Synapse

DLA44/43/41

8-channel (5.1/2.0) digital audio loudness control and upmixer/downmixer unit based on Linear Acoustic algorithms.

Installation and Operation manual

LINEAR ACOUSTIC

Committed.

AXON

TECHNICAL MANUAL

DLA44/43/41

8-channel (5.1/2.0) digital audio loudness control and upmixer/downmixer unit based on Linear Acoustic algorithms.

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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.

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Date created: 30-09-2009
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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
DLA44/43/41

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 RRC18, RRC10, RRC04, RRS18 and RRS04 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 (RRC18/RRC10/RRC04/RRS18/RRS04) manual. The method of connection to a computer using Ethernet is described in those manuals as well.

> **!** **CHECK-OUT:** "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 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

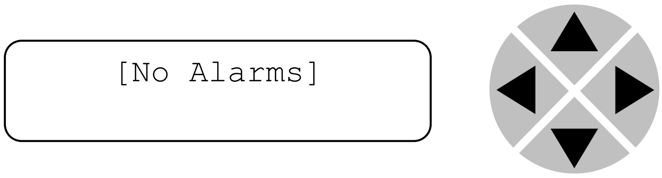
On powering up the Synapse frame, the card set will use basic data and default initialization settings. All LEDs will light during this process. After initialization, several LEDs will remain lit – the exact number and configuration is dependant upon the number of inputs connected and the status of the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

Changing settings and parameters

The front panel controls or 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.

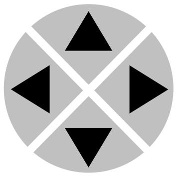
Press ►	T go forward through the menu structure.
Press ◄	To go back through the menu structure.
Press ▲	To move up within a menu or increase the value of a parameter.
Press ▼	To move down through a menu or decrease the value of a parameter.

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

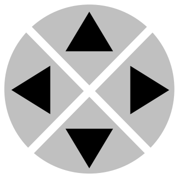
```
RRC18 [Select Card]
>S01=SFS10
```



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
```

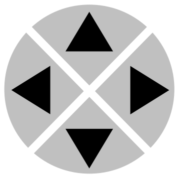


Pressing the ► selects the menu item shown, in this example `Settings`.

(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
```

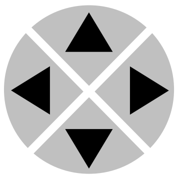


Pressing the ► selects the settings item shown, in this example `SDI-Format`.

(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.

Cortex

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. 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 Cortex, please refer to the 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 Cortex can be obtained from the rack controller manual and Cortex help files.

4 The DLA44/43/41 Card

Introduction

Inconsistent DTV audio loudness, or the so-called "loud commercial problem" is the number one complaint of television viewers and it is driving them away. It is clear that with the transition to digital (HD), a simple and cost effective solution is needed right now!

The DLA44/43/41 is based on third generation audio and loudness management technology by Linear Acoustic. The DLA protects your viewers from loudness shifts and loss of surround sound in a simple, cost effective, modular and hot-swap manner next to over 150 other Synapse modules.

The DLA accepts three pairs of PCM audio to handle a program stream containing 5.1 and two channel audio. The unit can apply multiband, multistage loudness control and upmixing to the applied audio. Loudness control is provided by the popular AEROMAX algorithm, while upmixing is provided by the air-proven and industry standard UPMAX algorithm. Both technologies are used world-wide to provide consistent and compelling 5.1 channel audio while remaining completely downmix compatible.

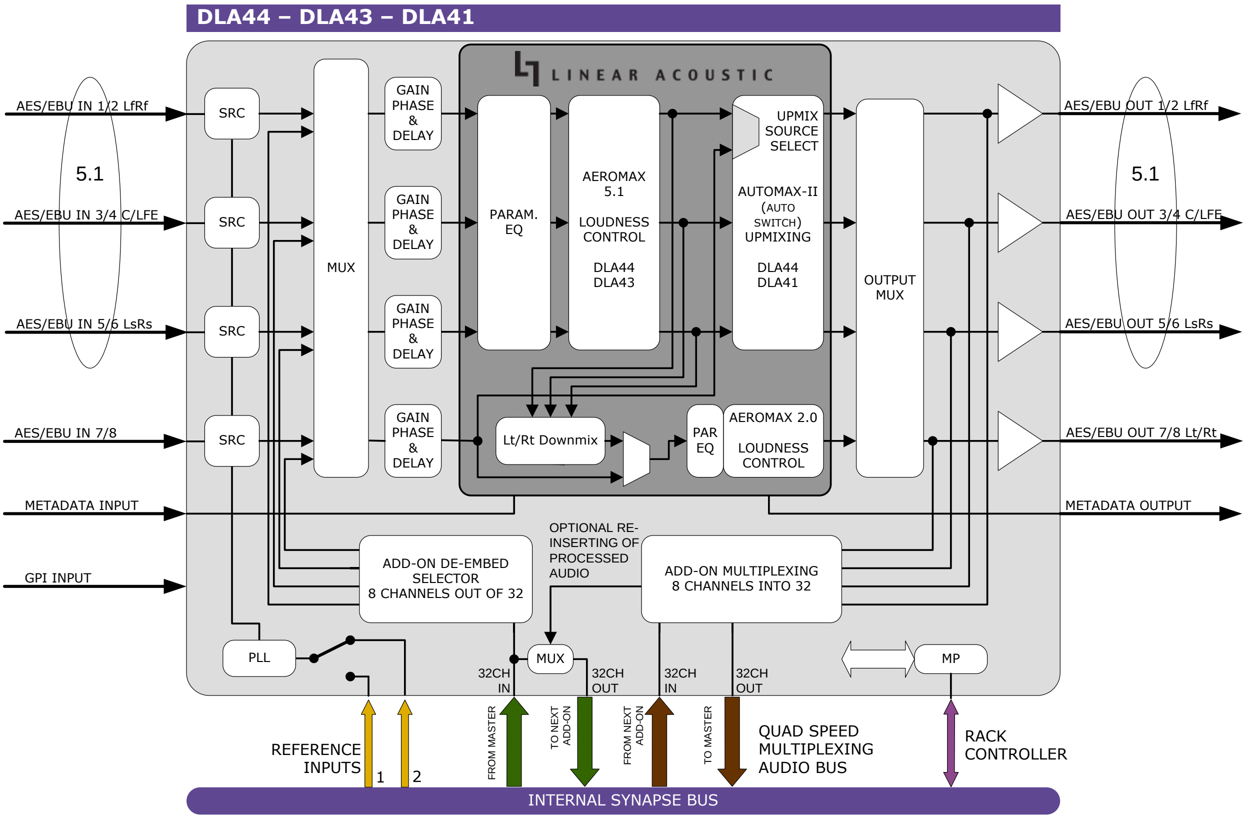
The DLA also includes the new AutoMAX-II auto-detection algorithm to smoothly and automatically bypass upmixing when applied content is 5.1 channels. The AutoMAX-II algorithm prevents any loss of dialogue or cause of switching artifacts. Upmixing and loudness processing modes can also be controlled the ACP protocol and Cortex or by GPI contact closures.

A full-time downmixed version of the main program is provided as the fourth AES output pair. This signal can be either a stereo LoRo downmix or an industry standard LtRt surround encoded mix compatible with all legacy consumer decoders.

- Quad speed ADD-On bus
- Input gain, phase and delay adjustments
- Parametric EQ for the 5.1 input and 2.0 sources
- 2.0 to 5.1 upmixing (DLA44 and DLA41 only)
- Downmix from 5.1
- 5.1 loudness control (DLA44/43 only)
- 2.0 loudness control of discrete or downmixed 2.0 input (DLA44/43 only)
- Metadata manipulation of external source to preset levels (DialNorm)
- Locks to Black & Burst, AES input and Mastercard.

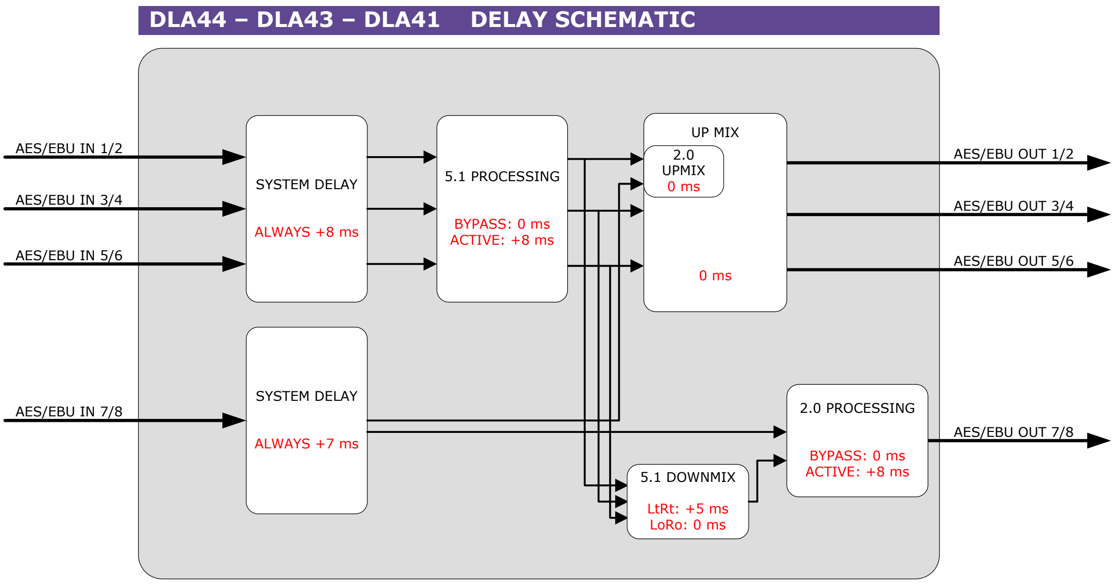
Full control and status monitoring through the front panel of the SFR04/SFR08/SFR18 frame and the Ethernet port (ACP)

Block Schematic



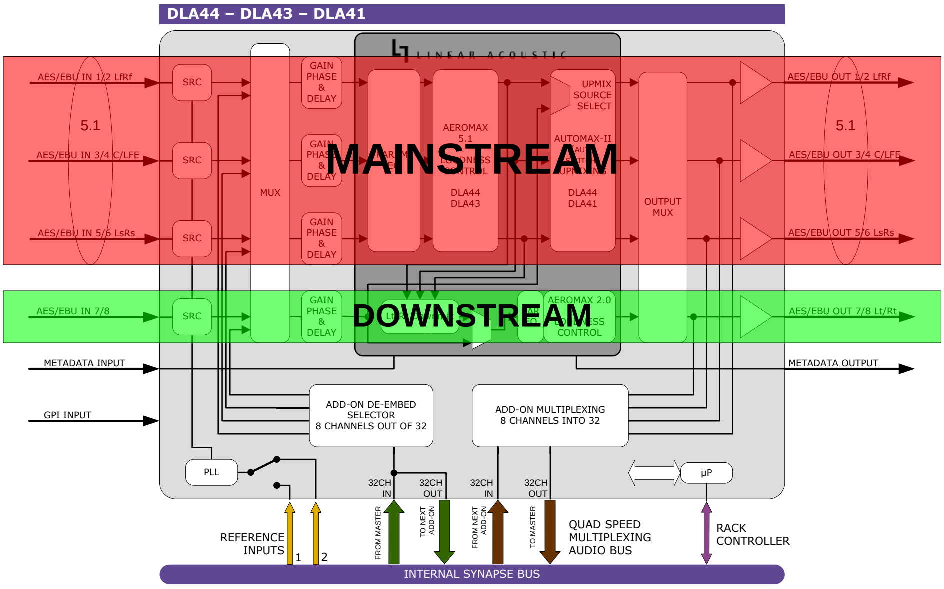
Delay Schematic

The processing of the various blocks in the DLA card cause delay. The following schematic shows what delay is caused by which blocks and settings:



Mainstream/ stereo stream distinction

In the settings (and the cortex interface) we distinguish 2 streams: the mainstream and the stereo stream (or downstream). The mainstream includes the 5.1 input/output. The stereo stream (or downstream) could contain the stereo stream for the downmix output. The following diagram shows these separated streams:



Program Procedure

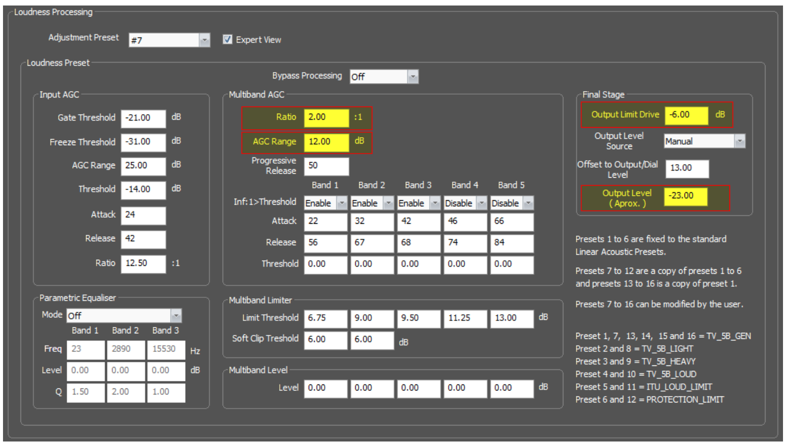
This card can be updated with new firmware when new firmware versions are released by Axon. You can download the .spf file from our website when new releases are announced. To upgrade the DLA44/43/41 you can follow the instructions as described in the 'reprogramming cards quick-guide', downloadable via our website.

EBU R128 configuration

To set up the DLA card to be conform the EBU R128 standard, set the `Preset_Actual` to one of the custom, non-fixed presets (7 till 14) and change the following settings to the given values:

- `#MB_Ratio` (under Multiband_Agc) set to 2:1
- `#MB_AGC_range` (under Multiband_Agc) set to 12.00dB
- `#Out_Lim_Drv` (under Final Stage) set to -6dB
- `#Master_out` (under Final Stage) set to -23dB

Using Cortex, please follow the following screenshot:



5 Settings Menu

Introduction

The settings menu displays the current state of each setting within the DLA44/43/41 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 Cortex.

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

SYSTEM CONTROL (DLA44/43/41)

LockMode

The DLA44/43/41 can be used as an ADD_ON card (in combination with an embedder/de-embedder card) In that case the card will extract the reference from the master card. In this case you are referred to the setting `MasterCard`. 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
`AES4` = Locks to the AES/EBU signal on input 4
`Reference` = The B&B reference input of the rackcontroller. With setting `Ref Source` you can choose between Ref input 1 or 2.
`Mastercard` = Locks to the ADD-ON bus input

Default setting is: `AES1`.

Ref Source

This settings has 2 functions. It specifies the reference input for the lock mechanism (when `LockMode` is set to `Reference`) and it enables Metadata generation if the plugged reference input is chosen.

SRC

This setting enables/disables the input Sample Rate Converter (SRC). When disabled (`Dolby-E`) the card is able to forward signals straight to the quad speed bus when they are of other formats than PCM.

Meters

The DLA44/43/41 features level meters that measure the audio level between -120dB and 0 dB. You can configure these meters to either measure the `input` of the Linear acoustic engine (so before the parameter EQ) or to measure the `output` of the linear acoustic engine (so before the upmix converter). Only the mainstream audio channels can be measured (see page 11). You can also decide to switch the meters `off` to avoid heavy ADD-ON bus data traffic.

Refreshrate

This item configures the meter refresh rate. Can be set between 25ms and 5000ms. To avoid heavy traffic on the bus, set it to a higher value.

GPI Control

The DLA44/43/41 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 (DLA44 only): 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

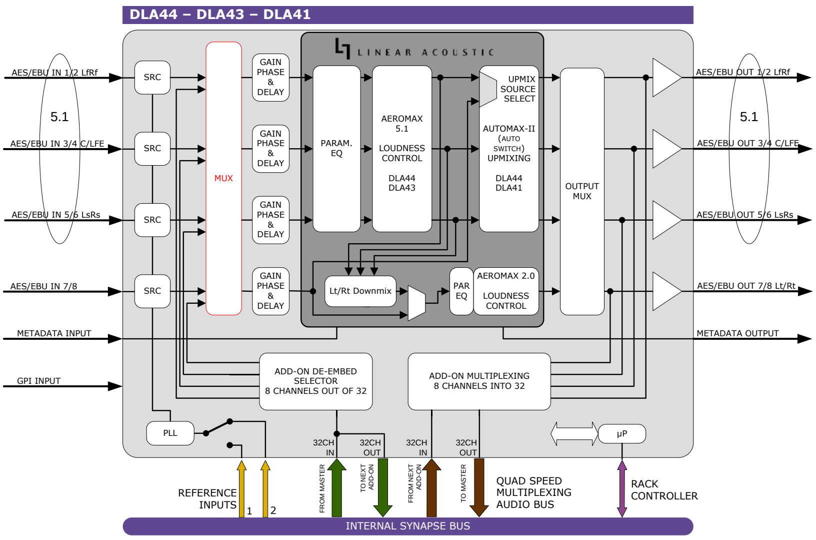
Pregain

Can be set to `manual` (uses Gain-CH_1 ~ Gain-CH_8 settings to attenuate the audio) or to `MD_Dialoglevel` (uses dialog level to attenuate the audio).

INPUT CONTROL (DLA44/43/41)

Sel_CH_1 ~ Sel_CH_8

With these settings you select which audio source you want to use for process channel 1 till channel 16. You can choose either `Local` audio (using the card's own AES/EBU inputs) or audio coming from the `Master` card via de Quad speed ADD-ON bus. Default is `Local`. The lines and blocks that are marked red in the following schematic are done by this setting:



Ch_1 ~ Ch_8

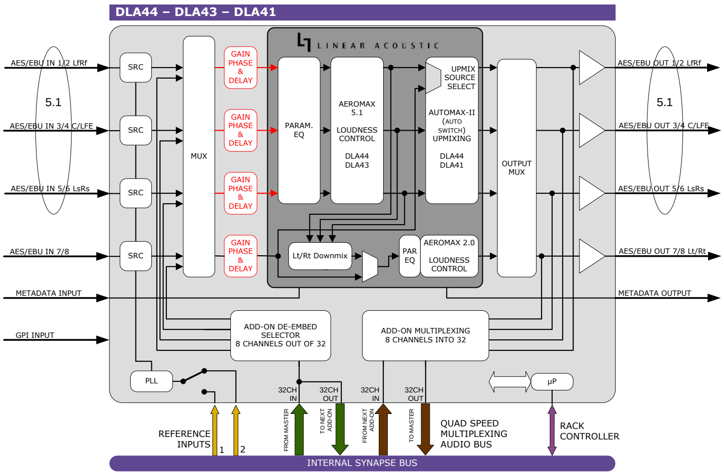
With these settings you select the actual source channel in the above selected source.

Note: When Local is selected, you can choose channel 1 till channel **16**. When Master is selected, you can choose channel 1 till channel **32**.

Gain-CH_1 ~ Gain-CH_8

These items allow you to gain the audio for each individual channel in a range from `-60dB` to `12 dB` in steps of `0.25 dB`. `-999dB` mutes this channel. Default is `0dB`. This setting does not work when `Pregain` is set to `MD_Dialoglevel`.

The lines and blocks that are marked red in the following schematic are done by this setting:



Phase-CH_1 ~ Phase-CH_8

These items allow you to gain the audio for each individual channel with 180 degrees. Default is `0 degree`. This setting does the same red blocks and lines as shown in the schematic above (`Gain-CH` setting explanation).

Delay_CH_1 ~ Delay_CH_8

These settings allow you to delay the audio of each channel in a range of `0` to `1300` ms. In steps of `0.01 ms`. Default is `0ms`. This setting does the same red blocks and lines as shown in the schematic above (`Gain-CH` setting explanation).

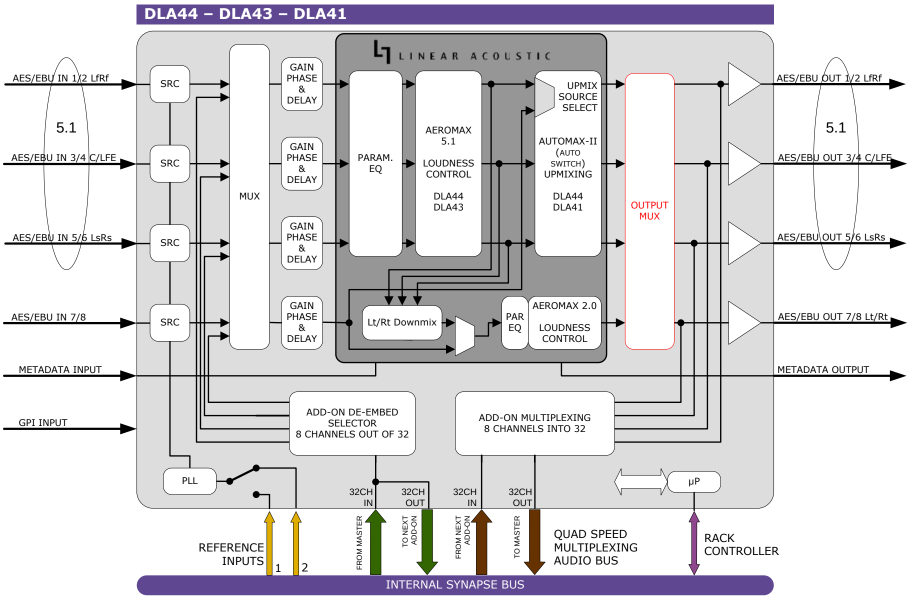
OUTPUT CONTROL (DLA44/43/41)

Out_1/2 ~ Out_7/8

With these settings you can select what sources should be used for the outputs. This can also be used for bypassing the Linear acoustic algorithms. The following settings are possible:

- `Loudness1/2` = Linear Acoustic output 1/2 (Lf Rf)
- `Loudness3/4` = Linear Acoustic output 3/4 (C/LFE)
- `Loudness5/6` = Linear Acoustic output 5/6 (LsRs)
- `Loudness7/8` = Linear Acoustic output 7/8 (LtRt)
- `Aes1/2` = First physical input
- `Aes3/4` = Second physical input
- `Aes5/6` = Third physical input
- `Aes7/8` = Fourth physical input
- `testtone` = Testone for testing the chain where the DLA is part of

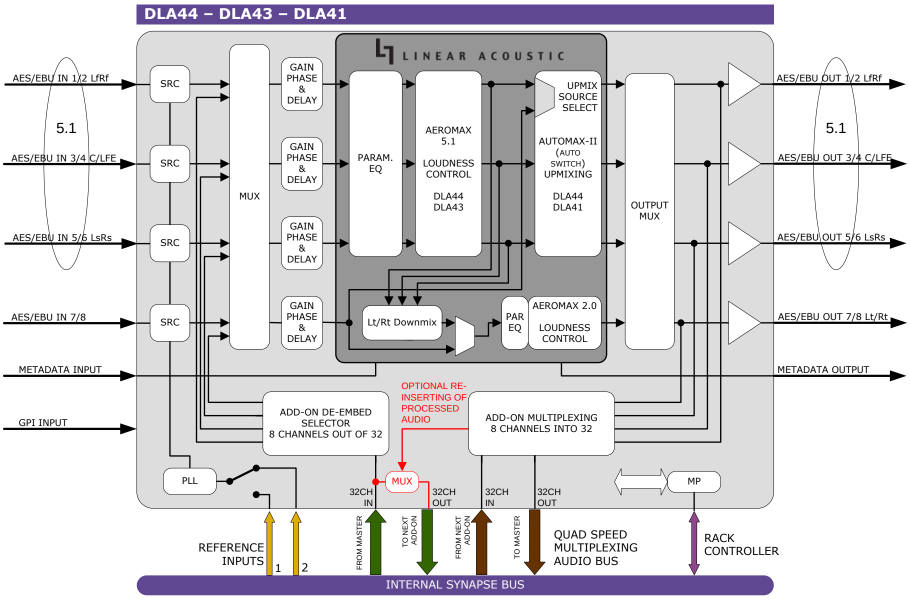
Defaults for `Out_1/2` till `Out_7/8` are respectively `Loudness1/2` till `Loudness7/8`. the following highlighted boxes perform these settings:



IN BUS CONTROL (DLA44/43/41)

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 Loudness controlled audio from this card to for instance a Dolby encoder 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. The lines and blocks that are marked red in the following schematic are done by this setting



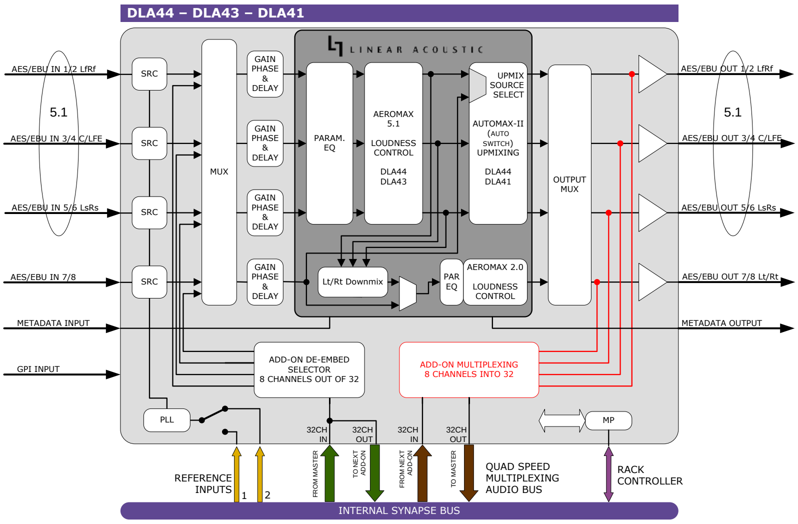
Override 25/32

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. Refer to above block schematic for a visualisation of this setting.

OUT BUS CONTROL (DLA44/43/41)

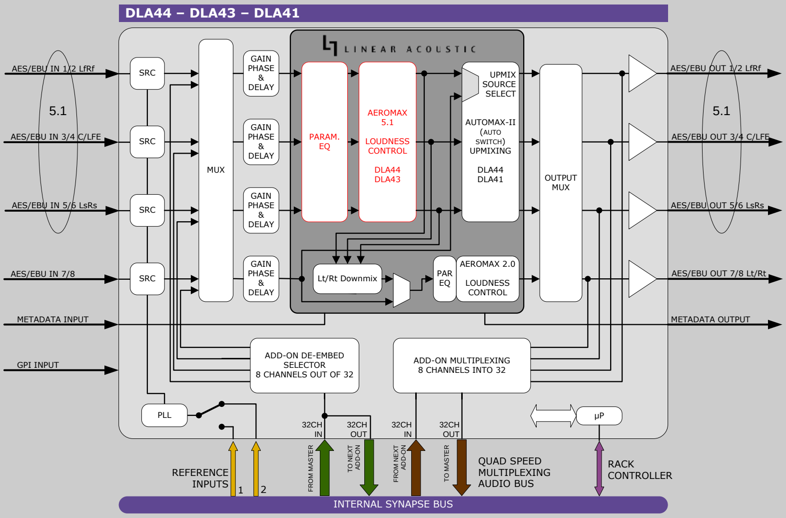
Slot1/2 ~ Slot31/32

These menu items are to fill the Quadspeed audio bus with the appropriate outputs. You can fill any of the 16 audio pairs (32 channels in total) with the audio that is set to `Out1/2`, `Out3/4`, `Out5/6` or `Out7/8` in the settings above these. You can also switch the concerning pair to `off`, making the concerning audio pair empty. the following highlighted boxes perform these settings:



5.1 LOUDNESS (DLA44/43 only)

The following settings are the loudness control settings for the Mainstream. The following highlighted boxes perform these settings:



#AMX Bypass

This option bypasses all the loudness processing options. Please note that this settings does not bypasses the final stage.

#Preset_Actual
Mainstream

With this item you can manually change the currently active preset. Can be any preset between 1 and 16 or GPI. With GPI selected, the active preset is controlled buy the GPI input (see `GPI control`). By default it is set to 1. All menu settings that are preceded with a '#'-prefix, up till the `**DWN_LOUD**` header, are part of this preset (in this manual marked with 'Mainstream').

#Preset_Adjust
Mainstream

Here you can select which of the 16 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, up till the `**DWN_LOUD**` header, are part of this preset (in this manual marked with 'Mainstream'). From factory default, the card's presets are pre-filled to industry standard default settings. We advise to make a backup of the card's settings before changing any presets. The default presets are the following:

- #1, #8, #14, #15, #16 = TeleVision 5-Band General. This is the most commonly used preset. It provides a moderate degree of dynamic range processing, and is appropriate for all types of content. It is the factory default for all core preset choices. Use of this preset is highly recommended as it will produce audio that will have an average dialog loudness of -27 as measured by a Dolby LM100.
- #2, #9 = TeleVision 5-Band Light. This is very similar to TV 5B general, however the ratio of the multiband compression has been reduced closer to 2:1 for a more gentle action. Please be aware that the lessening processing lessens the ability of the unit to control loudness tightly to a given target.
- #3, #10 = TeleVsion 5-Band Heavy. This is similar to TV 5-Band general, however the ratio of the multiband compression has been increased for a more dense and less dynamic sound.
- #4, #11 = TeleVision 5-Band Loud. Similar to TV 5-Band heavy, but louder and more punchy.
- #5, #12 = ITU loud LMT: Utilizes a specially tuned input AGC plus Multiband Limiters and the Final Limiter to slowly adjust the average program loudness to a given value and the multiband and final limiters wall act until the AGC catches up. This preset is appropriate for ingest or live applications but because the multiband AGC is bypassed it has less ability to manage spectral balance which is important for transmission
- #6, #13 = Protection Limit: Bypasses all processing except for the final output limiter which is set only to prevent overload.
- #7 = EBU-R128 mode: a preset specifically configured to be conform the EBU-R128 format. Output level -23, limit drive -6.

Presets 1 till 7 are fixed presets and cannot be adjusted, except for the output level setting. Presets 8 till 13 are copies of the first 6 presets but these can be adjusted. Presets 14, 15 and 16 are copies of preset 1 and also adjustable.

INPUT_AGC (DLA44/43 only)

The input AGC is a very slow acting front-end gain control with a 36dB gain range whose only purpose is to make sure that the following processing stages are fed with the correct average audio levels. It is basically the automatic equivalent of an operator slowly riding a gain control on a console to keep the audio close to reference level. Wideband in nature, the AGC is not meant to perform rapid gain reduction or expansion as its actions will be more audible, as with all wideband gain processors. As a slow gain rider, its actions are nearly inaudible thanks to the multiband processing that follows it. The AGC has two stages of gating where the gain expansion is slowed or stopped to prevent background noise increasing.

#Gate_Thresh
Mainstream preset

Gating Threshold. Set the point at which the AGC release time is made extremely slow to prevent increasing background noise and allow the AGC to return to unity gain. Can be set between 0 dB and -90 dB.

#Freeze_Thresh
Mainstream preset

Freeze stops all gain change (i.e. when the audio drops to silence), and remains frozen at its current gain value until the threshold is exceeded.

#AGC_range
Mainstream preset

AGC_Range sets how much gain expansion above unity is performed, and this amount is subtracted from the total AGC gain range of 36dB, so the default value allows for 24dB of expansion and 12dB of compression. This adjustment is reflected in real time by changing the AGC meter scale.

#Threshold
Mainstream preset

Defines the audio-level above which audio should be loudness controlled. Can be set in a range between -18dB and 0dB.

#Attack
Mainstream preset

Defines the attack level of the processed audio. Can be between 0 (slowest) and 150 (fastest).

#Release
Mainstream preset

Defines the level of release of the processed audio. Can be between 0 (slowest) and 150 (fastest)

#Ratio
Mainstream preset

This parameter configures the rate of processing done on the loudness controlled audio. Can be any value between 1:1 and 1000:1.

	PEQ (DLA44/43 only) Three bands of parametric equalization are provided for fine tuning if necessary. None of the factory-supplied presets use the parametric equalizers, but they are provided to create notch filters or other effects if necessary.
#B1_FREQ ~ #B3_FREQ Mainstream preset	This sets a center frequency for equalization. Can be set between 20Hz and 20.000Hz.
#B1_Level ~ #B3_Level Mainstream preset	Sets the gain of each band, applied to the center frequency. Can be set between -18 and +18dB.
#B1_Q ~ #B3_Q Mainstream preset	Sets the bandwidth to which the gain level should be applied, between 0 and 10.
	MULTIBAND_AGC (DLA44/43 only) This section is the heart of the dynamics processing engine. A multiband AGC (i.e. compressor) that allows for medium ratio adjustment of audio band.
#MB_Ratio	This item defines the amount of compression that is applied to the audio. This can be any value between 1:1 and 1000:1
#MB_AGC_range	MB_Range sets how much gain expansion above unity is able to be performed. Van be set between 0 and 36dB
#MB_Prog_R1	Sets the speed at which the release time is increased faster at very low gain values. This feature approximates a logarithmic release to help recovery from dramatic gain reduction more quickly. Can be set between 0 (slowest) and 150 (fastest)
#B1_Inf:1>thres ~ #B5_Inf:1>thres Mainstream preset	AGC can automatically increases ratio to Infinity:1 once a signal exceeds the threshold (set below), allowing for expansion below the threshold and limiting above the threshold. Useful for bass frequency control. This can be enabled or disabled for each band.
#B1_Attack ~ #B5_Attack Mainstream preset	These items set how fast an input signal is acted upon once is crosses the set threshold for each band. Can be set between 0 (slowest) and 150 (fastest).

#B1_Release ~ #B5_Release Mainstream preset	These items set how fast an input signal recovers from a gain change once that signal falls below the set threshold for each band. Can be set between 0 (slowest) and 150 (fastest).
#B1_Threshold ~ #B5_Threshold Mainstream preset	These set The reference point for the attack and release parameters to act on the audio signal present in that band. Can be set between -12dB and +12dB.
	MULTIBAND_LIM (DLA44/43 only) These settings perform multiband limiting of the signals coming from the multiband compressor.
#B1_Lim_Thres ~ #B5_Lim_Thres Mainstream preset	These set the point above which limiting action takes place at an Infinity:1 ratio for each band. Can be set between 0dB and +18dB.
#B1_SoftClp Mainstream preset	These items set the point above which band one (low bass) is very quickly limited, acting more like a clipper without the artifacts. This helps maintain a "tight" bass sound. Values can be set between 0dB and +18dB.
#B2_SoftClp Mainstream preset	These items set the point above which band two (mid bass) is very quickly limited, acting more like a clipper without the artifacts. This helps maintain a "tight" bass sound. Values can be set between 0dB and +18dB.
	MULTIBAND_LEV (DLA44/43 only) This is the section where each of the processing bands is summed and where overall frequency response can be tailored.
#MB_B1_Level ~ #MB_B5_Level Mainstream preset	Sets the mix level for each band summing all bands back together. These controls are prior to the final look ahead limiter and increasing gain may cause more final limiting (possibly more than desired).
	FINAL STAGE (DLA44/43 only)
#Out_Lim_Drv Mainstream preset	This Sets the level at which the wideband sum of all bands is fed to the final limiter. Can be set between -6dB and +6dB.

#Main Level Src
Mainstream preset

With this item you can choose to use the dialog level from external metadata to be used to control the Main out level. You can choose one of the external program's dialog levels, or choose to set it manually (by using the `#Master_out` setting)

#Offset Out Lvl
Mainstream preset

When the `Main level Src` is set to `Ext_meta` this values represents the offset to the dialog level in one of the programs in the metadata. The program can be selected with the setting `ExtprogID_x`. When `#Main Level Src` is set to `manual` this values represent the offset to the `output Level`.

In the first 6 presets these values are fixed and cannot be changed by the user. In the other presets they are adjustable and can be used to match the output level with the Loudness level measured by an external device for instance a Lambda Digital Audio and Metadata Monitor from Linear Acoustic.

#Master_out
Mainstream preset

This sets the output level for the current preset. Can be used to match the measured loudness of one preset to another. This is useful as more aggressive presets will measure differently from less aggressive versions. Can be set between -36dB and 0dB. This value is indicated with unit, but is actually an approximate value of the loudness level.

MD SETTINGS (DLA44/43/41)

#PrgmCnfg Id1
Mainstream preset

The DLA can make a new metadata bitstream. The program config is built out of PrgmCnfg Id1 + PrgmCnfg Id2. The output of this card can be:

- `2.0`
- `5.1`
- `5.1+2.0`
- `2.0+2.0`

With this setting you can set what should be in PrgmCnfg Id1. This can therefore be `2.0` or `5.1`.

#Metadata Set
Mainstream preset

With this setting you select one of the 8 metadata sets (see setting `#Metadata_set` further down the menu) to be used for Program 1 (mainstream data).

#Prg1 Reverts to
Mainstream preset

With this setting you can choose one of the Metadata sets to which the DLA should switch to in case the metadata coming from EXT(ernal) fails.

#Ext prog ID
Mainstream preset

With this setting you select the external source when the metadata set refers to EXT(ernal). Can be `Prog#1` till `Prog#4`.

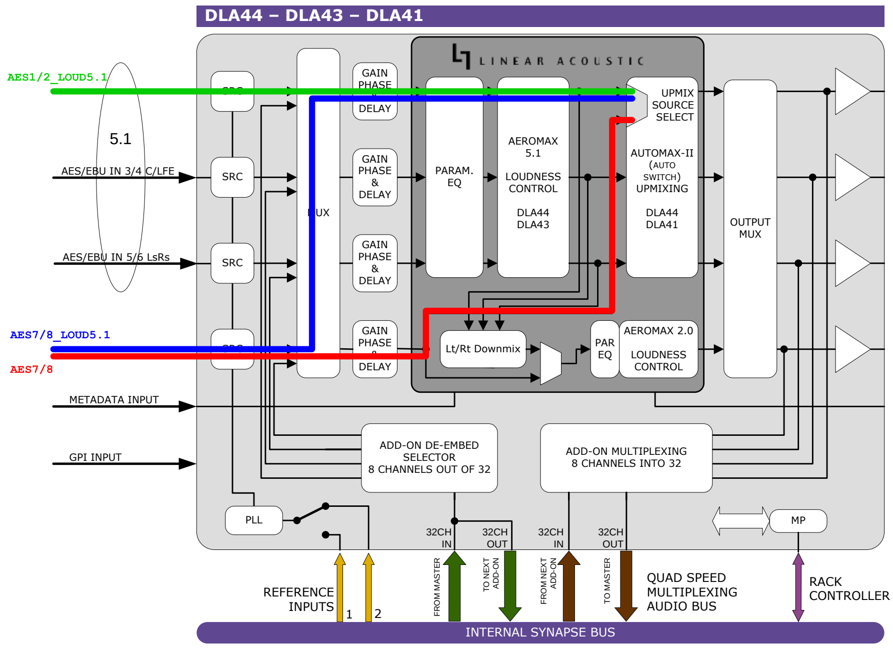
UPMIXER (DLA44/41 only)

Upmixer Input

When the `upmix` setting is set to `Automax`, then the `Automax` will use this setting as PCM input. Can be set to:

- `Chn1/2_Loud5.1`: Channel 1/2 will be used as upmix source. These channels have been processed by the Aeromax 5.1 block.
- `Chn7/8_Loud5.1`: Channel 7/8 will be used as upmix input. These channels have been processed by the Aeromax 5.1 block.
- `AES7/8`: AES7/8 will be used, only with this setting bypassing any loudness control processing blocks.

The following Block schematic indicates the 3 possible paths. The green line represents the signal path when set to `AES1/2_Loud5.1`. The blue line indicates the path when it is set to `AES7/8_Loud5.1`. The red line indicates the `AES7/8` setting.

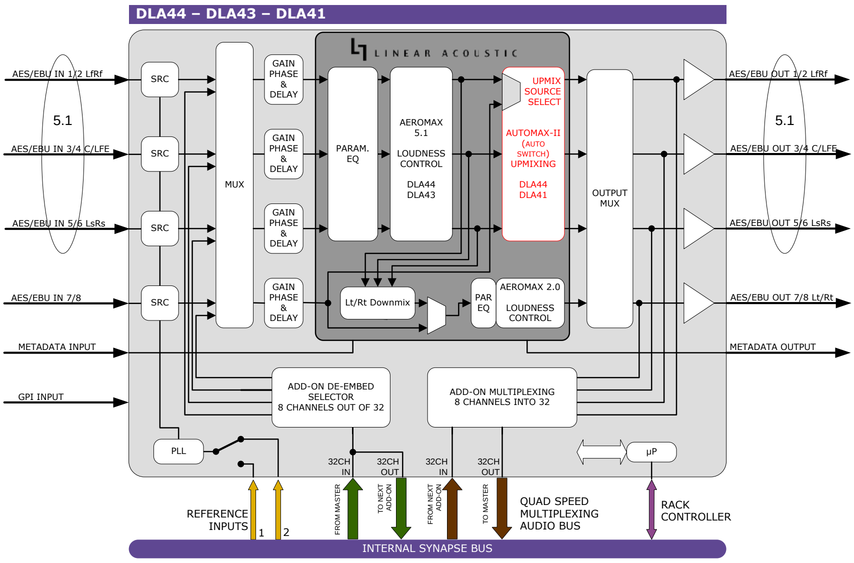


Up_Mix

With this setting controls the upmixer of the linear acoustic engine. The following settings are possible:

- `Upmix_Off`: disables upmixing
- `Upmix_On`: enables upmixing
- `AutomaxII_on`: with this setting the card decides whether to up mix or not, depending on the inputs (upmix off when 5.1 detected on input, upmix on when 2.0 is detected (depends on setting `backup chn` which input is used for PCM)).
- `MD_Upmix_On`: If external metadata is 5.1, the upmix will be switched off. If external metadata not 5.1 or when the metadata input is lost, the upmix will be switched on.
- `MD_AutoMax`: Same as MD_Upmix except when the Upmix should switched on, the DLA will check the input to be 2.0 before actually enabling upmix (depends on setting `backup chn` which input is used for PCM).
- `GPI_1_Upmix`: 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`.

The lines and blocks that are marked red in the following schematic are done by this setting:



Center_Width

Controls how much of center channel output of the upmixer is spread back into the Left and Right channel Main Outputs of the unit. The Default is 33% to prevent two channel mono material from being reproduced as stark center channel only audio, and to prevent center channel build-up with music content.

Surr_Depth

Controls how much of front left and front right channel output of the upmixer is spread back into the Left rear and Right rear Main Outputs of the unit. The Default is 100%.

LFE_ena

When enabled, this function creates a downmix compatible low pass filtered bass enhancement signal to feed the LFE channel.

LFE_Level

Sets the level of the created bass enhancement signal sent to the LFE output. Default is 12% which is compatible across the largest number of consumer systems.

Up_DSC_Thres

This sets the threshold for the AutoMAX-II algorithm to decide that the input audio is not 5.1. Audio present in the Center, Left Surround, or Right surround channels above this threshold will trigger the auto upmixer to start upmixing the LfRf (channels 1/2) audio. Can be set between -90dB and 0dB. Default is -75dB.

5.1>PCM_speed

With this setting you can adjust the speed of the switch between 5.1 and PCM when audio changes from 5.1 to PCM. Can be `Instant`, `VeryQuick`, `Quick`, `Medium`, `Slow` or `VerySlow`.

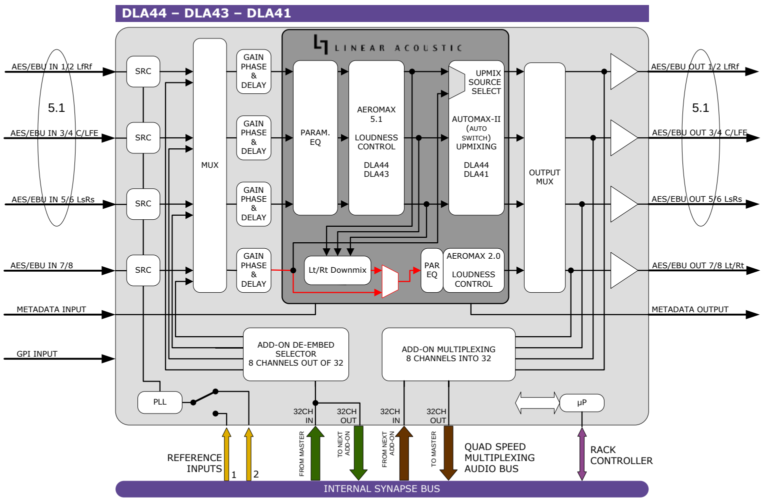
PCM>5.1_speed

With this setting you can adjust the speed of the switch between PCM and 5.1 when audio changes from PCM to 5.1. Can be `Instant`, `VeryQuick`, `Quick`, `Medium`, `Slow` or `VerySlow`.

DOWNMIXER (DLA44/43/41)

Input_Out7/8

With this setting you can bypass the downmix process and use AES input 7/8 to be loudness controlled (see highlighted diagram below). Set to `AES7/8` the bypass is enabled and the dowmixed audio will not be used. Set to `Downmix`, the output of the downmix processor is used and the AES input 7/8 is not used. The following highlighted blocks and lines show where this function takes place in the diagram:

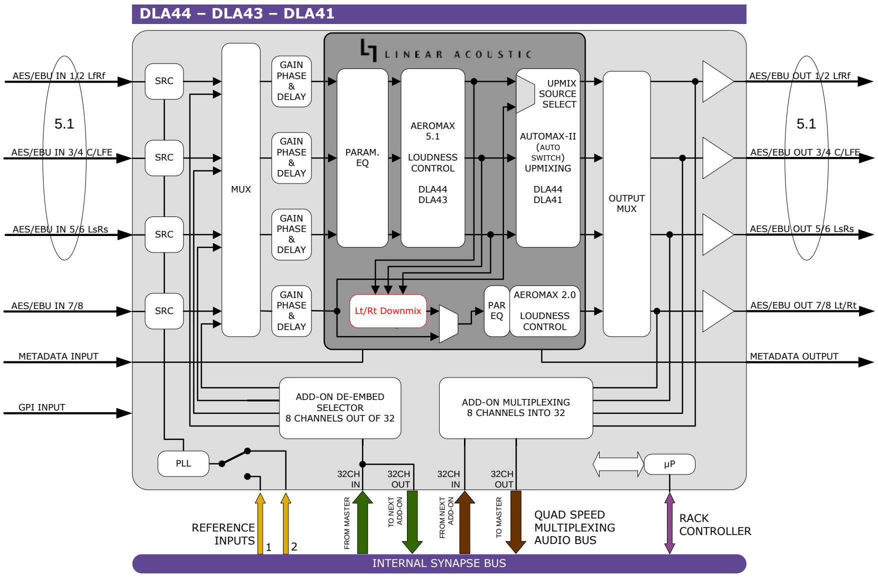


Input Control

When set to `manual` the input of outgoing channels 7/8 can be set with the `Input Control option`. When it is set to `Metadata Control` the input of 7/8 switches to Downmix when the metadata is valid. If there is no valid metadata it switches to AES7/8 as input.

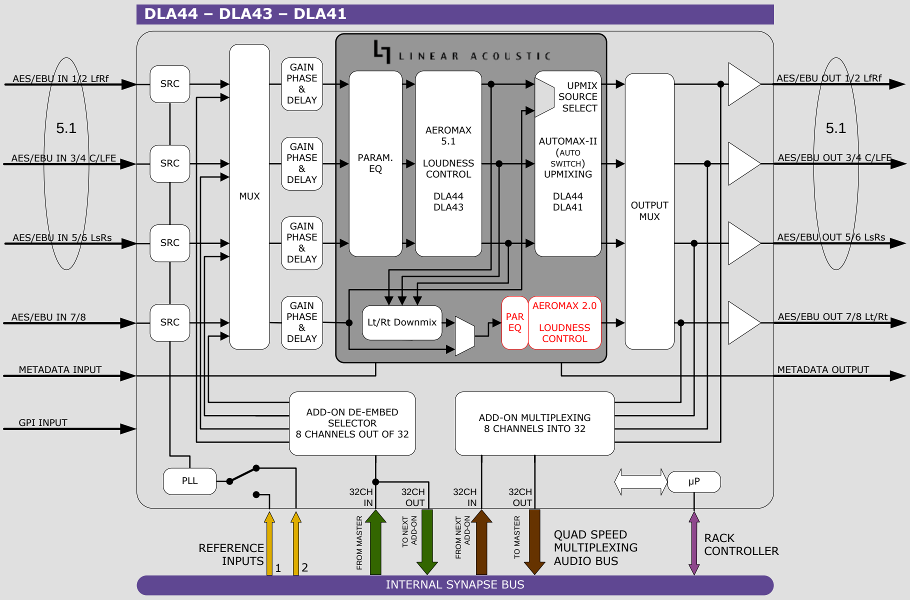
Dwn_Type

The downmix can be selected as stereo LoRo or surround compatible stereo LtRt. Note that due to the required phase encoding, selection of LtRt will increase the latency of the 7/8 output with 5 milliseconds.



2.0 LOUDNESS (DLA44/43 only)

The following settings are the loudness control settings for the downstream (or stereo stream). The processing path works the same as the mainstream path. The following highlighted box performs these settings:



#Dwn AMX Bypass

This option bypasses all the loudness processing options. Please note that this settings does not bypasses the final stage.

#DwnPrst_Actual
Downstream

With this item you can manually change the currently active preset. Can be any preset between 1 and 16 or GPI. With GPI selected, the active preset is controlled buy the GPI input (see setting `GPI control`). By default it is set to 1. All menu settings that are preceded with a ‘`#Dwn`‘-prefix are part of the preset, marked with ‘Downstream’.

#DwnPrst_Adjust
Downstream

Here you can select which of the 16 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 ‘`#Dwn`‘-prefix are part of the preset, marked with ‘Downstream’.
The from factory default, the card’s presets are pre-filled to industry standard default settings. We advice to make a backup of the card’s settings before changing any presets. The default presets are the following:

- #1 = TeleVision 5-Band General. This is the most commonly used preset. It provides a moderate degree of dynamic range processing, and is appropriate for all types of content. It is the factory default for all core preset choices. Use of this preset is highly recommended as it will produce audio that will have an average dialog loudness of -27 as measured by a Dolby LM100.

- #2 = TeleVision 5-Band Light. This is very similar to TV 5B general, however the ratio of the multiband compression has been reduced closer to 2:1 for a more gentle action. Please be aware that the lessening processing lessens the ability of the unit to control loudness tightly to a given target.
- #3 = TeleVsion 5-Band Heavy. This is similar to TV 5-Band general, however the ratio of the multiband compression has been increased for a more dense and less dynamic sound.
- #4 = TeleVision 5-Band Loud. Similar to TV 5-Band heavy, but louder and more punchy.
- #5 = ITU loud LMT: Utilizes a specially tuned input AGC plus Multiband Limiters and the Final Limiter to slowly adjust the average program loudness to a given value and the multiband and final limiters wall act until the AGC catches up. This preset is appropriate for ingest or live applications but because the multiband AGC is bypassed it has less ability to manage spectral balance which is important for transmission
- #6 = Protection Limit: Bypasses all processing except for the final output limiter which is set only to prevent overload.

Presets 7 till 16 are all loaded with the TeleVision 5-Band General preset (same as preset 1).

DWN_INPUT_AGC (DLA44/43 only)

#Dwn_Gate_thres
Downstream preset

Downstream Gating Threshold. Set the point at which the AGC release time is made extremely slow to prevent increasing background noise and allow the AGC to return to unity gain. Can be set between 0 dB and -90 dB.

#Dwn_Frz_thres
Downstream preset

Freeze stops all gain change (i.e. when the audio drops to silence), and remains frozen at its current gain value until the threshold is exceeded.

#Dwn_AGC_range
Downstream preset

AGC_Range sets how much gain expansion above unity is performed, and this amount is subtracted from the total AGC gain range of 36dB, so the default value allows for 24dB of expansion and 12dB of compression. This adjustment is reflected in real time by changing the AGC meter scale.

#Dwn_Threshold
Downstream preset

Defines the audio-level above which audio should be loudness controlled. Can be set in a range between -18dB and 0dB.

#Dwn_Attack Downstream preset	Defines the attack level of the processed audio. Can be between 0 (slowest) and 150 (fastest).
#Dwn_Release Downstream preset	Defines the level of release of the processed audio. Can be between 0 (slowest) and 150 (fastest)
#Dwn_Ratio Downstream preset	This parameter configures the rate of processing done on the loudness controlled audio. Can be any value between 1:1 and 1000:1.
	DWN_PEQ (DLA44/43 only)
#Dwn_B1_FREQ ~ #Dwn_B3_FREQ Downstream preset	This sets a center frequency for equalization. Can be set between 20Hz and 20.000Hz.
#Dwn_B1_Level ~ #Dwn_B3_Level Downstream preset	Sets the gain of each band, applied to the center frequency. Can be set between -18 and +18dB.
#Dwn_B1_Q ~ #Dwn_B3_Q Downstream preset	Sets the bandwidth to which the gain level should be applied, between 0 and 10.
	DWN_MBAND_AGC (DLA44/43 only)
#Dwn_MB_Ratio Downstream preset	This item defines the amount of compression that is applied to the audio. This can be any value between 1:1 and 1000:1
#Dwn_MB_AGC_rng Downstream preset	MB_Range sets how much gain expansion above unity is able to be performed. Van be set between 0 and 36dB
#Dwn_MB_Prog_Rl Downstream preset	Sets the speed at which the release time is increased faster at very low gain values. This feature approximates a logarithmic release to help recovery from dramatic gain reduction more quickly. Can be set between 0 (slowest) and 150 (fastest)
#DwnB1_Inf:1>th ~ #Dwn_B5_Inf:1>th Downstream preset	AGC can automatically increases ratio to Infinity:1 once a signal exceeds the threshold (set below), allowing for expansion below the threshold and limiting above the threshold. Useful for bass frequency control. This can be enabled or disabled for each band.

#Dwn_B1_Attack ~ #Dwn_B5_Attack Downstream preset	These items set how fast an input signal is acted upon once is crosses the set threshold for each band. Can be set between 0 (slowest) and 150 (fastest).
#Dwn_B1_Release ~ #Dwn_B5_Release Downstream preset	These items set how fast an input signal recovers from a gain change once that signal falls below the set threshold for each band. Can be set between 0 (slowest) and 150 (fastest).
#Dwn_B1_Thresh ~ #Dwn_B5_Thresh Downstream preset	These set The reference point for the attack and release parameters to act on the audio signal present in that band. Can be set between -12dB and +12dB.
	MULTIBAND_LIM (DLA44/43 only)
#Dwn_B1_LimThrs ~ #Dwn_B5_LimThrs Downstream preset	These set the point above which limiting action takes place at an Infinity:1 ratio for each band. Can be set between 0dB and +18dB.
#Dwn_B1_SoftClp Downstream preset	These items set the point above which band one (low bass) is very quickly limited, acting more like a clipper without the artifacts. This helps maintain a “tight” bass sound. Values can be set between 0dB and +18dB.
#Dwn_B2_SoftClp Downstream preset	These items set the point above which band two is very quickly limited, acting more like a clipper without the artifacts. This helps maintain a “tight” bass sound. Values can be set between 0dB and +18dB.
	DWN_MB_LEV (DLA44/43 only)
#Dwn_MB_B1_Level ~ #Dwn_MB_B5_Level Downstream preset	Sets the mix level for each band summing all bands back together. These controls are prior to the final look ahead limiter and increasing gain may cause more final limiting (possibly more than desired).
	DWN FINAL STAGE (DLA44/43 only)
#Dwn_Out_Lim_Drv Mainstream preset	This Sets the level at which the wideband sum of all bands is fed to the final limiter for the downmix output. Can be set between -6dB and +6dB.

#Stereo Level Src
Mainstream preset

With this item you can choose to use the dialog level from external metadata to be used to control the stereo out level, or to use the 5.1 output metadata setting for this. You can choose one of the external program's dialog levels, or choose to set it manually (by using the `#Stereo out Lvl` setting)

#Dwn_Off_Out Lvl
Mainstream preset

When the `Stereo level Src` is set to `Ext_meta` this values represents the offset to the dialog level in one of the programs in the metadata. The program can be selected with the setting `ExtprogID_x`. When `#Stereo Level Src` is set to `manual` this values represent the offset to the output Level.

In the first 6 presets these values are fixed and cannot be changed by the user. In the other presets they are adjustable and can be used to match the output level with the Loudness level measured by an external device for instance a Lambda Digital Audio and Metadata Monitor from Linear Acoustic.

#Stereo Out Lvl
Mainstream preset

This sets the stereo output level for the current preset. Can be used to match the measured loudness of one preset to another. This is useful as more aggressive presets will measure differently from less aggressive versions. Can be set between -36dB and 0dB. This value is indicated with unit, but is actually an approximate value of the loudness level.

DWN MD SETTINGS (DLA44/43/41)

#PrgmCnfg Id2
Downstream preset

The DLA can make a new metadata stream. The program config is built out of PrgmCnfg Id1 + PrgmCnfg Id2. The output of this card can be:

- 2.0
- 5.1
- 5.1+2.0
- 2.0+2.0

With this setting you can set what should be in PrgmCnfg Id2. This can therefore only be `2.0` or `off`.

#Dwn Metadata Set
Mainstream preset

With this setting you select one of the 8 metadata sets (see setting `#Metadata_set` further down the menu) to be used for Program 1 (mainstream data).

#Dwn Prg1 Revs to
Downstream preset

With this setting you can choose one of the Metadata sets to which the DLA should switch to in case the metadata coming from EXT(ernal) fails.

#Dwn Ext prog ID
Downstream preset

With this setting you select the external source when the metadata set refers to EXT(ernal). Can be Prog#1 till Prog#4.

METADATA (DLA44/43/41)

With the following setting you can define the Dolby metadata output of the DLA.

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).

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.

METADATA PRST (DLA44/43/41)

With the following setting you can define and fine tune the Dolby metadata presets for the MD output of the DLA.

#Metadata_set

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 (LCRS1Sr)`

#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.

#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`.

6 Status Menu

Introduction

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

Ref-Stat

This indicates the incoming reference status. Can be either 30Hz or 25Hz. When there is no reference, this item indicates NA (Not Available).

AES-In_1 ~ AES-In_8

If a valid signal is present on the corresponding input, `AES-IN_1` till `AES-In_8` indicate `OK`. If the signal is between –1 and 0dBFS, it is displayed as `> -1dBFS`. `NA` indicates that no AES signal is present.

Ext_Meta

Displays the status of the Metadata input. Can be `OK` or `NA`.

GPI

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

Upmix Stat

Indicates whether the DLA is currently upmixing or not (`on` or `off`).

Chn1_Input ~ Chn6_Input

These items are the indicators of the volume meters when `Meter` (in settings) is set to `Input` or `Output`. The setting `Refreshrate` defines how fast the indicators are refreshed.

MD Prgrm Config

This indicates whether or not there is a Program Config metadata.

MD FrameRate

This status displays Data rate metadata.

MD Dialog Lvl

This status displays the Dialogue level metadata (depends on the `MD_status_Src` setting which program stream is the source).

MD Bitstream

This status the Bitstream metadata (depends on the `MD_status_Src` setting which program stream is the source).

MD ChannelMode

This status displays the Channel mode metadata (depends on the `MD_status_Src` setting which program stream is the source).

MD LFE

This status displays the LFE metadata (depends on the `MD_status_Src` setting which program stream is the source).

MD Line Mode

This status displays the Line mode metadata (depends on the `MD_status_Src` setting which program stream is the source).

MD RF Mode	This status displays the RfMode metadata (depends on the `MD_status_Src` setting which program stream is the source).
MD D_Surnd	This status displays the Dolby surround metadata (depends on the `MD_status_Src` setting which program stream is the source).
MD Pref. Dwnmx	This status displays the Prefered Downmix metadata (depends on the `MD_status_Src` setting which program stream is the source).
MD Lt/RtCDwnmx	This status displays the Lt/Rt C Downmix metadata (depends on the `MD_status_Src` setting which program stream is the source).
MD Lt/RtSDwnmx	This status displays the Lt/Rt S Downmix metadata (depends on the `MD_status_Src` setting which program stream is the source).
MD Lo/RoCDwnmx	This status displays the Lo/Ro C Downmix metadata (depends on the `MD_status_Src` setting which program stream is the source).
MD Lo/RoSDwnmx	This status displays the Lo/Ro S Downmix metadata (depends on the `MD_status_Src` setting which program stream is the source).
MD D_Surnd EX	This status displays the Surround EX metadata (depends on the `MD_status_Src` setting which program stream is the source).
MD DC filter	This status displays the DC filter metadata (depends on the `MD_status_Src` setting which program stream is the source).
MD LFE filter	This status displays the LFE filter metadata (depends on the `MD_status_Src` setting which program stream is the source).
MD Lowpass fil	This status displays the Lowpass filter metadata (depends on the `MD_status_Src` setting which program stream is the source).
MD Sur3DB Att	This status displays the Surround 3db Attenuation metadata (depends on the `MD_status_Src` setting which program stream is the source).
MD Sur Phshift	This status displays the Surround Phase shift metadata (depends on the `MD_status_Src` setting which program stream is the source).

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.

DLA44/43/41 Events

The events reported by the DLA44/43/41 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

Audio-Data

`Error in Audio Data` can be selected between 0 .. 255. 0= no event, 1..255 are the priority setting. In case of a Dolby data ERROR an Event will be generated at the priority, or the audio carrier is falling away, or audio data is in the range of 0 dBFS and -1 dBFS, or the Encoder status is in error.

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.

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 DLA44/43/41 are:

Event Menu Item	Tag		Description
Announcements	0 or NA	0 or NA	Announcing of report and control values
Audio-Data	01_{hex}=AUDIO_ERROR	81_{hex}=AUDIO_OK	
Reference-Status	02_{hex}=REF_LOSS	82_{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/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

D+ OUT OK

This LED indicates a valid or not Dolby output.

ERROR

The error LED indicates an error if the internal logic of the DLA44/43/41 card is not configured correctly or has a hardware failure.

INPUT 1 ~ INPUT 5

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 DLA44/43/41 is locked to the master card.

DATA ERROR

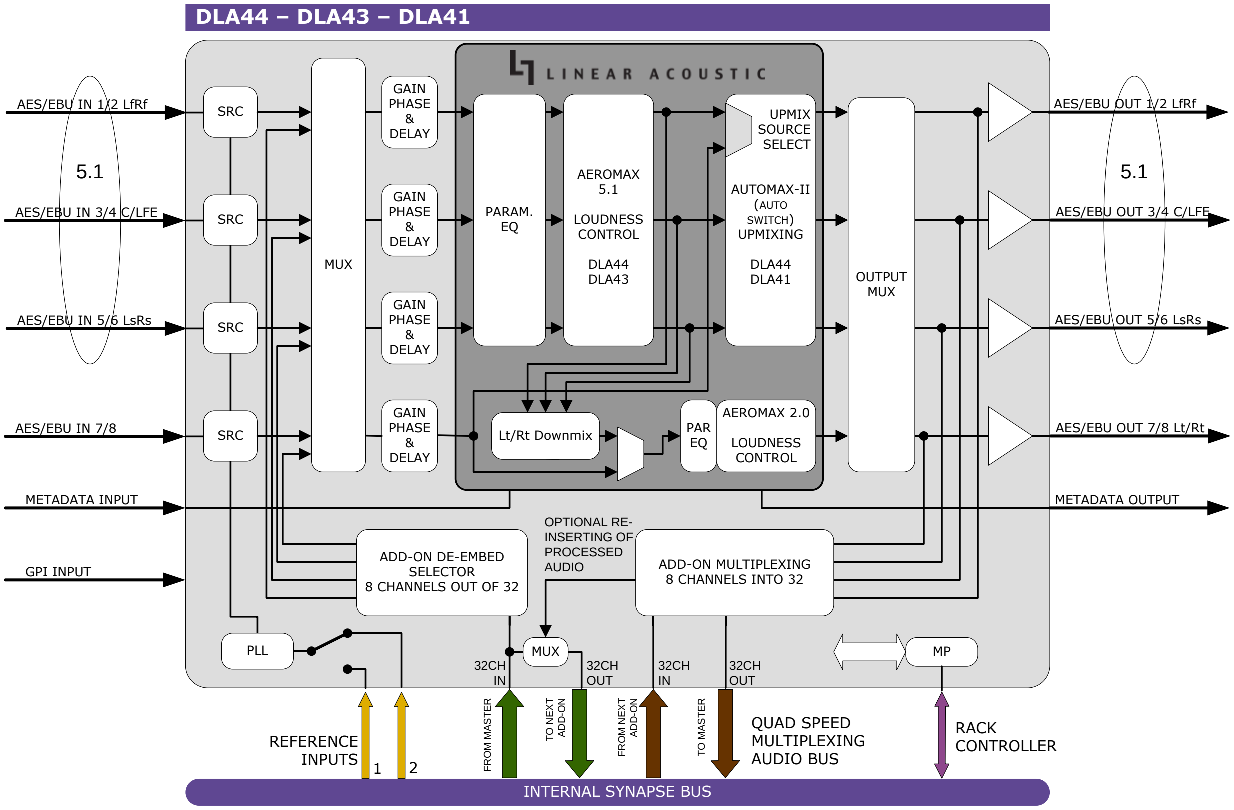
This led indicates different types of errors if there is an error in the Dolby encoding, or the audio carrier is falling away, or audio data is in the range of 0 dBFS and -1 dBFS.

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.

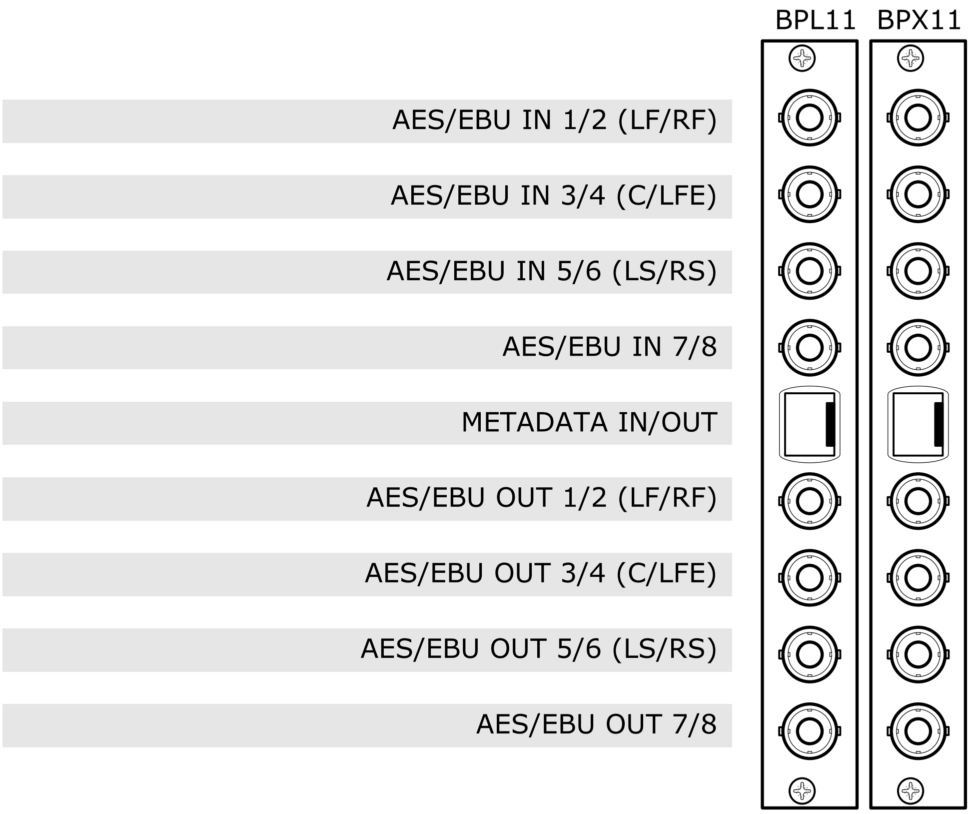
Note: When LEDS are blinking constantly, the card is still programming.

9 Block Schematic

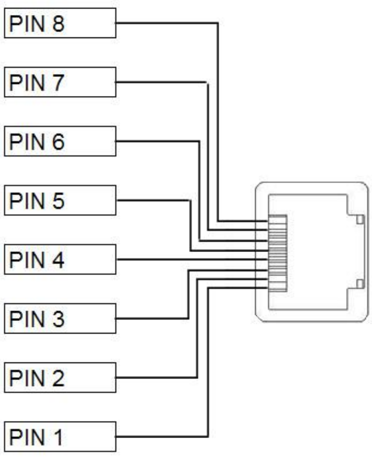


10 Connector Panel

The DLA44/43/41 can be used with the following back planes: BPL11 or BPX11:



Pin description of RJ45 connector:



Pin	Purpose
1	GPI 1
2	GPI 2
3	TXA(+)
4	RXA(+)
5	RXB(-)
6	TXB(-)
7	GPI3
8	GND

Appendix 1 Quad speed ADD-ON bus

Scope

The internal audio ADD-ON bus needs an upgrade. We want more channels (32 per video stream seems 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 needs to be compatible with all existing hardware (frames) and in the implementation of the master card it needs to be backward compatible with the original ADD-ON bus.

The master card will have two modes:

- Normal ADD-ON mode

  or

- Quad Speed audio ADD-ON mode

These modes are selectable on the Master Card. If a mode is selected all ADD-ON cards to that Master need to be in the same mode.
You can mix Master-Cards in one frame using the two different modes, but all cards to the right of the master must be in the same mode as the master.
A new Master breaks the chain and the Master Card ADD-ON mode can be selected again.

Features

The following features and rules will apply:

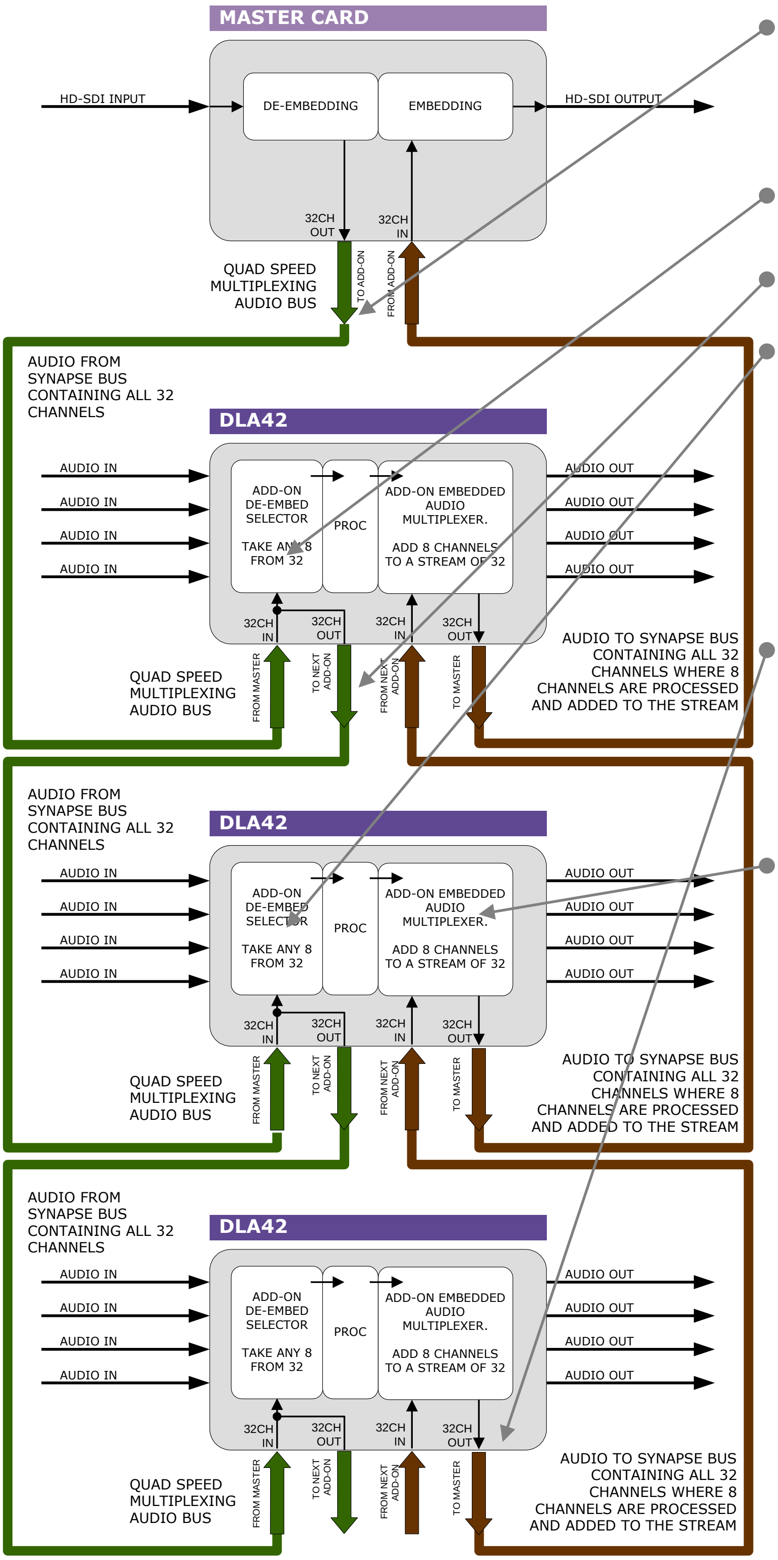
- Up to 32 channels output from the master card with looping to up to 3 ADD-ON cards
  - The ADD-ON card just picks the channels it wants to process
- Up to 32 channels input on the master card
  - If the master card can handle less then 32 channels, the lowest channel numbers will be used, as the ADD-ON card will always generate 32 channels (where some can be zero)
- 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
- 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 finaly 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

Example

The big difference between the new and old bus structure is the fact that it caries 4 times as much audio channels.

It is also bi directional by design. So half of the original physical infrastructure moves audio from the master card to the ADD-ON cards, and the other half is used to put the audio back

The following graphic shows how a typical quad speed bus chain works



The audio coming from the master card (dark green arrow) contains up to 32 channels.

The first ADD-ON card can select any of the 32 channels for internal processing

These channels are looped on to the next ADD-ON card.

This next ADD-ON (sitting in the next n+1 slot) Card can also free select any 8 from 32 channels. *(The DLA42 can also take 3 channels from the ADD-ON bus and 5 channels from its physical input)*

This looping works up to 3 times.

The brown arrow is the return path and sends the (processed) audio back to the master card.
This path is 32 channels wide and is clocked from the master card.

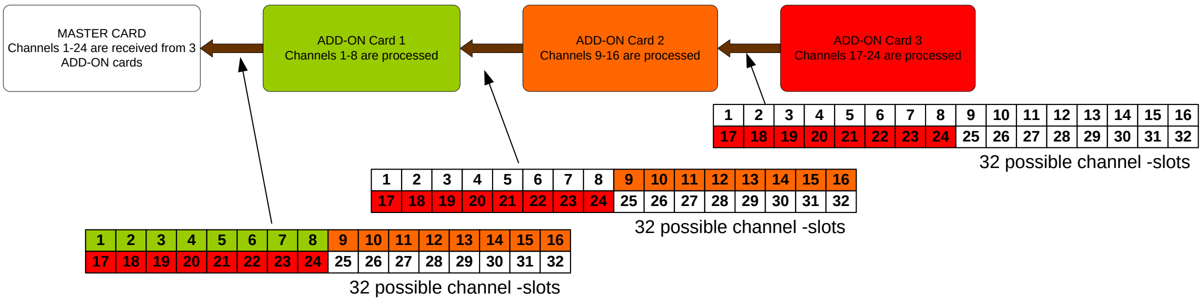
The ADD-ON card can overwrite for instance 8 channels of the 32. These 32 channels are then transported to the next ADD-ON card which overwrites another 8 channels.

Multiplexing

The injection of processed audio into the master card works differently then you were used to with the original audio ADD-ON bus. The brown large arrow will always carry 32 channels from ADD-ON to ADD-ON, or from ADD-ON to Master Card. If the actual channels are used or which channels are used is determined in the ADD-ON card.

In the example below you can see a 4 Card system. One Master Card, and 3 Quad speed ADD-ON cards (the maximum). The last (most right) ADD-ON card processes 8 channels. They are inserted (a menu selection) in slot 17-24 from 32 channel-slots. The second ADD-ON card also processes 8 channels, but they are inserted in slot 9-16 (of 32 slots). The first ADD-ON card inserts channels 1 to 8

This method allows for overwriting slots that come from the right hand Master Card. Channel-slot 25 to 32 are left empty in this example.



Note:
The top example shows a logical way of how the ADD-ON multiplexing could be performed. However; the insertion menu of for instance the DLA42 is much more flexible and allows putting every channel into any of the 32 channel-slots. The example below shows how the flexibility could be used.