D*AP4 VAP SLIM LINE

D*AP4 VAP Edition

Digital Voice Processor

Manual

www.jungeraudio.com

jünger Operating Manual **D*AP4 VAP**

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, QR code, 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
• **AUX program path**	extra 2Ch input for a program signal
• **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 output**	extra feed from the DSP to monitor DSP processing blocks
• **SNMP agent**	SNMP v1, see D*AP4 VAP-MIB
• **Remote control**	l-s-b EmBER plus protocol for VSM integration, 3rd party API
• **Mobile user interface**	graphical operator UI optimized for use on mobile devices

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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 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 (no device setup functions).

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.

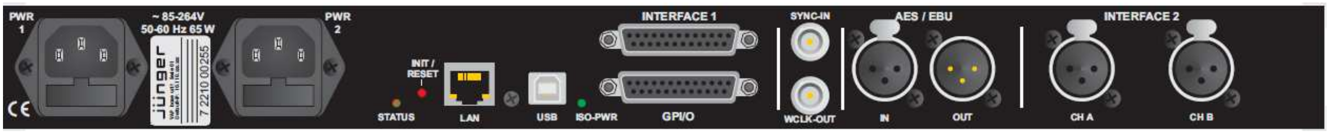
D*AP4 VAP front panel view

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 Red = device status ERROR 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).

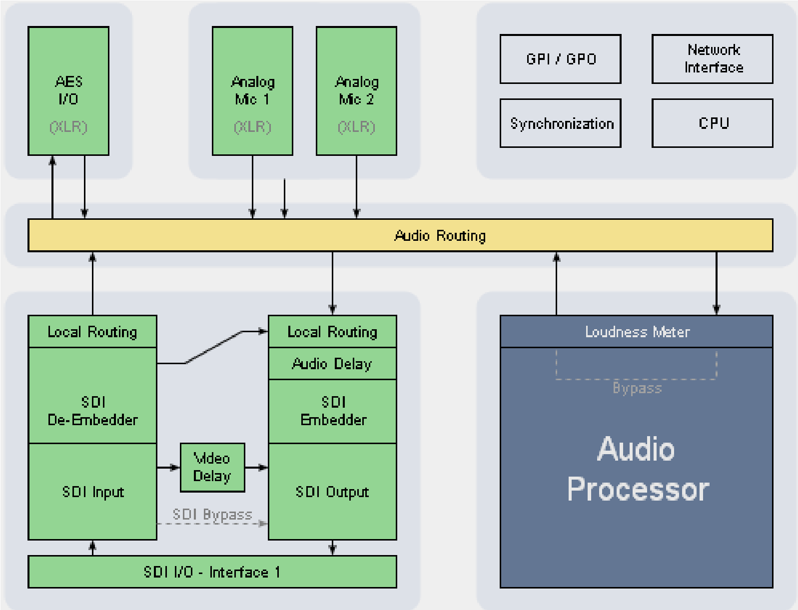
D*AP4 VAP rear view



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. 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 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.

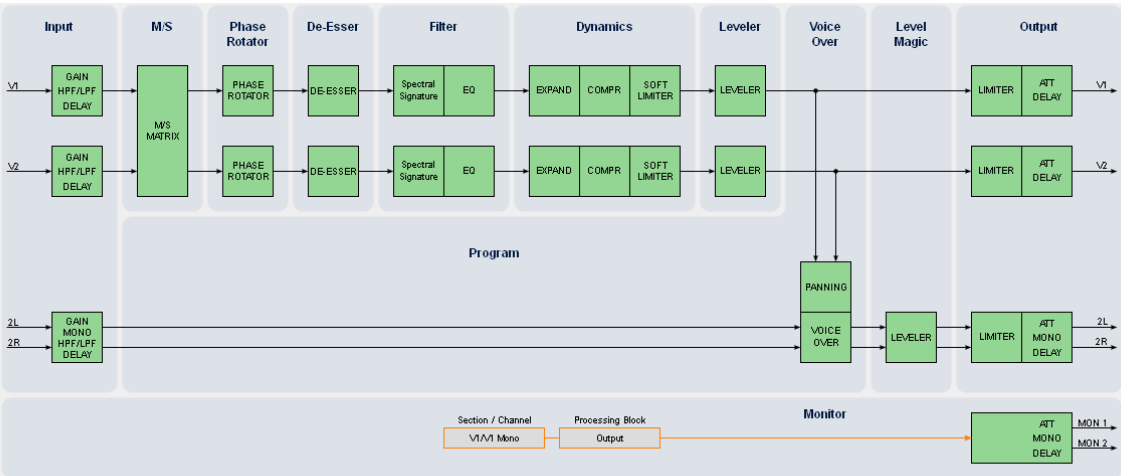
An interface slot is provided to carry optional 3G / HD / SD-SDI, AES I/O or even analog expansion modules. It allows for extremely flexible interfacing of the **D*AP4 VAP**, especially for video based voice over applications. The above schematic shows an SDI interface installed.

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 in order to process it. Similarly the **DSP** outputs 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 Firefox 20.x and higher.

The setup GUI can be complemented by other application programs running on MS Windows® XP, W7, W8 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 highly recommends using the **l-s-b Ember+** protocol which is widely distributed in the European broadcast industry. The user community is also increasing rapidly world wide.
By default, the **X*AP RM1** remote panel and the **D*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.
* **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 (FireFox 20.x 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

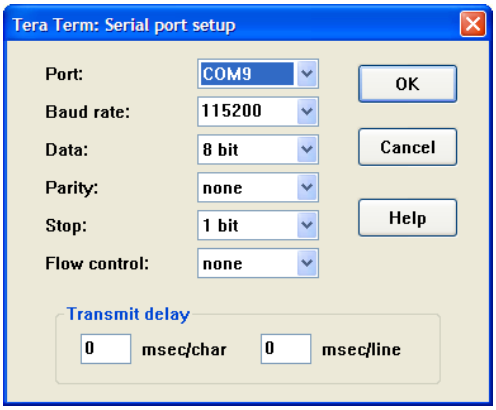
   You have 2 choices to assign the **D*AP4 VAP** an **IP address** :
   * From the serial console interface
   * Via Web browser
3. Assign the **X*AP RM1** remote panel a unique IP address configuration
4. Attach the **D*AP4 unit** to the **X*AP RM1** remote panel

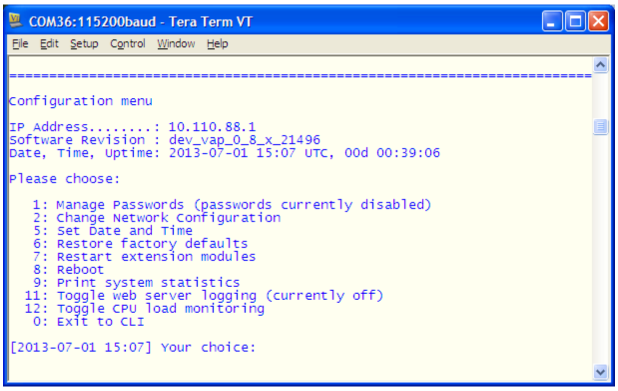
! Important Note: If you are not familiar with setting up devices for IP communication, we highly recommend you consult your system service or IT department to assist you.

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"** :

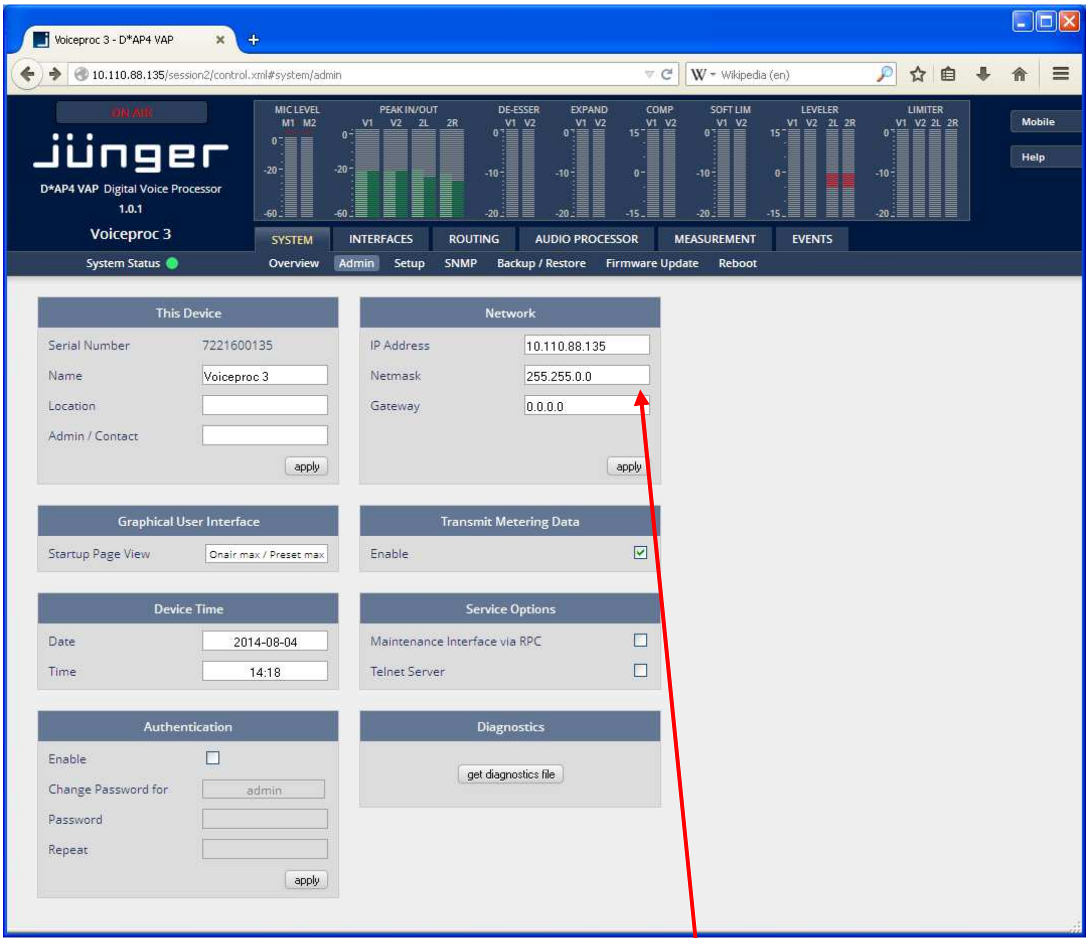
Do you want to reboot the device ? y **<ENTER>**

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 :



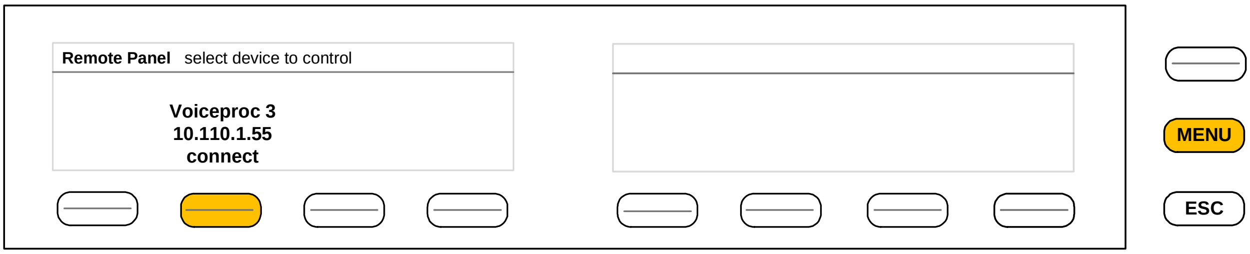
Enter the desired network configuration and press **<apply>**

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

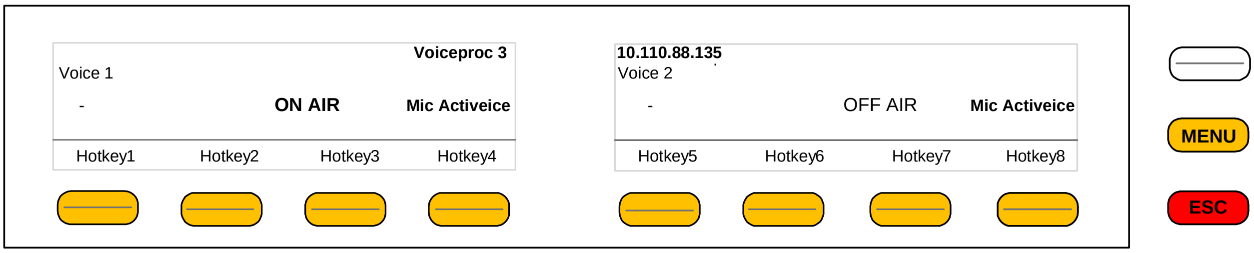
Important Note! After reboot neither the **web browser** nor the **X*AP RM1** remote panel may be able to communicate with the **D*AP4 device**. You must fill in the new IP address in the URL field and change the **X*AP RM1** remote panel settings to attach this device with its new IP address.

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

Power up display – may show up to four **D*AP4 VAP's** enabled for remote control for 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.



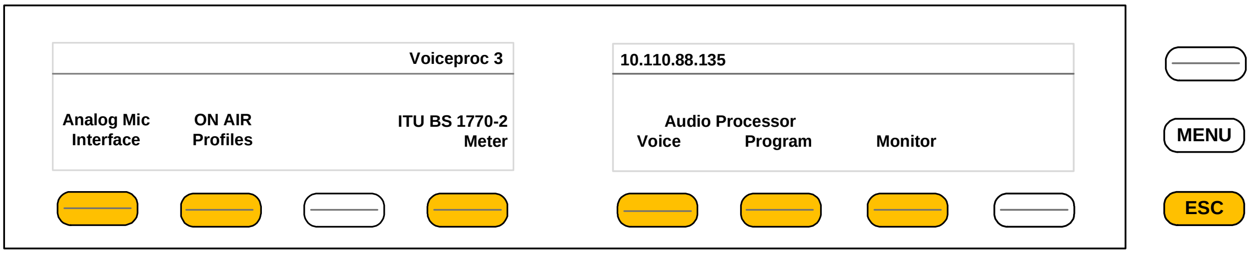
Pressing that button will connect with the **D*AP4 VAP.**
Now the **X*AP RM1** remote panel will gather all necessary information from that **D*AP4 VAP** unit (this may take a few seconds). When finished the **main operating display** opens up :



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

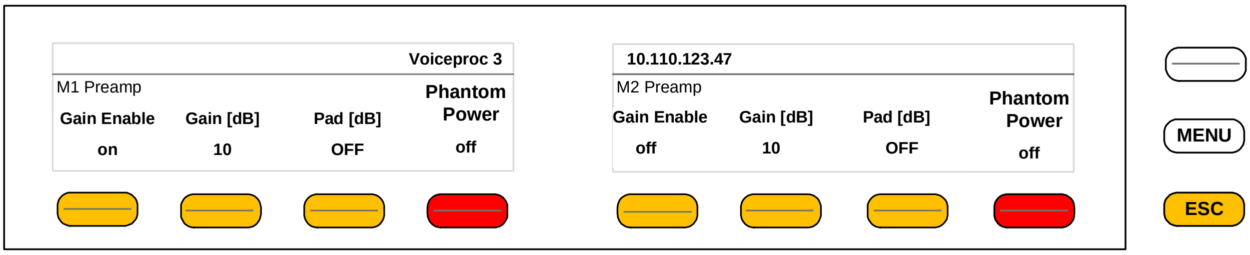
Operating – menu structure of the **X*AP RM1** remote panel – **operating displays**

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



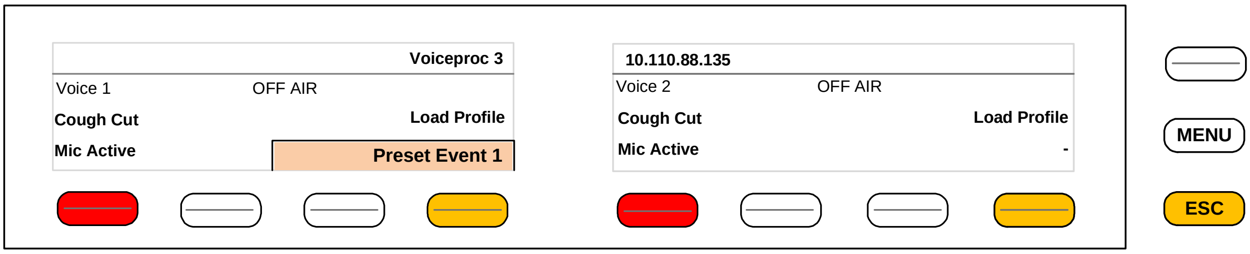
When pressing the **<ESC>** button you will return to the **main operating display**.

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



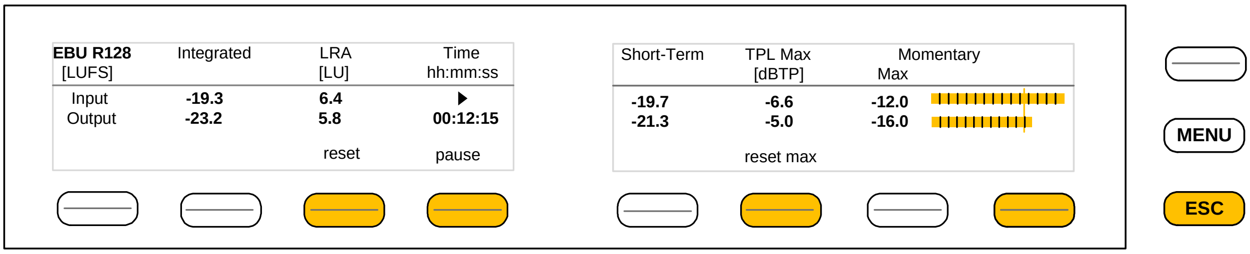
Here you can se tup both Mic preamps.

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



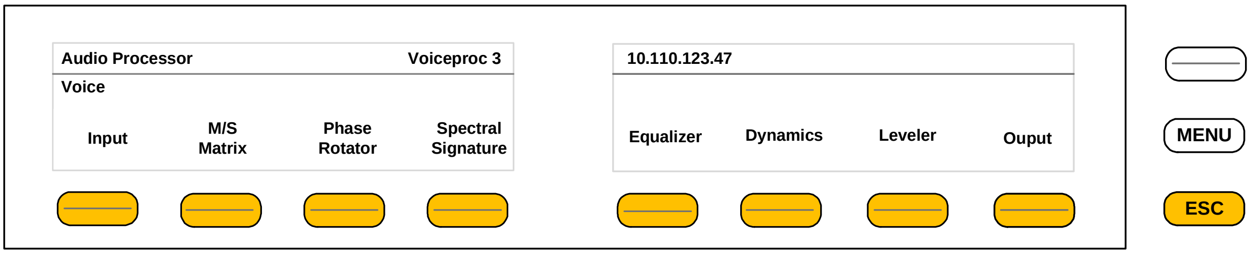
Here you may remote control the **cough cut** for the respective voice channel and load a pre-defined profile. A **profile** is a set of audio relevant presets that must be set-up in the :
EVENTS > Events > Preset Events section. Pressing the **<Load Profile>** button will highlight the area above the button (see above – the default display is a dash). You can now select a Preset Event by turning the **Rotary Encoder**. Here only **Preset Events** are available which contain presets that are relevant for the voice channel.

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



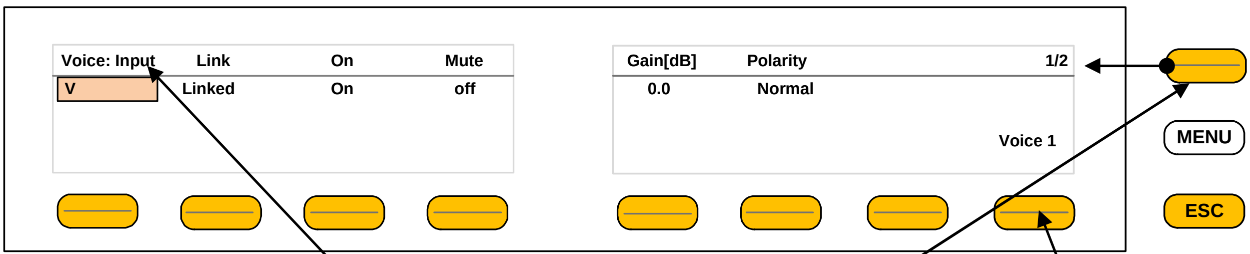
The meter style (ITU BS.1770-x / ATSC / EBU etc.) is defined by the settings of :
AUDIO PROCESOR > Level Magic > Loudness Mode. The above menu serves as a display of measurement values and offers the metering control buttons (reset & pause / continue).

Operating – menu structure of the **X*AP RM1** remote panel – operating displays – **Audio Processor > Voice**



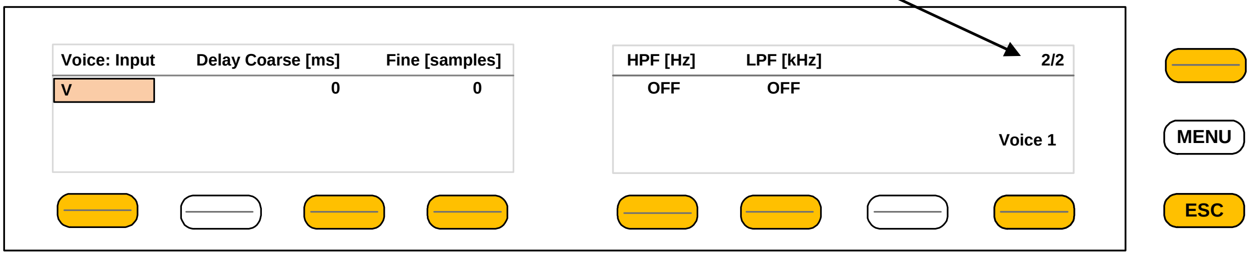
This menu gives access to tweak the voice channel(s) either in **2 x Mono** or **Stereo** mode.

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 the **2 x Mono** mode) :

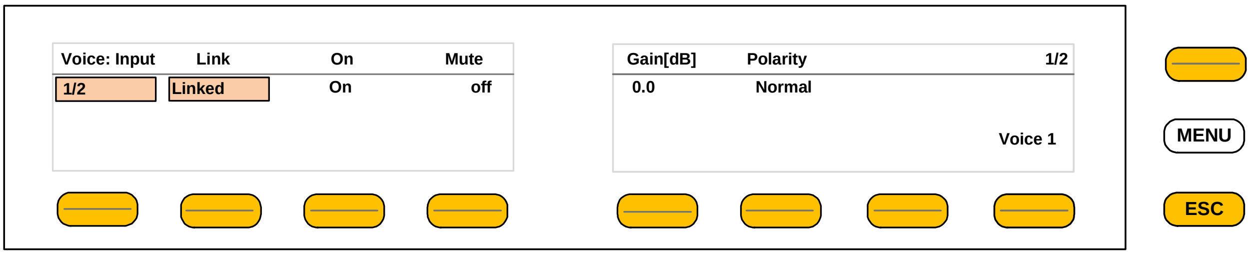


Here you are at the input section of the voice channel. The **<Page>** button toggles between two pages in this case and gains access to the remaining parameters. If the unit is set to **2 x Mono** mode, the **key #8** toggles between the two voice channels.

Here is the example for **page #2** after pressing the **<Page>** button :

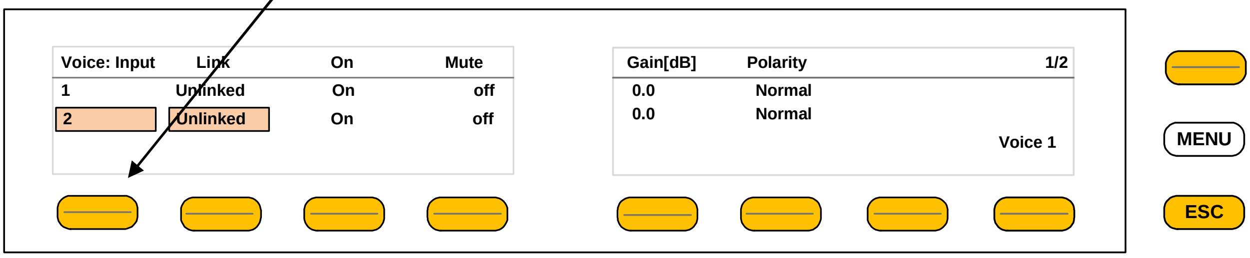


If the voice channels operate in stereo mode : **SYSTEM > Setup > Voice Channel Mode = Stereo,** you are able to **link / unlink** the respective processing blocks (see **AUDIO PROCESSOR > Input)** :



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

When you press **key #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 **key #1** you can toggle the voice channel that is under control :



In **Stereo** mode **key #8** is not active.

The example above demonstrates the general way how to se tup 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 **<Page>** 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 extra manual for details.

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

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

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

Hotkey # Status display of voice channels
1 user defined
2 user defined
3 user defined
4 user defined
5 user defined
6 user defined
7 user defined
8 user defined

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

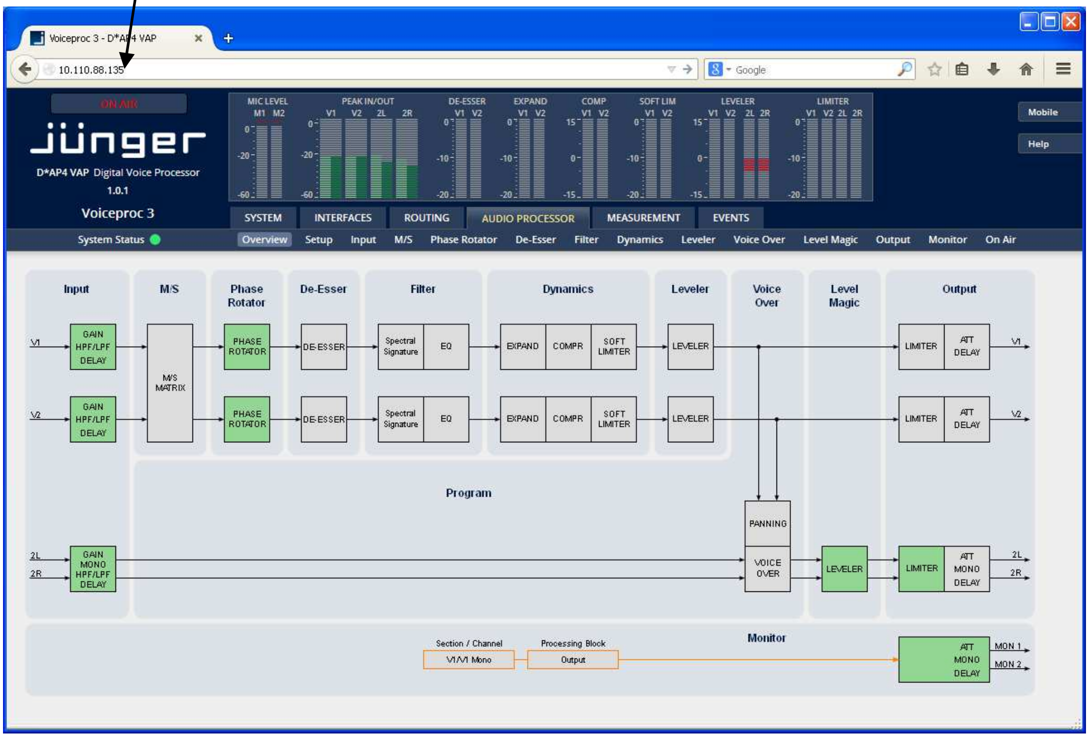
<MENU> opens the o**perating** displays :

Hotkey #
1 <Analog Mic Interface>
2 <ON AIR Profiles>
3 not active
4 <Loudness Meters> (see: AUDIO PROCESSOR > Level Magic > Loudness Mode)
5 <Voice> | parameter settings
6 <Program> | of the 3 signal paths Voice / Program / Monitor
7 <Monitor> | (see AUDIO PROCESSOR > Overview)
8 not active

<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.

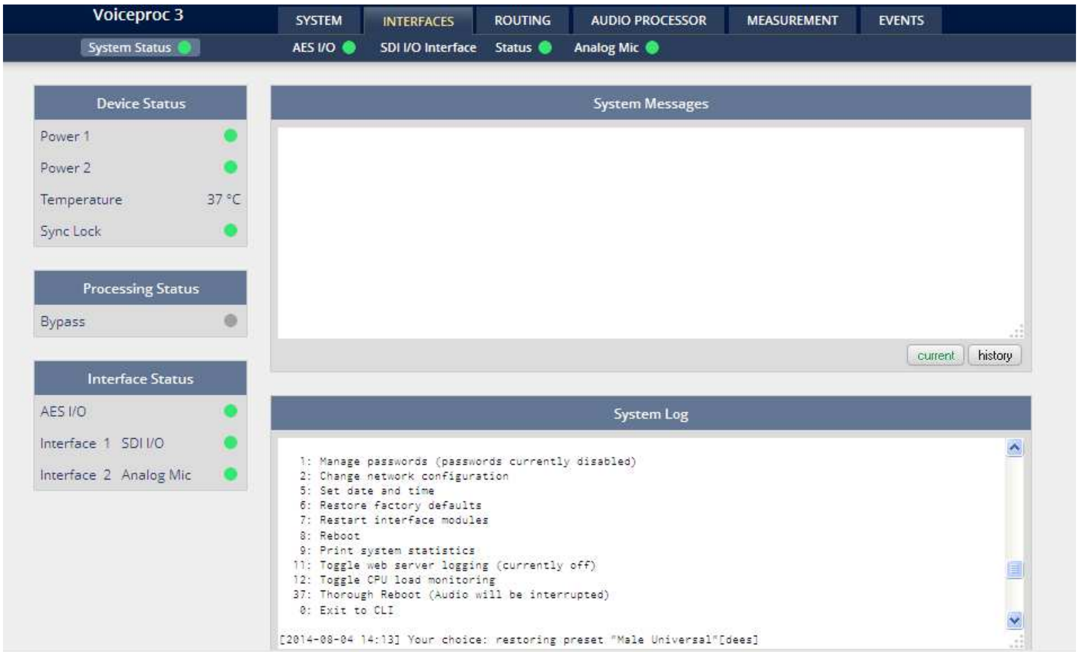
On the following pages we will go through the various panes to perform the basic setup of the device.

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.

Processing Status

- **Bypass** — Turns red if Bypass is activated.

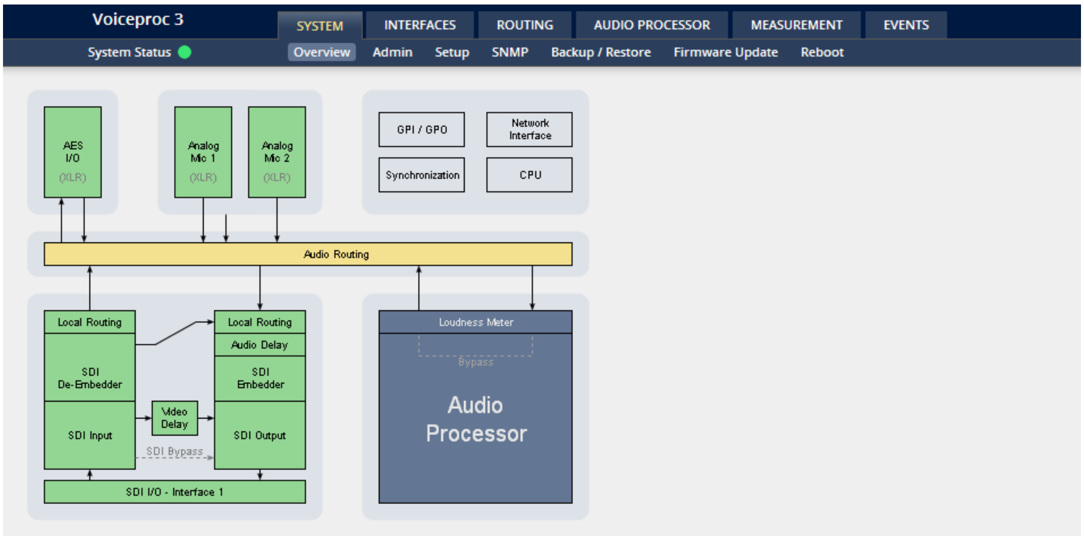
Interface Status

- **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 1 SDI I/O** — If an SDI I/O interface is installed it turns red if no or bad SDI signal is detected.
- **Interface 2 Analog Mic** — Turns red if a problem with the optional mic interface board has been detected.

System Messages — [current / history]
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 : SYSTEM > Admin > Diagnostics > **get diagnostics file**

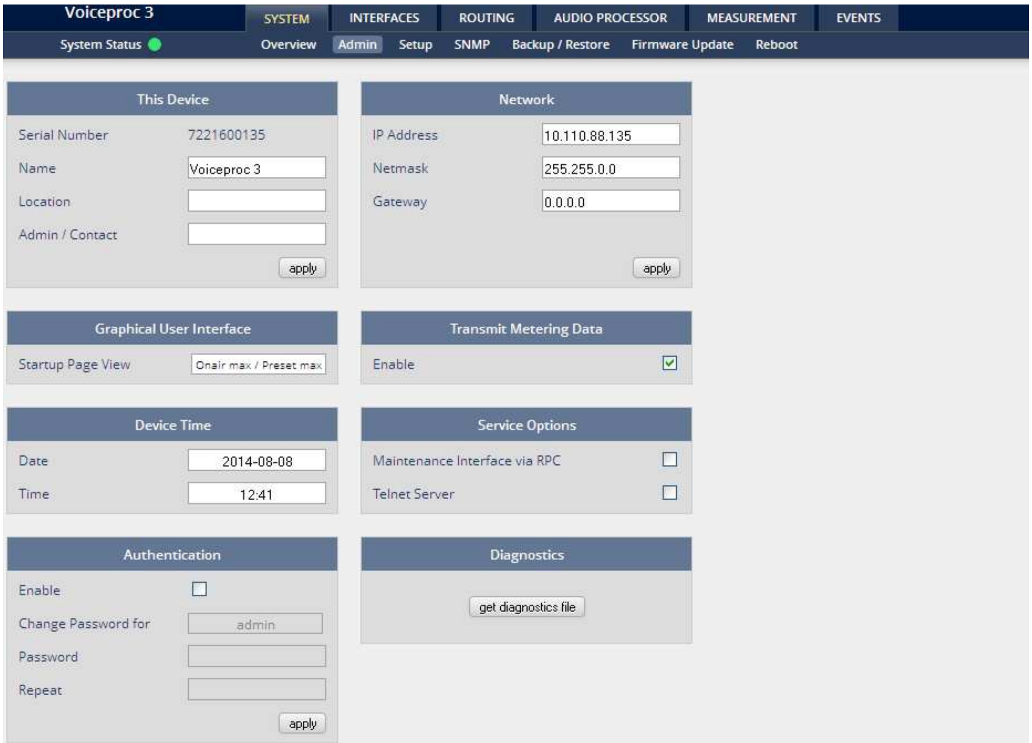
Setup GUI – SYSTEM – **Overview**



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

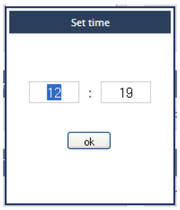
Graphical User Interface — Defines the appearance of the parameter panes regarding preset editor and on air parameter visibility (see below – for preset concept).

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

Device Time — Allows you to set the device clock. At the factory it is set to UTC (Coordinated Universal Time).

Date — If you click into the **Date** input field, you'll get a calendar tool :

Time — If you click into the **Time** input field, you will be able to set the device time :



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

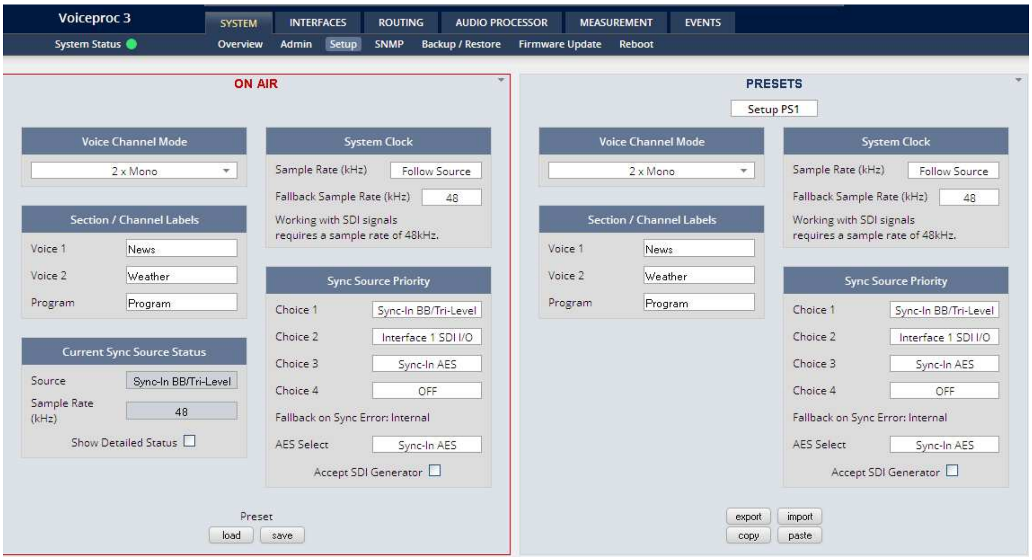
Maintenance Interface via RPC [OFF / ON]]
For administrative use to enable communication with factory tools**.**

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.

Setup GUI – SYSTEM – **Setup**



Voice Channel Mode [2 x Mono / Stereo]
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 se tup 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
Voice 1
Voice 2
Program

Each of the individual voice channels as well as the program path has a name that will be used as a reference for the display of parameters and their setup. You may use names of your choice.

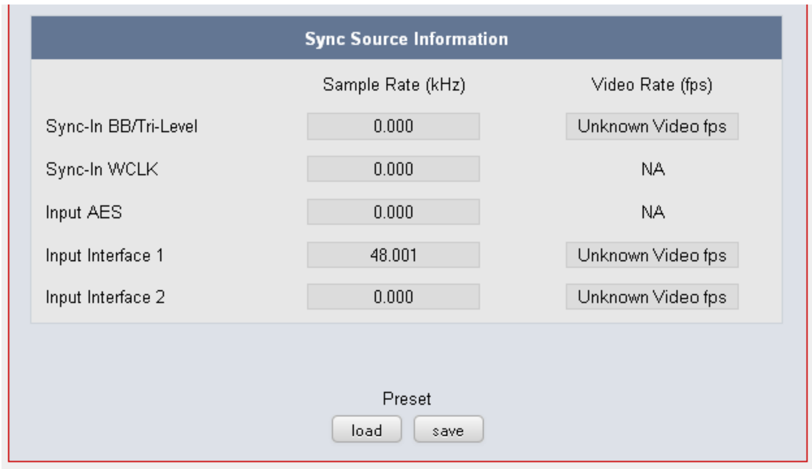
Current Sync Source Status shows the status of the 5 tier sync priority circuit

Source active sync source

Sample Rate measured sample rate

Show Detailed Status [ON / OFF]
If you enable the checkbox you will get this information :

Sync Source Information



System Clock

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

Sync Source Priority

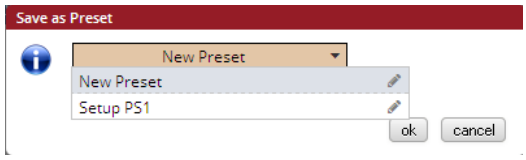
Choice 1 – 4	[OFF / Internal / Sync-In WCLK / Input AES 1/2 XLR / Sync-In Black Burst/Tri-Level]
Fallback on Sync Error :	Internal If the selected sync source is not available the next source will be selected. If none of the pre selected sync source is not available, the source will fall back to the internal clock oscillator.
AES Select	[OFF / Sync-In AES / Input AES 1/2 XLR]
Accept SDI Generator	[ON / OFF] 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**. 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 - the **preset concept** in detail

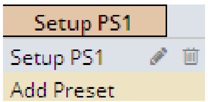
The example above shows the **preset concept** of the **D*AP4 VAP.** It is a general feature of the device and you will come across it in almost every area. For all relevant settings one set of **ON AIR** parameters and a practically unlimited number of **PRESETS** are available. The count depends on 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 on the little pencil icon the preset name turns ***italic*** and you may edit it.

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



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

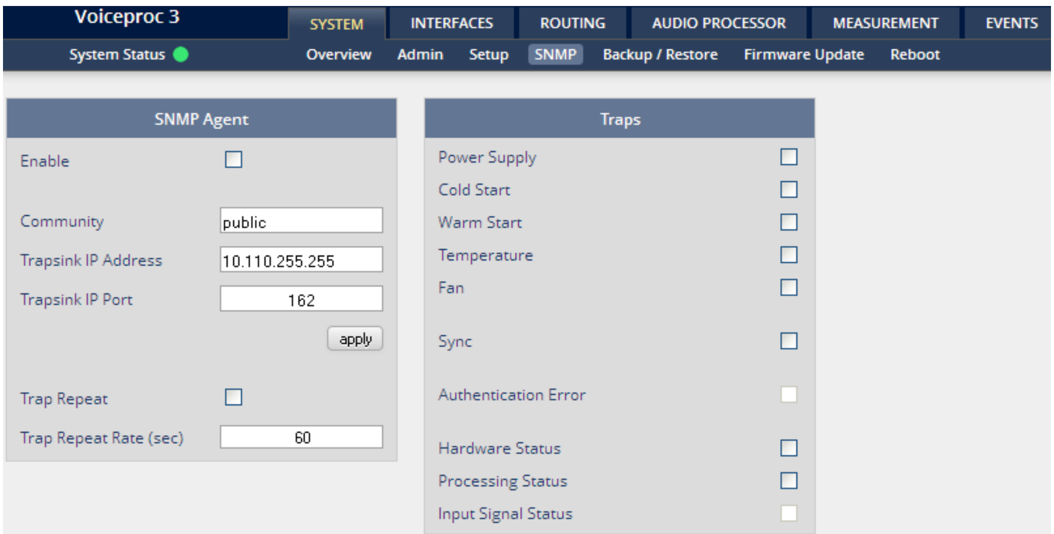
If you have selected the new preset or one of the existing presets indicated by the name displayed at the top, you may edit the parameter values.

Important Note! The presets of the **D*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 **overwrite the actual preset settings!** If you want to keep original values (e.g. from a factory preset) you must simply **copy** the content of 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 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.

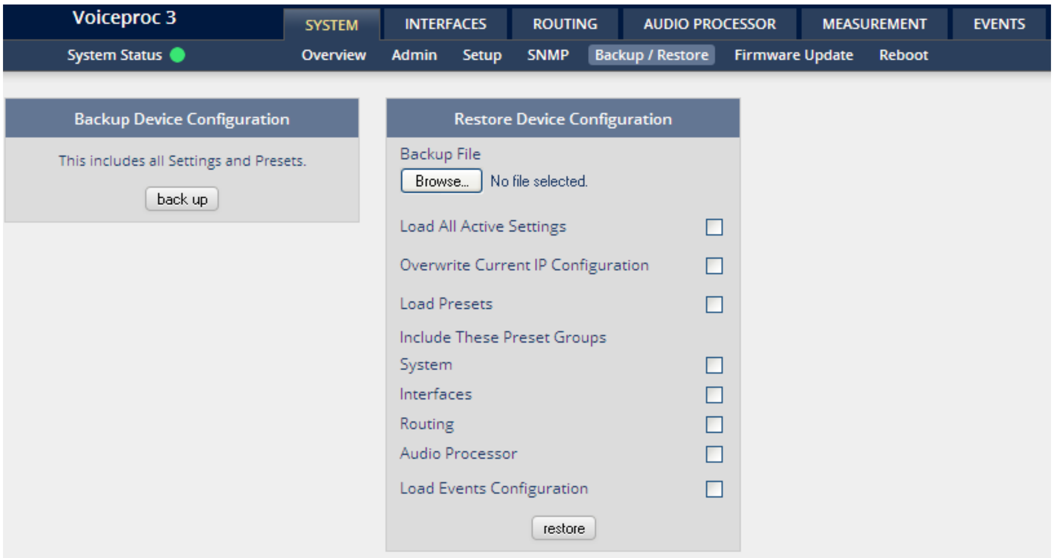
You may also **<export>** or **<import>** the preset content to / from a file.

Setup GUI – SYSTEM – **SNMP**

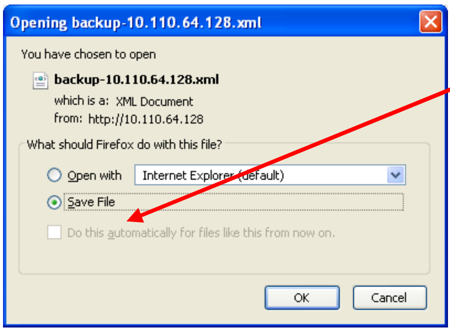


This pane is meant for basic settings of the **SNMP Agent** of the device. If you don't use SNMP based system monitoring, you must not enable the SNMP agent.

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

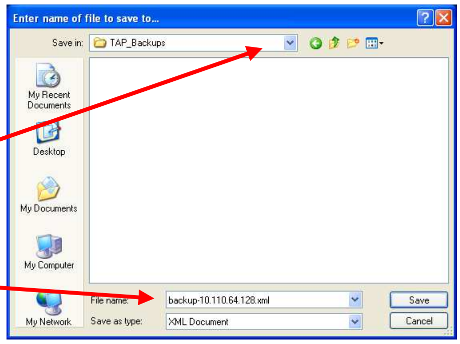


Here you can **back up** the complete **device** and **restore** parts or all of it. If you press **<back up>** the device controller will collect all necessary data and assemble it to an XML file.
Finally you will get a pop up message:



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

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

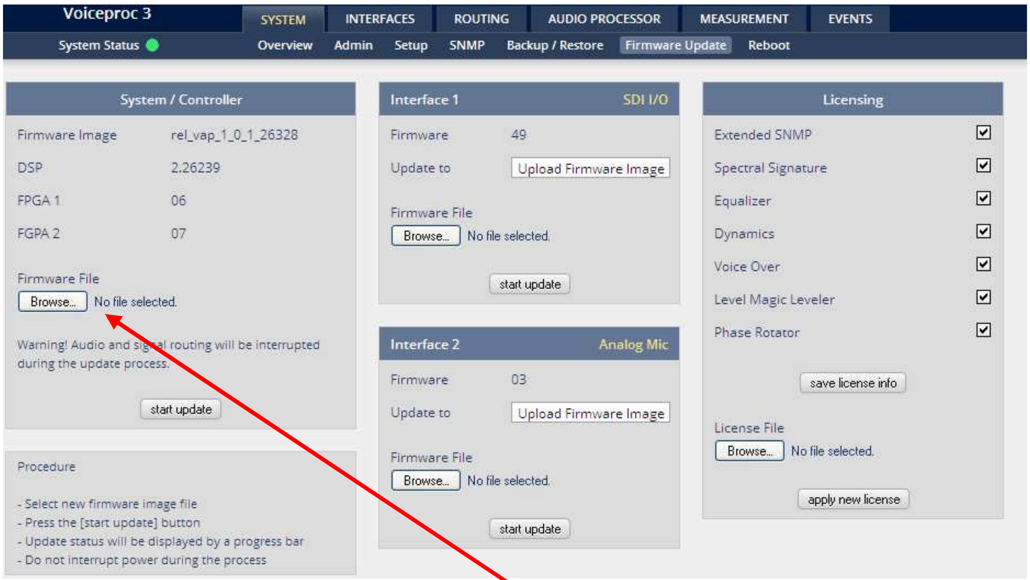


Similar applies to the restore process. You must select the desired backup file which you want to restore and check the necessary option(s) under "Restore Device Configuration".

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.

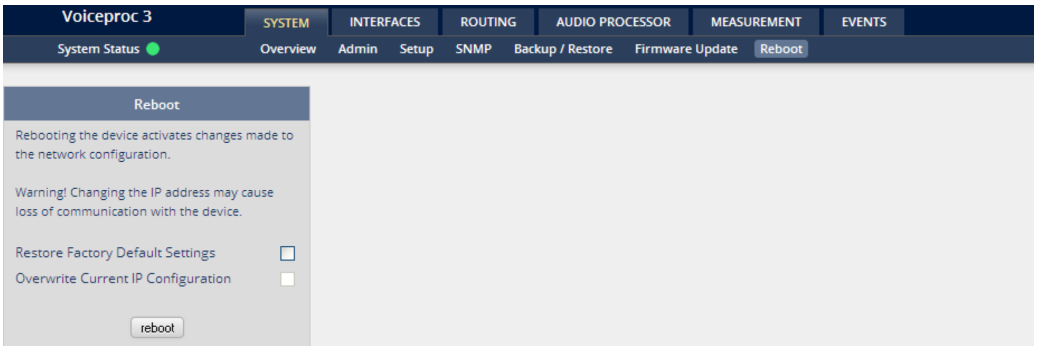


To update the **D*AP4 VAP**, you must **<Browse …>** ● to find the respective firmware file (which you have unzipped before) and press **<start update>**. After finishing the procedure the device will automatically reboot.

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".

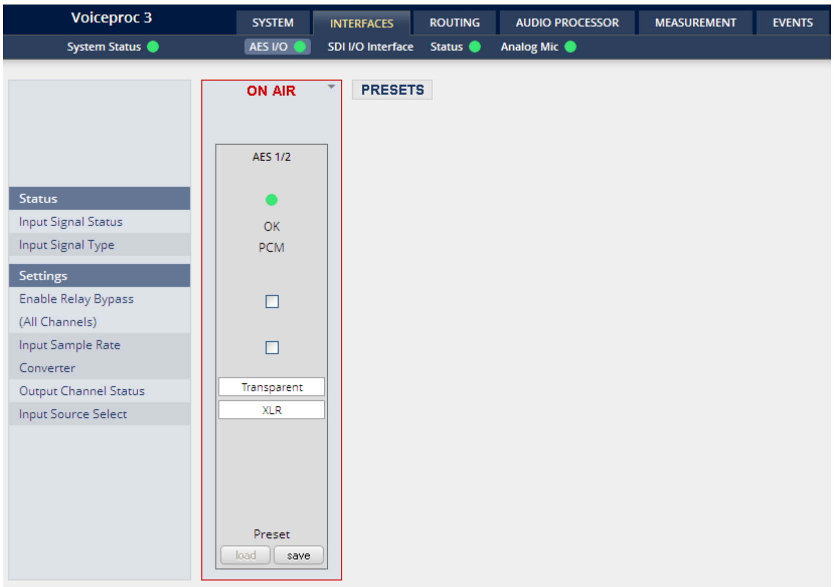
Finally you can see the options of your device which you have bought a license for. When you buy a license you must provide the **license info** file and you in return will get a **new license** file which you must apply to the device here.

Setup GUI – SYSTEM – **Reboot**



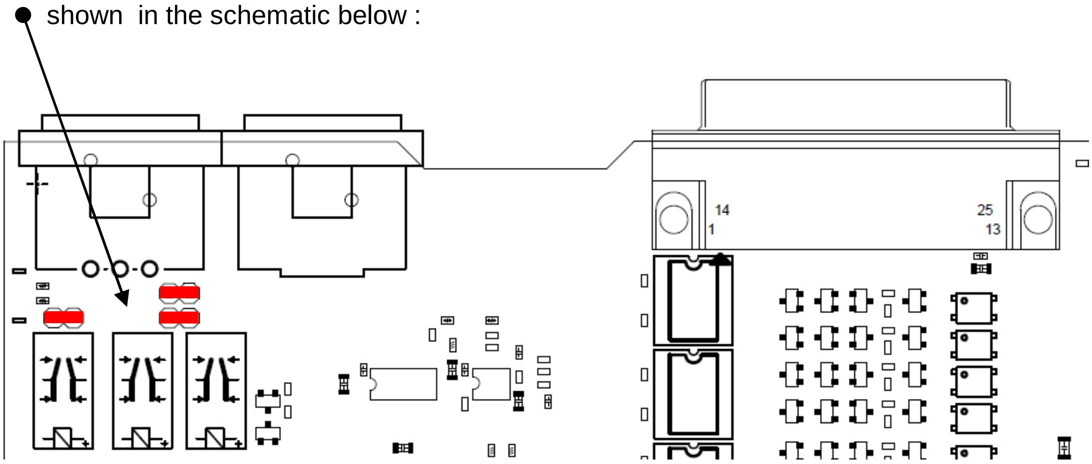
Restore Factory defaults	Will clean up the parameter and preset memory and will initialize all parameters to their factory default values and will reset passwords and turn authentication off.
Overwrite Current IP **IP Configuration**	You may exclude the current IP settings from this process to keep your existing settings.

Setup GUI – INTERFACES – **AES I/O**



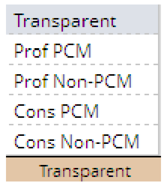
Status	[red / green]
Input Signal Status	[OK / Fail] Fail = no carrier, unlock, cranky [too much jitter]
Input Signal Type	[Mute / PCM / Non PCM] The **Non PCM** (e.g. Dolby encoded signal) status will be retrieved from a logical combination of the Validity flag and the channel status. If the input is not assigned by the **ROUTING** section, its status will not be incorporated into the **System Status**.
Settings	
Enable Relay Bypass (All Channels)	[ON / OFF] See next page for relay bypass setting.
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 like Dolby E.

Important note! The AES relay bypass circuit of the I/Os is activated inside the **D*AP4 VAP.** It is possible to deactivate it if necessary. You must open the cover plate from 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.

Output Channel Status [Transparent / Prof. PCM / Prof Non-PCM / Cons. PCM / Cons. Non-PCM]
The channel status can either be transparent from the input source of the **D*AP4 VAP** or may be overwritten.



Input Source Select [XLR]
The **D*AP4 VAP** has an XLR input only.

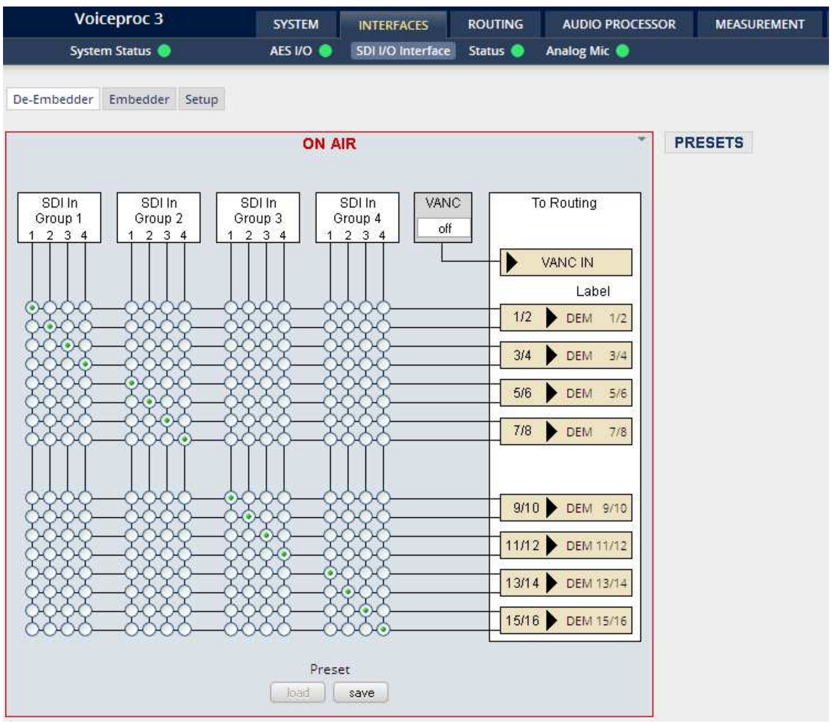
Set up GUI – INTERFACES – **Analog Mic**

Setup

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

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

If the **D*AP4 VAP** is equipped with an optional **SDI** interface the following settings will be available :



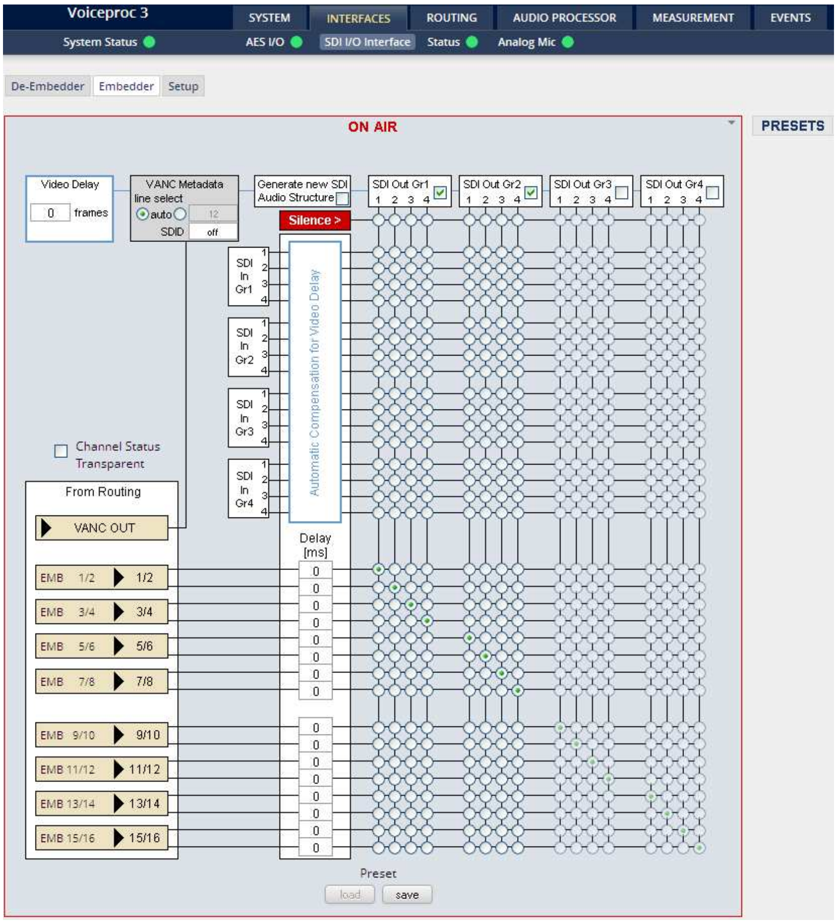
This pane has three more sub panes imbedded.

The **De-Embedder** has a 16 x 16 matrix to allow for any combination of audio signals to be presented to the **D*AP4 VAP** because inside the unit the signal routing is oriented in pairs. I.e. the label **"DEM 1/2"** represents two audio channels selected by the matrix.

An additional Dolby metadata stream may be de-embedded from the SDI.

For details see **SMPTE 2020-2** standard.

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



EMB 1/2 … 15/16
Signal labels from the **D*AP4 VAP** router that shows the origin of the signal pair presented to the embedder.

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.

Generate new SDI Audio Structure
If there is a need to replace the structure of the **Ancillary Audio Data Blocks** you can clean the whole area and generate a new structure. If the option is checked, there will be no signal available at the group output as long as no **SDI Out Grx** is checked.

SDI Out Grx
This check box enables each of the 4 SDI audio groups to be used individually by the embedder. If it is not checked and **"Generate new SDI Audio Structure"** is **not** enabled, the audio data from the input will travel directly from the SDI input to the output.

Silence
Mutes the respective audio channel on the embedder side.

Delay
The inputs of the embedder routing matrix can be taken either from the de-embedder or from the **D*AP4 VAP** in any combination. If audio channels are routed directly from the de-embedder to the embedder and a video delay is introduced between SDI input and SDI output, an audio delay that matches the setting of the video delay is automatically inserted in the audio paths.
For signals coming from the **D*AP4 VAP** routing an **independent delay** per single channel may be used.

Channel Status Bits Transparent

For the signals coming from the **D*AP4 VAP**, you can decide whether the **AES Channel Status Bits** are taken from their source or if you want to generate new ones.
In this case the **Channel Status** will be set to:

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

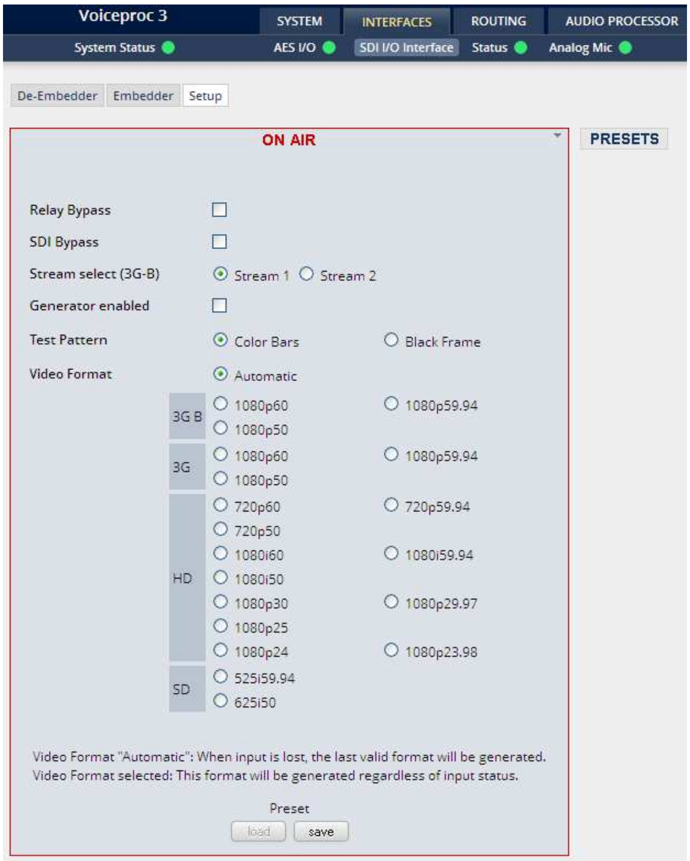
Important note! If you generate a new AES channel status the **Audio Mode** will be automatically set to **Non Audio** (AKA "other") for both channels, if an adjacent pair (1/2, 3/4 …) carries a Dolby E stream for example.

VANC Metadata

The **VANC Dolby Metadata** embedder allows you to embed a metadata stream. You may assign the stream an independent **SDID**.
You can select a line where the metadata must be embedded or leave it in auto mode so the next possible line will be selected.

For details see **SMPTE 2020** and **SMPTE 2020-2** standard.

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



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

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

Generator enabled The video generator may be enabled here. The **video format** it generates depends on the selection below.

Test Pattern — If the Generator is on, it will generate one of the two video test patterns, either black or 100% color bar.

Video Format — If the **Automatic** mode is selected and the Generator is enabled, it turns on if the SDI input signal fails. In this case it will generate the same video format as the previous input signal.
If **“Generator enabled”** is checked and if you have selected one of the **Video Formats** the Generator will be turned on using this format.

Important note! If the **generator is on**, either in manual or in automatic mode, it operates on an internal quartz reference. It is **not possible** to **genlock** it to an external reference or to the SDI input.

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

Video Standard — Display of the video standard detected by the SDI input.

SDI Bypass — Turns yellow if the SDI bypass function is activated.

Relay Bypass — Turns yellow if the power fail relay is deactivated manually.

Test Generator Active — Turns yellow if the Generator is turned on.

Video Delay Enabled — Turns green if the video delay is activated.

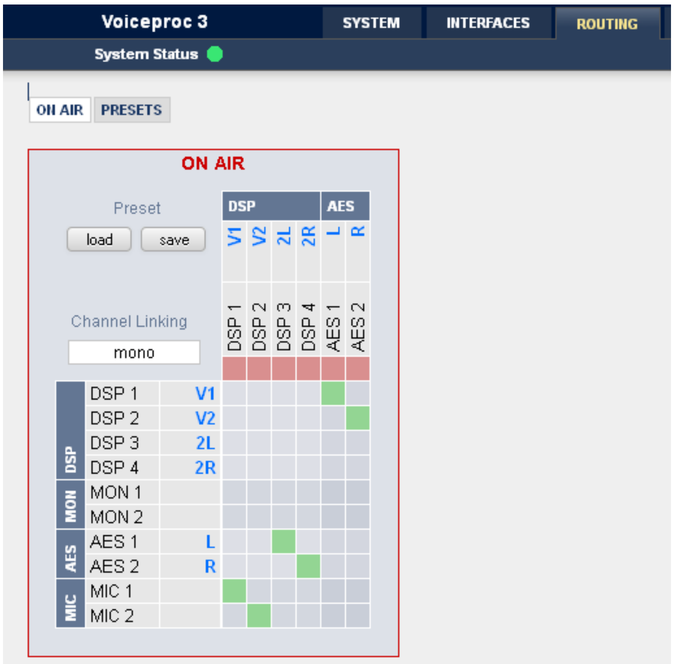
De-Embedder Audio Status — is grey if no audio is present.
Turns green if PCM audio is embedded.
Turns yellow if a non audio signal is present, an additional label shows the kind of signal if it is possible to gather the information.

ARIB B39 — Meta information standard

- **Data Available** — Turns green if ARIB B-39 meta information are detected.
- **Block Error** — Turns red if an error has been detected.
- **Audio Mode** — See **ARIB** Japanese standard "Structure of Inter-Stationary Control Data Conveyed by Ancillary Data Packets" : http://www.arib.or.jp/english/html/overview/doc/2-STD-B39v1_2.pdf

Setup GUI – **ROUTING**

This is the core of the **D*AP4 VAP** as it defines the audio signal flow inside the device :



The routing example shows both mic inputs connected to the voice channel inputs of the DSP. The AES input is connected to the DSP program input while the DSP output is connected to the AES output.

Important Note! If an optional SDI interface is installed the matrix will be expanded by the respective 16 I/Os: DEM 1 – DEM 16 and EMB 1 – EMB 16.

Each functional block of the device has a source- and a destination-label. Additional **blue** labels give an indication of the type of signal that is expected or issued by the respective function block or I/O (e.g. **L/R** for AES or DSP (DSP 1 **1L**, DSP 2 **1R** 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 / 2 indicates the voice channel and 2L, 2R the program path
(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 2R, 2L, V2, V1
where V1 / V2 indicate the voice channel and 2L, 2R the program output.

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

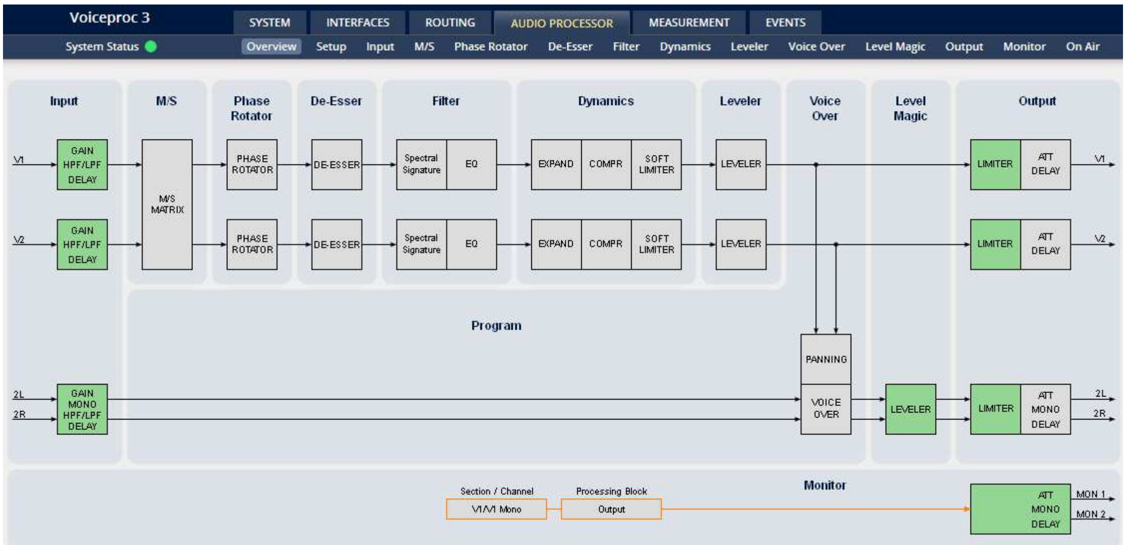
You select cross points by hovering with the mouse over the little squares. The color of the respective squares will change :

Mouse over	Color codes of cross points :
dark blue	Possible new cross point.
orange	You are about to reconnect a cross point.
grey	Cross point is not allowed (i.e. routing will cause a loop and will not be performed therefore).
red	You are about to disable a cross point

Clicking into the small squares will execute the routing.

Setup GUI – AUDIO PROCESSOR - **Overview**

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

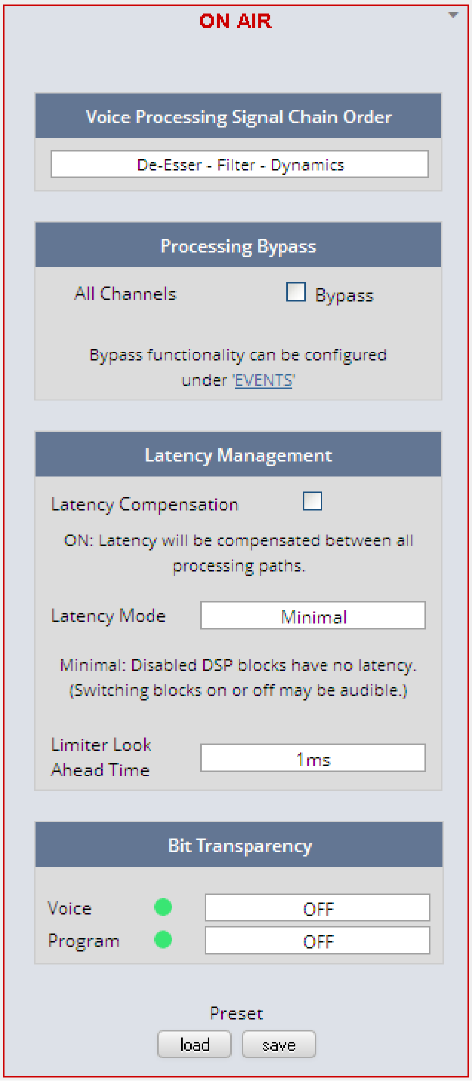


The processing blocks in use, which may be activated from their individual setup panes, will be indicated in green. I.e. blocks shown in grey are not activated by the user.

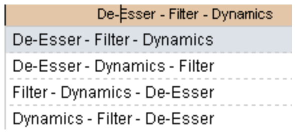
The order of the blocks depends on the setup of the audio processor (see next page).

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.

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 :

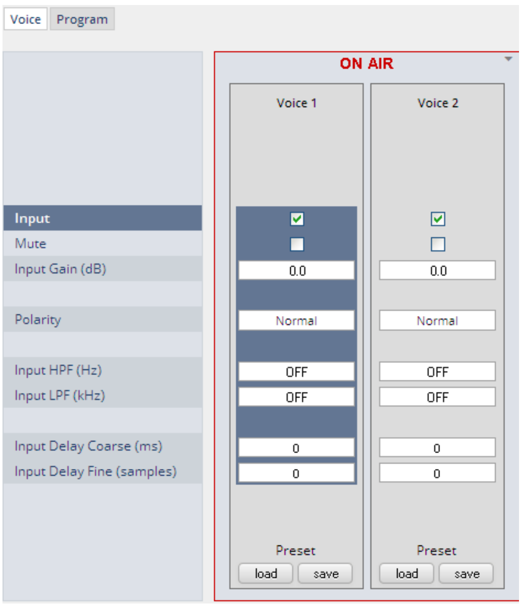
	44.100 kHz	48.000 kHz	88.200 kHz	96.000 kHz
Base Latency				
AES IN to AES OUT	3,33	3,06	2,12	2,04
Mic IN to AES OUT	4,25	3,9	2,59	2,44
Additional Latencies				
Spectral Signature	2,9	2,66	2,9	2,66
Dynamics Look Ahead	2	2	2	2
Limiter Look Ahead Long	1	1	1	1

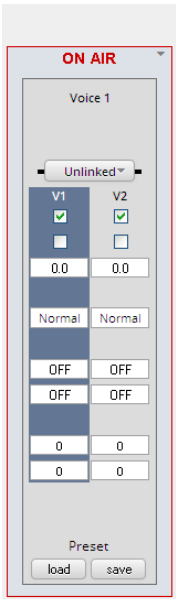
Base latency consists of 1 ms Limiter Look Ahead Time and all system inherent processing and input/output delays.
Activating sample rate converters will add additional latency (< 1ms).

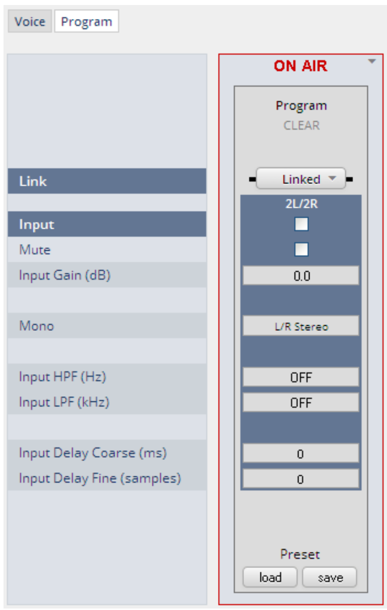
Setup GUI – AUDIO PROCESSOR – **Input**

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 :





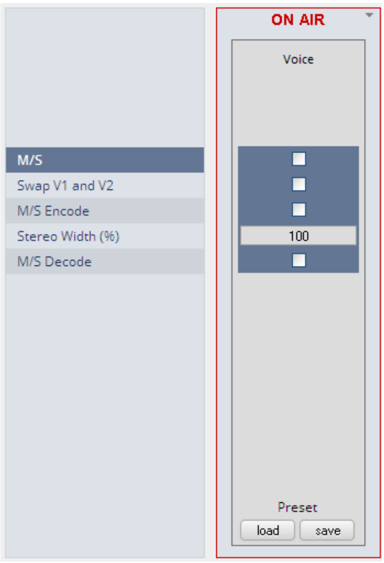


Link	[Unlinked, Linked] This function will only be available if the voice channel is set for "Stereo" (see SYSTEM > Setup > Voice Channel Mode).
Input	[Enable / Disable] Enables or disables the input section
Mute	[ON / OFF]
Input Gain (dB)	[-80.0 … 0.0 … 20.0]
Polarity (voice input)	[Normal / Inverted]
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).

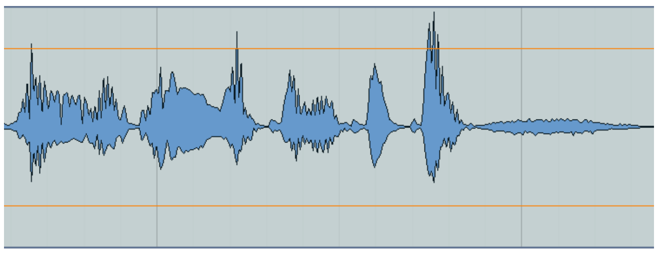
The **M** (mid) and **S** (side) signals may be processed to change the stereo width from 0% (mono) to100% (stereo) to 200% (excess width). If you want to process a L/R stereophonic signal you must first encode it to M/S and back to L/R after width correction.



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]

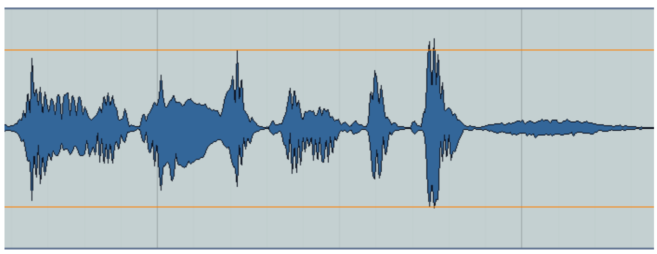
Setup GUI – AUDIO PROCESSOR – **PHASE ROTATOR**

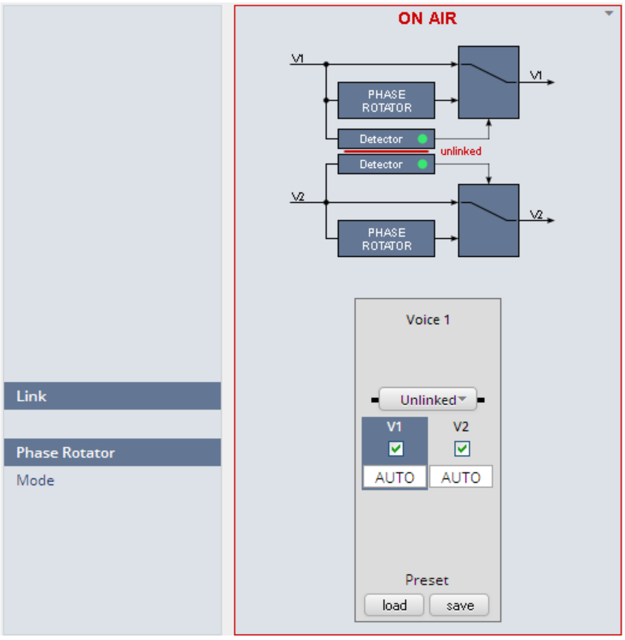
When working with human voice, one regularly experiences issues with imbalanced waveforms. Imbalanced in this context means that the positive or negative half of the alternating signal carries more power than the other.
The problematic result of this type of imbalance is unnecessarily applied dynamics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



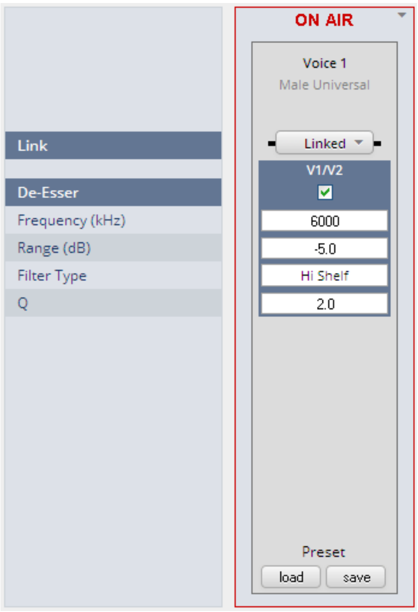


Each microphone channel has an independent Phase Rotator.

Here the display for stereo operation but unlinked

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

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

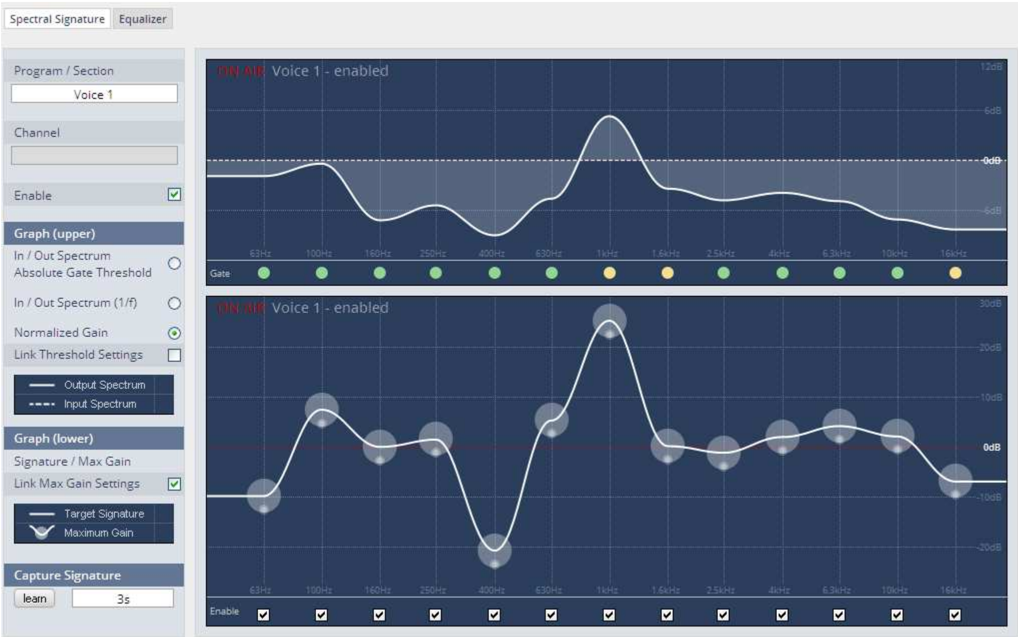


Link	[Unlinked, Linked] For stereo operation you may link the setup parameters for both voice channels.
De-Esser	[ON / OFF]
Frequency (Hz)	[1000 … 3000 … 16000]
Range (dB)	[-20.0 … 0.0]
Filter Type	[Peak / High Shelf]
Q	[0.4 … 1.0 … 8.0]

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 **– 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 [Voice 1 / Voice 2 / Preset]
Selects the source for which Spectral Signature will be displayed.
This selection depends on the Voice Channel Mode
(see SYSTEM > Setup > Voice Channel Mode).
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.
You can select which channel must actually be displayed (see also LI

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

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

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

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

Link Threshold settings [ON / OFF]
The absolute gate threshold can be set individually for every single band. However, in most cases this is not necessary. Checking this box links all gate thresholds. 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 (dB) [0 … 12]
Spectral Signature does not work with an absolute level reference. Its frequency response is based on level differences between bands only. Thus a signature is only represented on a relative graph showing the level positions related to the neighboring bands. In consequence, having a straight line does not mean Spectral Signature is not doing anything or is in a 'neutral' status. A straight line would cause Spectral Signature to modify the input signal towards the frequency response of white noise which is, in most cases, not desirable. 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.

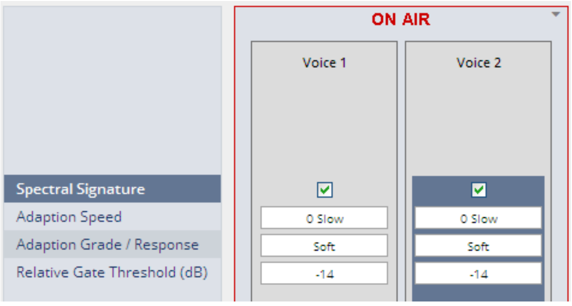
Link Max Gain Settings [ON / OFF]
Instead of dialing in all max gain settings individually per band, this link function is a handy tool for basic setup. This connection is absolute, differences between bands will be overwritten.

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

Capture Signature

<Learn>
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 time [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).



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.

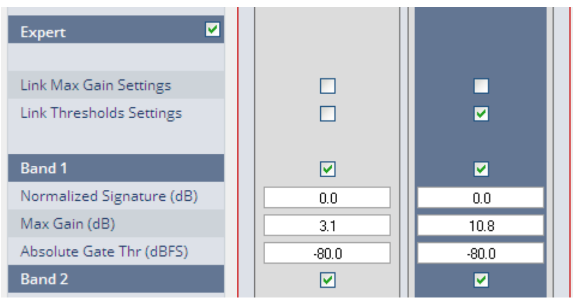
Adaption Grade / Response [Soft / Mid / Hard]
In order to achieve a stable and natural behavior, the intensity of the gain change needs to process according to a response curve. This curve is defined by a ratio. A high ratio means that a difference of 5 dB results in a gain change of almost the same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

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

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]

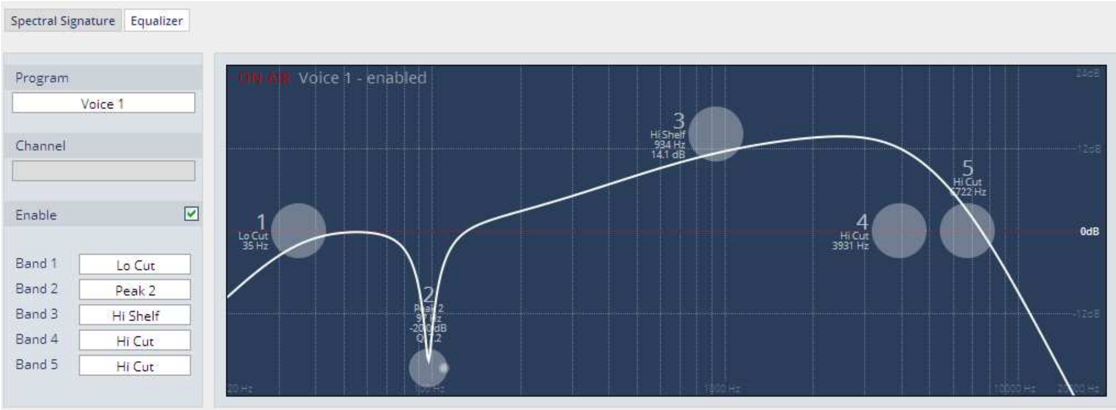
Normalized Signature level [-40.0 … 0 … 40.0]

Max Gain [0.0 … 12.0

Absolute Gate Threshold [-84.0 … 0.0]

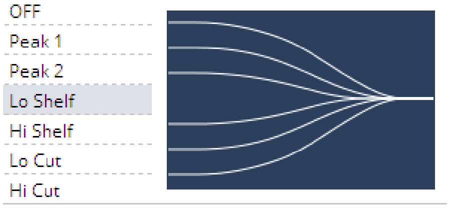
Band 2 … 16 similar parameters as Band 1

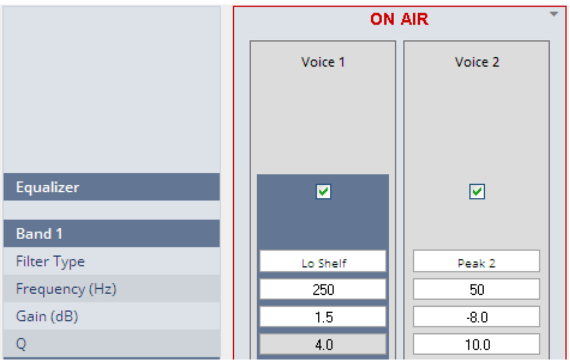
Setup GUI – AUDIO PROCESSOR – Filter **– Equalizer**



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	[Voice 1 / Voice 2 / Preset] or [Voice / 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	Applicable only if SYSTEM > Setup > Voice Channel Mode = Stereo to select which channel must actually be displayed.
Enable	[ON / OFF] Same function as <Equalizer>
Band 1 … 5	filter characteristic will be selected by this pop up :





Equalizer [ON / OFF]

Band 1

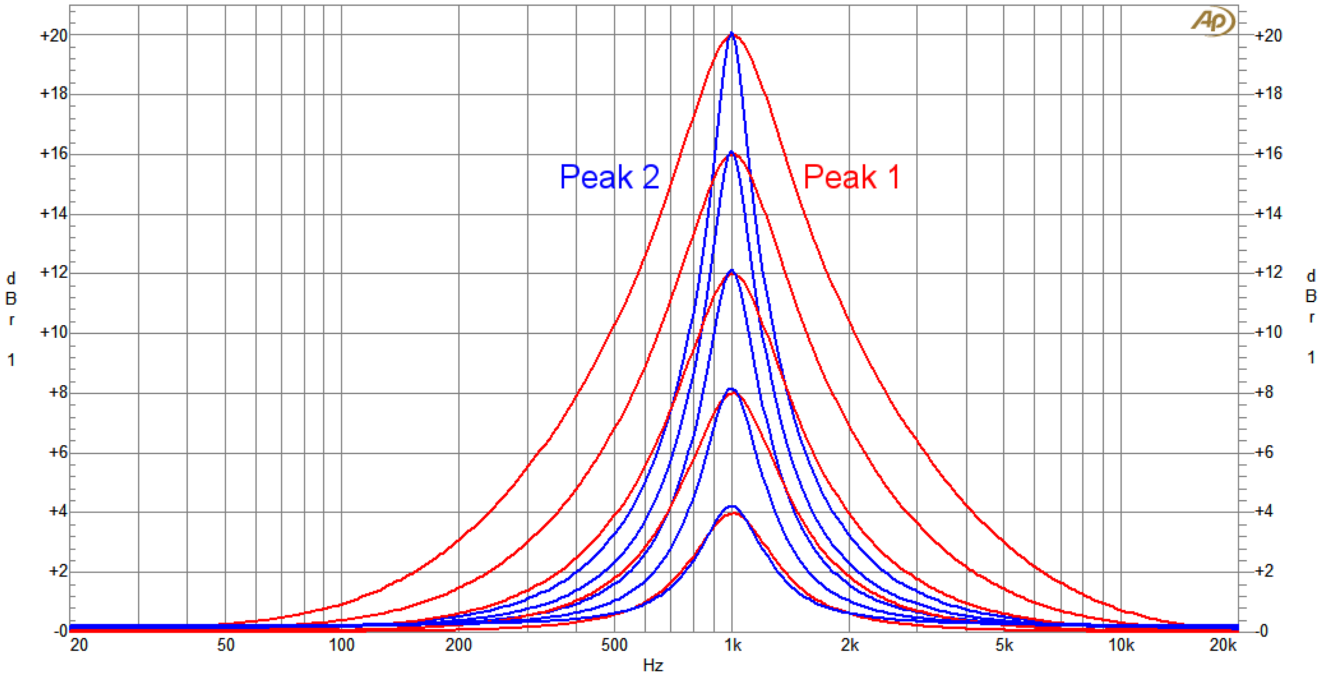
Filter Type	will be selected by the pop up above
Frequency (Hz)	[20 … 20000]
Gain (dB)	[-20.0 … 20.0]
Q	[0.4 … 4.0]

Band 2 … 5 same parameter set as Band 1

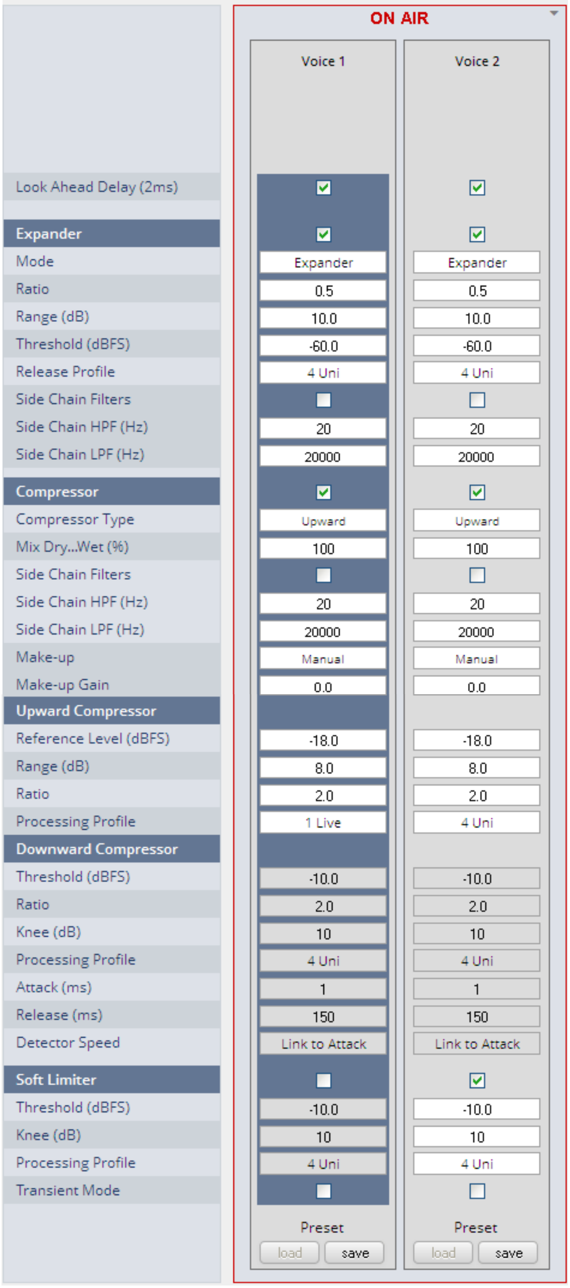
The **EQs** offer two different peak modes :

Peak 1 : The bell curves of the **Peak 1** filter features constant quality (Q) over gain. Q is defined at -3 dB 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 6-8 dB of gain.



Setup GUI – AUDIO PROCESSOR – **Dynamics**



Look Ahead Delay (2ms)	[ON / OFF]
Expander	[ON / OFF]
Mode	[Expander / Gate]
Ratio	[0.0 … 0.9]
Range (dB)	[0.0 … 40.0]
Threshold (dB)	[-80.0 … -10.0]
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]

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

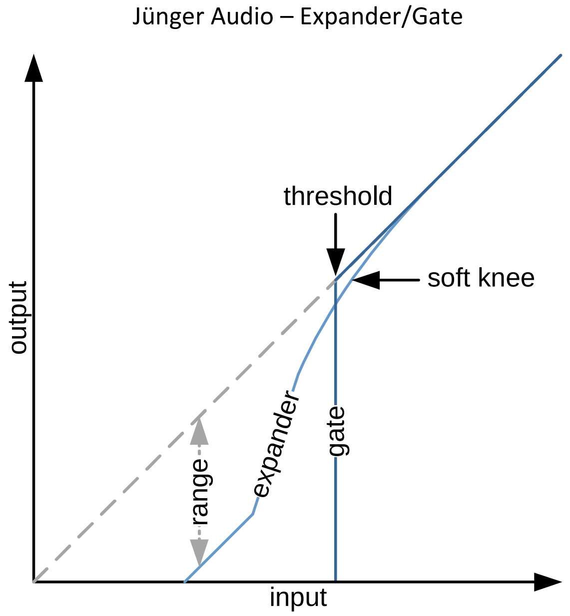
Some of the processing modules (De-Esser, Compressor, Expander/Gate) feature Side Chain Filters to shape the audio signal that feeds their detection system. It consists of independent high and low pass filters with tunable cutoff frequencies. Those filters are not audible within the actual signal chain. 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 -40 dB. Soft knee response with a transition range of 6 dB 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 1 dB below threshold will result in an output level of 2 dB below threshold. In the same way an input level of 4 dB below threshold results in an output level of 8 dB 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)

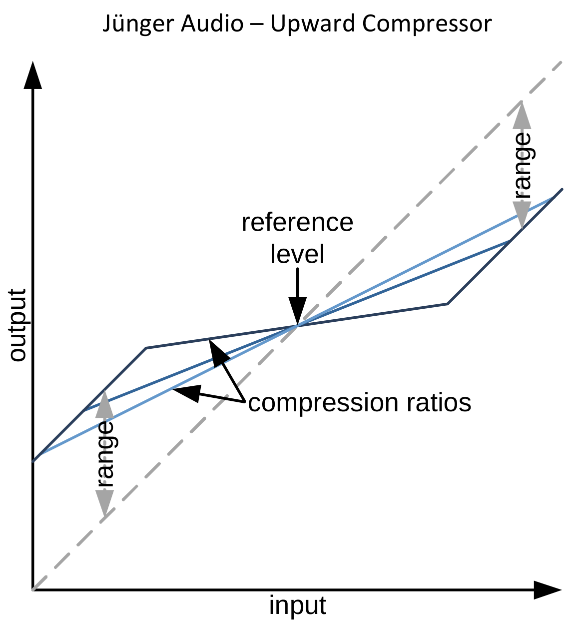
Compressor Type — The compressor features two different approaches to dynamic processing. In Upward mode all signals below reference level are amplified according to the ratio and range settings, all signals above reference level are reduced in the same way. This is the 'classic' approach of earlier 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 0.1 dB.

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 0 dBFS, 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 4 dB above threshold will result in an output level of 2 dB above threshold. In the same way an input level of 8 dB above threshold results in an output level of 4 dB above threshold and so on.

Upward compression ratio

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

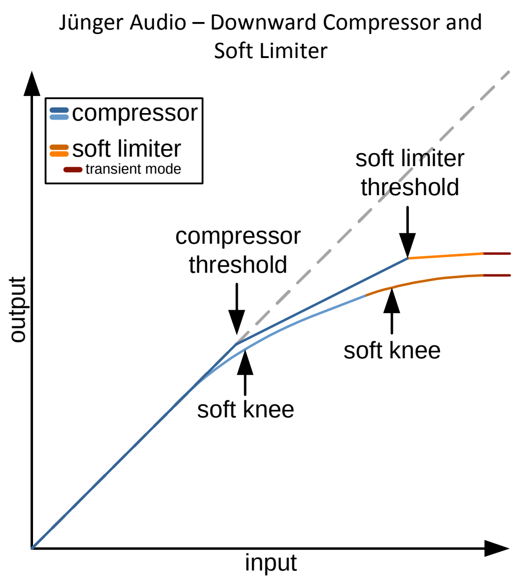
Signals above threshold are processed, signals below pass unaffected. Please be aware that this is only true when Knee is set to '0'.

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 8.6 dB 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 -9 dBFS) 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.

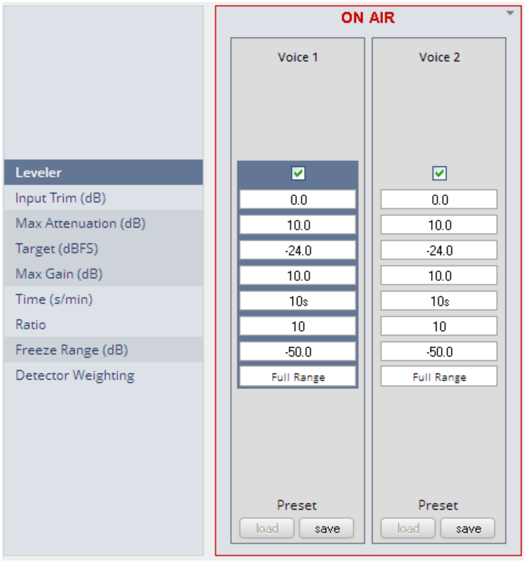
Threshold Signals above threshold are processed, signals below pass unaffected. Please be aware that this is only true when Knee is set to '0'.

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.

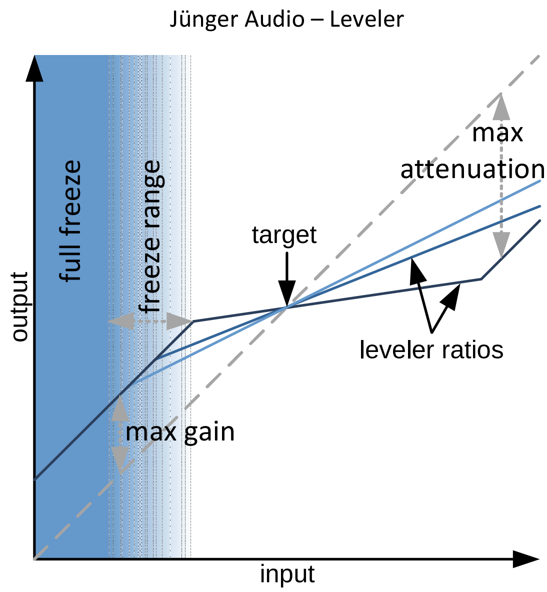
Transient Mode With the transient mode enabled the response curve of the Soft Limiter is reduced down to infinity to one. This is useful in settings where a wide knee may not have run out fully before full scale is reached.

Setup GUI – AUDIO PROCESSOR – **Leveler**



Leveler	[ON / OFF]
Input Trim (dB)	[-80.0 … 0.0 …. 20.0]
Max Attenuation (dB)	[0.0 … 40.0]
Target (dBFS)	[-50.0 … 0.0]
Max Gain (dB)	[0.0 … 40.0]
Time (s/min)	[1s … 30s … 2min]
Ratio	[1 … 40]
Freeze Range (dB)	[-60.0 … 0.0]
Detector Weighting	[Full Range / Proximity / Loudness]

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



Input Trim 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 Defines the maximum attenuation or damping of the Leveler process for signals that exceed the target level.

Target 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 Defines the maximum amplification of the Leveler process for signals that are below the target level.

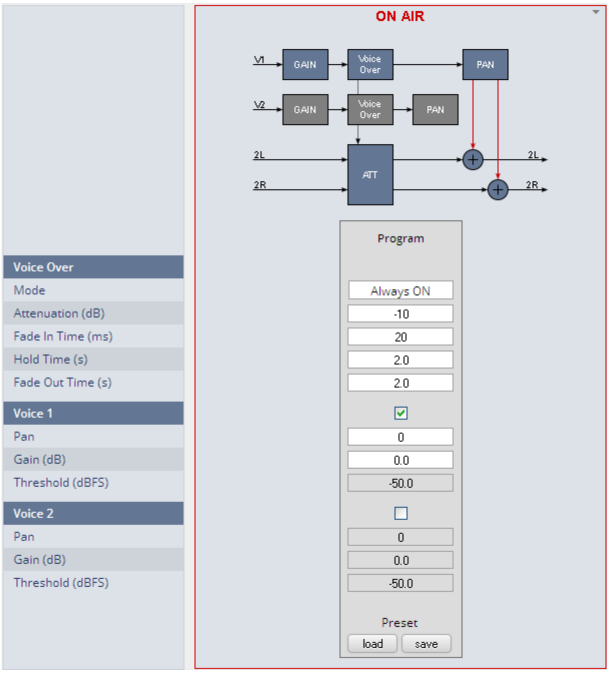
Time This defines the timing for the Leveler to reach target. Of course this is not an absolute value as it depends on the input level, signal structure, ratio and necessary amount of gain change.

Ratio 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 range	Instead of an absolute freeze level we implemented a Freeze Range within the Leveler where speed is constantly reduced. At the lower end of the range, the leveling process comes to a full stop. The Freeze Range begins at the maximum gain point. Example: 1. Target is set to -10 dBFS 2. Max Gain is set to 10 dB 3. Freeze Range is set to 10 dB 4. Freeze Range starts at -20 dBFS, full stop is reached at -30 dB Hint: Setting Freeze Range to 0 reduces its function to a freeze level depending on the Max Gain.
Detector Weighting	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.

Setup GUI – AUDIO PROCESSOR – **Voice Over**

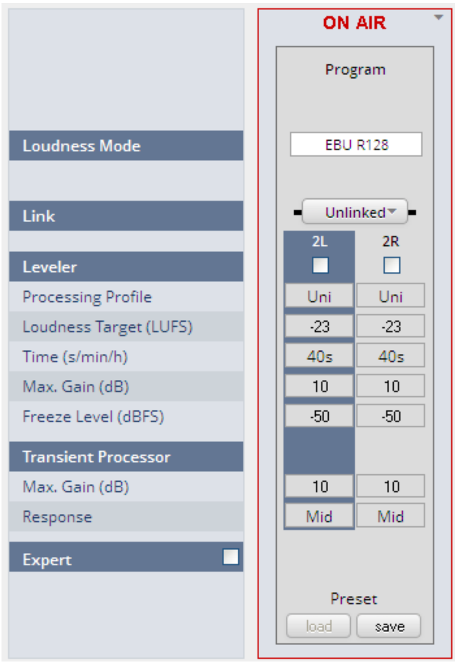
The voice over section allows for manual (mixing) / automatic (ducking) of a voice channel over the program feed. The dynamic schematic in the top of the pane shows the signal flow :



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

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).



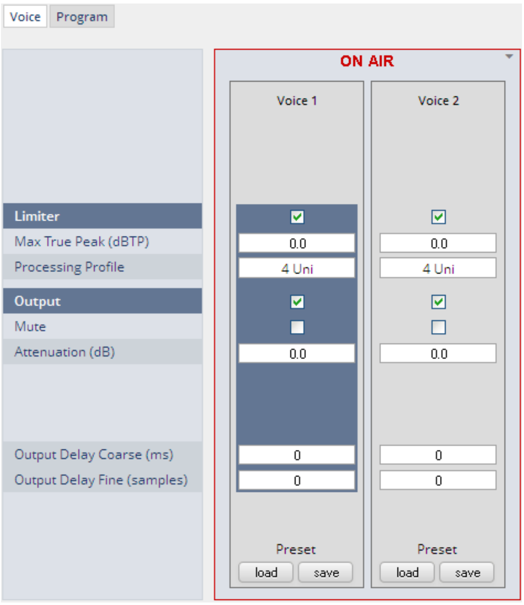
Loudness Mode	[Level / ITU BS.1770-1 / -2 / -3 EBU R128 / ARIB TR-832 / ATSC A/85 (2011) / ATSC A/85 (2013) / Free TV OP-59 / Portaria 354)
Link	[Linked, Unlinked] The program path may be unlinked for dual mono operation.
Leveler	[ON / OFF]
Processing Profile	[Live / Speech / Pop / Uni / Classic]
Loudness 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]

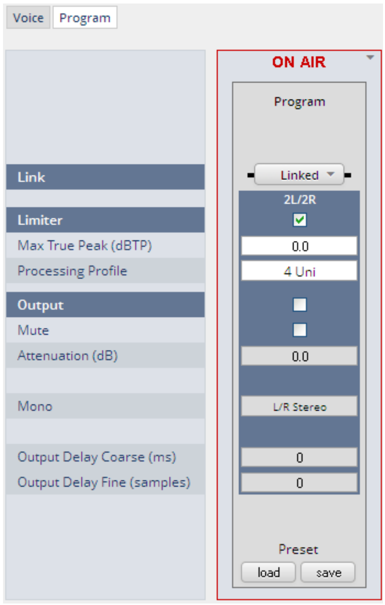
Expert	[ON / OFF]
Clear Processing History	<clear>
Initial Dynamic Gain (dB)	[-40 … 0 … 15]
AGC Recovery	[Normal / Fast]
Low Level Behavior	
Processing Threshold (dBFS)	[-80 … -70 … -20]
Below Threshold Mode	[Release / Hold]

For details regarding LevelMagic parameters see the bulletin : "Junger processing parameter description" 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.





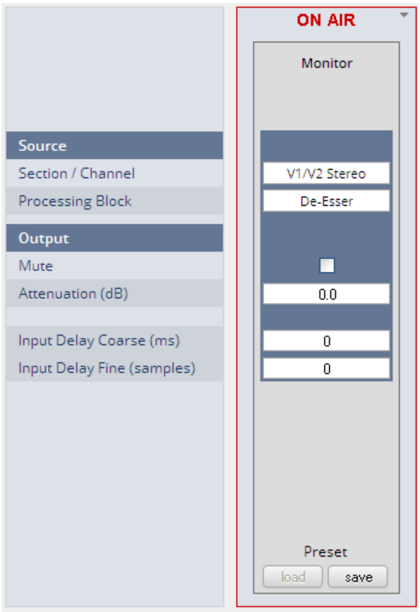
Link	[Unlinked / Linked] For voice channel only available if it is in stereo mode.
Limiter	[ON / OFF]
Max True Peak (dBTP)	[-20.0 … 0.0]
Processing Profile	[0 / 1 Live / 2 Speech / 3 Pop / 4 Uni 5 / 6 Classic / 7 / 8 / 9]
Output	[ON / OFF]
Mute	[ON / OFF]
Attenuation (dB)	[-80.0 … 0.0]
Mono	[L/R Stereo / 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 a monitor facility that may be connected to the function blocks of the audio processor (DSP).

For the example below the monitor is connected to both voice channels [V1/V2 Stereo] and to the output of the De-Esser section :



Source

Section / Channel [V1/V2 Stereo / V1/V1 Mono / V2/V2 Mono / Program]

Processing Block

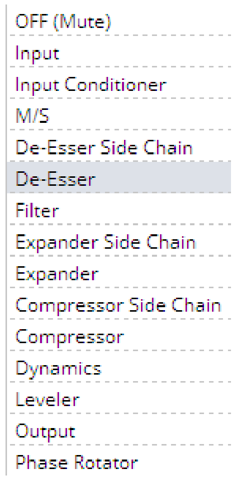
Output

Mute [ON / OFF]

Attenuation (dB) [-80.0 … 0.0]

Input Delay Coarse (ms) [0 … 2000]

Input Delay Fine (samples) [0 … 2000]

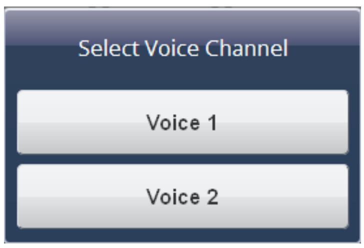


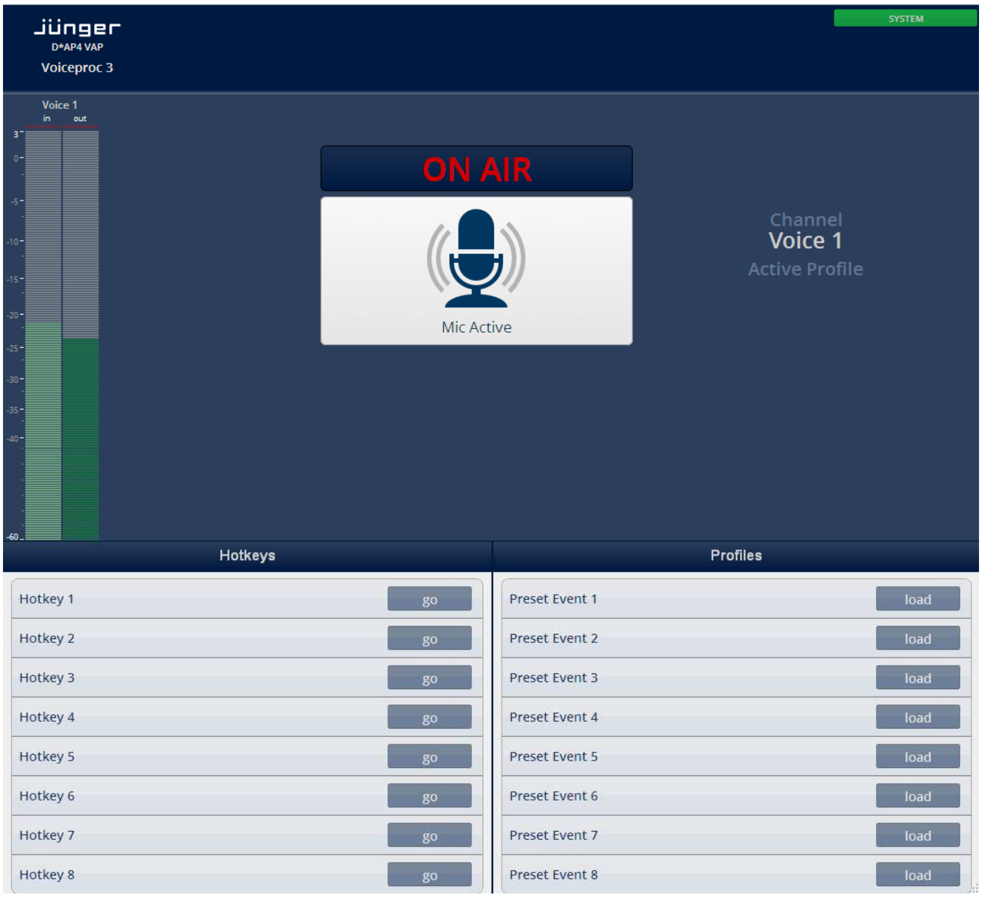
Setup GUI – AUDIO PROCESSOR – **On Air / Mobile UI**

The **V'*AP** provides an extra **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 network integration of the **D*AP4 VAP via** a WLAN.

First you will get a selection of one of the two voice channels that you may control, if the device is setup for **2 x Mono** mode (see SYSTEM > Setup Voice Channel Mode = 2 x Mono).
In case of 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 left the default names assigned as Factory defaults.



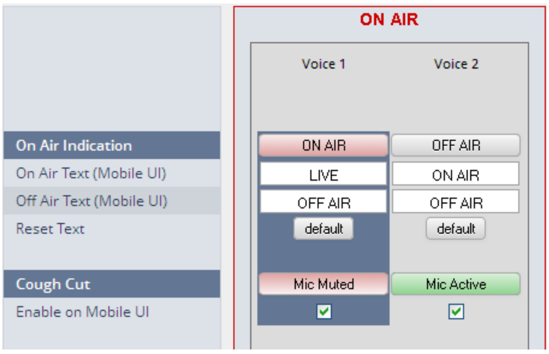


The content of this **operating UI** duplicates useful controls of the V*AP. You may activate one of the Hotkeys which are assigned to the **X*AP RM**$_1$ remote panel which in turn maybe setup for trigger actions of the event manager (see EVENTS for details).

But you may directly trigger one of the **Preset Events** in the right column at the bottom of the screen as well. The mobile UI will only display preset events which contain processing relevant parameters called **"Profiles"**.

On a touch screen (or via left mouse button) you may simply press the icon in the middle of the screen to temporarily mute that channel for a cough cut function. This will be indicated on the button.

The text at the top of the button 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

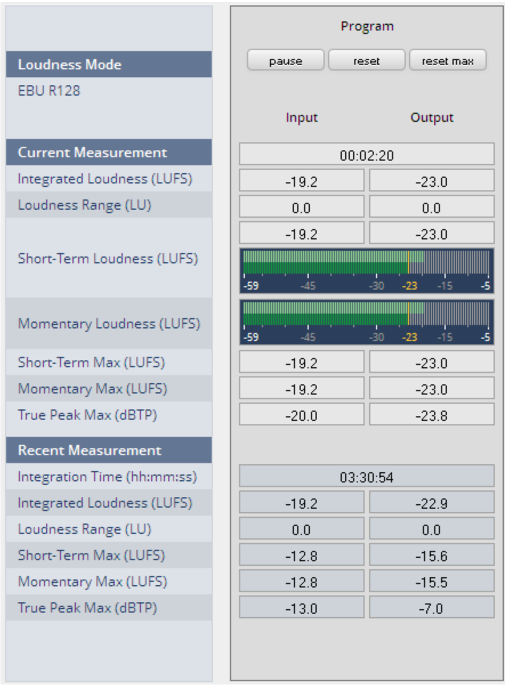
Reset Text will restore the default text for both the on air and the off air display

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

Enable on Mobile UI [ON / OFF]
if turned OFF the control via mobile UI is locked.

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.



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)**

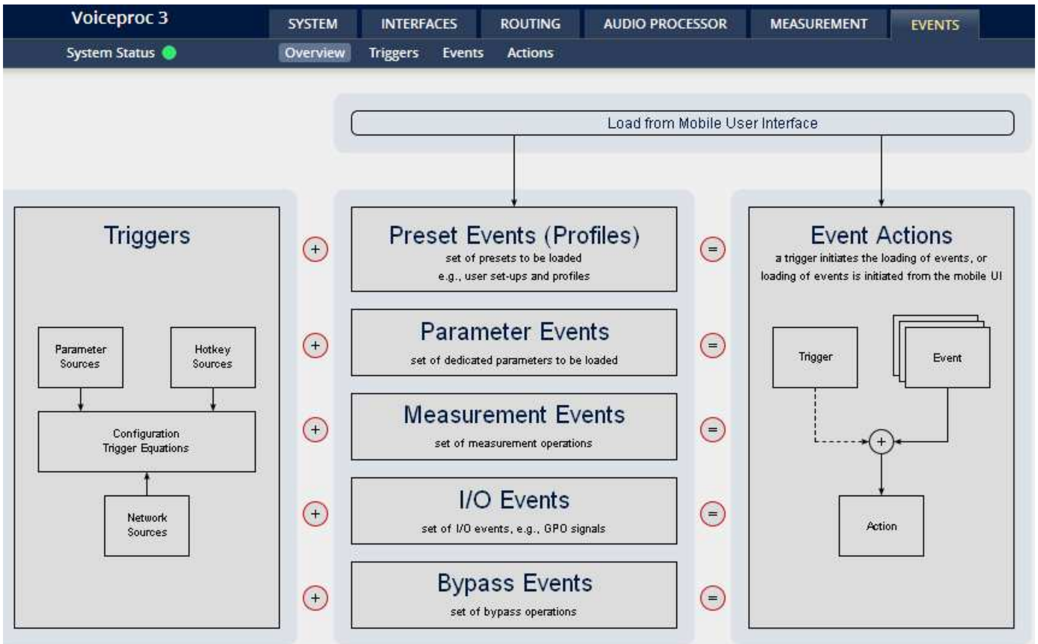
For the terms and details of loudness measurement we would ask you to consult the respective standards like EBU R128 issued by the EBU (EBU_tech3341, 3342, 3343, 3344). You will also find explanations here :

"Junger processing parameter description" 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)**, 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** :



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

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

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

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

Preset Events	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 Bypass of function blocks

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

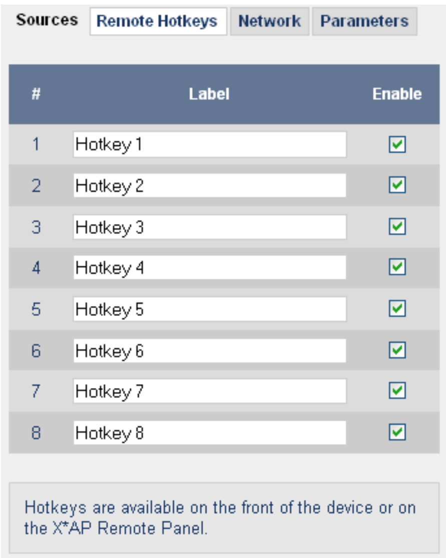
An action runs like a flip-book inside the **D*AP4 VAP**. This powerful technology spans from simply recalling a certain parameter over narrator specific parameter combinations (household name: "Preset") to the complete reconfiguration of the **D*AP4 VAP** including all signal routing, processing parameters 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 (see EVENTS > Actions > Bypass Action).

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 :

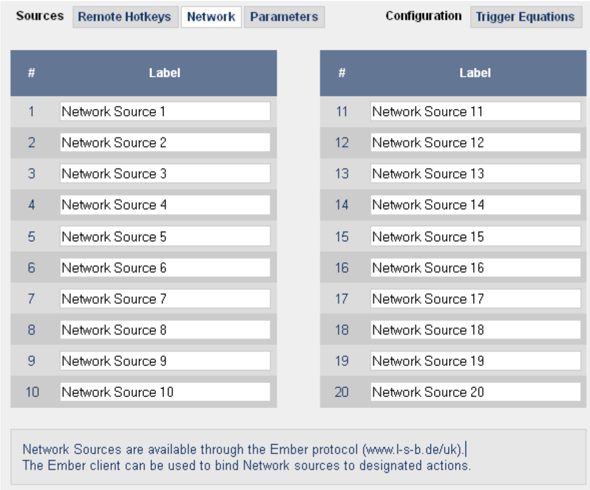


#	The number of the Hotkey on the **X*AP RM1** remote panel, counting from left to right.
Label	Each Hotkey may have a label that appears in the display of the **X*AP RM1** remote panel above that button.
Enable	[ON / OFF] If you turn it off the respective Hotkey on the **X*AP RM1** remote panel becomes 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**

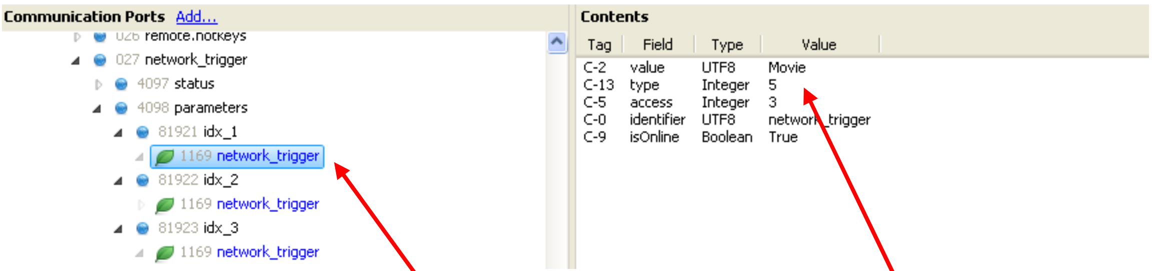
Network triggers are based on the **EmBER+** protocol. See [code.google.com/p/**ember**-plus/](code.google.com/p/ember-plus/) for details. The **D*AP4 VAP** receives such triggers over the TCP/IP network. The triggers are issued by a remote device or a broadcast automation system. You may assign these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



The 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.

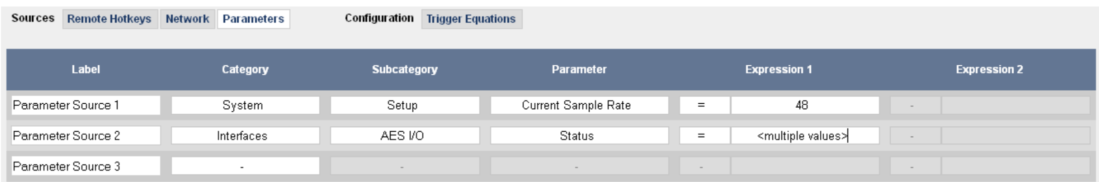
Below is a screen shot of the EmBER viewer tool :



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

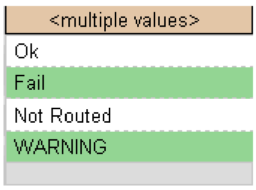
As a value you will receive the trigger name from the **D*AP4 VAP**. In this example it is the first trigger named : "Movie".

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



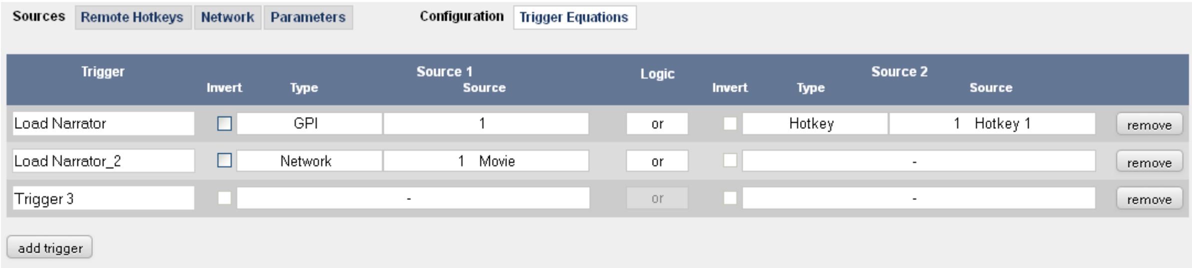
Above is an example of parameter trigger sources. **<multiple values>** indicates that more than one value of the parameter "Status" is bound to that trigger source :

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 :


Sources | Remote Hotkeys | Network | Parameters | Configuration | Trigger Equations

Trigger	Source 1 Invert	Source 1 Type	Source 1 Source	Logic	Source 2 Invert	Source 2 Type	Source 2 Source	
Load Narrator	☐	GPI	1	or	☐	Hotkey	1 Hotkey 1	remove
Load Narrator_2	☐	Network	1 Movie	or	☐	-		remove
Trigger 3	☐	-		or	☐	-		remove

add trigger

Trigger	Here you define a name for the trigger (Load Narrator).
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 [-].

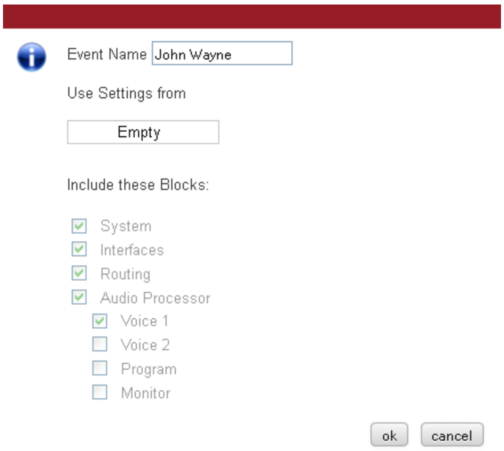
Setup GUI – EVENTS – Events – **Preset Events**

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 :

The example shows that the first preset (default name Preset 1) of a function block is assigned. If this preset event is executed, these presets will be loaded into that function block.

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

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" check boxes).

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

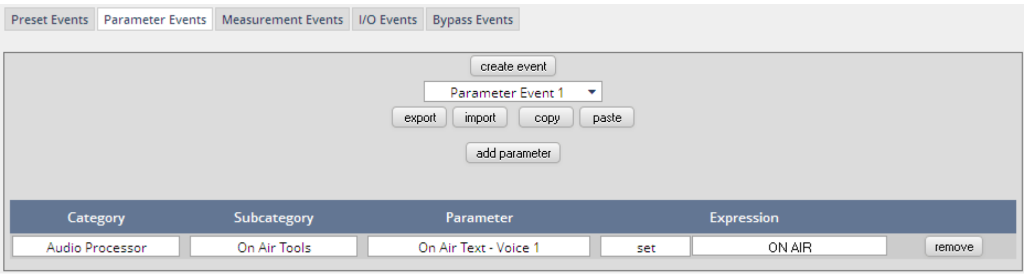
Empty Creates a set of empty fields where you may select the preset of your choice for the respective function block or leave it empty if no changes are needed …

Important note ! This is the way to create your own **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!

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

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

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



Category [Audio Processor / Measurement]

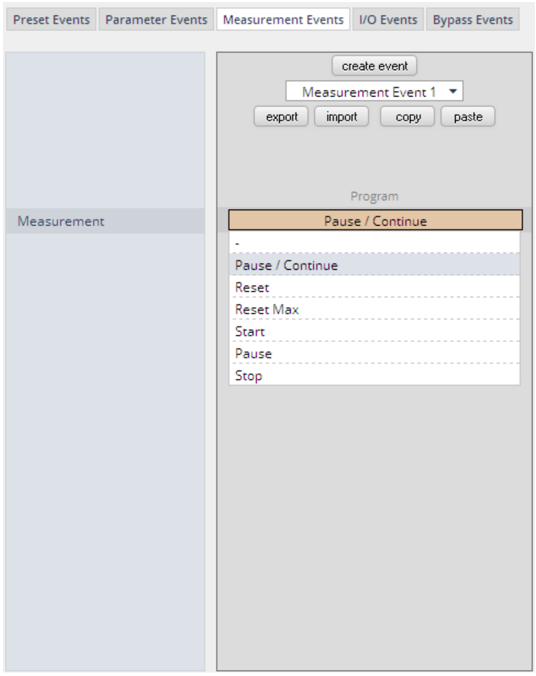
Subcategory [in case of Category = Audio Processor >> On Air Tools]
[in case of Category = Measurement >> Loudness]

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**

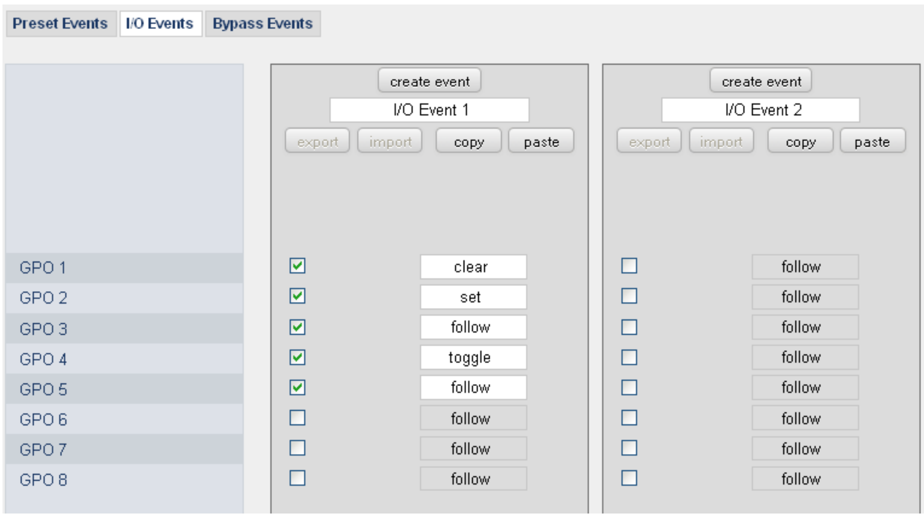
Measurement events can be used to control the integrated loudness measurement of the program path of the **D*AP4 VAP :**



For each measurement event you can assign one of the possible control functions.

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

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

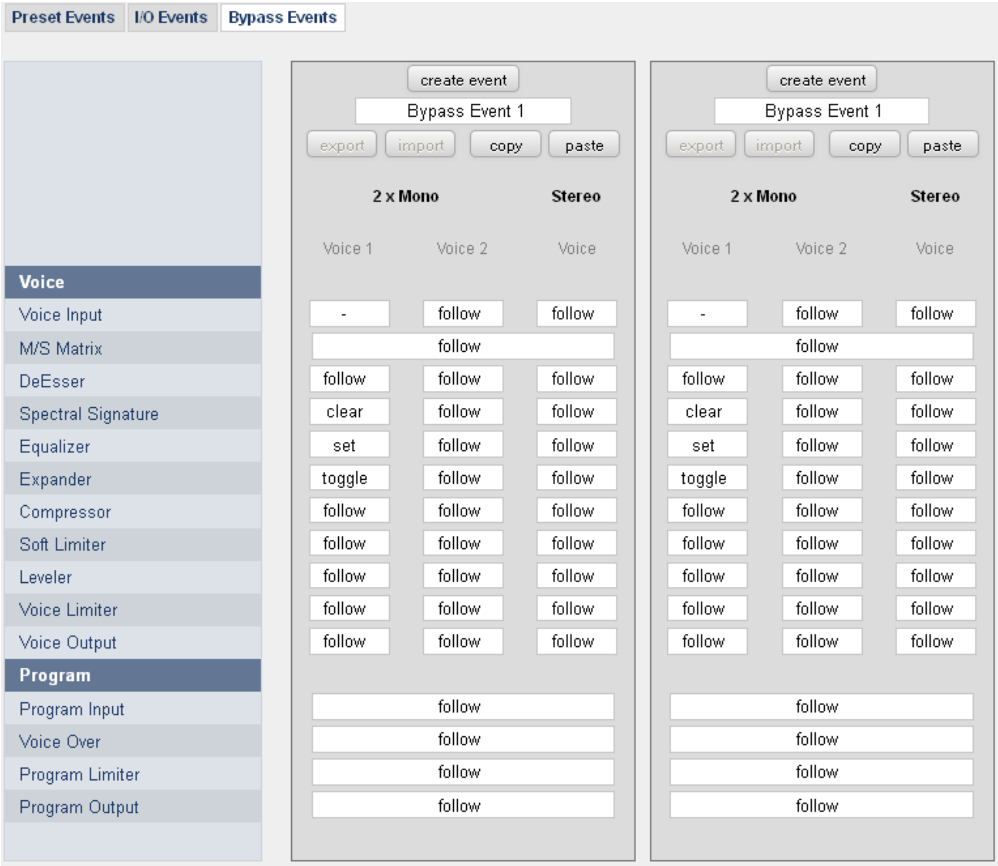


Each GPO (when incorporated into that I/O event) can be set to the behavior as follows :

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 Be careful because it needs a definite known starting condition to work properly.

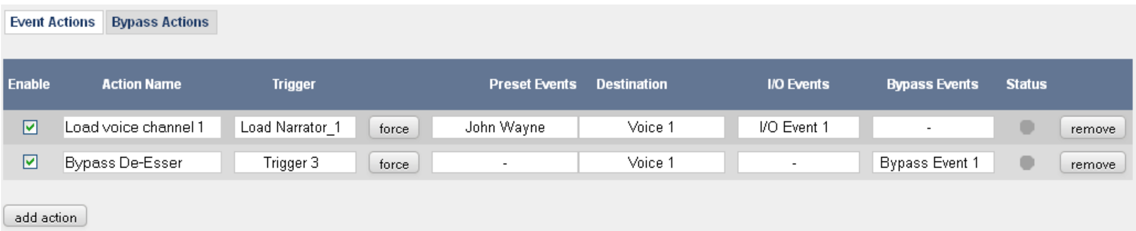
Setup GUI – EVENTS – Events – **Bypass Events**

The **D*AP4 VAP** allows you to bypass some or all of the function blocks to use them or for A/B comparison :



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

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



Here you create the action!
You should give the action a meaningful name, select a trigger (from one of the trigger equations) and select the respective event(s) you need to perform the desired action.

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 directly turn the bypass of one, some or all function blocks by simply clicking on the check boxes in the right hand panel :

Trigger

Lock Source	[ON / OFF] The **X*AP RM1** remote panel **<BYPASS>** button may be disabled here.
Force Trigger Active	[ON / OFF] Force the bypass function from the GUI instead of the **X*AP RM1** remote panel **<BYPASS>** button.

For an example setup of the **EVENTS** set-up pls. see the quick start guide that came with the **D*AP4 VAP**.

Technical data – **D*AP4 VAP**

• **Power supply**	dual power supply, auto fail over AC 85 V – 264 V, 50 Hz … 60 Hz 58W max
• **AES input**	AES3 110 Ohm balanced 0.3 V … 5.0 Vpp sample rate converter: 24 Bit, 32 kHz … 192 kHz, THD+N: < -130 dB @ 0 dBFS
• **AES output**	AES3 110 Ohm balanced 4.0 Vpp balanced power fail relay bypass
• **Sync internal**	44.1 / 48 / 88.2 / 96 kHz, +/- 25 ppm, 150ppm capture
• **Sync input**	AES3id: 48 kHz, 0,32 … 1,2 Vpp @ 75 Ohm Wordclock: 48 kHz, 1 … 3 V @ 75 Ohm Video: Black Burst or Tri Level, 0.5 … 1.0V @ 75 Ohm
• **Sync output**	Wordclock 48 kHz: > 2 V @ 75 Ohm
• **Network**	RJ45 rear connector 10/100MBit Ethernet auto sense, full duplex, auto MDI/X
• **USB**	USB 2.0 connector to internal console interface
• **GPI**	3 V – 30 V balanced, auto polarity
• **GPO**	relay change over contacts, 200mA/24V (DC/AC)
• **Environmental**	operating temperature 0 ºC to 50 ºC **D*AP4 VAP** - fan cooled non operating -20 ºC to 70 ºC humidity 90%, non condensing
• **Dimensions and Weight**	19", 1RU, depth 27 cm net weight approx. 5 kg shipping weight 7,5 kg

Technical data – interface boards – **2x MIC/LINE Input [O_D*AP4_AMIC_a]**

• **Max. input level**	14dBu (Mic) 24dBu (Line)
• **Frequency response**	20Hz (-0.1dB) … 22kHz (-0.1dB) @ SR=48kHz 20Hz (-0.1dB) … 43kHz (-0.1dB) @ SR=96kHz
• **Input impedance**	5.5kOhm, differential (Mic) 8.5kOhm, differential (Line)
• **Dynamic range**	> 109dB RMS, unweighted (20Hz … 22kHz)

Technical data – interface boards – **SDI De-Embedder / Embedder [O_D*AP4_SDI_a]**

- **SDI input**

  standards (auto sensing)
  3G - SMPTE 424/425M (Level A/B)
  HD - SMPTE 292M
  SD - SMPTE 259M
  formats
  1080p23.98, 24, 25, 29.97, 30, 50, 59.95, 60
  1080i50, 59.94, 60
  720p23.98, 24, 25, 29.97, 30, 50, 59.94, 60
  625i50
  525i59.94, …
  connector
  BNC IEC 169-8)
  75 Ohm
  return Loss
  > 15 dB (typ. > 18dB) from 5MHz to 1485 MHz
  > 10 dB (typ. > 11 dB) from 1485 MHz to 2970 MHz
  adaptive equalization, typical of Belden 1694A coaxial cable
  250 m at 270 Mbps
  250 m at 1.485 Gbps
  150 m at 2.97 Gbps
  jitter tolerance
  Timing: > 2UI, Alignment: > 0.7 UI

- **SDI output**

  standards
  3G - SMPTE 424/425M (Level A/B)
  HD - SMPTE 292M
  SD - SMPTE 259M
  formats
  1080p23.98, 24, 25, 29.97, 30, 50, 59.95, 60
  1080i50, 59.94, 60
  720p23.98, 24, 25, 29.97, 30, 50, 59.94, 60
  625i50
  525i59.94, …
  quantization
  10Bit
  connector
  BNC IEC 169-8)
  75 Ohm
  return loss
  > 15 dB (typ. > 18dB) from 5MHz to 1485 MHz
  > 10 dB (typ. > 11 dB) from 1485 MHz to 2970 MHz
  signal level
  800 mV +/- 10%
  D.C. offset
  0.0 V +/- 0.5 V
  rise and fall time
  < 135 ps at HD/3G, < 800 ps at SD
  overshoot
  < 10% of amplitude
  output jitter
  Timing: < 0.5 UI, Alignment: < 0.2 UI

• Special features relay bypass (manual or automatic on power fail)
320 ms video delay (number of frames depends on the video format)
16 chanel audio de-embedder / embedder
VANC (SMPTE 2020-2) de-embedder / embedder
16 x 16 de-embedder matrix (mono routing)
32 x 16 embedder matrix (mono routing)
320 ms audio delay per audio channel
automatic compensation of non processed audio signals for video delay

Technical data – interface boards – **4x AES I/O [O_D*AP4_AES_a]**

connector
- 25pin Sub-D female

inputs
- 110 Ohm balanced or 75 Ohm unbalanced jumper selection
- 0.3 V … 5.0 Vpp

sample rate converter
- 24 Bit, input sample rate 32 kHz … 192 kHz, THD+N < -130 dB @ 0 dBFS

outputs
- 110 Ohm balanced or 75 Ohm unbalanced jumper selection
- 4.0 Vpp balanced, 1.0 Vpp @ 75 Ohm

power fail relay bypass

Technical data – interface boards – **4x analog I/O [O_D*AP4_ADDA_a]**

connector
- 25pin Sub-D female

input
- impedance: > 10 kOhm, electronically balanced
- max input level: 0.0 dBu … +24 dBu adjustable in 0.5 dB steps
- dynamic range: 115 dB
- THD+N: @ -1 dBFS, 15 dBu: -90 dB
- frequency response: 20 Hz … 22 kHz (+/- 0.25 dB)
- crosstalk @ 20 kHz: > 100 dB
- calibration gain mismatch: < 0.3 dB

output
- impedance: 5 Ohm, electronically balanced
- max. output level @ 0 dBFS: 0.0 dBu … +24 dBu adjustable in 0.5 dB steps
- dynamic range: 110dB
- THD+N @ -1 dBFS: -92 dB
- frequency response: 20 Hz … 22 kHz (+/- 0.25 dB)
- crosstalk @ 20 kHz: > 100 dB
- gain mismatch balanced / unbalanced: < 0.3 dB

power fail relay bypass

*Technical data – interface boards – **2x digital mic input [O_D*AP4_DMIC_a]***
- tbd

Technical data - rear connectors - **pin assignment**

connector :	**GPI/O**
female	25-pin Sub-D
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 **-**

Technical data - optional interface modules – **pin assignment**

4x analog I/O [AN 144]

connector :	4 x analog I/O
female	25-pin Sub-D
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 [DD 188]

connector :	4x AES I/O
female	25-pin Sub-D
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 [AN 108]

connector :	8 x analog out
female	25-pin Sub-D
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

Safety Information

Electrical

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

Service safety

Only qualified personnel should perform service procedures.

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

To avoid fire or personal injury

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

Specifications are subject to change without notice

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