



IQDBD00/01

**HD/SD – SDI 16 Channel AES/EBU
Remultiplexer with Dolby Decoder**

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Contents

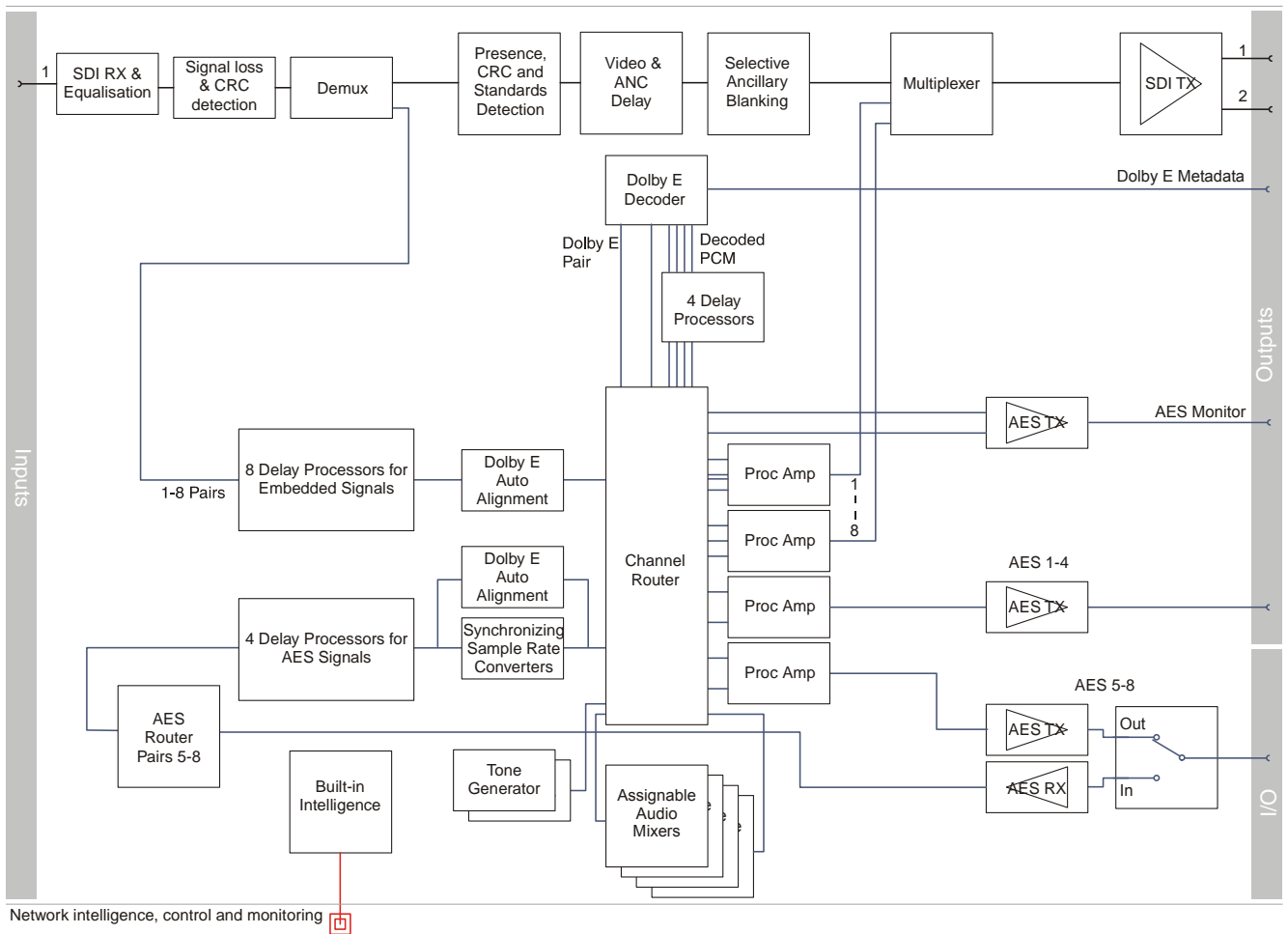
Module Description	4
Features	5
Rear Panel Views	6
Enclosures	7
Technical Profile	8
IQDBD00 Connections	10
Input Connections	10
Serial IN (IQDBD0043-2A and IQDBD0044-2A)	10
AES Unbalanced IN/OUT (IQDBD0043-2A)	10
AES Balanced IN/OUT (IQDBD0044-2A)	10
Output Connections	11
Serial Digital Video Outputs (IQDBD0043-2A and IQDBD0044-2A)	11
AES Unbalanced OUT (IQDBD0043-2A)	11
AES Unbalanced IN/OUT (IQDBD0043-2A)	11
AES Balanced IN/OUT (IQDBD0044-2A)	11
AES MON OUT (IQDBD0043-2A and IQDBD0044-2A)	12
RS485 DolbyE Metadata Output (IQDBD0043-2A and IQDBD0044-2A)	12
25 WAY D-Type Connection Details	13
Example of Connection to XLR Connectors	14
Card Edge Indicators	15
Controlling the IQDBD00 from the RollCall Control Panel	16
The Information Window	16
Video Input Settings	18
Specifying Valid Input Standards	18
Specifying Input Loss Conditions	18
Viewing CRC and EDH Errors	19
Video Output Settings	19
Specifying the Video Output Standard	20
Specifying a Test Pattern	20
Specifying the Video Output	21
Adjusting Gain and Black Levels	21
Controlling Monochrome Functions	22
Force Freezing the Picture	22
Audio Processing Overview	23
Audio Processing Block Diagram	23
SDI Demultiplexer	24
AES Processing	24
Audio Delay Processor	25
Synchronizing SRCs	25
Tone Generator	26
Mixers 1 to 4	26
Dolby Decoder	27
Embedder ProcAmps	27
SDI TX	28
AES ProcAmps	28
AES TX/RXs	28
AES Monitor	29
Dolby E Audio Handling	30
AES ProcAmp Settings	31
Controlling AES Output	31
Embedded ProcAmp Group Settings	32
Adjusting Embedded Audio Settings	32
Audio Mixer Settings	33
Adjusting Audio Mix	33
Changing a Mixer Name	34

Clip Indicators	34
AES Routing Settings	35
Specifying AES Routing	35
Monitor Routing Settings.....	36
Specifying Monitor Routing	36
Embedded Routing Groups Settings	37
Specifying Signal Sources.....	37
Video Delay/Dolby E Auto Line.....	38
Video Delay Modes	38
Minimum Audio.....	39
Frames	39
Manual H and V.....	39
Manual MS	40
Dolby E Auto Line.....	41
Dolby E Timing	42
Audio Delay Settings.....	44
Audio Delay Select-A/B	45
RollTrack	45
Total Delay	46
Dolby Decoded Channel Delay, AES Delay 5 to 8 and Embed Delay 1 to 8.....	46
Dolby Decoder	48
Dolby Decoder Source	48
Detect Mode	48
AES Channel Select.....	48
PCM Latency	48
Listening Mode Dolby Digital.....	49
Dynamic Range Control	49
Aux Channel Program Select.....	49
Aux Channel Dynamic Range	50
Aux Channel Output Mode	50
Dolby Decoder Outputs	50
Dolby Aux Channel Source	51
Dolby Decoder Metadata	51
Embed On/Off Settings	52
Embedder Priority.....	52
Group/Channel Active	52
Embedded I/P Setup.....	53
PCM.....	54
Data	54
Mixed	54
Embedded Audio Input Names	55
Embedded O/P Setup	56
Setting up Embedded Output.....	56
HANC and VBI Blanking Settings	57
Specifying Vertical Interval Data (VBI) Blanking Settings	57
Specifying Horizontal Ancillary Data (HANC) Blanking Settings.....	58
AES Output Port Setup 1-4.....	59
AES I/O Setup 5-8	60
PCM.....	61
Data	61
Mixed	61
RollTrack Settings.....	63
RollTrack Sources	63
RollTrack Addresses	63
RollTrack Commands.....	64
Using RollTracks	64
Viewing RollTrack Information.....	64
Memory Settings	65
Saving to Memory	65
Recalling Memory.....	66

Logging	67
Video Logging	67
Embedded Aud Logging.....	68
O/P Aud Logging	68
O/P Aud Level Logging	69
O/P Dolby-E Logging.....	69
Dolby Metadata 1 Logging	70
Dolby Metadata 2 Logging	70
Dolby Metadata 3 Logging	71
Dolby Metadata 4 Logging	71
AES Input Logging	72
AES Output Logging.....	72
RollCall Log Fields	73
The Setup Screen	76
Operation from an Active Control Panel	77
RollTrack Audio Delay Tracking	78
Dolby-E	84
What is Dolby E?	84
Dolby E and Metadata	84
Dolby E Partner Program.....	84
16 and 20-bit Dolby E	84
Dolby Digital and AC3	85
Metadata	85
Ancillary Passing and Audio Embedding	86
Manual Revision Record.....	88

Module Description

The IQDBD00 provides an integrated Dolby E decoding and remultiplexing solution for HD-SDI 1.5 Gbit/s or SD-SDI 270 Mbit/s signals. As well as providing multiplexing or demultiplexing for up to 16 PCM audio channels, eight AES/EBU streams, it can demultiplex and decode Dolby E data to output as AES or re-multiplex into the video stream. Dolby E features include automatic Dolby E alignment with the video signal, and metadata decoding and output to RS485. PCM audio processing features include tracking audio delay, gain, phase invert, mixing, Dolby E pair routing and separate channel level routing. Video features include proc. amp controls and up to 12 frames of delay.

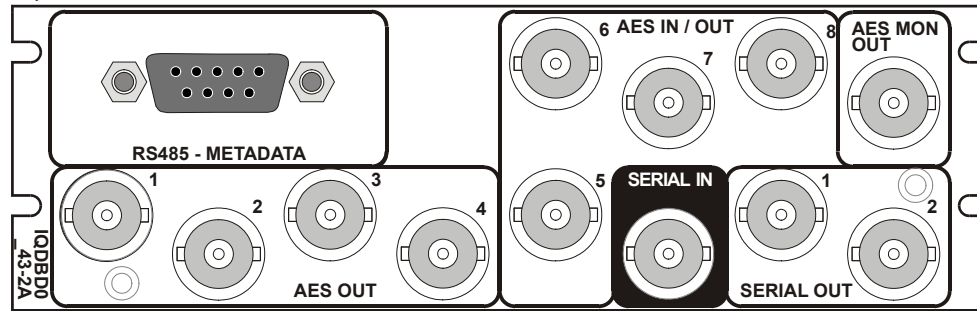


Features

- Multiplex unbalanced or balanced AES audio onto HD/SD-SDI video streams with channel-level control
- Demultiplex existing audio channels and output them to unbalanced or balanced AES
- Decode Dolby E data and either output to unbalanced or balanced AES or remultiplex into the HD/SD-SDI stream
- Decode Dolby E metadata and output as RS485
- Standards supported:
 - HD-SDI to SMPTE292M/274M/296M
 - SD-SDI to SMPTE259M-C
- Channel-level control allows up to 16 individual embedded audio channels to be swapped-over or swapped out
- 4 off 4 channel assignable audio mixers
- Audio proc-amp (with user programmable automatic level limiting)
- Audio delay channels include selectable fixed delay and tracking delays selectable for any pair
- Tracking audio delay which seamlessly tracks the video delay or external RollTrack inputs
- Dolby E support – pair routing and automatic re-alignment and synchronization to the video frame boundary
- Any group of embedded audio may be passed unchanged if not selected for processing
- Video delay feature, up to 12 frames
- Video controls including video gain and offset
- 16 x user memories
- Independent horizontal and vertical ancillary data blanking
- Input SDI, CRC, EDH and ANC data checking and reporting
- In-built test pattern generator
- Input loss detection – input pass through or black/pattern/freeze
- Naming of audio output channels for easy identification

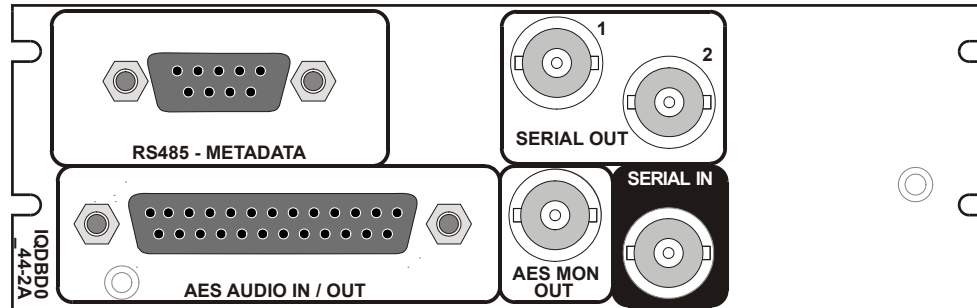
Rear Panel Views

IQDBD0043-2A



IQDBD0043-2A HD/SD-SDI 16 Channel demultiplexer with Dolby E decoder.
 2 HD/SD-SDI outputs, 4 AES/EBU unbalanced outputs, 4 AES/EBU unbalanced configurable input/outputs, 1 AES/EBU unbalanced monitor output, Dolby E Metadata output

IQDBD0044-2A

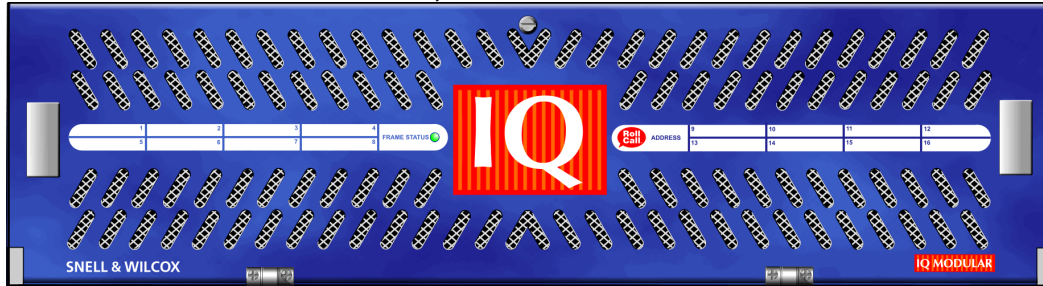


IQDBD0044-2A HD/SD-SDI 16 Channel demultiplexer with Dolby E decoder.
 2 HD/SD-SDI outputs, 4 AES/EBU balanced outputs, 4 AES/EBU balanced configurable input/outputs, 1 AES/EBU unbalanced monitor output, Dolby E Metadata output

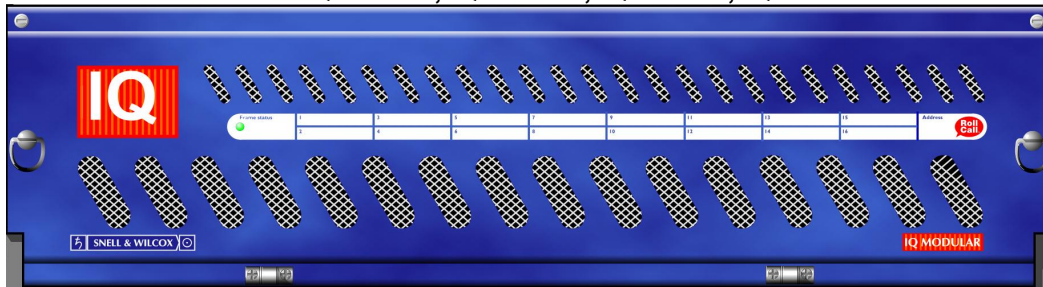
Enclosures

These modules can only be fitted into 'A' style enclosures as shown below.

Enclosure order codes IQH3A-S-0, IQH3A-S-P



Enclosure order codes IQH3A-E-0, IQH3A-E-P, IQH3A-0-0, IQH3A-0-P



Enclosure order code IQH1A-S-P



Technical Profile

Inputs & Outputs

Video Signal Inputs

Digital Video.....	1 x Serial Digital Input
Electrical	1.5 Gbit/s HD-SDI, SMPTE 292M, SMPTE 299M 270 Mbit/s SDI, SMPTE 259M-C
Connector / Format.....	BNC/ 75 ohm panel jack on standard S&W connector panel
Input Cable Length.....	Up to 140 m Belden 1694A @ 1.5 Gbit/s Up to 350 m Belden 1694A @ 270 Mbit/s
Return loss.....	> -15 dB

Video Signal Outputs

Digital Video.....	2 x Serial Digital Outputs
Electrical	1.5 Gbit/s HD-SDI, SMPTE 292M 270 Mbit/s SDI, SMPTE 259M-C
Connector / Format.....	BNC/ 75 ohm panel jack on standard S&W connector panel

Audio Signal Inputs/Outputs

Unbalanced AES/EBU

AES Audio I/O (software selectable)	4 Unbalanced
AES Audio Outputs	4 Unbalanced
AES Audio Monitor Output .	1 Unbalanced
Connector / Format	BNC/ 75 ohm panel jack

Balanced AES/EBU

AES Audio I/O (software selectable)	4 Balanced
AES Audio Outputs	4 Balanced
Connector / Format	25 Way D-Type / 110 ohm panel mounted
AES Audio Monitor Output .	1 Unbalanced
Connector / Format	BNC/ 75 ohm panel jack

RS485 Metadata

Connector.....	9 Way D-Type panel mounted
----------------	-------------------------------

Controls

Indicators

Power.....	OK (Green)
CPU	OK (Green flashing)
FPGA	OK (Orange flashing)
Status.....	OK (Green) Warning (Orange) Error (Red)
Lock	OK (Green)
SDI Error	Error (Red)

Dolby E Auto Line selection.....	Define Dolby E embed line for each video standard
Dolby E Decoder routing ...	channels routed from AES inputs 5 to 8, SDI 16 embedded channels from any group
Output side control proc. -	gain and polarity Independent Gain, Mute, & Polarity control over embedded output channels. +12 dB to -36 dB in 0.1 dB steps

RollCall Features

Audio Controls

Embedded Audio Types	PCM (to AES3)/ Data (SMPTE 337M inc. Dolby E)/ Mixed (Passes any channel status information present)
Channel routing	Output channels routed from Dolby E decoder, AES inputs 5 to 8, SDI 16 embedded channels from any group, test tone and silence
Embedder Priority	Normal distribution/Audio Prioritized
Embedded Group	Pass/Blank/Embed
Channel Status handling and checking	

Audio Delay

Audio Delay	Individual variable delay can be applied to the four AES Input pairs, four Dolby decoder output pairs and eight disembedded input pairs. The amount of delay comprises a combination of: Manual (coarse and fine), RollTrack, Video Delay.
Resolution	Approx. 0.5ms.

Maximum total delay2700ms for AES and embedded inputs.
2700ms plus one fram for Dolby decoder outputs.

Tone Setup:

Frequency 1 kHz, 2 kHz, 4 kHz, mute @ -20 dBFS or -18 dBFS

Video Controls

Output Standard.....Select, Follow Input

Standards List.....Select video standards for automatic follow

Black Level.....±200 mV in steps of 1 mV

Master Video Gain±6 dB in steps of 0.1 dB.

Y Gain±6 dB in steps of 0.1 dB.

Cb/Cr Gain±6 dB in steps of 0.1 dB.

Pattern Select 100% Color Bars, 75% Color Bars
SMPTE Bars, Tartan Bars, Pluge
Ramp, H Sweep, Pulse & Bar, Burst

Blank Ancillary DataBlank All, Blank HANC, Pass All, Pass when Output Standard equals Input Standard

VBI Line Blank.....Individual lines for each video standard

Manual FreezeOn/Off

FreezeField/Frame

Video Channel Control..... Y On/Off, C On/Off

Default Video Output..... Pattern / freeze/ black / run through

Metadata Controls

TBA

Other Controls

User Memories 16 x Save / Recall / Rename

Input/Output Names User configurable naming of the input and output AES/EBU, embedded audio and mixer channels

RollCall Features

Logging.....Video Status
Emb(eded) Audio Status
D/P Audio Status
D/P Audio Level Status
D/P Dolby E Status
AES Input Status
AES Output Status
Embedded audio output status, level & type (pairs 1-8)
Embedded Dolby E output timing status (pairs 1-8)
Misc

RollTrack ControlsSource, Address, Command, Status, Sending.

RollTrack Sources.....Unused, Video Delay, Input Present, Input Loss, Output Freeze, Output Unfreeze, Embedded Audio (Pairs 1-8)
AES Audio (Pairs 5-8)

Specifications

Video Standards

750(720)/59p, 750(720)/50p,

1125(1080)/29i, 1125(1080)/25i

525(480)/29i, 625(576)/25i

Horizontal Timing0 to 1 output line in steps of 1 pixel.

Delay Adjustment.....Horizontal and Vertical timing

Vertical Timing0 to 1 output frame in steps of 1 line.

Minimum DelayHD - 15 µs
SD - 42 µs

Video Delay.....HD - 1120 pixels to 11 Frames + 820 pixels
SD - 570 pixels to 11 Frames + 420 pixels

Internal audio processing....32 channels @ 24-bit

Embedded audio handling ..HD - 24-bit synchronous 48 kHz to SMPTE 299M
SD - 20-bit synchronous 48 kHz to SMPTE 272M-A

Audio Resolution Inputs: 32 kHz/44.1 kHz/48 kHz synchronous or asynchronous to video stream. Outputs: 48kHz synchronous to the video stream. Up to 24-bit, (20 MSBs embedded in SD-SDI stream).

Audio Delay Minimum: 0.75 ms for data signals and embedded input pairs; 3 ms for AES pairs
Maximum 2.5 s

Power Consumption

Module Power Consumption

TBA

IQDBD00 Connections

This section describes the physical input and output connections used by the IQDBD00.

Input Connections

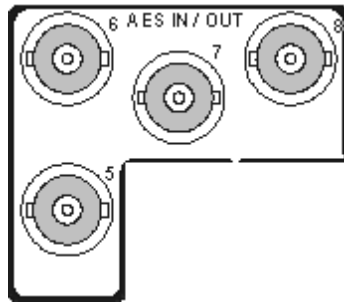
Serial IN (IQDBD0043-2A and IQDBD0044-2A)

Serial digital input connection to the module is made by means of a BNC connector, which terminates in 75 ohms.



AES Unbalanced IN/OUT (IQDBD0043-2A)

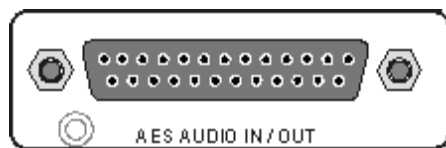
Connections 5 to 8 can be configured as either AES unbalanced inputs or outputs.



AES Balanced IN/OUT (IQDBD0044-2A)

The 25-way D type connector provides 4 AES Balanced outputs and 4 configurable AES Balanced input/output connections.

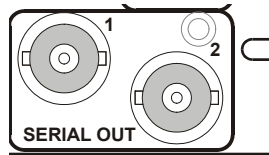
For connection details please see page 13.



Output Connections

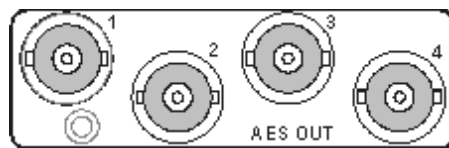
Serial Digital Video Outputs (IQDBD0043-2A and IQDBD0044-2A)

The unit has two serial digital video output connections, made by means of BNC connectors for 75 ohms.



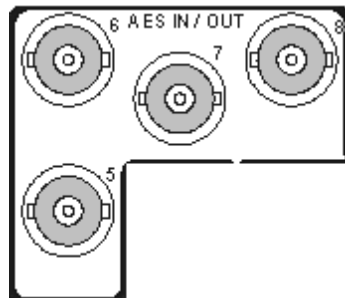
AES Unbalanced OUT (IQDBD0043-2A)

AES OUT connections 1 to 4 provide AES unbalanced output.



AES Unbalanced IN/OUT (IQDBD0043-2A)

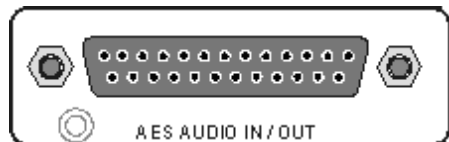
Connections 5 to 8 can be configured as either AES unbalanced inputs or outputs.



AES Balanced IN/OUT (IQDBD0044-2A)

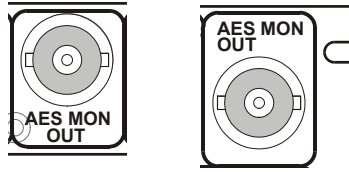
The 25-way D type connector provides 4 AES Balanced outputs and 4 configurable AES Balanced input/output connections.

For connection details please see page 13.

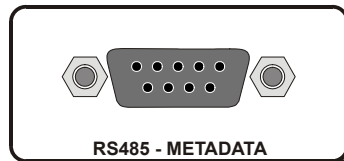


**AES MON OUT
(IQDBD0043-2A and IQDBD0044-2A)**

This is an AES monitoring output via a BNC connector for 75 ohms.

**RS485 DolbyE Metadata Output
(IQDBD0043-2A and IQDBD0044-2A)**

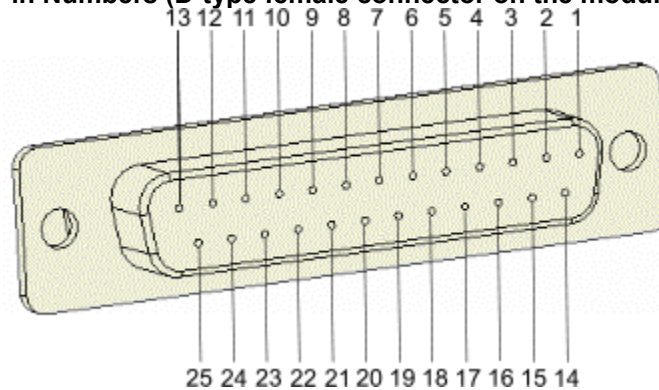
DolbyE metadata output is made via the RS485 9 pin D-type connector.



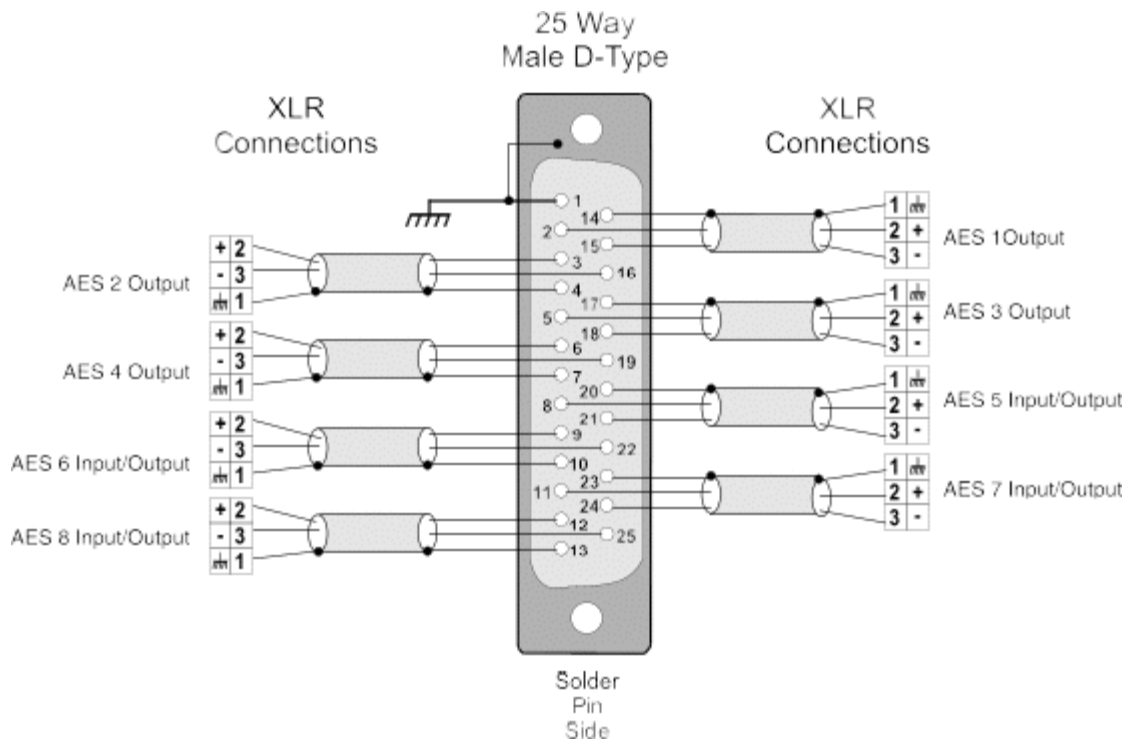
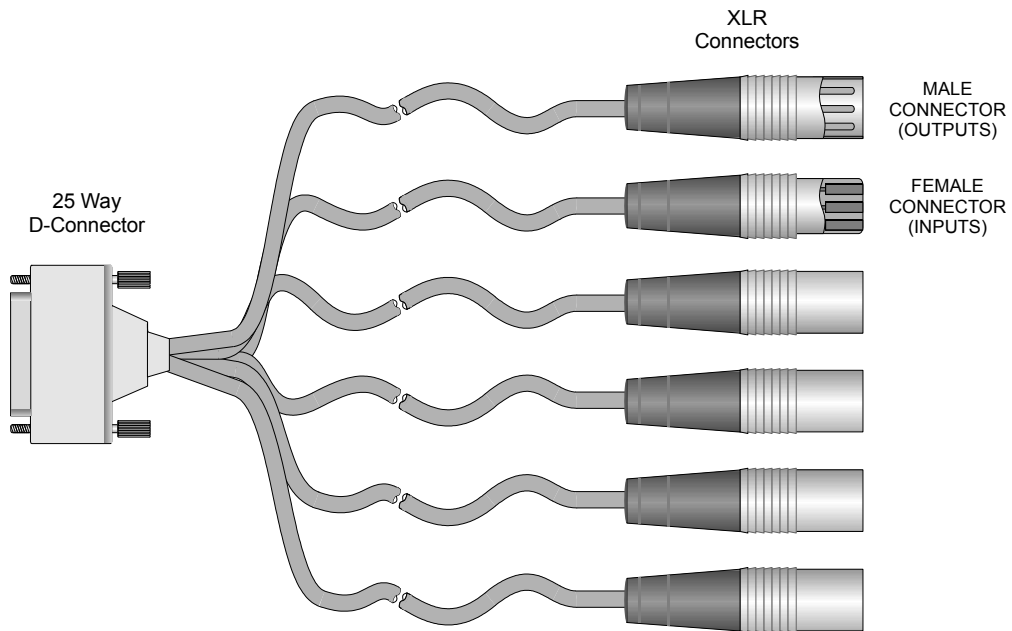
25 WAY D-Type Connection Details

By Pin Number			By Function		
Pin No	Description	Connection	Pin No	Description	Connection
1	Chassis Ground	Ground	1	Chassis Ground	Ground
2	Channel 1 +	AES 1+	2	Channel 1 +	AES 1+
3	Channel 2 +	AES 2+	15	Channel 1 –	AES 1-
4	Ground (2)	Screen 2	14	Ground (1)	Screen 1
5	Channel 3 +	AES 3+	3	Channel 2 +	AES 2+
6	Channel 4 +	AES 4+	16	Channel 2 –	AES 2-
7	Ground (4)	Screen 4	4	Ground (2)	Screen 2
8	Channel 5 +	AES 5+	5	Channel 3 +	AES 3+
9	Channel 6 +	AES 6+	18	Channel 3 –	AES 3-
10	Ground (6)	Screen 6	17	Ground (3)	Screen 3
11	Channel 7 +	AES 7+	6	Channel 4 +	AES 4+
12	Channel 8 +	AES 8+	19	Channel 4 –	AES 4-
13	Ground (8)	Screen 8	7	Ground (4)	Screen 4
14	Ground (1)	Screen 1	8	Channel 5 +	AES 5+
15	Channel 1 –	AES 1-	21	Channel 5 –	AES 5-
16	Channel 2 –	AES 2-	20	Ground (5)	Screen 5
17	Ground (3)	Screen 3	9	Channel 6 +	AES 6+
18	Channel 3 –	AES 3-	22	Channel 6 –	AES 6-
19	Channel 4 –	AES 4-	10	Ground (6)	Screen 6
20	Ground (5)	Screen 5	11	Channel 7 +	AES 7+
21	Channel 5 –	AES 5-	24	Channel 7 –	AES 7-
22	Channel 6 –	AES 6-	23	Ground (7)	Screen 7
23	Ground (7)	Screen 7	12	Channel 8 +	AES 8+
24	Channel 7 –	AES 7-	25	Channel 8 –	AES 8-
25	Channel 8 –	AES 8-	13	Ground (8)	Screen 8

Pin Numbers (D-type female connector on the module)

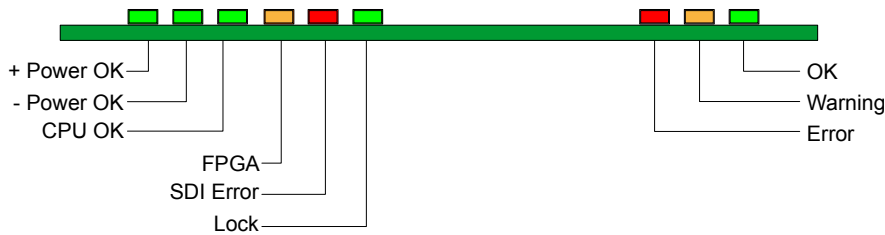


Example of Connection to XLR Connectors



Note: A line with a dot at the end (●—) indicates a connection to the cable screen.

Card Edge Indicators



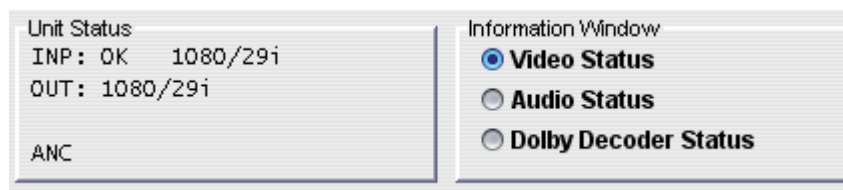
The LEDs on the edge of the IQDBD00 indicate the operating status of the module.

+ Power OK (Green)	Indicates that a positive power supply is present.
- Power OK (Green)	Indicates that a negative power supply is present.
CPU OK (Green)	Indicates that the CPU is running.
FPGA (Orange)	Flashes when the FPGA has been loaded.
SDI Error (Red)	This will become illuminated if any CRC, EDH or TRS errors are detected on the SDI input. It is also illuminated when the input is lost.
Lock (Green)	This LED is continuously illuminated when the unit is locked to its input. When the input is lost and the default output is selected as Input , this LED is extinguished and the Warning LED will illuminate.
Error (Red)	Indicates that there is a board internal fault and a service is required.
Warning (Orange)	When illuminated, this will indicate that there is an operational problem that could affect the output picture. This could be: <ul style="list-style-type: none"> • Embedder Status - ANC overflow. The embedder has run out of ancillary space over quite a few lines. Due to heavy use of ancillary space on the video input, particularly in SD. • Input Video - Incompatible input standard. Detected input standard is invalid. • Input Video - SDI problem. CRC or other SDI errors detected on selected input in the last whole field. • Reference - Lock Failure. Genlock failed to lock. Will generally become illuminated in transitional states like standard changes, but continuous illumination indicates a problem. More information is available in the status window.
OK (Green)	Indicates that the module is operating correctly.

Controlling the IQDBD00 from the RollCall Control Panel

The Information Window

The information window appears in the upper-right corner of each screen and displays basic information about the audio and video status of the unit.



Select either **Video Status**, **Audio Status**, or **Dolby Decoder Status** to display the corresponding information.

The following **Video Status** information is displayed:

Line One	
Displays the status of the video input, followed by the standard of the input, or last valid signal.	
INP:OK	The unit is receiving a valid input signal of a recognized standard.
INP:LOST	The unit is not receiving an input signal.
INP:FAIL	The unit is receiving an input signal, but the signal is not valid. For example, the frame rate differs between the input and output standards.
MISM	The unit is receiving a valid input signal, but in a different format to the selected output standard. For example, in 1080/29i, out 1080/29p.
Line Two	
Displays the operating standard of the unit.	
Unknown	The input signal standard is not recognized.
1080/29i (For example)	The operating standard
Line Three	
Displays the status of the ancillary data and the output picture,	
ANC	Ancillary present.
FRZ	Output frozen.
PAT	Output pattern.
MON	Monochrome.
HBL	Horizontal ancillary data is being blanked.
BLK	Output black.

When **Audio Status** is selected, the status of the embedded audio input is displayed where:

P	Indicates that the channel is a PCM audio input.
–	Indicates that no audio input is detected
D	Indicates that the signal is data (non-pcm, Dolby, etc...)
o	Indicates that the AES port is configured as an output.
CLIP	The word Clip will appear if any audio level reaches the 0dBFD point, i.e. the digital limit. The fader level should be reduced to prevent this occurring.
ANC FULL	Indicates the embedder / ancillary formatter has run out of ancillary space over quite a few lines and has thus lost data. Due to heavy use of ancillary space on the video input, particularly in SD.

When **Dolby Decoder Status** is selected, the following information is displayed:

Line One Displays the Dolby type and bit depth.	
Dolby D/E	Indicates whether the current Dolby signal is Dolby D or Dolby E.
16/20/24	Indicates the bit depth.
Line Two Indicates the Dolby program configuration.	
E.g. 5.1+2	This example shows a 5.1 channel program and a two channel secondary audio program (SAP).
Line Three Displays the current Dolby decoder status and video frame synchronization.	
<ul style="list-style-type: none"> • Status:OK • Status:Err • VFS:Val • VFS:Inv 	
Line Four Displays the Dolby Decoder output	
DDout:	
<ul style="list-style-type: none"> • P for PCM output • – for no output • A for aux channel 	
aux:	
<ul style="list-style-type: none"> • A indicates that auxiliary channel is being used. • CH4 indicates that channel 4 is being used. 	

Video Input Settings

Use the settings on the Video Input screen to specify the:

- Input standards that the unit will accept as valid.
- Condition that must exist in order for the unit to consider input to have been lost.

Additionally, this screen displays information about CRC/EDH.

The screenshot shows the 'Video Input' settings screen for unit IQDBD00. The top status bar displays 'IQDBD00 0000:06:01 - IQDBD00'. On the left is a navigation menu with options: Video Input (selected), Video Output, AES Proc.Amps 1-4, AES Proc.Amps 5-8, and Emb. Proc.Amps Groups 1-2. The main area is divided into several sections:

- Unit Status:** INP: OK 625/25i, OUT: 625/25i, ANC.
- Information Window:** Radio buttons for Video Status (selected), Audio Status, and Dolby Decoder Status.
- Input loss if:** Radio buttons for No Input and In/Out std. Mismatch (selected).
- Valid Input Standards:** A grid of checkboxes for various standards:

<input checked="" type="checkbox"/> 1080/23p	<input checked="" type="checkbox"/> 1035/29i	<input checked="" type="checkbox"/> 1035/30i
<input checked="" type="checkbox"/> 1080/23sF	<input checked="" type="checkbox"/> 1080/29p	<input checked="" type="checkbox"/> 1080/30i
<input checked="" type="checkbox"/> 1080/24p	<input checked="" type="checkbox"/> 1080/29i	<input checked="" type="checkbox"/> 1080/30p
<input checked="" type="checkbox"/> 1080/24sF	<input checked="" type="checkbox"/> 525/29i	<input checked="" type="checkbox"/> 720/59p
<input checked="" type="checkbox"/> 1080/25i	[Set All]	
<input checked="" type="checkbox"/> 1080/25p	[Clear All]	
<input checked="" type="checkbox"/> 625/25i	[720/60p]	
	[720/50p]	
- CRC/EDH Errors:**
 - CRC/EDH Errors: 406
 - Time Since Last Error: 0:23
 - ANC Errors: 115
 - Time Since Last Error: 0:23
 - Reset Counts: 3:35

Specifying Valid Input Standards

The input signal standards to be considered valid are defined on this screen.

To specify valid input standards:

In the **Valid Input Standards** section, select the input standards that will be accepted as valid.

To select all available standards, click **Set All**.

To deselect all available standards, click **Clear All**.

Note: *If any standards, other than those listed on this page are detected, they will be considered to be invalid.*

Specifying Input Loss Conditions

The module can be configured to take automatic action if the input signal is lost. The conditions that the module considers to be a loss of input are defined on this screen.

To specify the condition that will be considered an input loss:

In the **Input Loss if** section, select either:

- **No Input:** No input of any type is considered a loss of input.
- **In/Out std. Mismatch:** Input signal not matching the output standard is considered a loss of input.

Viewing CRC and EDH Errors

The **Video Input** screen also displays Cyclic Redundancy Checksum and Error Detection Handling errors.

The following CRC and EDH Error information is displayed in the **CRC/EDH Errors section**:

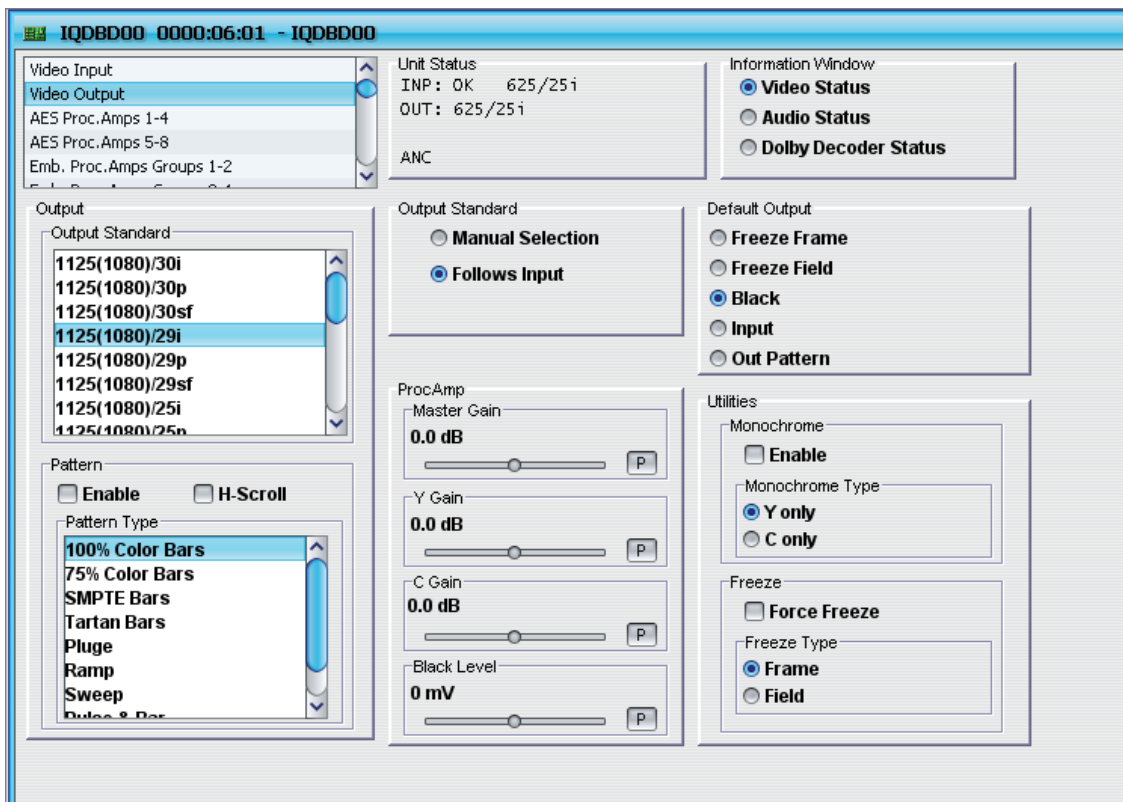
- **CRC/EDH Errors:** Displays the number of CRC and EDH errors that have occurred since the last reset.
- **Time Since Last Error:** Displays the time (in seconds) since the last CRC or EDH error was detected.
- **ANC Errors:** This displays the number of ANC errors since the last reset.

To reset the error counts to zero, click **Reset Counts**. **Time Since Reset** displays the time since the error counts were last reset.

Video Output Settings

Use the settings on the **Video Output** screen to:

- Specify the video output standard.
- Set up the test pattern, if any, to be generated.
- Specify the default output to be generated in response to a loss of input.
- Adjust the signal gain and black levels.
- Control the unit's monochrome functions.
- Control the unit's picture freeze function.



Specifying the Video Output Standard

There are two ways to specify the video output standard to be used by the unit. The video output can be set to follow the input, so that the output standard will always be the same as the input standard, or manually selected. When manually selected, the output standard will always be the one specified, regardless of the input signal.

To allow the output standard to follow the input standard:

In the **Output Standard** section, select **Follows Input**.

To manually specify the output standard:

1. In the **Output Standard** section, select **Manual Selection**.
2. From the **Output Standard** list, select the output standard to be used.
The available output standards are:

1125(1080)/30i	1125(1080)/25p	1125(1035)/29i
1125(1080)/30p	1125(1080)/25sf	750(720)/60p
1125(1080)/30sf	1125(1080)/24p	750(720)/59p
1125(1080)/29i	1125(1080)/24sf	750(720)/50p
1125(1080)/29p	1125(1080)/23p	525(480)/29i
1125(1080)/29sf	1125(1080)/23sf	625(576)/25i
1125(1080)/25i	1125(1035)/30i	

Specifying a Test Pattern

The **Video Output** screen also controls the test pattern. The test pattern options are to:

- Enable or disable the test pattern.
- Use a horizontal scrolling test pattern.
- Specify the test pattern to be used.

To enable or disable the test pattern:

In the **Test Pattern** section:

- Select **Enable** to enable the test pattern function.
- Clear **Enable** to disable the test pattern function.

To use a horizontal scrolling test pattern:

In the **Test Pattern** section, select **H-Scroll**.

When selected, the specified test pattern will scroll from left to right at a fixed rate. This mode is useful for differentiating between an input test pattern signal that has become a frozen picture (caused by loss of the input signal) and a test pattern that has been chosen as the output.

To specify the test pattern to be used:

From the **Pattern Type** list, select the pattern. The following pattern types are available:

100% Color Bars	75% Color Bars	SMPTE Bars
Tartan Bars	Pluge	Ramp
Sweep	Pulse & Bar	Burst
Black		

Specifying the Video Output

The **Default Output** setting defines the output that the unit generates in response to a loss of video input as specified by the **Input Loss If** setting on the **Video Input** screen.

To specify the default output:

In the **Default Output** section, select one of the following:

- **Freeze Frame:** On loss of input, the output becomes a non-interpolated frozen frame picture. All HANC and VANC are blanked.
- **Freeze Field:** On loss of input, the output becomes a frozen field one picture. All HANC and VANC are blanked.
- **Black:** On loss of input, the picture cuts to black.
- **Input:** The incoming signal will be displayed when ever possible.
- **Out Pattern:** On loss of signal, the specified test pattern is displayed.

Adjusting Gain and Black Levels

Use the **ProcAmp** functions to adjust the gain and black levels of the signal.

To adjust the gain and black levels:

In the **ProcAmp** section, adjust the following settings as required:

- **Master Gain:** Adjusts the overall gain (Y and Cb/Cr) over a range of ± 6 dB in steps of 0.1 dB. The preset value is 0.0 dB.
- **Y Gain:** Adjusts the Y (luminance) gain over a range of ± 6 dB in steps of 0.1 dB. The preset value is 0.0 dB.
- **C Gain:** Adjusts the Cb/Cr (color difference) gain over a range of ± 6 dB in steps of 0.1 dB. The preset value is 0.0 dB.

Note: *The total range of both Master + Y and Master + C controls is +6dB in steps of 0.1 dB. Preset value is 0dB.*

- **Black Level:** Adjusts the black level over a range of ± 200 mV in steps of 1 mV. The preset value is 0 mV.

Controlling Monochrome Functions

The unit can be set so that the output picture becomes monochrome using either the Y or Cb/Cr components of the signal.

To output a monochrome signal:

1. In the **Monochrome Section**, select one of the following:
 - **Y only**: The output picture becomes monochrome using only the Y component of the signal.
 - **C only**: the output picture becomes monochrome using only the Cb/Cr component of the signal.
2. Select **Enable**.

Force Freezing the Picture

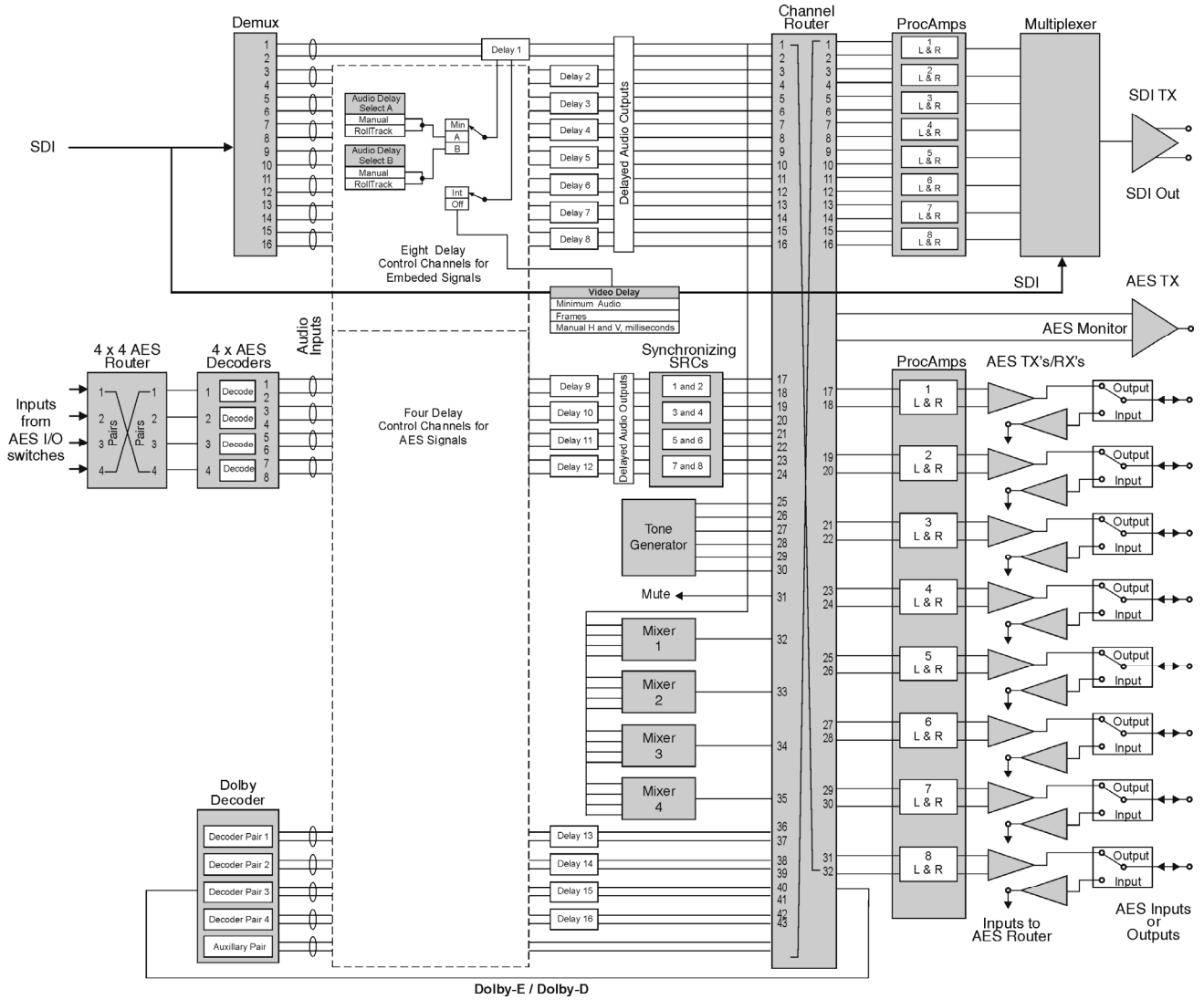
The unit can impose a force freeze on the picture. When this is done, the output becomes a frozen frame or frozen field.

To force freeze the output:

1. In the **Freeze Type** section select either
 - **Frame**: The picture becomes a frozen frame.
 - **Field**: The picture becomes a frozen field.
2. Select **Force Freeze**.

Audio Processing Overview

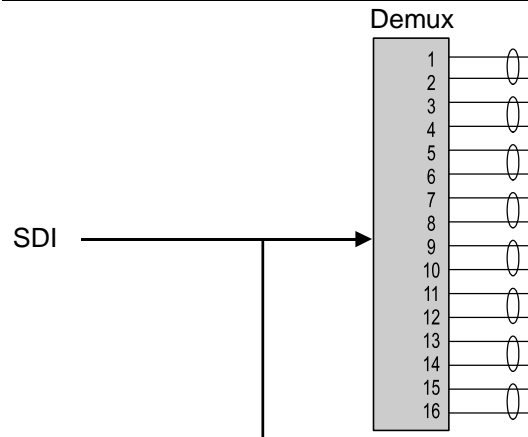
Audio Processing Block Diagram



SDI Demultiplexer

The audio channels are demultiplexed from the SDI signal as eight 2-channel pairs that are feed to the audio delay processor.

Note: *Note that embedded audio handling for HD is 24 bit synchronous 48 kHz to SMPTE 299 and SD is 20-bit synchronous 48 kHz to SMPTE 272M-A.*

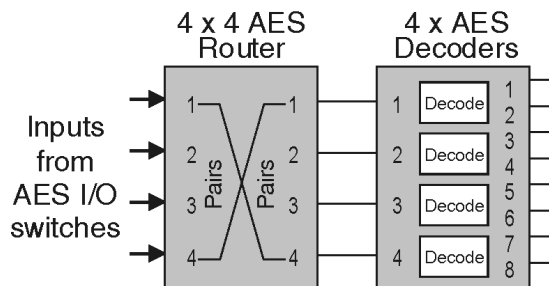


AES Processing

The AES ports may be configured as Inputs or Outputs. When a port is configured as an input the AES signal is passed to one of the inputs of the 4 x 4 AES router.

The router allows any of the 4 input pairs to be selected to appear on any of the 8 outputs.

The 4 pairs from the router are then fed to 4 decoders to produce four 2 channel pairs.



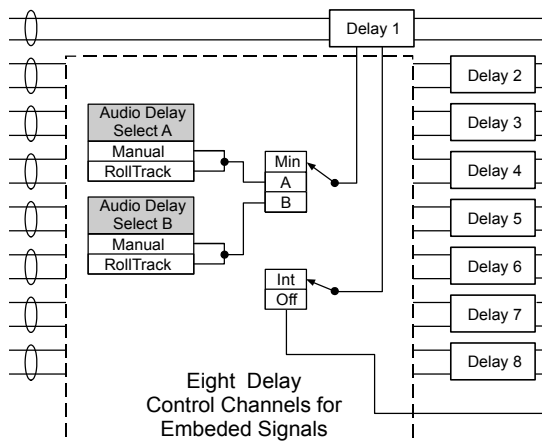
Audio Delay Processor

The SDI de-multiplexed and decoded AES signals and Dolby decoded signals (eight 2 channel pairs) are fed to sixteen separate delay blocks. Each block processes one audio pair (2 channels) at the same time, the delay time being the same for both channels.

The delay time may be set to Min (the minimum audio delay of approx. 0.75 ms or 3.0 ms if via a SRC) or to a time set by one of two (A and B) Audio Delay Selectors. A delay equal to the unit's video delay may also be added by selecting Int.

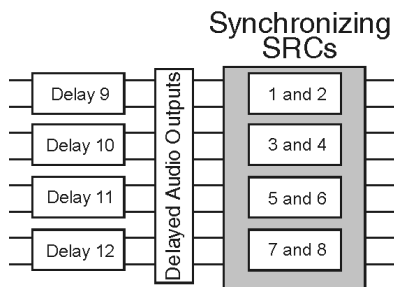
The delay time set by each of these selectors is controlled by a combination of the following:

- Manual** A delay set by the Manual Delay control will be applied.
- RollTrack** A delay set by external RollTrack commands will be applied.



Synchronizing SRCs

The outputs of delay blocks 1 to 8 (SDI de-multiplexed) are connected directly to inputs 1 to 16 of the channel router. The outputs of delay blocks 9 to 16 (Decoded AES) are passed through eight 2 channel synchronizing Sample Rate Converters to the inputs of the channel router.

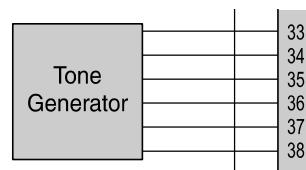


Tone Generator

A tone generator provides the following signals to the channel router.

- 1kHz at 20dBFS
- 2kHz at 20dBFS
- 4kHz at 20dBFS
- 1kHz at 18dBFS
- 2kHz at 18dBFS
- 4kHz at 18dBFS

Note: *The frequency and level of these signals is not adjustable.*



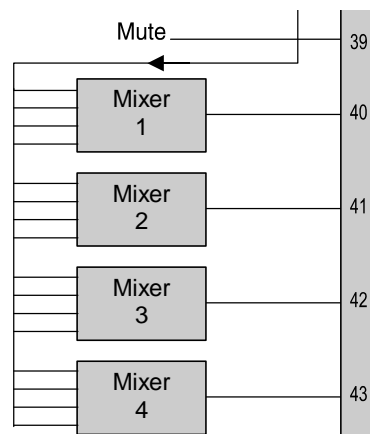
Mixers 1 to 4

There are four separate 4 input, 1 output audio mixers available.

Each mixer has four inputs with individual gain controls that allow the mixing levels for each of the input signals, to be adjusted.

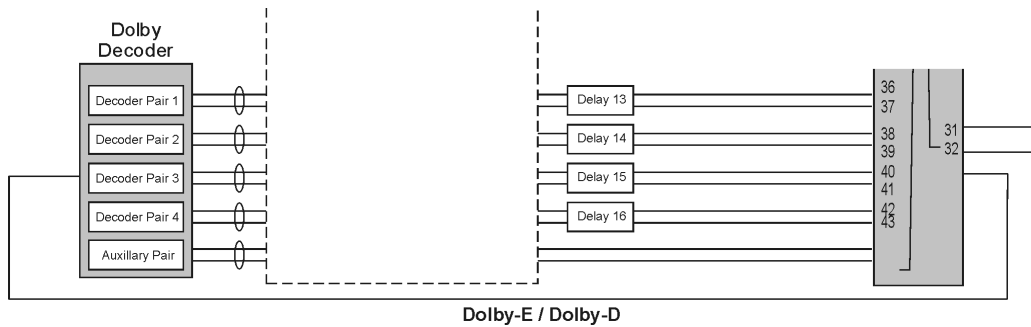
The inputs may be any of the following:

- Mute
- Disembed 1 to 8 Left/Right
- AES Bus 1 to 8 Left/Right
- Tone Generator outputs



Dolby Decoder

Dolby E or Dolby D source is passed from either one of the AES inputs or from embedded Dolby E in the SDI video stream.

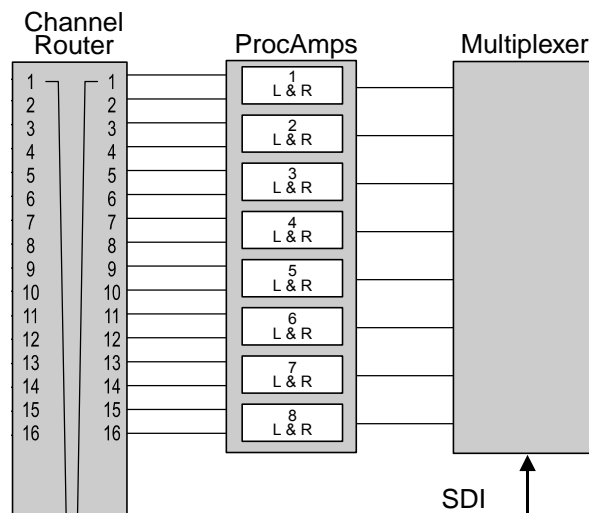


Embedder ProcAmps

There are eight 2 channel ProcAmps that receive their inputs from the output of the channel router.

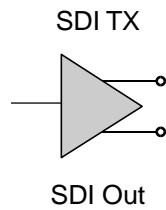
The two channels are processed as Left and Right with control of Gain, Inversion, Muting and a Stereo mode.

The outputs are configured as eight 2 channel signals in four groups and these are fed to the Multiplexer where they are embedded onto the SDI signal.



SDI TX

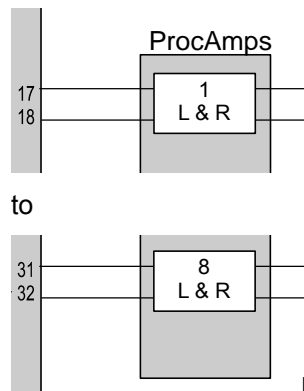
The embedded SDI signal is fed to the output amplifier that provides two SDI outputs.



AES ProcAmps

There are eight 2 channel ProcAmps that receive their inputs from the output of the channel router.

The two channels are processed as Left and Right with control of Gain, Inversion, Muting and a Stereo mode.



AES TX/RXs

There are eight AES transmitters and four AES receivers.

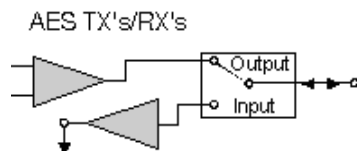
The AES ports 5-8 may be configured as inputs or outputs by setting the Input/Output switch.

When an AES Port is configured as an Output:

- The outputs from the AES ProcAmps are fed to the AES transmitter to provide an AES output.

When an AES Port is configured as an Input:

- The AES input signal is passed through an AES receiver and fed to an input of the AES router.

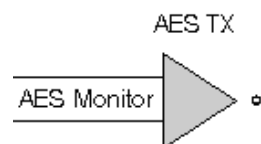


AES Monitor

An AES monitoring output is provided. This is an unprocessed signal derived from the output of the channel router.

Signal selections are as follows:

- Mute
- Dolby decoded pair 1 to 4/Aux
- Disembed 1 to 8 Left/Right
- Mixer 1 to 4 output
- AES Bus 1 to 8 Left/Right
- 1kHz -20dBFS Tone
- 2kHz -20dBFS Tone
- 4kHz -20dBFS Tone
- 1kHz -18dBFS Tone
- 2kHz -18dBFS Tone
- 4kHz -18dBFS Tone



Dolby E Audio Handling

This section describes how the IQDBD handles Dolby E audio. For a more general description of Dolby E and AC-3, see page 84.

Dolby E and Dolby Digital are registered trademarks of Dolby Laboratories, Inc.

Dolby E audio achieves compatibility with a digital video stream by virtue of a few fundamental features:

- The compressed audio data is broken into frame-duration packets to match the video frames it will be embedded in.
- Each frame is separated from neighboring frames by a guardband, which allows for clean video-type switching.
- Concealment of switches is possible at a decoder, based on redundant audio and block counts in each frame.

However, there is a burden in dealing with a number of parallel frame-based carriers (such as Dolby E) alongside, and often within, the video. Namely:

- Each Dolby E stream, within an AES-3 / SMPTE-337 data pair, must be synchronous and aligned with the video to allow a video stream to be switched or recorded cleanly. Incorrect alignment can lead to muting, or even high amplitude noise bursts.

As with SMPTE RP-168 switching, there is a small alignment tolerance. Upstream switched, or externally provided sources can present problematic deviations from the preferred Dolby E frame position.

- Each video standard has its own specified Dolby E frame alignment point. Thus, a standards-agile broadcast environment must arrange for correct referencing and delay adjustment in each standard.
- The embedding of audio within the video stream adds considerable timing uncertainty, and increases susceptibility to external factors such as loading of ancillary space and audio sample distribution produced by equipment in the video chain. In particular, standard definition is more prone to these issues than high definition.

The IQDBD is well equipped to deal with issues of carrying Dolby E audio within a broadcast environment. It has the tools required for the correct and easy handling of Dolby E in the following features:

- Transport enablers such as SMPTE-337 aware routing and channel status passing.
- Monitoring and logging of Dolby E frame timing for both passed and newly-embedded audio.
- Manual alignment facilities for consistently arranged environments.
- Automatic realignment of Dolby E frames to preferred or non-standard positions.

For more information about the Dolby E controls, refer to the following sections:

- Embed On/Off on page 48.
- Video Delay/Dolby E on page 38.
- O/P Dolby E Logging on page 67.

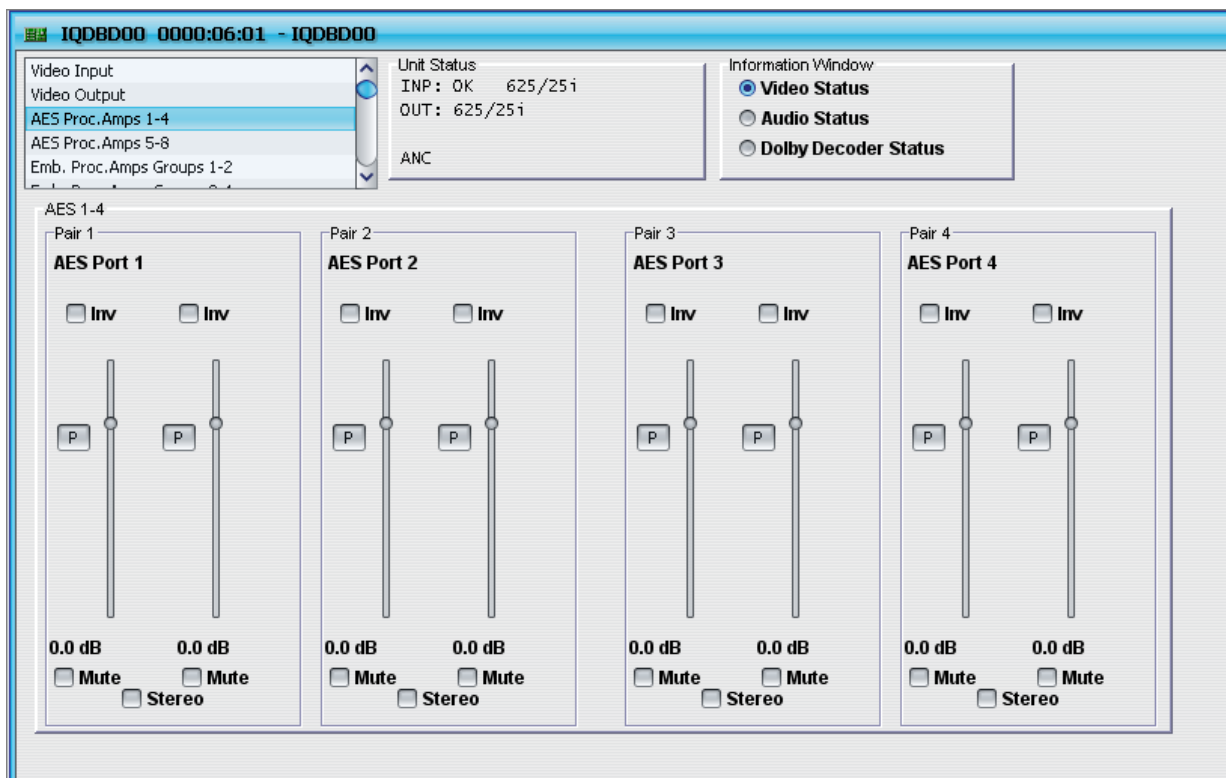
AES ProcAmp Settings

Use the settings on the **AES ProcAmps 1-4** and **AES ProcAmps 5-8** screens to control the units AES audio output. The channels are grouped into eight output pairs.

Note: *The ProcAmps are not available when the AES port is set as an input, or when the audio routed to that port is data.*

For both AES and embedded audio, the unit will either recognize the input as data only, or the input channels can be forced to act as data if they are data. The channels that have these inputs routed to their inputs will have their ProcAmp menus disabled.

For more information about how audio is processed by the module, refer to the Audio Processing Overview on page 23.



Controlling AES Output

To adjust the AES output:

For each pair in use, specify the following settings as required:

- **Inv:** Select this option to invert the signal polarity of the channel.
- **Mute:** Select this option to mute the channel.
- **Stereo:** Select this option to link the two channels in the pair together as a stereo pair. Any adjustments made to one channel in the pair will be applied to the other.
- **Gain Controls:** The gain controls adjust the gain of each channel, or stereo pair, over a range of +12 dB to -36 dB in 0.1 dB steps. The preset value is 0dB.

Embedded ProcAmp Group Settings

The 16 embedded audio channels are arranged into four channel groups, of two channel pairs each.

Use the settings on the **Embedded ProcAmps Groups 1-2** and **Embedded ProcAmps Groups 3-4** pages to adjust the gain, apply phase inversion, or mute any of the 16 embedded audio channels.

Note: *These controls are not available when an output is defined as data, the input is data, or the audio embedder is disabled.*

For more information about how audio is processed by the module, refer to the Audio Processing Overview on page 23.



Adjusting Embedded Audio Settings

To adjust embedded audio settings:

For each channel in use, specify the following settings as required:

- **Inv:** Select this option to invert the signal polarity.
- **Mute:** Select this option to mute the channel.
- **Stereo:** Select this option to link the two channels in the pair together, as a stereo pair. Any adjustments made to one channel in the pair will be applied to the other.
- **Gain Controls:** The gain controls adjust the gain of each channel, or stereo pair, over a range of +12 dB to -36 dB in 0.1 dB steps. The preset value is 0dB.

Audio Mixer Settings

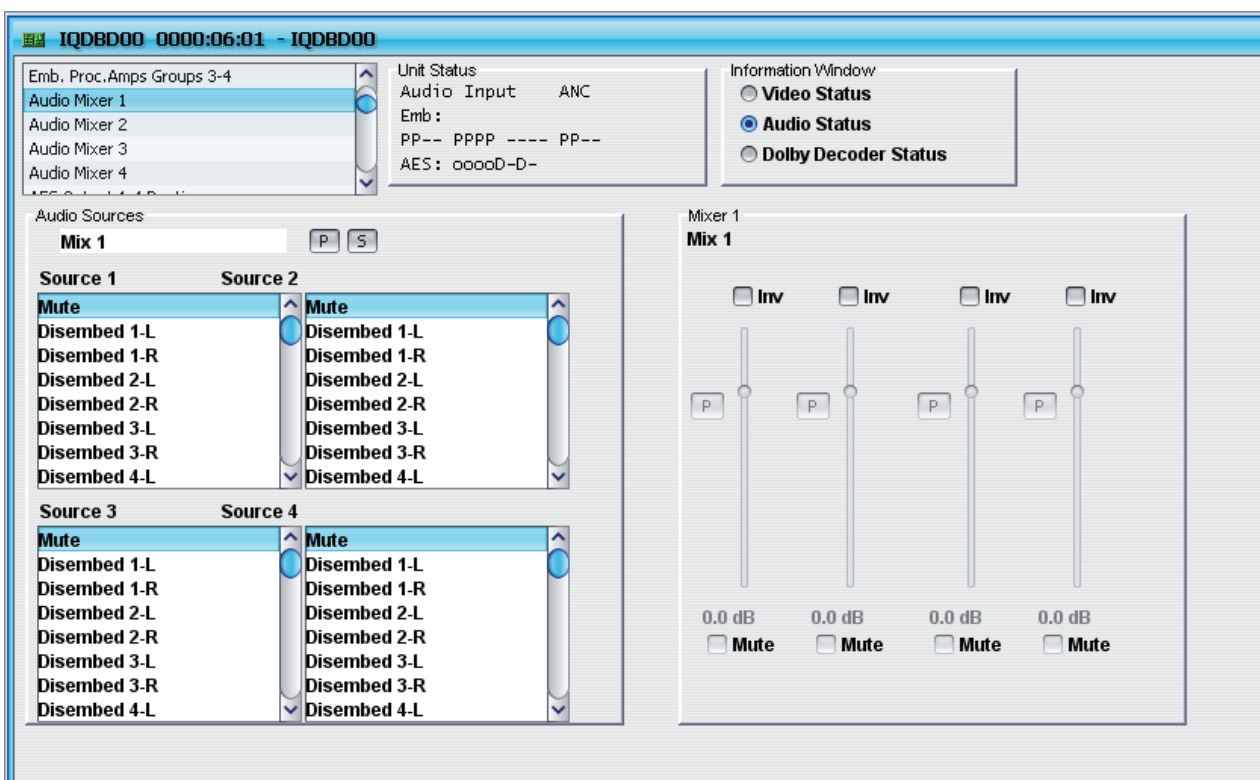
The audio mixers create new, mixed audio feeds from the incoming channels. This is useful when, for example, incoming stereo feeds are required in mono, or discrete surround sound feeds are required in stereo.

Not only can the incoming feeds be mixed together, but the exact balance can be set using the faders. This allows finer control over the resulting sound than a simple mono function.

Additional applications include mixing together incoming audio with local foreground, such as adding commentary to a sports feed. The invert function can be applied to a channel, to create a mix-minus channel for foldback.

For more information about how audio is processed by the module, refer to the Audio Processing Overview on page 23.

To access these settings, select **Audio Mixer 1**, **Audio Mixer 2**, **Audio Mixer 3**, or **Audio Mixer 4** from the function list.



Adjusting Audio Mix

Each of the four mixers has four inputs, with individual gain controls that allow the mixing level of each input signal to be adjusted. The settings for each mixer are the same, and as such, are only described once.

To adjust the audio mix

1. From the **Source** lists, select an input source.

Note: *If mute or one of the test tones is selected as the source, the corresponding gain control is disabled.*

2. Adjust the mix as follows:
 - **Inv:** Select this option to invert the signal.
 - **Mute:** Select this option to mute the channel.

- **Gain Controls:** The gain controls adjust the gain of each channel, or stereo pair, over a range of +12 dB to -36 dB in 0.1 dB steps. The preset value is 0dB.

Changing a Mixer Name

To change the name of a mixer:

1. In the text field at the top of the **Audio Sources** section, type the new mixer name..
2. Click the **S** button. The new name appears in the gain control section. To return to the default mixer name, click the preset button.

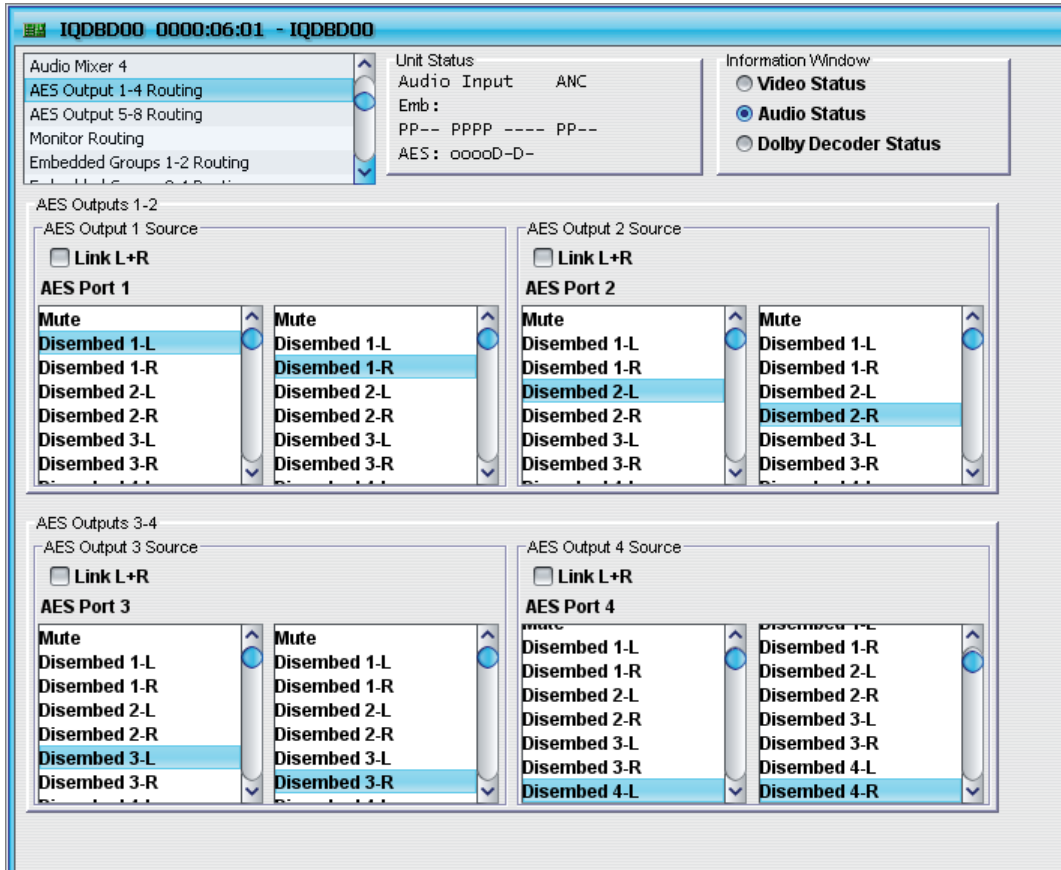
Clip Indicators

At the top of each fader, the word **Clip** appears briefly if the audio level reaches the 0 dBFS point – reduce the fader level to prevent this from happening.

AES Routing Settings

The **AES Output 1-4 Routing** and **AES Output 5-8 Routing** screens select the signal sources for the eight AES outputs.

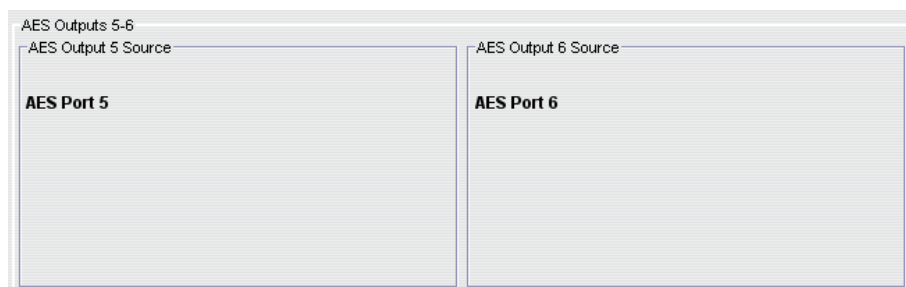
For more information about how audio is processed by the module, refer to the Audio Processing Overview on page 23.



Specifying AES Routing

For each channel within each pair, select the appropriate source. Select **Link L+R** to automatically pair a selected source with its corresponding left/right source, creating a stereo pair. This enables rapid selection in a stereo environment.

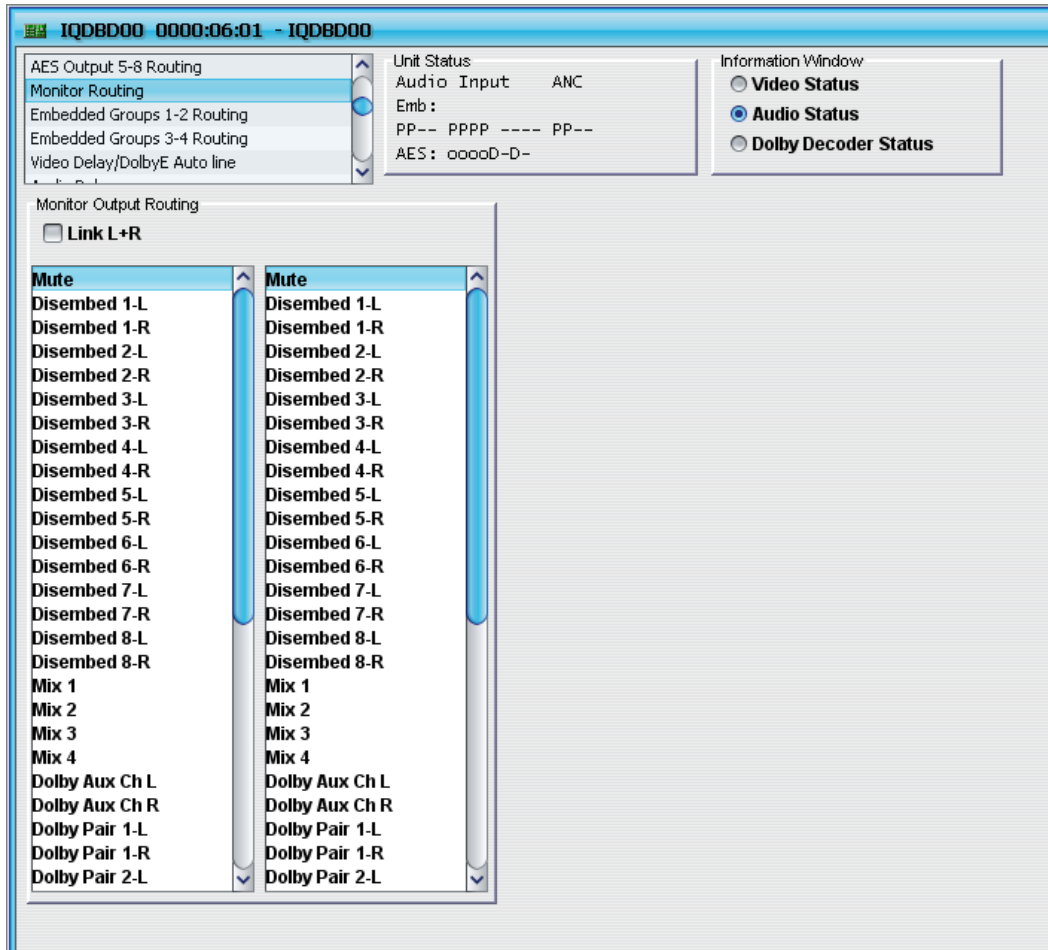
If any of the AES I/O ports are configured as inputs (on the AES I/O Port Setup screen), the entries on the AES Output 5-8 Routing screen will appear like this:



Monitor Routing Settings

Use **Monitor Routing** to select the signal sources for the additional AES output. This output can be used as an extra feed or a monitoring feed. Mute and audio tones can also be selected.

For more information about how audio is processed by the module, refer to the Audio Processing Overview on page 23.



Specifying Monitor Routing

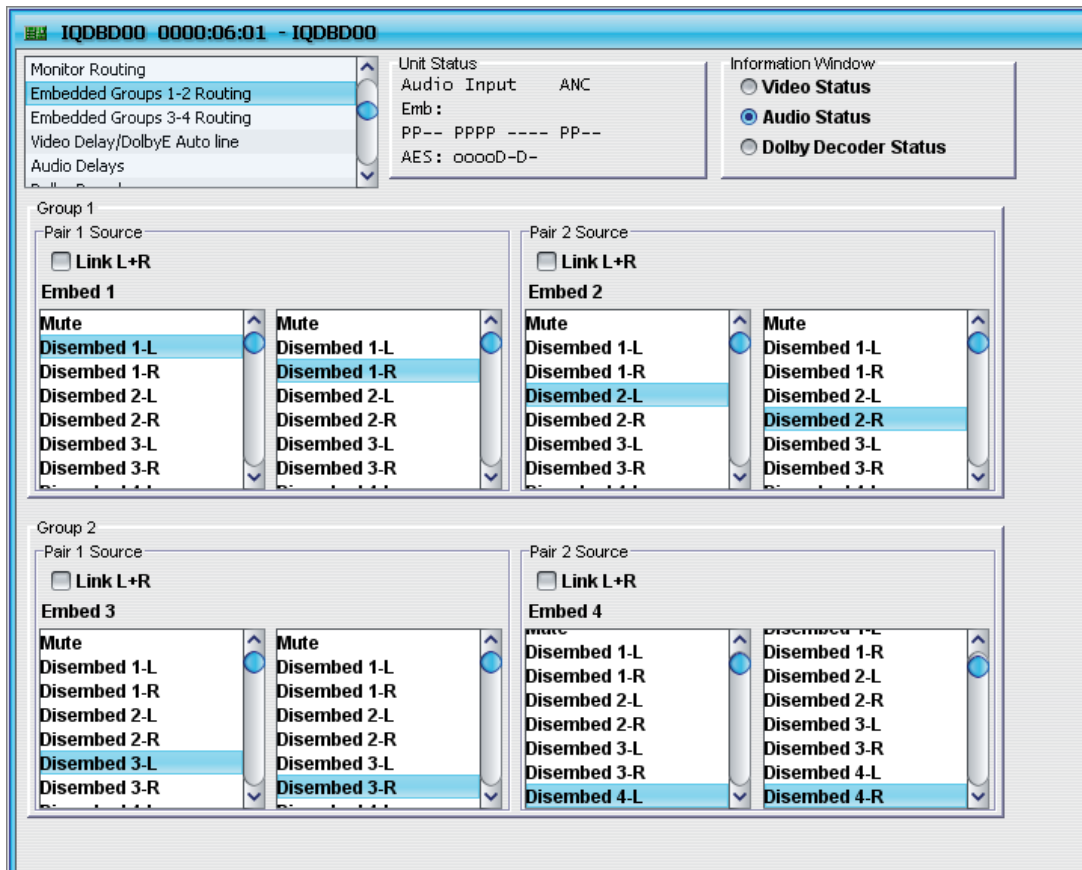
The procedure for specifying monitor routing is the same as for AES routing, described on the previous page.

Embedded Routing Groups Settings

Use settings on the **Embedded Routing Groups 1-2** and the **Embedded Routing Groups 3-4** screens to specify the signal sources for the SDI embedder. As well, mute and audio tones can be selected.

By default, there is a 1:1 relationship between the input source and the output. For example, channel one input is matched with channel one output, channel two input is matched with channel two output, and so on.

For more information about how audio is processed by the module, refer to the Audio Processing Overview on page 23.



Specifying Signal Sources

The settings for each source pair are the same, and as such, are only described once.

To specify a signal source:

For each source pair, select the required source; to configure the pair as stereo, select **Link L+R**.

Note: *A corresponding left/right source will be automatically chosen when available.*

If a mixer output is selected as a source, a different mixer output will automatically be chosen for the other channel. Mix 1 and 2, or mix 3 and 4 are paired.

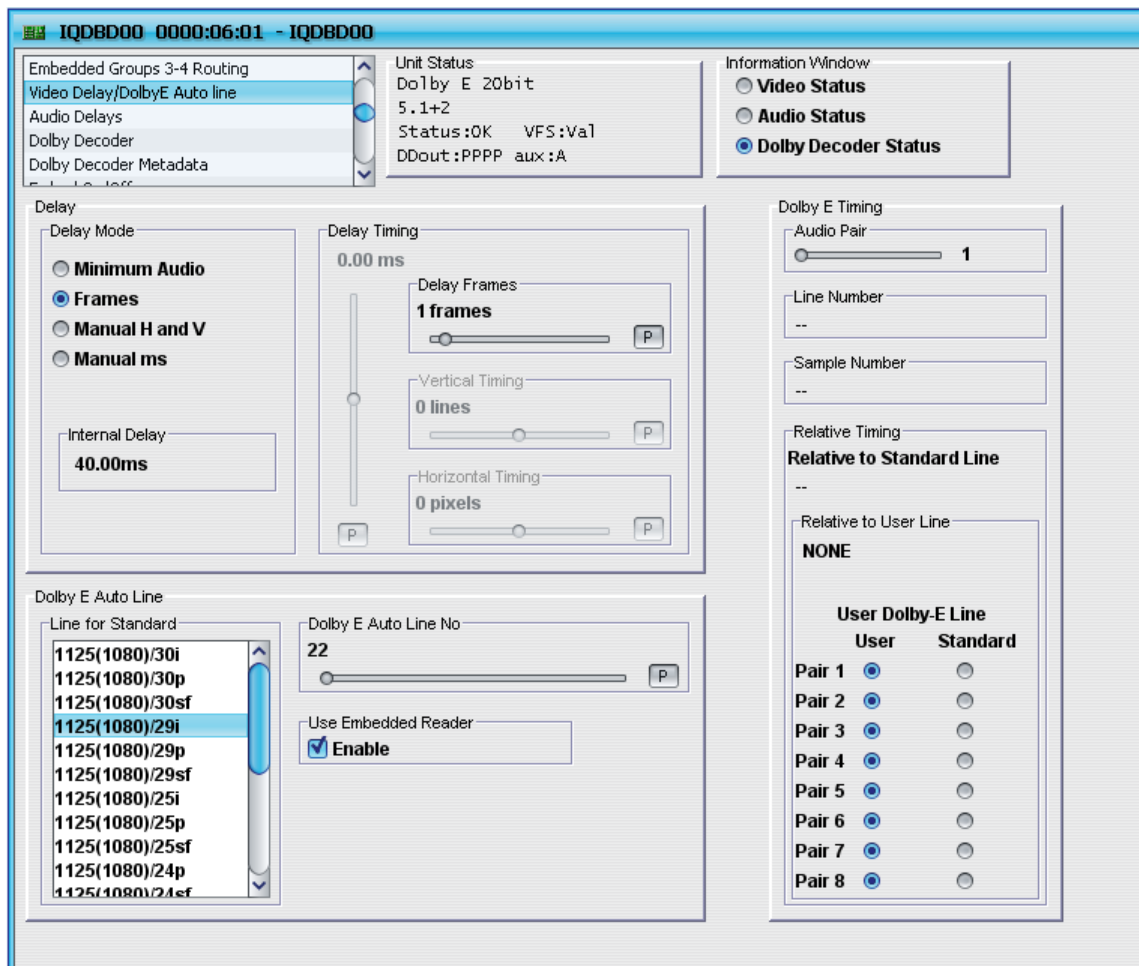
If a tone is selected as the source, the same tone will automatically be chosen for the other channel.

When an input pair is defined as data or mixed, an automatic pairing is established, similar to the Link L+R function. If either

channel of a data pair is selected, the other half will automatically follow in the correct L+R order.

Video Delay/Dolby E Auto Line

Use the Video Delay function to delay the video so that it is timed into the system, compensating for audio processing, or any other required delay. The delay can be defined in frames, lines and pixels, or in milliseconds. The total delay is cumulative, so that an accurate delay can be created. The total delay is displayed in the **Internal Delay** box in the **Video Delay / Dolby E** screen.



Video Delay Modes

There are four methods by which video delay can be specified:

- **Minimum Audio:** Sets the video I/O delay to a nominal 3 ms.
- **Frames:** Specifies the number of frames that the output signal will appear after the input signal.
- **Manual H and V:** Delays the video by changing the horizontal and vertical timing.
- **Manual ms:** Specifies the video delay in milliseconds (ms).

Internal Delay displays the total video delay applied.

Note: *When embedded audio is set to **Pass** or respective groups are not active on the **Embed On/Off** screen, the timing of the video to embedded audio will not be affected.*

Minimum Audio

The Minimum Audio option effectively provides zero delay but when no internal delay is added, it is limited to the minimum audio delay of approx. 0.75 ms or 3 ms if via a SRC.

This is a good choice when a simple mux or demux, or embedded audio shuffle is required because it allow the video and audio delays to be internally matched. When this option is selected all other timing controls become unavailable.

Frames

The **Frames** delay mode specifies the number of frames that the output signal appears after the input signal. This is a good choice for a timed video chain because using a multiple of whole frames may avoid the need to place a synchronizer later in the chain, even if the video standard changes (within the same frame rate family). All values other than 0 frames allow delay matching in the audio processing.

The range of adjustment is 0 to 11 frames in steps of 1 frame. The preset value is 1 frame.

Manual H and V

The **Manual H and V** delay mode adjusts the video delay by changing the horizontal and vertical timing. When a delay is required to be defined in terms of video lines, for example, to round up to the next frame boundary, this method can be used. However, it can introduce complications on changes of video standard, and the minimum delay can go below the 0.75 ms (3 ms for AES PCM pairs) audio minimum delay, which would prevent audio delay matching.

The range of adjustment for vertical timing is ± 1 output frame – 1 line in steps of 1 line. The preset value for vertical timing is 0 lines.

The range of adjustment for horizontal timing is ± 1 output line – 1 pixel in steps of 1 pixel. The preset value for horizontal timing is 0 pixels.

The total delay equals the sum of Delay Frames + Vertical Timing + Horizontal Timing. The sum of horizontal and vertical timing is post-limited to be under \pm a frame of the current standard. The overall effect being allow a delay of 1 frame – 5 lines, for example, to align an earlier 5-line delay device into the next frame boundary.

Note: *Horizontal and vertical timing can only be adjusted in the same direction. That is, either both positive, or both negative.*

Manual MS

The **Manual ms** delay mode specifies a delay in milliseconds. Use this method when a delay needs to be defined in terms of time, for example, to suit an audio or mixed interlace / progressive multi-standard environment. The overall effect is to allow a delay of 1 frame – 3 ms, to align an earlier 3 ms delay device to the next frame boundary.

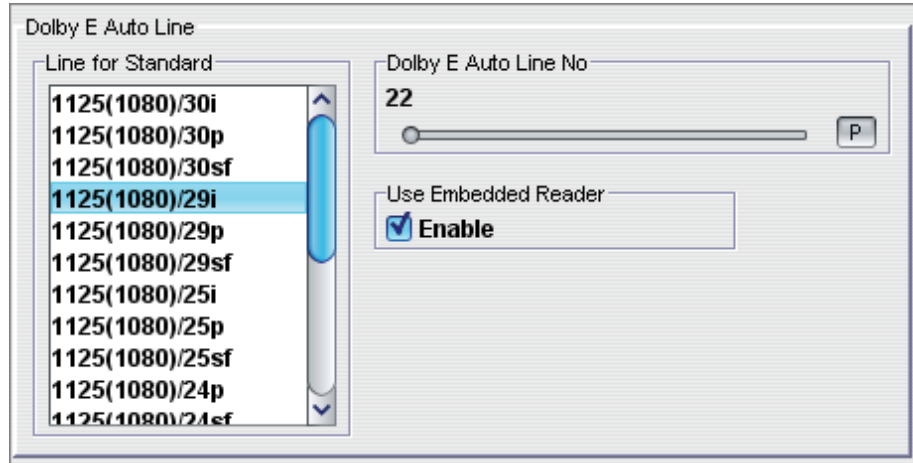
When a **Manual ms** delay is used, the **Vertical Timing** and **Horizontal Timing** controls become unavailable. However, the **Delay Frames** adjustment can still be used.

The total delay equals the sum of delay frame + the millisecond value specified.

Note: *The **Horizontal Timing** and **Vertical Timing**, even though they are disabled, will report this value in lines and pixels.*

Dolby E Auto Line

Use the settings in the Dolby E Auto Line section to specify a video line for Dolby E header alignment. For any pair with Auto DE enabled, the audio delay will attempt to align the output audio guardband on this video line, while keeping the delay within ± 0.5 frame of nominal. Each video standard may have its own line specified, so that a change of standard will automatically call up the appropriate line for alignment to.



Line for Standard / Dolby E Auto Line No

This allows access for defining and viewing the table of user specified lines for Dolby E auto alignment, status reporting and logging. Defaults for each standard are the published 525/29i and 625/25i positions mapped onto the supported standards, i.e. assuming the encoders / decoders are referenced to SD.

From the **Line for Standard** list, select the input video standard, this will automatically adjust the **Dolby E Auto Line No**, which can also be manually adjusted by means of the slider bar. The default Dolby E line numbers for the supported video standards are shown in the following table.

Line for Standard	Dolby E Auto Line No	Line for Standard	Dolby E Auto Line No
1125(1080)/29i	22	1125(1035)/29i	22
1125(1080)/29p	22	525(480)/29i	14
1125(1080)/29sf	22	625(576)/25i	11
1125(1080)/25i	20		
1125(1080)/25p	20		
1125(1080)/25sf	20		

Use Embedded Reader

This option allows the automatic Dolby E alignment adjustment to be made using the actual embedded data to improve the accuracy of the correction, and ensure the requested line is achieved. If Use Embedded Reader is not enabled, the correction is applied at the audio delay output, before the embedder and, as such, may not be as accurate. It is enabled by default.

Dolby E Timing

An indication of Dolby E header phasing at the outputs helps in making corrections and avoids clicking or other corruption on downstream switching due to the 5% inter-frame guardband being misaligned with the video switch point.

Each of the eight output embedded pairs, including passed input audio groups, can be selected for display and logging. The display will report (for both the standard and the user-defined line) a line offset, or indicate a missing Dolby E header, for example, indicating PCM audio.

Use the **Audio Pair** slider bar to select one of the eight embedded audio pairs.

Line Number reports the actual line number on which the Dolby E header occurs. Note that, depending on audio packet distribution, this may differ slightly from the expected line number. For example, if the expected line number is 22, line 21 or 23 may periodically be reported. This is a normal occurrence and does not necessarily indicate an error condition.

Sample Number reports the current audio sample number.

Dolby E position is reported Early, Late, or OK within the limits for each output standard as shown below. 'Frame ?' is reported if the video frame rate is at twice the audio frame rate. In this situation, the audio aligns to the nearest video frame. Since the number of video frames will be double the number of audio frames, the audio delay may require adjustment in order to align with the correct video frame.

Dolby E Timing

Audio Pair 1

Line Number

--

Sample Number

--

Relative Timing

Relative to Standard Line

--

Relative to User Line

NONE

User Dolby-E Line		
	User	Standard
Pair 1	<input checked="" type="radio"/>	<input type="radio"/>
Pair 2	<input checked="" type="radio"/>	<input type="radio"/>
Pair 3	<input checked="" type="radio"/>	<input type="radio"/>
Pair 4	<input checked="" type="radio"/>	<input type="radio"/>
Pair 5	<input checked="" type="radio"/>	<input type="radio"/>
Pair 6	<input checked="" type="radio"/>	<input type="radio"/>
Pair 7	<input checked="" type="radio"/>	<input type="radio"/>
Pair 8	<input checked="" type="radio"/>	<input type="radio"/>

There are two levels of limits for status and logging. For safe and reliable operation, the OK limit is ± 3 samples from the specified point, with WARN indications for up to the table limits, and ERROR indications beyond (this will definitely cause packet corruption on switching).

Standard	Switch Point	Early Line	Normal Line	Late Line
625/25i	6	<9	11	>30
525/29.97i	10	<12	14	>29
1080/25i 1080/25sf 1080/25p	7	<16	20	>55
1080/29.97i 1080/29.97sf 1080/29.97p 1035/29.97i	7	<17	22	>55
720/50p	7	<22	27	>73
720/59.94p	7	<23	29	>73

Note: All HD standards use line 7 for the frame switching line.
Unsupported frame rates are: 23.98, 24, 30, 60 in i or p or sf types

User Dolby-E Line

If the Dolby E delay has been manually configured, select **User** next to corresponding audio pairs to specify which of the eight audio pairs will monitor the user line for status and logging. Alternatively, select **Standard** to monitor the standard Dolby E line.

Audio Delay Settings

The IQDBD00 audio delay control system comprises 16 separate delay blocks. Each block processes a pair of PCM audio channels or one non-PCM data feed. The delay blocks can be controlled by one of two composite control feeds for external adjustment, or by an internal matching delay option. Each control feed can be driven by combination of both tracking and fixed delays. Tracking delays are those that can follow a variable timing value, such as the delay through a video synchronizer.

The selection of audio delay control A or B allows audio to be timed to, or synchronized with, one of two separate timing planes. Delays can be configured such that:

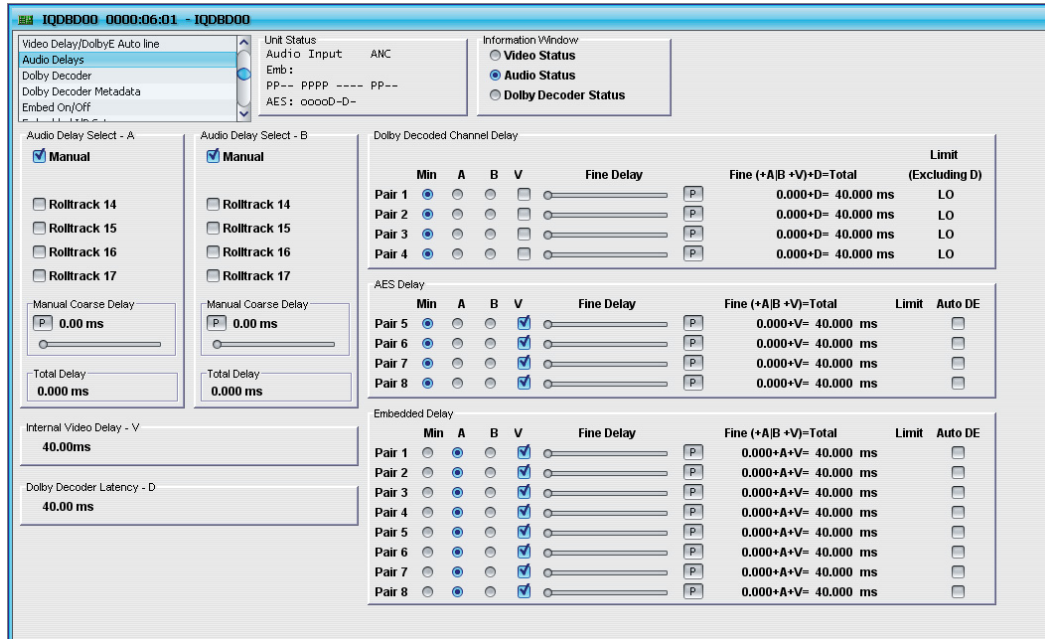
- A minimum of delay is applied.
- All audio is delayed together.
- Some audio is delayed, while some is not.
- Audio pairs have differing amounts of delay applied.

The delays are SMPTE-337 data aware, and as such will attempt to track any changes during the guardbands to avoid corrupting any data packets. This limits response to changes to a rate similar to the SRC filtered case, which is entirely adequate for tracking while synchronizing video.

For each delay block then the delay can be derived from one of the following settings:

Min	Effectively zero delay but when no Internal delay is added it is limited to the minimum audio delay of approx. 0.75 ms or 3 ms if via a SRC.
A	The delay time for this audio pair is the value set by the Audio Delay Select-A control.
B	The delay time for this audio pair is the value set by the Audio Delay Select-B control.
V	Adds an audio delay equal to the unit's video Internal Delay setting when enabled.
Fine Delay	The Manual Fine Delay control specifies manual delays finer than 0.25 ms (the adjustment increment of the Manual Coarse Delay control). The delay specified by this control is added to the value set with the Manual Coarse Delay control. The delay range is -0.244 ms to +0.244 ms in 0.05 ms steps. The preset value is 0.00 ms.
Auto DE	When selected, the unit alters the audio delay by up to half a frame to automatically place the Dolby E header on the same line as specified by the Dolby E Auto Line No control on the Dolby E Auto Line page.

RollTrack and GPI inputs must be static when contributing to delay controls using Auto DE.



For each audio delay control, **Audio Delay Select-A** and **Audio Delay Select-B**, a combination of the following delay drivers are used to set the value of the delay:

- **Manual:** Manually adjusts the delay.
- **RollTrack:** A delay set by external RollTracks will be applied. Delay values can be applied via RollTracks 14 to 17 inclusive.

RollTracks are signals sent between pieces of equipment so that they can work together in concert. Two modules can exchange delay values through the RollTrack system. For example, the delay used for audio passing through this remultiplexer could be set by the delay specified on a video synchronizer. For more information, see the RollTrack Audio Delay processing section.

Audio Delay Select-A/B

This function allows delay times (A and B) to be applied to the delay processor block.

For both Audio Delay A and Audio Delay B, you can use the following delay methods:

Manual Coarse Delay

To specify a manual delay, select the Manual Delay check box, and use the slider bar to specify the value of the delay. The delay range is -40.0 ms to 2000.00 ms, adjusted in 0.25 ms steps. For finer adjustments, use the Manual Fine Delay control, which is described below. The preset value is 0.00 ms.

RollTrack

RollTracks are signals sent between pieces of equipment so that they can work together in concert. For example two modules can exchange delay values through the RollTrack system. The delay used for the audio passing through this remultiplexer module could be set for example, by the delay through a video synchronizer.

Audio Delay values can be applied via RollTracks 14, 15, 16 and 17.

For more details please see the RollTrack Audio Delay Tracking section.

Total Delay

This indicator shows the total delay value that the control (A or B) is currently set. Any delay block being controlled by that control (A or B) will be set to this value plus the video Internal Delay (if selected). It is a useful check that the right combination of delays has been applied. It only indicates delays applied at this stage of the processing and not individual pair delays applied to the audio inputs. One of the useful features is being able to see whether tracking delays are changing their value.

The minimum amount of total delay shown here will be 3.0 ms, which is the system minimum delay in PCM. Data mode is made consistent with PCM by adding 88 samples worth of delay.

Dolby Decoded Channel Delay, AES Delay 5 to 8 and Embed Delay 1 to 8

These display the controls for each of the 4 Dolby Decoded, 4 AES and 8 Embed delay blocks.

For each delay block the delay can be derived from one of the following settings:

- Min** Effectively zero delay but when no Internal delay is added it is limited to the minimum audio delay of approx. 0.75 ms or 3 ms if via a SRC.
- A** The delay time for this audio pair is the value set by the Audio Delay Select-A control.
- B** The delay time for this audio pair is the value set by the Audio Delay Select-B control.
- V** Adds an audio delay equal to the unit's video Internal Delay setting when enabled.
- D** The delay within the Dolby decoder.
- Fine Delay** Use the Manual Fine Delay control to specify manual delays finer than 0.25 ms (the adjustment increment of the Manual Coarse Delay control). The delay specified by this control is added to the value set with the Manual Coarse Delay control. The delay range is -0.244ms to +0.244ms in 0.05ms steps. The preset value is 0.00ms.

Fine (+A|B +V)+D=Total

This displays the total of the A or B delay time, plus any manual fine delay, plus the internal video delay, plus the Dolby decoder delay.

Limit

If **LO** appears to the right of the check boxes, the delay set is lower than the minimum delay (0.75 ms or 3.0 ms).

Auto DE When selected, the unit alters the audio delay by up to half a frame to automatically place the Dolby E header on the same line as specified by the Dolby E Auto Line No control on the Video Delay/Dolby E Auto Line page. For more information, see page 38

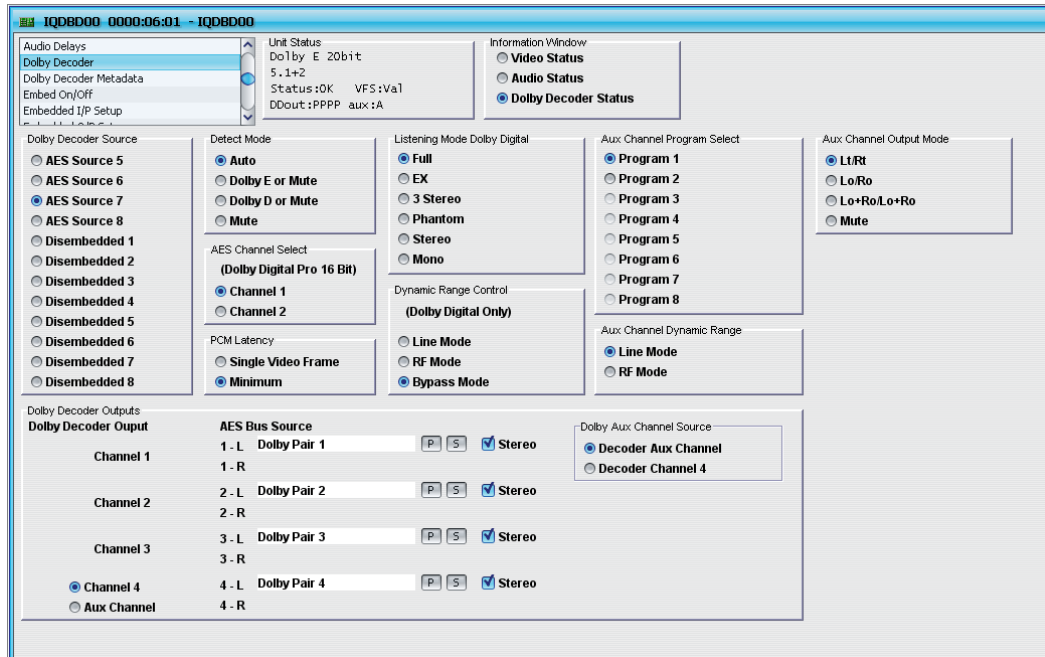
When an IQDBD00 is used as a post-switch firewall in an agile setting. Use of Mixed mode, whilst disabling procamps, provides for clean switching between PCM and Data such as Dolby E. Provided that the decoder can change modes cleanly it is possible to have a "hot" switch between types without any mute or disturbance. Since PCM has no alignment requirement the Auto DE controller will allow the audio delay to drift to the actual requested delay when PCM is detected, and a switch back to a Dolby E source will give some misalignment error during the Auto DE recovery time - and probably a decoder mute. For this reason it is essential that a pre-switch alignment firewall is used on the Dolby E source(s), and that the PCM delay is set to the nominal delay for correct alignment - probably an integer number of frames.

There are additional controls which should be selected for reliable operation in the presence of other ancillary data, in particular for standard definition video:

- **Embedder Priority (Embed On/Off page)** should be set to **Audio**. This will prioritize audio packet insertion so the requested line is always available.
- **Use Embedded Reader (Dolby E Auto Line page)** should be enabled. This allows the automatic alignment adjustment to be made using the actual embedded data to ensure the requested line is achieved.

Dolby Decoder

Use the settings on the Dolby Decoder page to specify how the module handles encoded Dolby source material.



Dolby Decoder Source

To specify which encoded Dolby source is passed to the Dolby Decoder, select it from the list. This can be from either one of the AES inputs (AES Source 5 to AES Source 8) or from Embedded Dolby E in the SDI video (Disembedded 1 to Disembedded 8)

Detect Mode

These options specify the type of Dolby to be decoded.

- Auto** In this mode, the module will automatically detect and decode any Dolby E or Dolby D bitstream. Any other bitstream is passed as a PCM signal.
- Dolby E or Mute** This mode will only decode Dolby E input. Non-Dolby E input will cause the Dolby Decoder's output to be muted.
- Dolby D or Mute** This mode will only decode Dolby D input. Non-Dolby D input will cause the Dolby Decoder's output to be muted.
- Mute** This option forces the Dolby Decoder's output to be muted.

AES Channel Select (Dolby Digital Pro 16 Bit)

This option selects which AES3 channel to use as the source for the Dolby Digital data bitstream if the Dolby Digital data is configured as Professional 16 Bit format.

PCM Latency

This control specifies the latency with which PCM data that is presented to the Dolby Decoder will be passed.

- Single Video** If this option is selected, when PCM data is present to the

Frame	decoder, if the decoder recognises the frame rate it will delay the PCM by one frame, otherwise it applies the minimum delay (approx 10ms).
Minimum	If this option is selected, when PCM data is presented to the decoder, it will pass through the decoder with the minimum delay possible. This is approximately 10mS.

Listening Mode Dolby Digital

When decoding Dolby Digital, the following listening modes can be selected:

- EX
- Full
- 3 Stereo
- Phantom
- Stereo
- Mono

Dynamic Range Control

(Dolby Digital Only)

When the unit is decoding Dolby Digital, this control allows the Dynamic Range mode to be specified.

Line Mode	Dynamic range control is applied to the decoded PCM suitable for line mode (dialog normalization and dynrng compression).
RF Mode	Dynamic range control is applied to the decoded PCM suitable for RF mode (dialog normalization and compr compression).
Bypass	No dynamic range control is applied to the decoded PCM.

Note: *If dynamic range control is selected and applied to the decoded PCM, the metadata remains unchanged.*

Aux Channel Program Select

Use this control to specify which program within the encoded Dolby is output on the auxiliary channel.

Aux Channel Dynamic Range

These options specify the type of dynamic range control applied to the auxiliary channel output.

Line Mode	Dynamic range control is applied to the decoded PCM suitable for line mode (dialog normalization and dynrng compression).
RF Mode	Dynamic range control is applied to the decoded PCM suitable for RF mode (dialog normalization and compr compression).

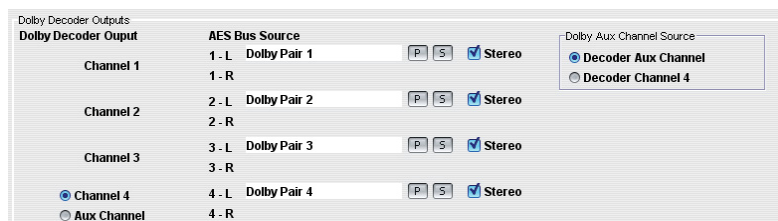
Aux Channel Output Mode

This control specifies the output mode that will be sent to the auxiliary output channel. Depending on the Dolby program type, these options change dynamically.

Dolby E 5.1-ch mode	Lt/Rt, Lo/Ro, Lo+Ro/Lo+Ro, mute
Dolby E 4-ch mode	Lt/Rt, Lo/Ro, Lo+Ro/Lo+Ro, mute
Dolby E 2-ch mode	L/R, L+R/L+R, mute
Dolby E 1-ch mode	L/R, L+R/L+R, mute
Dolby D 3/2 mode	Lt/Rt, Lo/Ro, Lo+Ro/Lo+Ro, mute
Dolby D 2/2 mode	Lt/Rt, Lo/Ro, Lo+Ro/Lo+Ro, mute
Dolby D 3/1 mode	Lt/Rt, Lo/Ro, Lo+Ro/Lo+Ro, mute
Dolby D 2/1 mode	Lt/Rt, Lo/Ro, Lo+Ro/Lo+Ro, mute
Dolby D 3/0 mode	Lt/Rt, Lo/Ro, Lo+Ro/Lo+Ro, mute
Dolby D 2/0 mode	L/R, L+R/L+R, mute
Dolby D 1/0 mode	Centre/Centre, mute
Dolby D 1+1 mode	L/R, L+R/L+R, mute

Dolby Decoder Outputs

The Dolby Decoder Output settings allow the main four Dolby decoded pairs to be renamed. The decoded pairs can be configured as separate left and right channels or designated as stereo pairs.



If the **Aux Channel** option is selected, the Aux Channel is used in place of the fourth decoded pair.

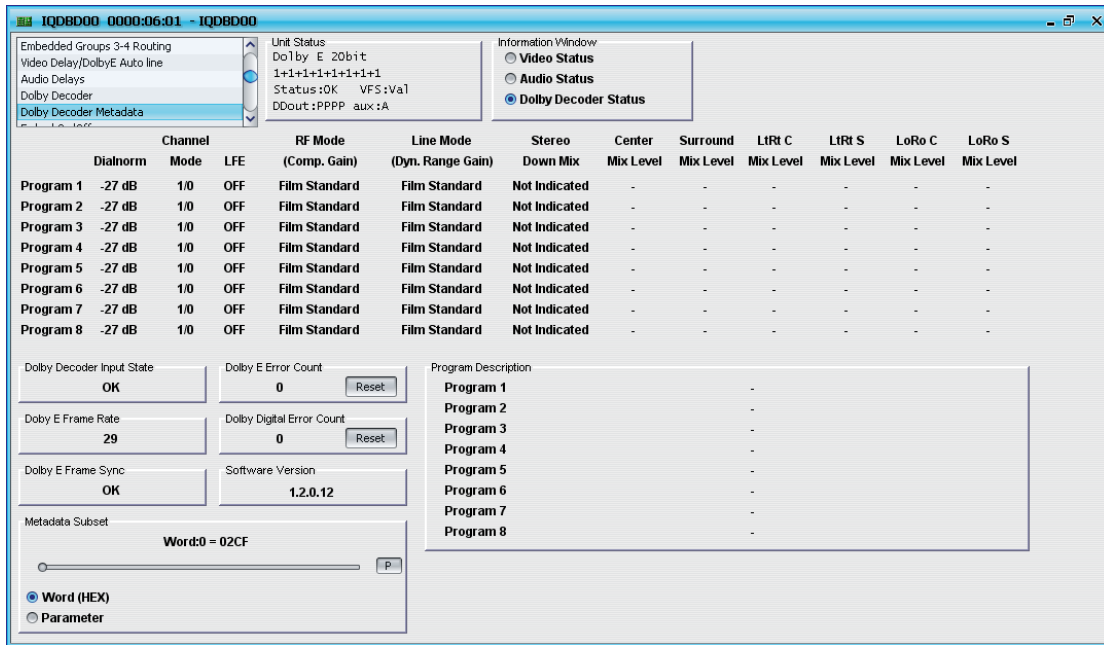
Note: *When the **Aux Channel** option is selected, the name given to decoded pair four will be applied to the auxiliary output. For example, the default name for the fourth decoded pair is “Dolby Pair 4”, if the Aux Channel option is selected, whenever Dolby Pair 4 is selected as a source in other menus, the Aux pair will actually be selected.*

Dolby Aux Channel Source

This option specifies the source for the Dolby auxiliary channel. It can be either the Decoder Aux Channel or Decoder Channel 4.

Dolby Decoder Metadata

The **Dolby Decoder Metadata** page displays general status information, and lists metadata parameters for all available programs within the current Dolby configuration.



Embed On/Off Settings

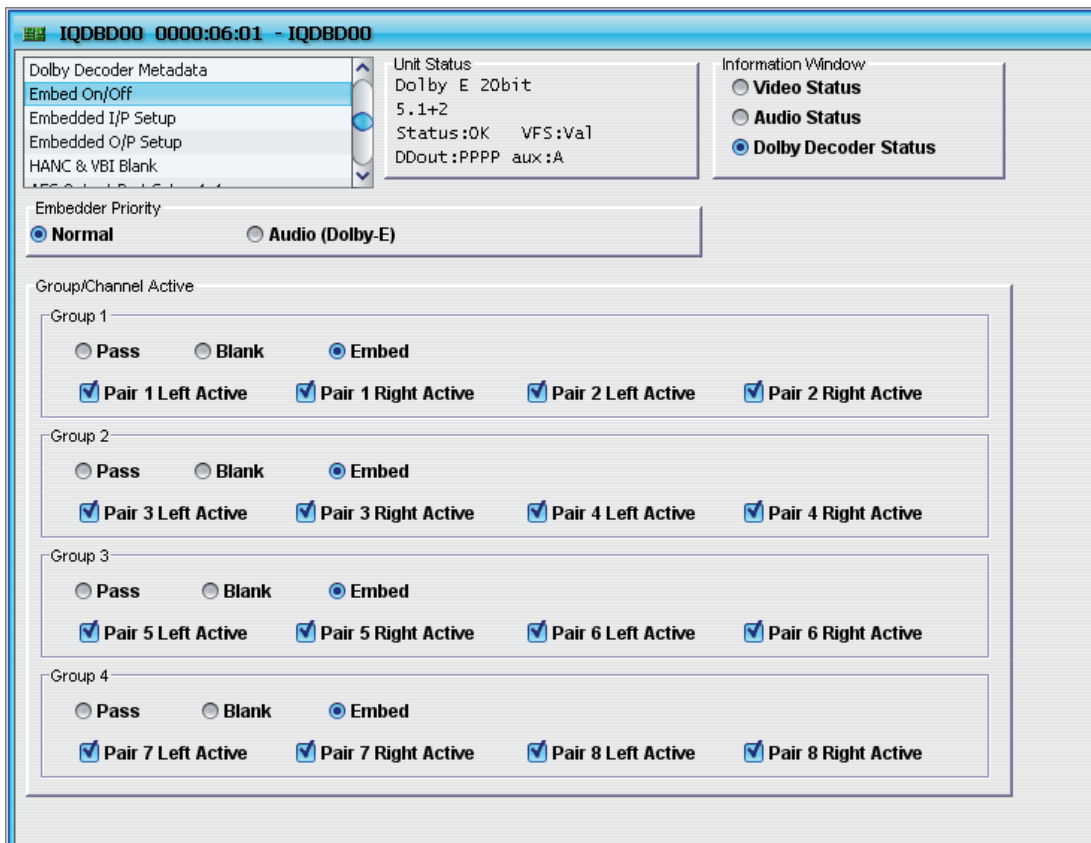
Use the Embed On/Off functions to specify how the module handles the embedding of audio into the output HD-SDI or SD-SDI video stream. The audio inputs are arranged into four groups, comprising two channel pairs each.

For each group, the module can be configured to:

- Pass audio unchanged.
- Remove audio data from the stream.
- Embed audio data into the stream.

By default, audio data is embedded in each group.

Within each group, specify whether each channel, for example, Pair 1 Left or Pair 1 Right, is active or disabled. A disabled channel is muted and flagged as absent, with the channel status indicating 48 kHz synchronous audio. This does not change the group behavior.



Embedder Priority

Selecting Audio (Dolby E) causes audio to be embedded first. This ensures that audio data, such as Dolby E is on a specific line. If processing Dolby E, it is recommended that this option is selected.

Group/Channel Active

For each Group (Group 1 to Group 4):

- To allow audio data to pass through the unit unchanged, select Pass.
- To remove audio data from the incoming stream, select Blank.
- To embed audio data into the stream, select Embed.

For each channel within each group:

- To activate the channel, select the corresponding check box.
- To disable the channel, clear the corresponding check box.

For more information about Audio Embedding, see Ancillary Passing and Audio Embedding on page 86.

Embedded I/P Setup

Use the Embedded I/P Setup screen to specify whether an input audio pair should be treated as:

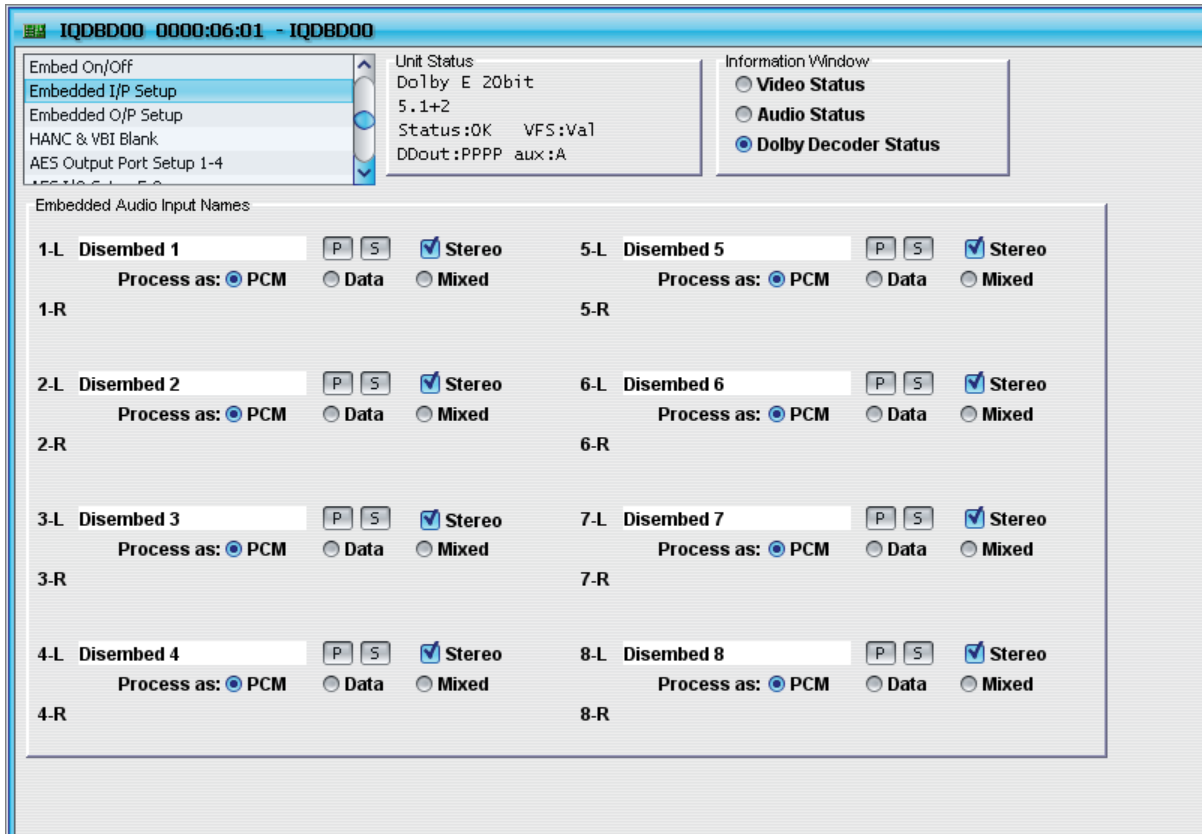
- PCM audio
- Data
- Mixed mode

This screen also allows the names of the disembedded audio inputs to be changed from the default names (Disembed 1 etc.).

The input embedded audio must be sampled at 48 kHz. Up to 20 bit resolution is supported in SD video standards, and up to 24 bit resolution is supported in all HD video standards.

The mixer, routing, and ProcAmp functions are channel-based and can pass or process on either channel. However, audio delays are pair-based and it is not possible to apply a different delay to the two channels of an input audio pair.

Embedded audio is a pair based system. This means that they carry the sound as two channels and it is not possible to carry just one channel. Of course, the unit can pass and process either channel as well as both. This is because the module is channel-based internally. When an embedded feed is used to carry non-PCM data then the whole data space is allocated to that function. In this mode the feed has to be treated together, it has to be treated as the equivalent of a pair. For example a non PCM feed at the input will occupy the same space as two audio channels and must be passed to the output as two channels of data.



PCM

If **PCM** mode is selected, the pair will be treated as PCM audio by the module. PCM processing can be either channel-based or stereo based. By default, channels are configured as stereo pairs. To switch to channel-based processing, clear the **Stereo** check box. PCM mode allows audio-specific operations, such as gain and tracking delay, to be adjusted by means of the Mixers and Proc Amps. In PCM mode, channel status is overwritten with filtered channel status from the input, and is assumed to be steady state 48 kHz synchronous professional, with audio or data type as indicated on the source. If Data is flagged by channel status the unit will still respond by disabling processing operations including forcing pair-based routing, thus giving a data-compatible audio path.

Data

If **Data** mode is selected, the pair will be treated as a non-PCM data feed through the circuitry within the module. This mode also protects the feed from PCM audio-specific operations such as gain and tracking delay. It also ensures that the audio is routed as a pair. In Data mode, channel status is overwritten with filtered channel status from the input, and is assumed to be steady state 48 kHz synchronous professional, with audio or data type as indicated on the source. One stream is equivalent to two PCM audio channels so in this mode there are no operations that can split the data. PCM audio will still be passed, and the channel status will follow the input type.

Mixed

If **Mixed** mode is selected, processing occurs as with Data mode; however, the channel status overwrite mechanism is bypassed and set to 'pass through' on a channel by channel basis. One stream is equivalent to two PCM audio channels so in this mode there are no operations that can split the data. This mode is suitable for processing Dolby E and Dolby AC3 (Dolby Digital) data, when transitions between data and PCM audio will be passed

cleanly with the source channel status, allowing decoders the best chance of handling the transition.

Embedded Audio Input Names

By default, channels are designated as stereo pairs. To name each channel individually, clear the **Stereo** check box.

To change a disembedded audio input name:

- Type the new name in the field, and click the save button.
- To return to the default name, click the preset button.

Note: *When a channel is designated as a stereo signal, the corresponding left/right channel does not appear.*

Names are stored in the Memory function so that a new set of names can be recalled by memory.

These names are used in the lists for the following functions:

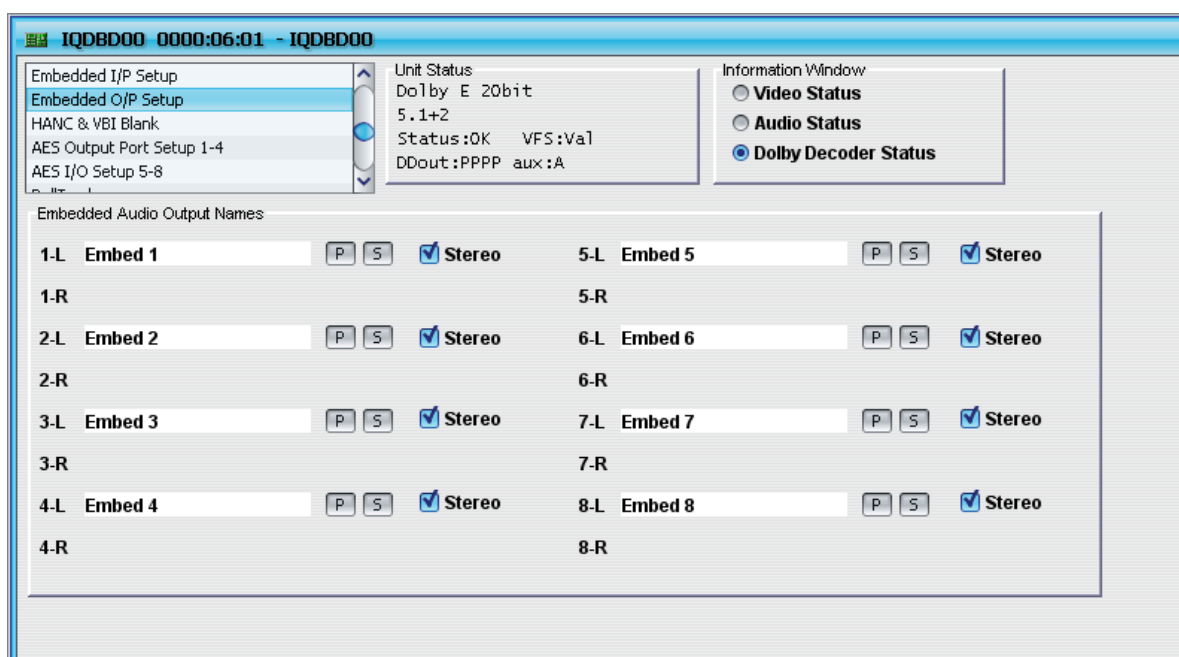
- Audio Mixer Inputs
- AES Routing
- Monitoring Routing
- Embedded Routing
- Embedded Input Setup

Embedded O/P Setup

Use the **Embedded O/P Setup** settings to change the names of the embedded audio outputs and designate stereo pairs.

The output embedded audio is always sampled at 48 kHz and synchronized to the output video. Up to 20 bit resolution is supported in SD video standards, and up to 24 bit resolution is supported in all HD video standards.

To access these settings, select **Embedded O/P Setup** from the function list.



Setting up Embedded Output

By default, channels are designated as stereo pairs. To name each channel individually, clear the **Stereo** check box.

To change an embedded audio output name:

- Type the new name in the field, and click the save button.
- To return to the default name, click the preset button.

Note: *When a channel is designated as a stereo signal, the corresponding left/right channel does not appear.*

Names are stored in the Memory function so that a new set of names can be recalled by memory.

These names are used in the lists for the following functions:

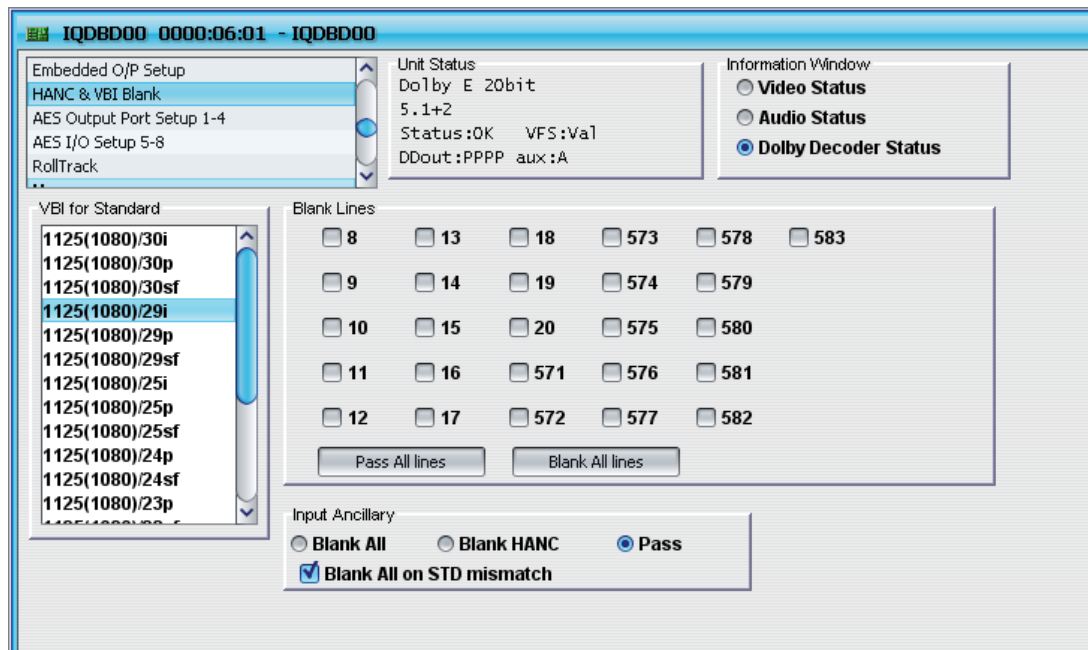
- Embedded Routing
- Embedded ProcAmps

HANC and VBI Blanking Settings

The settings on the **HANC & VBI blank** screen:

- Specify whether vertical interval data (VBI), is blanked or passed through the module. Note that the **Input Ancillary Blank All, Blank HANC, and Pass** settings allow ancillary blanking to be globally defined and do not change with output standard. Blanking can be applied to all lines or to specific lines, depending on the selected standard. Each standard can have different blanking applied.
- Specify whether, and under what circumstances, horizontal ancillary data is blanked or passed through the module.

To access these settings, select **HANC & VBI blank** from the function list.



Specifying Vertical Interval Data (VBI) Blanking Settings

Vertical Interval Data contained in the input stream can be blanked or passed through the module. To do this, specify the input standard and then select the lines to blank. Alternatively, you can choose to pass all lines, or blank all lines.

The input standards, for the purpose of VBI blanking, are:

1125(1080)/30i	1125(1080)/25i	1125(1080)/23sf	525(480)/29i
1125(1080)/30p	1125(1080)/25p	1125(1035)/30i	625(576)/25i
1125(1080)/30sf	1125(1080)/25sf	1125(1035)/29i	
1125(1080)/29i	1125(1080)/24p	750(720)/60p	
1125(1080)/29p	1125(1080)/24sf	750(720)/59p	
1125(1080)/29sf	1125(1080)/23p	750(720)/50p	

To configure VBI blanking settings:

1. In the **VBI for Standard** list, select the standard for which to configure VBI blanking.
2. In the **Blank Lines** section, do one of the following:
 - Select the lines to blank. The selected lines will not be passed to the output signal.
 - Click **Blank All Lines**. This selects all the check boxes in the **Blank Lines** section. All displayed vertical interval lines will be blanked.
 - Click **Pass All Lines**. This clears all the check boxes in the **Blank Lines** section. All displayed vertical interval lines will be passed.

Specifying Horizontal Ancillary Data (HANC) Blanking Settings

The **Input Ancillary Controls** specify whether to:

- Remove all ancillary data.
- Remove only horizontal ancillary data.
- Allow ancillary data to pass unaltered.
- Remove all ancillary data, if the input signal fails to match the output standard.

Note: *Embedded audio (as enabled for embedding) and EDH controls are not affected by these controls.*

To remove all ancillary data:

- In the **Input Ancillary** section, select **Blank All**.
This setting overrides the **VBI Blank Lines** control.

To remove only horizontal ancillary data (HANC);

- In the **Input Ancillary** section, select **Blank HANC**.

To allow ancillary data to pass unaltered:

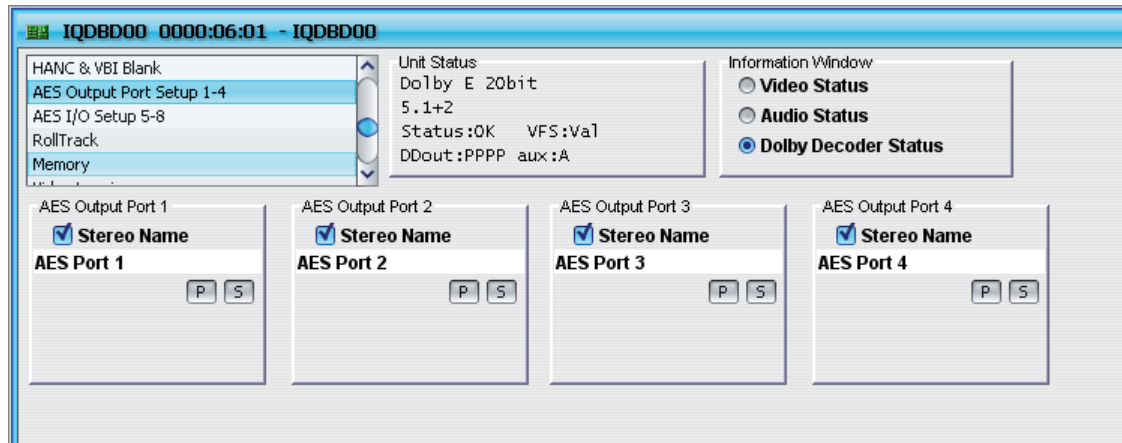
- In the **Input Ancillary** section, select **Pass**.
When **Pass** is selected, pass-through operations will not alter audio packets for groups that are not selected for embedding.

To remove all ancillary data if the input signal fails to match the output standard:

- In the **Input Ancillary** section, select **Blank All on STD mismatch**.

AES Output Port Setup 1-4

By default, the module's AES output ports are named AES Output Port 1 to AES Output Port 4. On the AES Output Port Setup screen, these port names can be changed as required. As well, the ports can be named as stereo pairs or each channel can be individually named.



By default, channels are designated as stereo pairs. To name each channel individually, clear the **Stereo** check box.

To change an AES Output Port name:

- Type the new name in the field, and click the save button.
- To return to the default name, click the preset button.

Note: *When a channel is designated as a stereo signal, the corresponding left/right channel does not appear.*

AES I/O Setup 5-8

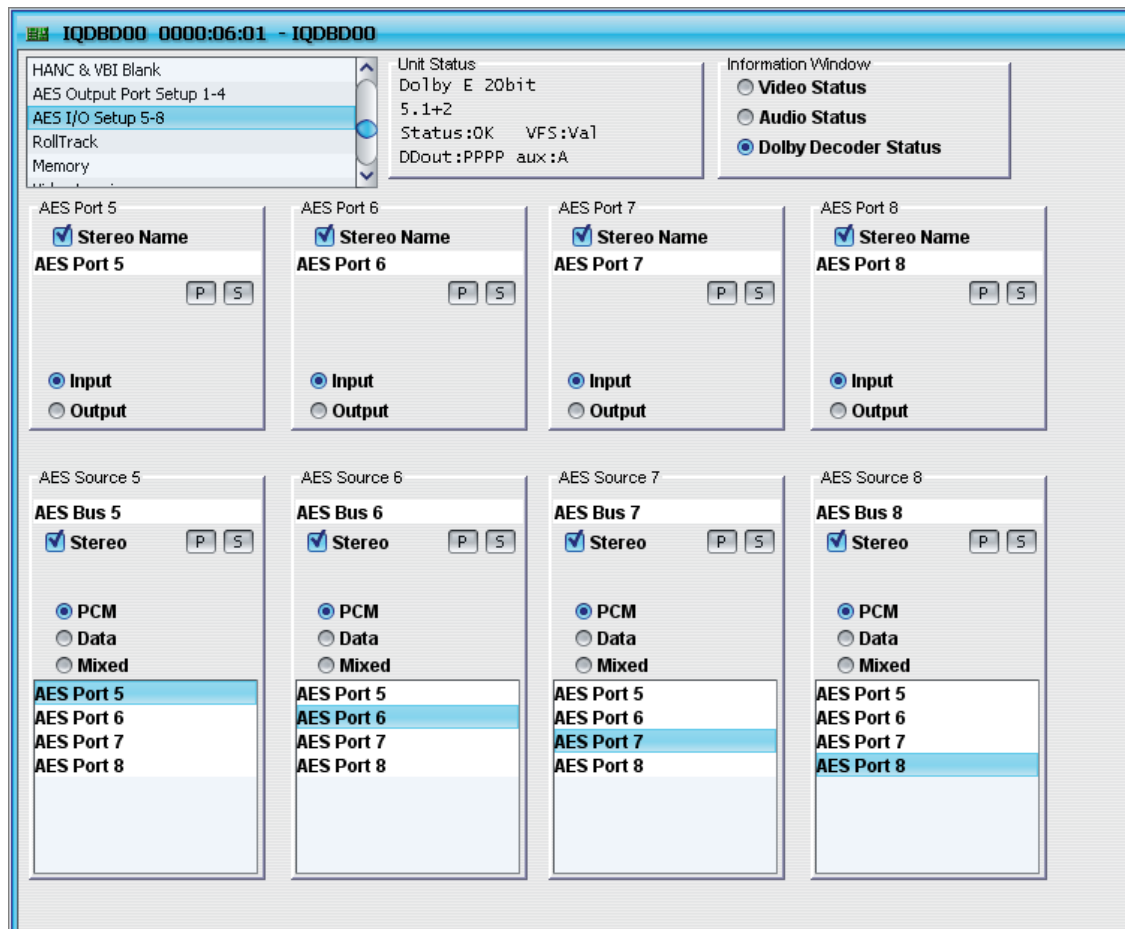
Use these settings to:

- Configure AES ports.
- Configure AES sources.

The module's AES inputs can accept a wide range of signal types. Sampling rates from 32 kHz, 44.1 kHz and 48 kHz can be accommodated because each input pair has a built in sample rate converter. PCM signals do not need to be synchronous to the output video because the input circuitry performs synchronization. Signals of 16-bit, 20-bit, or 24-bit resolution can be processed and embedded. Both standard professional AES signals and the consumer protocol IEC-958 type II (sometimes referred to as SPDIF) signals are accepted. Regardless of the signal, the AES outputs always operate at the standard 48 kHz sampling rate and are synchronized to the output video.

In addition to accepting a wide range of PCM signals, the AES ports accept non-PCM data, such as Dolby E and Dolby AC3 (Dolby Digital) feeds. Note that non-PCM data feeds must be synchronous to the video. In the case of Dolby E, this not only means that the data clock must be synchronized, but also that the Dolby E blocks must align to the video boundaries, otherwise, the signal becomes liable to corruption on video switching and recording.

To access these settings, select **AES I/O Setup 5-8** from the function list.



AES ports 5-8 can be individually configured as either an input or an output.

As well, each port can be renamed, for example, to reflect the signal that it is carrying. By default The AES ports are designated as stereo sources.

However, each channel can be named individually. These names are used for the following functions:

- AES ProcAmp
- AES Routing

To specify whether the port is an input or an output:

- To designate the port as an output, select **Output**.
- To designate the port as an input, select **Input**.

When a port is configured as an input, the AES port appears in the router listings at the bottom of the screen. This can be used to specify a different port routing if necessary.

To rename a port:

- Type the name of the port in the text field and click the save button.
To name each channel individually, clear the **Stereo** check box.
To return the port to its default name, click the preset button.

Configure each AES port to handle PCM audio, non-PCM data, or both. This selection is essential if Dolby E or Dolby AC3 (Dolby Digital) is being passed through an AES port.

Note: *In data and mixed modes, the Sample Rate Converter (SRC) is bypassed.*

SMPTE337 header detection is used to detect non-PCM.

PCM

If **PCM** mode is selected, the pair will be treated as PCM audio by the module. PCM processing can be either channel-based or stereo based. By default, channels are configured as stereo pairs. To switch to channel-based processing, clear the **Stereo** check box. PCM mode allows audio-specific operations, such as gain and tracking delay, to be adjusted by means of the Mixers and Proc Amps. In PCM mode, channel status is overwritten with filtered channel status from the input, and is assumed to be steady state 48 kHz synchronous professional, with audio or data type as indicated on the source. If Data is flagged by channel status the unit will still respond by disabling processing operations including sample rate conversion, and forcing pair-based routing, thus giving a data-compatible audio path.

Data

If **Data** mode is selected, the pair will be treated as a non-PCM data feed through the circuitry within the module. This mode also protects the feed from PCM audio-specific operations such as gain and tracking delay. It also ensures that the audio is routed as a pair. In Data mode, channel status is overwritten with filtered channel status from the input, and is assumed to be steady state 48 kHz synchronous professional, with audio or data type as indicated on the source. One stream is equivalent to two PCM audio channels so in this mode there are no operations that can split the data. PCM audio will still be passed, and the channel status will follow the input type.

Mixed

If **Mixed** mode is selected, processing occurs as with Data mode; however, the channel status overwrite mechanism is bypassed and set to 'pass through' on a channel by channel basis. One stream is equivalent to two PCM audio channels so in this mode there are no operations that can split the data. This mode is suitable for processing Dolby E and Dolby AC3 (Dolby

Digital) data, when transitions between data and PCM audio will be passed cleanly with the source channel status, allowing decoders the best chance of handling the transition.

As well, each source can be renamed. By default, the AES sources are designated as stereo sources. However, each channel can be named individually. These names are used for the following functions:

- Audio Mixer inputs
- AES Routing
- Monitor Routing
- Embedded Routing.

To rename a source:

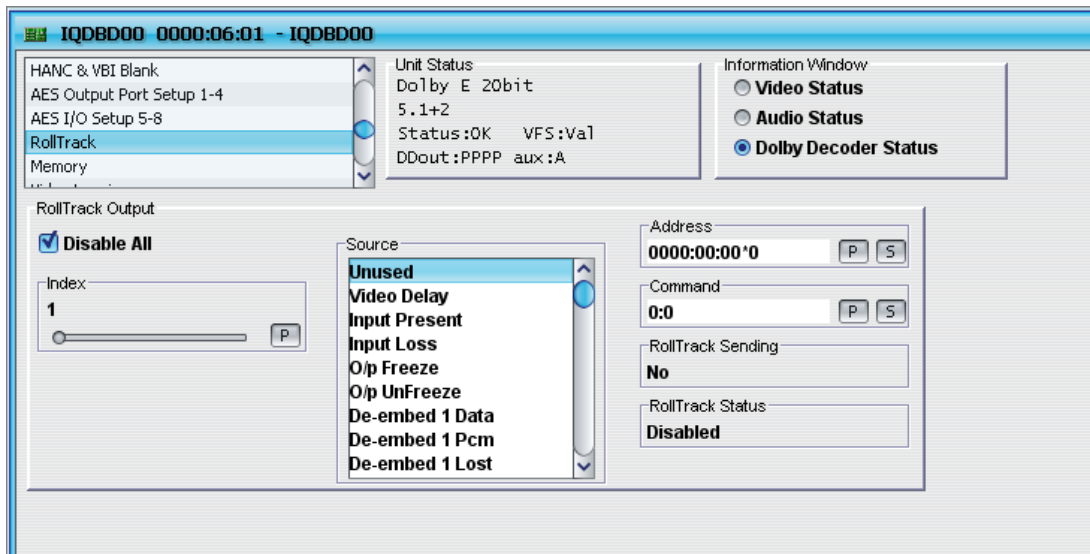
- Type the name of the source in the text field and click the save button.
To name each channel individually, clear the **Stereo** check box.
To return the source to its default name, click the preset button.

RollTrack Settings

The RollTrack settings allow information to be sent, by means of the RollCall network, to other compatible units on the same network.

Use the settings on the **RollTrack** screen to:

- Enable or disable the RollTrack functions.
- Configure up to 16 RollTrack outputs.
- Specify the conditions that trigger RollTrack data transmission.
- Set RollTrack destinations.
- Specify the RollTrack commands to be sent.



RollTrack Sources

The RollTrack Source specifies the source of the information that triggers the transmission of data.

The options are:

Unused	Video Delay	Input Present
Input Loss	O/p Freeze	O/p Unfreeze
De-embed 1-8 Data	De-embed 1-8 PCM	De-embed 1-8 Lost
DD Ch 1-4 PCM	DD Ch 1-4 Lost	AES 5-8 Data
AES 5-8 PCM	AES 5-8 Lost	DD Input Lost
DD PCM Input	DD Dolby D Input	DD Dolby E Input

RollTrack Addresses

The full RollTrack Address comprises four sets of numbers. For example, 0000:10:01*99.

The first set, 0000 in the example, is the network segment code number.

The second set, 10 in the example, identifies the (enclosure/mainframe) unit.

The third set, 01 in the example, identifies the slot number in the unit.

The fourth set, 99 in the example, is a user-configured number that uniquely identifies the destination unit in a multi-unit system. This ensures that only the correct unit responds to commands. If left at 00, an incorrectly fitted unit may respond inappropriately.

RollTrack Commands

Each RollTrack command comprises two sets of numbers, for example, 33039:3.

The first set, 33039 in the example, is the RollTrack command number, which identifies the command.

The second set, 3 in the example, is the value that is sent with the command.

Using RollTracks

To enable or disable RollTrack functions:

- To enable the RollTrack functions, clear the **Disable All** check box.
- To disable RollTrack functions, select the **Disable All** check box.

To configure a RollTrack action:

1. Select the **Index** number. This identifies the RollTrack action being configured. Up to 16 RollTrack actions can be created.
2. From the **Source** list, select the source of the information that will trigger RollTrack transmission.
3. Enter the RollTrack **Address** and click the save button. To return the address to its default value, click the preset button.
4. Enter the RollTrack **Command** and click the save button. To return the value to its default, click the preset button.

Viewing RollTrack Information

RollTrack Sending and **RollTrack Status** display information about the status of RollTracks.

RollTrack Sending displays the information when the unit is actively sending a RollTrack command:

- **String:** A string value is being sent.
- **Number:** A number value is being sent.
- **No:** The command is not being sent.
- **Yes:** The command is being sent.
- **Internal Type Error:** Inconsistent behavior. Contact your local Snell & Wilcox agent.

RollTrack Status displays the status of the currently selected RollTrack Index:

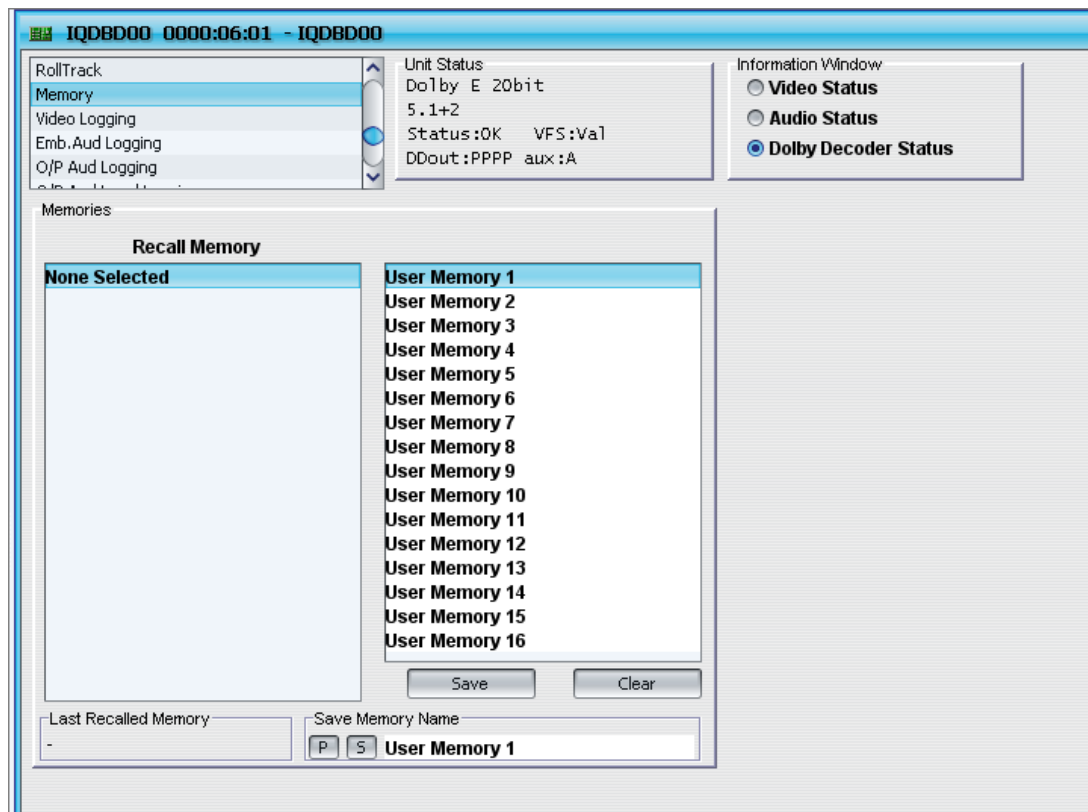
- **OK:** RollTrack message sent and received OK.
- **Unknown:** RollTrack message has been sent but has not yet completed.
- **Timeout:** RollTrack message sent, but acknowledgement not received. This could be because the destination unit is not at the specified location.
- **Bad:** RollTrack message has not been sent correctly, acknowledged at the destination unit. This could be because the destination unit is not of the type specified.
- **Disabled:** RollTrack sending is disabled.

Memory Settings

Use the **Memory** function to save up to 16 setups to be recalled later.

All of the module's controls are classed as either system or user. In general, system controls are the controls that are seldom changed once set up, and user controls are those controls that are likely to be changed frequently.

The **Memory** page is intended to store only user memories. On the other hand, a RollCall Saveset stores both the user memories and system memories. For more information about Savesets, refer to the RollCall Control Panel manual.



Saving to Memory

The **Memory Function** can save the settings of all items to memory. Default memory names can be changed to provide more meaningful descriptions.

To save settings:

- In the **Save Memory** column, select a memory location, and then click **Save**. The settings are saved and the memory appears the **Recall Memory** column.

To change a memory name:

- In the **Save Memory Name** field, type the new memory name, and then click the **S** button. To return the memory to its default value, click the preset button **P**.

Recalling Memory

Use the **Recall Memory** function to recall the settings saved in a memory location. **Last Recalled Memory** displays the most recently recalled memory.

An asterisk (*) appears after the last recalled memory name if adjustments have been made to any of the settings after it has been recalled.

If no memory choice has been made, or if the previous choice has been invalidated (because of a control change), **None Selected** is displayed.

To recall a memory:

- In the **Recall Memory** column, select the memory to recall. The recalled settings will be applied and the memory name will appear in the **Last Recalled Memory** section.

Logging

Information about several parameters can be made available to a logging device that is connected to the RollCall network.

Each logging screen comprises three columns:

- **Log Enable:** Select the check boxes that correspond to the parameters for which log information should be collected.
- **Log Field:** Displays the name of the logging field.
- **Log Value:** Displays the current log value.

Video Logging

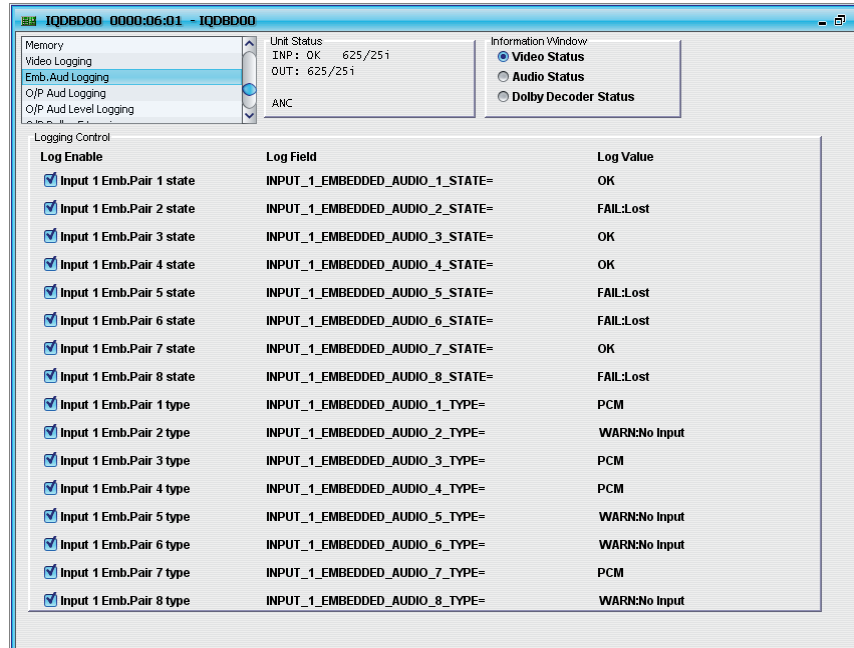
Video Logging records the status of video inputs and outputs.

The screenshot shows a software interface for video logging. At the top, there is a title bar with 'IQDBD00 0000:06:01 - IQDBD00'. Below the title bar, there are several status indicators and a navigation menu. The main area contains a table with three columns: 'Log Enable', 'Log Field', and 'Log Value'. The 'Log Enable' column contains checkmarks for various parameters. The 'Log Field' column contains the names of the logging fields. The 'Log Value' column contains the current values for these fields.

Log Enable	Log Field	Log Value
<input checked="" type="checkbox"/>	INPUT_1_TYPE=	HD / SD SDI
<input checked="" type="checkbox"/>	INPUT_1_STATE=	OK
<input checked="" type="checkbox"/>	INPUT_1_STANDARD=	625/25i
<input checked="" type="checkbox"/>	INPUT_1_SDI_ERRS=	OK
<input checked="" type="checkbox"/>	INPUT_1_SDI_ERRSEC=	0
<input checked="" type="checkbox"/>	INPUT_1_SDI_ANC_ERRS=	OK
<input checked="" type="checkbox"/>	INPUT_1_SDI_ANC_ERRSECS=	0
<input checked="" type="checkbox"/>	OUTPUT_1_STATE=	OK
<input checked="" type="checkbox"/>	OUTPUT_1_TYPE=	HD / SD SDI
<input checked="" type="checkbox"/>	OUTPUT_1_STANDARD=	625/25i
<input checked="" type="checkbox"/>	OUTPUT_1_ANC_STATE=	OK

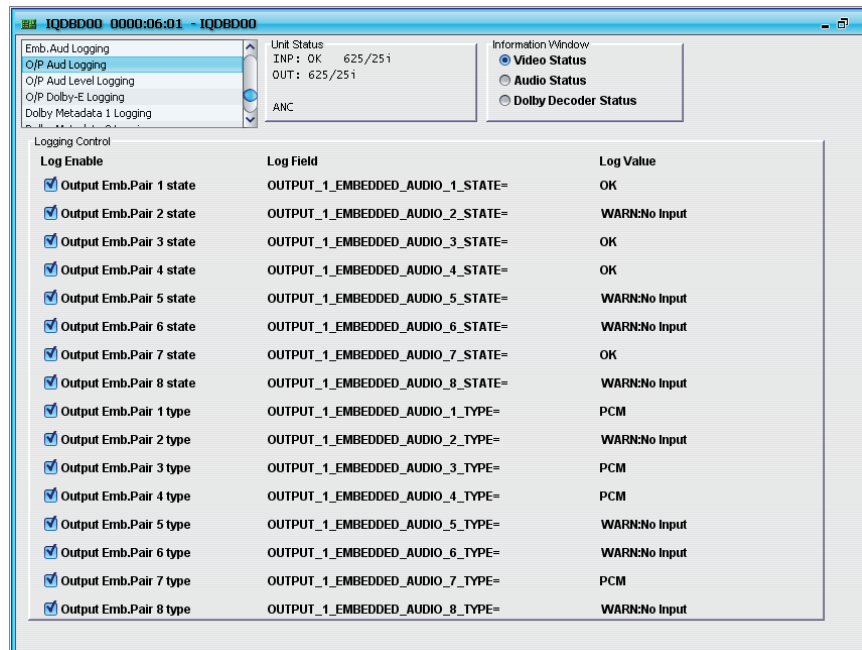
Embedded Aud Logging

Embedded Aud Logging records the status of the Embedded Audio inputs.



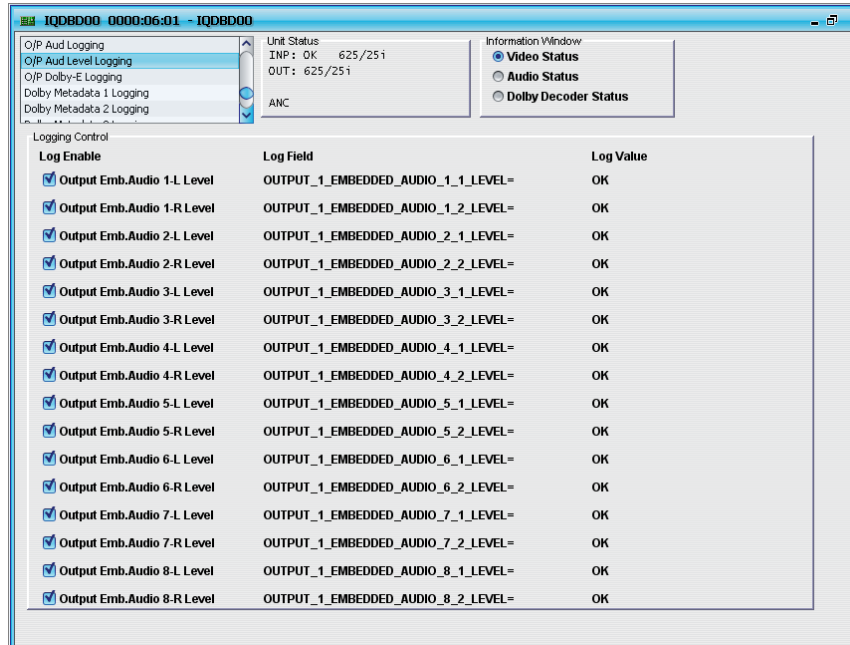
O/P Aud Logging

O/P Aud Logging records the status of the embedded audio outputs.



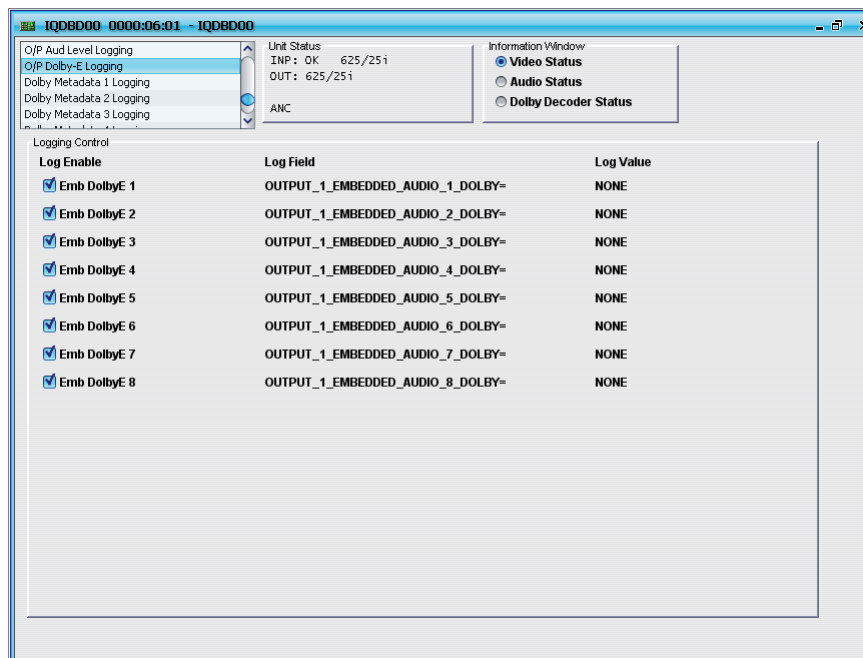
O/P Aud Level Logging

O/P Aud Level Logging records the level of the embedded audio outputs.



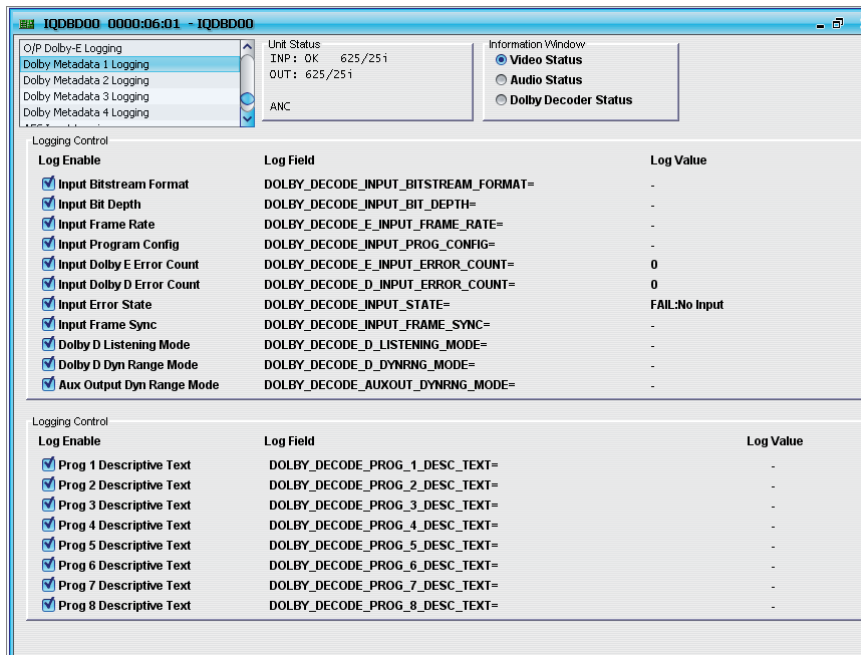
O/P Dolby-E Logging

O/P Dolby-E Logging records the presence of Dolby E signals, and their timing, with respect to the video.



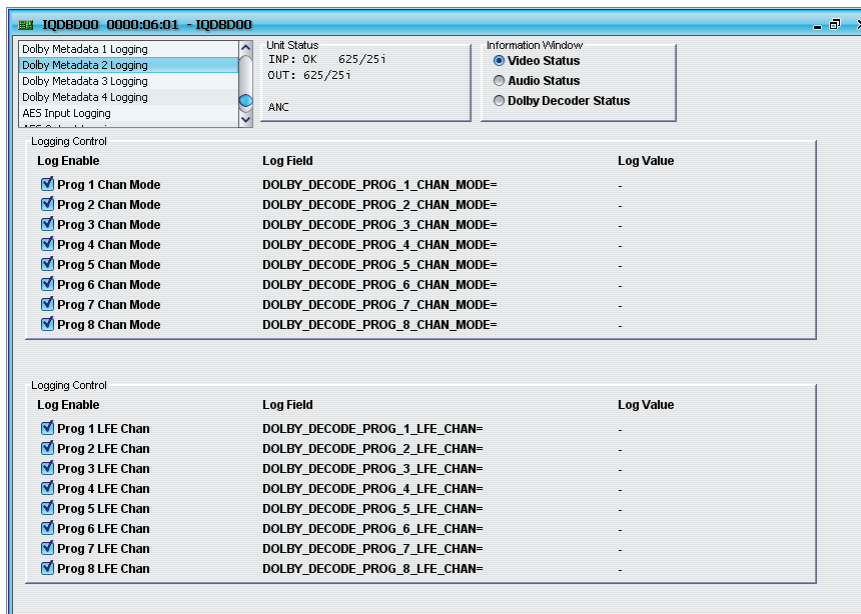
Dolby Metadata 1 Logging

Dolby Metadata 1 Logging records the status of Dolby Metadata information.



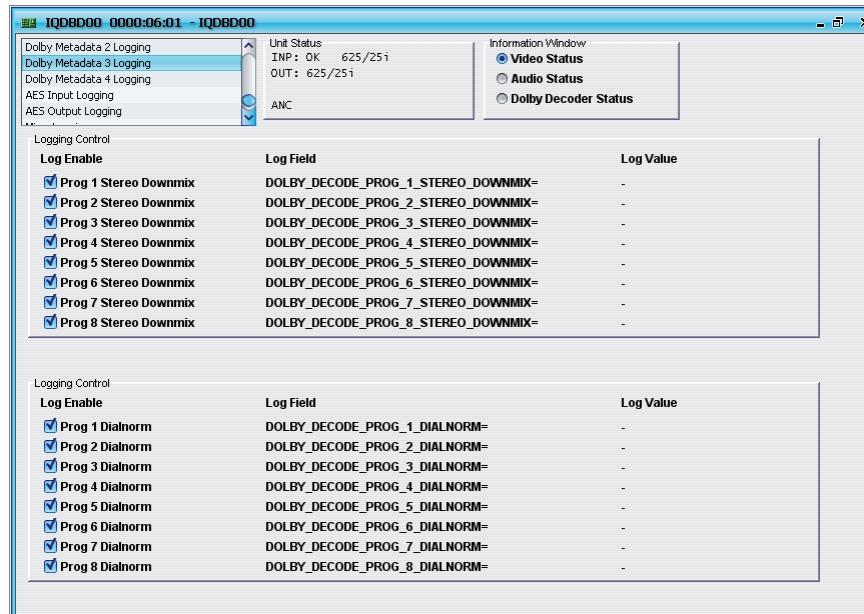
Dolby Metadata 2 Logging

Dolby Metadata 2 Logging records the status of Dolby Metadata information.



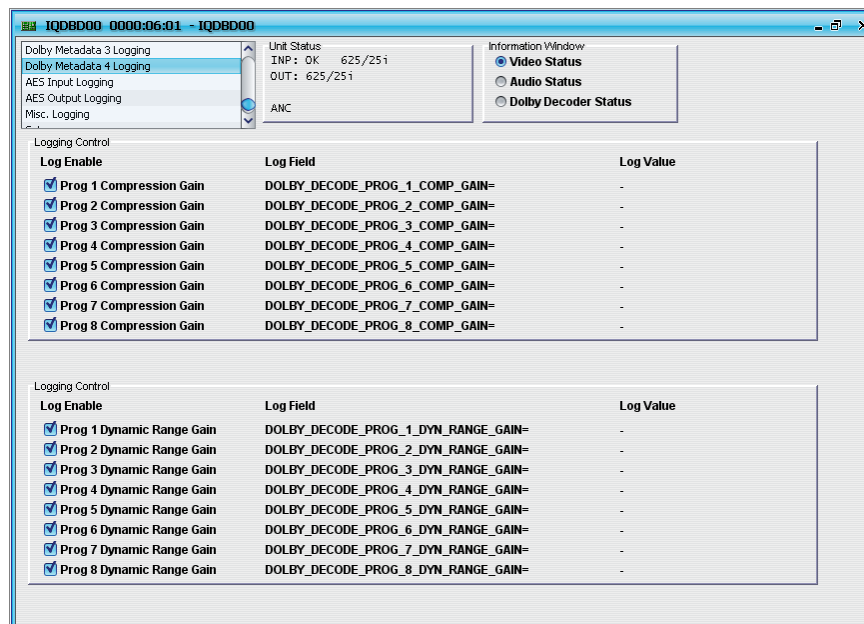
Dolby Metadata 3 Logging

Dolby Metadata 3 Logging records the status of Dolby Metadata information.



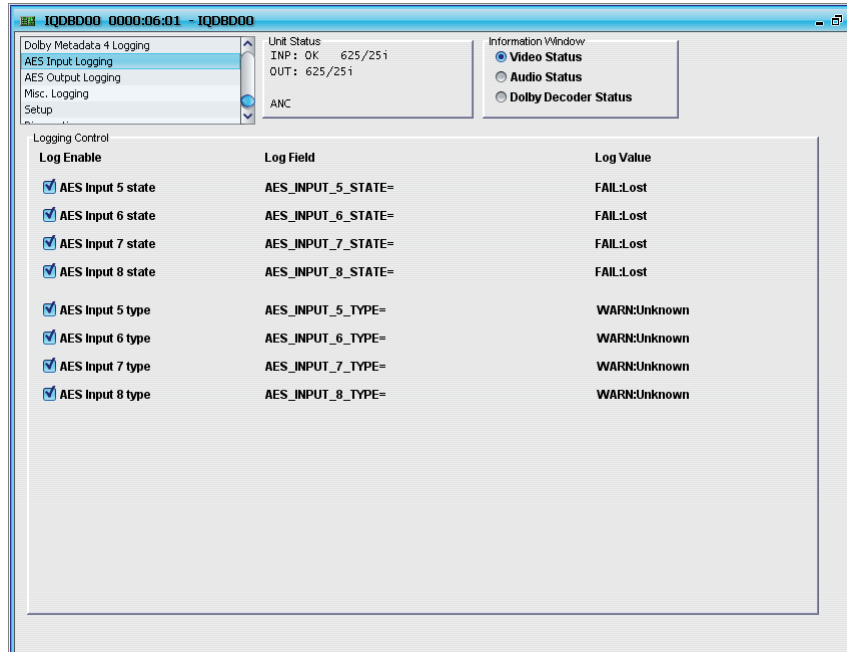
Dolby Metadata 4 Logging

Dolby Metadata 4 Logging records the status of Dolby Metadata information.



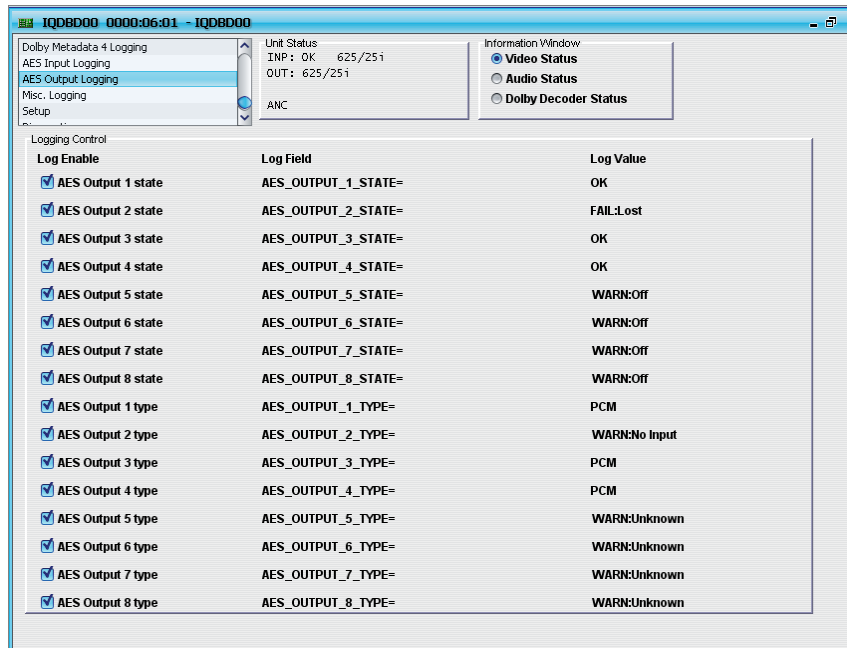
AES Input Logging

AES Input Logging records the status of the AES audio inputs.

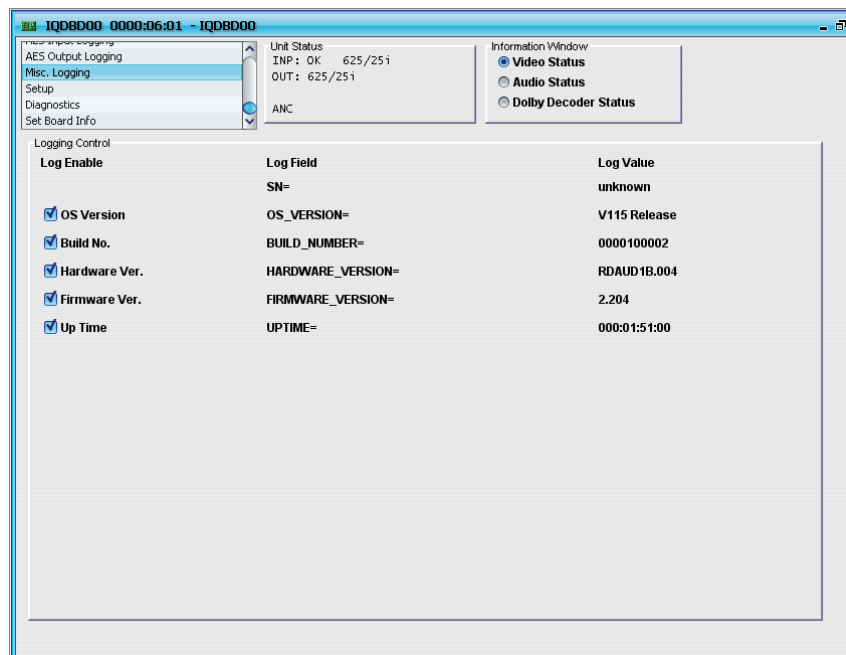


AES Output Logging

AES Output Logging records the status of the AES audio outputs.



Misc. Logging record miscellaneous information about the unit.



RollCall Log Fields

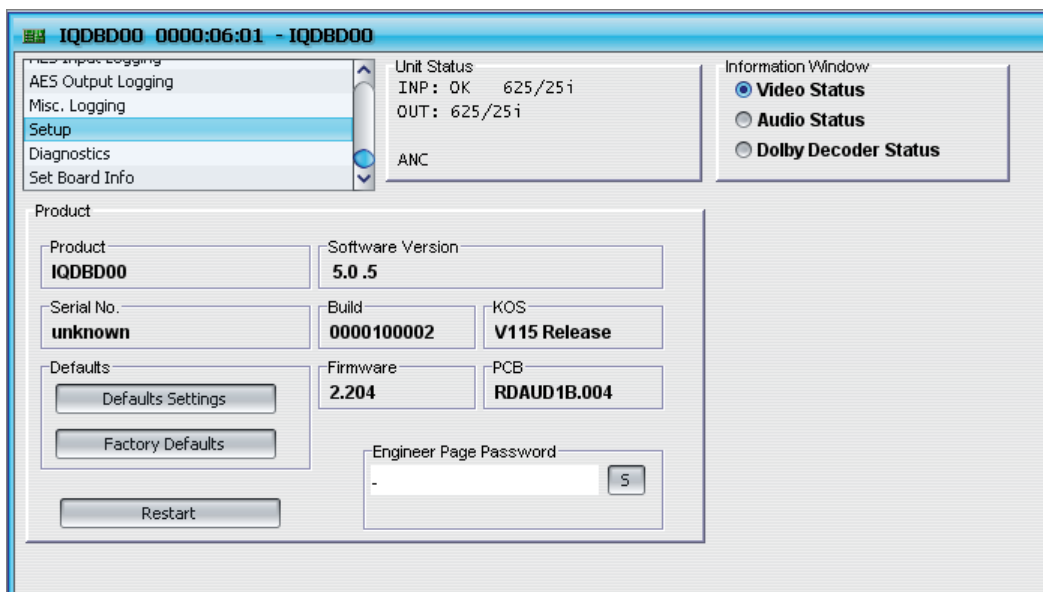
Log Field	Log Value
INPUT_1_TYPE	HD / SD SDI
INPUT_1_STATE	FAIL:Lost, OK, FAIL:Error
INPUT_1_STANDARD	STRING
INPUT_1_SDI_ERRS	NONE, OK, WARN
INPUT_1_SDI_ERRSEC	NUM
INPUT_1_SDI_ANC_ERRS	NONE, OK, WARN
INPUT_1_SDI_ANC_ERRSECS	NUM
OUTPUT_1_STATE	OK, FAIL, BLACK, PATTERN, FREEZE
OUTPUT_1_TYPE	HD / SD SDI
OUTPUT_1_STANDARD	STRING
OUTPUT_1_ANC_STATE	<Blank>, ANC FULL
INPUT_1_EMBEDDED_AUDIO_1_STATE to INPUT_1_EMBEDDED_AUDIO_8_STATE	FAIL:Lost, OK, LOST, Unknown
INPUT_1_EMBEDDED_AUDIO_1_TYPE to INPUT_1_EMBEDDED_AUDIO_8_TYPE	PCM, NONPCM, Unknown, WARN: No Input
OUTPUT_1_EMBEDDED_AUDIO_1_STATE to OUTPUT_1_EMBEDDED_AUDIO_8_STATE	FAIL:Lost, OK, LOST, Unknown

Log Field	Log Value
OUTPUT_1_EMBEDDED_AUDIO_1_TYPE to OUTPUT_1_EMBEDDED_AUDIO_8_TYPE	PCM, NONPCM, Unknown, WARN: No Input
OUTPUT_1_EMBEDDED_AUDIO_1_1_LEVEL OUTPUT_1_EMBEDDED_AUDIO_1_2_LEVEL to OUTPUT_1_EMBEDDED_AUDIO_8_1_LEVEL OUTPUT_1_EMBEDDED_AUDIO_8_2_LEVEL	OK, WARN:Overflow
OUTPUT_1_EMBEDDED_AUDIO_1_DOLBY to OUTPUT_1_EMBEDDED_AUDIO_8_DOLBY	NONE, OK, WARN:EARLY, WARN:LATE, ERROR:EARLY, ERROR:LATE
DOLBY_DECODE_INPUT_BIT_DEPTH	16, 20, 32
DOLBY_DECODE_E_INPUT_FRAME_RATE	<Input frame rate. E.g. 29>
DOLBY_DECODE_INPUT_PROG_CONFIG	<Dolby program configuration. E.g. 5.1+2>
DOLBY_DECODE_E_INPUT_ERROR_COUNT	NUM
DOLBY_DECODE_D_INPUT_ERROR_COUNT	NUM
DOLBY_DECODE_INPUT_STATE	OK, ERROR
DOLBY_DECODE_INPUT_FRAME_SYNC	OK, ERROR
DOLBY_DECODE_AUXOUT_DYNRING_MODE	LINE, RF, BYPASS
DOLBY_DECODE_PROG_1_DESC_TEXT to DOLBY_DECODE_PROG_8_DESC_TEXT	<Dolby program descriptive text>
DOLBY_DECODE_PROG_1_CHAN_MODE to DOLBY_DECODE_PROG_8_CHAN_MODE	<Dolby output mode. E.g. 3/2>
DOLBY_DECODE_PROG_1_LFE_CHAN to DOLBY_DECODE_PROG_8_LFE_CHAN	ON, OFF
DOLBY_DECODE_PROG_1_STEREO_DOWNMIX to DOLBY_DECODE_PROG_8_STEREO_DOWNMIX	Not Indicated, Lt/Rt, Lo/Ro
DOLBY_DECODE_PROG_1_DIALNORM to DOLBY_DECODE_PROG_8_DIALNORM	<Dialog normalization level. E.g. -27dB>
DOLBY_DECODE_PROG_1_COMP_GAIN to DOLBY_DECODE_PROG_8_COMP_GAIN	If compre = 0 NONE, Film Standard, Film Light, Music Standard, Music Light, Speech If compre = 1 <+48.16dB to -42.14dB>

Log Field	Log Value
DOLBY_DECODE_PROG_1_DYN_RANGE_GAIN to DOLBY_DECODE_PROG_8_DYN_RANGE_GAIN	If dynrnge = 0 NONE, Film Standard, Film Light, Music Standard, Music Light, Speech If dynrnge = 1 <+24.08dB to -18.08dB>
AES_INPUT_1_STATE to AES_INPUT_8_STATE	FAIL:Lost, OK, LOST, Unknown
AES_INPUT_1_TYPE to AES_INPUT_8_TYPE	PCM, NONPCM, Unknown, WARN: No Input
AES_OUTPUT_1_STATE to AES_OUTPUT_8_STATE	FAIL:Lost, OK, LOST, Unknown, OFF
AES_OUTPUT_1_TYPE to AES_OUTPUT_8_TYPE	FAIL:Lost, OK, LOST, Unknown
SN	<Serial number>
OS_VERSION	<Operating system version>
BUILD_NUMBER	<Software build number>
HARDWARE_VERSION	<Hardware version>
FIRMWARE_VERSION	<Firmware version>
UPTIME	<Uptime>

The Setup Screen

The **Setup** screen displays basic information about the unit. Use the **Setup** screen to reset the unit to factory defaults, and to reboot the unit.



The following information is displayed:

- **Product:** Displays the name of the module.
- **Software Version:** Displays the current software version.
- **Serial No.:** Displays the module's serial number.
- **Build:** Displays the factory build number. This number identifies all parameters of the unit (software version, build level, etc...).
- **Firmware:** Displays the version of the firmware system.
- **KOS:** Displays the version of the operating system.
- **PCB:** displays the PCB revision number.

To reset all of the unit's control to their preset values, except for those stored in memory, click **Default Settings**.

To reset the all of the unit's adjustments to their factory defaults, click **Factory Defaults**.

To reboot the unit, simulating a power-up/power-down cycle, click **Restart**.

Note: *The **Engineer Page Password** provides access to the engineer page. Access to this page is reserved for the use of Snell & Wilcox staff.*

Operation from an Active Control Panel

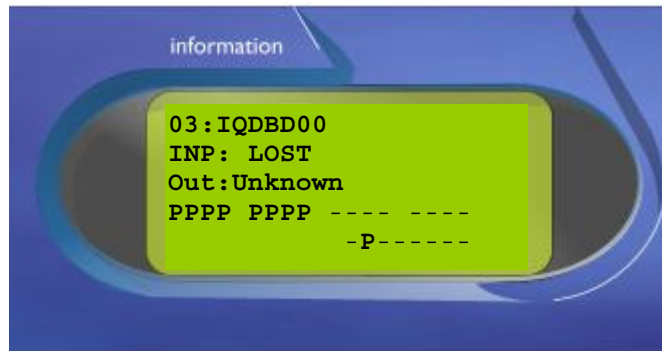
The card can be operated from an active control panel via the RollCall network.



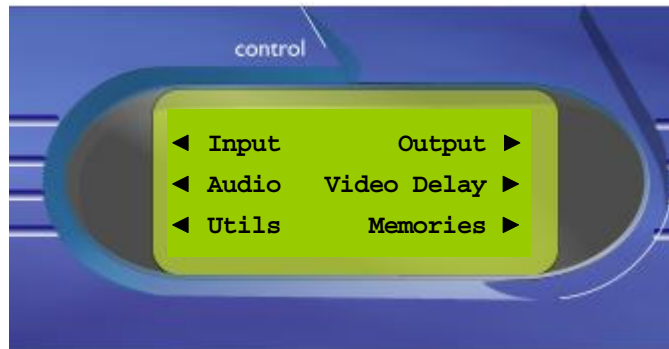
All operational parameters and selections are made using a system of menus displayed in two LCD windows. These menus duplicate the functions that are available via the RollCall control panel screens.

Operational details for the remote control panel can be found in the Modular System Operator's Manual.

The Information window has four lines of text indicating the current state of the unit.



The Control window displays all Selection Menus and sub-menus. Selections are made by pressing the button adjacent to the required item.



The main or top level menu allows various sub-menus to be selected by pressing the button adjacent to the required text line.

Note that where a menu item is followed by three dots (...) this indicates that a further sub-menu may be selected.

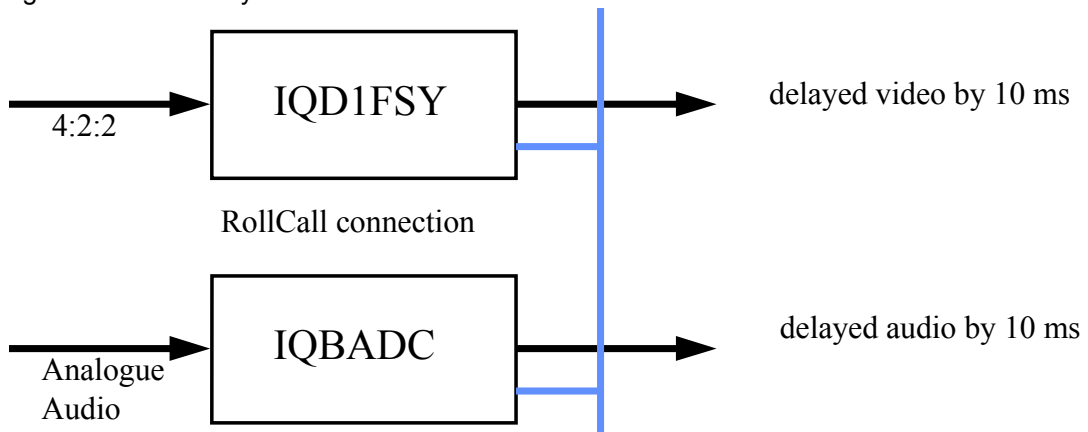
Whenever a menu item is selected the parameters of that selection will be displayed in the Information window of the front panel. Where the selection is purely a mode selection and does not enable a sub-menu, the text will become reversed (white-on-black) indicating that the mode is active. If the mode is not available for selection the text will remain normal.

RollTrack Audio Delay Tracking

RollTrack is a feature of RollCall™ (Snell & Wilcox's proprietary remote control system), that allows devices to communicate across the RollCall network with no direct user intervention.

RollTrack Audio Delay Tracking enables Snell & Wilcox RollCall™ compatible audio delay products to track delay introduced by RollCall™ compatible video processing products.

The simplest configuration is a single video unit and a single audio delay in a RollCall™ system. The audio delay will have the same delay as through the video path. If the delay changes the audio delay will track.



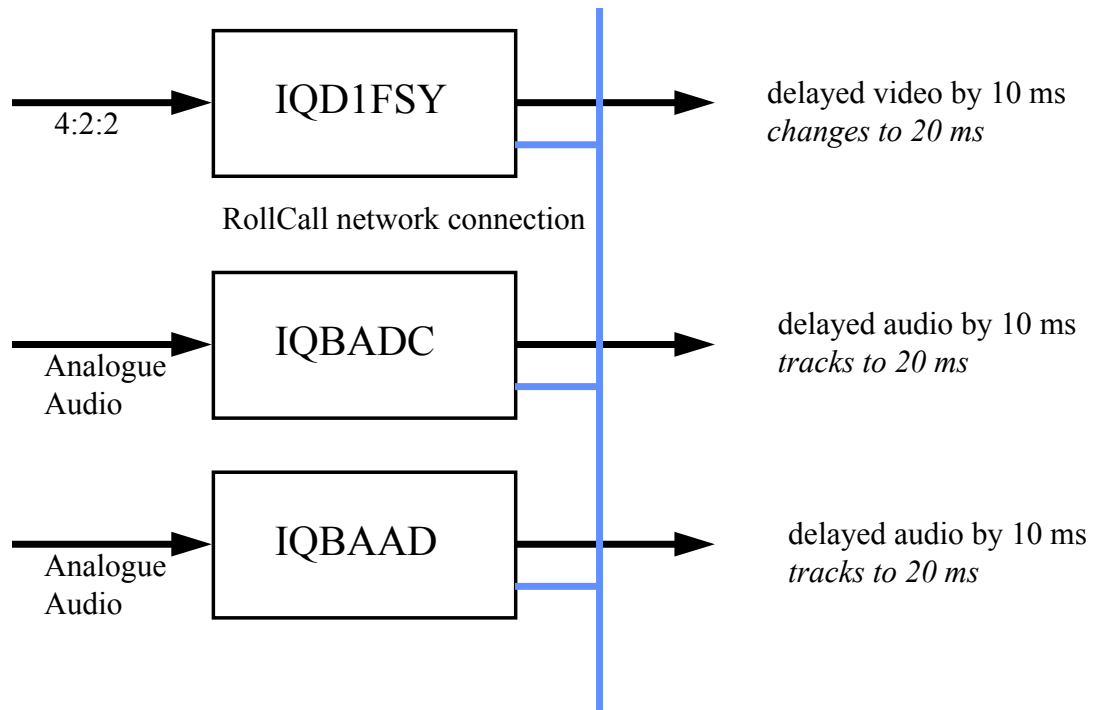
The next level of configuration is where there are multiple Frame Synchronizers (for example) each connected through RollCall™ to their own tracking Audio Delay. (It is worth stating that the synchronizers and audio delays do not have to be in the same enclosure; the addressing scheme, discussed later, allows for the units to be positioned anywhere in the RollCall™ domain.)

The maximum number of video units and audio delays in a RollCall™ system is set by the maximum limit of the number of modules in a RollCall™ network and is currently 3840 on a single network without bridges.

The unique identification of the destination unit (a decimal number) for various modules is as follows:

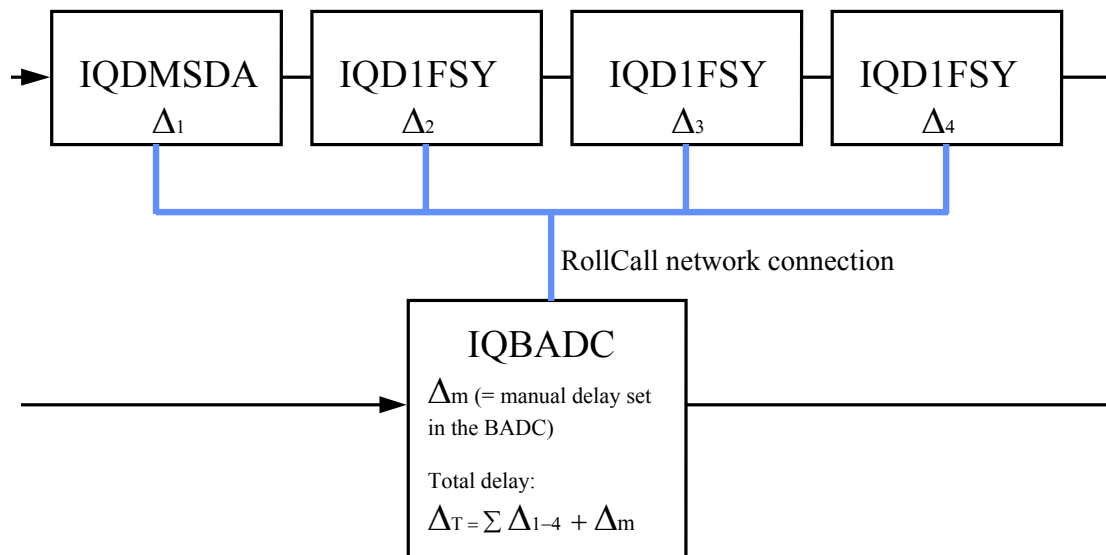
Module	ID
IQBADC	51
IQBDAC	52
IQBAAD	53
IQBDAD	54
IQBSYN	89
IQBADCD	107

The next level of complexity is a *vertical delay cluster* where a video unit can have up to eight audio delays tracking - of the same or different types.



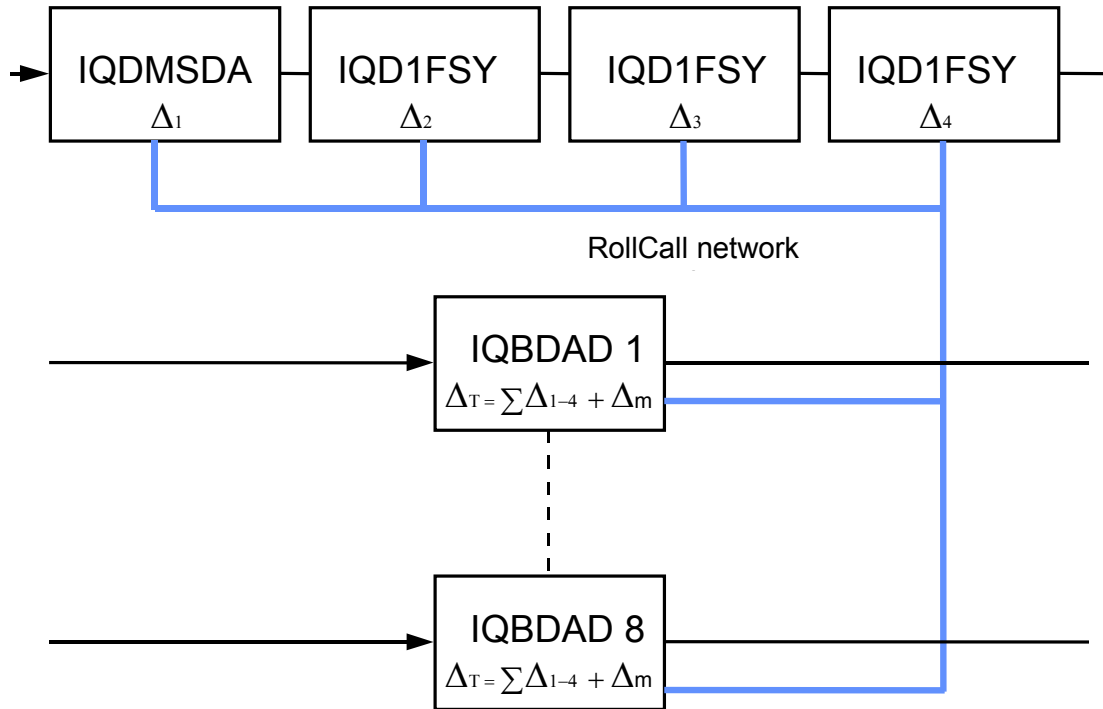
From one to eight audio delay products can be connected via RollCall™ to a single frame synchronizer, for example. If the synchronizer delay changes, then however many audio delays are connected will track the delay. The audio delays can also have a manual delay which will be added to the RollTrack delay.

The next level of complexity is a *horizontal delay cluster* where an audio delay can track up to four video units.



The total delay time through the audio delay is then the sum of the individual delays introduced by the video units plus the manual delay of the audio unit. The manual delay can be set to compensate for any fixed propagation delay in the video path or may be set to zero.

The next level of complexity is a *matrix delay cluster* where each audio delay (up to eight) can track up to four video units. This configuration is in effect a four by eight matrix of video units and audio delay units. The total delay time through the audio delay units is then the sum of the individual delays introduced by the video units plus the manual delay of the audio unit.



As any of the delay times change in the video path so will the audio delay time track this delay.

A virtual connection is made between from, say, an IQD1FSY to an IQBDAD by:

- selecting the *Setup...* Menu of the IQD1FSY
- then selecting the *Audio_Delay...* Menu
- then choosing from *Unit_1* to *Unit_8*
- then entering the unique network address of the IQBDAD in the form *nnnn:xx:yy*z*d* where
 - nnnn* = network address and in most cases will be 0000(hex);
 - xx* = IQ enclosure address (hex);
 - yy* = slot address of the IQBDAD (hex)
 - z* = the connection (or channel) number (decimal) - see table below.
 - d* = the unique identification of the destination unit (decimal) The ID entered

must match the

receiving units own ID or else the command will be ignored. If the ID value is set to 00, the receiving unit does not perform an ID match and will always accept the incoming command

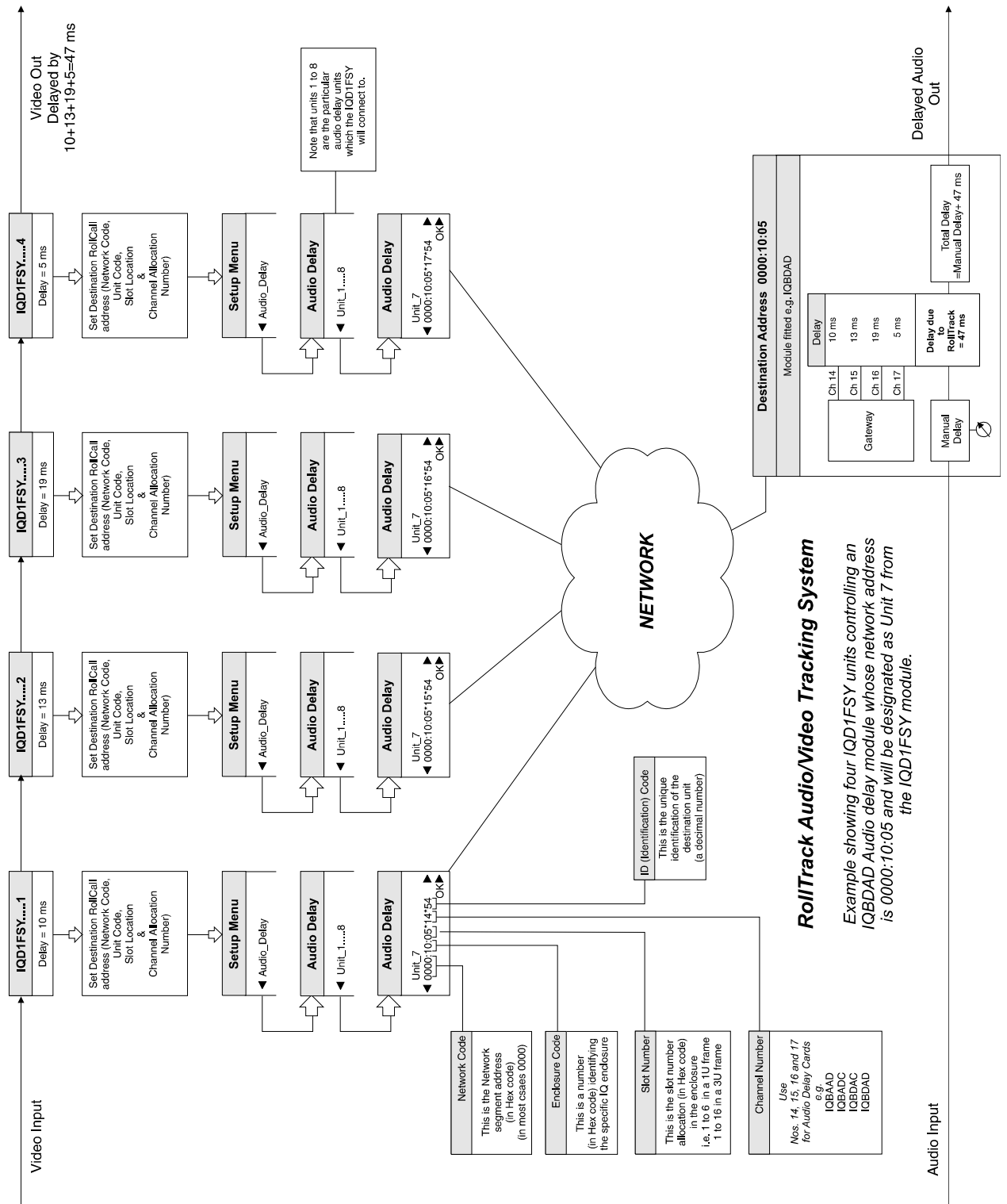
- then selecting the *Delay...* Menu of the IQBDAD
- then selecting *RollTrack*

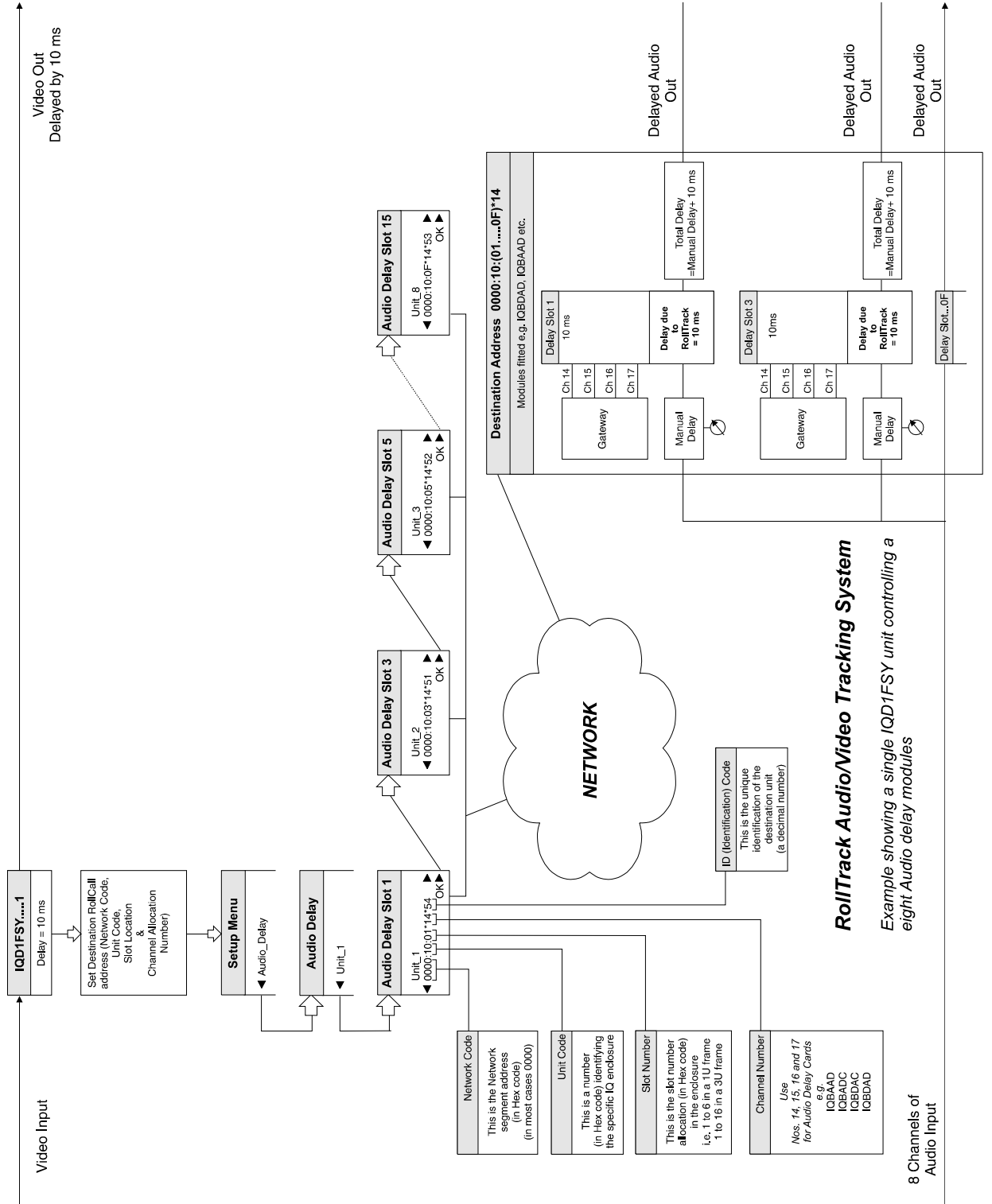
Example of Network Addresses with Channel Numbers and ID Numbers

	D1FSY 1	D1FSY 2	D1FSY 3	D1FSY 4
Audio delay 1	0000:10:01*14*54	0000:10:01*15*54	0000:10:01*16*54	0000:10:01*17*54
Audio delay 2	0000:10:03*14*54	0000:10:03*15*54	0000:10:03*16*54	0000:10:03*17*54
Audio delay 3	0000:10:05*14*54	0000:10:05*15*54	0000:10:05*16*54	0000:10:05*17*54
Audio delay 4	0000:10:07*14*54	0000:10:07*15*54	0000:10:07*16*54	0000:10:07*17*54

Audio delay 5	0000:10:09*14*54	0000:10:09*15*54	0000:10:09*16*54	0000:10:09*17*54
Audio delay 6	0000:10:0B*14*54	0000:10:0B*15*54	0000:10:0B*16*54	0000:10:0B*17*54
Audio delay 7	0000:10:0D*14*54	0000:10:0D*15*54	0000:10:0D*16*54	0000:10:0D*17*54
Audio delay 8	0000:10:0F*14*54	0000:10:0F*15*54	0000:10:0F*16*54	0000:10:0F*17*54

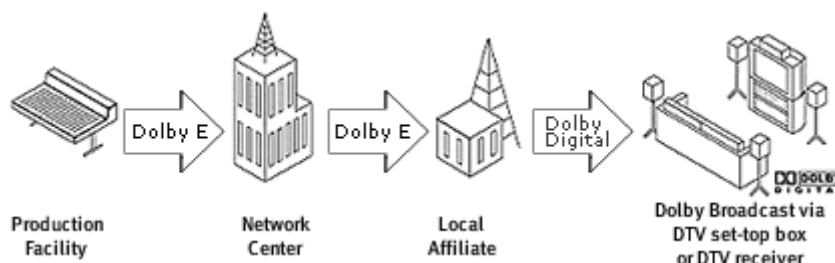
The most complex system would be an array of matrix delay clusters





Dolby-E

What is Dolby E?



Dolby E is a digital audio technology optimized for the distribution of surround and multichannel audio through digital two-channel postproduction and broadcasting infrastructures.

The Dolby E signal does not reach viewers at home. It is decoded back to baseband audio just prior to the final DTV transmission and then re-encoded into the final audio format specified by the various DTV emission systems (for example, Dolby Digital in ATSC, DVB, satellite, and cable systems; and AAC for ISDB in Japan).

With Dolby E, up to eight channels of audio, plus consumer and professional metadata, can be distributed via any stereo (AES/EBU) channel or recorded onto two audio tracks of conventional digital video tapes, video servers, communication links, switchers, and routers. Because the frame rate of Dolby E matches that of the video it accompanies, programs can be effortlessly switched, edited, and successfully encoded and decoded many times throughout the various stages of the broadcast chain. Audio/video synchronization is also simplified, with exactly one frame of delay added per Dolby E encode or decode stage.

Dolby E and Metadata

Metadata allows content providers unprecedented control over how a program will be reproduced in the home. Dolby E conveniently transports both consumer and professional metadata created during program production. Consumer parameters (transferred as a serial data stream from Dolby E to Dolby Digital codecs during the final audio encode, just prior to multiplexing with the digital video for final DTV transmission) are carried in both the Dolby E and the Dolby Digital bitstreams, while professional parameters are carried only in Dolby E and never reach viewers. All metadata parameters can pass unchanged through the various broadcast distribution stages.

Dolby E Partner Program

The Dolby E Partner Program provides broadcasters and systems designers with information about Dolby E compatibility of professional broadcast products. This information will help you plan your product purchase and system design decisions to create a clear path in your facility for the benefits and advantages of Dolby E technology.

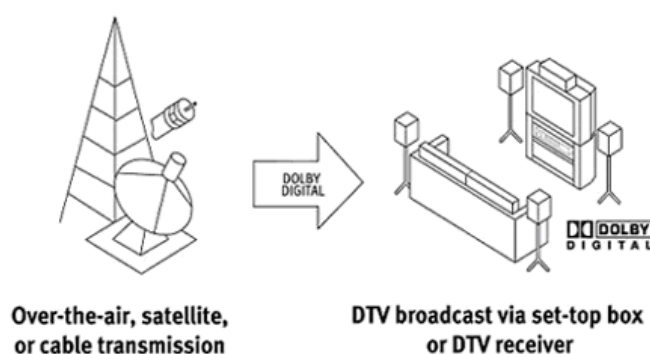
16 and 20-bit Dolby E

Any signal path that can carry a 16- or 20-bit audio signal, SMPTE 302M compatible, can carry Dolby E.

If the supported word length is 16 bits, the two channels together will offer a bandwidth of 1.536 Mbps, which Dolby E can use to carry six discrete audio channels plus metadata (descriptive and playback-control data related to the audio).

If 20-bit word length is available, the two channels will yield a combined bit rate of 1.92 kbps, which Dolby E uses to carry up to eight channels (perhaps a 6-channel surround mix plus a separate stereo mix), as well as metadata. A 24-bit mode is also specified for the Dolby E format but not yet implemented in encoders and decoders.

Dolby Digital and AC3



Also known as AC-3, Dolby Digital delivers up to 5.1 discrete channels of surround sound and is applied to the final broadcast transmission signal, just prior to multiplexing with the digital video. It is used extensively today on digital satellite (DBS), cable, and DTV and HDTV terrestrial services (including ATSC and DVB).

The flexibility of Dolby Digital enables broadcasters to deliver any number of audio channels, from all encompassing 5.1 surround sound down to mono audio, plus multiple-language programs and other specialized services. In the home, the Dolby Digital signal adapts to the viewer's playback system, delivering the best possible sound to all viewers, regardless of their equipment.

To enjoy the full 5.1 surround experience, viewers simply connect any DTV set-top box with a Dolby Digital output to a Dolby Digital home theater receiver.

Metadata

The key to the adaptability of Dolby Digital is metadata, "data about the audio data." Metadata is created during the production of DTV programming and is carried in the Dolby Digital bitstream. Metadata parameters maintain the "vision" of the audio created by a program's producer, make life easier for broadcasters, and give viewers the best audio DTV has to offer, whether viewers own mono, stereo, or 5.1-channel audio system

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Ancillary Passing and Audio Embedding

The SMPTE 272M and 299M standard specifications allow for up to four groups of AES/EBU digital audio to be embedded in a component digital video SD or HD stream respectively. Each group consists of two stereo pairs, comprising four channels, resulting in a total of sixteen audio channels for all four groups. This horizontal ancillary (HANC) data increasingly shares space with other types of ancillary data such as EDH, timecode and UMID labels making the insertion and passing a complex process.

Processing of audio by the module requires the reinsertion of the audio into the passed ancillary data. This is essentially separate from the extraction of audio from the input video, and allows the embedded audio to be firewalled when input disturbances force blanking of ancillary space.

For 625-line (or other 25Hz) video, 1920 audio samples occur in each frame. Whereas for 525-line (or other 29.97 Hz) video, 8008 audio samples occur over five video frames. Audio data is distributed evenly throughout each video frame, situated in the non-active picture regions between the end of one line and the start of the next. In SD, most lines contain three or four audio samples per active group. In HD, most lines contain one or two samples per active group. Additional lines are reserved after the SMPTE RP-168 switch points, which contain no samples. This is in contrast to most ancillary data, which resides in a fairly fixed HANC or VANC (vertical ancillary) space.

Audio Control and Extended Data packets, as defined in SMPTE 272M (SD), are not inserted by the module. Instead, they are handled by the 'passing' and 'blanking' functions. Audio Control packets, as defined in SMPTE 299M (HD), are always inserted. Block numbering as defined in SMPTE 291M is supported.

The SMPTE 291M standard defines the structure and space formatting for ancillary data within digital video streams. Within HANC data space, ancillary packets follow immediately after the end of active video (EAV) marker, including line numbers in HD. They are contiguous with each other until either the end of the last packet or the start of active video (SAV) marker. Unused space is filled with black level blanking. Likewise, VANC data space follows from the SAV to the EAV markers throughout vertical blanking, which is in principal similar to vertical blanking interval (VBI) use in legacy SD environments for typically analogue data waveforms.

The module allows separate blanking of HANC and VBI line spaces to completely remove data stored within those spaces for when specific applications demand it.

Complying with SMPTE 291M, the embedder uses a two-stage process, as follows:

- The embedder reformats the incoming ancillary to remove unwanted data which is already marked as deleted or audio packets that are marked for removal here, for example, when blanking or reinserting. Additional packets that are always removed are EDH packets (which are always reinserted) and Start and End Marker packets (which are never reinserted). This makes for the most efficient use of the data space and gives priority to passed data.
- The new audio data and control packets are appended in the remaining space.

The ancillary formatter passes any data space with no recognisable ancillary packets unchanged, but will operate when it finds a packet after EAV or SAV.

Data packets for passing are buffered while deleted packets are dropped. This requires a holding buffer and a delay so that all passed packets can be shuffled up earlier in the data space without gaps.

The following embed control section looks for the first free space after EAV. Once found, the embedding process is initiated with the first enabled group inserter. However, before starting the embedding for any of the inserter streams the remaining data space is checked to ensure that there is enough room for the ancillary packet. If not, the inserter is disabled to prevent corruption of the video format or generation of invalid ancillary data packets. This situation is a sample overflow which simply causes the data to be delayed until the following line. If this is repeated over many lines, there will be a full embed overflow, which is reported in the Audio Status window, on the card edge yellow Warning LED, and is logged.

Embedded ancillary data that does not conform to the space formatting requirements of SMPTE 291M may not be recognised, and so could be destroyed by the module. For example, if the data packets do not start immediately following the EAV marker the inserter will overwrite the packets. Similarly, if a data packet does start in the correct place but a second packet does not immediately follow the first one the second packet will be overwritten.

To summarize:

- Marked for Deletion and audio packets to be replaced, are removed from the ancillary data space for most efficient and clean use of data space.
- Ancillary packets are shuffled up within the data space during reformatting.
- Incoming ancillary data must be SMPTE 291M compliant.

