

Operating Manual

Precision Audio Monitoring



PAM1 MK2

Handbook 1. Issue 2

SAFETY

Installation.

Unless otherwise stated TSL equipment may be installed at any angle or position within an operating temperature range of 5°C - 30°C.

All TSL equipment conforms to the EC Low Voltage Directive:

EC Low Voltage Directive (73/23/EEC)(OJ L76 26.3.73)(LVD). Amendment: (93/68/EEC) (OJ L220 30.8.93).

In all cases the frame of the equipment must be earthed on installation.

Where appropriate, the earth pin on the IEC mains inlet connector is connected to the metal frame of the equipment, to 0 volts on the internal DC PSU and to signal ground unless otherwise stated. All metal panels are bonded together.

Check that the voltage selector setting (if fitted) and the fuse rating is correct for the local mains supply.

Due consideration for cooling requirements must be given when mounting the equipment. It is recommended that a 1RU of rack space, or a vent panel, should be left above and below the unit.

WARRANTY, MAINTENANCE AND REPAIR

- All TSL products have a one year warranty period starting from the date it leaves the factory.
- A repair warranty is to apply. That is, the product is to be returned for repair with no replacement and an exchange shipping policy is also to apply.
- TSL offers a seven day DOA policy together with an exchange shipping policy. That is, if a product has been declared 'dead on arrival' within a seven day period a warranty replacement will be shipped.
- A temporary replacement may be available where, for operational reasons, it is imperative that service is continued. The customer will be asked to enter into a 'loan agreement' for the duration of repair.
- All faulty equipment returned to TSL for repair will, where possible, be returned to the customer within seven working days.

TSL Returns Procedure

Please telephone +44 (0)1628 676221 (Fax: +44 (0)1682 676299) and ask for Customer Support, detailing the model and serial number of the equipment, who will provide a Returns Number. This will enable us to track the unit effectively and will provide some information prior to the unit arriving.

For each item, this unique Returns Number must be included with the Fault Report sent with the unit.

A contact name and telephone number are also required with the Fault Report sent with the unit.

Fault report details required.

- Company:
- Name:
- Address:
- Contact Name:
- Telephone number:
- Fax number:
- Email address:
- Returns Number:
- Symptoms of the fault (to include switch setting positions, input signals etc):

Packing

Please ensure that the unit is well packed as all mechanical damage is chargeable. TSL recommends that you insure your equipment for transit damage.

The original packaging, when available, should always be used when returning equipment.

If returned equipment is received in a damaged condition, the damage should be reported both to TSL and the carrier immediately.

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1.0 Introduction

The PAM1 is a 1RU x 280mm deep Audio Monitoring Unit (AMU) with two LCD displays for audio level measurement, video confidence and metadata status indication.

PAM1 has been designed in conjunction with some of the worlds leading broadcasters to produce arguably the most comprehensive, intuitive and feature packed product of its type.

As with any new product which relies on complex software, it is possible that you may find minor bugs or perhaps think of enhancements which would improve the operation of PAM1. In the event of either scenario, please feel free to contact TSL via your local reseller or directly on +44 1628-676221, asking for the PAM Product Manager.

TSL will be releasing upgrades and feature enhancements from time to time – as a purchaser of a PAM1 product you can receive these free of charge through your reseller or directly from TSL. Please refer to www.tslproducts.com for announcements.

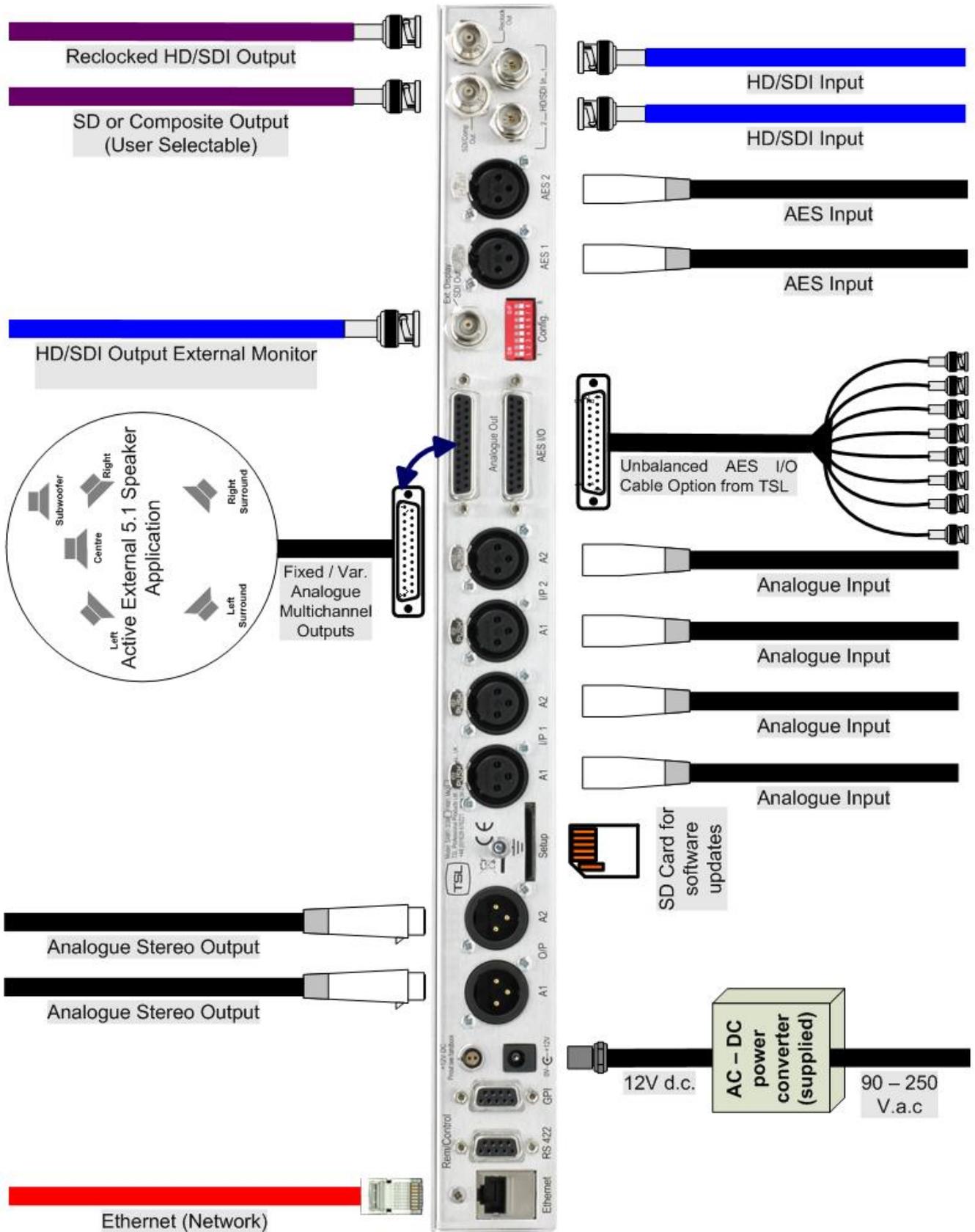
PAM1 MK2 is available in Dolby decoding and non-Dolby versions.

The following features are standard:

- Dual auto-sensing, 1080p (60, 59, 94 and 50Hz), HD/SDI video inputs
- De-embedded audio monitoring from video (HD/SDI) with intuitive selection from up to sixteen channels (four groups)
- 4 AES (4 Pairs/8 Channel) Inputs– 110 Ohm Balanced or 75 Ohm unbalanced via optional CAB-D25-BNC cable
- 2 Analogue Stereo Inputs
- Ethernet for Network connectivity (future use)
- USB connector for system updates and preset save/recall
- SD card slot for system updates and preset save/recall
- ‘Scroll To Hear’ – top level bargraph navigation using rotary encoder
- Downmix of discrete multichannel audio to stereo for compatibility monitoring
- Multichannel audio outputs for external stereo or 5.1 loudspeaker system integration
- ITU BS1770/71 loudness measurement and Histogram
- User Programmable ‘Assign Matrix’
- Audio Output Delay
- Automatic mode switching – depending on signal type
- Dolby E, Dolby Digital and Dolby Digital Plus decoding from HD.SDI and AES signal sources
- Lt Rt and Lo Ro Downmixing of Dolby Signals
- Hi Resolution Video confidence monitoring
- Re-clocked HD/SDI with down converted SDI or composite (PAL.NTSC) video outputs selected via rear panel Config. Switch.
- Dual high resolution LCD screens for 16 bargraphs, setup and metadata display.
- Choice of user selectable bargraph scales (BBC PPM, EBU PPM, EBU Digital, Nordic, VU and DIN)
- User programmable presets. 5 Hotkeys by hardware buttons, GPI and 24 internal, accessible by high level menu selection
- Fixed or variable analogue multichannel outputs (8 mono)
- Fixed or variable AES multichannel outputs (4 pairs/8 Channels)
- Variable stereo analogue outputs
- High quality internal full range loudspeaker system
- Dual 12V DC inputs
- Serial remote control
- Headphone output with LS muting
- Compact, lightweight (3.1Kg) 1RU case, 280mm deep

The PAM1 MK2 represents the natural evolution of the award winning PAM1 3G audio monitoring family. In purchasing the PAM1 MK2 you are investing in a solution which will evolve to meet your needs at no further cost to you the owner. As part of TSL company policy, software updates for the product will be free charge for the life of the unit (subject to conditions) and will unlock new features and improve existing ones.

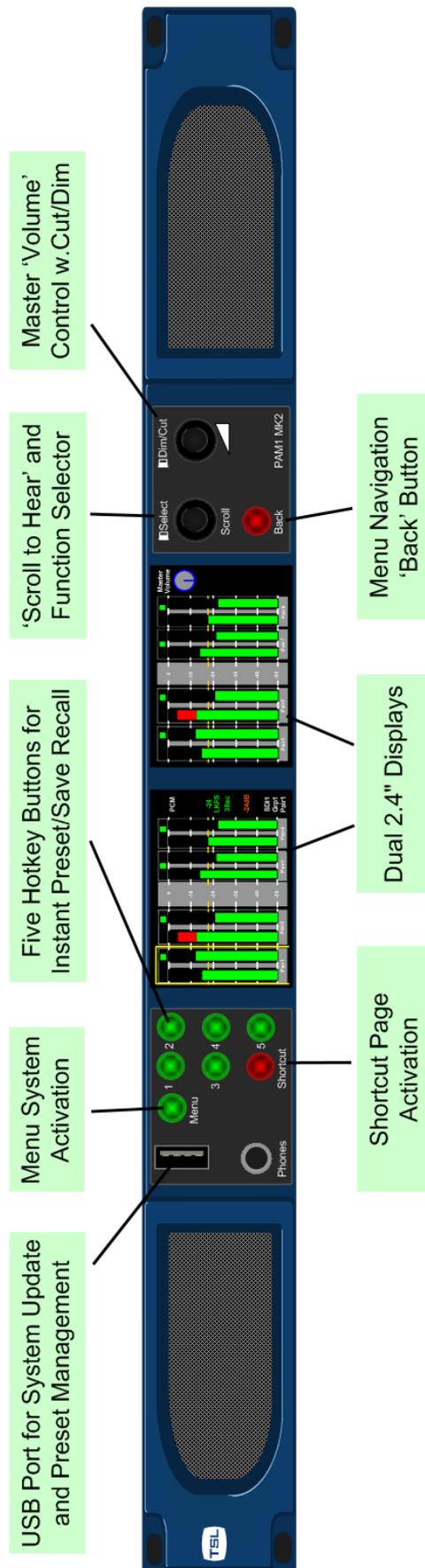
1.1 PAM1 MK2 Hookup Schematic



1.2 PAM1 MK2 Front and Rear Panel



1.3 PAM1 MK2 Front Panel Functions – Quick Lookup



2.0 Getting Started

The PAM1 is designed to be user friendly and intuitive to operate. The menus and functions will feel familiar to both users of TSL multichannel audio monitoring products and those buying a TSL solution for the first time.

The PAM1 MK2 comprises the complete operating system and features of the PAM2 MK2 however a number of additional features such as the 'Shortcut' button, which unique to PAM1 MK2 and whose functionality is described within this document.

Important Note: PAM1 ships with a default 0dBu reference level set to -18 dBFs, the default operating scale is EBU Digital. A simple power up sequence may be used to alter the operating reference level to -20 dBFs with a default EBU Digital operating scale.

For -20 dBFs default operation – apply DC power to the rear power connector whilst simultaneously pressing buttons Menu and '2'. The following message will appear on the right hand LCD signalling a successful operation.

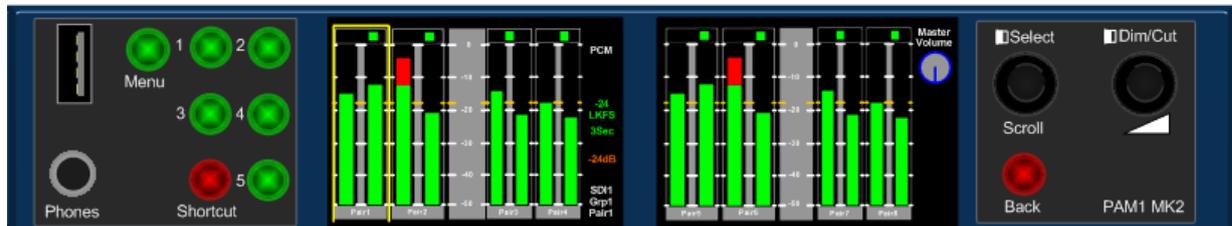
“Default memories cleared Restored Factory - 0dBu = -20dBFs”

For -18 dBFs default operation – apply DC power to the rear power connector whilst the menu button. A message will appear on the right hand LCD signalling a successful operation.

“Default memories cleared Restored Factory - 0dBu = -18dBFs”

2.1 Front Panel Controls and Displays

PAM1 MK2 menus and features are navigated via a shortcut button, multi-function encoder and a bank of five hotkey pushbuttons. No monitoring function is ever more than two clicks away from the front panel view.



- The left hand LCD screen displays bargraphs 1-8 in **Full-screen** mode and bargraphs 1-16 when Menu / Data or Video modes are activated.
- The left hand LCD also displays decoded Dolby bargraphs, Downmix components and Assign Matrix selections
- The right hand LCD screen displays bargraphs 9-16 in **Full-screen** mode and Menu / Data or Video screens when activated.
- The left hand rotary encoder is used to activate the '**Scroll To Hear**' feature and to make selections within **Menus** and **Presets**.
- The volume knob is used to adjust internal/external loudspeaker and headphone levels. The control also features a 'push' action which **Cut/Dim** loudspeaker audio.
- The '**Shortcut**' button is the key to simple operation of **PAM1 MK2**. Press **Shortcut** to access a unique menu page which includes the most commonly used menu commands and features.
- The USB port is used to save and load **User Presets** to a compatible memory storage device
- The 5 **Hotkey** buttons work like the presets on a car stereo. Push and hold a Hotkey to instantly save a favourite monitoring mode
- The Factory Default condition ships the following Presets pre-programmed on Hotkey buttons 1-5 respectively. These may be overwritten at any time: defaults can be restored via the reset procedure described in Section **2.20** of this handbook.
 1. SDI 1 Input Pair 1
 2. SDI 2 Input Pair 1
 3. AES Input Pair 1
 4. Analogue Input Pair 1
 5. Unassigned

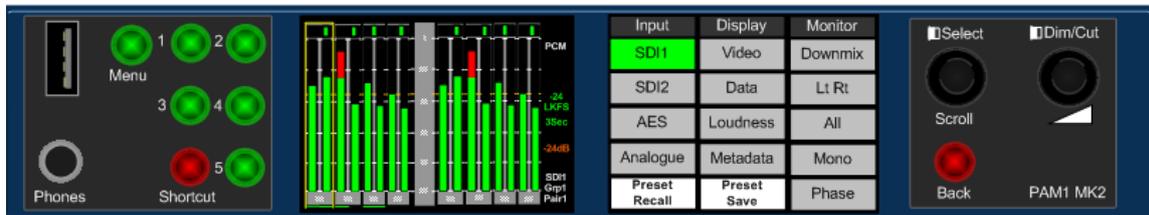
2.2 Rear Panel Connectivity

Unique to the PAM1 MK2, the rear panel features an External Monitor HD-SDI output BNC which can be connected to a compatible external display device directly via SDI or by use of a 3rd party convertor to HDMI or DVI equipped monitors.

The Ethernet port fitted to the rear of the unit is not active in the initial product release however it will be activated by a forthcoming software release and application software will follow in due course.

2.3 Shortcut Button and Menu

The Shortcut button is a feature unique to PAM1 MK2. It provides instant access to the most commonly used menu functions and enables the user to perform the majority of the device 'hands on' functions via just a single button push.



The Shortcut menu is divided into four sections:

- Input Select
- Display Mode
- Monitoring Mode
- Preset Save and Recall

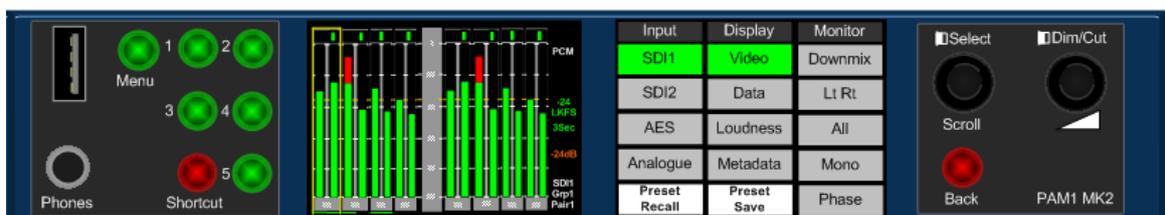
2.3.1 Input Select

The Shortcut menu provides instant access to the SDI1, SDI2, AES and Analogue sources. The chosen input is routed instantly to the bargraph display and the Shortcut menu exited automatically.

2.3.2 Display Mode Select

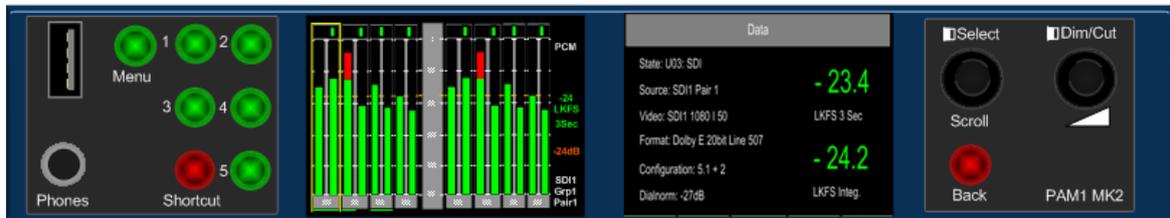
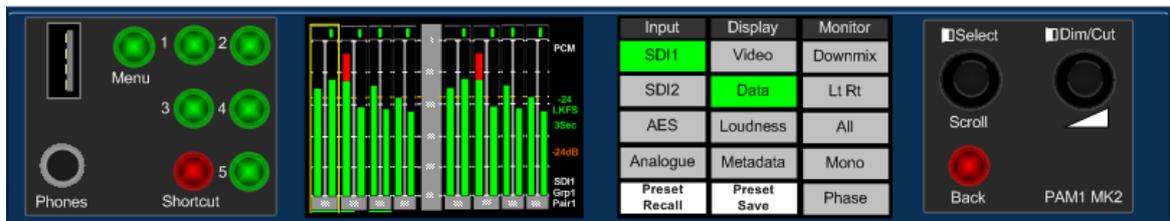
The Display Mode select provides instant access to four display options available for the right hand screen (other than bargraph). By using the encoder to choose one of Video, Data, Loudness or Dolby Metadata options instantly routes the chosen display mode to the right hand LCD screen. The shortcut menu is exited automatically.

- Display Mode - Video



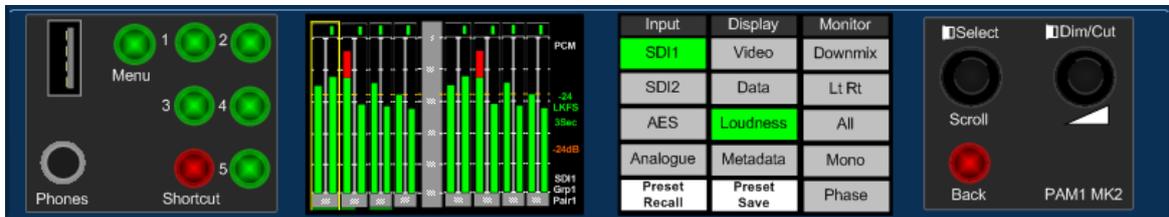
You can save this mode to a Hotkey switch for instant recall of the video preview display

- **Display Mode - Data**



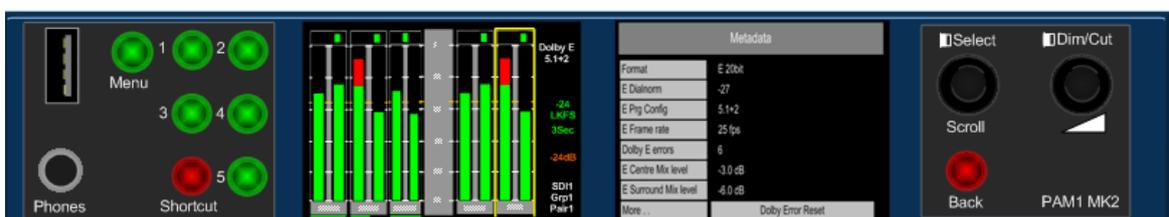
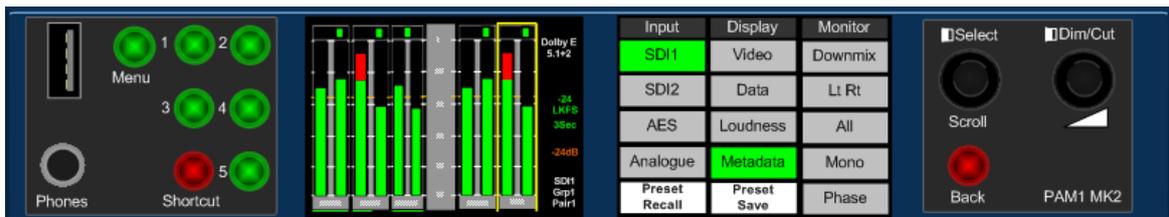
You can save this mode to a Hotkey switch for instant recall of the Data page

- **Display Mode – Loudness Histogram**



You can save this mode to a Hotkey switch for instant recall of the Loudness Histogram view

- **Display Mode – Dolby Metadata.** This mode is only available when a Dolby signal is decoded.



2.3.3 Monitoring Mode Select

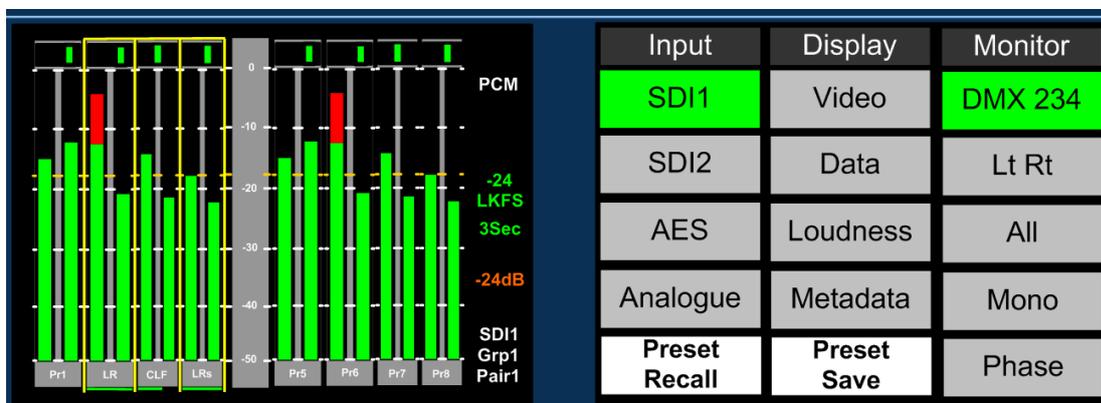
The Monitor Mode select accesses five of the most commonly used monitoring modes available to PAM1 MK2 users. Since the available monitoring modes are generally used dynamically during hands-on operation, the Shortcut Menu remains open even when a selection is made. The Back or Shortcut buttons are used to exit the Shortcut Menu after a monitoring mode selection has been made.

- **Monitoring Mode – Downmix**

The PAM1 MK2 **Downmix Mode** allows the operator to select an adjacent group of 3 audio pairs (6 audio channels) and create a stereo **Downmix**

By repeatedly selecting the **DnMix** button, the group of selected audio pairs move from left to right across the bargraph display enabling the user to **Downmix** discrete 5.1 or 5.0 audio from any location within the 16 channels displayed. Pressing and rotating the **Downmix** button also enables the user to scroll through the pair's selection in the sequence **123, 234, 345, 456, 567** and **678, downmixing** the appropriate surround sound audio signals as desired.

The **Downmix Mode** assumes that audio channels are presented in the correct **SMPTE** order (L, R, C, Lfe, Ls, and Rs). The **Downmix Mode** uses the ITU-R BS.775 algorithm standard.

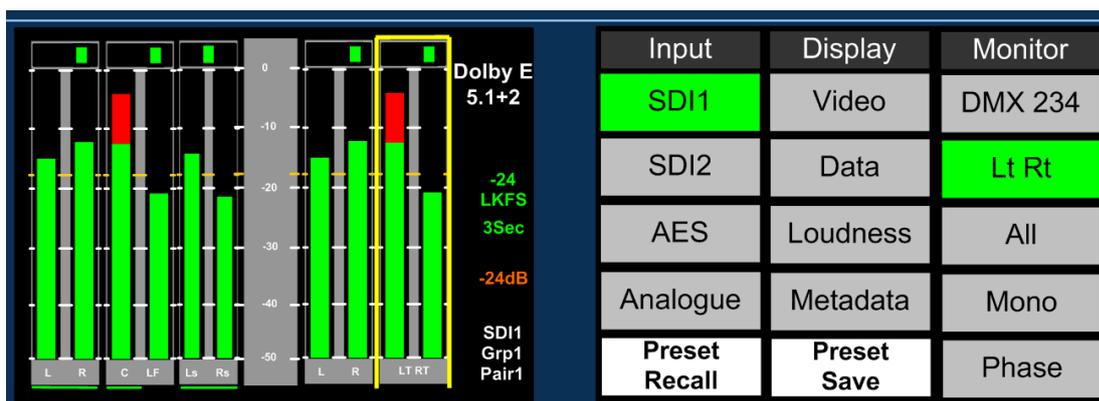


The example above shows how SDI1 pairs 2, 3 and 4 can be Downmixed to stereo

Once the 5.1 audio components are selected and **Downmixed**, the selection can be stored to a **Hotkey** for instant recall.

- **Monitoring Mode – Lt Rt / Lo Ro (Dolby Downmix)**

The PAM1 MK2 **Lt Rt / Lo Ro (Dolby Downmix) Mode** allows the operator to instantly select the Dolby Downmix mode when decoding a Dolby E, AC-3 or Dolby Digital Plus signal. This mode is only active when a Dolby signal is present.

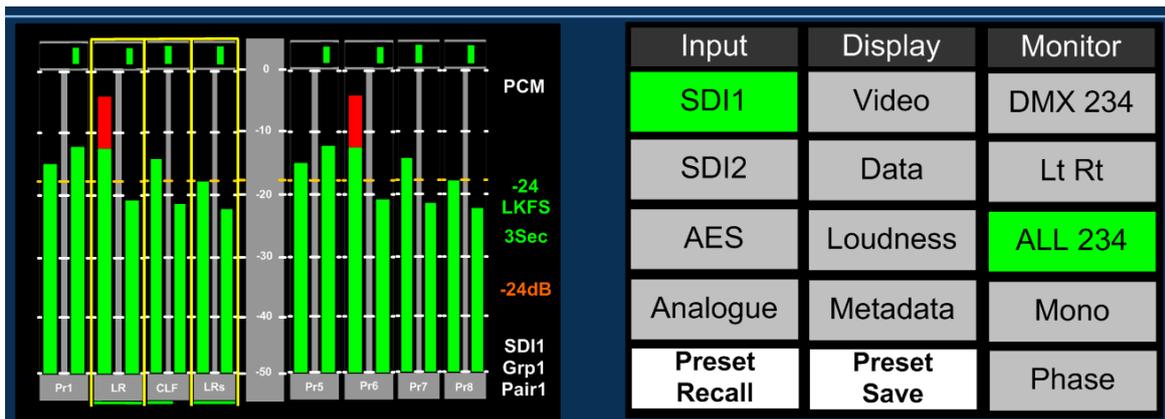


- **Monitoring Mode – ALL**

The **ALL** monitoring mode is a feature designed to be used when PAM1 MK2 is connected to an external active surround sound speaker system. The function defines a group of 5.1 or 5.0 audio signals that are to be routed to the analogue and AES multichannel outputs and level controlled via the rotary volume knob.

By selecting **ALL** the operator can select the group of 6 audios which corresponds to the 5.1 audio signals. Once a 5.1 group has been defined, the 5.1 audio stems are routed to the external speaker outputs in the correct **SMPTE** order (L, R, C, Lfe, Ls, and Rs).

By repeatedly selecting the **ALL** button, the group of selected audio pairs move from left to right across the bargraph display enabling the user to route discrete 5.1 or 5.0 audio from any location within the 16 channels displayed to the external variable audio connectors in order to feed external active 5.1 speaker systems. Pressing and rotating the **ALL** button also enables the user to scroll through the pair's selection in the sequence **123, 234, 345, 456, 567** and **678**.



This function ensures that an embedded discrete **5.1** signal can be monitored regardless of the location of the 5.1 stems. The user can store different settings for monitoring discrete 5.1 audio in their **User** and **Hotkey Presets**.

When **ALL** is activated, the PAM1 MK2 internal loudspeakers are muted automatically.

- **Monitoring Mode – Mono**

When the **Mono** button is selected, the monitored audio signal stereo signal pair is mono'd (sum = L+R). **Monitoring Mode – Mono**

When the **Mono** button is selected, the monitored audio signal stereo signal pair is mono'd (sum = L+R).

- **Monitoring Mode – Phase**

When the **Phase** button is selected, the left side of the monitored audio signal is reversed with respect to the right.

Monitoring Mode – Mono

When the **Phase** button is selected, the monitored audio signal stereo signal pair is mono'd (sum = L+R).

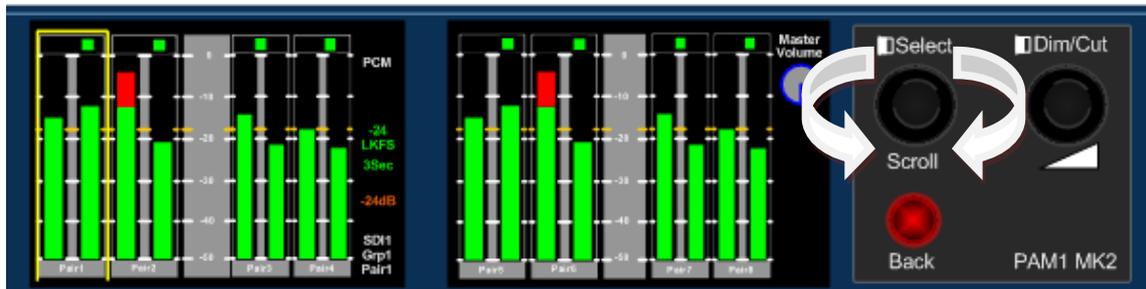
Input	Display	Monitor
SDI1	Video	DMX 234
SDI2	Data	Lt Rt
AES	Loudness	ALL 234
Analogue	Metadata	Mono
Preset Recall	Preset Save	Phase

2.4 Top Level Screen Mode

The example below shows PAM1 in dual screen **Metering Mode** listening to **Group 1 Pair 1** of an embedded SDI video signal.

2.4.1 Scroll to Hear

Push, hold and **rotate** the Encoder to move the yellow '**Scroll To Hear**' box to listen to any of the 16 embedded audio channels. This is the simplest means to navigate **PAM1** and can be used whenever the unit is not in a **Menu** or **Loudness Histogram** mode.



2.4.2 Dim Cut Control

The rotary volume control of **PAM1** also features a push switch cut/dim function. Push the volume control once to cut the volume to internal speakers, headphones and variable audio outputs. Pushing the volume control a second time will dim the output level by 12 dB. A final push will return the output volume to full level.

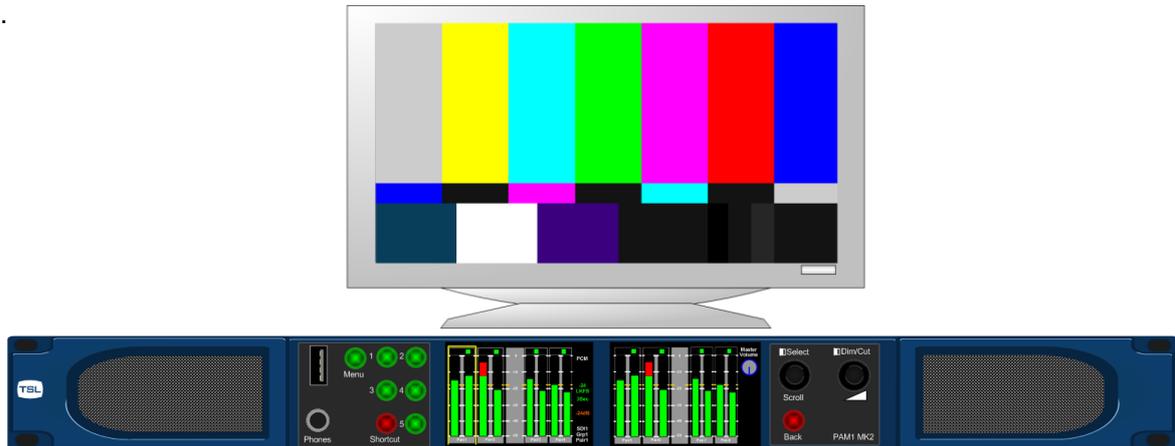
When the right hand LCD screen is in **Bargraph** mode the current condition of the **Master Volume** control is indicated via the graphical display as shown below.

- Blue denotes that volume control is **Normal**.
- Red denotes that volume control is **Cut**.
- Orange denotes that volume control is **Dim**.



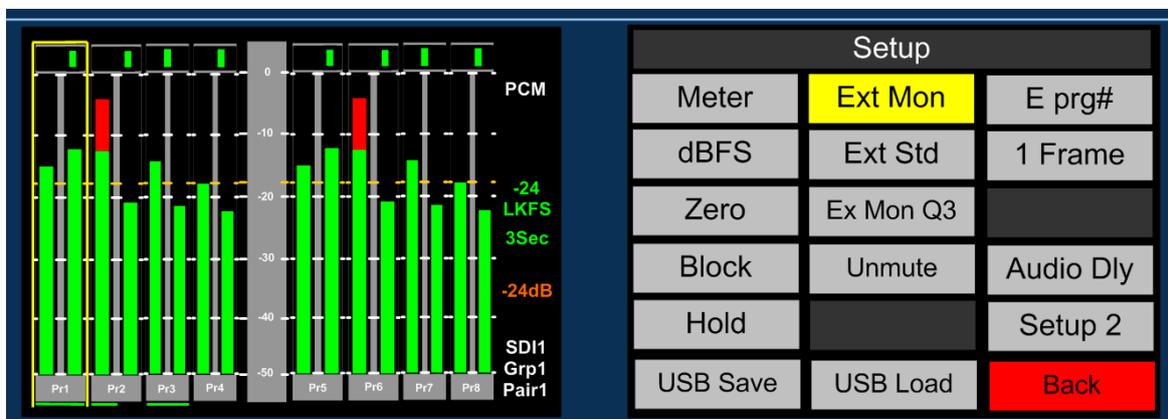
2.5 External Monitor Functionality

When an external monitor is first connected to PAM1 MK2 the screen will display a colour bar pattern signifying that the output is connected but not activated.



2.5.1 Activating the External Monitor Output

The External Monitor output is activated via the Setup menu. The menu is accessed by pressing the encoder, selecting **Setup** and then selecting **Ext Mon**



The default view upon activation will appear as per the image below with bargraph displays in the upper quadrants of the image.



2.5.2 Setting the External Monitor Output Format

The External Monitor output can be set to formats 1080i50, 1080i59.94 or 1080i60 with a 16:9 aspect ratio using the Setup Menu button Ext Std

Setup		
Meter	Ext Mon	E prg#
dBFS	50i	1 Frame
Zero	Ex Mon Q3	
Block	Unmute	Audio Dly
Hold		Setup 2
USB Save	USB Load	Back

2.5.3 Configuring the External Monitor 3rd Quadrant

The External Monitor comprises four quadrants of which the upper two permanently display the bargraph information from the PAM1 left and right LCD screens and the lower right quadrant, the currently active video signal image.

The lower right (3rd) quadrant can be configured to display Loudness Histogram, Data or Dolby Metadata pages 1 to 3 and is configured via the Ext Mon Q3 button from Setup Menu.

Setup		
Meter	Ext Mon	E prg#
dBFS	Ext Std	1 Frame
Zero	Histogram	
Block	Unmute	Audio Dly
Hold		Setup 2
USB Save	USB Load	Back

Setting the output to Dolby Metadata or Data will result in the following selections.

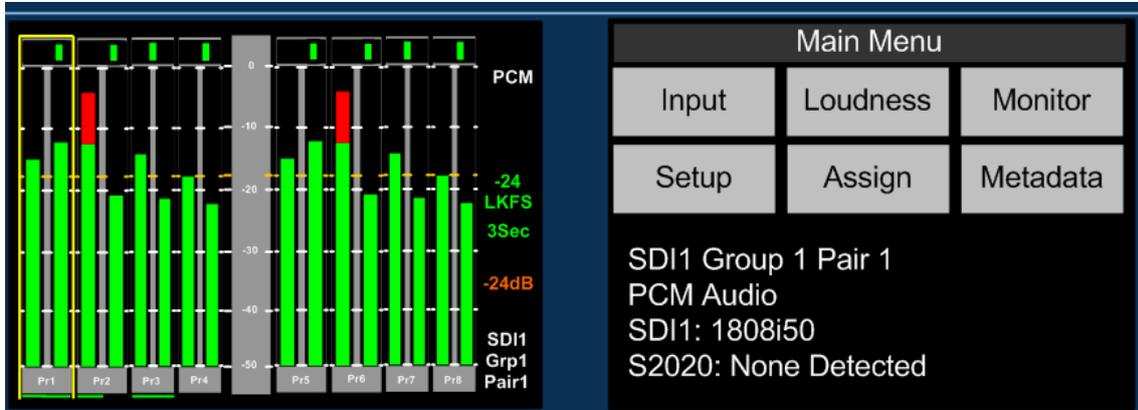
The left screenshot shows the 'Metadata' page with a table of audio parameters and a video image of fried chicken. The right screenshot shows the 'Data' page with numerical values for LKFS and Integ, and the same video image.

The External Monitor Output can be used with SDI monitors, Multiviewers or domestic PC screens via inexpensive SDI to HDMI/DVI/VGA converters.

2.6 Main Menu

Press the **Menu** button to access **Main Menu**. Use the **Scroll** encoder to highlight and select from the displayed **Menu** options. From any **Menu**, pressing **Back** (illuminated red) or the **Menu** button (**PAM1 MK2 only**) will either exit the current menu page or return directly to the top level screen view.

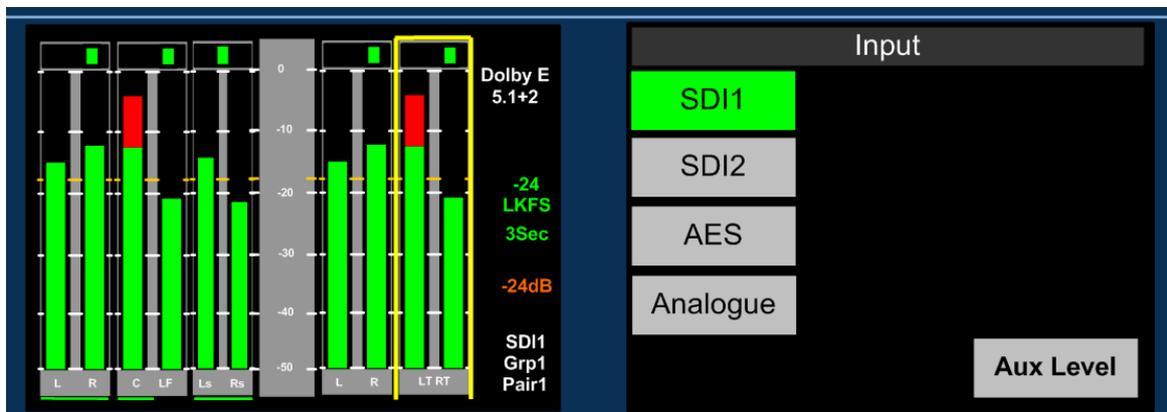
The Main Menu screen also displays basic data about the current selected input signal as shown.



2.7 Input Selection

PAM1 ships with **Hotkeys 1-4** pre-programmed to instantly access SDI1, SDI2, AES and Analogue inputs. Selecting these **Hotkeys** will directly route the selected input signal to the bargraph display. Since these **Hotkeys** are likely to be overwritten and customised as required, input selection can be made from the **Main Menu** screen or via the **Shortcut Menu** as previously described.

By highlighting and selecting the **Input** option, **SDI, AES and Analogue** inputs can be accessed. A **shortcut** button enables the user to 'jump' to the extended **Monitor Menu** once the desired input is selected. The **Aux Level** control is used in conjunction with the **Aux Mixer** described elsewhere in this document.



2.8 Monitor Menu – PCM (non-Dolby) signals

The **Monitor Menu** is generally used to activate listening modes not readily accessible via **‘Scroll To Hear’** selection or the **Shortcut Menu**. As PAM products have evolved over time since their introduction in 2008, more and more improvements to the operating system has meant that operators use the **Monitoring Menu** much more infrequently. Where alternative means to select **Monitoring** functions are available, they are described.

2.8.1 Phase Reverse

The **Monitor Menu** features a **‘Phase Reverse’** button which enables the user to reverse the phase of the right signal of a selected audio pair with respect to the left.

Note: This function can also be accessed directly via the **Shortcut Menu**

Monitor			
Pair 1	Pair 5		L-R
Pair 2	Pair 6		Phase
Pair 3	Pair 7		Mono
Pair 4	Pair 8		ALL
Mix	DnMix		

2.8.2 Left/Right/Swap

Pushing to select **L – R** enables individual selection of either **Left** or **Right** components of the highlighted stereo pair (or pairs).

The **L – R** function also enables **Left** and **Right** audio components to be **swapped**.

Monitor			
Pair 1	Pair 5		Left
Pair 2	Pair 6		Phase
Pair 3	Pair 7		Mono
Pair 4	Pair 8		ALL
Mix	DnMix		

2.8.3 Mono

The **Mono** mode button sums the left and right components of the selected audio channels.

Note: This function can also be accessed directly via the **Shortcut Menu**



2.8.4 Mix

When **Mix Mode** is activated, multiple bargraph pairs may be selected and monitored as a mixed stereo pair. Up to four pairs may be mixed together simultaneously.



2.8.5 Downmix

Downmix Mode allows the operator to select an adjacent group of 3 audio pairs (6 audio channels) and create a stereo **Downmix**.

Downmix Mode is described fully within the **Shortcut Menu** section of this document

2.8.6 All

ALL Mode allows the operator to select an adjacent group of 3 audio pairs (6 audio channels) and route them to a set of external 5.1 loudspeakers.

ALL Mode is described fully within the **Monitoring Using External Speakers** section of this document

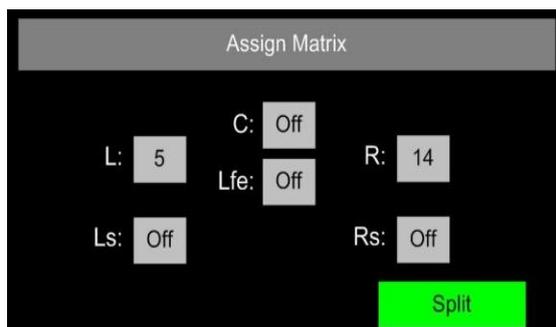
2.9 Split Mono Input Selection

The **Scroll to Hear** function of the **PAM1** features a special operational mode which enables the operator to select to listen to two non adjacent mono audio channels as a stereo pair.

Scroll the yellow **Hear Box** to the first of the desired mono channels using the usual 'push and turn' method and then release the encoder. For a subsequent period of 3 seconds the user is permitted to turn the encoder (without pressing) to select a second non adjacent mono audio signal. Once the selection is made the resultant selection is heard through the loudspeaker or headphone outputs with the first mono channel routed to the left output and the second to the right..

Note: A **Split Mono** selection is indicated on the front left hand screen in the lower right text window. The selection may be saved as a **Hotkey** or **User Preset** for instant recall

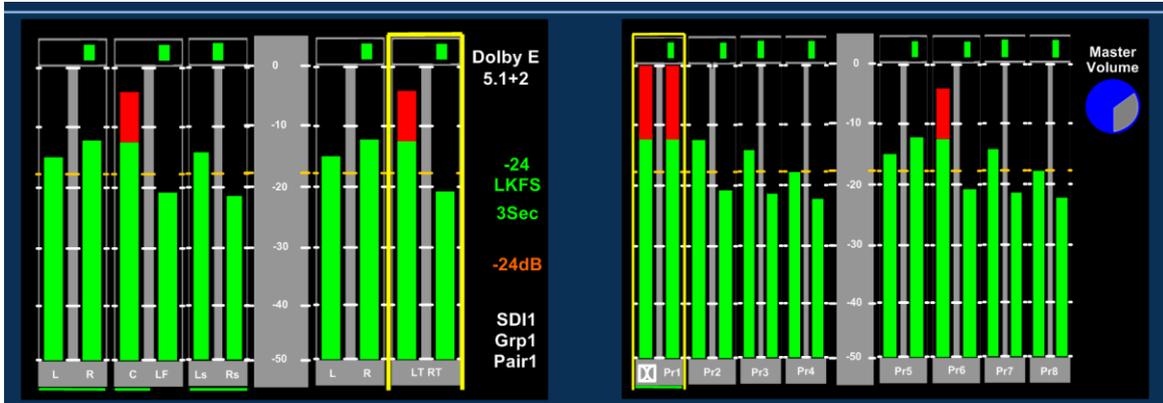
Split Mono selection uses the **Assign Matrix** section of the **PAM1** in order to create a pair of non consequential audio channels. When a **Split Mono** pair is defined, the Assign Matrix represents the resultant signal selection as illustrated below.



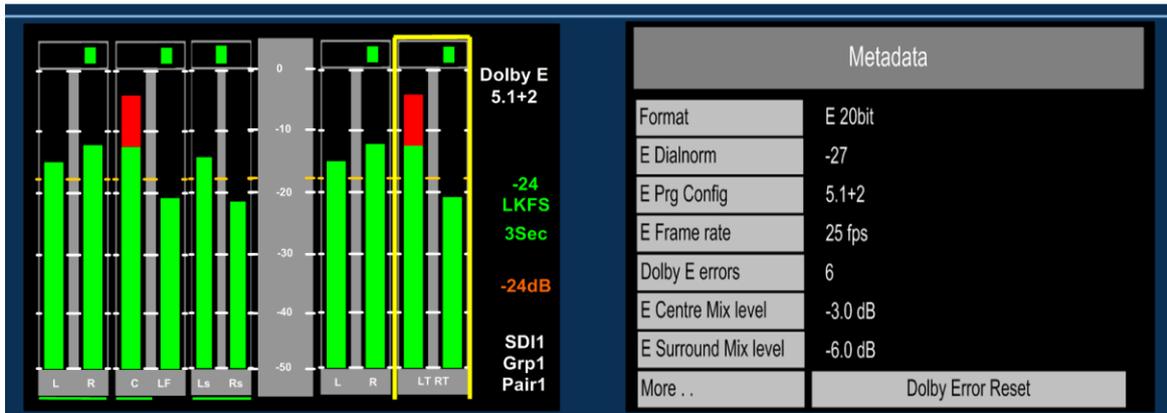
2.10 Dolby E, Dolby Digital and Dolby Digital Plus Monitoring

PAM1 MK2 with Dolby Option Fitted

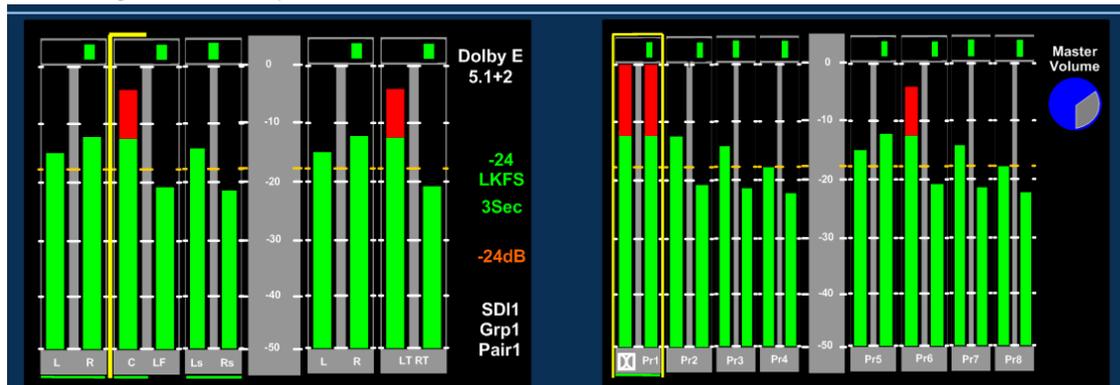
When a **Dolby E, Dolby Digital or Dolby Digital Plus** encoded audio pair is selected, **PAM1** will automatically decode the signal into its component parts and name the stems according to the Channel Coding information carried within the **Metadata**. The decoded audio component bargraph is displayed on the left **LCD** display and an Lt/Rt or Lo/Ro (user selectable) **Downmix** bargraph pair is created from the **Metadata** and shown as a 5th pair on the left screen (as shown below). In **Top Level Mode** the input bargraph display is shown on the right **LCD** display with a **Dolby** logo now superimposed below the decoded bargraph pair (as shown).



The right hand display can be altered using the **Shortcut Menu** to show **data, video, loudness or metadata** information as required when a Dolby signal is decoded.



Scroll to Hear Mode is used as normal when monitoring **Dolby E, Dolby Digital or Dolby Digital Plus** encoded signals. By pushing and turning the encoder the **'Hear Box'** can be moved across the decoded audio channels in the usual way, selecting to listen to **Dolby** components as desired. In the example below, the **Centre** channel has been 'soloed' and would be heard equally from the left and right stereo output channels.



To exit **Dolby Decoding Mode**, push and hold the encoder for several seconds and the **PAM1** will revert to the **Top Level Mode**. The encoder can be rotated to move the 'Hear Box' and select another signal from the **top level screen**.

2.10.1 Dolby E Timing and Rate Information

The **Data Screen** of the **PAM1** displays extended information about any decoded **Dolby E** and **Dolby Digital** sources. This shows the frame rate of the Dolby encoded signal and its timing relative to a video source where available.

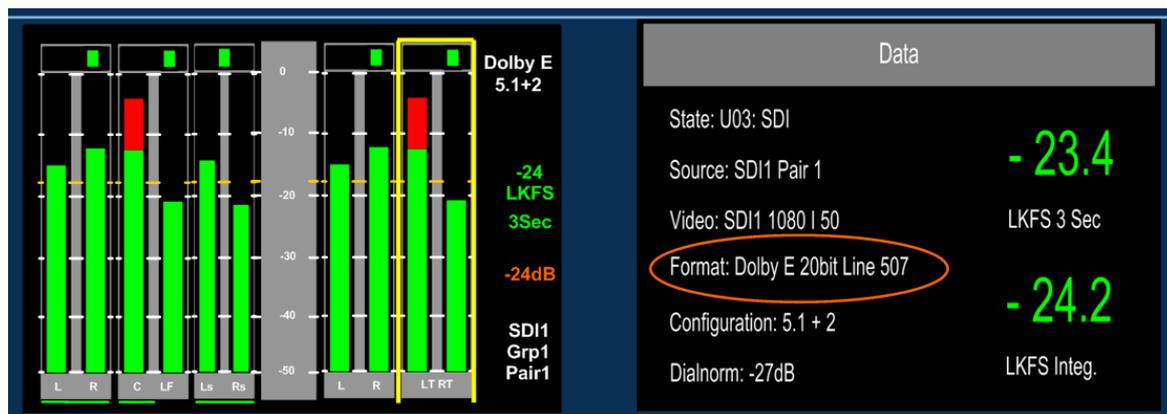
For AES inputs, the frame rate information is displayed as a minimum:

Dolby E 20bit 29.97 - or - **Dolby E 16bit 25Hz**

For embedded Dolby E sources, the start line position is displayed in the format:

Dolby E 16bit L11

The image below shows an example of a 20 bit **Dolby E** signal embedded at line 507



This functionality is also extended to mixed sources. If an AES source is selected for monitoring, whilst a video source is selected with the same frame rate, then the display will give the approximate line position of the AES **Dolby E** start relative to the video. This allows the timing of an AES signal to be checked either before embedding or after extraction (note the embedder / extractor delays need to be accounted for in a system). For sources of different frame rates, the display reverts to the **Dolby E** frame rate. The video frame rate is already displayed on the top level menu.

Dolby Digital and Digital Plus sources are also identified, although no further information is provided without a decoder.

All of the above functions are available both with and without on-board **Dolby** decoding, providing the source has a correctly formatted **SMPTE 337M** header

2.10.2 Dolby Metadata

Primarily for use with **Dolby E**, **Dolby Digital** and **Dolby Digital Plus** encoded audio signals: the **Metadata Menu** accesses a user selectable 3 pages each containing a group of 7 Dolby Metadata parameters. By highlighting and selecting any of the 7 options, the user can use the encoder to scroll through a list of all Metadata settings read by the CAT552 card for **Dolby E**, **Dolby Digital** and **Dolby Digital Plus** audio and create their own Metadata monitoring view.

Metadata - Page 1

Metadata	
Format	E 20bit
E Dialnorm	-27
E Prg Config	5.1+2
E Frame rate	25 fps
Dolby E errors	6
E Centre Mix level	-3.0 dB
E Surround Mix level	-6.0 dB
More . .	Dolby Error Reset

Metadata – Page 2

Page 2 Metadata fields include -

- Video rate
- Dolby E Programme Description
- Dolby E Stream mode
- Dolby E Surround mode
- Dolby E LFE On/Off
- Dolby E Mix Level
- Dolby E Room type

Metadata – Page 3

Page 3 Metadata fields include -

- Format
- Dolby D Errors
- Dolby D Coding
- Dolby D Centre mix
- Dolby D Surround mix
- Dolby D Data rate
- Dolby D Dialnorm

2.10.3 Dolby E programme number selection

The **Setup Menu** allows the user to select which programme the Metadata is read from. The screen below shows the **E prg#** selection on the Setup window. By selecting **E prg#** the user may scroll through **Dolby E** programme numbers 1 to 8 and select the appropriate programme number. This setting may be saved as part of a user preset.

Setup		
EBU Dig	Ext Mon	E prg#
dBFS		E vid dly
Zero		Video
Peak	UnMute	Audio dly
Hold		Setup 2
USB Save		USB Load

2.11 Monitoring using external loudspeakers

The **PAM1** features an intelligent monitoring matrix designed to optimise the use of both internal and external loudspeaker systems. Connection of either Stereo or 5.1 loudspeaker systems is accommodated and monitoring modes automatically configured to ensure that only the audio components selected are those that are heard using the appropriate output channels.

For example: when listening to a stereo **Downmix** through a 5.1 surround sound loudspeaker system, only the left and right speakers are active whilst the centre, LFE and surround speakers are muted. Dedicated multichannel monitoring modes **ALL** and **DRC** route surround audio components to 5.1 loudspeakers in the correct channel order without the need for complex routing configuration.

The table shown in **Section 2.12** explains the default monitoring modes based on the various menus and DIP switch selections detailed in the following descriptions.

De-embedded or decoded multichannel audio (Dolby or discrete PCM) is automatically routed to the AES and analogue outputs connectors. The rear panel located **DIP Switch** includes a selection to mute internal loudspeakers and to set AES and analogue outputs to variable level (adjusted by the front panel volume control).

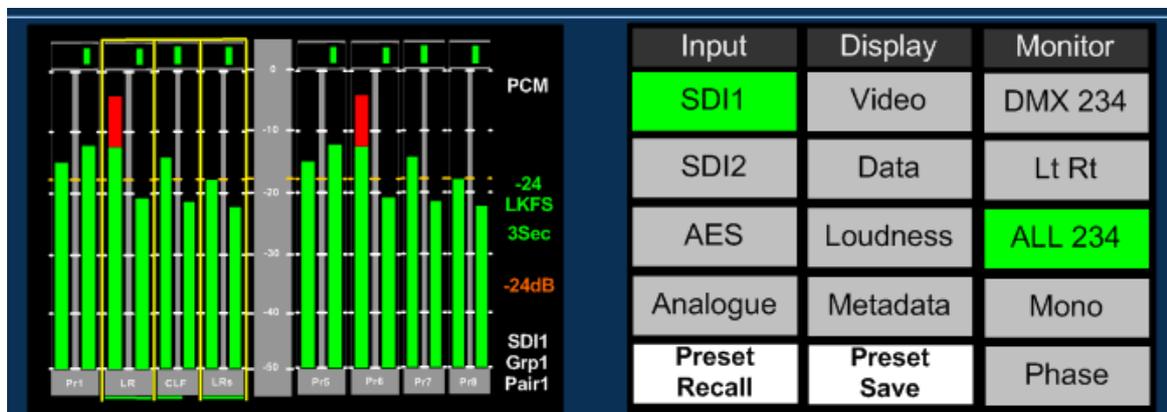
Stereo or 5.1 external loudspeakers can be connected (see the application diagram in front of this Handbook).

2.11.1 Discrete PCM 5.1 – Using ALL Mode

The **Shortcut** and **Monitoring** menus include an **ALL** option. By selecting **ALL** the operator can select any group of 6 audio channels from the bargraph display corresponding to the 5.1 audio signals. Once a 5.1 group has been defined, the 5.1 audio stems are routed to the external speaker outputs in the correct **SMPTE** order (L, R, C, LFE, Ls, and Rs).

By repeatedly selecting the **ALL** button, the group of selected audio pairs move from left to right across the bargraph display enabling the user to route discrete 5.1 or 5.0 audio from any location within the 16 channels displayed to the external variable audio connectors in order to feed external active 5.1 speaker systems. Pressing and rotating the **ALL** button also enables the user to scroll through the pair's selection in the sequence **123, 234, 345, 456, 567** and **678**.

The image below shows the **ALL** button selected from within the **Shortcut Menu** and the selection of audio pairs 2, 3 and 4 defined as an embedded 5.1 audio programme.



This function ensures that an embedded discrete **5.1** signal can be monitored regardless of the location of the 5.1 stems. The user can store different settings for monitoring discrete 5.1 audio using the **Hotkey** or **User Presets**.



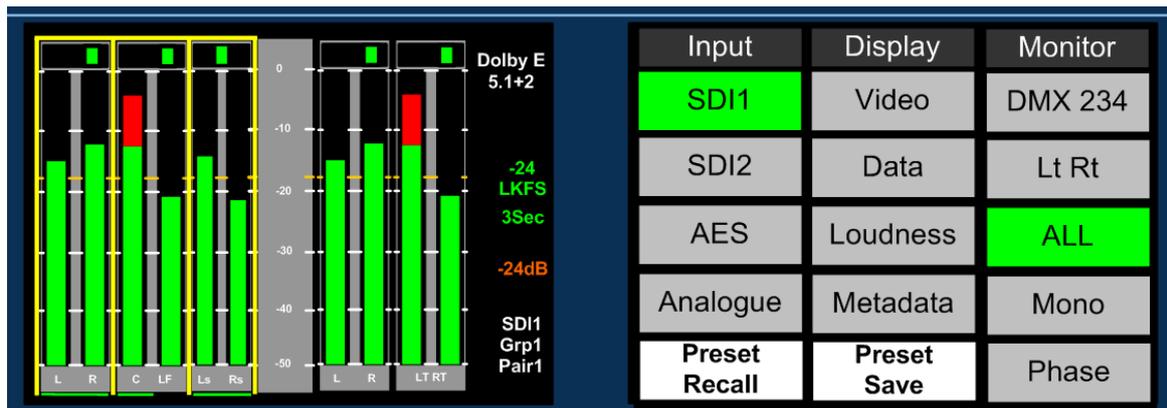
The **ALL** listening mode automatically **mutes** the internal loudspeakers and routes the selected audio to 5.1 external loudspeakers via the variable multichannel analogue and AES connectors. Selection can be saved as presets for instant recall.

2.11.2 Dolby E, Dolby Digital and Dolby Digital Plus

Using ALL and DRC Modes

The Dolby monitoring menu includes options called **ALL** and **DRC**. When used with a 5.1 loudspeaker monitoring system **ALL** routes the decoded 5.1 audio to the multichannel outputs. In **DRC** mode, the Dolby decoded 5.1 outputs is affected by Dolby DRC metadata and adjusted accordingly replicating the consumer/viewer listening experience. If the monitoring mode is switched away from **ALL** or **DRC** then the **Centre, LFE, Ls** and **Rs** speaker channels are muted and only the **Left** and **Right** channels are active. The **5.1** monitoring selections enable a user to 'solo' individual surround sound signals to check the integrity of their audio.

The image below shows how a Dolby decoded signal can be routed to external 5.1 speakers using the **Shortcut Menu**



The **DRC** listening mode can be used in conjunction with either **ALL** or **Lt Rt (Lo Ro)** modes and switches **Dynamic Range Compression** into the monitoring output signal path providing **DRC** is activated with the **Dolby Metadata** component of the encoded signal. **Line** and **RF DRC** monitoring modes can be selected via the **DRC** button enabling the user to hear the effect of either form of compression on the decoded signal and hence check the integrity of the audio heard by the consumer.

DRC mode can only be activated via the **Monitoring Menu** and is only available if a Dolby signal is being decoded. The images below show the selection of **ALL** and **DRC** modes using the monitoring menu.

Monitor			
Pair 1	Pair 5	L R	L-R
Pair 2	Pair 6	C LF	Phase
Pair 3	Pair 7	Ls Rs	Mono
Pair 4	Pair 8	7 8	ALL
Mix	DnMix	Lt Rt	DRC

Monitor			
Pair 1	Pair 5	L R	L-R
Pair 2	Pair 6	C LF	Phase
Pair 3	Pair 7	Ls Rs	Mono
Pair 4	Pair 8	7 8	ALL
Mix	DnMix	Lt Rt	Line

2.12 SMPTE 2020 (S2020) Metadata Monitoring and Downmix

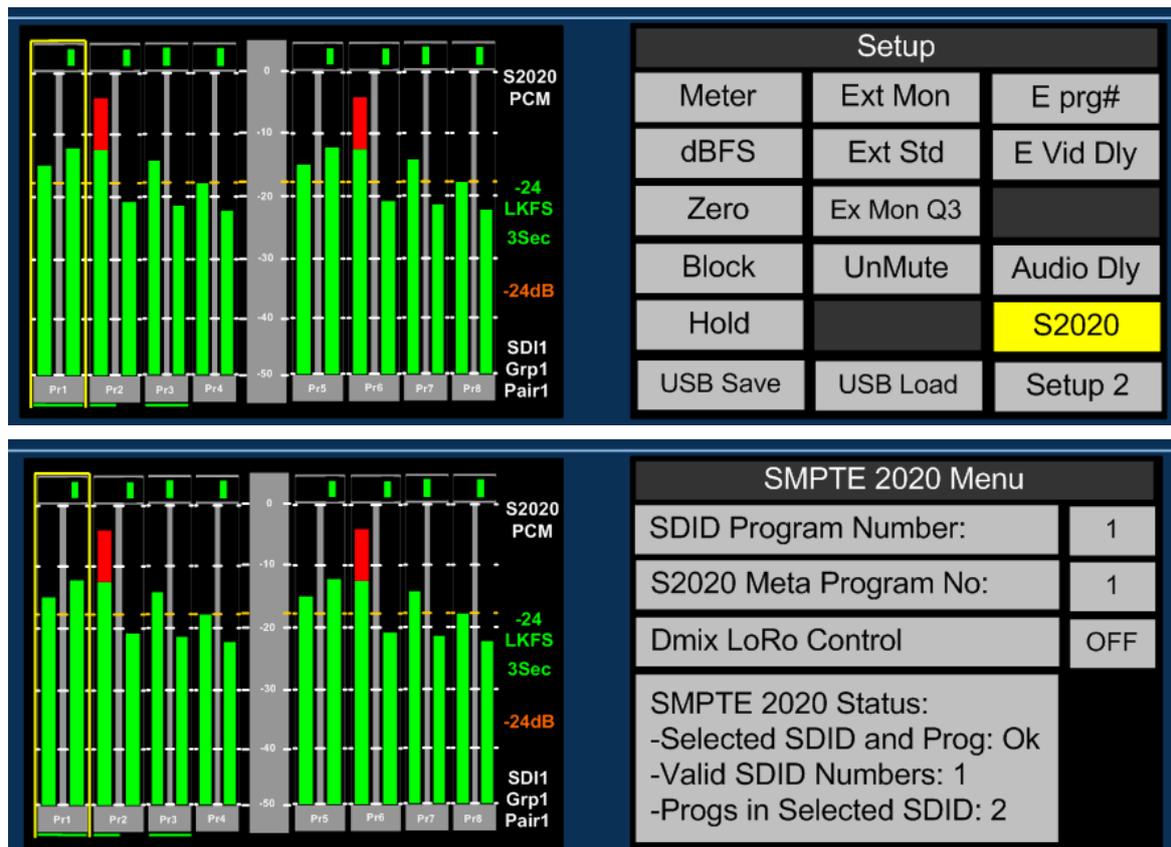
As of September 2014 the PAM 1 MK2 possesses the ability to extract ancillary SMPTE 2020 metadata from the VANC space of an SDI signal. The status of any embedded S2020 data is reported, metadata displayed and an automatic downmix capability can be activated to monitor the effect of the associated data values.

S2020 is a SMPTE defined standard method used to transport Dolby Digital metadata within an SDI based broadcast infrastructure removing the need for Dolby E encoding in favour of discrete PCM audio.

This manual assumes that the reader is familiar with the structure and terminology of S2020 data, if further information is required then please contact TSL Products Support Team or download a copy of the S2020 specification from www.smpte.org/standards

2.12.1 S2020 Menu

The S2020 menu is accessed via Main Menu/Setup/S2020



SDID Program Number: The SMPTE 2020 status indicator tells the user if any valid SDID are present and also the number (or numbers) from 1 to 9 where 1 is the default and 2 thru 9 represents audio pairs 1 to 8 respectively.

The user must set the SDID program number field to the appropriate value in order to extract the data which will be displayed within the metadata menus – described elsewhere in this handbook

S2020 Metadata Program Number: The SMPTE status indicator reports the number of programs within the selected SDID. In the configuration shown above there are 2 audio programs which might typically be 5.1 + 2 and as with Dolby E, there will be 2 sets of metadata for both the 5.1 and stereo program elements.

The user selects the program from which he/she wishes the metadata to be displayed.

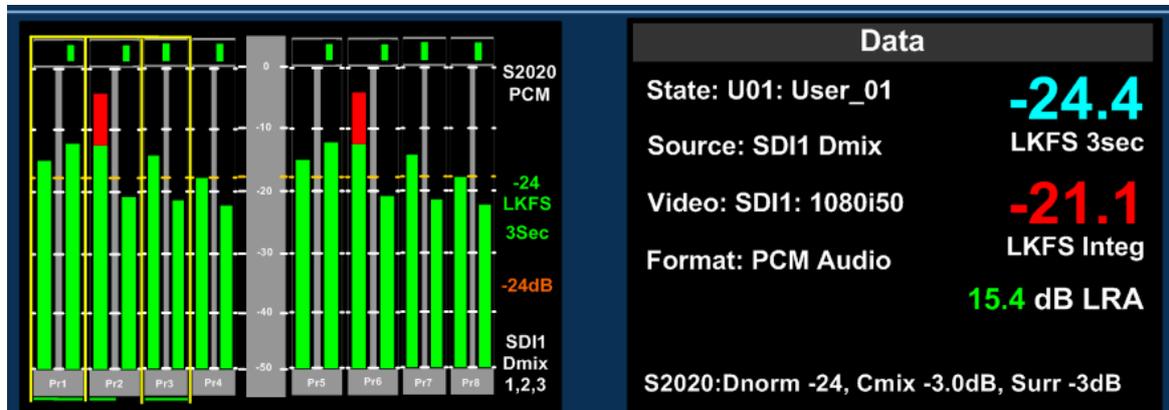
2.12.2 S2020 Downmix Lo Ro Control

PAM1 MK2 can use the LoRo downmix parameters held within the S2020 metadata to auto-downmix a 5.1 signal to stereo for compliance checking.

In the example below the operator has switched on the Dmix LoRo control for an S2020 program SDID number 2 (equating to discrete 5.1 audio channel pairs 1, 2 and 3 in an SDI signal).

SMPTE 2020 Menu	
SDID Program Number:	2
S2020 Meta Program No:	1
Dmix LoRo Control	ON
SMPTE 2020 Status: -Selected SDID and Prog: Ok -Valid SDID Numbers: 2 -Progs in Selected SDID: 1	

The operator must then activate the PAM downmix feature via the shortcut menu as described elsewhere. With the data page active on PAM1 MK2 the operator will see the following images across the two LED displays.



In this example the downmix parameters associated with the S2020 metadata are displayed at the bottom of the data screen alongside the dialnorm value. PAM1 MK2 is using these values to create a stereo LoRo downmix in order for the operator to check the compatibility of the stereo image that will be heard by the viewer at home.

2.12.3 **S2020 Metadata**

There are five pages of Metadata information available to the PAM1 MK2 user accessed via Main Menu/Metadata or through the Shortcut button. S2020 Metadata is displayed on pages 4 and 5, accessed by repeatedly pressing the **'More..'** button at the bottom of the screen.

The two pages of S2020 Metadata report the following information. Unlike the Dolby Metadata pages, the parameters are not user selectable.

SMPTE 2020 Metadata 1		SMPTE 2020 Metadata 2	
Selected S2020	SDID3, PGM1, OK	SDI Vid. Rate	SDI1: 1080i50
Dialnorm	-24	PGM Descr.	Pgrm 1
Pgm Config.	5.1 + 2	Bitstream Mode	Complete Main
Frame Rate	25Hz	Surround Mode	Not Indicated
Error Count	0	LFE On/Off	LFE off
Centre Mix Lvl	-3.0 dB	Mix Level	80 dB SPL
Surr Mix Lvl	-3.0 dB	Room Type	Large Rm, XCurve
More ..	S2020 Error Reset	More ..	S2020 Error Reset

2.13 Internal and External Monitoring Modes

The table below indicates the various monitoring modes available to PAM1 users and the audio signals presented to internal loudspeakers and external connectors when each mode is selected.

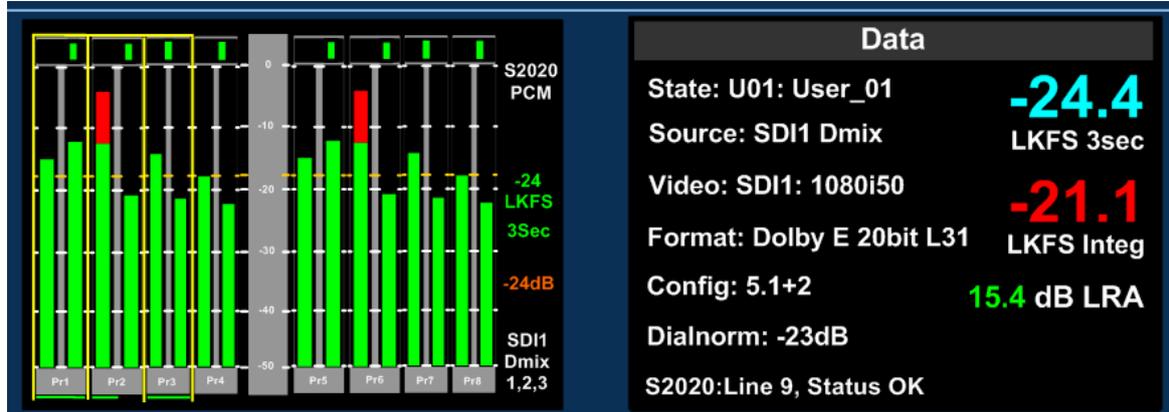
Ext Fixed/Variable D25 refers to both analogue and AES multichannel 1-8 output connectors whose modes can be altered between fixed and variable via the rear panel DIP switches.

AES outputs 9-16 are fixed level at all times and carry a direct feed of inputs 9-16 of the selected source whenever the **AES** outputs are in fixed mode. When **AES** outputs are set to variable mode, this connector carries a fixed level copy of the signal routed to the variable output connector.

Monitoring Mode	Dolby Encoded Audio				Discrete Audio			
	Internal LS	Ext Fixed - D25 (Int LS On)	Ext Fixed - D25 (Int LS Off)	External Variable	Internal LS	Ext Fixed - D25 (Int LS On)	Ext Fixed - D25 (Int LS Off)	External Variable
All	Muted	De-embedded AES and Analogue	De-embedded AES and Analogue	5.1 Audio from decoded source	Muted	De-embedded AES and Analogue	De-embedded AES and Analogue	5.1 Audio from selected channels
DRC + ALL (Line or RF)	Muted	De-embedded AES and Analogue	De-embedded AES and Analogue	5.1 Audio from decoded source	N/A	N/A	N/A	N/A
Lt Rt	Lt Rt only	De-embedded AES and Analogue	De-embedded AES and Analogue	Lt Rt only	N/A	N/A	N/A	N/A
Lo Ro	Lo Ro only	De-embedded AES and Analogue	De-embedded AES and Analogue	Lo Ro only	N/A	N/A	N/A	N/A
DM ix	Stereo Downmix of pairs 1,2,3	De-embedded AES and Analogue	De-embedded AES and Analogue	Stereo Downmix of decoded pairs 1,2,3	Stereo Downmix of selected channels	De-embedded AES and Analogue	De-embedded AES and Analogue	Stereo Downmix of selected channels
* Individual	Stereo output of selected channels	De-embedded AES and Analogue	De-embedded AES and Analogue	Stereo output of selected decoded channels	Stereo output of selected channels	De-embedded AES and Analogue	De-embedded AES and Analogue	Stereo output of selected channels
* Individual refers to single, paired or multiple audio channels selected from the monitoring menu.								

2.14 Data Screen.

The **Data Screen** view is activated from the **Shortcut Menu** and provides up to 7 fields of feedback regarding the signal being monitored and the operational status of **PAM1**. It also displays several key Loudness measurement parameters including Shortterm, Integrated and Loudness Range values when Loudness measurement is activated



State: The current User Preset name and number

Source: The selected input signal and channel

Video: The format of the selected video input signal or that of the most recently selected signal should an audio only source be routed to the bargraph display

Format: Data pertaining to the audio component of the selected signal (ie PCM or Dolby encoded). The Dolby line position is also displayed where appropriate: however, this is replaced with the Dolby signal frame rate when the encoding rate is incompatible with that of the host video signal.

Configuration: Channel configuration of a Dolby encoded signal

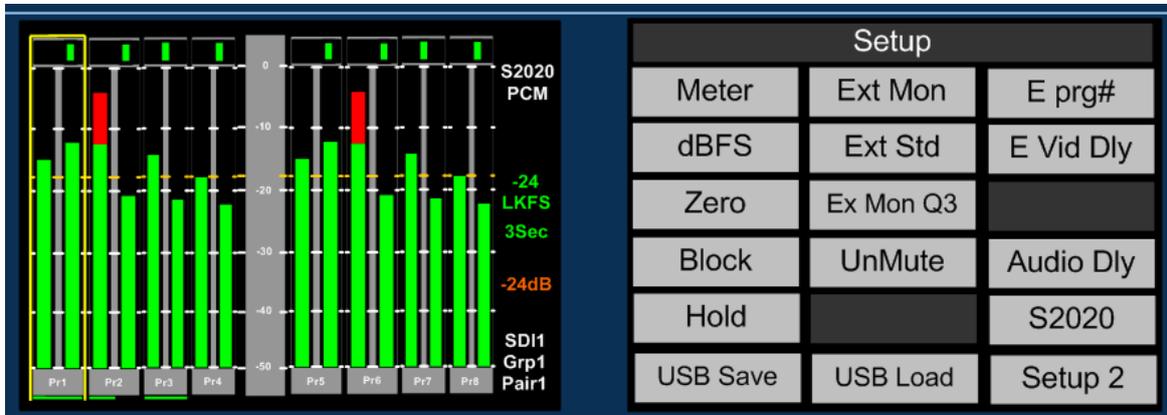
Dialnorm: Dialnorm value of a Dolby encoded signal

S2020: The line position and current status of any inserted SMPTE 2020 metadata. **This function was added to the software release in September 2014**

2.15 Setup and Setup 2 menus

Setup Menu 1 includes options for different **Scales, Reference Levels, Peak Hold, Dolby E programme, Dolby E Video compensation delay, Video Window** and **Audio delay**. By highlighting and selecting a parameter, the user can scroll through the available options and tailor **PAM1** to suit their needs.

From September 2014 onwards, the Setup menu has included a button used to access the **SMPTE 2020** menu. Not all screen images in this handbook reflect that change however all functionality described is unaffected by the minor adjustment to the Setup menu



2.15.1 Bargraph Settings and non-system default conditions

The standard operating conditions of **PAM1** are described during the Introduction to this **Handbook** in section **2.0**. Non system operating defaults for level and scale options may be set as follows:

From initial power up, go into the **SETUP** menu and choose the desired values for Meter type and alignment levels. These will be echoed on the left display as you choose them. Having chosen your defaults, move the yellow highlight box to "**SETUP2**", but do not go into the lower menu.

Then press and hold the encoder without turning it for 15 seconds (the screens will refresh after 1 second, which is the Dolby scroll exit timer finishing - ignore this)

After 15 seconds, you should get a message saying "Default Setup stored Back to proceed".

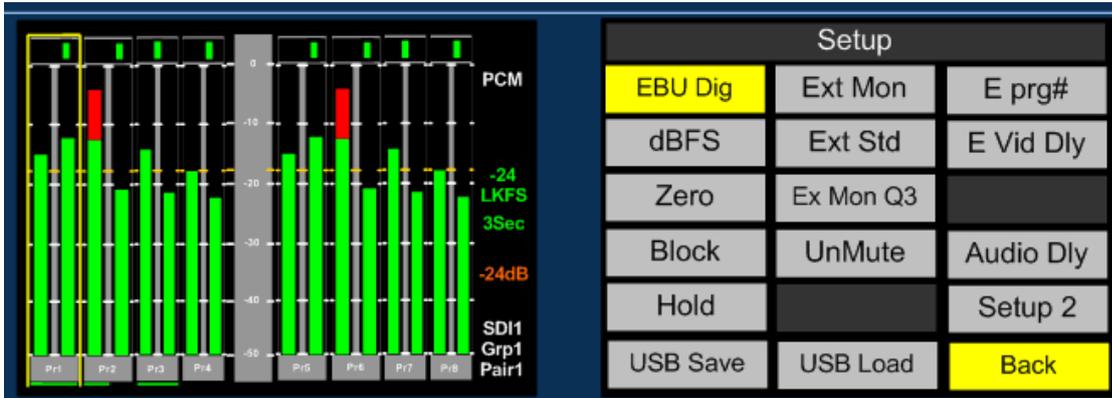
You should now find all memories, factory or user, have the chosen defaults

The factory memories can only be changed by repeating this process (or by a factory reset), but the user memories can be configured as desired (so you can have different metering types on different presets).

NOTE, however, that if you repeat this procedure to change the factory memories, then **ALL** user memories will have their metering defaults over-written.

2.15.2 Meter Scale Selection

The highlighted option at the top left of the Setup Menu is used to choose the bargraph scale type displayed by the current preset.

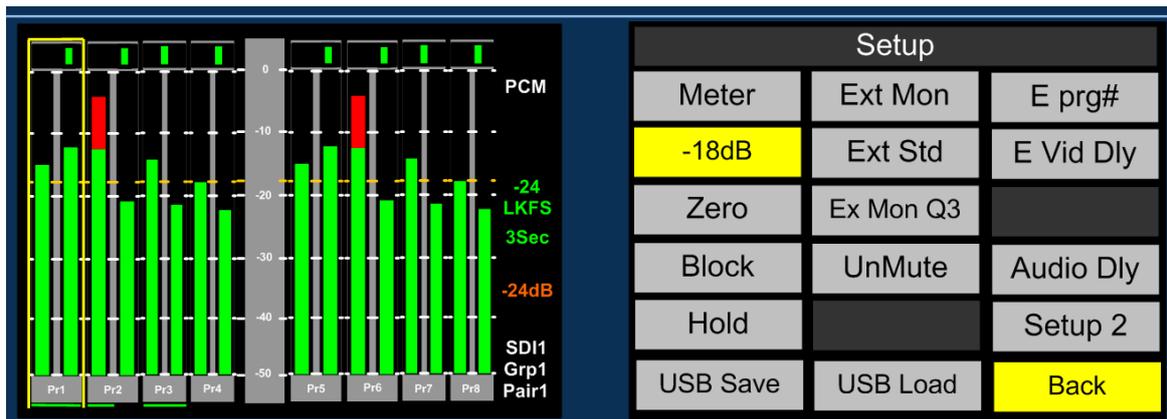


Seven International scales are supported by PAM1, they are as follows:

- EBU Digital (IEC 60268)
- BBC PPM (IEC IIa)
- EBU PPM (IEC IIb)
- Nordic (IEC I)
- DIN PPM
- Standard VU

2.15.3 Reference and Peak Level Selection

The dBFS parameter can be used to alter the 0dB reference level from between -12 and -24 dBFS. The Zero parameter selects the offset level between the Zero dBFS setting and the onset of Peak indication (the point where the bargraph changes colour to red).



2.15.4 Block Colour Mode Selection

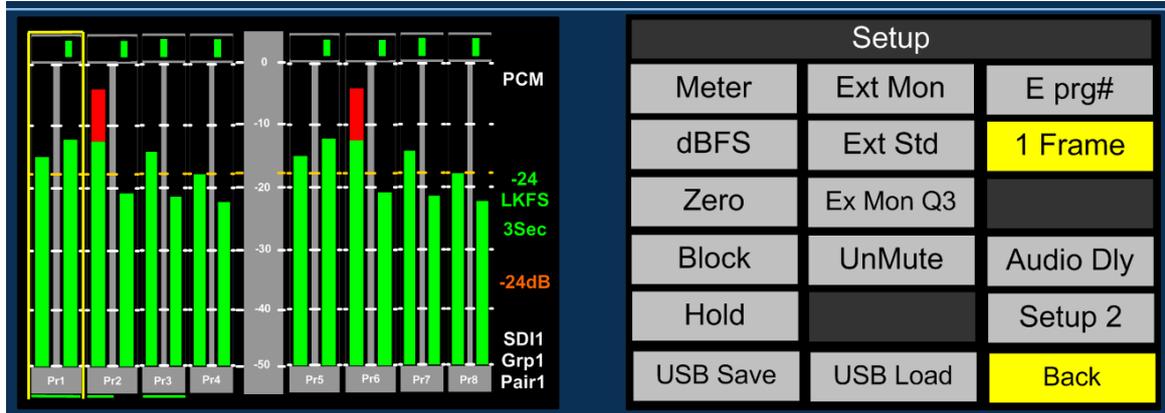
When set to Block mode, the Peak parameter enables the user to choose to illuminate the indication between zero dBFS and the peak level in yellow.

2.15.5 Hold

The Hold parameter activates a peak hold indication when switched on.

2.15.6 Dolby E Video Delay

The PAM1 features a 1 frame (40 ms) video delay setting which can be inserted into the Downconverted video display to compensate for the latency caused by decoding a Dolby E signal. If selected and activated, the delay will be switched into the video signal path automatically when a Dolby E input signal is detected and decoded. The Video Confidence window on PAM1 MK2 will be subject to a 1 frame delay when the video delay mode is active.

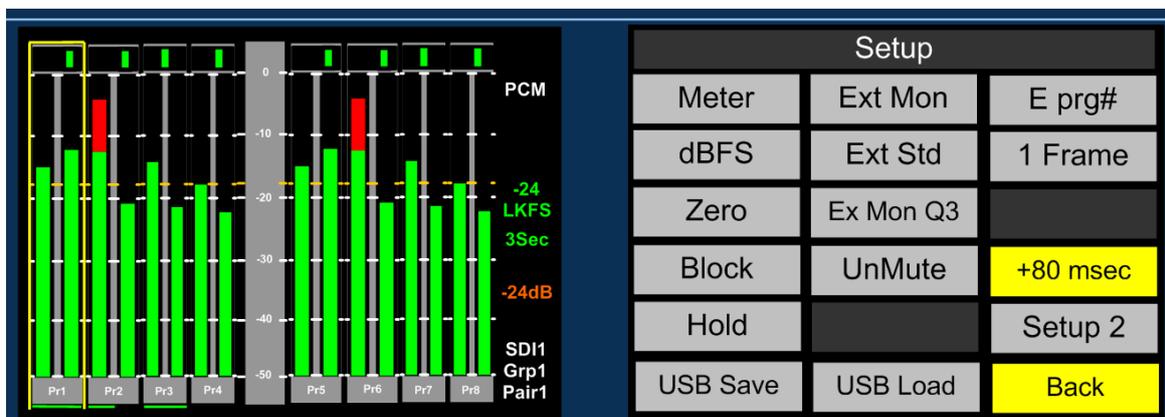


2.15.7 Audio Delay

The TSL PAM family of audio monitoring units is unique in that they feature the ability to insert up to 250ms of delay into the audio monitoring signal path (headphone, internal speaker or variable level outputs). This is designed for use in situations where the unit is being used in conjunction with a video system which introduces latency (for example: Multiviewer and LCD screen combinations) and so the audio monitoring system must be delayed to compensate.

By selecting ‘Audio dly’ from the **Setup Menu** whilst pushing and rotating the encoder, audio delay in 1ms steps up to a total value of 250ms may be selected. The selection must be stored as part of the associated **User Preset** and is not set as part of the system default. This allows the user to configure different settings for dissimilar signal types or for inputs and modes where audio delay is not required (such as analogue audio sources without associated video).

The delay feature auto-compensates for decoding latency introduced by Dolby D and E decoding. For example, if an audio monitoring delay setting has been configured for 80ms whenever the SDI1 input is selected and the embedded audio contains a Dolby E channel which is decoded, the audio delay algorithm will reduce the preset setting by 40ms so that the overall delay remains constant: 40ms (Dolby E decode latency) + 80ms (SDI 1 delay preset) – 40ms (auto compensation) = 80ms.



2.15.8 Signal Conditional UnMute

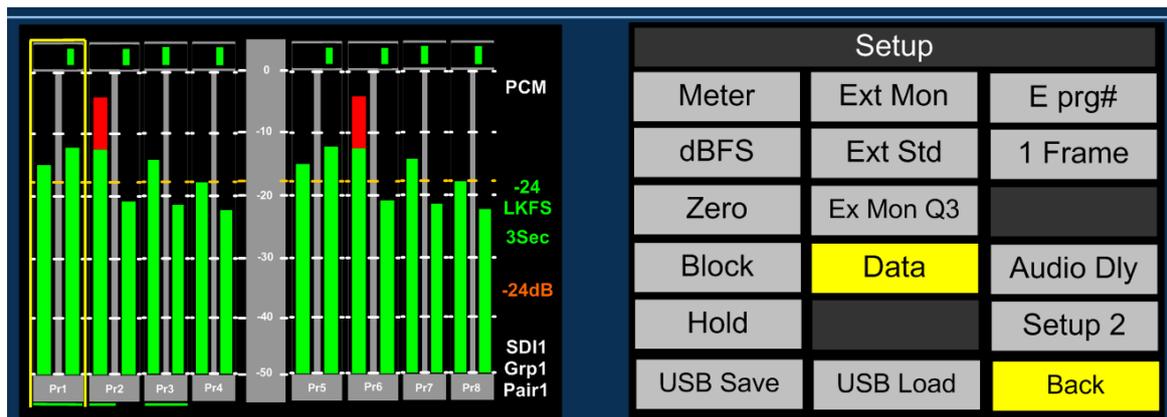
The **PAM1** is designed to protect the operator and his/her loudspeaker equipment against unexpected or hostile audio. It performs this task by monitoring the data contained within signal SMPTE headers and ancillary data which is compared to the audio signal.

Example: A PCM audio signal embedded within SDI may be accompanied by a SMPTE header declaring that it is data and not audio. Although the signal will be present on the **PAM1** bargraphs it will be reported within the data information fields as **Data** and the loudspeakers will be muted by default as a defence mechanism to protect against damage to hardware and hearing.

2.15.9 Un-Mute a PCM signal identified as 'Data'

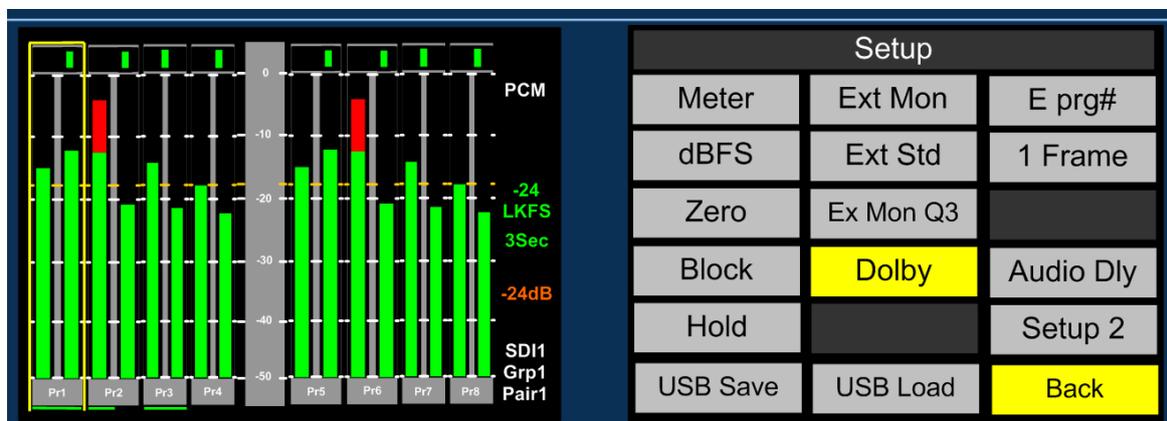
The button labelled **UnMute** provides a means to override the default speaker muting system designed to protect against hostile or unexpected audio signal conditions.

Use the encoder to select the **UnMute** button and push / turn to select **Data**. This will override the default speaker muting condition temporarily: the condition can be assigned to a user preset or **Hotkey** using the **User Preset Save** procedure described elsewhere in this handbook.



2.15.10 Un-Mute a Dolby signal identified as 'Data'

If a Dolby encoded signal is not identified as **Dolby** correctly or if there is unexpected header information then **PAM1** will mute by default. To override this condition please follow the procedure outlined previously and select **Dolby**. Save this condition as a preset or Hotkey.

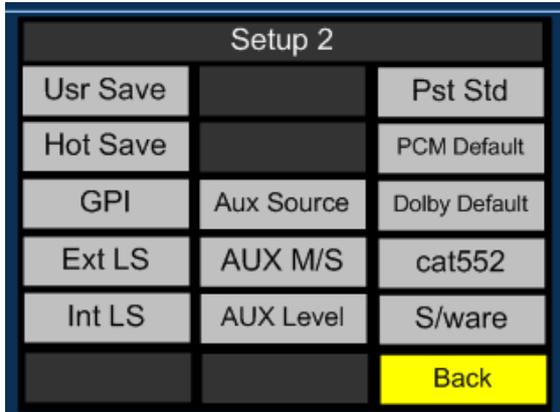


2.15.11 Override All conditional muting

Both Data and Dolby default mute conditions can be activated simultaneously by selecting **ALL** using the **UnMute** button. As previously described this selection must be saved to a preset or Hotkey for it to become a permanent state of operation.

2.15.12 Setup Menu 2

Setup Menu 2 is used to access the global system settings and software update processes (for information on software updating see **Section 2.20**). **Setup 2** is accessed from a button within the first level **Setup Menu**



2.15.13 Locking Presets (Usr Save)

Presets can be locked to prevent them from being accidentally overwritten. Select the **Usr save** button and push/turn to lock. The **Save** function will now become disabled and editing of the **User Preset** function de-activated.

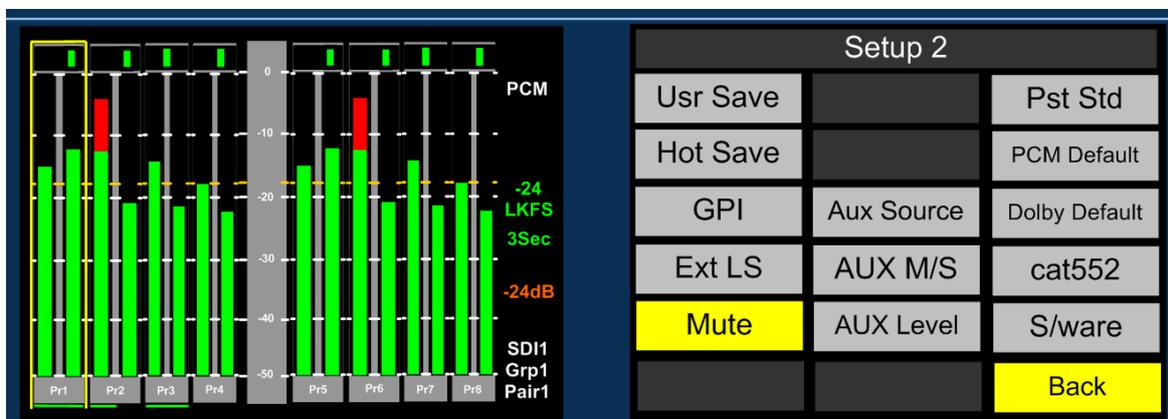
2.15.14 Locking Hotkeys (Hot Save)

Hotkeys can be locked to prevent them from being accidentally overwritten. Select the **Hot Save** button and push/turn to lock. The **Hotkey** function will now become disabled.

2.15.15 GPI

GPI functionality is described elsewhere within this handbook (see Section 2.21)

2.15.16 Internal Speaker Mute (Int LS)

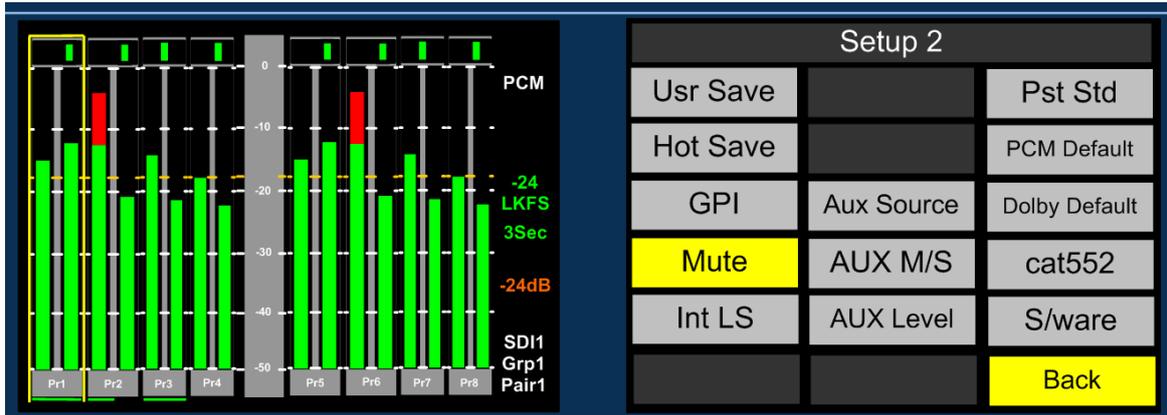


The **Internal loudspeakers** may be muted via either a rear panel DIP switch or by using the button located within **Setup 2** menu. If the DIP switch is set to 'off' then the function of the on screen button is defeated. **The action of the on screen Internal LS mute switch is preset dependant and not a system default.**

Coupled with the ability to mute the **External Speaker** outputs (see below) the **PAM1** can be used with both internal and external speaker systems switching between the two. This setup may suit applications such as **MCR** or **QC Suites** where specific programme content occasionally requires

the attention of full range or 5.1 listening or when two or more **PAM1** operators share a common external speaker system.

2.15.17 External Speaker Mute (Ext LS)



The **External loudspeaker (analogue and AES variable level outputs)** connections may be muted by using the button located within **Setup 2** menu. **The action of the on screen External LS mute switch is preset dependant and not a system default.**

2.15.18 Preset Standard Auto-Switching (Pst Std)

Preset Standard Auto-Switching functionality is described elsewhere within this handbook (see Section 2.17)

2.15.19 Aux Mixer (Aux Source, Aux M/S, Aux Level)

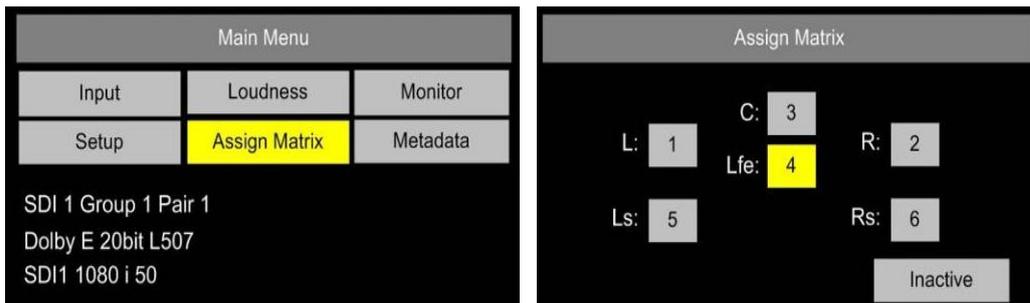
Aux Mixer functionality is described elsewhere within this handbook (see Section 2.19)

2.16 Assign Matrix

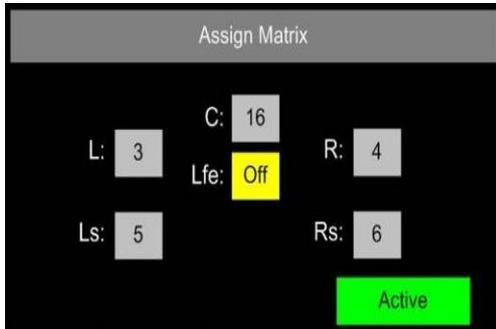
The **Assign Matrix** is intended for use in multichannel audio monitoring situations where discrete surround sound signals are being transported in non SMPTE or unconventional channel order.

The **Assign Matrix** takes the form of a **16 x 6** router where the inputs can be routed from **SDI1, SDI2, AES** or **Analogue** connections (exclusively). The outputs represent the 6 surround channels of a **5.1** programme and once the matrix is 'activated' are displayed as 6 bargraphs on the left **LCD**. Standard **PAM1** monitoring modes can be used to derive a **Downmix**, solo individual channels or route audio to external surround loudspeakers.

Access to the **Assign Matrix** is via the Main Menu.

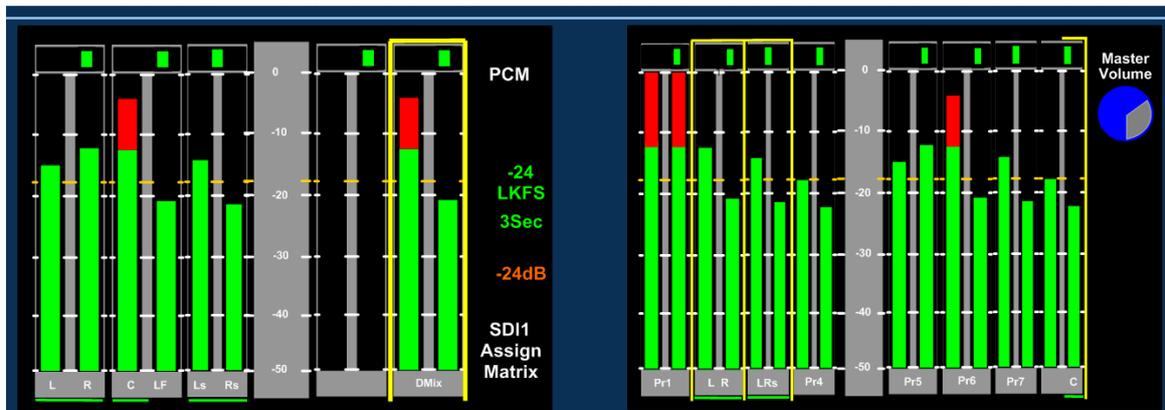


Application Example 1: a clean 4.0 surround mix of **Left, Right, Left** and **Right Surround** is embedded sequentially across channels 3 to 6 of an SDI signal while multiple language mono **Centre** channels are embedded elsewhere within the same signal. An operator it needs to be able to reconstruct each 5.0 surround sound mix for monitoring purposes – the **Assign Matrix** is designed to simplify the task and by using **User Preset** programming to enable instant recall of each monitoring setup. The images below show how this set up would be achieved using a Centre channel embedded on channel 16.



The icons within the **Assign Matrix** screen represent individual channels of a 5.1 programme – by selecting each individually via the encoder, any mono input audio signal can be routed to that output channel. Output channels can be muted if they are not required (as **LFE** in the example shown).

Once the **Assign Matrix** routing is completed and the matrix **activated** the following bargraph display is represented across the left and right LCD screens respectively.



The left hand screen shows a bargraph representation of the output channels with a default condition of a stereo **Downmix** whilst the right hand screen provides a reference to the status of the input channels as defined by the **Assign Matrix**. The sources of the surround sound components are denoted by yellow boxes and text identification below the bargraphs. The setup could then be stored directly to a **User Preset** button for instant recall.

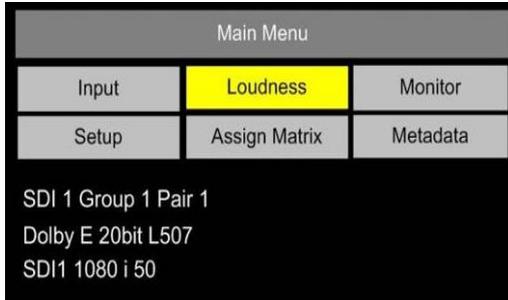
The **Assign Matrix** may be used for non-surround applications and is ideal for creating customised stereo or mono mixes from multiple mono sources (by using **Downmix**) or for applications such as **'Solo in Place'** and **LCR** monitoring.

It is important to note that the **Assign Matrix** will remain active when associated with a **User Preset** state until the **Inactive** button is pressed.

2.17 Loudness Measurement

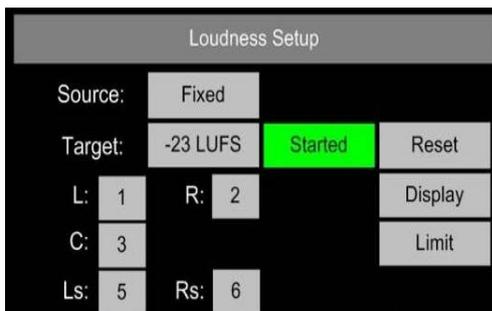
ITU-R BS1770 **Loudness Setup** menu is accessed from the **Main Menu**.

PAM1 aims to provide the user with a set of tools which comply with the recommendations of both the **ATSC (A85)** and **EBU (R128)** Loudness committees whilst giving the flexibility to tailor operation to suit local requirements.



The **Loudness Setup** menu may be used to define the parameters used to measure **Loudness** using the **ITU-R BS1770** algorithm, to select and set the specific audio channels under test and to start/stop measurement manually. The resultant **Loudness** measurement may be observed via either the **Bargraph Display** or from the **Loudness Histogram** which is accessed via the **Data** button.

The **Loudness** implementation does not require the operator to listen to the audio components being measured as the location of the **Hear Box** and **Loudness Cursor** may be independent of one another (**Fixed** and **Dolby** source modes).

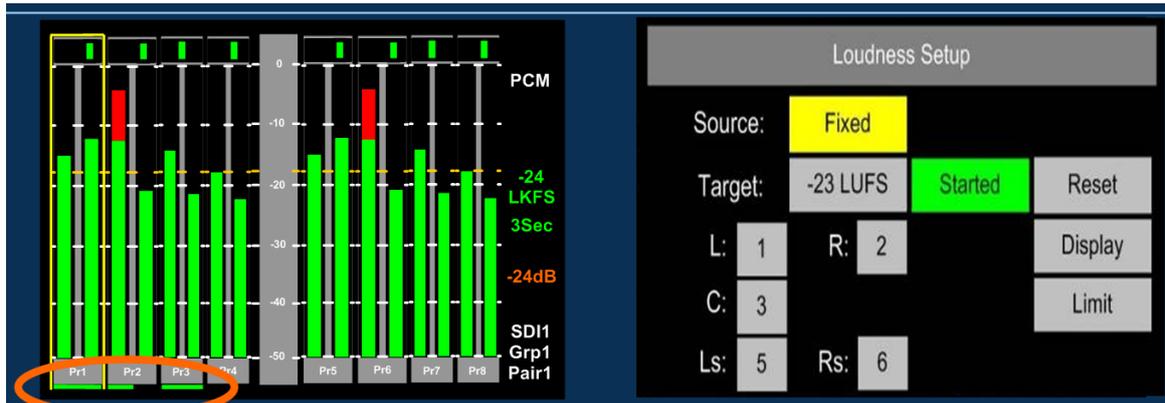


2.17.1 Source Selection.

The **PAM1** comprises 4 modes of operation for use when measuring Loudness from a User determined audio signal. In **Fixed mode** the user is able select whichever channels are subject to measurement, **Follow mode** takes a measurement of a 5.0 signal determined by the location of the yellow 'Hear' box and finally **Dolby mode** automatically selects the surround sound components of a decoded **Dolby E, Digital or Digital Plus** signal.

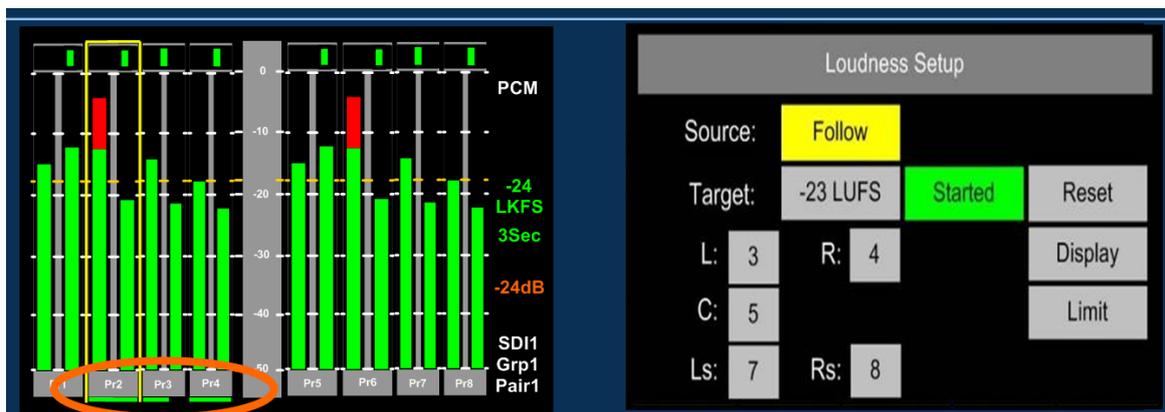
Off Mode: The **Loudness** menu **Source Selection** includes an **OFF** position for use when the **Loudness** measurement is not required. This switch position disables all onscreen representation of **Loudness** measurement including the **Target** value, **Measured** value, **Histogram** and **Menu** selection

Fixed Mode: Fixed Mode should be used when the audio signal under test should not change despite the position of the yellow 'Hear Box'. In the example shown below, channels 1, 2, 3, 5 and 6 are selected as the L, R, C, Ls and Rs components respectively.

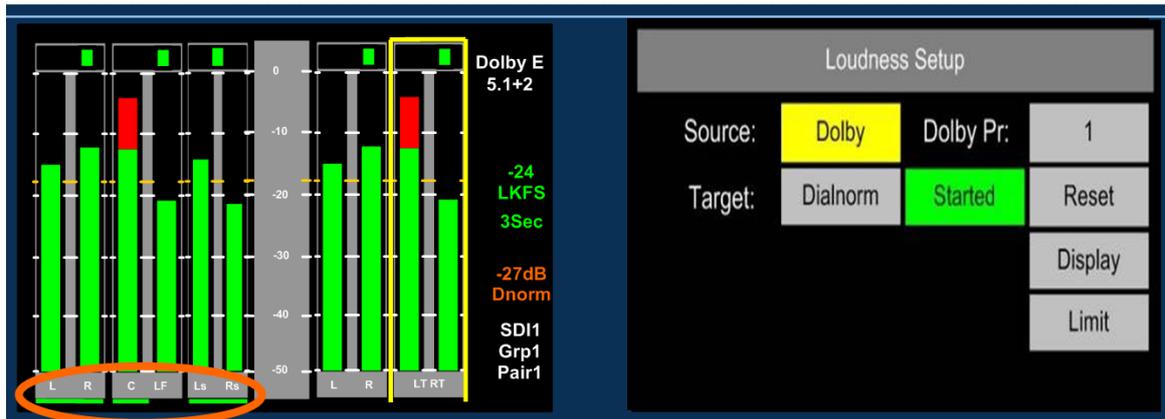


Their selection is denoted by the horizontal green cursor displayed below the 'pair' identification. Providing the user does not switch away from the selected signal input or **User Preset**, then the **PAM1** will continue to measure the Loudness of the highlighted audio components. In this mode of operation **PAM1** does not make the assumption that the audio signals are present in the correct **SMPTE** channel order for 5.1 sound – by selecting the channel components individually, non **SMPTE** order 5.1 audio can be chosen and measured. Unused channels can also be turned 'OFF' for **Stereo**, **4.0** or other custom applications. As per the **ITU-R** guidelines, the LF channel is not subject to measurement.

Follow Mode: Follow Mode should be used when the audio signal under test is determined by the position of the yellow **Hear Box**. In operation, as the user scrolls to hear audio components of the displayed audio signal, the green **Loudness** cursor will move accordingly and measure the 5 component signals to the right of the selected source. In this mode of operation **PAM1** makes the assumption that the audio signals are present in the correct **SMPTE** channel order for **5.1** audio. As per the **ITU-R** guidelines, the LF channel is not subject to measurement.

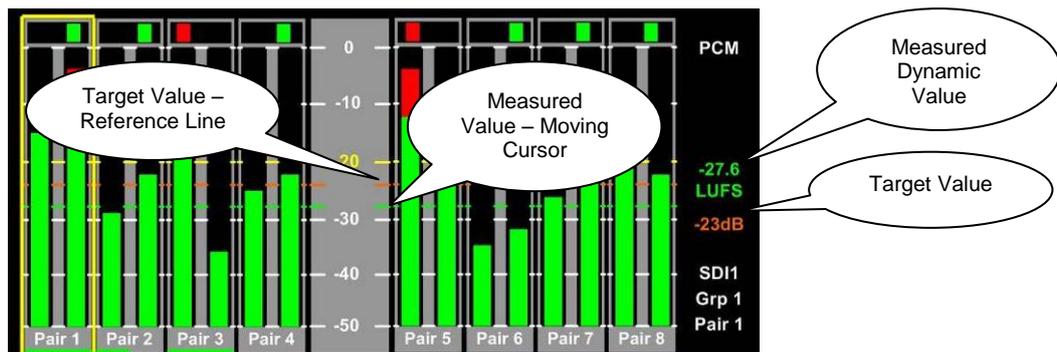


Dolby Mode: When decoding a Dolby signal, the **Loudness** measurement feature can be set to automatically select the appropriate audio components of the 5.1 programme. Since the **PAM1** identifies the channel configuration via the embedded **Dolby Metadata**, the Loudness feature uses this information to define the signal under test.



2.17.2 Target Level, Limit Threshold and Red Line Alarm

The purpose of Loudness measurement is to ensure that TV programme audio content is produced, distributed and ultimately transmitted at a level which does not inconvenience the listener and/or contravene any local or national regulations. **PAM1** helps the broadcaster to monitor the **Loudness** of programme audio content by comparing the measured level against a user defined **Target Level** (this might be figure determined by a regulator). A threshold **Limit** above or below the **Target** value determines the point at which the **Loudness** value displayed changes from green to red. The onscreen information is described below.



The **Target Level** can be set within a range of +13 to -30 or it can be defined by the **Dialnorm** of a decode **Dolby** signal. The value is displayed as an orange numerical value to the right of the left LCD screen. An orange reference cursor is also displayed across the bargraphs.

The derived **Loudness** figure is indicated as a dynamic red or green numerical value using either **LKFS** or **LUFS** units of measurement. A moving cursor follows the **LFS** value as it changes dynamically with both number and cursor changing colour from green to red as the **Target Value** plus **Limit** (if set) is exceeded.

The **Limit Value** at which a **Loudness** value 'red lines' can be set at up to +6 LU above the **Target Value**. At the point at which the '**Red Line**' limit is reached, **PAM1** triggers an output **GPI** via the D15 rear connector. This can be used for external **Alarm** notification and/or to control a third party **Loudness** control device.

2.17.3 Start, Stop and Reset Triggers

In order that **Loudness** measurement can be targeted towards specific programme content, **PAM1** is equipped with the ability to **Start** and **Stop** (pause) measurement cycles. These functions can either be triggered manually from the on screen button or from a **GPI** via the **D9** connector at the rear of the unit.

The action of restarting measurement after a 'stop' is executed recommences calculation of **Loudness** from the previous value or values. This methodology ensures that programme loudness measurement can be achieved outside of scheduled interruptions such as **commercial breaks** or **announcements**. A means to integrate this could be to use **studio** or **playout automation** to start/stop measurement via the **GPI**.

The **Reset** button discards any previously measured values enabling a new calculation to begin once the **Start** button is activated once more.

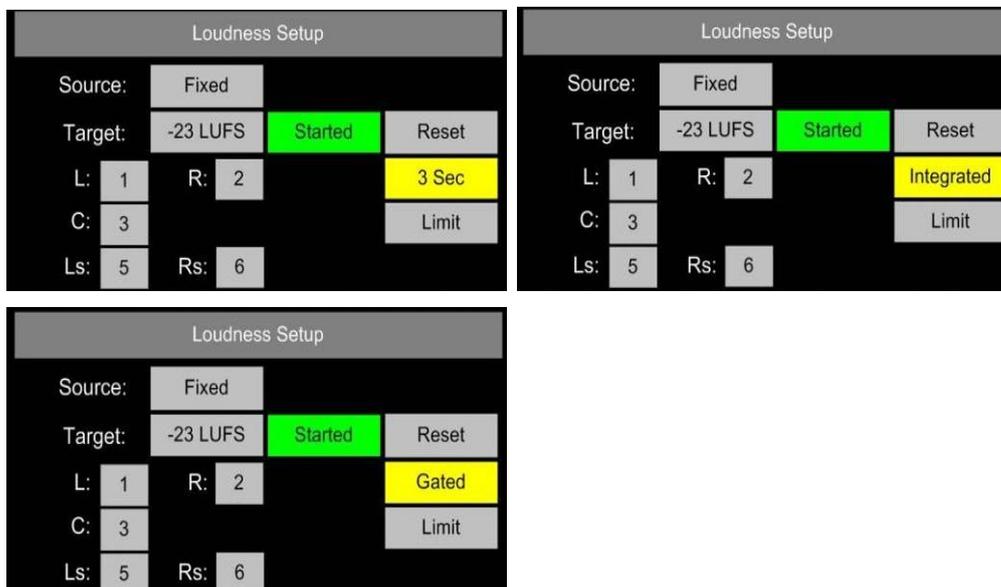


2.17.4 Histogram Display and Integration Settings

The **Display** button within the **Loudness Setup** menu provides access to three integration modes by which the resultant loudness level can be displayed as either a numerical value, moving line or via the **Histogram**.

The sample for all Loudness measurement is set internally at 400ms. The 3 modes of display are as follows:

- A **3 Second** sliding window.
- **Integrated Measurement** over the entire duration of any clip/programme. Control achieved via the Start/Stop/Reset buttons (soft or GPI)
- **Gated**: Similar to Integrated except that any sample lower than the -10dB gating level is discarded.



The **Loudness Histogram** is accessed via the **Shortcut Menu**. The **Histogram** view can display loudness against time over a **2, 7, 15, 30** and **60 minute** period. The scale shows variation of +/- 6 LU either side of the user defined **Target Level**. There are direct access buttons at the top of the display which can be used to toggle between the two time period displays, three Integration modes and the **Loudness Setup** menu (via the **Loudness History** button).



Any measured values which exceed the **Target** level are shown in red. Over **Target** values which then exceed **Limit** are then clipped and do not extend to show peak. This gives the user a comprehensive and instantaneous reference to the behaviour of audio within the tested programme material.

The diagram below shows a programme **Histogram** displaying a **Target level** measurement with a **Limit** level set at +4 LU which is being exceeded at three points

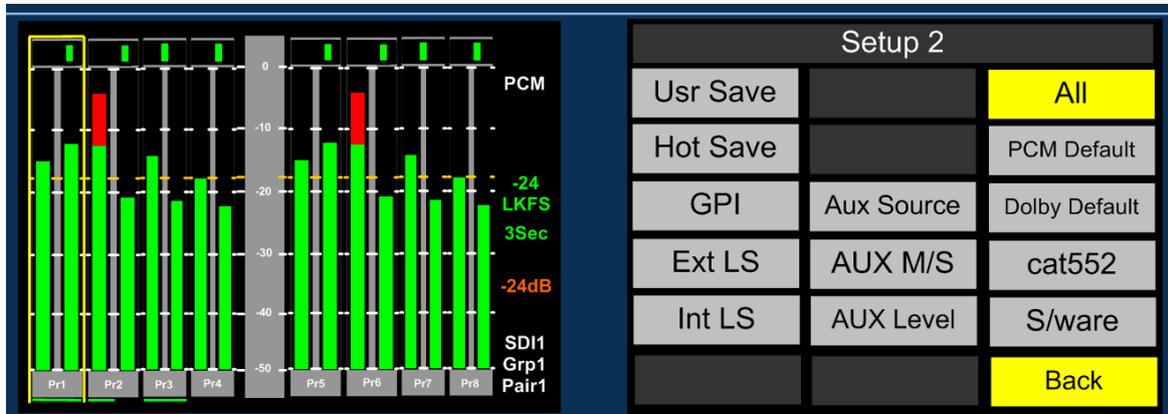


2.18 Preset Standard Rules

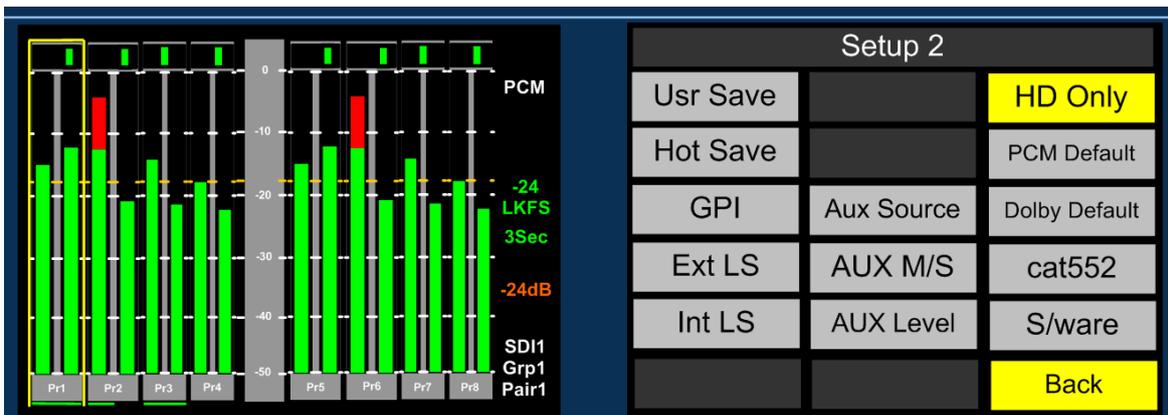
The **Preset Standard Rules** feature within the **Setup 2** menu enables **PAM1** to automatically switch between **Presets** depending on dynamic input signal format changes.

The **Preset Standard (Pst Std)** button has 3 user states.

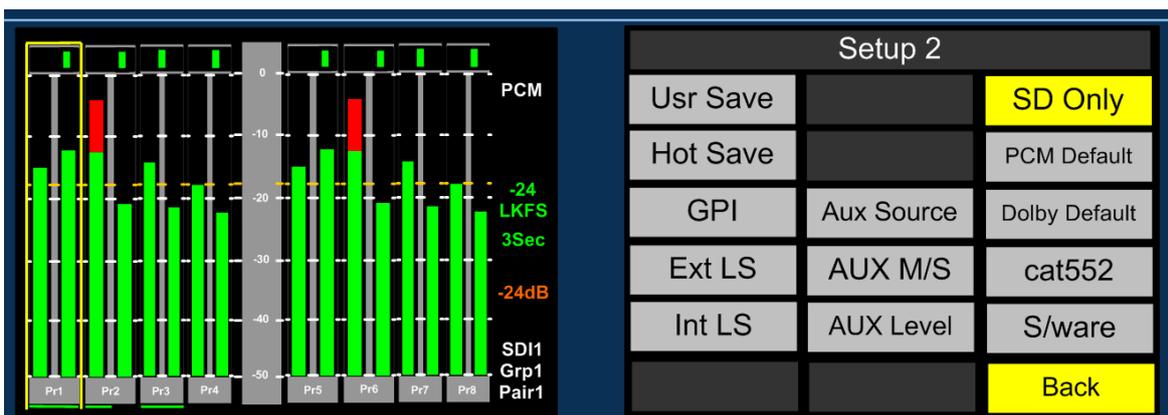
- **ALL** is the default condition and is used in normal operation. By definition, when **Preset Standard** is set to **ALL**, the **PAM1** operates as usual and is oblivious to changes to input signal types and functions as normal.



- **HD Only** should be selected as part of a **Preset** which is intended to be automatically activated whenever an **HD** signal is received.



- **SD Only** should be selected as part of a **Preset** which is intended to be automatically activated whenever an **SD** signal is received.

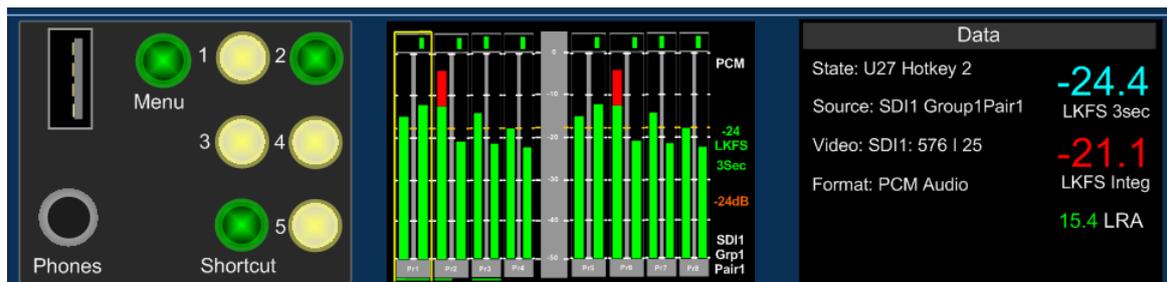
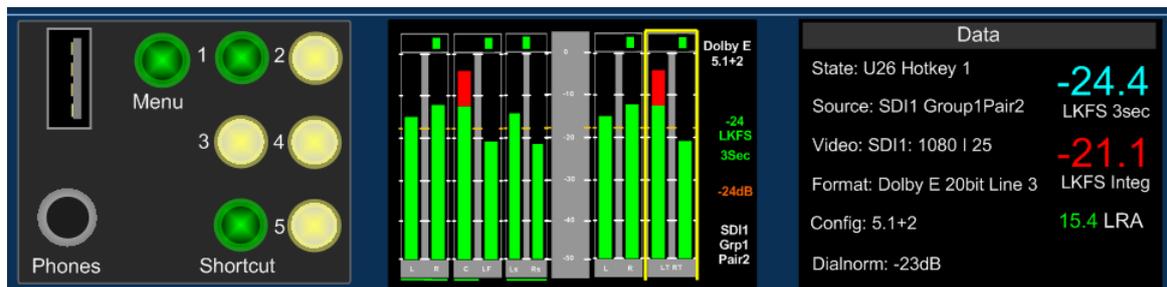


In use, when monitoring modes are being setup and saved as a **Preset** and will be subject to **Preset Standard** rules, the **Pst Std** selection must be made and saved as part of that preset or presets.

Important: For **Preset Standard** to work, the **SD** preset should be stored directly adjacent to the **HD** preset at either a pair of **User Preset** locations or two **Hotkey** buttons. The **SD** button must have first been pressed to recall said mode in order to activate automatic switching operation.

Application Example: **SDI 1** input is cabled across a TV Station transmission which switches between **SD** and **HD** material. The **SD** signal always contains an embedded stereo audio programme on Group 1 Pair 1: the **HD** signal comprises a **Dolby E** signal on pair 2 with an encoded 5.1 programme. The operator wants to hear the stereo programme when an **SD** signal is present and the **Lt Rt Downmix** decoded from the **Dolby E** signal when **HD** is present without having to remember to switch between listening modes manually. **Preset Standard Rules** can be used to configure two independent **Hotkey Presets** between which the **PAM1** will automatically switch as the input signal standard changes.

The images below show how the application example would appear in operation with the **Data** button active and **SD Only** and **HD Only** **Preset Standard Rules** applied to **Hotkey Preset** buttons 1 and 2.



2.19 PCM and Dolby Default

The PCM and Dolby Default functions provide a sympathetic conditional switching function to the **Preset Standard** feature described previously. By assigning default conditions for PCM and Dolby signal types the **PAM1** can automatically configure the audio monitoring characteristic appropriate to the operation on a preset by preset basis.

PCM Default can be set to 'OFF' or 'STEREO' where a detected incoming PCM signal will cause the **PAM1** to either remain in its current monitoring state (OFF) or default to STEREO

Dolby Default can be set to 'OFF', 'Lx Rx' or 'ALL' where a detected incoming Dolby signal will cause the **PAM1** to either remain in its current monitoring state (OFF), to automatically Downmix the signal (Lx Rx) or send the decoded 5.1 components to the external surround speaker output AES and analogue connections (ALL) providing they are set to 'Variable' mode.

Application Example: When the 'Hear Box' is set to monitor any given pair as part of a preset, the default monitoring condition for both internal and external speakers is to reproduce the audio in stereo. If the associated preset is set with a condition where the **Dolby Default** is set to ALL, the **PAM1** will automatically select the surround 5.1 audio components of an incoming Dolby encoded signal and route them to the external speaker outputs. This kind of application may be used in an

environment where PCM audio is always associated with stereo signals and Dolby encoding is used for 5.1 emission.

2.20 Auxiliary Input Mixer

The **Auxiliary Input Mixer (AIM)** is intended for use in environments where users of **PAM1** wear headphones during normal operation and need to listen to an additional audio input signal such as Intercom (Talkback) mixed into their monitoring feed.

The **Auxiliary Input Mixer** provides the ability to select exclusively from either an **Analogue** or **AES** audio input and to mix that audio signal into the **PAM1** monitoring output buss. The resultant signal can be heard through either headphone or the loudspeaker outputs. The selected auxiliary input signal can be defined as either mono or stereo in order that an associated input signal (such as a line level mono output from an **Intercom Panel** or **Matrix**) can be plugged into the left or right leg of the selected input and heard through the left, right or stereo outputs of the **PAM1**.

Auxiliary Input Mixer can be accessed via the **Setup 2** menu where 3 buttons allow the user to select the desired input, define whether it is mono or stereo and adjust an input level trim. In order that the input level can be quickly adjusted, the level trim control is duplicated on the **Input** menu screen.

Setup 2		
Usr Save		Pst Std
Hot Save		PCM Default
GPI	Aux Source	Dolby Default
Ext LS	AUX M/S	cat552
Int LS	AUX Level	S/ware
		Back

Auxiliary Input Mixer is a global feature whose parameters and settings are not saved as part of a **User Preset**.

Aux Source: By using the encoder to select the **Aux Source** button, the user can scroll through the inputs and select the required source to route to the **Auxiliary Input Mixer**. Analogue Stereo Inputs 1 to 6 and AES pairs 1 to 8 are available for exclusive selection.

Setup 2		
Usr Save		Pst Std
Hot Save		PCM Default
GPI	Anlg 1	Dolby Default
Ext LS	AUX M/S	cat552
Int LS	AUX Level	S/ware
		Back

Aux M/S: By using the encoder to select the **Aux M/S** button, the user can define whether the selected input is to be mixed within **AIM** as a mono or stereo signal. For example: a mono line level input from a local **Intercom** panel might be connected to the left channel XLR connector of analogue input 1. In order that the operator can hear Intercom audio through both left and right headphone earpieces he would select the input to be mono so that the left and silent right inputs are summed together with equal signal sent to both left and right outputs.

Setup 2		
Usr Save		Pst Std
Hot Save		PCM Default
GPI	Anlg 1	Dolby Default
Ext LS	Mono	cat552
Int LS	AUX Level	S/ware
		Back

Aux Level: The Aux Level control enables the user to trim the level of the selected Aux Input between 'Off' and +12dB of gain with respect to the signal monitored by the main **PAM1** monitoring function. The level adjustment is graduated in incremental steps as the user pushes and rotates the encoder control.

Note: As the encoder is pushed and turned, the change in trim level cannot be heard until the control is released.

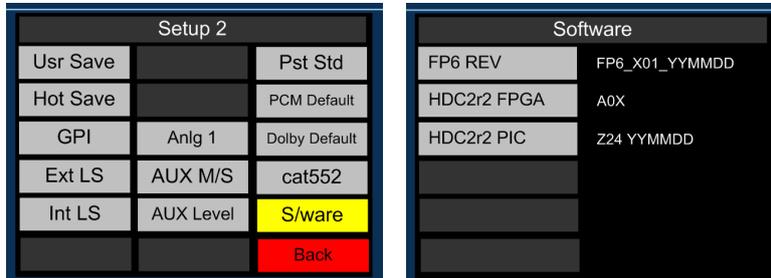
Aux Level control is duplicated within the **Input** menu screen in order to provide the user with rapid access to the trim control.

Setup 2		
Usr Save		Pst Std
Hot Save		PCM Default
GPI	Anlg 1	Dolby Default
Ext LS	AUX M/S	cat552
Int LS	+6dB	S/ware
		Back

2.21 System Upgrade, Default Restore and CAT552 Reset

PAM1 is designed to be field upgradeable. The user will need an SD memory card or USB stick, a PC and a card reader (for SD only).

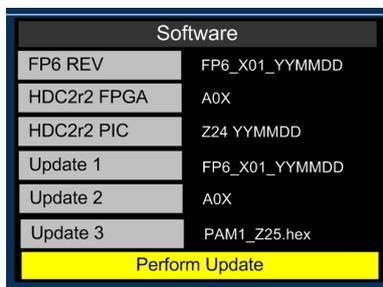
Format the memory card to FAT16 if used (or FAT as described in Windows XP). **PAM1** operating software and occasional **Dolby CAT552** updates will be sent to you by TSL in the form of a ZIP file or may be downloaded from the **Product Support** area at www.tslproducts.com. These files **must** be extracted directly to the memory device in order to construct the correct file structure for upload.



Once the upgrade file is saved, select the **S/Ware** function from the **PAM1 Setup menu 2** and view the current **FP** (front panel) and **FPGA** software versions.

Press **Back** to exit the **S/Ware** mode and insert the memory device.

Then re-enter **S/Ware** mode, a new menu option should have appeared at the bottom of the screen. Highlight and select **Perform Update**.



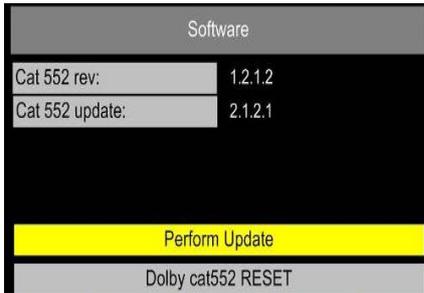
During an update, the left hand **LCD** screen displays the progress of each stage of the installation on a bargraph with a scale 0 to 100%. Front Panel (FP6), FPGA, PIC and Cat552 boards may all be updated via the SD card procedure. Depending on the file size of the update, the procedure may take up to 5 minutes to complete and may appear to have stopped – this is perfectly normal.

WARNING: Under no circumstances should power be interrupted during a software update. In the unfortunate event that this has occurred, PAM1 may have to be returned to TSL headquarters in the UK to have its operating system re-installed.

Once the update is complete, power PAM1 off and then re-apply power after 30 seconds to complete the procedure.

Note: Menu button should be pressed (Menu and Hotkey 2 to set -20 dBFs as operational default reference level) during a post update power cycle in order to clear the system memories and buffers. Please note that this will restore system default conditions and clear any customer programmed User Presets.

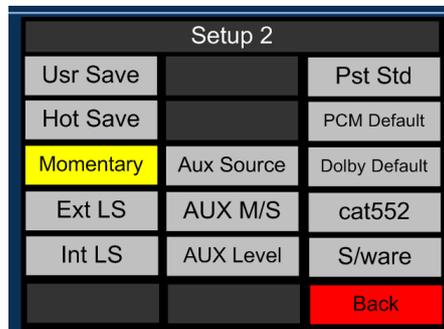
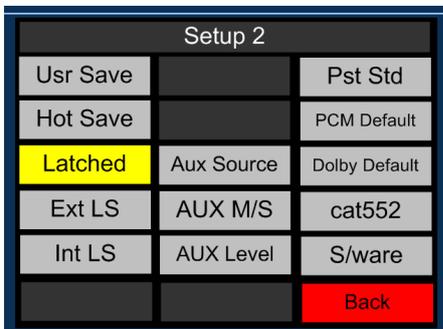
The optional **Dolby CAT552** module can be updated using a similar procedure that described for the **PAM1** operating system. From **Setup 2** select **Cat 552** to review and update the Dolby card operating software. Features such as Dolby Digital Plus decoding may be added to the **CAT552** module via the update procedure.



The **Dolby CAT552** card may need to be reset in the event that it 'locks'. Select the **CAT552** option from the **Setup 2** menu and then reset the card.

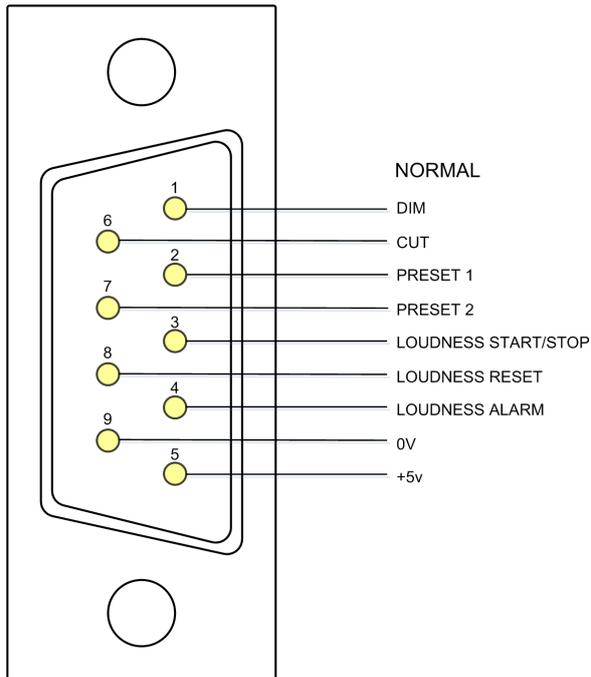
2.22 GPI Application

The rear panel GPI connector functionality can be used to **Dim/Cut** loudspeakers, to recall **User Presets** and to control **Loudness** measurement features. The Setup 2 menu page includes a button to access the GPI mode selector. The operator can choose to set the GPI connector to operate in either '**Latch**' or '**Momentary**' modes as described in the following section.



2.23 GPI Connectivity

The GPI operation is described pictorially by the pin out diagram shown below.



In **Momentary Mode**, **Dim** and **Cut** GPI's latch in a toggle manner, i.e. one closure to ground toggles the function ON: the next ground toggles it OFF. In an ON state: the GPI pin is driven low to allow an LED to be fed from the port. This LED drive is briefly pulsed high at about 100Hz to allow the port to be read whilst it is driving.

The preset recall GPI's in '**Latching**' mode are mutually exclusive.

! Note that the +5V power from this connector is intended to drive a "1-of-N" LED. If multiple LED's are to be used simultaneously, then a small external supply will be needed. An internal resistor within the PAM1 prevents the +5V rail from being shorted but limits the current available.

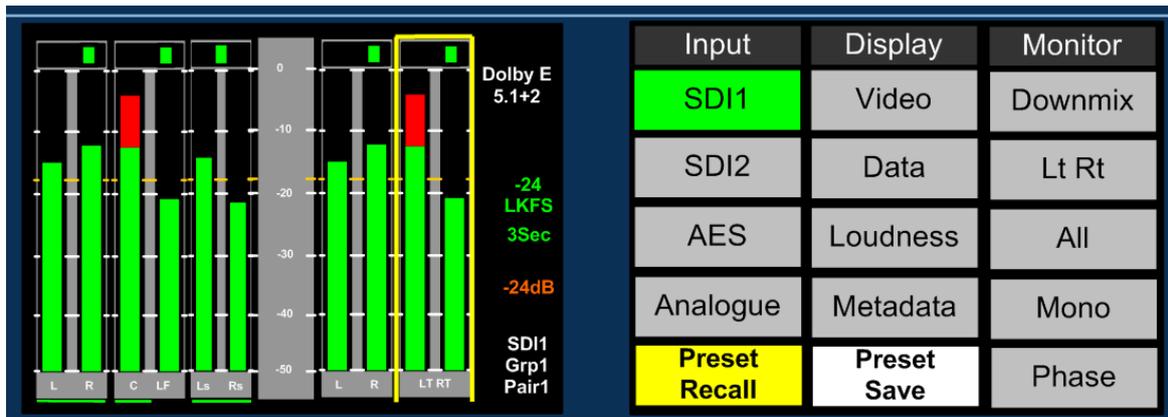
2.24 Hotkey and User Preset Memories

PAM1 features 5 Hotkey and 24 User Programmable Presets, for save and instant recall of default and favourite settings.

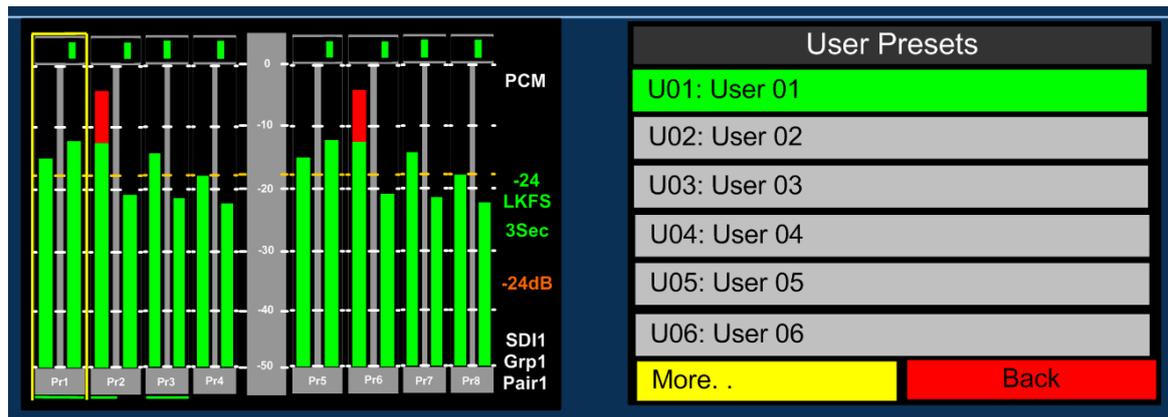
Hotkeys 1-5 are accessed directly from the front panel of PAM1 and User Presets 1-24 can be recalled and saved via the **Shortcut Menu**

2.24.1 Recalling User Presets

User Preset Save and Recall functions can only be accessed via the Shortcut Menu. Press 'Shortcut' and the recall and save buttons can be seen at the bottom of the screen as shown below.



Pressing Preset Recall brings up the 24 User Presets in four pages of 6 presets per page. Each preset is assigned the default user name in the form 'U01: User 01' to 'U24: User 24' however each preset can be allocated a unique user name.



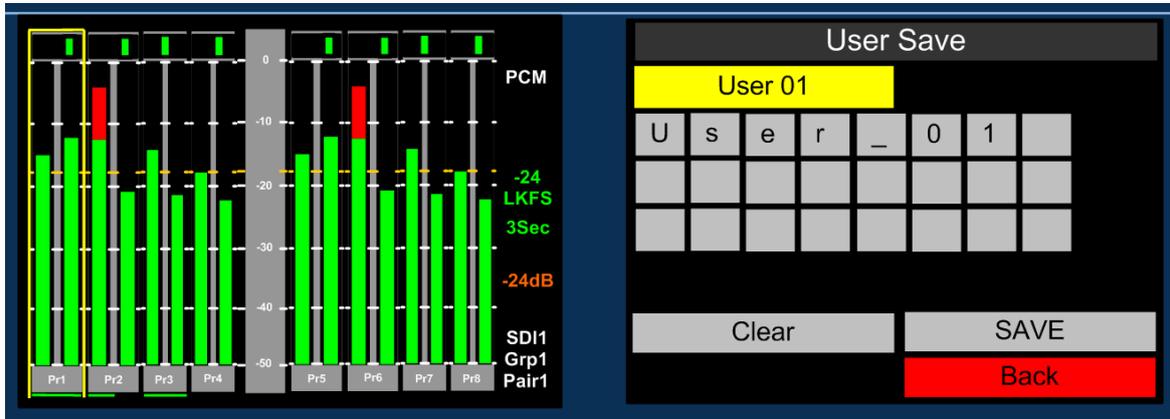
Recalling a preset instantly reconfigures the PAM1 to the settings previously stored.

Note: All User Preset defaults are set to SDI1 Group1 Pair1

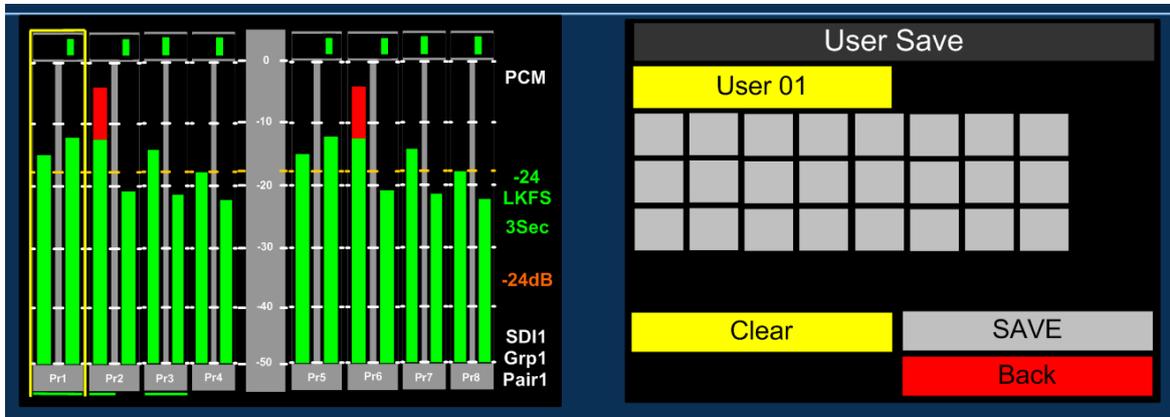
2.24.2 Saving User Presets Internally

The **PAM1** enables the operator to save favourite configurations as user presets simply and quickly. In any operational mode the user can save a user preset by accessing the **User Save** button via the **Shortcut Menu**

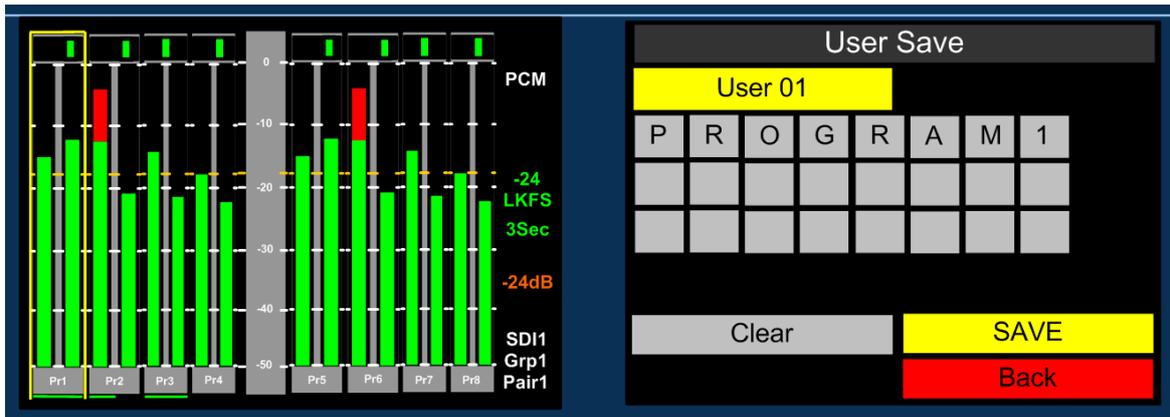
The User Save menu automatically appears on the right hand screen.



Select the **Clear** button to delete the default or existing text and use the text fields to create a preset or hotkey name to associate with the user setup.



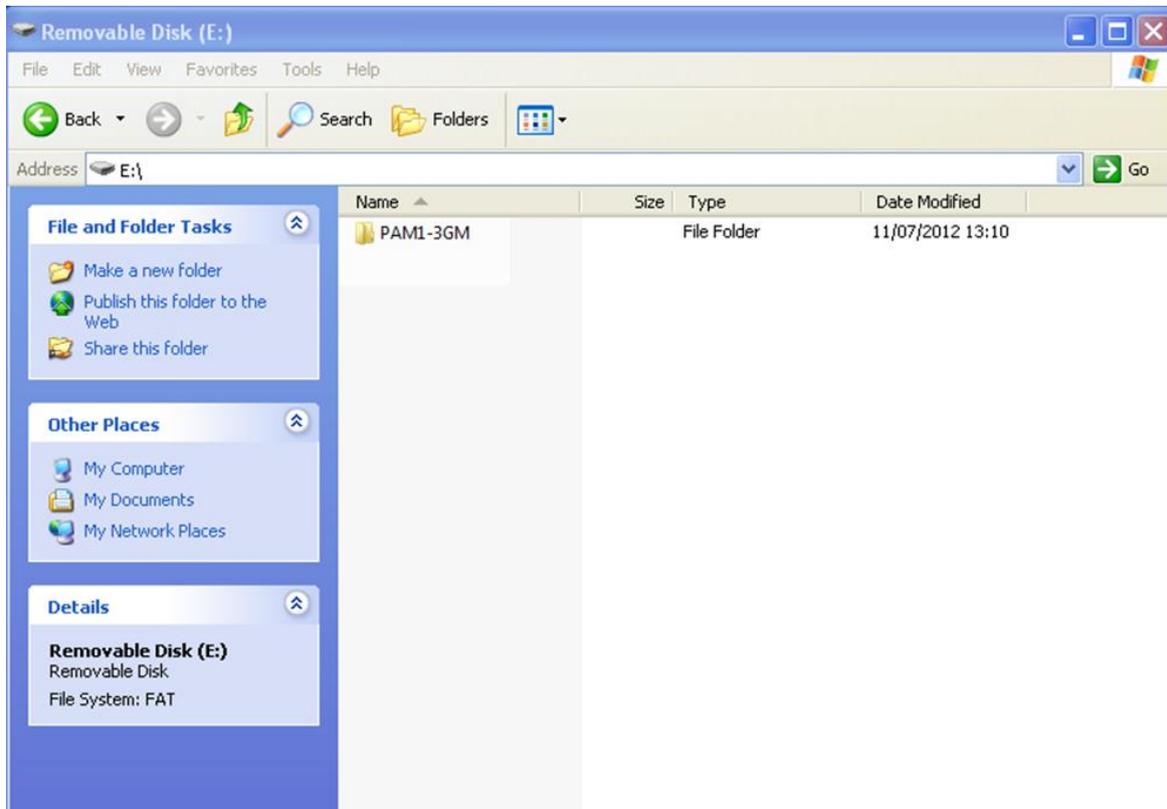
Select the text fields and push / turn the encoder to select alphanumeric or symbols to create the intended name. Once complete simply select **SAVE** and the setup will be saved to the associated memory location



2.24.3 Saving and Recalling Presets Externally

The **PAM1** configuration files can be backed up to an external memory device such as an SD card or USB drive. Both the front panel USB socket on PAM1 MK2 and rear panel SD card port (both models) can be used for backup and load procedures as well as a means to update new software releases.

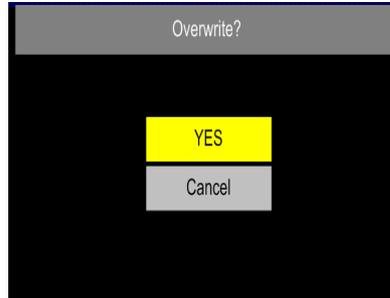
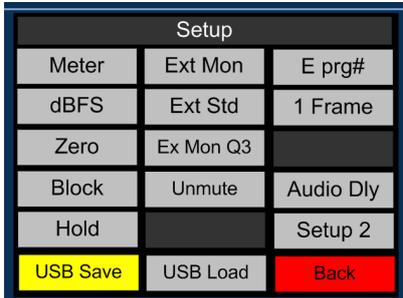
In order for a storage device to be recognised by PAM1 it must contain a folder named in a recognised format. The image below shows the default naming convention for a folder which can be seen and used by PAM1. This folder can be created manually using the 'Make a new folder' function within a Windows OS or via an equivalent command when using other programs or computer operating systems.



If a storage device is used to upgrade system software within **PAM1** the folder structure will automatically be created in the format shown above.

2.24.4 SD Card / USB Save

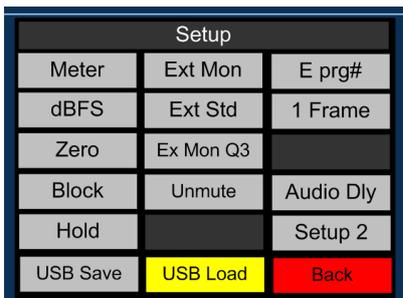
PAM1 will recognise a storage device containing a folder with the name **PAM1-3GM**. If the device is inserted into the USB or SD socket and the Setup Menu accessed then a button will be displayed labelled **SD or USB Save**. Selecting this button will cause the current status of **Hotkey** and **User Presets** 1 to 24 to be written to the device. If a preset backup already exists on the card then a request for confirmation prompt will be displayed before the save procedure commences.



2.24.5 SD Card / USB Load

Once a set of **Hotkey** and **User Presets** has been stored onto a USB storage device that device can be used to load the same presets onto other PAM1 units within the broadcast facility. Operators can also maintain copies of their own favourite settings and engineers can save and store backups of configurations to reload should a unit need to be reset during a software update or for operational reasons.

The SD / USB Load button is adjacent to the SD / USB Save key within the Setup Menu of PAM1. Selecting this function will automatically overwrite the User Presets on the device so please ensure these are backed up on another storage device if required.



3.0 Connectivity and pin-out details

PAM1 uses industry standard connectivity wherever possible. The D25 connectivity used for analogue and AES I/O adopts a pinning convention commonly used for Yamaha Commercial Audio equipment and breakout cables are readily available at low cost from companies such as www.cpc.farnell.com www.hosatech.com and many others.

For unbalanced AES I/O connectivity an optional BNC breakout cable, CAB-D25-BNC, is available from TSL or your local reseller. When used in conjunction with **PAM1**, DIP switch 2 (AES Impedance) must be switched to the 75 ohm position. For more information please refer to section 3.5 of this handbook.

3.1 Analogue XLR Connectors

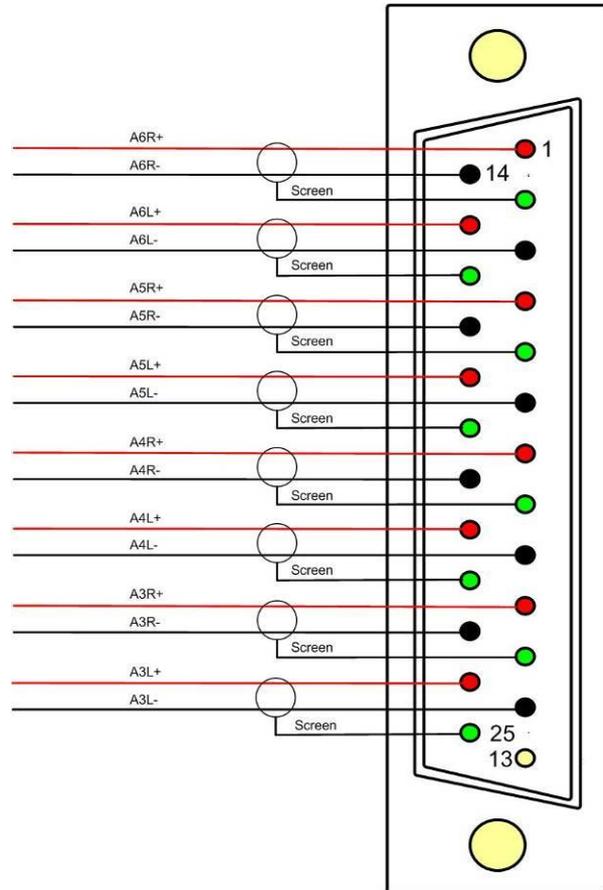
CONN	PIN	FUNCTION
ANALOG 1	1	GND
ANALOG 1	2	1 IN+
ANALOG 1	3	1 IN-
ANALOG 2	1	GND
ANALOG 2	2	2 IN+
ANALOG 2	3	2 IN-

3.2 AES XLR Connectors

CONN	PIN	FUNCTION
AES1	1	GND
AES 1	2	1 IN+
AES 1	3	1 IN-
AES 2	1	GND
AES 2	2	2 IN+
AES 2	3	2 IN-

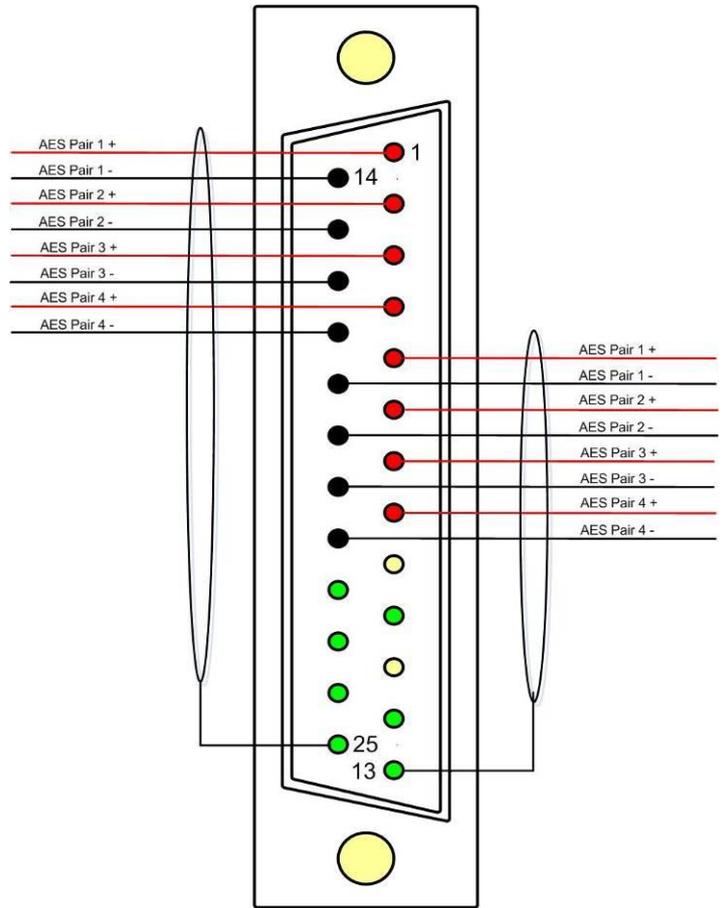
3.4 Analogue Input Connector – D25 Socket Pinout on unit, Plug on mating cable.

D 25 SOCKET ON AMU	AUDIO IN
PIN NO	FUNCTION
1	A6R+
14	A6R+
2	Ground
15	A6L-
3	A6L-
16	Ground
4	A5R+
17	A5R-
5	Ground
18	A5L+
6	A5L-
19	Ground
7	A4R+
20	A4R-
8	Ground
21	A4L+
9	A4L-
22	Ground
10	A3R+
23	A3R-
11	Ground
24	A3L+
12	A3L-
25	Ground
13	N/C



3.5 AES Input/Output Connector 1-4 – D25 Socket Pinout, Plug (shown) on mating cable.

D 25 SOCKET ON AMU	AES INPUTS/OUTPUTS
PIN NO	FUNCTION
1	Ch1&2 In 1+
14	Ch1&2 In 1-
2	Ch3&4 In 2+
15	Ch3&4 In 2-
3	Ch5&6 In 3+
16	Ch5&6 In 3-
4	Ch7&8 In 4+
17	Ch7&8 In 4-
5	Ch1&2 Out 1+
18	Ch1&2 Out 1-
6	Ch3&4 Out 2+
19	Ch3&4 Out 2-
7	Ch5&6 Out 3+
20	Ch5&6 Out 3-
8	Ch7&8 Out 4+
21	Ch7&8 Out 4-
9	N/C
22	Ground
10	Ground
23	Ground
11	N/C
24	Ground
12	Ground
25	Ground
13	Ground



AES connectors may be wired using unbalanced terminations for SPDIF and 75R coaxial systems.

Optional AES breakout cable CAB-D25-BNC-2 is available from TSL Product Sales (+44 1628 676221) and provides BNC Socket to D25 connectivity.

Please note that when using PAM1 with unbalanced AES audio connections that the 75/110 ohm DIP Switch must be selected prior to use.

When using the D25 for unbalanced AES, AES XLR connectors 1 and 2 may not be used for balanced AES connectivity.

3.6 Remote Control Connector/ RS 422 - D9 Socket

This is wired for RS422 slave operation.

D9	CONTROL
1	0V
6	0V
2	TX-
7	TX+
3	RX+
8	RX-
4	0V
9	0V
5	N/C

3.7 Metadata - D9 Socket

This is wired for RS422 master operation.

D9	CONTROL
1	0V
6	0V
2	RX-
7	RX+
3	TX+
8	TX-
4	0V
9	0V
5	N/C

3.8 DIP switch configuration functions

SWITCH	FUNCTION
1	Video aspect ratio (16:9 Up/4:3 Dn)
2	AES Impedance (75R Up/110R Dn)
3	Internal speaker mute - Up
4	Analogue variable – Up
5	AES variable - Up
6	XLR fixed Out (when set)
7	Not Used
8	Composite Up/SDI Dn

4.0 Notes

There are no user adjustable assemblies/components within this unit.

This unit requires rear support when rack mounted.

In order to affect status changes of the unit using the rear DIP switch, the unit will require re powering before the changes take effect.

Output analogue levels are adjustable over the following range:

0dBm = 0.775V into 600Ω i.e. 1mW power dissipation.

0dBu = 0.775V RMS = PPM 4.

Shipping condition, -18 dB ref 0FS = 0dBu output.

Typical European line up: -18 dBu

Typical American line: -20 dBu

4.1 General Notes

Please note that some American equipment has the function of the XLR pins 2 & 3 reversed.

TSL product is wired to the European standard

The screw locks on the D25 connectors use UNC 4-40 standard threads.

5.0 PAM1 - MK2 Technical Specifications

Power Supply

Supply Voltage	12V DC
Power Consumption	bc.

Physical Dimensions

Height	44mm (1RU)
Width	483mm (19")
Depth	280mm
Weight	3100g

Analogue Inputs 1 – 2 (Input 3-6 refer to section 3.4)

Connector Type	XLR Female 3 pin. Pin 1 Gnd, Pin 2 hot, Pin 3 cold.
Signal	Balanced line level audio.
Frequency Response	30Hz to 25kHz
Impedance	>20k Ω

Inputs AES 1&2

Connector Type	XLR Female 3 pin. Pin 1 Gnd, Pin 2 hot, Pin 3 cold.
Standard	AES3 (1994) at 48kHz, 44.1kHz or 32kHz
Impedance	110 ohm (balanced.)

Inputs AES 1 to 8

AES I/O, 25 way D type (See elsewhere for details)

Input, HD/SDV 1 &2

Connector Type	BNC.
Standard	4:2:2 component with embedded 48Khz audio. (SMPTE 259M, 292M and 424M)
Impedance	75ohm

Line Output.

Connector	XLR 3 pin Male
Impedance	50 Ω
Output Levels	Through level control with 0dB gain.
Fixed Line O/P	Available on D25 (If selected on front panel)

Headphone Output.

Connector	Stereo Jack socket type A
Impedance	50 Ω
Output Levels	Through level control with 0dB gain.

De embedded output

Connector	25 way D type
Impedance	110 Ohm
Output	Groups 1, 2, 3 and 4

Video Output

Connector	BNC
Impedance	75 Ohm
Output	Composite video or SDI (selectable)

Re-clocked Output

Connector	BNC
Impedance	75 Ohm
Output	Re-clocked serial output of the SELECTED input HD/SDV

AES Output

Connector	AES I/O, 25 way D type (See elsewhere for details)
Impedance	110 Ohm
Output	Selected SDI Group. decoded Dolby signal or AES I/P 1-

HD Standards Supported

1080i/50	1080p/23.98	1035i/30
1080i/59.94	1080p/24	1035i/29.94
1080i/60	1080p/25	1080sf/30
1080P/50	1080p/29.97	1080sf/29.97
1080P/59.94	1080p/30	1080sf/25
1080P/60	1080i/25	1080sf/24
720p/50	1080i/24.94	1080sf/23.98
720p/59.94	1080i/30	
720p/60		
480i/30.00 (SD - NTSC)		
576i/25.00 (SD - PAL)		

Performance

Response	70Hz to 20KHz
Electrical Distortion	Better than 0.1%
Hum and noise	Better than -80dB
SPL	>98dB at 0.6 m
Amplifier Output	40 watts total power output
Digital Sample Rate	32 to 48KHz auto select

6.0 Installed HDC-2T Audio Monitor Module Specification Overview

This specification describes the HDC-2T Audio Monitor Module.

This module has been designed to monitor a combination of analogue audio, AES3 digital audio and AES or Dolby E digital audio embedded in SMPTE 259M or SMPTE 292M video data streams, together with the video content which is output as composite and/or SDI. HD formats are passed through a simple down-conversion process to the monitoring output.

Mechanical

PCB: 4 layer, 120mm x 376mm with integral BNC and XLR connectors
 Component Height: <30mm above pcb surface, <2mm below
 <65mm above pcb surface with Dolby E fitted

Power

The module assumes the supply of regulated power will be made available via the power input connector. Poorly regulated or noisy supply rails may affect the quality of the analogue outputs. The HDC-2T will accept two feeds of +12V to +24V DC power, approximately 60W typical when using loudspeaker outputs. This allows dual redundant or external battery operation.

Inputs

HD/SDI

Connector Type: BNC
 Receiver type: AC coupled, auto equalising with clock regeneration
 Impedance: 75Ω, return loss ≥15dB to 1.5GHz
 Standards: SMPTE 259M-C with embedded 48kHz audio per SMPTE 272M-A
 SMPTE 292M with embedded 48kHz audio per SMPTE 299M
 Performance: ≥300m of high quality cable at 270Mbit (eg Belden 1694)
 ≥100m of high quality cable at 1.5Gbit

AES 3 or AES 3id

Connector type: Inputs 1 & 2, XLR 3 pin. (can be built for unbalanced BNC input)
 Inputs 1, 2, 3 & 4. 25way D-type 4 stereo pairs, pin-out as per Yamaha
 Impedance: 75Ω unbalanced or 110Ω balanced. Impedance is switch selected via DIP
 Switch 2. To obtain an unbalanced connection one line of the input needs to be grounded at an electrically convenient point.
 Input Sensitivity: < 200mV p-p per AES3.
 Standards: AES3-1992 at 96 kHz, 48kHz, 44.1kHz or 32kHz

Analogue Inputs

Connector type: 4 x XLR 3 pin, (Two stereo pairs)
 Board header Further 4 stereo pairs (8 channels)

Remote control

Connector type: Header, 10way to connect to 9pin D-type (RS422)

Outputs

Video

Connector Type: BNC
 Output 1 Equalised active loop-through
 Impedance: 75Ω
 Amplitude: 800mV p-p ±10%

Output 2 Composite SD (Down-converted when input is HD)
 Format: PAL or NTSC according to standard on SDI input
 Impedance: 75Ω
 Amplitude: 1V p-p ±5%

Output 3 - Optional *SDI version of image on composite output*
Impedance: 75Ω
Amplitude: 800mV p-p ±10%

AES

Eight AES (16 channels) may be output from analogue audio, embedded audio or from decoded Dolby E/D when the option is fitted

Connector type: 25way D-type 4 AES pairs (In and Out), pin-out as per Yamaha Standard
Impedance: 75 Ω unbalanced or 110 Ω balanced. Impedance is switch selected with on-board transformer balancing. To obtain an unbalanced connection one line of the output needs to be grounded at an electrically convenient point.
Amplitude: 1V into 75 Ω or >2V into 110 Ω

Analogue Audio

Eight analogue channels (4 stereo pairs) that may be output from AES, embedded audio or from decoded Dolby E/D when the option is fitted

Connector Type: XLR one pair fixed or variable
25 way D type 4 stereo pairs, pin-out as per Yamaha/Tascam
Format: Electronically balanced, centre ground.
D/A Conversion: 24 bit resolution.
THD+N: >80dB referred to 0dBFS

Loudspeakers

Connector Type: 0.156" board header
Format: Two active cross-over or 4 broad-band loudspeaker outputs
10 to 40W (4 Ω) per channel into depending on input power supply

GPI inputs

Connector type: Header to 9-way D-type plug

Control

Connector type: Header for current AMU-1 operator control board
Connector type: Header, serial bus for future operator control/display panels