration. The **X*AP RM1** provides the ability to control up to 4 units via hardware simultaneously, while **D*AP4 VAP**’s network interoperability is designed to allow full integration into broadcast scheduling and studio management systems. With this feature, preset changes will take place automatically according to your content schedule.

At the heart of the **D*AP4 VAP** is a sophisticated audio processor, powered by Analog Devices® Sharc DSPs. These DSPs provide signal processing, audio delays, monitoring facility as well as level measurements.

The AES I/Os on the motherboard may be added to by a variety of interface modules that can be installed as an option into the **D*AP4 VAP**’s interface slot.

A comprehensive routing matrix allows for almost every combination of audio signal flow from inputs to outputs.

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 in use. 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 operator and automatically executed by the event manager of the device.

The **D*AP4 VAP** provides a web based setup GUI and an **X*AP RM1** remote panel that displays status and metering information and allows user intervention. Due to the complexity of the device, the features of the **X*AP RM1** remote panel are limited to operating needs.

Junger Audio’s application manager **J*AM** is also available as an add on and can be attached by a few simple clicks to the **D*AP4 VAP** so that users can display level bar graphs.

The availability of an SNMP agent, which provides traps and status polling rounds up the feature set of the **D*AP4 VAP**.

As with most advanced tools, the **D*AP4 VAP** can be driven in a variety of ways, depending on requirements and ideas of the user. These can range from the simple and straightforward through to quite complex set ups.
Although this manual explains the functions and general operation of the **D*AP4 VAP**, it does not give detailed scenarios because the operational needs of today's productions vary so widely between organizations and their work flows and cover so many different parameters – from simple editorial work places, to complex database driven shift control for multiples of work places, through to semi-automatic operation controlled by broadcast automation systems.

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.

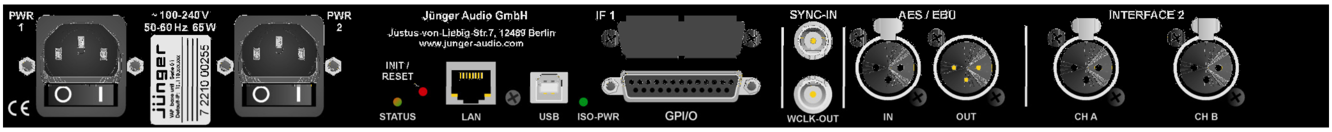
**D*AP4 VAP** front panel view


The front panel of the **D*AP4 VAP** 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** cycles through the multi level display:

| | |
|---|---|
| **Level 1** | Power up display [device type / firmware version] |
| **Level 2** | Status [OK / Error] / Device Name / IP address |
| **Level 3** | IN / OUT peak meter |
| **Level 4** | Monitor M1 / M2 peak meter |
| **Level 5** | Program Out short term loudness |
| **Level 6** | Program Out integrated loudness and integration time |
| **Display background color** | Green = device status OK<br>Red = device status ERROR<br>flashing red / green during boot up |

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

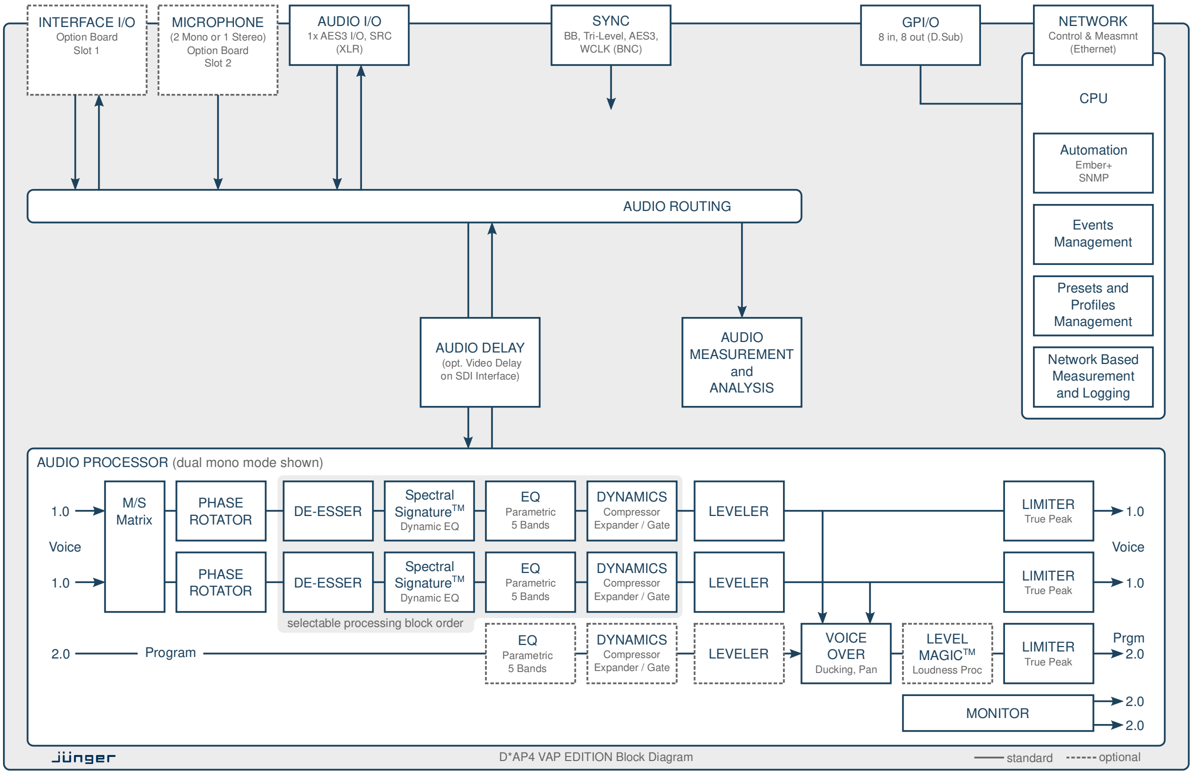
**D*AP4 VAP** rear view




For fail safe operation, the **D*AP4 VAP** 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 backlight color of the front panel display.

| | |
|---|---|
| **STATUS** | shows the status of the device controller |
| **INIT / RESET** | pressing the INIT button briefly will warm start the device controller.<br>Holding down the button until the **STATUS** LED flashes 5 times will initialize the **D*AP4 VAP** to factory default |
| **LAN** | RJ45 socket for Ethernet connection to a LAN |
| **USB** | USB 2.0 type B socket to connect the built in **USB >> serial** converter with an external PC |
| **ISO-PWR** | lights up if the isolated 5V power supply for GPI /O application is turned on |
| **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) |
| **SYNC IN** | 75Ohm BNC connector to connect with external sync sources |
| **WCLK-OUT** | 75Ohm BNC connector to synchronize external devices to the **D*AP4 VAP** internal word clock |
| **AES / EBU IN** | AES3 input |
| **AES / EBU OUT** | AES3 output |
| **Interface 2** | slot to mount the optional dual high end microphone pre amp module<br>or the optional dual AES42 module for digital microphones |

Block Diagram




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

The core of the unit is the audio processor with 4 inputs, 4 outputs and a 2ch monitor output.

An **AES I/O** on the motherboard is provided for digital line operation. The respective connectors have relay bypass for power fail operation. The bypass ciruit may be disabled by an internal jumper.

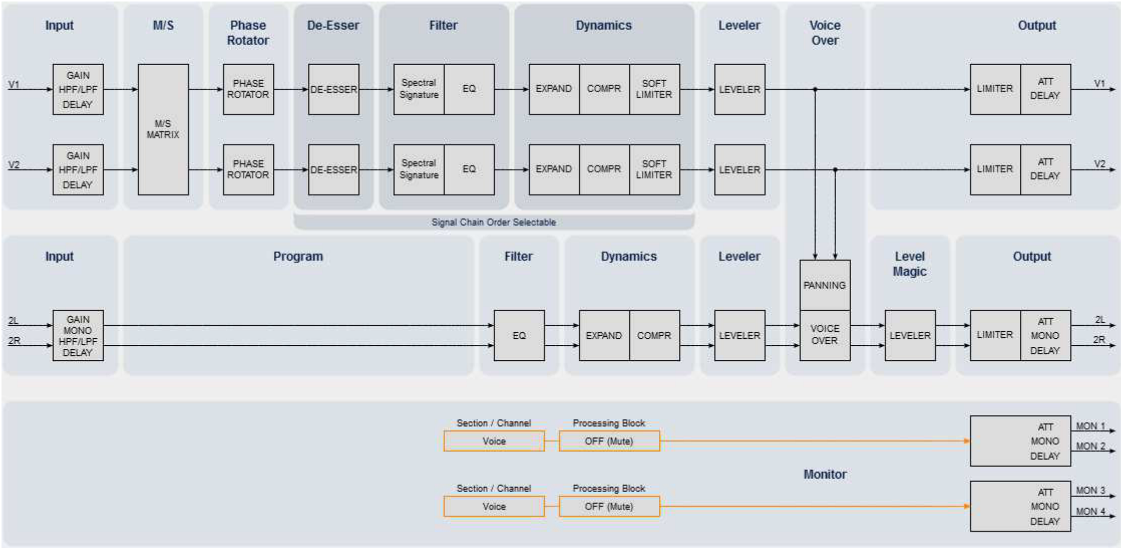
An interface slot is provided to carry optional 3G / HD / SD-SDI, AES I/O / MADI / DANTE or even analog expansion modules. It allows for extremely flexible interfacing of the **D*AP4 VAP**, especially for video based voice over applications. The above schematic shows a MADI interface installed.

On the rear righthand side is the location for an optional high end dual microphone pre amp with phantom power or an alternative optional dual AES42 input module.

The sync. circuit can deal with all formats to integrate the **D*AP4 VAP** into digital facilities with a sample rate from 44.1 to 96kHz. Other devices may be synchronized by the word clock output of the **D*AP4 VAP**.

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

## Audio Processing Blocks




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

It is important to understand that the physical input interfaces of the device must be routed to the **DSP** inputs (V1, V2, 2L, 2R) in order to process it. Similarly the **DSP** outputs (V1, V2, 2L, 2R, MON 1 … MON4) must be routed to output interfaces. You will find those settings by clicking on the **ROUTING** tab.

## Control Concept

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

The setup GUI utilizes web technology. At the time of editing this manual the functionality of the web GUI is optimized for actual Firefox.

The setup GUI can be complemented by other application programs running on MS Windows® W7/8.1/10 like the Junger Audio 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 incorporated into a monitoring system.

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

## Operating Concept

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

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

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

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

## Event Concept

The **D*AP4 VAP** incorporates  a sophisticated event management system.
Events may be combined to perform actions. The **D*AP4 VAP** offers  these event types:

* **Preset Events** for System set up, Interfaces, Routing, Audio Processing etc.
* **Parameter Events** to control specific parameters of the **VAP**
* **Measurement Events** to control the loudness measurement
* **I/O Events** for GPOs
* **Bypass Events** for pre-configured bypass scenarios

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 – quick start guide

Before the **D*AP4 VAP** can be used, there are some basic configuration steps which must be followed in the order set out below. This example assumes you will process one physical condenser microphone and feed the signal to your digital mixing desk that runs at 96kHz sampling rate.

* Connect the mic to the XLR CHA input of Interface 2
* Connect the XLR AES/EBU OUT connector to your digital desk
* Connect the BNC SYNC IN to the Word Clock output of your desk
* Hook up the device to your PC network
  Consult your IT administrator for assistance if you are not sure about this procedure
  - Connect it to a switch or hub or directly to a PC / LapTop by an Ethernet cable
    (some PCs need a cross over cable when connected 1:1)
  - Find an unused IP address - ask your administrator!
  - Assign it that IP address and set the network mask accordingly, a gateway is optional
    (see next page for details)
* Open a browser (Latest stable FireFox recommended) and connect with the device
  - Type in the IP address as an URL
* Set the **sync source**
  - SYSTEM > Setup > Sync Source Priority > **Choice 1=Sync-In WCLK**
    leave all other **Choices x=OFF** (for the beginning)
  - SYSTEM > Setup > System Clock > Sample Rate (kHz) = **Follow Source**
* Define the program configuration
  - SYSTEM > Setup > **Voice Channel Mode=2 x Mono**
* Setup the microphone input
  - INTERFACES > Analog Mic > M1 > **Input=Mic, Enable Preamp Gain=On (check box), Preamp Gain=40dB, Pad=OFF, Phantom Power=On (check box)**
* Set the routing to the Audio Processor (DSP)
  - ROUTING > MIC > **MIC 1=DSP 1**
* Set the routing from the Audio Processor (DSP)
  - ROUTING > DSP > **DSP 1=AES 1**

Now you should have the mic signal on your desk and you may start experimenting with the various parameters of the audio processing blocks.

## Getting Started – IP setup in general

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

1. Ask the system administrator for two unique IP addresses of the local area network, for the netmask used and if a gateway address is necessary.
2. Assign the **D*AP4 VAP** an IP address

   You have 2 choices to assign the **D*AP4 VAP** an **IP address**:
   * From the serial console interface
   * Via Web browser

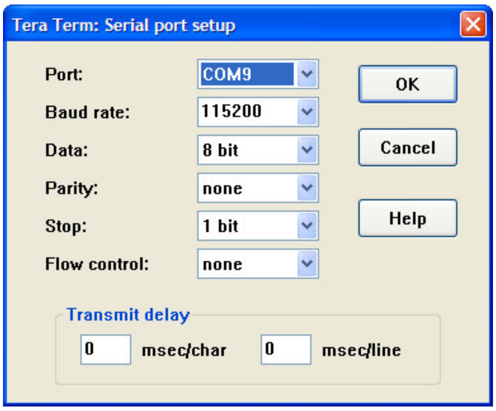
3. Assign the **X*AP RM1** remote panel a unique IP address configuration
4. Attach the **D*AP4 unit** to the **X*AP RM1** remote panel

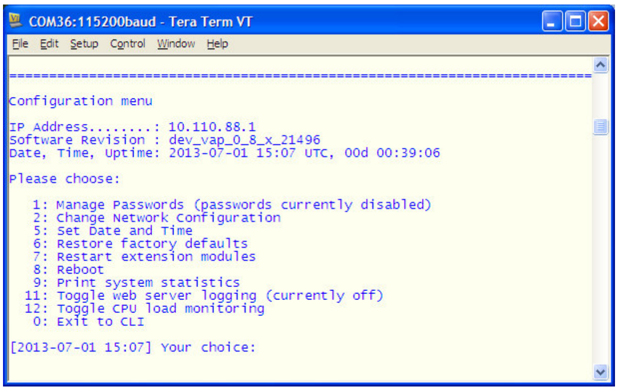
**Important Note!** If you are not familiar with setting up devices for IP communication, we highly recommend you consult your system service or IT department to assist you.

## Getting Started – IP setup – **via console interface**

The tool to change the IP configuration of the **D*AP4 unit** can be reached 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:




Go for item **2:**

**"Your choice: 2"** **<ENTER>**
**"Current network configuration**

**IP Address: 10.110.88.1**
**Netmask ...: 255.255.0.0**
**Gateway ...: 10.110.0.1**

You must enter the IP address and the netmask. Here is an an example:

**Enter new IP address, press ENTER to cancel: "192.168.176.78"** **<Enter>**
**Enter new netmask, press ENTER to cancel: "255.255.255.0"** **<Enter>**

**Important Note!** The gateway entry is optional but you must ensure that the gateway address matches the network mask related to the device IP address!
If you are not sure simply leave it at **0.0.0.0**.

**Enter new gateway, press ENTER to configure without gateway: <Enter>**
**Changing Network configuration**
**The network configuration has been changed. Please reboot the device to activate the new settings.**

Select item 8:

**Do you want to reboot the device ? <ENTER>**

Press small **"y":**

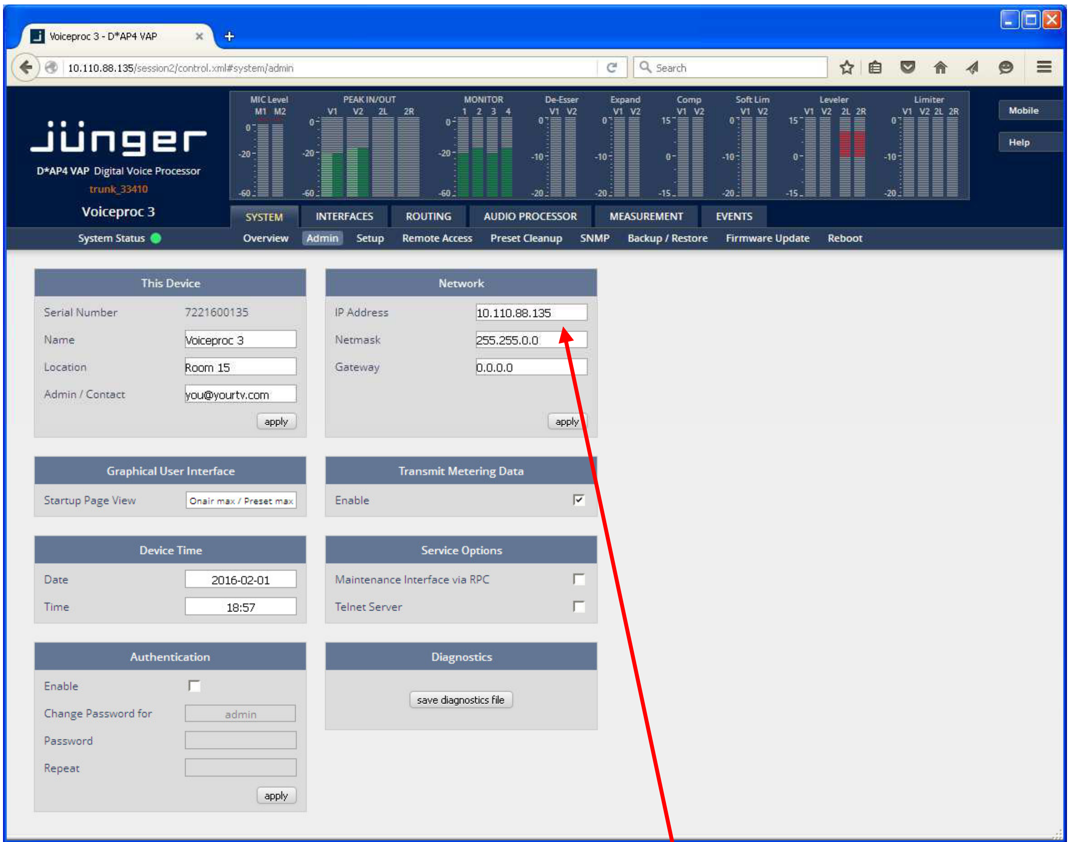
**Do you want to reboot the device ? y <ENTER>**

**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 – **via web browser**

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



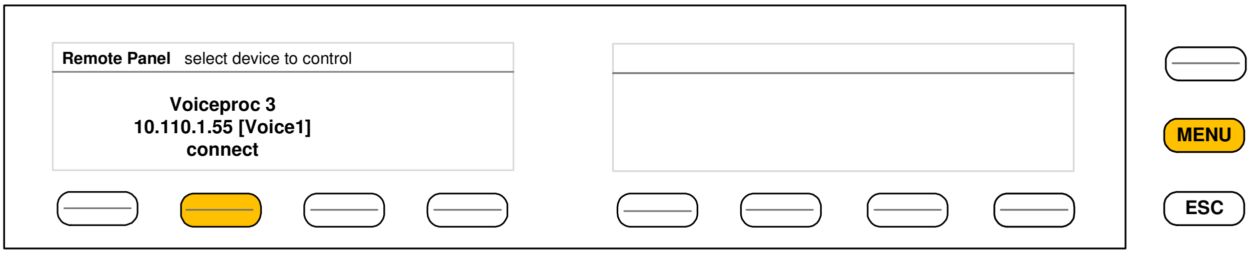

Enter the desired network configuration and press **<apply>**

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

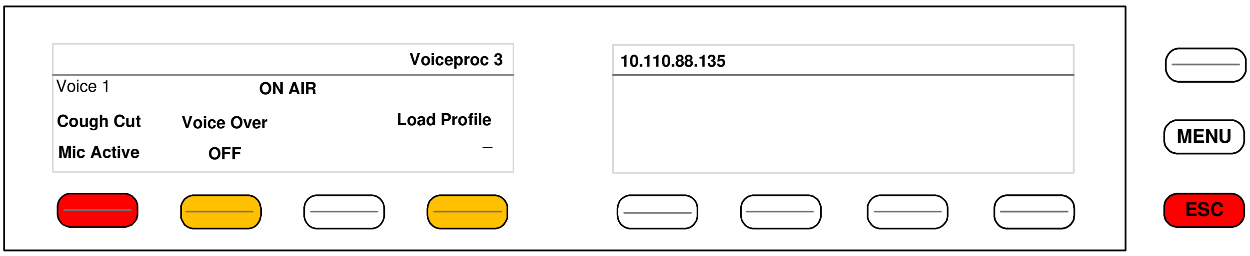
**Important Note!** After reboot neither the **web browser** nor the **X*AP RM1** remote panel may be able to communicate with the **D*AP4 device**. You must fill in the new IP address in the URL field and change the **X*AP RM1** remote panel settings to attach this device with its new IP address.

Operating - menu structure of the **X*AP RM1** remote panel – **power up display**

**Power up display** – may show up to four **D*APx** which are enabled for remote control via this **X*AP RM1** remote panel**.** The example below has just one **D*AP4 VAP** unit [given name "Voiceproc 3"] attached for remote control. The status is **"connect"** (i.e. you may connect with that device).
See **X*AP RM1** manual for details.

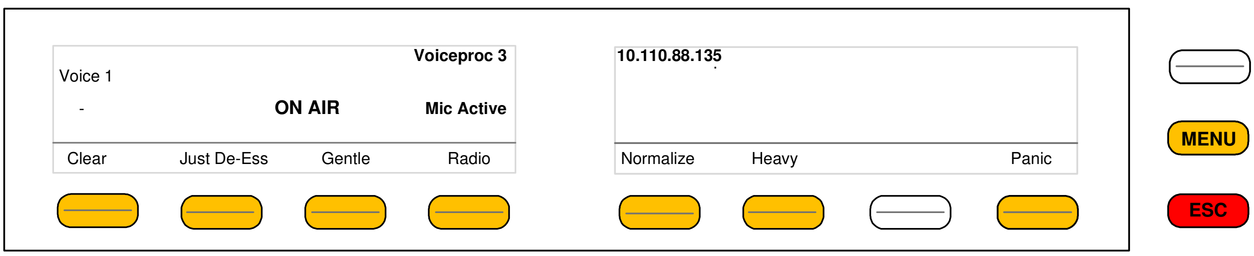



Pressing that button will connect with the **D*AP4 VAP.**
Now the **X*AP RM1** remote panel will gather all necessary information from that **D*AP4 VAP** unit (this may take a few seconds). When finished the **main operating display** opens up.
The appearance of that display depends on the setting found in:
**SYSTEM > Remote Access > X*AP Remote > X*AP Remote Feature Set**. If it is set to "Load Profiles [Voice x]" the capabilities are limited to load such profiles, to control the cough cut and to activate the voice over function manually (AUDIO PROCESSOR > Voice Over > Mode = "Manual"):




From here you may fire pre-defined hotkeys and observe the status of both voice channels.
"Voice 1" is the default name of the first voice channel.
If **SYSTEM > Setup > Voice Channel Mode = Stereo**, settings will be made in reference to the first voice channel. Because this is the main operating display, the **<ESC>** button lights **red** to indicate that the **power up display** is below the **main operating display**.
Pressing **<ESC>** sends you back to the **power up display** (device selection).

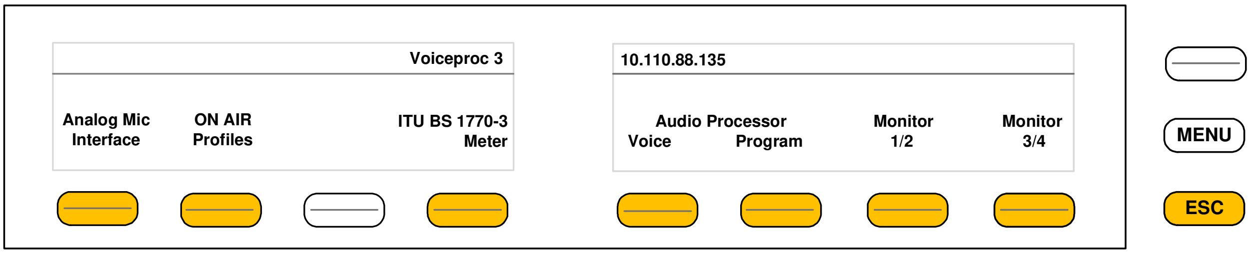
If the X*AP Remote Feature Set "Standard Set" is selected, this **main operating display** will be shown (example for Voice Channel Mode = Stereo):




Now you may fire the pre-set actions (see EVENTS > Actions > Event Actions) via the hotkeys.
You may configure these buttons via: EVENTS > Triggers > Remote Hotkeys **and** > Trigger Equations.

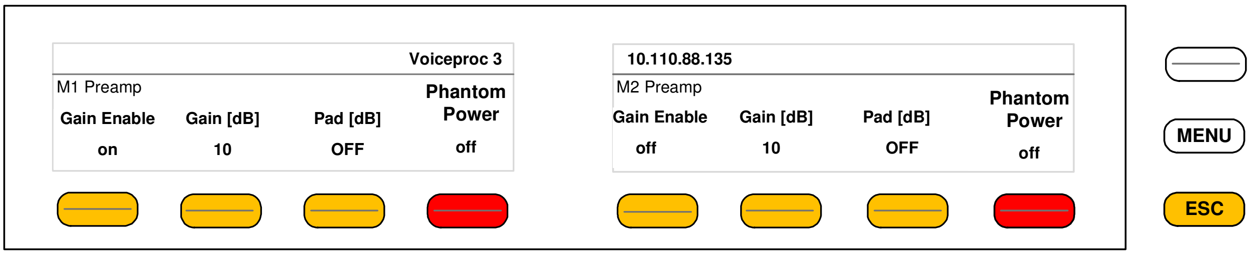
Operating – menu structure of the **X*AP RM1** remote panel – **operating displays**

When pressing the **<MENU>** button, the **upper operating display** opens up:



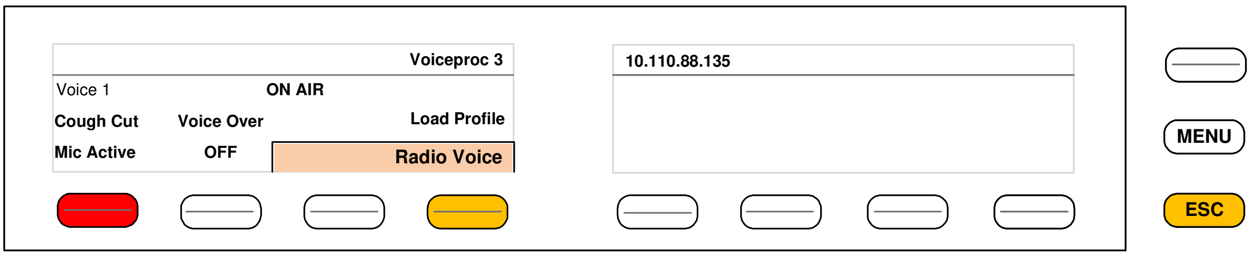

When pressing the **<ESC>** button you will return to the **main operating display**.

Operating display – **Analog Mic Interface**



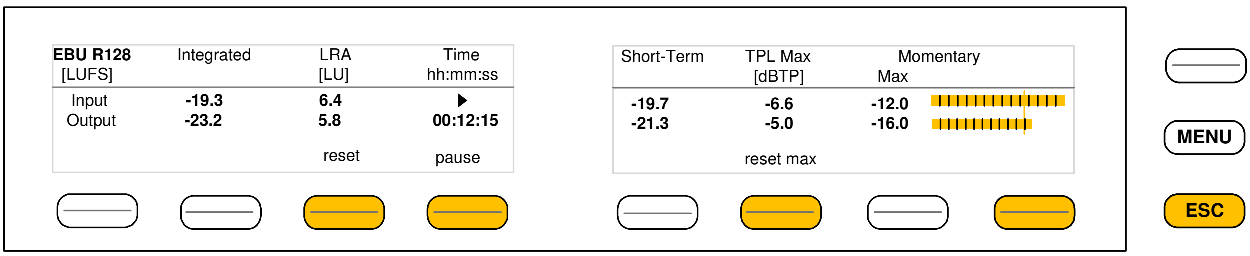

Here you can setup both mic preamps.

Operating display – **ON AIR profiles**



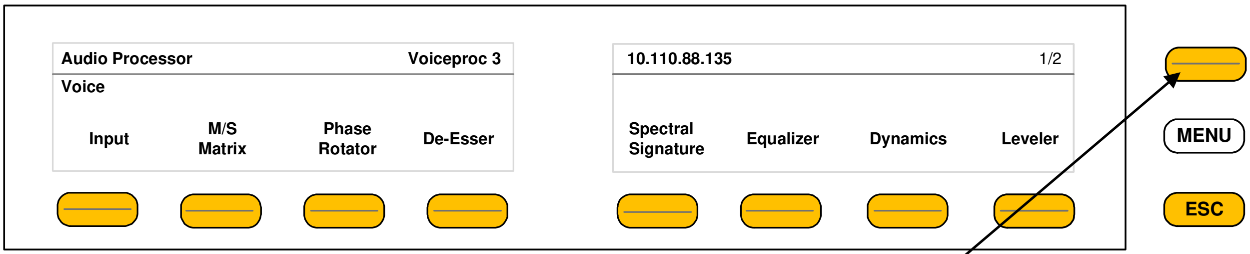

This is the same menu that you will reach if the feature set "Load Profiles" is selected (see previous page). Here you may remote control the **cough cut** for the respective voice channel and load a pre-defined profile. A **profile** is a set of audio relevant presets that must be set-up in the: **EVENTS > Events > Preset Events** section. Pressing the **<Load Profile>** hotkey will highlight the area above the button (see above – the default display is a dash). You can now select a profile by turning the **Rotary Encoder**. After the selection you must press the **Rotary Encoder** or the **<Load Profiles>** hotkey.

Operating display – **Loudness Meter**

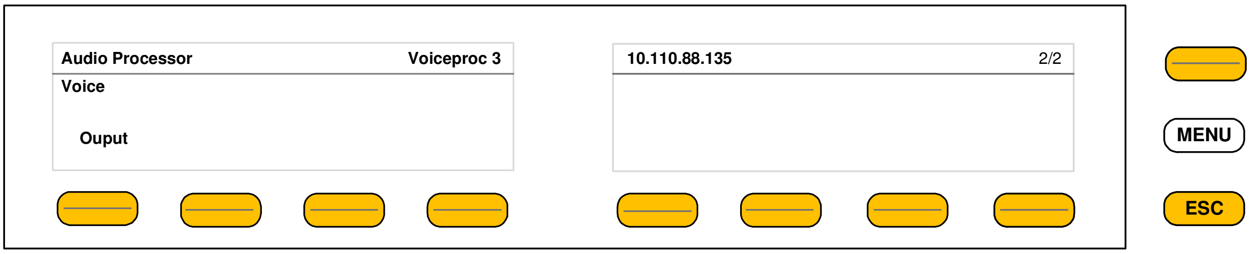



The meter style (ITU BS.1770-x / ATSC / EBU etc.) is defined by the settings of:
**AUDIO PROCESOR > Level Magic > Loudness Mode (example is for EBU R12().**
The above menu serves as a display of measurement values and offers the metering control buttons (reset & pause / continue).

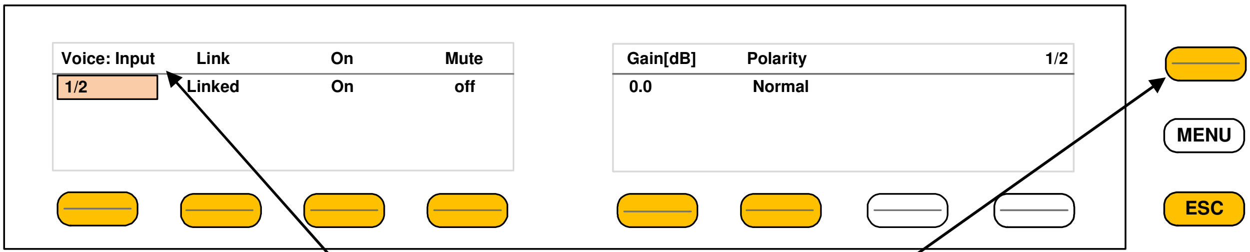
Operating display – **Audio Processor > Voice**




This menu gives access to tweak the voice channel(s). The active <**Shift**> button indicates that there is another page (2/2):

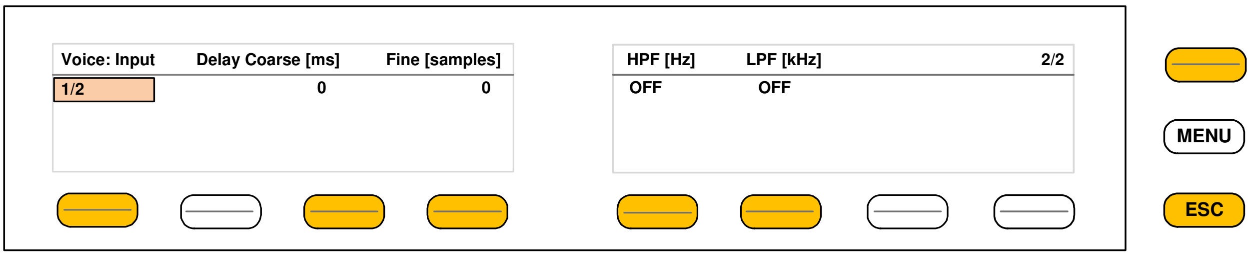



The example below explains how to set parameters via the **X*AP**. E.g. if you press <**Input**> all **p**arameters for the **Input** function block will be accessible (here we assume **Stereo** mode):

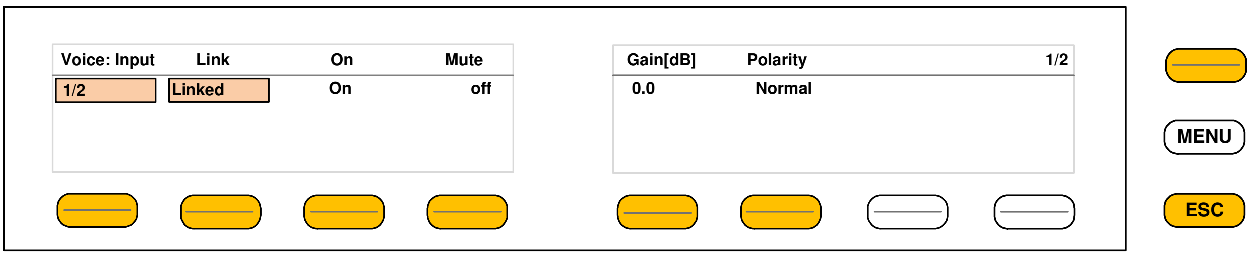



Here you are at the input section of the voice channel. The <**Shift**> button again toggles between two pages and gains access to the remaining parameters.

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

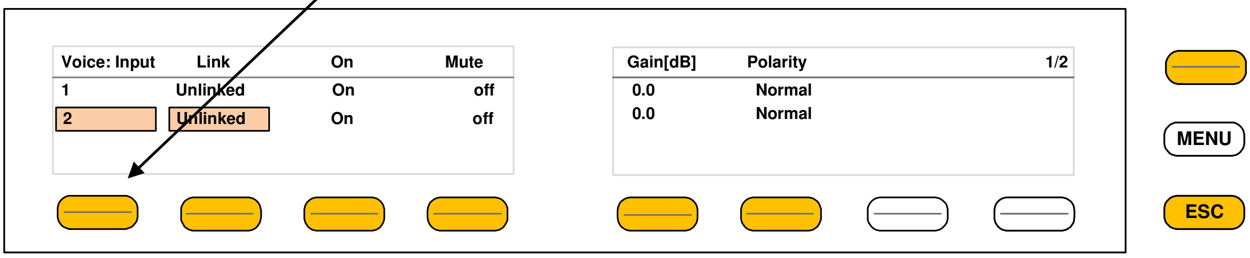



If the voice channels operate in stereo mode: **SYSTEM > Setup > Voice Channel Mode = Stereo**, you are able to **link** / **unlink** the respective processing blocks (see AUDIO PROCESSOR > Input):




The above example shows both voice channels in **Linked** mode.

When you press **hotkey #2** you are able to unlink both channels. Now you must simply push the rotary encoder (or turn it counter clockwise / clockwise) to toggle between **Linked** and **Unlinked** condition.
In case of **Unlinked**, the display shows two independent parameter sets.
By pressing **hotkey # 1** you can toggle the voice channel that is under control:




The examples above demonstrates the general way how to setup parameters of the **AUDIO PROCESSOR** of the **D*AP4 VAP**:
* Select a parameter
* Change it by using of the **Rotary Encoder**.
  - Push it to toggle states
  - Turn it to increment / decrement values.

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

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

**Power up** display

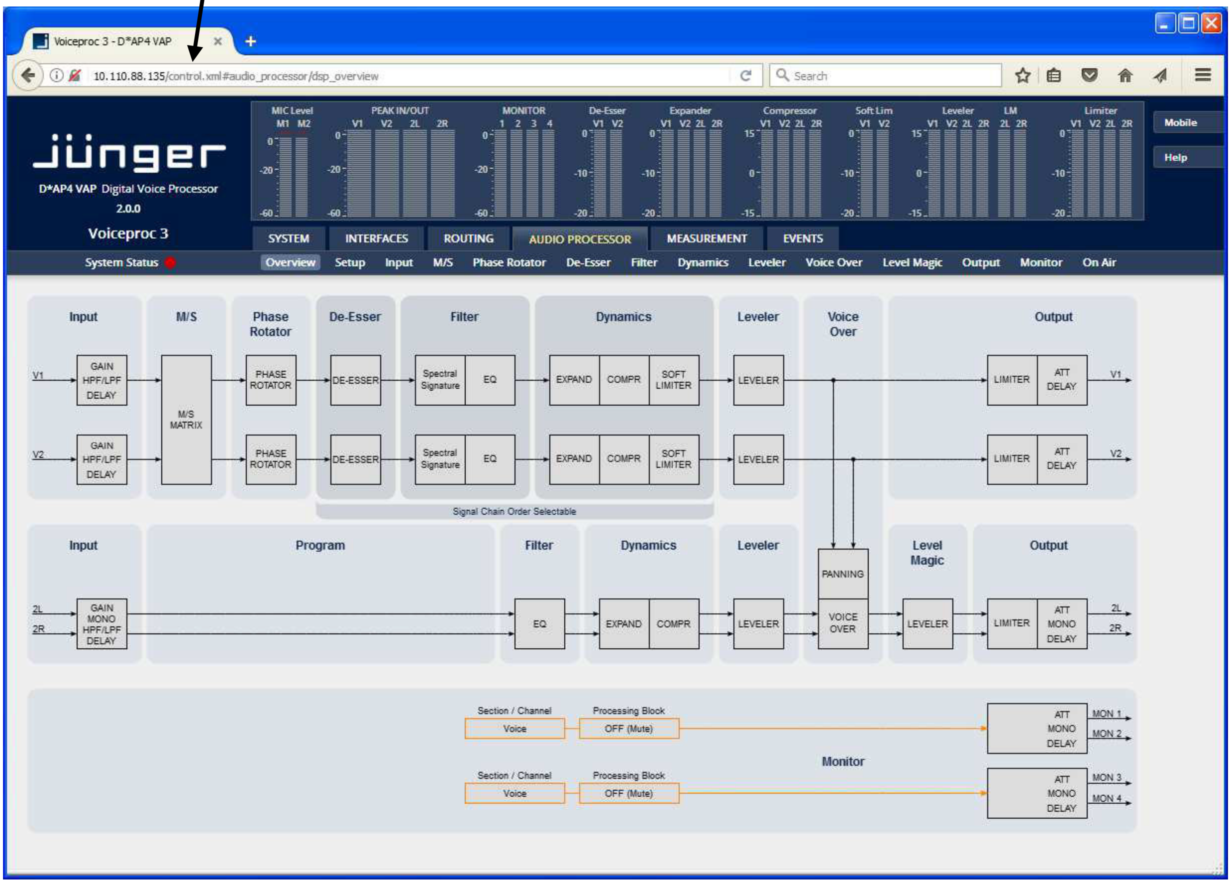
**<MENU>** opens **X*AP RM1** remote panel IP setup menu. See **X*AP RM1** manual for details.
<Address> setup
<Netmask> setup
<Gateway> setup
<empty>
Device 1 setup IP & ON / OFF
Device 2 setup IP & ON / OFF
Device 3 setup IP & ON / OFF
Device 4 setup IP & ON / OFF
**<ESC>** back to **power up** display

**<connect>** will connect with that particular **D*AP4 VAP** and opens the **main operating** display:
Hotkey #
1 <Clear>
2 <Just De-Ess>
3 <Gentle>
4 <Radio>
5 <Normalize>
6 <Heavy>
7 <empty>
8 <Panic>
**<ESC>** will jump back to **power up** display

**<MENU>** opens the **upper operating** displays:
Hotkey #
1 <Analog Mic Interface>
2 <ON AIR Profiles>
3 <not active>
4 <Loudness Meters>
5 <Voice>
6 <Program>
7 <Monitor 1/2>
8 <Monitor 3/4>
**<ESC>** back to **main operating** display

Setup GUI – connecting with the **D*AP4** unit **–** AUDIO PROCESSOR > **Overview**

You must open a browser and enter the **IP address** of the **D*AP4** unit into the **URL** field and press **<Enter>**. The browser will fetch the necessary information and will 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 section you see several bar graph displays for signal levels as well as for gain reduction display of several function blocks.

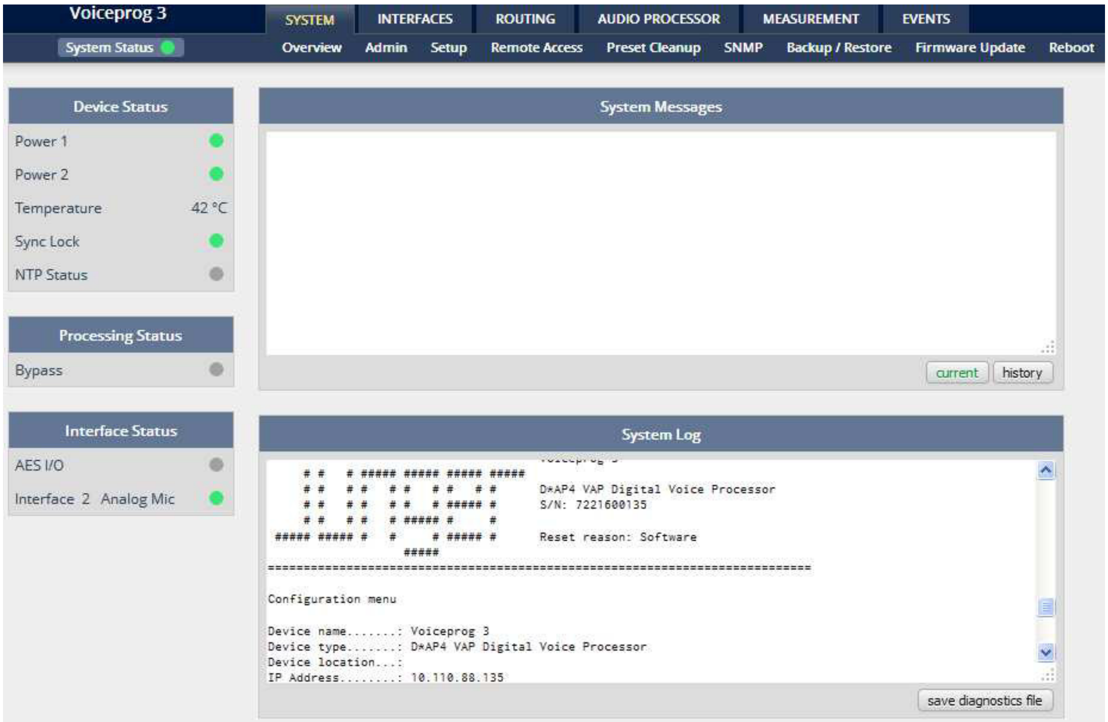
On the following pages we will go through the various panes to perform the basic setup of the device.

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

You must set up the installed interface module and finally set the signal routing.
You will find those settings under the **SYSTEM** link.

Setup GUI – SYSTEM – **System Status**

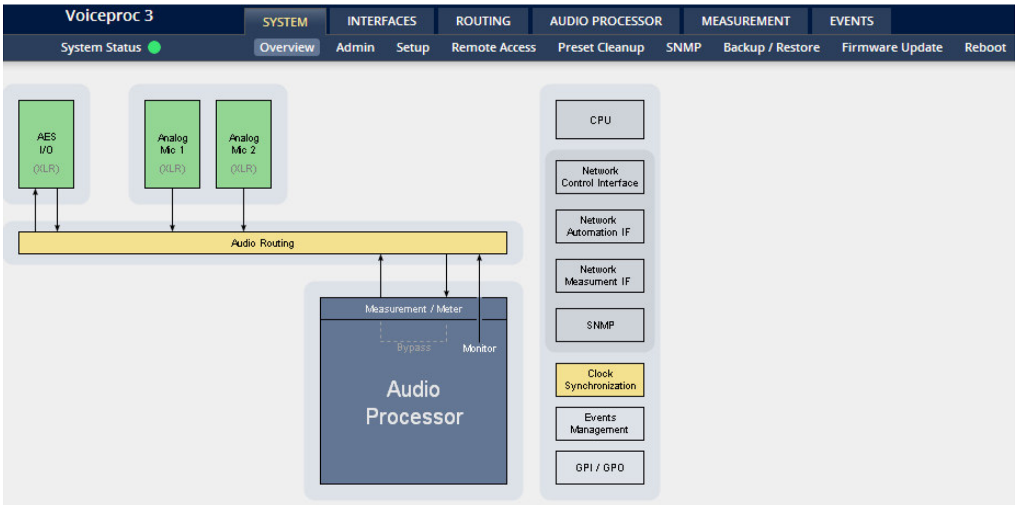
The system status is a special link you can reach independently from where you are:




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

| | |
|---|---|
| **Device Status** | Provides the hardware status of the **D*AP4 VAP.** |
| **Power 1** | Status of the first power supply (left hand side of rear). |
| **Power 2** | Status of second power supply (to the right of the first power supply) |
| **Temperature** | Measured on the surface of the main PCB. |
| **Sync Lock** | Turns red if the external sync source is lost or unstable. |
| NTP Status | Is grey if the NTP server synchronization is turned off. It is green if the clock is synchronized. It turns red, if the clock can't 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 (e.g. no carrier). |
| **Interface 2 Analog Mic** | Turns red if a problem with the optional mic interface board has been detected. |
| **System Messages** | [current / history]<br>Displays a list of messages produced by the system controller. |
| **System Log** | The system controller messages will be logged. This log information may be downloaded from the device and sent to Junger Audio. In case of a problem you can press: **<save diagnostics file>** from here or from: SYSTEM > Admin > Diagnostics. |

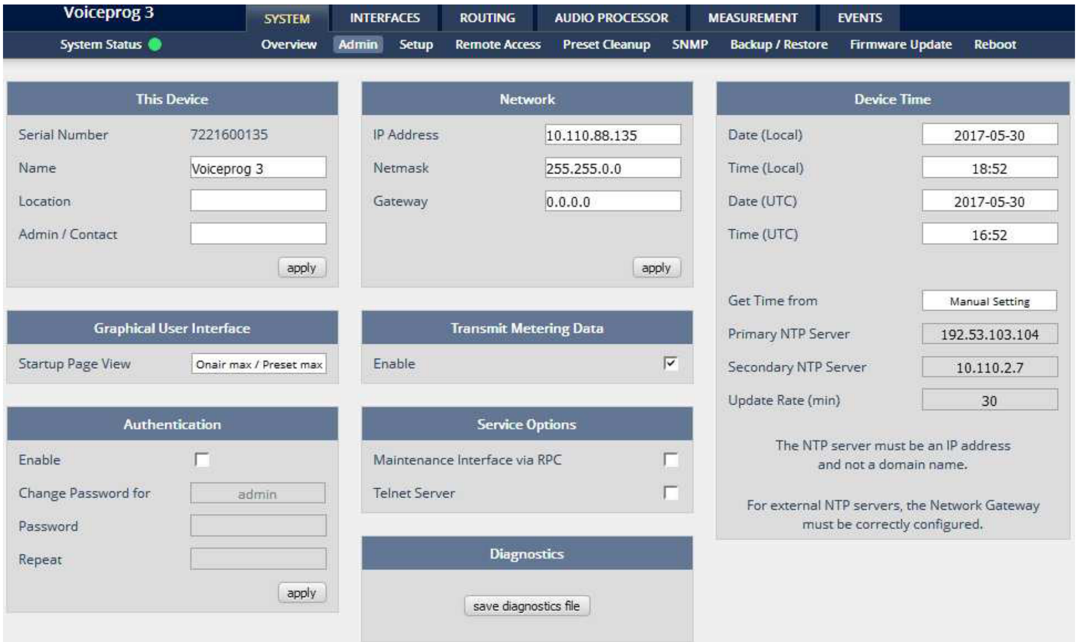
Setup GUI – SYSTEM – **Overview**




The graphical overview shows the main building blocks of the device including the options installed, in this example the microphone interface is placed into the INTERFACE 2 (see rear view) and a MADI I/O module is placed into INTERFACE 1 slot.

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

Setup GUI – SYSTEM – **Admin**



**This Device** Input fields for information utilized by higher level services.

**Serial Number** The electronic serial number. It is printed on a label at the rear of the device.

**Name** Give the device a meaningful name that may be used by name services and SNMP management.

**Location** The place where the **D*AP4 VAP** is located.

**Admin / Contact** e-mail address of a person in charge.

**Graphical User Interface** Defines the appearance of the parameter panes regarding preset editor and on air parameter visibility (see below – for preset concept).

**Startup Page View** Defines the appearance of the parameter panes regarding preset editor and on-air parameter visibility (see below – for preset concept).

**Authentication** To prevent non-authorized people from changing D*AP4 VAP settings the administrator may assign passwords for either the admin and/or an operator (same applies for talent/artist). 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** [enable / disable]
The administrator may turn authentication off.

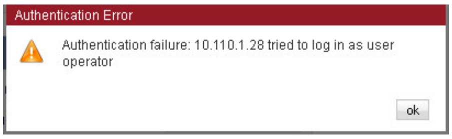
**Change Password for** [ON / OFF]
Select which password you will set / change.

**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*AP4 VAP – via web browser**

**IP Address** The address of your choice – default [10.110.xxx.yyy]

**Netmask** The net mask of your network – default [255.255.0.0]

**Gateway** The optional gateway address – default [0.0.0.0]

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

**Service Options**

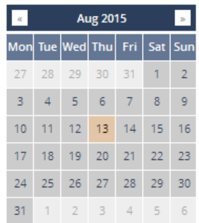
**Maintenance Interface via RPC** [OFF / ON]]
For administrative use to enable communication with factory tools**.**

**Telnet Server** [ON / OFF]
Enables a telnet server to connect the consol interface via IP (port 21).

**Diagnostics**

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

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




**Date (Local)** If you click into the **Date (local)** input field, a calendar tool: appears to select month and year.


**Time (Local)** If you click into the **Time (local)** input field, you will be able to set the device time.

**Date (UTC)** Similar as above for local date setting.

**Time (UTC)** Similar as above for local time setting.

**Get Time from** [Manual Setting / Browser / NTP Server]
If set to **NTP Server** the D*AP4 will look for the below servers to synchronize the internal clock.

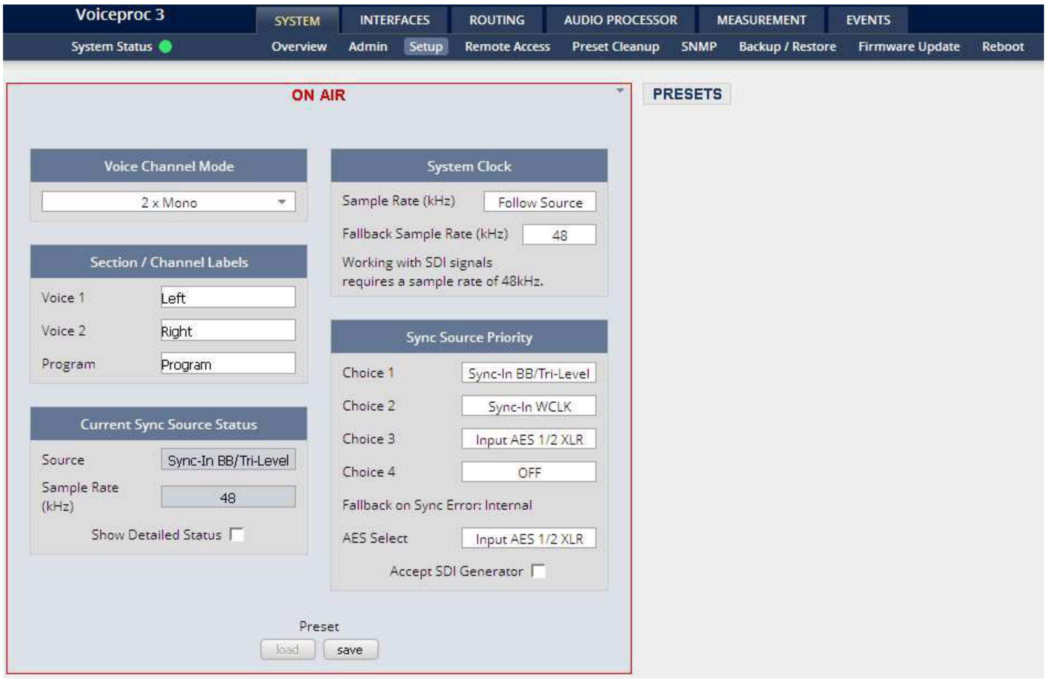
**Primary NTP Server** [5.9.110.236] default set to a publicly accessible NTP server via internet. This is used for device testing an may be overwritten at any time.

**Secondary NTP Server** [10.110.2.7] default set to an internal NTP server from Junger Audio. This is used for device testing and may be overwritten at any time.

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

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

Setup GUI – SYSTEM – **Setup**



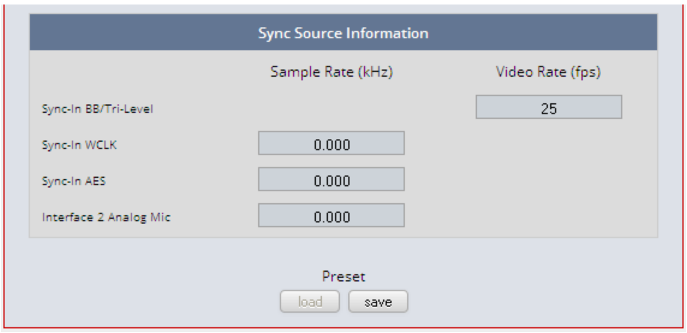

| | |
|---|---|
| **Voice Channel Mode** | [2 x Mono / Stereo]<br>Set according to the type of voice signal. This will automatically configure all relevant audio processing blocks. |

**Important Note!** If set to "Stereo" all relevant settings will provide an **unlink** function in case you need to setup both channels differently for any reason. I.e. the layout of several panes and the **X*AP RM1** will differ depending on this setting. If put into "Stereo" mode the label for Voice 1 (see below) will be used.

| | |
|---|---|
| **Section / Channel Labels** | Each of the individual voice channels as well as the program path |
| **Voice 1** | has a name that will be used as a reference for the display of |
| **Voice 2** | parameters and their setup. You may use names of your choice. |
| **Program** | Default names are Voice 1 / Voice 2 and Program. |

| | |
|---|---|
| **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]<br>If you enable the checkbox you will get this additional information: |

**Sync Source Information**




**System Clock**

| | |
|---|---|
| **Sample Rate (kHz)** | [Follow Source / 44.1 / 48 / 88.2 / 96] |
| **Fallback Sample Rate (kHz)** | [44.1 / 48 / 88.2 / 96] |

**Sync Source Priority**

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

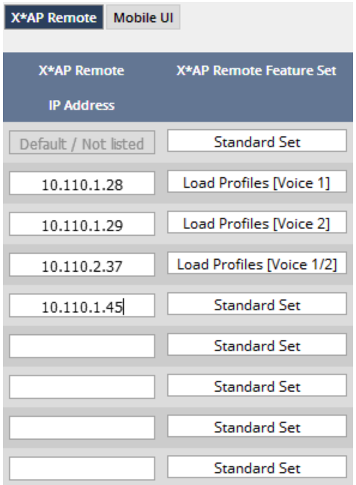
**Important note!** It is not possible to gen-lock the SDI generator. The generator will run on its own internal 27MHz crystal clock.

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

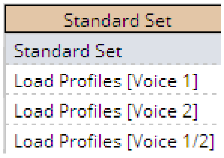
The **VAP** is designed for multi user applications where users will frequently alternate.
Since the VAP has two fully loaded processing channels one may use it for two independent mics from different sound booths. On the other hand the **X*AP** can control multiple **VAP**s one by one and a single **VAP** may be controlled from multiple **X*AP**s. This requires a flexible remote concept that allows you to recall pre-set configurations from the **X*AP** panel or from the mobile UI. You can control pre-settings of the **EVENTS** system via remote access from the **X*AP** remote panel or from a mobile **UI** on a tablet, a smart phone or even via a browser session from any PC in the network.

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

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




Multiple **X*AP** remote panels may gain access to the **VAP**.
For each **X*AP** you may pre-set **Feature Sets**:

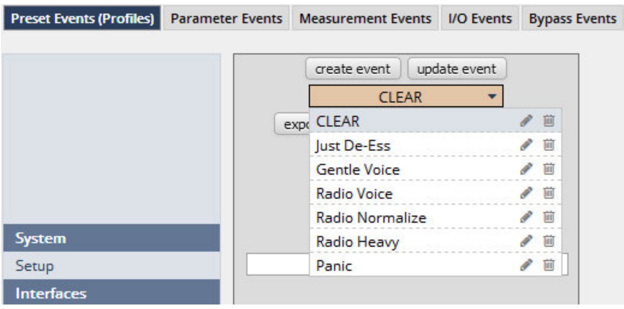



Profiles will be set up in the **EVENTS > Events > Preset Events (Profiles)** area. They combine a number of presets of several processing blocks.

**IP Address**

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

When you restore the factory defaults after an update to the latest **VAP** firmware via **SYSTEM > Reboot > Restore Factory Default Settings**, you will find a number of factory configured **preset events** also called **profiles**:

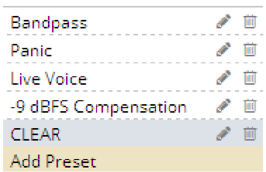



These profiles are based on factory default presets of various function blocks.

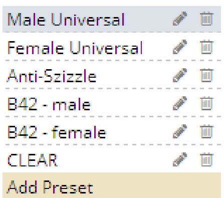
The **VAP** offers a variety of such pre-configured presets in all relevant function blocks to ease the use of the **VAP** in most day-to-day applications.

Here are a few examples of factory default presets of the **AUDIO PROCESSOR**:

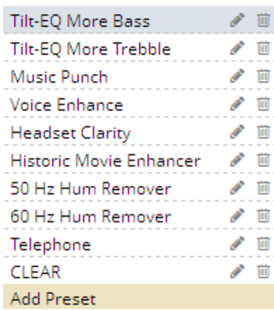
**> Input:**



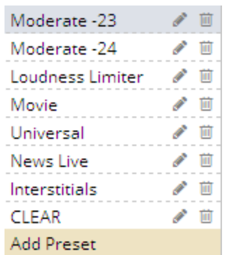

**> De-Esser:**




**> Filter > Equalizer:**

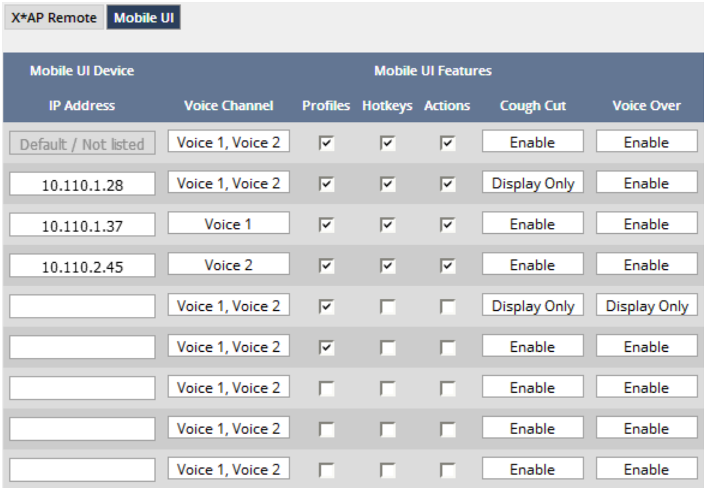



**> Level Magic:**



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

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



X*AP Remote | Mobile UI

| Mobile UI Device | Mobile UI Features | | | | | |
|---|---|---|---|---|---|---|
| IP Address | Voice Channel | Profiles | Hotkeys | Actions | Cough Cut | Voice Over |
| Default / Not listed | Voice 1, Voice 2 | ☑ | ☑ | ☑ | Enable | Enable |
| 10.110.1.28 | Voice 1, Voice 2 | ☑ | ☑ | ☑ | Display Only | Enable |
| 10.110.1.37 | Voice 1 | ☑ | ☑ | ☑ | Enable | Enable |
| 10.110.2.45 | Voice 2 | ☑ | ☑ | ☑ | Enable | Enable |
| | Voice 1, Voice 2 | ☑ | ☐ | ☐ | Display Only | Display Only |
| | Voice 1, Voice 2 | ☑ | ☐ | ☐ | Enable | Enable |
| | Voice 1, Voice 2 | ☐ | ☐ | ☐ | Enable | Enable |
| | Voice 1, Voice 2 | ☐ | ☐ | ☐ | Enable | Enable |
| | Voice 1, Voice 2 | ☐ | ☐ | ☐ | Enable | Enable |

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

**Voice Channel** [Voice1 / Voice 2 / Voice1, Voice2]

**Profiles** [ON / OFF]
Turn it on if the respective **UI** should show profiles to load.

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

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

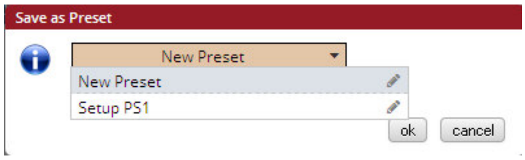
**Cough Cut** [Enable / Display Only / Hide] the **<Cough Cut>** button of the **UI**.

**Voice Over** [Enable / Display Only / Hide] the **<Voice Over>** button on the **UI**.
It will appear in the UI if manual voice over is selected
(see AUDIO PROCESSOR > Voice Over > Mode = Manual).

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

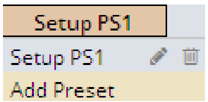
The example above shows the **preset concept** of the **D*AP4 VAP.** It is a general feature of the device and you will come across it in almost every area. For all relevant settings one set of **ON AIR** parameters and a practically unlimited number of **PRESETS** are available. The count depends on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

**<load>** or **<save>**:


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

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




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

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

**Important Note!** The presets of the **D*AP4 VAP** are persistent by nature. You are working directly on the preset memory, i.e. you need not worry about storing such presets. The **D*AP4 VAP** does it for you.
On the other hand you must be aware that you are **overwriting the actual preset settings!** If you want to keep original values (e.g. from a factory preset) you must simply **copy** the content of the 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 another preset which you have selected before pasting.

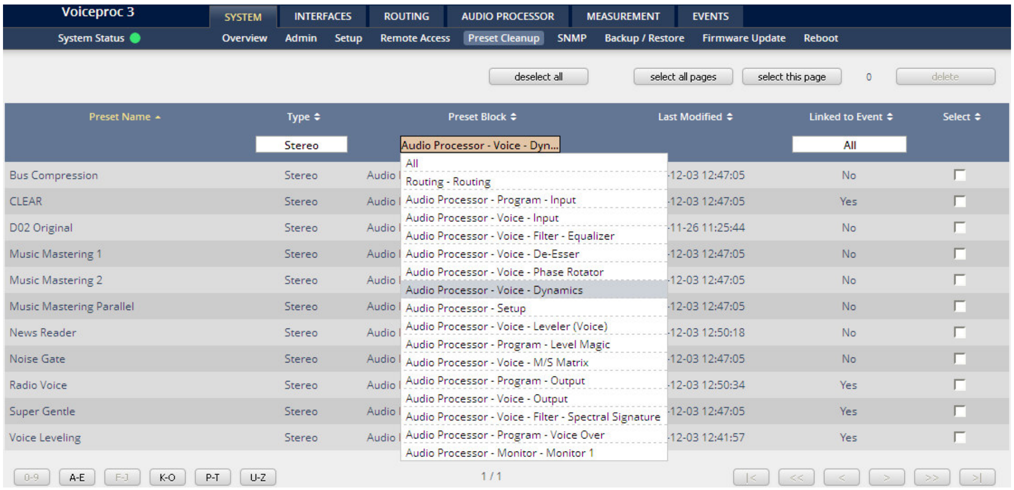
You may also **<export>** or **<import>** the preset content to / from a file.

Setup GUI – SYSTEM – **Preset Cleanup**

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


| Preset Name | Type | Preset Block | Last Modified | Linked to Event | Select |
|---|---|---|---|---|---|
| | Stereo | All | | All | |
| -9 dBFS Compensation | Stereo | Audio Processor - Input | 2014-12-02 10:44:34 | No | |
| -9 dBFS Compensation | Stereo | Audio Processor - Input | 2014-12-02 10:44:55 | Yes | |
| 50 Hz Hum Remover | Stereo | Audio Processor - Filter - Equalizer | 2014-12-02 10:55:44 | No | |
| 60 Hz Hum Remover | Stereo | Audio Processor - Filter - Equalizer | 2014-12-02 10:55:44 | No | |
| Anti-Szizzle | Stereo | Audio Processor - De-Esser | 2014-12-02 10:46:19 | No | |
| Auto | Stereo | Audio Processor - Phase Rotator | 2014-07-23 12:50:00 | Yes | |
| B42 Female | Stereo | Audio Processor - De-Esser | 2014-12-02 10:46:19 | No | |
| B42 Male | Stereo | Audio Processor - De-Esser | 2014-12-02 10:46:19 | No | |
| Bandpass | Stereo | Audio Processor - Input | 2014-12-02 10:44:34 | No | |
| Bandpass | Stereo | Audio Processor - Input | 2014-12-02 10:43:30 | No | |
| Bus Compression | Stereo | Audio Processor - Dynamics | 2014-12-03 12:47:05 | No | |
| CLEAR | Stereo | Audio Processor - De-Esser | 2014-12-02 10:46:19 | Yes | |

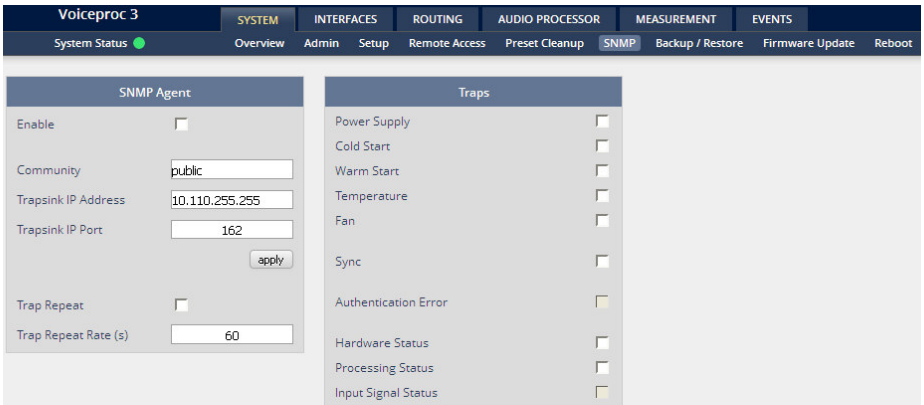
You can sort the table by pressing on one of the column headlines. You can qualify your selection by the "Type" selector and / or the "Preset Block", "Linked to Event", "Last Modified" column headlines.
The pull down lists allow to reduce the number of presets displayed:




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:

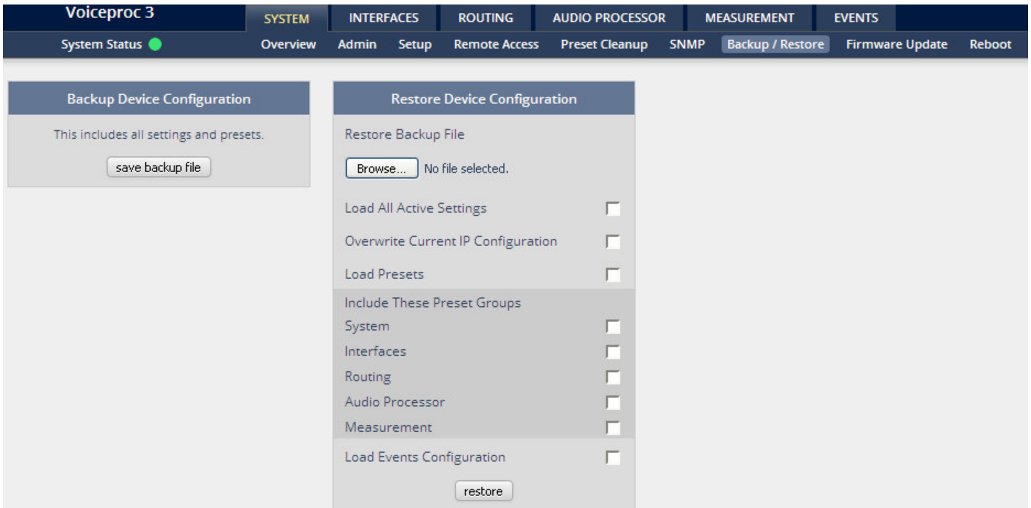

A selection is made by clicking on a line to activate the check box. Once you have made your selection (highlighted lines) you can press the **<delete>** soft button to execute the process. This will remove the selected presets permanently from the device.

## Setup GUI – SYSTEM – **SNMP**

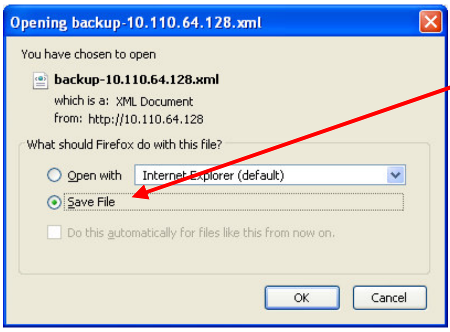



This pane is meant for basic settings of the **SNMP Agent** of the device. If you don't use SNMP based system monitoring, you should not enable the SNMP agent.

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

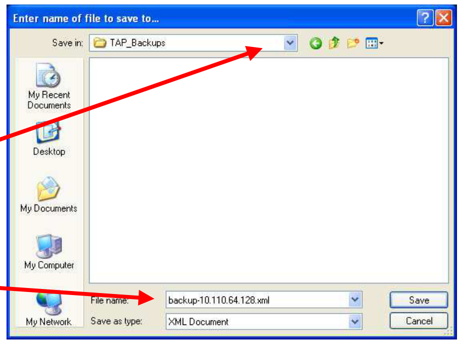



Here you can **back up** the complete **device** and **restore** parts or all of it. If you press **<back up>** the device controller will collect all necessary data and assemble it to an XML file.
Finally you will get a pop up message:




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

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

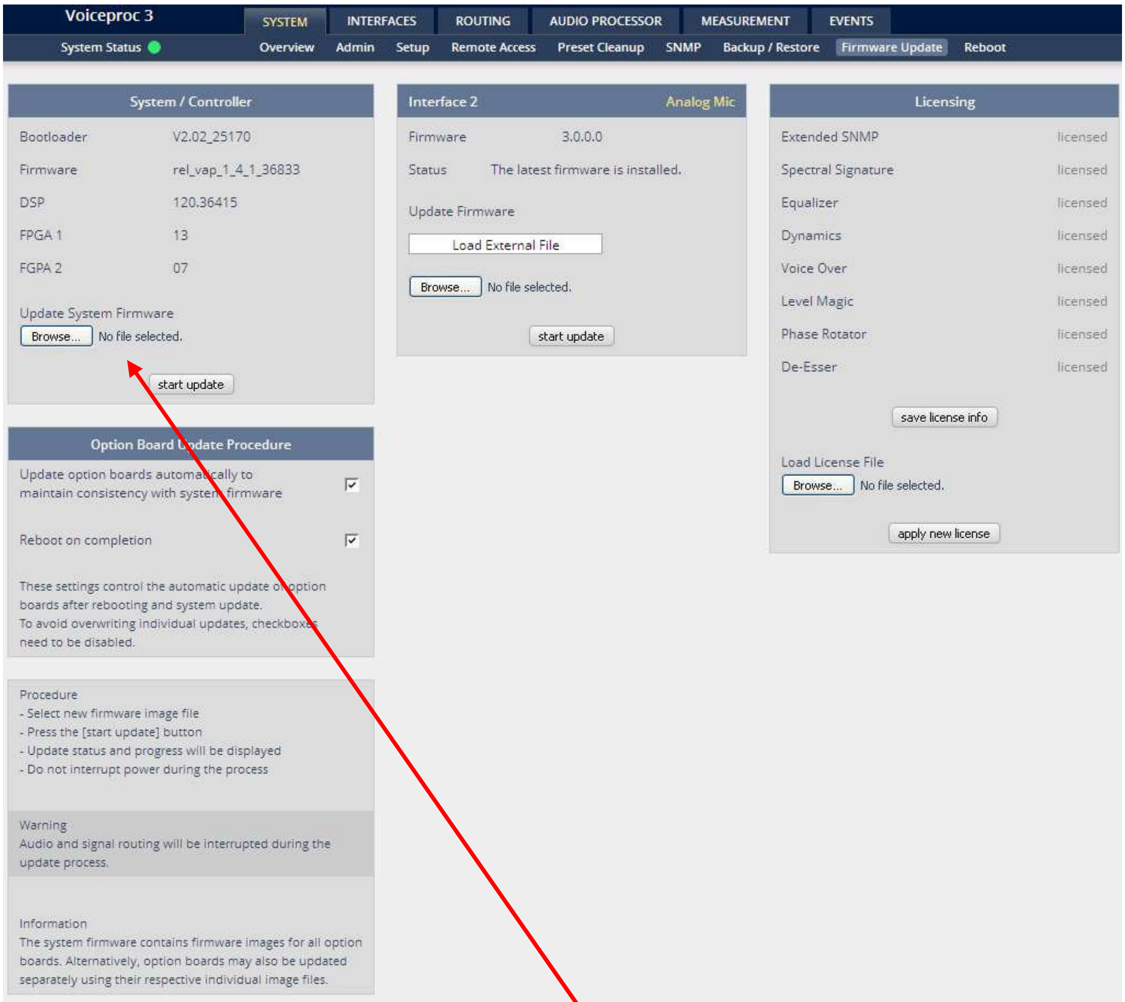



Similar applies to the restore process. You must select the desired backup file which you want to restore and check the necessary option(s) under "Restore Device Configuration".

Setup GUI – SYSTEM – **Firmware Update**

The files to update the **D*AP4 VAP** will be available in **ZIP** format. You must unpack them to your PC in order to access them for the update procedure.

You will find an image file for the **D*AP4 VAP** core system in the format (example): "rel_vap_1_0_1-26328.img" as well as update files for components, like the optional interface boards in the format: "rsdi150_v51.sdi" or for the **X*AP RM1** remote panel.

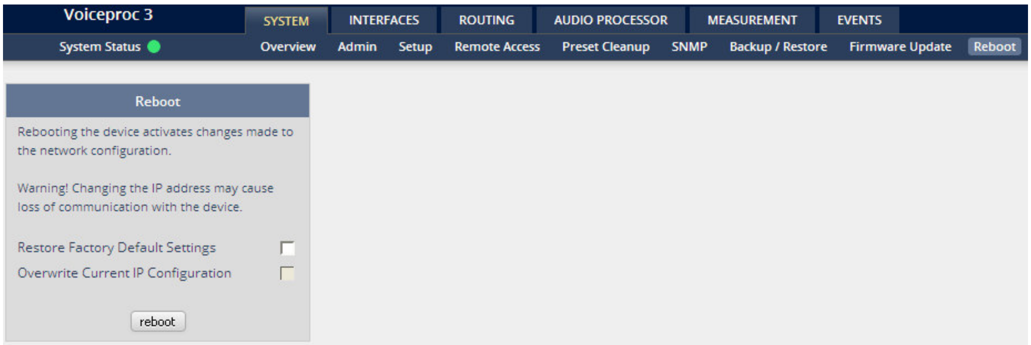



To update the **D*AP4 VAP**, you must **<Browse …>** to find the respective firmware file (which you have unzipped before) and press **<start update>**. After finishing the procedure the device will automatically reboot.

You may also update the firmware of an installed SDI board (Interface 1) or an Analog Mic board (Interface 2). The respective file(s) have been uploaded together with the system firmware so you can select an update file form the **Update Firmware** pull-down box. In case you provide an extra file you must select the option: "Load External File".

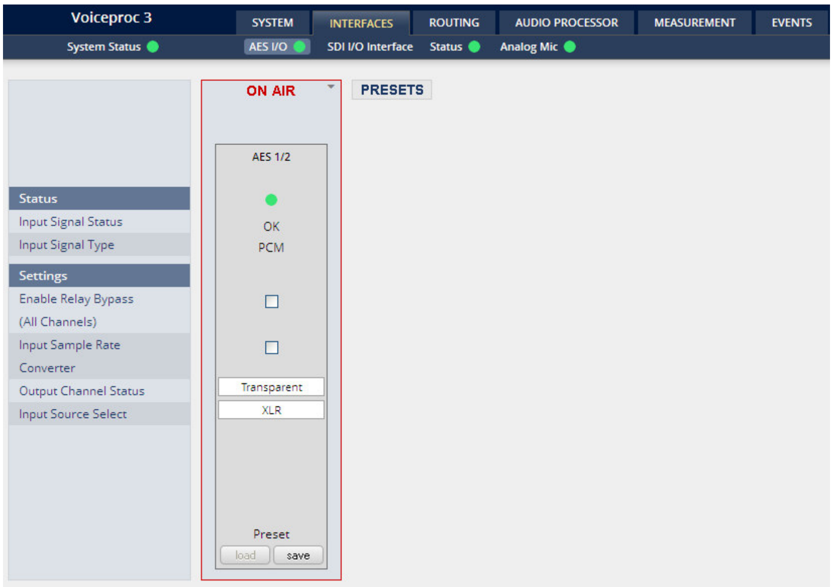
Finally you can see the options of your device which you have bought a license for. When you buy a license you must provide the **license info** file and you in return will get a **new license** file which you must apply to the device here.

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** [red / green]

**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.

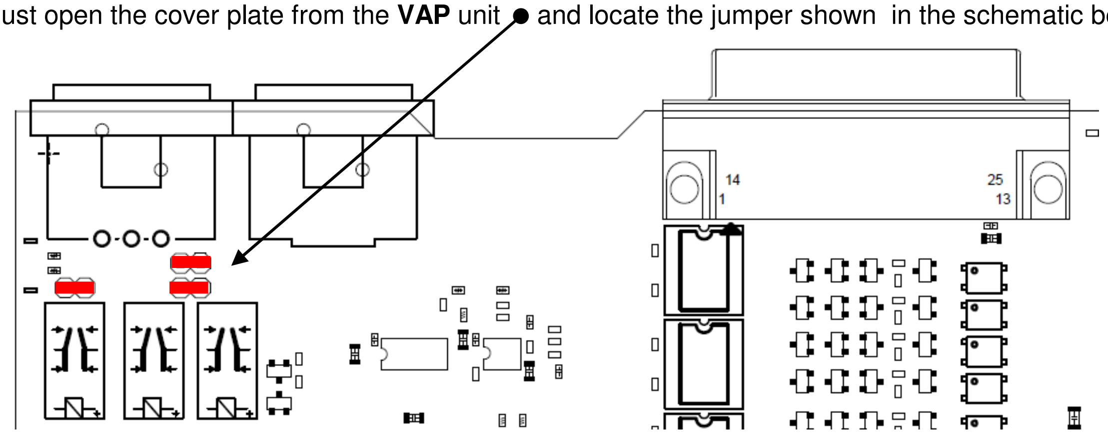
**Important Note!** The input signal status is logically combined and represented as part of the System Status. If one of the inputs is not assigned by the ROUTING matrix, its status will not be incorporated into the System Status. If non of the inputs is routed the Interface Status > AES I/O status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

**Settings**

**Enable Relay Bypass (All Channels)** [ON / OFF]
.

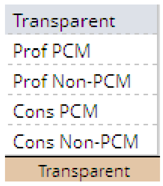
**Input Sample Rate Converter** For asynchronous sources it is possible to turn a **SRC** on.
If a **SRC** is turned on and the input status becomes **Non-PCM**, the **SCR** will be turned OFF automatically in order to maintain the original data structure of the encoded bit stream (e.g. Dolby E).

**Important note!** The AES relay bypass circuit of the AES I/Os may be deactivated inside the **VAP.** You must open the cover plate from the **VAP** 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.

| | |
|---|---|
| **Output Channel Status** | [Transparent / Prof. PCM / Prof Non-PCM / Cons. PCM / Cons. Non-PCM]<br>The channel status can either be transparent from the input source of the **D*AP4 VAP** or may be overwritten. |
| **Input Source Select** | [XLR]<br>The **D*AP4 VAP** has an XLR input only. |



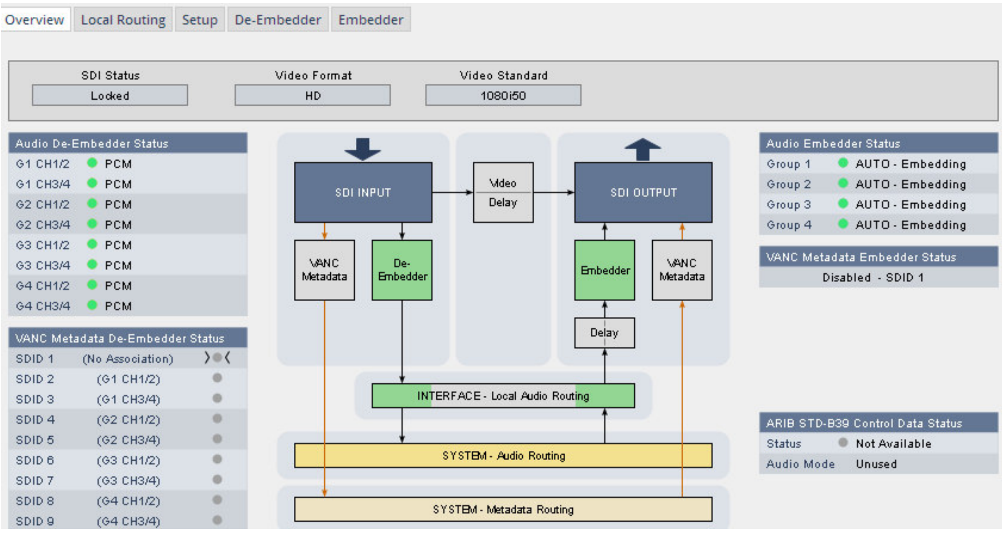

Setup GUI – INTERFACES – **Analog Mic**


**Setup**

| | |
|---|---|
| **Input** | [Mic / Line] |
| **Enable Preamp Gain** | [ON / OFF] |
| **Preamp Gain (dB)** | [10 … 65] |
| **Pad (dB)** | [OFF / -10] |
| **Phantom Power** | [ON / OFF]<br>Phantom power is available when Input = Mic is selected. |

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

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

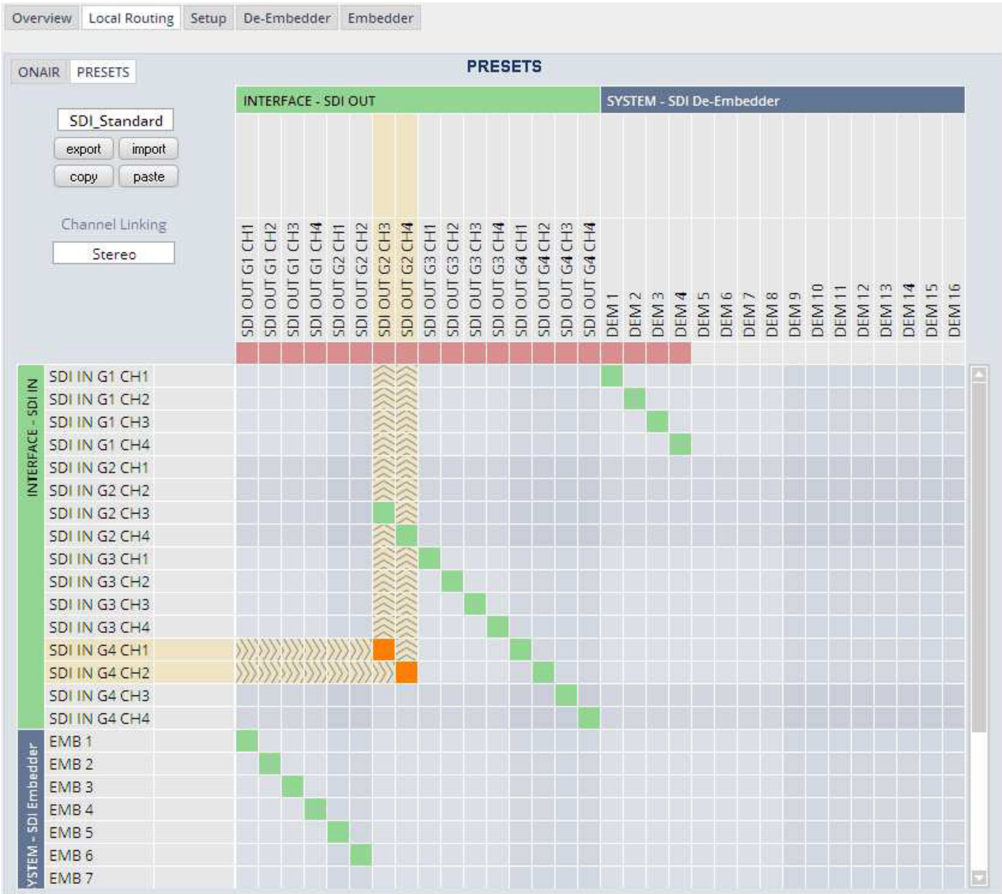



The overview pane shows all relevant information of that interface:

| | |
|---|---|
| **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:<br>**Embedding –** a new group will be built<br>**Replace –** the structure of the group from the input is kept and the audio content is simply replaced<br>**Delete –** the group from the input is deleted<br>**OFF –** the embedder from that group is turned off |
| **VANC Metadata Embedder Status** | [Enabled / Disabled & selected SDID#]<br>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"<br>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. Below you see an example routing that sends all signals 1:1 from the physical de-embedders [INTERFACE – SDI IN G2 CH3 … SDI IN G4 CH4] to the physical embedders [SDI OUT G2 CH3 … SDI OUT G4 CH4]. The signals from the physical de-embedders [SDI IN G1 CH1 … SDI IN G1 CH4] are sent to the device router [DEM 1 … DEM4] while the device router outputs [SYSTEM – SDI Embedder EMB 1 … EMB 6] are routed to the first 6 SDI channels [SDI OUT G1 CH1 … SDI OUT G2 CH2]:

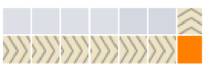



You must use the scroll bar to navigate through the matrix. In the upper left corner you can select between the **ONAIR** and the **PRESETS** view of the matrix. On the **ON AIR** page you will also see the device signal labels (see ROUTING section further below for details).

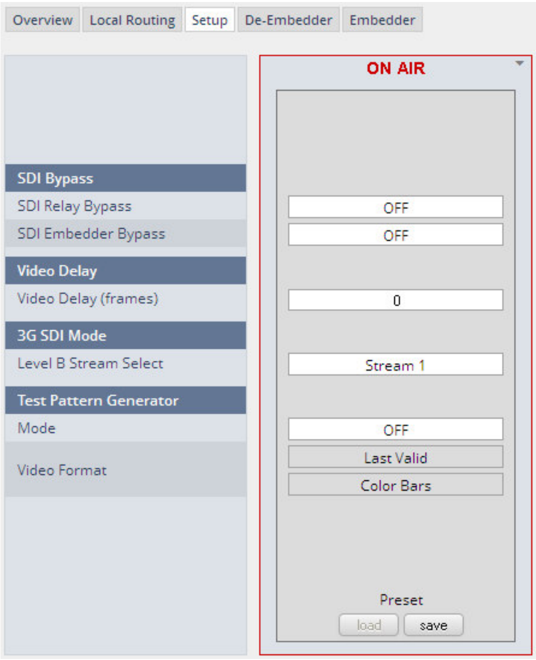
| | |
|---|---|
| **Channel Linking** | [mono / stereo]<br>You can decide if the routing must be performed in mono or stereo mode (where adjacent odd/even channels are routed at once). |

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** | <u>Color codes of cross points:</u> |
|---|---|
| **dark blue** | Possible new cross point. |
| **orange** | You are about to reconnect a cross point. |
| **grey** | Cross point is not allowed<br>(i.e. routing will cause a loop and will not therefore be performed). |
| **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**

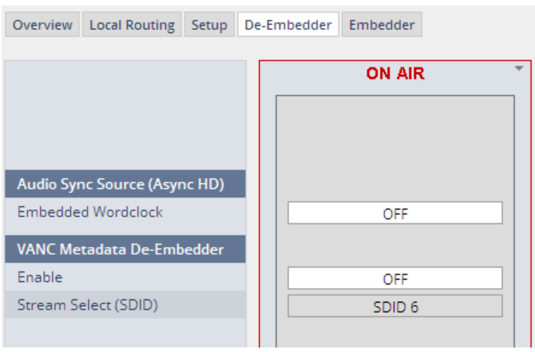
**Level B Stream Select** A 3G-SDI signal may have two HD sub streams (e.g. for 3-D TV), AKN as 3G-B standard select between stream 1 or 2 for embedded audio. See SMPTE 425M for details.

**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**



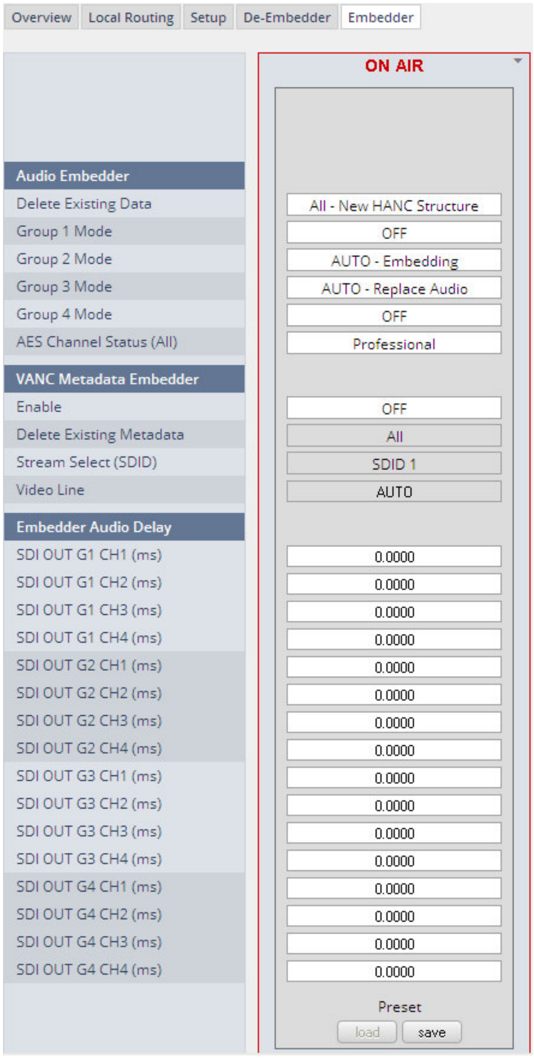

**Audio Sync Source (Async HD)** The HD SDI standard allows for asynchronous audio. This is critical if you have decided to synchronize the device on such signal. Here you find a solution. You may either use the embedded word clock or the SDI carrier itself as a reference.

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

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

DEM1= From de-embedder group 1 channel 1.

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]
If Professional these values are used:

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

**Important note!** If you generate a new AES channel status the **Audio Mode** will be automatically set to **Non Audio** (AKA "other") for both channels, if 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 VANC lines are available for data insertion.

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

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

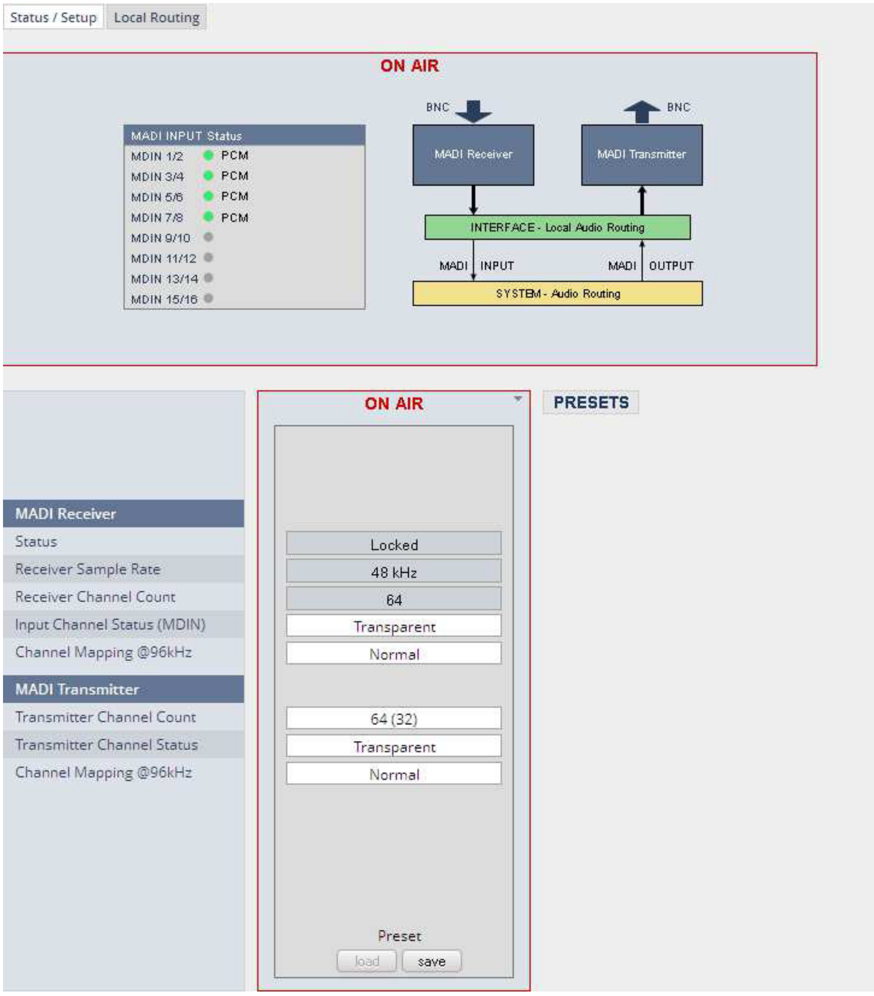
**SDI OUT G1 CH1 (ms)** [0.0000 … 340.000]

**to**

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

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

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




**MADI Receiver**

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

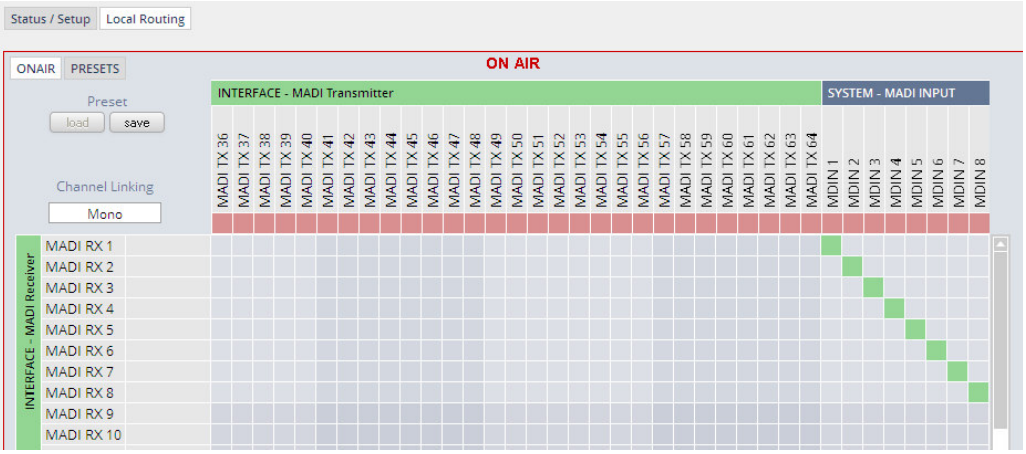
**MADI Transmitter**

| | |
|---|---|
| **Transmitter Channel Count** | [64 (32) / 56 (28)]<br>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] |

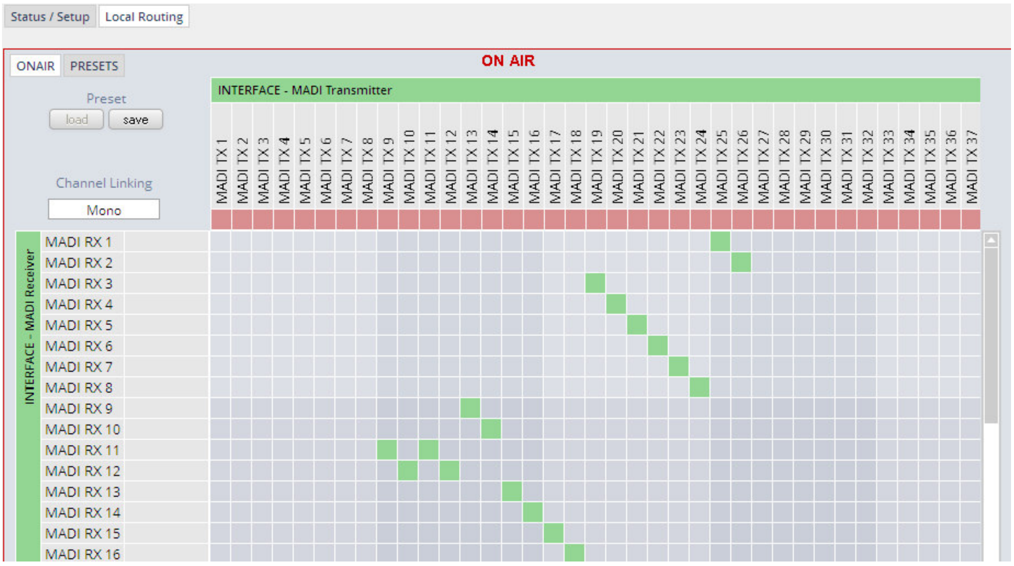
The connection for fiber cable is made by a LC connector. Looking at the rear panel the transmitter is the left one and the receiver the right one.

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

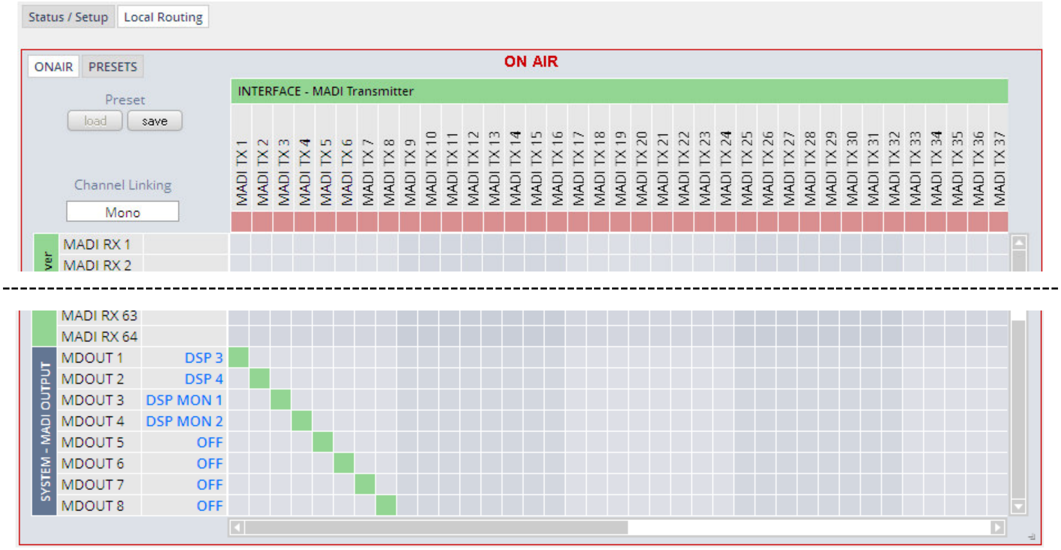
Below are some excerpts from the local routing pane. Single channels from or to the **VAP** may be connected with the MADI transmitter or MADI receiver respectively.
The example below shows the first 8 MADI channels from the receiver (MADI RX 1 … MADI RX 8) connected with the device inputs **SYSTEM MADI INPUT** (MDIN 1 … MDIN 8):




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



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




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*AP4** to an audio over IP (AoIP) network. Junger Audio has committed itself to the quasi industry standard **Dante** developed by the company **Audinat:**

"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 **A**udio **o**ver **IP** must be able to handle the **IP multicast**. So it needs a bit of care when it comes to network gear. The recommendation is to separate the control network from the audio network.

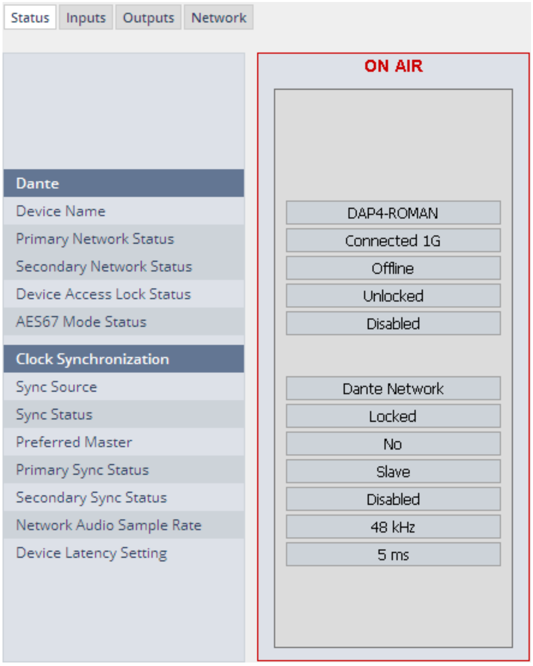
For details pls. refer to the Audinate web-site: https://www.audinate.com. Here you will find many useful application videos and FAQs.

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

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

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

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




**Dante**

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

**Primary Network Status** [Offline / Connected + bandwidth]

**Secondary Network Status** [Offline / Connected + bandwidth]

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

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

**Clock Synchronization**

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

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

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

**Preferred Master** [No / Yes]
The **Dante** algorithm automatically looks for the best clock master inside the network but one may force a **Dante** module to become the clock master.

**Primary Sync Status** [Slave / Master]

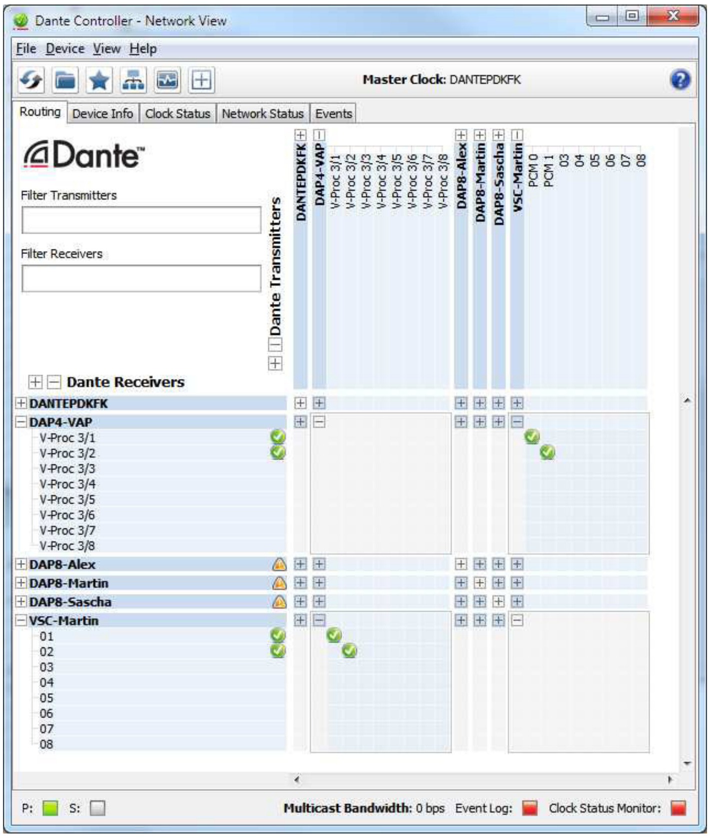
**Network Audio Sample Rate** [44.1 kHz / 48 kHz / 88.2 kHz / 96 kHz]
Depending on the A*P device type the sample rate is limited to the device specification.

**Device Latency Setting** [5ms]
You can allow for a certain transmission latency if you face network problems of any kind.

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*AP VAP** and will be displayed there.

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



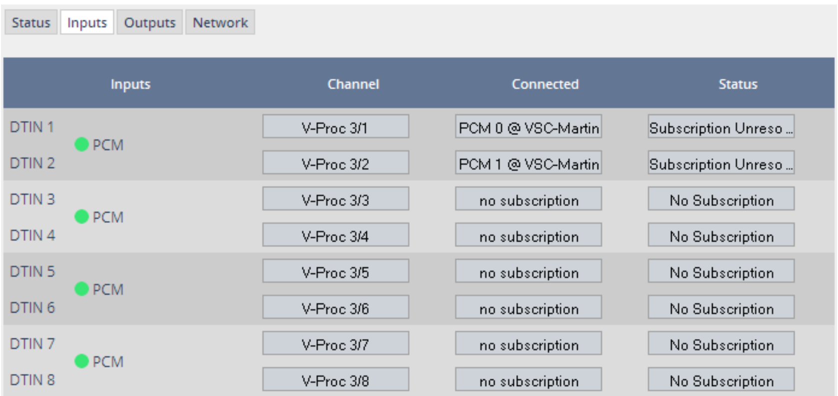

As an example you see here a "DAP4-VAP" (name given by the Dante Controller) that has assigned the labels V-Proc 3/1 … 3/8 for both the inputs and the outputs.

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

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

The outputs PCM 0 and PCM 1 from the VCS are assigned to the **VAP** inputs V-Proc 3/1 and 3/2 while two outputs from the "DAP4-VAP" are assigned to the VSC inputs "01" and "02".

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



Status | Inputs | Outputs | Network

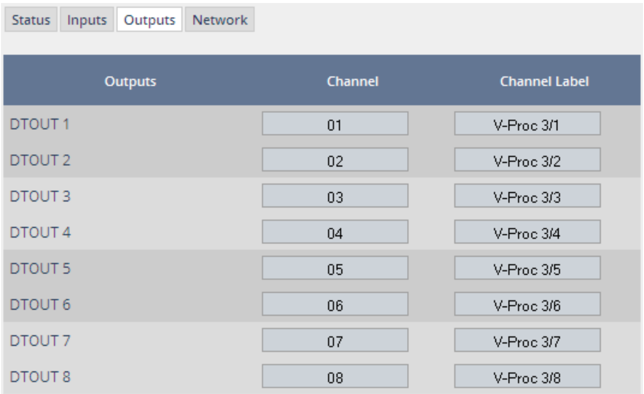
| Inputs | | Channel | Connected | Status |
|---|---|---|---|---|
| DTIN 1 | PCM | V-Proc 3/1 | PCM 0 @ VSC-Martin | Subscription Unreso... |
| DTIN 2 | | V-Proc 3/2 | PCM 1 @ VSC-Martin | Subscription Unreso... |
| DTIN 3 | PCM | V-Proc 3/3 | no subscription | No Subscription |
| DTIN 4 | | V-Proc 3/4 | no subscription | No Subscription |
| DTIN 5 | PCM | V-Proc 3/5 | no subscription | No Subscription |
| DTIN 6 | | V-Proc 3/6 | no subscription | No Subscription |
| DTIN 7 | PCM | V-Proc 3/7 | no subscription | No Subscription |
| DTIN 8 | | V-Proc 3/8 | no subscription | No Subscription |

**Inputs**

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

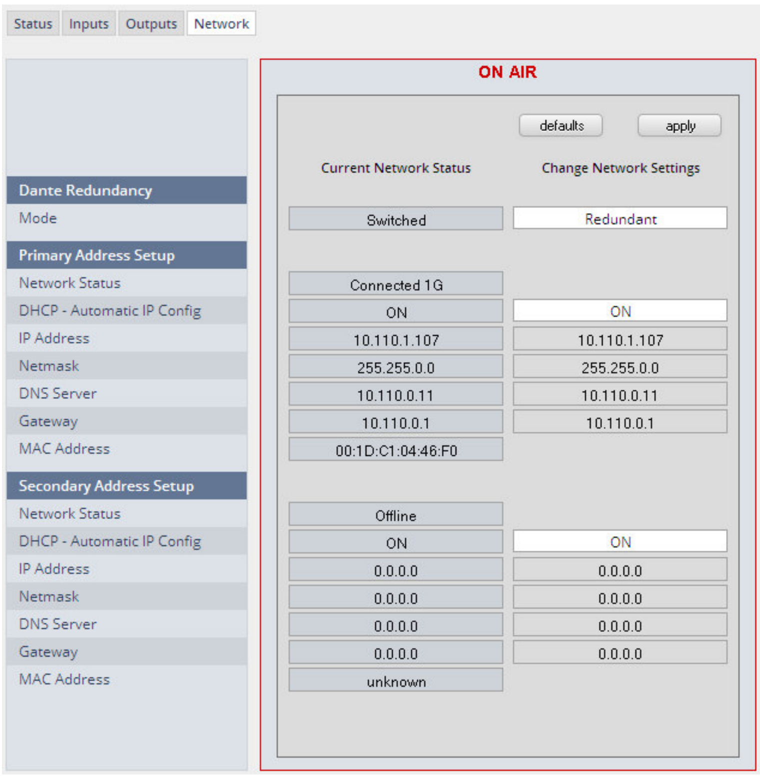
| | |
|---|---|
| **Channel** | The labels assigned to that channel by the **DanteController** |
| **Connected** | The source of the audio signal. |
| **Status** | [No Subscription / Subcription Unresolved / Wait / Naming Problem / Loopback / Idle / Subscription in Progress / Connected (Unicast) / Connected (Multicast) / Manual Config / Format Problem / QoS Problem / Latency Problem / Clock Domain Problem / Link Down / Fail / Unknown]<br>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 eight labels can be configured for each stream from the interface to the network. This allows configuring multi layer routing. |

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




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

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

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

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

**Important Note!** It may happen by accident that the update of the Dante module fails. E.g. if the firmware update option: SYSTEM > Firmware Update > Option Board Update is set to "Update option boards automatically …." and the device loses power during this process, the Dante module will be in the fail-save state. This is indicated in the Dante Controller software.
In this case you must repair it by aid of a Dante tool. You can download it from the website:
https://www.audinate.com/content/dante-firmware-update-manager-v31009-windows

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

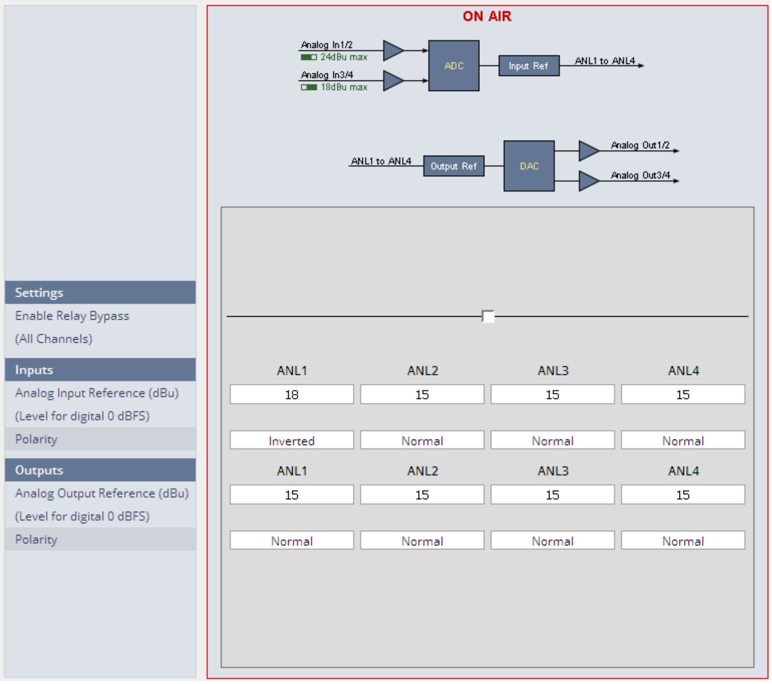
The update manager performs two tasks, the recovery from the fail-safe state and the update of an valid Junger basic firmware for the Dante module.

After you have managed to recover from fail-safe you must power cycle the **D*AP4** 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_vap_1_4_3.zip > junger_vap_firmware > Dante_recovery_image > DT-100-v1.0.3-7.dnt

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:




The graphics shows the setting of the input reference level set via the jumpers on the PCB (see sketch below).

**Settings**

**Enable Relay Bypass**
(All Channels)
[ON / OFF]

**Inputs**

**Analog Input Reference (dBu)**
**(level for digital 0dBFS)**
[0.0 … 15.0 … 18.0 / 24.0]

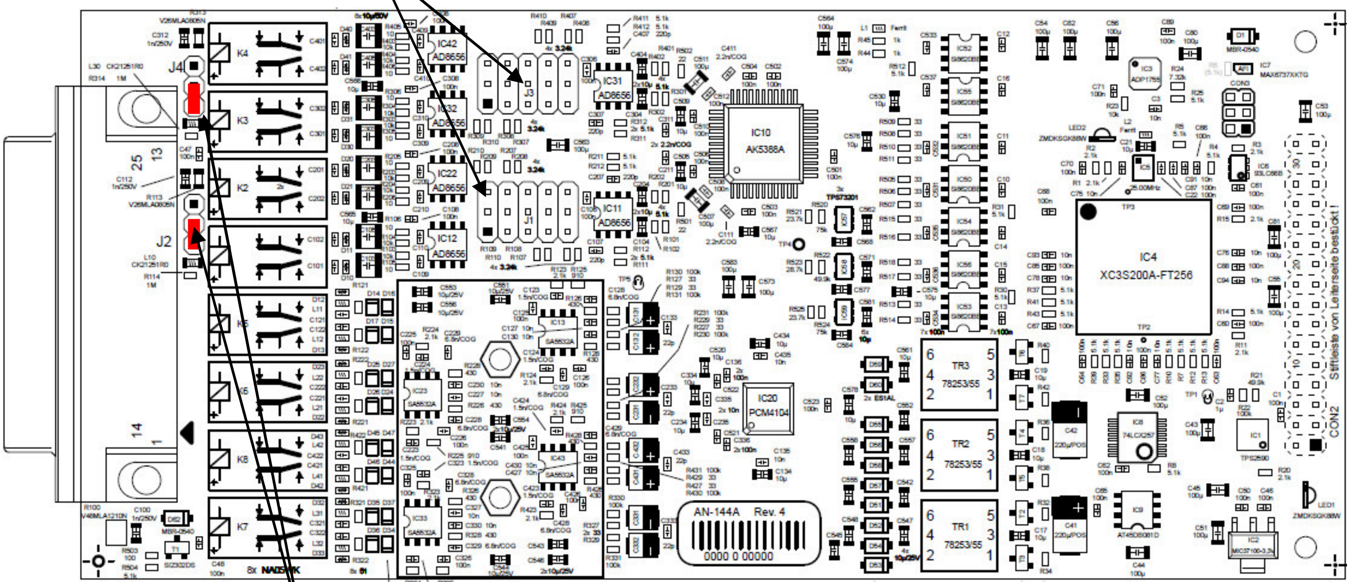
**Polarity**
[Normal / Inverted]

**Outputs**

**Analog Output Reference (dBu)**
**(level for digital 0dBFS)**
[0 … 15.0 … 24.0]

**Polarity**
[Normal / Inverted]

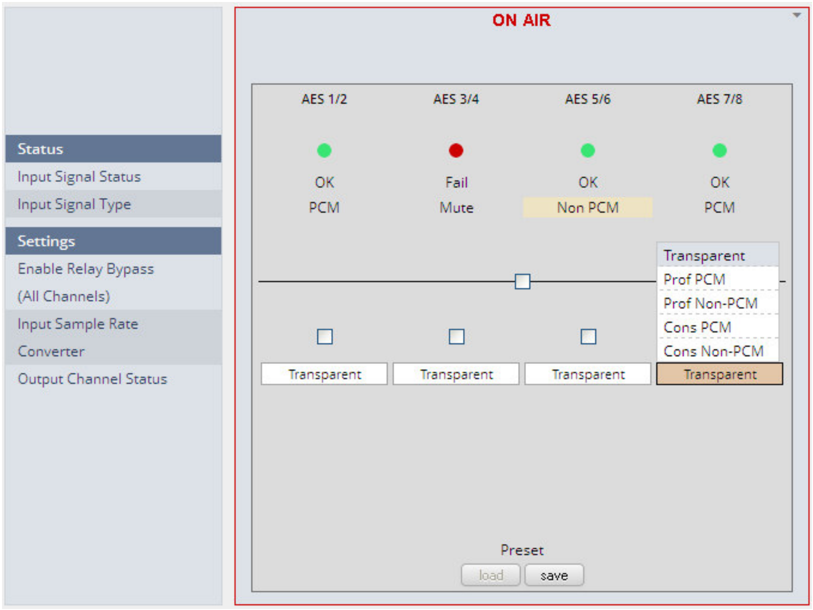
**Important Note!** The maximum input level for the card must be set by two jumper blocks labeled CH1/2 and CH3/4 ● (see legend printed on the PCB).



The two jumpers ● above may lift the shield pins for CH1/1 and CH3/4 from analog ground to HF coupling.

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:



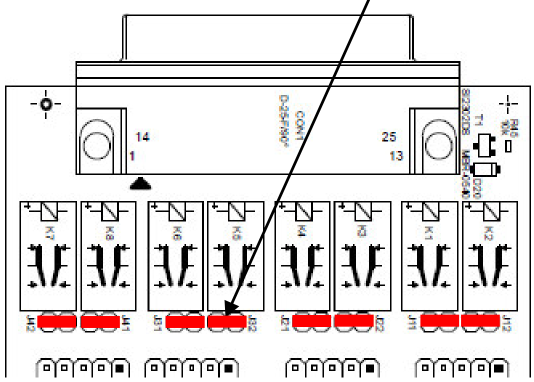

**Status**

| | |
|---|---|
| **Input Signal Status** | green [OK] / red [Fail] |
| **Input Signal Type** | [Mute / PCM / Non PCM]} |

**Settings**

| | |
|---|---|
| **Enable Relay Bypass (All Channels)** | [ON / OFF]<br>Power fail bypass relay that may be activated by the GUI |
| **Input Sample Rate Converter** | [ON / OFF] |
| **Output Channel Status** | [Transparent / Prof PCM / Prof Non-PCM / Cons PCM / Cons Non-PCM]<br>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). |

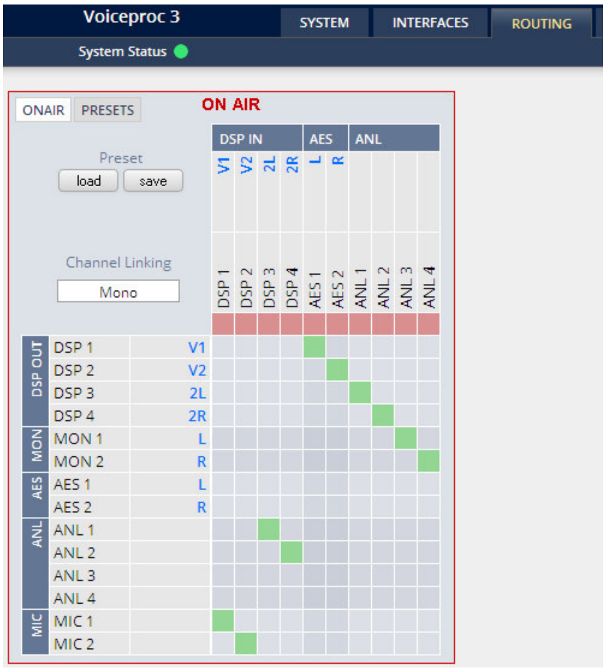
**Important note!** The AES relay bypass circuit of the I/Os is activated on the option board. It is possible to deactivate it if necessary. You must open the cover plate of the **D*AP4 VAP** unit and locate the jumper shown in the schematic below. You must remove the jumpers ● to de-activate the AES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



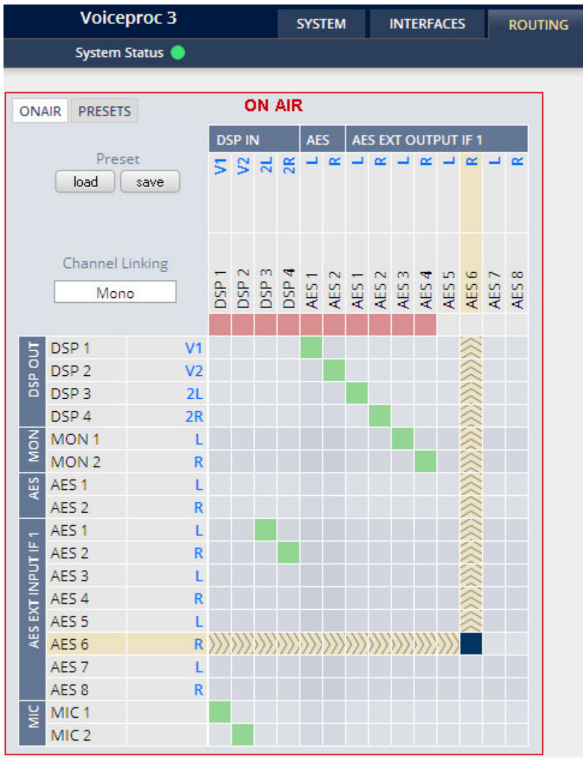
Setup GUI – **ROUTING**

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

Example for an analog interface board:




example for an AES interface board:




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

Top / horizontal (column headlines) = signal destinations

**DSP** [DSP 1 … DSP 4]
The DSP inputs carrying the signal type labels **V1**, **V2**, **2L**, **2R** where **V1/V2** indicate the voice channel input and **2L/2R** the program path input (see AUDIO PROCESSOR > Overview).

**AES** [AES 1 / AES 2]
The AES output of the device.

Left hand / vertical (line headlines) = signal sources

**MIC** [MIC 1 / MIC 2]
The inputs of the optional mic interface.

**AES** [AES 1 / AES2]
The AES input.

**MON** [MON 1 / MON 2]
The audio processor (DSP) has an independent monitor output.
It may be connected with the internal processing blocks.
(see AUDIO PROCESSOR > Overview)

**DSP** [DSP 1 … DSP4]
The DSP outputs carrying the signal type labels **V1, V2, 2L**, **2R** where **V1/V2** indicate the voice channel and **2L/2R** the program output.

The routing example on the left hand side shows both mic inputs connected to the voice channel inputs **V1**/**V2** of the DSP. The analog input ANL1 / ANL2 is connected to the DSP program input while the DSP voice channel outputs **V1**/**V2** are connected to the AES output and the DSP program output **2L**/**2R** and the DSP monitor output **L**/**R** are connected to the analog outputs ANL 1 … ANL 4.

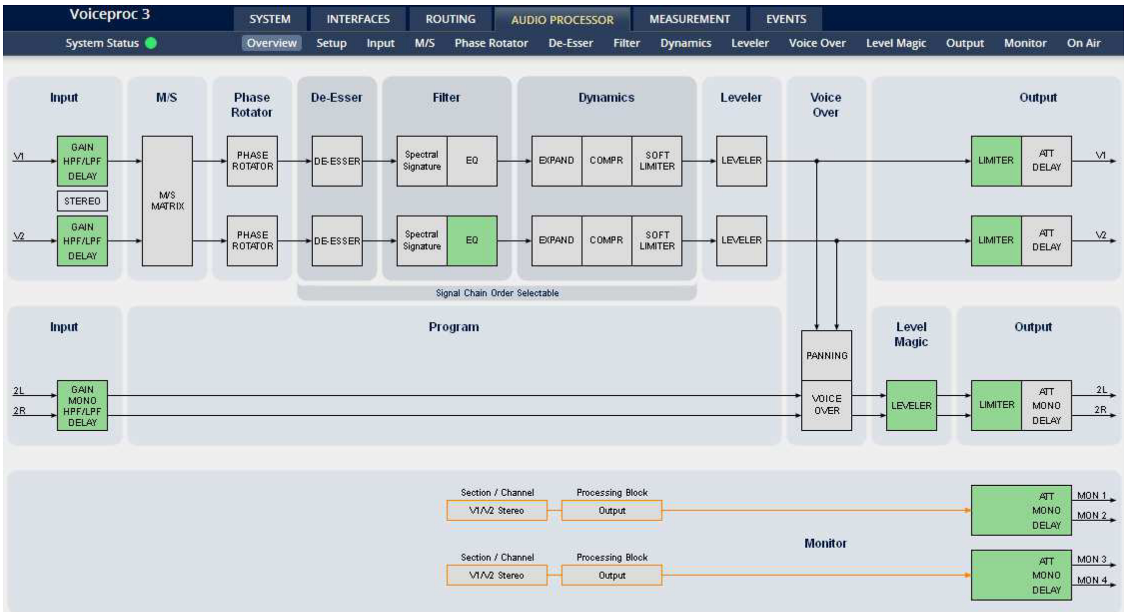
**Important Note!** If a different optional interface board is installed the matrix will be expanded by the pre-defined number of I/Os with their labels:

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

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

## Setup GUI – AUDIO PROCESSOR - **Overview**

The overview shows the actual signal routing of the audio processor blocks, rendered by the DSPs.

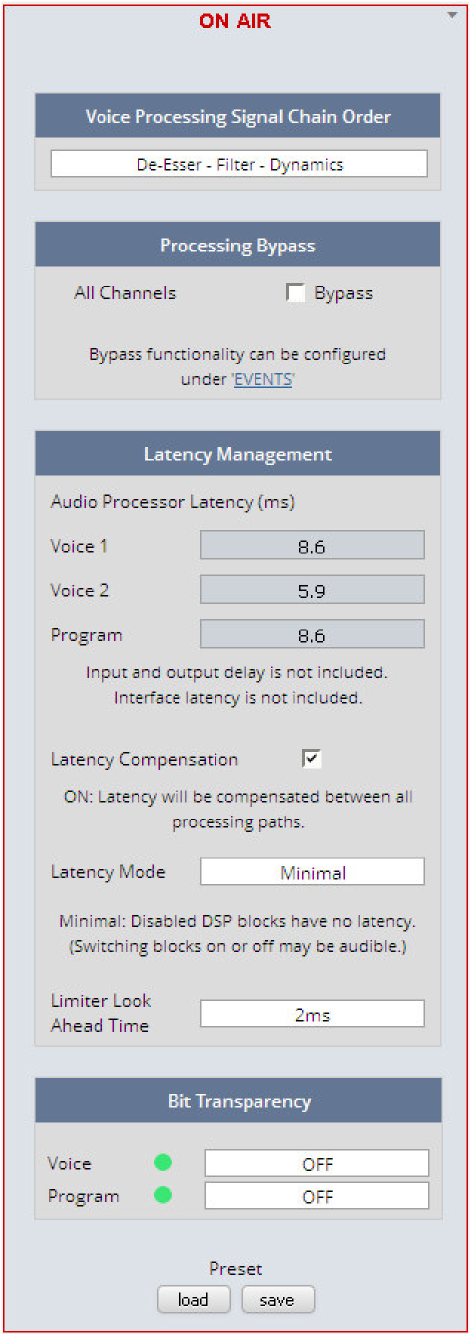



The processing blocks in use, which may be activated from their individual setup panes, will be indicated in green. I.e. blocks shown in grey are not activated by the user.

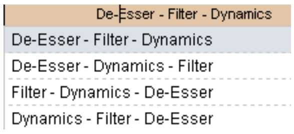
The order of the blocks of the voice channels depends on the setup of the audio processor (see next page).

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

Setup GUI – AUDIO PROCESSOR - **Setup**




**Voice Processing Signal Chain Order**




The order of the processing blocks can be re-arranged:
De-Esser, Filter, Dynamic
Individually to match your preference.

**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.

**Audio Processor Latency (ms)**

**Voice 1**

**Voice 2**

**Program**

**Latency Compensation**

[ON / OFF]
"vertical" compensation to match two channels

**Latency Mode**

[Minimal, Nominal]
"horizontal" compensation for one channel**.** 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.

**Limiter Look Ahead Time**

[1ms, 2ms]
Set the True Peak Limiters Look Ahead Time to 1ms to reduce audio latency. Full True Peak limiting is guaranteed with both settings. The audio quality in most cases will not be affected. However, when in doubt leave it at 2ms for assured maximum sonic performance.

**Bit Transparency**

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

System Latency:

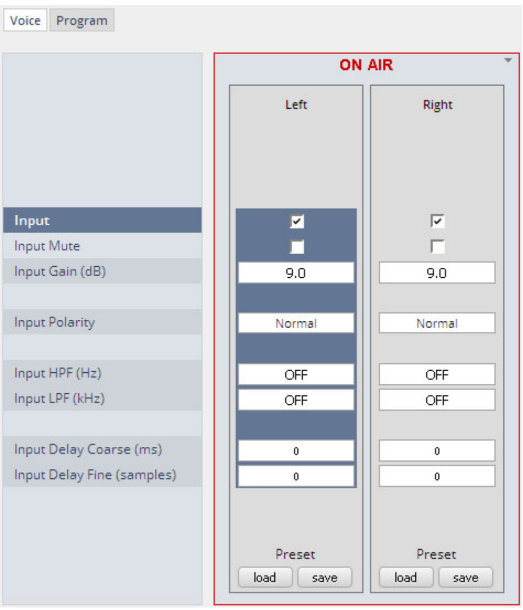
| | 44.100kHz | 48.000kHz | 88.200kHz | 96.000kHz |
|---|---|---|---|---|
| Base Latency | | | | |
| AES IN to AES OUT | 3,33 | 3,06 | 2,12 | 2,04 |
| Mic IN to AES OUT | 4,25 | 3,9 | 2,59 | 2,44 |
| Additional Latencies | | | | |
| Spectral Signature | 2,9 | 2,66 | 2,9 | 2,66 |
| Dynamics Look Ahead | 2 | 2 | 2 | 2 |
| Limiter Look Ahead (option) | 1 | 1 | 1 | 1 |

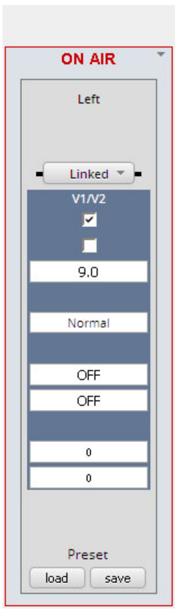
Base latency consists of 1 ms Limiter Look Ahead Time and all system inherent processing and input/output delays.
Activating sample rate converters will add additional latency (< 1ms).

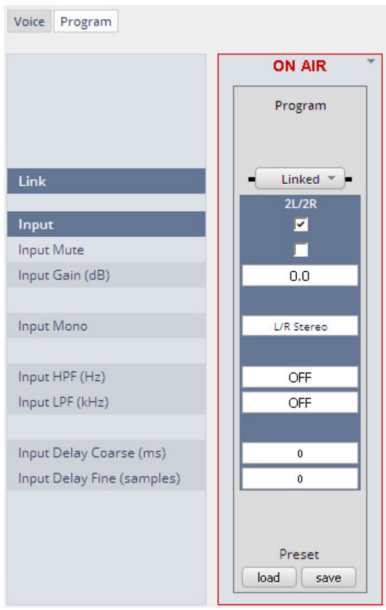
## Setup GUI – AUDIO PROCESSOR – **Input**

You may set the input conditions for both signal paths - voice (V1 / V2) and program (2L / 2R) via the page embedded tab sheets. The layout of the embedded "Voice" pane differs depending on the general setup (see SYSTEM > Setup > Voice Channel Mode):

Voice Channel Mode = "2 x Mono" = "Stereo" (unlinked) Program pane:




**Link** [Unlinked, Linked]
This function will only be available if the voice channel is set for "Stereo" (see SYSTEM > Setup > Voice Channel Mode).

**Input** [Enable / Disable]
Enables or disables the input section

**Mute** [ON / OFF]

**Input Gain (dB)** [-80.0 … 0.0 … 20.0]

**Polarity** (voice input) [Normal / Inverted]

**Mono** (program input) [L/R Stereo / L+R Mono / L/L Mono / R/R Mono]

**Input HPF (Hz)** [OFF / 20 / 40 / 80 / 120]

**Input LPF (Hz)** [OFF / 15 / 20 / 22]

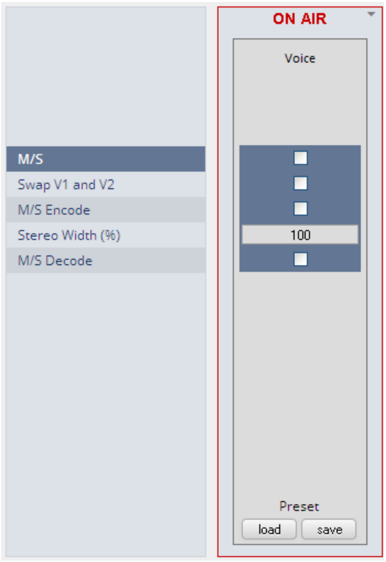
**Input Delay Coarse (ms)** [0.0 … 2000.0]

**Input Delay Fine (samples)** [0 … 2000]

Setup GUI – AUDIO PROCESSOR – **M/S**

The **M/S** block allows for transformation of the voice channel signals from L/R to M/S and vice versa (if you have a M/S mic connected).

The **M** (mid) and **S** (side) signals may be processed to change the stereo width from 0% (mono) to100% (stereo) to 200% (excess width). If you want to process a L/R stereophonic signal you must first encode it to M/S and back to L/R after width correction.




**M/S** [ON / OFF]
Turns the M/S block ON and OFF

**Swap V1 and V2** [ON / OFF]

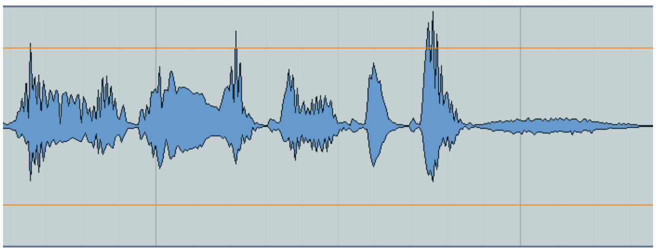
**M/S Encode** [ON / OFF]

**Stereo Width (%)** [0 … 100 … 200]

**M/S Decode** [ON / OFF]

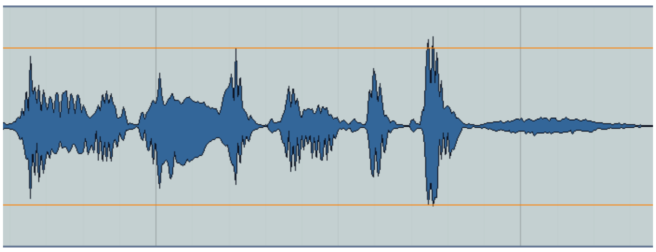
Setup GUI – AUDIO PROCESSOR – **PHASE ROTATOR**

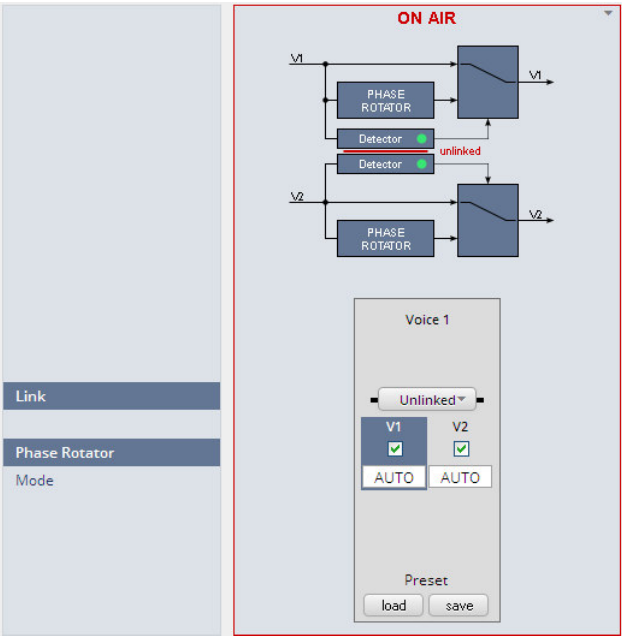
When working with human voice, one regularly experiences issues with imbalanced waveforms. Imbalanced in this context means that the positive or negative half of the alternating signal carries more power than the other.
The problematic result of this type of imbalance is unnecessarily applied dynamics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



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

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



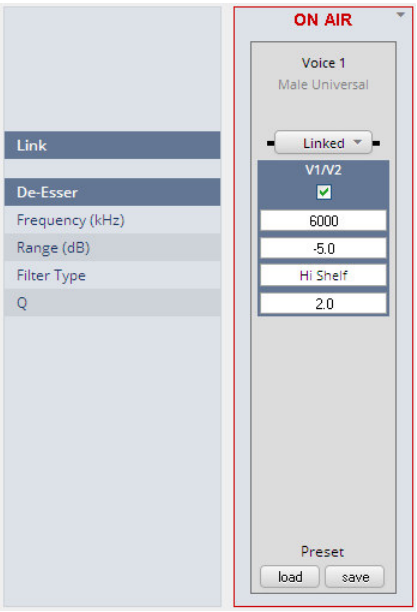



Each microphone channel has an independent Phase Rotator.

Here the display for stereo operation but unlinked

| | |
|---|---|
| **Link** | [Unlinked / Linked]<br>For stereo operation you may link the setup parameters of both voice channels. |
| **Phase Rotator** | [ON / OFF] |
| **Mode** | [OFF / ON / AUTO]<br>OFF<br>System is inactive<br>ON<br>System always applies phase wrapping<br>AUTO<br>Unbalanced waveforms are automatically detected and phase wrapping is applied only if necessary. |

Setup GUI – AUDIO PROCESSOR – **De-Esser**

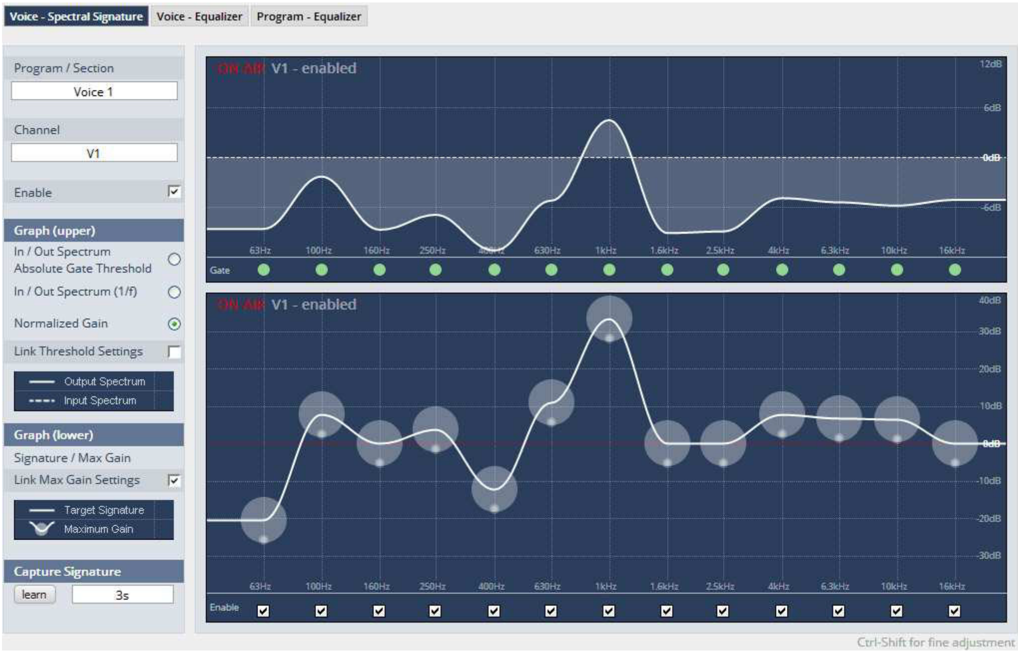



| | |
|---|---|
| **Link** | [Unlinked, Linked]<br>For stereo operation you may link the setup parameters for both voice channels. |
| **De-Esser** | [ON / OFF] |
| **Frequency (Hz)** | [1000 … 3000 … 16000] |
| **Range (dB)** | [-20.0 … 0.0] |
| **Filter Type** | [Peak / High Shelf] |
| **Q** | [0.4 … 1.0 … 8.0] |

**Important Note!** For the following explanations we assume that the **D*AP4 VAP** is set up for 2 x Mono operation mode (see SYSTEM > Setup > Voice Channel Mode). I.e. there are always two voice channels displayed and no link option.

Setup GUI – AUDIO PROCESSOR – Filter **– Voice - 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** | ["Left" / "Right" / Preset]<br>For stereo operation the name assigned to "Voice 1" will be displayed.<br>Selects the source for which Spectral Signature will be displayed.<br>This selection depends on the Voice Channel Mode<br>(see SYSTEM > Setup > Voice Channel Mode).<br>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** | Applicable only if SYSTEM > Setup > Voice Channel Mode = Stereo.<br>You can select which channel must actually be displayed (see also LI |
| **Enable** | [ON / OFF]<br>Enables / disables Spectral Signature for the selected section.<br>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.<br>This window can be used in three different ways: |
| **Input / Output Spectrum**<br>**Absolute Gate Threshold** | <br>[alternative selection]<br>The spectrum is shown in absolute values (related to digital full scale).<br>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 sphere.<br>The **Gate** soft LED row at the bottom indicates whether the absolute or relative gate of the band is closed (yellow) or open (green).<br>A gray LED indicates that the band is switched out. |
| **In / Out Spectrum (1/f)** | [alternative selection]<br>1/f (3dB/octave) weighted spectrum. |

**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. 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).

**Graph (lower)**

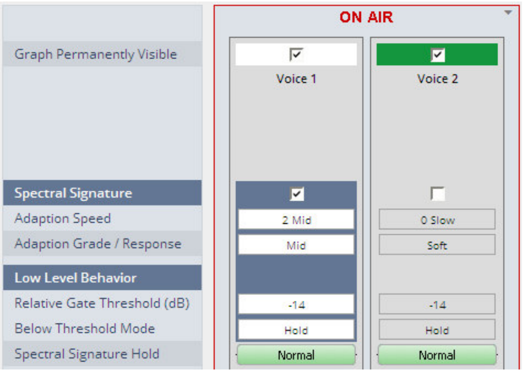
**Signature / Max Gain** [0 … 12]
Spectral Signature does not work with an absolute level reference. Its frequency response is based on level differences between bands only. Thus a signature is only represented on a relative graph showing the level positions related to the neighboring bands. In consequence, 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. 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), Spectral Signature will 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 voice channel selected for display (see upper display).
White color represents the selected voice channel (Voice 1 for example) while the other channel shows dark green.

**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'.

**Low Level Behavior**

**Relative Gate Threshold (dB)**

[-10 …. -14 … -20 / OFF]
To prevent a band from amplifying noise (especially hum), a relative gate can be set. If the energy within one band is lower than this gate, no amplification will take place. This is especially useful, when mixed content with highly varying frequency response is processed (for example a radio station output with alternating presenter voice and music).

**Below Threshold Mode**

[Release / Hold]

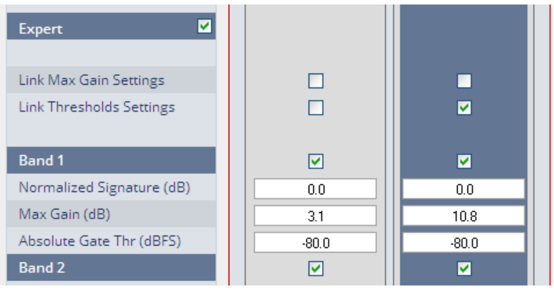
In continuous operation the 'Below Threshold Mode' should remain in 'release'. In this case the dynamic gain slowly returns to its neutral state in case of signal absence. In this mode a returning signal would start a new processing period with its lead in attack time. This can be undesired, especially in production applications where transport operations introduce unnatural gaps. In those cases setting the 'Below Threshold Mode' to 'hold' will pause the dynamic processing at the last value until the signal returns. Returning signals are treated just like continuous signals. This function has some similarities to the 'Freeze Level' but works with a different designation as it is meant to keep processing fluent over signal loss.

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

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

**Link Max Gain Settings** [ON / OFF]

**Link Threshold Settings** [ON / OFF]




**Band 1** [ON / OFF]

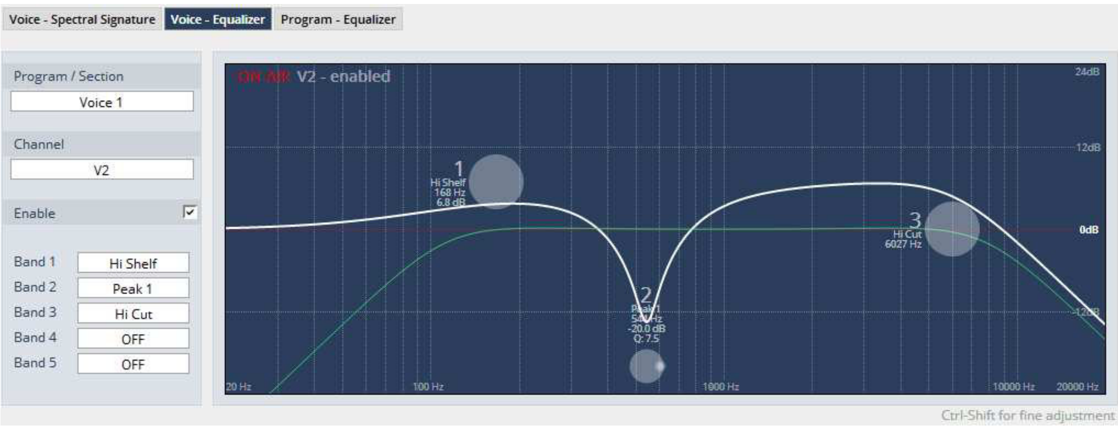
**Normalized Signature level** [-40.0 … 0 … 40.0]

**Max Gain** [0.0 … 12.0

**Absolute Gate Threshold** [-84.0 … 0.0]

**Band 2 … 16** similar parameters as Band 1

Setup GUI – AUDIO PROCESSOR – Filter **– Voice - Equalizer**



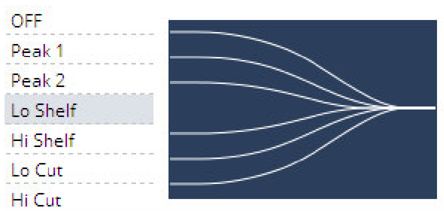

The graphical EQ offers 5 bands. The characteristic of each band can be setup either left hand side of the graph or alternatively for each band further below.

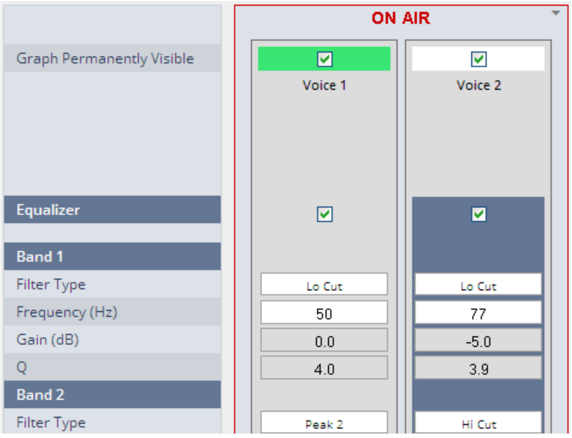
**Program / Section** ["Voice 1" / "Voice 2" / Preset] or ["Voice 1" / Preset]
Selects the source for which the curve will be displayed.
This selection depends on the Voice Channel Mode (see SYSTEM > Setup) and whether or not the channels are linked for stereo operation.

**Channel** [V1 / V2]
select which channel is under control
Not applicable if SYSTEM > Setup > Voice Channel Mode = Stereo.

**Enable** [ON / OFF]
Same function as **<Equalizer>** further below

**Band 1 … 5** the filter characteristic will be selected by this pop up:






**Graph Permanently Visible** The color code of the column headers will change depending on the selected voice channel. White color represents the actual selected channel (Voice 2 for example) while the other channel shows light green.

**Equalizer** [ON / OFF]

**Band 1**

**Filter Type** will be selected by the pop up Above.

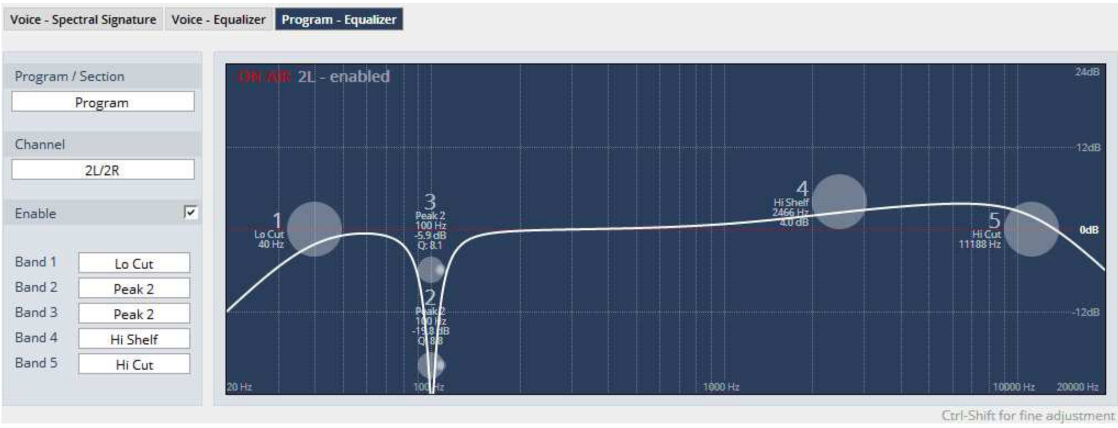
**Frequency (Hz)** [20 … 20000]

**Gain (dB)** [-20.0 … 20.0]

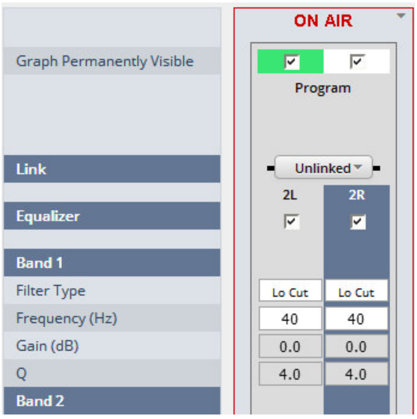
**Q** [0.4 … 10.0]

**Band 2 … 5** same parameter set as Band 1.

Setup GUI – AUDIO PROCESSOR – Filter **– Program - Equalizer**




The parametric EQ for the program path offers 5 bands in a similar way as for the voice EQ. Pls. refer to the explanation above. **Program** / **Selection** has no option because this is for the **"Program"** path. **Channel** depends on the Link mode of the program path. If linked **2L/2R** appears in that field.




**Graph Permanently Visible** The color code of the column headers will change depending on the selected voice channel. White color represents the actual selected channel (2R for example) while the other channel shows light green.

**Equalizer** [ON / OFF]

**Band 1**

**Filter Type** will be selected by the pop up Above.

**Frequency (Hz)** [20 … 20000]

**Gain (dB)** [-20.0 … 20.0]

**Q** [0.4 … 10.0]

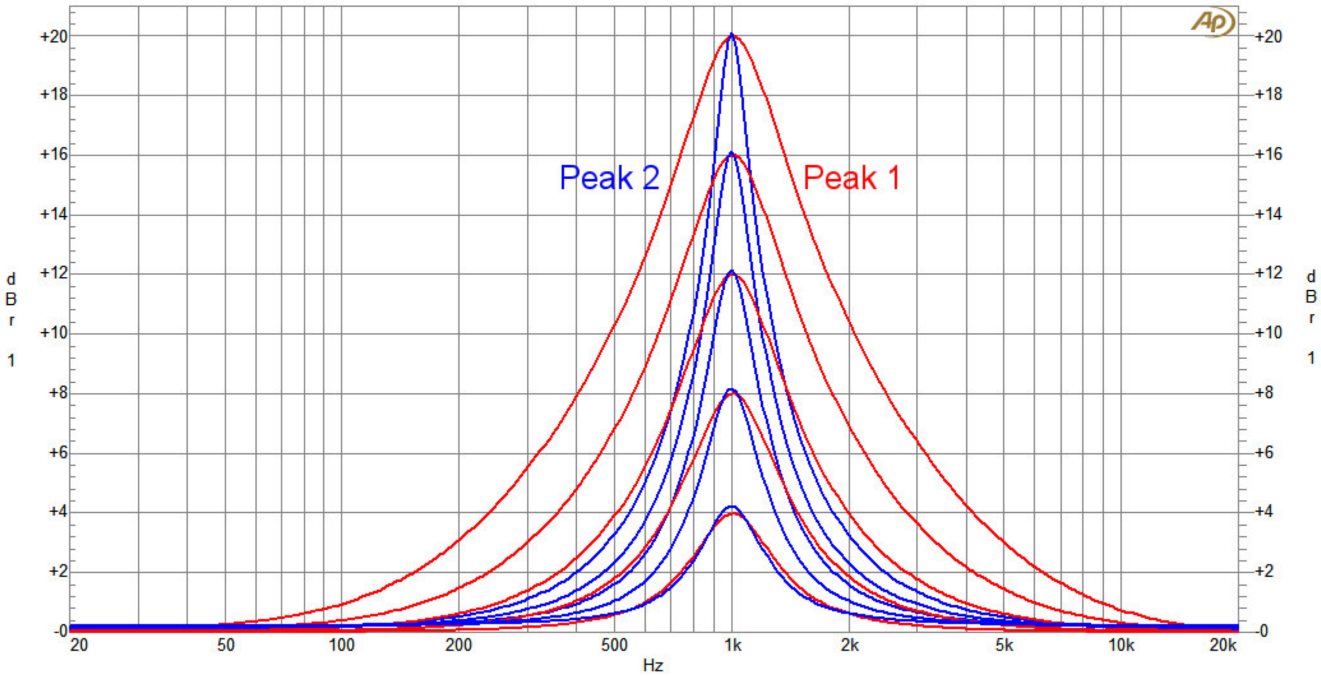
**Band 2 … 5** same parameter set as Band 1.

**Important Note!** For numeric input double click into the parameter field. You must use the period as a decimal separator. When done press <ENTER>.
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.

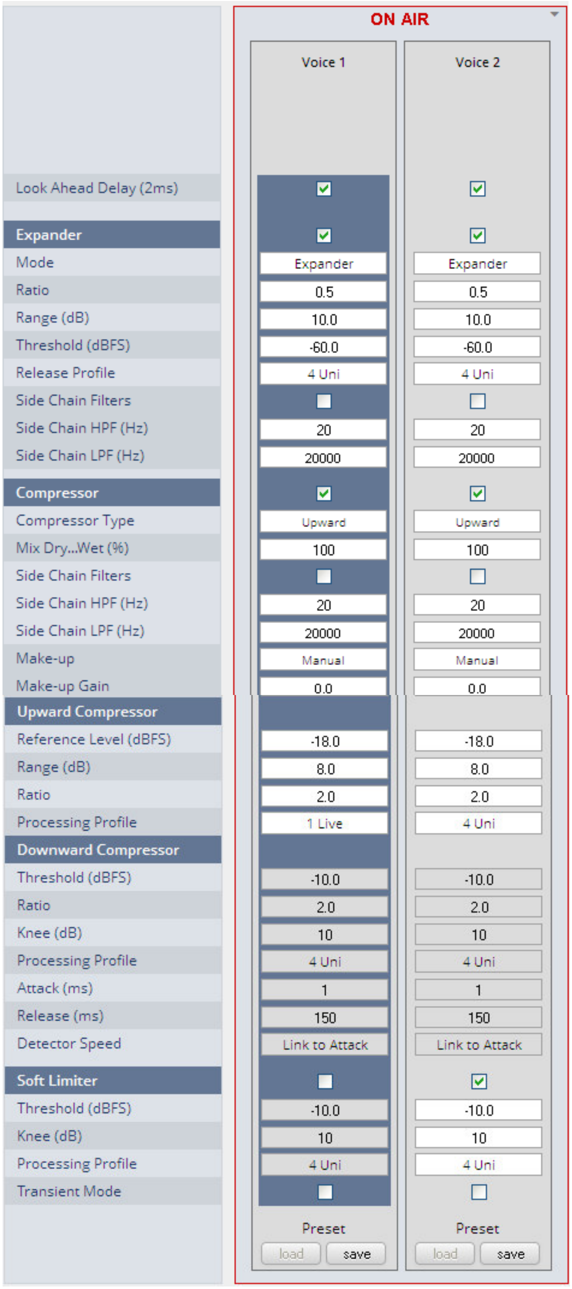
The **EQs** offer two different peak modes:

**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.



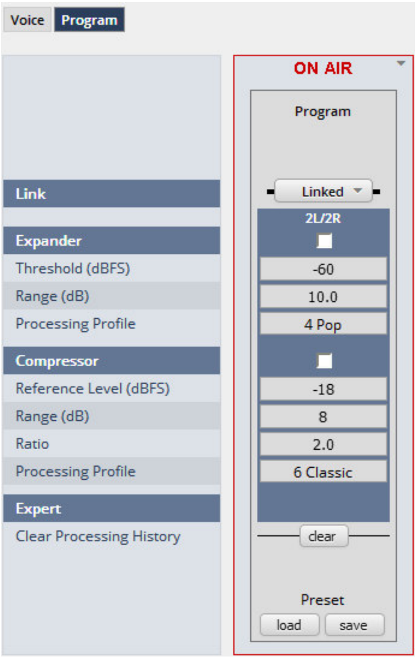
Setup GUI – AUDIO PROCESSOR – Dynamics - **Voice**




| | |
|---|---|
| **Look Ahead Delay (2ms)** | [ON / OFF] |
| **Expander** | [ON / OFF] |
| **Mode** | [Expander / Gate] |
| **Ratio** | [0.0 … 0.9] |
| **Range (dB)** | [0.0 … 40.0] |
| **Threshold (dBFS)** | [-80.0 … -10.0] |
| **Release Profile** | [0 / 1 Live / 2 Speech / 3 Pop / 4 Uni / 5 / 6 Classic / 7 / 8 / 9] |
| **Side Chain Filters** | [ON / OFF] |
| **Side Chain HPF (Hz)** | [1 … 5000] |
| **Side Chain LPF (Hz)** | [1000 … 20000] |
| **Compressor** | [ON / OFF] |
| **Compressor Type** | [Upward / Downward] |
| **Mix Dry Wet (%)** | [0 … 100] |
| **Side Chain Filters** | [ON / OFF] |
| **Side Chain HPF (Hz)** | [1 … 5000] |
| **Side Chain LPF (Hz)** | [1000 … 20000] |
| **Make-up** | [Manual / Auto] |
| **Make-up Gain** | [-40.0 … 40.0] |
| **Upward Compressor** | |
| **Reference Level (dBFS)** | [-60.0 … 0.0] |
| **Range (+/- dB)** | [0.0 … 20.0] |
| **Ratio** | [1.1 … 8.0] |
| **Processing Profile** | [0 / 1 Live / 2 Speech / 3 Pop / 4 Uni 5 / 6 Classic / 7 / 8 / 9] |
| **Downward Compressor** | |
| **Threshold (dBFS)** | [-60.0 … 0.0] |
| **Ratio** | [1.1 … 8.0] |
| **Knee (dB)** | [0 … 20] |
| **Processing Profile** | [0 / 1 Live / 2 Speech / 3 Pop / 4 Uni / 5 / 6 Classic / 7 / 8 / 9 / Manual] |
| **Attack (ms)** | [1 … 100] |
| **Release (ms)** | [10 … 1000] |
| **Detector Speed** | [Peak / RMS / Link to Attack] |
| **Soft Limiter** | [ON / OFF] |
| **Threshold (dBFS)** | [-60.0 … 0.0] |
| **Knee (dB)** | [0 … 20] |
| **Processing Profile** | [0 / 1 Live / 2 Speech / 3 Pop / 4 Uni / 5 / 6 Classic / 7 / 8 / 9] |
| **Transient Mode** | [ON / OFF] |

Setup GUI – AUDIO PROCESSOR – Dynamics - **Program**

Compared to the voice channel, the program channel has less parameters:




| | |
|---|---|
| **Link** | [Linked / Unlinked] |
| **Expander** | [ON / OFF] |
| **Threshold (dBFS)** | [-80.0 … -10.0] |
| **Range (dB)** | [0.0 … 40.0] |
| **Processing Profile** | [0 / 1 / 2 Live / 3 Speech / 4 Pop / 5 Uni / 6 / 7 Classic / 8 / 9] |
| **Compressor** | [ON / OFF] |
| **Reference Level (dBFS)** | [-60.0 … 0.0] |
| **Range (dB)** | [0 … 8 … 20] |
| **Ratio** | [1.1 … 2.0 … 4.0] |
| **Processing Profile** | [0 / 1 Live / 2 Speech / 3 Pop / 4 Uni / 5 / 6 Classic / 7 / 8 / 9] |
| **Expert** | |
| **Clear Processing History** | <clear> |

The dynamics section includes some technical features that are explained below:

**Look Ahead Delay**

Traditionally, all Junger Audio dynamics feature a look ahead delay (LAD) of 2 milliseconds. This allows the system to process fast transients without missing even the steepest peaks. As the live audio signal must be delayed for two milliseconds, this lag needs to be considered when measuring overall latency. The LAD can be disabled, if the advantage of the resulting lower latency prevails. Please keep in mind that disabling LAD in one channel causes an offset between both channels. This can be compensated for by the delay compensation parameter in the Audio Processor Setup, but then again the latency advantage disappears.

**Side Chain Filters**

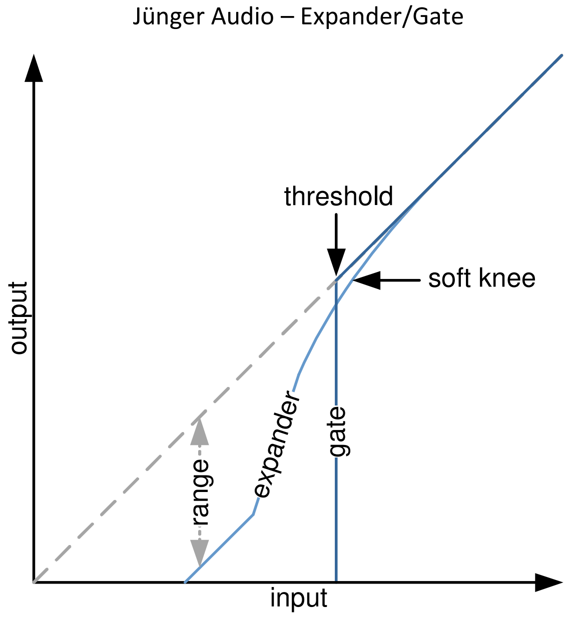
Some of the processing modules (De-Esser, Compressor, Expander/Gate) feature Side Chain Filters to shape the audio signal that feeds their detection system. It consists of independent high and low pass filters with tunable cutoff frequencies. Those filters are not audible within the actual signal chain. Its purpose is to make the detection more or less sensitive to certain frequency ranges. As an example to understand the benefits, with live speech the amount of pop noise coming from the microphone varies highly. As it is not advisable to generally reduce the amount of bass to preserve the voice character, it is necessary to at least keep the bass thumps from forcing the Compressor into heavy gain reduction. By reducing the bass in the Compressor side chain, one can keep it from overreacting (it does not 'know' of the thump, thus is not reacting to it) while preserving the original frequency response.

**Expander**

**Mode** — The Expander can be switched to either working as an Expander or a Gate. Both modes differ in two parameters:

**Mode=Gate** — Fixed reduction ratio of infinite to one. All signals below threshold are muted. No range available. Hard knee response at threshold.

**Mode=Expander** — Selectable reduction ratio of 0:1 up to 0.9:1 with a selectable maximum reduction of down to -40dB. Soft knee response with a transition range of 6dB above and below threshold




**Ratio** — Expansion ratio from 0:1 (heavy reduction) up to 0.9:1 (slight reduction). A ratio of 0.5:1 means that an input level of 1dB below threshold will result in an output level of 2dB below threshold. In the same way an input level of 4dB below threshold results in an output level of 8dB below threshold and so on.

**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.

**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.

**Release profile** — The release profile controls the timing of the closing of the Gate/Expander. Release profile 0 is a very fast profile and even short gaps or signal intermissions lead to gain reduction. At the other end of the scale, 9 is a very slow profile with a relaxed handling of gaps and low level periods. All profiles feature the same super fast opening when the signal returns above threshold.

**Compressor (general parameters)**

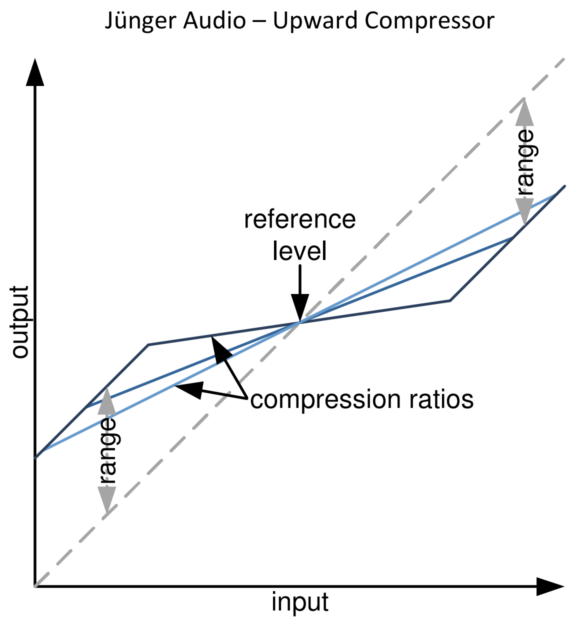
**Compressor Type** — The compressor features two different approaches to dynamic processing. In Upward mode all signals below reference level are amplified according to the ratio and range settings, all signals above reference level are reduced in the same way. This is the 'classic' approach of earlier Junger Audio compressor designs. The Downward mode is the more common way of dynamic range compression. Here all signals above threshold are reduced according to the ratio while all signals below threshold remain untouched.

**Mix Dry...Wet** — In most settings, the full signal is fed to a compressor to achieve a certain level of gain reduction. Sometimes it is useful to add a portion of the original, uncompressed signal to the output to restore some micro dynamics. This technique is called 'parallel compression'. The ratio of dry (unprocessed) and wet (compressed) signal can be dialed in with this Mix parameter.

**Make-Up**

To set up the desired output level of the compressor, Make-Up Gain (or attenuation) needs to be applied. This is a simple and static output level adjustment without any dynamic content. In Auto mode the amount of Make-up Gain is automatically determined depending on the threshold and ratio settings. When set to manual, its value can be set in steps of 0.1dB.

**Upward Compressor**




**Reference Level**

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**

This defines the range over which dynamic compression is applied as defined by the ratio setting. Signals outside 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:

**Downward compression ratio**

An example ratio of 2:1 means that an input level of 4dB above threshold will result in an output level of 2dB above threshold. In the same way an input level of 8dB above threshold results in an output level of 4dB above threshold and so on.

**Upward compression ratio**

A ratio of 2:1 means that an input level of 4dB below reference level will result in an output level of 2dB below reference level. In the same way an input level of 8dB below reference level results in an output level of 4dB below reference level and so on.

**Processing Profile**

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. For profile "Manual" one may set "Attack", "Release" and "Detection Speed".

**Downward Compressor**

**Threshold**

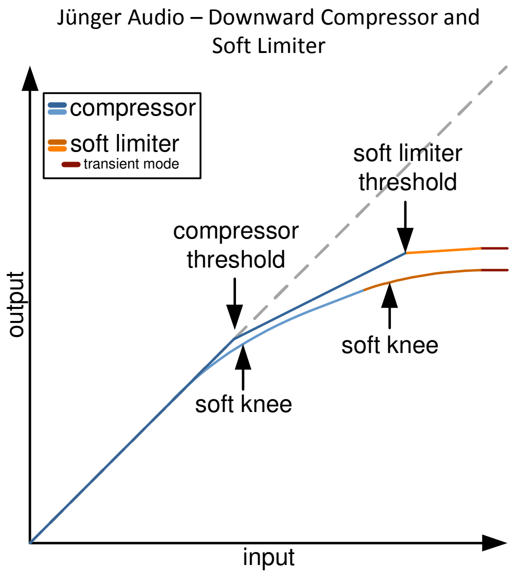
Signals above threshold are processed, signals below pass unaffected. Please be aware that this is only true when Knee is set to '0'.

**Ratio**

Please see ratio description in Upward Compressor section.

**Knee**

The Knee parameter allows the user to decide whether the transition from linear to processed happens immediately at threshold (so called hard knee) or if a transition range (soft knee) is applied in which the ratio is steadily raised from 1:1 at the lower knee end up to the defined ratio at the upper range (see picture).




**Processing Profile** 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. Alternatively the timings of Attack and Release (return to neutral) can be set up manually. In this case they are not adaptive to the input.

**Attack** Settling time after exceeding threshold. Defined as the time period to achieve 63% of full reduction according to signal level.

**Release** Time constant for the process to return to zero after signal fallback below threshold. The system returns with 8.6dB per time constant.

**Detector Speed** The internal level detector of the compressor is configurable with three settings [Peak / RMS / Link To Attack]:

**Peak** the system detects every single signal peak and reacts accordingly with fast and appropriate gain reduction.

**RMS** Instead of 'riding on peaks' the detector analyses the energy of the signal and reacts with a more moderate and more 'musical' reduction.

**Link To Attack** The length of analysis is coupled to the setting of Attack. With a slower attack the RMS analysis is based on a longer portion of the signal. On the other hand a fast Attack setting brings the analysis very close to the Peak behavior.

**Soft Limiter** When working in a non-loudness based audio environment, it became common practice to use the output limiter (typically set to -9dBFS) as a creative tool for compression. In a modern, loudness based studio this option got lost. As a substitute, the new Soft Limiter brings back the option of using a dedicated sample peak limiter with soft knee characteristics for the microphone processing chain.

**Threshold** Signals above threshold are processed, signals below pass unaffected. Please be aware that this is only true when Knee is set to '0'.

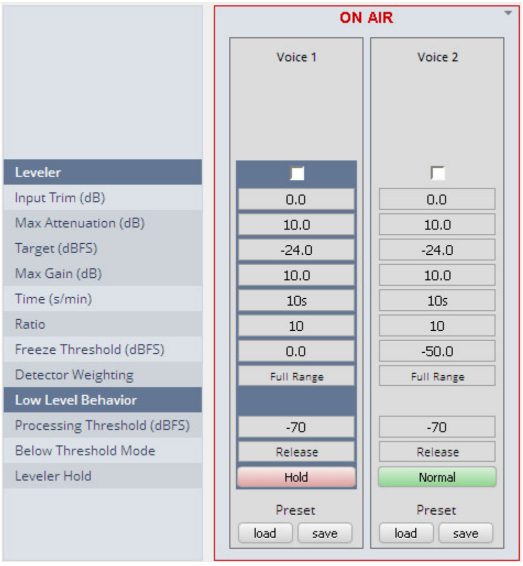
**Knee** The Knee parameter allows the user to decide whether the transition from linear to processed happens immediately at threshold (so called hard knee) or if a transition range (soft knee) is applied in which the ratio is steadily raised from 1:1 at the lower knee end up to the defined ratio at the upper range (see picture).

**Processing Profile** The timing characteristics of the Soft Limiter 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 release timings. The names behind some of the numbers may help to easily find adequate values to your content.

**Transient Mode** With the transient mode enabled the response curve of the Soft Limiter is reduced down to infinity to one. This is useful in settings where a wide knee may not have run out fully before full scale is reached.

Setup GUI – AUDIO PROCESSOR – Leveler - **Voice**

The new Leveler is optimized for single channels. Its main purpose is to balance an already processed signal to a certain target level.



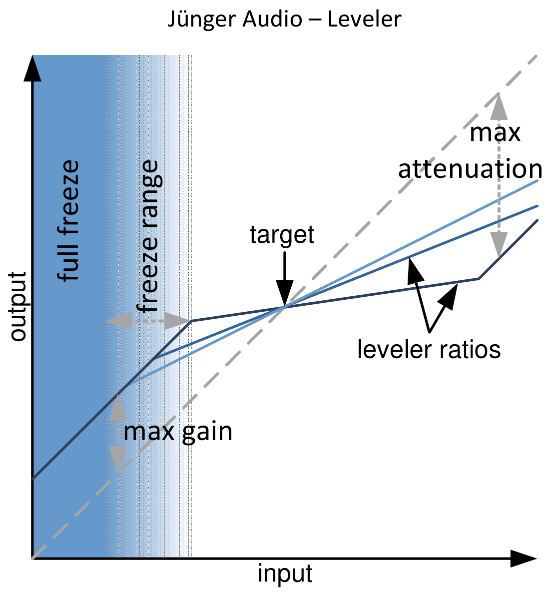

**Leveler** [ON / OFF]

**Input Trim (dB)** [-80.0 … 0.0 …. 20.0]
Simple input level trim to prevent the Leveler from reducing static offset. Hint: To use the Input Trim without the Leveler, simply reduce its Max Attenuation and Max Gain down to 0.

**Max Attenuation (dB)** [0.0 … 10.0 … 40.0]
Defines the maximum attenuation or damping of the Leveler process for signals that exceed the target level.

**Target (dBFS)** [-50.0 … -24.0 … 0.0]
This is the balance point of the Leveler. Signals that exceed the target are reduced, signals below target are amplified to bring the overall level to this center of gravity.

**Max Gain (dB)** [0.0 … 10.0 … 40.0]
Defines the maximum amplification of the Leveler process for signals that are below the target level.




**Time (s/min)** [1s … 30s … 2min]
This defines the timing for the Leveler to reach target. Of course this is not an absolute value as it depends on the input level, signal structure, ratio and necessary amount of gain change.

**Ratio** [1 … 10 … 40]
As opposed to a classic compressor, the Leveler ratio is normally fixed to infinity to one. With this Leveler you can bring down the ratio to a much lower value to achieve a more relaxed compensation instead of heavy steering towards target. Setting it to 1:1 disables the leveling process.

**Freeze Threshold (dBFS)** [-60.0 … -50.0 … 0.0]

**Detector Weighting** [Full Range / Proximity / Loudness]
The Leveler features a Side Chain Filter with special characteristics to adapt the leveling process to three major applications.

**Full Range** The Side Chain is not filtered and the Leveler is running on full bandwidth detection.

**Proximity** The Side Chain uses a low shelf filter to compensate for microphone proximity effects.

**Loudness** In modern broadcast production environments the final product is played out in accordance with current loudness standards. Those standards and recommended practices always refer to the output signal and do not consider the condition of the source channels. This is correct, but never the less, it can be very useful to consider loudness for these single channels. With this loudness filtering you can bring the output to a consistent level, based on modern loudness recommendations. The output will integrate seamlessly into your loudness normalized product. In many situations, no additional loudness correction is necessary. This approach is compatible with all international loudness recommendations.

Technical remark: K-Filtering is used as described in ITU-R BS 1770-3.

**Low Level Behavior**

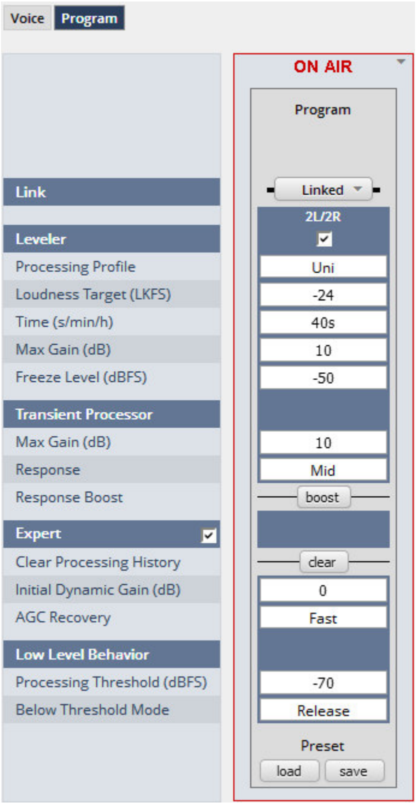
**Processing Threshold** [-80 … -70 … -20]

**Below Threshold Mode** [Release / Hold]
In continuous operation the 'Below Threshold Mode' should remain in 'Release'. In this case the dynamic gain slowly returns to its neutral state in case of signal absence. In this mode a returning signal would start a new processing period with its lead in attack time. This can be undesired. In those cases setting the 'Below Threshold Mode' to 'hold' will pause the dynamic processing at the last value until the signal returns. Returning signals are treated just like continuous signals. This function has some similarities to the 'Freeze Level' but works with a different designation as it is meant to keep processing fluent over signal loss.

**Leveler Hold** [Normal / Hold]
Leveler Hold freezes the levelers dynamic gain and preserve its current state, for as long as the hold function is activated. This can be useful for example to prevent loud sounds like sneezing or coughing from causing leveler action.

Setup GUI – AUDIO PROCESSOR – Leveler - **Program**

The program leveller is a **LevelMagic** algorithm running in level mode that is optimized for program signals. Its main purpose is to balance the program signal to a certain target level for smooth auto voice over operation.




**Link** [Linked, Unlinked]
The program path may be unlinked for dual mono operation.

**Leveler** [ON / OFF]

- **Processing Profile** [Live / Speech / Pop / Uni / Classic]
- **Loudness Target (LUFS)** [-50 … 0]
- **Time (s/m/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>

**Expert** [ON / OFF]

- Clear Processing History <clear>
- Initial Dynamic Gain (dB)
- AGC Recovery

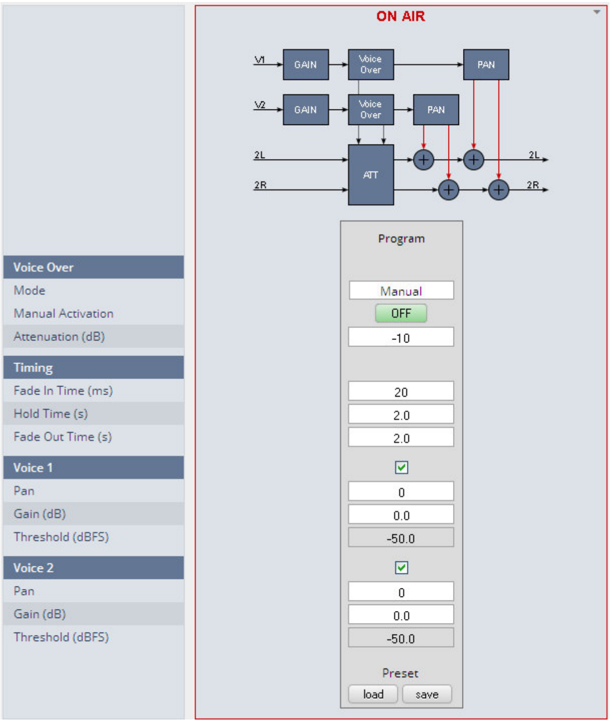
**Low Level Behavior**

- **Processing Threshold (dBFS)** [-80 … -70 … -20]
- **Below Threshold Mode** [Release / Hold]

For details regarding LevelMagic parameters see the bulletin: "Junger_LevelMagic-Leveler-Dynamics_Parameters_161128.pdf" on the Junger web site http://junger-audio.com/downloads.

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 signal flow.
(SYSTEM > Setup > Voice Channel Mode = Stereo).

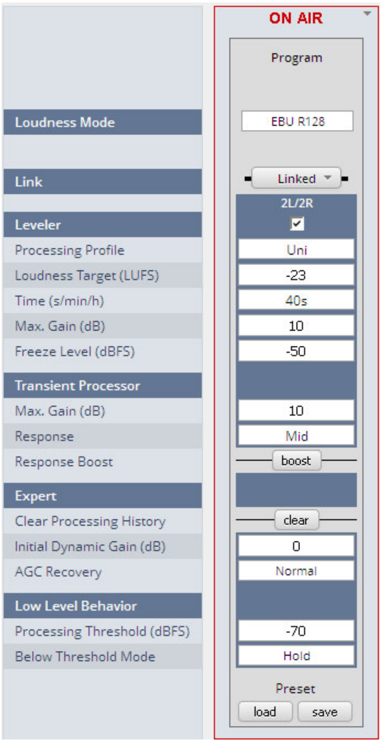



| | |
|---|---|
| **Voice Over** | |
| **Mode** | [OFF / Always ON / AUTO / Manual] |
| **Manual Activation** | [ON / OFF] |
| **Attenuation (dB)** | [-40 … 0] |
| **Timing** | |
| **Fade In Time (ms)** | [10 … 20 … 1000] |
| **Hold time (s)** | [0.0 … 2.0 … 10.0] |
| **Fade Out Time (s)** | [0.0 … 2.0 … 10.0] |
| **Voice 1** | [ON / OFF] |
| **Pan** | [-50 … 0 … 50] |
| **Gain (dB)** | [-80.0 … 0.0 … 20.0] |
| **Threshold (dBFS)** | [-60.0 … -50.0 … 0.0] |
| **Voice 2** | [ON / OFF] |
| **Pan** | [-50 … 0 … 50] |
| **Gain (dB)** | [-80.0 … 0.0 … 20.0] |
| **Threshold (dBFS)** | [-60.0 … -50.0 … 0.0] |

The Pan values represent the direction of the voice over signal in the program output from left [-50] over mid [0] to right [50].

Setup GUI – AUDIO PROCESSOR – **Level Magic**

This function block is used for loudness control of the program path. It can be used to control an independent program signal or a program signal including voice over (see Overview diagram).

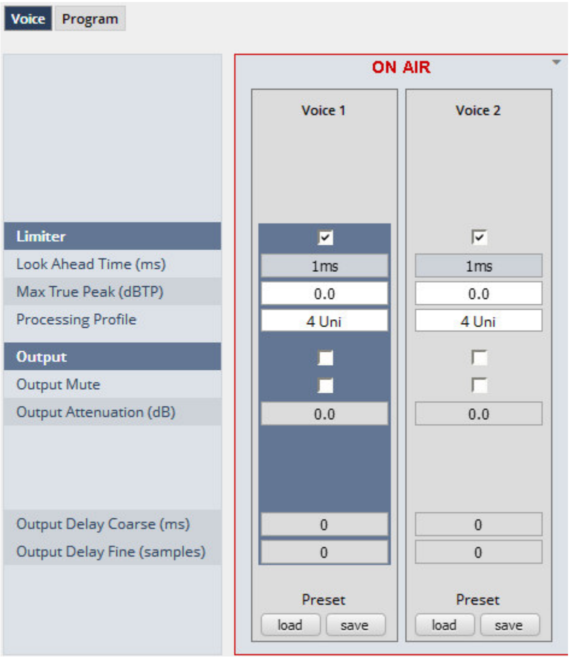


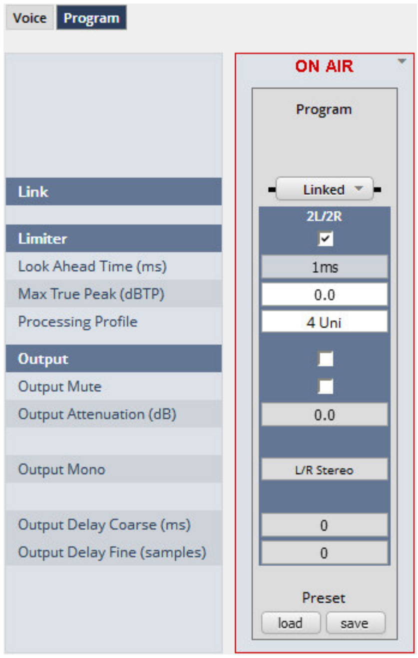

| | |
|---|---|
| **Loudness Mode** | [Level / ITU BS.1770-1 / -2 / -3 EBU R128 / ARIB TR-832 / ATSC A/85 (2011) / ATSC A/85 (2013) / Free TV OP-59 / Portaria 354) |
| **Link** | [Linked, Unlinked] The program path may be unlinked for dual mono operation. |
| **Leveler** | [ON / OFF] |
| **Processing Profile** | [Live / Speech / Pop / Uni / Classic] |
| **Loudness Target (LUFS)** | [-50 … 0] |
| **Time (s/m/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] |
| **Expert** | [ON / OFF] |
| **Clear Processing History** | <clear> |
| **Initial Dynamic Gain (dB)** | [-40 … 0 … 15] |
| **AGC Recovery** | [Normal / Fast] |
| **Low Level Behavior** | |
| **Processing Threshold (dBFS)** | [-80 … -70 … -20] |
| **Below Threshold Mode** | [Release / Hold] |

For details regarding LevelMagic parameters see the bulletin: "Junger_LevelMagic-Leveler-Dynamics_Parameters_161128.pdf" on the Junger web site http://junger-audio.com/downloads.

Setup GUI – AUDIO PROCESSOR – **Output**

The **Output** block allows you to use a **True Peak** limiter, **Mute** and **Attenuate** the output signals from the DSP, do a mono conversion for stereo channels and add delay. This may be set independently for both the voice and the program channel.



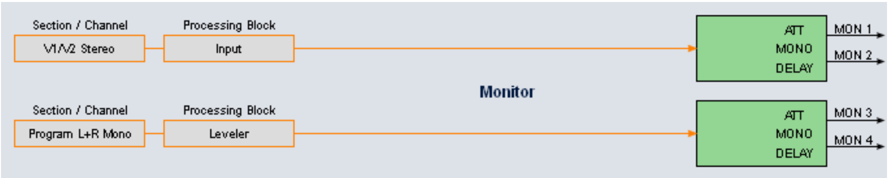

| | |
|---|---|
| **Link** | [Unlinked / Linked]<br>For voice channel only available if it is in stereo mode. |
| **Limiter** | [ON / OFF] |
| **Max True Peak (dBTP)** | [-20.0 … 0.0] |
| **Processing Profile** | [0 / 1 Live / 2 Speech / 3 Pop / 4 Uni<br>5 / 6 Classic / 7 / 8 / 9] |
| **Output** | [ON / OFF] |
| **Mute** | [ON / OFF] |
| **Attenuation (dB)** | [-80.0 … 0.0] |
| **Mono** | [L/R Stereo / L+R Mono / L/L Mono / R/R Mono] |
| **Output Delay Coarse (ms)** | [0 … 2000] |
| **Output Delay Fine (samples)** | [0 … 2000] |

If the voice channel mode is set to stereo (see SYSTEM > Setup > Voice Channel Mode = Stereo) a mono circuit will be available for the voice channel as well.

Setup GUI – AUDIO PROCESSOR – **Monitor**

As seen on the **AUDIO PROCESSOR > Overview** page the **D*AP4 VAP** provides two monitor facilities which may be connected to the function blocks of the audio processor (DSP).

For the example below (see SYSTEM > Setup) the VAP is in **Voice Channel Mode = "Stereo"**. The first monitor is connected to the **Input** section of the voice channels in **Stereo [V1/V2 Stereo]** while the second monitor is connected to the **Leveler** output of the **Program** path in **L+R Mono [Program L+R Mono]** (see AUDIO PROCESSOR > Overview):

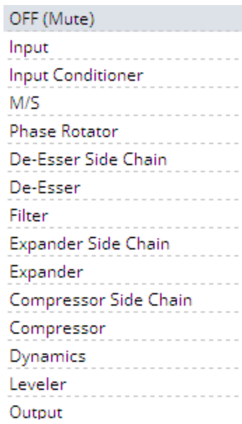



The settings must be done on the Monitor pane:


**Source**

**Section / Channel** [V1/V2 Stereo / V1+V2 Mono / Program Stereo / Program L/L Mono Program R/R Mono / Program L+R Mono]

**Processing Block** Select here which processing block should be monitored




**Monitor Output**

**Mute** [ON / OFF]

**Attenuation (dB)** [-80.0 … 0.0]

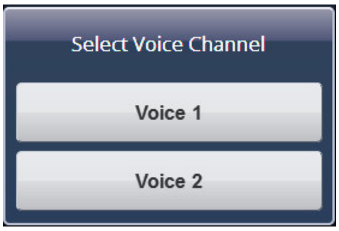
**Delay Coarse (ms)** [0 … 2000]

**Delay Fine (samples)** [0 … 2000]

Setup GUI – AUDIO PROCESSOR – **On Air** / **Mobile UI**

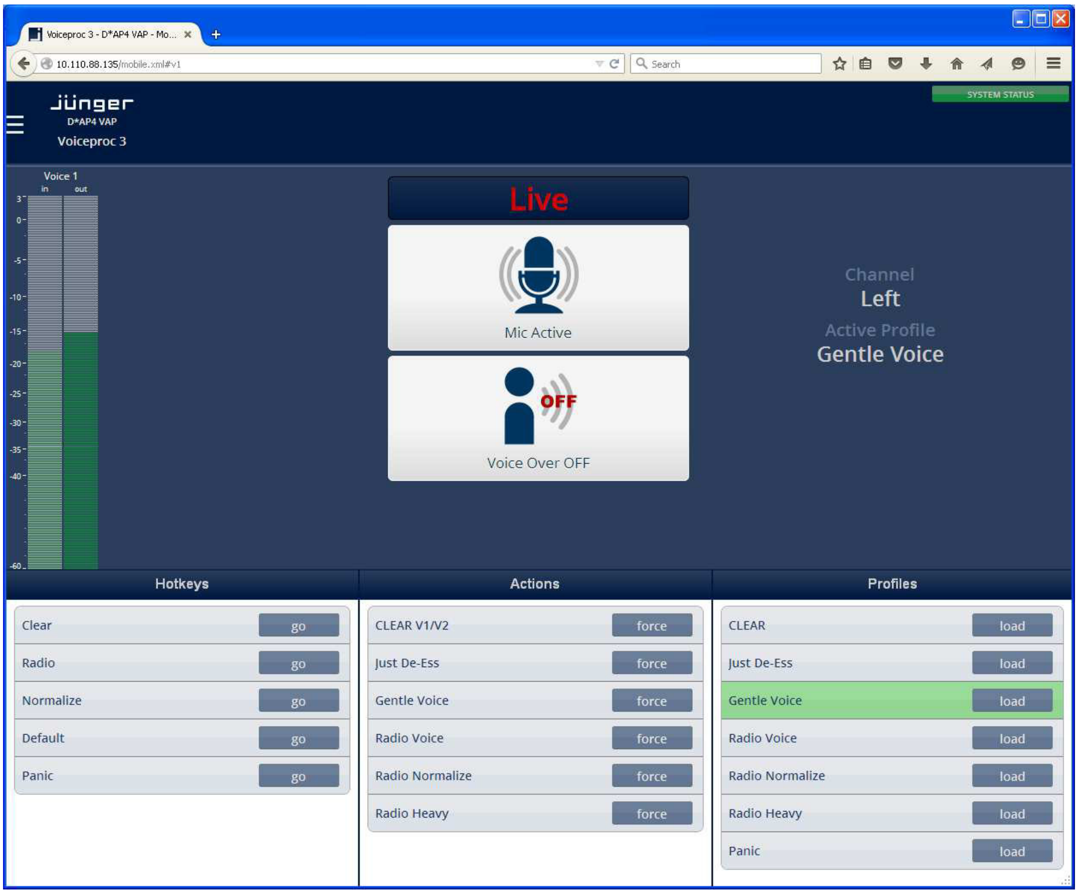
The **V'*AP** provides an **operator UI** for live application that may be used on tablets or mobile phones but may also be displayed on a PC's web browser. When you click on the **<Mobile>** soft button in the upper right area of the GUI above the **<Help>** button, a new tab in your current browser 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 wireless accessibility of the **D*AP4 VAP via** a WLAN.




If the device is setup for two independent voice channels (see SYSTEM > Setup Voice Channel Mode = 2 x Mono) you must select one of the two voice channels firstly.

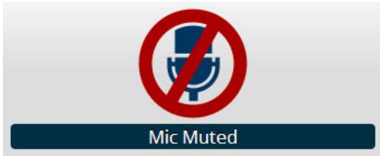
In case of Voice Channel Mode = Stereo both channels will be controlled via **Voice 1** channel.
Keep in mind that you may assign meaningful names to the voice channels (see SYSTEM > Setup > Section / Channel Labels).
For this explanation we named them "Voice 1" and "Voice 2".

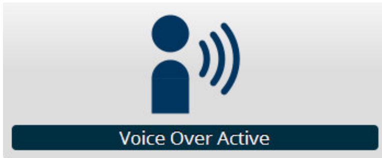



The **mobile UI** features (e.g. which type of buttons will be displayed) may be configured at SYSTEM > Remote Access > Mobile UI.

The content of this **operating UI** duplicates useful controls of the **V*AP**. You may activate Hotkeys which are assigned to the **X*AP RM$_1$** remote panel which in turn maybe setup for trigger actions of the event manager (see EVENTS > Triggers > Remote Hotkeys) for details).

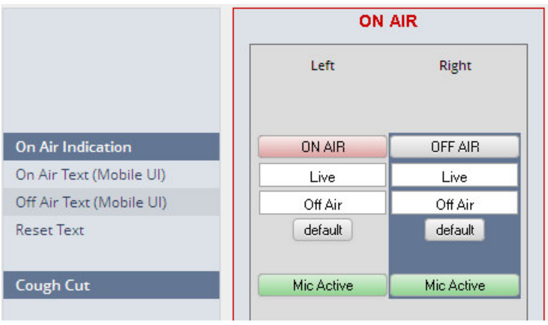
But you may directly trigger **Actions** or **Preset Events** at the bottom of the screen as well. The **mobile UI** will only display preset events which contain processing relevant parameters called **"Profiles"**. I.e. you can not reconfigure the **VAP** from here.




On a touch screen (or via left mouse button) you may simply press the icons (if enabled via SYSTEM > Remote Access > Mobile UI) in the middle of the screen to temporarily mute that channel for a cough cut function or to turn voice over on / off (if set to: **Voice Over** > Voice Over > Mode > “Manual”

The text at the top of the button(s) can be defined via the web GUI. The system distinguishes between two possible texts.




**On Air Indication** [OFF AIR / ON AIR] turns that mic channel permanently on or off

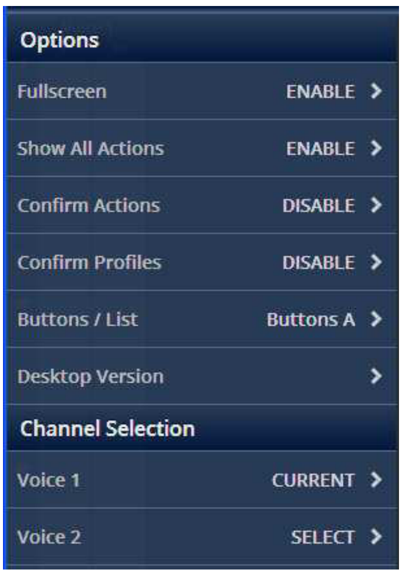
**On Air Text (Mobile UI)** will be displayed if the mic is on air

**Off Air Text (Mobile UI)** will be displayed if the mic is off air

**Reset Text** will restore the default text for both the on air and the off air display

**Cough Cut** temporary mic mute button. It works in parallel to the mobile UI's **<Mic Active>** button.

The option menu at the upper left hand side offers some additional settings from the mobile GUI:


The option selection field > shows what happens if you press it.

Set the browser to full screen

Sow all actions (not only the actions of the actual voice channel)

Confirm Actions will force a confirmation pop up

Confirm Profiles will force a confirmation pop up

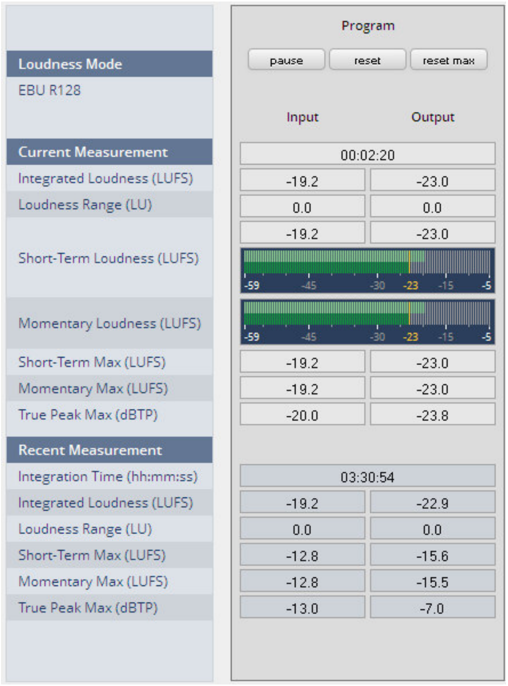
Selects between a button or list stile display

Will load the GUI of the VAP

Finally you may change the voice channel that is under control from the **mobile UI**.

Setup GUI – MEASURMENT – **Loudness**

The **D*AP4 VAP** offers a sophisticated loudness measurement tool for the input and output of the program path of the device. The three control buttons **<pause>**, **<reset>**, **<reset max>** may be used to manually control the actual measurement. The measurement method follows the general settings of the **D*AP4 VAP**: AUDIO PROCESSOR > Level Magic > Loudness Mode




**Loudness Mode** Defined at the Level Magic setup pane.

**Current Measurement** [hh:mm:ss]

**Integrated Loudness (LUFS)**

**Loudness Range (LU)**

**Short Term Loudness (LUFS)**

**Momentary Loudness (LUFS)**

**Short Term Max (LUFS)**

**Momentary Max (LUFS)**

**True Peak Max (dBTP)**

**Resent Measurement**

**Integrated Time (hh:mm:ss)**

**Integrated Loudness (LUFS)**

**Loudness Range (LU)**

**Short Term Max (LUFS)**

**Momentary Max (LUFS)**

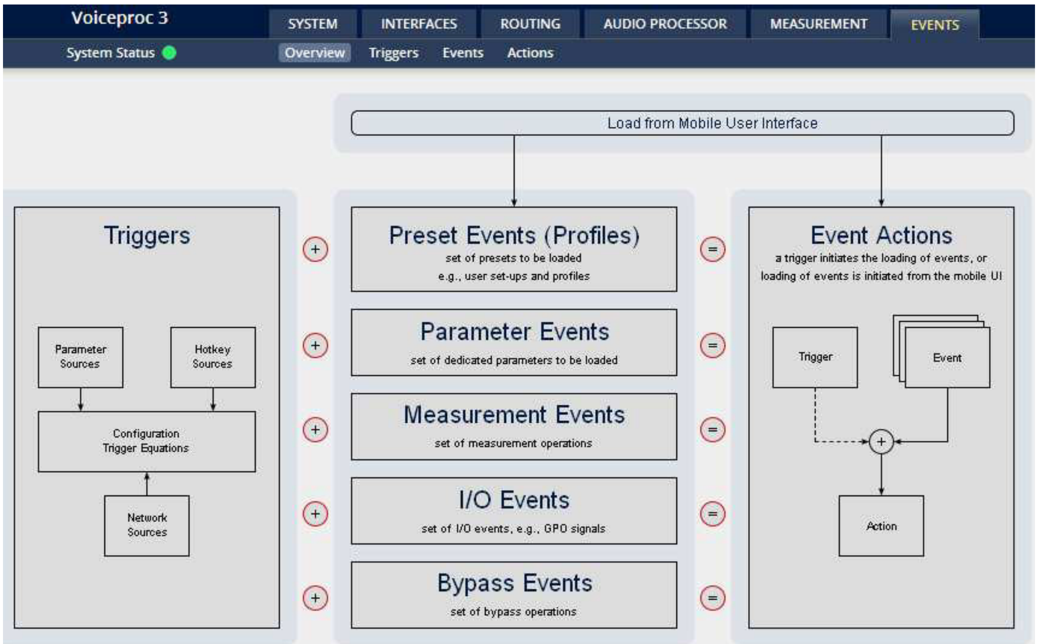
**True Peak Max (dBTP)**

For the terms and details of loudness measurement we would ask you to consult the respective standards like EBU R128 issued by the EBU (EBU_tech3341, 3342, 3343, 3344). You will also find explanations here: "Junger_LevelMagic-Leveler-Dynamics_Parameters_161128.pdf " on the Junger web site http://junger-audio.com/downloads.

Setup GUI – **EVENTS** – **Overview**

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

The overview shows the building blocks of the action management of the **D*AP4 VAP.**
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 EmBER+ protocol. |
| **Parameter Sources** | Device parameters / status information grouped into systems and Interfaces. |

The triggers will finally be defined by a trigger equation that may be the logical combination of two trigger sources.

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

| | |
|---|---|
| **Preset Events (Profiles)** | System / Interfaces / Routing / Audio Processor / Voice / Program / Monitor |
| **Parameter Events** | System / Audio Processor / Measurement |
| **Measurement Events** | Pause / Continue / Reset / Reset Max / Start / Pause / Stop |
| **I/O Events** | GPOs |
| **Bypass Events** | Voice / Program |

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

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

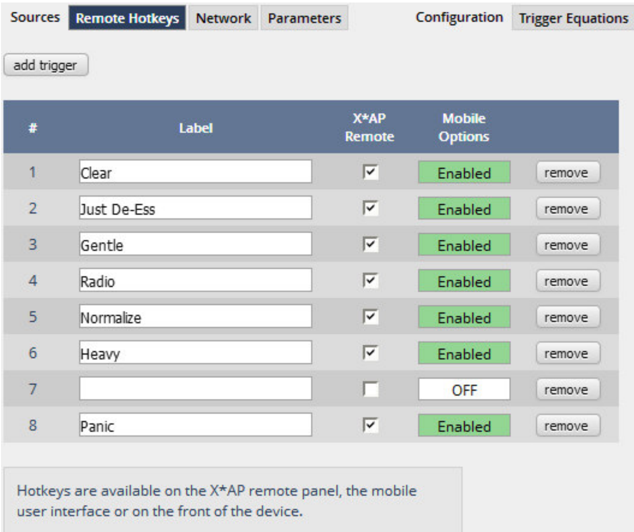
An action runs like a flip-book inside the **D*AP4 VAP**. This powerful technology spans from simply recalling a certain parameter over narrator specific parameter combinations (household name: "Preset") to the complete reconfiguration of the **D*AP4 VAP** including all signal routing, processing parameters and so forth. It also 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*AP4 VAP** will automatically react to the system and/or parameter status.

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

1. Define - trigger sources
2. Configure - triggers by logical combination of pre defined trigger sources
3. Set up events - by selecting presets for function blocks
4. Create actions - what will happen - which trigger will launch which event? Or what will happen in case of some one presses the **<BYPASS>** button at the **X*AP RM1** or is engaging the **<Force Trigger Active>** check box or ignite an action from the **mobile UI**.

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

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

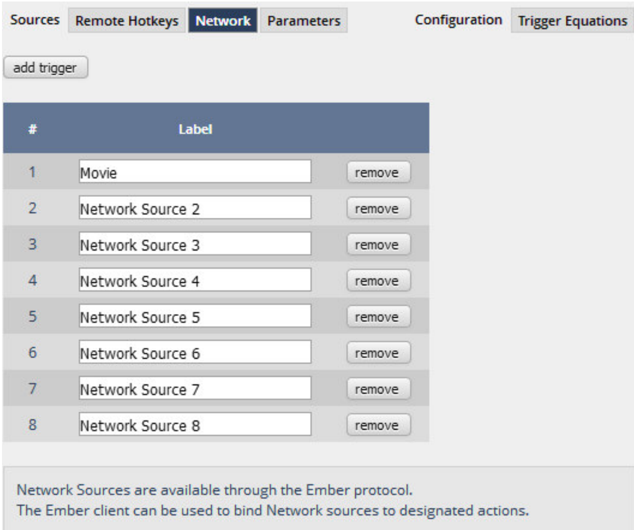



| | |
|---|---|
| **<add trigger>** | You can add lines here. |
| **#** | The number of the Hotkey on the **X*AP RM1** remote panel, counting from left to right. |
| **Label** | Each Hotkey may have a label that appears in the display of the **X*AP RM1** remote panel above that button. |
| **Enable**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

The number of hotkey triggers is not limited. You may also add virtual hotkeys which can be used by a 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. See code.google.com/p/**ember**-plus/ for details. The **D*AP4 VAP** receives such triggers over the TCP/IP network. The triggers are issued by a remote device or a broadcast automation system. You may assign these triggers to virtual panels, physical buttons or play list events of Ember+ enabled control instances. The **D*AP4 VAP** has 20 Network Triggers.

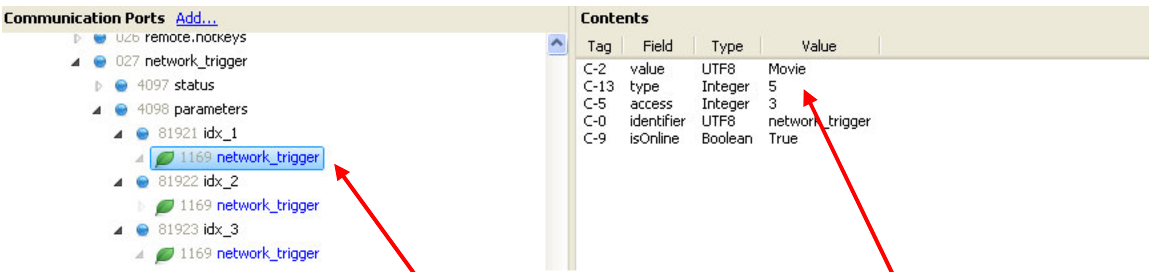



**#** The number of the network trigger.

**Label** Label of that network trigger. It appears on the **Configuration** pane as well as in the **EmBER+** tree of the setup interface of a control instance.

**<remove>** will remove a line from the list.

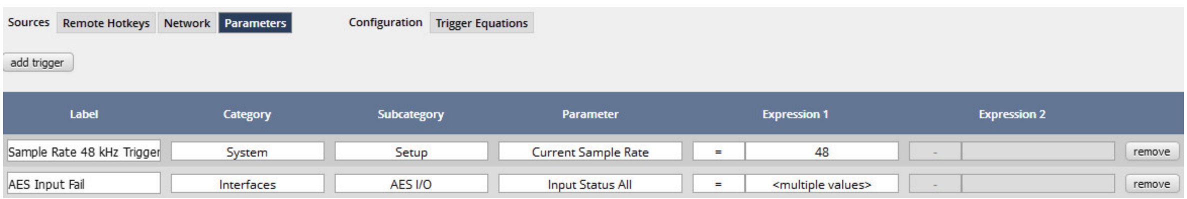
Below is a screen shot of the EmBER+ viewer tool:



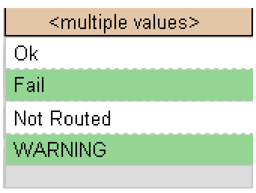

In the EmBER+ tree you go to:
"Device" > controller_dsp > network_trigger > parameters > e.g. "idx_1"

As a value you will receive the trigger name from the **D*AP4 VAP**. In this example it is the first trigger named: "Movie".

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



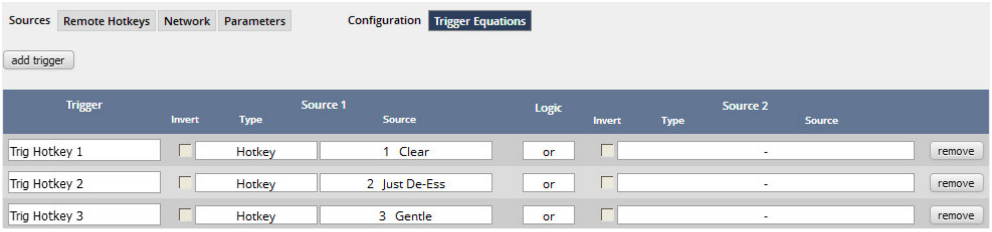

Above is an example of parameter trigger sources. **<multiple values>** indicates that more than one value of the parameter "Status" is bound to that trigger source:




If you click into the "Expression 1" field you see two entries marked greenish, i.e. if one of these values is true, "Expression 1" is true. You must uncheck both in order to select a different setting afterwards.

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.

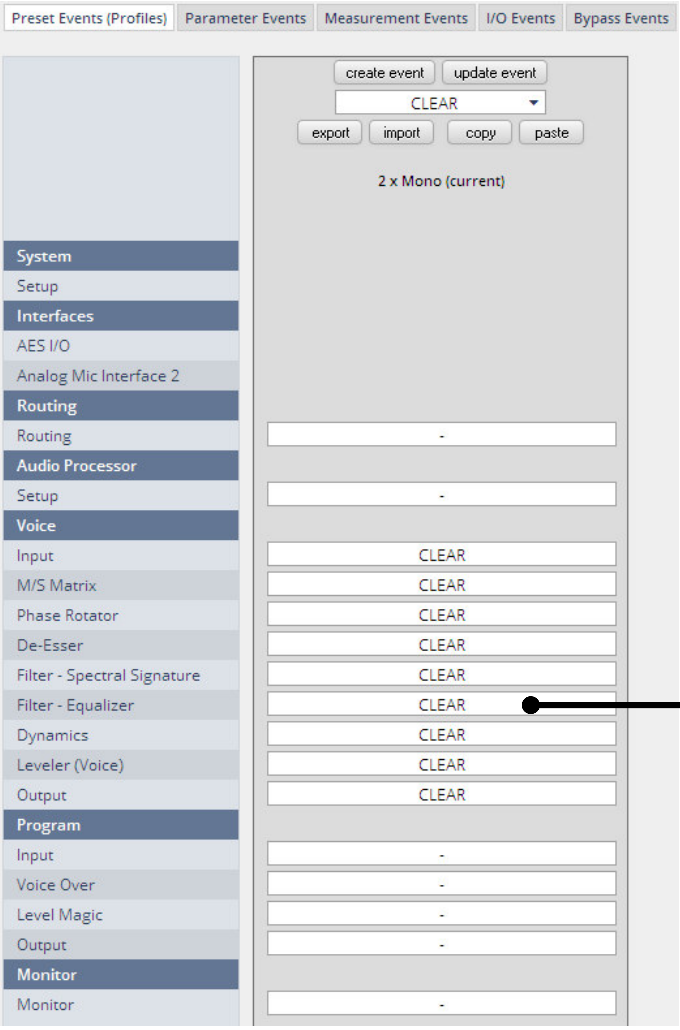
- **Invert** — [ON / OFF]
  If the type of trigger allows an inverted operation it can be defined here.
- **Type** — [GPI / Hot Key, Network / Parameter / Event active / Trigger effective/ Bypass / Sync Lock]
- **Source** — [e.g. for GPIs it will be 1 … 8]
  It acts like an index for the trigger type (In case of GPI it is the physical GPI number or in case of **X*AP** Hotkeys the key number (1 of 8)

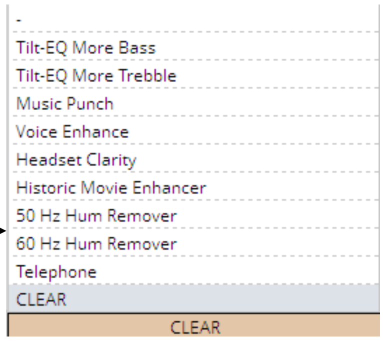
**Logic** — Kind of logical operation [and, or, xor].

**Source 2** — Second source for the logical combination of two trigger sources. If only one source exists, you may leave it unassigned [-].

Setup GUI – EVENTS – Events – **Preset Events (Profiles)**

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 De-Esser, 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:



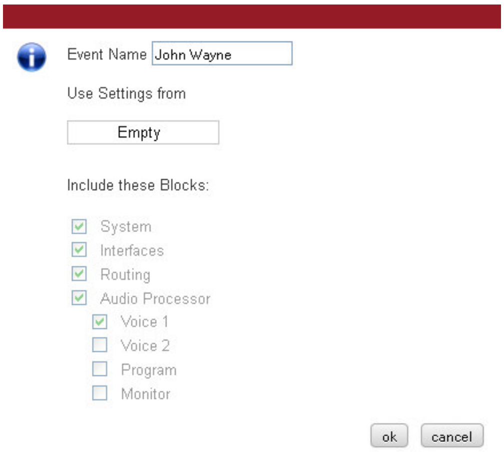




Pull down list of all factory default presets of the Filter-Equalizer section where **CLEAR** is one of it

The example shows that the factory default profile **CLEAR** is selected.
It will load the presets **CLEAR** for each function block of the voice channel.
I.e. each function block has a preset with the name CLEAR. Don't be confused by the same name of the presets!

This is the tool to reconfigure the **D*AP4 VAP** completely, partially or to change a few audio parameters marginally. At the top of the page you see the button **<create event>**. It may save all actual parameters in presets of the respective function block and assigns it the same name as the event itself.

You are able to create a new preset event by pressing **<create event>**:




**Event name** — [John Wayne]
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 marginally tuned afterwards to form a slightly different event.

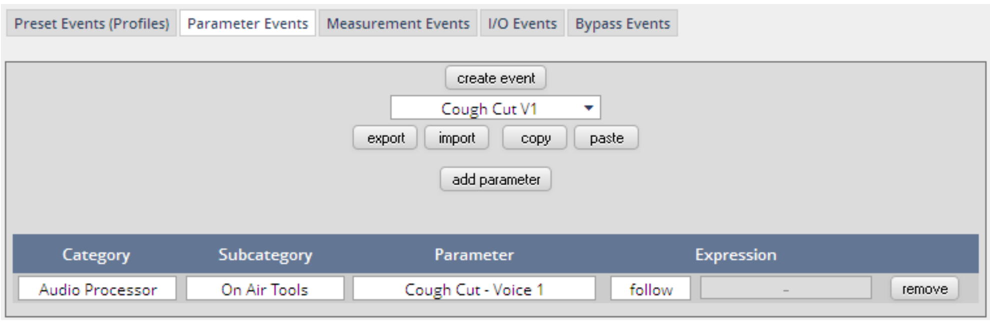
**Empty** — Creates a set of empty fields where you may select the preset of your choice for the respective function block or leave it empty if no changes are needed …

**Include these Blocks** — [System / Interface / Routing / Audio Processor / Voice 1 / Voice 2 / Program / Monitor]
You can tell the event manager which function blocks must be included in this event (or not).

**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**

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




**Category** — [Audio Processor / Measurement]

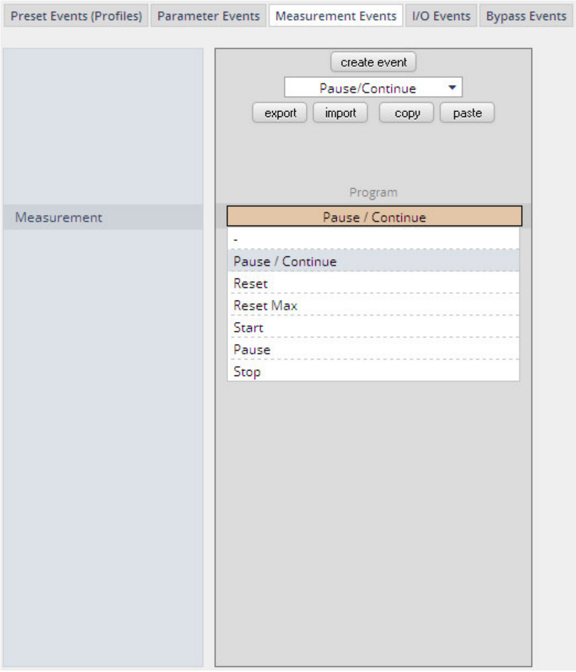
**Subcategory** — [in case of Category = Audio Processor >> On Air Tools]
[in case of Category = Measurement >> Loudness]

**Parameter** — depending on the pre-selected Subcategory you may define the relevant parameter which you want to set.

**Expression** — If applicable, the value of the parameter that will be set if the parameter event is triggered during an event action.

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

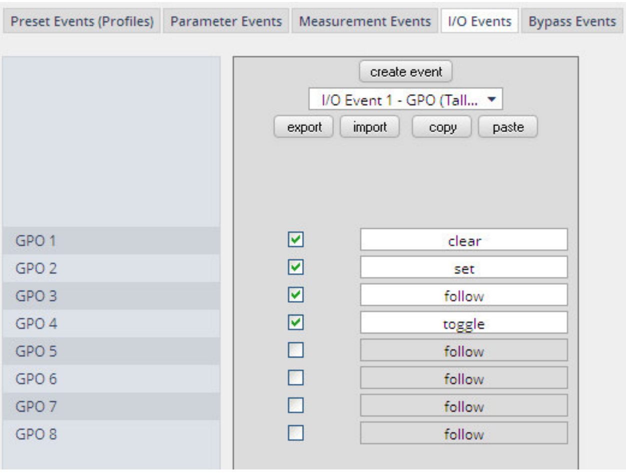
Measurement events can be used to control the integrated loudness measurement of the program path of the **D*AP4 VAP:**




For each measurement event you can assign one of the possible control functions from the pull-down.

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

At the moment I/O events are restricted to control the **GPOs** of the **D*AP4 VAP**:

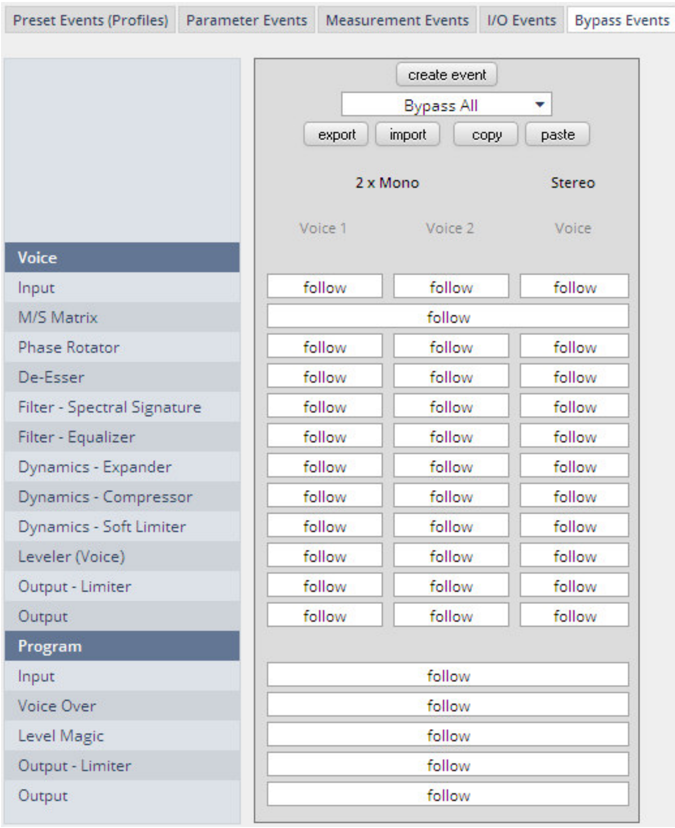



Each GPO (when incorporated into that I/O event) can be set to the behavior as follows:

| | |
|---|---|
| **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<br>Be careful because it needs a definite known starting condition to work properly. |

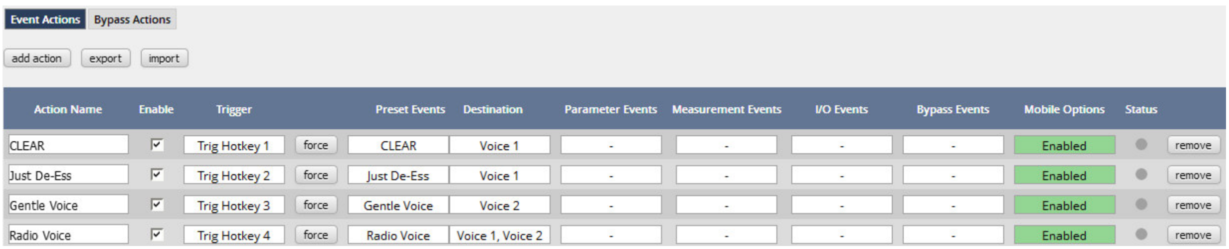
Setup GUI – EVENTS – Events – **Bypass Events**

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




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

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




You should give actions a meaningful name, select a trigger (from one of the trigger equations) and select the respective type of events you need to perform the desired action.
The column **Mobile Options** allows you to select whether or not the action will be available at the Mobile UI and what color a respective button will have.

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

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

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


**Trigger**

| | |
|---|---|
| **Source** | "X*AP Remote Bypass Button" |
| **Lock Source** | [ON / OFF]<br>The **X*AP RM1** remote panel <**BYPASS**> button may be disabled / enabled here. |
| **Force Trigger Active** | [ON / OFF]<br>Force the bypass function from the GUI instead of the **X*AP RM1** remote panel <**BYPASS**> button. |
| **Bypass Event** | "2 x Mono / Stereo"<br>This is a generic setup page where both modes (2x Mono / Stereo) can be pre-set. |
| **Voice** | Audio processing blocks of the voice path. |
| **Input … Output** | [follow / -]<br>Define the voice function block that is part of the bypass action. |
| **Program** | Audio processing blocks of the program path. |
| **Input … Output** | [follow / -]<br>Define the program function block that is part of the bypass action. |

Technical Data - 2 Channel Voice Audio Processor Edition **[D*AP4 VAP EDITION]**

| | | |
|---|---|---|
| General | • 2 channel voice processor (mono or stereo)<br>• 2 channel program path processing<br>• 2 channel monitor output<br>• Expandable by hard and software options | |
| Audio Sample Rate | 44.1, 48, 88.2, 96kHz, (32 … 196kHz @ input with SRC)<br>±150ppm sync input capture, ±25ppm master-sync stability | |
| AES/EBU Input | Relevant specifications comply with AES3-X-2009, IEC 60985 and AES11-2009 | |
| | 2 channels (1 stereo input), XLR-3 connector | |
| | 24bits, transparent forwarding of PCM and compressed audio (w/o SRC)<br>24bits, PCM, sample rate converter (SRC) activated | |
| | Impedance | 110Ohm, differential |
| | Input level | 0.3 … 5Vpp @ 110Ohm differential |
| | Sample Rate Converter (SRC) | THD+N -120dB @ 0dBFS, 1kHz<br>Latency < 0.3ms |
| AES/EBU Output | Relevant specifications comply with AES3-X-2009, IEC 60985 and AES11-2009 | |
| | 2 channels (1 stereo output), XLR-3 connector | |
| | 24bits, transparent forwarding of PCM and compressed audio | |
| | Impedance | 110Ohm, differential |
| | Output voltage | 3Vpp (typ.) @ 110Ohm differential |
| | Power fail relay bypass between AES/EBU input and output (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.) @ R = 75Ohm single-ended |
| Network Interface | RJ45 connector, 10/100Mbit Ethernet auto sense, full duplex, auto MDI/X | |
| USB Interface | USB 2.0 connector to internal console interface | |
| GPI Signals | 8 general purpose inputs (GPI), divided into 2 groups with separate common signal, isolated | |

| | | |
|---|---|---|
| | Connector type | D-Sub25 connector female, same for GPO |
| | Input conditions | 3 … 24Vdc, < 5mA |
| | Auxiliary supply | 5V (nom.), 200mA (max.), isolated |
| GPO Signals | 8 general purpose outputs (GPO), SPST, divided into 2 groups with separate common signal, isolated | |
| | Connector type | D-Sub25 connector female, same for GPI |
| | Output conditions | 24Vac/dc (max.), 120mA (max.) |
| Expansion Slots | 1 general purpose expansion slot for option boards,<br>1 dedicated expansion slot for 2 microphone inputs [O_DAP_AMIC_a] | |
| Power Supply | Dual power supply, automatic fail over,<br>85 … 264Vac, 50 … 60Hz, 58W (max.) | |
| Environmental | Operating temperature 0 … 50ºC, fan cooled,<br>Non-operating -20 … 70ºC,<br>Humidity < 90%, non-condensing | |
| Physical | 19", 1 RU, 27 cm 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.<br>Audio embedding and de-embedding complies with SMPTE 299M (3G, HD) or SMPTE 272M-AC (SD).<br>Metadata embedding and de-embedding complies with SMPTE 2020-2. | |
| Video Data Rate | 2970/2967Mbps (3G), 1485/1483.5Mbps (HD), 270Mbps (SD) | |
| Video Formats | 1080p23.975, 24, 25, 29.97, 30, 50, 59.94, 60<br>1080i50, 59.94, 60<br>720p23.975, 24, 25, 29.97, 30, 50, 59.94, 60<br>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<br>> 10dB, 1485 … 2970MHz |
| | Cable length (max.) | 250m @ SD for Belden 1694A cable<br>230m @ HD for Belden 1694A cable<br>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<br>> 10dB, 1485 … 2970MHz |
| | Output jitter | < 0.2UI (Alignment), < 0.5UI (Timing) |
| Audio Latency | Input to Output | Embedder and de-embedder combined<br>HD, 3G < 0.6ms<br>SD typ. 1.5ms (< 2ms) |
| General Features | • Power fail relay bypass (may be activated via GUI)<br>• Lip-Sync compensation for processed and non-processed audio signals<br>• Dedicated routing for non-processed channels, all channels (max. 16) can be routed to/from the device or looped through<br>• Test pattern generator<br>• Master-sync capable<br>• ITU-R BT.1685 / ARIB STD-B39 metadata support | |

Technical Data – Option Board Analog Line-In and/or Mic-In **[O_DAP_AMIC_a]**

| | | |
|---|---|---|
| Audio | 24bit sigma-delta A/D-converter | |
| Audio Sample Rate | 44.1, 48, 88.2, 96kHz | |
| Inputs | 2 channels, selectable for microphone or line level | |
| | Connector type | XLR-3 |
| | Input level (max.)<br>(0dBFS equiv.) | Mic: 14dBu @ 0dB gain<br>Line: 22dBu @ 0dB gain, pad attenuation -10dB |
| | Impedance | Mic: 9kOhm (typ.), differential<br>Line: 20kOhm (typ.), diff., pad attenuation -10dB |
| | THD+N | -88dB @ -1dBFS, 1kHz, 0dB gain |
| | Dynamic range | > 110dB (RMS) |
| | Crosstalk attenuation | > 110dB @ 1kHz (typ.) |
| | CMRR | > 60dB @ 1kHz (typ.) |
| | Equivalent input noise | -126dBu (RMS) (typ.) @ 65dB gain |
| | Frequency response | 20Hz … 22kHz (< ±0.1dB) @ 48kHz<br>20Hz … 43kHz (< ±0.1dB) @ 96kHz |
| | Preamp gain | 0dB, 10 … 65dB in 1dB steps |
| | Pad attenuation | -10dB |
| Phantom Power | 48V (nom.) enable per input channel, individual short-circuit protection | |
| General Features | • Floating, balanced analog inputs<br>• Electrical isolation between both channels and device<br>• Digitally controlled input gain 0 … 65dB | |

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

| | | |
|---|---|---|
| Audio | 24bit D/A-converter | |
| Audio Channels | 8 output channels (e.g. for speakers) | |
| Audio Sample Rate | 44.1, 48, 88.2, 96kHz | |
| Analog Outputs | 8 channels | |
| | Connector type | D-Sub25 connector female |
| | Output Level (max.) (0dBFS equiv.) | 0 … 24dBu, adjustable in 0.5dB steps |
| | Impedance | 50Ohm (typ.), differential |
| | THD+N | -91dB @ 0dBFS = 15dBu, 1kHz |
| | Dynamic range | > 103dB (RMS) |
| | Crosstalk attenuation | > 103dB @ 0dBFS = 15dBu, 1kHz |
| | Frequency response | 20Hz … 22kHz (< ±0.3dB) @ 48kHz<br>20Hz … 43kHz (< ±0.3dB) @ 96kHz |
| General Features | • Power fail glitch prevention<br>• Balanced analog outputs<br>• Electrical isolation between outputs and device | |

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

| | | |
|---|---|---|
| Audio | 24bit sigma-delta A/D-converter, 24bit D/A-converter | |
| Audio Channels | 4 input channels, 4 output channels | |
| Audio Sample Rate | 44.1, 48kHz | |
| Analog Inputs | 4 channels | |
| | Connector type | D-Sub25 connector female, same for outputs |
| | Input Level (max.) (0dBFS equiv.) | 0 … 24dBu, adjustable in 0.5dB steps |
| | Impedance | 20kOhm (typ.), differential |
| | THD+N | -93dB @ 0dBFS = 15dBu, 1kHz |
| | Dynamic range | > 110dB (RMS) |
| | Crosstalk attenuation | > 93dB @ 0dBFS = 15dBu, 1kHz |
| | CMRR | > 71dB @ 0dBFS = 15dBu, 1kHz |

| | | |
|---|---|---|
| | Frequency response | 20Hz … 22kHz (< ±0.1dB) @ 48kHz<br>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<br>20Hz … 43kHz (< ±0.3dB) @ 96kHz |
| General Features | • Power fail relay bypass between inputs and outputs<br>• Balanced analog inputs and outputs<br>• 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)<br>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<br>0.3 … 5Vpp @ 75Ohm single-ended |
| | Sample Rate Converter (SRC) | THD+N -120dB @ 0dBFS, 1kHz<br>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<br>1Vpp (typ.) @ 75Ohm single-ended |
| General Features | • Power fail relay bypass (can be deactivated by jumper)<br>• Input sample rate converters (SRC)<br>• Electrical isolation between inputs, outputs and device (if configured for differential mode, 110Ohm)<br>• AES3 channel status management, non-audio detection<br>• 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<br>Processable by D*AP8: 16 channels @ 44.1, 48kHz<br>Processable by D*AP4: 8 channels @ 44.1, 48, 88.2, 96kHz | |
| | Impedance | 75Ohm |
| | Input level | 0.15 … 0.8Vpp @ 75Ohm |
| | Cable length (max.) | 150 m (Belden 1694A) |
| BNC Output | 64/56 channels @ 44.1 and 48kHz, 32/28 @ 88.2 and 96kHz<br>Processable by D*AP8: 16 channels @ 44.1, 48kHz<br>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<br>• Reference grade word clock recovery, master-sync capable<br>• Dedicated routing for non-processed channels, all channels (max. 64) can be routed to/from the device or looped through<br>• 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<br>Processable by D*AP8: 16 channels @ 44.1, 48kHz<br>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]:<br>-31 … -8dBm, OM2 multimode (50/125µm)<br>[O_DAP_MO_SM_a]:<br>-23 … -8dBm, singlemode (9/125µm)<br>(standard values, others on request) |

| | | |
|---|---|---|
| | Cable length (max.) | [O_DAP_MO_MM_a]: 1.5km, OM2 multimode<br>[O_DAP_MO_SM_a]: 2km, singlemode<br>(standard values, others on request) |
| Optical Output, LC | 64/56 channels @ 44.1 and 48kHz, 32/28 @ 88.2 and 96kHz<br>Processable by D*AP8: 16 channels @ 44.1, 48kHz<br>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]:<br>-23 … -14dBm, OM2 multimode (50/125µm)<br>[O_DAP_MO_SM_a]:<br>-15 … -8dBm, singlemode (9/125µm)<br>(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)<br>• Reference grade word clock recovery, master-sync capable<br>• Dedicated routing for non-processed channels, all channels (max. 64) can be routed to/from the device or looped through<br>• AES3 channel status management, non-audio detection<br>• 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),<br>primary and secondary port |
| Inputs | Processable by D*AP8: 16 channels @ 44.1, 48kHz<br>Processable by D*AP4: 8 channels @ 44.1, 48, 88.2, 96kHz |
| Outputs | Processable by D*AP8: 16 channels @ 44.1, 48kHz<br>Processable by D*AP4: 8 channels @ 44.1, 48, 88.2, 96kHz |
| General Features | • AES67 compliant (when available)<br>• Network master-sync can be provided by D*AP device<br>• Master-sync capable (for D*AP device)<br>• Non-audio detection for input channels<br>• Glitch-free Dante™ audio redundancy using dual Ethernet networks |

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

8x GPI/O

| 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 **-** |

Mic / Line IN

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

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

4x analog I/O [O_DAP_ADDA_a]

| 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]

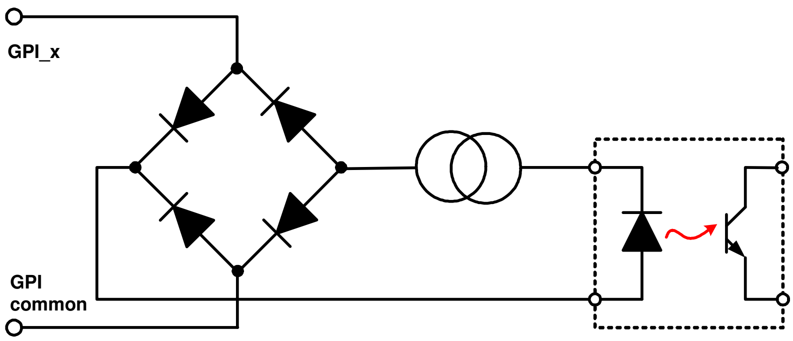
| connector: | 4x AES I/O |
|---|---|
| female | 25-pin D-Sub |
| | |
| 1 | OUT-4 + |
| 2 | GND |
| 3 | OUT-3 - |
| 4 | OUT-2 + |
| 5 | GND |
| 6 | OUT-1 - |
| 7 | IN-4 + |
| 8 | GND |
| 9 | IN-3 - |
| 10 | IN-2 + |
| 11 | GND |
| 12 | IN-1 - |
| 13 | |
| 14 | OUT-4 - |
| 15 | OUT-3 + |
| 16 | GND |
| 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:

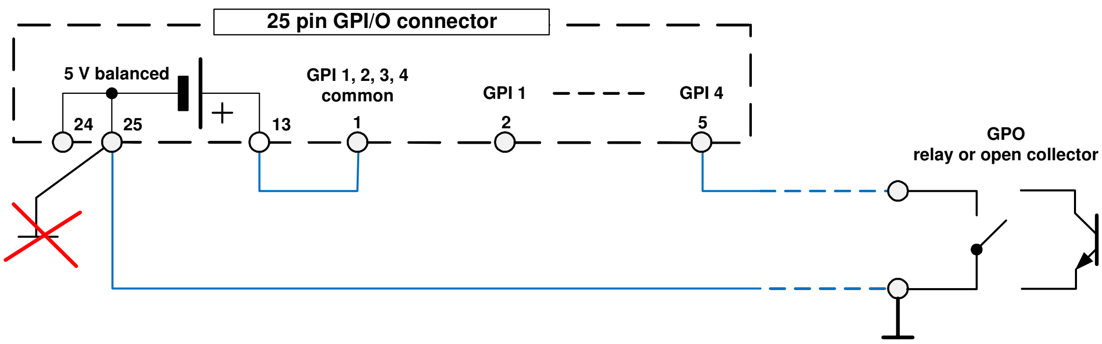



At the **GPI** input is a **bridge rectifier** I.e. you do **not** need to care about the polarity of the input voltage.
A **constant current source** in line with the **optical coupler** limits the current.
You must simply provide a voltage in the range from 5V to 30V to activate a **GPI**.

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<br>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.<br>The ON / OFF position is indicated by engravings [I] / [o] on the lever. It must be reached without difficulty.<br>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.<br>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.<br>It is recommended to use metal brackets (sheet steel angle) to support the device. |
| Provide proper Ventilation | this case and if the device has a built in fan, a gap of at least 1cm must be left between the device edge and the steel angle. It is highly recommended to leave a gap of at least 1RU above and below the device. |
| Use proper power cord | Use only the power cord specified for this product and certified for the country of use. |
| Do not operate without covers | Do not operate this product with covers or panels removed. |
| Do not operate with suspected failures | If you suspect that there is damage to this product, have it inspected by qualified service personnel. |
| Risk of explosion: | The device contains a lithium battery. If replaced incorrectly or by a different or inadequate type an explosion may occur. |

**Warranty**

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

Specifications are subject to change without notice

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

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

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

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

**www.facebook.com/jungeraudio** **www.twitter.com/jungeraudio**

© 2017 Jünger Audio GmbH
All rights reserved. All trademarks are the property of their respective owners. Product specifications are subject to change without notice.
This material is provided for information purposes only; Jünger assumes no liability related to its use