



ROYAL TELEVISION SOCIETY  
INNOVATIONS AWARDS 2009 - WINNER



# PAM2-3G16 AUDIO MONITORING UNIT

**Handbook – Issue 4**

**TSL**  
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**SAFETY**

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**Installation.**

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

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## WARRANTY, MAINTENANCE AND REPAIR

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- 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 676200 (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.**



### EC DECLARATION OF CONFORMITY

Application of Council Directives Nos:  
EC Low Voltage Directive (73/23/EEC)(OJ L76 26.3.73)(LVD).  
Amendment: (93/68/EEC) (OJ L220 30.8.93).  
Conformity Standards Declared:  
EN 60950  
EMC Directive: 89/336/EEC, Amended 92/31/EEC.  
Conformity Standards Declared:  
EN 50081-1: 1992- EMC- Generic Emissions, Part 1.  
EN50082-1: 1997- EMC- Generic Immunity, Part 1.  
EN61000-3-2:1995- Current Harmonic Emissions.  
EN61000-3-3:1995- Voltage Fluctuations & Flicker.

Manufacturer's Name: TSL  
Manufacturer's Address: Vanwall Road  
Maidenhead SL6 4UB  
England  
United Kingdom

Type of Equipment: Audio Monitoring Unit

Model No: PAM2-3G16 Series

Date CE Mark Affixed: 26/01/09

I, the undersigned, declare that the equipment specified above conforms to the quoted Directives and Standards.

Place: Maidenhead, England

Signature: \_\_\_\_\_

Date: \_\_\_\_\_

Print: T. Orme

Position: DEVELOPMENT MANAGER

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**Contents**

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**1.0 Introduction****2.0 Operation****2.1 Control and Displays****2.2 Top Level Screen Mode**

- 2.2.1 Scroll to Hear
- 2.2.2 Video Confidence Display
- 2.2.3 Data Screen Mode
- 2.2.4 Dolby E Timing and Rate Information
- 2.2.5 Cut Dim Control
- 2.2.6 LED Brightness Settings

**2.3 Main Menu Mode****2.4 Input Selection**

- 2.4.1 SDI 1 Input
- 2.4.2 SDI 2 Input
- 2.4.3 AES Input
- 2.4.4 Optical Input
- 2.4.5 Analogue Input

**2.5 SDI Input and Video Confidence Display****2.6 Monitor Menu**

- 2.6.1 Phase Swap
- 2.6.2 Left/Right/Stereo
- 2.6.3 Mono
- 2.6.4 Mix Mode
- 2.6.5 Downmix Mode
- 2.6.6 Split Mono Mode

**2.7 Front Switch Monitor Select Mode****2.8 Dolby E or Dolby Digital Monitoring (Dolby version only)**

- 2.8.1 Dolby Decoding
- 2.8.2 Scroll to Hear Mode – Dolby Encoded Audio

**2.9 Monitoring Using External Speakers**

- 2.9.1 Discrete PCM 5.1 Audio
- 2.9.2 Dolby E, Dolby Digital and Dolby Digital Plus (Dolby version only)

**2.10 Internal and External Monitoring Modes****2.11 Data and Metadata Menus**

- 2.11.1 Dolby Metadata (Dolby version only)
- 2.11.2 Data Screen

## **2.12 Setup Menu 1 and 2**

- 2.12.1 Bargraph Settings and non-system default conditions
- 2.12.2 Dolby E Video Delay (**Dolby version only**)
- 2.12.3 Audio Delay
- 2.12.4 Setup Menu 2
- 2.12.5 Setup Menu 2 – Locking Presets
- 2.12.6 Setup Menu 2 – LED/OLED Brightness
- 2.12.7 Setup Menu 2 – GPI
- 2.12.8 Setup Menu 2 – Internal Speaker Mute
- 2.12.9 Setup Menu 2 – External Speaker Mute
- 2.12.10 Setup Menu 2 – Energy Saving Screen Saver Mode
- 2.12.11 Setup Menu 2 – Preset Standard Auto-Switching

## **2.13 Assign Matrix**

## **2.14 Loudness Measurement**

- 2.14.1 Source Selection
- 2.14.2 Target Level, Limit Threshold and Red Line Alarm
- 2.14.3 Start, Stop and Reset Triggers
- 2.14.4 Histogram Display and Integration Settings

## **2.15 Preset Standard Rules**

## **2.16 Auxiliary Input Mixer**

## **2.17 System Upgrade and CAT552 reset**

## **2.18 GPI Application**

## **2.19 GPI Connectivity**

## **2.20 Preset Memories**

- 2.20.1 Factory Presets
- 2.20.2 User Presets

## **3.0 Pin-out Details**

### **3.1 Analogue XLR Connectors**

### **3.2 AES XLR connectors**

### **3.3 Analogue Output connector – D25 Socket Pinout**

### **3.4 Analogue Input connector – D25 Socket Pinout**

### **3.5 AES input/Output connector – D25 Socket Pinout**

### **3.6 GPI connector – D15 Plug Pinout**

### **3.7 Remote Control connector/RS422 - D9 Socket Pinout**

### **3.8 Metadata - D9 Socket Pinout**

### **3.9 DIP switch configuration functions**

## **4.0 Notes**

## **5.0 Power Supply Mounting Bracket**

## **6.0 General Notes**

## **7.0 Specifications**

## **8.0 HDC-2T Audio Monitor Module Specification**

## 1.0 Introduction

The PAM2-3G16 is a 2RU x 320mm deep Audio Monitoring Unit with two OLED displays for audio level measurement, video confidence and metadata status indication.

PAM2-3G16 has been designed in conjunction with some of the worlds leading broadcasters to produce arguably the most comprehensive, intuitive and feature packed product of it's 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 PAM2-3G16. In the event of either scenario, please feel free to contact TSL via your local reseller or directly on +44 1628-676200, asking for the PAM Product Manager.

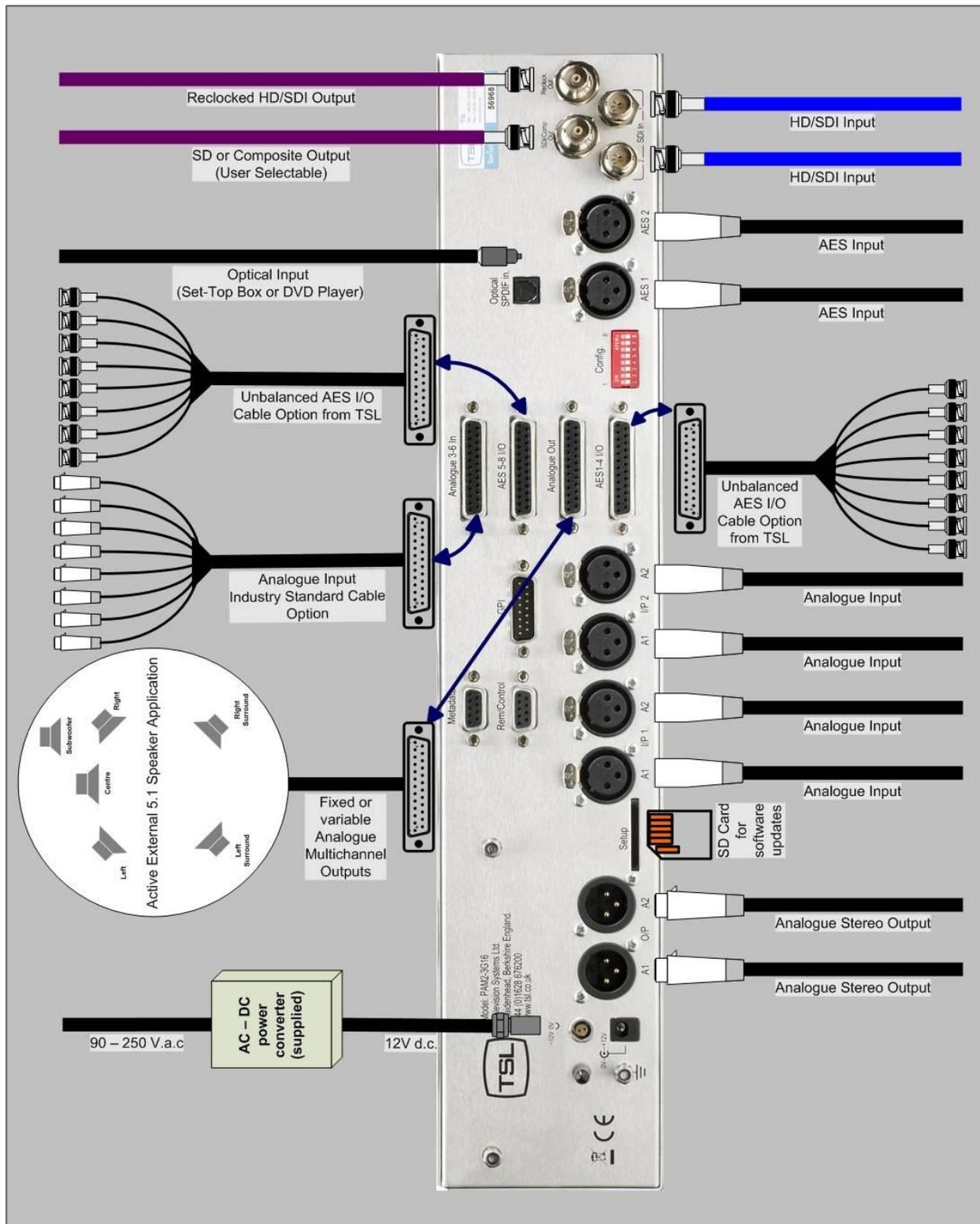
TSL will be releasing upgrades and feature enhancements from time to time – as a purchaser of PAM2-3G16 you should receive these directly, free of charge, through your reseller or directly from TSL. Please refer to [www.tsl.co.uk](http://www.tsl.co.uk) for announcements.

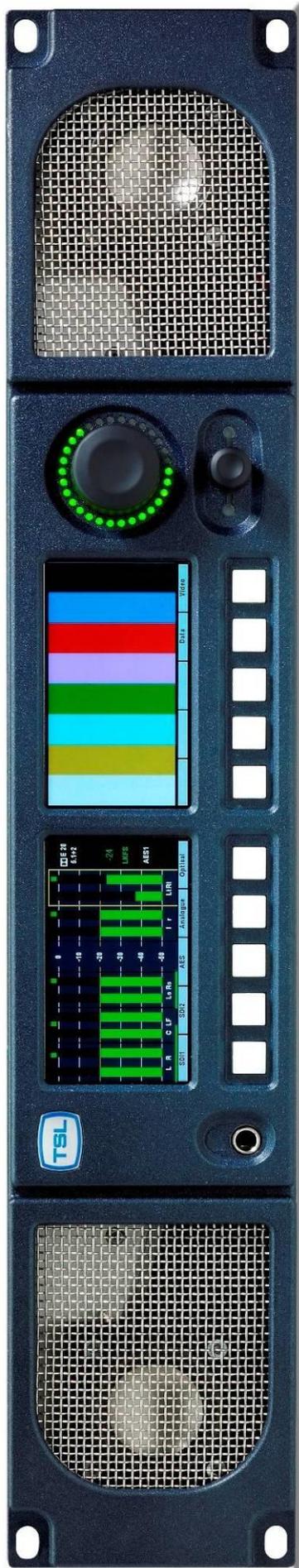
PAM2-3G16 is available in Dolby and non-Dolby versions.

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)
- 8 AES (8 Pairs/16 Channel) Inputs– 110Ohm Balanced or 75Ohm unbalanced via optional CAB-D25-BNC cable
- SPDIF Optical Input
- 6 Analogue Stereo Inputs
- 'Scroll To Hear' – top level bargraph navigation using rotary encoder
- Downmix of discreet 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
- Energy Saver Screen Mode
- 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 OLED 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. 8 by hardware buttons, GPI and 24 internal, accessible by high level menu selection
- Factory preset recall – 5 presets selecting SDI1, SDI2, AES, Analogue and Optical Inputs
- Fixed or variable analogue multichannel outputs (8 mono)
- Fixed or variable AES multichannel outputs (8 pairs/16 Channels)
- Variable stereo analogue outputs
- High quality internal full range loudspeaker system
- Variable LED brightness control
- Dual 12V DC inputs
- Serial remote control
- Headphone output with LS muting
- Compact, lightweight (5.1Kg) 2RU case, 320mm deep

PAM2-3G16 Hookup Schematic





PAM2-3G16



User Handbook Issue 4

## 2.0 Operation.

The PAM2-3G16 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.

**Important Note: PAM2-3G16 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 6 and 8. The following message will appear on the right hand OLED signalling a successful operation.**

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

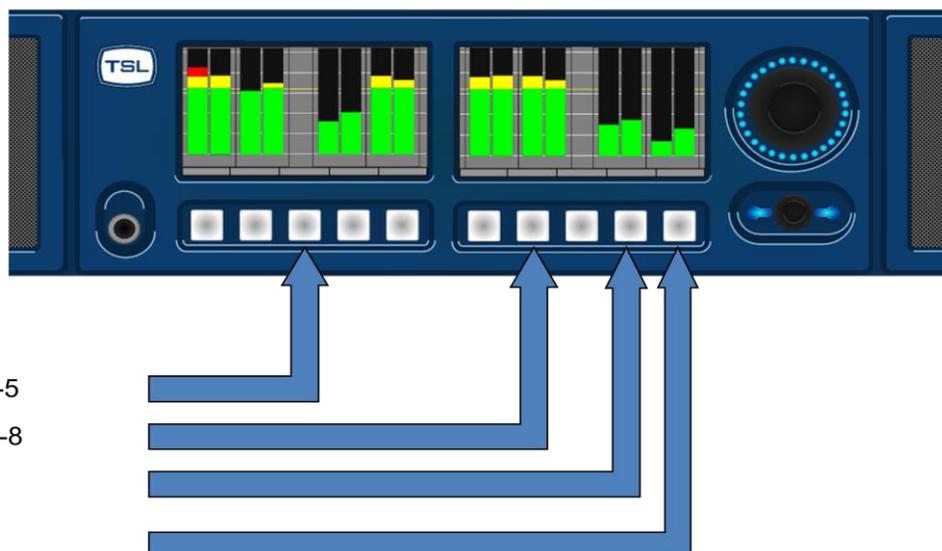
**For -18 dBFS default operation – apply DC power to the rear power connector whilst simultaneously pressing buttons 6, 7 and 8. A message will appear on the right hand OLED signalling a successful operation.**

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

**To set global reference level and meter parameters outside of the standards described, please refer to section 2.10.1 of this handbook.**

## 2.1 Controls and Displays

PAM2-3G16's menus and features are navigated via a multi-function encoder and a bank of ten pushbuttons. The diagrams below give an overview of the front control panel.



- User preset recall buttons 1-5
- User preset recall buttons 6-8
- Preset Save / Data button
- Back / Video button

- The left hand OLED screen displays bargraphs 1-8 in **Full-screen** mode and bargraphs 1-16 when Menu / Data or Video modes are activated.
- The right hand OLED screen displays bargraphs 9-16 in **Full-screen** mode and Menu / Data or Video screens when activated.
- The lower rotary encoder is used to activate the '**Scroll To Hear**' feature and to access **Menus** and **Presets**.
- The volume knob is used to internal/external loudspeaker and headphone levels. The control also features a 'push' action which **Cut/Dim** loudspeaker audio.

- Volume / Dim / Cut Control

- 8 Bargraph / Menu / Data / Video

- 8/16 Bargraph Display



- Encoder – Preset Recall / Menu / Scroll

- Scroll – Push and Rotate
- Menu – Push and Release
- Preset – Rotate to Recall

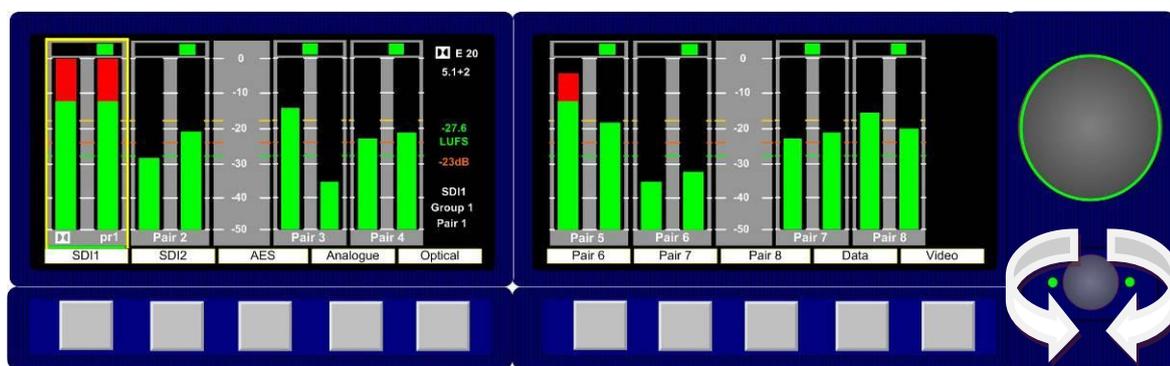
- The 5 buttons located below the left hand OLED screen are used to recall presets 1-5
- The Factory Default condition ships the following User Presets pre-programmed on buttons 1-5 respectively. These may be overwritten by customer programmed presets; defaults can be restored via the reset procedure described in Section 2.11 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. Optical Input (Digital Audio Pair 8)
- The first 3 buttons located below the right hand OLED screen are used to recall presets 6-8 and to restore system default conditions during a power up sequence.
- The 9th button which located below the right hand OLED screen is used to toggle the **Data and Loudness Histogram Screens** when in **Full-screen mode** and to save **User Presets** when in **Menu mode**.
- The 10th button located below the right hand OLED screen is used to toggle the **Video Screen** when in **Full-screen mode** and to navigate **'Back'** from **Menu modes**.

## 2.2 Top Level Screen Mode

The example below shows PAM2-3G16 in dual screen **Metering Mode** listening to **Group 1 Pair 1** of an embedded SDI video signal.

### 2.2.1 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 **PAM2-3G16** and can be used whenever the unit is not in **Menu mode**.



### 2.2.2 Video Confidence Display

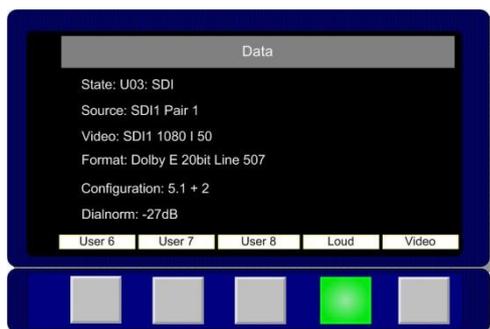
Press the **Video** button to toggle the right hand OLED display and view the input video signal. The bargraph display is compressed onto the left hand OLED screen.

The aspect ratio of the video display is selected via the **Setup Menu**.



### 2.2.3 Data Screen Mode

Press the **Data** button (button 9) to toggle the right hand OLED display and view the input signal data screen. The following image shows the **Data** screen in operation with a Dolby E signal connected via the **Optical** connector.



### 2.2.4 Dolby E Timing and Rate Information

The **Data Screen** of the **PAM2-3G16** displays extended information about any **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**

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

### 2.2.5 Dim Cut Control

The rotary volume control of **PAM2-3G16** 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 – this is indicated by a single LED dot cursor in cut and dim conditions.

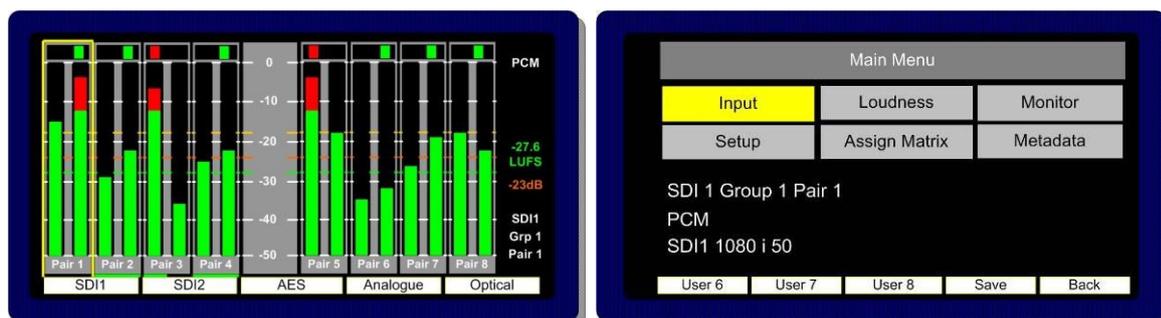
### 2.2.6 LED Brightness Settings



The **Setup 2 Menu** contains a button to select the operational brightness of the switch and encoder LEDs. By selecting and rotating the encoder the overall brightness of the LEDs may be changed from low to high level across 5 graduations. 1 is low, 5 is high.

## 2.3 Main Menu

Press the **Encoder** to access **Main Menu**. The encoder is then rotated and pressed to highlight and select from the displayed **Menu** options. From any **Menu**, pressing **Back** (illuminated red) will exit the current menu page.



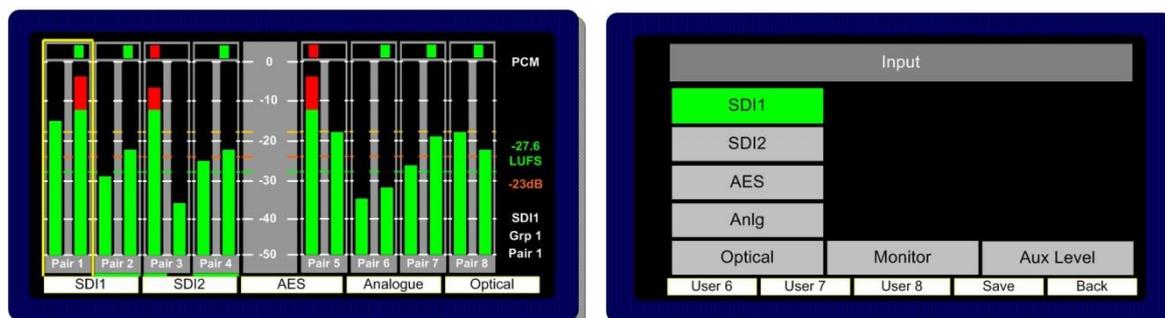
The Main Menu screen also displays information about the current selected input signal as shown in the photo above.

## 2.4 Input Selection

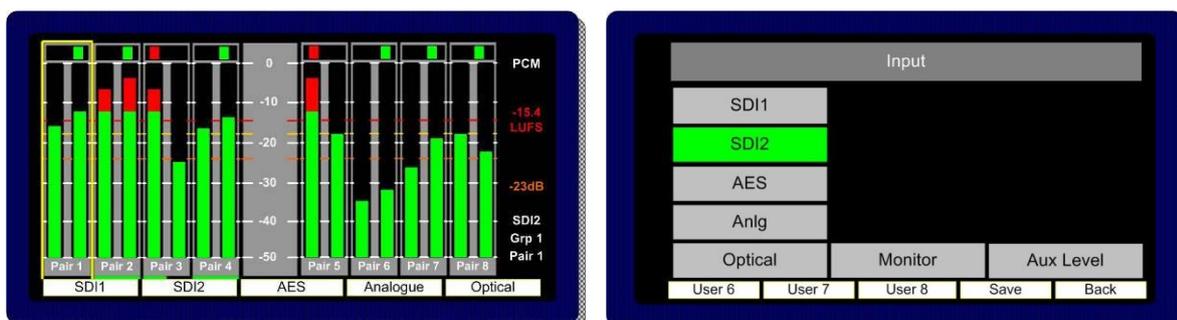
PAM2-3G16 ships with **User Presets 1-5** programmed to instantly access SDI1, SDI2, AES, Analogue and Optical TOSLINK inputs. Selecting these presets will directly route either input signal to the bargraph display. These **User Presets** may be overwritten however and customised as required. From the **Main Menu** screen, **Input Menu mode** may be selected.

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 **Monitor Menu** once the desired input is selected.

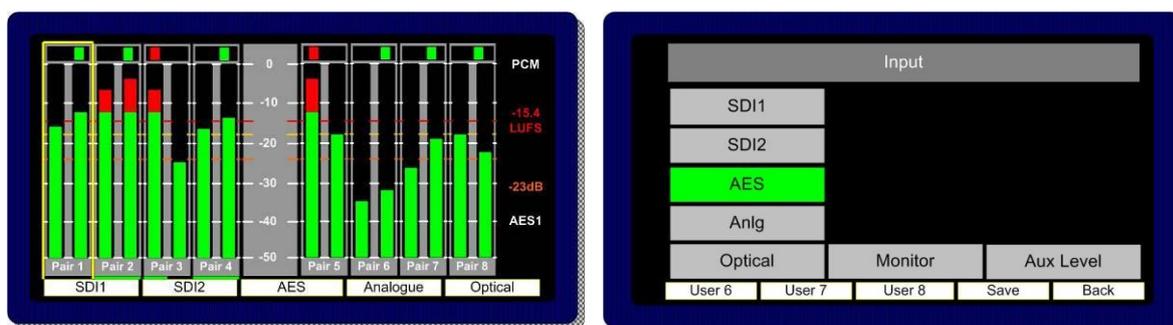
### 2.4.1 SDI 1 Input



### 2.4.2 SDI 2 Input



### 2.4.3 AES Input



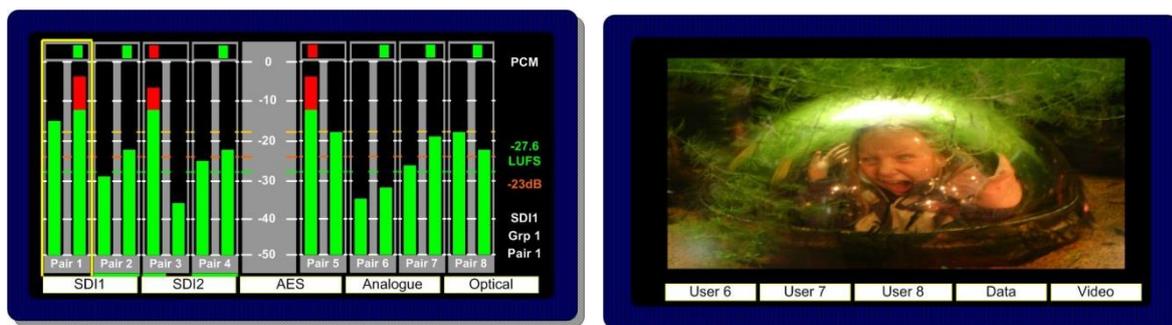
2.4.4 **Optical Input** – PAM2-3G16 is unique amongst rack mount audio monitoring products in that it features a TOSLINK Optical Input. The **Optical** Input is selected via Factory Preset 5. When **Optical** is selected, **AES channel 8** is replaced by the signal received via the **Optical** connector. This input may be used to connect directly to a Set Top Box, DVD or CD player and can decode **Dolby Digital** or Dolby Digital Plus encoded signals.

### 2.4.5 Analogue Input

**PAM2-3G16** features 6 stereo analogue audio inputs. These may be selected simultaneously by selecting **Analogue** from either the monitoring menu or preset buttons. Bargraph pairs 1-6 will display the respective input audio, pairs 7 and 8 are not used in this mode.

## 2.5 SDI Input and Video Confidence Display

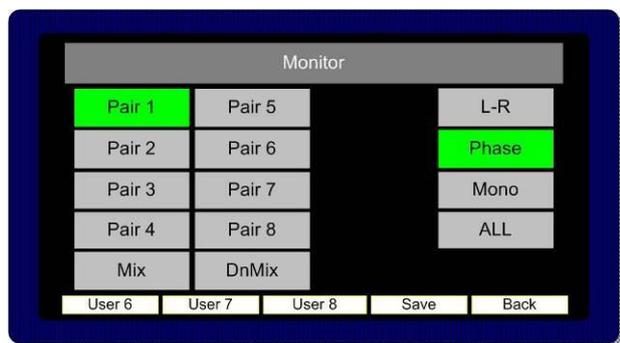
The **Video Display** may be activated using button 10 when not in **Menu mode**. The input video signal of the **SDI Input** selected last will be routed to the screen when the audio input selected is either **AES** or **Analogue**.



## 2.6 Monitor Menu

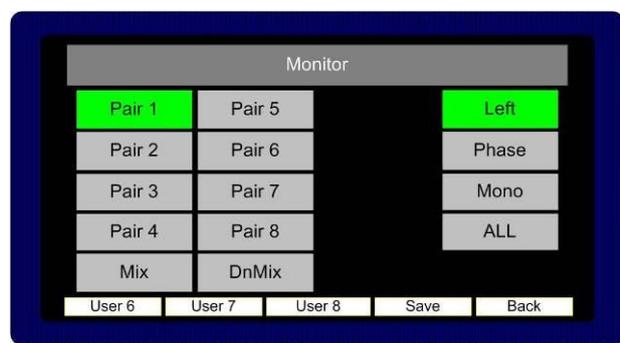
The **Monitor Menu** is used to activate listening modes not accessible via '**Scroll To Hear**' selection.

2.6.1 The **Monitor Menu** features a '**Phase Swap**' button which enables the user to reverse the phase of the right signal of a selected audio pair with respect to the right.

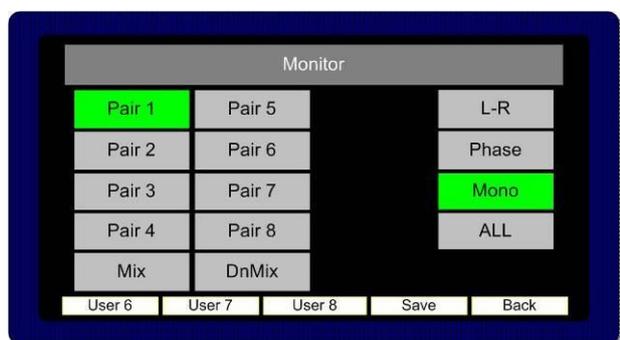


2.6.2 Pushing to select **L – R** enables individual selection of either **Left** or **Right** components of the highlighted stereo pair (or pairs).

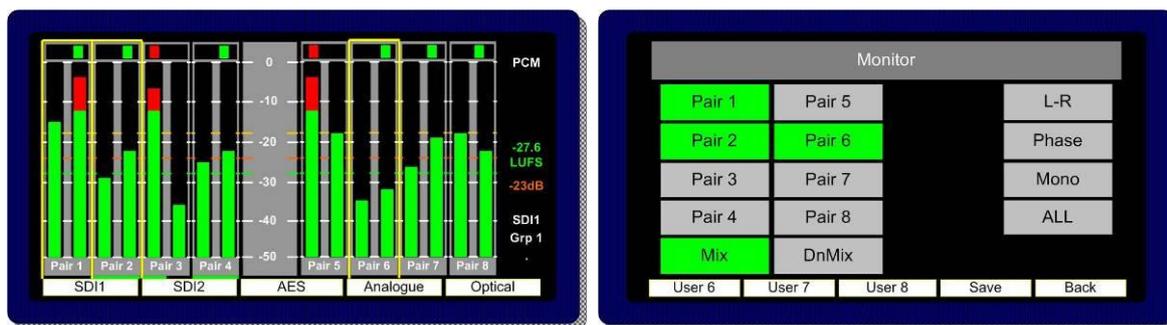
The **L – R** function also enables **Left** and **Right** audio components to be swapped.



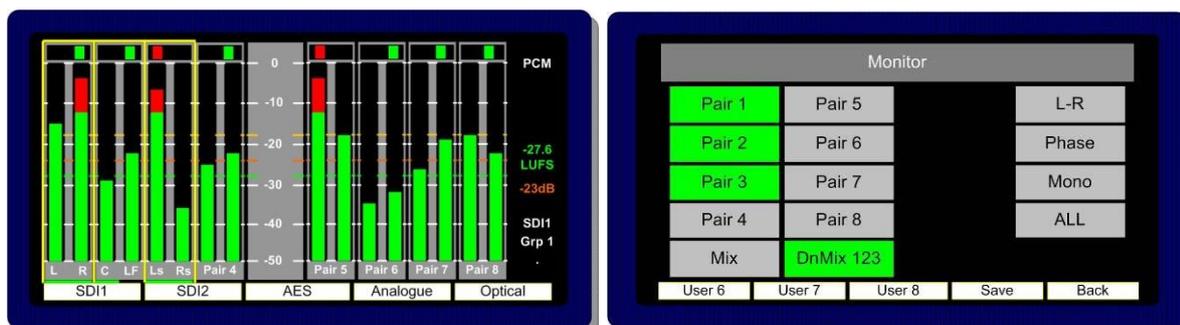
2.6.3 The **Mono** mode button mono's the selected bargraph pair



2.6.4 When **Mix Mode** is activated, multiple bargraph pairs may be selected and monitored as a mixed stereo pair.

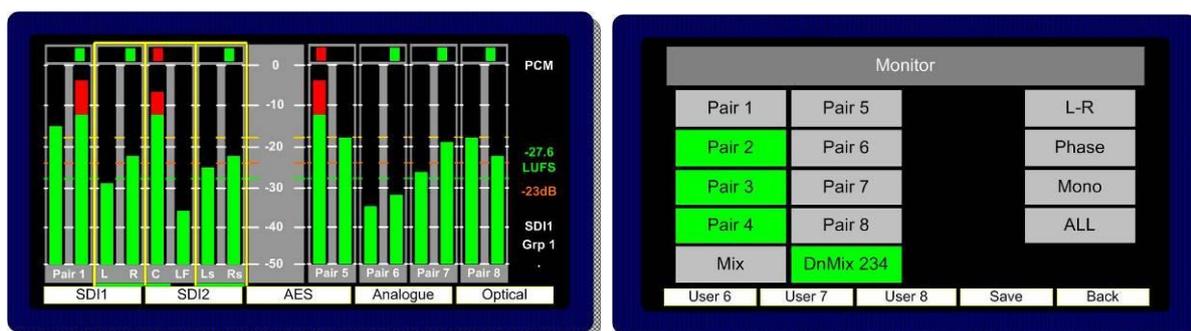


2.6.5 **Downmix Mode** allows the operator to select an adjacent group of 3 audio pairs (6 audio channels) and create a stereo **Downmix**.



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 **SMPT**E order (L, R, C, Lfe, Ls, and Rs). The **Downmix Mode** uses the ITU-R BS.775 algorithm standard.



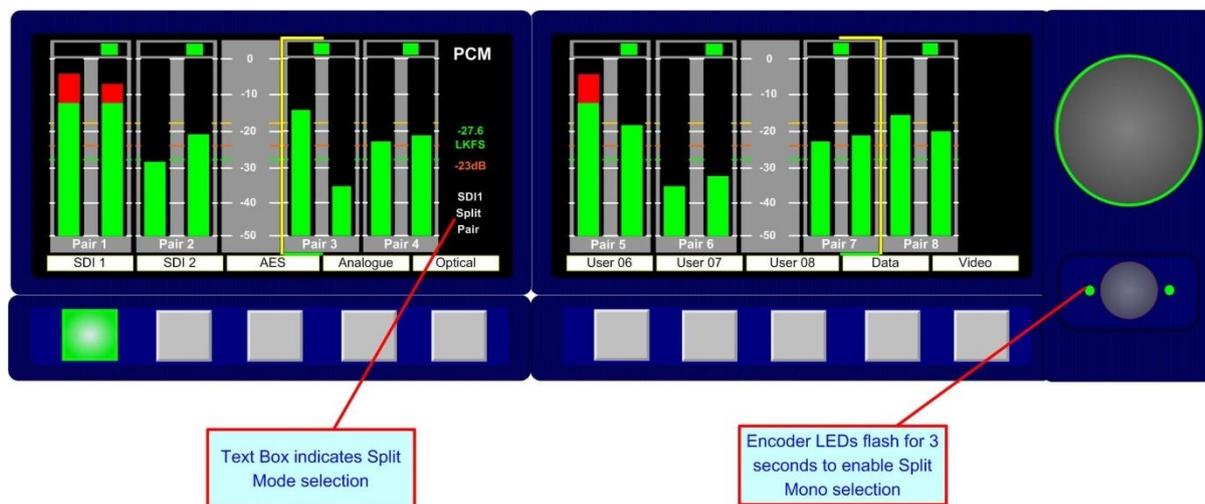
Once the 5.1 audio components are selected and **Downmixed**, the selection can be stored as a **User Preset** to a **Hotkey** (see **Storing User Presets**) and recalled at the press of a button.

### 2.6.6 Split Mono Input Selection

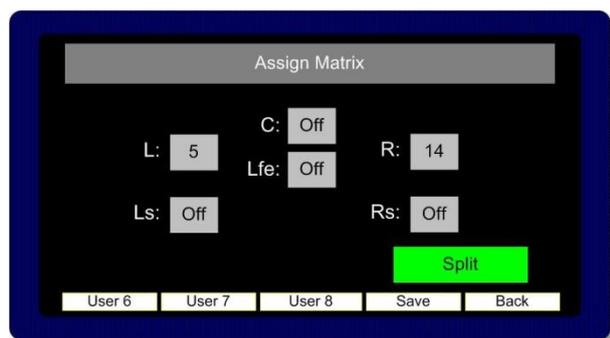
The **Scroll to Hear** function of the **PAM2-3G16** features a special operational mode which enables the operator to select to listen to two non adjacent mono audio channels.

By scrolling the yellow **Hear Box** to the first of the desired mono channels the LEDs on either side of the rotary encoder will flash for a period of 3 seconds. During that time 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 LEDs will stop flashing and the resultant selection is heard through the loudspeaker or headphone outputs.

**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 **User Preset** for instant recall (see **Saving User Presets**).



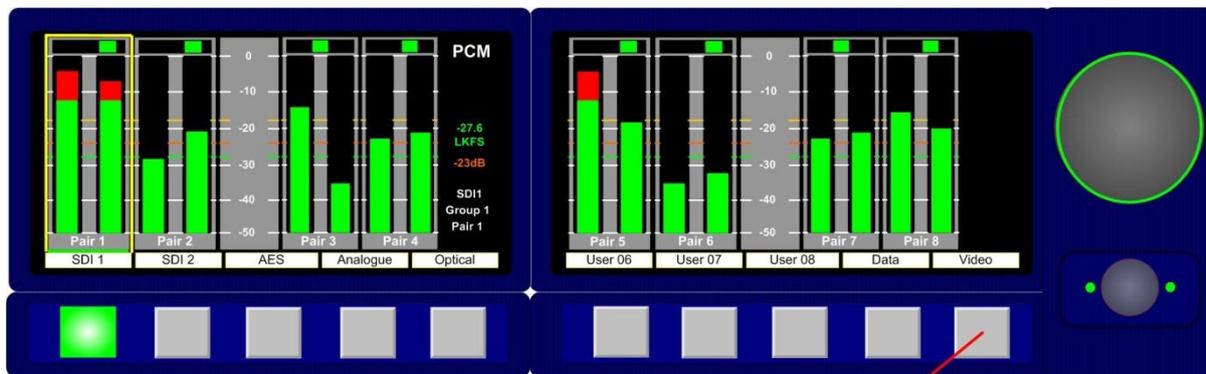
**Note;** **Split Mono** selection uses the **Assign Matrix** section of the **PAM2-3G16** in order to create a pair of non consequential audio channels. When a **Split Mono** pair is defined, the Assign Matrix represents the resultant signal selection as illustrated below.



## 2.7 Front Switch Monitor Select Mode

In order to further enhance hands on operation of **PAM2-3G16**, an operational mode exists which enables the user to toggle between the standard front panel operation of 8 user preset recall switches below the OLED displays and direct access to up to 6 monitoring mode selector buttons.

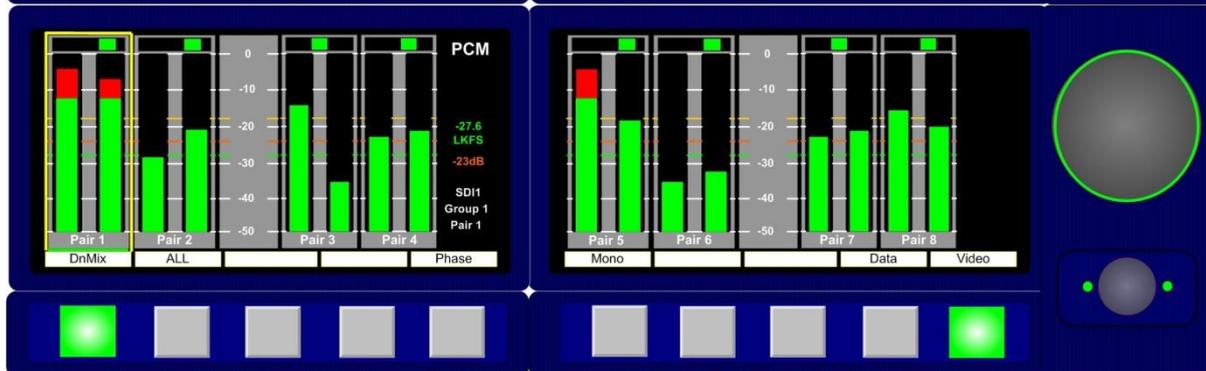
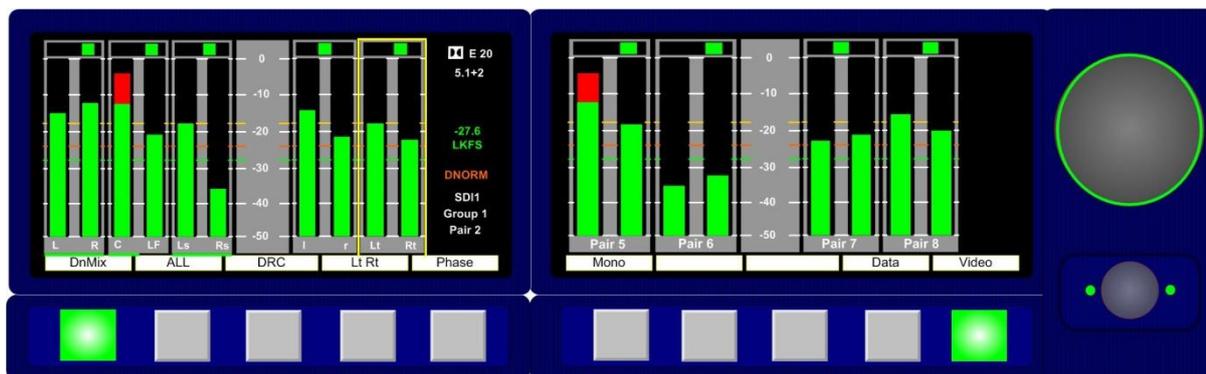
By simply holding button 10 (**Video Button**) for a period of more than one second, the display of buttons 1-8 switches from the standard operating view of User Preset recall selection to one giving direct and instantaneous access to a number of **Monitoring Menu** features.



Push and Hold Button 10 for 1 second to switch modes

The display changes depending on the whether the signals currently monitored are **Dolby E, D, DD+** or **PCM** audio. The images below represents the front panel view for Dolby encoded signals and **PCM** audio respectively.

### Dolby



### PCM

**DnMix**

**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. Repeatedly pressing **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

**ALL (Non Dolby)**

By selecting **ALL**, the operator can select the group of 6 audios from 2 groups which corresponds to the 5.1 audio signal. 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**

**ALL and DRC (Dolby Only)**

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 output is affected by Dolby DRC metadata and adjusted accordingly. 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 **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. **DRC** monitoring mode can be selected via the **DRC** button in the enabling the user to hear the effect of dynamic range compression on the decoded signal and hence check the integrity of the audio heard by the consumer.

**Lt Rt (Dolby Only)**

By pressing the **LtRt** button when **PAM2-3G16** is decoding a Dolby E, D or DD+ signal, a stereo **Downmix** will be selected. This is the default listening/monitoring condition for a Dolby signal and is derived using the originally authored Dolby metadata thus enabling the user to hear the effect of stereo downmixing as heard by the consumer.

**Phase**

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

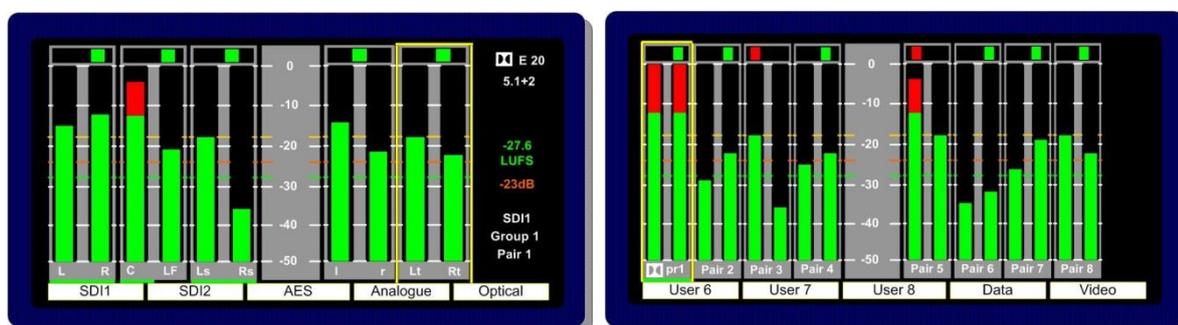
**Mono**

The **Mono** mode button mono's the selected bargraph pair.

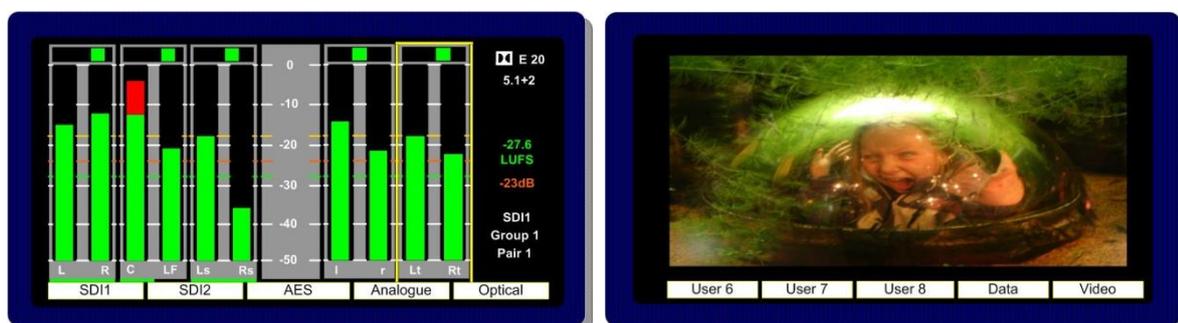
## 2.8 Dolby E, Dolby Digital and Dolby Digital Plus Monitoring

2.8.1 When a **Dolby E, Dolby Digital or Dolby Digital Plus** encoded audio pair is selected, **PAM2-3G16** 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 **OLED** display – an Lt/Rt (or Lo/Ro) **Downmix** bargraph pair is created from the **Metadata** and shown as a 5<sup>th</sup> pair on the left screen (shown below).

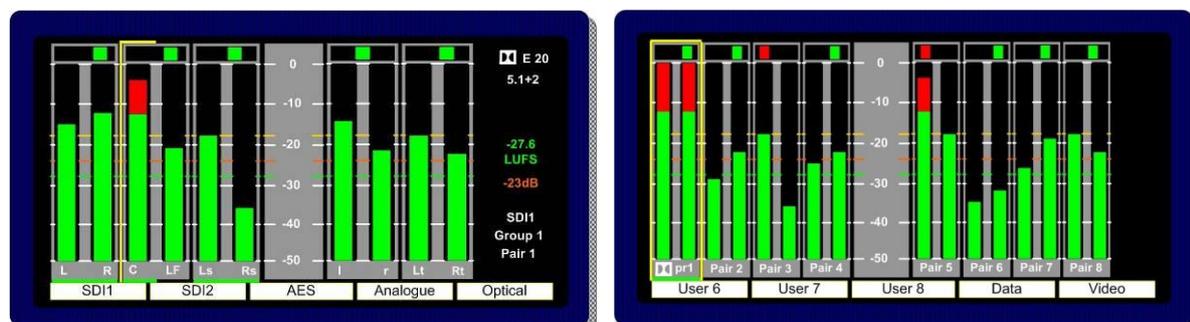
In **Top Level Mode** the input bargraph display is shown on the right **OLED** display with a **Dolby** logo now superimposed below the decoded bargraph pair (as shown).



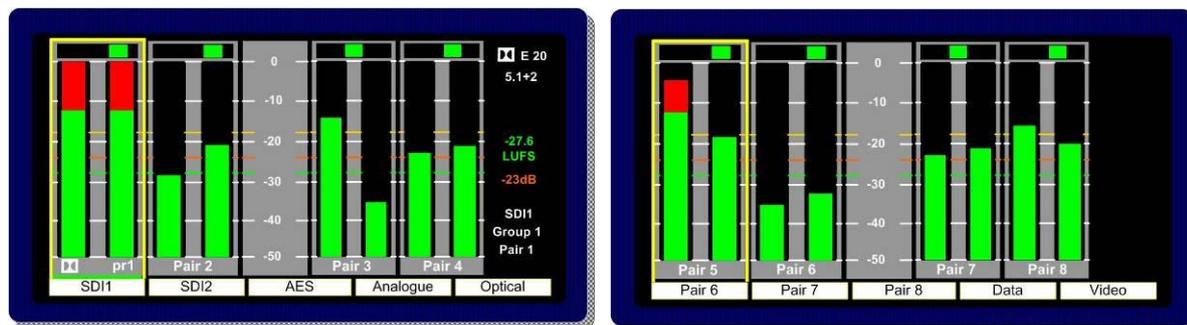
The **Video Display** may be activated using button 10.



2.8.2 **Scroll to Hear Mode** may be used 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 **PAM2-3G16** will revert to the **Top Level Mode** as shown below. The encoder can be rotated to move the ‘**Hear Box**’ and select another signal from the **top level screen**.



## 2.9 Monitoring using external loudspeakers

The **PAM2-3G16** 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 2.9** explains the default monitoring modes based on the various menus and DIP switch selections detailed in the following descriptions.

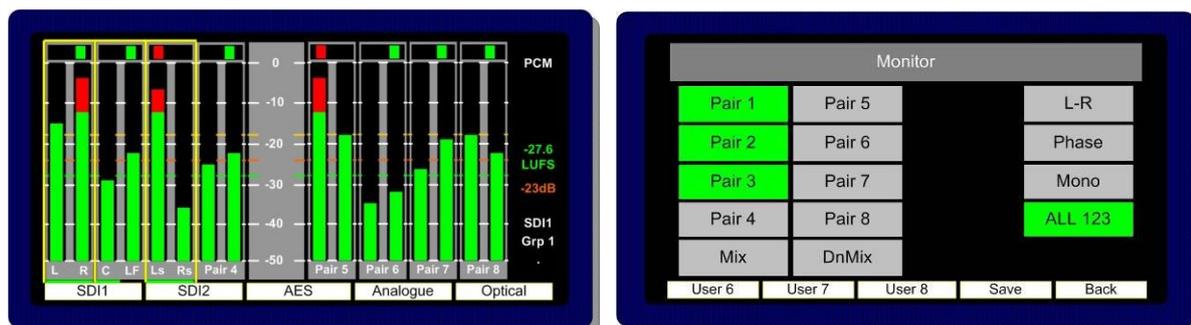
De-embedded or decoded multichannel audio (Dolby or discrete PCM) is automatically routed to the AES and analogue outputs connectors. The rear panel located **DIP Switch** includes a selection to mute internal loudspeakers and to set AES and analogue outputs to variable level (adjusted by the front panel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

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

### 2.9.1 Discrete PCM 5.1

The non-Dolby monitoring menu also includes an **ALL** option. By selecting **ALL** the operator can select the group of 6 audios from 2 groups which corresponds to the 5.1 audio signal. 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**.



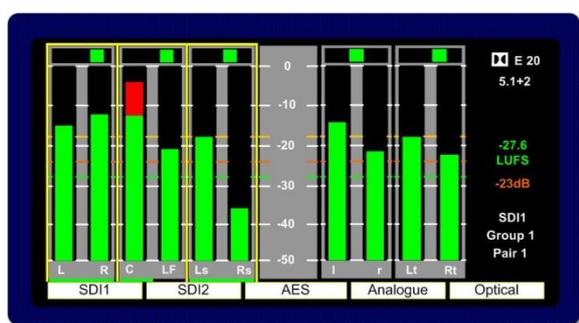
This function ensures that an embedded discrete **5.1** signal can be monitored regardless of the location of the 5.1 stems. The user can store different settings for monitoring discrete 5.1 audio in their **User Presets**.



The **ALL** listening mode **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 user presets for instant recall.

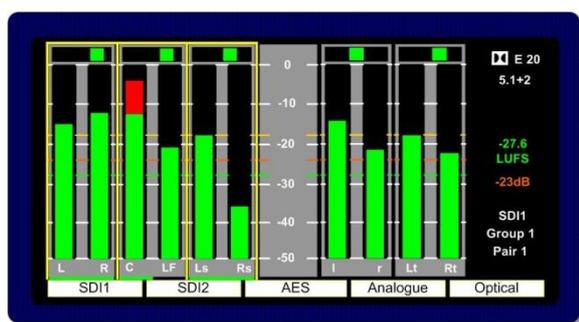
### 2.9.2 Dolby E, Dolby Digital and Dolby Digital Plus

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 output is affected by Dolby DRC metadata and adjusted accordingly. 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.



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
User 6	User 7	User 8	Save Back

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.



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
User 6	User 7	User 8	Save Back

## 2.10 Internal and External Monitoring Modes

The table below indicates the various monitoring modes available to PAM2-3G16 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 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** or **Analogue** 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.

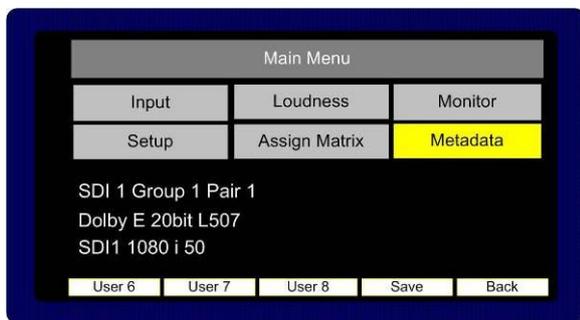
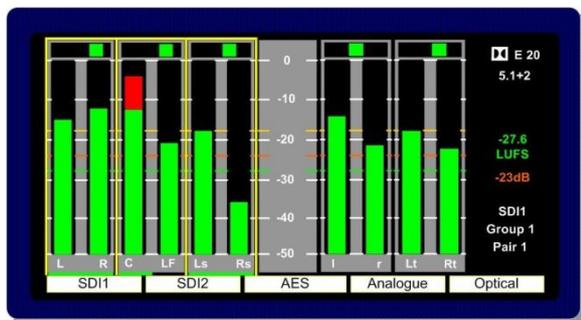
Monitoring Mode	Dolby Encoded Audio				Discrete Audio			
	Internal LS	Ext Fixed - D25 (Int LS On)	Ext Fixed - D25 (Int LS Off)	External Variable	Internal LS	Ext Fixed - D25 (Int LS On)	Ext Fixed - D25 (Int LS Off)	External Variable
All	Muted	Full	Full - AES Only	Full	Muted	Full	Full	Full
DRC (Line or RF)	Muted	Full	Full - AES Only	Full	Muted	Full	Full	Full
Lt Rt	Lt Rt	Full	Full - AES Only	Lt Rt (1st pair only) 2-4 muted	N/A	N/A	N/A	N/A
Lo Ro	Lo Ro	Full	Full - AES Only	Lo Ro (1st pair only) 2-4 muted	N/A	N/A	N/A	N/A
DMix	DMix	Full	Full - AES Only	DMix (1st pair only) 2-4 muted	DMix	Full	DMix (1st pair only) 2-4 muted	DMix (1st pair only) 2-4 muted
* Individual	Individual	Full	Full - AES Only	Indiv. (1st pair only) 2-4 muted	Individual	Full	Indiv. (1st pair only) 2-4 muted	Indiv. (1st pair only) 2-4 muted

\* **Individual** refers to single, paired or multiple audio channels selected from the monitoring menu.

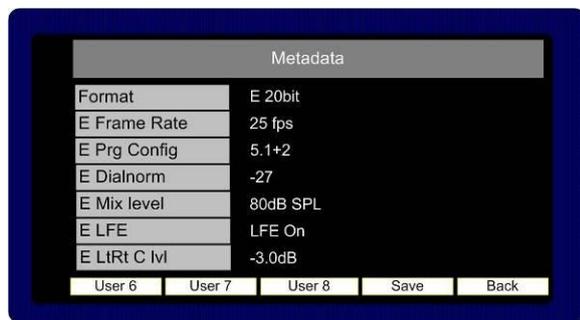
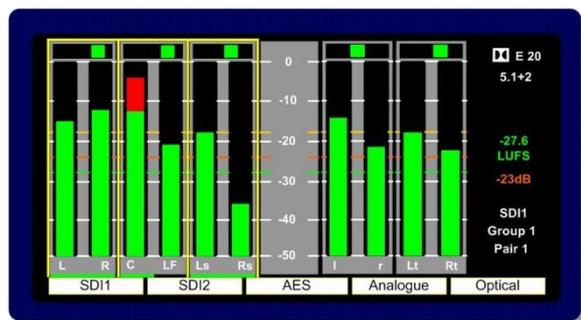
## 2.11 Data and Metadata Menus

**2.11.1 Dolby Metadata.** Primarily for use with **Dolby E**, **Dolby Digital** and **Dolby Digital Plus** encoded audio signals; the **Metadata Menu** accesses a user selectable 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.

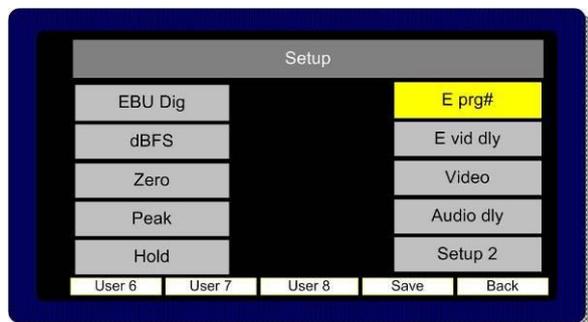
**Dolby E** signals might consist of up to 8 separate programmes (ie: 5.1+2 contains 2 programmes, 2+2+2+2 contains 4 programmes etc) and each programme may be configured with different Metadata parameters.



The example below shows the selection of Dolby E PGM 1 as the default for the Metadata screen associated with the selected 5.1 audio signal.

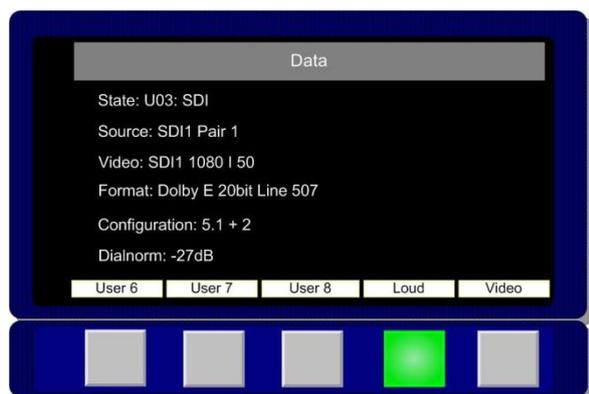


The **Setup Menu** allows the user to select which programme the Metadata is read from. The screen below shows the **E prg#** selection on the Setup window. By selecting **E prg#** the user may scroll through **Dolby E** programme numbers 1 to 8 and select the appropriate programme number. This setting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



### 2.11.2 Data Screen.

The **Data Screen** can be toggled on/off from the 9<sup>th</sup> button on the front panel when in **Top Level Mode**. When the **Loudness** feature is activated, the **Data** selection will toggle between **Bargraph/Data/Loudness**.



The **Data Screen** provides up to 6 fields of feedback regarding the signal being monitored and the operational status of **PAM2-3G16**.

**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

## 2.12 Setup Menu 1 and 2

**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 **PAM2-3G16** to suit their needs .

### 2.12.1 Bargraph Settings and non-system default conditions.

The standard operating conditions of **PAM2-3G16** 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)

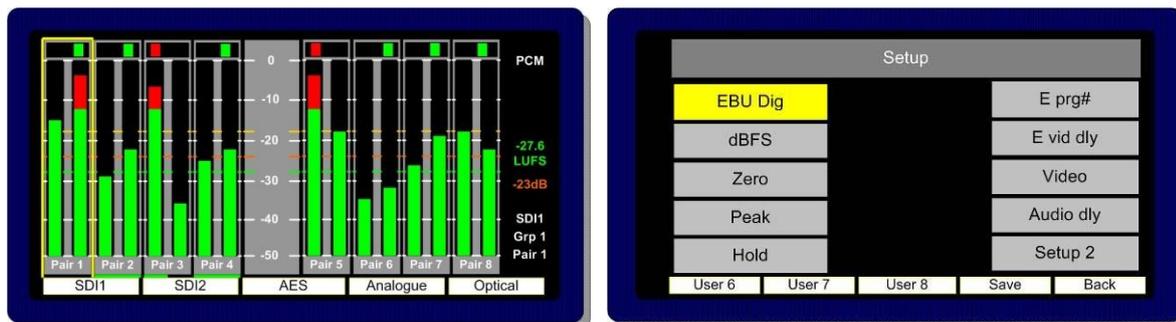
After 15 seconds, you should get a message saying "Default Setup stored SW10 to proceed".

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

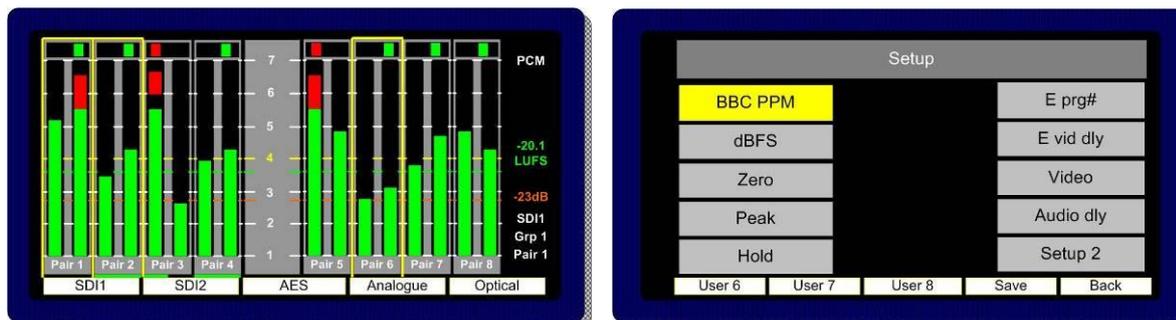
The factory memories can only be changed by repeating this process (or by a factory reset), but the user memories can be configured as desired (so you can have different metering types on different presets).

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

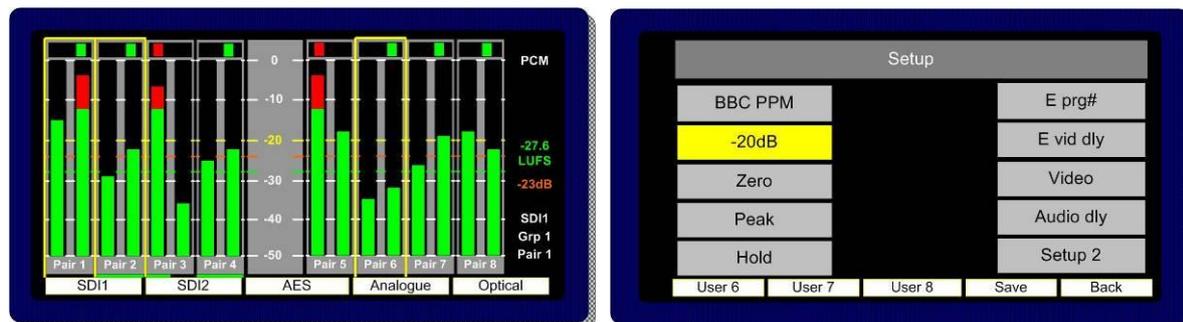
The bargraph example below shows a selected **EBU Digital** scale with **-18dBFS** reference level (yellow indicator line on **Bargraph** display).



The example below shows the previous **EBU Digital** bargraph scale changed to display a **BBC PPM** bargraph.



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).

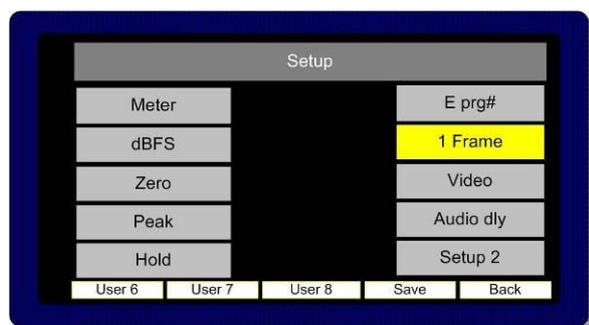


When set to **Block** mode, the **Peak** parameter enables the user to choose to illuminate the indication between zero **dBFS** and the peak level in yellow.

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

### 2.12.2 Dolby E Video Delay

The **PAM2-3G16** 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 **PAM2-3G16** will be subject to a **1 frame delay** when the video delay mode is active.

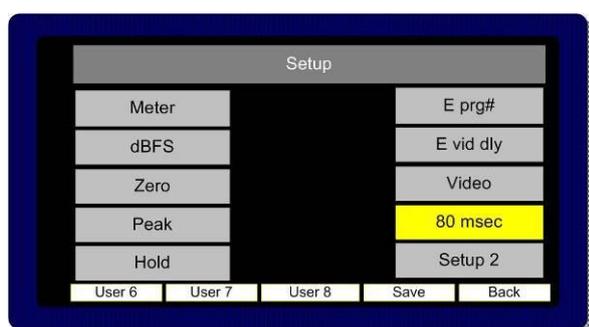


### 2.12.3 Audio Delay

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.

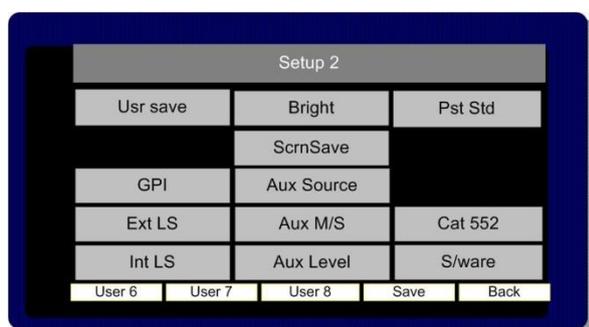
By selecting '**Audio dly**' from the **Setup Menu** whilst pushing and rotating the encoder, audio delay in 1ms steps up to a total value of 250ms may be selected. The selection must be stored as part of the associated **User Preset** and is not set as part of the system default. This allows the user 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.



### 2.12.4 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 2.15**). **Setup 2** is accessed from a button within the first level **Setup Menu**



### 2.12.5 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.

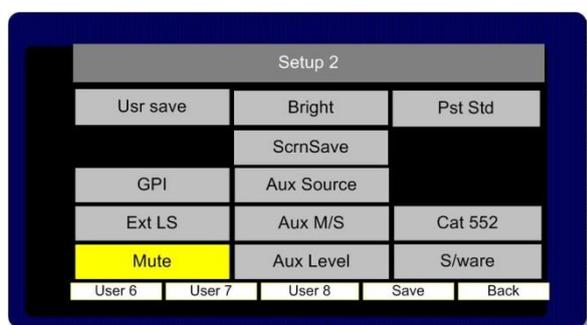
### 2.12.6 LED and OLED Brightness (Bright)

The default brightness level of the OLED displays, button and encoder LEDs can be altered on a scale of 1 to 5 (brightest) to suit operational conditions. Default setting is 3.

### 2.12.7 GPI

GPI functionality is described elsewhere within this handbook (see Section 2.17)

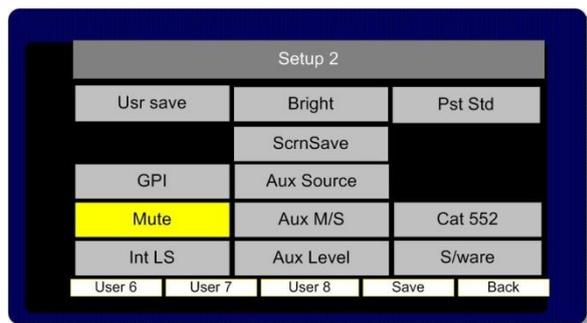
### 2.12.8 Internal Speaker Mute (Int LS)



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 **PAM2-3G16** 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 **PAM2-3G16** operators share a common external speaker system.

### 2.12.9 External Speaker Mute (Ext LS)

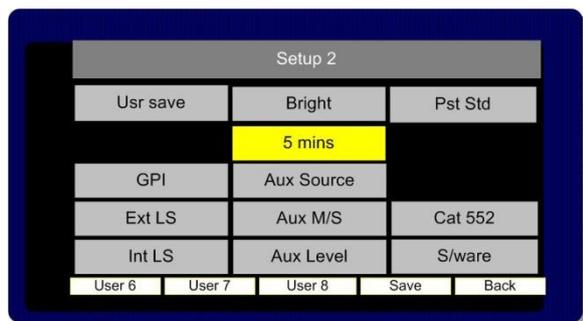


The **External loudspeaker (analogue and AES variable level outputs)** connections may be muted by using the button located within **Setup 2** menu. **The action of the on screen External LS mute switch is preset dependant and not a system default.**

### 2.12.10 Preset Standard Auto-Switching (Pst Std)

Preset Standard Auto-Switching functionality is described elsewhere within this handbook (see Section 2.14)

### 2.12.11 Energy Saver Mode (ScrnSave)



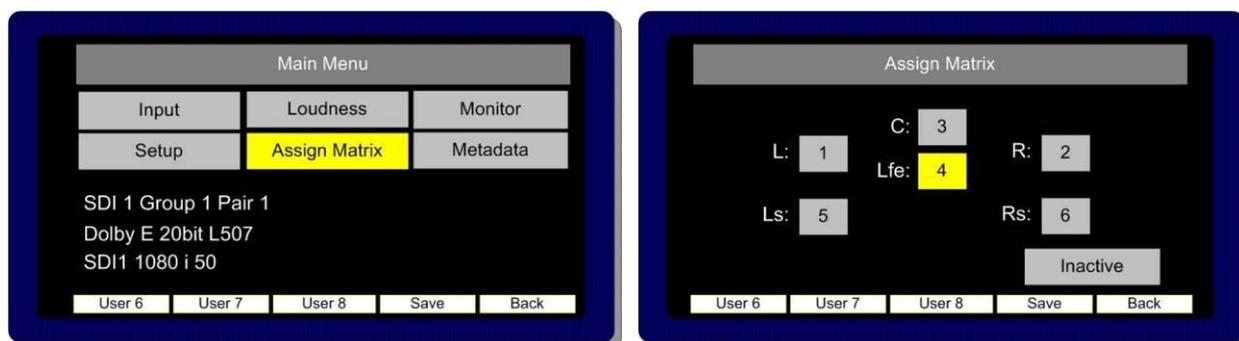
**PAM2-3G16** features an energy saving mode which dims the OLED displays after a defined period of time when activated. The **ScrnSave** button within **Setup2** menu access 4 preset time settings after which the OLED displays will dim, reactivated once a button is pressed or an encoder turned by one click. The time out settings available are **5, 10, 20 mins** and **None**.

## 2.13 Assign Matrix

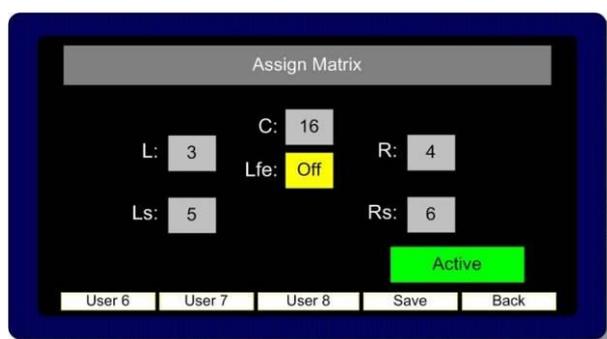
The **Assign Matrix** is intended for use in multichannel audio monitoring situations where discrete surround sound signals are being transported in non SMPTE or unconventional channel order.

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 **OLED**. Standard **PAM2-3G16** 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.

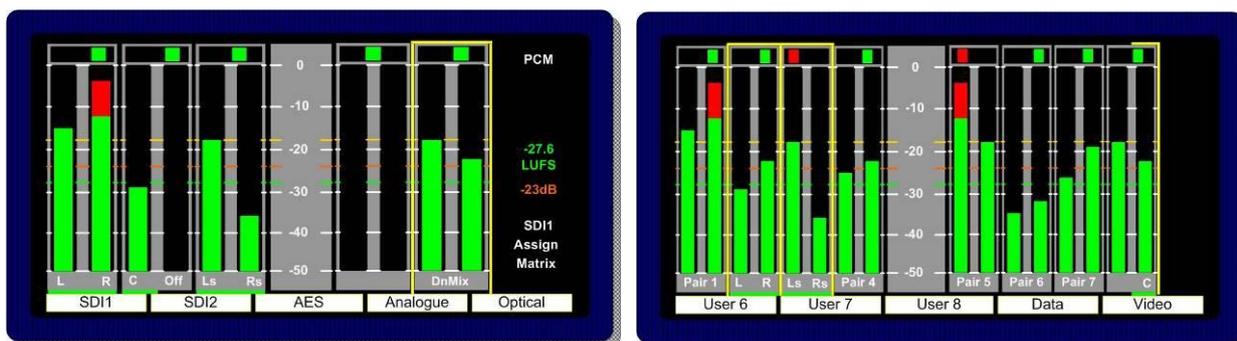


**Application Example 1**; a clean 4.0 surround mix of **Left, Right, Left** and **Right Surround** is embedded sequentially across channels 3 to 6 of an SDI signal while multiple language mono **Centre** channels are embedded elsewhere within the same signal. An operator it needs to be able to reconstruct 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 OLED 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.

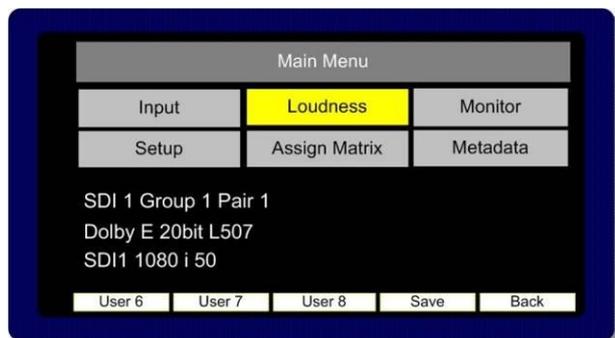
The **Assign Matrix** may be used for non-surround applications and is ideal for creating customised stereo or mono mixes from multiple mono sources (by using **Downmix**) or for applications such as **'Solo in Place'** and **LCR** monitoring.

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.

## 2.14 Loudness Measurement

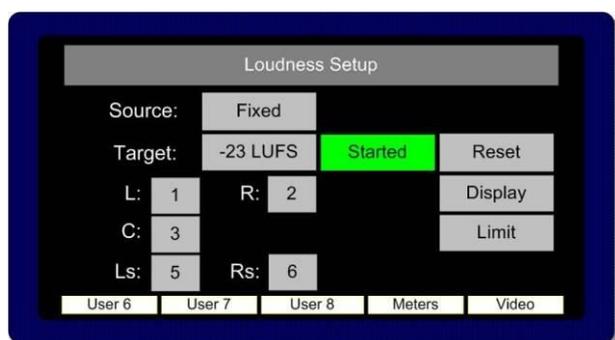
ITU-R BS1770 **Loudness Setup** menu is accessed from the **Main Menu**.

**PAM2-3G16** 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).

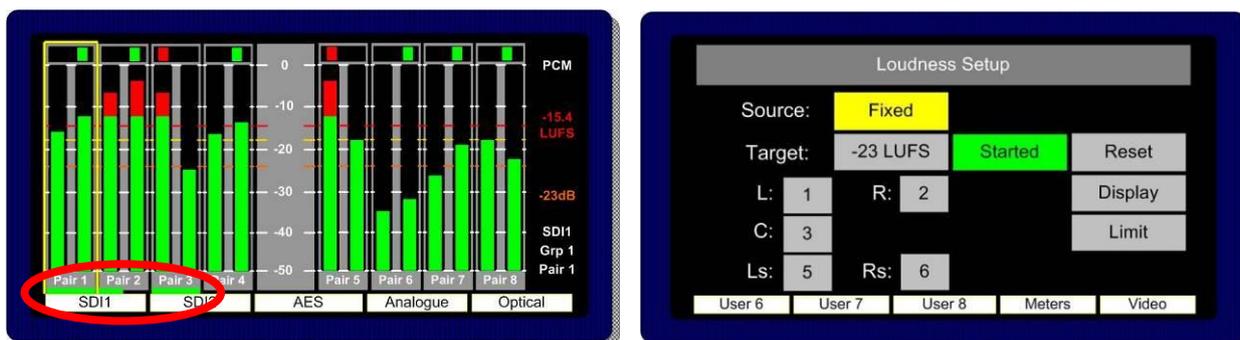


### 2.14.1 Source Selection.

The **PAM2-3G16** 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.

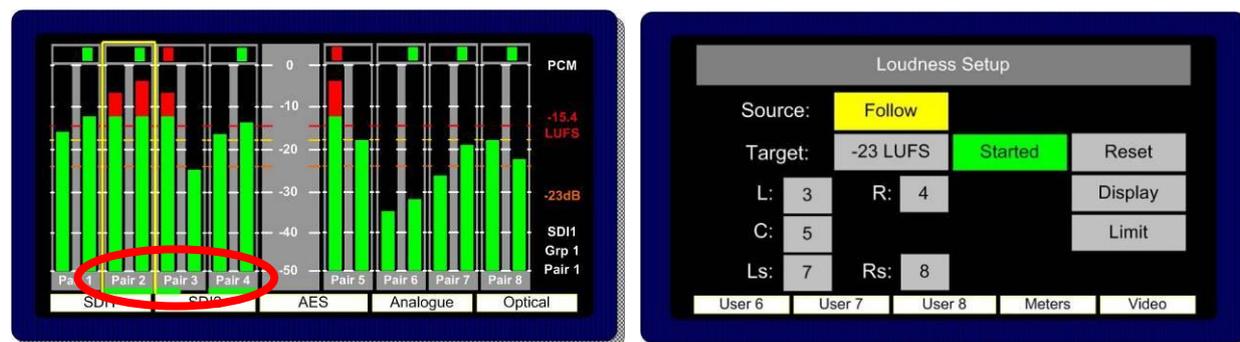
**Off Mode:** The **Loudness** menu **Source Selection** includes an **OFF** position for use when the **Loudness** measurement is not required. This switch position disables all onscreen representation of **Loudness** measurement including the **Target** value, **Measured** value, **Histogram** and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

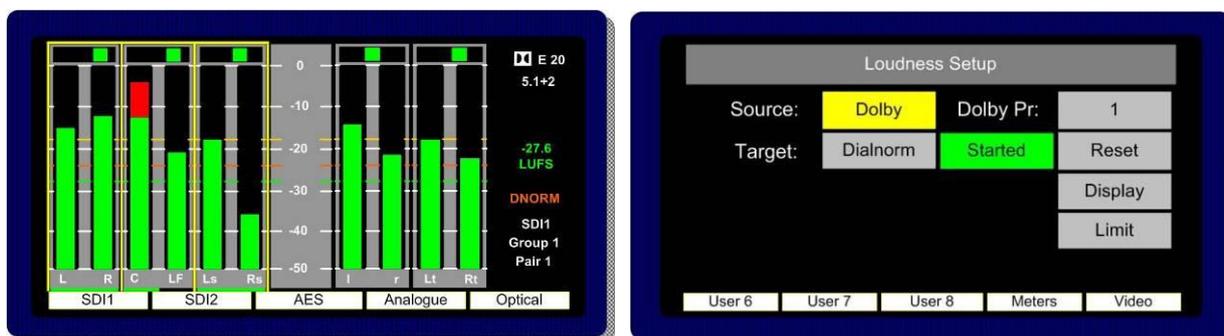


Their selection is denoted by the horizontal green cursor displayed below the ‘pair’ identification. Providing the user does not switch away from the selected signal input or **User Preset**, then the **PAM2-3G16** will continue to measure the Loudness of the highlighted audio components. In this mode of operation **PAM2-3G16** 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 **PAM2-3G16** makes the assumption that the audio signals are present in the correct **SMPTE** channel order for **5.1** sound. As per the **ITU-R** guidelines, the LF channel is not subject to measurement.

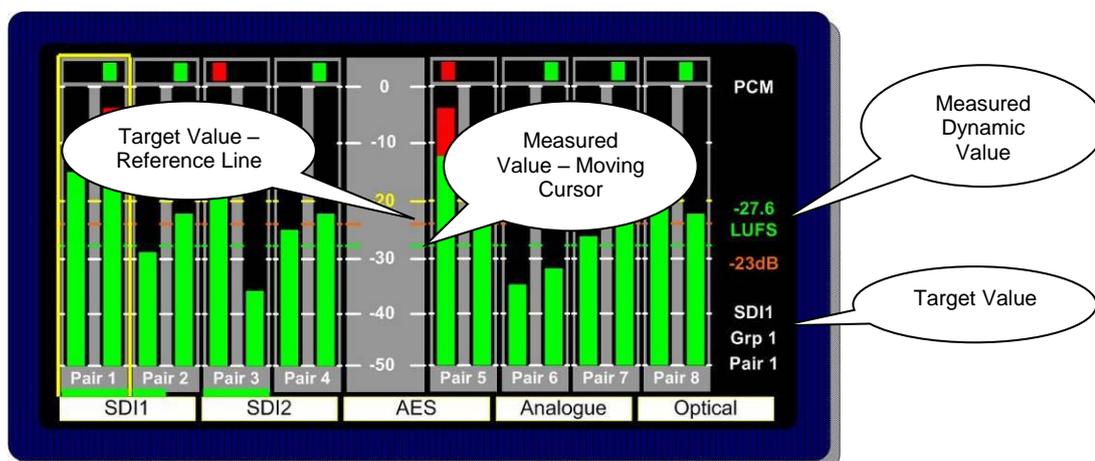


**Dolby Mode:** When decoding a Dolby signal, the **Loudness** measurement feature can be set to automatically select the appropriate audio components of the 5.1 programme. Since the **PAM2-3G16** identifies the channel configuration via the embedded **Dolby Metadata**, the Loudness feature uses this information to define the signal under test.



### 2.14.2 Target Level, Limit Threshold and Red Line Alarm

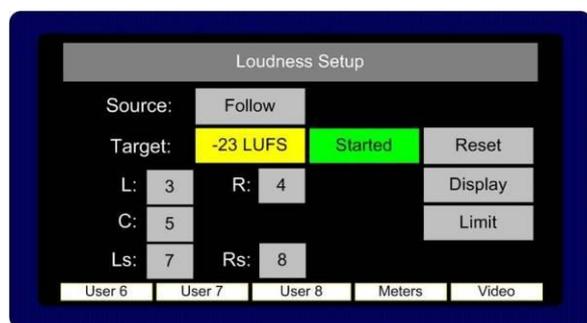
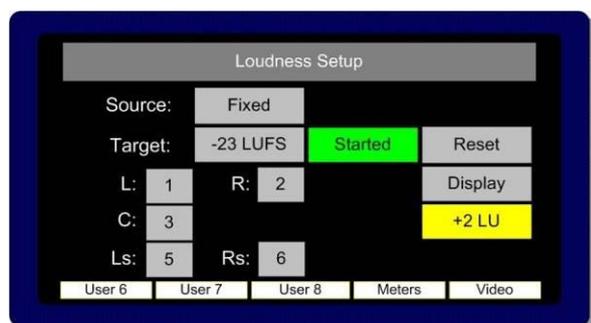
The purpose of Loudness measurement is to ensure that TV programme audio content is produced, distributed and ultimately transmitted at a level which does not inconvenience the listener and/or contravene any local or national regulations. **PAM2-3G16** helps the broadcaster to monitor the **Loudness** 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 OLED 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, **PAM2-3G16** 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.

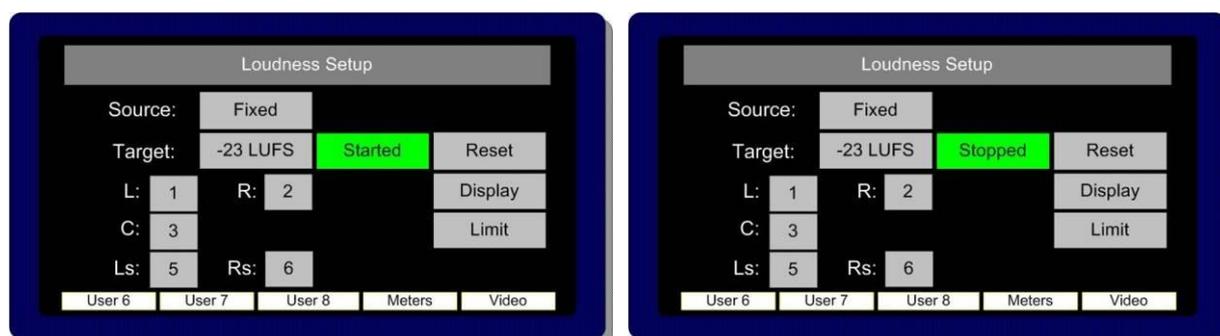


### 2.14.3 Start, Stop and Reset Triggers

In order that **Loudness** measurement is targeted towards specific programme content, **PAM2-3G16** 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 **D15** 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.

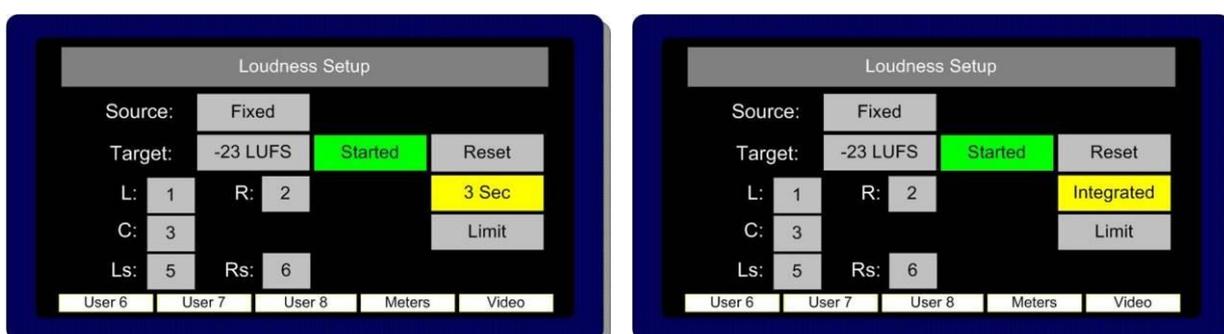


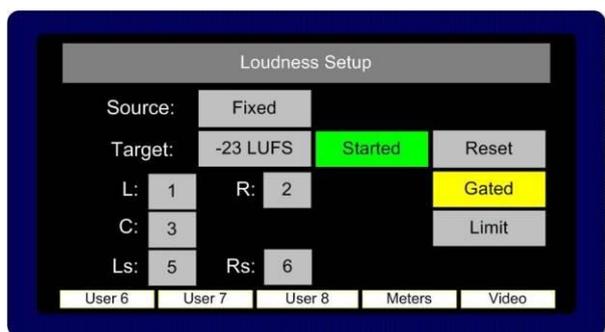
### 2.14.4 Histogram Display and Integration Settings

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**.

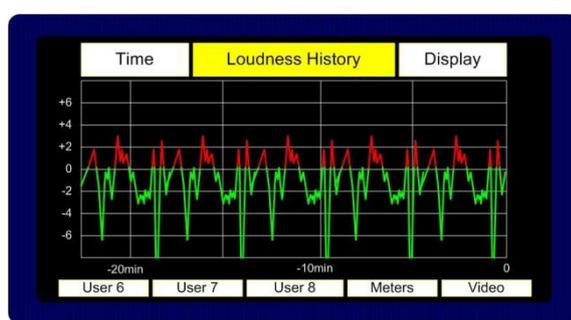
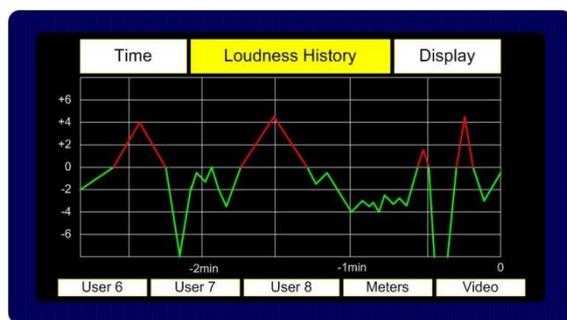
The sample for all Loudness measurement is set internally at 400ms. The 3 modes of display are as follows;

- 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 a -8dB gating level is discarded. At the time of writing (September 2010) the use of gating for ITU-R BS1770 loudness measurement was being discussed and 8dB suggested as a suitable threshold value pending agreement.



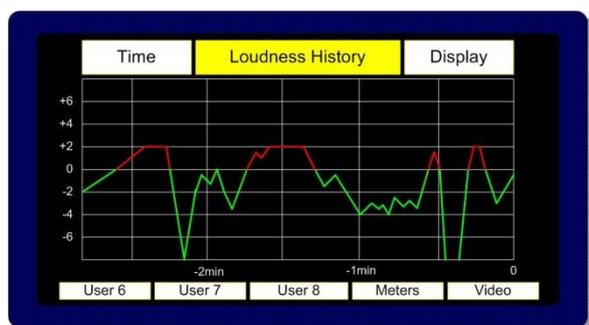


The **Loudness Histogram** is recalled via the 9<sup>th</sup> button at the **Top (Bargraph) Level** by toggling the **DATA** button between **Data/Loud/Meter** modes. The **Histogram** view can display loudness against time over a **2, 10, 20, 40** and **80 minute** period. The scale shows variation of  $\pm 6$  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



## 2.15 Preset Standard Rules

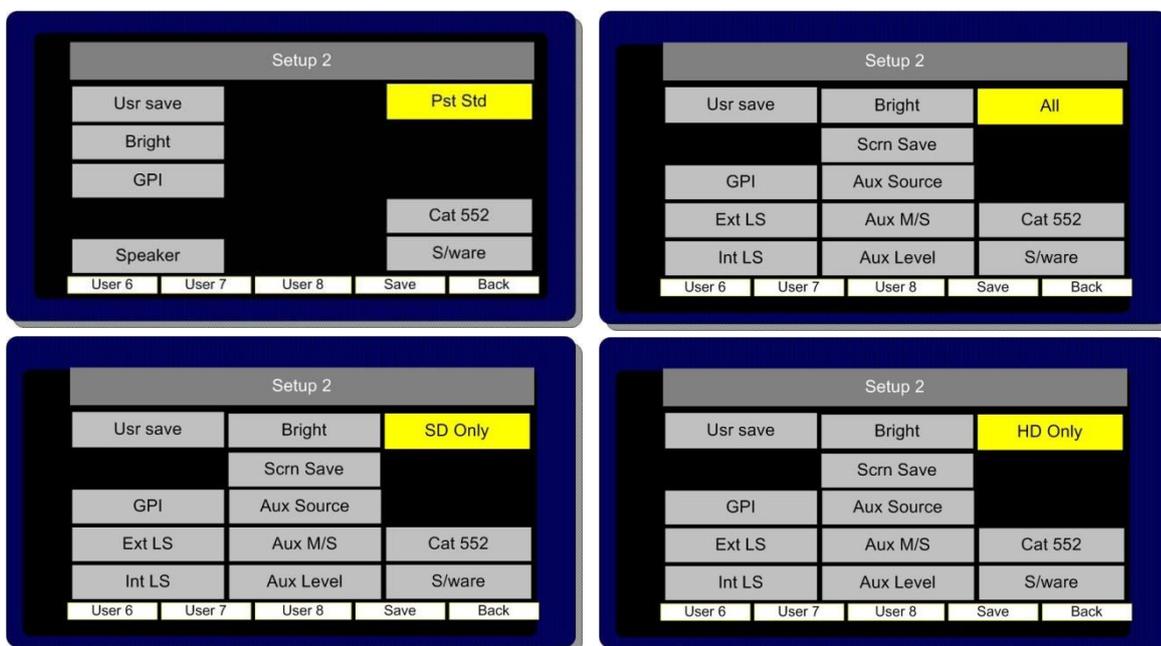
The **Preset Standard Rules** feature within the **Setup 2** menu enables **PAM2-3G16** to automatically switch between **User Presets** depending on the input signal format. Each **SDI** monitoring mode or **User Preset** is subject to the action defined by the **Preset Standard (Pst Std)** button however the default operating condition renders the setting irrelevant unless the button is pushed and a change activated.

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 **PAM2-3G16** operates as usual and is oblivious to changes to input signal types, auto sensing and functions as normal.
- **HD Only** should be selected as part of a **User Preset** which is intended to be automatically activated whenever an **HD** signal is received.
- **SD Only** should be selected as part of a **User Preset** which is intended to be automatically activated whenever an **SD** signal is received.

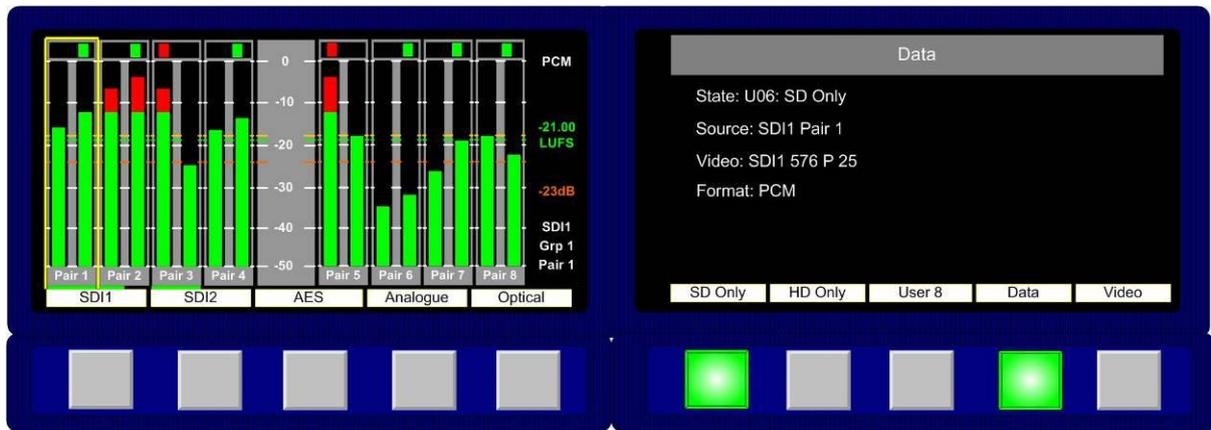
In use, when monitoring modes are being setup and saved as a **User Preset** which will be subject to **Preset Standard** rules, the **Pst Std** selection must be made and saved as part of the preset or presets.

**Important:** For **Preset Standard** to work, the **SD** preset should be stored to the left and directly adjacent to the **HD** preset on a pair of front panel **User Preset** buttons. The **SD** button must first be 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 **User Presets** between which the **PAM2-3G16** 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 **User Preset** buttons 6 and 7.



## 2.16 Auxiliary Input Mixer

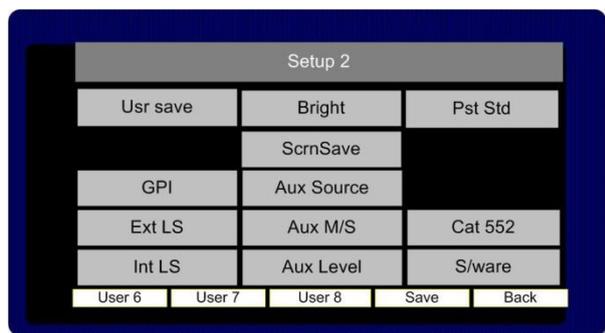
The **Auxiliary Input Mixer (AIM)** is intended for use in environments where users of **PAM2-3G16** wear headphones during normal operation and need to listen to an additional audio input signal such as Intercom (Talkback) mixed into their monitoring feed.

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 **PAM2-3G16** 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 **PAM2-3G16**.

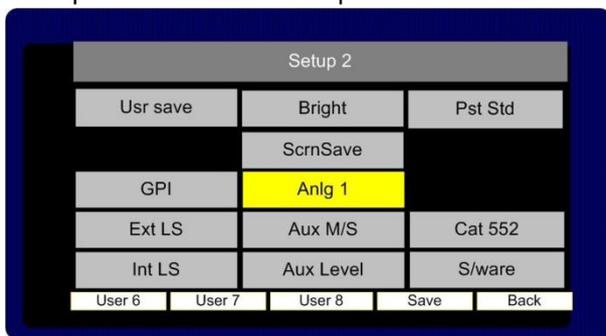
**Auxiliary Input Mixer** can be accessed via the **Setup 2** menu where 3 buttons allow the user to select the desired input, define whether it is mono or stereo and adjust an input level trim. In order that the input level can be quickly adjusted, the level trim control is duplicated on the **Input**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

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

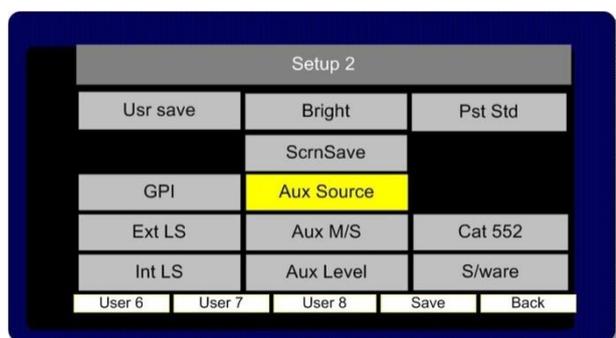
Access to the **Auxiliary Input Mixer** is via the **Setup 2** menu.



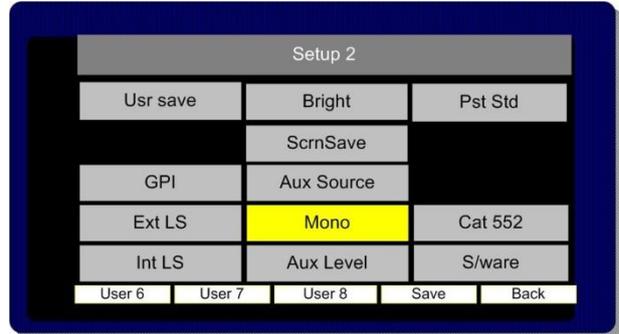
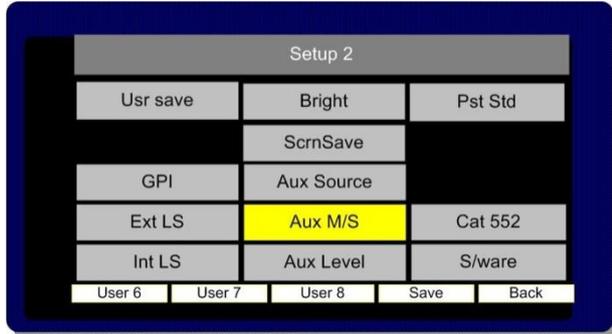
**Aux Source;** By using the encoder to select the **Aux Source** button, the user can scroll through the inputs and select the required source to



route to the **Auxiliary Input Mixer**. Analogue Stereo Inputs 1 to 6 and AES pairs 1 to 8 are available for exclusive selection.



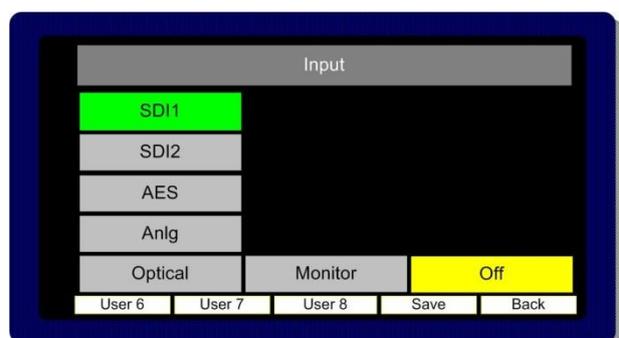
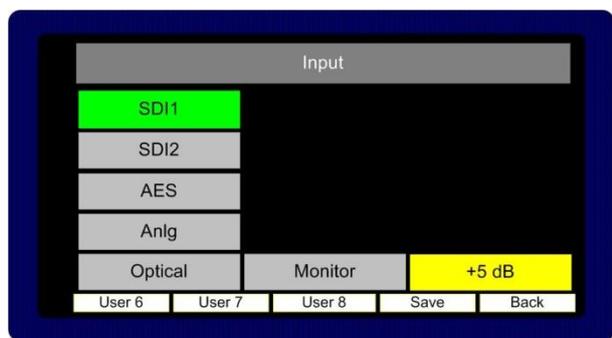
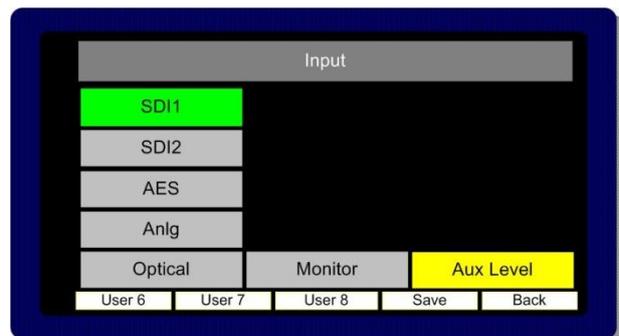
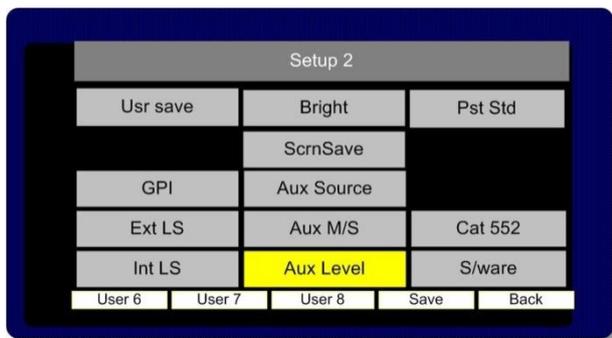
**Aux M/S;** By using the encoder to select the **Aux M/S** button, the user can define whether the selected input is to be mixed within **AIM** as a mono or stereo signal. For example; a mono line level input from a local **Intercom** panel might be connected to the left channel XLR connector of analogue 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.



**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 **PAM2-3G16** 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.



## 2.17 System Upgrade, Default Restore and CAT552 Reset

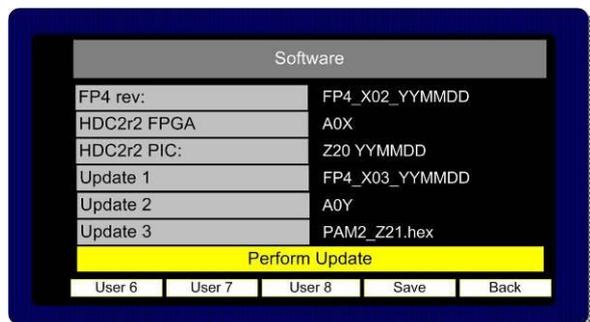
PAM2-3G16 is designed to be field upgradeable. The user will need an SD memory card, a PC and a card reader.

Format the memory card to FAT16 (or FAT as described in Windows XP). **PAM2-3G16** 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.tsl.co.uk](http://www.tsl.co.uk). These files **must** be extracted directly to the memory card in order to construct the correct file structure for upload.

Once the upgrade file is saved, select the **S/Ware** function from the **PAM2-3G16 Setup menu 2** and view the current **FP** (front panel) and **FPGA** software versions.



Press button 10 to exit the **S/Ware** mode and insert the SD card. 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 **OLED** screen displays the progress of each stage of the installation on a bargraph with a scale 0 to 100%. Front Panel (FP4), 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, PAM2-3G16 may have to be returned to TSL headquarters in the UK to have its operating system re-installed.**

**Once the update is complete, power PAM2-3G16 off and then re apply power after 30 seconds to complete the procedure.**

**Note: Buttons 6, 7 and 8 should be pressed (6 and 8 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.**

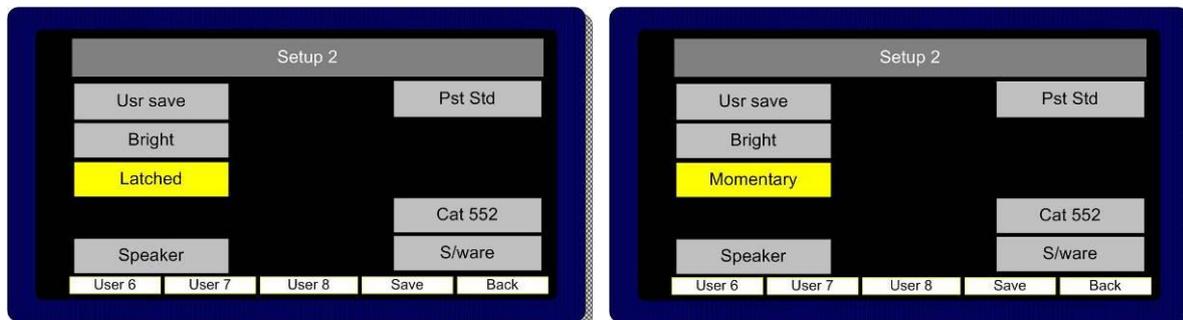
The optional **Dolby CAT552** module can be updated using a similar procedure that described for the **PAM2-3G16** operating system. From **Setup 2** select **Cat 552** to review and update the Dolby card operating software. Features such as Dolby Digital Plus decoding may be added to the **CAT552** 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.



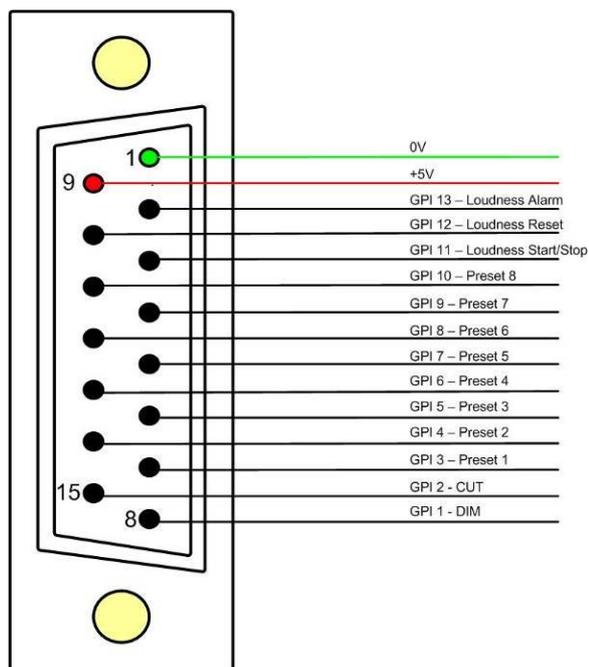
## 2.18 GPI Application

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.



## 2.19 GPI Connectivity

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



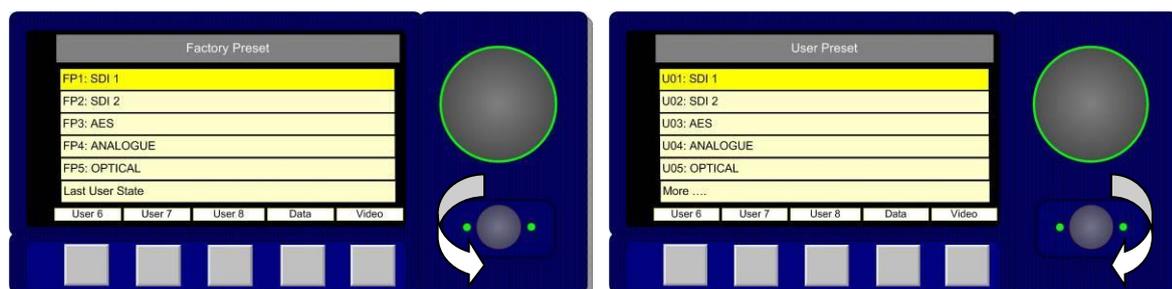
In **Momentary Mode**, **Dim** and **Cut** GPI's latch in a toggle manner, i.e. one closure to ground toggles the function ON; the next ground toggles it OFF. In an ON state; the GPI pin is driven low to allow an LED to be fed from the port. This LED drive is briefly pulsed high at about 100Hz to allow 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 PAM2-3G16 prevents the +5V rail from being shorted but limits the current available.

## 2.20 Preset Memories

PAM2-3G16 features **5 Factory** and **24 User** Programmable Presets, for save and instant recall of default and favourite settings.



### 2.20.1 Factory Presets

Rotating the **Encoder** counter-clockwise activates the **Factory Preset** window in the right hand OLED screen. Highlight and select a preset, **PAM2-3G16** will revert to **Full Screen** mode displaying the selected audio from the preset list shown below.

**Last User State** returns the **PAM2-3G16** to the operational mode last used when the operator entered the **User** or **Factory Preset** condition.

1.	<b>SDI 1</b>
2.	<b>SDI 2</b>
3.	<b>AES</b>
4.	<b>Analogue</b>
5.	<b>Optical</b>
	<b>Last user State</b>

### 2.20.2 User Presets

In any **Main Menu** mode, **Button 9** is illuminated **Green** and can be used to save the current monitoring condition to one of **24 User Presets**. By pressing **Button 9** the **User Preset** list is displayed in the right hand OLED screen. The image below show how a **User Preset** location may be selected from a **Main Menu** screen.



Once the **Green** save button (button 9) is pressed the **User Save** menu is activated. Select and highlight a **User Preset** location from the list **U1** to **U24**.

**Note:** **U1** to **U8** are automatically stored on **Preset Recall** buttons **1-8**.

The text boxes in the **User Save** menu can be used to name a **User Preset**. Up to 18 letters or numbers are available. The example below shows a memory being named as **Dolby**. Selecting and pressing **Done** saves the memory to the chosen location.

**Note;** the text displayed above the **Preset Recall** button will display either the first word or 10 characters of text, whichever is shortest.



To recall a **User Preset** which isn't stored on **Preset Recall** buttons 1-8, rotate the **Encoder** clockwise to activate the **User Preset** window in the right hand OLED screen. Highlight and select to recall a **User Preset** from locations **U01** to **U24**.

### 3.0 **Connectivity and pin-out details**

**PAM2-3G16** 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](http://www.cpc.farnell.com) [www.hosatech.com](http://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 **PAM2-3G16**, 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**

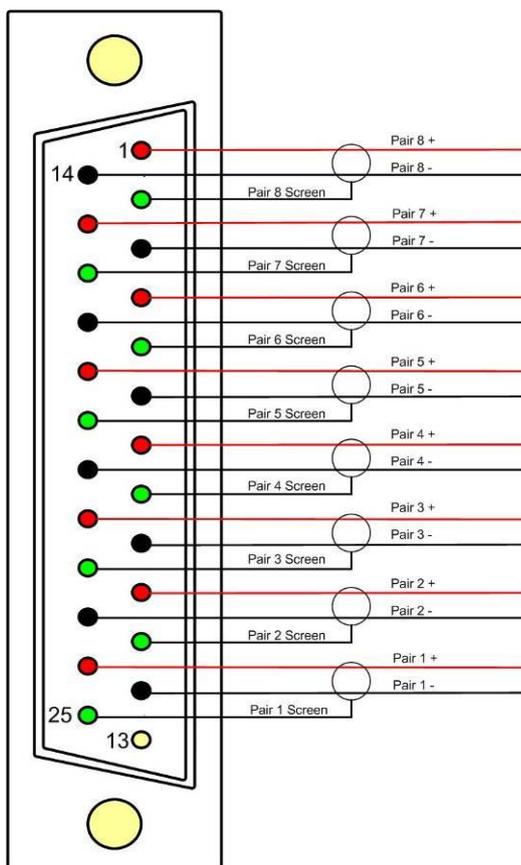
CONN	PIN	FUNCTION
ANALOG 1	1	GND
ANALOG 1	2	1 IN+
ANALOG 1	3	1 IN-
ANALOG 2	1	GND
ANALOG 2	2	2 IN+
ANALOG 2	3	2 IN-

#### 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-

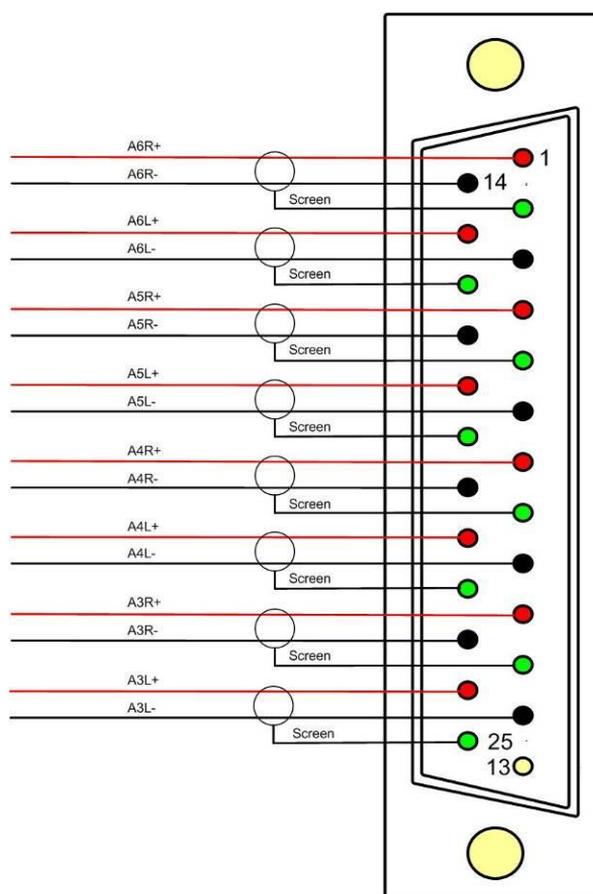
### 3.3 Analogue Output Connector – D25 Socket Pinout on unit, Plug (shown) on mating cable.

D 25 SOCKET ON AMU	AUDIO OUT
PIN NO	FUNCTION
1	A8+ (7.1)
14	A8- (7.1)
2	Ground
15	A7+ (7.1)
3	A7- (7.1)
16	Ground
4	A6+ (RS)
17	A6- (RS)
5	Ground
18	A5+ (LS)
6	A5- (LS)
19	Ground
7	A4+ (LFE)
20	A4- (LFE)
8	Ground
21	A3+ (Centre)
9	A3- (Centre)
22	Ground
10	A2+ (FR)
23	A2- (FR)
11	Ground
24	A1+ (FL)
12	A1- (FL)
25	Ground
13	N/C



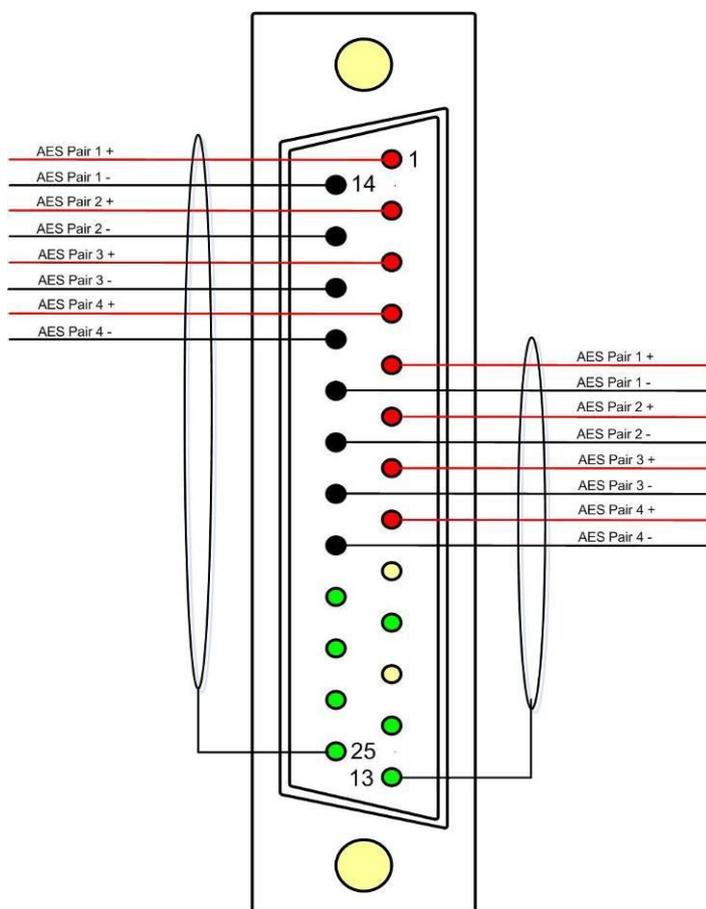
### 3.4 Analogue Input Connector – D25 Socket Pinout on unit, Plug on mating cable.

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 Connectors 1-4 and 5-8 – D25 Socket Pinout, Plug (shown) on mating cable.

D 25 SOCKET ON AMU	AES INPUTS/OUTPUTS
PIN NO	FUNCTION
1	Ch1&2 / 9&10 In 1+
14	Ch1&2 / 9&10 In 1-
2	Ch3&4 / 11&12 In 2+
15	Ch3&4 / 11&12 In 2-
3	Ch5&6 / 13&14 In 3+
16	Ch5&6 / 13&14 In 3-
4	Ch7&8 / 15&16 In 4+
17	Ch7&8 / 15&16 In 4-
5	Ch1&2 / 9&10 Out 1+
18	Ch1&2 / 9&10 Out 1-
6	Ch3&4 / 11&12 Out 2+
19	Ch3&4 / 11&12 Out 2-
7	Ch5&6 / 13&14 Out 3+
20	Ch5&6 / 13&14 Out 3-
8	Ch7&8 / 15&16 Out 4+
21	Ch7&8 / 15&16 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 Sales (+44 1628 676200) and provides BNC Socket to D25 connectivity.

Please note that when using PAM2-3G16 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 GPI Connector – D15 Plug Pinout, Socket on mating cable.

D 15 PLUG ON AMU	GPI
PIN NO	FUNCTION
1	0V
9	+5V
2	GPI 13 – Loud Alarm
10	GPI 12 – Loud Reset
3	GPI 11 – Loud Start/Stop
11	GPI 10 – Preset 8
4	GPI 9 – Preset 7
12	GPI 8 – Preset 6
5	GPI 7 – Preset 5
13	GPI 6 – Preset 4
6	GPI 5 – Preset 3
14	GPI 4 – Preset 2
7	GPI 3 – Preset 1
15	GPI 2 – CUT
8	GPI 1 – DIM

### 3.7 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

### 3.8 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

### 3.9 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.

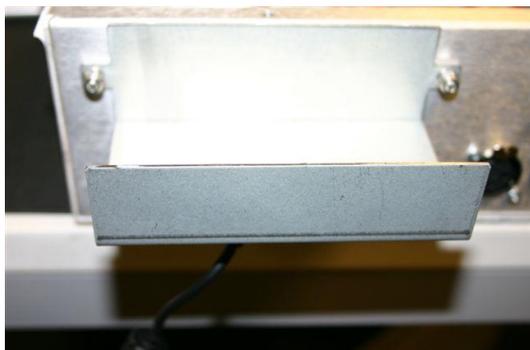
Typical European line up: -18 dBu

Typical American line: -20 dBu

#### 5.0 Power Supply Mounting Bracket

**PAM2-3G16** is powered via a supplied external 12 volt, DC supply. For your convenience a rear panel mounting bracket is also included with the product in order that the power supply can be held in place. Threaded studs are built onto the back panel of **PAM2-3G16** so that the bracket can be attached simply using a pair of 'nyloc' nuts.

Please see the photograph sequence below for details of how to mount the bracket and fit the power supply.



## 6.0 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.

## 7.0 PAM2 - 3G8 Technical Specifications

### Power Supply

Supply Voltage	12V DC
Power Consumption	bc.

### Physical Dimensions

Height	88mm (2RU)
Width	483mm (19")
Depth	320mm
Weight	5100g

### 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 $\Omega$

### 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 $\Omega$
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 $\Omega$
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		
<b>480i/30.00 (SD - NTSC)</b>		
<b>576i/25.00 (SD - PAL)</b>		

**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

## 8.0 Installed HDC-2T Audio Monitor Module Specification (including 'add on' expansion board).

### 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: 75 $\Omega$ , return loss  $\geq$ 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:  $\geq$ 300m of high quality cable at 270Mbit (eg Belden 1694)  
 $\geq$ 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: 75 $\Omega$  unbalanced or 110 $\Omega$  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: 75 $\Omega$   
 Amplitude: 800mV p-p  $\pm$ 10%

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

*Output 3 - Optional* SDI version of image on composite output  
 Impedance: 75 $\Omega$   
 Amplitude: 800mV p-p  $\pm$ 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 $\Omega$  unbalanced or 110 $\Omega$  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 $\Omega$  or >2V into 110 $\Omega$

**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 $\Omega$ ) 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