DSF66

SoundField® dual digital audio upmixer and downmixer.

Installation and Operation manual

Quad speed ADD-ON

SOUNDFIELD

Committed.

®AXON

TECHNICAL MANUAL

DSF66

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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, SFR08 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: 29-06-2010
Date last revised: 13-11-2013

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 DSF66

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 controller 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 described in those manuals as well.

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

Although not required to use Axon 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.

Locating the card

The Synapse card can be placed vertically in an SFR18 frame or horizontally in an SFR08 or SFR04 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 LEDs will light for a few seconds, this is the time it takes to initialise the card.

3 A Quick Start

When Powering-up

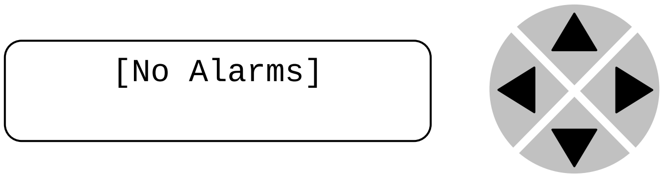
On powering up the Synapse frame, the card set will use basic data and default initialization settings. All LEDs will light during this process. After initialization, several LEDs will remain lit – the exact number and configuration is dependant upon the number of inputs connected and the status of the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

Changing settings and parameters

The front panel controls or 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.

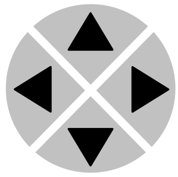
Press ►	To go forward through the menu structure.
Press ◄	To go back through the menu structure.
Press ▲	To move up within a menu or increase the value of a parameter.
Press ▼	To move down through a menu or decrease the value of a parameter.

REMARK: 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

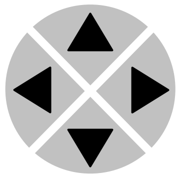
```
RRC18 [Select Card]
>S01=SFS10
```



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
```

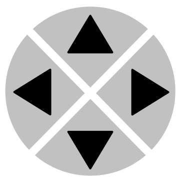


Pressing the ► selects the menu item shown, in this example `Settings`.

(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
```

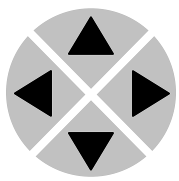


Pressing the ► selects the settings item shown, in this example `SDI-Format`.

(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]
```



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

Axon Cortex 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 operation of Axon Cortex, please refer to the Axon Cortex help files (press F1 in any window).

Menu Structure Example

Slot	Module		Item		Parameter		Setting
▲							
▲							
S02			Identity				
▲			▲				
S01	SFS10	▶	Settings	▶	SDI-Format	▶	Auto
▼			▼		▼		▼
S00	RRC18		Status		Mode		625
			▼		▼		▼
			Events		Ref-Input		525
					▼		
					H-Delay		
					▼		
					▼		

REMARK: Further information about Front Panel Control and Axon Cortex can be obtained from the rack controller manual and Axon Cortex help files.

4 The DSF66 Card

Introduction

The DSF66 is a dual hardware stereo-to-5.1 upmix and downmix processor, designed for HD broadcasters who use a lot of archived stereo material and wish to generate acceptable 5.1 broadcast mixes from stereo soundtracks. The DSF66 has a shared metadata input and shared physical I/O.

Software and hardware upmixing tools have existed for some years, but most of them create material for the extra three channels in a 5.1 mix by using processing, for example adding reverb or applying phase-shifts to the stereo material to create information for the rear surround channels. Instead, the DSF66 generates the material for the extra channels by closely analyzing the source stereo signal over time. Using a unique algorithm developed for the purpose, the DSF66 can detect reverberant content in the stereo signal, differentiate it from the direct sounds in the mix, and separate it out.

Users can adjust the details of the processing directly from the DSF66 GUI in Cortex, with control offered over a variety of different parameters including the level of the direct and ambient components in the front and rear channels, and the divergence of the Centre channel in the generated 5.1 mix, with options from a discrete Centre channel at one extreme to a phantom Centre at the other. Output level controls are also offered for each of the channels in the final 5.1 mix.

The I/O configuration of the DSF66 can be changed to allow for convenient connection of external inputs, outputs or a combination. In standalone mode the DSF66 has 4 AES/EBU inputs and 4 AES/EBU outputs (8 mono in and 8 mono out). In (quad speed) ADD-ON mode the unit can be configured as 8in (16 mono in) or 8 out (16 mono out) to connect either 16 mono external source channels listen to 16 external channels in 8-out mode. If processing of embedded audio from a master card is required the unit can be used without physical I/O and all channels are routed from and to the Quad Speed Audio bus

- 3 physical I/O modes 8-in, 4-in + 4-out, or 8-out
- Output gain and delay adjustments
- 2x Upmix stereo to 5.1
- 2x Downmix from 5.1
- Cross fading between upmixed and discrete 5.1 (5.1/2.0 input auto-sensing)
- Cross fading between downmixed and discrete 2.0 (5.1/2.0 input auto-sensing)
- 8 presets for convenient storing of use cases.
- Locks to Black & Burst, AES input and Mastercard.
- Full control and status monitoring through the front panel of the SFR04/SFR08/SFR18 frame and the Ethernet port (ACP)

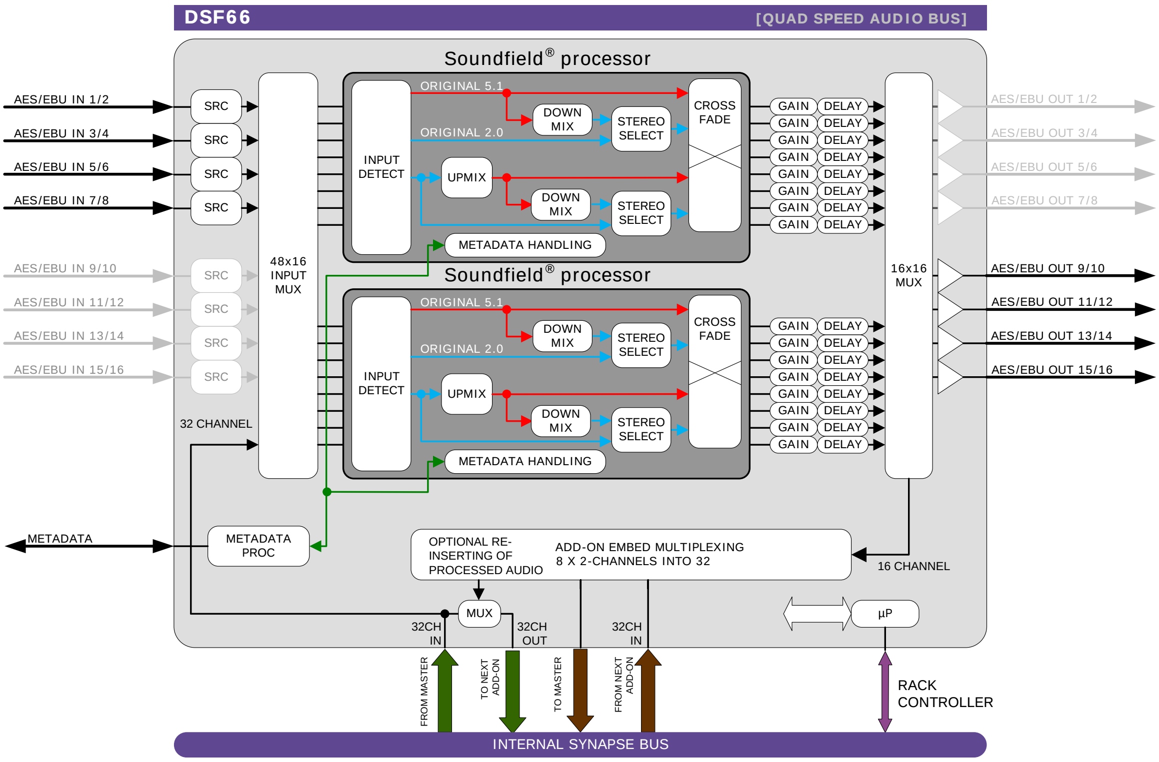
Applications

- High-End upmixing and downmixing of broadcast related program audio streams
  - To ensure a constant 5.1 + Stereo output regardless of input format
  - Simultaneous independent Upmixer and Downmixer (LoRo or LtRt)

Program Procedure

This card can be updated with new firmware when new firmware versions are released by Axon. You can download the .spf file from our website when new releases are announced. To upgrade the DSF66 you can follow the instructions as described in the 'reprogramming cards quick-guide', downloadable via our website.

Block schematics



5 Settings Menu

Introduction

The settings menu displays the current state of each setting within the DSF66 and enables the item to be changed or adjusted.

Settings can be changed using the front panel of the Synapse frame (SFR18, SFR08 or SFR04) or Axon Cortex.

Please refer to chapter 3 for information on the Synapse front panel control and Axon Cortex.

SYSTEM CONTROL

GPI Control

The DSF66 has 3 physical GPI contacts to control the card. You can use the GPI contact to either control the Presets, or to control the Upmix functions. These are the possible settings:

- `Preset_Latch`: Latching GPI mode to control presets, when a contact is closed momentarily (edge triggered). Refer to the following table for all possible preset triggers:

GPI 3	GPI 2	GPI 1	Preset value
0	0	1	#1
0	1	0	#2
1	0	0	#3

- `Preset_Non-Latch`: Non-latching GPI mode to control presets, when a contact is closed all the time (level triggered). Refer to the following table for all possible preset triggers:

GPI 3	GPI 2	GPI 1	Preset value
0	0	0	#1
0	0	1	#2
0	1	0	#3
1	0	0	#4

- `Preset_BCD` mode: Binary mode to control presets. When nothing is closed, the value is 0. When all 3 contacts are closed then the value is 7 and preset 8 is selected (since value 0 = preset 1). Refer to the following table for all possible combinations:

GPI 3	GPI 2	GPI 1	Preset value
0	0	0	#1
0	0	1	#2
0	1	0	#3
0	1	1	#4
1	0	0	#5
1	0	1	#6
1	1	0	#7
1	1	1	#8

- `Upmix_Latch`: Latching GPI mode to control the upmixer, when a contact is closed momentarily (edge triggered). Can be switched on or off by use of the GPI's contacts in the following manner:

GPI 3	GPI 2	GPI 1	Upmix value
0	0	1	On
0	1	0	Off
1	0	0	Off
0	0	0	Off

- `Upmix_Non_Latch`: Non-Latching GPI mode to control the upmixer, , when a contact is closed all the time (level triggered). Can be switched on or off by use of GPI contact 1 only in the following manner (GPI2 and 3 are discarded):

GPI 3	GPI 2	GPI 1	Upmix value
x	x	1	On
x	x	0	Off

- `Preset_Upmix_BCD` mode: Binary mode to control loudness presets including upmix on/off settings. When nothing is closed, the value is 0. When all 3 contacts are closed the GPI BCD value if 8 and preset 4 + Upmix ON is selected (since value 0 = preset 1). Menu item `UP_Mix` has to be set to `GPI-Upmix` and `#Prest_actual` or `#DwnPrst_Actual` set to `GPI` in order for this setting to function properly. Refer to the following table for all possible combinations:

GPI 3	GPI 2	GPI 1	GPI BCD value	Preset settings
0	0	0	1	Loudness preset 1
0	1	0	2	Loudness preset 2
1	0	0	3	Loudness preset 3
1	1	0	4	Loudness preset 4
0	0	1	5	Loudness preset 1 + Upmix On
0	1	1	6	Loudness preset 2 + Upmix On
1	0	1	7	Loudness preset 3 + Upmix On
1	1	1	8	Loudness preset 4 + Upmix On

Lock-Mode

The DSF66 can be used as an ADD_ON card (in combination with an embedder/de-embedder card). In this case you are referred to the setting `MasterCard`, which will extract the reference from the master card. It is also possible to use an external signal to lock to. In that case you are referred to the setting:

`AES1` = Locks to the AES/EBU signal on input 1(default)
`Ref1` = The B&B reference input of the rackcontroller
`Ref2` = The second B&B reference input of the rack controller (if available)
`Mastercard` = Locks to the ADD-ON bus input (always use this setting when using quad speed add-on bus functionality).

In_Out

`In_Out` determines how the synapse bus inputs and outputs work. There are 3 modes:

- `4In-4Out`: 4 AES/EBU input and 4 AES/EBU outputs
- `8In`: 8 AES/EBU input and no outputs
- `8Out`: no inputs and 8 AES/EBU outputs

Refer to the block schematics for more details.

INPUT CONTROL

Sel_Chn_Lf_1

With this setting you select the source for the left front speaker channel of the first Soundfield processor. This can either be `local` (physical AES inputs) or `Master` (inputs from the master card via the ADD-ON bus). Default is `Local`.

Chn_Lf_1

Here you select a channel out of the above selected source which will be the left front speaker channel of processor 1. Default is channel 1.

Sel_Chn_Rf_1

With this setting you select the source for the right front speaker channel of the first Soundfield processor. This can either be `local` (physical AES inputs) or `Master` (inputs from the master card via the ADD-ON bus). Default is `Local`.

Chn_Rf_1

Here you select a channel out of the above selected source which will be the right front speaker channel of processor 1. Default is channel 2.

Sel_Chn_C_1

With this setting you select the source for the center speaker channel of the first Soundfield processor. This can either be `local` (physical AES inputs) or `Master` (inputs from the master card via the ADD-ON bus). Default is `Local`.

Chn_C_1

Here you select a channel out of the above selected source which will be the center speaker channel of processor 1. Default is channel 3.

Sel_Chn_LFE_1

With this setting you select the source for the low frequency effects channel of the first Soundfield processor. This can either be `local` (physical AES inputs) or `Master` (inputs from the master card via the ADD-ON bus). Default is `Local`.

Chn_LFE_1

Here you select a channel out of the above selected source which will be the low frequency effects channel of processor 1. Default is channel 4.

Sel_Chn_Ls_1

With this setting you select the source for the left surround speaker channel of the first Soundfield processor. This can either be `local` (physical AES inputs) or `Master` (inputs from the master card via the ADD-ON bus). Default is `Local`.

Chn_Ls_1	Here you select a channel out of the above selected source which will be the left surround speaker channel of processor 1. Default is channel 5.
Sel_Chn_Rs_1	With this setting you select the source for the right surround speaker channel of the first Soundfield processor. This can either be `local` (physical AES inputs) or `Master` (inputs from the master card via the ADD-ON bus). Default is `Local`.
Chn_Rs_1	Here you select a channel out of the above selected source which will be the right surround speaker channel of processor 1. Default is channel 6.
Sel_Chn_L_1	With this setting you select the source for the left stereo channel of the first Soundfield processor. This can either be `local` (physical AES inputs) or `Master` (inputs from the master card via the ADD-ON bus). Default is `Local`.
Chn_L_1	Here you select a channel out of the above selected source which will be the left stereo channel of processor 1. Default is channel 7.
Sel_Chn_R_1	With this setting you select the source for the right stereo channel of the first Soundfield processor. This can either be `local` (physical AES inputs) or `Master` (inputs from the master card via the ADD-ON bus). Default is `Local`.
Chn_R_1	Here you select a channel out of the above selected source which will be the right stereo channel of processor 1. Default is channel 8.
Sel_Chn_Lf_2	With this setting you select the source for the left front speaker channel of the second Soundfield processor. This can either be `local` (physical AES inputs) or `Master` (inputs from the master card via the ADD-ON bus). Default is `Local`.
Chn_Lf_2	Here you select a channel out of the above selected source which will be the left front speaker channel of processor 2. Default is channel 1.
Sel_Chn_Rf_2	With this setting you select the source for the right front speaker channel of the second Soundfield processor. This can either be `local` (physical AES inputs) or `Master` (inputs from the master card via the ADD-ON bus). Default is `Local`.
Chn_Rf_2	Here you select a channel out of the above selected source which will be the right front speaker channel of processor 2. Default is channel 2.

Sel_Chn_C_2

With this setting you select the source for the center speaker channel of the second Soundfield processor. This can either be `local` (physical AES inputs) or `Master` (inputs from the master card via the ADD-ON bus). Default is `Local`.

Chn_C_2

Here you select a channel out of the above selected source which will be the center speaker channel of processor 2. Default is channel 3.

Sel_Chn_LFE_2

With this setting you select the source for the low frequency effects channel of the second Soundfield processor. This can either be `local` (physical AES inputs) or `Master` (inputs from the master card via the ADD-ON bus). Default is `Local`.

Chn_LFE_2

Here you select a channel out of the above selected source which will be the low frequency effects channel of processor 2. Default is channel 4.

Sel_Chn_Ls_2

With this setting you select the source for the left surround speaker channel of the second Soundfield processor. This can either be `local` (physical AES inputs) or `Master` (inputs from the master card via the ADD-ON bus). Default is `Local`.

Chn_Ls_2

Here you select a channel out of the above selected source which will be the left surround speaker channel of processor 2. Default is channel 5.

Sel_Chn_Rs_2

With this setting you select the source for the right surround speaker channel of the second Soundfield processor. This can either be `local` (physical AES inputs) or `Master` (inputs from the master card via the ADD-ON bus). Default is `Local`.

Chn_Rs_2

Here you select a channel out of the above selected source which will be the right surround speaker channel of processor 2. Default is channel 6.

Sel_Chn_L_2

With this setting you select the source for the left stereo channel of the second Soundfield processor. This can either be `local` (physical AES inputs) or `Master` (inputs from the master card via the ADD-ON bus). Default is `Local`.

Chn_L_2

Here you select a channel out of the above selected source which will be the left stereo channel of processor 2. Default is channel 7.

Sel_Chn_R_2

With this setting you select the source for the right stereo channel of the second Soundfield processor. This can either be `local` (physical AES inputs) or `Master` (inputs from the master card via the ADD-ON bus). Default is `Local`.

Chn_R_2

Here you select a channel out of the above selected source which will be the right stereo channel of processor 2. Default is channel 8.

ENGINE1_PRST

Prst_Eng_act1

With this item you can manually change the currently active preset of Soundfield processor 1. This is an overarching preset involving the engine mode preset, the upmix preset, the downmix preset and the metadata presets. Can be any preset between 1 and 8. Can also be set to `GPI`, in which case the Presets are triggered by GPI contacts (requires `GPI_control` to be set to a GPI mode. Default is set to 1.

Prst_Eng_adj1

Here you can select which of the 8 selectable presets of Soundfield processor 1 you want to adjust. Changing this will not change the active preset, unless the currently active preset is the same you are going to edit.

#Prst_Mode_act1

This is the actual engine mode preset for Soundfield processor 1. The settings `#Engine_mode` and `#Surr_Det_mode` are part of the preset. Default is preset 1.

#Prst_Up_act1

This is the actual upmix parameters preset for Soundfield processor 1. The settings with a `#UP-`prefix are part of this preset. Default is 1.

#Prst_Down_act1

This is the actual downmix parameters preset for Soundfield processor 1. The settings with a `#DM-`prefix are part of this preset. Default is 1.

#Prst_Md_act1

This is the actual metadata set that will be used for Soundfield processor 1. All metadata parameters are part of this preset. Default is set A.

#Prst_Md_rev1

When the metadata source is lost, this menu items select what metadata will must be used instead by soundfield processor 1. You can set it to switch to internal parameter sets `A` to `H`. Default is set A.

#Ext prog ID1

This setting lets you select a program inside the external metadata source which will be used by Soundfield processor 1 whenever a metadata parameter is set to `Ext_meta`. Default is `Prog#1`.

	ENGINE2_PRST
Prst_Eng_act2	With this item you can manually change the currently active preset of Soundfield processor 2. This is an overarching preset involving the engine mode preset, the upmix preset, the downmix preset and the metadata presets. Can be any preset between 1 and 8. Can also be set to `GPI`, in which case the Presets are triggered by GPI contacts (requires `GPI_control` to be set to a GPI mode. Default is set to 1.
Prst_Eng_adj2	Here you can select which of the 8 selectable presets of Soundfield processor 2 you want to adjust. Changing this will not change the active preset, unless the currently active preset is the same you are going to edit.
#Prst_Mode_act2	This is the actual engine mode preset for Soundfield processor 2. The settings `#Engine_mode` and `#Surr_Det_mode` are part of the preset. Default is preset 1.
#Prst_Up_act2	This is the actual upmix parameters preset for Soundfield processor 2. The settings with a `#UP-`prefix are part of this preset. Default is 1.
#Prst_Down_act2	This is the actual downmix parameters preset for Soundfield processor 2. The settings with a `#DM-`prefix are part of this preset. Default is 1.
#Prst_Md_act2	This is the actual metadata set that will be used for Soundfield processor 2. All metadata parameters are part of this preset. Default is set A.
#Prst_Md_rev2	When the metadata source is lost, this menu items select what metadata will must be used instead by soundfield processor 2. You can set it to switch to internal parameter sets `A` to `H`. Default is set A.
#Ext prog ID2	This setting lets you select a program inside the external metadata source which will be used by Soundfield processor 2 whenever a metadata parameter is set to `Ext_meta`. Default is `Prog#1`.

	ENGINE_MODE
Prst_Mode_adj	Here you can select which of the 8 selectable engine mode presets you want to adjust. Changing this will not change the active engine mode preset, unless the currently active preset is the same you are going to edit. The menu setting `#Engine_mode` and `#Surr_Det_mode` are part of this preset.

#Engine_Mode

To simplify the process of setting up the required processing and signal routing the DSF66 has 6 engine modes to choose from. `Simple Upmix`, `Simple Downmix`, `Upmix with Autodetect`, `Downmix with Autodetect`, `Upmix and Downmix with Autodetect`, `Stand Alone Upmix and Downmix`.

#Surr_Det_Mode

With this setting you can select how the Soundfield processor must detect surround sound. This can be done by only detecting a center speaker channel (`C_Detect`), by only detecting left and right surround channels (`Ls/Rs_Detect`) or both center as well as left and right surround channels (`C/Ls/RS_Detect`). Default is `C_Detect`.

UPMIX_PARAMS

Prst_Up_adj

Here you can select which of the 8 selectable upmix presets you want to adjust. Changing this will not change the active upmix preset, unless the currently active preset is the same you are going to edit. The settings with an `#UP-`prefix are part of this preset

#Up_Trigger

With this setting you set how the upmixer of the SoundField engine is controlled. Can be:

- `Always On:` Upmixing is manually set to be always on (Default)
- `Metadata Loss:` Upmixing is triggered when metadata is lost
- `GPI1:` When this setting is used, the Upmixer is switched on or off by means of GPI. Note that this setting requires setting `GPI_control` to be set to `Upmix_Latch` or `Upmix_nonlatch`.

#UP_Matrix_mode

The DSF66 has two modes of operation, Upmix and Matrix Decode

- `Upmix Mode`: Upmix mode should be selected when processing standard non-matrix encoded stereo material. In Upmix mode the DSF66 sends only ambient sounds to the rear surround channels. The only exception to this is when the Width control is enabled to the point where a 'wrap around' effect is created by feeding varying degrees of front direct sound to the rear channels.
- `Matrix Decode Mode`: Matrix Decode mode should be selected when processing matrix encoded stereo material (Dolby® Pro Logic® etc) .In this mode the DSF66 will detect any stereo material that has been matrix encoded and any direct sound intended by the encoding process for the rear surround channels will be sent there.

#UP_Drct_Snd	The Direct Sound control increases or decreases the level of direct sound by + or - 6dB. In most applications direct sound is only present in the front three channels.
#UP_Front_Amb	The Front Ambient Sound control increases or decreases the level of ambient sound in the front Left and Right channels by + or - 6dB.
#UP_Rear_Amb	The Rear Ambient Sound control increases or decreases the level of ambient sound in the rear left and right surround channels by + or - 6dB.
#UP_Width	The Width control enables the Direct Sound in the original stereo image to be made wider - at its most extreme the Direct Sound will be in the rear channels.
#UP_Cntr_Div	The Centre Divergence control takes the centre channel and diverges it to front Left and Right by the amount set on the control. When no divergence is set all mono material will appear exclusively in the centre channel (hard centre). When divergence is set to full, all mono material will be sent to the front Left and Right channels equally and the centre channel will be muted (phantom centre). Any other divergence setting will create a mix between hard and phantom centre.

DOWNMIX_PARAMS

Prst_Down_adj	Here you can select which of the 8 selectable downmix presets you want to adjust. Changing this will not change the active downmix preset, unless the currently active preset is the same you are going to edit. The settings with a `#DM-`prefix are part of this preset
#DM_Matrix_mode	This parameter selects whether the downmix is a matrix encoded downmix (Lt/Rt) or not (Lo/Ro).
#DM_Adapt_EQ	Enabling the Adaptive EQ parameter will activate a gentle adaptive EQ, this EQ will counter any frequency coloration that may occur due to downmix incompatibilities in the 5.1 material.
#DM_Frnt_Lvl	This setting sets the volume level of the front (left and right) channels. Can be adjusted between -12dB and +12dB. Default is 0dB.
#DM_Cntr_Lvl	This setting sets the volume level of the center speaker channel. Can be adjusted between -12dB and +12dB. Default is 0dB.

#DM_Surr_Lvl

This setting sets the volume level of the surround (left and right) channels. Can be adjusted between -12dB and +12dB. Default is 0dB.

#DM_LFE_Ena

With this setting you can enable (`on`) or disable (`off`) the LFE channel for the downmix. Default is `off`.

#DM_LFE_Lvl

This setting sets the volume level of the low frequency effects channel (if enabled). Can be adjusted between -12dB and +12dB. Default is 0dB.

#DM_Amb_Ena

When enabled we can select separate direct sound and ambient sound downmix values for Ls and Rs.

#DM_Amb_Lvl

Sets the Ambient downmix level for the Ls and Rs signals, this will only have an effect when #DM_Amb_Ena is active.

METADATA_PARAMS

MD_Status_Src

In the status menu, the metadata values of a metadata stream can be displayed. With this setting you can choose one out of the maximum of 8 program streams (`pgr1` ~ `prg8`) to be used as source for this status items.

Program Config

The program config metadata describes the type of audio that is inside the bitstream to which this program is assigned. Can be set to `5.1+2` or to `Ext_Meta` (in which case the `program_config` data will be taken from the external metadata program).

FrameRate

With this you can set the metadata ‘framerate’ value. Can be `23.98`, `24`, `25`, `29.97`, `30` or set to use the setting in the external metadata input (`Ext_meta`).

Prst_Meta_adj

Here you can select which of the 8 selectable metadata sets you want to adjust. Changing this will not change the active engine mode preset, unless the currently active preset is the same you are going to edit. All metadata parameter settings are part of this preset

	METADATA_PARAMS
MD_Status_Src	In the status menu, the metadata values of a metadata stream can be displayed. With this setting you can choose one out of the maximum of 8 program streams (pgr1 ~ prg8) to be used as source for this status items.
Program Config	You can set this item to Int_meta or Ext_meta. Set to Int_meta, the #PrgmCnfg Id1 and #PrgmCnfg Id2 values will be used to fill the Program config metadata. Set to Ext_meta, the external metadata input value will be used.
FrameRate	With this you can set the metadata 'framerate' value. Can be 23.98, 24, 25, 29.97, 30 or set to use the setting in the external metadata input (Ext_meta).
Prst_Meta_adj	With this item you can select which metadata set you want to adjust parameter setting of. Possible are A till H. Default is set to parameter set A. All following items preceded with '#' are slaves of this set. Note: Unless this setting is set to a currently in use metadata set, changing metadata settings will not have a direct effect on the output.
#Dialogue-src	This item lets you select which metadata source to use to set the dialogue level. Choices are between external program (Ext_Meta), manually set dialogue level via the card's own metadata settings (Int_Meta) or to use the value of setting Master_Output to be used as source for the dialogue level.
#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
#Bitstrm	Bitstream describes the audio service contained within the Dolby Digital. A complete audio program may consist of a main audio service (a complete mix of all program audio), an associated audio service comprising a complete mix, or one main service combined with an associated service. To form a complete audio program, it may be (but rarely is) necessary to decode both main service and an associated service using a maximum total bit rate of 512 kbps, Refer to the guide to use of the ATSC digital television standard, documentA/54 for further information. Although a detailed descriptions follows.

Bitsteam	Description
`Complete`	CM flags the bitstream as the Main Audio service for the program and all elements are present to form a complete audio program. Currently, this is the most common setting. The service may contain one (mono) to six (5.1) channels.
`M&E`	The bitstream is the main audio service for the program, minus a dialogue channel. The dialogue channel, if any, is intended to be carried by an associated dialogue service. Different dialogue services can be associated with a single ME service to support multiple channels.
`Visual`	This is typically a single channel program intended to provide a narrative description of the picture content to be decoded along with the main audio service. The visual service may also be a complete mix of all program channels, comprising up to six channels.
`Hearing`	This is typically a single channel program intended to convey audio that has been processed for increased intelligibility and decode along with the main audio service. The Hearing service may also be a complete mix of all program channels.
`Dialogue`	This is typically a single program intended to provide a dialogue channel for a Main service. If the main service contains more than two channels, the dialogue is limited to only one channel. If the ME service is two channels, the Dialogue can be a stereo pair: the appreciate channels of each service are mixed tighter (requires special decoders)
`Commentary`	This is typically a single channel program intended to convey additional commentary that can be optionally decoded along with the main audio service. This service differs from dialogue services because it contains an optional, rather than required, dialogue channel. The service may also be complete mix of all program channels, comprising up to six channels.
`Emergency`	This is a single channel service that is given priority in reproduction. When the E-service appears in the bitstream, it is given priority in the decoder and the main service is muted.
`VO_Karaoke`	This is a single channel service intended to be decoded and mixed to the center channel. (requires special decoders)
`Ext_meta`	Use the Bitstream metadata settings from an external program.

#Ch-Mode

This parameter instructs the encoder as to which inputs to use for this particular program: it tells the decoder what channels are present in this program so the decoder can deliver the audio to the correct speakers.

The setting is described as X/Y, where X is the number of front channels (left, Center, Right) and Y the number of rear (surround) channels.

Channel mode setting	Description
`1/0 ( C )`	Centre
`2/0 (LR)`	Left, Right
`3/0 (LCR)`	Left, Centre, Right
`2/1 (LRS)`	Left Right Surround
`3/1 (LCRS)`	Left Center Right Surround
`2/2 (LRS1Sr)`	Left Right Surround_Left Surround_right
`3/2 (LCRS1Sr)`	Left Center Right Surround_Left Surround_right
`Ext_meta`	Use the Channel mode metadata setting of the external program (`Ext_meta`).

Default is `3/2 (LCRSlSr)`

#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`.

#Line

`Line` sets the Dynamic range metadata of presets.

- `NONE`, no dynamic range compression is applied unless downmixing could cause overload, in which case protection dynamic range is automatically applied.
- `Film stnd`, Applies more compression to a subjectively loud film that requires dynamic range restriction.
- `Film Light`, Applies light compression to a subjectively quiet film that does not require dynamic range restriction.
- `Music Stnd`, Applies more compression to music that is not compressed and requires dynamic range restriction.
- `Music light`, Applies light compression to music that is already compressed and does not require excessive dynamic range restriction.
- `Speech`, Appropriate for programs with predominantly dialogue.

You can also choose to use the metadata settings in the external program (`Ext_meta`). Default is `None`.

#RfMode

`RfMode` has the same options as `Line`, but each option is `11 dB` more sensitive to avoid overloading the RF input of a television. `None`, `Film stnd`, `Film light`, `Music stnd`, `Music light` and `speech`. You can also choose to use the metadata settings in the external program (`Ext_meta`). Default is `None`.

#D Srnd

`Dolby Surround`. Determines when a Dolby Digital decoding product also contains a Dolby Pro Logic decoder, whether the two-channel encoded bistream contains a Dolby Surround (Lt/Rt) program that requires Pro Logic decoding. Decoders can use this flag to automatically switch on Pro-logic decoding as required.

- `Not indic`, Not Indicated
- `Not Srnd`, Not Dolby surround; the bitstream contains information that was not Dolby Surround encoded.
- `Dolby Srnd`, Dolby Surround; the bitstream contains information that was Dolby Surround encoded. After Dolby Digital decoding, the bitstream is pro logic decoded.

You can also choose to use the metadata settings in the external program (`Ext_meta`). Default is `Not Srnd`.

#Pref dwnmx

Preferred Down mix. This parameter allows the user to select either Lt/Rt or the Lo/Ro downmix in a consumer decoder that has stereo outputs. Consumer receivers are able to override this selection, but this parameter provides the opportunity for a 5.1 channel soundtrack to play in `Lo/Ro` mode without user intervention. This is especially useful on music material. `NOT indicated`, `Lt/Rt` and `Lo/Ro` are the possible mix types. You can also choose to use the metadata settings in the external program (`Ext_meta`). Default is `Lt/Rt`.

#Lt/Rt C dwnmx

Lr/Rt Center Mix Level. This setting indicates the level shift applied to the center channel when adding to the left and right outputs when downmixing to an Lt/rt output. Its operation is similar to the surround downmix level in the Universal metadata. `0dB`, `-1.5dB`, `-3.0dB`, `-4.5dB`, `-6.0dB` and `–999dB`. You can also choose to use the metadata settings in the external program (`Ext_meta`). Default is `–3dB`.

#Lt/Rt S dwnmx

LtRt Surround Mix level. This setting indicates the level shift applied to the surround channels when downmixing to an Lt/Rt output. Its operation is similar to the surround downmix level in the universal metadata. `-1.5dB`, `-3.0dB`, `-4.5dB`, `-6.0dB` and `-999dB.` You can also choose to use the metadata settings in the external program (`Ext_meta`). Default is `–3dB`.

#SF_AutoLoRo

This enables the auto generation of downmix parameters for the metadata based on the upmix parameters.

#Lo/Ro C dwnmx

Lo/Ro Center mix level. This setting indicates the level shift applied to the center channel when adding to the left and right outputs when downmixing to a Lo/Ro output. When Extended BSI parameters are active, this parameter is used and the Center Mix Level parameter in the universal parameters is not.+3dB, +1.5dB, 0dB, -1.5dB, -3.0dB, -4.5dB, -6.0dB and -999dB. You can also choose to use the metadata settings in the external program (Ext_meta). Default is –3dB. This setting will automatically change the C_downmx metadata setting (which can not be set separately anymore) according to the following table:

Lo/Ro C dwnmx:	Sets C_downmx automatically to:
+3.0dB	-3.0dB
+1.5dB	-3.0dB
0.0dB	-3.0dB
-1.5dB	-3.0dB
-3.0dB	-3.0dB
-4.5dB	-4.5dB
-6.0dB	-6.0dB

#Lo/Ro S dwnmx

Lo/Ro Surround Mix level. This setting indicates the level shift applied to the surround channels when downmixing to a Lo/Ro output. When extended BSI parameters are active, this parameter is used, and the surround mix level parameter in the universal parameters is not. -1.5dB, -3.0dB, -4.5dB, -6.0dB and -999dB. You can also choose to use the metadata settings in the external program (Ext_meta). Default is –3dB.

This setting will automatically change the S_downmx metadata setting (which can not be set separately anymore) according to the following table:

Lo/Ro S dwnmx:	Sets S_downmx automatically to:
-1.5dB	-3.0dB
-3.0dB	-3.0dB
-4.5dB	-6.0dB
-6.0dB	-6.0dB
-999dB	-999dB

#Dolby Srnd EX

Surround EX. This setting is used to identify the encoded audio as surround EX encoded material. This parameter is only used if the encoded audio has two surround channels. An amplifier or receiver with Dolby Digital EX decoding can use this parameter as a flag to switch the decoding on or off automatically. The behavior is similar to the Dolby Surround Mode parameter. `Not Indic.`, `NotDolbySrnd`, `DolbySrnd`. You can also choose to use the metadata settings in the external program (`Ext_meta`). Default is `Not Srnd`.

#DC filter

DC filter. This setting determines whether a DC blocking 3Hz highpass filter is applied to the main inputs channels of a Dolby Digital encoder prior encoding. This parameter is not carried to the consumer decoder. It is used to remove DC offsets in the program audio and would only be switched off in exceptional circumstances. `On` this function is active, `OFF` this function is not active. You can also choose to use the metadata settings in the external program (`Ext_meta`). Default is `ON`.

#LFE filter

LFE lowpass filter. This setting determines whether a 120hZ 8 order lowpass filter is applied to the LFEE channel input of a Dolby Digital encoder prior to encoding. It is ignored if the LFE channel is disabled. This parameter is not sent to the consumer decoder. The filter removes frequencies above 120Hz that would aliasing when decoded. This filter should only be switched off if the audio to be encoded is known to have no signal above 120 Hz. `On` this function is active, `OFF` this function is not active. You can also choose to use the metadata settings in the external program (`Ext_meta`). Default is `ON`.

#Lowpass Filter

Lowpass Filter. This setting determines whether a lowpass filter is applied to the main input channels of a Dolby Digital encoder to encode. This filter removes high frequent signals that are not encoded. At the suitable data rates this filter operates above 20 kHz. In all cases it prevents aliasing on decoding and is normally switched on. This parameter is not passed to the consumer decoder. `On` this function is active, `OFF` this function is not active. You can also choose to use the metadata settings in the external program (`Ext_meta`). Default is `ON`.

#Srnd 3dB atten

Surround 3dB attenuation. This setting determines whether the surround channels are attenuated 3 dB before encoding. The attenuation actually takes place inside the Dolby Digital encoder. It balances the signals levels between theatrical mixing rooms (dubbing stages) and consumer mixing rooms (dvd or tv studios) Consumer mixing rooms are calibrated so that all five main channels are at the same sound pressure level (SPL). For compatibility reasons with older film formats, theatrical mixing rooms calibrate the surround channels 3dB lower in SPL that the front channels. The consequence is that signal levels on tape are 3dB louder. Therefore, to convert to a consumer mix from theatrical calibration it is necessary to reduce the surround levels by 3dB. `On` = this function is active, `OFF =` this function is not active. You can also choose to use the metadata settings in the external program (`Ext_meta`). Default is `OFF`

#Srnd Ph Shift

Surround Phase Shift. This setting takes care that the Dolby Digital encoder applies a 90-degree phase shift to the surround channels. This allows a Dolby Digital decoder create an Lt/Rt downmix simply. For most material the phase shift has a minimal impact when the Dolby Digital program I decoded to 5.1 channels, but provides an Lt/Rt output that can be Prologic decoded to L, C, R ,S if desired. However, for some phase-critical material (such as music) this phase shift is audible when listening in 5,1 channels. Likewise some material downmixes to a satisfactory Lt/Rt signal without needing this phase shift. It is therefore important to balance the needs of the 5.1 mix and the Lt/Rt downmix for each program. `On` this function is active, `OFF` this function is not active. You can also choose to use the metadata settings in the external program (`Ext_meta`). Default is `ON`.

Gain_Chn_Lf_1 ~ Gain_Chn_R_1

These items allow you to gain the audio for each individual channel of Soundfield processor 1 in a range from `–60dB` to `12 dB` in steps of `0.25 dB`. `–999dB` mutes this channel. Default is `0dB`.

Gain_Chn_Lf_2 ~ Gain_Chn_R_2

These items allow you to gain the audio for each individual channel of Soundfield processor 2 in a range from `–60dB` to `12 dB` in steps of `0.25 dB`. `–999dB` mutes this channel. Default is `0dB`.

Delay_Chn_Lf_1 ~ Delay_Chn_R_1

These settings allow you to delay the audio of each individual channel of Soundfield processor 1 in a range of `0` to `4000` ms. In steps of `0.01 ms`. Default is `0ms`.

Delay_Chn_Lf_2 ~ Delay_Chn_R_2

These settings allow you to delay the audio of each individual channel of Soundfield processor 2 in a range of `0` to `4000` ms. In steps of `0.01 ms`. Default is `0ms`.

QUADSPEED_OUTPUT

Override 17/24

If you want to pass processed audio from one quad speed add-on card to the other (for instance if you want to pass processed audio from this card to for instance a Loudness Control add-on card next to this card) you have to use this setting. You can choose to override input channels 17/24 on the add-on bus of the next card (right side) with output channels 1 to 8 or pass the master-card audio.

Override 25/32

With this setting you can choose whether you want to override input channels 25/32 on the add-on bus of the next add-on card (right side) with output channels 9 to 16 or pass the master-card audio.

OUT BUS CONTROL

Slot1/2 ~ Slot31/32

These menu items are to fill the Quad speed audio bus with the appropriate outputs. You can fill any of the 16 audio pairs (32 channels in total) with the following audio pair sources:

- `LfRf1` - Left front and right front channels of processor 1.
- `CLFE1` - Center and LFE channels of processor 1.
- `LSRs1` - Left surr. and right surr. channels of processor 1.
- `LR1` – Stereo output channels of processor 1.
- `LfRf2` - Left front and right front channels of processor 2.
- `CLFE2` - Center and LFE channels of processor 2.
- `LSRs2` - Left surr. and right surr. channels of processor 2.
- `LR2` – Stereo output channels of processor 2.
- `AES1/2` – Physical AES input 1
- `AES3/4` – Physical AES input 2
- `AES5/6` – Physical AES input 3
- `AES7/8` – Physical AES input 4
- `AES9/10` – Physical AES input 5 (only in `8in` mode)
- `AES11/12` – Physical AES input 6 (only in `8in` mode)
- `AES13/14` – Physical AES input 7 (only in `8in` mode)
- `AES15/16` – Physical AES input 8 (only in `8in` mode)

You can also switch the corresponding pair to `off`, making that specific audio pair empty.

6 Status Menu

Introduction

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

Reference_Stat

This status item displays the status of the reference lock of the card. Can be `30Hz` or `25Hz` (when locked to ref1 or ref2), can be `Input-Lock` (locked to AES1) or can be `MasterCard` (locked to the ADD-ON inputs). If there's no lock, `NA` is displayed.

Temp_status

The card has its own temperature sensor. This item displays whether or not the temperature of the card is OK or not (`Error`).

GPI

This item displays which GPI is currently triggered. Can be #`1` till #`8`.

Input_Chn_Lf_1 ~ Input_Chn_R_1

Displays the status of each individual input of Soundfield processor 1. Can be `OK`, `NA` (not available), `Silence` (audio silence is detected) or `Clipped` (full scale is detected).

Frmt_Chn_Lf/Rf1 ~ Frmt_Chn_L/R1

Displays the format of each corresponding audio pair of the Soundfield processor 1 inputs. Can be one of the following:

- `Unlocked`
- `PCM`
- `Null`
- `AC-3`
- `TimeStmp`
- `MPEG-1`
- `MPEG-2`
- `SMPTE-KLV`
- `Dolby E`
- `Caption data`
- `User Def`
- `Reserved`

Input_Chn_Lf_2 ~ Input_Chn_R_2

Displays the status of each individual input of Soundfield processor 2. Can be `OK`, `NA` (not available), `Silence` (audio silence is detected) or `Clipped` (full scale is detected).

Frmt_Chn_Lf/Rf1 ~ Frmt_Chn_L/R1

Displays the format of each corresponding audio pair of the Soundfield processor 1 inputs. Possible detectable formats are listed under `Frmt_Chn_Lf/Rf1`.

Aes_Input_0102 ~ Aes_Input_1516	If a valid signal is present on the corresponding input then `OK` is indicated. `NA` indicates that no AES signal is present.
Eng1_Mode_Stat	With this status item you can see in what mode Soundfield processor 1 is running. Can be: ▪ `Upmix` ▪ `Downmix` ▪ `Upmix with Auto Xfade` ▪ `Downmix with Auto Xfade` ▪ `Up and Downmix with Auto Xfade` ▪ `Up and Downmix`
Upmix_Eng1	Displays the status of the Soundfield processor 1 Upmixer. Can be `on` or `off`.
Downmix_Eng1	Displays the status of the Soundfield processor 1 Downmixer. Can be `on` or `off`.
Eng1_Mode_Stat	With this status item you can see in what mode Soundfield processor 2 is running. Can be: ▪ `Upmix` ▪ `Downmix` ▪ `Upmix with Auto Xfade` ▪ `Downmix with Auto Xfade` ▪ `Up and Downmix with Auto Xfade` ▪ `Up and Downmix`
Upmix_Eng1	Displays the status of the Soundfield processor 2 Upmixer. Can be `on` or `off`.
Downmix_Eng1	Displays the status of the Soundfield processor 2 Downmixer. Can be `on` or `off`.
Ext_Meta	Displays the status of the external metadata source. Can be `OK` or `NA`.

The source of the following status items (preceded with '`MD`' prefix) is dependent on the `MD_Status_Src` setting.

MD_Prgm_Config	This status indicates the program config as present on the metadata preset. Can be one of the following values: ■ `5.1+2` ■ `5.1+1+1` ■ `4+4` ■ `4x2` ■ `8x1` ■ `5.1` ■ `3x2` ■ `6x1` ■ `2+2` ■ `7.1` ■ `Other` ■ `NA`
MD FrameRate	Indicates the value of the frame rate metadata parameter.
MD Dialog Lvl	Indicates the value of the dialogue level 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 LFE	Indicates the value of the LFE channel metadata parameter.
MD Line Mode	Indicates the value of the line mode metadata parameter
MD RF Mode	Indicates the value of the RF mode metadata parameter.
MD D_Surnd	Indicates the value of the Dolby surround 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 DC Filter	Indicates the value of the DC filter metadata parameter.
MD LFE Filter	Indicates the value of the LFE filter metsdata parameter.
MD Lowpass Fil	Indicates the value of the Low pass filter metadata parameter.
MD Sur3d Att	Indicates the value of the surround 3dB attenuate metadata.
MD Sur PhShift	Indicates the value of the surround phase shift metadata.

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.

DSF66 Events

The events reported by the DSF66 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

Reference

`Reference` can be selected between 0 .. 255. 0= no event, 1..255 are the priority setting. If the reference is lost an Event will be generated at the priority.

Temprature

`Reference` can be selected between 0 .. 255. 0= no event, 1..255 are the priority setting. If a temperature error is detected an Event will be generated at the priority.

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 DSF66 are:

Event Menu Item	Tag		Description
Announcements	0 or NA	0 or NA	Announcing of report and control values
Reference-Status	02_{hex}=REF_LOSS	82_{hex}=REF_RETURN	reference lost or returned
Temprature	$0a_{hex}$=TEMP_TOO_HIGH	$8a_{hex}$=TEMP_OK	Temprature error or OK

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/RRS 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

The error LED indicates an error if the internal logic of the DSF66 card is not configured correctly or has a hardware failure.

INPUT 1 ~ INPUT 8

These LEDs indicated the presence of a valid AES/EBU signal on the inputs 1 till 5.

REFERENCE

This LED indicates the presence of a valid reference signal and that the DSF66 is locked to the master card.

DATA ERROR

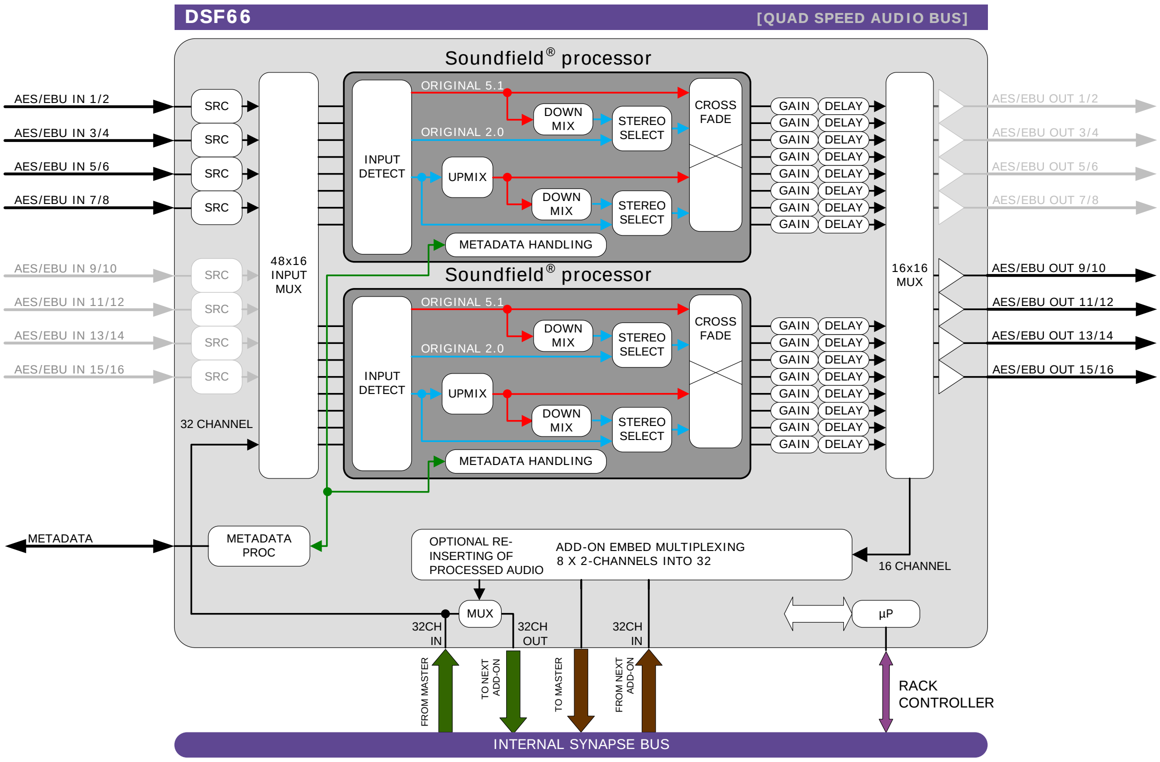
This led indicates different types of errors if there is an error in the Dolby encoding, or the audio carrier is falling away, or audio data is in the range of 0 dBFS and -1 dBFS.

CONNECTION

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

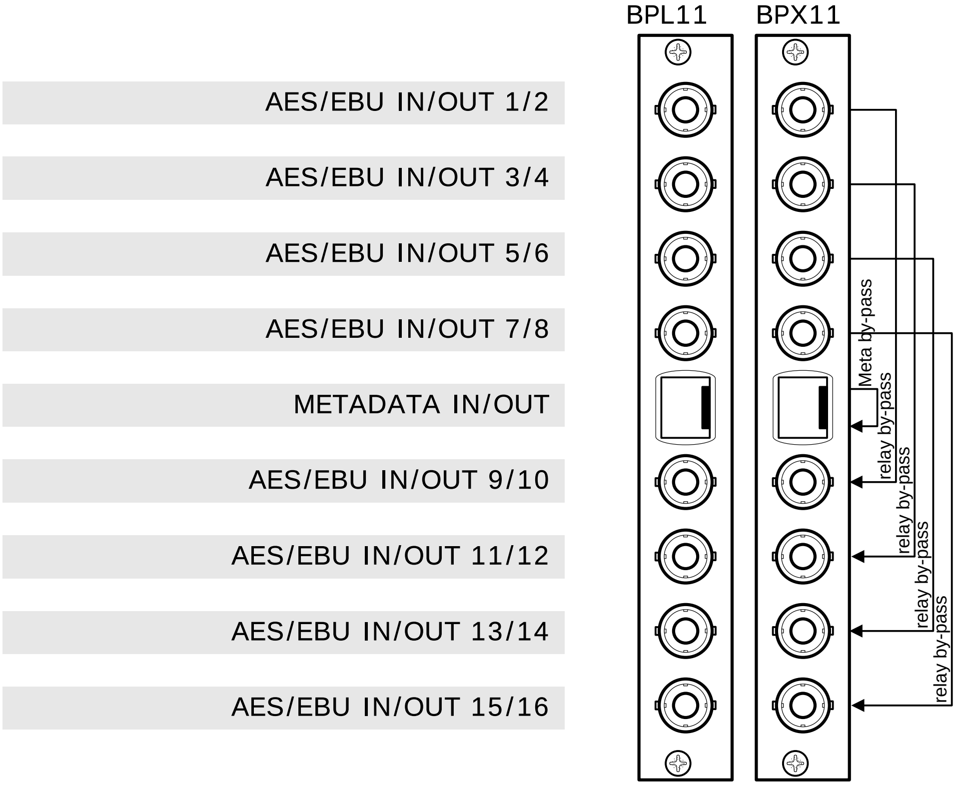
Note: When LEDS are blinking constantly, the card is still programming.

9 Block Schematics



10 Connector Panel

The DSF66 can be used with the BPL11 and BPX11 back planes:



Appendix 1 Soundfield Recommended settings

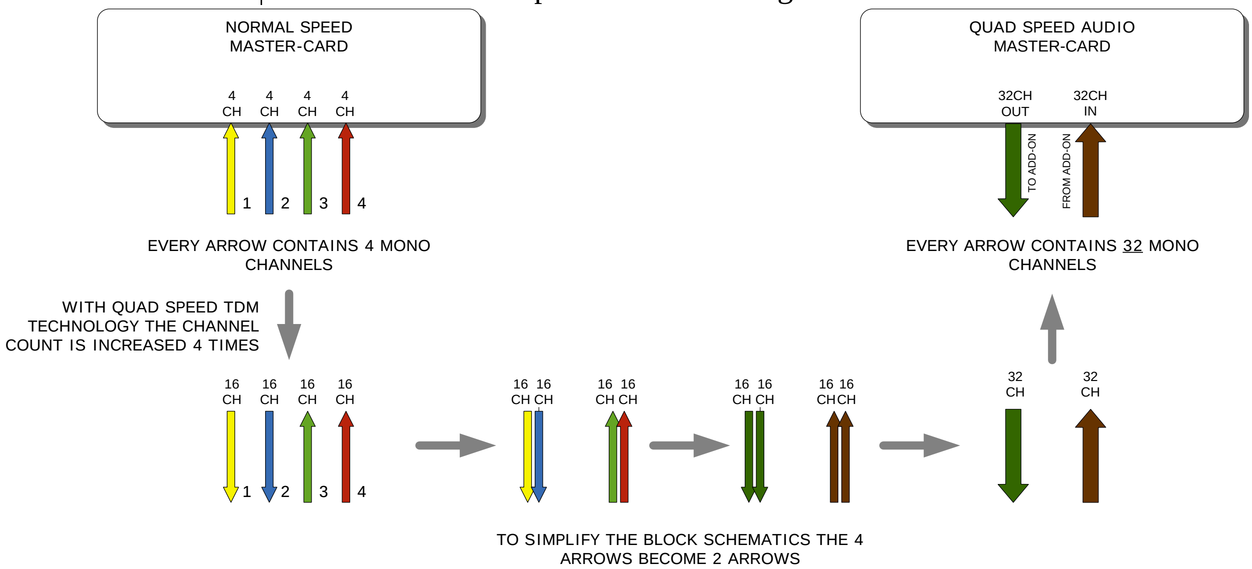
The Soundfield upmixer will work with the default settings across a wide range of material. But below you will find a few “Presets”. In order to keep it simple, the examples only tweak the 5 main parameters and keep everything else set to default.

1. News:
    a. Direct Sound – 0
    b. Front Ambient Sound - 0
    c. Rear Ambient Sound - 0
    d. Width - 0
    e. Centre Divergence – 0.25
2. Sports:
    a. Direct Sound - 0
    b. Front Ambient Sound - 0
    c. Rear Ambient Sound - +2
    d. Width – 0.25
    e. Centre Divergence – 0.00
3. Movies:
    a. Direct Sound - 0
    b. Front Ambient Sound - 0
    c. Rear Ambient Sound - +2
    d. Width - 0
    e. Centre Divergence - 0
4. Pop Music:
    a. Direct Sound - 0
    b. Front Ambient Sound - 0
    c. Rear Ambient Sound - 0
    d. Width – 0.40
    e. Centre Divergence – 0.00
5. Classical Music:
    a. Direct Sound - 0
    b. Front Ambient Sound - 0
    c. Rear Ambient Sound - 0
    d. Width - 0
    e. Centre Divergence – 0.25

Appendix 2 Quad speed ADD-ON bus

The internal audio ADD-ON bus needed an upgrade for some applications. We wanted more channels (32 per video stream seem possible in the near future). And we want the bus to be bidirectional, so 32 channels in and 32 channels out at the same time.

The new interface needed to be compatible with all existing hardware (frames) and in the implementation of the master card it sometimes needed to be backward compatible with the original ADD-ON bus.



So the MASTER-CARD is now firmware enhanced to run 32 channels in either direction (64 channels total) instead of 16 channels in one direction

Some MASTER-CARD's will have two modes and some MASTER-CARD's will only have the Quad Speed mode [where the logical ADD-ON cards are only available in Quad Speed mode:

Dual mode MASTER-CARD's have a menu item to select the appropriate mode are. If a mode is selected all ADD-ON cards to that Master need to be in the same mode.

The following features and rules will apply:

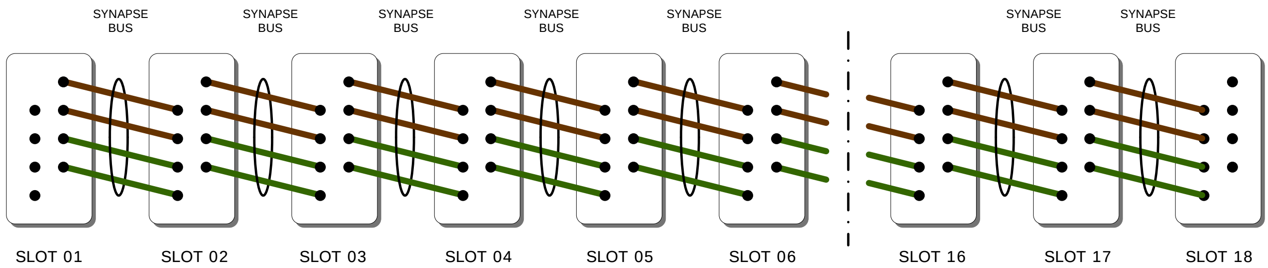
- Up to 32 channels output from the master card with looping to up to 3 ADD-ON cards
  - The ADD-ON card just picks the channels it wants to process
  - Some ADD-ON cards will have the possibility to re-inject processed audio onto the next ADD-ON card
- Up to 32 channels input on the master card
  - If the master card can handle less than 32 channels, the lowest channel numbers will be used, as the ADD-ON card will always generate 32 channels (where some channels can be empty or silent)
- Channel shuffling is done in the ADD-ON card
  - The Master Card has only one setting to enable the quad speed audio bus

- Every Quad-Speed ADD-ON card takes 32 channels from the ‘right hand ADD-ON card’ and adds (or overwrites) the local processed channels.
    - This can be done for any of the channels that are processed in the ADD-ON card
- Some Master Cards are switchable between normal and quad-speed bus
- Channel designations on the block schematics:
    - Channel 1-32 (or less) are injected into the dark green large arrow from Master Card to ADD-ON card and looped on to the next ADD-ON card via the dark green arrow
    - The ADD-ON card injects up to 32 channels into the brown large arrow
    - An ADD-ON card will also actively loop extra processed channels into the next ADD-ON card, and finally into the Master Card
- The cross looping of the original design is now a straight loop
- The quad speed bus can also work in one direction
    - You can use a Quad Speed audio bus to de-embed audio from the master and present on the ADD-ON card as AES/EBU, Bitstream (like Dolby) or analog audio

If applicable the ADD-ON card can also be used as in injection point of physical audio streams

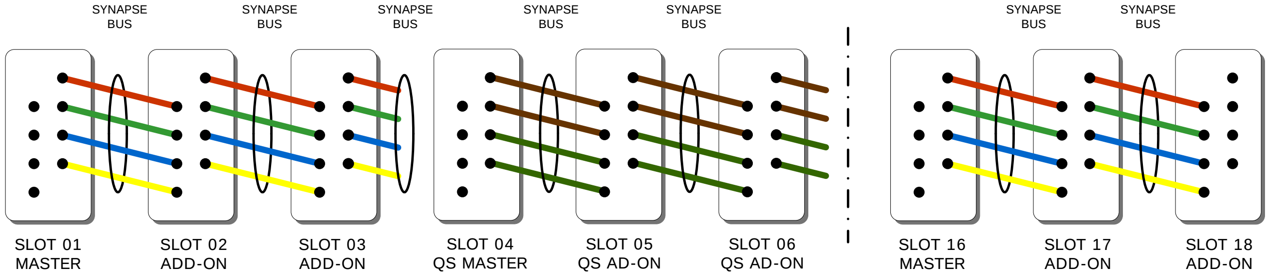
The ADD-ON cards also provide a looping function from one ADD-ON to the next ADD-ON card. This is however a more intelligent looping with optional re-insertion and multiplexing of signals.

Cascading of Quad Speed cards works identical to normal add-on cards. Every connection in the example below transports 16 mono audio channels (= 32 channels per color). It shows the inter slot connections ‘in quad Speed mode’ as part of the frame bus PCB.



The system makes use of the same passive copper traces on the internal bus PCB as normal add-on bus cards.

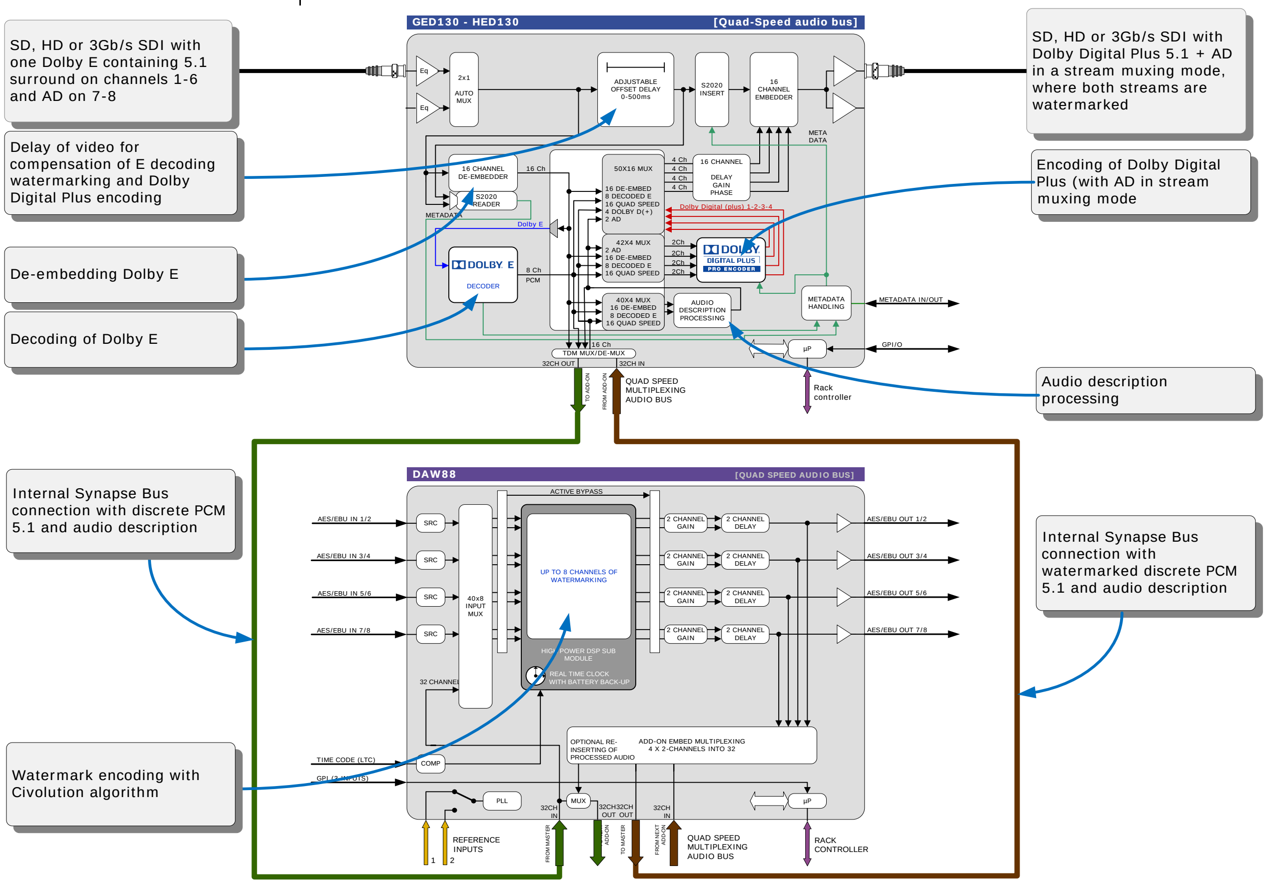
The maximum amount of ADD-ON cards in Quad Speed mode is 3. These 3 ADD-ON cards will run all on the same clock in the same phase as the MASTER-CARD. This guarantees that audio channels that are processed in different ADD-ON cards will still operate in the same phase, something very important when processing multiple discrete surround channels.
You can mix normal speed Master-Cards with Quad Speed MASTER-Cards in one frame as the MASTER-CARD breaks the connection to the left hand card. All cards to the right of the master must be in the same mode as the master.



Mixing normal ADD-ON with Quad Speed ADD-ON combo's in one frame is allowed

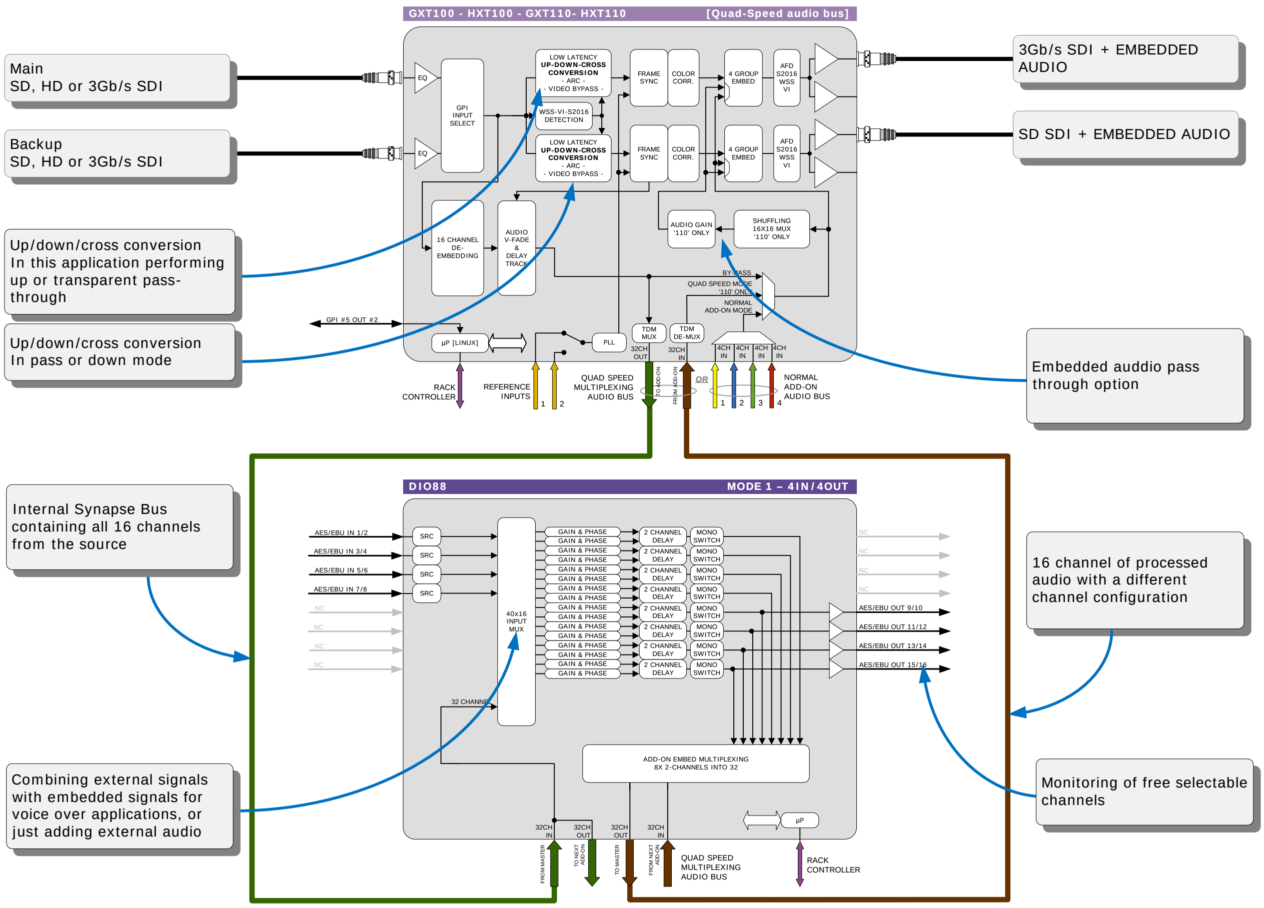
Some examples

This is an Example where we combine a MASTER-CARD that performs embedded domain Dolby E to Dolby Digital Plus encoding. Between the E-decoding and Dolby Digital Plus encoding we want to watermark the left, right and center channel of a the decoded discrete 5.1 surround channels and watermark a PCM channel used as a voice over for audio description.



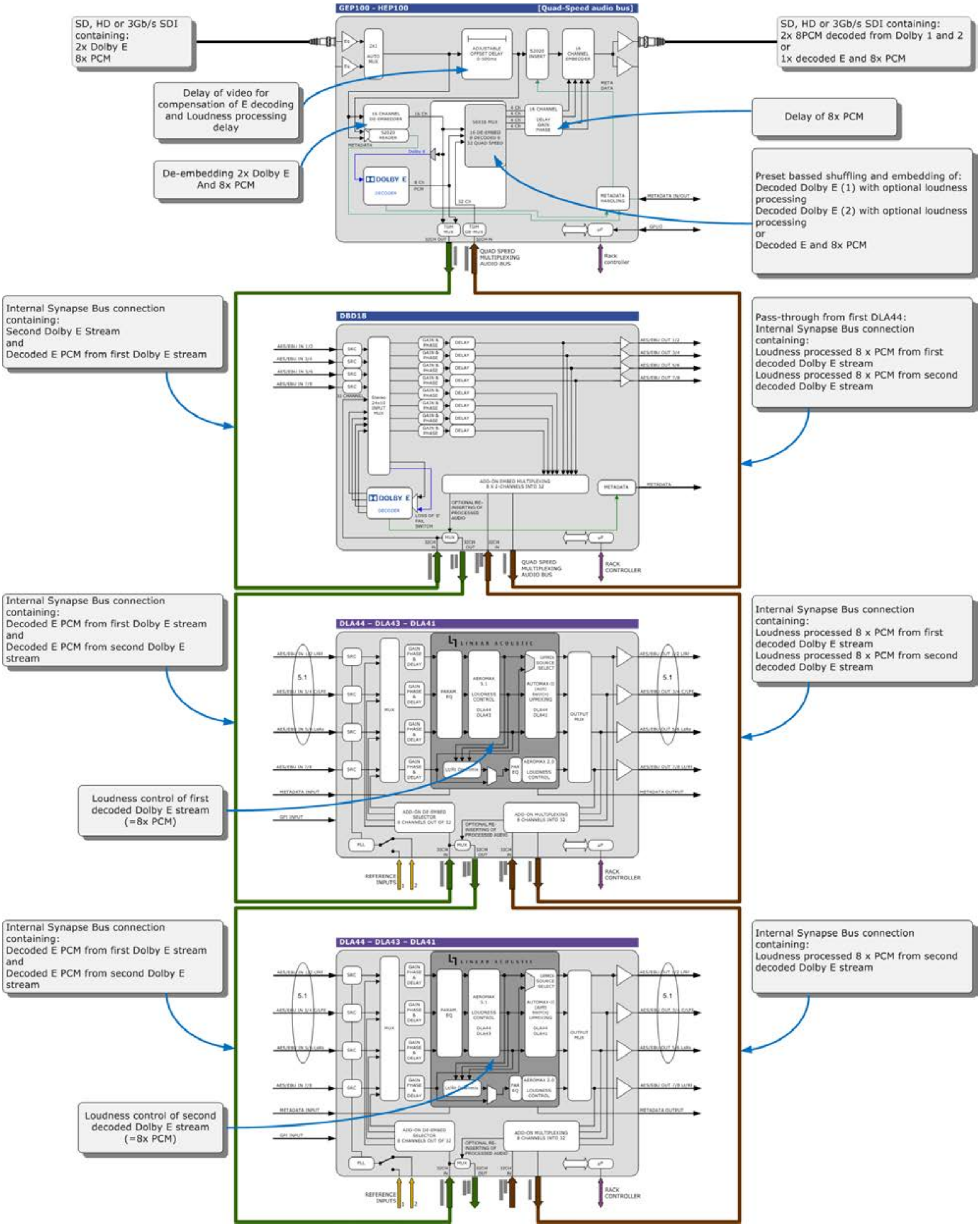
Embedded domain Dolby E to Dolby Digital Plus with Watermarking. The only connection to the outside world are two BNC cables.

Another example of the Quad-Speed audio ADD-ON bus shows a transmission application where a dual up/down/cross output card is connected to a DIO88 in a setup where the embedded audio combined with external audio and a convenient PCM monitoring is available.



In the following example (next page) you will see a 4 card application that performs a massive amount of processing divided over 1 MASTER-CARD and 3 ADD-ON cards. This is a typical ‘ingest’ configuration and is used where the infrastructure does not use Dolby E (two in this example) but PCM+s2020. The input is a SD, HD or 3Gb/s SDI containing 2 Dolby E streams and 8 mono PCM streams. The output is the same SDI stream but with a selection of 16 channels selected out 8 original PCM channels and 16 PCM channels that are decoded from the Dolby E streams. The combo performs the following processing:

- De-embedding of 8x PCM and 2x Dolby E
- Decoding of two independent Dolby E streams
- Loudness processing of up to 16 channels sourced by any of the 8x PCM or decoded Dolby E streams
- Upmixing of a 2.0 to 5.1 if a Dolby E stream is not available
- Physical monitoring of all processed PCM streams
- Preset based shuffling of all source channels into 16 channels with the appropriate offset delays
- S2020 metadata insertion sourced from the E decoders, embedded s2020, generated presets or an external feed
- Video delay to compensate for audio propagation delay
- Embedding of up to 16 channels


GEP100 - HEP100
[Quad-Speed audio bus]
SD, HD or 3Gb/s SDI containing:
2x Dolby E
8x PCM
SD, HD or 3Gb/s SDI containing:
2x 8PCM decoded from Dolby 1 and 2
or
1x decoded E and 8x PCM
Delay of video for compensation of E decoding and Loudness processing delay
Delay of 8x PCM
De-embedding 2x Dolby E And 8x PCM
Preset bassed shuffling and embedding of:
Decoded Dolby E (1) with optional loudness processing
Decoded Dolby E (2) with optional loudness processing
or
Decoded E and 8x PCM
DBD18
Internal Synapse Bus connection containing:
Second Dolby E Stream
and
Decoded E PCM from first Dolby E stream
Pass-through from first DLA44:
Internal Synapse Bus connection containing:
Loudness processed 8 x PCM from first decoded Dolby E stream
Loudness processed 8 x PCM from second decoded Dolby E stream
DLA44 - DLA43 - DLA41
Internal Synapse Bus connection containing:
Decoded E PCM from first Dolby E stream
and
Decoded E PCM from second Dolby E stream
Internal Synapse Bus connection containing:
Loudness processed 8 x PCM from first decoded Dolby E stream
Loudness processed 8 x PCM from second decoded Dolby E stream
Loudness control of first decoded Dolby E stream
(=8x PCM)
DLA44 - DLA43 - DLA41
Internal Synapse Bus connection containing:
Decoded E PCM from first Dolby E stream
and
Decoded E PCM from second Dolby E stream
Internal Synapse Bus connection containing:
Loudness processed 8 x PCM from second decoded Dolby E stream
Loudness control of second decoded Dolby E stream
(=8x PCM)