Synapse

GJA420/440/840/880
HJA420/440/840/880

3Gb/s, HD, SD embedded domain Loudness controller
based on Jünger Audio algorithms

Installation and Operation manual

Powered by LINUX

Quad speed MASTER

Embedded Metadata S2020

3 TRIPLE RATE Gb/s, HD, SD

Powered by jünger

Upgradable to 3Gb/s

® AXON
THE HEART OF BROADCAST

Synapse

TECHNICAL MANUAL

GJA420/440/840/880
HJA420/440/840/880

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WARNING: TO REDUCE THE RISK OF FIRE OR ELECTRICAL SHOCK, DO NOT EXPOSE THIS APPLIANCE TO RAIN OR MOISTURE

● ALWAYS disconnect your entire system from the AC mains before cleaning any component. The product frame (SFR18 or SFR04) must be terminated with three-conductor AC mains power cord that includes an earth ground connection. To prevent shock hazard, all three connections must always be used.
● NEVER use flammable or combustible chemicals for cleaning components.
● NEVER operate this product if any cover is removed.
● NEVER wet the inside of this product with any liquid.
● NEVER pour or spill liquids directly onto this unit.
● NEVER block airflow through ventilation slots.
● NEVER bypass any fuse.
● NEVER replace any fuse with a value or type other than those specified.
● NEVER attempt to repair this product. If a problem occurs, contact your local Axon distributor.
● NEVER expose this product to extremely high or low temperatures.
● NEVER operate this product in an explosive atmosphere.

Warranty: Axon warrants their products according to the warranty policy as described in the general terms. That means that Axon Digital Design BV can only warrant the products as long as the serial numbers are not removed.

Date created: 09-01-2014
Date last revised: 06-11-2015

This product complies with the requirements of the product family standards for audio, video, audio-visual entertainment lighting control apparatus for professional use as mentioned below.

EN60950	Safety
EN55103-1: 1996	Emission
EN55103-2: 1996	Immunity

Axon Digital Design GJA420/440/840/880 HJA420/440/840/880 Tested To Comply With FCC Standards FOR HOME OR OFFICE USE	This device complies with part 15 of the FCC Rules Operation is subject to the following two conditions: (1) This device may cause harmful interference, and (2) This device must accept any interference received, including interference that may cause undesired operation.

Table of Contents

1 Introduction to Synapse

An Introduction to Synapse

Synapse is a modular system designed for the broadcast industry. High density, intuitive operation and high quality processing are key features of this system. Synapse offers a full range of converters and processing modules. Please visit the AXON Digital Design Website at www.axon.tv to obtain the latest information on our new products and updates.

Local Control Panel

The local control panel gives access to all adjustable parameters and provides status information for any of the cards in the Synapse frame, including the Synapse rack controller. The local control panel is also used to back-up and restore card settings. Please refer to the rack controler manuals for a detailed description of the local control panel, the way to set-up remote control over IP and for frame related settings and status information.

Remote Control Capabilities

The remote control options are explained in the rack controller manual. The method of connection to a computer using Ethernet is also described in the ERC/ERS/RRC/RRS manual.

! **CHECK-OUT:** "AXON CORTEX" SOFTWARE WILL INCREASE SYSTEM FLEXIBILITY OF ONE OR MORE SYNAPSE FRAMES

Although not required to use Cortex with a Synapse frame, you are strongly advised to use a remote personal computer or laptop PC with Axon Cortex installed, as this increases the ease of use and understanding of the modules.

2 Unpacking and Placement

Unpacking

The Axon Synapse card must be unpacked in an anti-static environment. Care must be taken NOT to touch components on the card – always handle the card carefully by the edges. The card must be stored and shipped in anti-static packaging. Ensuring that these precautions are followed will prevent premature failure from components mounted on the board.

Placing the card

The Synapse card can be placed vertically in an SFR18 frame or horizontally in an SFR04 and SFR08 frame. Locate the two guide slots to be used, slide in the mounted circuit board, and push it firmly to locate the connectors.

Correct insertion of card is essential as a card that is not located properly may show valid indicators, but does not function correctly.

NOTE: On power up all LED's will light for a few seconds, this is the time it takes to initialise the card.

3 A Quick Start

When Powering-up

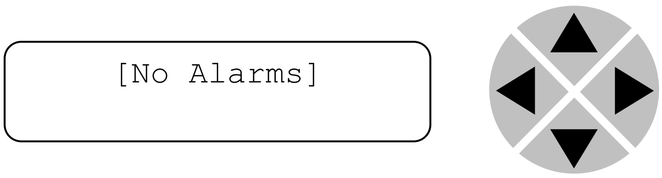
On powering up the Synapse frame, the card set will use basic data and default initialisation settings. All LED's will light during this process. After initialisation, several LED's will remain lit – the exact number and configuration is dependant upon the number of inputs connected and the status of the inputs.

Changing settings and parameters

The front panel controls or the Axon Cortex can be used to change settings. An overview of the settings can be found in chapter 5, 6 and 7 of this manual.

Front Panel Control

Front Panel Display and Cursor



Settings are displayed and changed as follows;

Use the cursor 'arrows' on the front panel to select the menu and parameter to be displayed and/or changed.

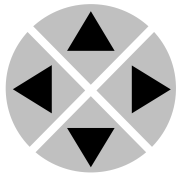
Press ►	To go forward through the menu structure.
Press ◄	To go back through the menu structure.
Press ▲	To move up within a menu or increase the value of a parameter.
Press ▼	To move down through a menu or decrease the value of a parameter.

NOTE: Whilst editing a setting, pressing ► twice will reset the value to its default.

Example of changing parameters using front panel control

With the display as shown below

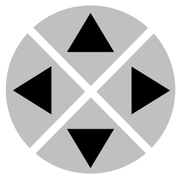
```
RRC18 [Select Card]
>S01=SFS10
```



Pressing the ► selects the SFS10 in frame slot 01.

The display changes to indicate that the SFS10 has been selected. In this example the `Settings` menu item is indicated.

```
SFS10 [Select Menu]
>Settings
```

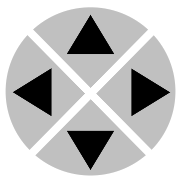


Pressing the ► selects the menu item shown, in this example `Settings`.

(Pressing ▲ or ▼ will change to a different menu eg `Status`, `Events`).

The display changes to indicate that the SFS10 `Settings` menu item `SDI-Format` has been selected and shows that it current setting is `Auto`.

```
SFS10 [Settings]
>SDI-Format=Auto
```

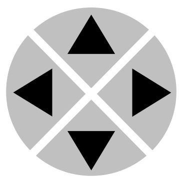


Pressing the ► selects the settings item shown, in this example `SDI-Format`.

(Pressing ▲ or ▼ will change to a different setting, eg `Mode`, `H-Delay`).

The display changes to indicate that the SFS10 `Edit Setting` menu item `SDI-Format` has been selected.

```
SFS10 Edit Setting]
SDI-Format>Auto
```



To edit the setting of the menu item press ▲ or ▼.

All menu items can be monitored and/or changed in this way. Changing a setting has an immediate effect.

Axon Cortex Software

Axon Cortex Software can be used to change the settings of Synapse modules from a PC, either locally or remotely. The software enables communication based on TCP/IP between the setup PC and Synapse frames/modules.

Each Synapse frame is addressed through its rack controller's unique IP address, giving access to each module, its menus and adjustment items. Axon Cortex has access to data contained within the Synapse module and displays it on a GUI. The software has an intuitive structure following that of the module that it is controlling.

For downloading Axon Cortex, please refer to our website: www.axon.tv. For instruction about how to use Axon Cortex, please check the Axon Cortex help files for details (press F1 in any window)

Menu Structure Example

Slot	Module		Item		Parameter		Setting
▲							
▲							
S02			Identity				
▲			▲				
`S01`	`SFS10`	▶	`Set-tings`	▶	`Standard_dig`	▶	`Auto`
▼			▼		▼		▼
S00	RRC18		Status		Mode		625
			▼		▼		▼
			Events		Ref-Input		525
					▼		
					H-Delay		
					▼		
					▼		

NOTE: Further information about Front Panel Control and Axon Cortex can be obtained from the RRC and RRS operational manuals and the Cortex help files.

4 The GJA-HJA Card

Introduction

The GJAxx0 and HJAxx0 are embedded domain dual audio stream hardware processors, designed for broadcasters who need automatic loudness control and optional upmixing.

Based on the popular and well respected LEVEL MAGIC II™ processing these cards can perform a high quality loudness adjustment completely conform the CALM and R128 standards

Users can adjust all the Jünger based settings of the processing and embedded handling directly from the G/JAxx0 GUI in Cortex, with control offered over a variety of different parameters. Output level controls and delay adjustment are also offered for each of the channels in the final 5.1 mix.

The Quad Speed audio bus allows for implementation of additional audio processing. This means that an additional processing card like for instance a DDP24 or DBD28 can be added to perform Dolby processing, without any additional wiring. The ADD-ON card often does not need a connector panel and all audio routing is performed inside the Synapse frame by just placing these cards in adjacent slots.

Features

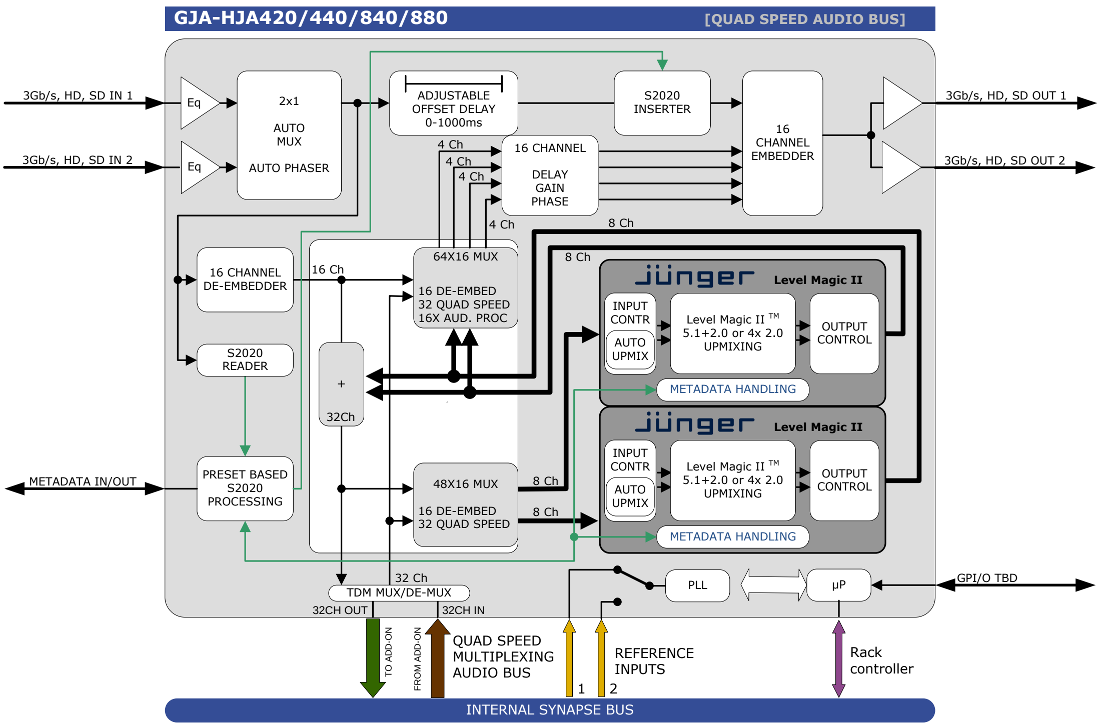
- GJA/HJA420 = 4x 2.0 loudness control for SD, HD and 3Gb/s (G only) embedded I/O
- GJA/HJA440 = 5.1 + 2.0 loudness control and auto upmix for SD, HD and 3Gb/s (G only) embedded I/O
- GJA/HJA840 = 8x 2.0 loudness control for SD, HD and 3Gb/s (G only) embedded I/O
- GJA/HJA880 = 2x 5.1 + 2.0 loudness control and auto upmix for SD, HD and 3Gb/s (G only) embedded I/O
- LEVEL MAGIC II™ loudness management according to: EBU R128, ITU.1770
  (all versions), ATSC A/85 and ARIB TR-B32
- Dynamics with compressor and expander
- Surround up mix functionality
- DOLBY® metadata generator
- Loudness logging via Cortex
- Output gain and delay adjustments
- Cross fading between upmixed and discrete 5.1 (5.1/2.0 input auto-sensing)
- 16 channels of audio gain
- 16 channel audio delay up to 5000ms just prior to the embedding stage
- 2 SDI inputs (with auto switch on carrier loss, and switch back function)

- Compatible with the following input formats (auto selecting) (1080p only for GAWxxx):
  - 1080p/59.94
  - 1080p/50
  - 1080i/59.94
  - 1080i/50
  - 1080p/29.97
  - 1080p25
  - 1080psf/23.98
  - 720p/59.94
  - 720p50
  - SD525
  - SD625
- Video offset delay between 0 and 1000ms
- Quad Speed Audio ADD-ON bus for bidirectional audio processing
- 2 SDI + embedded audio outputs
- 7 presets that configure all 16 input channels at once, controlled by ACP (Cortex)
- Append and overwrite modes
- Silence detection and peak detection (0dBFS)
- Transparent for ATC time code RP188, RP196, RP215
- Locks to Tri-level, Bi-level syncs or input
- Full control and status monitoring through the front panel of the SFR04/SFR08/SFR18 frame and the Ethernet port (ACP)
- Optional 1 or 2 fiber inputs, 1 or 2 fiber outputs or a fiber in and output (replacing 1 SDI in and output) on the I/O panel
- Optional relay bypass (BHX18D)

Applications

- Optional 1 or 2 fiber inputs, 1 or 2 fiber outputs or a fiber in and output (replacing 1 SDI in and output) on the I/O panel
- Optional relay bypass (BHX18D

Block schematic



5 Settings Menu

Introduction

The settings menu displays the current state of each GJA-HJA setting and allows you to change or adjust it. Settings can be changed using the front panel of the Synapse frame (SFR18, SFR08 or SFR04) or with Cortex. Also the SCP08 control can be used. Please refer to chapter 3 for information on the Synapse front panel control and Cortex.

Note: All items preceded with a #-sign are part of the presets.

VIDEO

Inp_Select

With this item you can decide which of the 2 inputs is used and how the card will switch between the 2 inputs. Choices are:

- `Auto`: The card chooses input 1 if there is a source. If there is no input 1, the card will automatically switch to input 2.
- `SDI-1`: only input 1 is used (disables detection of input 2)
- `SDI-2`: only input 2 is used (disables detection of input 1)

Switch-Back

With `Inp_Select` set to Auto, the card will automatically switch to the other input when the first input was lost. With `Switch-Back` set to `On`, the card will switch back to the first input if this it is back up again. Set to `Off` the card will keep using the other input even if the first input is back up again.

Lock-Mode

`Lock-Mode` determines whether the card is locked to input 1 (`SDI1`), input 2 (`SDI2`) or to the reference (`Ref1` or `Ref2`).

Can also be set to `Auto-SDI`, in which case SDI1 has priority. When SDI1 is not present, locking will switch to SDI2. When the signal is back again at SDI1, lock will switch back to SDI1 (The locked-to status item, in the status menu, shows which SDI channel the card is locked to).

By default this setting is is set to `SDI1`.

Out-Frmt

With `Out-Frmt` you can fix the output should be. Possible settings are:

- `Atuo` (default),
- `1080i60, 1080i50`
- `1080p30`, `1080p25`, `1080p24`
- `720p60`, `720p50`
- `SD525`, `SD625`
- `1080p50`, `1080p60` (GJA only)

Phaser1-Offset

Sets the offset of the auto phaser of input 1 (see block schematic) between 0 and 4124px. Default is 0px.

Phaser2-Offset

Sets the offset of the auto phaser of input 2 (see block schematic) between 0 and 4124px. Default is 0px.

Phaser-status

It is possible to display the function of the autophasers in the status menu of the card. This setting enables or disables the status-items: `Phaser1_H_Pos`, `Phaser2_H_Pos`, `Phaser1_Stat` and `Phaser2_Stat`. Default setting is `Off`.

Delay-Bypass

You can bypass the delay block entirely by setting this to `on`. By default it is switched `off`.

Delay-mode_1

With this setting you decide whether the card should apply delay by means of `time` in milliseconds (defined with `Time-Delay_1`) or to apply delay by means of frames, lines and pixels (`Fr-Ln-Px`). Default is `Fr-Ln-Px`.

Time-Delay_1

This setting is only used when `Delay-mode_1` is set to `Time`. It defines the delay that should be applied to the video in milliseconds between 0 and 10000ms.

F-delay_1

`F-Delay_1` sets the amount of delayed Frames. The available range is from 0 to 250 frames (dependent on the input format). When Out-Frmt is SD, the maximum is 250 frames, when it is 720p50/60 the maximum is 120 frames. All other HD formats can be delayed a maximum of 60 frames.

V-delay_1

`V-Delay_1` setting allows adjustment of the vertical phase of the output signal with respect to the selected reference input.

The `V-Delay_1` setting gives a delay in addition to the reference timing. For example: if the `V-Delay_1` is set to 10 TV HD lines, the output signal will be delayed by reference timing + 10 TV HD lines. The signal is delayed (advanced) with respect to the phase of the reference signal. The available range is from 0 to a maximum of 1124 lines (dependent on I/O format). The default setting is `0ln`.

H-delay_1

The H-Delay_1 setting allows adjustment of the Horizontal phase of the output signal with respect to the selected reference input.

The H-Delay_1 setting gives a delay in addition to the reference timing. For example: if the H-Delay_1 is set to 10 pixels, the output signal will be delayed by reference timing + 10 pixels. The signal is delayed (advanced) with respect to the phase of the reference signal. The available range is from 0 to a maximum of 4124 pixels (dependant on I/O format). The default setting is 0px.

Delay-Status

It is possible to display (in the status menu IODelay2) the processing time of the card in the status menu. This setting allows you to switch this function ON or OFF. Default setting is OFF

PRESET

GPI-Ctrl

The GJA-HJA has several physical GPI contacts to control the card's presets (if presets are set to be GPI controlled)

Latch: Latching GPI mode. When a contact is closed momentarily (edge triggered).

Non-Latch: Non-latching GPI mode. When a contact is closed all the time (level triggered). Refer to the following table for examples of possible preset triggers:

Ext-Mode

With this item you set the purpose of pins 5 till 8 of the RJ45 connector on the backpanel. The pupose can be either additional GPIO contacts (resulting in 7 GPI contacts instead of 3) or to use those pins for a dolby metadata I/O. Default is GPIO.

Active-Preset

With this item you can manually change the currently active preset . Can be any preset between 1 and 7. By default it is set to 1. All menu settings that are preceded with a '#'-prefix are part of the preset.

Edit-Preset

Here you can select which of the 7 selectable presets you want to edit. Changing this will not change the active preset, unless the currently active preset is the same you are going to edit. All menu settings that are preceded with a '#'-prefix are part of the preset.

PrstEditView

With this setting set to Follow Active, the edit preset settings will follow the active preset when the active preset is changed. This to avoid confusion when changing the active. Set to Independent the edit preset will not automatically follow active preset changes. By default set to Follow Active.

#Preset_Name

Sets/displays the name of the currently displayed preset.

#Source_lmJ1Ch01~ #Source_lmJ1Ch08

With these settings you can select where the input audio channels for the Junger loudness engine J1 are coming from:

- `SDI1`: Audio comes from SDI input 1 (embedded audio)
- `AddOn01/16`: Audio comes from addon channels 1 to 16
- `AddOn17/32`: Audio comes from addon channels 17 to 32

#lmJ1_Ch01 ~ #lmJ1_Ch08

With this setting you decide which audio channel of the above selected source is used for the loudness engine J1, respectively channel 1 till 8. Can be any of the available 16 channels or set to `off`.

#Source_lmJ2Ch01~ #Source_lmJ2Ch08

With these settings you can select where the input audio channels for the Junger loudness engine J2 are coming from:

- `SDI1`: Audio comes from SDI input 1 (embedded audio)
- `AddOn01/16`: Audio comes from addon channels 1 to 16
- `AddOn17/32`: Audio comes from addon channels 17 to 32

#lmJ2_Ch01 ~ #lmJ2_Ch08

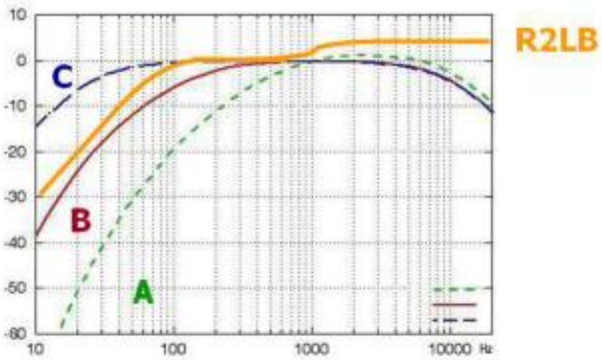
With this setting you decide which audio channel of the above selected source is used for the loudness engine J2, respectively channel 1 till 8. Can be any of the available 16 channels or set to `off`.

#lmJ1_control

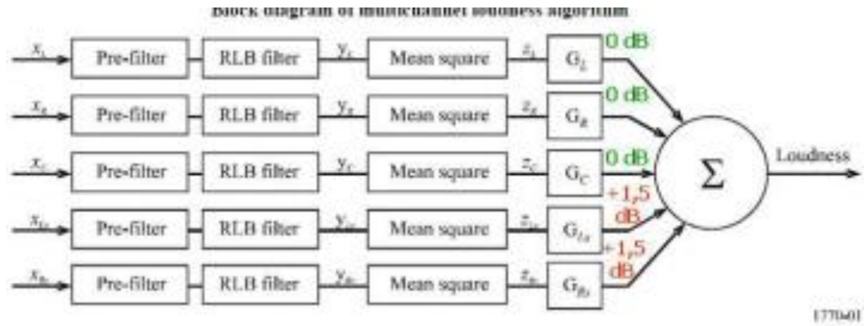
Level Magic Process Control for loudness engine 1: The Junger Audio proprietary, level based process. The aim is to maintain a desired operating level. Curves and algorithms are the intellectual property of Junger Audio and will not be disclosed.

- `ITU-BS.1770-1 (A/85:2011)`

Loudness based measurement. Several filters and RMS weighting are used to get a loudness equivalent result. Starting from the well known A, B, C weighting curves (DIN-IEC 651), the ITU did further research into the relationship of frequencies, their overall levels, their peak levels and the duration of signals to develop the best representation of human loudness perception. The result was the RLB curve [Revised Low frequency B] The combination of the RLB-filter and the Pre-filter is called R2LB [Secondly Revised Low frequency B curve. AKA K-Weighting curve :



E.g. K-Weighting filters are used as in the example below for the measurement of loudness of a surround signals (LFE must not be included).



- ITU-BS.1770-2
- ITU-BS.1770-3

The frequency weighted measurement has been extended by a gating function. The EBU PLOUD-Group developed a gating function to exclude quiet sections from the measurement to prevent possible loudness underread. Examples are background noise and atmo which do not add to the loudness perception as much as louder signals do. This gating function consists of an absolute threshold at -70LUFS and a relative threshold 10LU below the absolute-gated level. This involves the necessity to recalculate the whole measurement with every subsequent block. The ITU approved the gating function and included it into its recommendations ITU-R BS. 1770-2 and later.

Important note! Systems working in ITU 1770-1 mode do not feature a gating function. Thus its output readings may vary a bit from meters compliant to EBU R128 or ITU 1770-2/-3. Further keep in mind that the gate is only applicable to the integrated or program loudness measurement (from start to stop) andnot used for short-term or momentary measurements.

- EBU R 128

This is a work to rule, based on ITU-BS.1770-2. To characterise an audio signal the measures of Programme Loudness, Loudness Range and Maximum True Peak Level are used.

The Program Loudness Level is normalized to -23LUFS[Loudness Units referenced to Full Scale] with a permitted deviation of +/- 1LU. The measurement includes a gating method as specified in ITU-BS.1770 (summarized in EBU Tech Doc 3341).

Loudness Range LRA measures the variation of loudness on a macroscopic timescale. It is supplementary to the measure of overall (integrated) loudness. Units are LU. The algorithm for calculating it can be found in EBU Tech Doc 3342.

The Maximum Permitted True Peak Level of a program during production is -1dBTP, measured with a meter compliant with both ITU-BS.1770 and EBU Tech Doc 3341.

#lm_J1_Bypass	Set the engine J1 into bypass mode
#lmJ1_linking_01~ #lmJ1_linking_04	The link function connects all parameters of two channels in a stereo pair. In situations where mono or mid-side transmission is utilized, it is recommended to unlink channels. More importantly it links the control circuits of the processing blocks in order to maintain the sound balance of these channels. Several link options are available depending on the operating and process control modes. While 4 x 2 mode is straight forward (linked or unlink), the 5.1 + 2 mode offers different modes for the surround channels, depending on the device. For EBU mode we have two options: `All without LFE` (LFE is not linked) and `All and LFE` (LFE is linked).
#ProgramJ1_01 ~ #ProgramJ1_04	Select a predefined loudness program A till P for this (linked) channel (group)
#lmJ1_Delay	Post loudness engine audio delay.
#lmJ2_control	Level Magic Process Control for loudness engine 2. See `#lmJ1_control` for exmplanation.
#lm_J2_Bypass	Set the engine J2 into bypass mode
#lmJ2_linking_01~ #lmJ2_linking_04	The link function connects all parameters of two channels in a stereo pair. In situations where mono or mid-side transmission is utilized, it is recommended to unlink channels. More importantly it links the control circuits of the processing blocks in order to maintain the sound balance of these channels. Several link options are available depending on the operating and process control modes. While 4 x 2 mode is straight forward (linked or unlink), the 5.1 + 2 mode offers different modes for the surround channels, depending on the device. For EBU mode we have two options: `All without LFE` (LFE is not linked) and `All and LFE` (LFE is linked).
#ProgramJ2_01 ~ #ProgramJ2_04	Select a predefined loudness program A till P for this (linked) channel (group)
#lmJ2_Delay	Post loudness engine audio delay.

LOUDNESS PROGRAM

lm_program

Here you predefine the loudness program. This can be any preset between A and P. If the preset is active editing the preset will be live. All menu settings that are preceded with a '#'-prefix are part of the preset.

#lm_program_name

To ease remembering which preset is used for what cases, you can name your active preset with this setting (maximum of 16 characters). The active program names are shown in the status items ActiveProgJ1_1~8 and ActiveProgJ2_1~8

#lm_input_gain

The input level can be altered by +/- 20dB to match level diagram needs or for static loudness offset control.

#lm_leveler_en

Enable or disable the Leveler function of the loudness engine

#lm_target_level

This is the target level of the whole system. All processes within the LevelMagic™ algorithm are designed to aim for the target level. It is crucial to understand that the target level is not a threshold and is not a reference for peak levels of any kind. For easier understanding imagine the target level as the balance point or center of gravity of the signal. Level Magic is balancing the signal around this centre, thus achieving a consistent loudness impression for the listener. Single peaks are not affected by this balancing process so that, as far as possible, the natural dynamics of the program are preserved.

Leveler – ITU mode – Loudness Target [0 … -50LKFS]

ITU has defined the unit of measure to LKFS (loudness K-Weighted referenced to digital Full Scale)

Leveler – EBU mode – Loudness Target [0 … -50LUFS] (Loudness Units referenced to digital Full Scale)

EBU has defined the unit of measure to LUFS

Important Note! LKFS and LUFS are different units for the same measure. They are fully compatible.

#lm_agc_time

This controls the speed at which LevelMagic™ tries to reach target level. This setting should not be confused with the attack time of a conventional sound processor. As the leveling process is a self-adjusting system this time is not an absolute term but rather an initial value that could exceed the numerical value many times. When setting it, it is necessary to take the overall function of the system into account. Production duties may require faster time settings, while ingest or play-out correction systems may need slower settings.

#lm_agc_max_gain

This parameter controls the maximum permitted gain change to reach the target level. It can be useful to limit the maximum amount of gain so as not to overly boost noise and other unwanted signals. The maximum attenuation is not affected by this setting. The system regulates the maximum attenuation adaptively to the signal structure.

#lm_freeze_lvl

The Freeze Level function holds the amount of gain or attenuation if the signal level drops below this threshold. It works in a similar way to a Hold function in other sound processors. Although this sounds difficult, it is in fact easy to understand with an example. Assuming the process applies a gain change of 10 dB to achieve target loudness, the input level will suddenly drop below freeze level. The gain change remains in its last state until the signal returns above Freeze Level. This behavior is different to the Processing Threshold (see below) where the gain change would return to its neutral state if the level falls below threshold. It is necessary to always set Freeze Level above the Processing Threshold to prevent unwanted release behavior.

#lm_trp_max_gain

The Transient Processor can be limited to a maximum processing gain range. Sometimes a hard setting with a very limited gain range can sound more natural than a softer response at full gain range. Adjusting the Transient Processor according to the designated overall behavior of the Level Magic process will improve its neutral processing character.

#lm_trp_response

The response of the Transient Processor is a highly self-adjusting process which reacts adaptively to the incoming signal structure. Its response can be adjusted in three presets from a more vital to a more relaxed setting but it also depends on the Limiter Processing setting. That means that the overall handling of transients and peaks is determined by the parameters of the Transient Processor and the Limiter.

#lm_limiter_en

Enable or disable the Limiter function of the loudness engine.

#lm_max_tpl

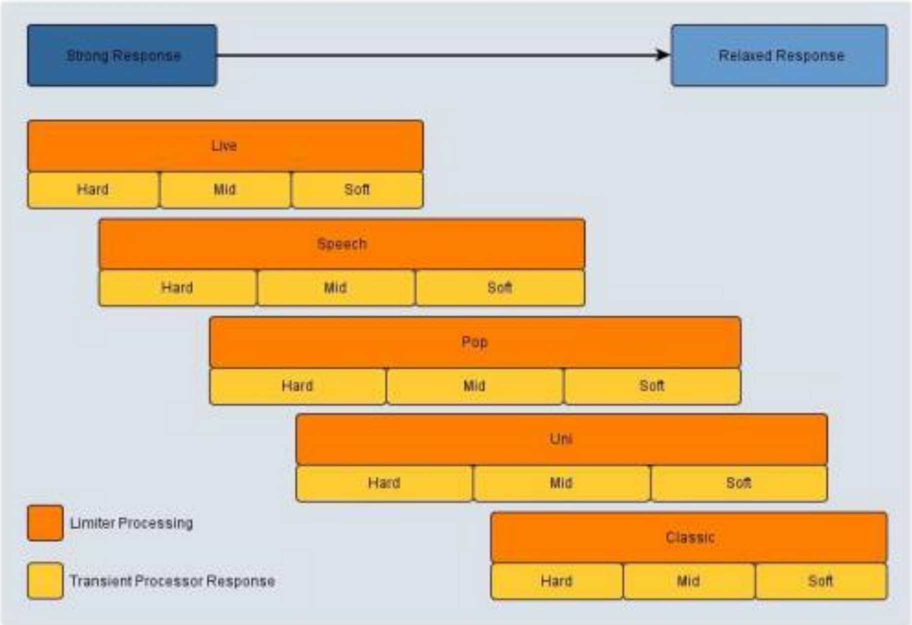
The Maximum Peak Level sets the threshold for the system's true peak limiter. Its fast detection system with a 2ms look-ahead time characterizes its response as a full brick wall limiter, not only for the obvious sample peaks but also for the hidden inter-sample peaks. Its precision fulfills all criteria defined in ITU 1770.

#lm_processing

Although generally speaking the limiting process will never sound bad, it is possible to further improve its neutrality by selecting one of the five given presets to match the actual content of the processed audio signal. The limiter setting has some impact on the Transient Processor.

- `live` fastest response
- `speech` fast response
- `pop` medium response
- `uni` slow response
- `classic` slowest response

The following picture illustrates this interaction between transient processor and Limiter depending on the parameter settings:



#exp_bypass

Expander: The classic Jünger expander reduces noise and unwanted sounds in a very unobtrusive and effective way. When the signal falls below threshold it is steadily reduced over a level range of 6 dB. In consequence 6 dB below threshold the maximum reduction is reached no matter which ratio is used. In Gate mode, its ratio is set to infinite to one.

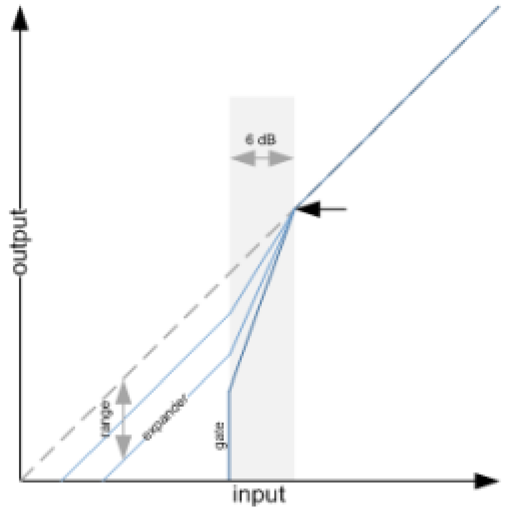
The bypass of the Expander function of the Loudness engine can be turned on or off with this item

#exp_threshold

Signals below threshold are reduced, signals above pass unaffected.

#exp_range

Maximum reduction range. Its value determines the maximum reduction of the input signal. This parameter is sometimes called 'floor', but differs in terminology. A floor level is defined as an absolute value in dBFS, no matter where the threshold is set. Range defines the relative range of reduction in dB below threshold and is thus independent from absolute values. When set to 'Gate', the input signal is muted.



#exp_release

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

#comp_bypass

Compressor: All signals below reference level are amplified according to the ratio and range settings, all signals above reference level are reduced in the same way. This is the 'classic' approach of earlier Jünger Audio compressor designs.

The bypass of the Compressor function of the Loudness engine can be turned on or off with this item

#comp_ref_lvl

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

#comp_ratio

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

#comp_range

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

#comp_processing

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

#lm_agc_recovery

All gain changes are processed adaptively to the incoming audio signal. Under normal conditions this adaptive reaction is working fine. However in special configurations it can be necessary to have a faster recovery or release time. Again, an example helps to explain the effect. If Level Magic is configured to work without gain or just a very small amount of positive gain, returning to unity from heavy attenuation can take quite some time. If a very loud part (above target) is followed by a quiet section (right at or below target) the recovery from damping leads to an unnatural 'fade in' effect for the quieter part. To decrease this effect 'AGC Recovery' can be set to 'fast' to accelerate this 'fade in time'. It is accelerated up to fifteen times its normal speed. The result sounds almost the same as if an audio engineer is riding the fader to correct unwanted level jumps and thus is very natural and well accepted by the listener. Please note that this setting is most helpful for setups where no positive gain (AGC amplification) is allowed. The effect works relative to the 'Leveler Time' setting and hence is more obvious for short 'Leveler Time' values.

#lm_init_gain

This parameter directly depends on the 'Clear Processing History' trigger. Instead of resetting the dynamic gain, it can be preloaded to a desired value the moment 'Clear Processing History' is triggered. The preload value is specified by the 'Initial Dynamic Gain' parameter. This helps reduce attack time artifacts, if switching programs incorporates a known and undesired level jump. It is not necessary to exactly predict the level difference between the programs but already helps to set up a few dB in the right direction. It is easy to understand with an example: The level jump from a movie to a commercial break is usually around 6 dB. Resetting Level Magic at the transition point helps to even out the difference. Nevertheless Level Magic needs some attack time to build up damping of 6 dB, which can be audible depending on the program structure. If the process is reset to a damping value between -4 to -6 dB instead, the attack time is much shorter and artifacts fall below the perception threshold. In many cases an Initial Dynamic Gain value of +/-3 dB is sufficient to create transitions with seamless loudness.

#lm_proc_thres

The 'Low Level Behavior' parameters define what happens if the level drops below the 'Processing Threshold'.

#lm_below_thres	In continuous operation the 'Below Threshold Mode' should remain in 'release'. In this case the dynamic gain slowly returns to its neutral state in case of signal absence. In this mode a returning signal would start a new processing period with its lead in attack time. This can be undesired, especially in production applications where transport operations introduce unnatural gaps. In those cases setting the 'Below Threshold Mode' to 'hold' will pause the dynamic processing at the last value until the signal returns. Returning signals are treated just like continuous signals. This function has some similarities to the'Freeze Level' but works with a different designation as it is meant to keep processing fluent over signal loss.
#ClrProcHistory	This is a triggered action that resets the dynamic processing without any release time. Imagine it as a short circuit to the timing circuits of an analog dynamic processor which discharges the whole system and immediately returns the dynamic gain to its neutral state. This function is useful to reset the process when switching programs (e.g. from movie to commercial breaks).
#upmix_enable	With this setting you can enable the upmix function. (H/GJA880/440 Only)
#upmix_source	With this setting you select between surround L/R (1L/1R) and independent stereo input (2L/2R) as the feed for the upmix module. Default is 1L/1R.
#upmix_profile	Set the upmix profile to Balanced or Front Projection. Default is Balanced.
#upmix_switch	You may turn on or off the upmix permanently or involve the surround detector for auto switching. Default is Auto.
#surr_det_source	The Surround Detect circuit observes the input channels to decide if the surround signal has disappeared in order to do an automatic upmix if desired. Here you set up which signals must be observed to detect a surround source: Only Center channel (C), Surround channels (S), or both (C+S). Default is Center channel (C)
#failover_thres	trigger threshold for the fail detector
#failover_wait	time from detection of an audio loss to the moment of switch over

#upmix_mode	Tells the algorithm if the input signal may have correlated components (Stereo) or not (Mono). Default is Stereo.
#upmix_process	Reaction time of the upmix processes. For news, sports, shows with permanently changing content (e.g. applause) setting "fast" is recommended while mid / slow is recommended for music, movies.
#upmix_proc_time	Jünger Audio has developed its own proprietary 5.1 upmix algorithm to get a good quality surround image from a stereo input signal. 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. The Look-Ahead Delay has great influence on the quality of the upmix process in regard to the latency of the process. The more time you have to analyze the stereo signal the better the result for the upmix signal will be. Depending on the system latency requirements (ingest vs. live broadcast) you may change the processing time accordingly.
#upmix_C_div	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/C/R (1.0). The effect will be a wider presentation of center signals in a surround sound image.
#upmix_sur_bal	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.
#upmix_sur_gain	Sets the level of Ls/Rs channels
#upmix_lfe_mode	You may turn this option on if the upmix process must generate a subwoofer signal that will appear in the LFE channel.
#upmix_lfe_gain	you can set the LFE level here
#downmix_gain	output gain of the stereo downmix signal
#downmix_C_lvl	Setting for the Center Mix level of the downmix
#downmix_sur_lvl	Setting for the Surround Mix level of the downmix
#stereo_source	Source of the +2 stereo channel.

EMBEDDING

#Emb_GrpSel

With this setting you select which audio groups of embedder 1 should be enabled for embedding audio into video output 1 and 2. The groups `group1`, `group2`, `group3` or `group4` can be separately set to be ON or OFF in the selection list. You can also choose to not enable any of the audio groups by setting this item to "____". By default it is set to "`1234`", All groups active.

#Emb-Mode

With `Emb-Mode` you select how the audio in all groups should be embedded into the video: `overwrite` the existing audio, or `Append`. Can also be set to `off` (switching off embedding entirely). Default is `overwrite`.

EMB AUDIO OUT

#SourceEmb-A1 ~ #SourceEmb-A4

With these settings you can select where the corresponding audio channels of embedder A (channel A1 till channel A4) are coming from:

- `SDI1`: Audio comes from SDI input 1 (embedded audio)
- `SDI2`: Audio comes from SDI input 2 (embedded audio)
- `AddOn01/16`: Audio comes from addon channels 1 to 16
- `AddOn17/32`: Audio comes from addon channels 17 to 32
- `J1Out`: Audio comes from Jünger loudness engine 1, ch. 1 to 8
- `J2Out`: Audio comes from Jünger loudness engine 2, ch. 1 to 8 (H/GJA880/840 Only)

#Emb-A1 ~ #EmbA4

With this setting you decide which audio channel of the above selected source is used for embedder A, respectively channel 1 till 4. Can be any of the available 16 channels or set to `off`.

#SourceEmb-B1 ~ #SourceEmb-B4

With these settings you can select where the corresponding audio channels (channel B1 till channel B4) of embedder B are coming from:

- `SDI1`: Audio comes from SDI input 1 (embedded audio)
- `SDI2`: Audio comes from SDI input 2 (embedded audio)
- `AddOn01/16`: Audio comes from addon channels 1 to 16
- `AddOn17/32`: Audio comes from addon channels 17 to 32
- `J1Out`: Audio comes from Jünger loudness engine 1, ch. 1 to 8
- `J2Out`: Audio comes from Jünger loudness engine 2, ch. 1 to 8 (H/GJA880/840 Only)

#Emb-B1 ~ #EmbB4

With this setting you decide which audio channel of the above selected source is used for embedder B, respectively channel 1 till 4. Can be any of the available 16 channels or set to `off`.

#SourceEmb-C1 ~ #SourceEmb-C4

With these settings you can select where the corresponding audio channels (channel C1 till channel C4) of embedder C are coming from:

- SDI1: Audio comes from SDI input 1 (embedded audio)
- SDI2: Audio comes from SDI input 2 (embedded audio)
- AddOn01/16: Audio comes from addon channels 1 to 16
- AddOn17/32: Audio comes from addon channels 17 to 32
- J1Out: Audio comes from Jünger loudness engine 1, ch. 1 to 8
- J2Out: Audio comes from Jünger loudness engine 2, ch. 1 to 8 (H/GJA880/840 Only)

#Emb-C1 ~ #EmbC4

With this setting you decide which audio channel of the above selected source is used for embedder C, respectively channel 1 till 4. Can be any of the available 16 channels or set to off.

#SourceEmb-D1 ~ #SourceEmb-D4

With these settings you can select where the corresponding audio channels (channel D1 till channel D4) of embedder D are coming from:

- SDI1: Audio comes from SDI input 1 (embedded audio)
- SDI2: Audio comes from SDI input 2 (embedded audio)
- AddOn01/16: Audio comes from addon channels 1 to 16
- AddOn17/32: Audio comes from addon channels 17 to 32
- J1Out: Audio comes from Jünger loudness engine 1, ch. 1 to 8
- J2Out: Audio comes from Jünger loudness engine 2, ch. 1 to 8 (H/GJA880/840 Only)

#Emb-D1 ~ #EmbD4

With this setting you decide which audio channel of the above selected source is used for embedder D, respectively channel 1 till 4. Can be any of the available 16 channels or set to off.

EmbA1_Gain ~ EmbD4_Gain

Adjusts the gain for the corresponding incoming audio channel between -60 and 12dB. Everything below -60dB (indicated as -999 dB) means the audio will be muted.

EmbA1_Phase ~ EmbD4_Phase

Adjusts the audio phase of the corresponding individual audio channel to 0 deg or 180 deg.

#EmbA1_Delay ~ #EmbD4_Delay

Adjusts the delay of the corresponding audio channel between 0 and 5000ms. These settings are part of the main preset. Processing delay for the loudness engine is 5.3 ms.

MISC

NonPCM-Bypass

With this setting you can switch to bypass audio proccessing for all non-PCM audio on or off.

Fade-Time

Fade/time is locked to 2 parameters: channel-switch and gain-change. It is used as the fade-in/out time of the channel-switch of audio channels. The old channel will be fade-out and the new channel will be fade in according to the time chosen with fade-time. Fade-Time is also used for smooth transitions when gain-values are changed. These smooth transitions are triggered by a change in Gain settings or a Preset change. With this setting you can manually set this fade time between `0ms` and `10.000ms`. The default is `500ms`.

Audio-Phase

If this setting is set to *Align*, the card ensures audio-phase alignment between multiple audio channels and audio groups, which is necessary for multi-channel (surround) purposes. If errors in the signal-chain occur the de-embedder blocks reset synchronously to maintain audio-phase-alignment. If this setting is set to *Off*, the card *eats-all* audio including errors. Even if there are DBN/ANC/ECC or channel-sequence errors, the de-embedder will pass them. Be aware that audio-phase-alignment between multiple audio channels and audio groups can not be maintained if this setting is set to *Off*.

Note: This setting can be helpful to solve problems in the field using equipment which doesn't follow the standards correctly.

AudioStatusBits

With this setting you select whether the audio status bits should be `Transparant` (same status bit on the outputs as on the inputs) or to `overwrite` them with new status bits.

Silence-Level

With this setting you set a loudness threshold for the silence detection. Can be set between -100 and -20 dBFS. When the audio goes below this value, a silence alert is triggered.

Silence-Time

Here you can set the threshold in time when an audio signal will be reported as silent. Can be set between 1 and 255 seconds. Default is 10 seconds.

METADATA

Extract_Line

With this item you set a line between line 0 and line 1125 from where you want to extract the metadata from the input when `S2020-Soucre` is set to `Rail1` or `Rail2`. By default set too line 0, in which case the S2020 is in auto-mode.

Extract_Ass_Ch

One attribute of the S2020 metadata is the association channel. The association channel is the channel to which the metadata is connected. You can select the S2020 metadata to be extracted from one of the possible associated channel pairs ranging from `Ch01/02` to `Ch15/16`. Can also be set to `None` (in case there is no association set in the S2020 source or to `Auto` (in which case the S2020 is extracted from the first available associated channel).

S2020-Emb

With this setting you decide whether you want to `overwrite` or to switch `off` metadata (S2020) inserting.

Insert_Line

With this setting you set a line to which the S2020 data should be inserted. Can be set between line 1 and line 1125. Default is line 9.

Insert_Method

There are 2 methods to insert S2020 (refer to the S2020 SMTPE document). Can be set to `Method A` or `Method B`. Default is `B`.

Insert_Ass_Ch

With this setting you select one of the 8 channel pairs (`Ch1/2` till `Ch15/16`) to which the metadata should be associated. Can also be set to `None` (which is also a valid value of the metadata item).

Shuffler_MD_Src

With this setting you decide which of the metadata sources you want to use as input of the Metadata Generator/Metadata Shuffler block in the above schematic. You can select the `SDI` input, or the `Local` metadata input. Default is `SDI`.

LocalOut_MD_Src

With this setting you decide which of the metadata sources you want to use as the metadata output on the I/O-panel. Can be the metadata from the SDI input (`SDI`, default), from the I/O-panel RJ45 metadata input (`Local`), or from the card's internal metadata generator (ShufflerOut).

ProgramConfig

The program config metadata describes the type of audio that is inside the bitstream to which this program is assigned. Can be one of the following values:

- `5.1+2`
- `5.1`
- `4x2`
- `3x2`
- `2+2`

Default is `5.1+2`.

FrameRate	With this you can set the metadata 'framerate' value. Can be 23.98, 24, 25, 29.97 or 30. Default is 25.
Program Sel	With this setting you select which Program preset you want to activate. Editing the preset will be live. Can be set between program 1 and program 4. Default is 1.
#ProgramText_Src	This item lets you select which metadata source to use to set the program text. Choices are between external program (Ext_Meta) or manually set program text via the card's own metadata settings (Int_Meta). Default is Int_Meta.
#ProgramText	You can describe the program in your own words in this field. Can be a string of maximum 16 characters.
#Dialogue_Src	This item lets you select which metadata source to use to set the dialogue level. Choices are between external program (Ext_Meta) or manually set dialogue level via the card's own metadata settings (Int_Meta). Default is Ext_Meta.
#Dialogue_Lev	Dialogue level sets the average loudness of a dialogue in a presentation. The range is from -31dB to -1dB. This item will only influence the output if #Dialogue_src is set to Int_Meta. The default setting is -27dB.
#LFE	The status of the LFE Channel parameter indicates to a Dolby Digital encoder whether an LFE Channel is present within the bitstream. Channel mode determines whether the LFE Channel parameter can be set. You must have at least three channels in order to be able to add an LFE channel. Can be either enable or disable. You can also choose to use the metadata settings in the external program (Ext_meta). Default setting is enable.
MD_Status_Src	In the status menu the status of all the metadata parameters of one metadata set can be monitored. With this setting you select which metadata set you want to monitor. Can be the ShufflerOut, Local or SDI. Refer to schematic in the Metadata header of the metadata settings for a visual explanation. Can also be switched to off, in which case there will be no status monitoring of metadata (default).
MD_Status_Pgm	With this item you select which program out of the above selected metadata set you want to monitor. Can be 1 till 8. Default is 1.

6 Status Menu

Introduction

The status menu indicates the current status of each item listed below.

SDI-Input_1

This status item indicates the presence and format of a valid signal in input 1. This is displayed as:

- 1080p60
- 1080p50
- 1080i60
- 1080i50
- 1080p30
- 1080p25
- 1080p24
- 1035i60
- 720p60
- 720p50
- SD525
- SD625
- NA

SDI-Input_2

This status item indicates the presence and format of a valid signal in input 2. This is displayed as listed under SDI-Input1.

SDI-Freq_1

Inidcates the frequency of SDI input 1. Can be 1:1, 1:1.001 or NA.

SDI-Freq_2

Inidcates the frequency of SDI input 2. Can be 1:1, 1:1.001 or NA.

CRC-Stat_1

Displays if there are CRC errors on input 1.

CRC-Stat_2

Displays if there are CRC errors on input 2.

Ref-Format

Displays the reference format. Can be one of the following:

- NA
- NTSC/480i
- PAL/576i
- 480p
- 576p
- 720p
- 1080i
- 1080p

Locked-To	Displays to what the card is locked: `Ref`, `SDI1`, `SDI2` or `Not Locked`.
Active-Out1	Indicates what the current source is of output 1, can be `SDI1` or `SDI2`.
Active-Out2	Indicates what the current source is of output 2, can be `SDI1` or `SDI2`.
IO-Delay_1	This indicates the delay of the input compared to the output. Displayed in ms.
GPI	Indicates the current GPI value
ANC-In1-Stat	Shows the status of the ancillary data in SDI input 1. Can be NA, OK or error.
ANC-In2-Stat	Shows the status of the ancillary data in SDI input 2. Can be NA, OK or error.
GrpInUse-In1	Displays which groups are in use in input 1. Displayed as for instance 1_3_ when groups 1 and 3 contain audio and for instance _234 when groups 2, 3 and 4 contain audio.
GrpInUse_In2	Displays which groups are in use in input 2. Displayed as for instance 1_3_ when groups 1 and 3 contain audio and for instance _234 when groups 2, 3 and 4 contain audio.
ATC -Stat	Indicates any ATC errors. can be `NA` (not available), `Present` or `Error`.
Grp-Ins	Indicates the status of the audio groups on the addon bus. Can be Error or OK.

SDI1DemFrmt01/02 ~ SDI1DemFrmt15/16

Displays the format of the corresponding de-embedded audio of input 1. Can be one of the following:

- NA
- PCM
- Null
- AC-3
- TimeStmp
- MPEG-1
- MPEG-2
- SMPTE-KLV
- Dolby E
- Caption data
- UserDef
- Rsvd

SDI2DemFrmt01/02 ~ SDI2DemFrmt15/16

Displays the format of the corresponding de-embedded audio of input 2. Can be one of the formats listed under SDI1DemFrmt01/12.

AddOnFrmt01/02 ~ AddOnFrmt31/32

Displays the format of the corresponding addon bus channels. Can be one of the formats listed under SDI1DemFrmt01/12.

EmbStatOutA1 ~ EmbStatOutD4

Display the status of the individual audio channels of the embedder output. Can be OK, Silence, Clipped or NA (not available)

EmbFrmtOutA1/2 ~ EmbFrmtOutD3/4

Displays the format of the corresponding audio channels of the embedder output. Can be one of the following formats:

- NA
- PCM
- Null
- AC-3
- TimeStmp
- MPEG-1
- MPEG-2
- SMPTE-KLV
- Dolby E
- Caption data
- UserDef
- Rsvd

SDI1S2020Stat

This item indicates the status of the S2020 (embedded metadata) signal on input 1. Can be OK, Error or NA (not available)

SDI1S2020Prog	This status indicates the program config as present in the S2020 signal of SDI1. Can be one of the following values: ■ 5.1+2 ■ 5.1+1+1 ■ 4+4 ■ 4x2 ■ 8x1 ■ 5.1 ■ 3x2 ■ 4 ■ 2+2 ■ 7.1 ■ Other ■ NA
SDI2S2020Stat	This item indicates the status of the S2020 (embedded metadata) signal on input 2. Can be `OK`, `Error` or `NA` (not available)
S2020-Src_Method	This status indicates the S2020 Mapping Method as present on the current SDI S2020 source. Can be `NA` (not available), `Method A` or `Method B`.
SDI2S2020Prog	This status indicates the program config as present in the S2020 signal of SDI2. Can be one of the values listed under `SDI1S2020Prog`.
LocMetaStat	Indicates the status of the metadata input on the backpanel. Can be `Ok`, `NA` or `Error`.
LocMetaProg	This status indicates the program config as present in metadata input on the backpanel. Can be one of the values listed under `SDI1S2020Prog`.
MD_ProgramConfig	This status indicates the program config as present on the metadata preset selected with `MetaDet`. Can be one of the values listed under `SDI1S2020Prog`.
MD FrameRate	Indicates the value of the frame rate metadata parameter.
MD PitchShift	Indicates the pitch shift of the metadata.
MD ProgramText	Displays the program's text field (set with #Program_txt).
MD AC3Datarate	Indicates the value of the AC3 bitrate metadata parameter.

MD Bitstream	Indicates the value of the bitstream mode metadata parameter.
MD ChannelMode	Indicates the value of the channel mode metadata parameter.
MD CenterMixLvl	Indicates the value of the Center downmix level metadata parameter.
MD SrndMixLvl	Indicates the value of the surround downmix level metadata parameter.
MD D_Surnd	Indicates the value of the Dolby surround metadata parameter.
MD LFE	Indicates the value of the LFE channel metadata parameter.
MD Dialog Lvl	Indicates the value of the dialogue level metadata parameter.
MD LanguageCode	Indicates the value of the language code metadata parameter.
MD AudioProdInfo	Indicates the value of the audio production info metadata parameter.
MD ProdMixLvl	Indicates the value of the audio production mix level metadata parameter.
MD ProdRoomType	Indicates the value of the audio production room type metadata parameter.
MD AC3Copyright	Indicates the value of the AC3 copyright metadata parameter.
MD AC3OrigBitstr	Indicates the value of the AC3 original bitstream metadata parameter.
MD Pref. Dwnmx	Indicates the value of the preferred downmix metadata parameter.
MD Lt/RtCDwnmx	Indicates the value of the Lt/Rt center downmix metadata parameter.
MD Lt/RtSDwnmx	Indicates the value of the Lt/Rt surround downmix metadata parameter.
MD Lo/RoCDwnmx	Indicates the value of the Lo/Ro center downmix metadata parameter.
MD Lo/RoSDwnmx	Indicates the value of the Lo/Ro surround downmix metadata parameter.

MD D_Srnd Ex	Indicates the value of the Dolby surround EX metadata parameter.
MD D_HeadPhone	Indicates the value of the Dolby headphone metadata parameter.
MD ADConvType	Indicates the value of the A/D conversion type metadata parameter
MD DC Filter	Indicates the value of the DC filter metadata parameter.
MD Lowpass Fil	Indicates the value of the Low pass filter metadata parameter.
MD LFE Filter	Indicates the value of the LFE filter metsdata parameter.
MD Sur PhShift	Indicates the value of the surround phase shift metadata.
MD Sur3d Att	Indicates the value of the surround 3dB attenuate metadata.
MD RFPreEmph	Indicates the value of the RF pre emphasis metadata parameter.
MD RF Mode	Indicates the value of the RF mode metadata parameter.
MD Line Mode	Indicates the value of the line mode metadata parameter
FPGA-Stat	Displays the status of the FPGA chip. Can be error or OK.
DM-D_Status	Indicates the status of daughter I/O board D. can be OK, NA or Error.
DM-D_Type	Displays which type of input or output board is currently detected on circuit D. For the GJA/HJA cards this should always be `AXD469`.
Mode_J1	Shows the operating mode of the Junger loudness engine J1, which is dependent of the Junger License. Without a correct license for this engine it is set to `Bypass`. Other possible modes are `5.1+2` or `4x2.0`.
Mode_J2	Shows the operating mode of the Junger loudness engine J2, which is dependent of the Junger License. Without a correct license for this engine it is set to `Bypass`. Other possible modes are `5.1+2` or `4x2.0`.

7 Events Menu

Introduction

An event is a special message that is generated on the card asynchronously. This means that it is not the response to a request to the card, but a spontaneous message.

What is the Goal of an event?

The goal of events is to inform the environment about a changing condition on the card. A message may be broadcast to mark the change in status. The message is volatile and cannot be retrieved from the system after it has been broadcast. There are several means by which the message can be filtered.

Events

The events reported by the card are as follows;

Announcements

`Announcements` is not an event. This item is only used for switching the announcement of status changes on/off. 0=off, other =on

Input_A

Input_A can be selected between 0 .. 255. 0= no event, 1..255 is the priority setting.

Input_B

Input_B can be selected between 0 .. 255. 0= no event, 1..255 is the priority setting.

Ref-Status

Reference can be selected between 0 .. 255. 0= no event, 1..255 is the priority setting.

What information is available in an event?

The message consists of the following items;

1) A message string to show what has happened in text, for example: “INP_LOSS”, “REF_LOSS”, “INP_RETURN”.
2) A tag that also shows what happens, but with a predefined number: e.g. 1 (= loss of input), 2 (= loss of reference), 129(= 1+128 = return of input). For a list of these predefined tags see the table on the next page.
3) A priority that marks the importance of an event. This value is defined by the user and can have any value between 1 and 255, or 0 when disabled.
4) A slot number of the source of this event.

The Message String

The message string is defined in the card and is therefore fixed. It may be used in controlling software like Synapse Set-up to show the event.

The Tag

The tag is also defined in the card. The tag has a fixed meaning. When controlling or monitoring software should make decisions based on events, it is easier to use the tag instead of interpreting a string. The first implementation is the tag controlled switch in the GPI16.

In cases where the event marks a change to fault status (e.g. 1 for Loss of Input) the complement is marked by the tag increased by 128 (80_{hex}) (e.g. 129 (81_{hex}) for Return of Input).

Defining Tags

The tags defined for the card are:

Event Menu Item	Tag		Description
Announcements	0 or NA	0 or NA	Announcement of report and control values
Input_A	01_{hex}=INPA_LOSS	81_{hex}=INPA_RETURN	input A lost or returned
Input_B	02_{hex}=INPB_LOSS	82_{hex}= INPB_RETURN	input B lost or returned
Reference	03_{hex}=REF_LOSS	83_{hex}=REF_RETURN	reference lost or returned

The Priority

The priority is a user-defined value. The higher the priority of the alarm, the higher this value. Setting the priority to Zero disables the announcement of this alarm. Alarms with priorities equal or higher than the Error Threshold setting of the RRC will cause the error LED on the Synapse rack front panel to light.

The Address

Together with the message string or the tag, the slot number or address of the card is relevant to be able to assign the event to a certain card.

8 LED Indication

Error LED

The error LED indicates an error if the internal logic of the GJA-HJA card is not configured correctly or has a hardware failure.

Input_A LED

This LED indicated the presence of a valid SDI video signal on input A.

Input_B LED

This LED indicated the presence of a valid SDI video signal on input B.

ANC Data LED

Indicates the presence of embedded audio within the input signal.

Reference LED

Indicated the presence of a valid reference signal on the selected reference input connector (ref-1 or ref-2).

Data Error LED

This LED indicates a CRC error.

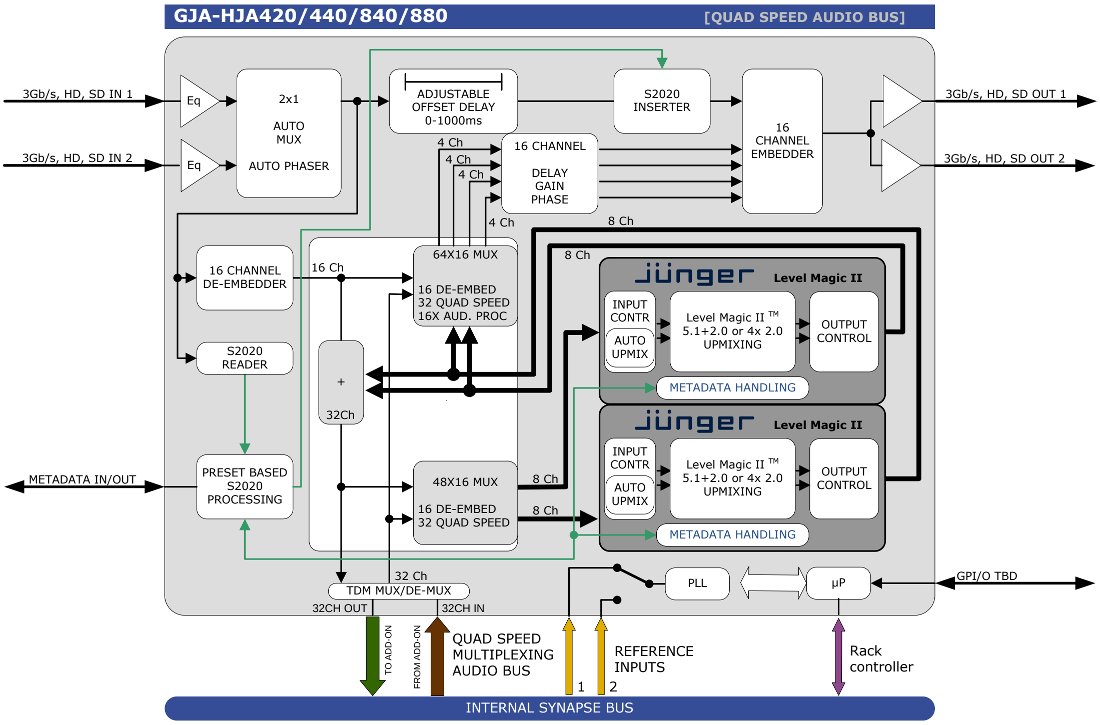
Connection LED

This LED illuminates after the card has initialized. The LED lights for 0.5 seconds every time a connection is made to the card.

Error LED

The error LED indicates an error if the internal logic of the card is not configured correctly or has a hardware failure.

9 Block Schematic



10 Connector Panels

The GJA and HJA can be used with the BPH18, the BPH18D or the bypass relay equivalents. The following table displays the pinout of these backpanels in combination with the card.

BPH18D BHX18D

3Gb/s, HD, SD SDI INPUT 1 (OPTIONAL FIBER INPUT)

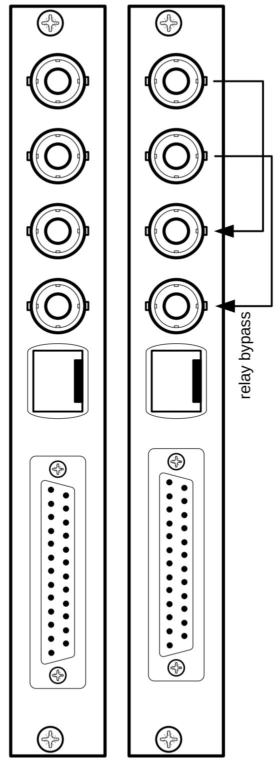
3Gb/s, HD, SD SDI INPUT 2 (OPTIONAL FIBER INPUT)

3Gb/s, HD, SD SDI OUT 1 (OPTIONAL FIBER OUTPUT)

3Gb/s, HD, SD SDI OUT 2 (OPTIONAL FIBER OUTPUT)

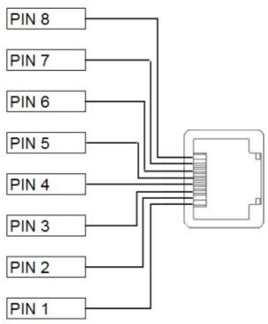
GPI/METADATA INPUT/OUTPUT

NOT CONNECTED



!Unused inputs and outputs must be terminated with the correct impedance!

GPI pinning



Pin	Function
1	Ground
2	GPI 1
3	GPI 2
4	GPI 3
5	GPI 4 / TXA(+)
6	GPI 5 / TXB(-)
7	GPI 6 / RXA(+)
8	GPI 7 / RXB(-)

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