audient

ASP 2802

DUAL LAYER CONTROL

OPERATING MANUAL

Version: 1.00

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Welcome to ASP2802

Thank you for selecting this Audient product.

We hope that your new ASP2802 console will provide an intuitive, ergonomic and flexible analogue recording and mixing platform in a powerful yet small package.

We are sure that ASP2802 will deliver a wealth of routing, control and production flexibility coupled with outstanding audio quality.

ASP2802 provides an impressive array of connectivity for its size, featuring high quality signal conditioning, routing, summing, processing and monitoring capabilities. Add to this a fully fledged moving fader automation system and integrated HUI™ control surface and this becomes a very powerful console featuring our all-new Dual Layer Technology.

ASP2802 will adapt to become the centre of your creative workflow, ideally suited for use in project studio production spaces, location recording or high quality hybrid mixing systems.

Let's take a look!

How to use this manual.

This manual has been divided into sections for your convenience. Along the way a number of handy tips and suggestions will offer possible uses and applications for the console during creative use.

Look out for the "tips and suggestions" boxes!

✱ TIPS & SUGGESTIONS

Any numbers in brackets (xx) refer to control numbers as found in the overview illustrations in each section.

Mackie HUI™ (Human User Interface) is a registered trademark of LOUD Technologies Inc.

WARNING

THIS EQUIPMENT MUST BE EARTHED
DO NOT EXPOSE TO RAIN OR MOISTURE

CAUTION

RISK OF
ELECTRIC SHOCK
DO NOT OPEN

ATTENTION

RISQUE DE CHOC
NE PAS ENLEVER

Unpacking

Your ASP2802 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.

Your 2802 packaging should include the console, an IEC power cord, quick start guide and a CD containing in this manual, signal flow block diagrams and drivers. Please note that rack ears for ASP2802 are optional.

Important Safety Instructions

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

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

2802 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. Therefore 2802 will work happily anywhere in the world but 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.

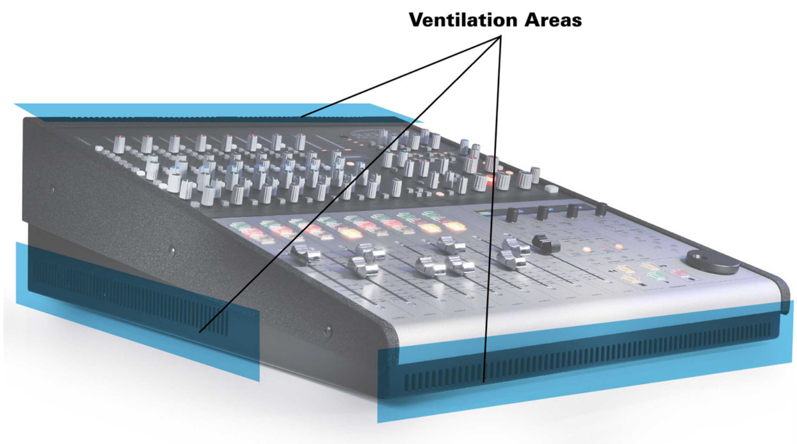
See the warranty section provided at the end of this manual for details of your cover.

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.

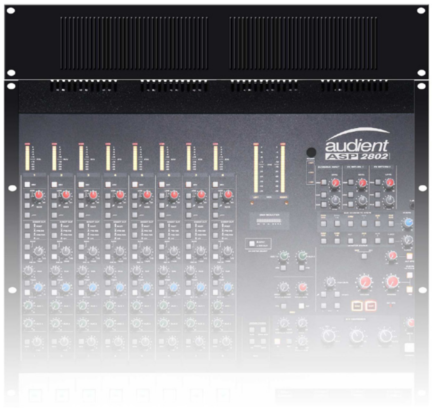
These can be found along the left hand bottom edge on the side panel (below the cheek), the rear top wrap-around ventilation panel and below the faders at the foot of the console.

If mounting the console into some form of studio furniture or desk, **please ensure that there are sufficient air gaps at these locations**.



Rack Mounted Ventilation

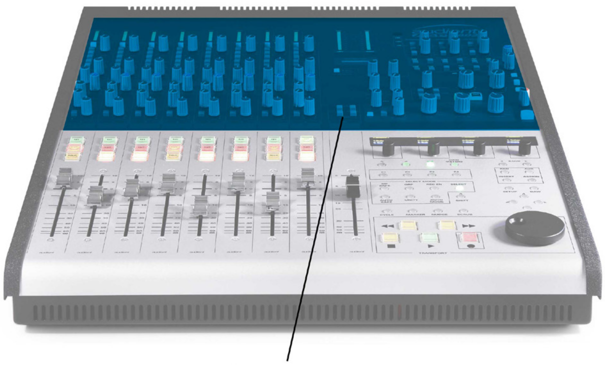
Rack Ventilation Blank



The console can also be rack mounted with the rack ear kit and will occupy 13RU plus connector space (suggest 15RU total) - **please ensure that these ventilation holes are not restricted.**

A vented 19” blanking plate is recommended to cover the connector space, providing adequate air flow to the top ventilation plate.

Lexan Overlay



Lexan Area

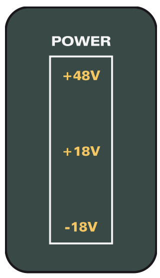
The front panel at the top of ASP2802 (dark section only) features a rugged under-surface printed polycarbonate overlay.

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.

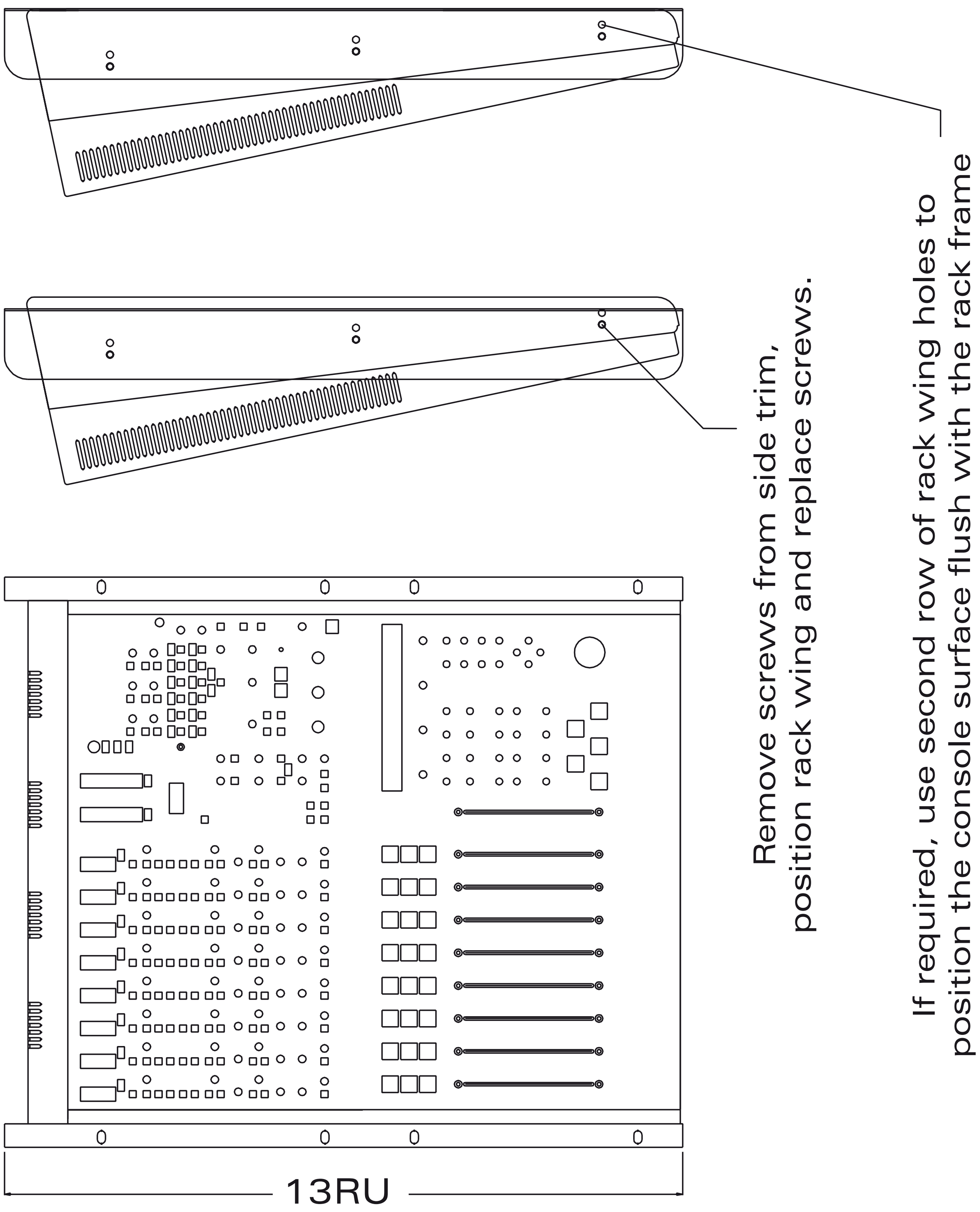
However at least the panel legending will not wear off like some of the silk screened or engraved products of yesteryear!

Power Up

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

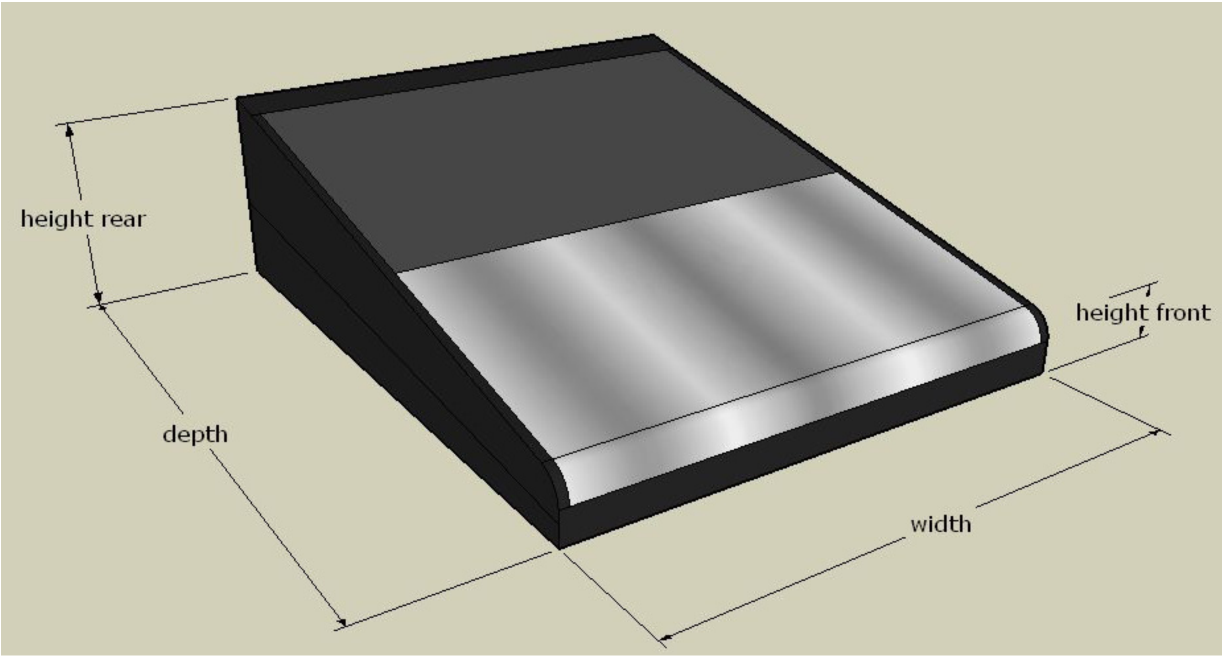


Optional Rack Ear Kit - Installation



ASP2802 Dimensions (including rubber feet)

Height Front -	60 mm	Height Rear - 170 mm
Width -	450 mm	
Depth -	550 mm	Height (depth) when racked 13RU plus connector space 15RU



ASP2802 Specifications

Some general performance figures from the console - taken with Prism D-Scope test equipment.

Frequency Response	+/- 0.3 dB 20 Hz to 20 kHz
THD + Noise	<0.015% @ 1 kHz
Noise	Mic EIN (20 Hz to 20 kHz, 150 ohm source < -127.5 dBu
Bus Noise	All inputs routed < -80 dBu
Mic CMRR	70 dB (min. gain)

ASP2802 has many features including 8 channels of Class-A hybrid discrete microphone preamplifier as used in our time honoured ASP8024 console and ASP008 8-channel microphone preamplifier, guaranteeing outstanding sonic transparency and detail.

When combined with useful LED bargraph metering, input signal conditioning, freely assignable direct outputs and comprehensive cue monitoring options, ASP2802 is right at home in a small location recording or tracking environment.

A rear panel full of high quality locking XLR, TRS and 25-pin D-Sub connectors (wired to Tascam DA-88 pinout specification, page 42) ensures it is simple and quick to integrate your AD/DA converters, favourite outboard and existing microphone preamplifiers.

ASP2802 is equally comfortable as a hybrid mixing tool - featuring a DAW HUI™ control surface combined with 8 channels of analogue fader automation and up to 30 main inputs on mixdown.

The high number of inputs in such a small package is achieved via 16 inputs in the channels (alternative cue input used) plus a further 14 inputs (4 stereo summing input pairs, 2 stereo FX returns and stereo DAW Mix input) in the master section.

Combined with 2 aux sends (plus stereo cue depending upon cue destination or alternative input usage), inserts, dedicated DAW returns, mix bus parallel insert mode, classic Audient summing, patchable VCA bus compressor and a high quality centre section complete with mono, dim, phase and talkback communications - any eventuality is covered.

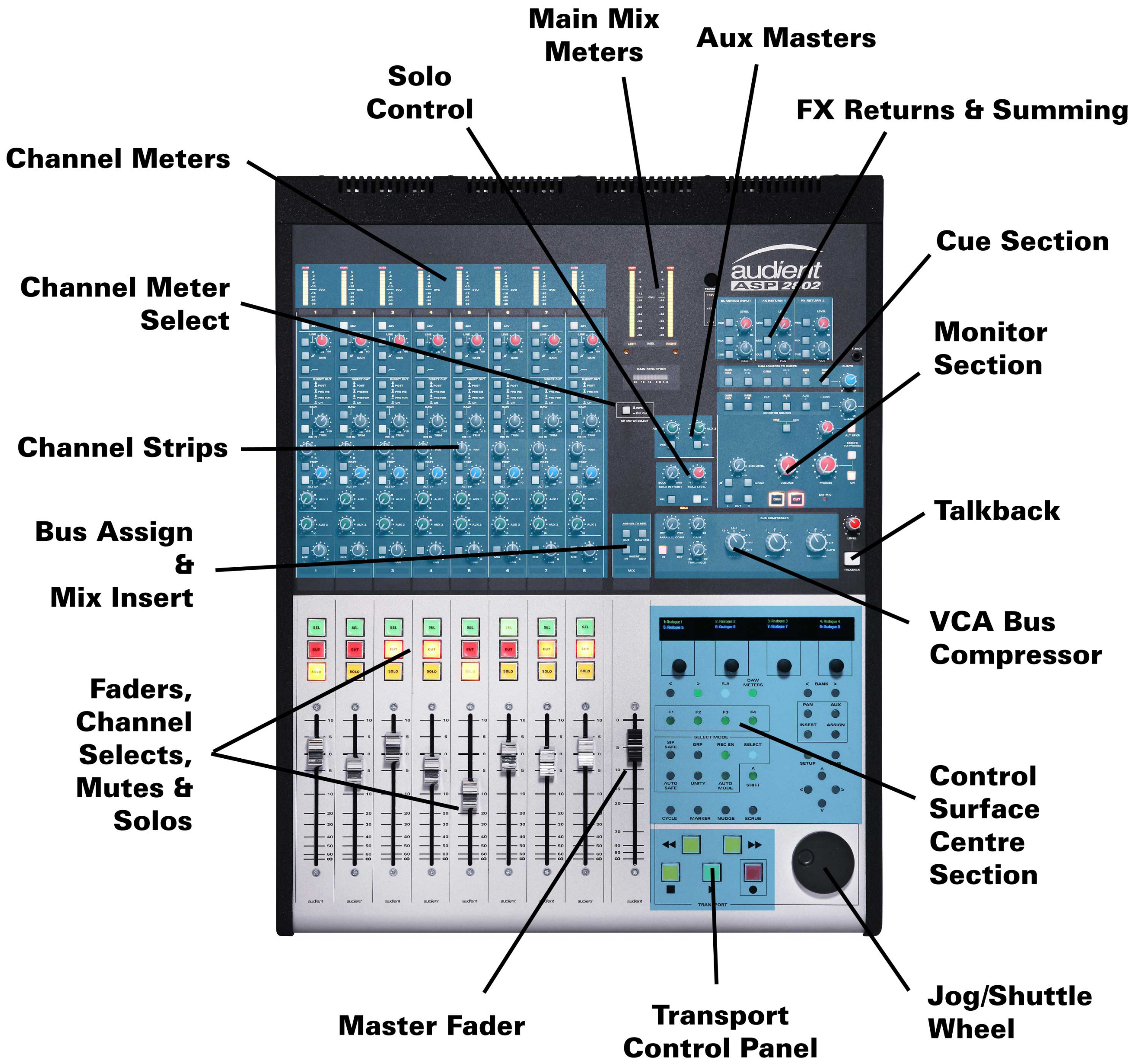
Get ready to unleash the power of this formidable but neatly packaged mini console.

Introduction

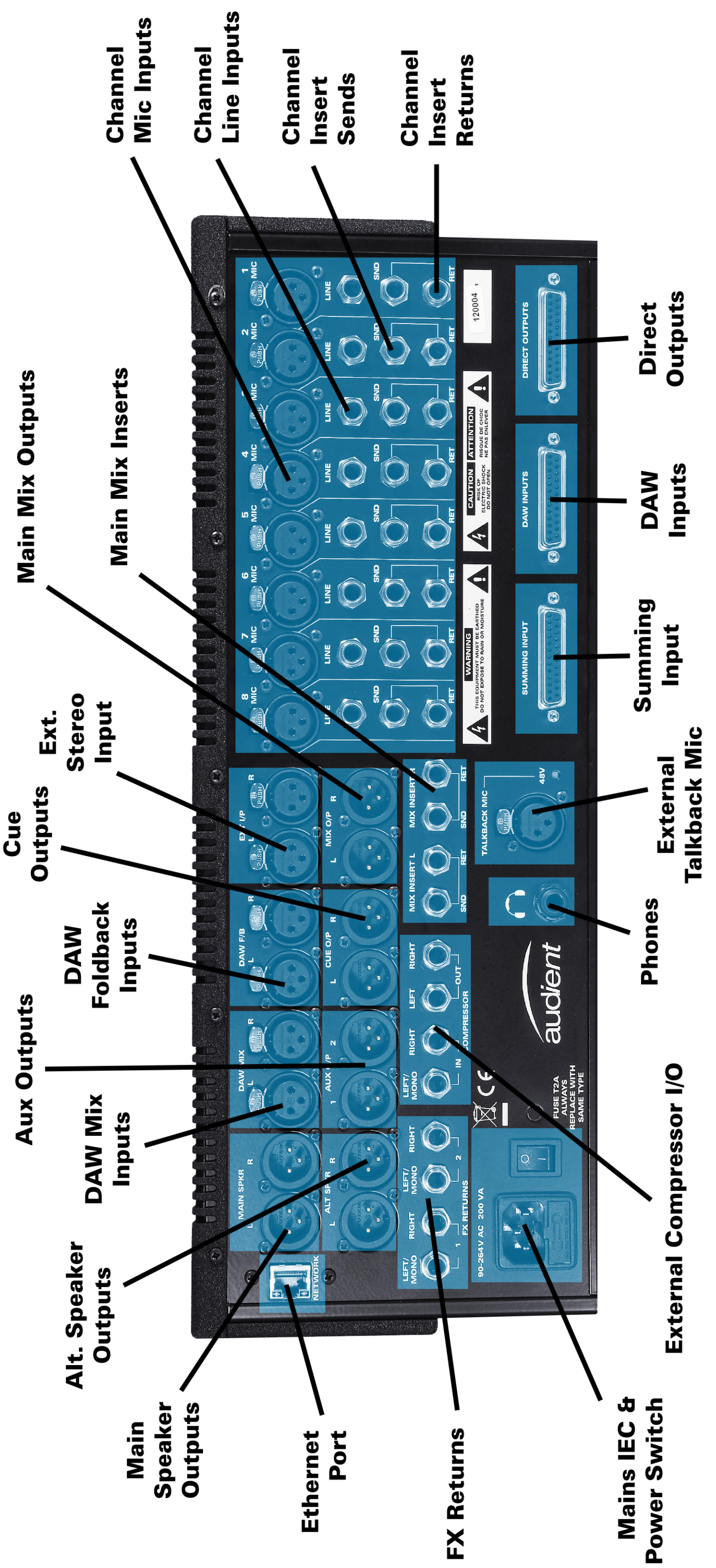
ASP2802 Feature Set

- 8 class-A hybrid discrete microphone preamplifers
- 8 balanced line inputs
- 8 balanced DAW inputs
- 8 balanced assignable direct outputs
- Input conditioning functions such as 48V phantom power, polarity reverse and a 75Hz high pass filter
- Balanced, switchable inserts on channels 1-8 and main mix. Main mix insert has a sum feature to facilitate parallel processing. Note that all insert sends are permanently active
- 2 Auxiliary sends globally switched for pre or post fader operation
- Channels 1-8 feature VCA automation that can be controlled directly from your DAW
- Channels 1-8 feature click-free silent muting
- LED bargraph channel metering which can follow either channel input, channel output or DAW channels as provided from the control surface
- Channel 1-8 'unity' function for quick set up if mixing 'in the box' but summing via ASP2802
- Stereo cue with alternative input functionality to expand number of channel inputs from 8 to 16
- Flexible cue section with summed source selection, built-in talkback communications with provision for an external talkback microphone and 48V phantom power if desired
- Fully featured 8-fader HUI™ based control surface with jog/shuttle wheel, transport panel, 4 rotary encoders and OLED channel display
- On-board VCA bus compressor with wet/dry blend and patchable inputs / outputs
- 2 Stereo FX returns with mono mode, balance and level controls, plus routing to either cue or mix bus
- 8 channel summing input (4 stereo pairs) with mono mode, balance and level controls plus routing to either cue or mix bus
- Fully featured monitor section with source selection, including i-jack front panel input. Monitor section includes mono, polarity reverse, cut left, cut right, dim, volume and speaker selection
- ALPS 100mm faders (channels are motorized and master fader is manual)
- i-Jack consumer level front panel monitor input for MP3 players etc
- Solo-in-front
- Solo-in-place
- Solo safe
- Channel automation safe function
- Attractive black livery
- Internal passively cooled power supply
- Audient Dual Layer Technology

Operational Surface Overview



Rear Panel Connections Overview



Channel Strip Overview

audient ASP 2802

DUAL LAYER CONTROL

ASP2802 features 8 high quality, flexible channel strips with a multitude of routing options.

The channel strips are located on the left side of the console and the ergonomic, inclined profile (approx. 12°) of ASP2802 ensures all controls are clearly visible and within arms reach.

There are three main channel inputs:

- Mic
- Line
- DAW

Each of these inputs is fed from their own rear panel connectors (Mic - Female XLR, Line - 1/4" TRS Jack and DAW - DB25 Female).

For more information and specifications see the connections section or block diagrams.

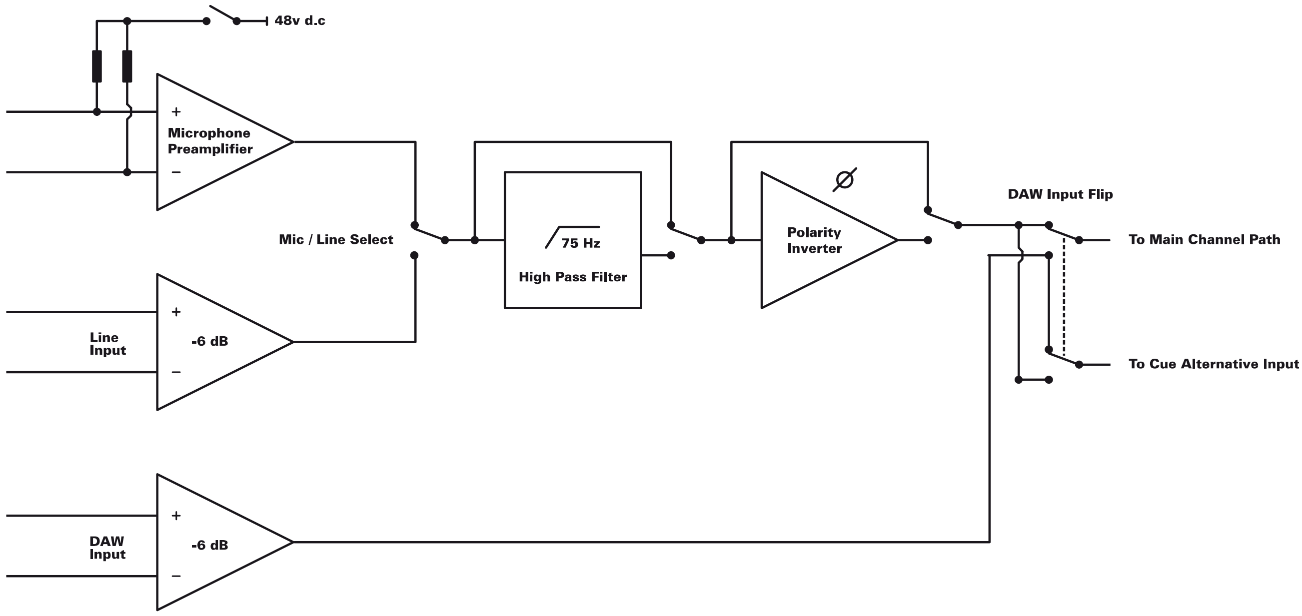
Input Conditioning

The microphone preamplifier provides 6 to 60 dB of clean gain, delivered from our classic hybrid discrete class-A transistor and integrated circuit design. 48v phantom power and an input impedance of >2k5 ohms ensure that you will get the most from your microphones.

Two features that should be very useful during tracking are available on either the microphone preamplifier or line input paths (if using boutique external microphone preamplifiers) are the 75 Hz high pass filter and polarity reverse.

The HPF enables you to remove any rumble or subsonic noise and polarity invert provides alignment of sources (for example snare top and snare bottom). These features are tied to the mic/ line path and are not available on the DAW input - however these are easily performed 'in the box' during mixing.

The three input stages and basic signal flow of the ASP2802 input conditioning stage is illustrated below:



The mic/line path normally feeds the main channel path, this path features the configurable direct output and as such is the default recording path.

In typical operation it would be common to use the DAW input as the return path from your workstation outputs and use the mic/line path for recording (as psuedo in-line functionality is entirely possible with the direct output options).

It is possible to monitor the post record path (commonly referred to as tape) via the DAW input when assigned to the alternative cue input, providing a psuedo in-line short and long fader style of operation for those familiar with traditional in-line consoles.

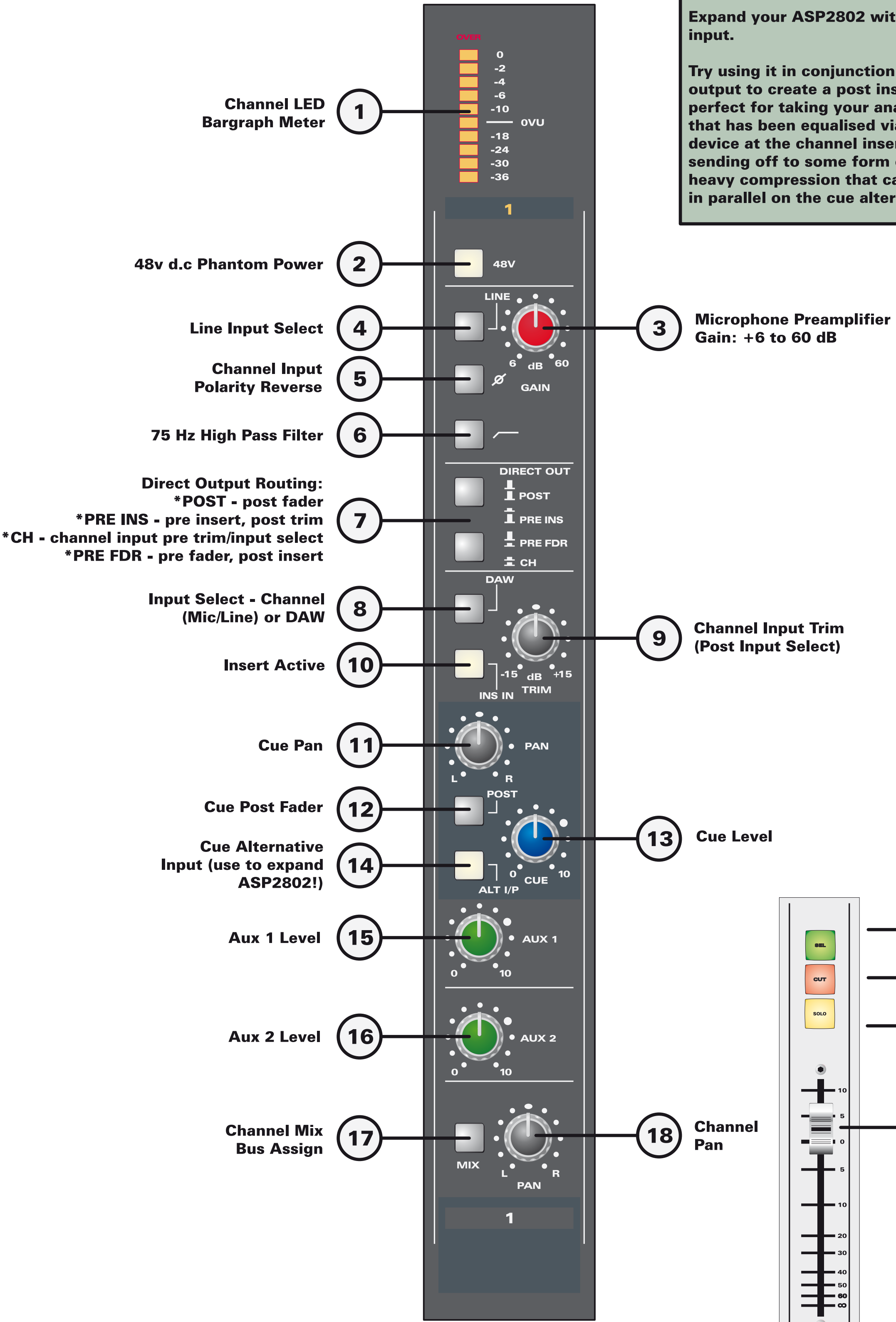
With two line inputs per channel (DAW and line), and a flexible cue send with alternative input (fed from whichever of these is not feeding the main channel path) an 8 channel ASP2802 can provide 16 dedicated line inputs for mixdown along with a large number of extra inputs in a dedicated FX return section.

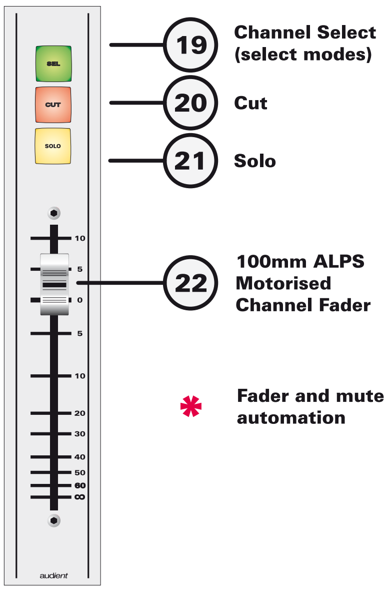
Channel Strip

✱ TIPS & SUGGESTIONS

Expand your ASP2802 with the Cue Alt input.

Try using it in conjunction with the direct output to create a post insert mult - perfect for taking your analogue signal that has been equalised via an analogue device at the channel insert point and sending off to some form of distortion or heavy compression that can be returned in parallel on the cue alternative input!





Peak Reading Bargraph Meter 1

A 10-segment LED bargraph meter with peak detection and a separate overload indication LED, suitable for use in today's modern digital recording environment.

6 dB per segment in the lower region of the meter (-36 to -18 dB) and 4 dB per segment in the middle region (-18 to -6 dB) with 2 dB per segment in the top region of the meter for greater resolution.

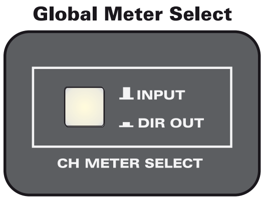
0VU is the centre of the scale and calibrated to a nominal operating level of +4 dBu. Which in turn is relative to a typical EBU calibration of +18 dBu = 0 dBFS. 0 dBFS is your absolute maximum digital level (full scale) and should be avoided at all costs - digital clipping is not a pleasant sound.

As such the bargraph meter scale on ASP2802 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. Therefore ASP2802 provides some headroom for your beautifully crisp transient sources.

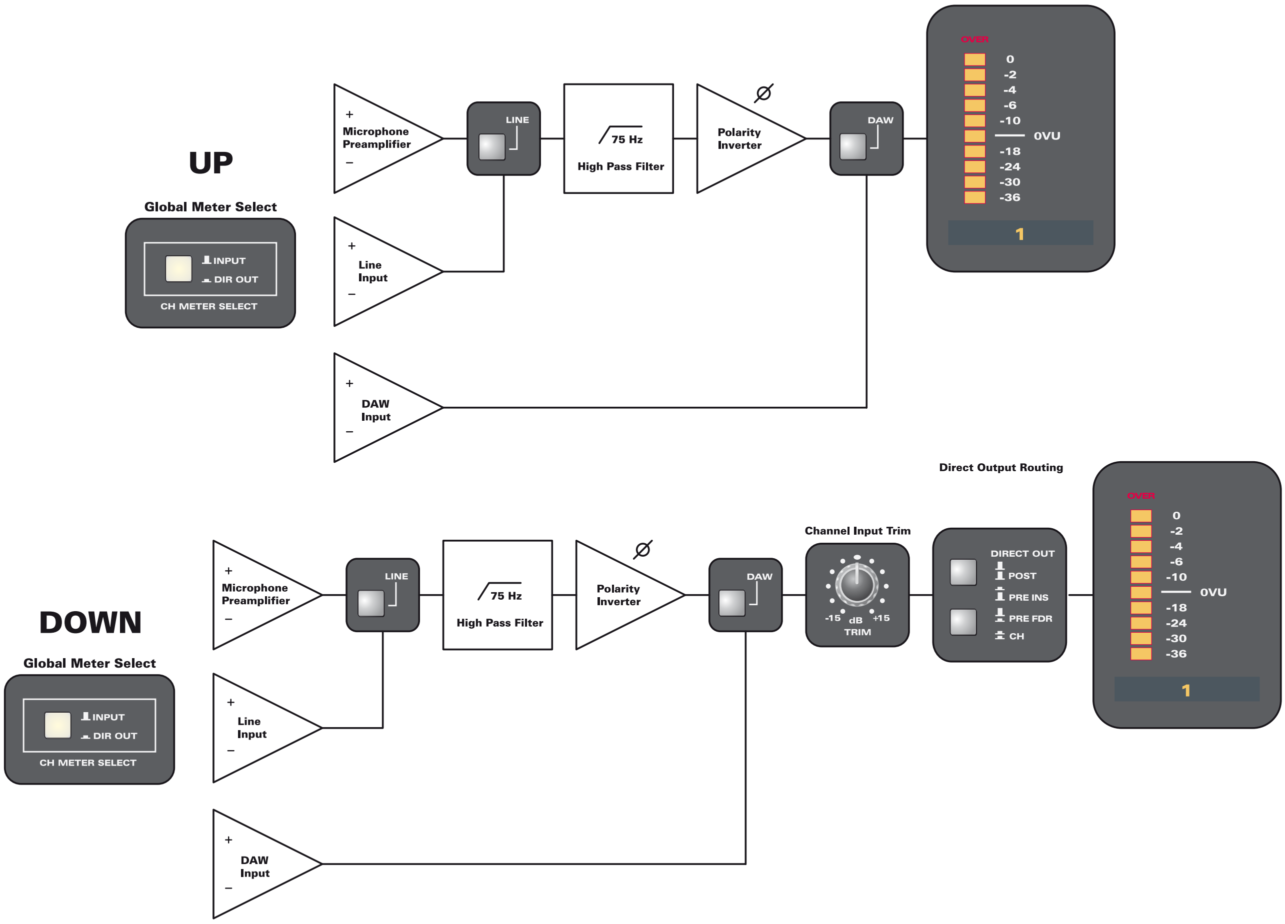
DIGITAL CLIP		
0 dBFS	+18 dBu	
-2 dBFS	+16 dBu	
-4 dBFS	+14 dBu	
-6 dBFS	+12 dBu	
-10 dBFS	+8 dBu	
-14 dBFS	**+4 dBu**	**0VU**
-18 dBFS	0 dBu	
-24 dBFS	-6 dBu	
-30 dBFS	-12 dBu	
-36 dBFS	-18 dBu	

Channel Meter Global Select - (MTR)



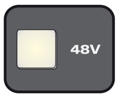
This switch is located in the centre section of the console below the main mix and compression gain reduction metering. This globally alters the source to the peak reading bargraph meter at the top of each channel strip. Note that this can be overridden if using the meters to display DAW channel metering values in control surface operation (DAW Meters function).

In the UP position, the channel input path feeds the meter, taken before the channel trim control (9) but after the mic/line and DAW flip (4 & 8) routing selections. When engaged into the DOWN position, the meter takes its feed from the the direct output feed. Thus, its exact location is determined by the routing selections made in the direct output assign section (7) - see later in this chapter.



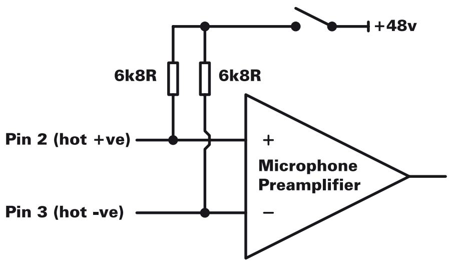
48v Phantom Power (2)

Engaging this switch provides 48 volt D.C (direct current) phantom power to the microphone input. Phantom power is applied via a pair of current limiting resistors to XLR pin 2 (hot +ve) and XLR pin 3 (cold -ve). ASP2802 phantom power meets the DIN 45956 specification of 48v, +/-4v at 10mA per channel.



With individual phantom power switches on each channel you have total control over which microphones receive power. Rest assured that the powerful switch-mode power supply can deliver all of the current needed for the hungriest microphones.

Phantom power wiring:



Microphone Preamplifier Gain (3)

The microphone preamplifier fitted to ASP2802 is the same class-A, low-noise hybrid design made popular with the ASP8024 large format in-line console and the ASP008 8-channel rackmount preamplifier.

The amount of gain on offer here is 6 to 60 dB of clean amplification and features a low-noise, differential discrete transistor input stage.

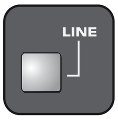
With a minimum gain of 6 dB and the option of taking a post-fader direct output, ASP2802 is capable of dealing with even the most aggressive of tracking sessions, ensuring headroom and clarity with total freedom from clipping your recording device.

The configurable direct output also allows you to ride levels to “tape” or “disk” by providing a post fader option.

Explore the direct output routings further (see channel strip item 7) as ASP2802 can function in several useful ways for tracking.

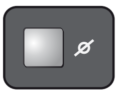
Line Input Select (4)

Engaging this switch overrides the microphone preamplifer input with the line input - this is the channel source, which can later be swapped for the DAW input.



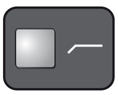
Polarity Invert (5)

A 180° polarity invert function available for the microphone preamplifier or line input - useful for checking phase when tracking with multiple microphones or flipping an existing track during mixdown if fed to the line input. This is achieved with a unity gain inverting amplifier.



High Pass Filter (6)

A 75 Hz HPF with a 12 dB per octave roll-off rate can be engaged on the mic/line source - useful for removing low frequency content that clutters up your mix and robs the console of headroom.



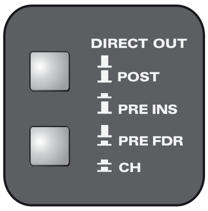
Considering that many listeners use systems that can barely reproduce 60-70 Hz accurately - a well placed HPF on certain sources can clean up mix elements that contain subsonic energy, such as microphone stand rumble, traffic noise, or some plosive material without overly affecting the final mix.

Direct Output Routing 7

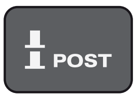
This section is probably the most flexible element in the channel strip, providing 4 different signal path configurations for the entire channel strip, switched on a per channel basis.

There are 4 modes to remember here - and they will seriously aid your workflow.

- POST
- PRE INS
- PRE FDR
- CH

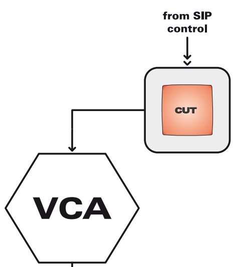


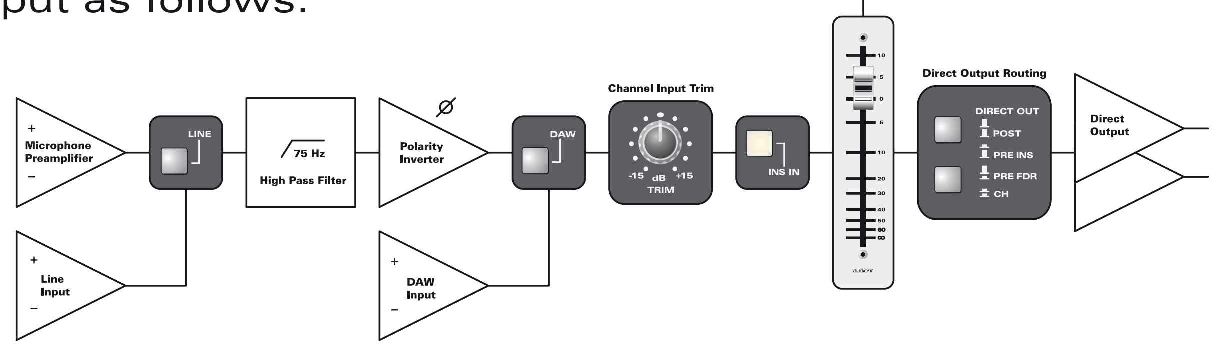
The first mode - POST requires both switches in the direct output routing section to be in the UP position.



The direct output is taken POST VCA FADER and POST VCA MUTE.

This configures the direct output as follows:





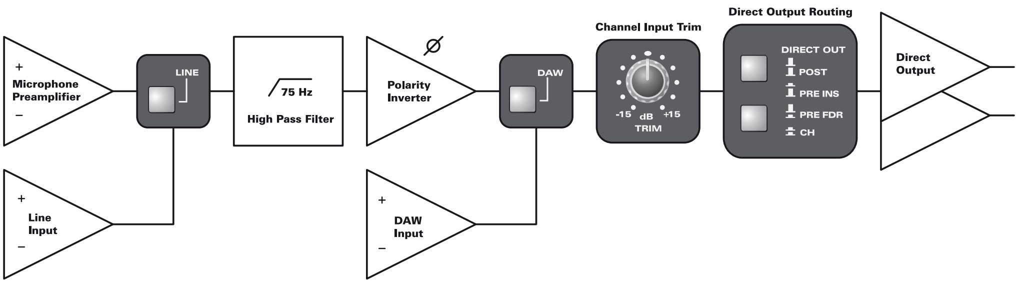
Post fader is the most appropriate mode to use for the following tasks:

- Riding levels to “tape” or “disk” when tracking (especially when microphone levels are hot
- Processing line level signals via the insert and “printing back” the effected sound to “tape” or “disk” with final level control

The second mode - PRE INSERT requires the first switch in the direct output routing section to be in the DOWN position.

The direct output is now taken PRE INSERT and PRE FADER.

This mode configures the direct output as follows:



This mode is useful when you want to take a “clean feed” during tracking, yet still monitor your signal with processing via the insert and set balances in real time, perhaps for headphone mix purposes with the processed signal on aux sends or the channel fader.

This mode also provides extra gain during tracking via the channel trim control - a convenient place to ride levels and arguably a cleaner way of gain staging as opposed to varying the microphone preamplifier gain during a take.

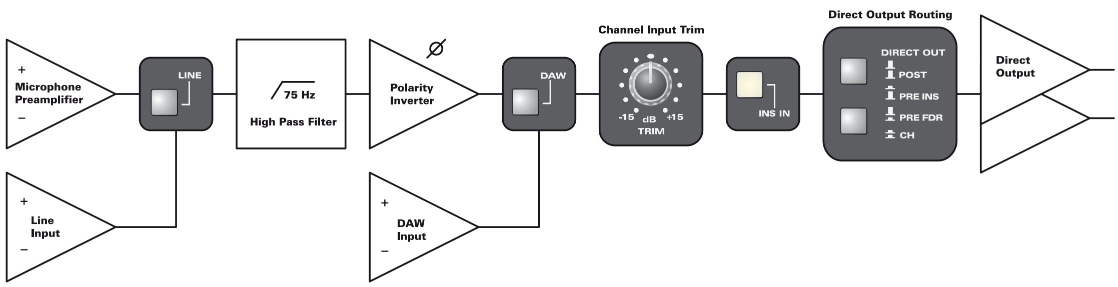
The third mode - PRE FADER requires the second switch in the direct output routing section to be in the DOWN position.

Direct Output Routing cont.

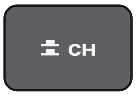
The PRE FADER mode is useful when you want to take a pre fader record output with the addition of extra gain from the trim section and perhaps some processing (dynamics perhaps) on the insert.

Pre fader routing may be the most useful if you are using the console faders to set a monitor balance and therefore do not want to have the faders affecting your recording levels.

In this mode signal at the direct output is taken POST INSERT but PRE FADER.



The fourth mode - CHANNEL requires both switches in the direct output routing section to be in the DOWN position.



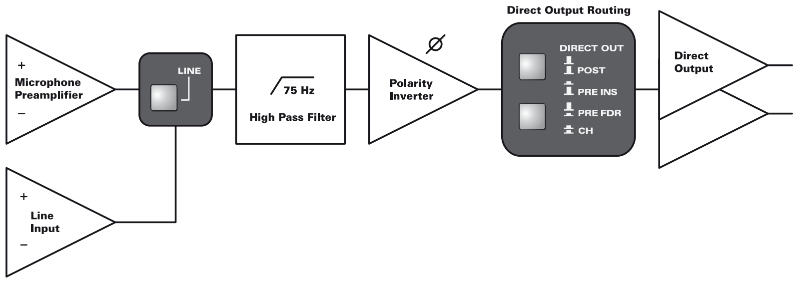
This is quite possibly the most useful mode for tracking if you are used to working with an in-line console architecture.

In normal use, CHANNEL mode allows the direct output to be taken direct from the microphone preamplifier, HPF and polarity inverter stage with no added gain stages such as channel trim. This provides the cleanest signal path for recording.

By using the line input this mode provides the cleanest path to feed your favourite outboard microphone preamplifiers to your recording platform - but with the added benefit of monitoring direct from the latency free environment of an analogue console.

Direct Output Routing cont.

The signal path when in CHANNEL mode is as follows:



Using CHANNEL mode it is possible to combine it with the CH/DAW flip switch (8) to create a “pseudo in-line” architecture.

When operating in “psuedo in-line” mode you can effectively take the microphone preamplifier or line input (which could be fed from your external microphone preamplifiers), HPF and polarity inverter output straight to the direct output and recording device.

Whilst the line or microphone preamplifier input feeds the direct output, the channel can be reconfigured to allow monitoring of the signal pre recording device and/or post.

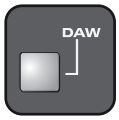
One possible path is via the alternative cue input (14) which allows the signal to be recorded to be monitored (prior to reaching its destination) by sending the channel input (line or mic) signal to the cue level and pan control (11 & 13) and finally summing the cue bus into the main mix bus (via the global control in the centre section above the master fader).

*NOTE that all cue sends will feed the main mix bus if this control is activated, as it is a global assign feature. Therefore this operation cannot be selected on a channel by channel basis.

This is similar to monitoring your short fader record path on an in-line console architecture.

Direct Output Routing cont.

By pressing the DAW switch (8) DOWN, the main channel path (equivalent of the long fader) is fed by the DAW line input.



This would allow you to monitor the recorded signal from your workstation or tape machine, making simple record and playback (before and after) monitoring very quick and efficient.

If you are using a tape machine with a delay between record and reproduction heads, or a workstation with noticable latency from input to output - the benefit here is huge.

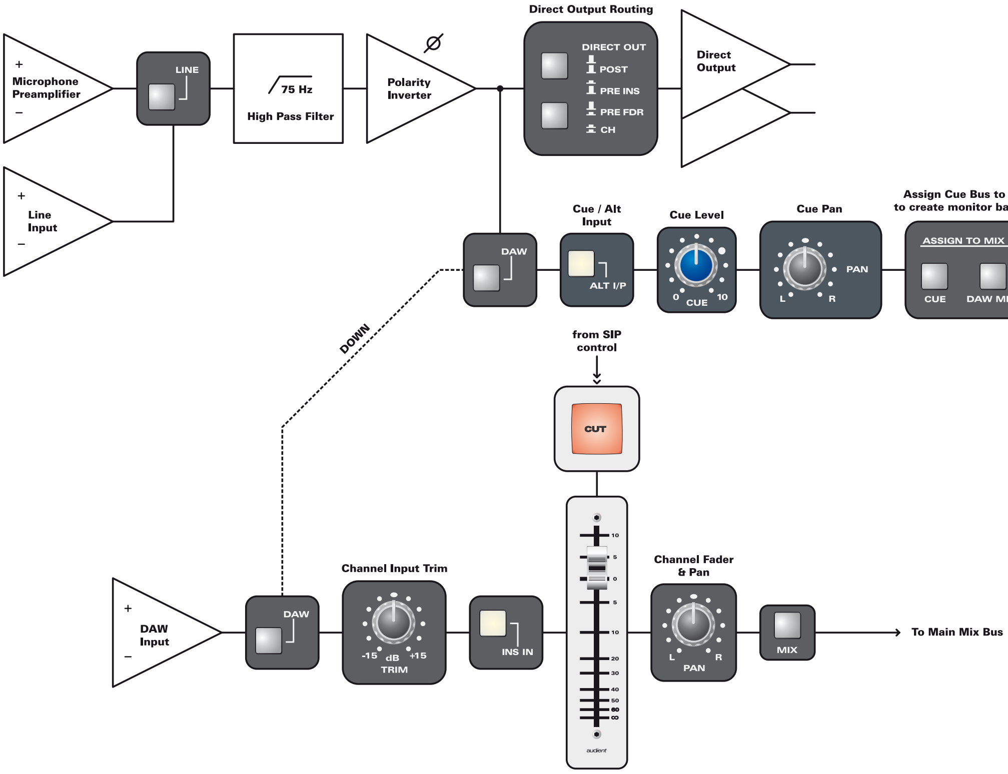
It is possible to feed your microphone preamplifier signal direct to tape or disk from the channel input. Meanwhile, use the cue alternative input to feed a balance of the undelayed signal to your artist's headphone cue mix - whilst at the same time monitoring the main fader path fed from tape or disk so you can hear the colouration provided by your recording medium (see diagram on the next page).

✱ SUGGESTION

Here are several other useful ways of setting up the console and direct outputs for monitoring during tracking and/or stem mixing:

- Use a post fader direct output to enable riding of input sources or printing stems with automation data during mixdown.
- Monitor back the recorded signals (printback or live feeds) via the channel DAW inputs and alternative cue input by assigning the cue bus to either the main mix bus or monitoring its output by selecting it in the monitor source selection.

Psuedo In-Line Architecture



A Note On Building Monitor Mixes

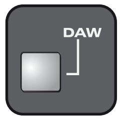
Instead of building monitor mixes the traditional way using the console auxiliaries, consider building your rough / monitor mix in the DAW utilising the ASP2802 HUI™ control surface functionality.

Bring this mix back into the console either via the alternative cue inputs or (if using the cue to feed external artist cues or fx devices) the DAW Mix and DAW FB stereo rear panel XLR inputs, which can be combined in the monitor and cue section to provide hybrid headphone mixes containing some live feeds from console auxiliaries along with DAW foldback mixes which may include backing or click tracks for example.

This way of working allows your cue mixes to be completely recalled very quickly by saving them in your DAW session.

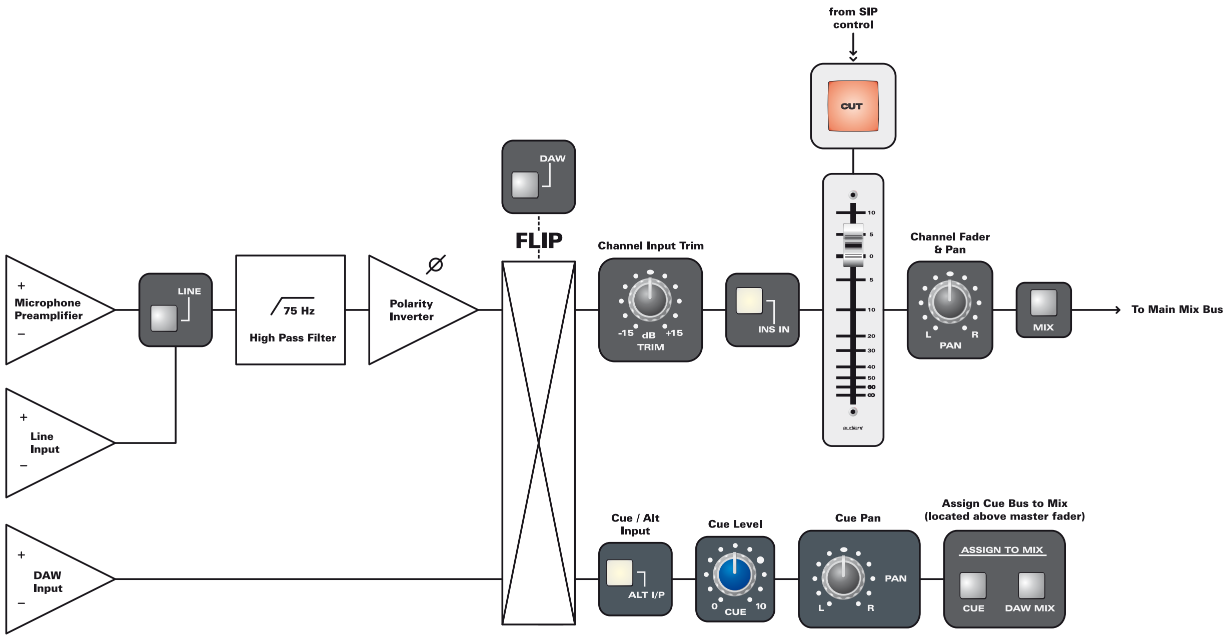
DAW Input / Channel Flip (8)

This function flips the sources to both the main channel path and the cue section alternative input.



When the switch is in the UP position, the mic / line input feeds the main channel path, while the DAW line input feeds the cue section alternative input (which can optionally be selected in the cue section to turn the cue into an extra input with level, pan and global bus routing).

When the switch is engaged into the DOWN position - the sources to each section flip over - therefore the mic / line input will feed the cue section alternative input and the DAW input will feed the main channel path.



DAW channel flip can be used to transform the console from a tracking arrangement where the mic/line source feeds the main channel path, into a mixdown situation where the DAW input feeds the main channel path, providing access to the 100mm linear faders and VCA automation system.

Channel Trim Control 9

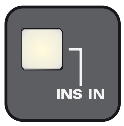
The potentiometer provides a trim range of +/-15 dB and depending upon your direct output routing, can be used as an extra gain boost added to the microphone preamplifier when tracking. This is useful for low output microphones such as ribbon transducer designs.

The trim control is also very useful for setting correct gain staging at the input of the console when utilising the mic / line input (or the DAW input with the DAW channel flip switch engaged.

Insert Active 10

When pressed DOWN, this switch engages the insert send and return loop for each channel (insert send and return balanced 1/4" TRS jack connectors found on the rear panel).

This is a great place to insert your favourite outboard processors into the channel path.



Please bear in mind that with no TRS connectors plugged into the insert points (rear of the console) - the send and return jacks are normalled to prevent loss of signal during accidental engagement of the insert switch.

✱ SUGGESTION

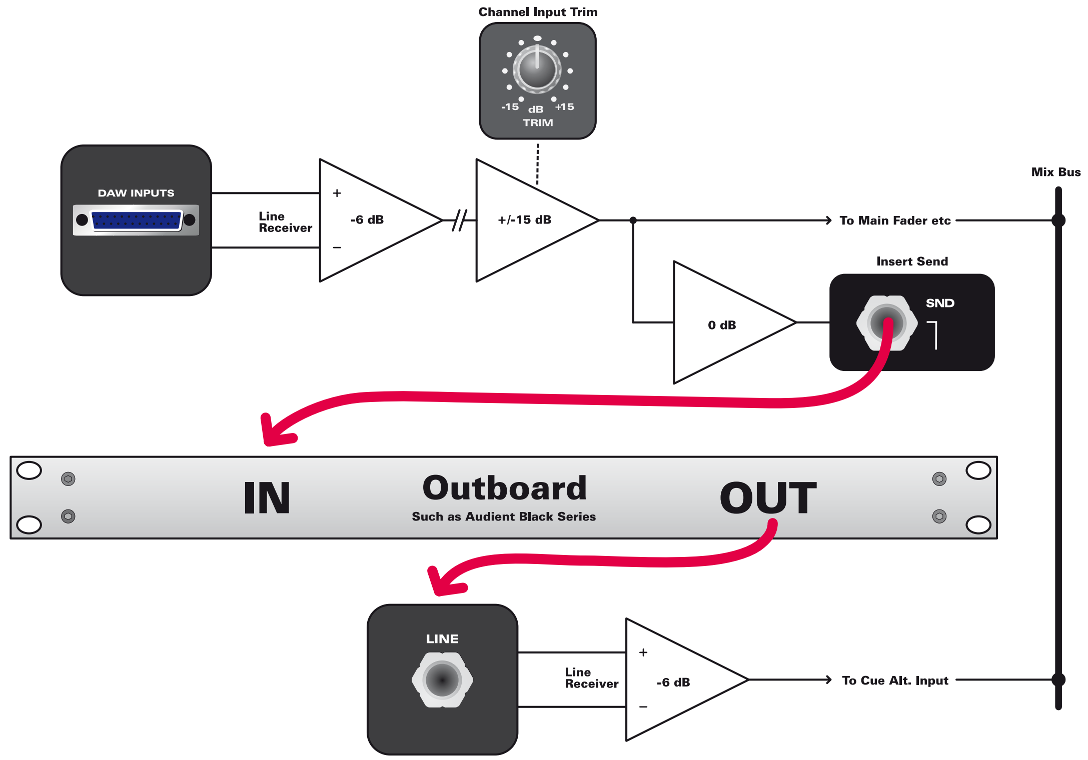
To obtain the most flexibility from ASP2802 consider cabling the console into a patchbay system (this will provide the ability to patch external outboard between your DAW outputs and the DAW input to process signals entering the alternative cue input.

Active Insert Send & Parallel Processing Techniques

Please bear in mind that the insert send is always active - the switch only engages the return path. Thus it is possible to "mult" signals from the insert send to set up parallel compression or feed other external destinations in parallel (a back-up recording system perhaps).

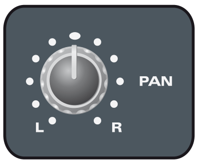
> Try taking a feed from the insert send jack and returning a processed version of the signal (such as a distorted or heavily compressed version) into another channel line input, or for simplicity the alternative input on each channel (depending upon DAW channel flip status this could either be the line input or DAW input.
>
> ✱ TIP!

The diagram below illustrates how to connect such a path - in this example the main channel path input is taken from the DAW input (typical mixing scenario) by depressing the DAW switch. This leaves the line input free for the return path from your outboard device - which can be returned to the mix bus via the cue section alternative input and bus assign.



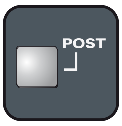
Cue Pan (11)

The cue section pan control can be used to position sources into the stereo field of either the CUE bus or the main MIX bus depending upon the state of the global 'assign to mix cue' button found above the master fader. The pan control provides a smooth taper with an approximate pan law of -4 dB, providing a decent compromise between constant power and constant voltage pan law design ideals.



Cue Post Fader Assign (12)

Engaging this switch feeds the cue section with a post fader signal. With the switch in the UP position a pre fader signal is used.



Primarily it is expected this will be used during mixdown where the cue section may be sent to feed a reverb, in which case you will want the reverb level to drop relative to the main fader position.

During tracking, it is likely you will use the send as a pre-fade feed, so that control room fader balances do not alter cue balances for performers being recorded. In this case the switch should be in the UP position.

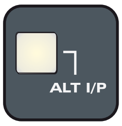
This switch has no effect if the alternative input (14) is engaged (as it overrides the send functionality).

Cue Level (13)

The cue level control can be used as the send level to the cue bus - or can be used as a "2nd" line input channel fader when the alternative input function (14) is used.

Cue Alternative Input Select (14)

Engaging this switch expands ASP2802 with another 8 input channels with access to the main mix bus with their own pan and level controls - effectively increasing the channel count from 8 to 16.



As already mentioned in this manual - there are many uses for this extra input path, from expanding your input quota when summing to returning effects processors and compressors.

Cue Bus Assign - NOTE* GLOBALLY ASSIGNED

In order to fully utilise the alternative cue input as extra channels to the main mix bus, the output of the cue bus has to be re-directed into the main mix bus.

This is globally set and accessed in the master section of the console, just above the master fader.

Aux Level Controls (15) (16)

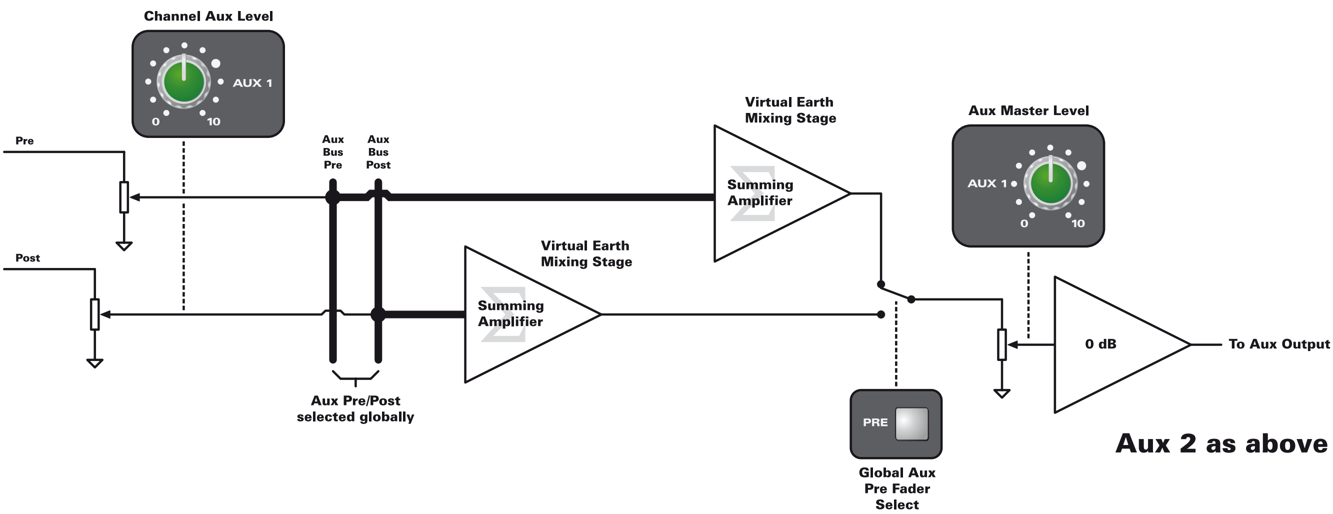
Aux 1 and Aux 2 level potentiometers are used to control the individual send level to each mono auxiliary bus.

Aux Sends Pre / Post Fader Assign - *NOTE GLOBALLY SWITCHED

ASP2802 features two aux send controls - which feed two aux buses (Aux 1 & Aux 2).

The aux send section of the console can be used to feed extra cue mixes and/or reverb units etc during tracking or mixing.

For this reason, as with most consoles - ASP2802 includes the option of a pre fade feed, however please note that the pre-fade and post-fade buses are both fed simultaneously from the send potentiometer. The pre/post switch located in the master section determines which of these buses is fed to the actual auxiliary output, thus pre or post fader auxiliary operation is globally selected here (see section on aux masters for more information).



Monitoring of Aux and Cue Buses - A Helpful Note

✱ TIP!

When tracking it is often faster and more useful to build a monitor mix for musician foldback on your control room monitors.

For this reason it is useful to be able to solo or listen exclusively to the balance being created and fed into any of these buses.

In order to provide such functionality on ASP2802, the aux buses feed not only their own hardware outputs but ASP2802's capable monitor section. Here the Aux 1 and Aux 2 buses can be selected and combined as control room sources, allowing auditioning of headphone mixes direct in the control room, however to monitor the cue bus, it must be summed into the main mix bus via the assign control located above the master fader.

For more information about how to achieve this please read the aux and cue masters section of this manual.

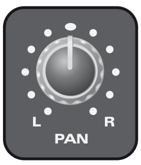
Mix Bus Assign

In order to feed any of the 8 main channels (output of the VCA fader and pan pot) into the main stereo mix bus, it is necessary to engage the mix assign switch located next to the pan pot.

Channel Pan

The pan control provides a smooth taper with an approximate pan law of -4 dB, providing a decent compromise between constant power and constant voltage pan law design ideals.

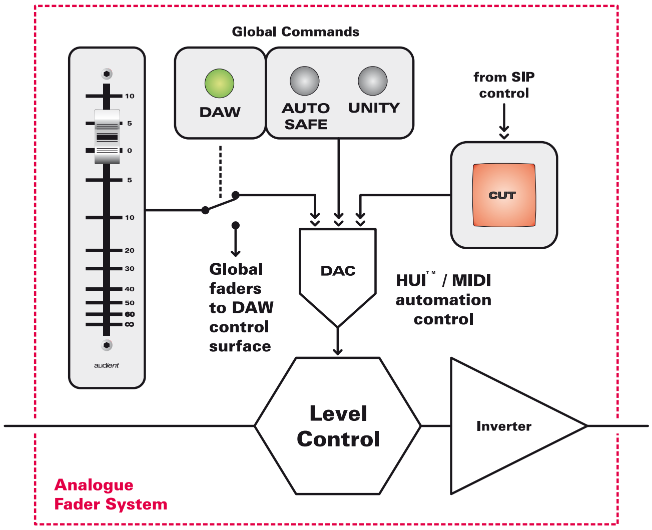
Both the cue and channel pan controls are centre detented for easy mono positioning.

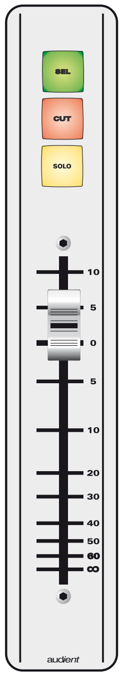


Channel Fader & Control (22)

Each of the 8 main channels on ASP2802 uses a high-quality ALPS 100mm linear fader which serves two purposes. This fader is not in the audio path of the console but rather acts as a control fader, sending D.C. control voltage changes that are translated and utilised by the analogue layer fader system which provides attenuation, mute and +10 dB of gain "in-hand" above unity for the analogue audio path.

The fader can also be used to provide D.C. control voltages which are processed by the on-board microprocessors providing DAW HUI™ control surface functionality and fader automation.





The level control devices used in ASP2802 provide a clean, accurate and repeatable automation platform with very fine resolution and are also used to provide a click-free mute system for each channel.

The level control system used can vary the amount of amplification or attenuation it provides by responding to a modulating control signal. Linear changes in D.C. control voltage - which in this case are delivered by a microprocessor / DAC working in tandem - correspond to logarithmic changes in level (dB) at the fader system output.

Channel Fader Modes

The microprocesser responds to mute / solo-in-place commands, fader position and MIDI automation control data.

There are also several other useful functions associated with the control of the fader that are provided to make sure the user finds this part of the console as friendly and transparent as possible (see page 38).

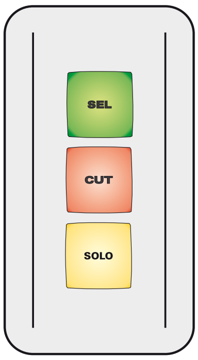
Fader Scales - DAW Specific

Please bear in mind that the scale provided next to the fader is for analogue level indication only and the actual fader scale when in DAW control surface mode is dictated by the host workstation.

For example Logic is +6 dBFS above unity, yet Pro Tools is +12 dBFS - therefore the resultant scale and in particular the range of motion at the top of the fader will differ depending upon which host platform is in use. This is a limitation of the HUI™ protocol.

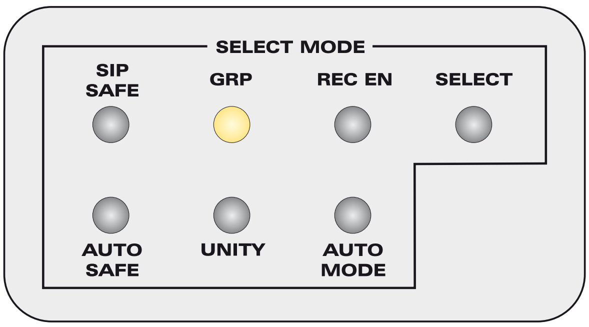
For more information please see the platform specific control surface manuals, found online at **www.audient.com**.

Select Switch & Select Modes (19)



Above each fader are three illuminated push switches.

The first switch, SELECT (green LED) can be used for several functions on both analogue and DAW layers when used in combination with the select modes available in the middle of the control surface panel.

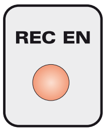


Select Switch & Select Modes cont.

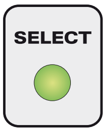
There are seven select modes available on ASP2802 (see below), of these seven, three are used in the analogue layer:

Solo-In-Place Safe - used in the analogue layer for isolating channels that you do not want muted when using solo-in-place mode - reverb returns for example.

Group - currently only used within DAW control surface mode for grouping tracks.



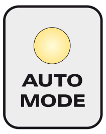
Record Enable - only used within DAW control surface mode for arming tracks.



Channel Select - currently provides channel select functionality in DAW control surface mode.

Automation Safe - used in the analogue layer for isolating channels from any automation control data, ideal for trying out a new ride before recording the pass of automation.

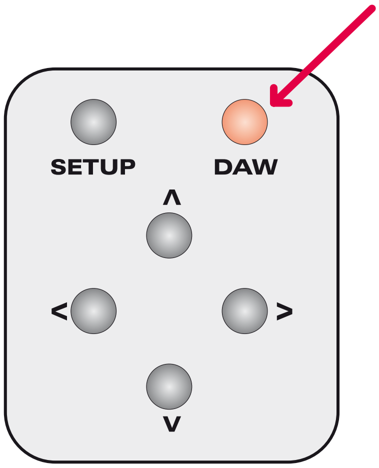
Unity - used in the analogue layer to provide a quick and easy way to position the channel fader at the unity gain (0 dB) position. This is very useful if you are setting up a stem session for analogue summing etc.



Automation Mode - although used to control the analogue automation system, this control is used in the DAW layer to set host specific automation modes like read, touch and latch etc as ASP2802 uses the DAW automation system and MIDI to control the on-board fader system.

To use these functions press the function required in the select mode panel and then press the large green channel select switches (19) on each and every channel that requires the application of the function selected.

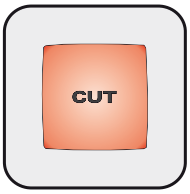
Cut & Solo Controls



Each channel features large illuminated cut and solo switches above the fader. These cut and solo switches provide control functionality in both the analogue and the DAW layer so be aware of which layer you are controlling by observing the status of the DAW layer switch in the control surface panel. For more information on DAW control please see the separate platform specfic manuals available online at **www.audient.com**.

When the DAW layer button is not illuminated, you are operating in the analogue layer.

Channel Cut (20)

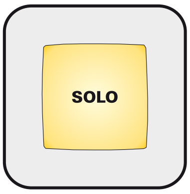


The cut switch is used to provide click-free silent muting in conjunction with the channel fader control system. Mutes can be automated from a MIDI controller capable host DAW - for more information concerning this matter please see the separate automation & control surfaces manuals provided online at **www.audient.com**.

The channel cut is also activated when using the solo-in-place system, allowing channels to be soloed without routing them to the solo bus. This effectively mutes all other channels that are not in solo - therefore preserving all bus routing and access to main mix insert processing etc.

If you wish to isolate the channel cuts so that specific channels do not mute when another is soloed use the solo safe feature in the select mode panel.

Channel Solo (21)



The solo switch on each channel places the channel into SOLO - either AFL or PFL (after fade or pre fade listen) or Solo-In-Place depending upon master section settings.

For more information concerning these solo modes, please see the section in this manual outlining the monitor section functionality (page 50 onwards).

Summing Input & FX Returns

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ASP 2802

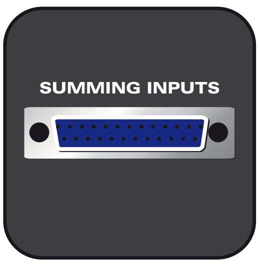
DUAL
LAYER CONTROL

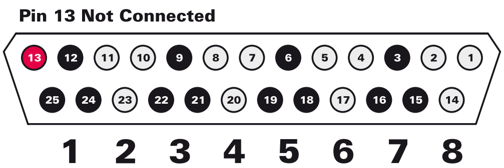
Summing Input

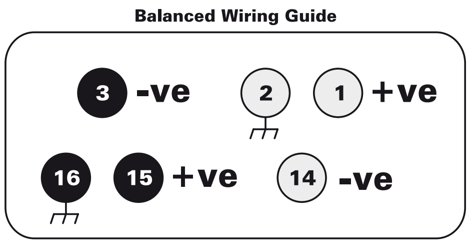
Instead of providing just the common stereo effects returns for processing devices such as reverbs and delays to join the mix, ASP2808 features a novel 8-channel summing input as well as a pair of stereo FX returns.

This summing input is fed from a rear panel DB25 connection that conveniently allows easy integration with a D-SUB based patchbay system or any DB25 capable device including many AD/DA converters that are available on the market.

The summing input conforms to the Tascam DA-88 wiring specification for 8-channel balanced analogue 25-pin D-Sub connectivity.







Channel Number	DSUB Pin Number		
	+ve	-ve	Shield
1	24	12	25
2	10	23	11
3	21	9	22
4	7	20	8
5	18	6	19
6	4	17	5
7	15	3	16
8	1	14	2

Shield connections are bussed and grounded to the chassis within the unit

This 8 channel balanced input is configured as four stereo pairs feeding a stereo mix bus. Therefore channels are paired such as: Sum1L - Sum2R, Sum3L - Sum4R etc.

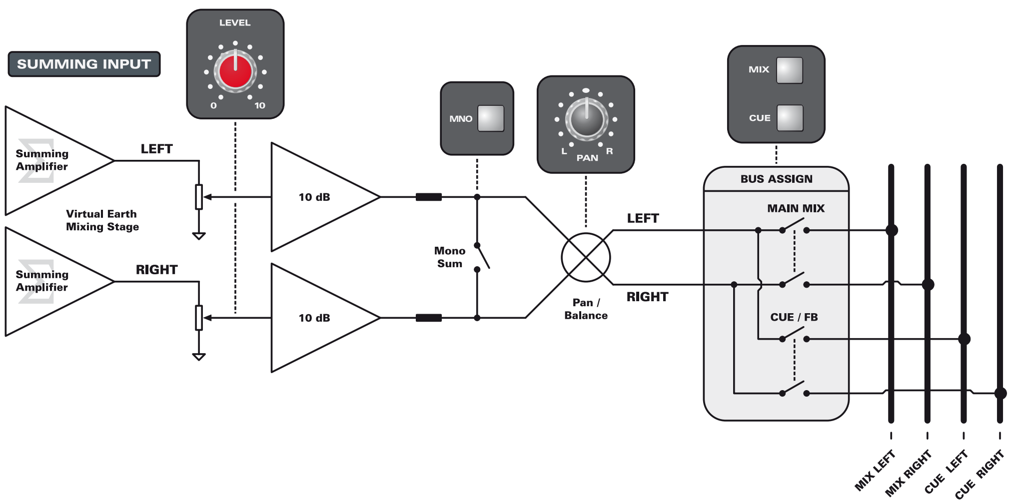
Although the 8 channels are summed as four pairs - provision is made to mono the output of the summing input mix bus before it reaches the main ASP2802 mix bus, therefore providing a further 8 mono inputs upon mixdown.

This summing input would be very useful in scenarios such as the following:

- Bringing 4 extra stereo submixes into ASP2802 from your DAW utilising the HUI™ control surface functionality to create said submixes.
- Bringing in 4 stereo inputs from external synthesisers and samplers and feeding to the cue section for practice sessions.
- Returning the analogue outputs of 4 stereo or dual mono FX processors which you may send to digitally from your DAW using your workstation auxiliary sends (typically for riding into vocal delays and reverbs etc) - assuming said processors have an output level control so you can balance into the summing bus.

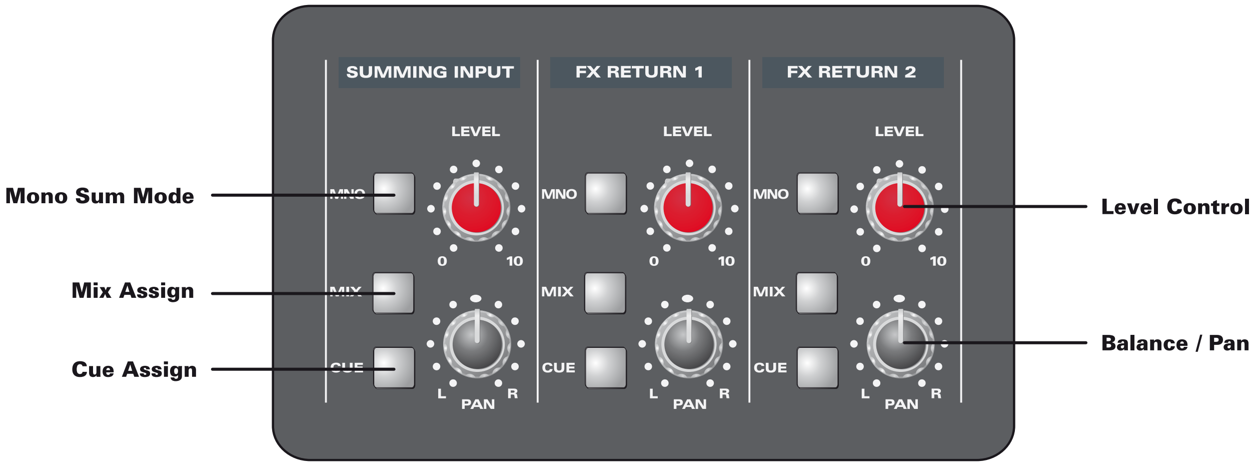
The summing input section provides overall level control for the combined 8 channels, a stereo balance or pan control, mono sum mode and assigns for both the cue and mix buses.

Therefore any external effects processing can be added to the main mix or monitored in the cue section.



Alongside the summing input, there are a further two stereo fx returns, providing a total of 12 inputs in this section of the console.

These are fed from rear panel 1/4" TRS balanced jack connections and are most likely to be used when feeding mono in / stereo out reverb units from aux sends 1 and 2 and returning them here.



Again as with the summing input, the two stereo fx returns offer level control, balance or pan (depending upon stereo or mono sum operation), mono sum and assigns for both the mix and cue buses. Therefore it is feasible to place your reverb returns into artist headphone mixes ensuring a comfortable performance as many prefer a little reverb for inspiration and some singers may hold pitch better with this added ambience.

The rear panel inputs can be located here:

Aux & Cue Masters

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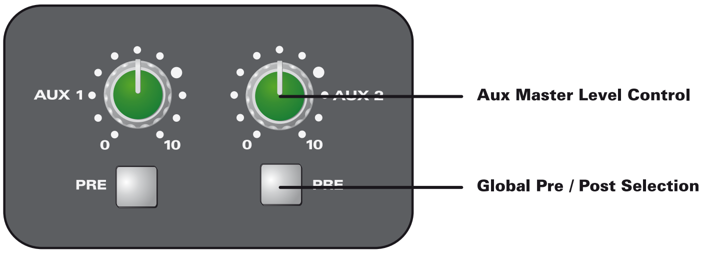
Aux Masters

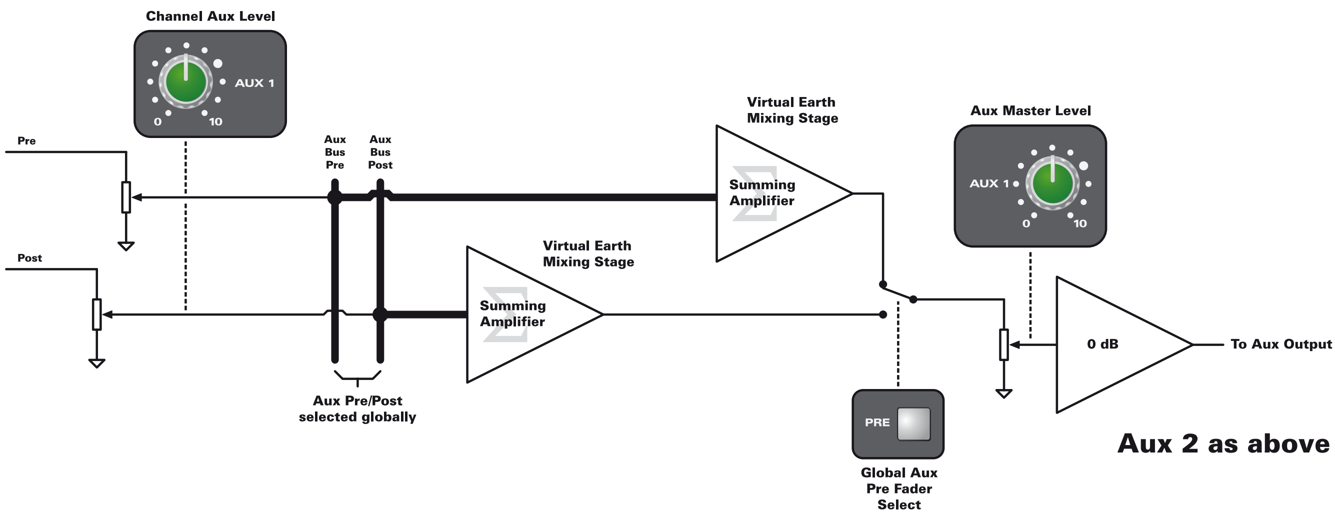
ASP2802 has two aux buses and a simple set of controls are provided for the master output of these buses.

The two aux masters feature a level control - useful for that final trim of output level, especially if reducing overload in some vintage digital reverb processors that may be connected here.

Each of the two aux masters can be globally switched to pick-up either the pre-fade or post-fade auxiliary buses from the channels.



The following flow diagram illustrates how the signal arrives at the aux master bus from each channel and how the global selection works at the aux master location.

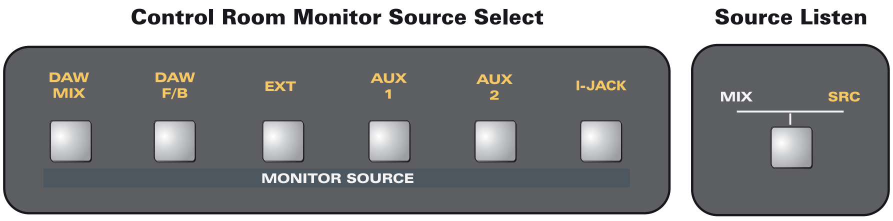


The signal leaves after the master level control via the rear panel balanced male XLR outputs.

Monitoring Aux Buses

The aux buses feed the monitor source selection panel in both the cue and control room monitor sections. Therefore headphone mixes can be created via aux sends 1 & 2 and can be sent to performers directly via the cue source selections. They can also be monitored in the control room to build the foldback mixes by ear (as opposed to performer direction) and could aid you in quickly auditioning a send to a reverb or parallel compressor.

In order for this to be achieved please select the desired source in the monitor source panel. If you wish to monitor one of these sources in the control room make sure you depress the MIX/SRC switch to allow the source selection to feed the monitor path of ASP2802. Please be aware that these sources sum when more than one is selected.

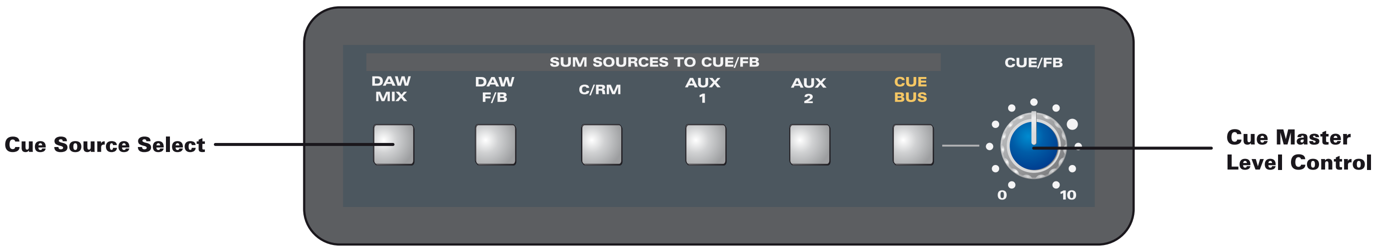


Cue Master

ASP2802 features a very flexible cue section located in the top right or the master section.

The CUE/FB section allows you to select whatever sources you want to feed the main cue outputs on the rear of the console (XLR Male).

The cue bus is naturally an option here and must be selected, but so are a number of other sources - again please be aware that these sources sum when more than one is selected.



Cue Master cont.

There are two external stereo inputs - DAW MIX and DAW FB (foldback).

These two stereo inputs (found as rear panel line level XLR inputs) are intended to be used to monitor either the mix output of your DAW or a separate stereo foldback mix from your DAW.

This is useful as headphone mixes are entirely possible at low latency with newer computers and low latency DSP devices. As such your headphone mixes can be saved within your session and even include effects for monitoring purposes.

ASP2802 provides the perfect hybrid platform as the control surface layer can be used to balance these mixes in superfast time and the Dual Layer Technology allows you to seamlessly flip between your DAW submixes and the analogue layer of ASP2802.

Along with the two external DAW inputs (both of which are also suitable for use as two-track returns) - the cue master section offers the ability to send the output of the AUX 1 and AUX 2 master summing stages to the cue outputs.

This is useful if you are using the last option - C/RM (control room) as your main cue mix. A scenario that may often be encountered when you require that last absolutely essential vocal overdub during the final stages of mixing.

If you are using all input channels including the alternative cue input for mixdown, you may decide to send the artist your control room balance. This can be achieved by engaging the C/RM switch. In this case you may use an aux send on a channel to send the artist more of themself without affecting your control room balance.

To do this, just engage both the C/RM and whichever aux bus you are using (AUX1 or AUX2).

Monitoring the Cue Bus

The rear panel XLR male balanced cue outputs of the console are fed from whichever sources are selected in the cue panel. In order for the cue sends of each channel to reach the cue outputs, and therefore your performers headphone distribution system, you must select the cue bus as a source in the cue panel.

However, in order to monitor the cue bus in the control room to create headphone mixes by ear you must feed the cue bus into the main mix bus so that you can hear any changes being made to the cue balance.

You will also likely need to de-assign the 8 ASP main channels from the mix bus so that you do not sum the same signal twice into the mix bus.

The CUE/FB sources can also be monitored directly via the built in headphone output located on the rear panel of ASP2802.

This headphone output can be used by the engineer to audition cue mixes on headphones (the most likely playback medium for the performer), therefore allowing cue mix balance judgements to be made in the confidence that they will translate to headphones. After all, a great cue mix ensures comfort in the studio, which directly translates into a better performance.

The ability to assign the CUE/FB output to the headphones also benefits a single operator / performer project studio environment, where the artist and engineer are the same person!

Also if tracking in a space that lacks a lot of iso-booths, placing a singer or guitar player in the control room often allows you to lay guide tracks along with drums etc while maintaining some degree of isolation.

Monitor Section

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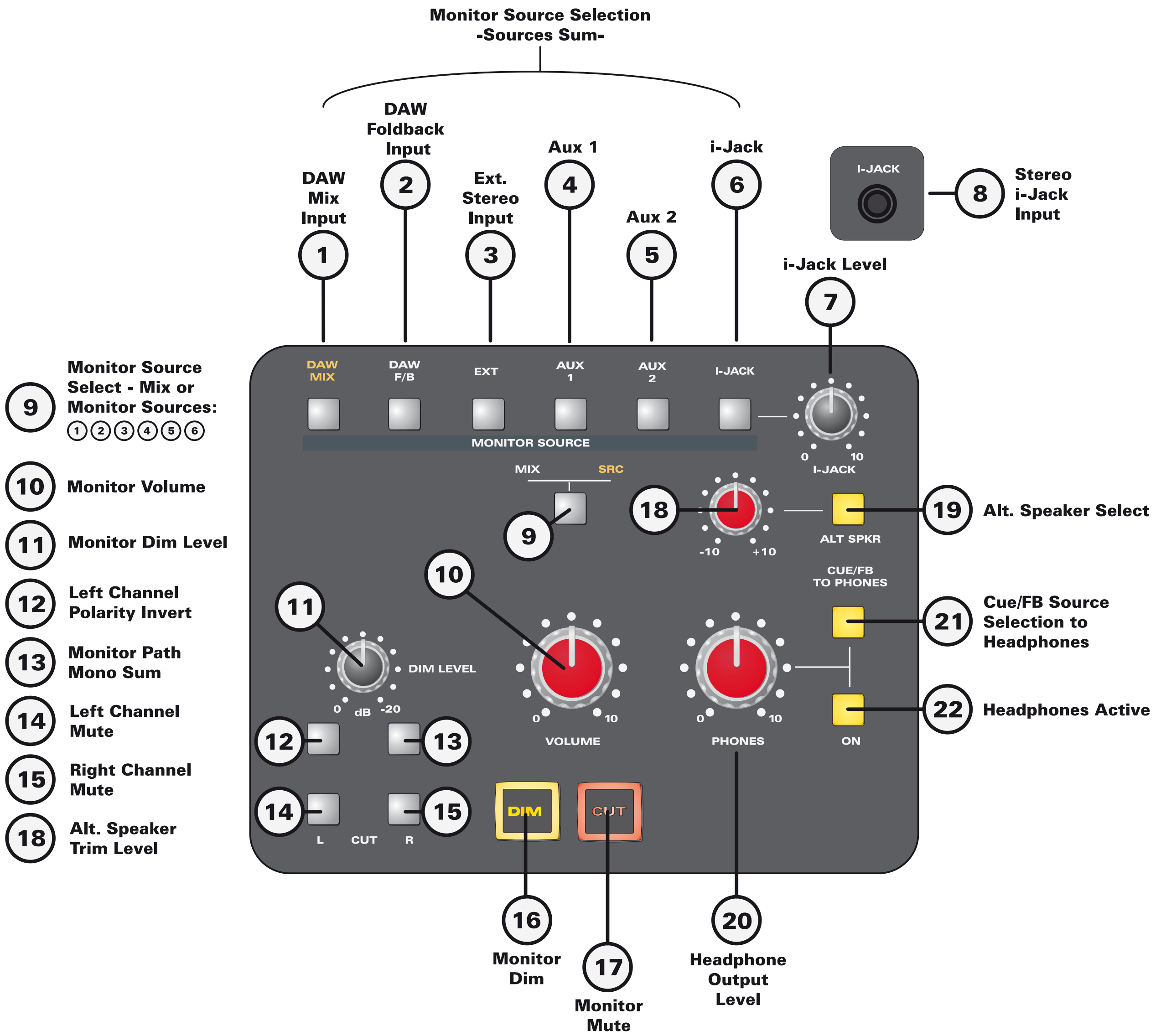


✱ TIPS & SUGGESTIONS

Use the DAW Mix input to quickly audition an ‘in the box’ rough mix or monitor the output of your DAW easily.

Use the DAW Foldback input to return ‘in the box’ artist headphone mixes to the console. Audition the balances by activating the DAW FB switch and selecting SRC via the Monitor Source switch (9). This can provide you with full recall of cue mixes saved inside your DAW sessions. Beware of latency though.

The external stereo input provides a way to return 2-track devices such as a mastering tape machine, CD player, DAT or digital recording system.

Use the conveniently located front panel mini jack socket to audition portable MP3 and personal A/V players. Very useful for checking rough mixes, MP3 mix translation or reference material.

However we recommend the use of high resolution lossless audio as reference material.

Use polarity invert and mono sum to check for mono compatibility and phase issues. Polarity invert combined with mono sum facilitates difference (sum & difference) monitoring, great for checking stereo content.

Cut L and Cut R ensure you can monitor mono correctly on one speaker - mono is more accurate coming from a single source as it minimises phase cancellation in the listening environment.

Alternate monitor outputs provide you with the option of checking mixes on various playback systems.

Audition cue balances for performers using the built-in beefy headphone amplifier and CUE/FB to Phones switch (21).

The monitor section of ASP2802 provides comprehensive control room monitoring options.

The circuitry is mostly passive with only one buffer stage before the balanced loudspeaker outputs - thus providing you with a tonally neutral and accurate monitor path.

There are a number of features that are very useful for checking the translation and content of your mixes.

Please refer to the diagram and numbering on the previous page when reading this section.

Source Selection 1–6

There are 5 sources that can feed the monitor section of 2802.

DAW Mix - a stereo line input found as balanced XLR connectors on the rear panel. This stereo input can be used to audition "in the box" mixes quickly and can also be fed into the main mix bus for more inputs at mixdown time! Useful!

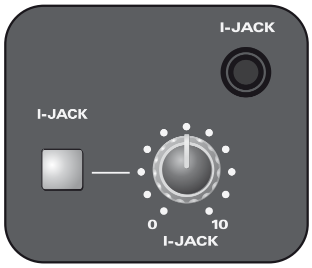
DAW Foldback - a 2nd stereo line input found as rear panel balanced XLR connectors. You may find this input very useful for auditioning headphone mixes that are created in the DAW.

EXT. Stereo - a 3rd dedicated external stereo input found as balanced rear panel XLR connectors. Intended for use as a two-track return for mixdown devices such as DAT machines, 1/2" and 1/4" tape machines and other devices.

AUX 1 & AUX 2 - an internal feed from both auxiliary masters, use to audition headphone mixes or balances into send effects processing.

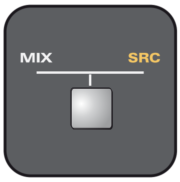
i-JACK - a front panel located stereo mini jack connector (8). Designed for use with portable music players and the like - great to audition rough mixes as MP3 etc. A level control is available for this input also (7).

I-Jack Level Control & I jack Input 7 – 8



A stereo level control for the i-Jack input. Note that this is a stereo unbalanced input with approximately 10k ohms input impedance.

Monitor Mix / SRC Select 9



The monitor source switch configures the monitor path to listen to either the output of the main mix bus or whatever is selected in the monitor source panel (1-6).

Main Monitor Volume 10

A stereo level control for setting the main listening level. You may find it useful to calibrate your listening environment such that a particular location on this potentiometer corresponds to a loudness level from your studio monitors that is not harmful or fatiguing for long listening periods. This should also ensure the human hearing system is listening at a level where our natural frequency response is flattest.

Monitor Calibration

By setting your studio monitor amplifers to produce approximately 83 to 85 dB_{SPL} for a known and typical source level - it is possible to achieve a listening level that should ensure the ears are operating at their flattest frequency response due to aural perception and equal loudness contours.

We recommend that you use the DAW MIX input and feed a -20 dBFS pink noise signal out from your DAW (or other suitable noise source), and set the input level such that it sits at 0VU.

If you have the choice, use a calibrated SPL meter set to read with a slow response time, RMS detection and C-weighted filter.

Monitor Calibration cont.

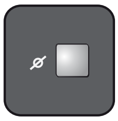
C-weighting is the most wideband weighting filter and most similar to the human ear at 85 dB_{SPL}. Place the microphone in the listening position and adjust the monitor level pot until you read 85 dB_{SPL} from each loudspeaker. By returning to this level on program material you have a good chance of obtaining a constant and accurate reference level.

Dim Level (11)

The monitor section includes a user-specified dim level that will lower the level of the output feeding the speakers when the dim function is engaged (17). Dim can be set anywhere from 0 to -20 dB

Polarity Invert (12)

Engaging the polarity invert function inverts the polarity of the left channel of the mix, allowing you to check for stereo “difference” content when summed to mono, essentially creating a L-R instead of a L+R sum. The switch is momentary to ensure it is not left on by accident!

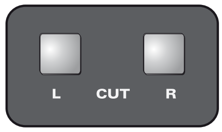


Mono Sum (13)

Engaging this function sums the left and right signals in the monitor section to produce a mono output.

This is a useful feature for checking any issues that may arise with out of phase stereo content (especially at low frequency) and mono compatibility. It is also very useful to engage during the microphone placement process (when tracking), by adjusting stereo pair microphones in mono - until they provide the best tonal balance - thus hopefully minimising any phase cancellation when switched back into stereo.

Cut L and Cut R (14)–(15)



Individual left and right output mutes can be engaged here - perfect for checking mono sum on a single speaker. This is often the most accurate method as it minimises the acoustic interaction from two (likely non-perfectly matched) loudspeaker sources.

Dim Select (16)

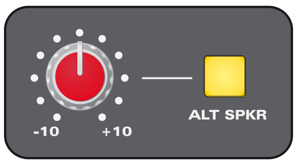
Engaging this switch dims the loudspeaker output by an amount set by the dim level control (11).

Please note that the DIM function is also engaged automatically when utilising the on-board talkback communication system to prevent howl-round feedback loops.

Main Monitor Cut (17)

A global main monitor cut that disconnects both the main and alternative loudspeaker outputs.

Alt. Monitor & Output Trim Level (18)–(19)



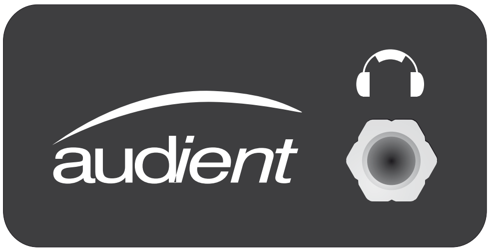
ASP2802 features two monitor outputs - typically used to feed a nearfield and either a mini 'grot-box' style monitor set, a large farfield arrangement or public address type system for checking 'club' bottom end!

The alternative monitor output also features a trim control (overall volume is still determined by the master volume control (10).

These can be used to match levels or set comfortable levels across both loudspeaker sets.

Headphone Output (20)—(22)

There is a high quality stereo headphone output available on ASP2802.



The level control (20) is located on the front panel in the monitor section and you will find plenty of level available here for driving most studio headphones to sufficiently loud SPL levels.

The headphone output is typically fed from the monitor section of the console and thus its source is determined by the source select panel found in the monitor section of the console.

However the headphone output can be fed alternatively with the Cue / FB signal by depressing the Cue/FB to Phones switch (21).

Please note that the cue signal provided here is determined by the cue source select panel and therefore can be made up of several different combinations if needed.

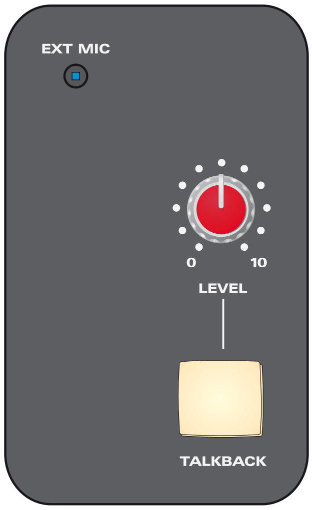
The headphone output can be engaged with the ON switch (22) located to the right hand side of the monitor section, which is also where the headphones level control is found.

Monitor Section

The master section of ASP2802 also contains all communication functions for use during tracking and overdubs.

Talkback

An integrated talkback microphone can be found in the master section above the power indicators and to the right of the main mix bus metering. This built-in electret microphone when activated feeds the cue section of the console and thus the hardware cue outputs.



There is a talkback level control which sets the gain of the talkback microphone preamplifier - which has a gain range of +20 to +55 dB.

An external talkback microphone can be selected with a separate XLR input found on the rear panel.

To select this microphone and override the small electret in the compressor panel, engage the small recessed switch located above the talkback level control by using a long thin non-conductive object to depress the switch.

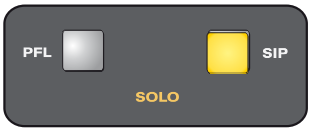
To engage 48v d.c phantom power for the external microphone, engage the small recessed switch found next to the XLR on the rear panel.

The talkback switch is momentary to avoid leaving it on during delicate discussions! When activated the talkback switch will also activate the dim function in the monitor section to prevent feedback.

External Talkback Microphone Input

Solo Functionality

ASP2802 has four modes of solo which can be globally set from the master section of the console. When a channel is placed into solo, the SOLO indicator will light in the master section.



AFL - After Fade Listen

This mode is the default setting when the PFL switch is in the UP position. Solo feeds are taken after the channel fader and pan. The signal is sent to the stereo AFL solo bus and the monitor section switches to listen to this bus.

PFL - Pre Fader Listen

This mode is selected by depressing the PFL switch in the master section. Solo feeds are taken before the channel fader. The signal is sent to the mono PFL solo bus and the monitor section switches to listen to this bus.

SIP - Solo In Place

This mode does not use the solo bus to listen to channels in solo, instead the centre section returns a d.c control voltage command back to the channel mutes. Using this method engaging solo on a channel still uses the main mix bus as the listen path, muting all other channels that are not in solo. This is useful as channels in solo will still pass through mix bus insert processing etc.

Also this mode can be combined with the solo safe function found in the control surface select mode section. Solo SIP safe enables channels to not be muted during SIP activity - great for preserving vocal reverbs etc.

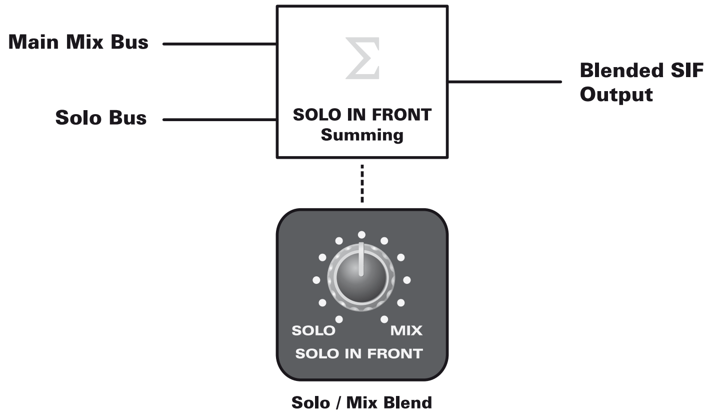
Solo Bus Level Trim

The solo bus level trim provides a +/-10 dB trim range to enable you to control any level drops or jumps that may occur when shifting back and forth between full mix and soloed channels.

SIF - Solo In Front

This mode is very useful as it allows you to listen in isolation or with some of the main mix blended in.

This ensures that you keep a reference point in ear shot at all times. The SIF control is a continuously variable potentiometer that blends the main mix bus back in with the solo bus.

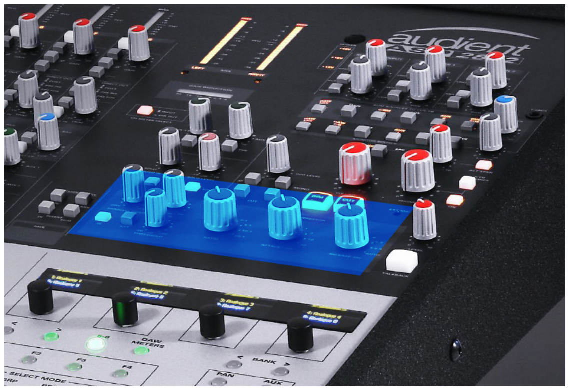


Please note that SIF does not work in SIP mode.

VCA Bus Compressor

audient
ASP 2802
DUAL
LAYER CONTROL

Bus Compressor



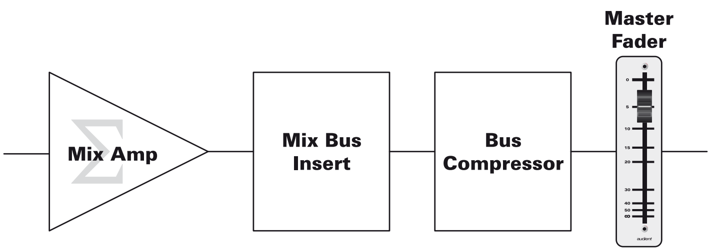
ASP2802 features an integrated high class, patchable VCA bus compressor. It is located in the master section of the console on the right hand side underneath the monitor section.

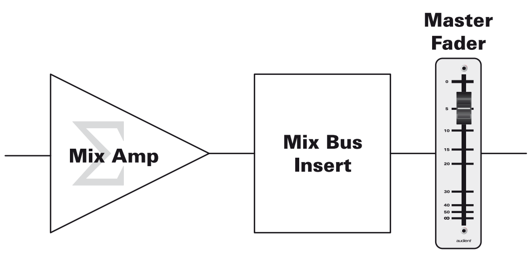
An adaptation of the compressor featured in the ASP8024 large format console, this is a soft-knee VCA design optimised for mix bus use.

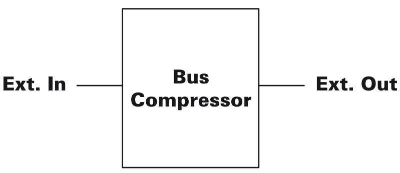
However with the addition of a wet/dry blend control for parallel compression and external patch capability, this compressor really works well on the drum bus (among other uses which we are sure you will find).

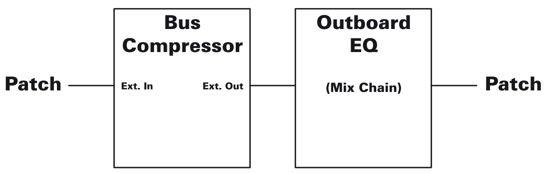
Please note that by default it is located after the mix insert points. However, it can be externally patched and therefore removed from the mix bus path and freely assigned to any new destination via its own balanced rear panel I/O.

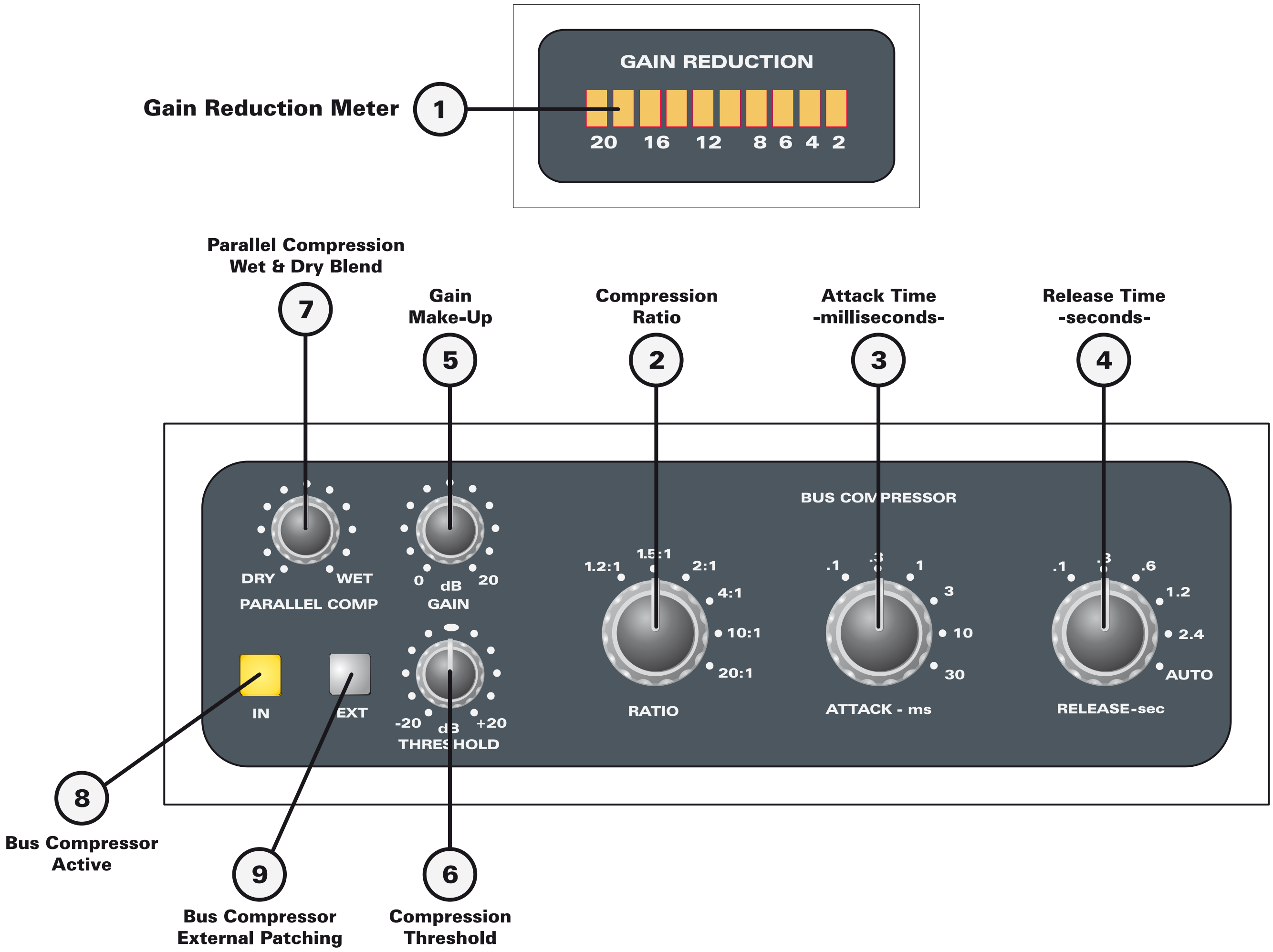
This way it is entirely possible to create a chain at the mix bus insert point with the VCA compressor followed by or preceeded by an outboard equaliser (a popular choice!) Alternatively, you could deploy the bus compressor on a vocal subgroup across a pair of ASP2802 channels.











✱ TIPS & SUGGESTIONS

The bus compressor is a VCA (voltage controlled amplifier) gain cell design and can be set to gently glue your main mix or stun your drum bus - a flexible tool that can be used on a variety of sources.

For that very reason, we have provided you with the ability to externally patch the bus compressor to its own input and output jacks located on the rear panel.

Perhaps you dislike overall mix compression and would rather leave it to your mastering engineer, or perhaps you have your own favourite outboard mix bus processor.

The ASP compressor can be employed anywhere you see fit - individual channels or a stereo group - the choice is yours.

With the addition of a wet and dry compression blend it is possible to create parallel compression New York style.

Try this on your drum bus, it is fantastic at preserving your transient information while adding punch and body.

When using the wet and dry blend, you may find that you do not need to use as much make-up gain as there is a slight amount of gain in the sweep of the pot.

This makes for a great way to increase output level post compression and achieve a nice sense of punch!

We have found a great starting place for a wet/dry blend is around 1 to 4 o'clock towards wet.

Adjust make-up gain to taste!

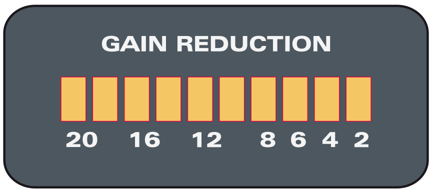
Please see the example settings page in the manual for guidance.

Please also bear in mind that these are only examples and, due to the dynamic nature of program material, will require adjustment to achieve their best performance on your material.

Bus Compressor

Gain Reduction Meter 1

A 10-segment LED bargraph meter indicating the amount of gain reduction. Due to the scale of the meter, sometimes you will only want the first few LEDs flickering to keep compression to musical levels on the mix bus.



The scale on the gain reduction meter ranges from 2dB to 20dB of compression.

Compression Ratio 2

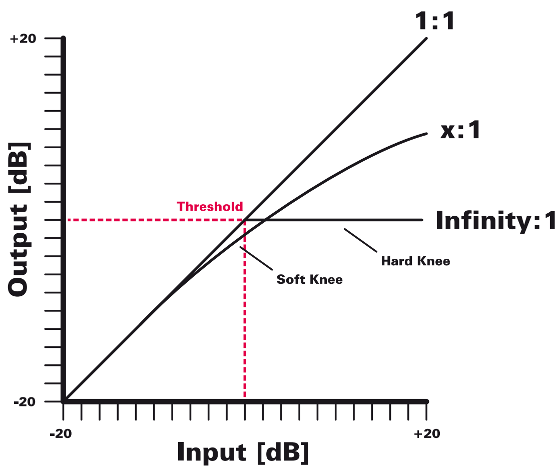
Specified as a ratio of input vs output (in:out), the ratio control sets how much compression takes place once a signal has crossed the threshold point set in the sidechain circuitry.

A ratio of 2:1 indicates that when an input signal crosses the threshold by 2 dB, an increase of only 1 dB would be present at the output, thus 1 dB of gain reduction would be produced.

Generally, low ratios such as 1.5:1 and 2:1 are used over the main mix bus, but feel free to experiment. As the ratio increases, the compressor acts more like a limiter (10:1 and 20:1) and can be very useful or powerful on dynamics sources such as drums or vocals. The soft knee design provides a gradual entrance to compression from below the threshold, and is very smooth.

Ratios can be set to:

- 1.2:1 (very gentle)
- 1.5:1
- 2:1
- 4:1
- 10:1
- 20:1



Attack Time (3)

Specified in milliseconds (ms) - this is the time taken for the compressor sidechain to react to the input signal above the threshold and trigger gain reduction (compression).

Attack times can be set to:

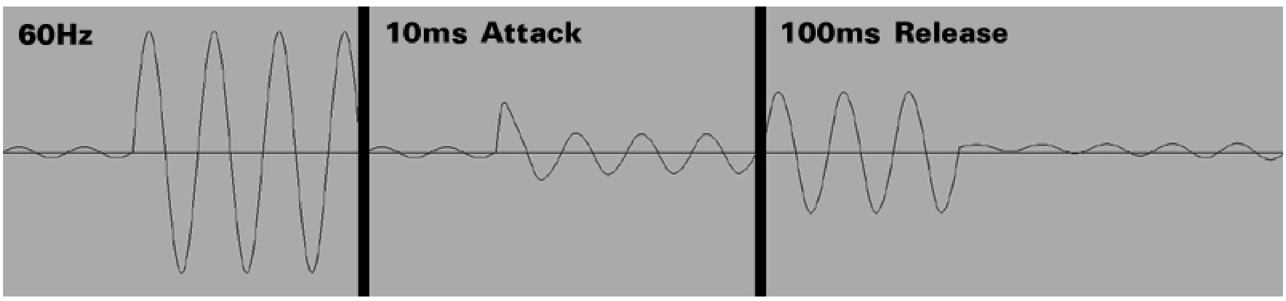
- 0.1 ms
- 0.3 ms
- 1 ms
- 3 ms
- 10 ms
- 30 ms

Typically on program material and mix bus use, slower attack times (try 10 or 30 ms for a punchy starting point) are preferred as they will not mangle transient material or distort low frequency content as much as faster times.

If the attack or the release time is faster than one cycle of the lowest frequency to be reproduced, distortion will occur as the compressor will alter the waveshape during one cycle.

Remember that the period of one cycle, T(sec) = 1/f (Hz).

The following diagram illustrates typical waveshaping at 60Hz (approximately 17ms period) as created by relatively fast attack and release times (10ms attack, 0.1 sec release at 4:1). Note that aside from some attack waveshaping, the ASP2802 compressor is very clean!



Release Time 4

Specified in seconds (s) - this is the time taken for the compressor sidechain to release from gain reduction once the input signal has passed below the threshold.

Release times can be set to:

- 0.1 sec (100 ms)
- 0.3 sec (300 ms)
- 0.6 sec (600 ms)
- 1.2 sec
- 2.4 sec
- Auto Release

Typically faster release times are required when you want the compressor to get out of the way "quickly" which can lead to pumping and very obvious compression.

On vocal bus or piano, longer release times and auto release are very useful. However if you want colourful compression on your drum bus, faster release times are worth trying. Start at 0.1 sec and work backwards until you get a suitably balanced effect.

Auto release utilises an adapative circuit that alters its release time in a program dependent manner. It is fantastically smooth and well suited to mix bus duty and also excels on vocal duties.

Gain Make-Up 5

With a range of 20 dB, the gain make-up control can be used to restore program material level after compression, thus reducing the dynamic range but preserving the overall output level.

Bus Compressor

Compression Threshold 6

The threshold control is specified in dB and sets the point at which compression begins.

There is a range of +/-20 dB on this control.
This compressor has a relatively smooth knee but the threshold point is similar for each ratio, making it predictable to set an operational threshold and then audition ratios without re-adjusting.

Wet / Dry Parallel Compression 7

Parallel compression has been a popular mix technique that allegedly started in New York. Globally adopted and adapted, parallel compression is a fantastic way of preserving transient punch whilst simultaneously reducing dynamic range, through a form of upward expansion.

✱ TIP!

We have added this functionality to the ASP2802 bus compressor where there is a dedicated control, however this can also be achieved on the mix bus insert point by using the insert sum mode which places the insert in parallel with the dry signal.

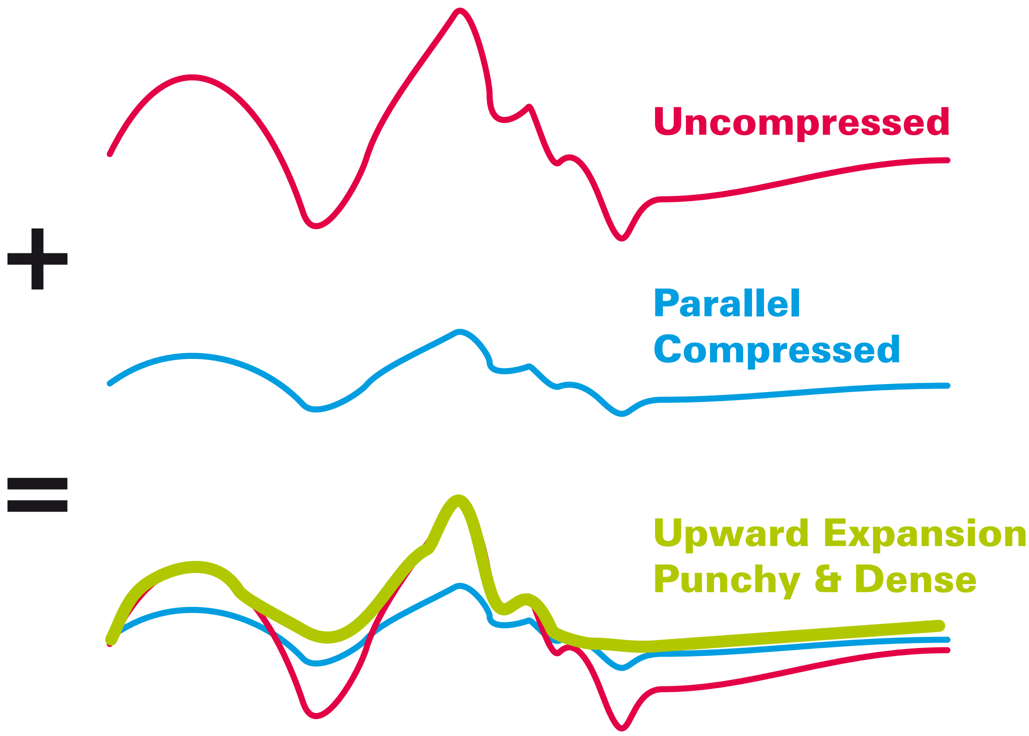
The wet and dry blend control on the bus compressor allows you to “crossfade” between the uncompressed and compressed signal. A very useful feature which allows more aggressive compression to be used without squashing all of the life out of the mix - parallel compression = **energy.**

There are two useful starting methods when using parallel compression.

One would be to set some gentle compression (low ratio) up 100% wet and then just edge back with a small amount of dry signal to add some punch back.

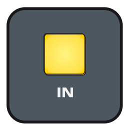
Wet / Dry Parallel Compression cont.

The other school of thought involves setting up some rather aggressive compression (faster attack times and high ratios) and running the dry balance quite high so the smaller amount of compressed signal adds energy and dynamic colour.



Compressor In 8

This illuminated push switch engages the compressor and also the gain reduction meter, so gain reduction can only be viewed when compression is active.



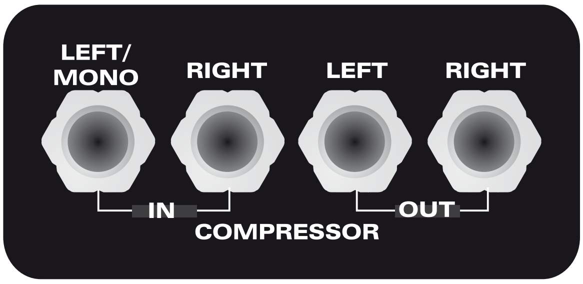
External Input 9

The compressor can be patched to its own rear panel balanced I/O, thus removing it from the main mix bus path and freeing it for use elsewhere in your mix.

The rear panel compressor I/O can be found in the bottom left hand corner of the console if facing it from the rear.

External Connections

The compressor can be fed from mono sources if so desired, just connect into the left input and output.

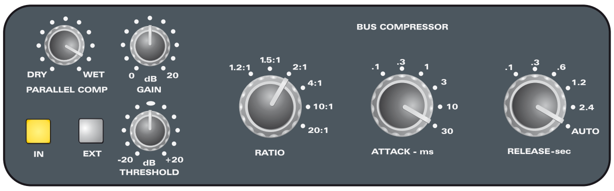


External Compressor I/O

Please see the next page of this manual for some example settings, but remember - your ears are the best monitors!

Bus Compressor Settings

Mix Bus



Adjust threshold for 2-3 dB G.R
Adjust make-up gain accordingly

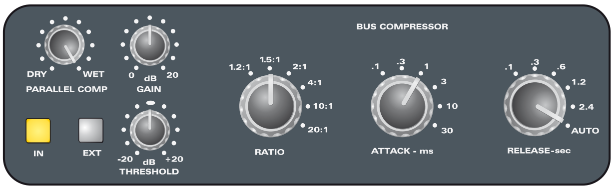
Drum Bus

Adjust threshold for 6-8 dB G.R
Adjust make-up gain accordingly

Drum Squash

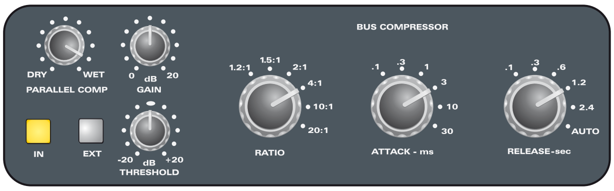
Adjust threshold for 8-10 dB G.R
Adjust make-up gain accordingly

Vocal Bus



Adjust threshold for 4-5 dB G.R
Adjust make-up gain accordingly

Percussion Bus

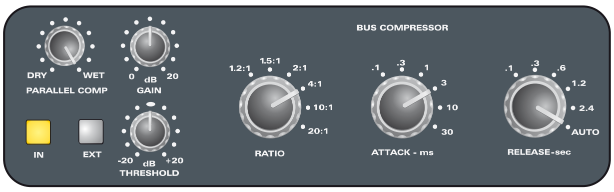


Adjust threshold for 4-5 dB G.R
Adjust make-up gain accordingly

Parallel Lift

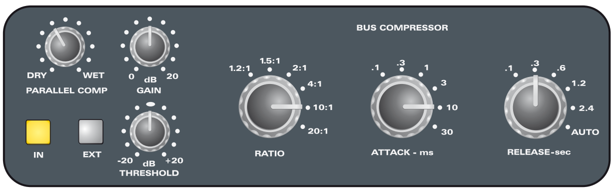
Adjust threshold for 6-8 dB G.R
Adjust make-up gain accordingly

Brass or Electric Guitar Bus



Adjust threshold for 4-8 dB G.R
Adjust make-up gain accordingly

Crazy Smash - Electronica



Adjust threshold for >10 dB G.R
Adjust make-up gain accordingly

Main Mix Overview

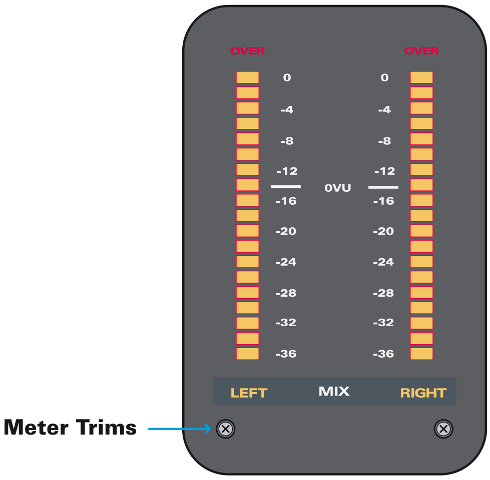
audient
ASP 2802
DUAL
LAYER CONTROL

Main Mix Bus

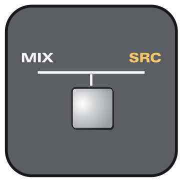
The main mix bus and all signal combining stages on ASP2802 are optimised for clean summing with high headroom.

Mix Bus Metering

To gauge this headroom, large 20-segment LED peak meters calibrated to 0VU = +4 dBu = -14 dBFS are provided on the main mix path output. These meters provide 36 dB of range.

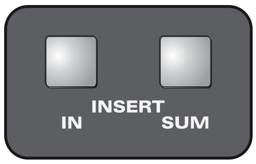


Please note these meters follow the selection made with the Mix / SRC switch in the monitor section.



Mix Insert

The mix bus features a switchable insert point. The insert send is always active regardless of switch state, and therefore can be used a pre master fader pick off point if desired.



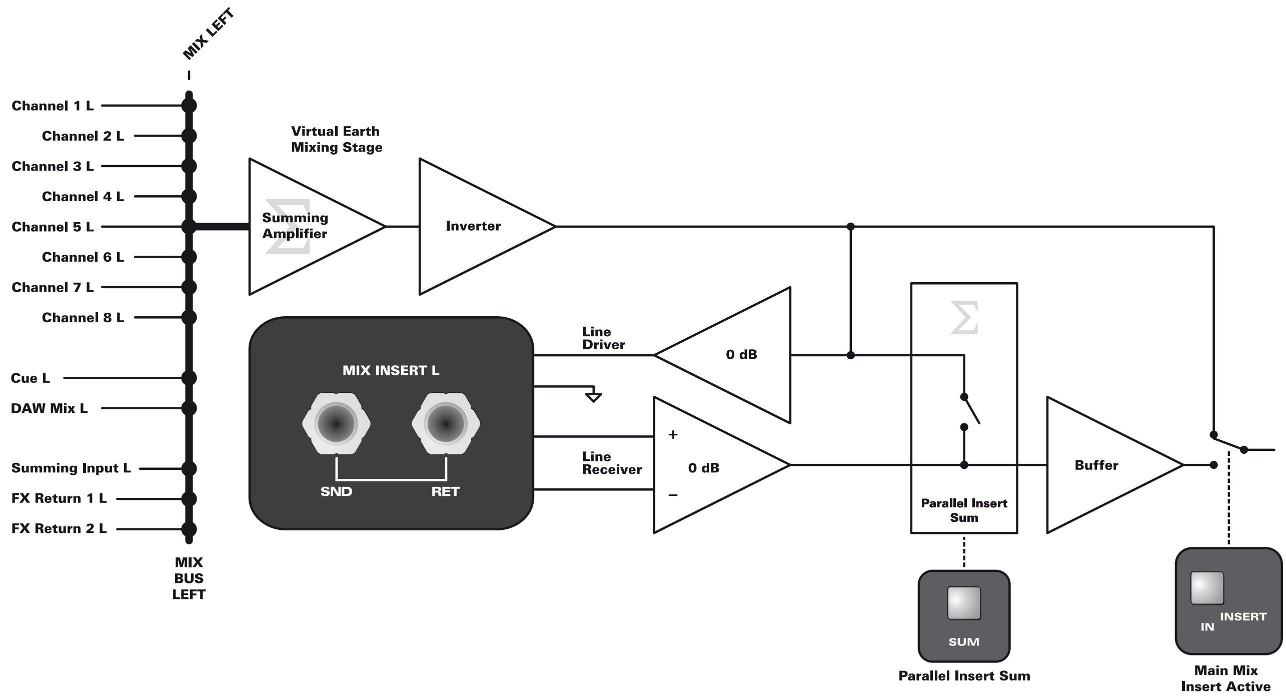
Mix Insert Parallel Mode

Using the insert SUM functionality, anything inserted into the main mix path will now sit in parallel with the main mix, facilitating creation of parallel processing techniques such as subtle colouration / enhancing, equalisation and compression.

The parallel insert can also operate as handy balanced line level expansion inputs. External summing devices can be used to add extra inputs by returning their output to the insert return point and engaging the insert sum switch.

Mix Insert Parallel Mode cont.

The following part of the block diagram illustrates how this parallel sum mode is achieved at the main mix insert point.



Input Expansion - Cheating the 2802 to become 3202!

✱ TIP!

This console features 8 main channel inputs plus an ability to use the cue sends as an extra 8 line level inputs to the mix bus. Coupled with the 12 inputs available in the FX returns section, there are 28 inputs available at mixdown... or are there?

If you want to get tricky there are two ways to gain a couple of extra stereo inputs to the main mix bus - this may come in handy during a busy mixdown!

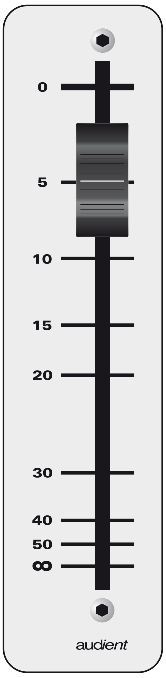
Firstly, you can assign the DAW MIX rear panel XLR inputs to the main mix bus, therefore providing one further stereo input (ASP3002?) and secondly, you can use the main mix insert (return connection only) in conjunction with the insert active and insert sum modes to provide another stereo expansion input, therefore 2802 is actually capable of 3202!

Main Mix Fader

The main mix fader on ASP2802 is a non automated, fully passive, high quality ALPS 100mm stereo linear fader.

There is no gain in hand and the fader is very suitable for fade in and fade out use during final mix printback. Feel your fade outs, stop drawing them in with a mouse!

MASTER
FADER



Manual 100mm
ALPS Fader
-STEREO-

Connections Overview

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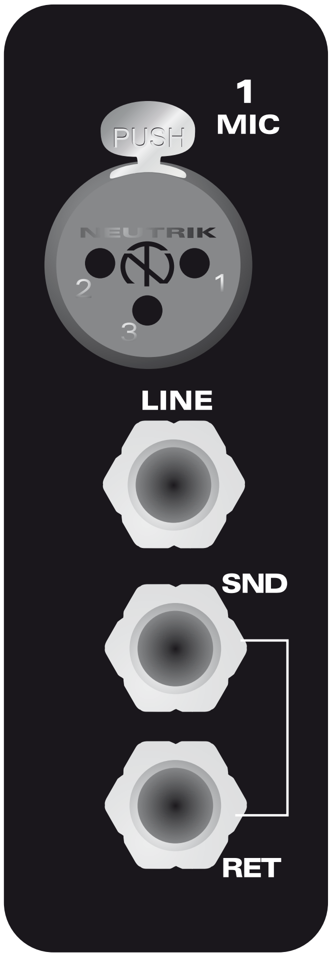
ASP2802 features a plethora of connections on the rear panel - as highlighted during the introduction of this manual.

All XLR balanced input and outputs are wired as Pin 1 (shield - chassis / shield ground), Pin 2 (hot +ve) and Pin 3 (cold -ve).

Channel Strip Inputs and Outputs

The first input from (top to bottom) is the MICROPHONE XLR INPUT (female). The input impedance of the microphone input is greater than 2.5k ohms and the maximum input level is approximately +16 dBu (where 0 dBu is referenced to 0.775 V_{rms}).



The next connection is the channel LINE INPUT, a balanced TRS with an input impedance of >10k ohms and a maximum operating level of +27 dBu. This and all other TRS jacks are wired as Tip (+ve), Ring (-ve) and Sleeve (shield).

The INSERT SEND jack comes next, a balanced TRS connector fed from an impedance balanced output with an output impedance of 150 ohms and a drive capability of +21 dBu.

The INSERT RETURN TRS jack is also fully balanced with a >10k ohm input impedance and a maximum input level of +21 dBu.

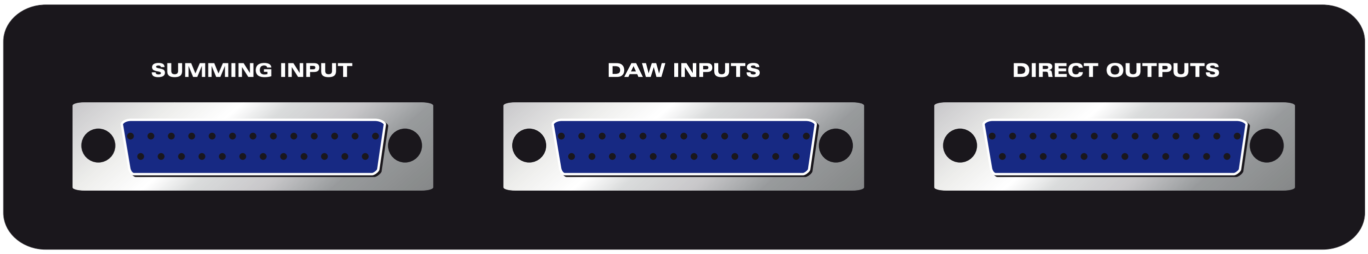
The remaining channel strip connections (DAW Inputs and Direct Outputs) are found on 2 8-way DB25 female connectors at the bottom of the rear panel. These are wired to Tascam DA-88 25-pin D-SUB standard (see next page). The DAW Inputs and Summing Inputs have a balanced input impedance of >10k ohms and a maximum input level of +27 dBu.

The direct outputs have a balanced output impedance of <75 ohms and a maximum output level of +27 dBu.

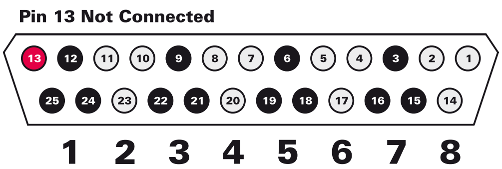
DB25 Tascam DA-88 Wiring Specification

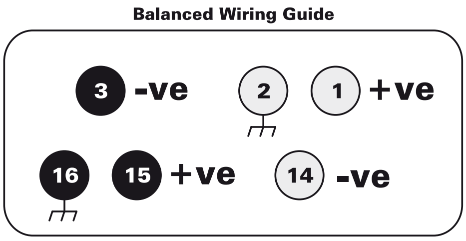
There are 3 DB25 connectors located on the rear of ASP2802.

- DAW Inputs 1-8
- Direct Outputs 1-8
- Summing Inputs (1-8)



The wiring pinout specification for these connections is as follows:

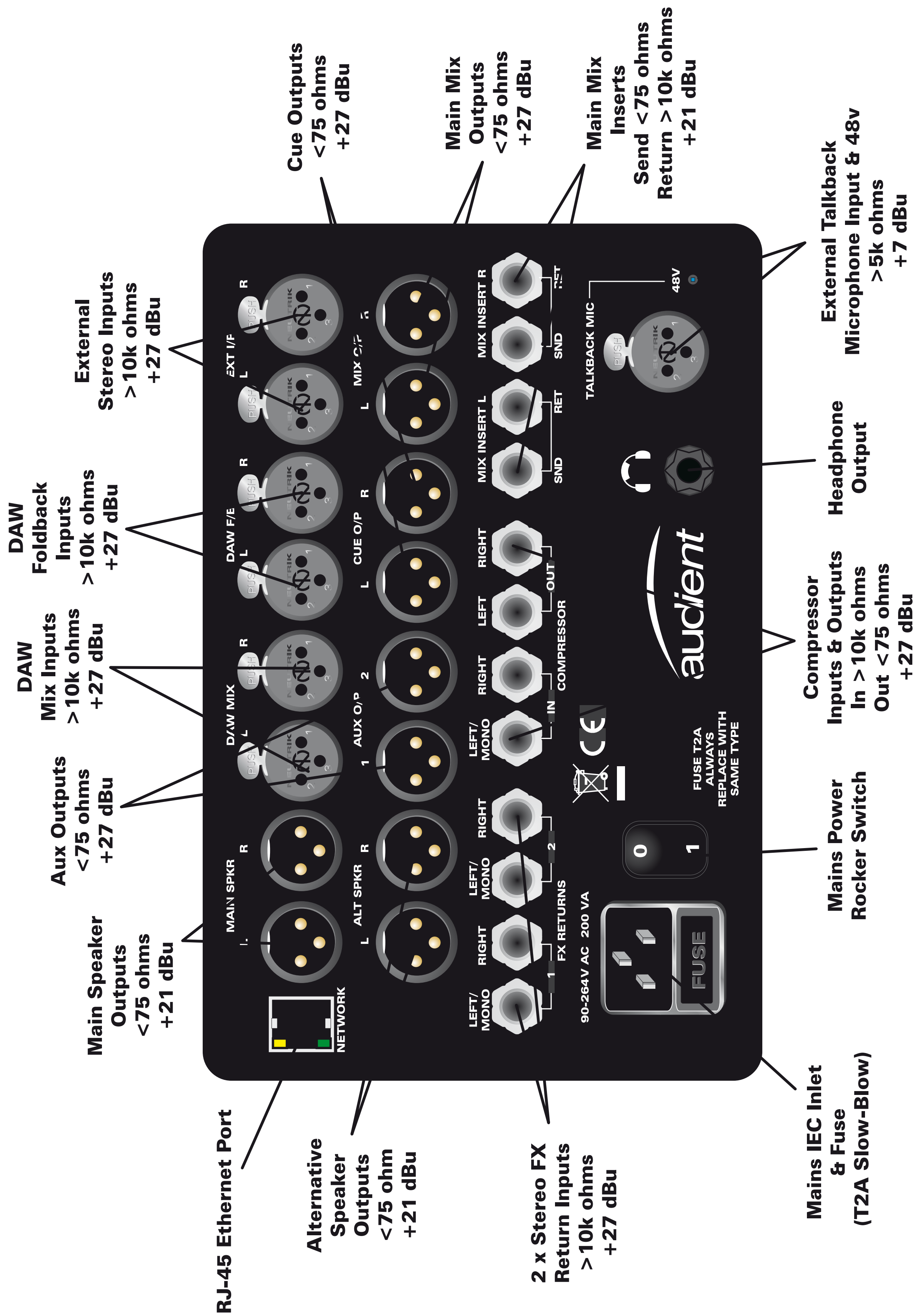




Channel Number	DSUB Pin Number +ve	-ve	Shield
1	24	12	25
2	10	23	11
3	21	9	22
4	7	20	8
5	18	6	19
6	4	17	5
7	15	3	16
8	1	14	2

Shield connections are bussed and grounded to the chassis within the unit

Master Section Connections



Master Section Connections (top left to right)

ASP2802 connects to your DAW via IP based networking protocols, and as such the first connector of interest in the master section is a standard RJ-45 CAT-5e Ethernet port, which provides bi-directional high speed communication to your DAW and carries MIDI automation data as well as control surface protocols. There are two LEDs that indicate network connection and activity.

The Main Speaker, Alt Speaker, Aux 1 & 2, Cue and Main Mix outputs are on balanced male XLR connectors with <75 ohms output impedance and a maximum output level of +27 dBu.

The DAW Mix, DAW F/B and External Stereo inputs are on balanced female XLR connectors with >10k ohms input impedance and a maximum input level of +27 dBu.

Two stereo FX Returns and the External Compressor inputs are provided on balanced 1/4" TRS jack, with an input impedance of >10k ohms and a maximum input level of +27 dBu.

The External Compressor outputs are provided on balanced 1/4" TRS jack and have an output impedance of <75 ohms with a maximum output level of +27 dBu.

The Main Mix Inserts are also on balanced 1/4" TRS jack with the send jacks providing an output impedance of <75 ohms and a maximum drive capability of +21 dBu, while the return jacks offer >10k ohms input impedance with +21 dBu of maximum input level.

The Mains Power input is a stand 3-pin IEC connector with integrated fuse, it will accept line voltages from 90-264v. The fuse is a T2A slow-blow.

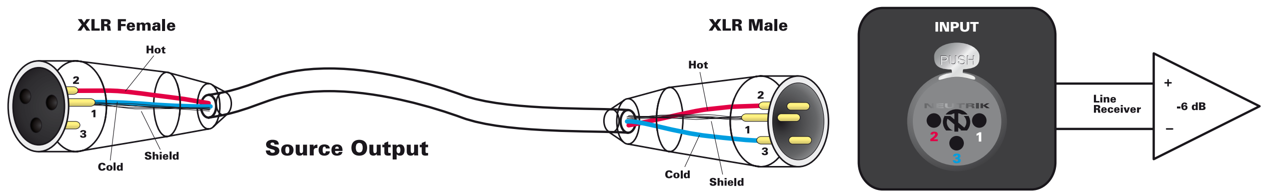
The Headphone output is an unbalanced, stereo TRS 1/4" jack connection with enough current drive to cause hearing damage if driven hard - BE CAREFUL, protect your ears!

Master Section Connections (top left to right) cont.

The External Talkback microphone input is a balanced female XLR connector with >5k ohms input impedance and a maximum input level of +7 dBu. Please note 48v phantom power can be provided to this input to incorporate whatever talkback microphone you fancy!

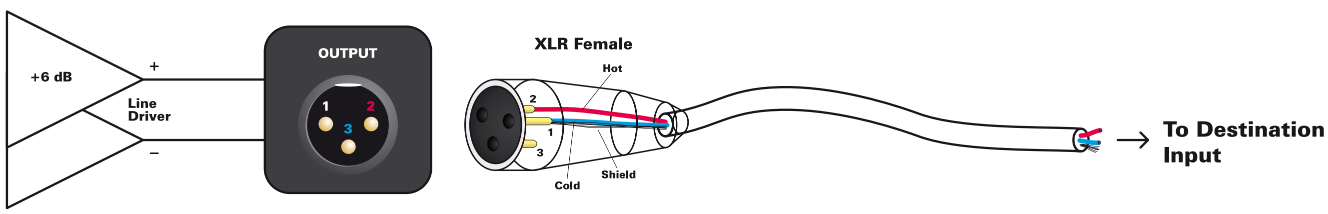
Unbalancing ASP2802

To unbalance an ASP2802 line level input, link pin 3 to pin 1 at the source. Please note that this must be achieved at the source end, therefore the ASP2802 line input must be left wired as balanced to preserve some of the common-mode rejection performance (CMRR) of the input stage.



To unbalance an output from ASP2802, link pin 3 to pin 1 at the output of the console.

Please note that this can also be achieved in a 1/4" TRS Jack connector by connecting the ring to the sleeve within the jack housing. To achieve an unbalanced output from the DB25 channel direct outputs you will have to do it at the terminating end, assuming that the cable breaks out into tails.



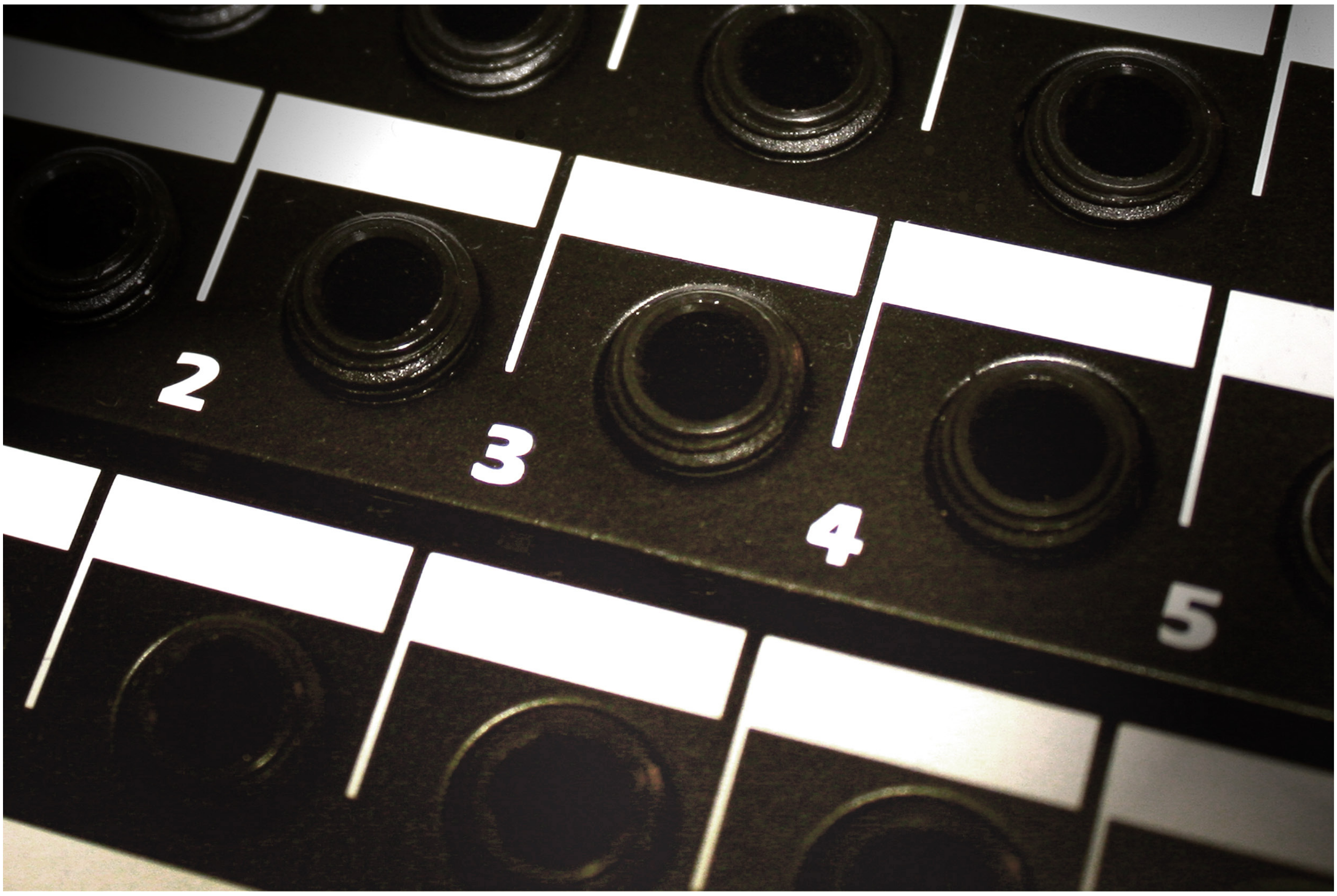
Please note that when wiring your cables do not leave as much exposed bare shield as can be seen in these diagrams, they have been exaggerated for clarity.

Total Number of Connections

To provide a starting point for patchbay integration and cable purchasing, you may find the following table of information helpful.

Connection	Amount	Type
Microphone Input	8	XLR Female
Line Input	8	1/4" TRS Jack
Channel Insert Send	8	1/4" TRS Jack
Channel Insert Return	8	1/4" TRS Jack
Direct Output	8	DB25 Female x 1
DAW Input	8	DB25 Female x 1
Summing Input	8	DB25 Female x 1
FX Returns	4	1/4" TRS Jack
Ext. & DAW Monitor Inputs	6	XLR Female
Aux & Cue Outputs	4	XLR Male
Main Mix Outputs	2	XLR Male
Main Mix Insert Send	2	1/4" TRS Jack
Main Mix Insert Return	2	1/4" TRS Jack
Loudspeaker Outputs	4	XLR Male
Bus Compressor Input	2	1/4" TRS Jack
Bus Compressor Output	2	1/4" TRS Jack
External Talkback Input	1	XLR Female
Headphones Output	1	1/4" TRS Stereo Jack
Mains Power	1	IEC Socket
Control Surface / Ethernet	1	RJ-45 Ethernet Socket
i-Jack Front Panel Input	1	3.5mm Stereo Mini-Jack
Total	89	

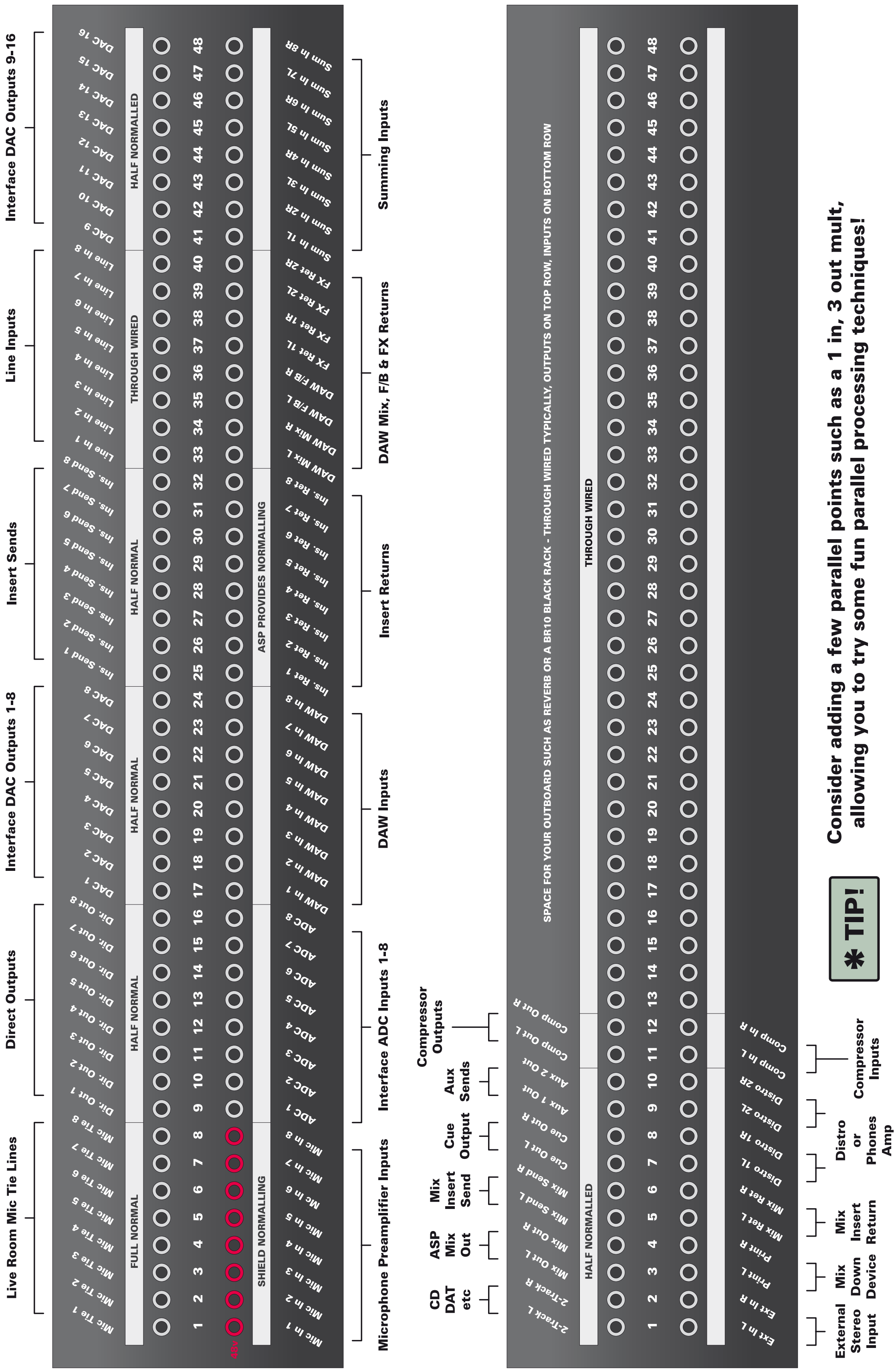
Patchbay Layout



audient
ASP 2802

DUAL
LAYER CONTROL

Possible Patchbay Layout (Bantam)



Consider adding a few parallel points such as a 1 in, 3 out mult, allowing you to try some fun parallel processing techniques!

Patchbay Wiring

In the example provided on the previous page, two 96-point bantam patchbays are used, with plenty of spare points left over.

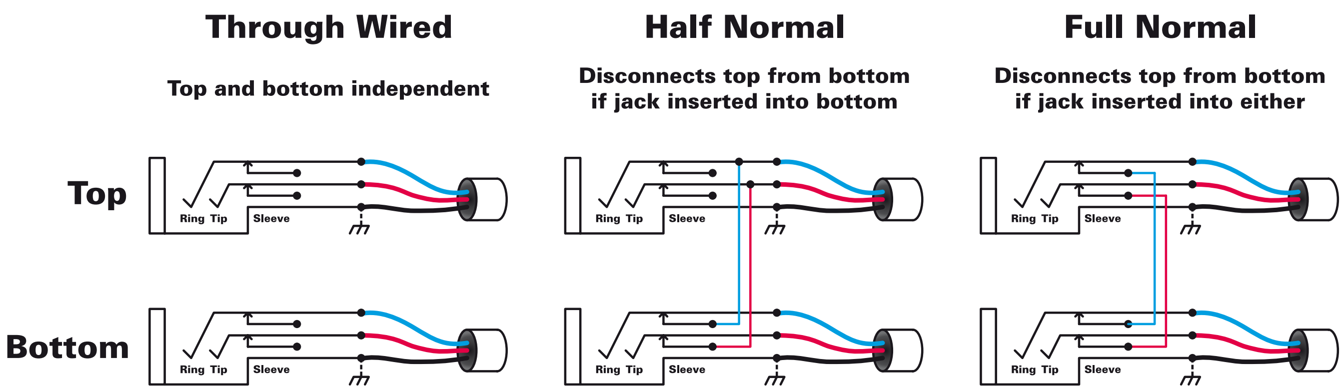
It also featured a 16 channel DAW interface that provided 16 outputs for mixdown using the summing input and 8 main channels. Obviously you can juggle this around and adapt the ideas for your optimal workflow and exact I/O count.

Although 2 bantam bays may look excessive, with a few quality pieces of outboard gear and a set of hardware synthesisers and samplers etc - this bay will fill very quickly.

You may notice that the loudspeaker outputs are not on the bay, this is quite common - as they would typically be hardwired to your monitors, however adding them to the patchbay may add flexibility for anyone who likes to switch playback systems around, or travel with a pair of their favourite monitors.

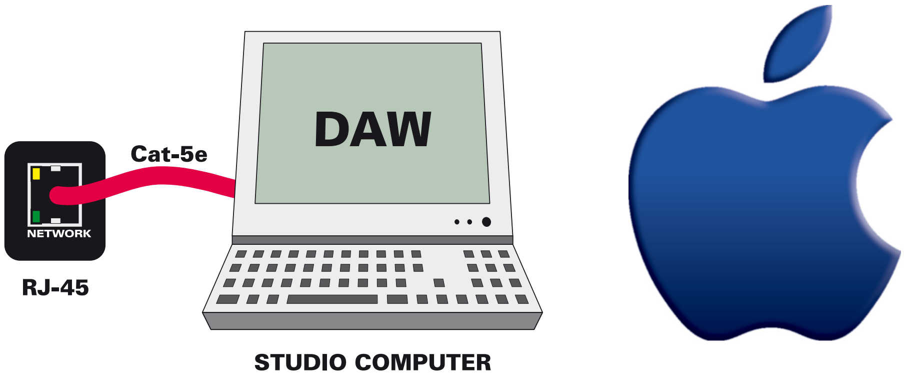
If you need to squeeze a layout onto a single 96-way bay, it will fit if you hardwire your microphone tie lines to the ASP2802 microphone preamplifiers and also hardwire your computer interface or recording platform input and output to the DAW or Line inputs and the direct outputs.

Installing your console with a good patchbay system will increase your workflow, provide the ultimate flexibility and last a lifetime.



DAW Ethernet Connection (Mac)

You can use standard Cat-5e Ethernet cable to connect ASP2802 to your studio DAW computer. The interface in ASP2802 is compatible whether you use a straight through or cross-over cable.



Please ensure that you have downloaded and installed the latest version of our AuNet.dmg application from **www.audient.com**.

On Apple computers please ensure that AuNet is added to your applications folder via drag and drop.

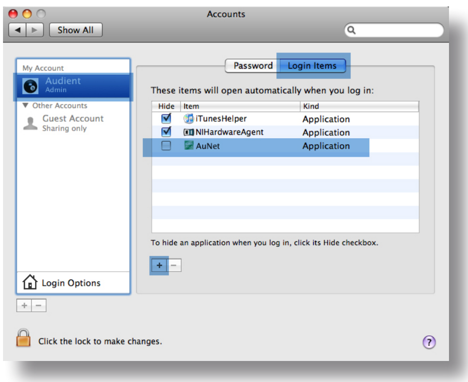
Macintosh HD > Applications

You can enable the application to always boot upon user log-in, therefore preventing any forgetfulness!

To achieve this, do the following:

System Preferences > Accounts > (select the account you wish to alter on the left hand side) > Login Items Tab > Add New Application (+ Button) > Browse and select AuNet

Everytime this user account opens, AuNet should now boot.



Once you have connected ASP2802 to your studio computer via either a direct Ethernet link or through a router and installed AuNet you are ready to set up the networking side of the console.

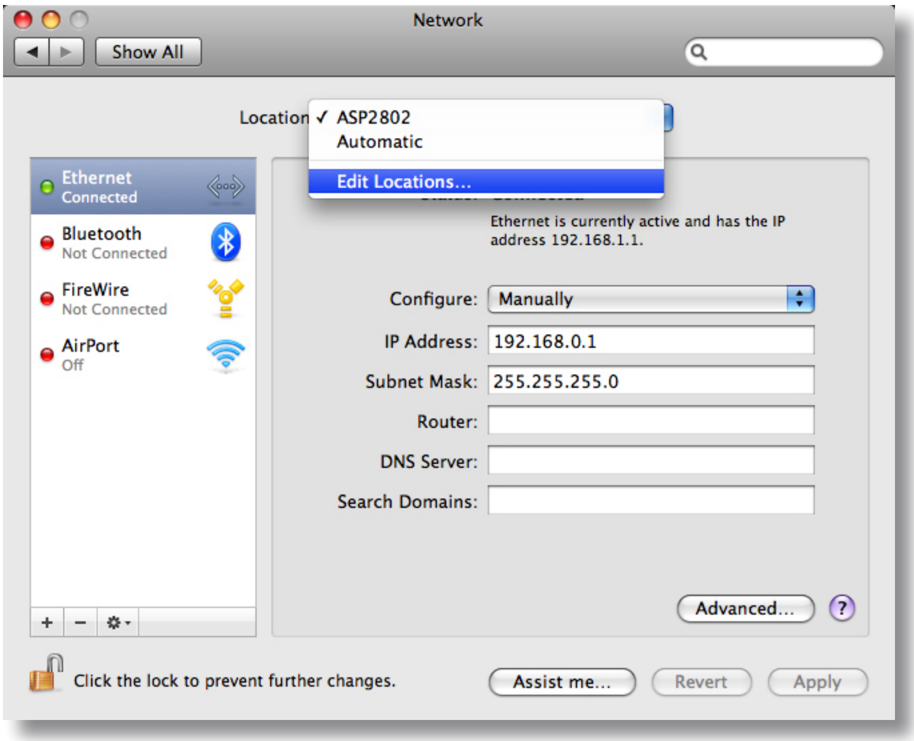
IP Address Setup for Direct Connection

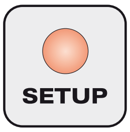
Unless you have a router in your studio, it is most likely that you will directly connect ASP2802 to your DAW via a direct Ethernet link.

First you must set your DAW IP address (please note if you do have a router, this should not be adjusted and you should skip to the next section).

On Apple computers please do the following:

> System Preferences > Network > Ethernet Tab > Locations > Edit Locations > Add New Location (+ Button) > Name ASP2802 > Select this Location > Set Configure to Manually > Set IP Address to 192.168.0.1 > Set Subnet Mask to 255.255.255.0

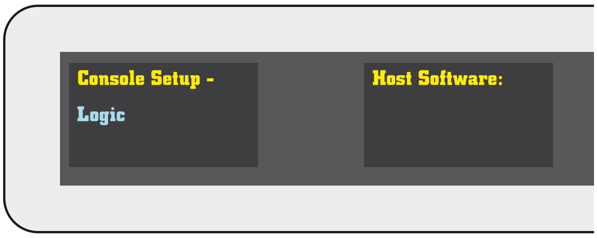




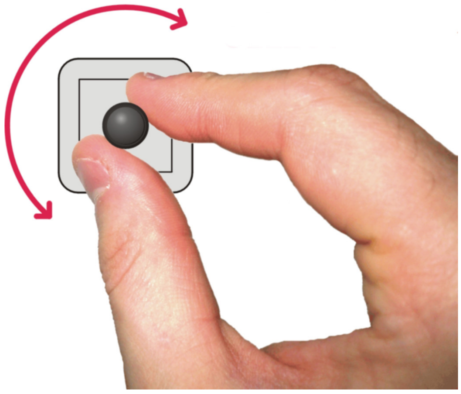
Moving across to ASP2802, power up the console and once booted, press the SETUP button. The OLED displays will show the first page of console setup.

First set the application you are using with ASP2802. The options at the time of release include:

- Apple Logic 9
- Avid Pro Tools 8
- Steinberg Cubase 5 (Nuendo)

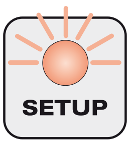


Use the first rotary encoder to select the required DAW application.

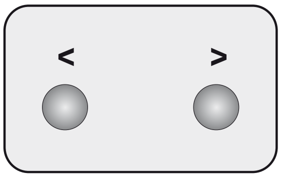


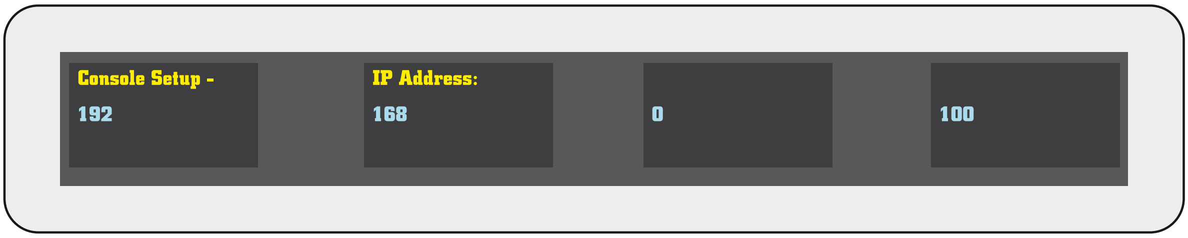
Note that once a change has been made, the setup switch LED will flash to indicate that a setting has changed.

If you exit setup by pressing the switch once more and then return by entering setup again, you will notice that your setting has been automatically saved.

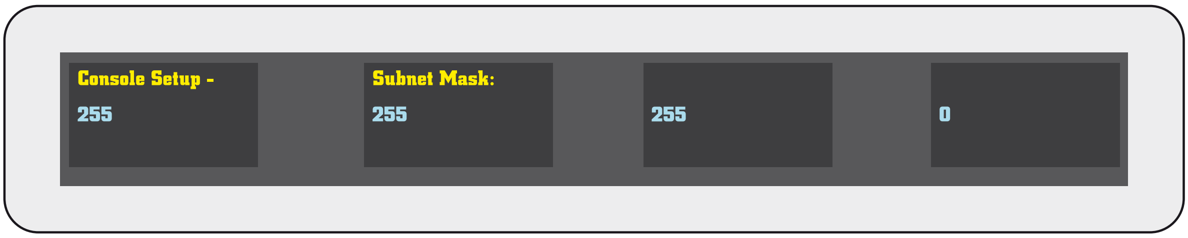


Using the page keys, page to the right and ensure that Use DHCP is set to NO. Page right again to reach the IP address screen and set the IP address to 192.168.0.100 (default) using the encoders, then move to the fourth page, and check that the subnet mask is set to 255.255.255.0.

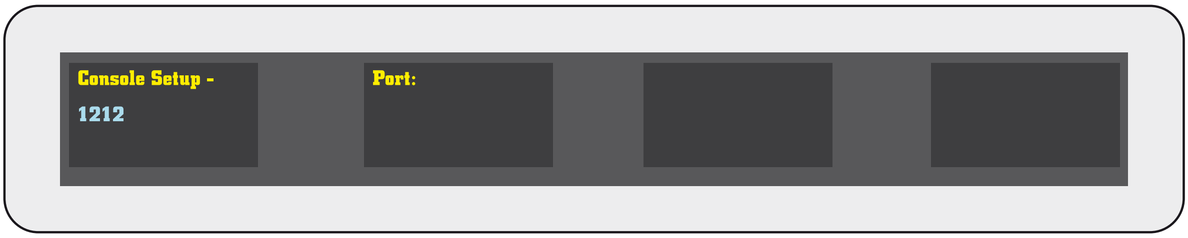




Subnet Mask set to 255.255.255.0:



Port set to 1212:



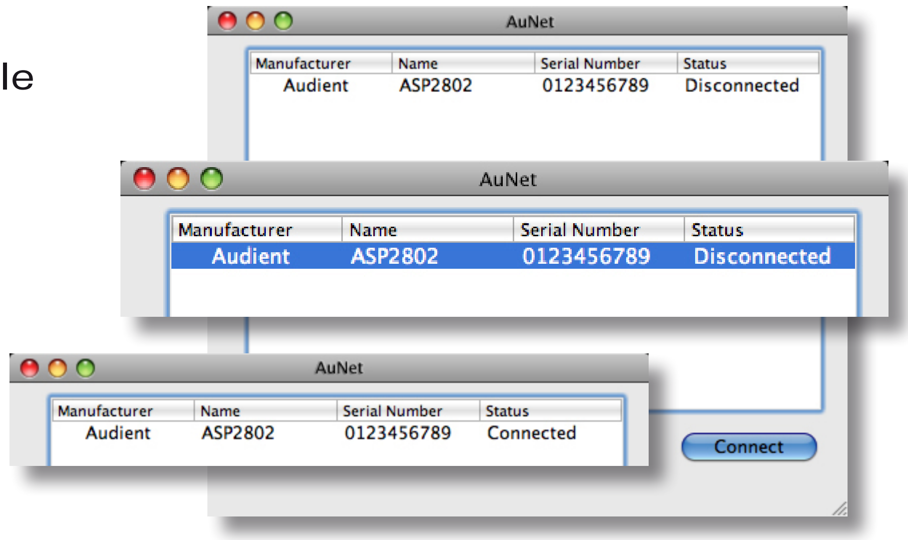
Press the setup key to exit setup and your setting will be saved.

Now power down the console and restart your Apple computer. Once booted, ensure that the AuNet application is running and open on your DAW machine. Finally power up the console.

After the console boots, it should be picked up by AuNet (you may have to repeat the power down / restart process if you are experiencing difficulties).

AuNet should indicate detection of the console and should also show you the serial number of the console.

Select the console and click on the connect button to the bottom right >>



Networking (Mac)

Your console should now be connected to your studio DAW computer and ready for configuration as a HUI™ control surface and analogue automation platform.

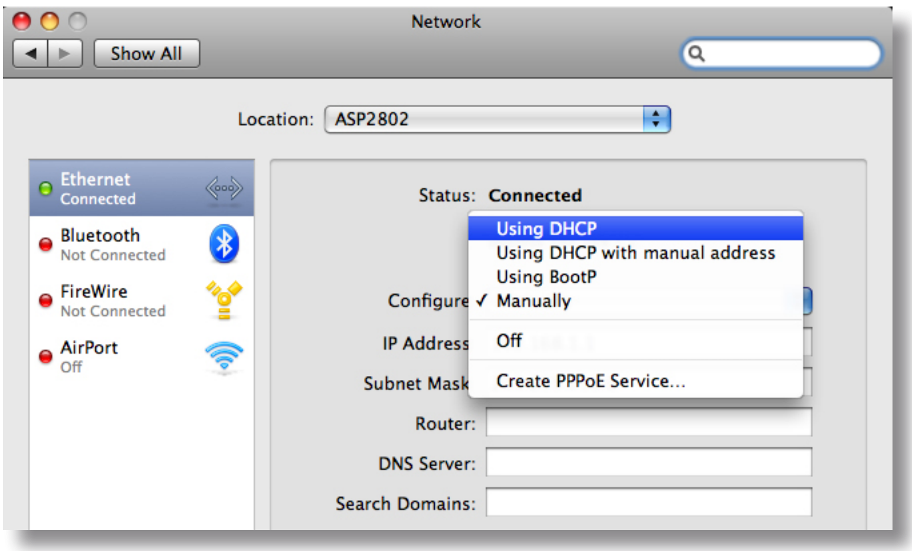
Setting up the console with DHCP (for use with routers)

If you have more than one device connected to your DAW computer, you will most likely be using a network router to talk to all of these devices.

If you have connected ASP2802 to your DAW computer via a network router, you will need to use DHCP to enable automatic configuration of IP address, providing fast and smarter networking.

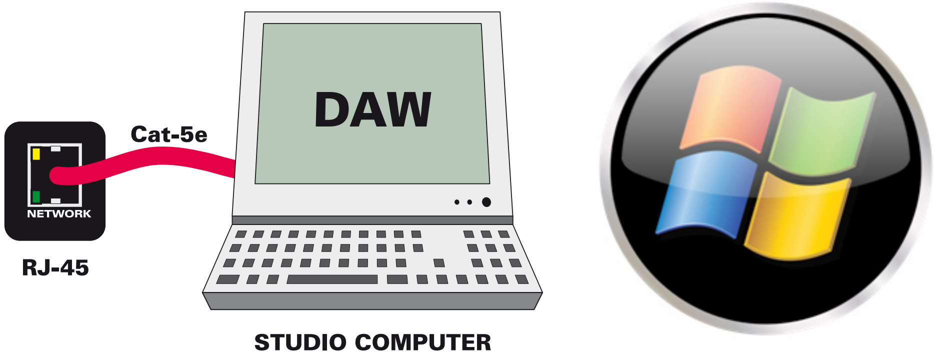
Ensure that you enable DHCP in your DAW computer system preferences (you should already know how to do this if you are running an existing network).

You will most likely be running DHCP with automatic IP hand out.



Ensure that the console is set to use DHCP on the second page of the setup menu, once set reboot the console and follow the same AuNet connection procedure as outlined previously once the console is detected.

You can use standard Cat-5e Ethernet cable to connect ASP2802 to your studio DAW computer. The interface in ASP2802 is compatible whether you use a straight through or cross-over cable.

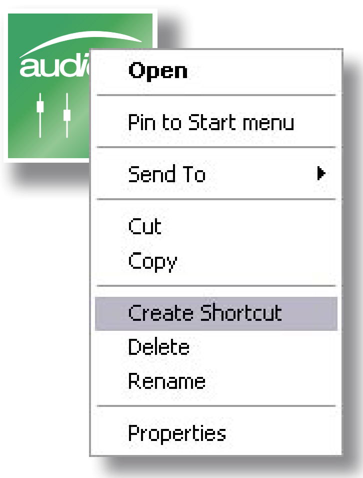


Please ensure that you have downloaded and installed the latest version of our AuNet.msi application from **www.audient.com**.

On PC computers please ensure that AuNet is installed using the provided .msi package. The program will be added to your local system drive (C:) and a folder will be created under:

C:\ Program Files \ Audient

You can enable the application to always boot upon user log-in using the Windows Start Up folder in XP. To do this create a new shortcut for AuNet by right clicking on the application icon and selecting 'Create Shortcut'.



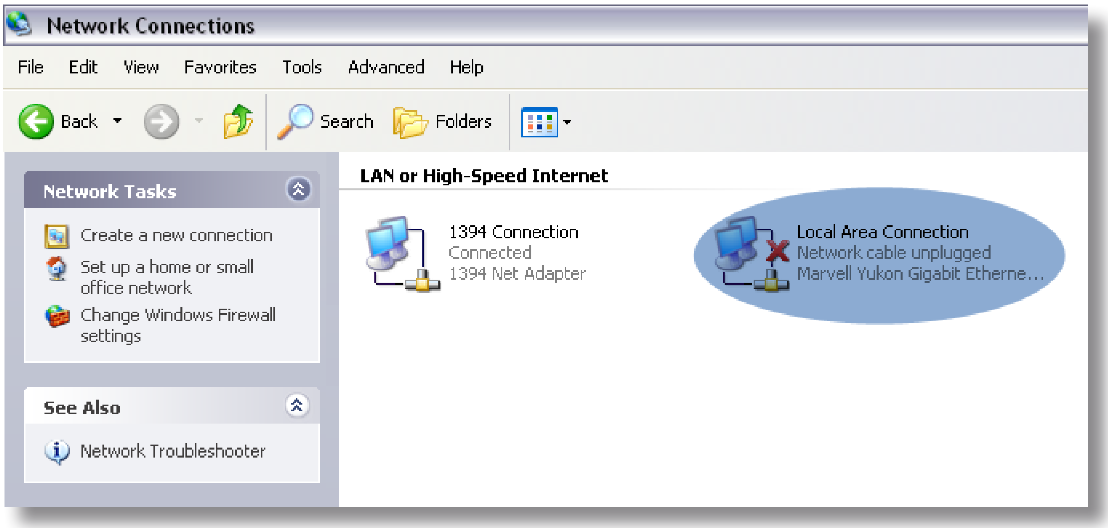
Drag and drop this shortcut into the following location:

C:\ Documents and Settings \
All Users \ Start Menu \
Programs \ Startup

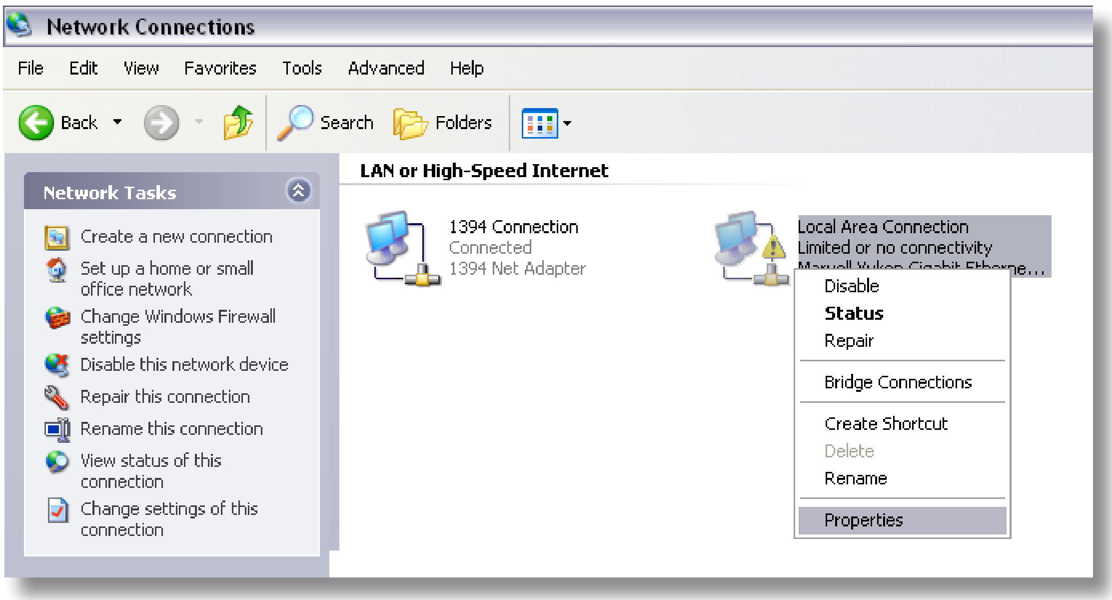
Networking (PC)

Once you have connected ASP2802 to your studio computer via either a direct Ethernet link or through a router and installed AuNet you are ready to set up the networking side of the console.

Until your console is connected (or any other LAN device to your PC), the LAN status indicated in Control Panel > Network Connections will be unplugged.

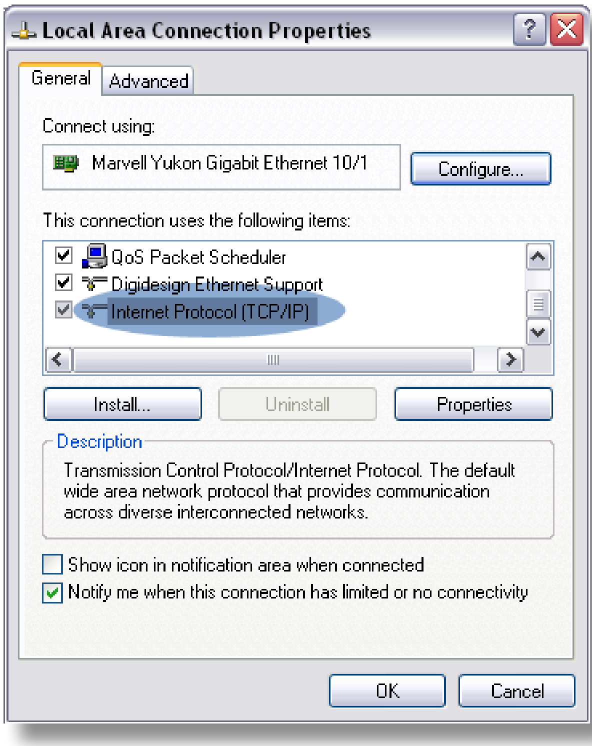


Right click on the Local Area Connection icon and select Properties.

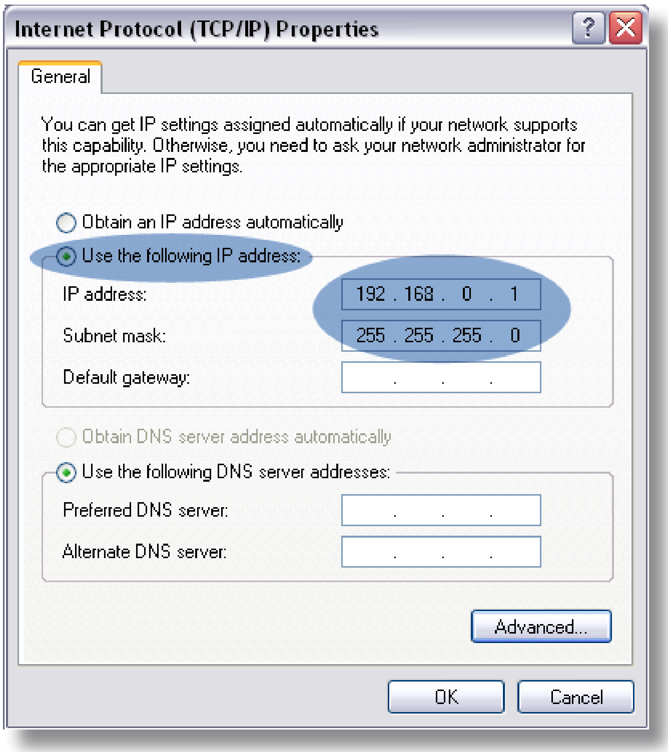


Networking (PC)

In the Local Area Connections properties dialogue window scroll down and select Internet Protocol TCP/IP and then open the properties dialogue box with the button to the right.



Here select the 'use the following IP address' radio button.

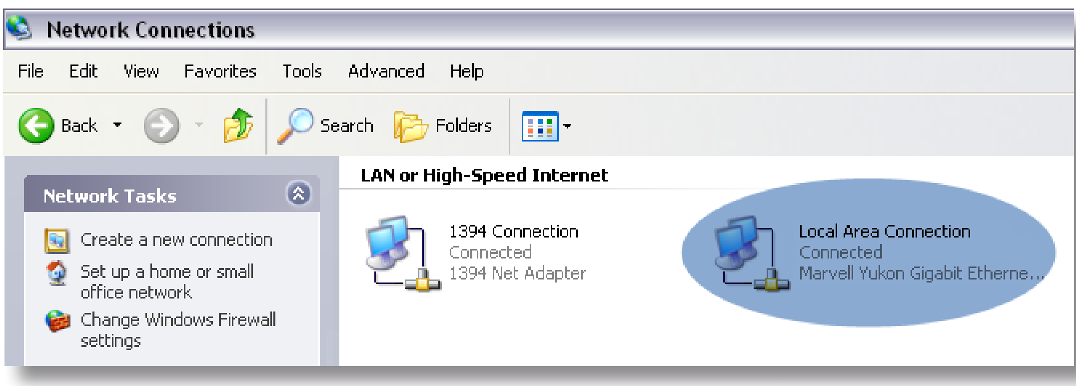


Set the IP address of your PC to 192.168.0.1

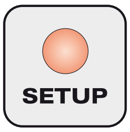
Set the subnet mask to 255.255.255.0

Leave the default gateway blank and click OK.

Your PC may take some time to apply the settings so please be patient.



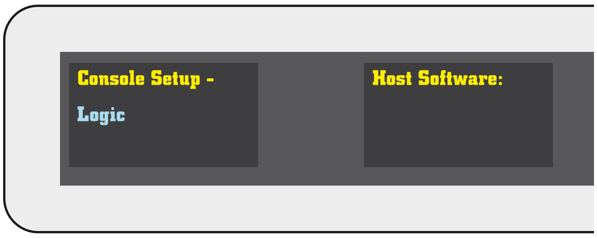
Power the console on and once booted, check that a connection has been opened in the Network Connections window via the Control Panel.



Moving across to ASP2802, press the SETUP button. The OLED displays will show the first page of console setup.

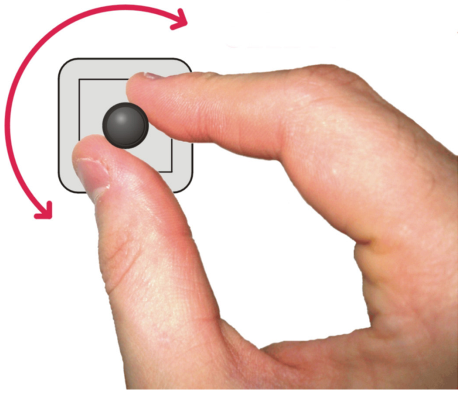
First set the application you are using with ASP2802. The options for PC at the time of release include:

- Avid Pro Tools 8
- Steinberg Cubase 5 (Nuendo)

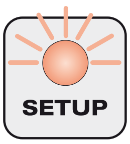


Use the first rotary encoder to select the required DAW application.

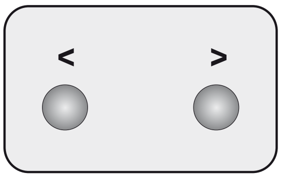
Note that once a change has been made, the setup switch LED will flash to indicate that a setting has changed.

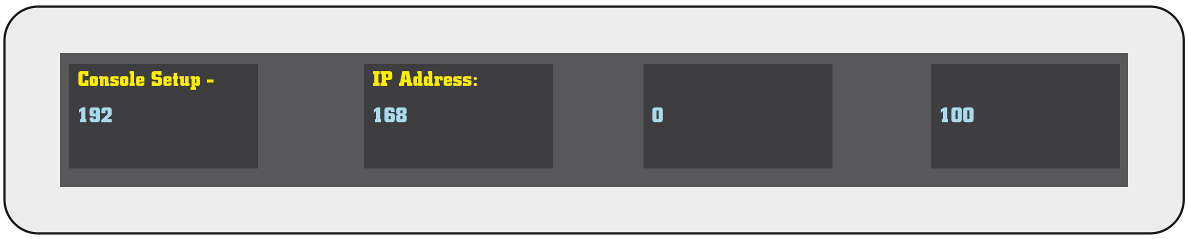


If you exit setup by pressing the switch once more and then return by entering setup again, you will notice that your setting has been automatically saved.



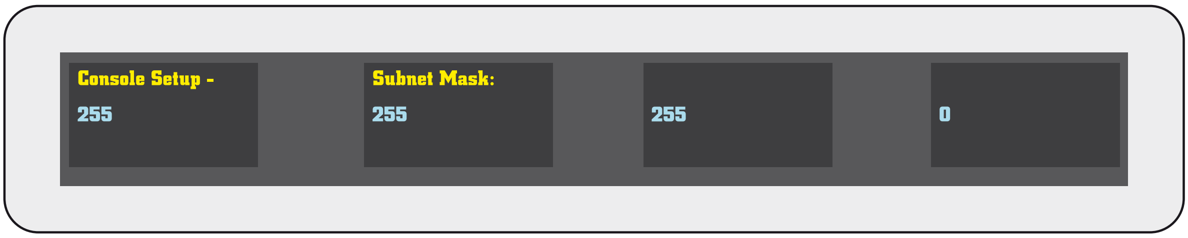
Using the page keys, page to the right and ensure that Use DHCP is set to NO. Page right again to reach the IP address screen and set the IP address to 192.168.0.100 (default) using the encoders, then move to the fourth page, and check that the subnet mask is set to 255.255.255.0.



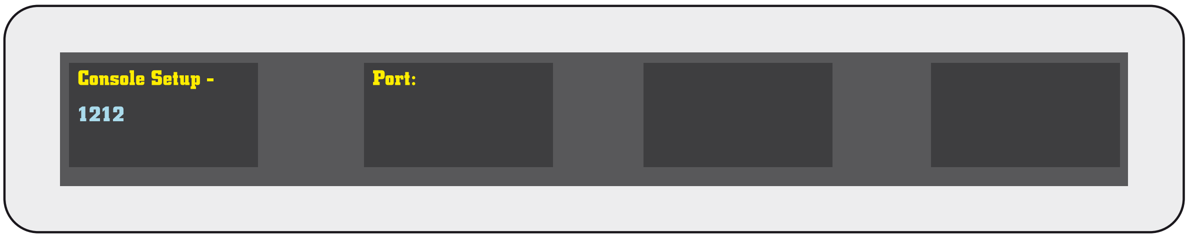


Networking (PC)

Subnet Mask set to 255.255.255.0:



Port set to 1212:



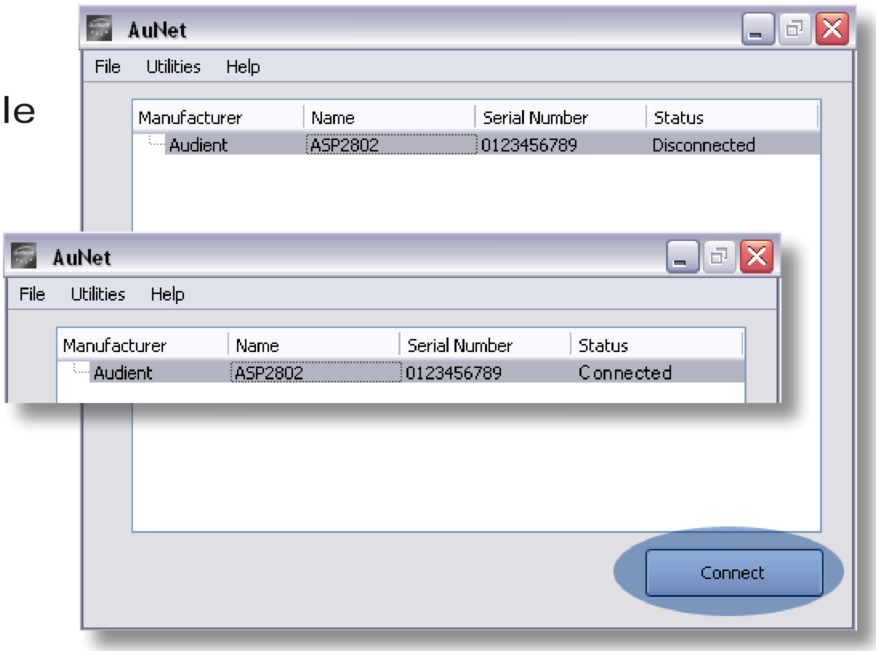
Press the setup key to exit setup and your setting will be saved.

Now power down the console and restart your PC. Once booted, ensure that the AuNet application is running and open on your DAW machine. Finally power up the console.

After the console boots, it should be picked up by AuNet (you may have to restart your computer if experiencing difficulties and repeat).

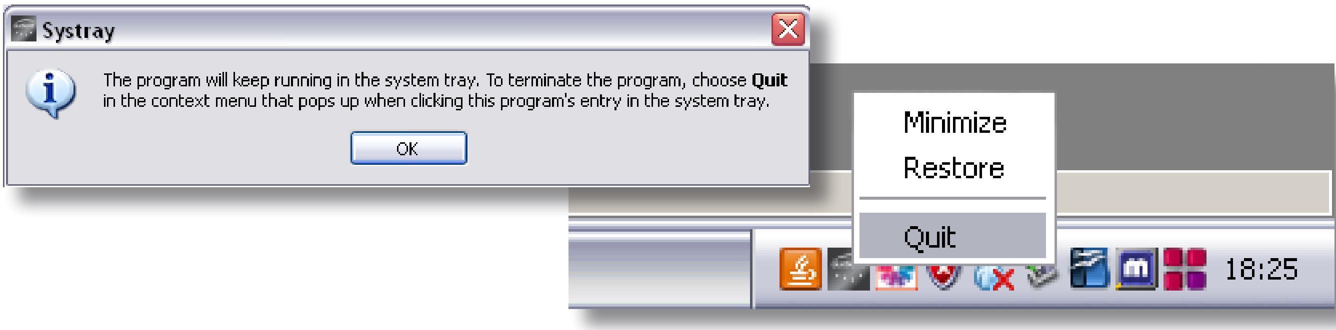
AuNet should indicate detection of the console and should also show you the serial number of the console.

Select the console and click on the connect button to the bottom right >>>



Your console should now be connected to your studio DAW computer and ready for configuration as a HUI™ control surface and analogue automation platform.

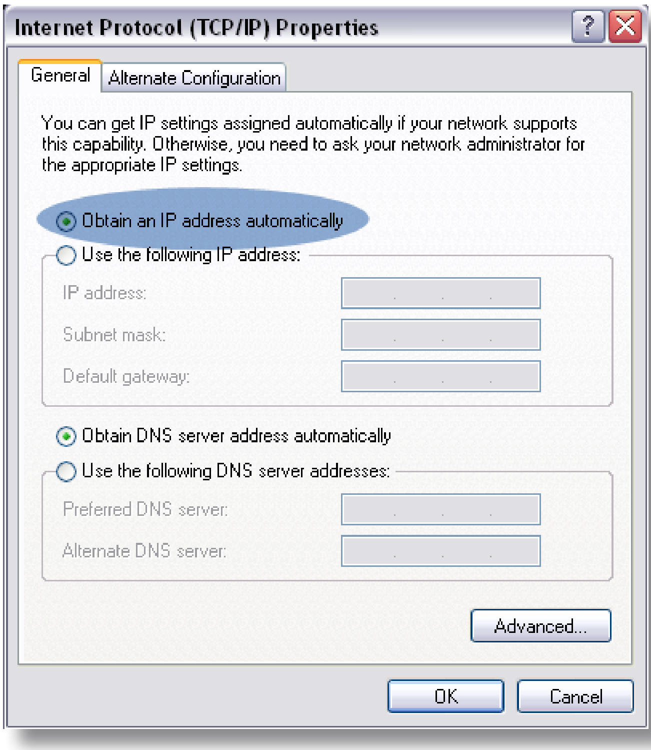
If you need to exit the AuNet program it will still run in the systray until you manually quit by right clicking on the icon in the system tray and select quit.



Setting Up Automatic IP Assign (for use with routers)

If you have more than one device connected to your DAW computer, you will most likely be using a network router to talk to all of these devices.

If you have connected ASP2802 to your DAW computer via a network router, you will need to use DHCP to enable automatic configuration of IP address, providing fast and smarter networking.

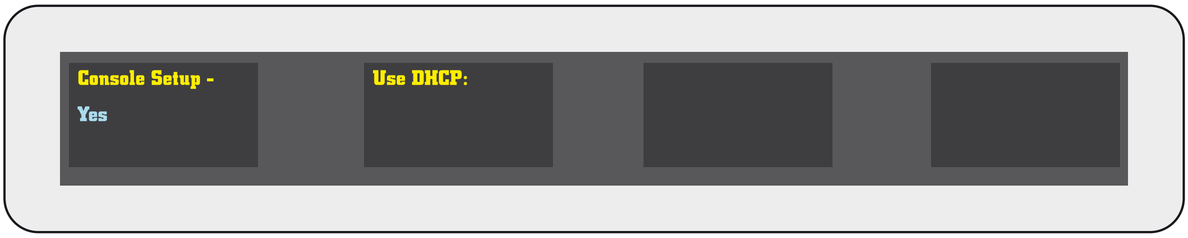


Ensure that you enable 'obtain IP addresses automatically' in Network Connections > Local Area Connections > Internet Protocol TCP/IP Properties (you should already know how to do this if you are running an exisiting network).

Networking (PC)

You will most likely be running DHCP with automatic IP hand out.

Ensure that the console is set to use DHCP on the second page of the setup menu, once set reboot the console and follow the same AuNet connection procedure as outlined previously once the console is detected.



Firmware Updates

When updates are released for your console to improve control surface workflow, bug fixes and add new features, you will need to update the console firmware.

To do this please visit **www.audient.com** to obtain the latest firmware from the ASP2802 webpage and follow the simple instructions provided along with the firmware download to flash the new upgrade to the console memory.

If you experience any issues please read the ASP2802 FAQ found online at **www.audient.com**.

ASP2802 firmware developed with FreeRTOS, uIP & lwIP.

www.freertos.org
www.sics.se/~adam/uip/index.php/Main_Page
www.sics.se/~adam/lwip/index.html

Fader Automation

audient ASP 2802

DUAL LAYER CONTROL

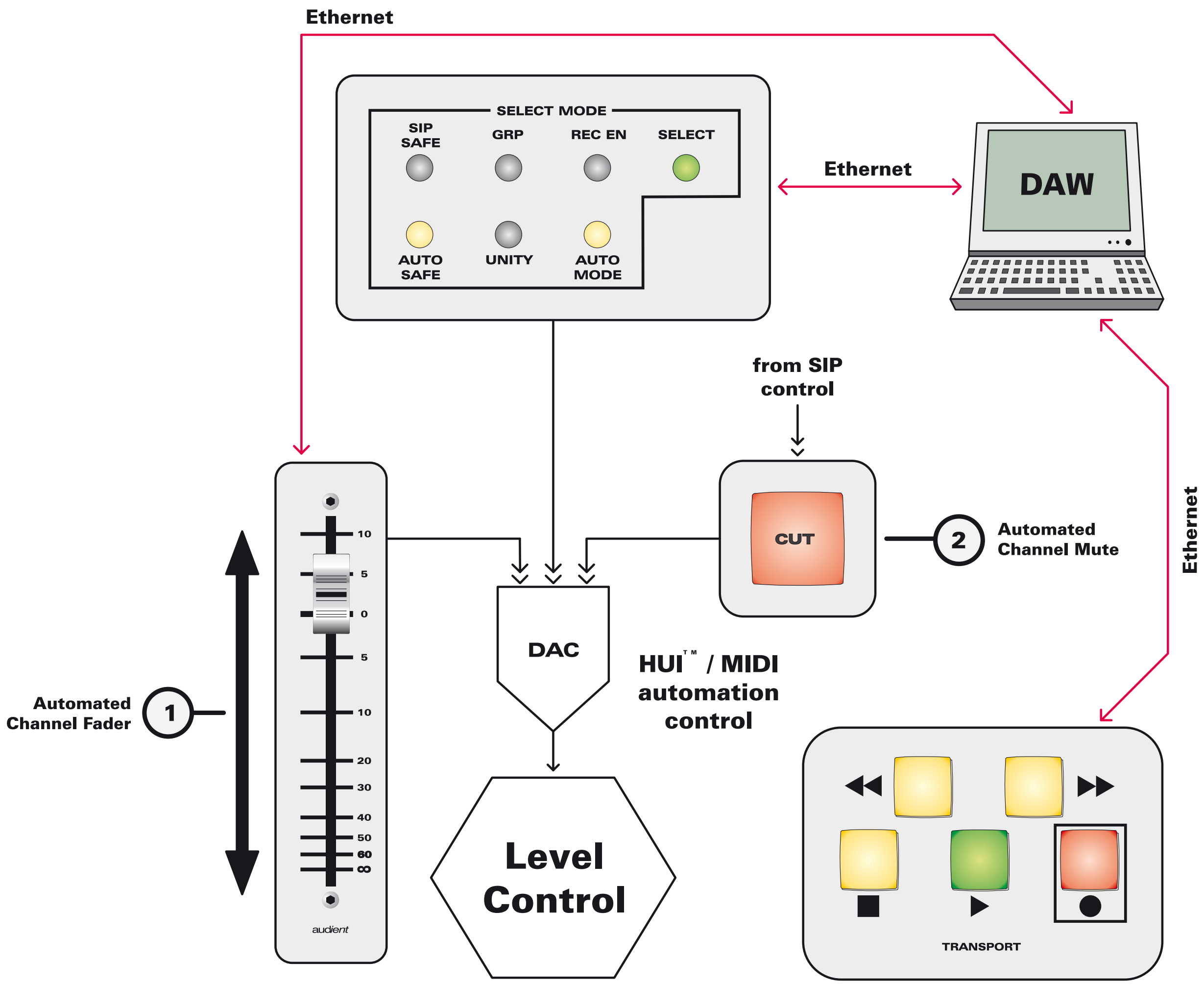
Overview of Functions

The diagram below provides an overview of the automation system and the parameters that can be controlled.

ASP2802 provides fader and mute automation along side its full DAW control surface layer.

The sections of most interest when using the console automation are as follows:

- Faders
- Cut (Mute) Switches
- Select Mode Panel - Auto Mode, Select and Auto Safe
- DAW Transport



A Note on Automation Safe

Automation safe should be used when you want to isolate a particular channel from automation - for example - to audition rides without "fighting" existing automation data or without "printing" the rides if channels are still in write enabled.

Please note that when the console boots, automation safe is engaged by default. If you wish to automate a channel you must first press the automation safe switch (in the select mode panel) and then turn off automation safe from each channel you wish to automate.

Note that all select mode layers are stored and function simultaneously. By toggling the select mode switches in conjunction with the large green channel select switches it is possible to obtain rapid control of SIP safe, DAW record enable, select, unity and automation safe channel settings.

All select mode layers are stored even after a power down, so remember to clear them manually if the next session requires a different set-up.

Automation & Control Surface DAW Specific Manuals

Further platform specific DAW setup procedures for both the automation system and control surface layer can be found in the DAW specific manuals located online at **www.audient.com**.

Currently Logic 9, Pro Tools 8 and Cubase / Nuendo 5 are suported on Apple Mac computers.

Pro Tools 8 and Cubase / Nuendo 5 are supported on PC computers.

Control Surface Overview

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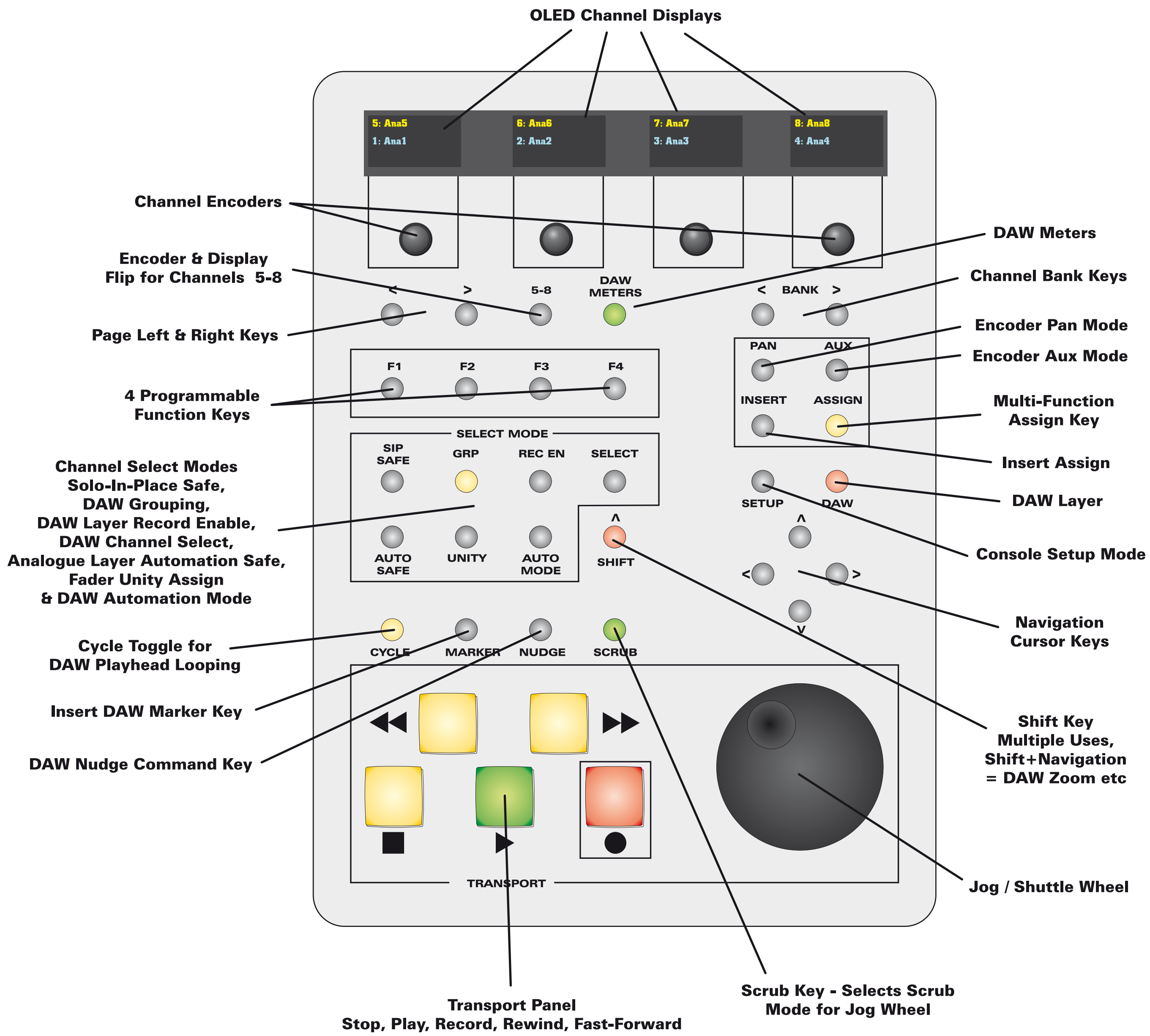
DUAL

LAYER CONTROL



Control Surface Operation

Please refer to the separate DAW specific control surface manuals found online at **www.audient.com** to guide you through the finer points of operation within each DAW platform.

Example Systems

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DUAL LAYER CONTROL

Example System 1 - 8 Channel DAW Platform

The first example system shown on the next page illustrates a basic system with a typical 8-in and 8-out soundcard / interface as commonly available. In this example there are 8 microphone channels, some basic outboard processing, one set of monitors and a single reverb unit hooked up to the aux sends.

To print back your final mix, a stand-alone CD recorder is used, however this could be a tape machine or as simple as feeding back the mix outputs to your soundcard / interface inputs upon mixdown.

Example System 2 - Hybrid Production Platform

As example 1, however a standalone sampler / synth is fed into the channel line inputs to enable monitoring and recording of its outputs.

Also the cue outputs now feed a headphone distribution system, useful for feeding multiple performers a mix.

Example System 3 - 16 Channel Tracking Platform

In this example a second pair of monitors are added to provide a useful reference for judging midrange and balances.

In order to provide 16 microphone inputs for tracking, an extra Audient ASP008 preamplifier is used. This is routed digitally to a 16 channel capable soundcard / interface.

These 8 feeds are sent back to the 8-channel summing input on ASP2802, allowing monitoring of these channels and access to the cue section.

In this example, the analogue line outputs of the ASP008 are also routed to the ASP2802 line inputs, allowing optional access to the direct output system here (note that they would override the console microphone inputs at that point).

Example System 3 cont.

If you wish to work it another way, an all analogue, zero-latency monitor mix could be formed by feeding the analogue output of ASP008 (or whatever standalone preamplifiers you are using) into the DB25 DAW Input connection. Using the alternative cue input, you could balance these extra 8 inputs using the cue pan and level controls accessing both the cue output section and main mix bus for headphone and control room monitoring.

Note that in this arrangement, actual recording of the extra 8 channels (9-16) must either happen via a mult at the output of the preamplifiers or via a preamplifier with a digitial output fitted such as an ASP008.

Example System 4 - 24 Channel Mixing Platform

The final example illustrates a heavily connected ASP2802 ready for medium to large scale mixdown purposes.

There are 24 outputs from the DAW interfaces which are used for mixing / line-level summing with a few useful tricks.

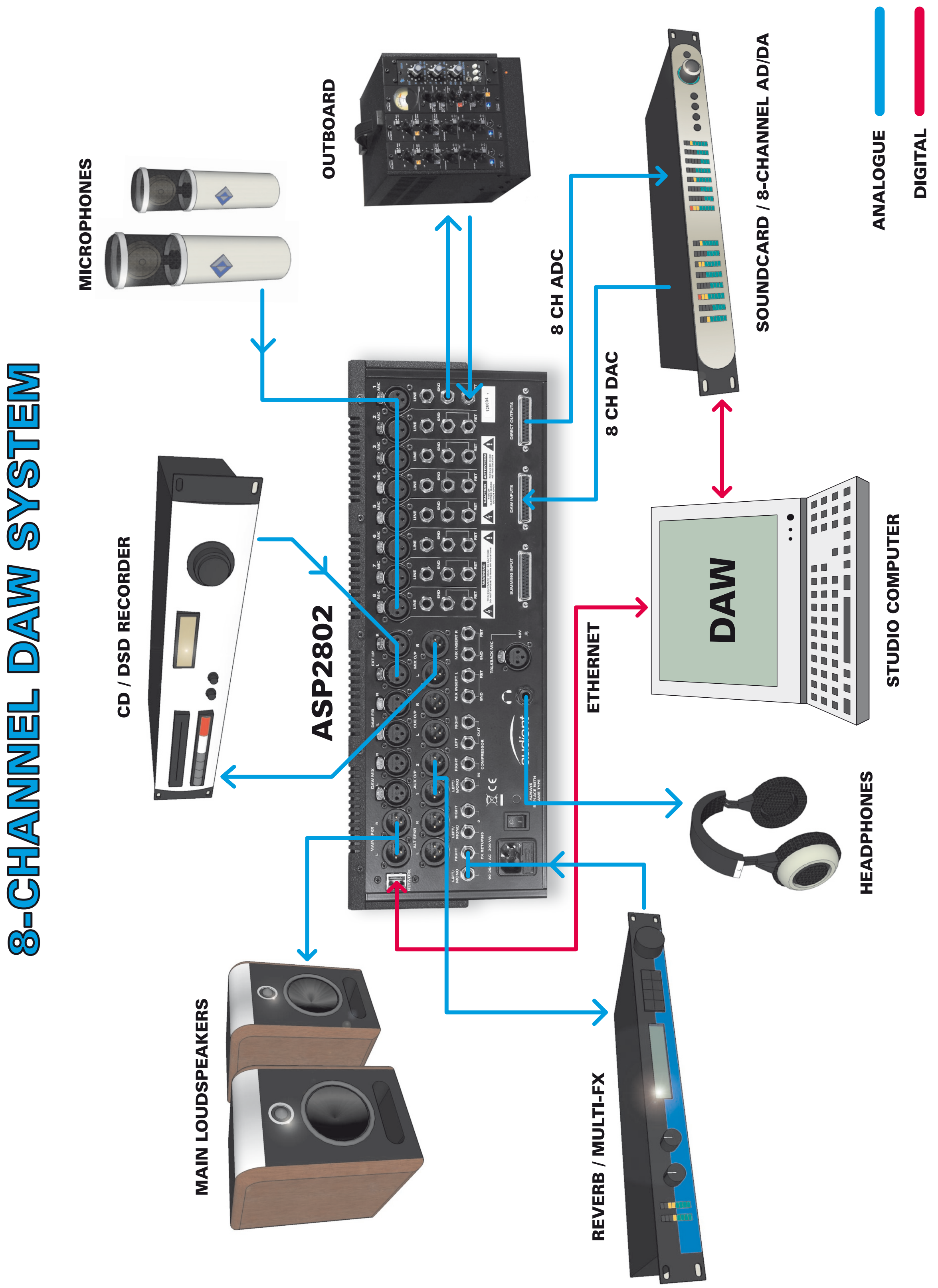
The first 8 channels feed the standard DAW Inputs and can be used to access the main automation channels on ASP2802 when selecting DAW input mode. These 8 channels can also access the outboard selection via the switched insert points.

The following 8 outputs (9-16) feed the ASP2802 line inputs and will feed the alternative cue input.

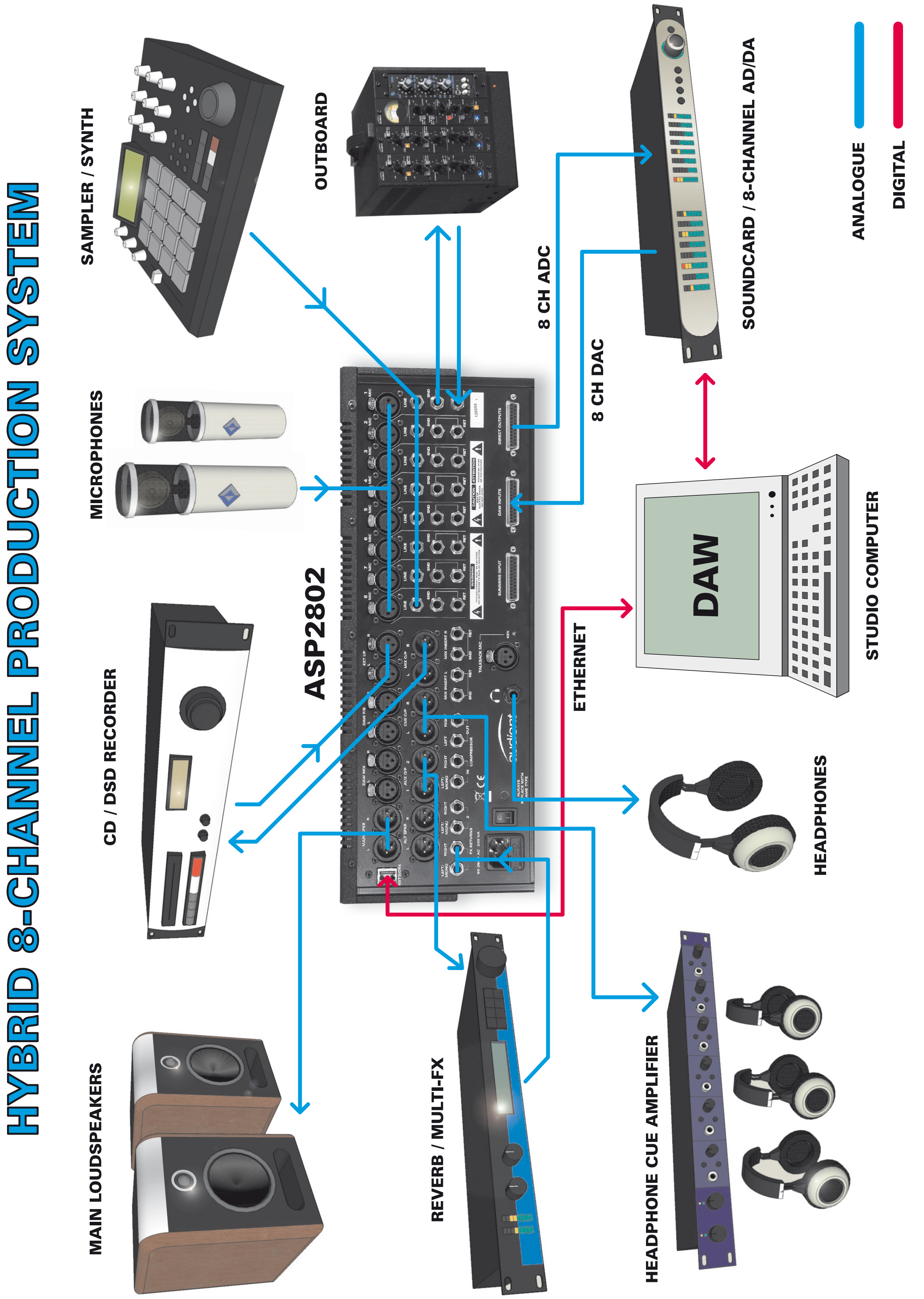
Here these do not go directly to the main mix bus via the cue assign control, instead they are fed into a subgroup compressor via the main cue outputs (channels 9-16), which is then returned into the DAW Mix Input to re-join the main mix bus.

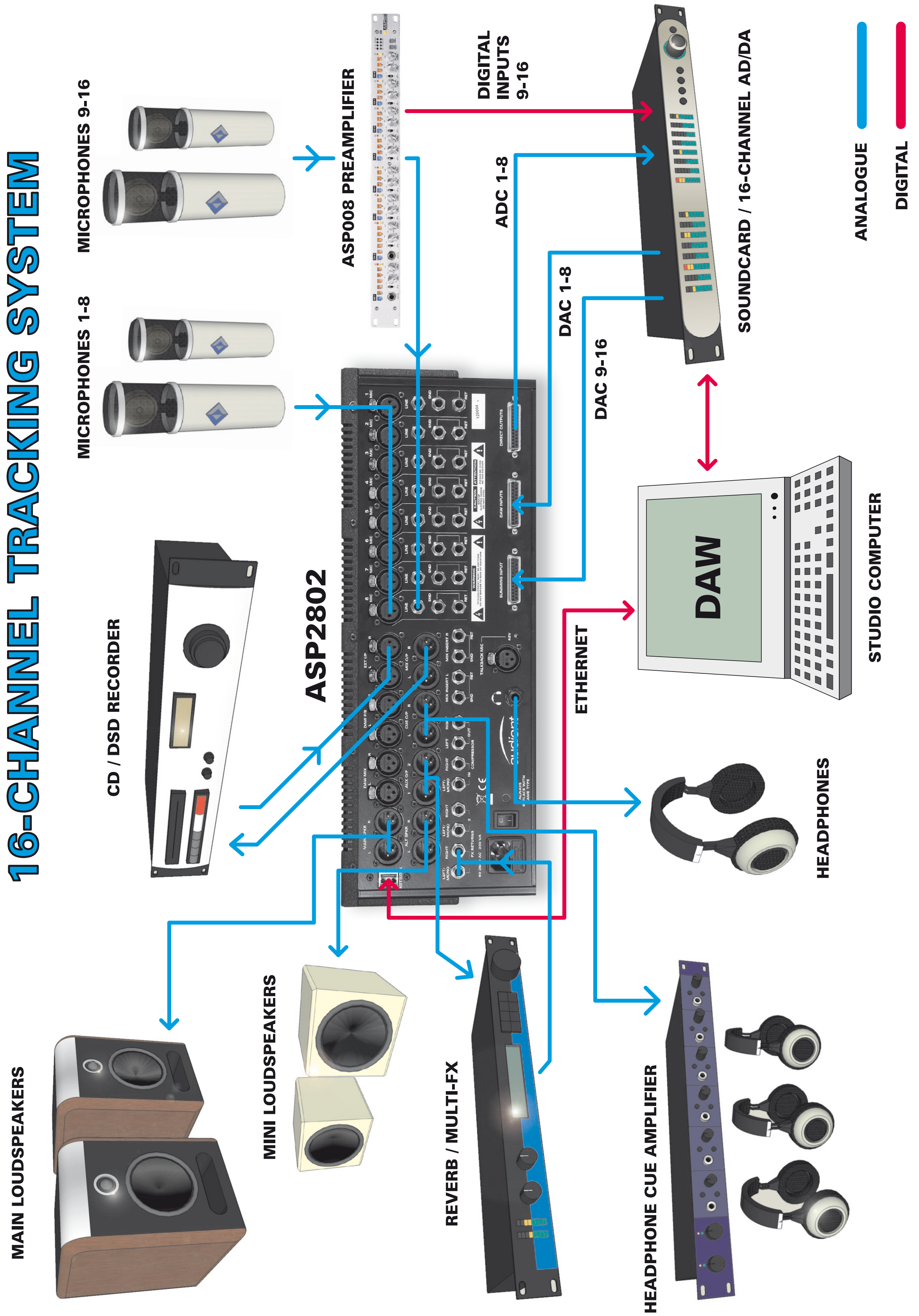
Here you could compress your drum mix for example.
The remaining channels (17-24) feed the summing input which routes directly into the main mix bus.

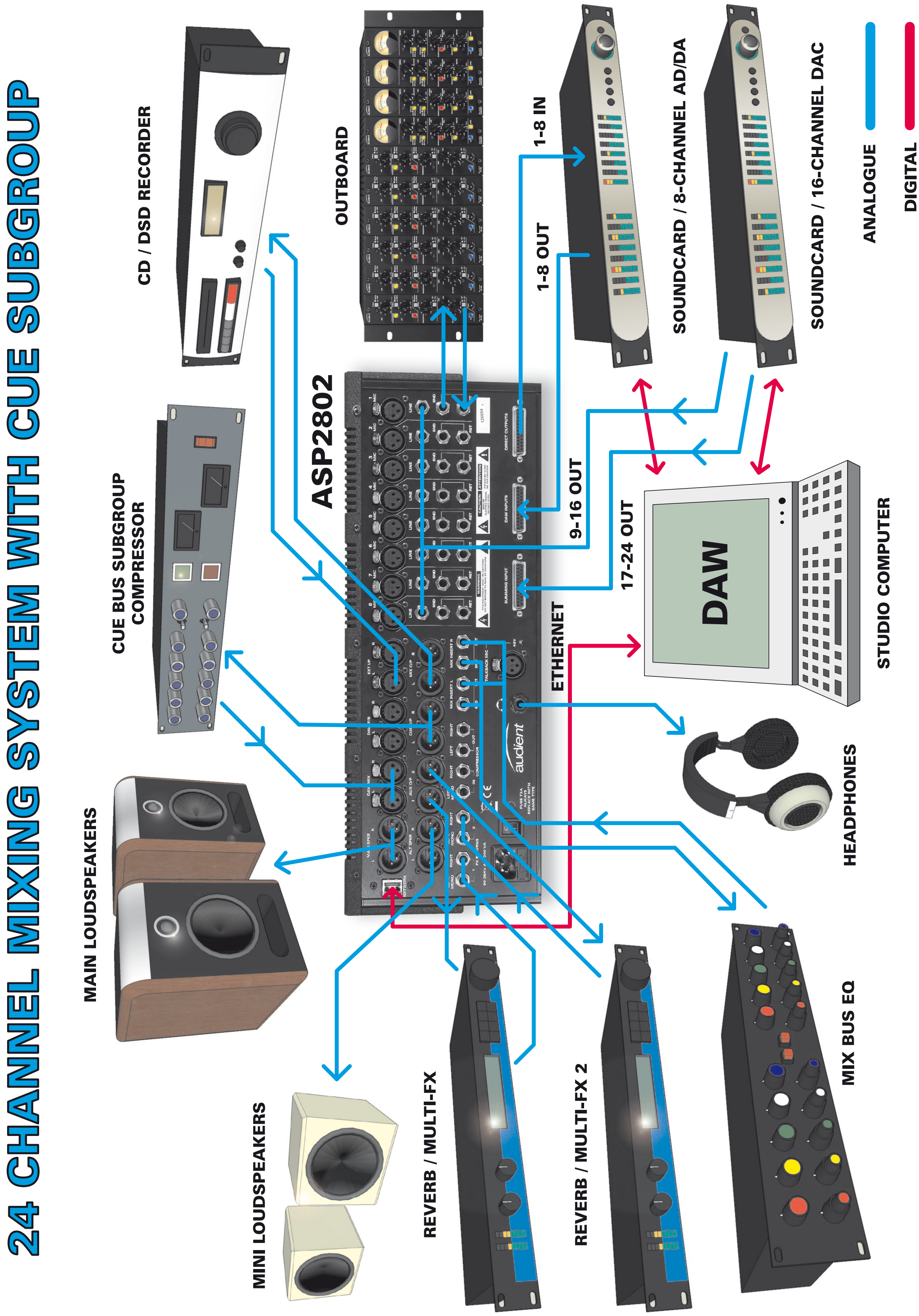
Example System 1



Example System 2




16-CHANNEL TRACKING SYSTEM
MICROPHONES 9-16
MICROPHONES 1-8
ASP008 PREAMPLIFIER
DIGITAL INPUTS 9-16
ADC 1-8
DAC 1-8
DAC 9-16
SOUNDCARD / 16-CHANNEL AD/DA
ANALOGUE
DIGITAL
CD / DSD RECORDER
ASP2802
ETHERNET
DAW
STUDIO COMPUTER
HEADPHONES
MAIN LOUDSPEAKERS
MINI LOUDSPEAKERS
REVERB / MULTI-FX
HEADPHONE CUE AMPLIFIER



24 CHANNEL MIXING SYSTEM WITH CUE SUBGROUP
CD / DSD RECORDER
OUTBOARD
1-8 IN
1-8 OUT
SOUNDCARD / 8-CHANNEL AD/DA
SOUNDCARD / 16-CHANNEL DAC
ANALOGUE
DIGITAL
CUE BUS SUBGROUP COMPRESSOR
ASP2802
9-16 OUT
17-24 OUT
DAW
ETHERNET
STUDIO COMPUTER
HEADPHONES
MAIN LOUDSPEAKERS
MINI LOUDSPEAKERS
REVERB / MULTI-FX
REVERB / MULTI-FX 2
MIX BUS EQ

Panel Visualisations

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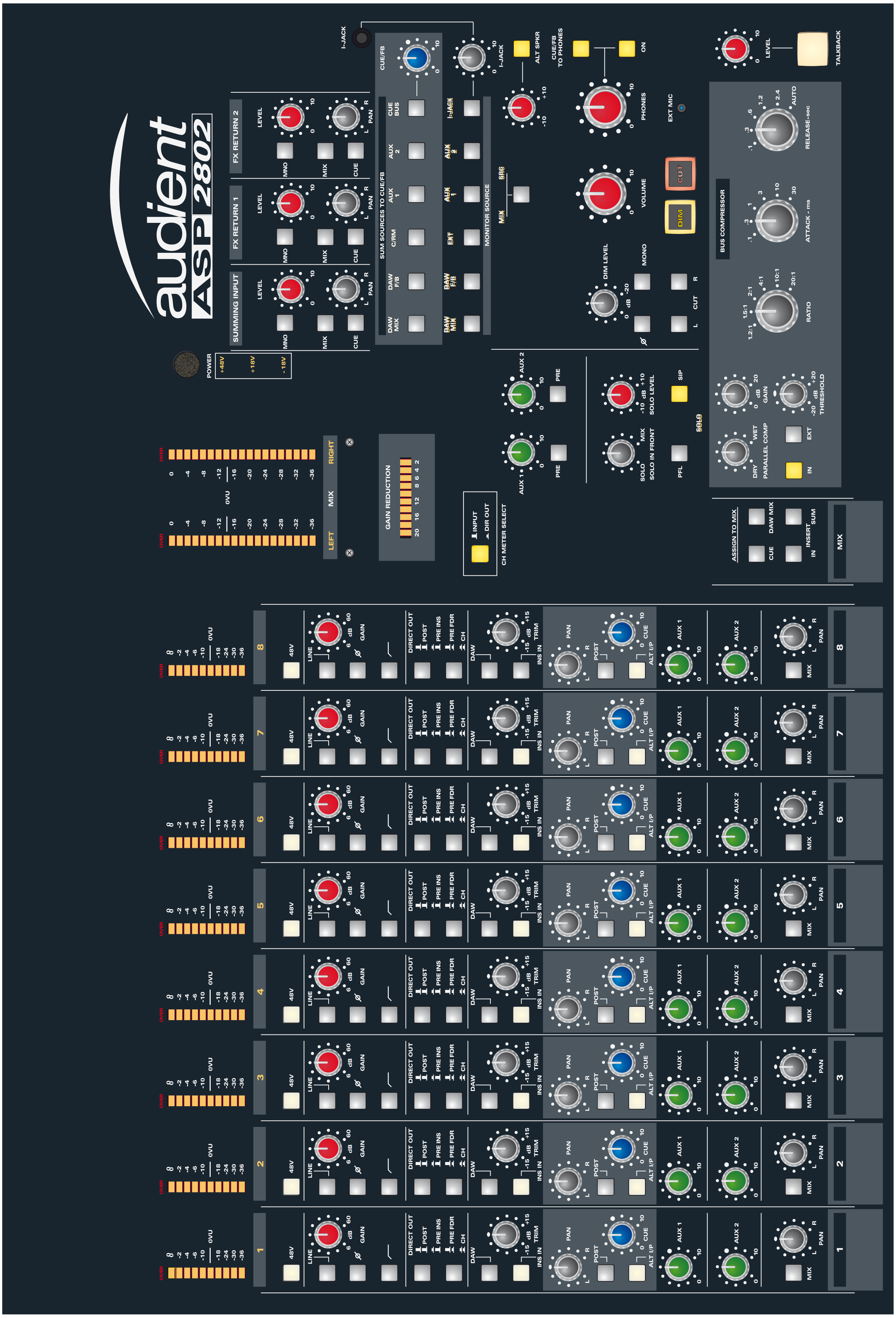
DUAL LAYER CONTROL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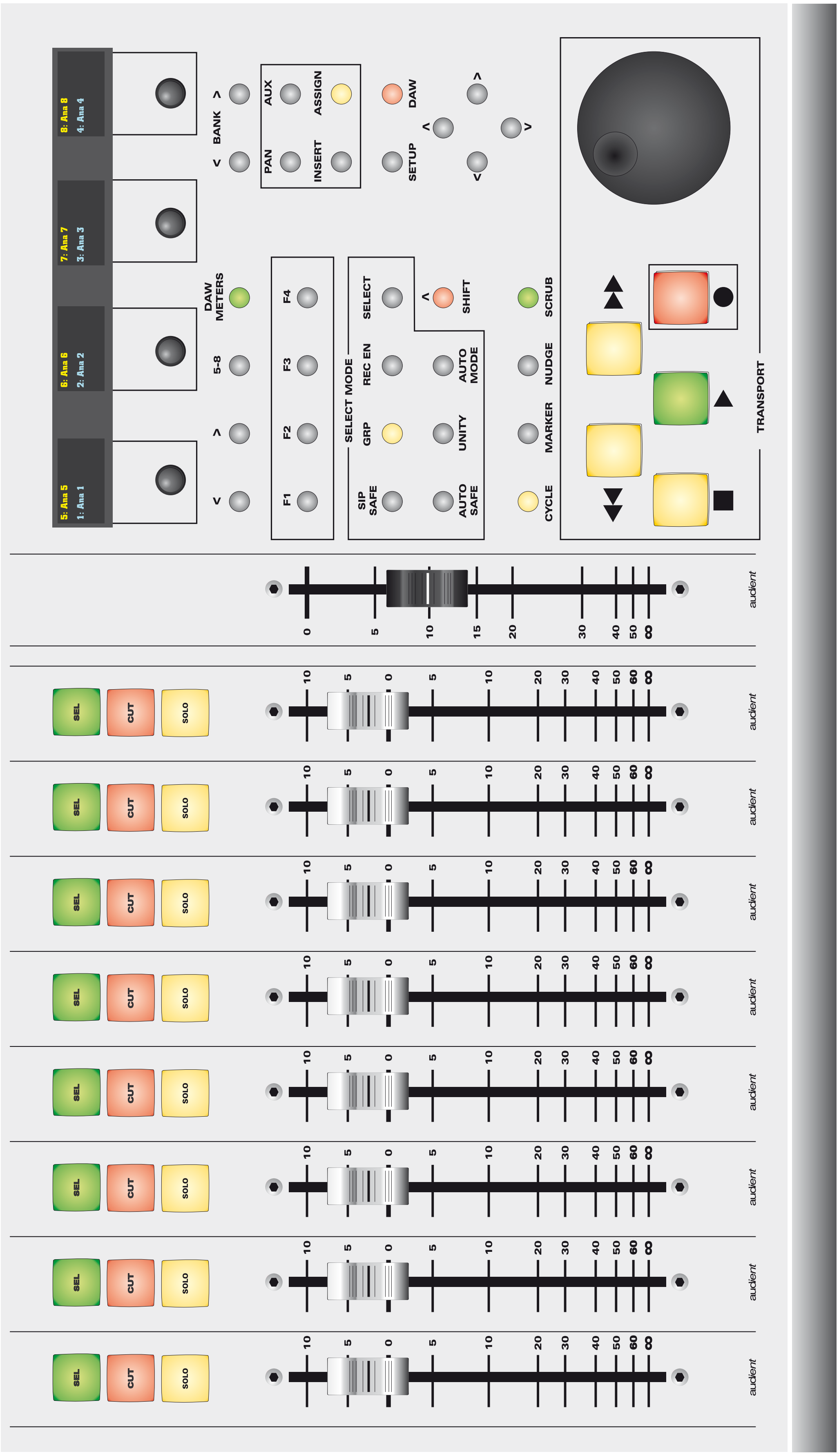
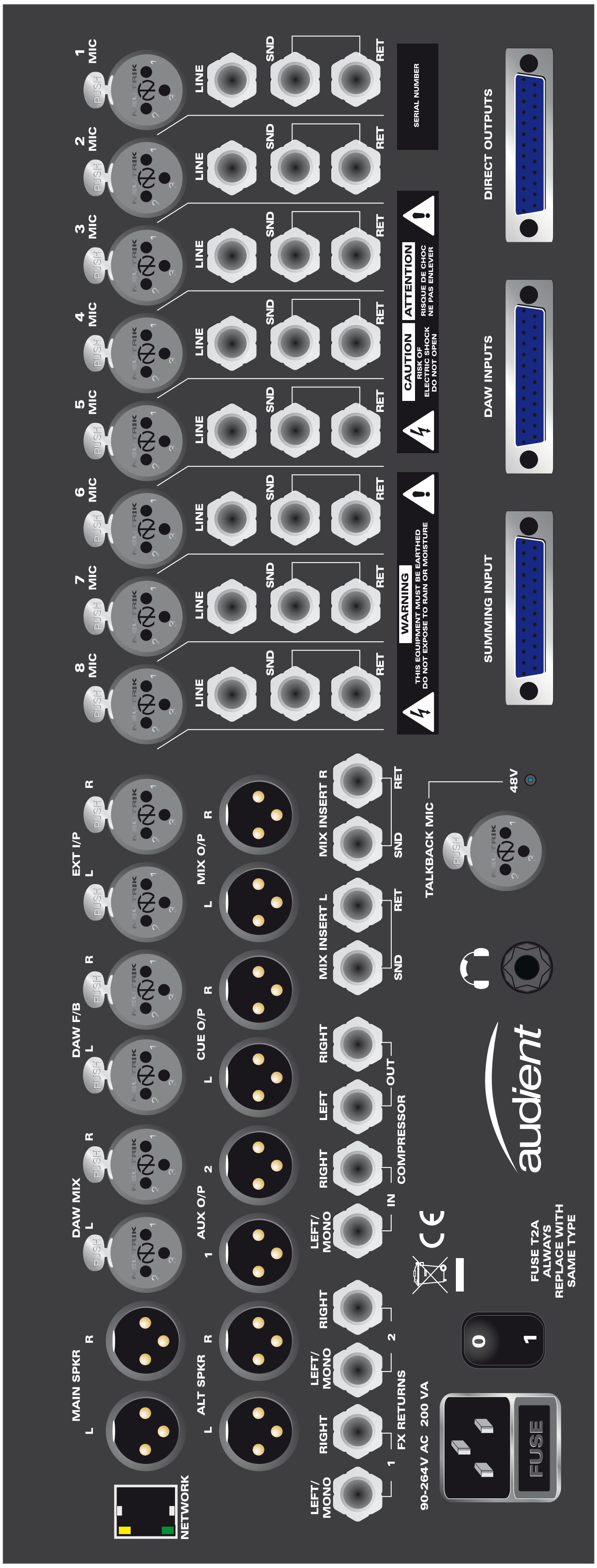
ASP 2802
POWER
SUMMING INPUT
FX RETURN 1
FX RETURN 2
SUM SOURCES TO CUE/FB
MONITOR SOURCE
GAIN REDUCTION
CH METER SELECT
BUS COMPRESSOR
PARALLEL COMP
RELEASE - sec
ATTACK - ms
RATIO
THRESHOLD
ASSIGN TO MIX
PHONES
VOLUME
DIM LEVEL
SOLO LEVEL
SOLO IN FRONT
TALKBACK
LEFT
MIX
RIGHT

Panel Visualisations





1 MIC
2 MIC
3 MIC
4 MIC
5 MIC
6 MIC
7 MIC
8 MIC
LINE
SND
RET
SERIAL NUMBER
CAUTION
RISK OF ELECTRIC SHOCK DO NOT OPEN
ATTENTION
RISQUE DE CHOC NE PAS ENLEVER
WARNING
THIS EQUIPMENT MUST BE EARTHED
DO NOT EXPOSE TO RAIN OR MOISTURE
DIRECT OUTPUTS
DAW INPUTS
SUMMING INPUT
EXT I/P
DAW F/B
DAW MIX
MAIN SPKR
ALT SPKR
MIX O/P
CUE O/P
AUX O/P
MIX INSERT R
MIX INSERT L
TALKBACK MIC
48V
COMPRESSOR
IN
OUT
LEFT
RIGHT
LEFT/ MONO
FX RETURNS
NETWORK
audient
FUSE T2A
ALWAYS REPLACE WITH SAME TYPE
90-264V AC 200 VA
FUSE

Warranty

Your ASP2802 comes with a manufacturer's warranty for one year from the date of dispatch 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 ASP2802 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.

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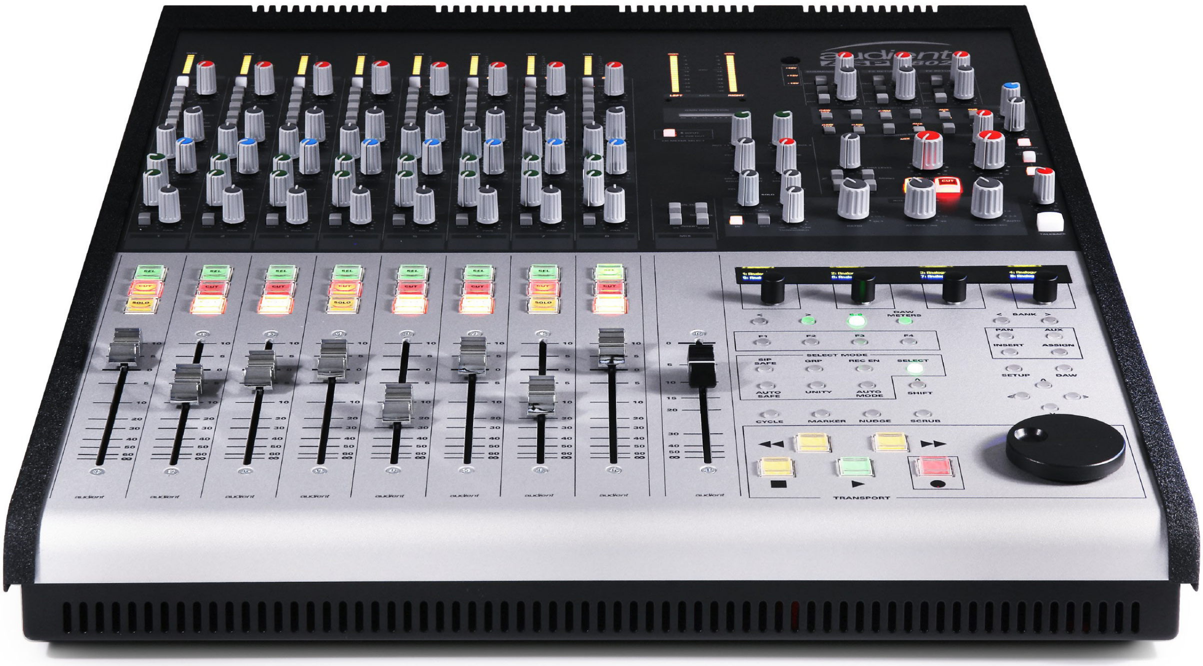
ASP 2802

DUAL
LAYER CONTROL



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