

# Owner's Manual

**ASP 4816**



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V 1.04

E&OE

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## **Introduction.**

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Thank you for selecting this Audient product.

We hope that your new ASP4816 console will provide an intuitive, ergonomic and tactile path to analogue enlightenment.

ASP4816 is a small format analogue console with a big heart. Small in stature, yet not in functionality, it provides an impressive array of connectivity, signal conditioning, routing, summing, processing and monitoring capability in one beautiful, tidy package.

ASP4816 will adapt to become the centre of your creative workflow, equally suited to the project studio or high-end professional environment.

Lets take a look!

### **How to use this manual.**

This manual has been divided into sections for your convenience.

The first section introduces ASP4816 and provides an overview of the control surface and where to find things. The following sections provide detailed information regarding each function, operation of controls and how they relate to each other. Along the way a number of handy tips and suggestions will offer up possible uses and applications for the console during creative use.

## Unpacking

Your Audient ASP4816 Console has been carefully and meticulously tested and inspected before dispatch.

Please check for any signs of transit damage. If any signs of mishandling are found please notify the carrier and inform your dealer immediately. The packaging should include the console, an IEC power cord and this manual.

### IMPORTANT SAFETY INSTRUCTIONS

Please read all of these instructions and save them for later reference before connecting ASP4816 to the mains and powering up the console. To prevent electrical shock and fire hazard follow all warnings and instructions marked on the ASP4816.

This unit is connected via its IEC power cord to the mains safety earth.

NEVER OPERATE THIS CONSOLE WITH THIS EARTH CONNECTION REMOVED.

## Internal Switch Mode Power Supply & Mains Fuse

ASP4816 utilises an internal switch-mode power supply that is very quiet and passively cooled with plenty of current capability and headroom. This switch-mode design will accept any A.C line voltage from 90V to 264V. Please ensure your A.C mains line voltage is within this specification. Consult a qualified technician if you suspect difficulties.

DO NOT ATTEMPT TO TAMPER WITH THE POWER SUPPLY OR MAINS VOLTAGES  
- HAZARDOUS TO HEALTH.

Always replace the mains fuse with the correct value - T2A slow blow.

## Service and Repair

The console uses a complex internal pcb sandwich arrangement making field service only possible by a qualified technician. If any technical issues do arise with your console, please contact your dealer as soon as possible to arrange for technical support. Do not attempt to fix the console unless qualified to do so. See the warranty section provided at the end of this manual for details of your cover.

However, if you do need to access the inside of the console, the mechanical design allows for easy entry. If you do need to access the inside of the console for any reason please power down the device and disconnect the IEC power cord before proceeding.

All audio path PCBs are panel mounted to the control surface, with the I/O connector PCBs mounted to the rear of the chassis base. The switch mode power supply, and power distribution PCB are mounted to the chassis base. PCBs are linked via ribbon cables or small cable runs. As such the console can be opened without stress or strain on internal wiring or components.

4 x M6 bolts located underneath the armrest allow this to be lifted off. The control surface module is also held in place with 4 M6 bolts located on the underside of the chassis. Once these are removed the control surface module can be hinged up using the hand hold cut-outs in the front surface of the module. (just like a car bonnet!). A 45cm long prop is recommended to support the control-surface module, freeing both hands for whatever issue you may be attempting to solve!

## Ventilation

Care should be taken not to obstruct the series of ventilation holes in the metalwork of the console. The desk is designed to release heat and take adequate air flow via these holes to ensure longevity of performance. These are located underneath the armrest, in the sides of the console and meterbridge.

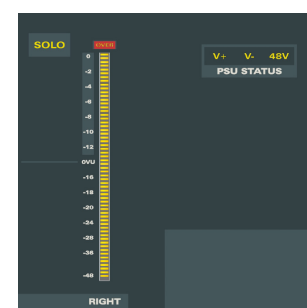
If mounting the console into some form of studio furniture or desk, please ensure a sufficient air gap at these locations.

## Lexan overlay

The front panel of the ASP4816 features a rugged under-surface printed polycarbonate overlay. Exposure to direct sunlight for extended periods of time should be avoided as this can have a detrimental effect on the overlay panel and the control knobs. However at least the panel legending will not wear off like some of the silk screened or engraved products of yesteryear!

## Power-up

On power up, please check that the power rail indicators for +48v d.c and +/-18v d.c light on the right hand side of the console.



## Audio interfaces

The ASP4816 has been designed and developed to provide highly robust system integration interfaces, allowing worry-free system hook-up under the most demanding situations.

Inputs and outputs are implemented using advanced electronically balanced or ground sensing topologies and are fitted with extensive RFI rejection networks. All signal interfaces are also fully protected against accidental misuse e.g. by the connection of phantom powered cables.

## Pin conventions

Signal interfaces are provided on either metal shell locking XLR, TRS Jack or 25 pin D Sub connectors. XLR Pin 1 is connected to the chassis safety ground. XLR Pin 2 is designated as signal positive and Pin 3 as signal negative.

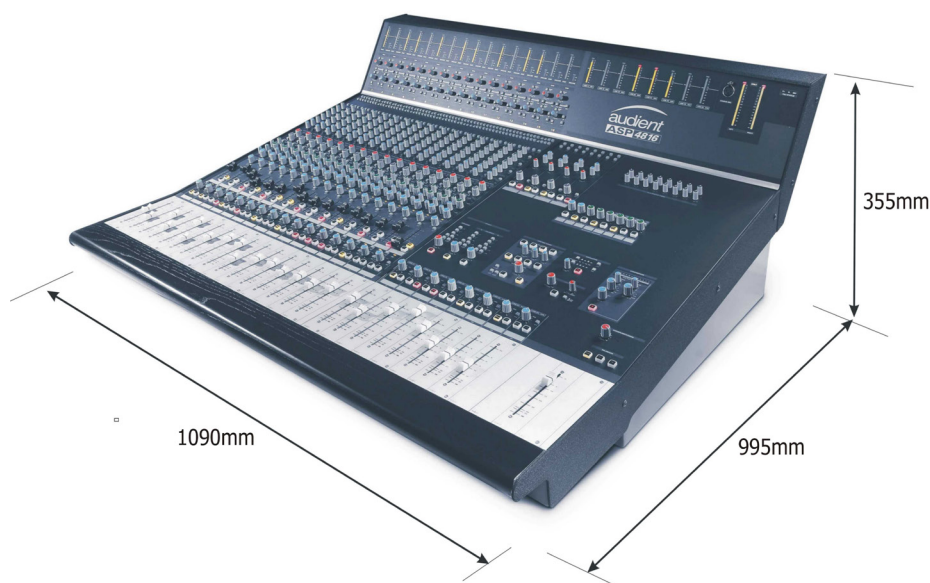
To unbalance the outputs of the ASP4816 Pin 3 should be connected to Pin 1 at the output of the console. Similarly, inputs from unbalanced sources should be connected via twin screened cables with the Pin 3 connection tied to the screen at the unbalanced source.

## Screen connections

In order to maintain optimum EMC performance it is important that screens are properly connected at both ends of cable runs. In this way the electromagnetic shield provided by the equipment chassis and the cable screens will be optimised to reject interference.

It is recommended that only high quality braided screen cables are used to avoid compromising EMC performance.

## Dimensions



## Frequency Response

Mic input to Mix output	<+0,-0.3dB	20Hz-20kHz	@6-40dB gain.
Line input to Mix output	<+0,-0.3dB	20Hz-20kHz	@0dB gain.

## THD AND NOISE AT +20dB OUTPUT

Mic XLR input to any output	<0.005% at 1kHz
Line input to any output	<0.005% at 1kHz
Tape input to any output	< 0.003% at 1kHz

## NOISE

Mic EIN (20-20kHz, 150R source)	<-127.5dBu
Bus noise (no inputs routed)	<-93dBu
Bus noise (36 inputs routed)	<-78dBu

## CROSSTALK AND MUTE ATTENUATION AT 1kHz

Short fader Mute	>90dB
Long fader Mute	>90dB
Mix assign	>90dB
Bus assign	>90dB

MIC CMRR	70dB (Min gain) 75dB (Max gain)
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## MAXIMUM INPUT

Mic	>+21dBu (min gain)
Line	+30dBu (min gain)
insert return	>+21dBu

## MAXIMUM OUTPUT INTO 2K OHMS

Mix output	>+26dBu
Bus output	>+26dBu
Aux output	>+26dBu
Insert send	>+20dBu
Monitor, Studio, F/B outputs	>+20dBu

ASP4816's many features include 16 channels of Class-A discrete microphone preamplifiers as used in our world renowned ASP8024 console and ASP008 8-channel microphone preamplifier, guaranteeing outstanding sonic transparency and detail.

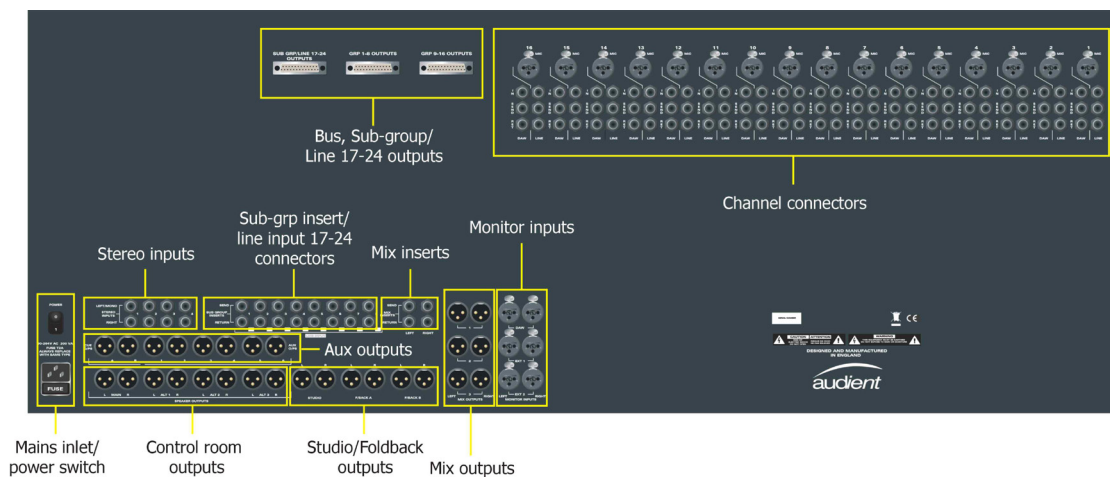
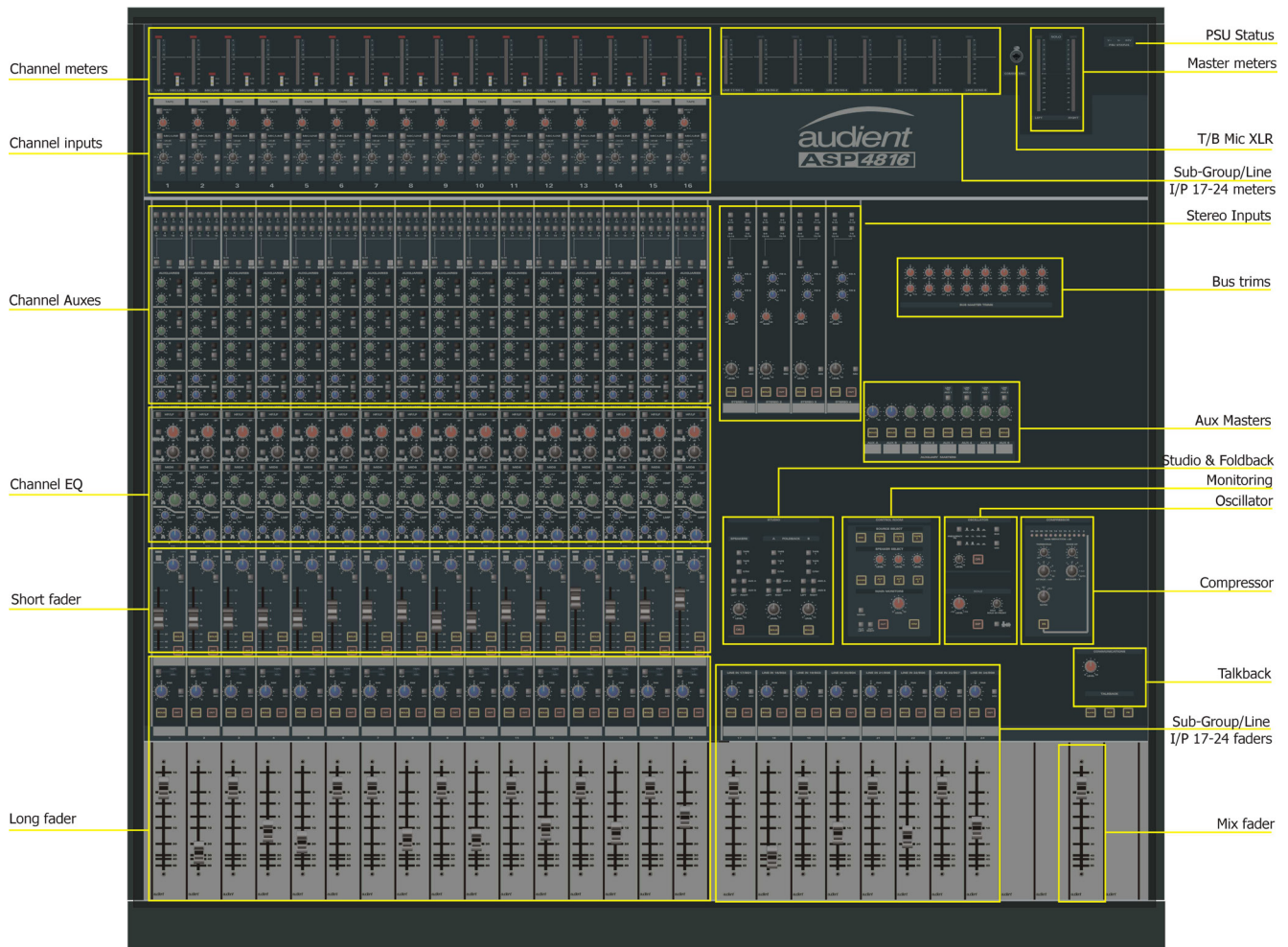
When combined with high resolution bargraph metering, input conditioning, comprehensive cue monitoring options, ASP4816 is right at home in a tracking environment.

A rear panel full of high quality locking XLR and TRS connectors ensures it is simple and quick to integrate your favourite outboard and existing microphone preamplifiers.

However, ASP4816 is equally comfortable as a mixing tool - with up to 6 selectable aux sends, inserts and dedicated DAW returns, 8 sub-groups with switchable inserts, 16 main output buses, classic Audient summing, VCA bus compressor and a high quality centre section complete with mono, dim, phase and comprehensive monitoring configurations - any eventuality is covered.



## Console layout



## In-line architecture

ASP4816 has an In-line architecture. This means that the ‘channel path’ and the ‘monitor path’ are both included in the same physical strip.

Because the two signal paths are integrated in this way, in-line consoles have sometimes been seen as confusing. However great care has been taken in the cosmetic and ergonomic design of the ASP4816 to make the two paths easily distinguishable from one another. Dark areas of the control surface and dark switch buttons are associated with the short fader (SF) or channel path while lighter areas of the control surface are associated with the long fader (LF) or monitoring path.

For the remainder of this manual the signal paths will be referred to as the LF and the SF paths. As a default condition the Mic/Line input feeds the SF path while the DAW return signal uses the LF path.

This assignment can be reversed using the FLIP switch. To identify which mode has been selected back lit legends indicate the signal source for each path.



## Path swapping

As just discussed, the ASP4816’s In-Line architecture provides huge operational flexibility and power.

The block diagram on the next page shows the SF and LF (short and long fader) paths. The FLIP switch is at the input of both paths and determines whether the MIC/LINE or the DAW input is used for a particular path. The normal position for all switches on the console is UP and with the FLIP switch in this position the DAW signal is routed through the LF path as shown.

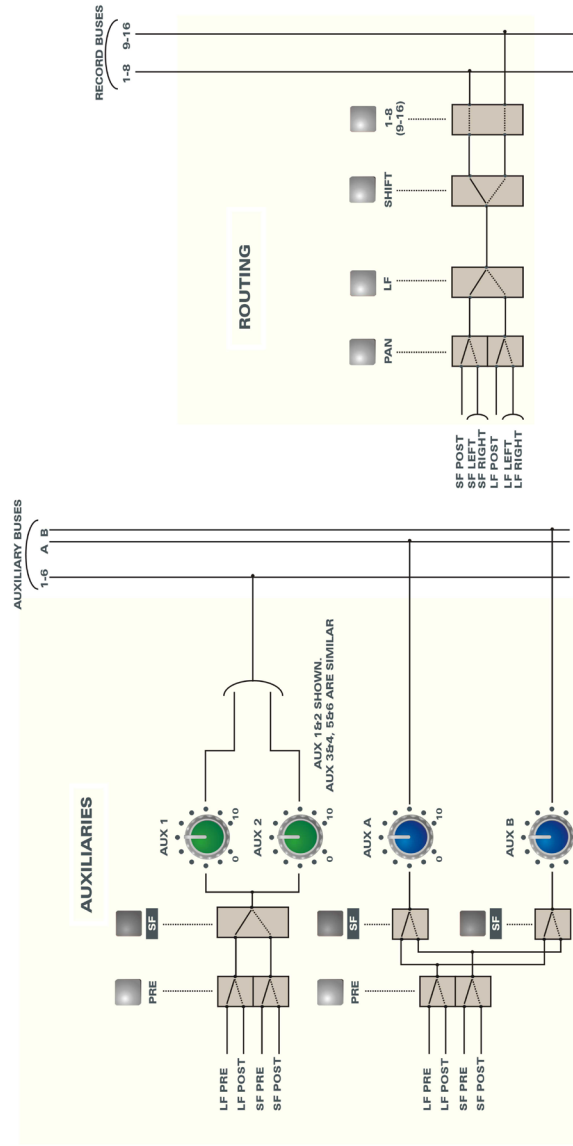
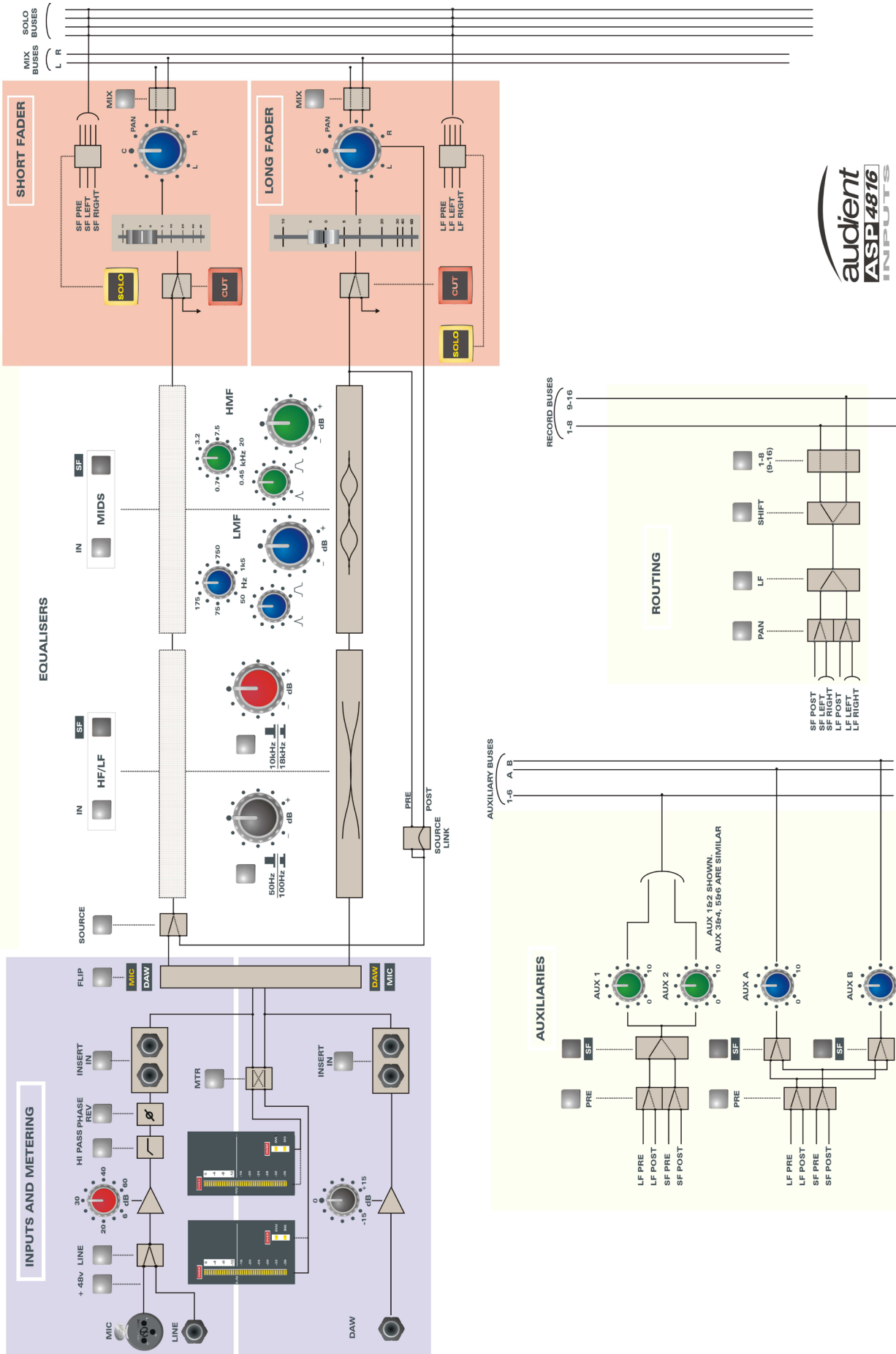
By studying the diagram it can be seen that the MIC/LINE input is routed through the SF path but that when the FLIP switch is pressed this situation is reversed and the DAW signal is now routed through the SF path.

There are switches labelled SF associated with both the HF/LF equaliser and the MID equaliser. Again the normal position of these switches is UP, placing the equalisers in the LF path. By pressing the SF switches the equaliser sections can be moved over to the SF path. Thus it is possible to have the entire equaliser in either the LF or SF path or one section of the equaliser in each path. This situation could be useful when both the LF and SF paths are used for mixdown. If recordings are to be made with equalisation then the equalisers should be placed in the SF path.

Note in this situation that you will hear the effect of the equalisation and it will be recorded. When the equaliser is in the LF path you will hear the changes but they will not be recorded. They will only be recorded if a stereo recording is made, as will be the case when mixing the title.

It is possible to source the auxiliaries from the SF path by pressing the SF switch associated with each pair of auxiliaries. The auxiliaries can further be selected to be either pre or post fader for whichever path they are in.

# Overview



## Solo

Pressing a solo button on a channel with either PFL or AFL selected allows either the PFL or AFL signal for that channel to be heard on the monitors and viewed on the stereo output meters.

Associated with the solo switch is the SOLO-IN-FRONT control and this allows the relative level of the solo'd signal and the stereo mix to be adjusted. It is therefore possible to hear a channel in isolation or with some amount of the mix behind it. Solo-in-Place is an extension of the AFL and PFL facilities. If SIP is selected on the master module the signal on the main stereo bus will be replaced by the AFL (Post Pan) signal of the solo'd channel. This is the equivalent of cutting all other channels except the one you want to check, but is achieved by just a single button push.

Note that if you are recording when Solo in Place is used this will be recorded !

For this reason the SIP switch is illuminated alerting users to possible danger. The most likely use for SIP will be just before a mix when equalisation is being set up. It is often easier to adjust the equalisation on a solo'd channel but it is essential to check how it sounds when in the mix.

## Metering

Metering is extremely important as it can determine whether or not a signal is too low in level, in which case it may be noisy, or too high in level in which case it may suffer distortion. The aim of the meters is to assist in setting the signal level between the two extremes of noise and distortion.

A signal level that is too high clips meaning that the smooth waveform abruptly changes when the electronics runs out of headroom. Normally a visual indication of impending clipping is given before the audible effects become apparent.

At the other extreme, if there is insufficient signal, any noise present will be amplified along with the signal when it is eventually brought up to the correct level. There are no indicators warning you of this condition - only the fact that the signal is not "peaking" to any extent on the meter.

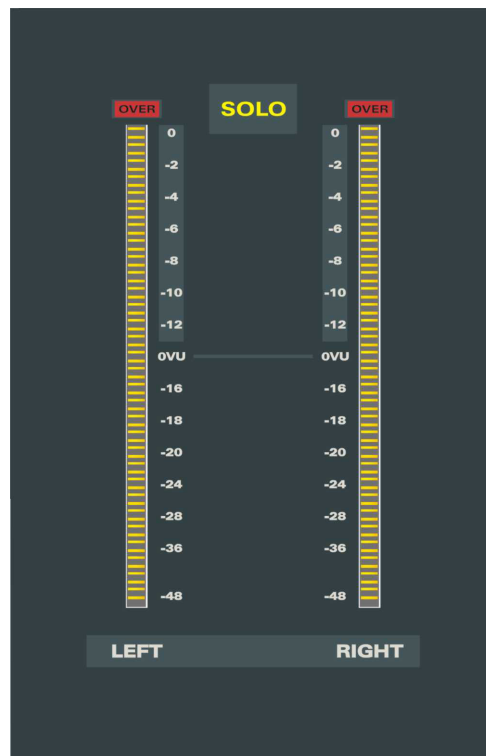
To aid this situation the ASP4816 has input and output meters. DAW input levels can be monitored on a 20 segment meter, while microphone input levels can be monitored on a 3 segment meter. Provision is made for the function of these meters to be reversed, as, particularly when tracking, it may be desirable to observe the input signal on the large meter.

## *Metering - continued*

0VU is calibrated to a nominal operating level of +4dBu. Which in turn is relative to a typical EBU calibration of +18dBu = 0 dBFS. 0 dBFS is your absolute maximum digital level (full scale) and should be avoided at all costs - digital clipping is not a pleasant sound.

As such the bargraph meter scale on ASP4816 is calibrated in dBFS, such that +4 dBu = 0VU = -14 dBFS. Therefore, a nominal operating level of 0VU will result in 14 dB of headroom in your digital recording platform (providing your AD/DA converters are aligned to 0 dBFS = +18 dBu).

14 dB of headroom is very useful as most music sources (except for the huge dynamic range of an orchestra) have crest factors (the difference between average (RMS) and peak level) of around 12-20 dB. ASP4816 therefore provides ample headroom for your beautifully crisp transient sources.



### **Meters are provided for :-**

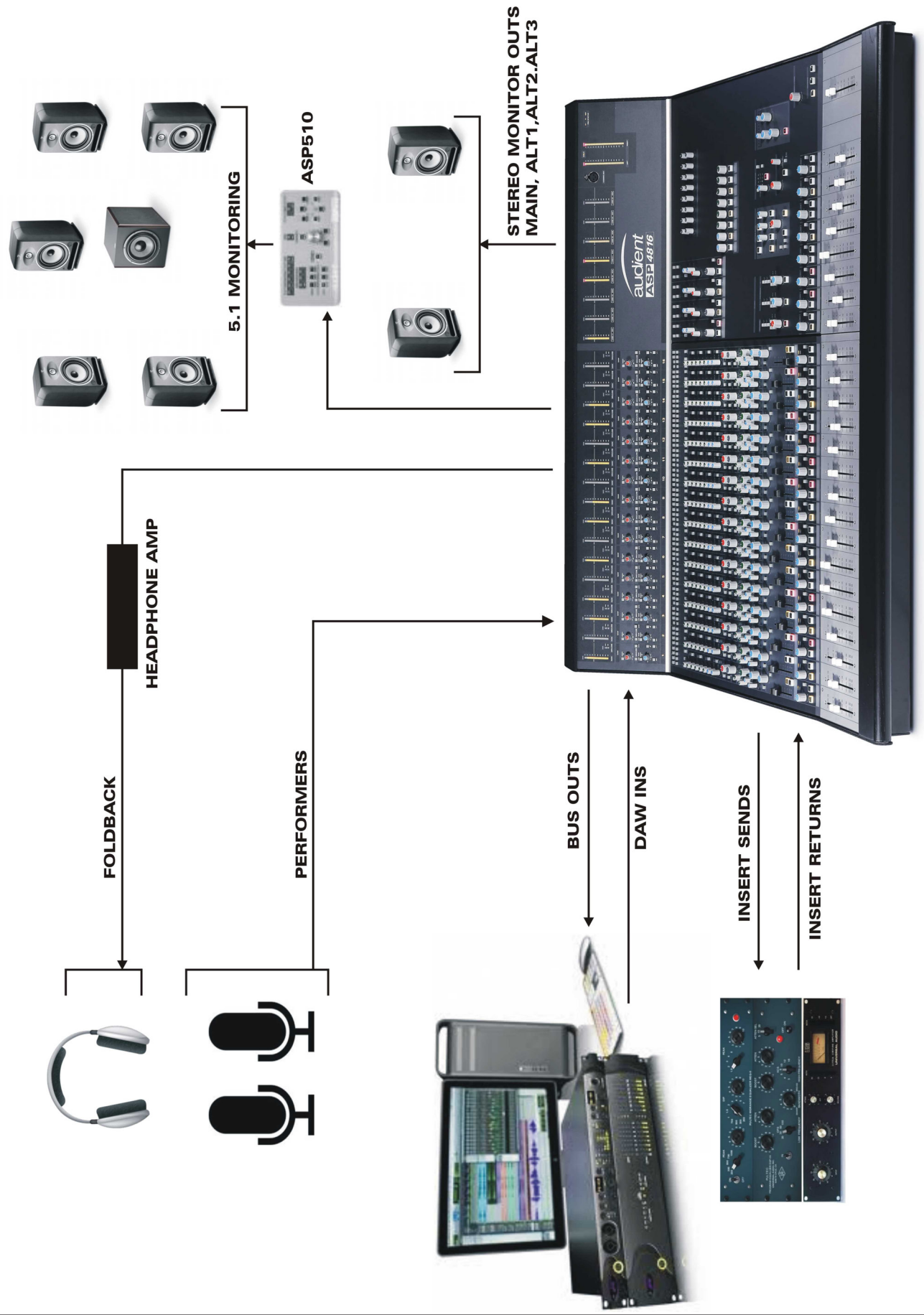
Main Stereo Output (30 segment)

Line Inputs 17-24/Sub-group Outputs (20 segment)

DAW Inputs (20 segment)

MIC/LINE Inputs (3 segments)

# Typical configuration



## **AFL**

This allows after fade (post fade) signals to be heard on the monitors and viewed on the main stereo meters.

## **Auxiliaries**

Sometimes known as auxiliary sends these are used as secondary mix buses. The mixes created on these buses are then used to feed effect units or are fed back to the performers as a Foldback feed. Every channel has access to the auxiliary mixes and the contribution of any channel can be varied by using the appropriate auxiliary level control. Auxiliaries can be switched pre or post fader and be allocated to either the SF or LF signal paths.

## **Auxiliary Master**

Associated with every auxiliary there is an auxiliary master control to give overall level control rather than having to adjust the contribution from every channel.

## **Bus Master**

Inputs are assigned to bus outputs through a routing matrix, enabling one or many inputs to be assigned to the same bus. The bus outputs usually correspond to inputs on the DAW or other recording device. Each bus then has a mixing amplifier whose gain can be controlled by the Bus Master Trim. This allows the level to a multi-track input to be raised or lowered without having to adjust the individual level of each channel routed to that track. The faders can still be used to adjust the relative levels of channels.

## **Cut**

The cut or mute control is used to silence (mute) a signal path when it is not in use. This removes the noise contribution from that source leading to a quieter mix. There are cut switches in both the SF and LF signal paths.

## **Dim**

This allows the control room loudspeaker levels to be reduced by a preset amount. Dim will be brought into action automatically when talkback is used preventing howl round.

## **Equaliser**

Equalisers are what would be referred to as tone controls on consumer equipment. Equalisers are divided into a number of bands - 4 in this case. There is scope to adjust high and low frequencies and two bands of middle (mid) frequencies. The high and low frequency sections are shelving and the turnover frequency is switchable. The middle frequency sections are peaking and the frequency of the peak (or dip) is adjustable. It is also possible to alter the Q of the mid sections with a pot, making the Q continuously variable between two values. In-line consoles often have the facility for the equaliser to be split such that it can be used partly in the channel path and partly in the monitor path. The SF switches on ASP4816 equalisers allow the HF/LF and MID equalisers to be independently switched into the SF signal path.

## **Flip**

Flip allows the inputs to the signal paths to be swapped. Normally the LF path will carry the DAW input, however, with FLIP pressed it will carry the MIC/LINE input while the DAW input will travel through the SF path.

## **Foldback**

Foldback is a mix that is returned to the performers in the studio in order that they can play in time with what is already recorded. It could simply be the console stereo output although more usually it is taken from a pair of auxiliary buses allowing a different mix to be created. Talkback may also be included on the foldback outputs enabling communication with the artists.

## **In-line**

This refers to a type of console which contains two signal paths within a channel strip. The channel signal path is used to feed a DAW or other recording device while the monitor path is used to carry the output of the DAW (or other recording device) through to the stereo mix bus. In-line consoles can be more compact than split consoles or can carry more channels for a given size. The possibility of switching signals between the two paths and of sharing facilities between the paths makes the in-line concept a very attractive one.

## **Insert points**

Insert points allow the signal path to be broken allowing the insertion of some signal processing device. The device inserted is then in series with the signal path. When not required the device can either be switched out using the INSERT switch or unplugged from the console.

## **LF**

The Long fader is normally used to feed the mix. The lighter areas of the channel strips are areas used for the LF signal path. There may be a dark switch labelled SF allowing that facility to be switched into the SF or short fader path.

## **Line**

The line input is a high level, high input impedance input intended for high level sources such as the outputs of a DAW or other recording device, sampler etc.

## **Mic**

The microphone input is a low level, low impedance input intended for use by low output devices such as microphones. This contrasts with the line input which is intended for use by equipment with high output levels, such as a DAW line output for example.

## **Mix**

This allows signal to be routed to the stereo mix bus which is the main output of the console. This routing can be applied to both the SF and LF signal paths and is particularly useful during mix down when as many inputs as possible are often required.

## **Pan**

Short for panoramic potentiometer this control places a mono source signal onto the stereo bus. The proportion of signal fed to the left and right buses is variable (using the pan control) and alters the spatial position of an instrument within the mix. Thus a number of channels can all be panned to different spatial positions. Generally low frequency instruments such as kick drums are panned centrally as they are omnidirectional and for a given SPL the speakers are being driven at a lower level leading to less distortion. Signals can also be panned across odd and even bus outputs allowing them to be recorded in stereo.

## **Q**

Q is an indication of the frequency range or bandwidth over which a peaking equaliser will be effective. Low Qs affect a wide range of frequencies while high Qs affect a much narrower range of frequencies. It thus allows an equalisation adjustment to be targeted to maximise the effect where required while at the same time minimising changes where they are not wanted. Generally high Qs sound less pleasant than low Qs.

## **Routing**

This is the process of selecting to which bus output of the console the signal should be routed. Routing can be to multiple tracks and if an odd/even combination is selected then panning can be used to record a stereo signal.

## **SF**

The Short fader is normally used to feed the inputs to a DAW or other recording device. It therefore controls the recorded level of the signal. When SF appears by a control or group of controls it means that these functions can be switched into the short fader (or channel fader) path.

## **Shift**

This allows the number of routing switches to be reduced by doubling the function of each switch. With Shift unpressed routing is possible to tracks 1 through 8. With Shift pressed routing to tracks 9 through 16 is possible.

## **Solo in place**

This is a method of previewing the signal in a channel and works by cutting all the signals feeding the stereo bus other than the one(s) being solo'd. This is a destructive process and does affect the stereo or mix output of the console.

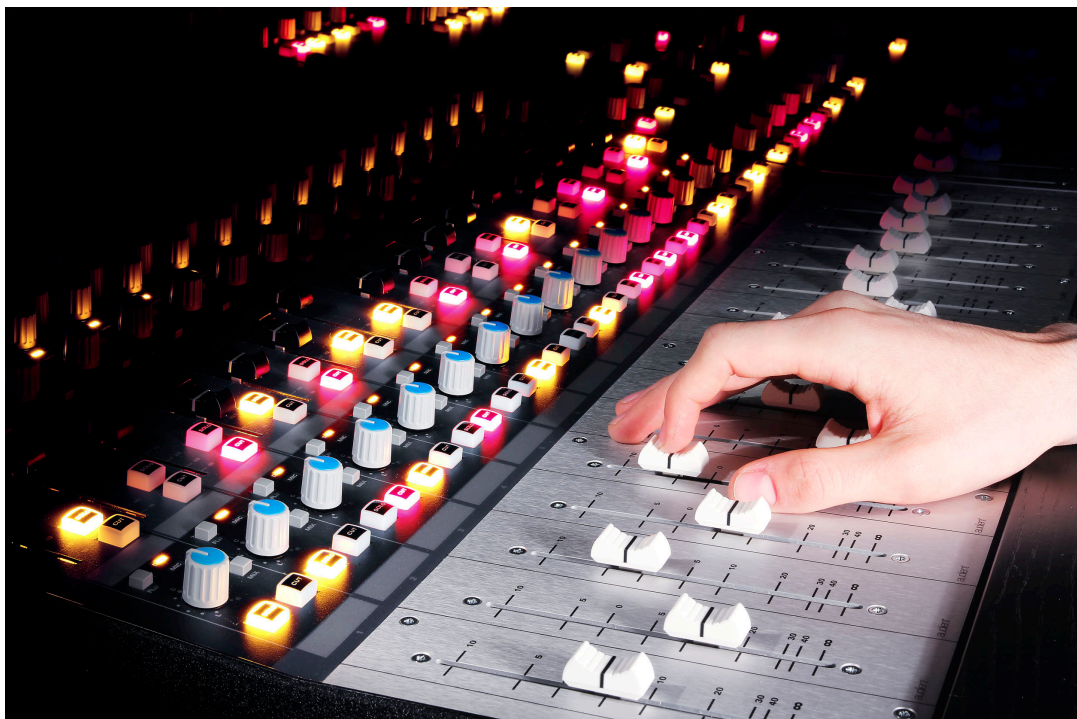
## **Stereo Bus**

This is usually the main bus in the console and provides the output to whatever stereo recording device is in use. The stereo output is also used as the main monitor source allowing the output of the multi-track to be heard and the balance of the individual tracks in the mix to be adjusted.

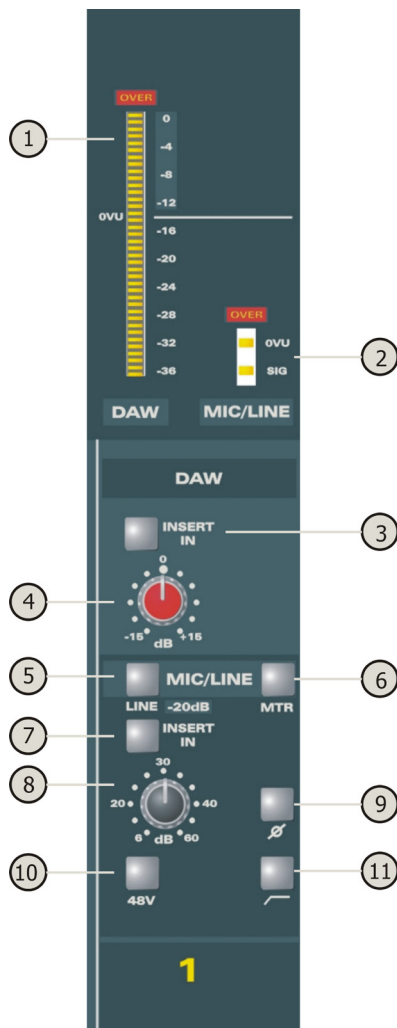
## **Talkback**

Talkback is a means of communication from the mixing console to the performer.

## Console functions.



## CHANNEL STRIP



- |   |  |
|---|--|
| 1 20 Segment meter (normally monitoring the DAW input)  | 7 INSERT IN - places an insert point in the MIC/LINE path.   |
| 2 3 Segment meter (normally monitoring the Mic/Line input).                                       | 8 Mic/Line GAIN Control  |
| 3 INSERT IN places an insert point in the DAW path  | 9 $\emptyset$ Polarity (Phase) Reverse Switch  |
| 4 DAW Input TRIM control  | 10 48V Phantom Power Switch. Turn the loudspeakers down before switching this on or off!   |
| 5 MIC/LINE switch - press down to select the line input.  | 11 High Pass Filter Switch. A high pass filter can be used to get rid of any unwanted low frequencies that may be present such as air conditioning rumble. |
| 6 MTR - press to show the mic/line input on the large meter and the DAW input on the small meter. |  |

## INPUT POD

The input pod is the gateway to the remainder of the signal processing of the console. There are three inputs, a mutually exclusive microphone and line input, and a tape input.

Note the different colouring used to identify the different signal paths of the pod. Anything on a light background is associated with the LF path while anything on a dark background is associated with the SF path.

The mic/line input normally sends signal to the SF or short fader signal path of the console while the DAW input normally sends signal through the LF or long fader path of the console unless this is reversed by the FLIP switch.

There are two meters associated with the inputs, a 20 segment 0dBFS peak reading meter showing the DAW input signal and a 3 segment peak reading meter gives an indication of the mic/line level.

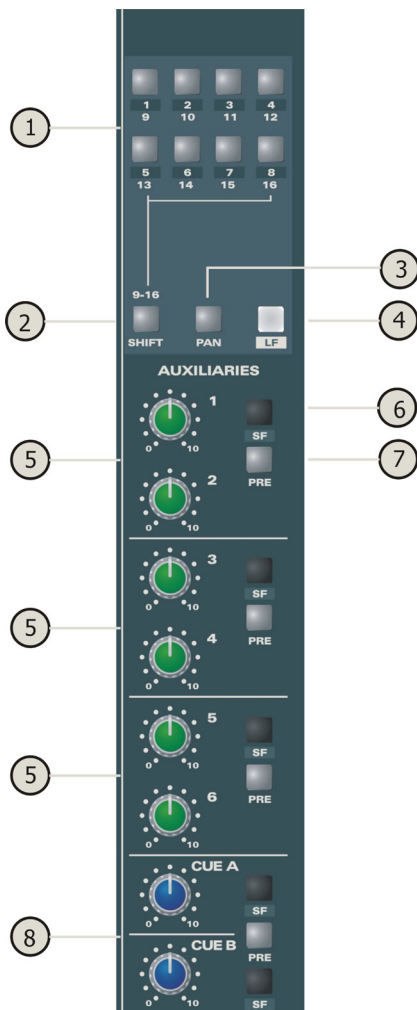
Both the MIC/LINE and the DAW inputs have insert points which can be switched in and out of circuit.

The Mic Input has a gain control range of +6dB to +60dB and the Line Input has a range of -14dB to +40dB. Switches allow for Phantom Power, Polarity Reversal and High Pass filtering - enabling rumble and sources alignment issues to be easily dealt with.

The DAW input has a trim control with a range of  $\pm 15$ dB.

There is a back illuminated number at the bottom of the pod for channel identification.

## CHANNEL STRIP



- |  |  |
|--|--|
| <p>1 ROUTING SWITCHES select the bus outputs that the SF signal is sent to. The first eight switches also route to the 8 sub-groups</p> <p>2 SHIFT gives access to bus outputs 9-16.</p> <p>3 PAN allows the signal to be panned across odd and even buses.</p> <p>4 LF replaces the SF signal being sent to the routing switches with the signal from the LF signal path.</p> | <p>5 LEVEL adjusts the level sent to an auxiliary output.</p> <p>6 SF allows the signal feeding the auxiliary to be taken from the SF signal path.</p> <p>7 PRE allows the auxiliary signal to be taken pre fader instead of the normal situation where it is taken post fader.</p> <p>8 CUE A and B are identical in operation to the others but can be individually switched into the SF path.</p> |
|--|--|

## ROUTING and AUXILIARIES

The routing section takes the signal from the SF path and routes it to the bus outputs which in turn are usually connected to the inputs of a DAW or other recording device. Buses 1 to 8 also have a parallel path and feed the 8 sub-groups. These can be used to pre mix channels together for final mix down or as sends to a DAW or other recording device.

There are 16 bus outputs accessed by 8 routing buttons and a SHIFT button.

Routing can be to multiple outputs and if odd and even outputs are selected panning can be used to place the signal within a stereo image.

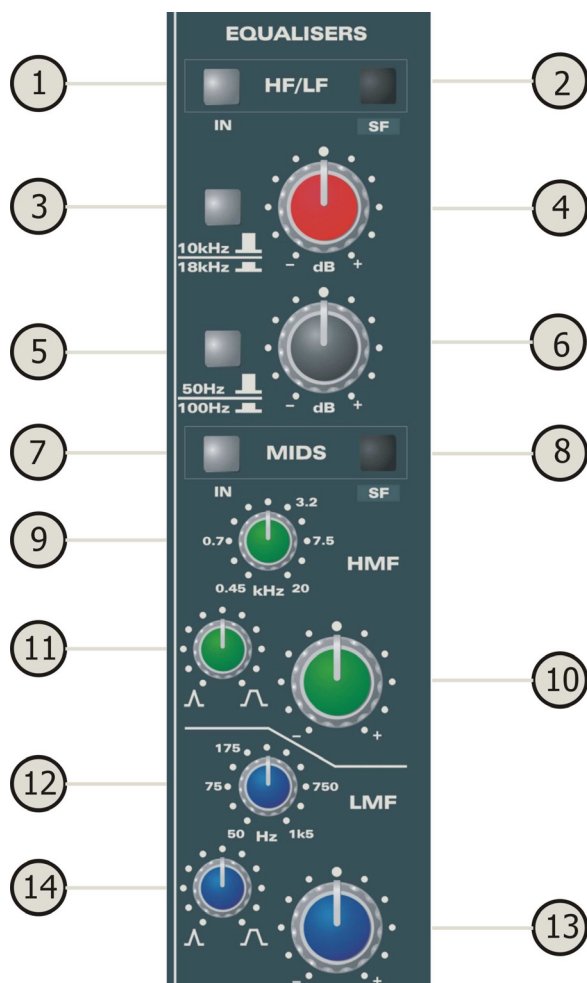
With multiple buses available it is possible to achieve a variation on the multi-bus compression techniques made popular by mix engineers like Michael Brauer. (See [www.mbrauer.com/articles](http://www.mbrauer.com/articles)).

The LF switch moves the routing from the SF path over to the LF path and can be useful for track bouncing. If PAN is down and routing is again selected for odd and even buses then the post fade post pan LF signal is sent to the bus outputs selected.

There are 8 auxiliary outputs. The Auxiliaries can be switched to the SF path.

Cue A and B work in the same manner as the other auxiliaries but are intended mainly for use as sends to the FOLDBACK system.

## CHANNEL STRIP



- |   |  |
|---|--|
| 1 HF/LF Section. This places the HF/LF equaliser in circuit.  | 8 SF places the MIDS equaliser in the SF signal path where it can be used to treat a signal before being recorded. |
| 2 SF places the HF/LF equaliser in the SF signal path where it can be used to treat a signal before it is recorded. | 9 This controls the centre FREQUENCY of the high mid equaliser.  |
| 3 10kHz/18kHz allows the frequency of the HF equaliser to be selected.  | 10 HIGH MID boost/cut control.   |
| 4 HF boost/cut control.   | 11 HIGH MID Q control.   |
| 5 50Hz/100Hz allows the frequency of the LF equaliser to be selected.   | 12 This controls the centre FREQUENCY of the low mid equaliser.  |
| 6 LF boost/cut control.   | 13 LOW MID boost/cut control.  |
| 7 MIDS Section. This places the MIDS equaliser in circuit.  | 14 LOW MID Q control.  |

## EQUALISER

Note the dark background indicating that the equaliser is normally associated with the LF signal path.

The equaliser is split into two sections - one for high and low frequencies (HF/LF) and the other for middle frequencies (MIDS). Both sections can be switched in and out independently and switched into the SF signal path independently.

The HF section has a boost/cut range of 15dB. The frequency can be switched - 10kHz with the switch up and 18kHz with the switch depressed.

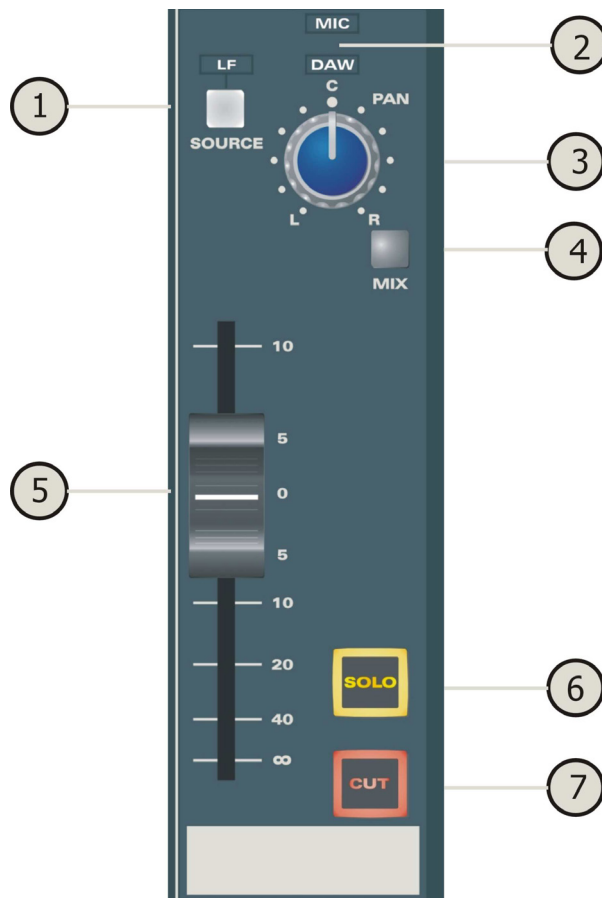
The LF section has a boost/cut range of 15dB and a shelving characteristic. The frequency can be switched - 50Hz with the switch up and 100Hz with the switch depressed.

The high mid section (HMF) has three controls. The boost/cut range is 15dB and the frequency of operation can be varied from 450Hz to 20kHz. A Q control allows for a very sharp and narrow characteristic or for a more gentle characteristic covering a wider range of frequencies.

The low mid section (LMF) has three controls. The boost/cut range is 15dB, and the frequency of operation can be varied from 50Hz to 1.5kHz. A Q control allows for a very sharp and narrow characteristic or for a more gentle characteristic covering a wider range of frequencies.

The actual Q range is between 3.8 (0.4 Octave) and 0.65 (2 octaves).

## CHANNEL STRIP



- |  |  |
|--|--|
| <p>1 SOURCE selects the post fade LF signal as the input to the SF signal path, over-riding the MIC or DAW selection from the Flip switch. By altering a link on the circuit board this signal can be made pre fade. The LF legend will illuminate when the switch is pressed and the MIC/DAW indicator will blank.</p> <p>2 MIC/DAW. These back lit indicators show whether the MIC/LINE or the DAW input is selected to the SF path. Only one will be illuminated and it can be changed by using the FLIP switch.</p> <p>3 This is the PAN control for the SF signal enabling it to be panned across odd and even buses.</p> | <p>4 MIX routes the SF signal to the stereo mix bus. It is good practice to unrout any channels which are not needed. This will reduce mix amp noise.</p> <p>5 This is the SHORT FADER which controls the level of the SF signal.</p> <p>6 SOLO allows the SF signal to be heard on the monitors and viewed on the master meters. If Solo In Place is selected it will replace the console output.</p> <p>7 CUT allows the SF signal path to be muted. This may help to reduce noise in a mix if a channel is not in use for a period of time.</p> |
|--|--|

### SHORT FADER (SF)

There are 3 back illuminated indicators showing the selected input to the SF path. This can be changed between Mic/Line and DAW using the FLIP switch.

LF Source allows the source for the SF path to be taken from the LF path. This could be used during mixdown to send the LF signal through the SF path up to the routing matrix where the bus outputs can be used as additional effect sends. Normally this signal is derived after the long fader (POST LF) but it can be made PRE LF by changing an internal link on the circuit board.

The source switching occurs before the equaliser so it is possible to equalise the SF signal which has been taken from the LF path by switching either one or both equaliser sections into the SF path. (If this is done then the equaliser section is no longer available in the LF path).

The SF PAN control allows panning across the bus outputs when pan is selected on the routing section of the strip.

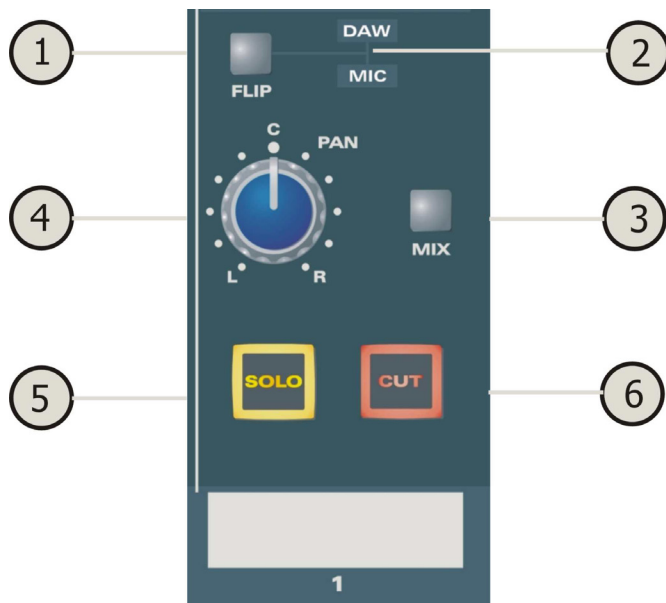
MIX allows the SF signal to be routed to the stereo mix bus and allows the SF path to be used as an additional input during mixdown.

The short fader is designed for use with 10dB of gain in hand allowing the signal to be boosted or reduced in level if required.

The SOLO switch allows the channel to be auditioned through the AFL/PFL or Solo in Place facilities.

CUT allows the SF signal path to be muted.

## CHANNEL STRIP



- 1 FLIP swaps the inputs between the LF and SF paths.
- 2 These back lit indicators show whether the MIC/LINE or the DAW input is selected to the LF path. Only one will be illuminated and it can be changed by using the FLIP switch.
- 3 MIX allows the LF signal to be routed to the stereo mix bus. Normally this switch should be pressed but it is good practice to unroute any channels which are not needed. this will help to reduce mix amp noise.
- 4 This is the PAN control for the LF signal enabling it to be panned across the stereo bus
- 5 SOLO allows the LF signal to be heard on the monitors and viewed on the master meters or, if Solo in Place is selected, it will replace the console output.
- 6 CUT allows the LF signal path to be muted. This may help to reduce the noise in a mix when a channel is not in use for a period of time.

### FLIP and PAN

The FLIP switch allows the MIC/LINE input and the DAW input to be reversed. Normally the DAW input feeds the LF signal path - with FLIP pressed it will feed the SF signal path. The MIC/LINE input normally feeds the SF signal path and with FLIP pressed it will feed the LF signal path. Illuminated indicators in each section show which input is selected to the LF and SF paths.

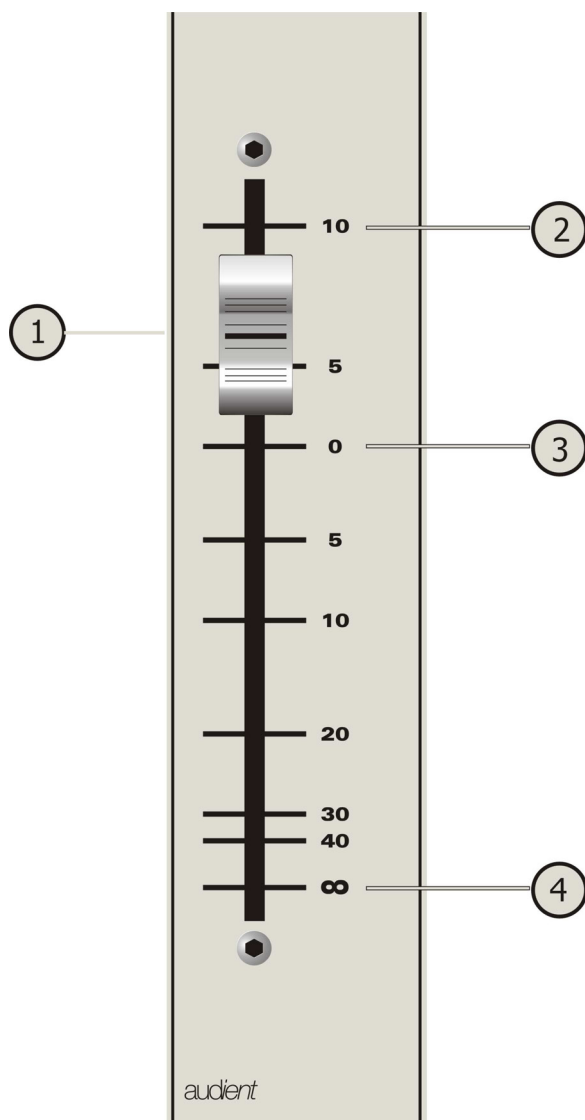
The PAN control pans the LF signal across the stereo mix bus and the MIX switch assigns the LF signal to the stereo bus.

The SOLO switch allows the long fader signal to be auditioned through the AFL/PFL or Solo in Place facilities.

CUT allows the LF signal path to be muted.

Page 33 gives more detail regarding the long fader used to control the level of the LF signal path.

## CHANNEL STRIP



### LONG FADER (LF)

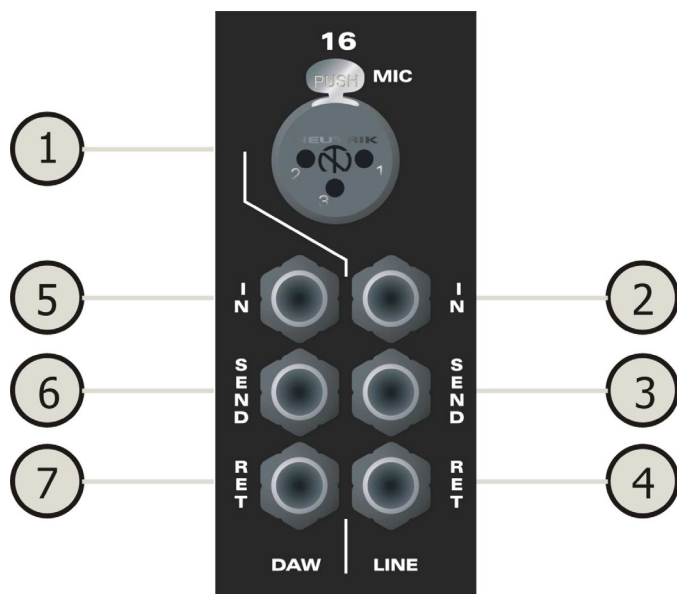
Located close to the operator this fader operates on the LF signal path and is therefore mainly used for creating the monitor mix and the final stereo mix for the title.

The fader is expected to operate around the 0dB mark with 10dB of gain in hand allowing the signal to be increased or decreased in level.

When level setting start with the fader in this position then adjust the input sensitivity control for the correct level to optimise the gain structure.

- |  |  |
|--|--|
| 1 This is the LONG FADER which controls the level of the LF signal.  | 3 This is the 0dB mark. This is the normal operation position for the fader. |
| 2 This is the +10dB mark. The fader is fully open at this point and introducing 10dB of gain into the signal path. | 4 With the fader here the signal path is closed and no signal will pass.     |

## CHANNEL STRIP



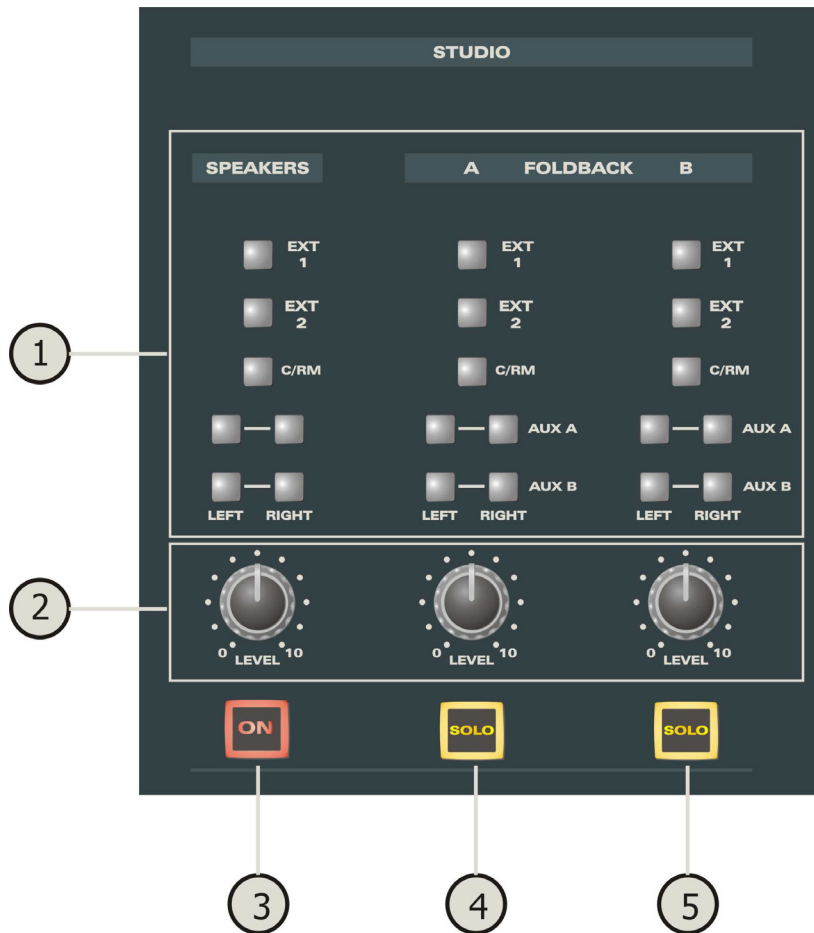
- 1 MICROPHONE INPUT. Microphones or other low levels inputs can be connected to this input.
- 2 LINE INPUT. This input can be selected in place of the microphone input. Like the DAW input it is suitable for high level line sources.
- 3 MIC/LINE INSERT SEND . This is the insert point send output for the microphone and line input. Signal is always present here and can be used as an additional output. Only the Mic/Line signal will appear here and it is not affected by the Flip switch.
- 4 MIC/LINE INSERT RETURN. When the insert point is in use the signal from the external processing equipment should be connected here.
- 5 DAW INPUT. The output of the DAW should be connected here although the input is suitable for any high level line source.
- 6 DAW INSERT SEND. This is the insert point send output for the DAW input. Signal is always present here and this can be used as an additional output. Only the DAW signal will appear here and it is not affected by the Flip switch.
- 7 DAW INSERT RETURN. When the insert point is in use the signal from the external processing equipment should be connected here.

## INPUT CONNECTORS

The rear mounted connector panel is where the input, output and insert point connectors are located.

The microphone input uses an XLR connector while the line input, DAW input and the insert sends and returns use Tip, Ring and Sleeve jacks.

## MASTERS



- 1 SOURCE SELECT. These select the signals to be sent to the studio loudspeakers and foldback.
- 2 The LEVEL controls set the Studio Loudspeaker and Foldback levels.
- 3 Studio Loudspeaker ON switch. This activates the Studio Loudspeaker.

- 4 FOLDBACK 1 SOLO switch. Allows foldback 1 to be checked on the control room monitoring system.
- 5 FOLDBACK 2 SOLO switch. Allows foldback 2 to be checked on the control room monitoring system.

## STUDIO SPEAKERS and FOLDBACK

This section of the master module looks after the STUDIO LOUDSPEAKER and FOLDBACK outputs of the console. In every case the same sources are available -

EXT 1

EXT 2

CONTROL ROOM (C/RM)

AUXILIARY A (AUX A)

AUXILIARY B (AUX B)

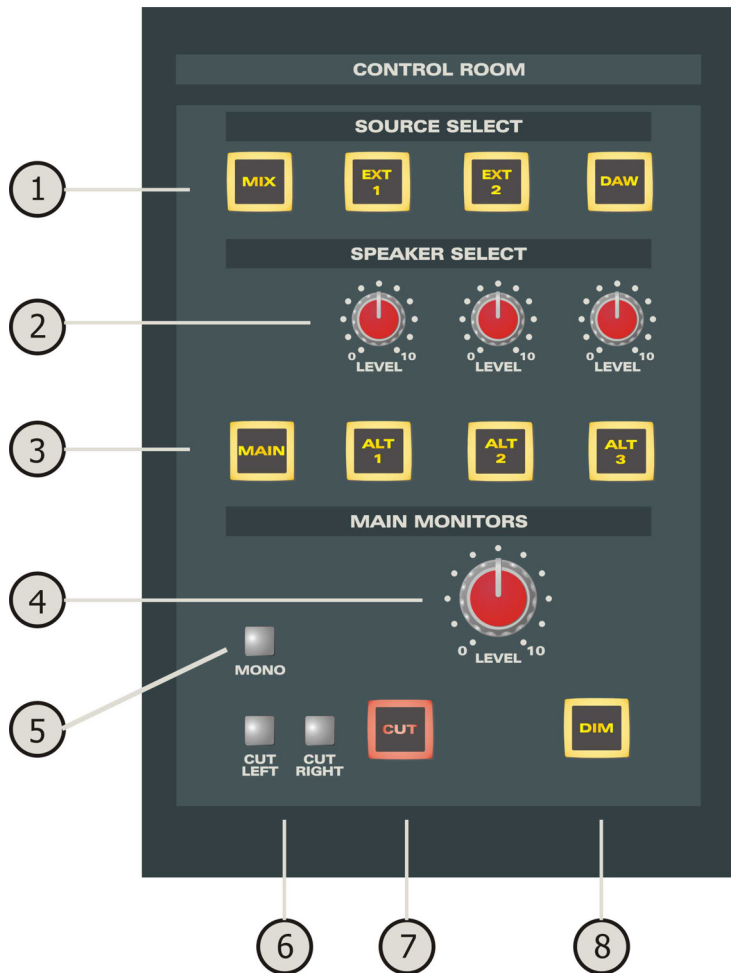
EXT 1 and 2 are external inputs to the console and could be anything you care to plug in.

C/RM selects the control room monitor output as the source. This is used to provide a playback of the MIX to the studio speakers, or to set up a quick foldback mix, perhaps with a little help from the Aux A and Aux B buses.

CUE A and CUE B are mono sources which can be independently selected to the left and right sides of the studio speaker or foldback outputs.

Each output has a level control. The studio loudspeaker has a ON switch and the FOLDBACK outputs have SOLO switches. These enable the foldback mix to be created or checked in the control room. This is always AFL.

## MASTERS



- |   |   |
|---|---|
| 1 SOURCE SELECT buttons.  | 6 CUT buttons allowing independent cutting of the left and right speakers.  |
| 2 LEVEL controls for the alternate loudspeakers.                        | 7 DIM allows the speakers to be reduced in level by a preset amount (15dB). |
| 3 SPEAKER SELECT buttons.   | 8 CUT allows the speakers to be completely silenced.                        |
| 4 Main Loudspeaker LEVEL Control.                                       |   |
| 5 MONO button allowing the signal to be checked for mono compatibility. |   |

## CONTROL ROOM MONITORING

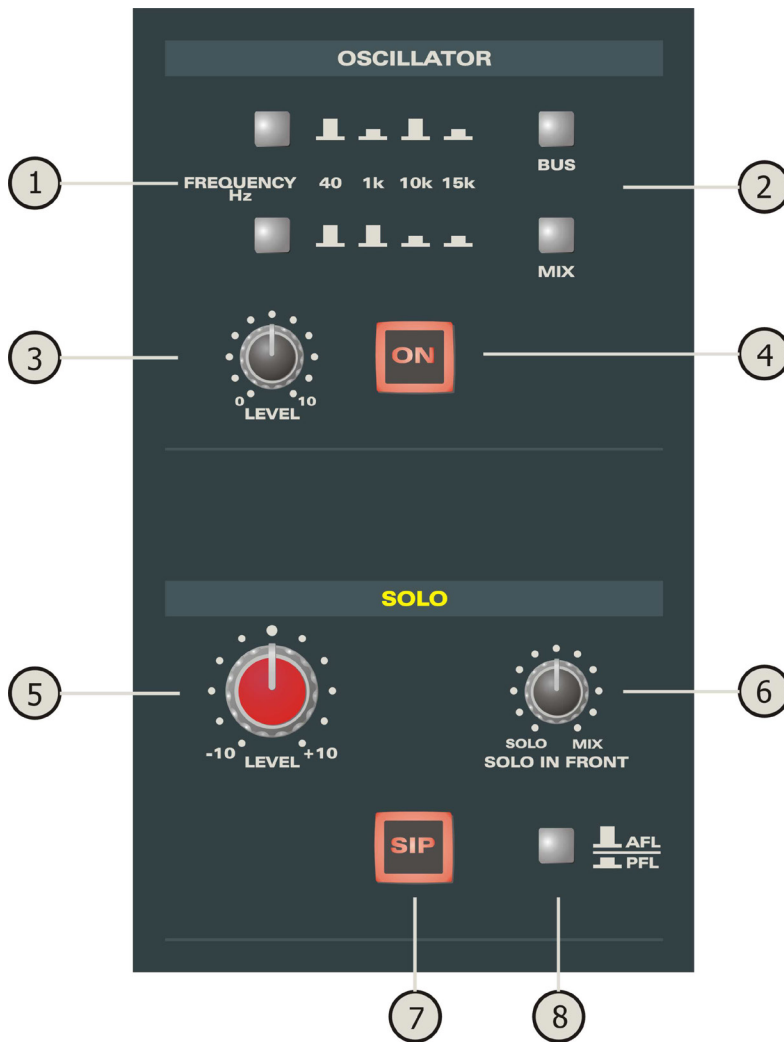
The Control Room Monitoring system is one of the most used systems in a studio. Typically there are several sets of loudspeakers with a main pair and one or more alternate pairs. These are sometimes referred to as Near, Far and Mid field monitors depending on their proximity to the listening position.

The ASP4816 provides for 4 pairs of loudspeakers, a MAIN pair and ALTERNATE 1, 2 and 3. Source selection is made from either the MIX or DAW, EXT 1 and 2.

Each loudspeaker pair has its own level control. The control for the main pair is larger than the controls for the alternate monitors. The level controls are independent of each other.

Associated with the main loudspeakers are some further controls allowing the left and right outputs to be independently or simultaneously CUT, the monitoring system to be placed in MONO mode, or for the output to be dimmed by a preset amount.

## MASTERS



- |  |  |
|--|--|
| <p>1 Oscillator FREQUENCY selection switches. Frequencies of 40Hz, 1kHz, 10kHz or 15kHz can be selected using the two switches.</p> <p>2 Oscillator ROUTING switches. These route the oscillator signal directly to either the buses 1-16 or the stereo mix bus.</p> <p>3 Oscillator LEVEL Control.</p> <p>4 Oscillator ON switch. During normal operation of the console ensure that the oscillator is switched OFF.</p> <p>5 SOLO level control.</p> | <p>6 SOLO IN FRONT alters the ratio of the SOLO signal to the MIX signal making it possible to listen to the SOLO'd signal in isolation or combined with some amount of the stereo mix.</p> <p>7 The SIP switch switches the console into SOLO IN PLACE mode. This means that the solo signal will replace the mix signal at the stereo output of the console.</p> <p>8 This selects either PFL or AFL mode. This switch is not effective in SOLO IN PLACE mode.</p> |
|--|--|

## OSCILLATOR

The 4 frequency oscillator can be assigned either to buses 1-16 or to the stereo mix bus. The level is adjustable and when not in use the oscillator should be completely turned off.

## SOLO

This is the master control area for the AFL/PFL and SOLO IN PLACE system. If SIP is not illuminated then the console will be in either AFL or PFL mode depending upon the PFL/AFL switch.

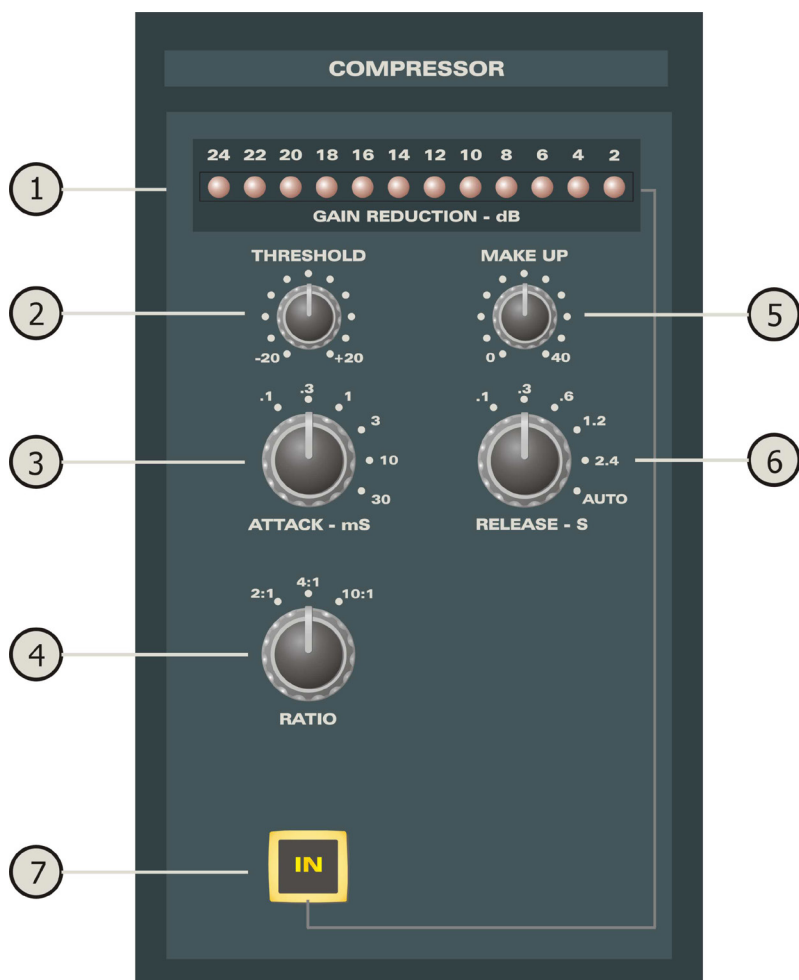
PFL allows pre fade signals to be auditioned on the monitors and viewed on the master meters by pressing the SOLO button on the appropriate channel strip.

AFL allows the post fade post pan signals to be auditioned on the monitors and viewed on the master meters by pressing the SOLO button on the appropriate channel strip.

A control adjusts the relative level of the solo signal and the mix signal allowing the solo'd channel to be heard in isolation or with some amount of the mix.

SIP mode allows the post pan signal to be auditioned by switching the solo'd channels onto the console output and muting the stereo mix for the duration of the solo. Since this happens on the mix output, solo in place should not be used during mixing but is very helpful when setting up the mix.

## MASTERS



- 1 A 12 segment bargraph METER indicating the amount of gain reduction taking place.
- 2 THRESHOLD control for determining the level where compression starts. Threshold can be set between -20dBu and +20dBu.
- 3 ATTACK controls the speed with which the compressor reacts. Attack is adjustable in 6 steps from 0.1 to 30ms.
- 4 RATIO determines the amount of compression used once the signal is above the threshold level. For example a ratio of 2:1 means that the output will rise 1dB for every 2dB rise in input level.
- 5 MAKEUP is a gain control. When a signal is compressed the level is reduced. A gain make up stage restores the peak level although of course the dynamic range of the signal has now been reduced. 40dB of gain is available.
- 6 RELEASE controls the speed at which the compressor allows the gain to return to normal when the signal drops below the threshold level. Release is available from 0.1s to 2.4s in 5 steps with an additional AUTO setting.
- 7 The IN switch puts the compressor into circuit.

## COMPRESSOR

A compressor can be switched into the main stereo signal path when required. Note that it is located after the mix insert point but before the main fader.

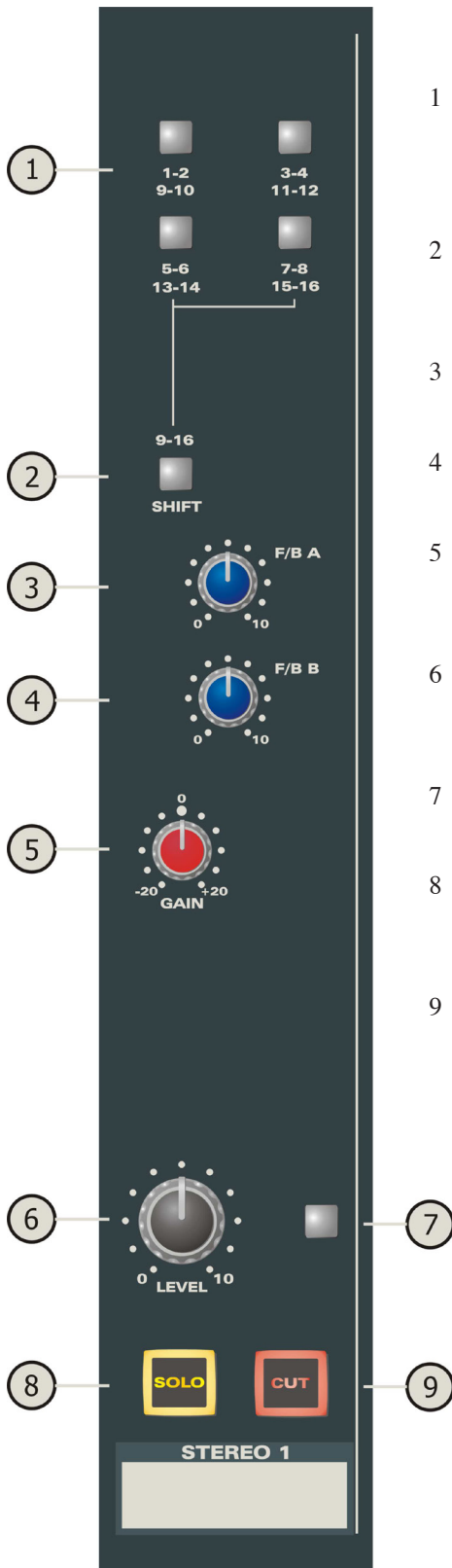
The compressor characteristics are optimised for use in mix processing while many of the parameters remain under the control of the engineer.

Threshold, Gain Make Up, Attack, Release and Ratio are all adjustable while the entire processor can be switched out of circuit when not required.

A bargraph meter indicates the amount of compression applied to the signal. When compression is taking place this should show some very healthy activity. Please do not try to over compress the mix output as the results will sound terrible. Always use your ears to check the effect that you are trying for!

A good use of the compressor may be to limit peak signals by setting a highish threshold with a high ratio. Thus when the threshold is reached the signal is barely allowed to increase beyond it.

## MASTERS



- 1 ROUTING SWITCHES allowing the input signal to be sent to the DAW or other recording device.
- 2 SHIFT button working with the routing switches to give access to tracks 9-16.
- 3 F/B A allows the input signal to be fed to Foldback A.
- 4 F/B B allows the input signal to be fed to Foldback B.
- 5 TRIM control to compensate for different input levels. This has a range of  $\pm 20$ dB.
- 6 Rotary FADER allowing the input signal to be faded up or down as required.
- 7 MIX allows the input signal to be sent to the stereo mix bus.
- 8 SOLO allows the input signal to be auditioned depending upon the selected solo mode.
- 9 CUT is used to mute the signal or simply remove any noise contribution from the return

## STEREO INPUTS

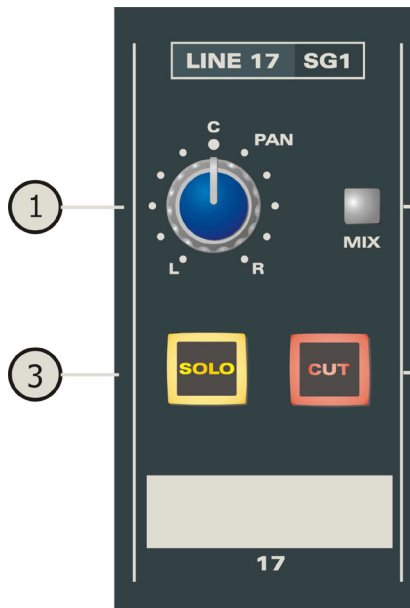
The Stereo inputs allow signals to be brought back into the console (returned) from an effect unit without using up a complete channel strip. Often stereo signals can only be routed to the stereo mix, however, the ASP4816 stereo inputs allow routing back to the multi-track in addition to the mix. This means that effects can be recorded on the multi-track.

Routing to the foldback system is also possible using F/B A and F/B B allowing performers to hear any reverberation or other effect.

A gain trim with 20 dB of range, rotary fader, solo and cut switch complete the facilities on the input.

The stereo channels effectively increase the number of inputs on the console that are available especially during a mix down.

## MASTERS



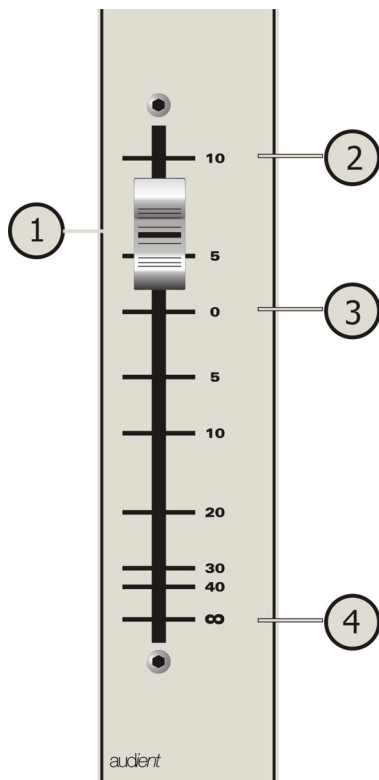
- 1 The PAN knob pans the sub-group signal across the stereo bus.
- 2 MIX allows the sub-group to be assigned to the stereo mix bus and hence the main console output.
- 3 SOLO allows the sub-group signal to be auditioned depending upon the selected pan mode.
- 4 CUT mutes the sub-group output.

### LINE I/Ps 17-14 and SUB-GRP OUTPUTS.

In addition to routing signal to the bus outputs for recording, the first eight buses are also fed to 8 sub-group outputs. Each sub-group has a PAN control, SOLO and CUT switches and a fader controlling the bus output level. The sub-groups have insert points located on the rear panel of the console allowing external processing to be patched in when called for. The MIX switch assigns the sub-group to the stereo mix bus and allows the sub-groups to be used in mix down.

The sub-group INSERT RETURNS are also used as Line inputs 17-24.

### LINE I/Ps 17-14 and SUB-GRP FADERS.



- 1 The SUB-GROUP FADER sets the level of the sub-group output signal.
- 2 This is the +10dB mark. The fader is fully open at this point and introducing 10dB of gain into the signal path.
- 3 This is the 0dB mark which is the normal operation position for the fader.
- 4 With the fader here the signal path is closed and no signal will pass.

The 8 sub-group/Line input 17-24 faders are located in the centre of the console, close to the operator, and control the sub-group output levels. These outputs are directly available at the rear of the console or they may be used to feed the stereo mix bus simply by pressing the mix buttons.

## MASTERS



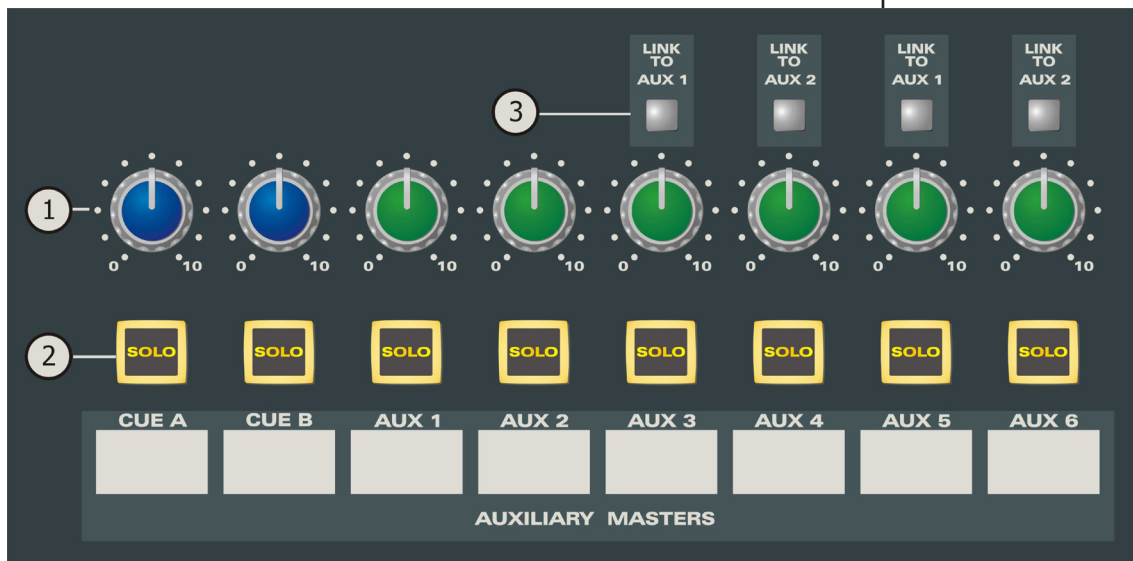
- 1 The LEVEL control adjusts the talkback level.
- 2 Pressing SLATE allows talkback to the bus outputs. This allows track identification information, for example, to be recorded.
- 3 Pressing SLS lets you talk to the studio loudspeaker.
- 4 Pressing FB lets you talk to the Foldback outputs.  
  
In all cases pressing a talkback key will cause the control room monitors to dim. This helps prevent feedback, makes the talkback more intelligible and does not affect the console outputs.

## TALKBACK

Talkback is used to communicate with the STUDIO, the FOLDBACK system or the Bus outputs of the console. Note that the talkback to the foldback system will work even when the foldback levels are turned down. The talkback microphone may be phantom powered by connecting Link 1 on the PC10801 board.

## MASTERS

## AUX MASTERS



- 1 AUXILIARY MASTER LEVEL controls the overall auxiliary output.
- 2 SOLO button used to audition the auxiliary output. This is always AFL.
- 3 The LINK button is used when it is required to combine signals from different auxiliaries into a common signal. The leftmost auxiliary becomes the overall master, thus if auxiliary 3 is linked to auxiliary 1 then the output of auxiliary 1 should be used as the master output.

The auxiliary masters control the overall level of the auxiliary outputs. A balance or mix can thus be created by using the controls on the channel strips and the overall level adjusted by using the auxiliary master control. The auxiliary outputs can be solo'd so that a balance can be created by listening to the output.

A typical mixing situation may require that the LF and SF paths feed the same effect device.

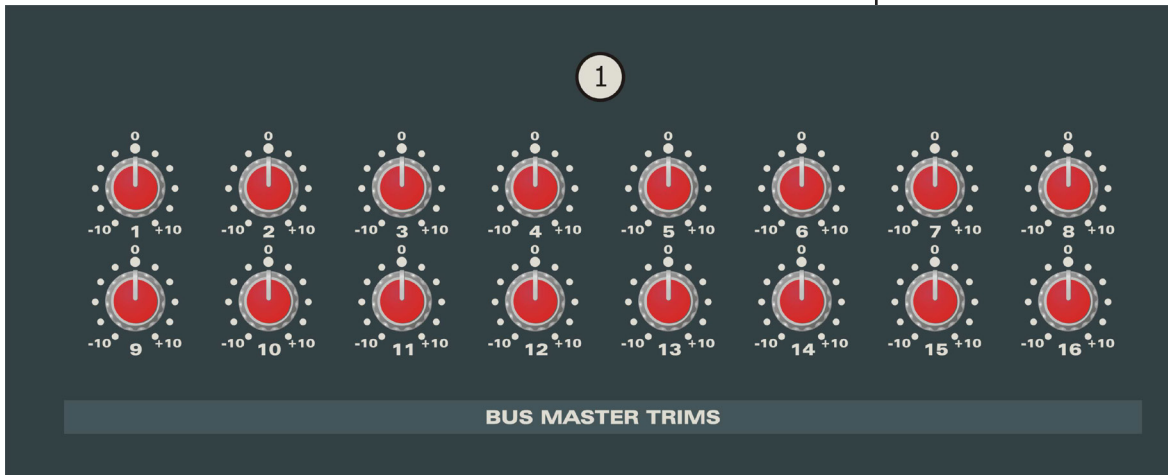
Although an auxiliary send can be assigned to either of the two paths they naturally cannot access both paths at the same time. By linking auxiliaries this problem is overcome and if the auxiliary 3 signals need to be combined with those on Auxiliary 1 this can be achieved by simply using the link facility. The following combinations are possible:

AUXs 3 and 5 can be linked to AUX 1.

AUX s 4 and 6 can be linked to AUX 2.

## MASTERS

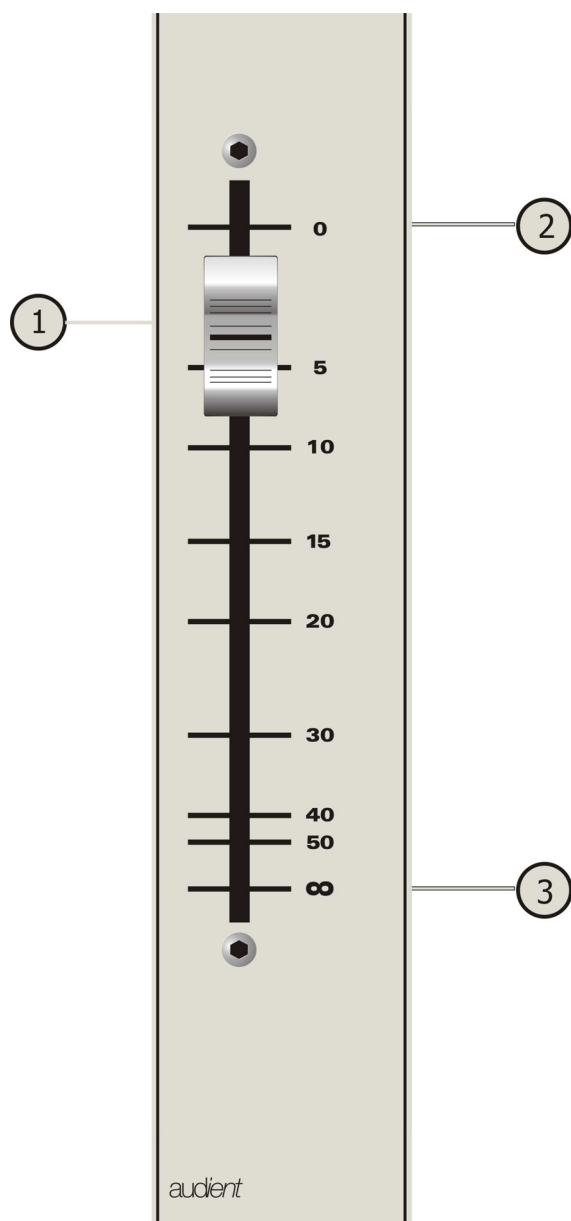
### BUS MASTER TRIM



- 1 BUS MASTER TRIM controls the overall level of a bus output.

The BUS MASER TRIMS are the final stage of level control over the signals routed to the bus outputs. Each bus output has a trim with a range of  $\pm 10\text{dB}$ .

## MASTERS



- 1 The MASTER FADER controls the level of the stereo output signal.
- 2 This is the 0db mark. The fader is fully open at this point.
- 3 With the fader here the signal path is closed and no signal will pass.

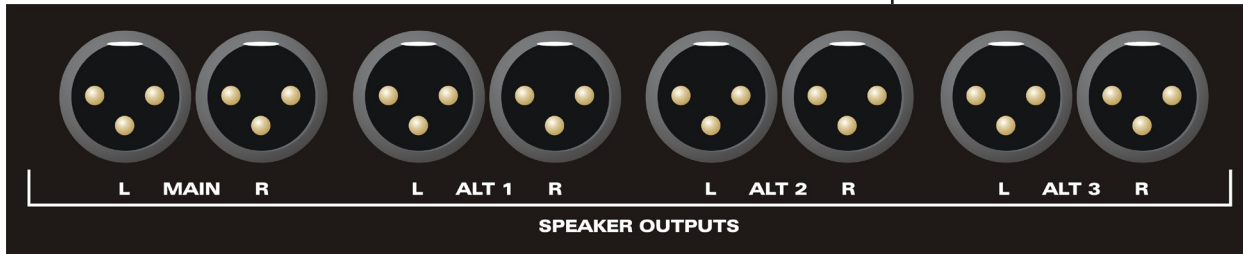
### MASTER FADER

The master fader is used to control the stereo output of the console. Unlike the channel faders it is calibrated with the 0dB mark at the top as the main purpose of this fader is to create a fade out at the end of a title.

Under normal operating conditions the fader should always be set at maximum. If it has to be pulled back a significant distance it indicates that the levels to the mix bus are too high and should be reduced.

## MASTERS

### CONTROL ROOM CONNECTORS



- 1 MAIN Control room loudspeaker output.
- 2 ALTERNATIVE 1 loudspeaker Output.
- 3 ALTERNATIVE 2 loudspeaker Output.
- 4 ALTERNATIVE 3 loudspeaker

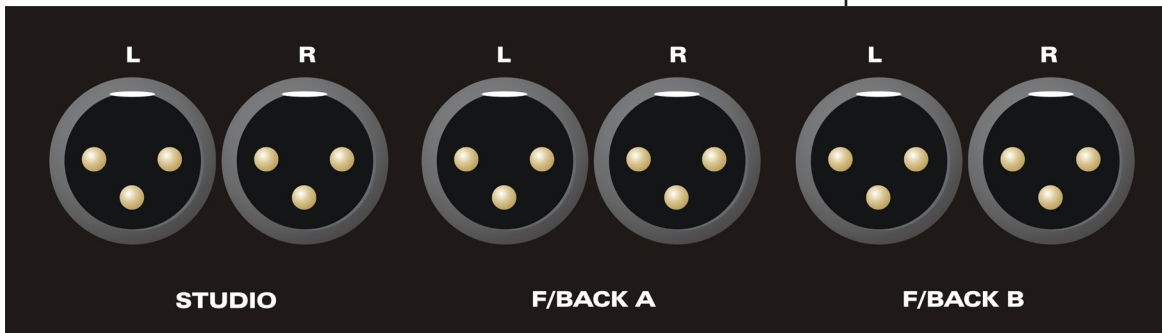
These are the connectors for the control room loudspeakers. There is a main output and 3 alternative outputs which can be selected from the control surface of the console. Each output is stereo, having a left and right connector.

The console does not contain any power amplification and these outputs should be connected into a suitable power amplifier for the loudspeakers in use.

The speakers may of course be self powered in which case connection should be made to the inputs on the loudspeakers.

## MASTERS

### STUDIO/ FOLDBACK CONNECTORS



- 1 **STUDIO LOUDSPEAKER .** These carry the outputs to the studio loudspeaker system. The output depends upon the selection made on the control surface of the console.
- 2 **FOLDBACK A output.** Foldback is normally used as a feed to headphones. The output depends upon the selection made on the control surface of the console.
- 3 **FOLDBACK B output.** Foldback is normally used as a feed to headphones. The output depends upon the selection made on the control surface of the console.

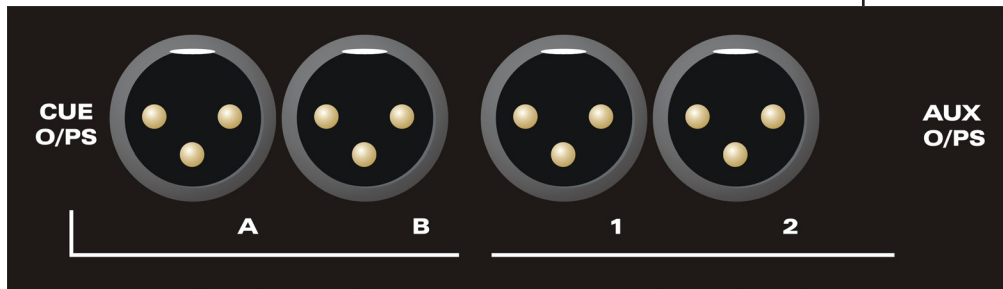
These are the XLR connectors for the Studio Loudspeaker and the two foldback outputs. Each output is stereo, having a left and right connector.

The console does not contain any power amplification and these outputs should be connected into a suitable power amplifier for the loudspeakers or headphones in use.

The speakers may of course be self powered in which case connection should be made to the inputs on the loudspeakers.

## MASTERS

### AUX CONNECTORS



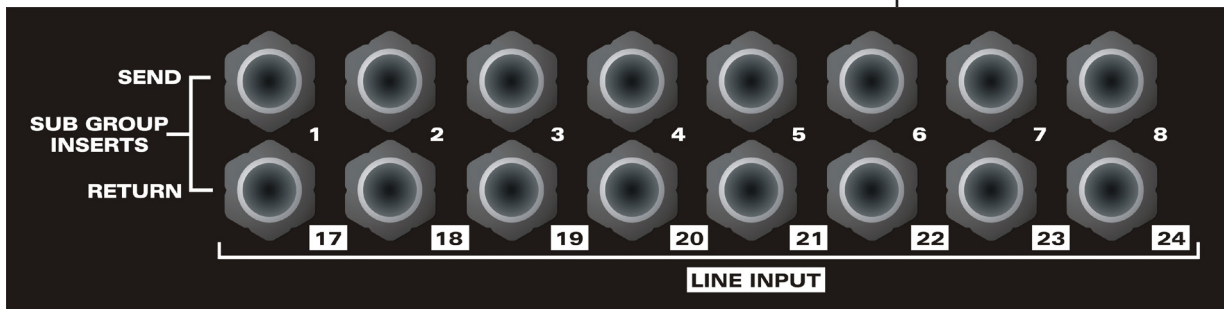
- 1 CUE A and CUE B outputs. These auxiliaries are normally used as feeds to the foldback system by selecting them on the control surface of the console. They can also be used as additional effects sends.
- 2 AUXILIARY 1 and 2 outputs. The remaining auxiliary outputs are identical.

Not all of the outputs are shown for clarity.

The auxiliary outputs are typically used to send to effect units such as reverberation units. They are line level fully balanced outputs.

## MASTERS

### SUB-GRP INSERT/ LINE INPUT 17-24 CONNECTORS



This panel carries the connectors for the sub-group insert points. There are 8 sub-groups and each insert point has a send output and a return input.

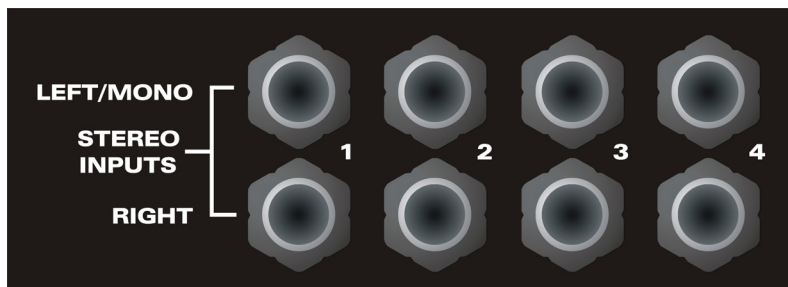
Signal is always present on the send output. If required the insert returns could be used as very basic inputs to the stereo mix bus during mixdown, from a submixer or sampler etc.

The insert return jacks can also be used to provide 8 additional line inputs to the mix. These are designated as Line inputs 17-24.

Note that the additional line inputs are not available if the insert return is also being used.

## MASTERS

### STEREO INPUT CONNECTORS

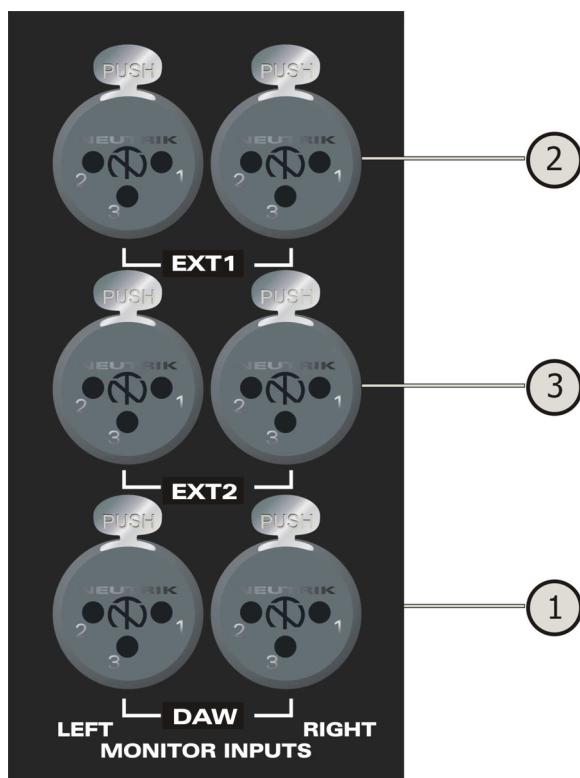


This panel carries the input TRS jacks for the stereo inputs located on the master section of the console.

There are 4 returns each with a left and a right input.

If a mono source is used then it should be plugged into the left input. This is normalled over to the right input causing the signal to travel through the left and right signal paths of the return. If a plug is inserted into the right input the normalled connection is broken.

## MASTERS



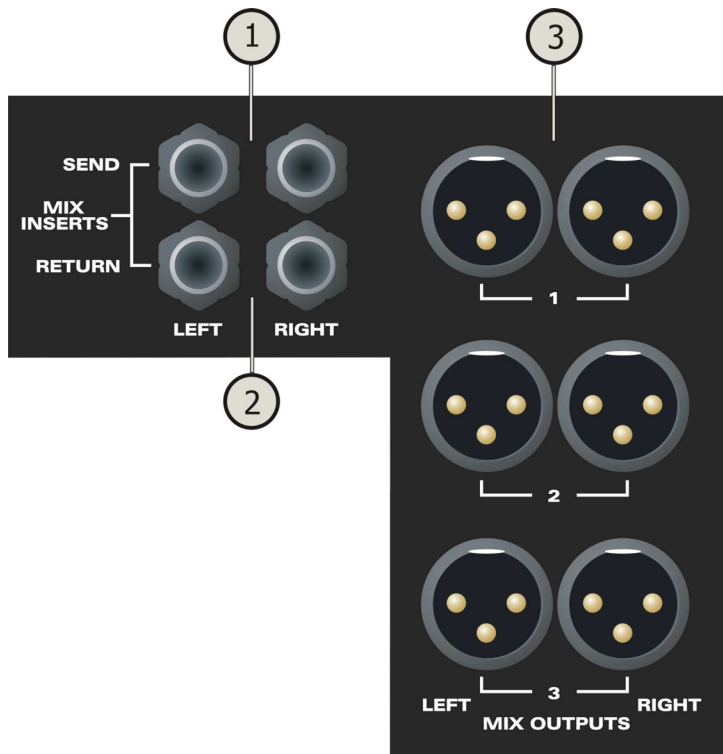
- 1 Input for DAW
- 2 Input for EXT 1
3. Input for EXT 2.

## MONITOR INPUTS

This panel carries the connectors for the external inputs to the monitoring system.

All three inputs are available as sources to the control room monitoring system while the first two can be used as sources for the studio monitoring system and the foldback system.

## MASTERS



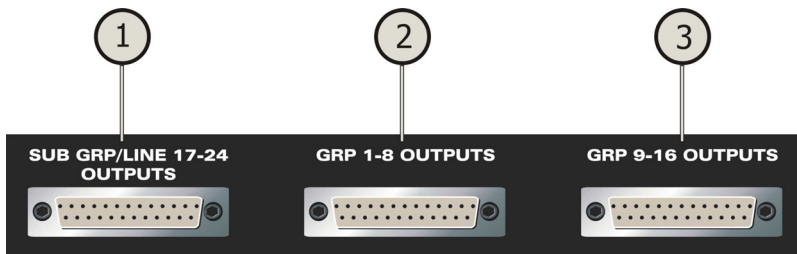
- 1 MIX INSERT SEND jacks. There is a separate send output for the left and right mix.
- 2 MIX INSERT RETURN jacks. There is a separate return input for the left and right mix.
- 3 MIX OUTPUT (Stereo Output) connectors. There are three sets of connectors which can all be used simultaneously.

## MIX OUTPUTS

There are 3 sets of stereo output connectors enabling the connection of 3 stereo recording machines. The outputs can be used for many purposes of course.

Associated with the outputs are the mix insert points. They allow a final mix processor to be inserted into the MIX path, and are located before the MIX compressor/limiter.

## MASTERS



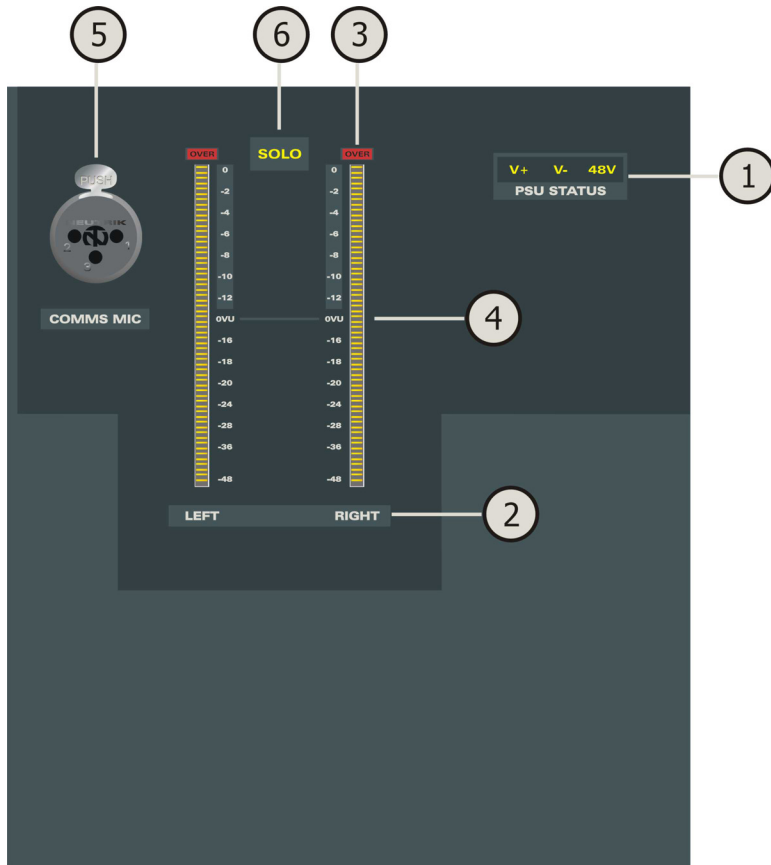
- 1 SUB-GROUP/LINE 17-24 OUTPUT CONNECTOR
- 2 CONNECTOR for BUS outputs 1 through 8.
- 3 CONNECTOR for BUS outputs 9 through 16.

### BUS, SUB-GROUP/ LINE 17-24 OUTPUT CONNECTORS

Rather than having individual connectors for each bus and sub-group output it is much more convenient (and quicker to make a connection) if they are on D-sub multi-pole connectors.

The bus sends are split across 2 connectors with each carrying the bus outputs for 8 tracks. The sub-group/line 17-24 outputs are on a third connector.

## MASTERS



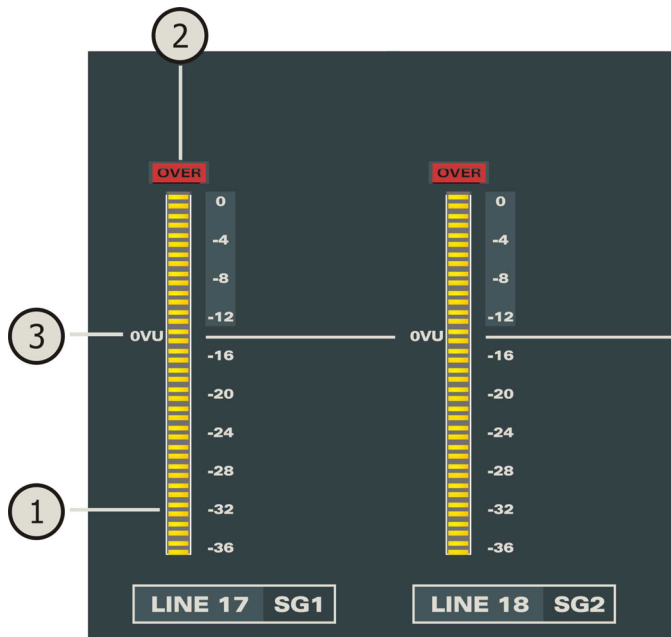
### THE MASTER METERS

Metering has been addressed on the input modules although there are also meters associated with the master functions on the console.

The output meters show the output levels of the sub-groups and also the stereo output. If problems are indicated here with the level being either too high or low then it is most likely that the problem is further back in the signal path. Follow our procedure for gain setting to see if this remedies the situation.

- |  |   |
|--|---|
| <p>1 Power Supply Status Indicators for the +18V, -18V and 48V power rails.</p> <p>2 Left and Right Peak Reading Main Output Meters.</p> <p>3 Over Indicators, when illuminated the signal is in danger of being clipped and the level should be reduced. A level of +20dBu will bring the OVER indicator on. Maximum or 0dBFS is indicated at a signal level of +18dBu, corresponding to full modulation on a digital recorder.</p> | <p>4 0VU. When the signal is at a level of +4dBu the bars should be illuminated up to this point. Note that this is not meant to be normal operating level because the meters are peak reading.</p> <p>5 Talkback Microphone Connector. If talkback is to be used then a microphone should be plugged in here.</p> <p>6 Solo Indicator. This lights when a SOLO button has been pressed on the console.</p> |
|--|---|

## MASTERS



- 1 Sub-group Meter. There are 8 peak reading sub-group meters - 1 for each sub-group output.
- 2 Over Indicator. When illuminated the signal is in danger of being clipped and the level should be reduced. Like the main meters this will illuminate with a signal level of +20dBu. 0dBFS is indicated at a level of +18dBu, which normally corresponds to full modulation on a digital recorder.
- 3 OVU. When the signal is at a level of +4dBu the bars should be illuminated up to this point. Note that this is not meant to be normal operating level because the meters are peak reading.

### THE Sub-group/ LINE INPUT 17-24 METERS

There 8 sub-group outputs meters on the console (only 2 have been shown for clarity). The sub-groups are accessed in parallel with the main bus outputs and only the first 8 buses therefore have a corresponding sub-group.

The sub-groups are able to feed the stereo mix and also have their own outputs located on the rear panel of the console.

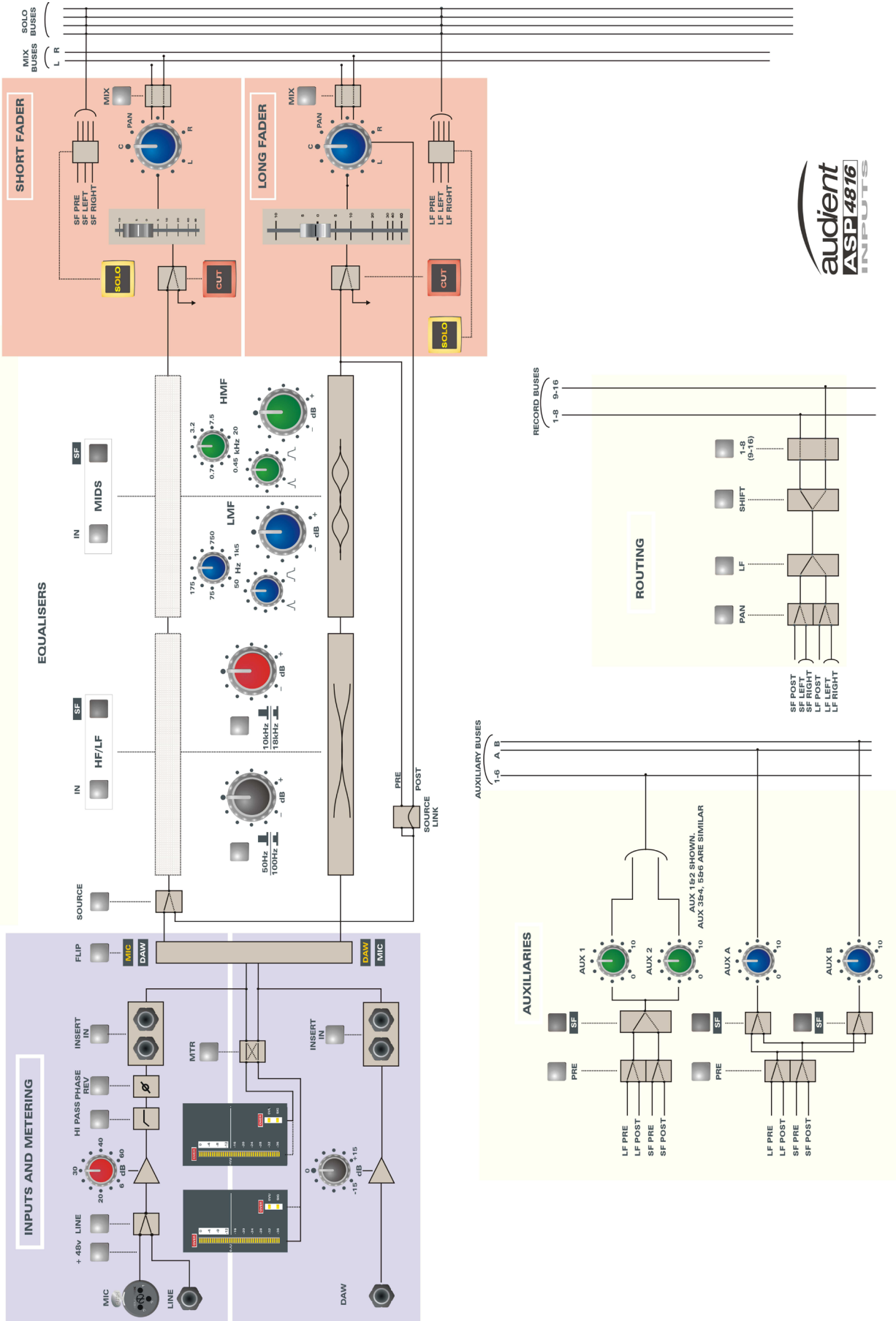
The sub-group outputs can be directly used to feed a DAW or other recording device.

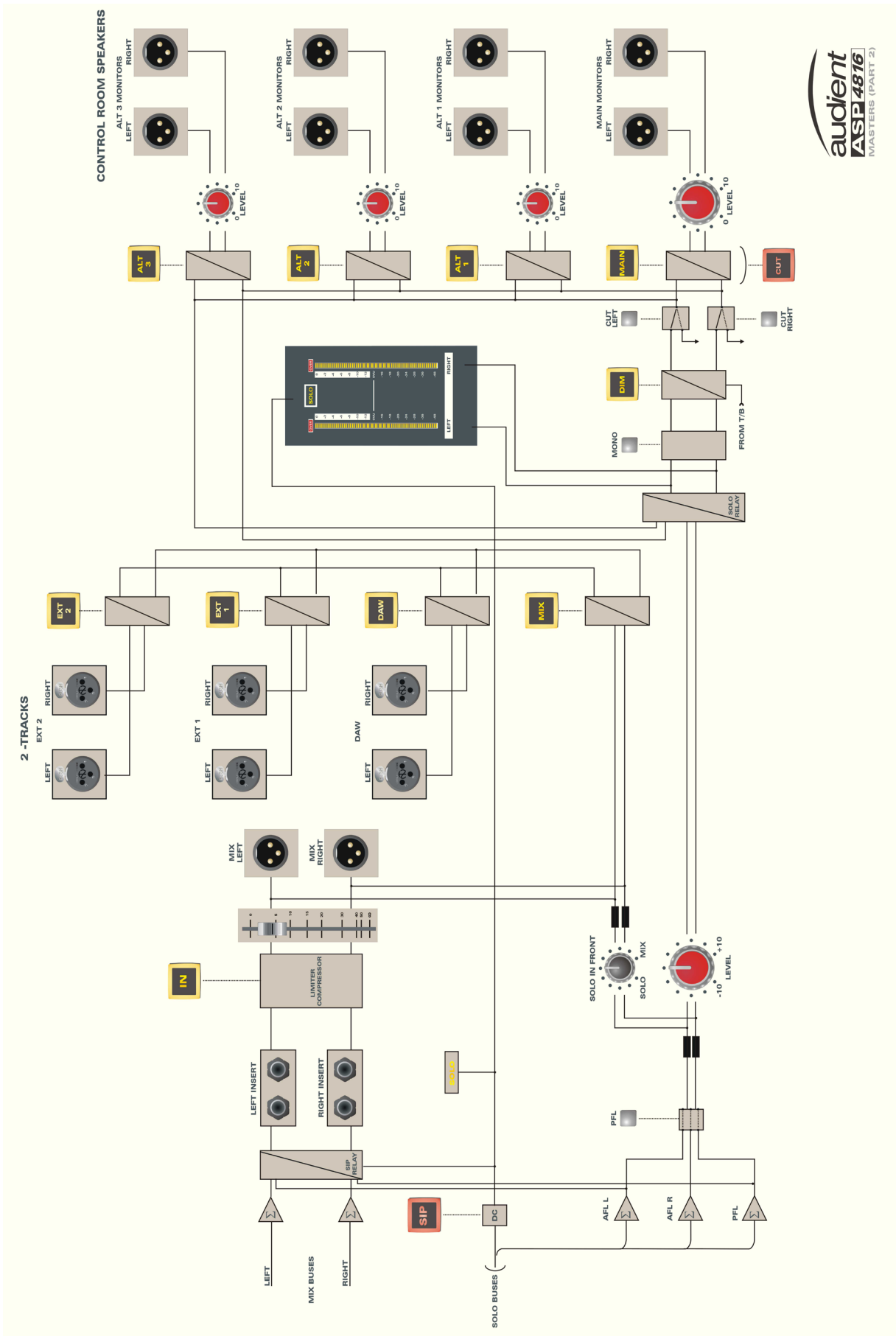
The sub-group insert returns can also be used to access line inputs 17-24, which share access to the same routing and level controls as the sub-groups themselves.

A typical use for the sub-groups would be during mixdown when a number of signals are to be combined to one fader for simpler control. If the signals are on the LF path then they must first of all be sent to the routing matrix by pressing the button in the routing section. By routing the signals to buses 1 through 8 they will also be routed to the sub-groups.

## Block diagrams









## Warranty.

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Your ASP4816 comes with a manufacturer's warranty for one year from the date of despatch to the end user.

The warranty covers faults due to defective materials used in manufacture and faulty workmanship only.

During this warranty period Audient will repair or at its discretion replace the faulty unit provided it is returned carriage paid to an authorised Audient service centre. We will not provide warranty repair if in our opinion the fault has resulted from unauthorised modification, misuse, negligence, act of God or accident.

We accept a liability to repair or replace your ASP4816 as described above. We do not accept any additional liability. This warranty does not affect any legal rights you may have against the person who supplied this product – it is additional to those rights.

### Limitations

This warranty does not cover damage resulting from accident or misuse. The warranty is void unless repairs are carried out by an authorised service centre. The warranty is void if the unit has been modified other than at the manufacturer's instruction. The warranty does not cover components which have a limited life, and which are expected to be periodically replaced for optimal performance. We do not warrant that the unit shall operate in any other way than as described in this manual.