the reference in loudness management

T*AP

T*AP

Television Audio Processor

Manual

www.jungeraudio.com

hardware features

• **1RU Base Unit**	compact 19" processing device with front side controls and displays
• **1RU Remote Panel**	detachable panel with case, powered by POE (Power Over Ethernet)
• **4x 2 channel audio delay**	up to 2sec.delay time each
• **Dolby® decoder**	built in optional Dolby® E or D or D+
• **Dolby® encoder**	built in optional Dolby® E or D or D+ or AAC or HE-AAC
• **Dolby® metadata I/O**	two RS485 9-pin Sub-D connectors
• **4x AES3id I/O + SRC**	on board AES I/O with relay bypass and SRC (selectable) per input
• **Two interface slots**	expansion slots for optional I/O boards : 3-G/HD/SD-SDI, AES, analog
• **RJ45 POE connector**	for connecting the X*AP Remote Panel
• **RJ45 network connector**	100BaseT full duplex Ethernet interface
• **USB connector**	built in USB < > serial adapter to access the service port
• **8x GPI**	balanced inputs on 25pin Sub-D
• **8x GPO**	relay change over contacts on 25pin Sub-D
• **Aux power supply**	isolated 5V supply for external GPI/O wiring
• **External sync IN**	BNC input (Word Clock, AES, Black Burst, Tri-Level)
• **Sync OUT**	BNC Word Clock output

software features

• **8 channel LevelMagic™**	Junger Audio level, loudness and limiter control algorithm
• **5.1 Upmix**	Junger Audio upmix algorithm
• **Downmix**	stereo downmix from 5.1 source
• **Filter**	HP/LP filter, 5x parametric EQ, SpectralSignature
• **Voice Over**	manual or automatic ducking functions
• **Fail Over**	switching of alternative signals to maintain audio for a specific program
• **Delay**	separate delay for each processing channel up to 2000 ms.
• **Dynamics**	compressor / expander
• **Monitor**	to check paths inside the DSP processor, separate downmix
• **SNMP agent**	SNMP v1 get (no set) and configurable traps (see TAP-MIB)
• **EmBER protocol**	supports the l-s-b Ember and Ember+ protocol for VSM integration, and 3rd party API

Content

Content

Introduction

At the heart of the **T*AP** is a sophisticated audio processor, powered by Analog Devices® Sharc DSPs. These DSPs provide the 10 channel audio processing and monitoring facility.
They are surrounded by several I/O interfaces, audio delay lines and an optional Dolby® decoder and encoder.

The four AES3id I/Os on the motherboard may be rounded up by a variety of interface modules that can be installed as an option into the **Base Unit**'s interface slots.

A comprehensive routing matrix allows almost every signal flow - from inputs to outputs, from and to Dolby® encoder / decoder, the built in audio delay lines and the audio processor itself.

The routing architecture uses the industry's most advanced event management. Triggered by GPIs, Hot Keys on the **X*AP** Remote Panel, internal status information or network based remote control, the **T*AP** may be reconfigured from surround to multiple stereo operation on the fly.

Routing paths, the enabling and disabling of audio processing blocks and the setting of processing parameters can be pre configured by individual presets dedicated to each function block. The content of the presets can be displayed and edited off line while the device is on air. These presets may either be recalled on demand by the operator via the GUI, the X*AP Remote Panel Hot Keys or play-out automation systems, but may also be part of complex scenarios defined by the operator and automatically executed by the event manager of the device.

The **T*AP** provides a web based setup GUI and a **X*AP** 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 Remote Panel are limited to operating needs.

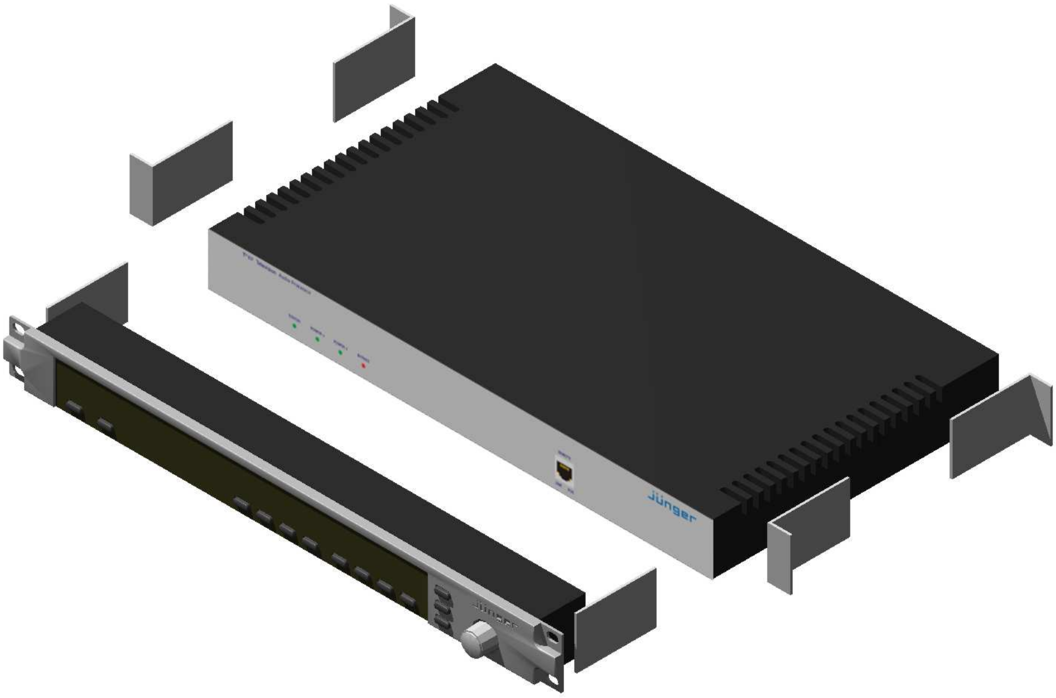
Junger Audio's LoudnessLogger is also available as an add on and can be attached by a few simple clicks to the **T*AP** so that users can log loudness data as well as display it as a plot on a PC screen in real time.

Completing the feature set of the **T*AP** is the availability of an SNMP agent, which provides traps and status polling. As an option, it can also control the internal loudness measurement and the retrieval of measurement data.

As with most advanced tools, **T*AP** 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 **T*AP**, it does not give detailed scenarios because the operational needs of today's broadcasters vary so widely between organizations and their work flows and cover so many different parameters – from ingest to studio operation, from master control rooms to play-out or even rebroadcast applications.

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

hardware concept



The **T*AP** consists of a **Base Unit** that carries all relevant connectors and a detachable **X*AP** Remote Panel both in 19" 1RU format.

The **X*AP** Remote Panel is powered by POE (Power Over Ethernet) and designed to control multiple **Base Unit**s one at a time.

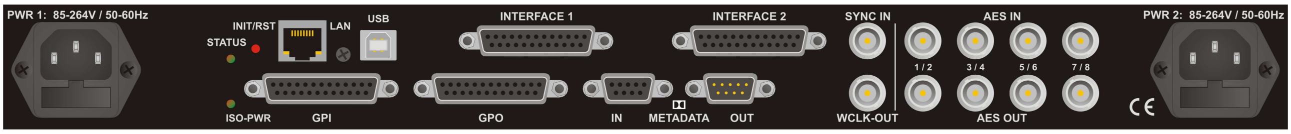
For a stand alone installation the **X*AP** Remote Panel may be attached to a dedicated **Base Unit** by brackets.
In this case we highly recommend to support the chassis by additional brackets screwed to the rear as shown above or by metal angles supporting the device from the bottom.

Base Unit front panel view

The front panel of the **Base Unit** shows 4 status LEDs :

STATUS	general representation of the device status. It is a sum display of all relevant status information
Power 1	status of power supply # 1
Power 2	status of power supply # 2
BYPASS	shows if one of the audio processing parts of the **T*AP** is put into bypass mode

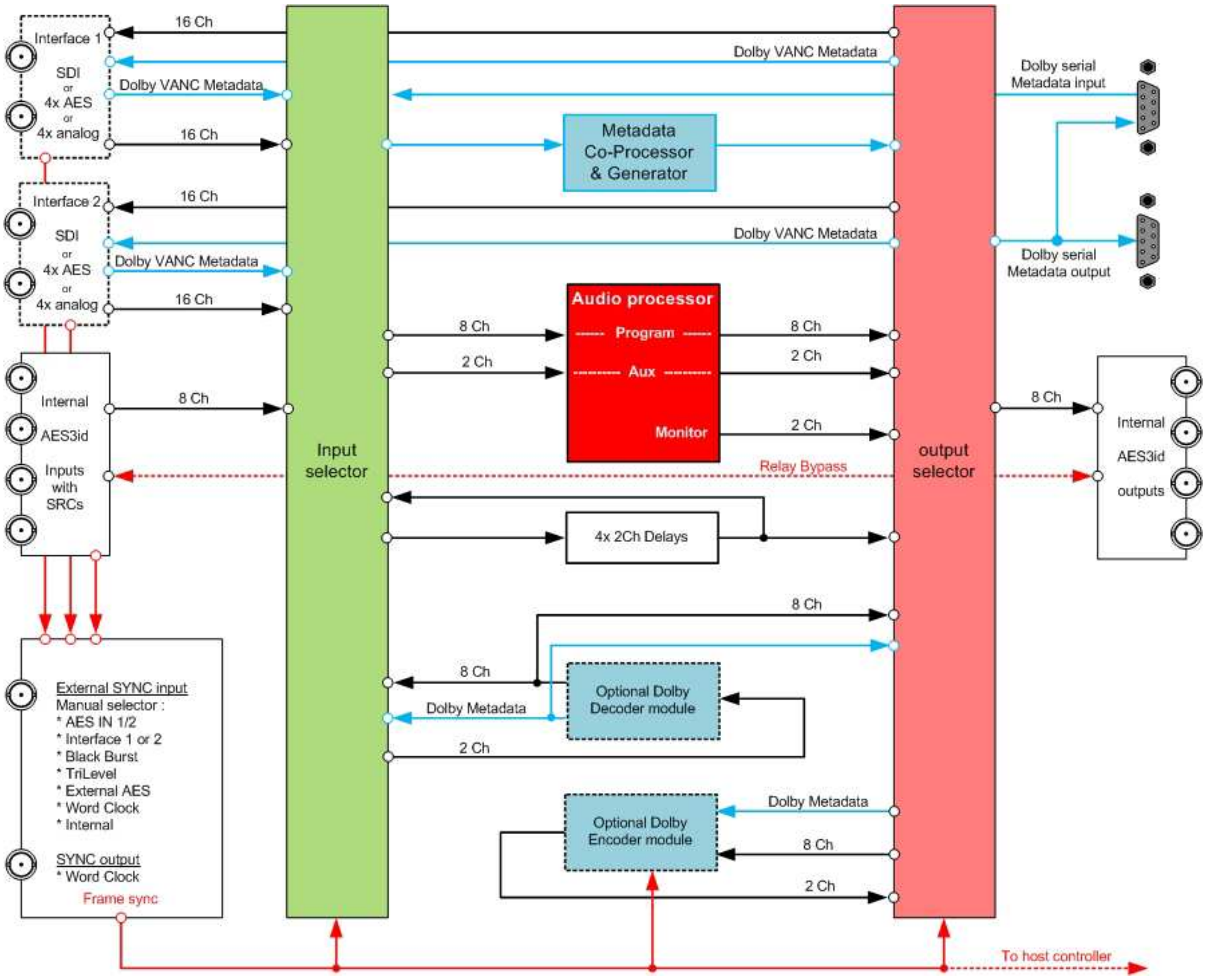
Base Unit rear view



For fail safe operation the **Base Unit** provides two independent power supplies. These power supplies operate in load balance. The status of both PS are displayed on the **Base Unit** front panel as well as on the **X*AP** Remote Panel.

STATUS LED	shows the status of the device controller
INIT	pressing the INIT button briefly will warm start the device controller. Holding down the button until the **STATUS LED** flashes 3 times will initialize the **Base Unit** 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 LED	lights up if the isolated 5V power supply for GPI /O application is turned on
GPI	25pin Sub-D female connector to interface with the 8 optical isolated general purpose inputs
GPO	25pin Sub-D female connector to interface with the 8 switch over relay general purpose outputs
Interface 1	slot to mount one of the optional interface boards (SDI, AES, analog)
Interface 2	slot to mount one of the optional interface boards (SDI, AES, analog)
METADATA IN	9pin Sub-D female connector to receive and send Dolby® serial metadata
METADATA OUT	9pin Sub-D male connector to send Dolby® serial metadata
SYNC IN	75Ohm BNC connector to connect with external sync sources
WCKL-OUT	75Ohm BNC connector to synchronize external devices to the **T*AP** internal word clock
AES IN 1/2 – 7/8	AES3id inputs
AES OUT 1/2 – 7/8	AES3id outputs

block diagram



The above schematic shows the principal blocks of the **T*AP**.
The core of the unit is the 10 channel Audio Processor with 2ch Aux inputs and a 2ch monitor output.

On the motherboard you will find **4x AES3id** I/Os which are bridged by relays in case of a power failure.
Two I/O slots which may carry option boards allow for extremely flexible interfacing of the **T*AP**. I.e. you may process the audio signals of two independent TV programs.

The unit may also be fitted with Dolby E/D/D+ decoder and encoder. For comprehensive metadata processing the unit has serial metadata I/O connectors. All metadata functions are centralized in a metadata Co-processor.

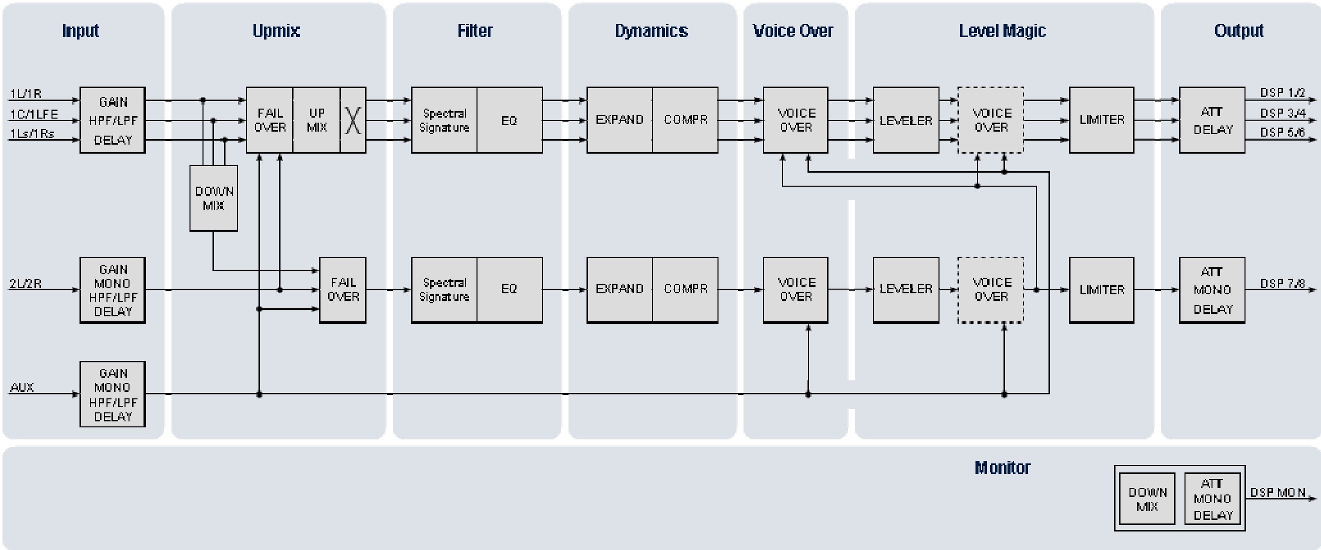
The sync circuit provides all features to integrate the **T*AP** into digital processing environments. Other devices may be synchronized by the word clock output of the **T*AP**.

Beside the option to delay all DSP outputs, the **T*AP** has 4x 2Ch delay lines that may be routed into any signal path of the device.

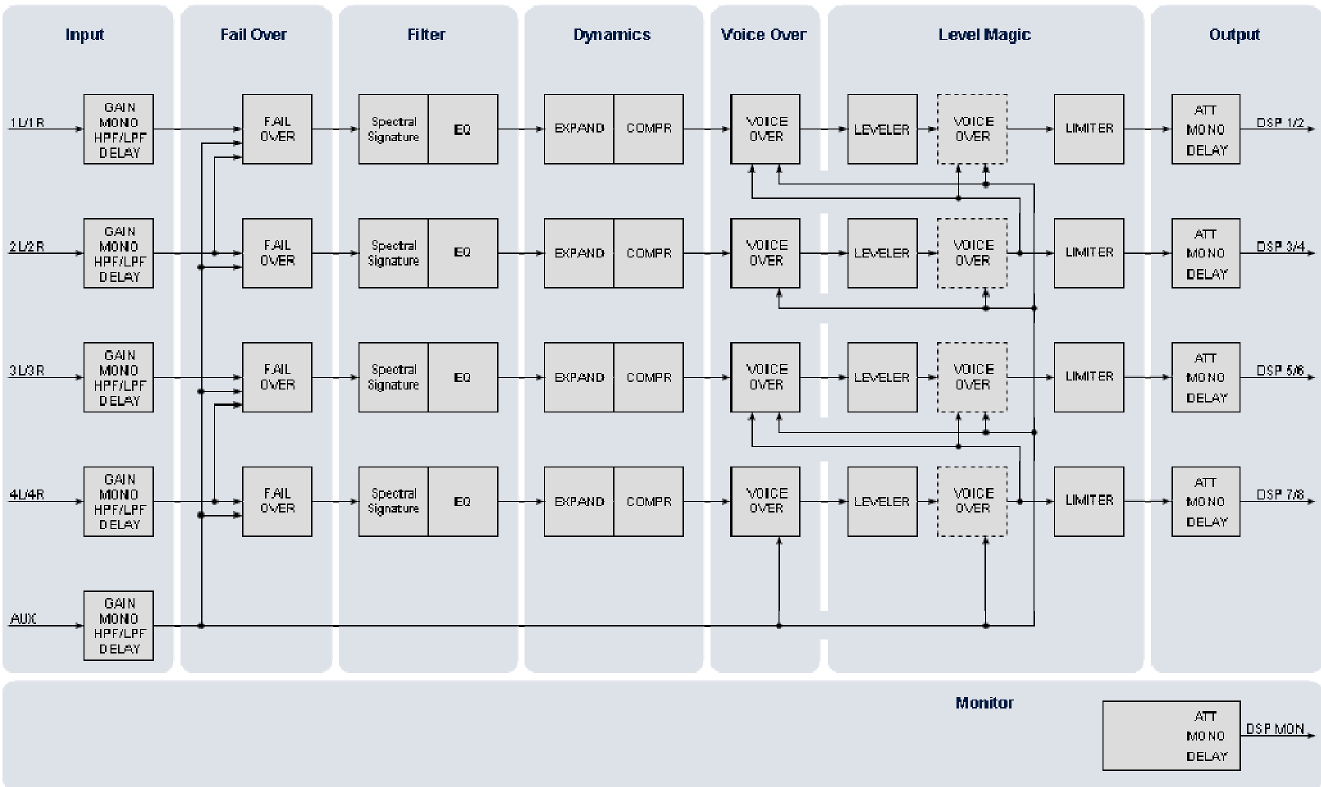
audio processing blocks

Speaking in D-E terminology, the **T*AP** may be configured as a surround sound processor with additional stereo program processing (5.1 + 2) or as a four times stereo processor (4 x 2).

Audio processor block diagram 5.1 + 2 program configuration :



Audio processing block diagram 4 x 2 program configuration :



Important Note! In a **4 x 2** configuration, the processing links between stereo channels may be disabled via the respective function block, to perform full or partial mono processing if required. You must keep in mind that such a configuration is still treated as 4 programs if it comes to program related setups and information such like the Dolby Metadata.

control concept

The communication between the **X*AP** Remote Panel, the **Base Unit**, setup and operating tools, is based on **TCP/IP over Ethernet**.

The setup GUI utilizes web technology. At the time of editing this manual the functionality of the web GUI is developed for Firefox 15.1.

The setup GUI will be completed by several application programs running under MS Windows® XP, W7 like the JA **Application Manager**.

An **SNMP** agent is also available on the device and may be explored by a monitoring system.

Junger highly recommends using the l-s-b **Ember+** protocol which is widely distributed in the European broadcast industry where the user community is increasing rapidly world wide. By the way, the **X*AP** Remote Panel and the **Base Unit** "talk" Ember natively. For backwards compatibility the T*AP supports both the Ember (on TCP port 9999) and Ember+ (on TCP port 9000).

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 page embedded soft buttons for groups of parameters.

Each parameter area has a set of presets. The presets can be recalled at any time during operation, either by manual intervention, automatically by the internal event manager or by external authorities.

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 **T*AP**. You may recall such presets at any time manually, or automatically.

The presets of the **T*AP** are persistent by nature. You are working directly on the preset memory, i.e. you must not worry about storing such presets. The **T*AP** does it for you.

event concept

With the **T*AP** you have a sophisticated event management system on hand.

Events are bound to **Trigger** which may be nested and are defined by the logical combination (AND, OR, XOR) of two random trigger sources. Such a trigger source may be device status information (e.g. sync lost), GPIs, network commands, hotkeys of the **X*AP** Remote Panel, status (true or false) of parameters.

The pre defined trigger may ignite events which will recall presets from the several function blocks of the **T*AP** :

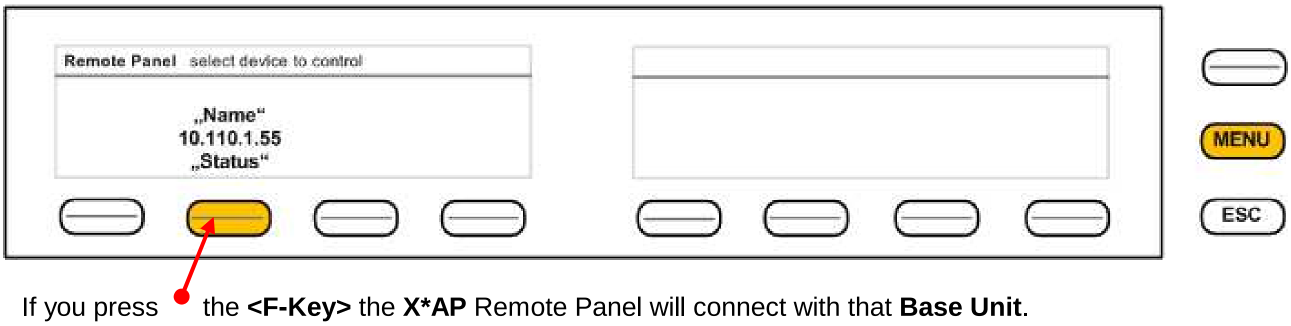
* **Preset Events** for System, Interfaces, Routing, Dolby Processing, Audio Processing
* **Action Events** for GPOs, Loudness Measurement
* **Bypass Events** for pre configured bypass scenarios

getting started – basic **X*AP** Remote Panel operation

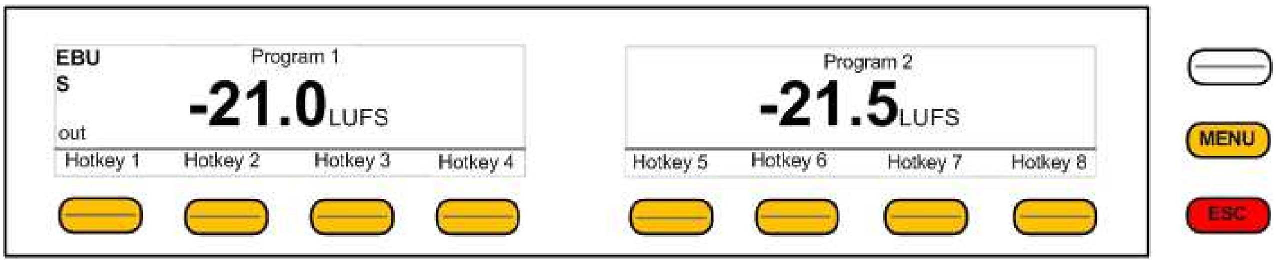
The communication interface of the **T*AP** is based on TCP/IP over Ethernet. The **X*AP** Remote Panel as well as the **Base Unit** must have unique IP addresses in order to "talk" to each other as well as to other devices within the Local Area Network. An **X*AP** Remote Panel may for now control up to **4** Base Units, one at a time.

If the **X*AP** Remote Panel is attached mechanically to a **Base Unit** it should be connected via the Ethernet socket on the front panel of the **Base Unit**. If the **X*AP** Remote Panel is detached from the **Base Unit**, one may use the CAT5 cabling of a facility or the OB-Van to connect it to the distant **Base Units** front socket in order to get power. If it must be connected via a router of the network, this router must have a **POE** (Power Over Ethernet) port. If this is not the case, you must use a wall plug **POE** power supply.

After power up and booting, the **X*AP** Remote Panel shows the **T*AP Base Units** which are "attached" to it. The display shows the respective device **"Name"**, the **IP address** and the connect **"Status"**.
Options are "connect", "can't connect" and "unknown device". In case of "connect" you may press one of the highlighted buttons.



If you press the **<F-Key>** the **X*AP** Remote Panel will connect with that **Base Unit**.
(The above example has just one **T*AP Base Unit** attached for remote control). Now the **X*AP** Remote Panel will gather all necessary information from that **Base Unit** (may take a few seconds) and open up the main operating display :



Because this is the main operating display, the **<ESC>** button light **red** to indicate that the power up display is above the **main display**. Pressing **<ESC>** returns you back to the device selection.

getting started – IP setup in general

The process of installing a **T*AP** into an **IP network** is as follows :

1. Ask the system service people for two unique IP addresses of the network, netmask and gateway address
2. Assign the **Base Unit** an IP address
3. Assign the **X*AP** Remote Panel an IP address
4. Attach the **Base Unit** to the **X*AP** Remote Panel

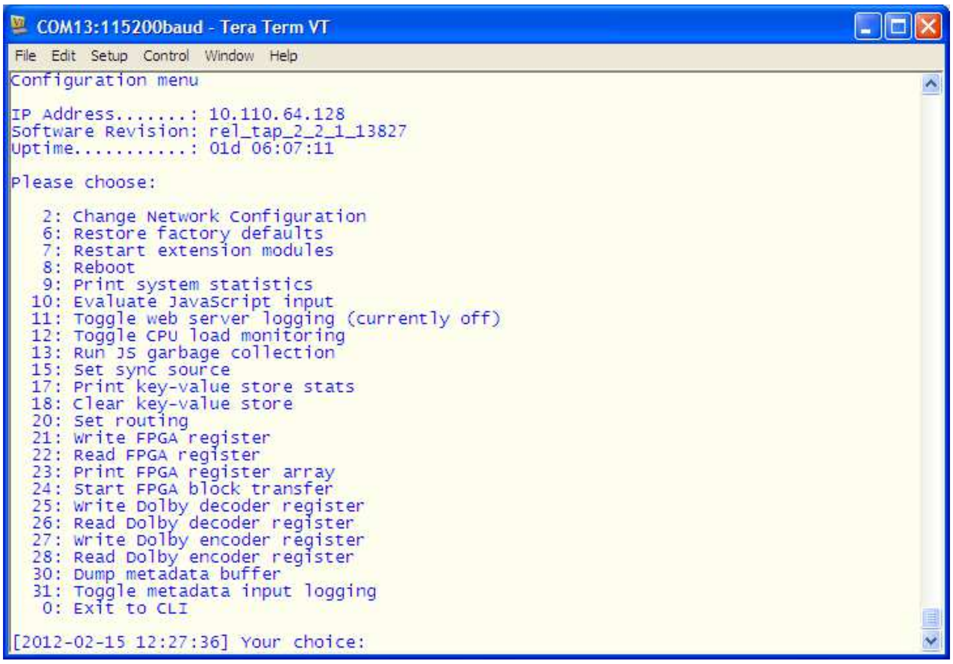
You have 2 choices to assign the **Base Unit** an **IP address** :

* From the serial console interface
* Via Web browser

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

getting started – IP setup of the **Base Unit** – **via console interface**

The tool to change the IP configuration of any **Base Unit** will be reached via the console interface. You must connect the **Base Unit** 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** and press **<ENTER>** :

"Your choice: 2"
"Current network configuration

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

You must enter the IP address and the netmask.

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 take care that the gateway address matches the network mask related to the device IP address!
If you re not sure simply enter **0.0.0.0**.

Enter new gateway, press ENTER to configure without gateway : "0.0.0.0" <Enter>
Network configuration has been changed. Please reboot the device
To activate the new settings.

Select item 8 and press **<ENTER>** :

Do you want to reboot the device ?

Press **small "y"** :

Do you want to reboot the device ? y

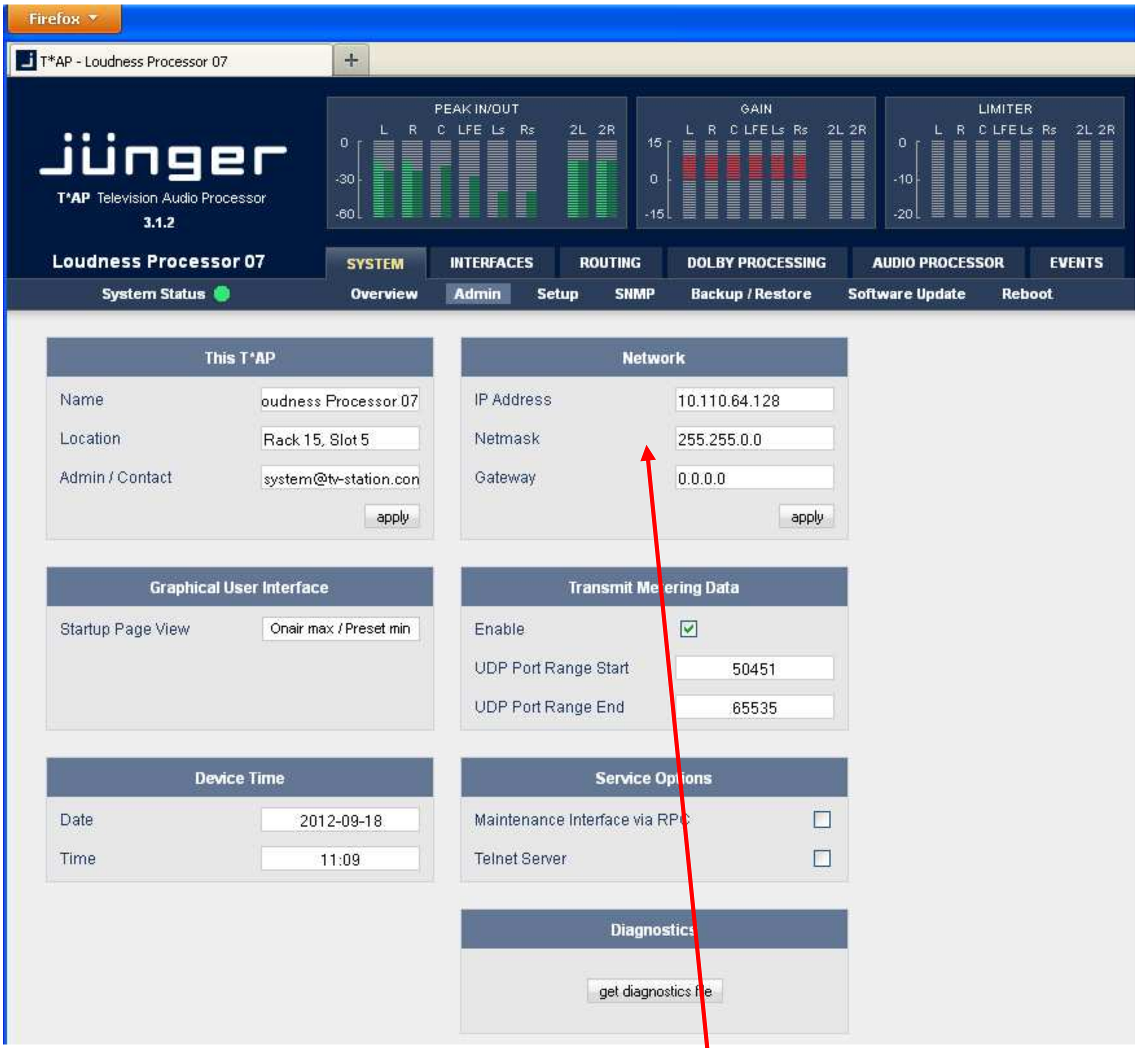
Press **<ENTER>**

Rebooting the device ……..

After reboot has finished the new IP configuration is active.

getting started – IP setup of the **Base Unit** – **via web browser**

* Read the **default IP address** printed on a label above the RJ45 Ethernet connector.
* Set up network parameters of the PC which meet the default IP address of the **Base Unit** (net mask = 255.255.0.0).
* Connect the **Base Unit** with the PC either by an Ethernet cross over cable or by a switch.
* Open a browser and type IP address of the **Base Unit** into the URL field and press **<ENTER>**. This will open the **AUDIO PROCESSOR** tab sheet of the GUI.
* Click on **<SYSTEM>** and the "**Admin"** tab will open automatically :

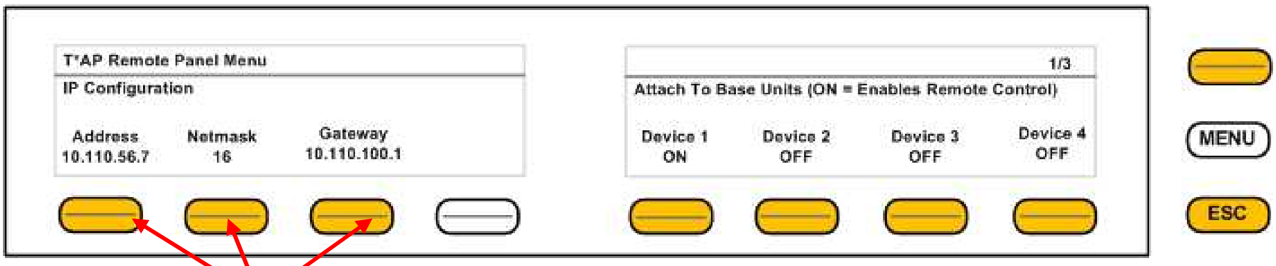


Enter the desired network configuration and press **<apply>**
Afterwards you must reboot the **Base Unit** in order to activate the new IP configuration.
Regarding Gateway address see above.

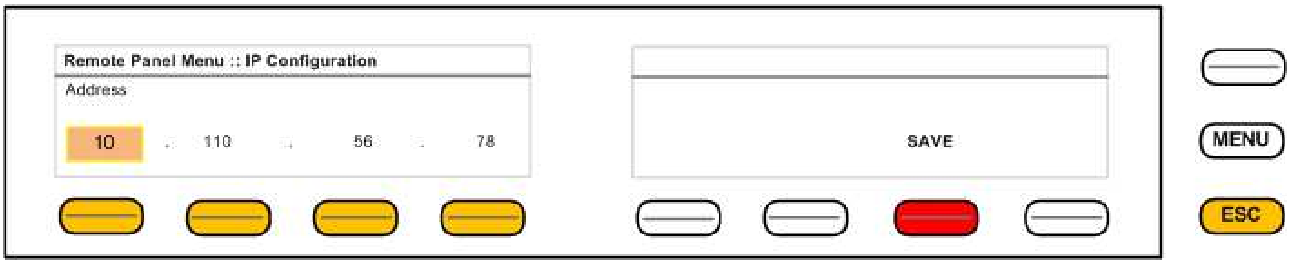
Important Note! After reboot neither the **web browser** nor the **X*AP** Remote Panel may be able to communicate with the **Base Unit**. You must key in the new IP address in the URL field and change the **X*AP** Remote Panel settings to attach this device again.

getting started – IP setup of the **X*AP** Remote Panel

By pressing the **<MENU>** button after power up or by pressing the red **<ESC>** button from the main display, you will enter the **"T*AP X*AP** Remote Panel **Menu"** page 1/3 to set up the IP configuration of the X*AP Remote Panel and to attach up to 4 devices to this X*AP Remote Panel :



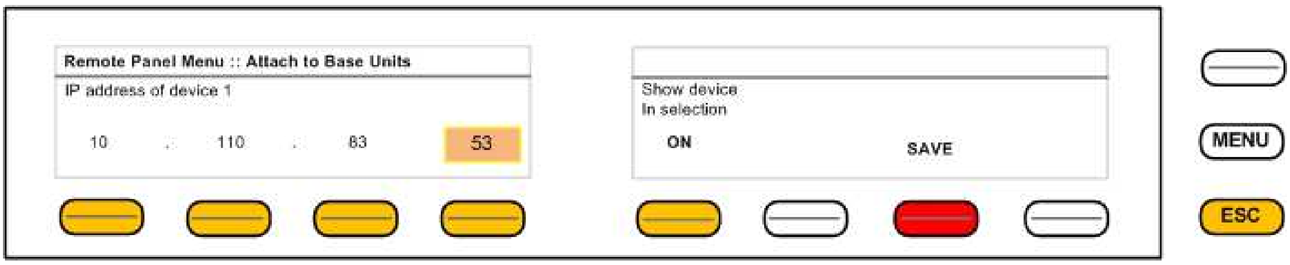
You may press the relevant **<F-Keys>** and separate windows will appear for comfortable set up.
Here an example for the address field :



You must press one of the relevant **<F-Keys>** and that field will be highlighted as well as the Rotary Encoder. Now you can change the value by turning the knob. When the setting of all fields is finished, you must press **<SAVE>**. The display will return to the initial "**T*AP** X*AP Remote Panel **Menu"** page 1/3.

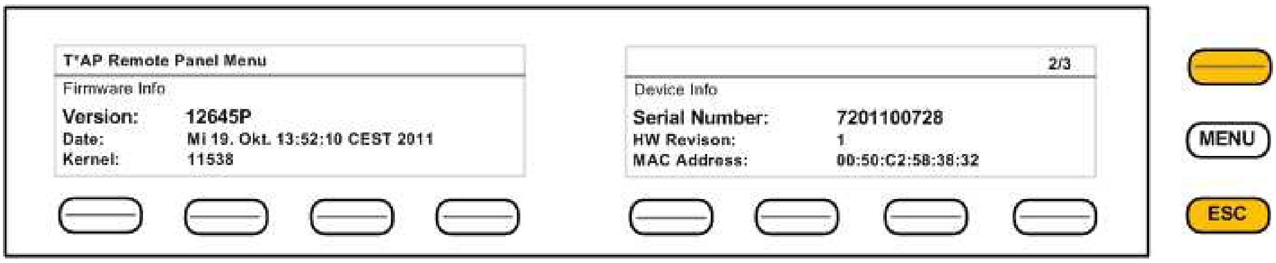
getting started – attach a **Base Unit** to a **X*AP** Remote Panel

You must press one of the **"Device x" <F-Keys>** and a different window will open :



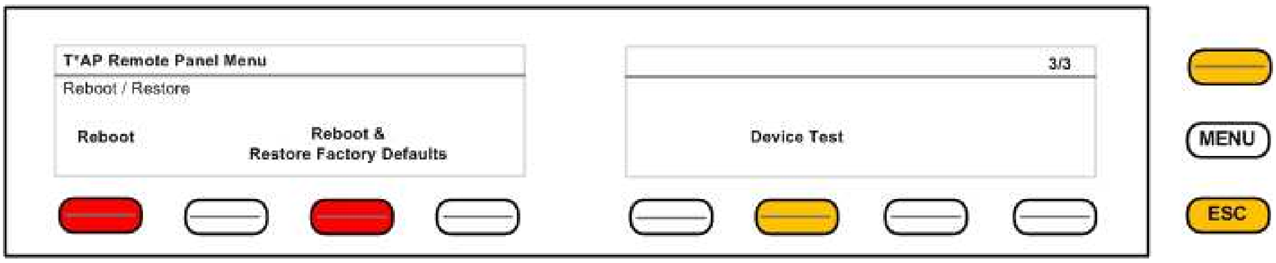
Same procedure: Set up the IP address of the **Base Unit** you are about to attach.
You must turn "**Show device in selection"** to **ON** in order to reach the device via the initial display later on.
Pressing **<SAVE>** will return to the "**T*AP** X*AP Remote Panel **Menu"** page 1/3.

getting started – **X*AP** Remote Panel menu page 2/3 – **firmware display**



This page shows static information regarding firmware versions and device infos.

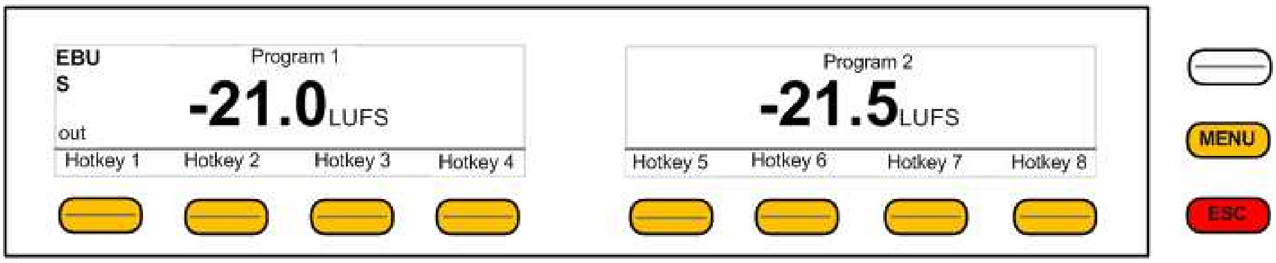
getting started – **X*AP** Remote Panel menu page 3/3 – **reboot, restore factory default, device test**



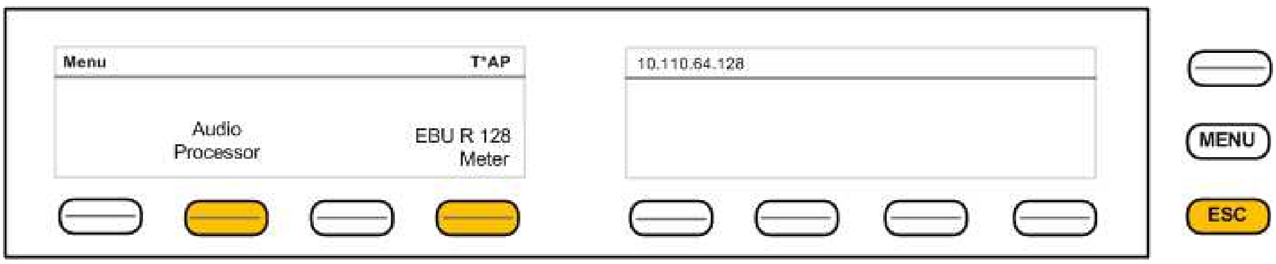
Page 3 allows for reboot, restoring of factory defaults and function test of the X*AP Remote Panel LEDs, buttons and the rotary knob. Pressing the Device Test button opens up further menus to test the respective items.

operating - menu structure of the **X*AP** Remote Panel

Power up display – may show up to 4 **Base Units** enabled for remote control for this X*AP Remote Panel**.**
Pressing these buttons connects with the respective **Base Unit**.
After gathering all **Base Unit** settings the **Main Display** opens up :

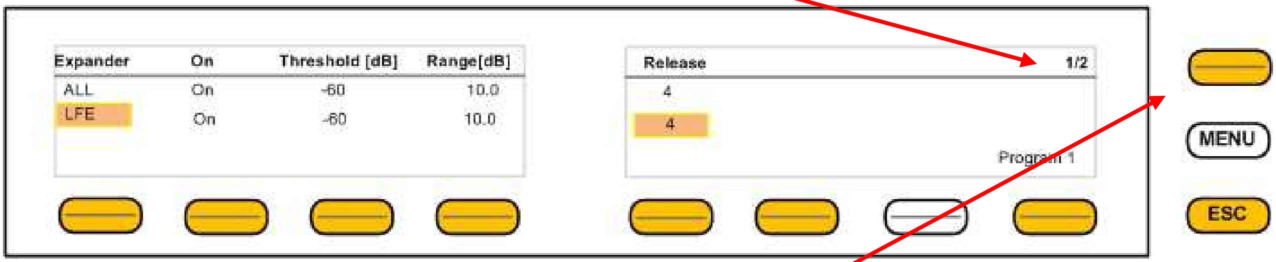


When pressing the **<MENU>** button, the main operating menu opens up:



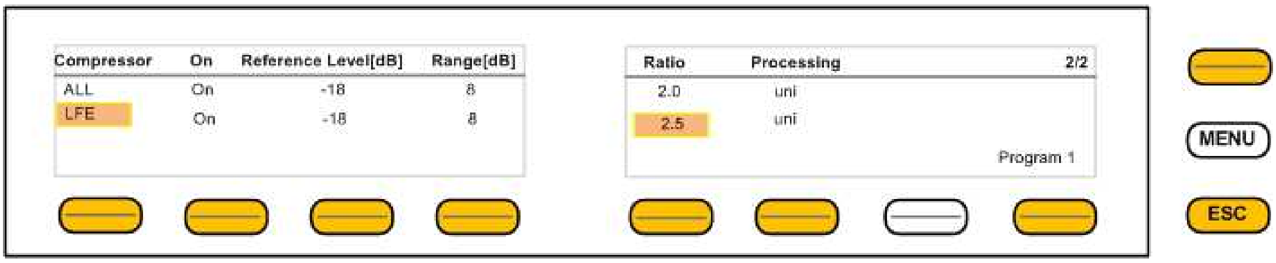
operating – menu structure of the **X*AP** Remote Panel - **principle of operation**

If you are in a specific parameter menu the display structure may change due to the program configuration of the **T*AP**. Below is an example for setting the parameters for the **Dynamics** while the **T*AP** is in **5.1 + 2** program configuration and operates in ITU mode. In this case you have two parameter sets for the first program: **ALL** and **LFE** (if the **LFE** is not linked**)**.
Since the Dynamics have two subsections: **Expander** and **Compressor**, this menu has two pages, indicated by the number in the top right hand corner :

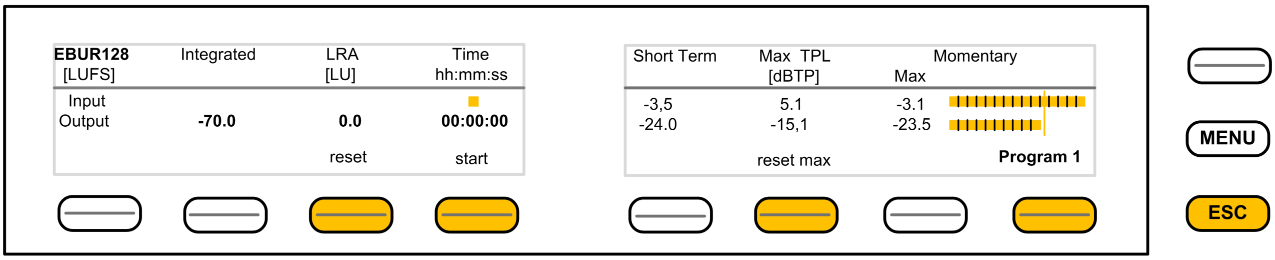


You may switch between both pages with the <page> button
<Hotkey 1> toggles between the two parameter sets ALL / LFE. The parameter set under control is highlighted. If for example you now press **<Hotkey 5>,** the **Release** setting for the **LFE** will be enabled and the **Rotary Encoder** is also illuminated. You may now change the **Ratio** by turning the knob.
<Hotkey 8> toggles between **Program 1** (5.1) and **Program 2** (1x2).

Next page shows the **Compressor** parameters



Here another example for **<EBU Meter>**



In this case the **<Hotkeys>** will control the program based loudness measurement process defined by **EBUR128**. The display represents the measurements of **Integrated-** / **Short Term**- and **Momentary-Loudness** as well as the **LRA** (Loudness Range) [LU] and **Max TPL** [dBTP],
the **Maximum True Peak** level.

The measure for the EBU Meter display is **[LUFS]** (Loudness Units Full Scale) as long as not defined differently.

For details pls. refer to the EBU-Tech 3341 document.

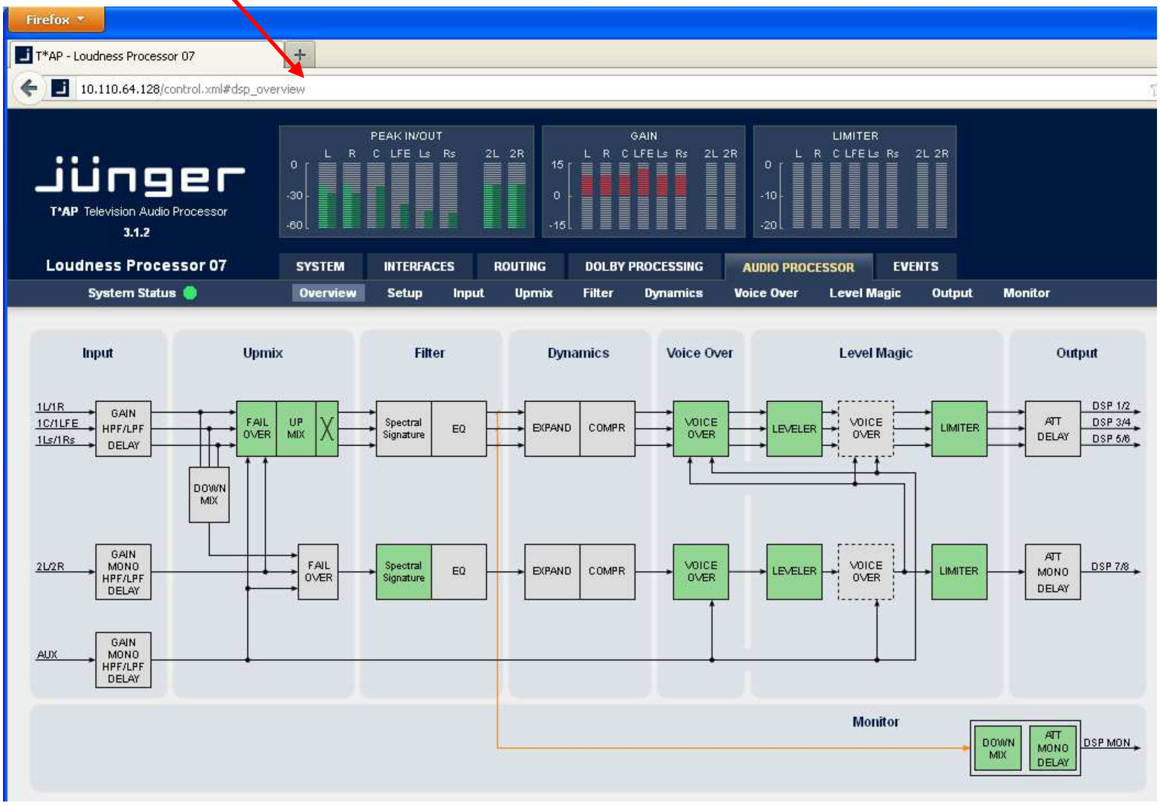
operating – menu structure of the X*AP Remote Panel – **menu tree**

Power Up Display
<MENU> opens X*AP Remote Panel IP setup menu.
<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

After connecting with a **Base Unit** the Main Display opens up :
Main Display
<ESC> will jump back to **power up** display
<MENU> opens **Operating** display:
Hotkey #
1 <Empty>
2 <Audio Processor>
1 <Input>
2 <Upmix> [page 1 - 2]
3 <Equalizer> [page 1 – 5]
4 <Spectral Signature>
5 <Dynamics> [page 1 - 2]
6 <Level Magic> [page 1 - 3]
7 <Output>
8 <Monitor> [page 1 - 2]
<ESC> back to Menu
3 <Empty>
4 <EBU Meter>
1 <empty>
2 <empty>
3 <reset>
4 <pause/continue>
5 <empty>
6 <reset max>
7 <empty>
8 <Program_x>
<ESC> back to Menu
5 <empty>
6 <empty>
7 <empty>
8 <empty>
<ESC> back to **Main** display

setup GUI – connecting with the **Base Unit** – AUDIO PROCESSOR > **Overview**

You must open a browser and enter the **IP address** of the **Base Unit** into the **URL** field and press **<Enter>**. The browser will fetch the necessary information and opens the entrance page :



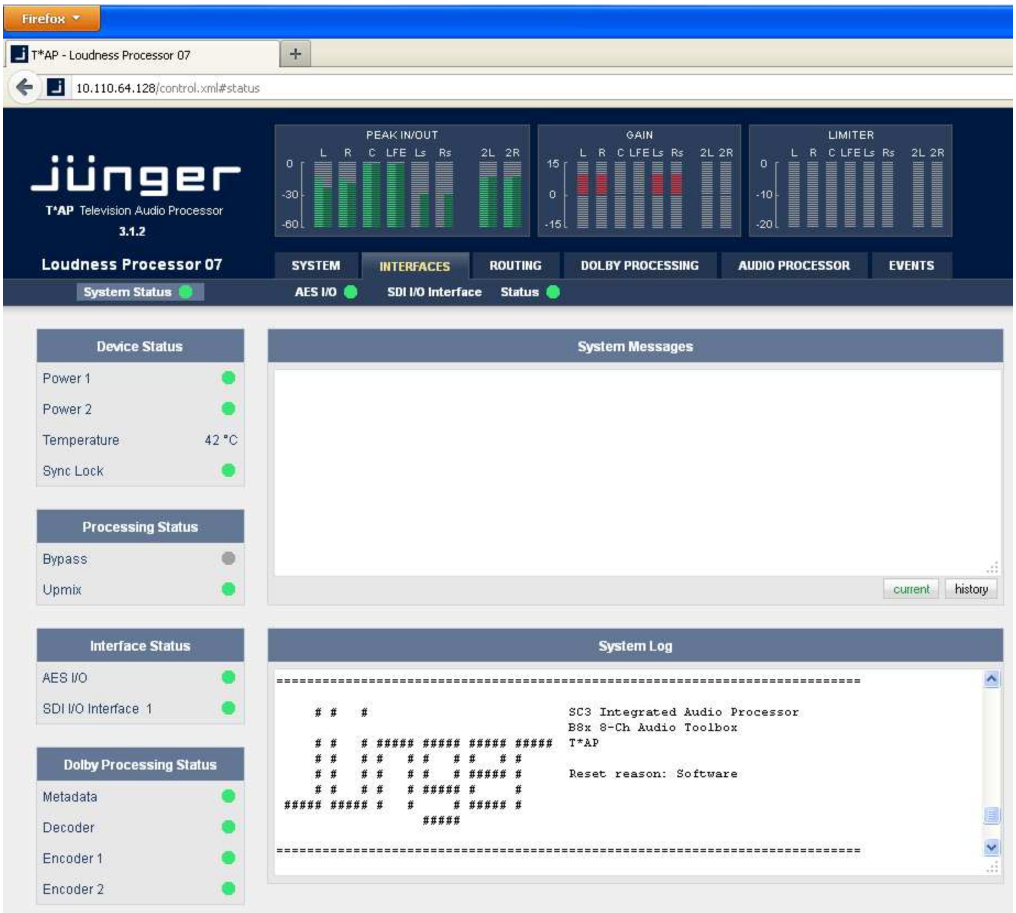
It is the **AUDIO PROCESSOR** pane with its sub pane **Overview.**

On the following pages we will go through the various panes of the setup GUI.

Firstly you must set up basic things such as program configuration, give the programs meaningful names and set the synchronization source. You may also give the device a name, tell it its location and define an administrative contact which may be used by monitoring systems of your house (e.g. via SNMP).

These settings you will find under the **SYSTEM** link.

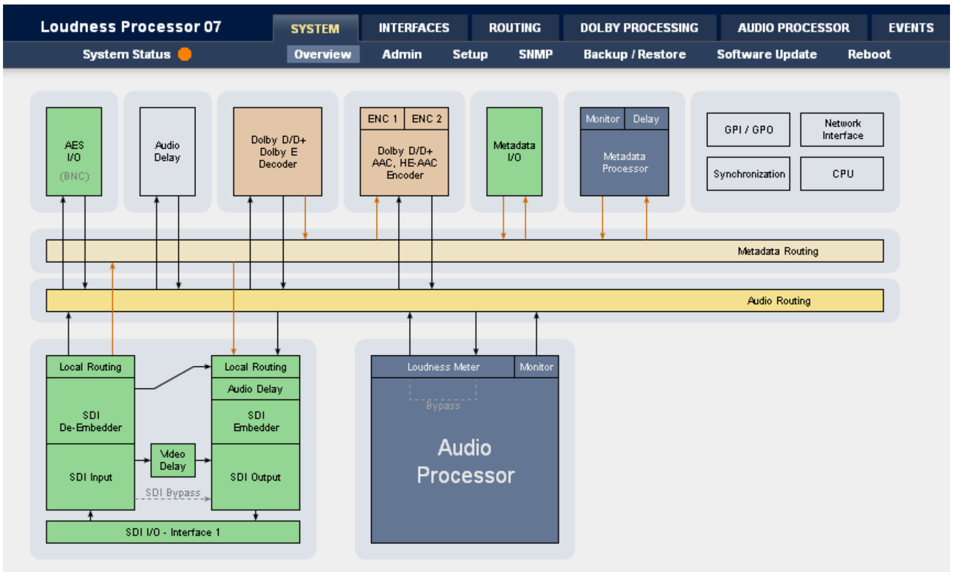
setup GUI – SYSTEM – **System Status**



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 **Base Unit**
Power 1	status of the first power supply (left hand side from rear)
Power 2	status of second power supply (right hand side from rear)
Temperature	measured on the surface of the main PCB
Sync Lock	turns red if the external sync source is removed or unstable
Processing Status	
Bypass	turns red if Bypass is activated
Upmix	turns green if Upmix 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
SDI I/O Interface	turns red if the SDI input is not locked (no or bad SDI signal)
Dolby Processing Status	
Metadata	turns red if Metadata are not valid
Decoder	turns red if the input signal is **not** Dolby encoded (PCM or corrupted)
Encoder 1	status of the first encoder (if optional CAT561 is installed) status of the D-E encoder (if optional CAT569 is installed)
Encoder 2	status of the second encoder (if optional CAT561 is installed)

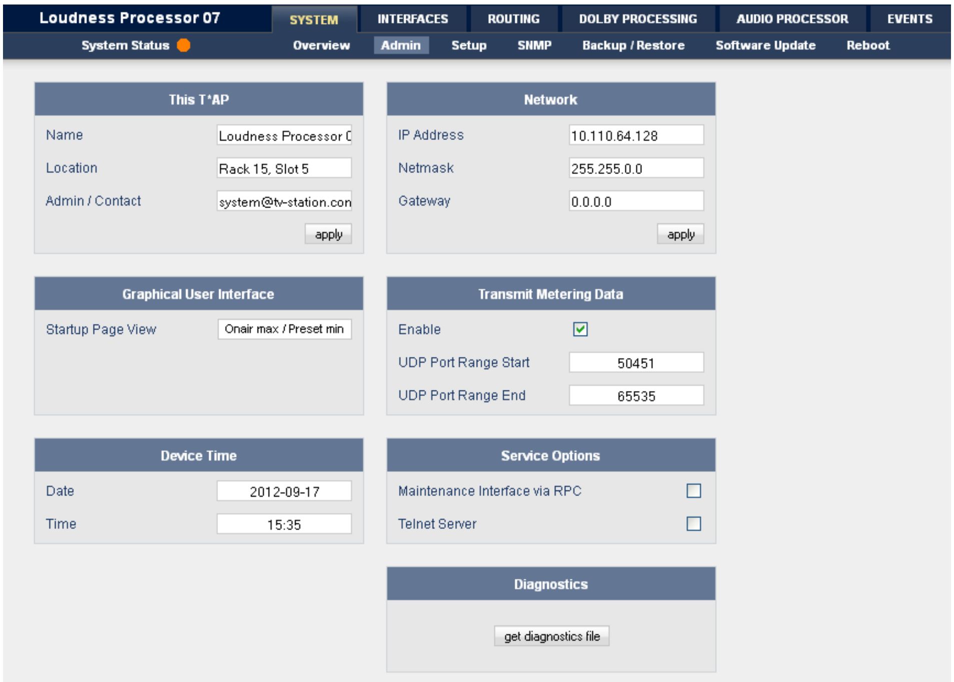
setup GUI – SYSTEM – **Overview**



The graphical overview shows the main building blocks of the device including the options installed such as Dolby OEM modules or interface modules.

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

setup GUI – SYSTEM – **Admin**



This T*AP input fields for information utilized by higher level services.

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

Location the place where the **T*AP** 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).

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, a comfortable calendar tool will pop up :

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

Network IP address setup, see above: getting started – IP setup of the **Base Unit** – **via web browser**

IP Address

Netmask

Gateway

Transmit Metering Data metering data will be streamed via UDP protocol. In order to receive such data by external applications you must define ports (port range) for matching fire wall definitions.

Enable enables UDP port range use by the device for transferring meter data from the **Base Unit** to the PC where the browser resides.

UDP Port Range Start lowest port number.

UDP Port Range End highest port number.

Service Options

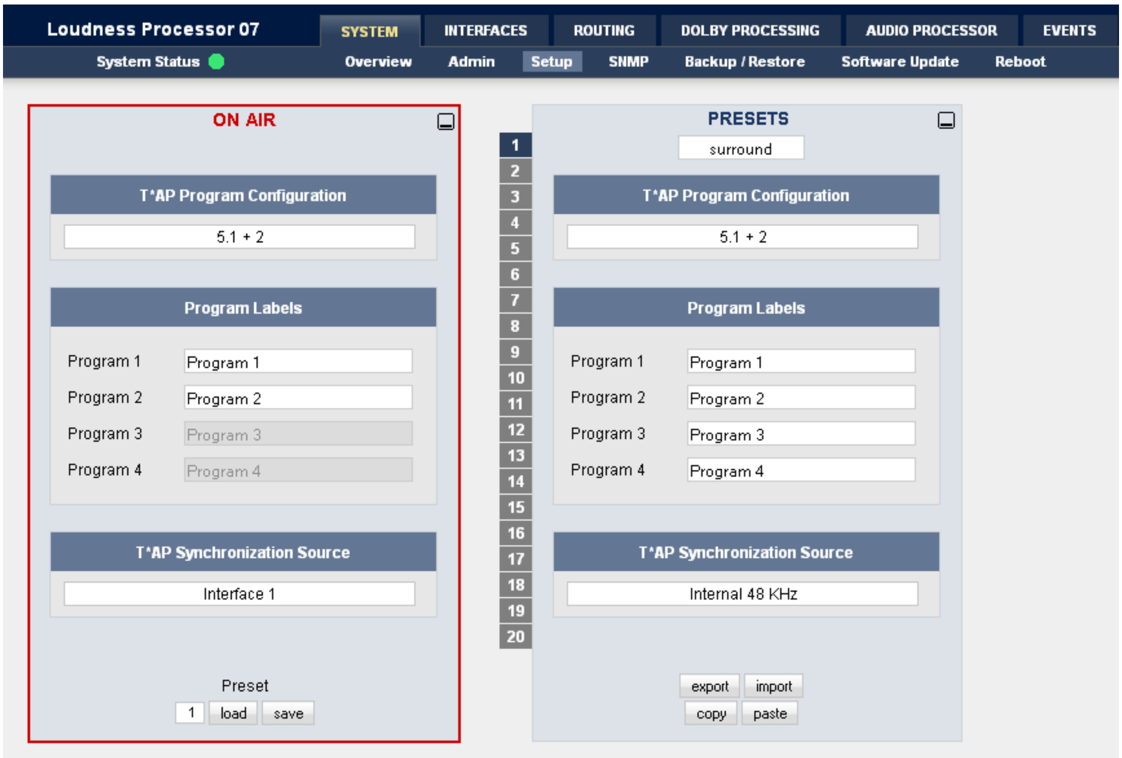
Maintenance Interface via RPC for in house use to enable communication with factory tools**.**

Telnet Server 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 service team to send such file by e-mail for analysis.

setup GUI – SYSTEM – **Setup**



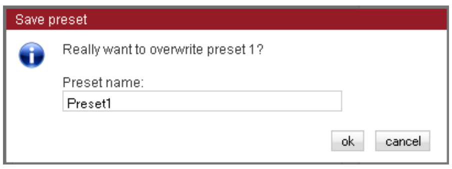
T*AP Program Configuration here you may select between 5.1 + 2 and 4 x 2. This will automatically configure all audio processing parts processors

Program Labels each of the individual programs (two in 5.1 +2 and four in 4 x 2 has a name that will be used as a reference for the display of parameters and its setup.

T*AP Synchronization Source with this pull down you may select between the available sync sources :
Internal 48kHz, External AES, Input AES 1/2, External WCL, Interface x (if an option board is installed Black Burst or TriLevel).

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

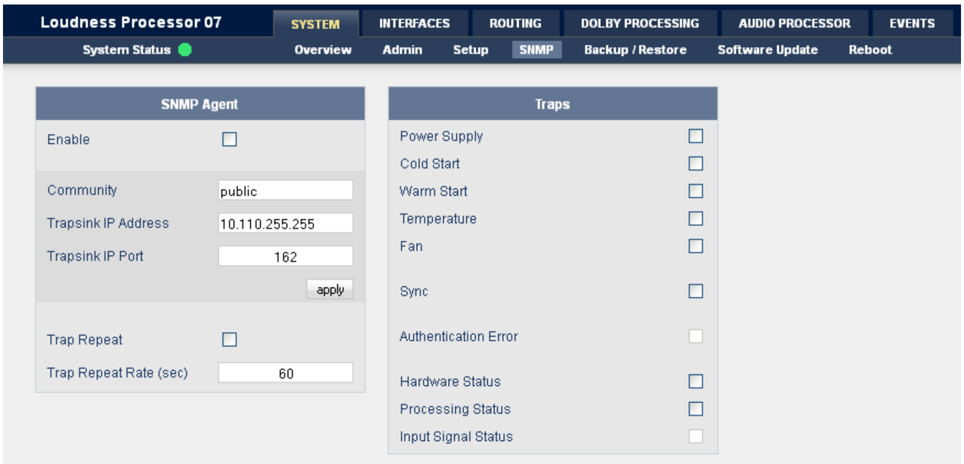
The example above shows the **preset concept** of the **T*AP.** It is the central theme of the device.
For all relevant setting of the device one set of **ON AIR** parameters and **20** sets of **PRESETS** are available
If you want to load parameters from a preset or save parameters from the **ON AIR** area to a preset, you must first select a preset number at the bottom of the **ON AIR** page.
You must press [1] to open the pull down list to select the desired preset.
Pressing [load] will execute it. When you press [save], you will be asked in a pop up :



to overwrite the selected preset and to give it a (new) name.
[copy] [paste] acts as a clip board for the parameters of individual presets,
while [export] [import] will allow you to store / recall the set of **20 presets** to / from the PC file system.

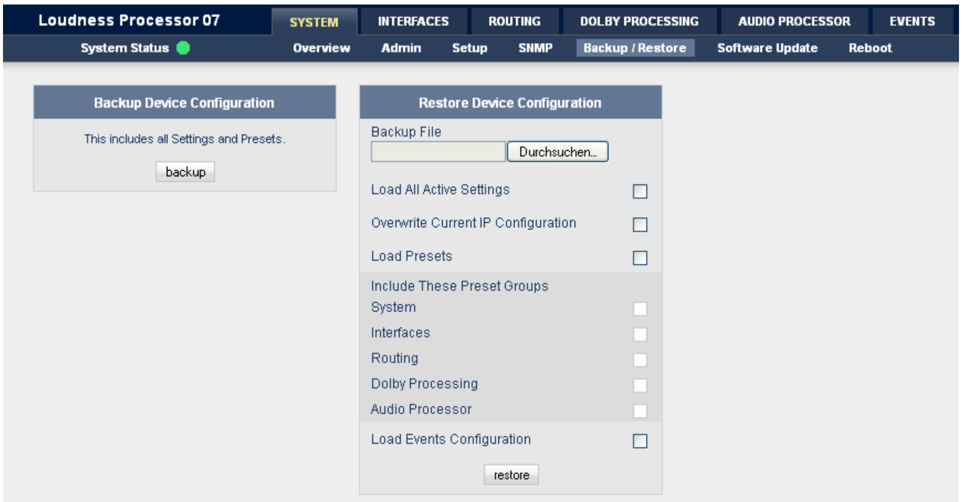
Important note! The presets of the **T*AP** are persistent by nature. You are working directly on the preset memory, i.e. you must not worry about storing such presets. The **T*AP** does it for you.

setup GUI – SYSTEM – **SNMP**

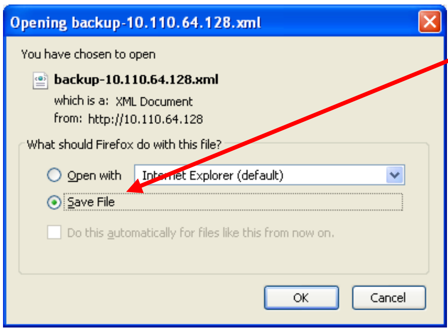


This pane is meant for basic settings of the **SNMP Agent** of the device. If you are not familiar with the use of SNMP protocol for system monitoring you should not enable the SNMP agent.

setup GUI – SYSTEM – **Backup / Restore**

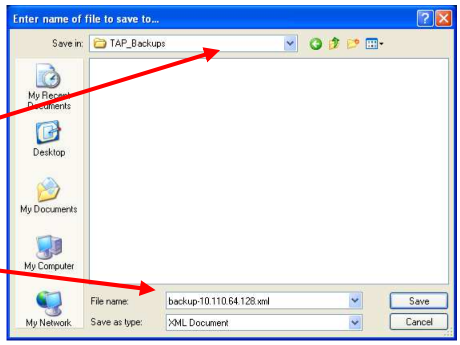


Here you can **backup** the complete **device** and **restore** parts or all of it .If you press backup the device controller will collect all necessary data and assemble it to an XML file. Finally you will get a pop up message:



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

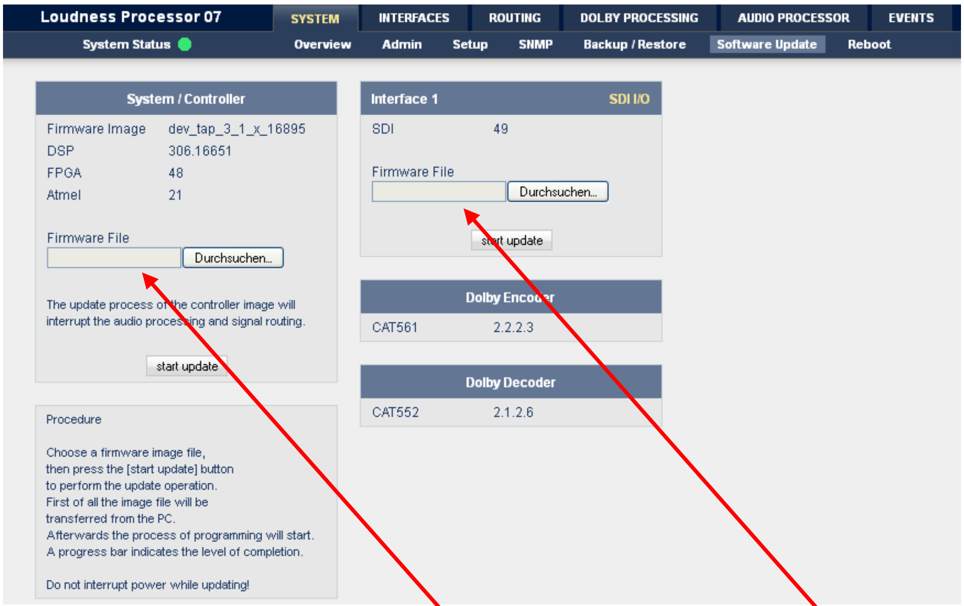
Select a folder
and alter that default file name if needed.



set up GUI – SYSTEM – **Software Update**

The files to update the TAP 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 TAP core system in the format : "rel_tap_x_y_z.img"
as well as update files for components, like the optional interface boards in the format : "rsdi150_v47.sdi" or for **Dolby** CAT (OEM) modules or for the **X*AP** Remote Panel.

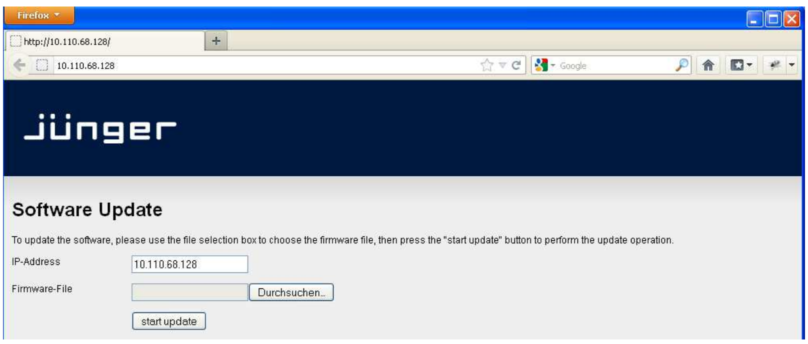


To update the **Base Unit**, you must **<Browse …>** for the respective Firmware File(which you have unzipped before) and press start update
After finishing the procedure the device will reboot.

You may also update the firmware of an SDI board installed in one of the two interface slots (or both). The example above shows one SDI I-O board installed into interface 1 slot. You must select the appropriate file from the firmware update bundle (ZIP file) and press start update afterwards.

browser based set up – **firmware update of the X*AP Remote Panel**

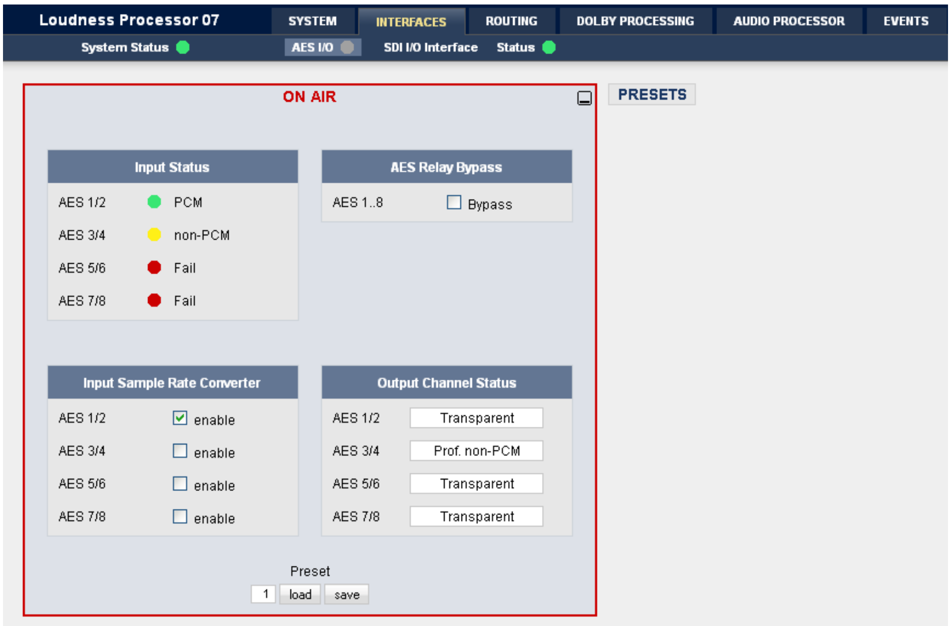
You must open a browser and enter the **IP address** of the **X*AP** Remote Panel into the **URL** field :



You must select the respective file and press :
start update

After finishing the procedure the **X*AP Remote Panel** will reboot and you must manually reconnect the **Base Unit** you are about to control.

set up GUI – INTERFACES – **AES I/O**



Input Status

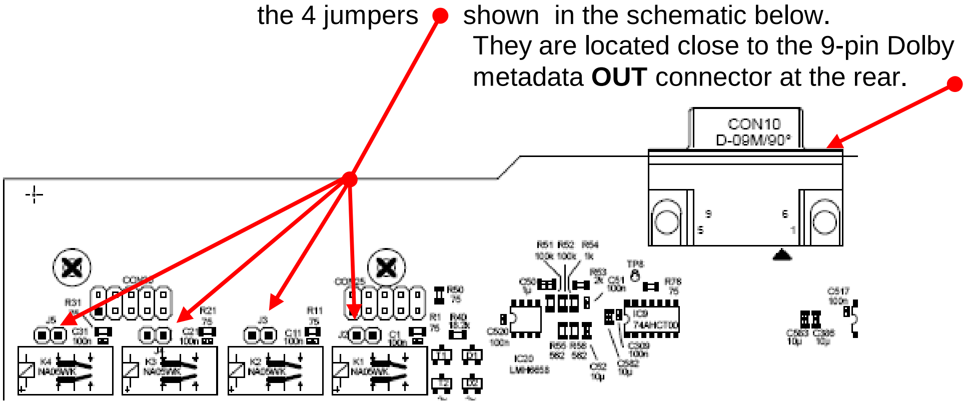
each AES input has a status detection that may show :
PCM, **Non PCM** (e.g. Dolby encoded signal) or
Fail (no carrier, unlock, cranky [too much jitter]).
The **non PCM** status will be retrieved from a logical combination of the Validity flag and the channel status.
If one of the inputs is not assigned by the ROUTING section, its status will not be incorporated into the System Status.

Input Sample Rate Converter

for asynchronous sources it is possible to turn a **SRC** on per input.
If a **SRC** is turned on and the input status becomes Non-PCM, the respective **SCR** will be turned off automatically in order to maintain the original data structure of an encoded bit stream like Dolby E.

AES Relay Bypass

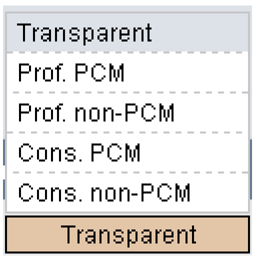
the power fail bypass relays of all 4 I/Os may be activated manually.
It is possible to exclude AES I/Os from the relay bypass circuit.
You must open the cover plate from the **Base Unit** and locate the 4 jumpers shown in the schematic below.
They are located close to the 9-pin Dolby metadata **OUT** connector at the rear.



You must remove the jumper to exclude the respective AES I/O.

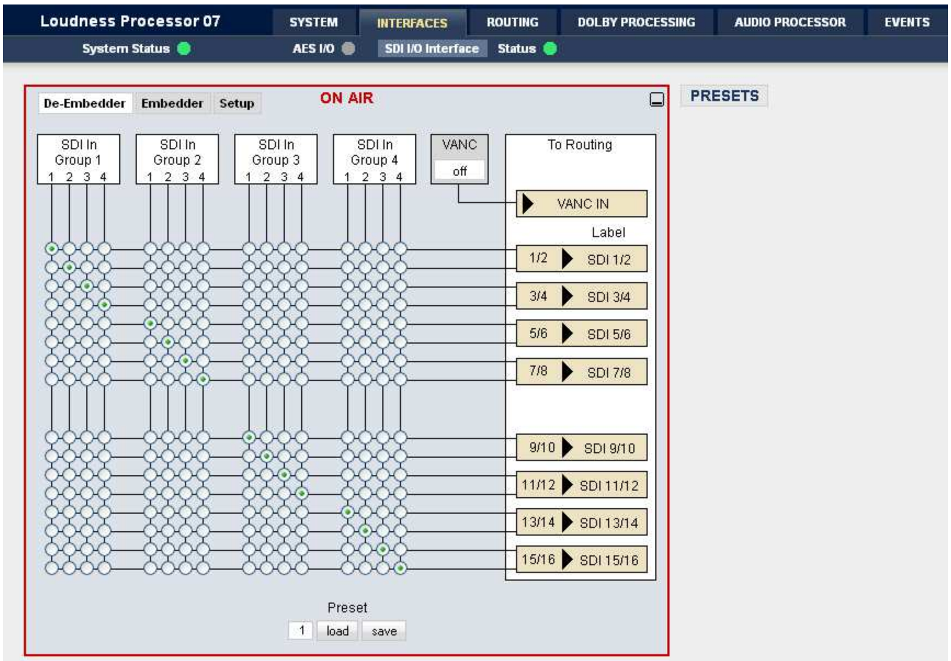
Output Channel Status

the channel status can be either transparent from the input source of the **T*AP** or may be overwritten.
The pull down offers these options :



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

This pane has three more sub panes implemented

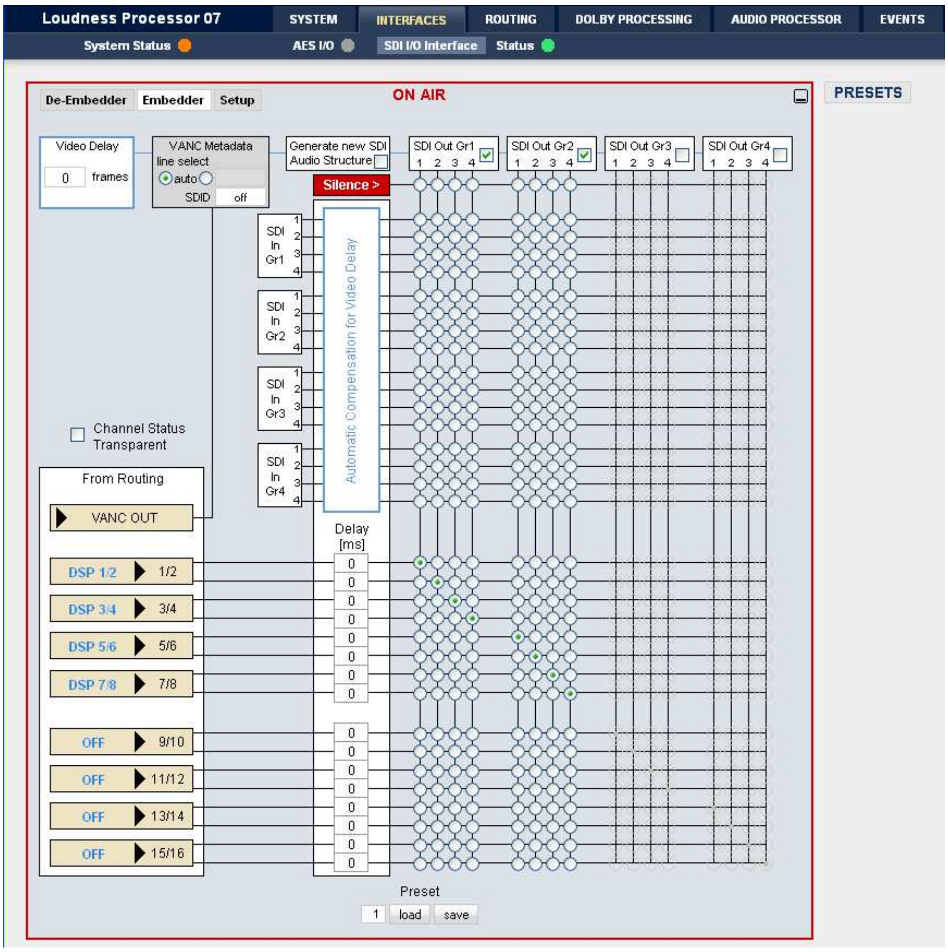


The **De-Embedder** has a 16 x 16 matrix to allow for any combination of audio signals to be presented to the **T*AP** because inside the **T*AP** the signal routing is oriented in pairs. I.e. the label "SDI 1/2" represents two audio channels selected by the matrix.

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

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

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



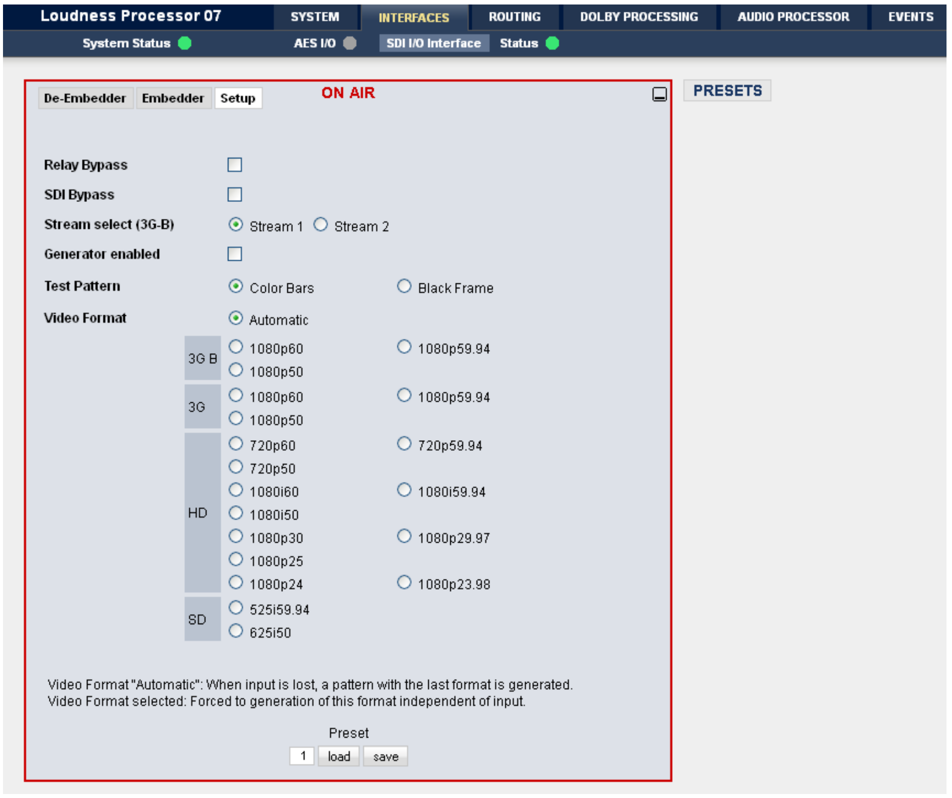
"DSP 1/2" …. "OFF"	signal labels from the **T*AP** router that shows the origin of the signal pair presented to the embedder.
Video Delay	For compensation of any kind of audio processing delay within the chain of devices you may use a **Video Delay**. Position "0" turns off the delay function.
Generate new SDI Audio Structure	If there is the 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 untouched 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 **T*AP** in any combination. If they are taken from the de-embedder and a **Video Delay** is introduced, the time of that Video Delay will be **automatically compensated** for those signals. For signals coming from the **T*AP** routing an **independent delay** per single channel may be used.
Channel Status Bits Transparent	For the signals coming from the C8k audio busses, 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** 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. For details see **SMPTE 2020** 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 substreams (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.

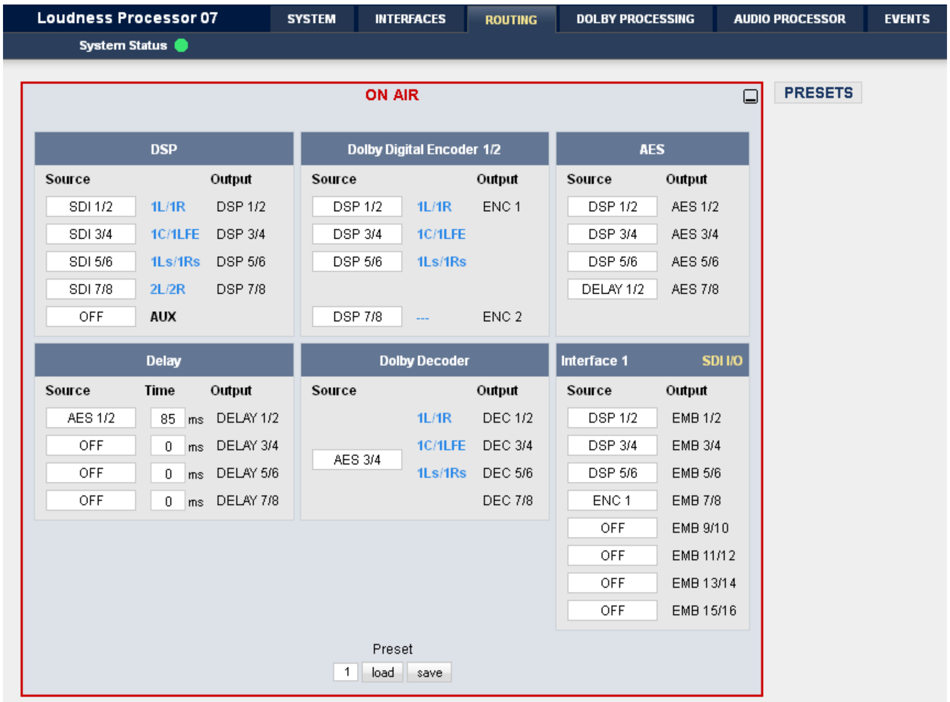
set up GUI – INTERFACES – (SDI I/O Interface) **Status**

This pan shows the status of the SDI interface (if one is installed) :

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.
VANC Metadata De-Embedder	turns green if Dolby metadata is present in the input SDI stream.
VANC Metadata Embedder	turns green if VANC embedder is enabled
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.
De-Embedder VANC Metadata	shows which **SDID** was found and gives the operator an indication which audio signals are related to that **SDID**. See **SMPTE 2020** for details.
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 **T*AP** because it defines the audio signal flow inside the machine :



Each functional block of the device has an input- and an output-label. The output-labels are pre-defined, while the label of an input must be selected by the administrator in order to route the signals.
Additional blue labels give an indication of the type of signal that is expected by the respective function block input (e.g. **1L/1R** for the DSP). The labels depend on the **Program Configuration**.

The above screen shot shows an example configuration :

DSP	the de-embedder outputs [SDI 1/2 to 7/8] are connected to the DSP 1/2 [1L/1R], 3/4 [1C/1LFE], 5/6 [1Ls/1Rs], 7/8 [2L/2R] inputs. After processing by the DSP, these signals will leave it at the outputs DSP 1/2 to 7/8.
Dolby Digital Encoder	an optional Dolby Digital encoder receives DSP 1/2 to 5/6 as an input, while the 2nd encoder has DSP 7/8 assigned. After encoding the signals appear at ENC 1 and ENC 2 outputs.
AES	the first three outputs AES 1/2 to AES 3/4 are connected with DSP 1/2 to 5/6 (e.g. for monitoring purposes), while AES 7/8 is connected to the delay output DELAY 1/2.
Delay	a signal pair from the AES 1/2 input will be delayed by 85ms.
Dolby Decoder	an external signal from the 2nd AES input AES 3/4 will be decoded. When the signal is present, the decoder reads the program configurations and sets the labels [1L/R, 1C/LFE, 1Ls/Rs] accordingly at the decoder outputs DEC 1/2, 3/4, 5/6.
Interface 1	DSP 1/2 … DSP 5/6 are connected with the embedder input EMB 1/2 … EMB 5/6 while encoder output ENC 1 is connected with the embedder input EMB 7/8. Where these signals will be embedded must be defined on the respective setup pane : INTERFACES > SDI I/O Interface > Embedder.

setup GUI – **DOLBY PROCESSING**

The Dolby® metadata system is too complex to describe it in a product manual like this.
We recommend to those who are not familiar with it, to study the many publications from **Dolby® Inc.** which you will probably find here (we can't guarantee that the link is active forever) :

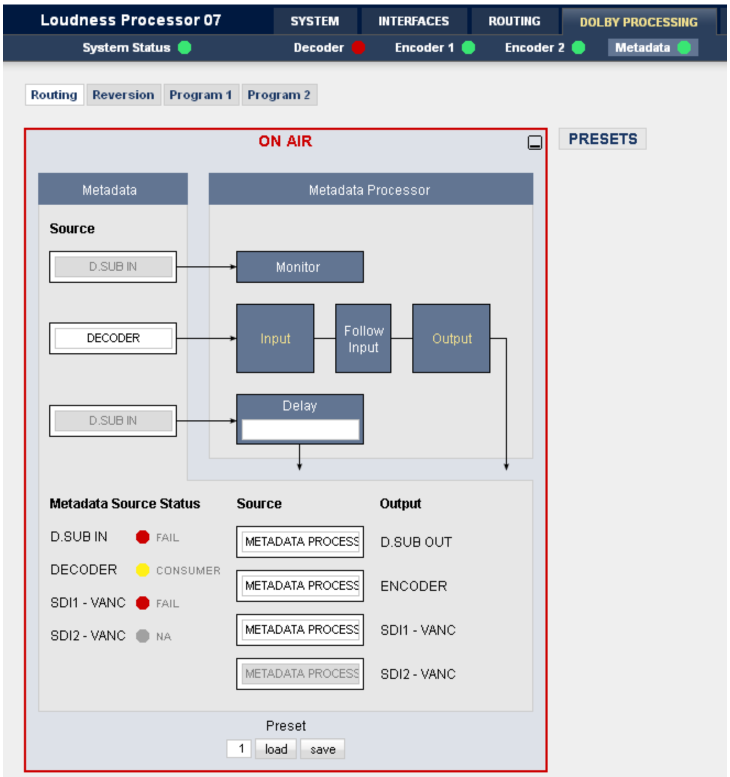
http://www.dolby.com/gb/en/professional/technology/landing.html

But so much for the beginning: The system is designed to squeeze multiple audio signals into standard 2 channel transmission / recording lines. The codecs used for this purpose are intellectual property of **Dolby® Labs. Inc**. Additional metadata has been implemented to control the customer experience at home when listening through TV set top boxes, DVD players, gaming machines, mobile devices.
Dolby distinguishes between consumer and professional metadata. While consumer meta data will travel to the consumer equipment, the professional metadata is used for the setup of encoders.

The appearance of the following pages depends on the number and type of Dolby decoders / encoders installed. These modules are an option which may be ordered with the **T*AP** or later on for field installation.

setup GUI – DOLBY PROCESSING – Metadata – **Routing**

The center of the **T*AP** Dolby processing is the built in **Metadata processor**. It can be the point of origin of metadata but it may also modify existing metadata from an available source :



The metadata system of the **T*AP** has three paths. An independent monitor will allow for the evaluation of an independent metadata set, selected by its **Source** drop down box.

The **Metadata Processor** in the middle may take it from an available **Source**, may manipulate it in the **Follow Input** section and present it to the router for further distribution at the **Output**.

The Metadata Source Status
The soft LED turns **red** If no metadata is present or the metadata is corrupted.

It turns **green** and the word **OK** will be displayed if a **RDD 6** compliant metadata stream is detected.

It turns **yellow** and the word **CONSUMER** will be displayed to indicate that only a metadata subset is provided if an AC3 or similar (D-D+) signal is decoded.

At the bottom right hand side you see the metadata output router. Here you may decide about the **Source** that feeds the respective **Output**.

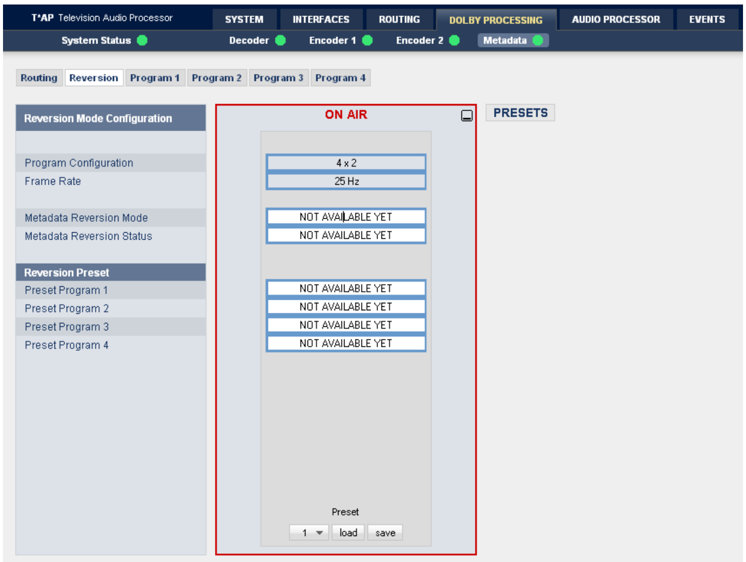
Source	[OFF / D SUB IN / SDI1 VANC / DECODER / METADATA PROCESSOR]

Important note! The metadata processor generates a full set of **SMPTE RDD 6** compliant metadata. Since the **T*AP** is designed to handle two (5.1+2) or 4 (4x2) programs all related parameters will be generated by the processor.

The function blocks Monitor, Follow Input and Delay are **not** implemented for T*P firmware release **3.1.x**

setup GUI – DOLBY PROCESSING – Metadata - **Reversion**

This pane defines the metadata sets in case of a loss of metadata from the input :

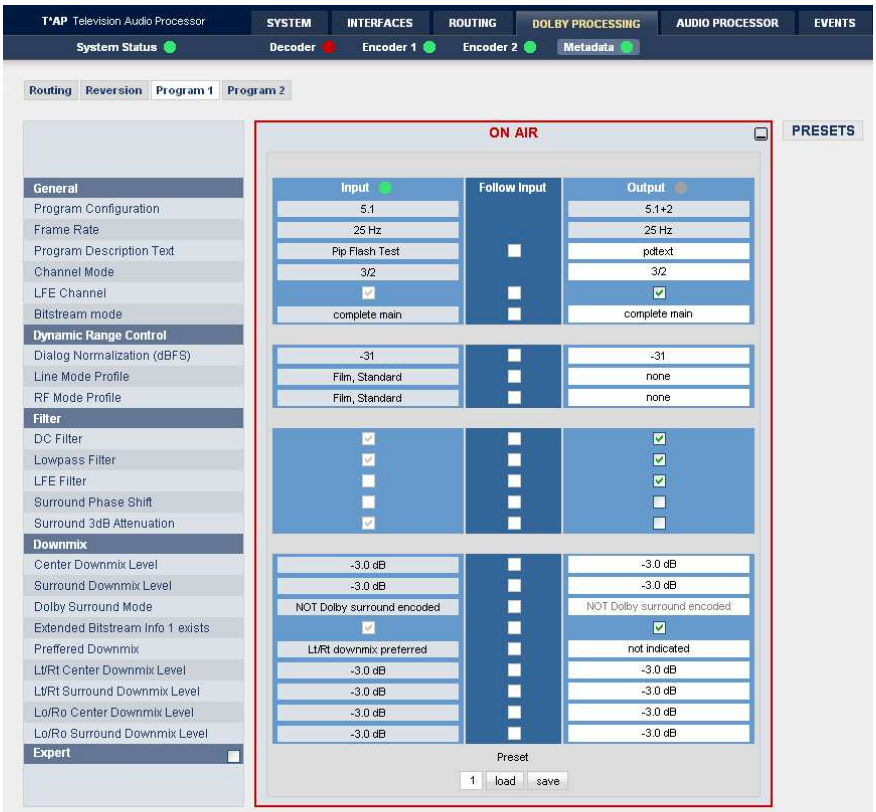


Beside D-E encoder settings, it will provide a set of alternative metadata.

This function is not available yet (release 3.2.x)

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

This pane displays input meta data and allows for setting of the various metadata separately for each program :



The follow input checkboxes are not active yet (release 3.2.x).

setup GUI – DOLBY PROCESSING **– Decoder**

Depending on the type of input stream provided to the decoder, this pane shows the incoming stream parameter and it allows to configure the decoder :

setup GUI – DOLBY PROCESSING – **Encoder(s)**

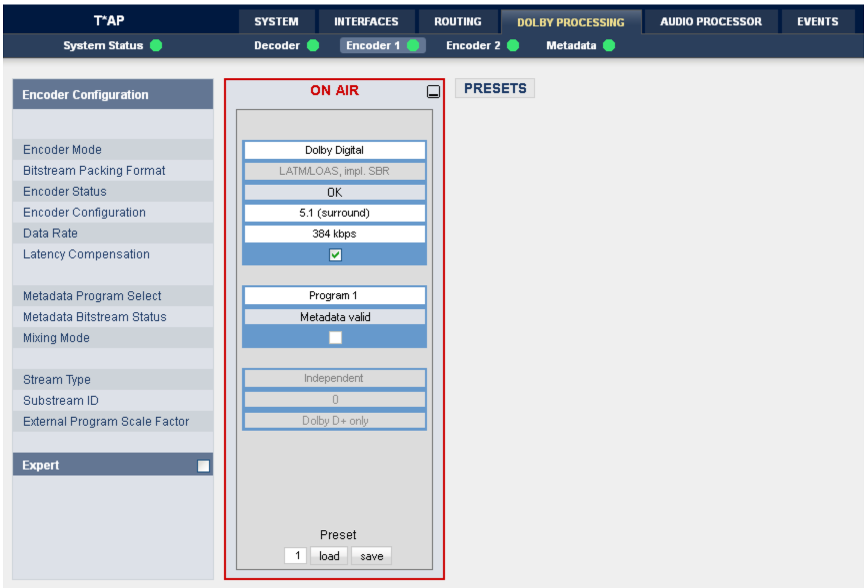
The encoder installed in the **T*AP** may either be a Dolby E encoder or a multifunctional encoder, that allows for several consumer format encoding :
Dolby Digital plus [D-D+]
Dolby Digital [D-D]
Dolby Pulse [HE-AAC v1 and v2 and AAC]

The multi format encoder has 8 physical PCM audio inputs and may be configured for **3 different operating modes** :

A) Two independent encoders (independent audio inputs Independent metadata	stream 1 – 3/2L or 2/0 stream 2 – 2/0	(D-D+, D-D, AAC, HE-AAC) (D-D+, D-D, AAC, HE-AAC)
B) Two encoders with different formats (but same audio inputs)	stream 1 – 3/2L stream 2 – 3/2L	(D-D, D-D+, AAC, HE-AAC) (D-D, D-D+, AAC, HE-AAC)
C) Two encoders multiplexed Into one output	stream 1 – (i0 + i1)	one output stream (multiplexed from two encoded signals. May be used for ATSC / DVB single PID transport multiplexes (e.g. for Visually Impaired - Audio Description – applications).

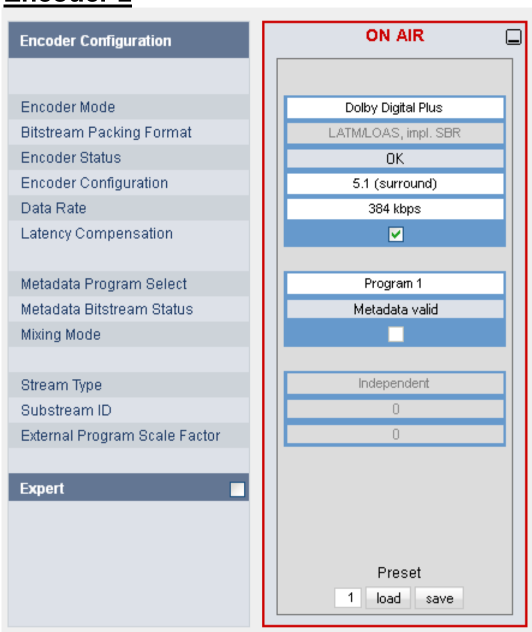
Since it is possible to run encoders in different modes (D-D, D-D+, AAC …) the latency of the encoding process will be different due to the algorithms used. If it is necessary to align the latency of the encoded outputs, you may turn on "Latency Compensation". In this case a latency of 305ms will be found for each encoder. This process is performed between encoders, i.e. you must cross check if the check boxes are **enabled** for **both** encoders!

A) Two independent encoders - configuration for **Encoder 1** :

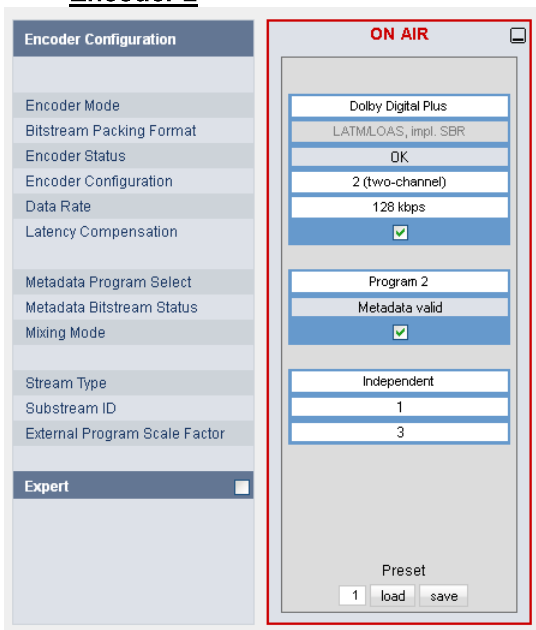


B) Two encoders mixed for **Dolby Digital plus** encoding and for **AD** (audio description) application :

Encoder 1



Encoder 2

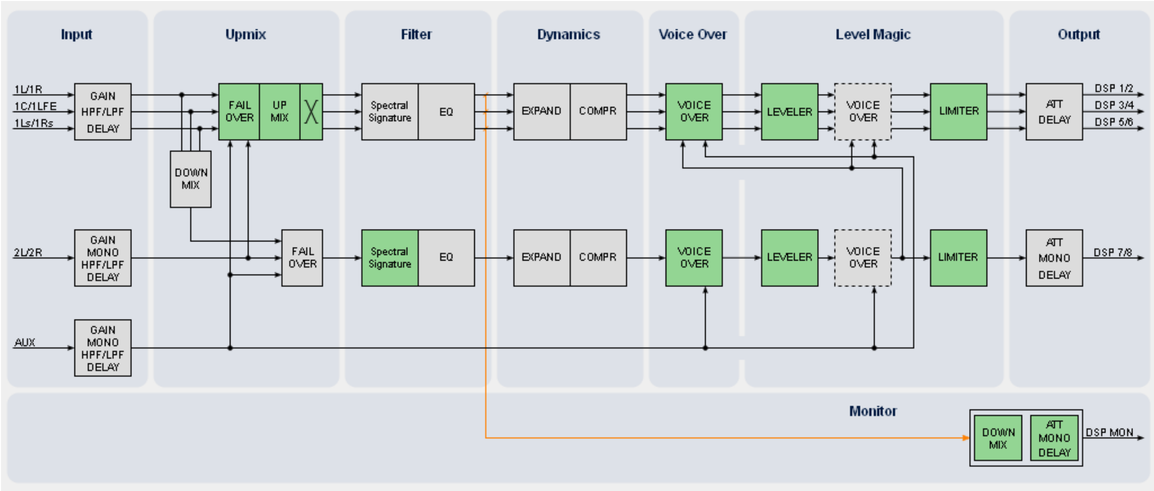


The different channel modes 3/2L [5.1 (surround)] and 2/0 [2 (stereo)], the enabled **Mixing Mode** (see expert parameters) and the different stream IDs for both encoders will implicitly set up such stream that can be used for single PID multiplexing in a DVB or ATSC DTV system.

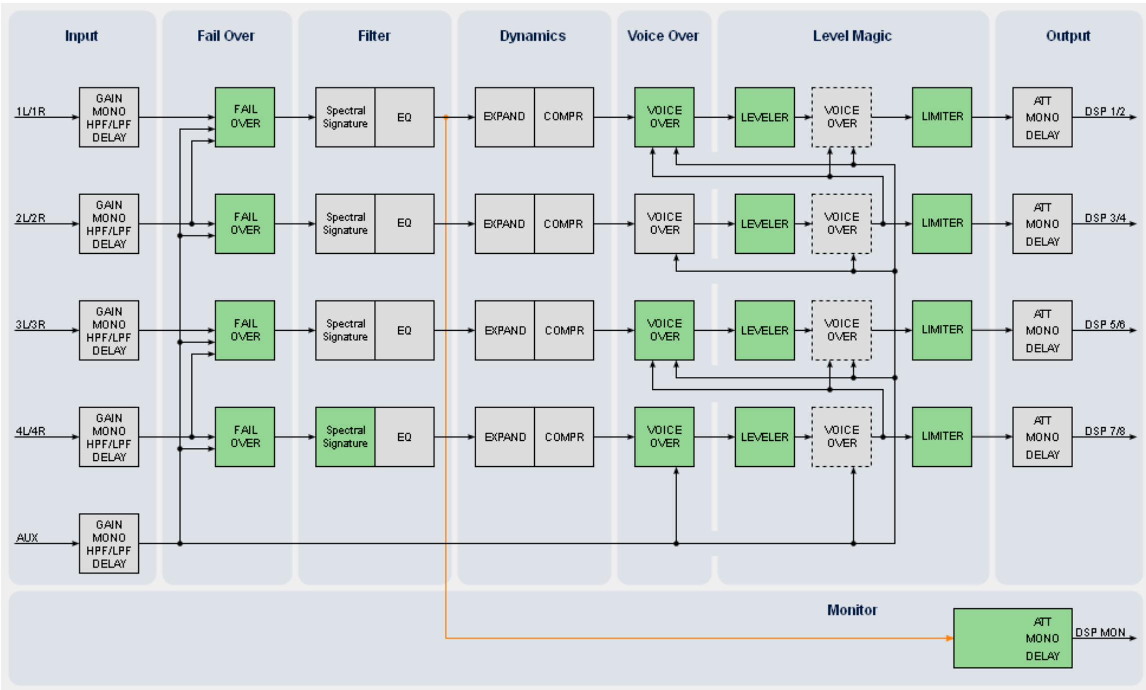
setup GUI – AUDIO PROCESSOR - **Overview**

The overview shows the actual signal routing of the audio processor blocks, rendered by the DSPs. This overview depends on the program configuration of the **T*AP**.

5.1 + 2 program configuration :



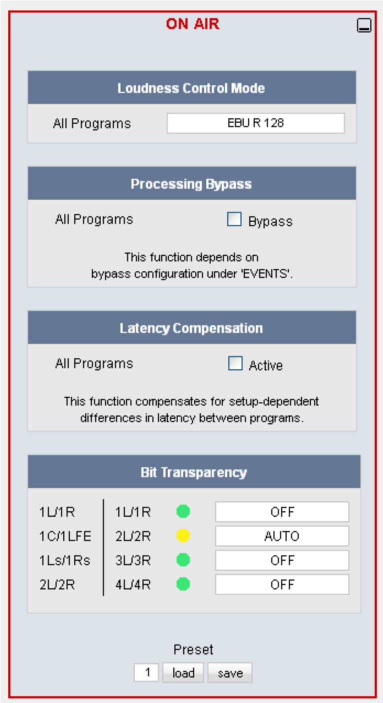
4 x 2 program configuration :



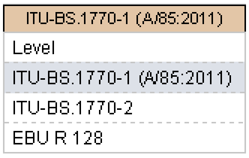
The processing blocks in use, which may be activated from their individual setup panes, will be indicated in green. Blocks shown in grey are not activated by the user. The location of blocks with dotted lines within the signal path depends on the block setup.

To navigate through the various processing blocks you may either use the mouse over function or the tabs provides in the navigation bars below the bar graph displays.

setup GUI – AUDIO PROCESSOR - **Setup**



Loudness Control Mode



the pull down offers the selection of these algorithms for the **LevelMagic™** process as well as for the loudness measurement:

Level	the **Jünger Audio** proprietary level based algorithm to achieve the same program level for different programs.
ITU-BS.1770-1	defined by the ITU and found in ATSC standard A/85:2011
ITU-BS.1770-2	enhanced ITU standard
EBU R128	defined by EBU-TECH 3341. Became the de facto standard for loudness based level control and metering in TV broadcast.

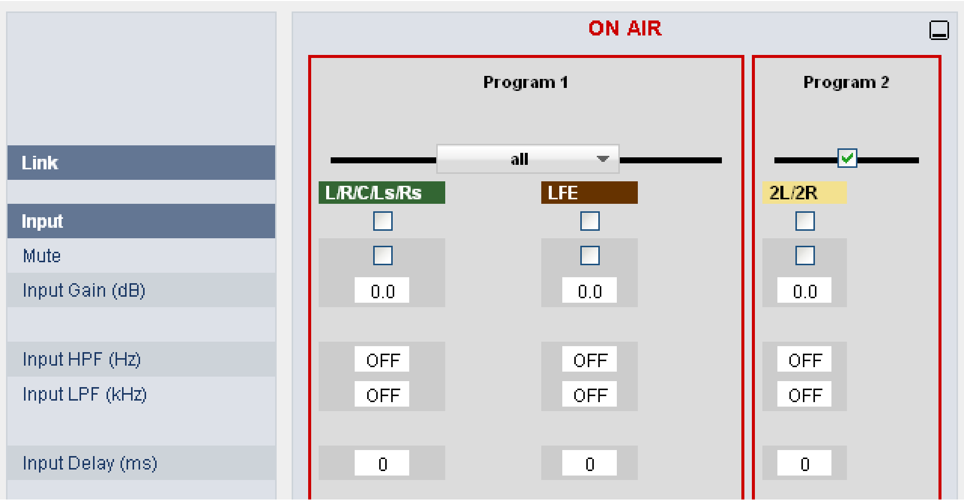
Processing Bypass will deactivate all predefined processing parameters. Which parameters are bound to the Processing Bypass function must be defined in the EVENTS section.
The audio signals still travel through the **DSP** but they are not processed, except the compensation for the processing delay of the Upmix (if Upmix is enabled).

Latency Compensation some processes like **Upmix** or **Spectral Signature** have an adjustable latency to increase the performance of such a processes. This may be compensate for program paths without latency introduced.

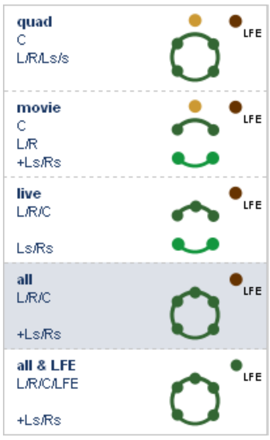
Bit Transparency **ON** will physically bypass the audio signals related to the labels on the left hand side. This function preserves the integrity of such signals if they appear in a signal path (e.g. Dolby encoded streams).
In case of **AUTO** the channel status will be observed and if **Non Audio** is detected bit transparency will be enabled.

The next pages will briefly explain the individual processing blocks.

setup GUI – AUDIO PROCESSOR – **Input**



Link



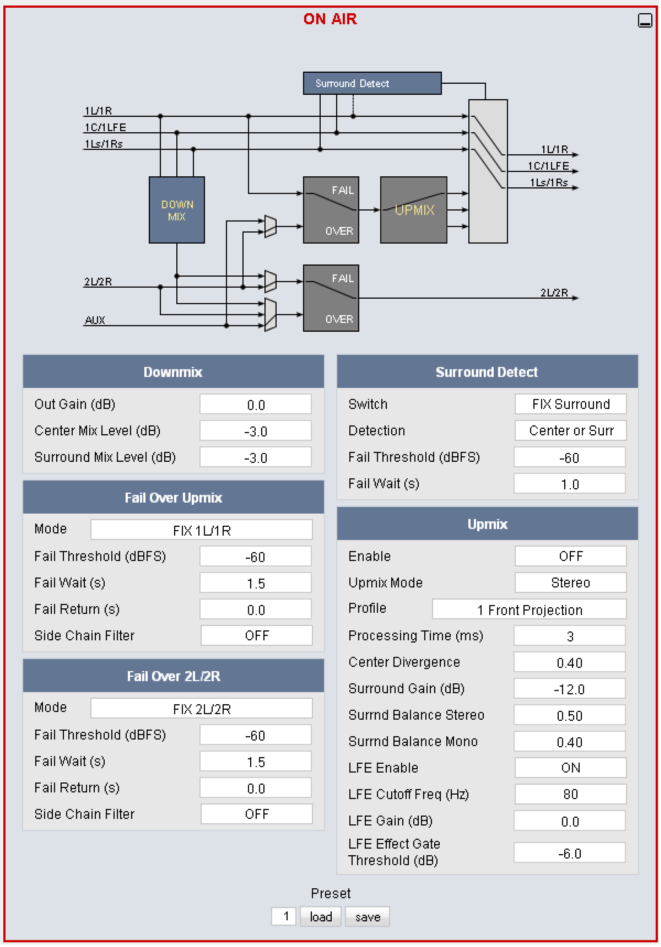
defines the coupling of the control circuits in order to maintain the listening balance for correlated signals or to provide a grouping of the setup parameters for multi channel signals. To the left is an example that shows the different link modes. This example applies in general to all other link settings for the **T*AP**.

Depending on the function block and the control mode (ITU vs. EBU) the number of possible link settings will differ.
Curves and dots of the same color indicate the link condition.

Input enables the control of the respective column

- **Mute** will mute all channels controlled by the respective column
- **Input Gain (dB)** sets the gain [-80 … +20]
- **Input HPF (Hz)** high pass filter (6dB/oct) cut off frequency [OFF, 2, 20, 40, 80, 120]
- **Input LPF (kHz)** low pass filter (6dB/oct) cut off frequency [OFF, 15, 20, 22]
- **Input Delay (ms)** [1 … 2000]

setup GUI – AUDIO PROCESSOR – **Upmix** & 2ch Fail Over (5.1+2 program configuration)



With firmware version 3.5 Jünger Audio introduced a new 5.1 upmix algorithm for upmixing stereo or even mono sources to multichannel surround sound while remaining acoustically downmix compatible. This is a real-time process which does a frequency analysis of the input signal. As known from the mathematical theory, the longer the time for such an analysis the better the result. But this will introduce more delay for the audio path, compared to the video. This delay, if acceptable in general, may be compensated by the video delay of the SDI embedder.

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

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

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

Downmix

Out Gain (dB)	output gain of the downmix signal [-20.0 … 20.0]
Center Mix level (dB)	[0.0 … -12.0]
Surround Mix Level (dB)	[0.0 … -12.0]

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

Mode

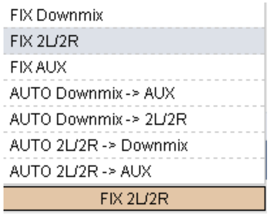
the switch may be permanently [FIX] connected with either the 1L/1R, 2L/2R or AUX input but it may also perform an [AUTO] switch over from 1L/1R to AUX or 1L/1R to 2L/2R if the first signal fails.
Both options may also be set to turn the upmix off [no Upmix] if the switch over takes place.

Fail Threshold (dBFS)	[-60 … -40]
Fail Wait (s)	[1.5 … 10.0]
Fail Return (s)	[0.0 … 10.0]

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

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

Mode



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

Fail Threshold (dBFS) [-60 … -40]

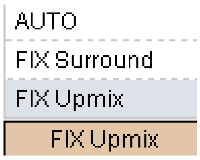
Fail Wait (s) [1.5 … 10.0]

Fail Return (s) [0.0 … 10.0]

Side Chain Filter [OFF / ON] see **Fail Over Upmix** at previous page

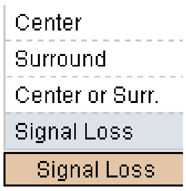
Surround Detect to perform an automatic upmix in case the main surround signal fails.

Switch



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

Detection



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

Fail Threshold (dBFS) [-80 … -40]

Fail Wait (s) [0.0 … 10.0]

Upmix

Enable [OFF / ON]

Upmix Mode [Stereo / Mono / Auto]

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

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

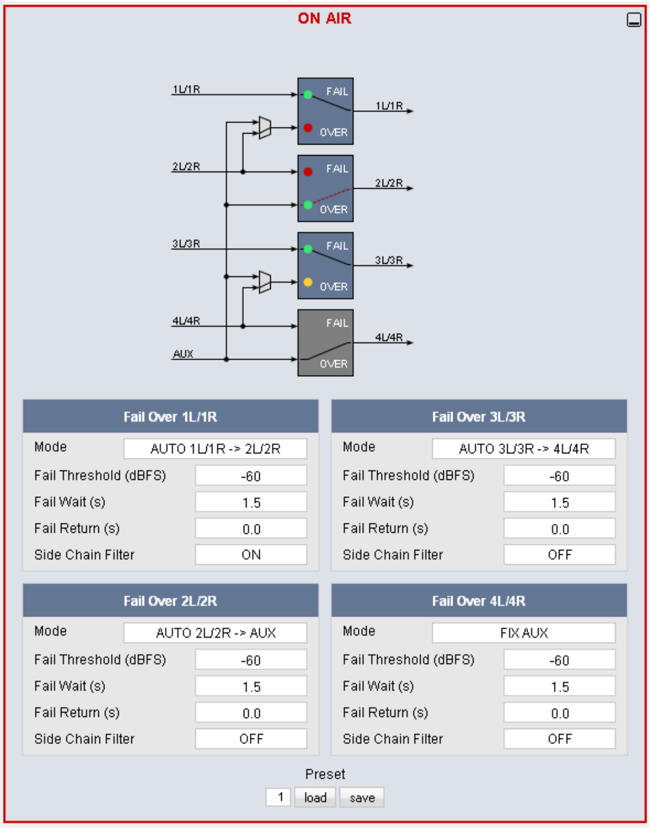
2 Emphasize Front – Based on setting 1 with a less strict front projection.

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

	4 Emphasize Surround – The distribution between the front and rear channels is highly dependent on the correlation of the input signal. Highly uncorrelated signals may create emphasized surround channels. 5 Wrap Surround – Even distribution of the signal between all channels, to create a feeling of being 'wrapped in sound' for creating spectacular effects.
Processing Time (ms)	[3 … 100] the processing time has great influence on the quality of the upmix process but of course alters the latency of the audio signal. It is highly recommended to allow as much processing time as possible. One can e.g. rise the processing time instead of adding audio delay to compensate for a delayed video line. Depending on the system latency requirements (ingest vs. live broadcast) you may change the processing time accordingly.
Center Divergence	[0.0 … 1.0] the upmix process assembles a center signal from the input stereo. It may either be fed to the center channel only (0.0) or spread between L/R (1.0). The effect will be a wider presentation of center signals in a surround sound image. Please note that the signal does not completely disappear from one source (L/R or C) depending on the selected profile.
Surround Gain (dB)	[0 … -24.0] sets the level of Ls/Rs channels.
Surround Balance Stereo	[0.0 … 1.0] defines the amount of direct sound mixed into the surround channels. 0.0 provides pure ambient sound while 0.1 to 1.0 will increase the amount of direct sound. Works only, when upmix mode is set to Stereo or switched to Stereo in Auto mode.
Surround Balance Mono	[0.0 … 1.0] defines the amount of direct sound mixed into the surround channels. 0.0 provides pure ambient sound while 0.1 to 1.0 will increase the amount of direct sound. Works only, when upmix mode is set to Mono or switched to Mono in Auto mode. For Auto mode lower values (0.2 – 0.4) are recommended to prevent unwanted effects when auto switching between Mono and Stereo.
LFE	[OFF / ON / Effect Gate] you may turn this option on if the upmix process shall generate a subwoofer signal that will appear in the LFE channel. When using the Effect Gate function the system interactively processes the subwoofer signal and generates a signal that comes very close to a real LFE signal, without creating permanent rumble and bass excitation.
LFE Cutoff Freq (Hz)	[60, 80, 100, 120] set the cutoff frequency for the generated LFE signal.
LFE Gain (dB)	[-20.0 … 20.0] you can set the LFE level here
LFE Effect Gate Threshold (dB)	[0.0 … -20.0] set the relative threshold of the Effect Gate processor.

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

For the **4x2 Program Configuration** (SYSTEM > T*AP Program Configuration) the **T*AP** offers **four** independent **Fail Over** circuits (see Overview sketch on page 34).



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

See the example above for the four program outputs :

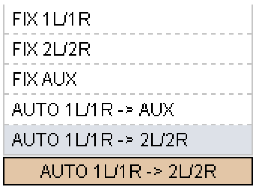
program 1 (1L/1R) has a valid input signal and is prepared for auto switch over to the second program input **2L/2R.**

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

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

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

MODE



The Fail Over output can be permanently connected to :

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

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

* or to the **AUX** input.

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

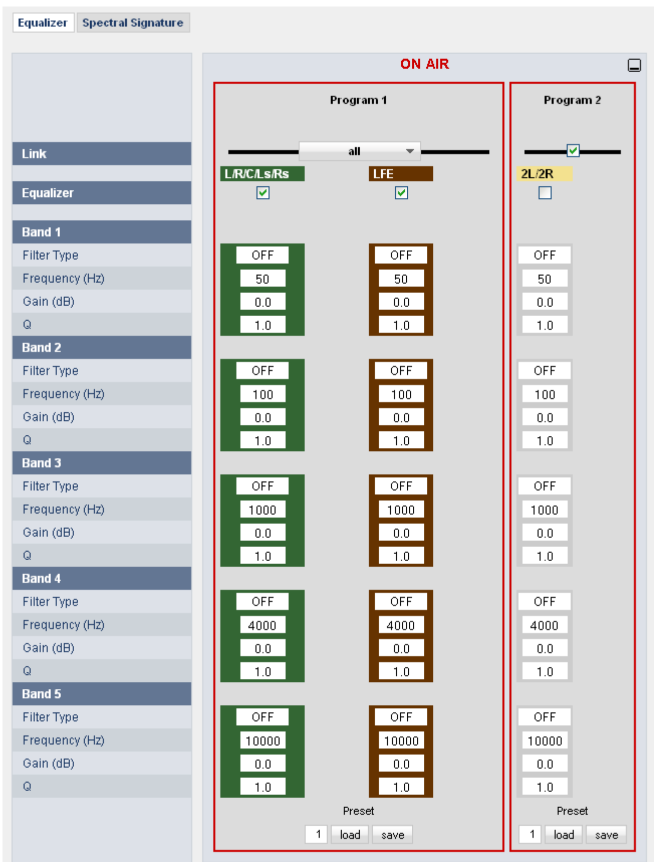
Fail Threshold (dBFS) [-80 … -40]
RMS weighted input level for fail detection

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

Fail Return (s) [0.0 … 10.0]
elapsed time after detection of a proper input signal until the switch back to the program input

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

setup GUI – AUDIO PROCESSOR – Filter - **Equalizer**



The filter section has two tabs.
The one on the left side allows the setting of five parametric EQs :

Link defines the coupling of the control circuits (see Input)

Equalizer [Enable / Disable]

Band 1

Filter Type [OFF, Lo Shelf, Peak, Hi Shelf]

Frequency (Hz) [20 … 2000]

Gain (dB) [-20 … +20]

Q [0.4 … 4.0]

Band 2 same as Band1

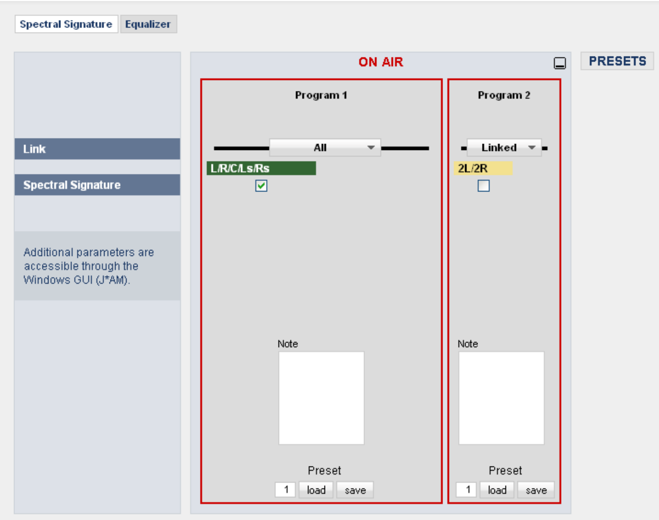
Band 3 same as Band1

Band 4 same as Band1

Band 5 same as Band1

For the 4x2 program configuration similar applies setup GUI – AUDIO PROCESSOR – Filter – **Spectral Signature**

Spectral Signature is a dynamic multiband filter. It is rendered on the **T*AP** DSP :

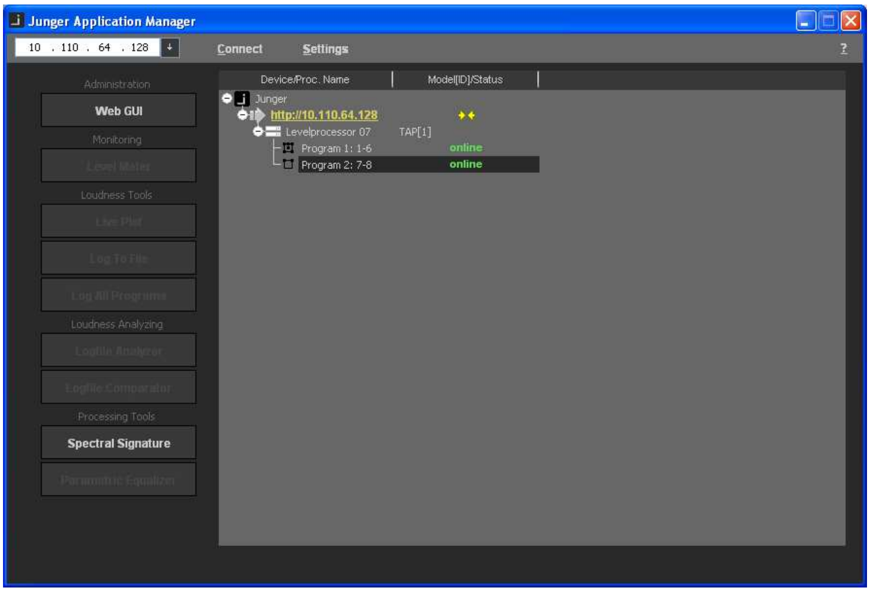


Link	defines the coupling of the control circuits (see Input).
Spectral Signature	[Enable / Disable] it must be enabled here in order to run the filter.
Note	each preset contains a reference curve

- The Signature -

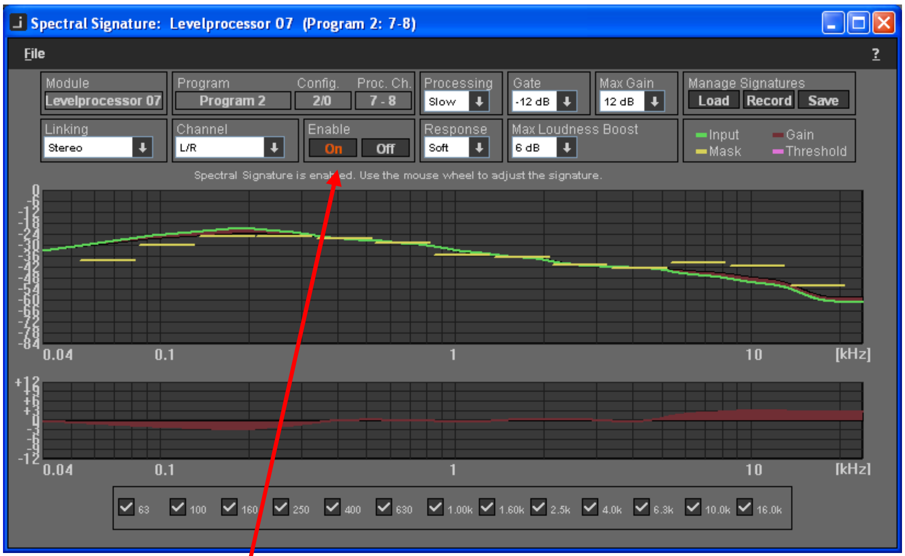
For each preset you may attach a note to describe the feature of that Signature mask.

To set up this application and to display the momentary behavior of it, you must run the Junger Application Manager **J*AP,** in order to connect it with the program channels which are processed by **Spectral Signature**. The application manager may be downloaded from the Junger website : www.junger-audio.com/download/soft-firmware



You must enter the IP address of the device and press **<Connect>** afterwards. The Application Manager will gather necessary information from the device and will display the IP address, the device name and the programs including the channels which are used by the respective program.
If you highlight a program that is enabled for **Spectral Signature** the soft button **<Spectral Signature>** becomes active.

When you press the soft button this window shows up on the PC screen :



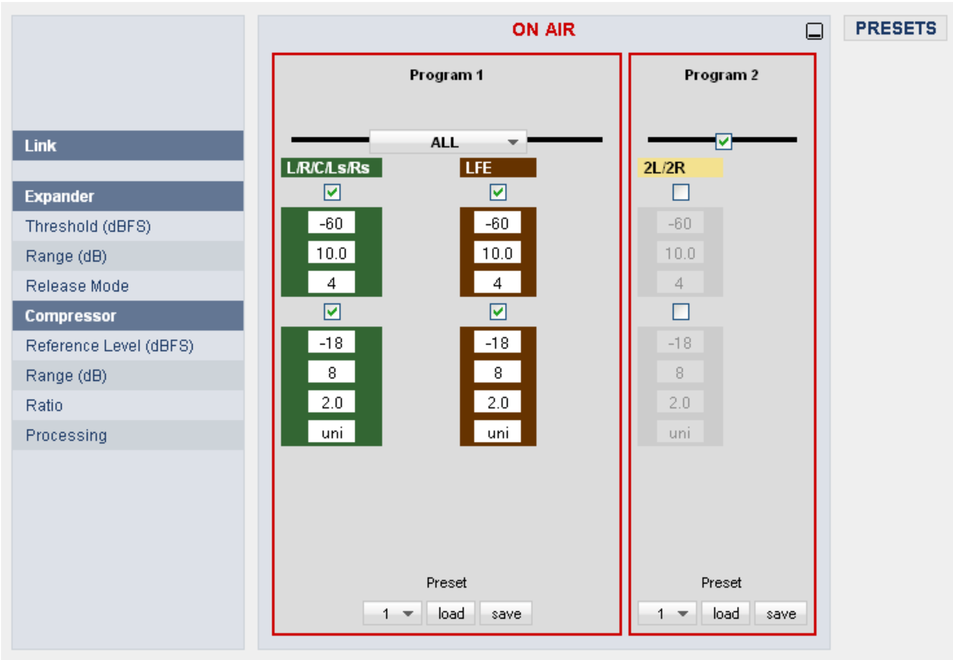
The process must be enabled in order to get the correct display. You can do it either from the **T*AP** GUI or from here. When starting this application the settings will be read from the **T*AP** and will be used and displayed here. Pay attention that **Max Gain** is not set to 0dB.

If you change settings you must store them in the **T*AP** by first selecting a preset number and pressing the **<save>** button in the ON AIR section of the **Spectral Signature** pane afterwards.

See separate manual for **J*AM** for more details.

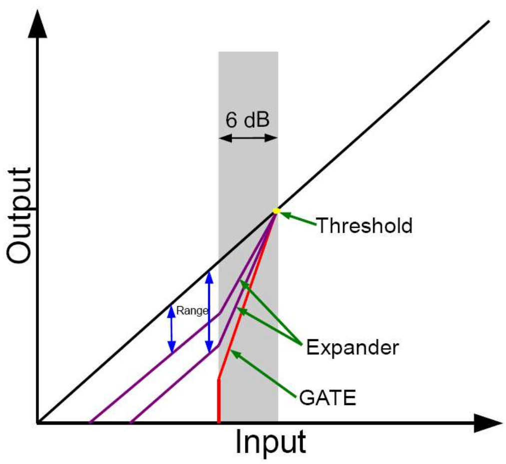
setup GUI – AUDIO PROCESSOR – **Dynamics**

An independent compressor / expander is available from here :

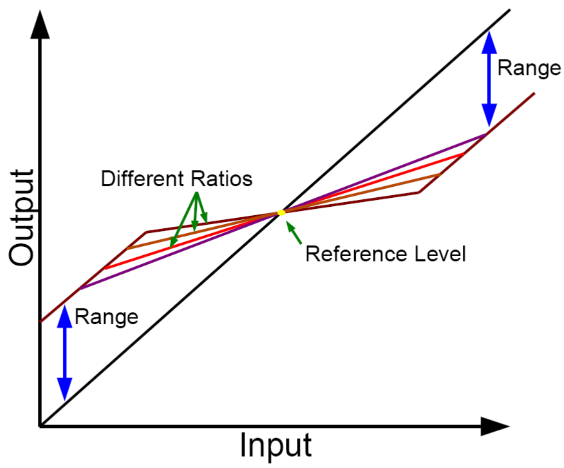


Link	defines the coupling of the control circuits (see Input)
Expander	[Enable / Disable]
Threshold (dBFS)	[-60 … -20]
Range (dB)	[0.0 …. 20, Gate]
Release Mode	[0 … 9]

simplified EXPANDER curve



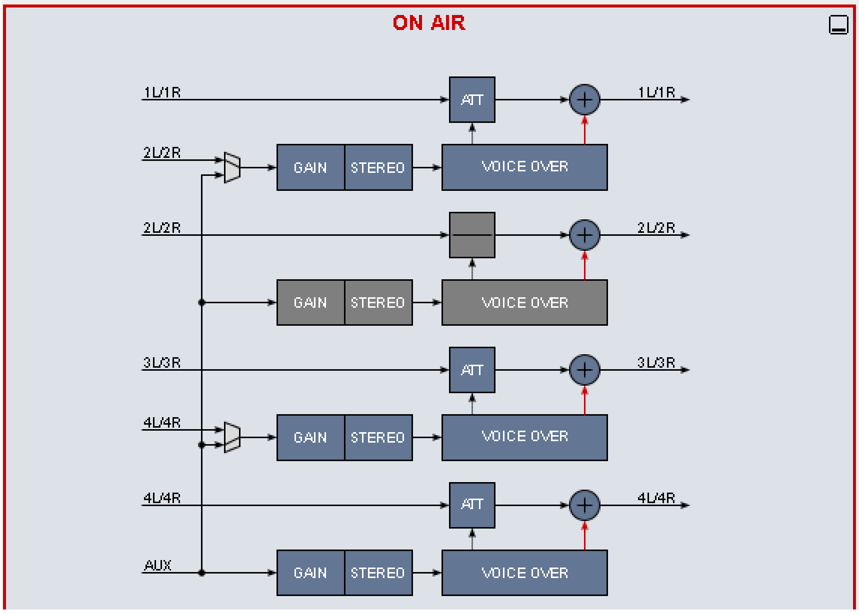
simplified COMPRESSOR curve



Compressor	[Enable / Disable]
Reference Level (dBFS)	[0 … -40]
Range (dB)	[0.0 … 20.0, Gate]
Ratio	[1 : 1.1 … 1 : 4.0]
Processing	[Live, Speech, Pop, Uni, Classic]

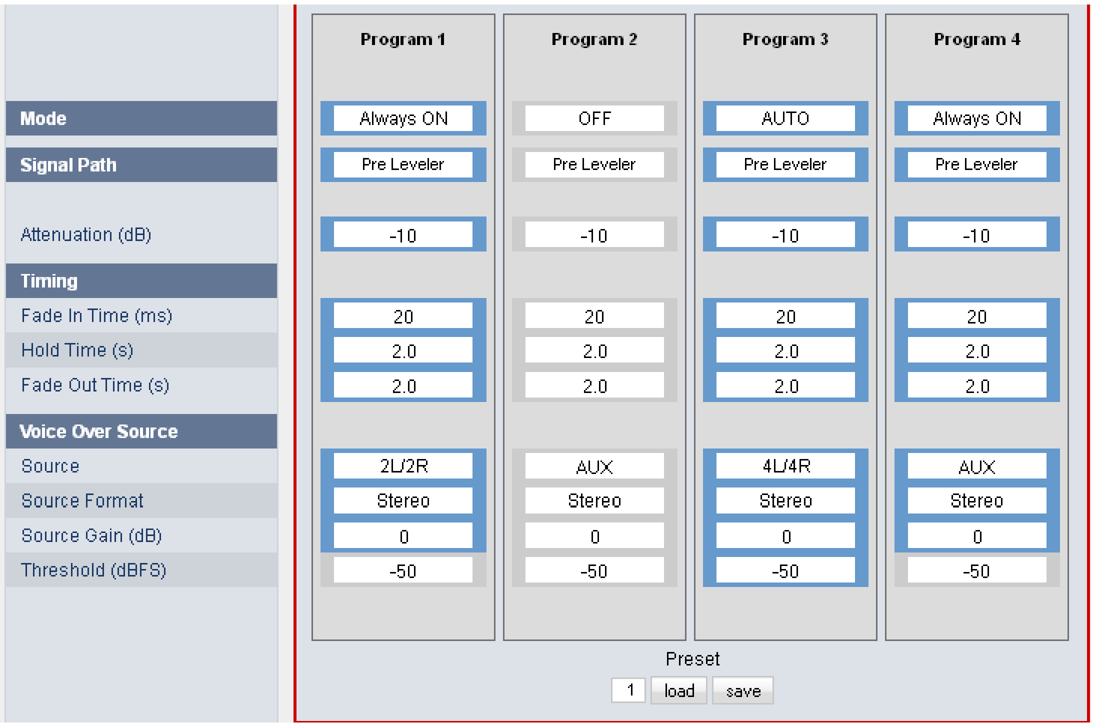
setup GUI – AUDIO PROCESSOR – **Voice Over** (4 x 2 program configuration)

Depending on the **Program Configuration** (SYSTEM > Setup > 5.1+2 or 4x2) the **T*AP** offers 2 or 4 voice over circuits. The example below shows a 4x2 program configuration :



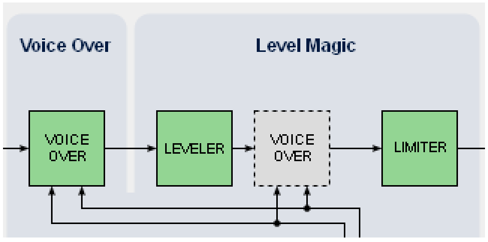
The program signal path (e.g. 1L/1R) includes an attenuator to reduce the program level and a node to mix the voice over signal with the respective program. As a source for the voice over signal you may select either the adjacent program input **2L/2R** or the **AUX** input. This allows you to build 2 fully independent voice over paths or up to 4 voice over paths with a common voice signal driven from AUX input.

Find below the parameter list for the example above :



Mode [OFF, Always ON, AUTO]
sets the voice over operating mode

Signal Path [Pre Leveler, Post Leveler]
the position in the signal path regarding the leveler processing block. You will also see it in the AUDIO PROCESSOR > Overview sketch, highlighted in green and surrounded by a solid line that surrounds the Voice Over processing block in use :



Attenuation (dB) [-30 … 0dB]
the attenuation of the program signal in case of active voice over

Timing

Fade In Time (ms) [10 … 1000]

Hold Time (s) [0.0 … 10.0]

Fade Out Time (s) [0.0 … 10.0]

Voice Over Source

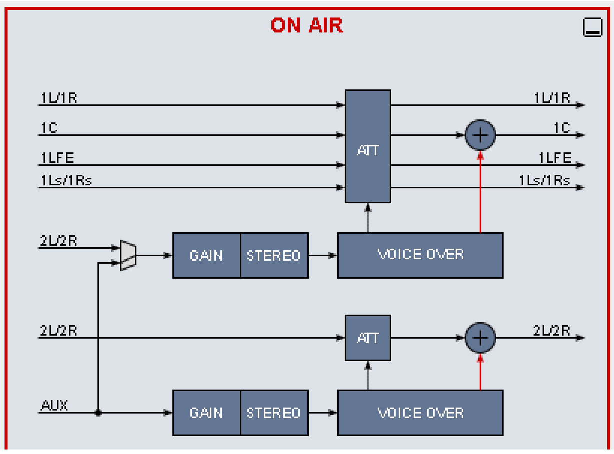
Source [2L/2R or AUX]

Source Format [Stereo, Mono LL, Mono RR, Mono L+R]
the voice feed of the Voice Over circuit is a two channel signal. You may select here, from which input channel the voice feed will be taken. LL for example means the voice signal is taken from the first input channel and it will be mixed into both program channels. Mono L+R means that a mono signal is built from a stereo input signal and is mixed to both (stereo) program channels.

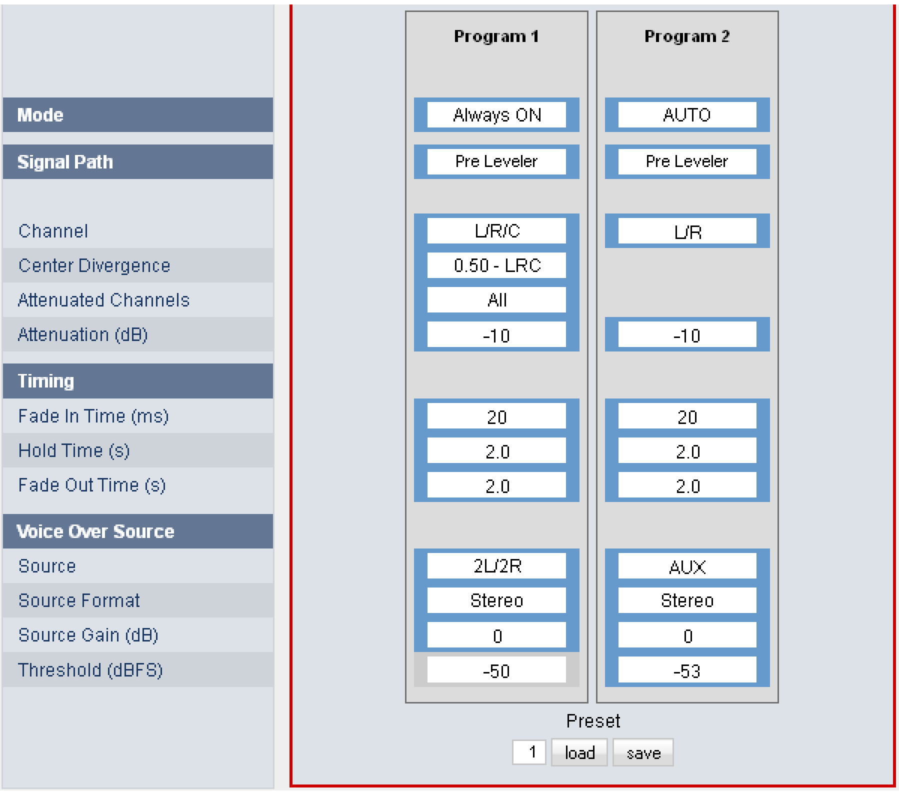
Source Gain (dB) [-20 … 20]
sets the gain for the voice signal prior to mixing.

Threshold (dBFS) [-60 … -40]
the threshold for the voice signal in AUTO mode.

setup GUI – AUDIO PROCESSOR – **Voice Over** (5.1 + 2 program configuration)



The program signal path for **program 1** is **5.1** while the **program 2** path is **stereo**.
The **AUX** input is used for the voice over signal.
This setup allows you to perform a manually or automatically controlled voice over for a surround and a stereo program with the same voice signal :

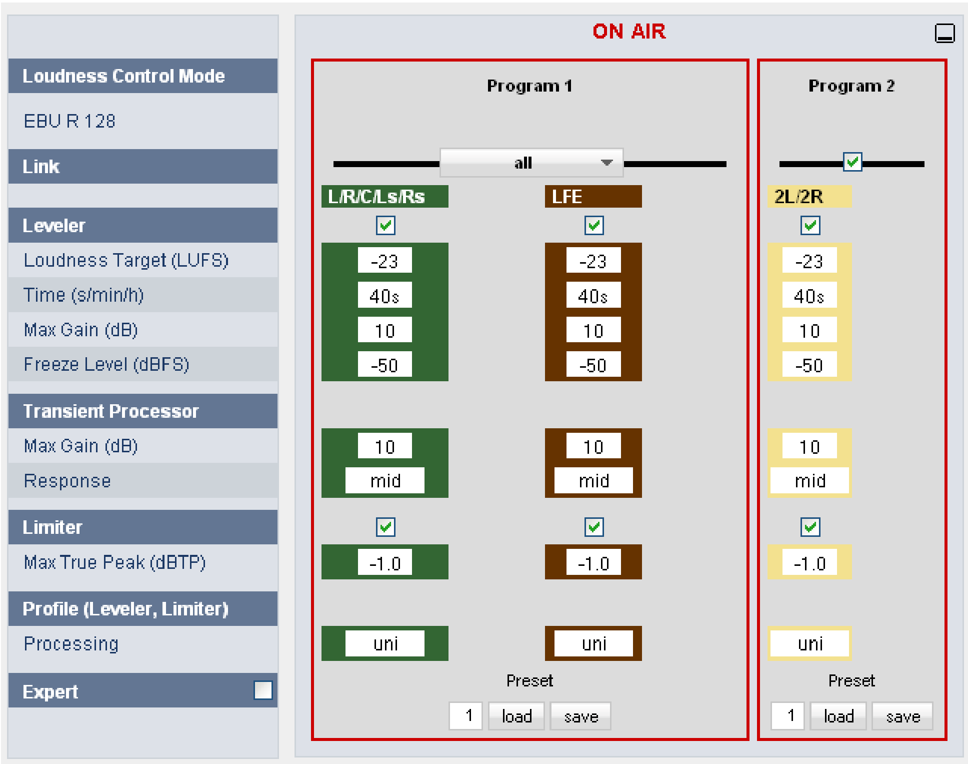


In addition to the parameters for 2Ch voice over the 5.1 circuit has these extra parameters :

Channel	[C, L/R, L/RC] selects the channel where the voice over will be mixed to.
Center Divergence	[0.00 – C only … 0.50 – LRC … 1.0 – LR only] allows to widen the projection of the voice over signal.
Attenuated channels	[ALL, Selected]
Attenuation (dB)	[-30 … 0]

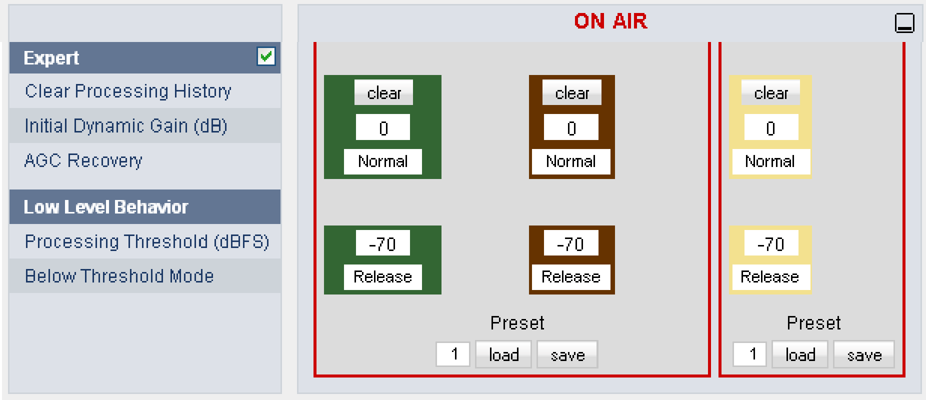
setup GUI – AUDIO PROCESSOR – **LevelMagic™**

Pls. keep in mind that the appearance of that pane depends on the respective loudness control mode (see Input). For description of the **LevelMagic™** parameters see engineering bulletin : "LevelMagic-2_Parameters_yymmdd.pdf", which is available for download from our web site.



Link	defines the coupling of the control circuits (see Input)
Leveler	[enable / disable] turns off Transient Processor as well.
Loudness Target	Level mode [0 … -50dBFS] ITU mode [0 … -50LKFS] EBU mode [0 … -50LUFS]
Time (s/min/h)	[10, 20, 40 / 1, 2, 5, 10, 20, 40 / 1, 2]
Max Gain (dB)	[0 … 40]
Freeze Level (dBFS)	[-20 … -60]
Transient Processor	
Max Gain (dB)	[0 … 15]
Response	[soft, mid, hard]
Limiter	[enable / disable]
Max True Peak (dBTP)	[0 … -20]
Profile (Leveler, Limiter)	
Processing	[live, speech, pop, uni, classic]

Expert [on / off]



The expert mode offers the possibility for manual intervention into the adaptive behavior of the **LevelMagic** process for critical material. For details pls. see the above mentioned document.

Clear Processing History manual or preset controlled

Initial Dynamic Gain (dB) [-40 … 15]
start value for the LevelMagic process after Clear Processing History

AGC Recovery [Normal / Fast]

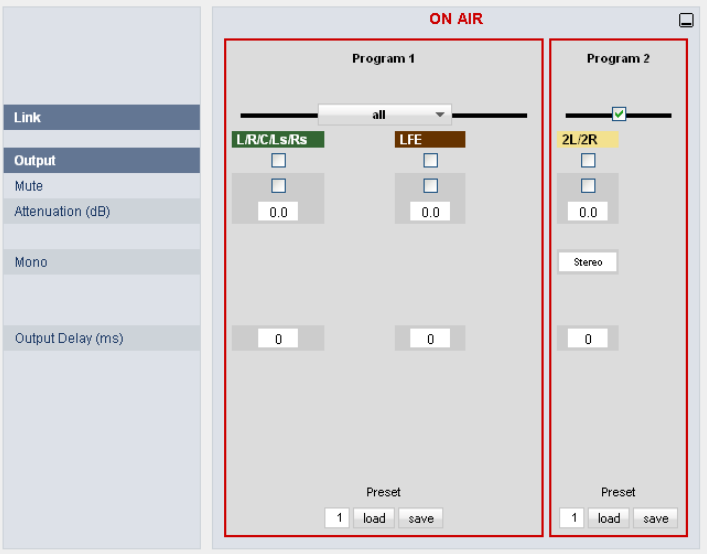
Low Level Behavior

Processing Threshold (dBFS)
[-80 … -20]
the threshold from where the processing gain will behave as defined by Below Threshold Mode.

Below Threshold Mode [release, hold]
returns slowly to 0dB gain change or stays at the Processing Threshold

setup GUI – AUDIO PROCESSOR – **Output**

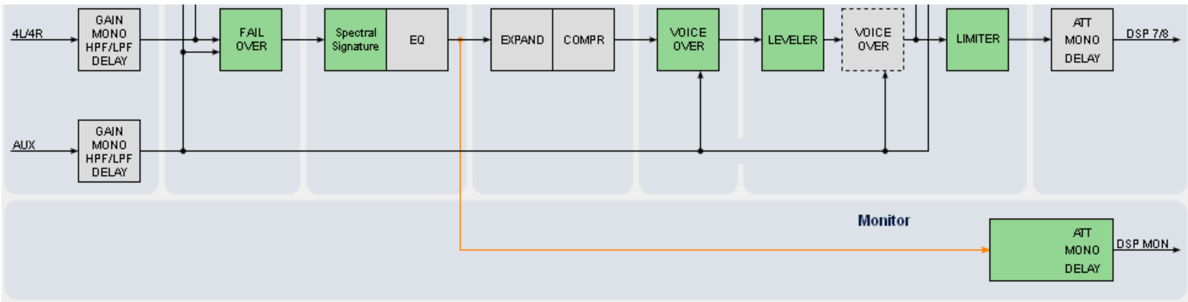
The **Output** block allows you to **Mute** and **Attenuate** the output signals from the DSP, do a mono conversion for stereo channels and add delay.

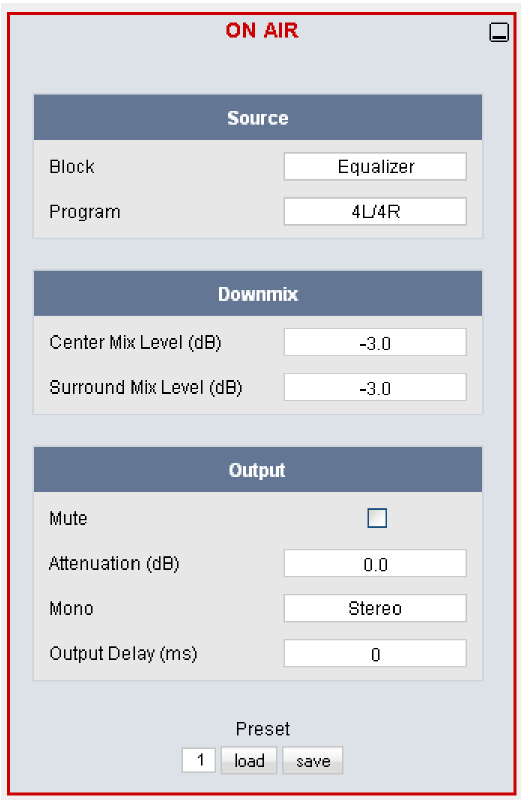


Link	defines the coupling of the control circuits (see Input)
Output	[enable / disable]
Mute	[on / off]
Attenuation (dB)	[-80.0 … 0.0]
Mono	[Stereo, L+R, LL, RR]
Output Delay (ms)	[0 … 2.000]

setup GUI – AUDIO PROCESSOR – **Monitor**

This Monitor is part of the audio processor and meant to monitor the individual processing blocks.
The monitor tap must be selected with the **Source Block** & **Source Program** parameters.
The actual tap will appear when you open the Overview pane.:





Source

Block	the processing block that should be monitored : [OFF, Input, Input Conditioner, Equalizer, Level Magic]
Program	[Surround, 2L/2R, Aux] for 5.1 +2 [1L71R … 4L74R] for 4x2 program associated audio channels processed by the specified block

Downmix if you are about to monitor a surround program. For 4 x 2 mode this parameter is not used.

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

Output

Mute	[Enable / Disable]
Attenuation (dB)	[-80.0 … 0]
Mono	[Stereo, L+R, LL, RR]
Output Delay (ms)	[0 … 2.000].

setup GUI – **EVENTS**

As mentioned in the initial paragraphs, you have a sophisticated **event management** system on hand, that allows you to initiate events manually (via the **X*AP** Remote Panel **Hotkeys)**, semi-automatically (by external commands or GPIs) and automatically (by changes to the internal status or parameters).

The **T*AP** knows three different event types, which will recall pre-defined presets :

Preset Events	System / Interfaces / Routing / Dolby Processing / Audio Processing
Action Events	GPOs / Loudness Measurement
Bypass Events	bypass of respective function blocks

A **Trigger** is the logical combination of up to two trigger sources. The **Trigger** will launch one or more events. An event runs like a flip-book inside the T*AP. This powerful technology spans from simply recalling a certain parameter to the complete reconfiguration of the **T*AP** from 4x2 to 5.1 +2 program configuration, including all signal routing, Dolby metadata handling, processing parameters and so forth. But it also enables several fail over scenarios where the T*AP will automatically react to the system and/or parameter status.

The way to set up the **EVENT** system is as follows :
1. Define trigger sources
2. Configure a trigger by logical combination of up to two trigger sources
3. Assign trigger to event(s)
4. Decide what shall happen (select presets for the events)

setup GUI – EVENTS – Trigger - **Trigger Configuration**

This is an excerpt from the Trigger Configuration pane :

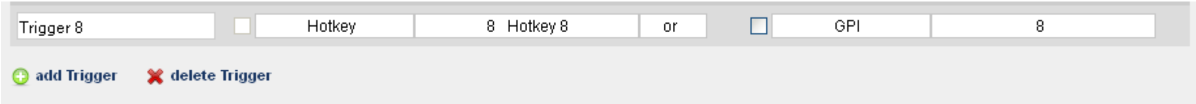
Trigger Configuration | Remote Hotkey Sources | Network Sources | Parameter Sources

Trigger #	Source 1 invert	Source 1 type	Source 1 source	logic	Source 2 invert	Source 2 type	Source 2 source
LM Gain	☐	GPI	1	or	☐	Network	1 Omnibus Ad start
SDI lost	☐	Parameter	1 SDI input fails	or	☐	-	
D-E fail over	☐	Parameter	2 Dolby E not present	or	☐	-	
Program Config	☐	Parameter	3 ARIB audio status	or	☐	-	
Trigger 5	☐	-		or	☐	-	

Trigger #	here you define a name for the trigger (Preset Load).
Source 1	the first source of a logical combination of two trigger sources.
invert	if the type of trigger allows an inverted operation it can be defined here.
type	the device knows different types of triggers [GPI, Hot Key, Network, Event active, Trigger active, Bypass, Sync Lock].
source	if the selected trigger type has multiple possible sources you must define it here [e.g. 1 … 8 for GPIs]. It acts like an index for the trigger source type.
logic	kind of logical operation [and, or, xor].
Source 2	second source of a logical combination of two trigger sources. If only one source exists, you may leave it unassigned [-].

Important Note! By accident you are able to set up a recursive behavior (same trigger is used for trigger setup or in an event that may activate this trigger). The plausibility is not checked so you may experience strange things if you are not careful.

At the bottom of the **Trigger** table we have two icons :



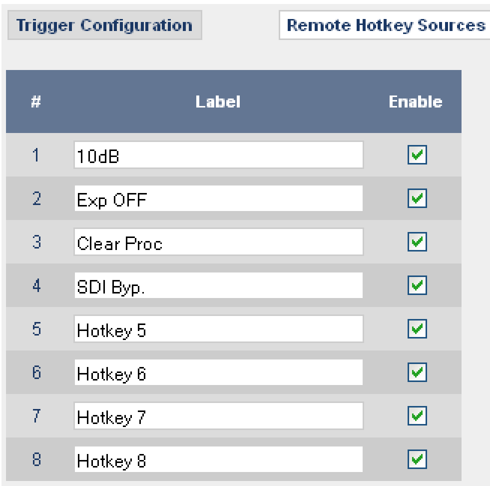
When you click on one of these icons you may add or delete a line of the above table.

When adding a trigger you may give it a name :

When removing a trigger you may select it by its name and press <OK>

setup GUI – EVENTS – Trigger – **Remote Hotkey Sources**

Hotkeys are the 8 buttons of an **X*AP** Remote Panel. You may give them names and enable them to show up as active on the **X*AP** Remote Panel :



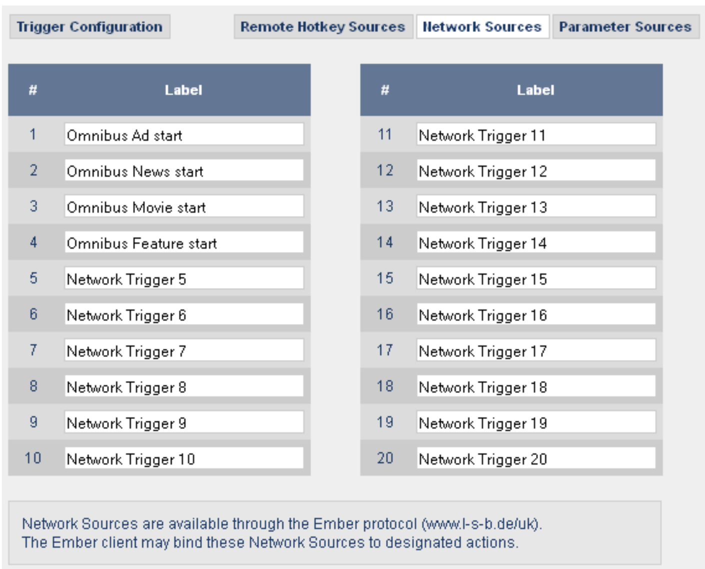
#	Label	Enable
1	10dB	☑
2	Exp OFF	☑
3	Clear Proc	☑
4	SDI Byp.	☑
5	Hotkey 5	☑
6	Hotkey 6	☑
7	Hotkey 7	☑
8	Hotkey 8	☑

the number of the Hotkey on the **X*AP** Remote Panel, counting from left to right.

Label each Hotkey may have a label that appears in the display of the **X*AP** Remote Panel above that button.

Enable [on / off]
if you turn it off the respective Hotkey on the **X*AP** Remote Panel becomes inactive - no label is displayed and the button background light turns off.

setup GUI – EVENTS – Triggers – **Network Sources**



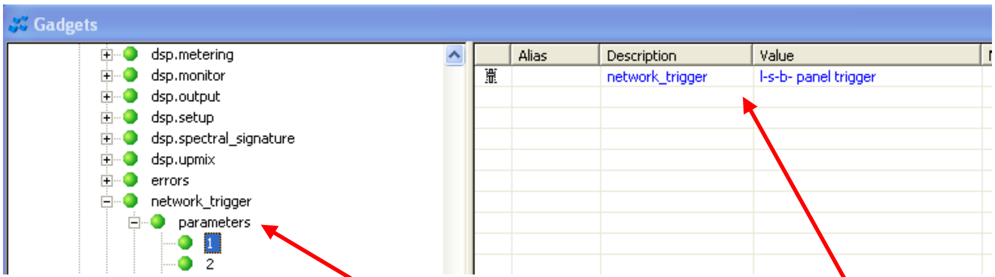
#	Label	#	Label
1	Omnibus Ad start	11	Network Trigger 11
2	Omnibus News start	12	Network Trigger 12
3	Omnibus Movie start	13	Network Trigger 13
4	Omnibus Feature start	14	Network Trigger 14
5	Network Trigger 5	15	Network Trigger 15
6	Network Trigger 6	16	Network Trigger 16
7	Network Trigger 7	17	Network Trigger 17
8	Network Trigger 8	18	Network Trigger 18
9	Network Trigger 9	19	Network Trigger 19
10	Network Trigger 10	20	Network Trigger 20

Network Sources are available through the Ember protocol (www.l-s-b.de/uk).
The Ember client may bind these Network Sources to designated actions.

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

#	number of the network trigger
Label	label of that network trigger. It appears on the **Trigger Configuration** pane as well as in the **EmBER+** tree of the VSM Studio gadget connector.

Below is a screen shot of the **VSM gadget connector** :



For the Ember tree you must select :
"Device" > controller_dsp > network_trigger > parameters > #.
As a value you will receive the trigger name from the **T*AP**.
In this example it is the default network trigger name.

setup GUI – EVENTS – Trigger – **Parameter Sources**

Below is an example of a few parameter trigger sources :

Trigger Configuration | Remote Hotkey Sources | Network Sources | Parameter Sources

Label #	Category	Subcategory	Parameter		Expression 1		Expression 2
SDI input fails	INTERFACES	SDI I/O Interface 1	SDI Lock	=	false	-	
Dolby E not present	DOLBY PROCESSING	Decoder	Status	=	Fail	-	
ARIB audio status	INTERFACES	SDI I/O Interface 1	ARIB B39 Audio Mode	=	3/2+LFE (5.1)	-	
Parameter Trigger 4	-	-	-	-		-	

Label	input field for a label of a parameter trigger source
Category	[INTERFACES / DOLBY PROCESSING / AUDIO PROCESSING]
Subcategory	e.g. If Category = DOLBY PROCESSING, possible Subcategories are : [Metadata Routing / Metadata Program / Decoder / Encoder] check all possible combinations with your T*AP
Parameter	e.g. if Subcategory = Metadata Routing, possible parameters are: [D SUB Metadata Input Status / Decoder Metadata Status / SDI1 – VANC Metadata Input Status SDI2 – VANC Metadata Input Status]
Expression 1	e.g. if Parameter = Status, possible expressions are: [NA / FAIL / CORRUPT / PCM / CONSUMER / OK]. The Expression allows **multiple values**. I.e. you may select PCM & CONSUMER. Since the drop down box is too small, both status expressions are marked green and the word **<multiple values>** will be used.
Expression 2	will be implemented soon

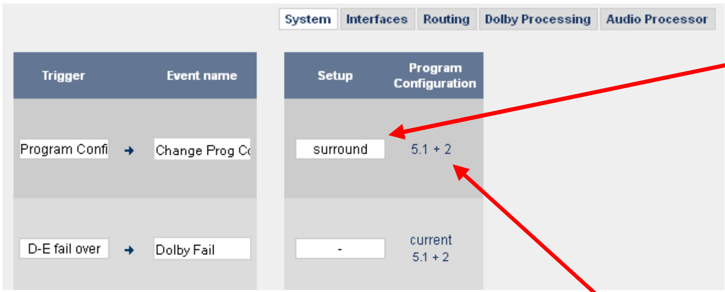
setup GUI – EVENTS – **Preset Events**

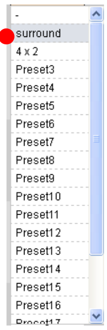
The change of configurations and / or parameters of the **T*AP** is based on presets in general.
Each setup pane offers a set of 20 presets for the respective function block. See on page 21 :
"setup GUI – SYSTEM - the **preset concept** in detail".

Important Note! The **Preset Events** tab controls multiple preset categories which are represented by the page embedded tabs. You must be aware that one trigger is valid for an entire line from System over Routing to Audio Processor. If you change the Trigger on one of the embedded pages it will be valid for all other pages.

setup GUI – EVENTS – Preset Events – **System**

On the **SYSTEM > Setup** pane you may change the program configurations manually or you may setup a respective presets (surround). The **Preset Event > System** will later on load this one.



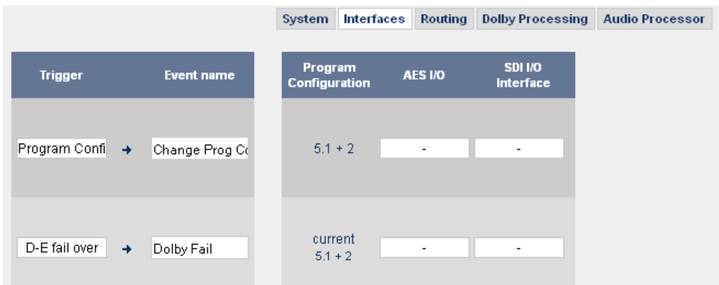


After selecting that preset the event manager reads the program configuration from this preset and prints it right beside the selection box.
The event manager now uses this program configuration for all other **Preset Events** sharing the same event name.

Trigger	when you click into the pull down box, you can select from one of the previously defined triggers (e.g. Program Config).
Event name	It is advisable to give this "System Preset Event" a name (e.g. Change Prog Config).
Setup	select one of the 20 presets from the SYSTEM > Setup menu (e.g. "surround").
Program Configuration	gives you an indication which mode the **T*AP** will be in when you load that preset.

setup GUI – EVENTS – Preset Events – **Interfaces**

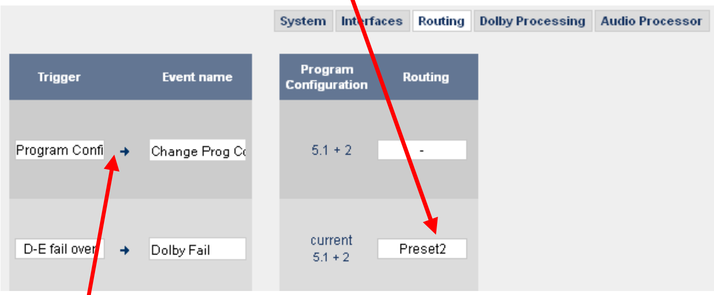
The **T*AP Base Unit** has four AES I/Os and for example an optional SDI I/O interface is installed. Therefore this pane offers two pull down boxes to select one of those preset for the respective interface per event.



The "Change Prog Conf" as well as the "Dolby Fail" events do not change things for an interface.
I.e. the selection is empty "**-**".

setup GUI – EVENTS – Preset Events – **Routing**

The **ROUTING** system has no sub panes and therefore only one set of 20 presets. They may be allocated to preset events. For this example it is **Preset2**:



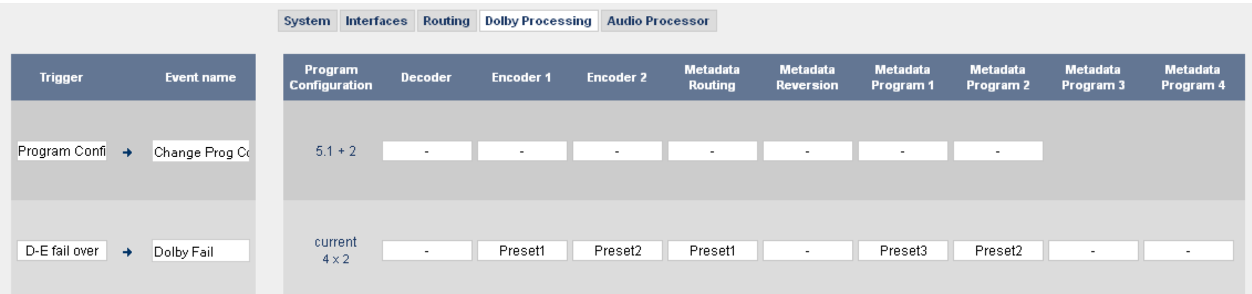
System | Interfaces | Routing | Dolby Processing | Audio Processor

Trigger		Event name	Program Configuration	Routing
Program Confi	→	Change Prog Co	5.1 + 2	-
D-E fail over	→	Dolby Fail	current 5.1 + 2	Preset2

The "Change Prog Conf" Event is not used to change routing.

setup GUI – EVENTS – Preset Events – **Dolby Processing**

The Dolby processing system has several setup sections for the hardware parts (decoder, encoders) as well as for the metadata routing and the program specific sets of metadata. That is why the **Preset Events > Dolby Processing** pane has **nine** columns:



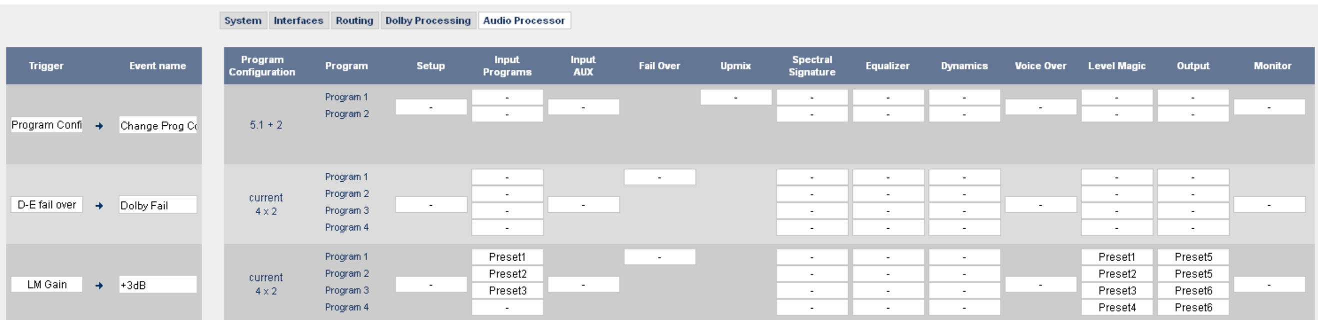
System | Interfaces | Routing | Dolby Processing | Audio Processor

Trigger		Event name	Program Configuration	Decoder	Encoder 1	Encoder 2	Metadata Routing	Metadata Reversion	Metadata Program 1	Metadata Program 2	Metadata Program 3	Metadata Program 4
Program Confi	→	Change Prog Co	5.1 + 2	-	-	-	-	-	-	-		
D-E fail over	→	Dolby Fail	current 4 x 2	-	Preset1	Preset2	Preset1	-	Preset3	Preset2	-	-

The "Dolby Fail" event will load several presets (see above)

setup GUI – EVENTS – Preset Events – **Audio Processing**

The most comprehensive set of presets is offered for the **Audio Processer**, which has **eight** different processing sections, a setup page, input / output and monitor pages, which you may (or may not) alter by triggering the event named **"+3dB"** via the **"LM Gain"** trigger :

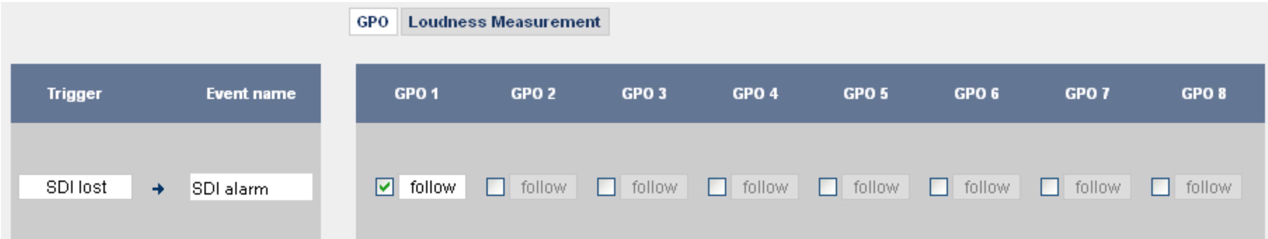


System | Interfaces | Routing | Dolby Processing | Audio Processor

Trigger		Event name	Program Configuration	Program	Setup	Input Programs	Input AUX	Fail Over	Upmix	Spectral Signature	Equalizer	Dynamics	Voice Over	Level Magic	Output	Monitor
Program Confi	→	Change Prog Co	5.1 + 2	Program 1	-	-	-		-	-	-	-	-	-	-	-
				Program 2		-				-	-	-		-	-	
D-E fail over	→	Dolby Fail	current 4 x 2	Program 1	-	-	-	-		-	-	-	-	-	-	-
				Program 2		-				-	-	-		-	-	
				Program 3		-				-	-	-		-	-	
				Program 4		-				-	-	-		-	-	
LM Gain	→	+3dB	current 4 x 2	Program 1	-	Preset1	-	-		-	-	-	-	Preset1	Preset5	-
				Program 2		Preset2				-	-	-		Preset2	Preset5	
				Program 3		Preset3				-	-	-		Preset3	Preset6	
				Program 4		-				-	-	-		Preset4	Preset6	

For the other two events "Change Prog Conf" and "Dolby Fail" there is no need to change anything in the Audio Processor for this example so we do not select presets there setup GUI – EVENTS – Action Events – **GPO**

Action Events are independent from **Preset Events**. That is why you must define a new event name (e.g. SDI alarm). This event should also be triggered by the "**SDI lost"** trigger.
The **T*AP** has 8 physical GPO's (relay change over contacts) which may be incorporated into an action event. The example below will only activate **GPO 1** if the **"SDI alarm"** event is triggered by the "SDI lost" trigger.

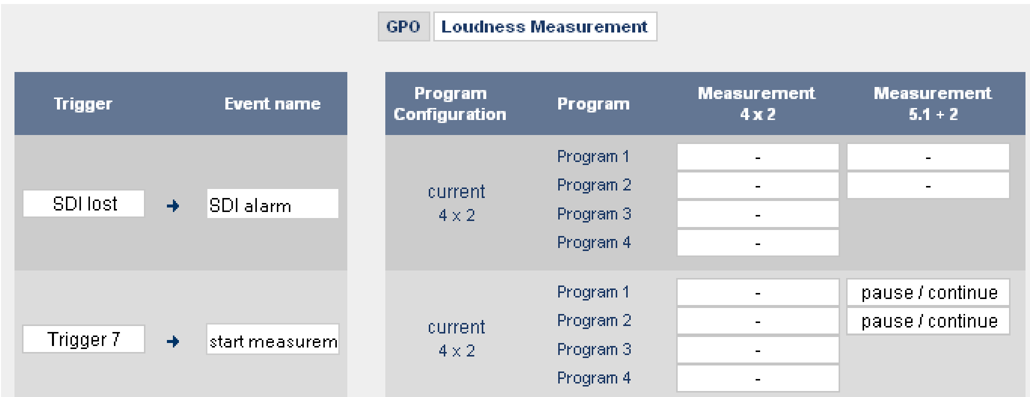


The options to switch the respective GPO are:

clear	will deactivate the previously activated GPO
set	will activate the GPO
follow	the GPO state will follow the trigger : turns on, if the trigger is activated, turns off, if the trigger is deactivated
toggle	turns on, on the rising edge of the trigger, turns off on the next rising edge. Toggle functions are always tricky because you must guarantee a known starting condition.

setup GUI – EVENTS – Action Events – **Loudness Measurement**

The **EBU R128** implements the possibility to start, pause, continue, reset a loudness measurement.

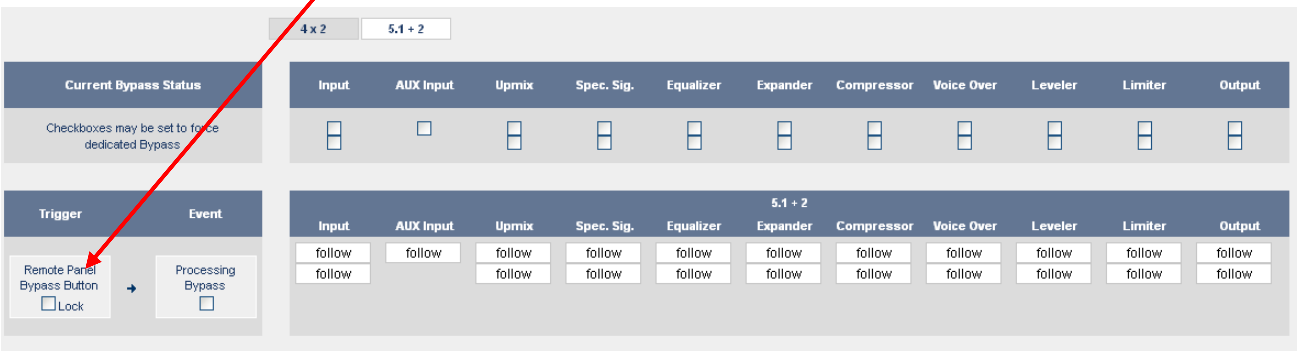


The example above defines one **Action Event > Loudness Measurement** where "pause / continue" will be activated by **"Trigger 7"** that starts the Action Event named **"start measurement"**.

The above pane must be set up for both the 4 x 2 and 5.1 + 2 program configuration. The event manager will take the respective actions depending on the actual program configuration.

setup GUI – EVENTS – **Bypass Events**

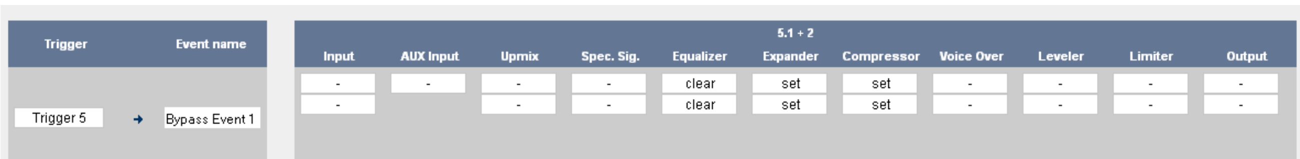
The **T*AP** has a dedicated **<BYPASS>** button on the **X*AP** Remote Panel. The function of this button may be configured in the upper section of the **Bypass Events** pane.
You may lock the button and ou may also control it with the **Processing Bypass** check box :



The top two rows of check boxes represent the bypass switches of the individual function blocks of the DSP. They may be used to force the bypass function of individual blocks manually. The number of lines (4 [4x2] or 2 [5.1+2] in this example) depends on the program configuration.

If you turn the **<BYPASS>** button of the **X*AP** Remote Panel **ON** the Processing Bypass check box will show it. But you may also use the check box to turn the button **ON / OFF**.

In the lower rows you may configure the bypass function of the individual function blocks to be controlled by an **Bypass Events** trigger :



The Event named "Bypass Event1" may be triggered by **"Trigger 5"**. It will turn the bypass **ON** for the function blocks: Expander, Compressor, and **OFF** for the Equalizer section.

Example EVENTS configuration

Finally an example for a field application. This shall demonstrate the steps which are needed to setup the **EVENTS** system.

In Japan it is common practice to change the configuration of processing devices depending on additional meta information. This meta information is standardized by the ARIB standard.
We will now demonstrate how to change the program configuration of the T*AP, controlled by such a meta information received via SDI embedded ancillary data packets.

On the **EVENTS > Trigger > Parameter Sources** pane we **define** a parameter **source** :

Label #	Category	Subcategory	Parameter	Expression 1		Expression 2	
SDI input fails	INTERFACES	SDI I/O Interface 1	SDI Lock	=	false	-	
Dolby E not present	DOLBY PROCESSING	Decoder	Status	=	Fail	-	
ARIB audio status	INTERFACES	SDI I/O Interface 1	ARIB B39 Audio Mode	=	3/2+LFE (5.1)	-	

To do so we have to look in the Category **Interfaces** for the Subcategory **SDI I/O Interface1** and there for the parameter **ARIB B39 Audio Mode**. The parameter expression **"3/2+LFE (5.1)"** will be the trigger source. We give it the label : "**ARIB audio status".**

The next step is to **configure** the **trigger** :

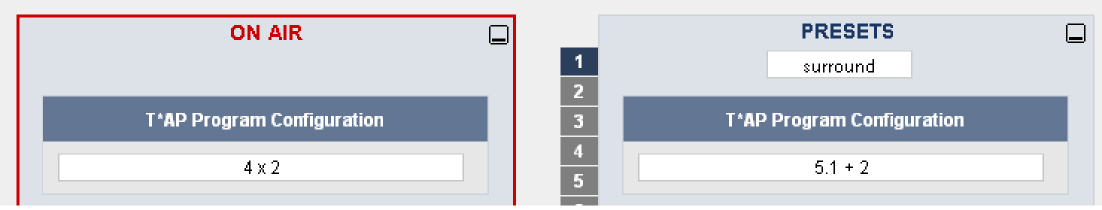
Trigger #	Source 1				Source 2		
	invert	type	source	logic	invert	type	source
LM Gain	☐	GPI	1	or	☐	Network	1 Omnibus Ad start
SDI lost	☐	Parameter	1 SDI input fails	or	☐	-	
D-E fail over	☐	Parameter	2 Dolby E not present	or	☐	-	
Program Config	☐	Parameter	3 ARIB audio status	or	☐	-	

We call the Trigger : **"Program Config."**. it is of the type **Parameter** and the source label (defined in step one) is **"ARIB audio status"**.

Now we must **assign trigger to events**. The trigger **"Program Config."** will launch the Preset Event **"Change Prog Conf"** :

Trigger		Event name	Setup	Program Configuration
Program Confi	→	Cange Prog Cor	surround	5.1 + 2

Finally we **decide** what shall happen if the example event "Change Prog Conf" becomes active. For our example it will load the preset called **"surround".** When you have a look at the **SYSTEM > Setup** pane you will see that the preset "surround" will reconfigured the T*AP to 5.1 + 2 program configuration :



In order not to get too complicated we will stop here. But you must keep in mind that the trigger **"Program Config"** may also be used to change ROUTING paths and AUDIO PROCESSOR settings as well as other things if it is appropriate.

technical data – **Base Unit**

- **Power supply** dual power supply, auto fail over
AC 85 V – 264 V, 50 Hz … 60 Hz
58W max
- **AES input** AES3id
24 Bit, 48 kHz, 0,32 … 1,2 Vpp
sample rate converters:
24 Bit, 32 kHz … 192 kHz, THD+N: < -130 dB @ 0 dBFS
- **AES output** AES3id
24 Bit, 48 kHz, nominal 1 Vpp @ 75 Ohm
power fail relay bypass
- **Sync internal** 48 kHz, +/- 10 ppm
- **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
RJ45 front panel connector
10/100MBit Ethernet auto sense, full duplex, auto MDI/X
Power Over Ethernet IEEE 802.3af
- **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
Base Unit - 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 – **X*AP** Remote Panel

- **Power supply** POE (Power Over Ethernet), IEEE 802.3af
- **Consumption** 8 W
- **Max cable length** if connected with the **Base Unit**, 30m distance CAT.5E (26AWGx4P)
- **Dimensions** 19", 1RU, depth 6 cm
- **Environmental** operating temperature 0 ºC to 50 ºC
non operating temperature -20 ºC to 70 ºC
humidity 90%, non condensing

technical data – interface boards – **SDI De-Embedder / Embedder [SDI 150]**

- **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 [DD 188]**

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 [AN 144]**

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 – **8x analog I/O [AN 108]**

connector
  25pin Sub-D female
output
  impedance: 5 Ω, electronically balanced
  max. output level @ 0 dBFS: 0.0 dBu … +24 dBu adjustable in 0.5 dB steps
  dynamic range: 110 dB
  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

technical data - **Base Unit** rear connectors - **pin assignment**

connector :	**GPI**
female	25-pin Sub-D
1	GPI_1a
2	
3	GPI_2a
4	GPI_3a
5	GPI_3b
6	GPI_4a
7	
8	GPI_5b
9	GPI_6a
10	GPI_6b
11	GPI_7a
12	GPI_7b
13	GPI_8b
14	GPI_1b
15	GPI_2b
16	
17	
18	
19	GPI_4b
20	GPI_5a
21	
22	
23	Isolated 5V **+**
24	Isolated 5V **-**
25	GPI_8a

connector :	**GPO**
female	25-pin Sub-D
1	GPO_1_NC
2	GPO_1_NO
3	GPO_2_common
4	GPO_3_NC
5	GPO_3_NO
6	GPO_4_common
7	GPO_5_NC
8	GPO_5_NO
9	GPO_6_common
10	GPO_7_NC
11	GPO_7_NO
12	GPO_8_common
13	
14	GPO_1_common
15	GPO_2_NC
16	GPO_2_NO
17	GPO_3_common
18	GPO_4_NC
19	GPO_4_NO
20	GPO_5_common
21	GPO_6_NC
22	GPO_6_NO
23	GPO_7_common
24	GPO_8_NC
25	GPO_8_NO

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

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

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

4x analog I/O [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 :	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

T*AP

jünger

the reference in loudness management

Jünger Audio GmbH
Justus-von-Liebig-Straße 7
12489 Berlin
Germany

phone: +49 30 6777 21 0
fax: +49 30 6777 21 46
info@jungeraudio.com
www.jungeraudio.com