

Control 2802

Operating Manual

Warranty

All Focusrite products are built to the highest standards and should provide reliable performance for many years, subject to reasonable care, use, transportation and storage.

In the event of a Manufacturing Defect becoming evident within 12 months from date of purchase Focusrite undertakes that the product will be repaired or replaced free of charge if the product is returned to the authorised dealer from whom it was purchased. In these circumstances, or if you need an out-of-warranty repair to your Focusrite product, please contact your local Focusrite distributor in your country of residence or business. Find your local distributor and you will be advised of the correct return procedure. Alternatively contact the Focusrite Reseller from which you purchased the product. If you purchase a Focusrite product outside your country of residence or business you will not be entitled to ask your local Focusrite distributor to honour this Limited Warranty, although you may request a chargeable repair.

Alternatively, the unit may be returned at your cost to the dealer you purchased the unit from so that they can organise a Warranty repair with their Focusrite distributor. This Warranty does not include cost of shipping to and from the authorised dealer from whom it was purchased. In every case it will be necessary to provide the original invoice or store receipt to accompany the defective product to the supplying dealer.

This Limited Warranty is offered solely to the first purchaser of the product from an Authorised Focusrite Reseller (defined as a reseller which has purchased the Product directly from Focusrite Audio Engineering Limited in the UK or its Authorised Distributors outside the U.K.) and is not transferable. This Warranty is in addition to your Statutory Rights in the country of purchase.

Please note: A Manufacturing Defect is defined as a defect in the performance of the product which may be expected from a reasonable interpretation of the published description and performance specifications as published by Focusrite Audio Engineering Limited. This does not include damage caused by post-purchase transportation, storage or careless handling, nor damage caused by misuse. A significant proportion of products returned under Warranty (which are very few in number compared to numbers sold) are found not to exhibit any fault at all. Please check that the mains voltage is correctly set for your local supply and that your connecting cables are in good order and correctly connected. If in doubt about the product functions please read the appropriate section of the user guide, and if necessary contact your dealer for advice before returning the product to the supplying dealer. You can also contact the Focusrite Technical support team should you require further assistance.

Accuracy

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Welcome to Control 2802

Thank you for choosing this Focusrite product. **Control 2802** is an analogue recording console and DAW (Digital Audio Workstation) control surface in one.

To cover both parts of the system there are two separate manuals:

- **Control 2802 Operating Manual** - this manual introduces the system and covers the analogue console. Please read this first.
- **Control 2802 Automation & Control Surface Manual** - covers the DAW control layer and analogue fader automation system. You will find separate documents for each of the major DAWs: ProTools, Logic Pro and Cubase.

All the documentation is included on the CD packed with the console, or may be downloaded from our website at: www.focusrite.com

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Important Safety Instructions

1. Read these instructions.
2. Keep these instructions.
3. Heed all warnings.
4. Follow all instructions.
5. Do not use this apparatus near water.
6. Clean only with dry cloth.
7. Do not block any ventilation openings. Install in accordance with the manufacturer's instructions.
8. Do not install near any heat sources such as radiators, heat registers, stoves, or other apparatus (including amplifiers) that produce heat.
9. Do not defeat the safety purpose of the polarized or grounding-type plug. A polarized plug has two blades with one wider than the other. A grounding type plug has two blades and a third grounding prong. The wide blade or the third prong are provided for your safety. If the provided plug does not fit into your outlet, consult an electrician for replacement of the obsolete outlet.
10. Protect the power cord from being walked on or pinched particularly at plugs, convenience receptacles, and the point where they exit from the apparatus.
11. Only use attachments/accessories specified by the manufacturer.
12. Use only with the cart, stand, tripod, bracket, or table specified by the manufacturer, or sold with the apparatus. When a cart is used, use caution when moving the cart/apparatus combination to avoid injury from tip-over.



13. Unplug this apparatus during lightning storms or when unused for long periods of time.
14. Refer all servicing to qualified service personnel. Servicing is required when the apparatus has been damaged in any way, such as power-supply cord or plug is damaged, liquid has been spilled or objects have fallen into the apparatus, the apparatus has been exposed to rain or moisture, does not operate normally, or has been dropped.

WARNING: To reduce the risk of fire or electric shock, do not expose this apparatus to rain or moisture.

- Do not expose this apparatus to drips or splashes.
- Do not place any objects filled with liquids, such as vases, on the apparatus.
- Do not install this apparatus in a confined space such as a bookcase or similar unit.
- Slots and openings in the cabinet are provided for ventilation and to ensure reliable operation of the product and to protect it from overheating. Please ensure adequate space around the apparatus for sufficient ventilation. Ventilation should not be impeded by covering the ventilation openings with items such as newspapers, tablecloths curtains etc.
- The apparatus draws nominal non-operating power from the AC outlet with its POWER switch in the off position.
- The apparatus should be located close enough to the AC outlet so that you can easily grasp the power cord plug at any time.
- An apparatus with Class 1 construction shall be connected to an AC outlet with a protective grounding connection.
- The MAINS plug or the appliance coupler is used as the disconnect device. Either device shall remain readily operable when the apparatus is installed for use.
- No naked flames, such as lighted candles, should be placed on the apparatus.

WARNING: excessive sound pressure levels from earphones and headphones can cause hearing loss.

GB

This equipment must be earthed by the power cord

FIN

Laite on liitettävä suojavaadoituskoskettimilla varustettuun pistorasiaan

NOR

Apparatet må kun tilkoples jordet stikkontakt

Important Safety Precautions



CAUTION
RISK OF ELECTRIC SHOCK
DO NOT OPEN



CAUTION: TO REDUCE THE RISK OF ELECTRIC SHOCK, DO NOT REMOVE COVER (OR BACK). NO USER-SERVICEABLE PARTS INSIDE. REFER SERVICING TO QUALIFIED SERVICE PERSONNEL.



The lightning flash with arrowhead symbol, within equilateral triangle, is intended to alert the user to the presence of uninsulated "dangerous voltage" within the product's enclosure that may be of sufficient magnitude to constitute a risk of electric shock to persons.



The exclamation point within an equilateral triangle is intended to alert the user to the presence of important operating and maintenance (servicing) instructions in the literature accompanying the appliance.

WARNING: TO PREVENT FIRE OR SHOCK HAZARD, DO NOT EXPOSE THIS APPLIANCE TO RAIN OR MOISTURE

Environmental Declaration

Compliance Information Statement: Declaration of Compliance procedure

Product Identification:	Focusrite Control 2802
Responsible party:	American Music and Sound
Address:	5304 Derry Avenue #C Agoura Hills, CA 91301
Telephone:	800-994-4984

This device complies with part 15 of the FCC Rules. Operation is subject to the following two conditions: (1) This device may not cause harmful interference, and (2) this device must accept any interference received, including interference that may cause undesired operation.

For USA to the User:

- Do not modify this unit! This product, when installed as indicated in the instructions contained in this manual, meets FCC requirements. Modifications not expressly approved by Focusrite may void your authority, granted by the FCC, to use this product.
- Important: This product satisfies FCC regulations when high quality shielded cables are used to connect with other equipment. Failure to use high quality shielded cables or to follow the installation instructions within this manual may cause magnetic interference with appliances such as radios and televisions and void your FCC authorization to use this product in the USA.
- This equipment has been tested and found to comply with the limits for a Class A digital device, pursuant to the part 15 of the FCC Rules. These limits are designed to provide reasonable protection against harmful interference when the equipment is operated in a commercial environment. This equipment generates, uses and can radiate radio frequency energy and, if not installed and used in accordance with the instruction manual, may cause harmful interference to radio communications. Operation of this equipment in a residential area is likely to cause harmful interference in which the user will be required to correct the interference at his own expense.

For Canada to the User:

This Class A digital apparatus complies with Canadian ICES-003.

Cet appareil numérique de la classe A est conforme à la norme NMB-003 du Canada.

RoHS Notice

Focusrite Audio Engineering Limited has conformed and this product conforms, where applicable, to the European Union's Directive 2002/95/EC on Restrictions of Hazardous Substances (RoHS) as well as the following sections of California law which refer to RoHS, namely sections 25214.10, 25214.10.2, and 58012, Health and Safety Code; Section 42475.2, Public Resources Code.

Installation

Package Contents

Your packaging should contain the following:

- **Control 2802** console
- IEC power cord
- CD containing the documentation and drivers.

Check for any signs of transit damage. If any signs of mishandling are found please notify the carrier and inform your dealer immediately.

Important Safety Instructions

Please read all of the safety instructions at the beginning of this manual 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 if 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

The console 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 100-240v $\pm 10\%$. Therefore, the console will work anywhere in the world provided that your A.C mains line voltage is within this specification.

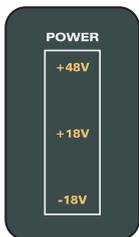
Consult a qualified technician if you suspect difficulties.

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.

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:



Service and Repair

The console uses a complex internal PCB sandwich arrangement making field service only possible by a qualified technician.

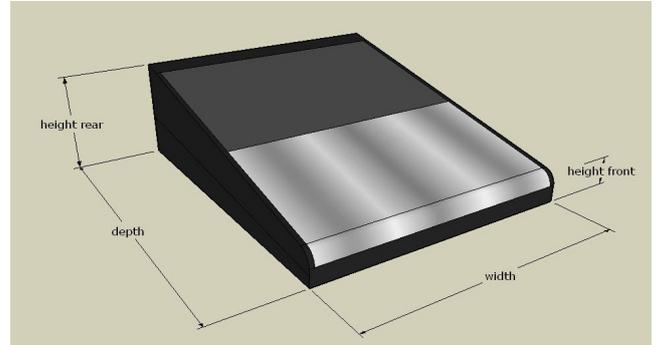
If any technical issues do arise with your console, please contact your dealer as soon as possible to arrange for technical support.

Do **NOT** attempt to fix the console unless qualified to do so.

See the warranty information at the beginning of this manual for details of your cover.

Dimensions (including rubber feet)

- Height Front: 60 mm
- Height Rear: 170 mm
- Width: 450 mm
- Depth: 550 mm

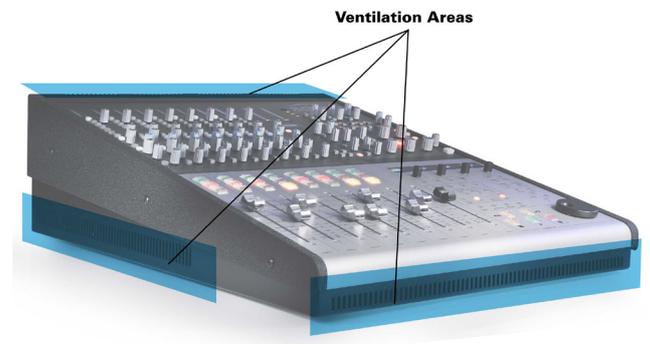


Ventilation

Care should be taken not to obstruct the series of ventilation holes in the metalwork of the console. The unit 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.

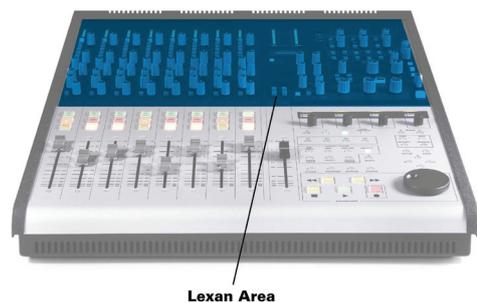
If mounting the console into some form of studio furniture or desk, **please ensure that there are sufficient air gaps at these locations:**



Lexan Overlay

The front panel at the top of the console (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.



Introduction

Control 2802 is both an analogue recording console and DAW control surface.

Analogue Console - 8 Channel Strips and Master Section:



With eight class-A hybrid discrete mic preamps, LED bargraph metering, input signal conditioning, assignable direct outputs and cue monitoring, **Control 2802** is ideal as a location recording or tracking console.

Control 2802 is equally comfortable as a hybrid mixing tool – featuring a DAW control surface layer and up to 32 analogue inputs on mixdown.

The high number of inputs is achieved via 16 inputs in the channels (mic/line + alternative cue input), plus a further 16 inputs in the master section (8-channel Summing Input, two stereo FX returns, stereo DAW Mix input & stereo mix insert running in “parallel sum” mode).

Each mic/line channel has two aux sends, a stereo cue (or alternate input), switchable insert and DAW return for in-line recording. The stereo mix bus includes a master fader, insert point and bus compressor, which may be removed from the main mix and patched elsewhere using its own external IO.

The monitor section includes mono, dim, phase and talkback, plus an i-Jack input for monitoring MP3 players. Outputs connect to main and alternate speakers plus headphones.

DAW Controller - 8 dual-layer Fader Strips and Control Panel:



The 8 Fader Strips (faders, **SEL**, **CUT** & **SOLO**) flip between controlling analogue channel levels and track faders within the DAW. This dual-layer technology makes it easy to switch from analogue to DAW control in an instant.

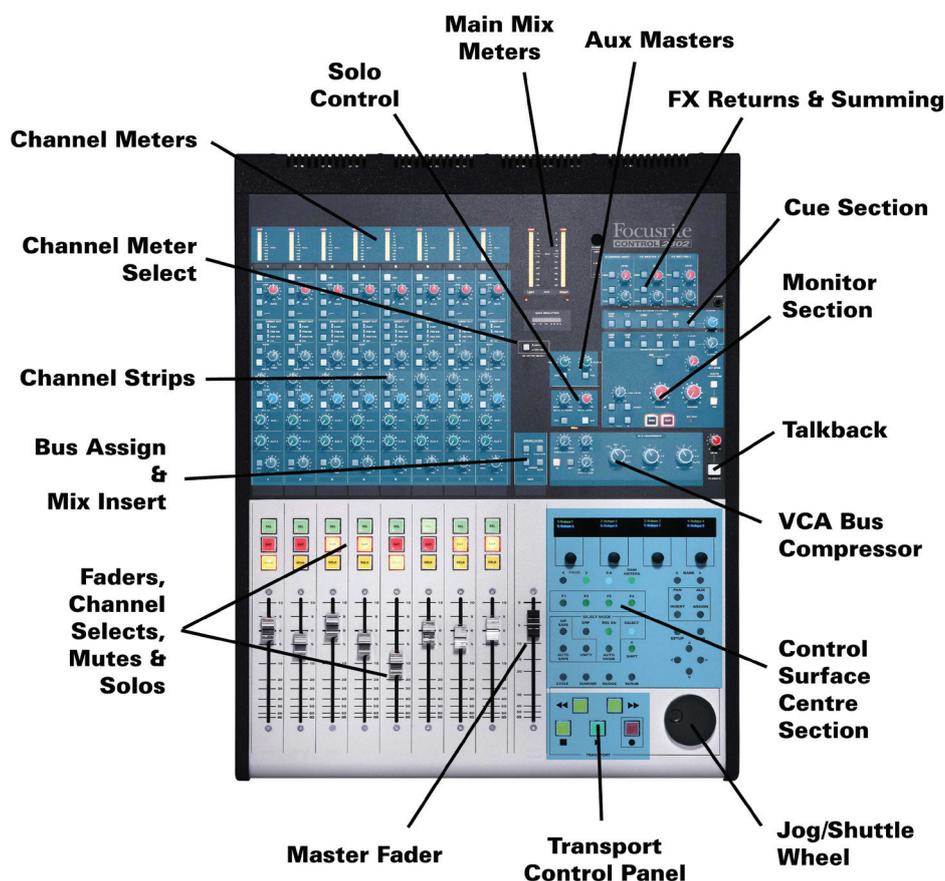
When working in the DAW layer, the console remotely adjusts parameters within the DAW software via an Ethernet connection (using the HUI™ control surface protocol).

At any one time, 8 DAW tracks may be adjusted from the console’s Fader Strips. These are the tracks “in focus” within the DAW software (paged to the HUI™ controller).

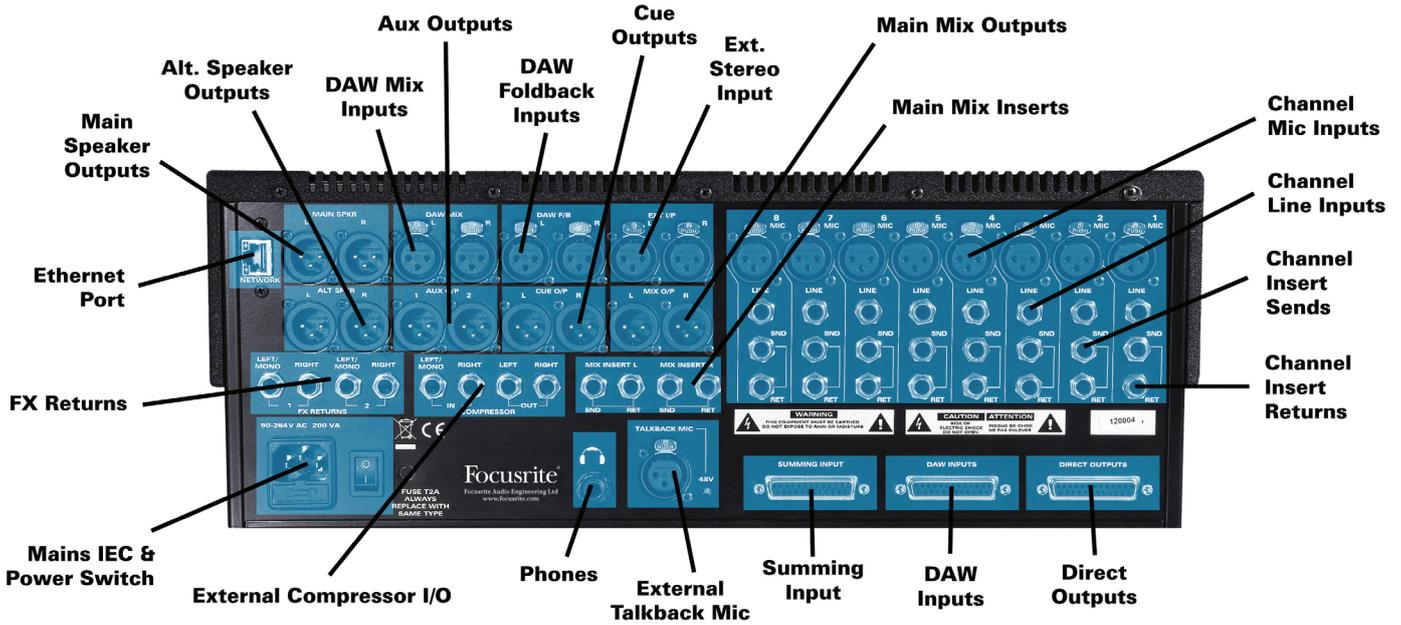
Each of the 8 channel faders is motorised, and both the analogue and DAW layers can be fully automated.

To the right of the channel faders is the master fader (not automated or dual-layer), which controls the main mix bus.

The Fader Strip layer switching, and other DAW parameters, such as auxes, plug-ins, transport and automation, are controlled from Control Surface Centre Section - this includes a jog/shuttle wheel, transport buttons, 4 rotary encoders and OLED channel displays.



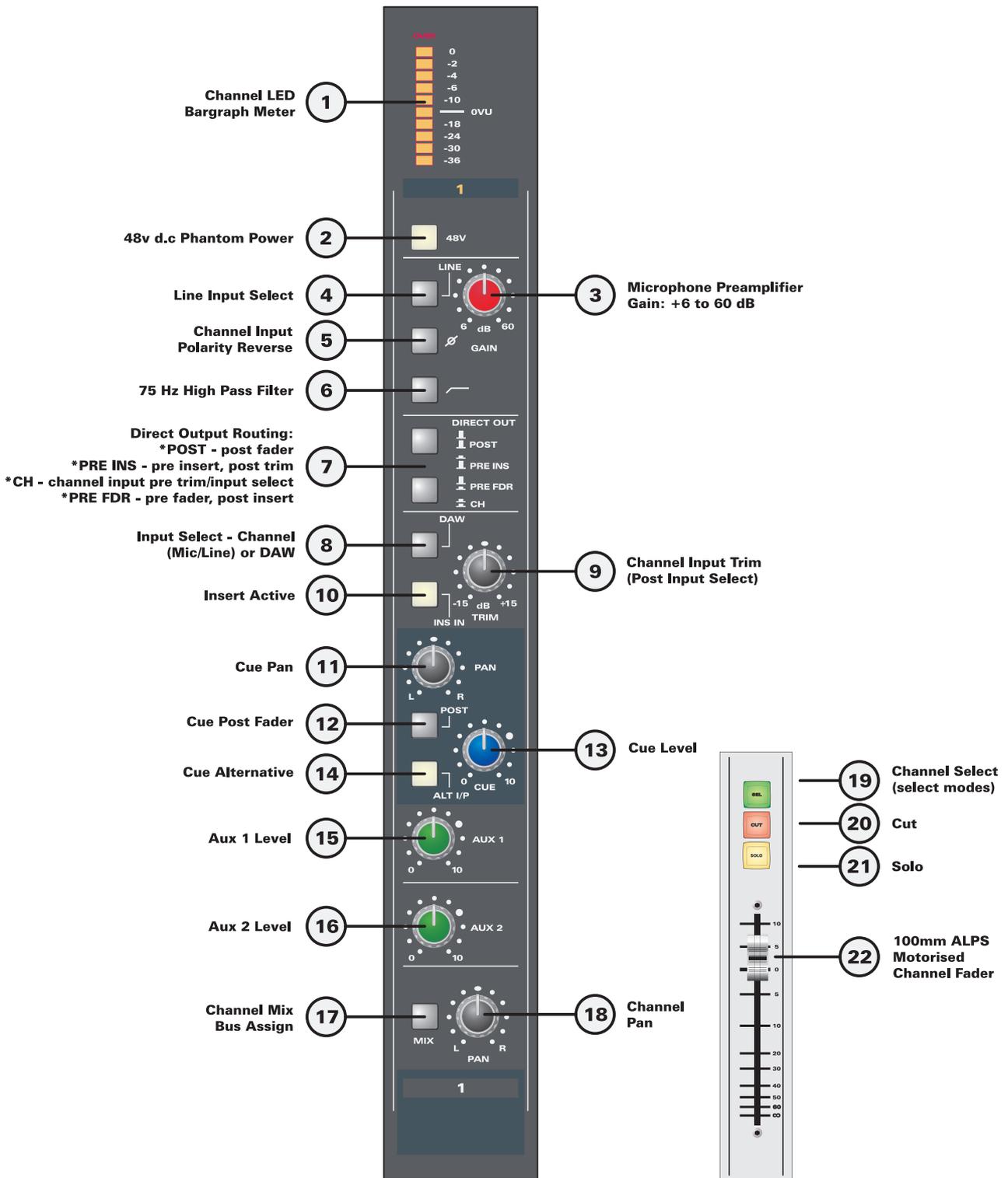
Rear Panel Connections



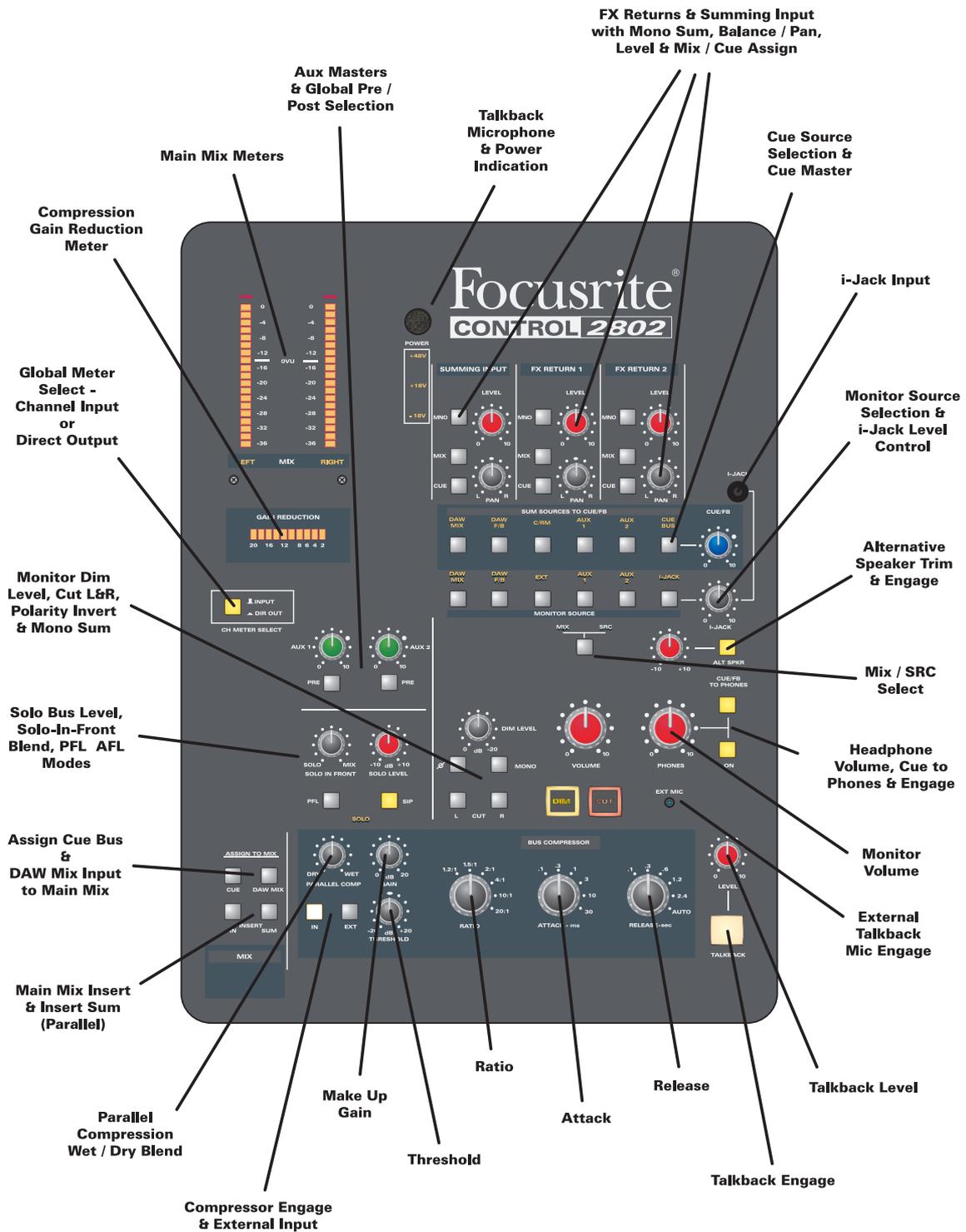
The following table summarises the connections. Please also refer to the example systems on page 42 and page 46 for wiring and pin-out information. For information on cabling the console, DAW and outboard devices into a patchbay system, see page 51.

Connector Name	Type	Typical Application
Channel MIC	XLR Female	Channel microphone inputs.
Channel LINE	1/4" TRS Jack	Channel line level inputs - for line inputs such as synths, samplers or external mic preamps.
Channel SND	1/4" TRS Jack	Channel insert sends - to outboard processing.
Channel RET	1/4" TRS Jack	Channel insert returns - from outboard processing.
DIRECT OUTPUTS	DB25 Female	Channel direct out - this is the main record path from the channel; connect to your DAW tracks.
DAW INPUTS	DB25 Female	Channel DAW input - for playback post-recording through the channel; connect from your DAW tracks.
SUMMING INPUT	DB25 Female	8 Summing Inputs - for additional mix inputs such as submixes from your DAW or extra effects.
NETWORK	RJ45	Ethernet connection to/from your DAW.
MAIN SPKR L&R	XLR Male	Main speaker outputs - to main control room loudspeakers.
DAW MIX L&R	XLR Female	DAW Mix input - connect from your DAW's internal mix bus; this input may feed the console's mix bus, cue and monitor section.
DAW FB L&R	XLR Female	DAW Foldback input - connect from headphone mixes generated within your DAW; this input may feed the cue and monitor sections.
EXT I/P L&R	XLR Female	External Input - this input may feed the monitor section; connect your stereo recorder to check printed mixes.
ALT SPKR L&R	XLR Male	Alt speaker outputs - to a second set of control room speakers.
AUX O/P 1&2	XLR Male	Aux outputs - send to effects such as reverb & delay.
CUE O/P L&R	XLR Male	Cue output - send to your artists headphones or a stereo effect.
MIX O/P L&R	XLR Male	Main Mix output - to a master recorder, or DAW, to print mixes.
FX RETURN 1 and 2 L&R	1/4" TRS Jack	FX Return inputs - for extra mix inputs such as reverb returns.
COMPRESSOR IN and OUT L&R	1/4" TRS Jack	External Compressor IO - patches the bus compressor.
MIX INSERT SND and RET L&R	1/4" TRS Jack	Main Mix insert - send/return main mix outboard processing.
TALKBACK	XLR Female	External Talkback mic - connect an alternate mic for talkback.
Phones	Headphone socket	For control room headphones.

Channel Strip Overview



Master Section Overview



DAW Control / Fader Automation Overview

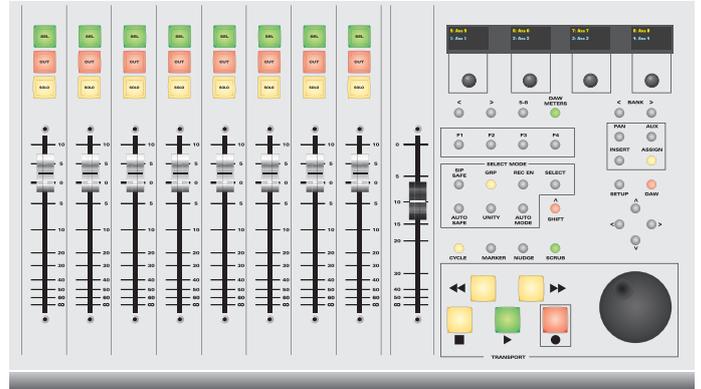
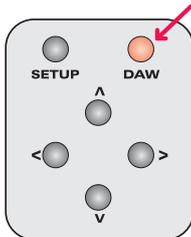
The lower part of the control surface can remotely adjust parameters within the DAW software, and respond to HUI™ / MIDI automation allowing you to automate both the DAW and analogue fader layers. To use these features you must connect **Control 2802** to your DAW computer via Ethernet, and configure the DAW software to support **Control 2802** as a HUI™ control surface.

You can find full details on the setup and operation in the “Automation & Control Surface Manual” which can be downloaded from our website at www.focusrite.com. Here you will find separate documents for each of the major DAWs: ProTools, Logic Pro and Cubase.

Dual Layer Fader Strips

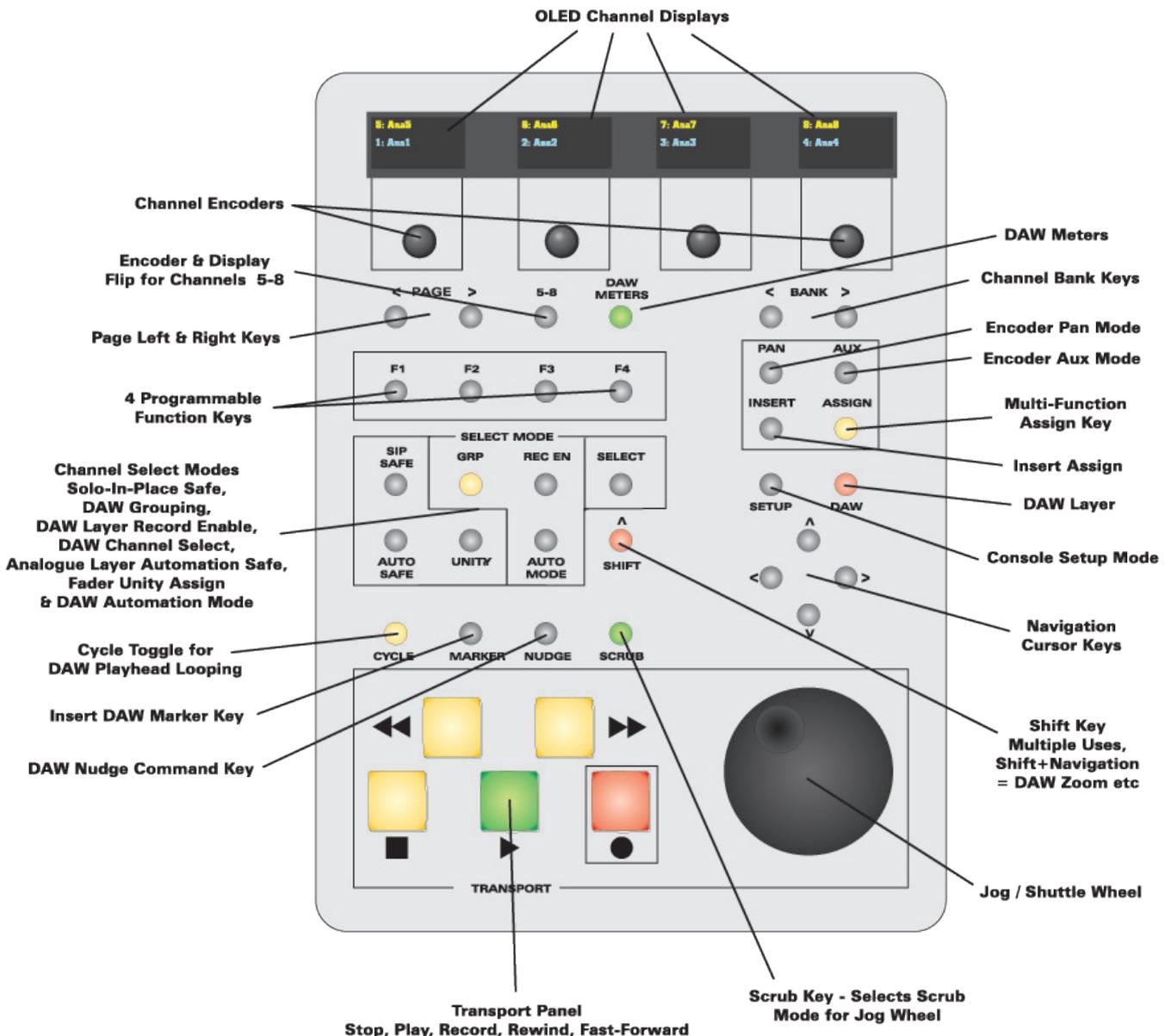
The 8 Fader Strips (faders, **SEL**, **CUT** & **SOLO**) flip between controlling analogue levels and track faders within the DAW.

Select the **DAW** switch, on the Control Surface Panel, to flip all 8 Fader Strips to the DAW layer:



Deselect **DAW** (not illuminated) to control the analogue layer.

Control Surface Panel Overview



Monitoring Audio (Quick-start)

Each channel has 3 inputs: **MIC**, **LINE** and **DAW**.

The default input is mic. To change input, select either **LINE** (4) or **DAW** (8).

DAW overrides **MIC/LINE** as the DAW return is inserted at a later point in the channel signal flow.

To get signal from the **MIC** input to the **MAIN SPKR L/R** outputs (on the rear connector panel):

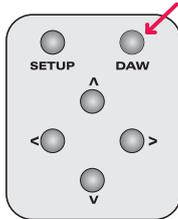
1. With the mic input selected, adjust the mic gain (3) and 48V phantom power (2) - you should see input level on the channel meter (1).

If not, check that **CH METER SELECT** in the centre of the console is set to **INPUT** (in the UP position):

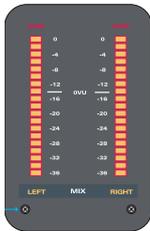


2. Select **MIX** bus assign (17) to route the channel to the main mix bus, and open the channel fader (22).

The fader can flip between controlling the analogue channel level or a DAW fader. So make sure that the **DAW** switch, located on the Control Surface Panel, is **NOT** illuminated so that you are controlling the analogue layer:



3. Then open the master fader - you should see level on the **LEFT & RIGHT MIX** meters:



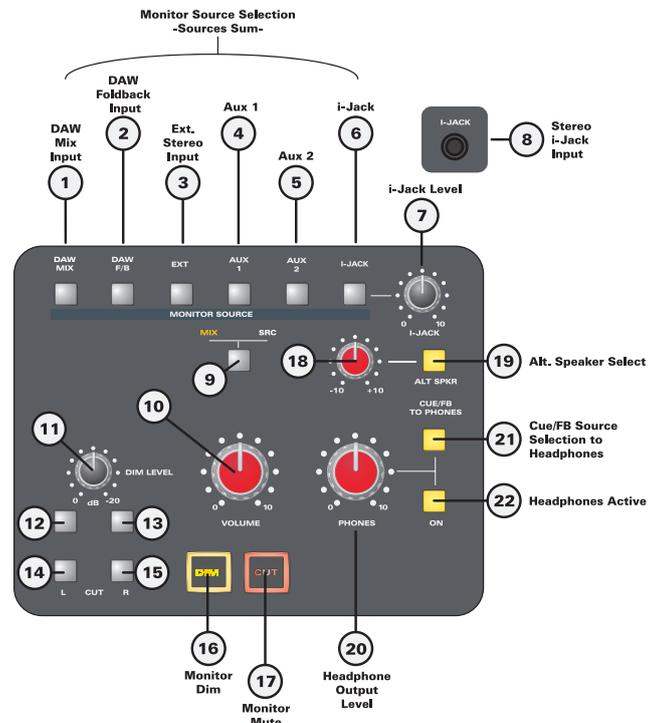
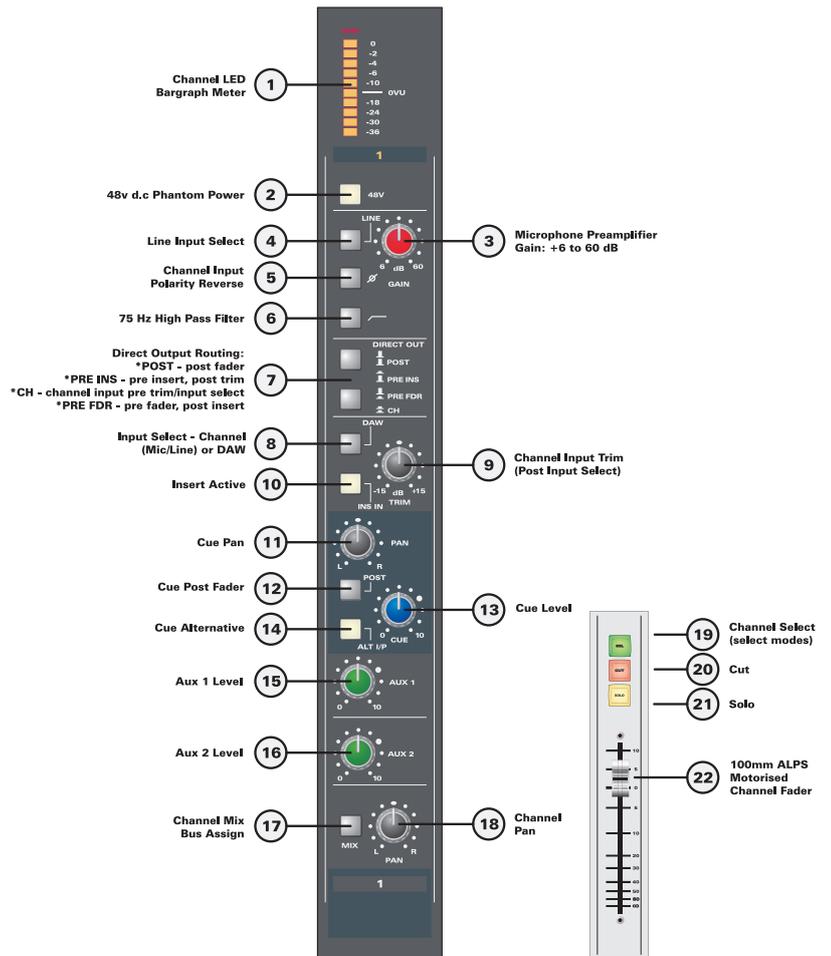
If not, check that the channel is not **CUT** (20) or that a different channel is not in **SOLO** (21).

Also check that **MIX** is selected as the **MONITOR SOURCE** in the Monitor Section (9).

4. Open the monitor **VOLUME** control (10) and you should be hearing audio!

If not, check that:

- The **MONITOR SOURCE** is not switched to **SRC** (9).
 - The monitor **CUT** and **DIM** (16 & 17) are not active.
 - The individual **L R** monitor cuts are not active (14 & 15).
 - The **ALT SPKR** switch (19) is not selected.
5. To listen on headphones, engage the headphones **ON** switch (22), make sure that **CUE/FB TO PHONES** (19) is off, and turn up the **PHONES** volume (20) - the control room monitor output is now sent to the headphone jack connector on the rear connector panel.



Signal Flow Overview

Channel Inputs

The console has eight channel strips, each with three main inputs:

- **Mic** – XLR (female).
- **Line** – 1/4" TRS jack
- **DAW** – 25-pin D-sub (female)

Input Gain & Conditioning

The microphone preamplifier provides 6 to 60 dB of gain, delivered from our 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.

The 75Hz high pass filter (HPF) and polarity reverse (phase invert) are available on either the microphone preamplifier or line input paths. 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.

Typical Recording Path

The mic/line path normally feeds the main channel, which in turn feeds the direct output and the recorder. The direct output can be taken from the channel input, pre-insert, pre-fader or post-fader providing different workflow options.

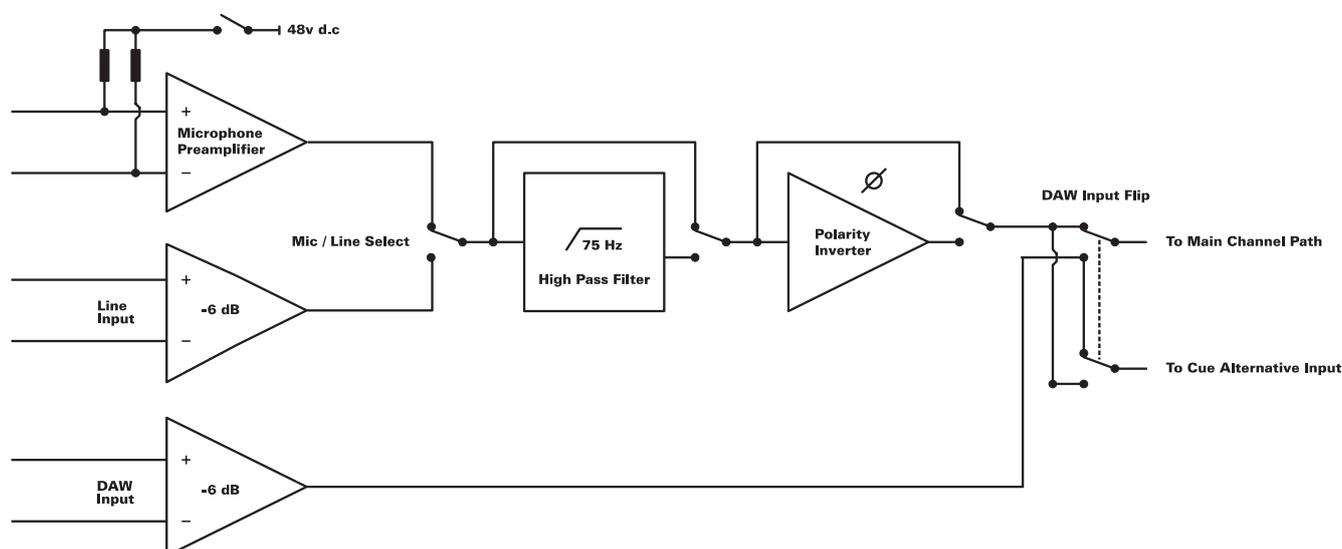
The return from the recorder feeds back into the console via the DAW input. The main channel path can be flipped from mic/line input to DAW return to monitor the recorded signal.

* TIP!

By combining the **CHANNEL** direct output routing mode with the **DAW** input flip and alternative cue input (**ALT I/P**), you can simultaneously:

- Record mic preamp or line inputs direct to disk (clean).
- Generate a cue send directly from mic/line inputs for your artist's headphones (with zero latency).
- Monitor the DAW returns (post record) through the main channel path so that you can hear any colouration provided by your workstation.

Channel Input Signal Flow:



Mixdown Summing Options

For mixdown, there are two possible summing options:

- **Mix in the analogue domain (inside Control 2802)** - for fast access to your recorded tracks, switch the DAW input to the main channel path. You are now playing back from the DAW through the analogue console, and summing through the **Control 2802**'s analogue mix bus.

Using this option you can sum up to 8 DAW returns through the main channels. (Combine these with the console's other inputs to mix up to 32 analogue inputs as described below).

* TIP!

Use the analogue fader automation to apply fader and cut automation to the 8 main channel paths.

- **Mix "in the box" (inside the DAW)** - using the DAW software, route your tracks onto the internal summing bus. Tracks are now summed inside the DAW and fed back to the console either via a pair of DAW inputs or the stereo DAW MIX input connector.

Using this option you can sum any number of DAW tracks "in the box", and combine these with the console's other inputs.

* TIP!

Flip the Fader Strips to the DAW layer to remotely control faders and cuts on any 8 DAW tracks.

Up to 32 Mix Inputs

In addition to the eight main channels, the cue send may be used as an alternative input path. With the **ALT I/P** switch active, the stereo cue bus is fed from either the mic/line or DAW input (depending on the status of the **DAW** input flip). Input level and pan are controlled from the cue "send" controls, and the cue bus is added to the main mix in the master section.

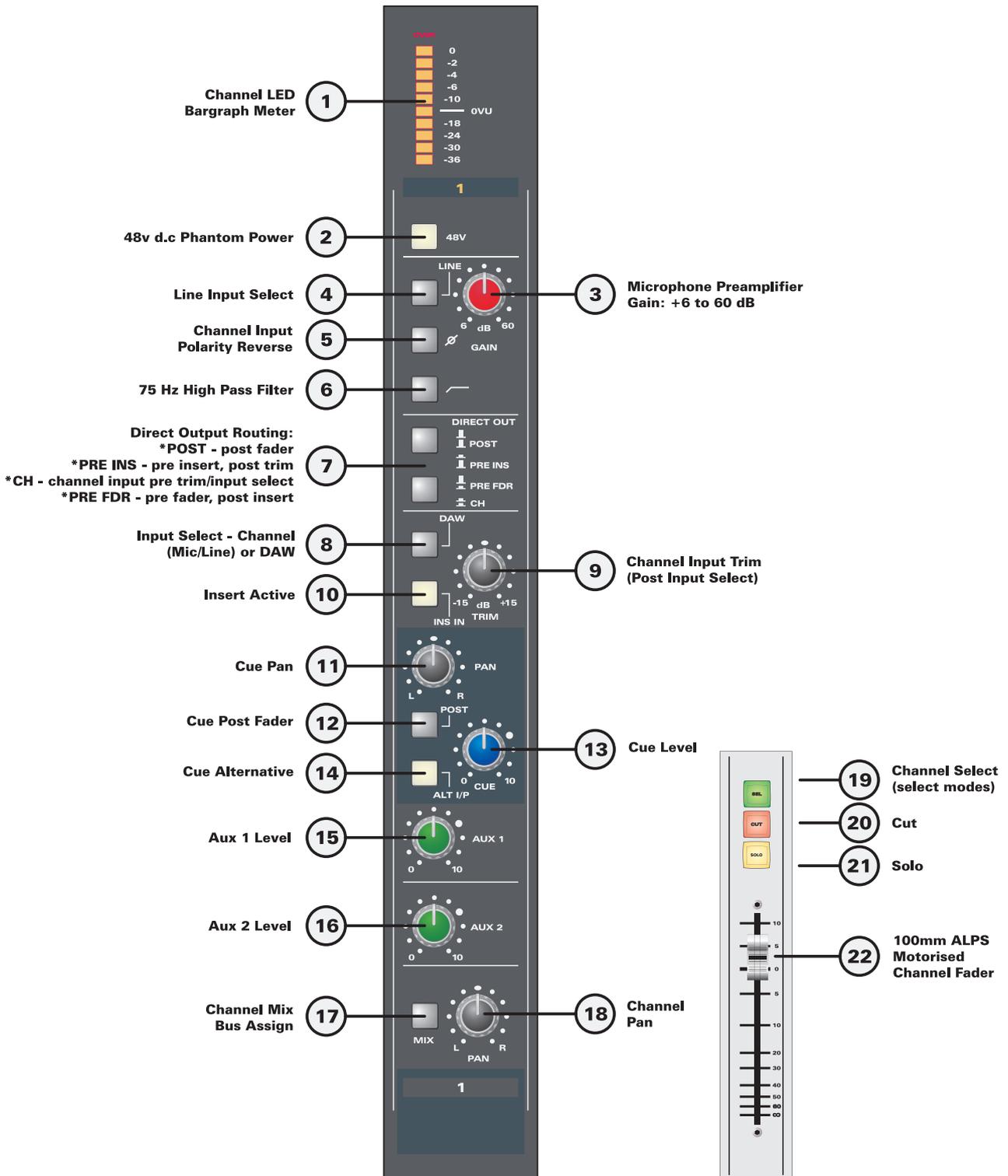
In addition to these 16 individual channel inputs, a further 16 inputs are available from the master section:

- Summing Inputs 1-8 - treated as four stereo input pairs, or 8 mono inputs, with a single level and balance/pan control.
- Stereo FX Returns 1 & 2 - with level and balance/pan controls for each return.
- Stereo DAW Mix input - with routing control. (To control the level or pan this input, adjust the output from the DAW).
- Stereo Mix Insert Return - working in "parallel sum" mode.

The Channel Strip

This section deals with the analogue channel strip operation. For details on using the faders to control parameters within the DAW software, please see the "Automation & Control Surface Manual".

*** TIP!** You may find it useful to print this page so that you can refer to the control numbering while reading about each channel strip section.



Channel LED Bargraph Meter 1

The meter provides a 10-segment LED bargraph meter with peak detection, plus a separate overload indication LED.

The 10 LEDs are utilised as follows:

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

0 VU is the centre of the meter scale and calibrated to a nominal operating level of +4dBu.

Relative to a typical EBU AD/DA converter calibration of +18dBu = 0dBFS, this means that +4dBu = 0 VU = -14dBFS:

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 —	OVU
— -18 dBFS —	0 dBu —	
— -24 dBFS —	-6 dBu —	
— -30 dBFS —	-12 dBu —	
— -36 dBFS —	-18 dBu —	

As a result, a nominal operating level of 0 VU provides 14dB of headroom in your digital recording platform. This is ideal for most music sources which have crest factors of around 12 to 20dB.

[The crest factor is the difference between average (RMS) and peak level.]

CH METER SELECT

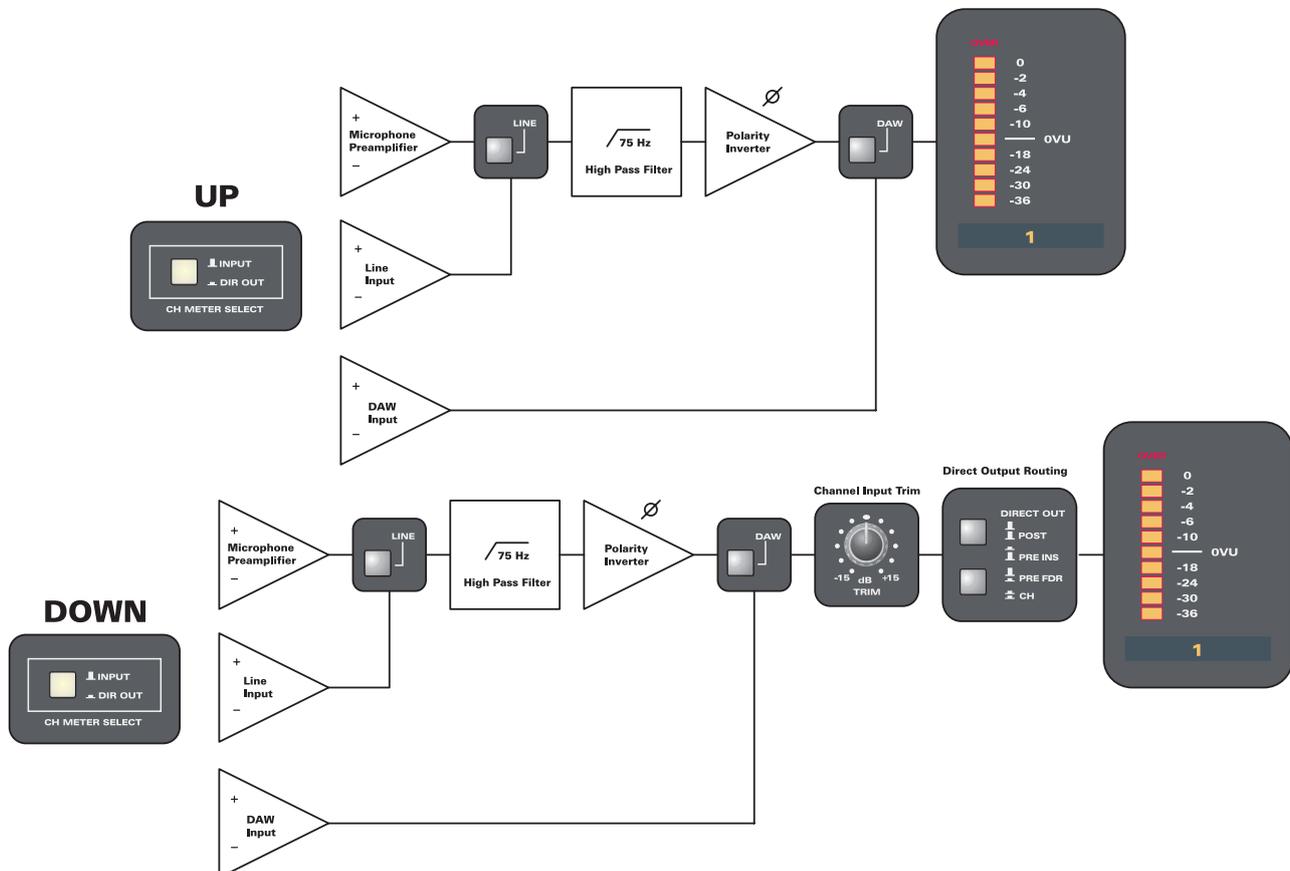


This switch selects the meter source globally across all 8 channels:

- **INPUT** - in the UP position, the channel input feeds the meter. Signal is taken before the channel trim control (9) but after the mic/line and DAW input selections (4 & 8).
- **DIR OUT** - in the DOWN position, the meter takes its feed from the direct output. Thus, its exact location is determined by the direct output routing (7) – see page 16.

*** NOTE** This selection is overridden when displaying DAW channel metering. (See the **DAW Meters** function in the “Automation & Control Surface Manual”).

Bargraph Meter Signal Flow Options:



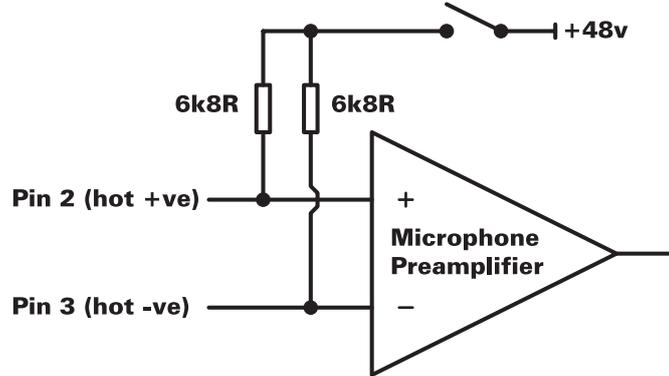
48v Phantom Power (2)



Engaging this switch provides 48 volt D.C phantom power to the microphone input.

Individual phantom power switches on each channel provide control over which microphones receive power.

Phantom power wiring



Microphone Preamplifier Gain (3)



Mic gain may be adjusted from 6 to 60dB.

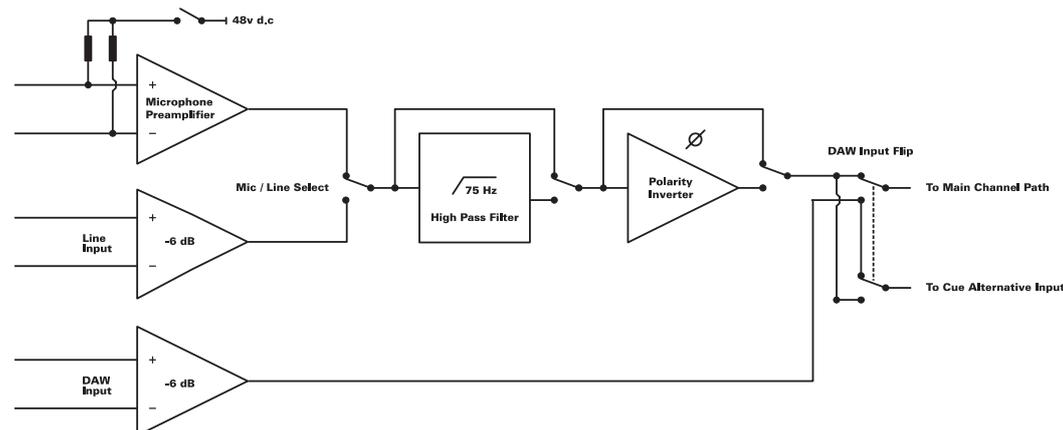
The gain on offer is clean amplification and features a low-noise, differential discrete transistor input stage.

Line Input Select (4)



Engaging this switch overrides the microphone preamplifier, as the channel source, and replaces it with the line input.

The mic/line path may also be replaced by the DAW input (8).



Polarity (Phase) Invert (5)



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

This function is only available for the mic or line input, and not for the DAW input - to invert the phase of a DAW input, use the phase invert within your workstation software.)

High Pass Filter (6)

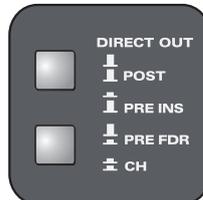


A 75Hz high pass filter (HPF), with 12dB per octave roll-off, can be engaged on the mic/line input path – useful for removing unwanted low frequency content.

Many listeners use playback systems that can barely reproduce 60-70Hz. Therefore, a well placed HPF on certain sources can clean up unwanted mix elements, such as microphone stand rumble and traffic noise.

This function is only available for the mic or line input, and not for the DAW input - to filter a DAW input, use the equalisation within your workstation software.

Direct Output Routing (7)



The direct out is normally used as the record send to the DAW. The feed is taken from one of four different points in the channel path, as illustrated on the next page:

- **POST** - post-fader
- **PRE INS** - pre-insert
- **PRE FDR** - pre-fader
- **CH** - from the channel input (mic/line path)

Direct Out: POST



This mode requires both switches to be in the UP position. The direct output is taken **POST FADER** and **POST MUTE**. This mode is ideal for:

- Using the faders to ride record levels (especially if microphone levels are hot).
- “Printing back” stem mixes to the recorder with fader automation.

Direct Out: PRE INSERT



This mode requires the first switch to be in the DOWN position. The direct output is taken **PRE INSERT** and **PRE FADER**. This mode is ideal for:

- Taking a “clean feed” to the recorder while monitoring post-insert, fader and pan. This mode is often used when you need a processed headphone mix (via the cue or aux), but wish to record a “clean” unprocessed signal.

The feed is taken after the **DAW** input flip (8) and channel input trim (9). This allows you to use the channel input trim as an additional gain control – arguably, a cleaner way of adjusting gain during a take (as opposed to varying the microphone preamplifier gain).

Direct Out: PRE FADER



This mode requires the second switch to be in the DOWN position. The direct output is taken **POST INSERT** but **PRE FADER**. This mode is ideal for:

- Using the faders to set a monitor balance without affecting the record level, while recording a processed signal. For example, to apply outboard dynamics to the record signal via the channel insert (10).

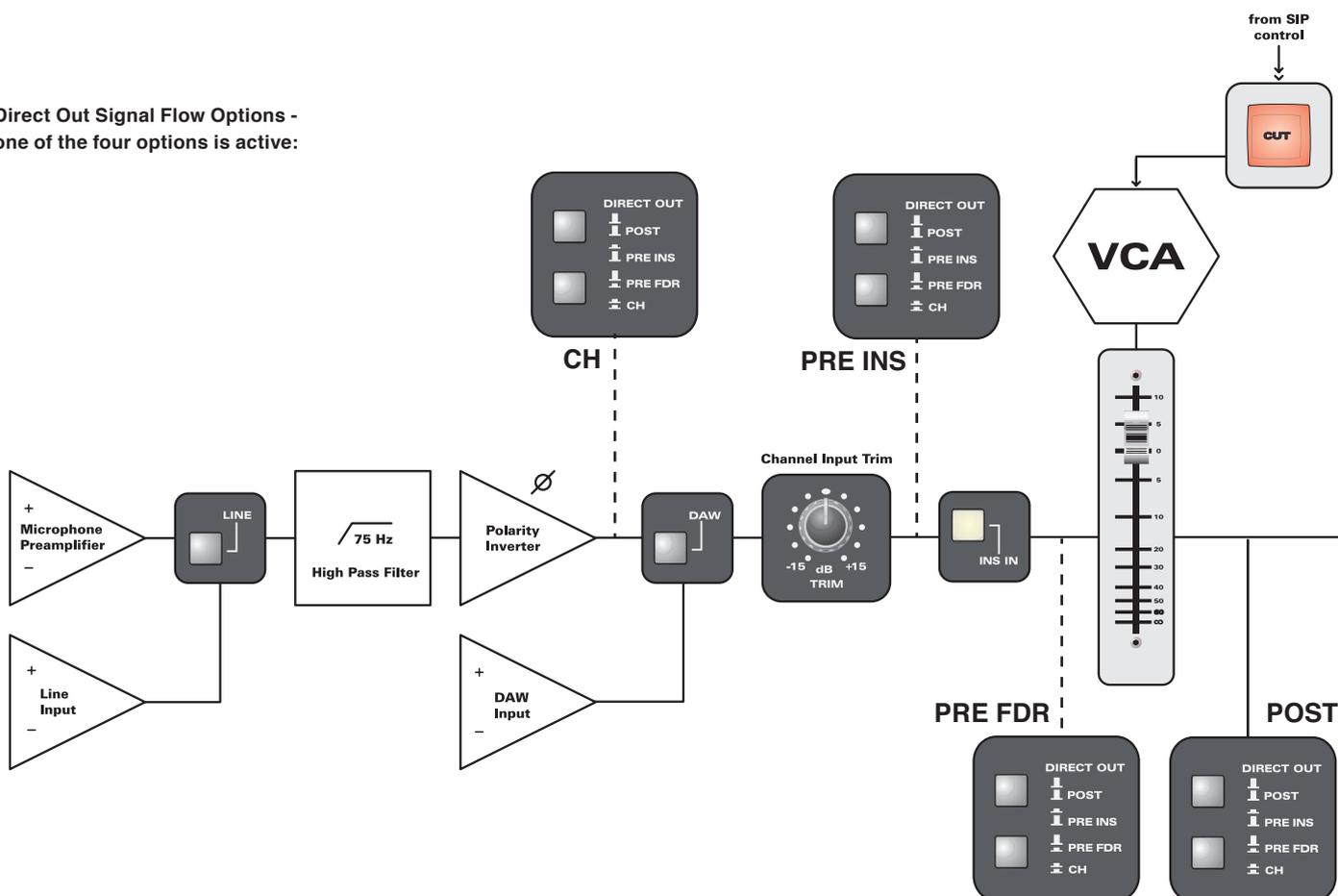
Direct Out: CHANNEL



This mode requires both switches to be in the DOWN position. The direct output is taken directly from the mic/line input, after the HPF and polarity inverter stage. This mode is ideal for:

- Taking a “clean feed” to the recorder, as an alternative to **PRE INSERT**. The feed is taken from the input stage, with no channel input trim (9), providing the cleanest signal path for recording.
- Recording with an “in-line” console architecture - where the main channel path can be used to monitor the send or return without affecting the recorded signal. This is explained in more detail on page 19.

Direct Out Signal Flow Options - one of the four options is active:



DAW - Mic/Line Input Flip 8



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

When the switch is in the UP position, the mic/line input feeds the main channel path, while the DAW input feeds the alternative cue input.

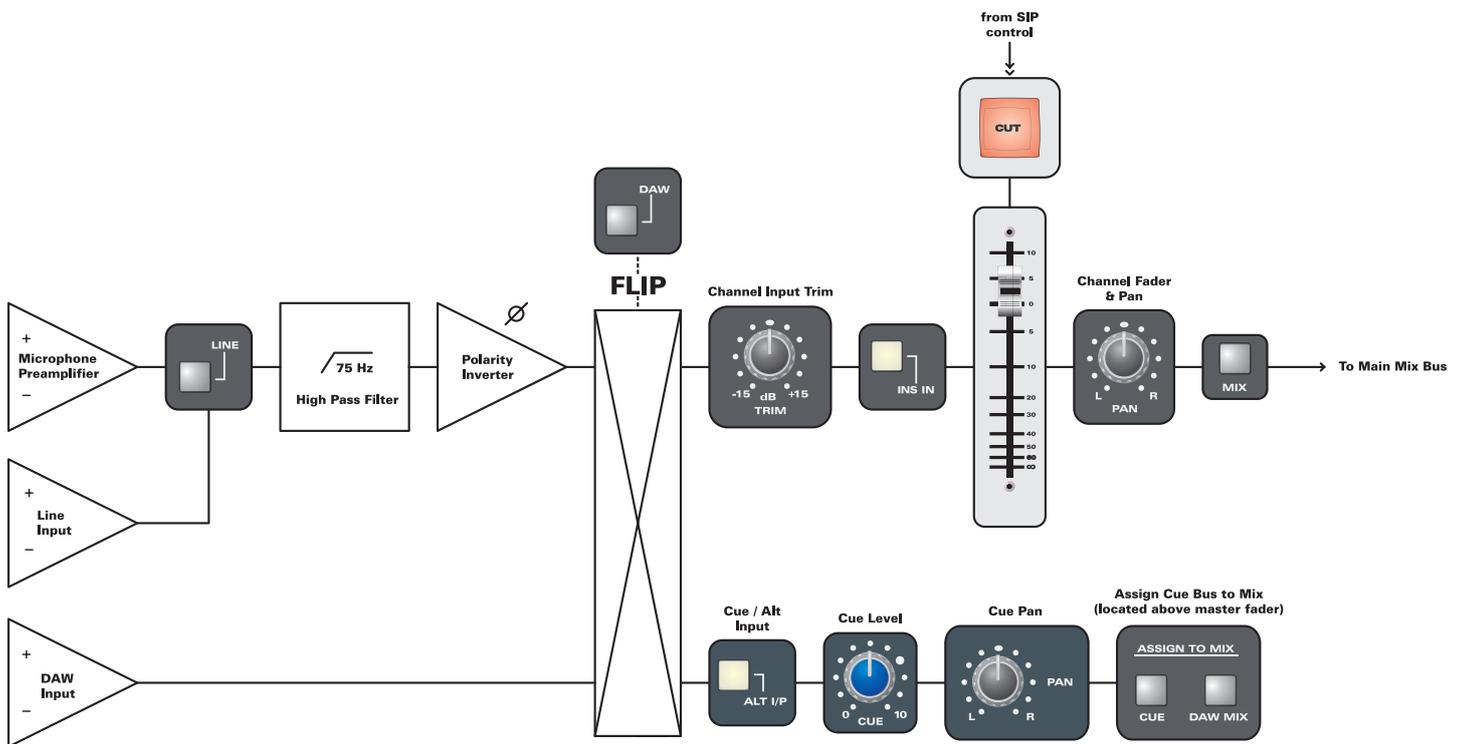
When the switch is in the DOWN position, the sources flip over. Therefore the DAW input feeds the main channel path, while the mic/line input feeds the alternative cue input.

In both cases, for the alternative cue input to be active, you must engage the **ALT I/P** switch (14).

On its own, the **DAW** switch flips the main channel path from mic/line input to DAW input for fast playback or mixdown of your recordings.

Combine the **DAW** switch (8) with the **ALT I/P** (14) to utilise both the main channel and cue paths:

- **ALT I/P DOWN + DAW UP** - the mic/line input feeds the main channel while the DAW return feeds the cue section. By adding the stereo cue bus to the main mix in the master section (**ASSIGN TO MIX - CUE**), the DAW return can be used as an extra mix input. The DAW input level and pan are controlled from the cue “send” (11 & 13). See page 21 for details.
- **ALT I/P DOWN + DAW DOWN** - the mic/line input feeds the cue bus while the DAW return feeds the main channel. This can be used in the same manner as above (with the inputs reversed). Or, it can be used to generate a zero latency cue send for your artist’s headphones. See page 19 for details.



Recording (with an In-Line Architecture)

The direct output **CHANNEL** routing mode (7) can be combined with the **DAW** input flip (8) to create a “pseudo in-line” architecture.

Using this workflow, you can simultaneously:

- Record mic preamp or line inputs direct to disk (clean).
- Generate a cue mix directly from the mic/line input for your artist’s headphones (with zero latency).
- Monitor the DAW returns (post record) through the main channel path so that you can hear any colouration provided by your workstation.

Recorder Send

Select the **CHANNEL** direct output routing mode (7) so that either the microphone preamplifier or line input is fed straight to the direct output and recording device as shown below.

Use the **LINE** input switch (4) to choose mic or line:

- For mic inputs, adjust the record level using the mic preamp gain control (3).
- For line inputs, adjust the record level on the outboard device. (If using external mic preamps, connect them here and adjust their mic gain.)

Recorder Return

The return from the recorder feeds back into the console via the DAW INPUTS 25-pin D-Sub connector on the rear panel.

Monitoring the Send and Return

Assign the main channel path onto the main mix bus, using the **MIX** bus assign switch (17).

Then use the **DAW** input flip (8) to switch the main channel between:

- Mic/line input (recorder send) - **DAW** in UP position
- DAW input (recorder return) - **DAW** in DOWN position

This makes simple record and playback monitoring very quick and efficient, and allows you to start setting fader levels, cuts and panning without affecting the record path.

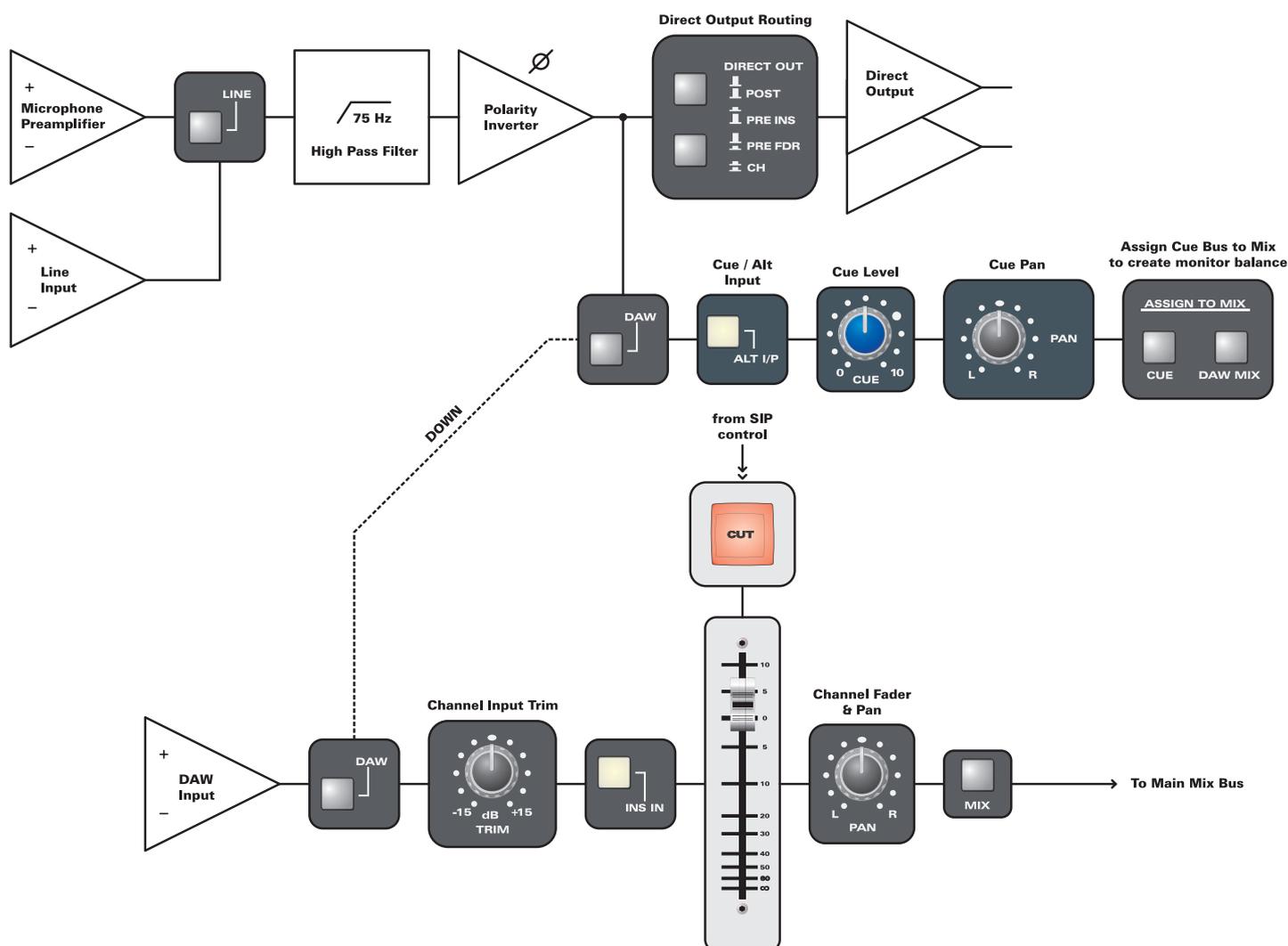
Generating a Zero Latency Cue Send

If the cue send for an artist’s headphone mix is taken directly from the mic/line input, then any delays introduced by your recorder or AD/DA converters can be avoided (“zero latency”). This is of huge benefit when dealing with digital systems.

On the artist’s input channel, select both the **DAW** input flip (8) and cue **ALT I/P** (14). This feeds the DAW return to the main channel and the mic/line input to the cue send as shown below.

Use the cue level and pan controls (11 & 13) to control the mix. You can add playback channels by turning off the cue **ALT I/P** (14) and selecting either a pre or post-fader send (12).

You can monitor the cue mix by sending it to your headphones (**CUE/FB TO PHONES**). Or, add it back to the main mix from the master section (**ASSIGN TO MIX - CUE**).



Channel Input TRIM 9



This potentiometer provides a trim range of +/-15dB.

It can be used to boost the recording level providing that your direct output (7) is taken post channel trim (**PRE INS**, **PRE FDR** or **POST**). This is useful for low output microphones such as ribbon transducer designs.

It is also very useful for setting correct gain staging at the input of the console.

Insert Active 10



In the DOWN position, this switch engages the channel insert send and return loop - used to insert outboard processors into the main channel path.

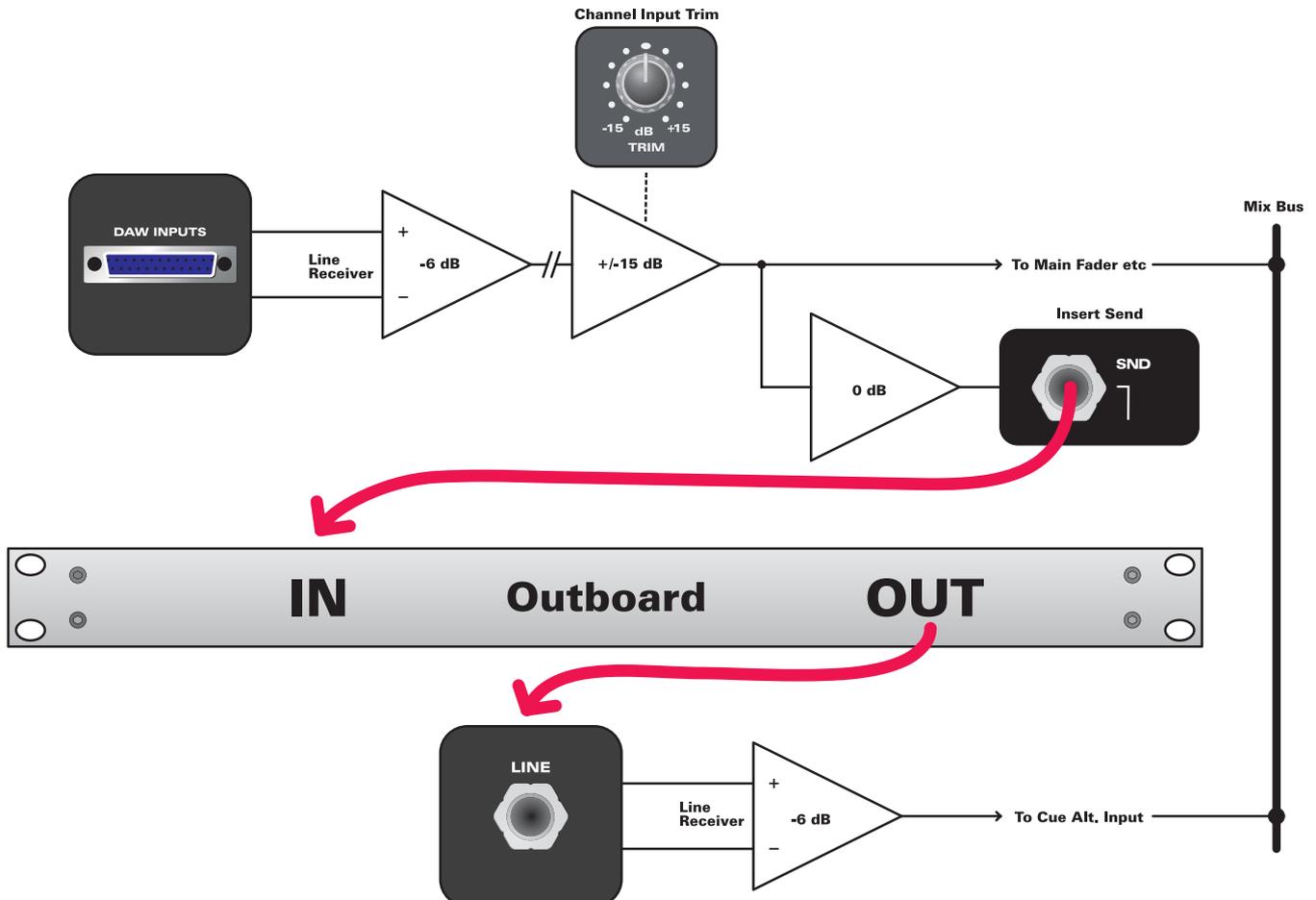
The insert send and return are available as balanced 1/4" TRS jack connectors for each channel. If nothing is connected, then the TRS jacks are normalled. This prevents signal loss if **INS IN** is accidentally turned on.

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 split or “mult” signals from the insert send to set up parallel compression or feed external destinations (such as a back-up recording system).

*** TIP!** Try taking a feed from the insert send and then returning a processed version (distorted or heavily compressed) into another input. Mix together the “clean” and “processed” versions to create an effect.

The diagram below illustrates this example using the main channel and alternative cue inputs of the same channel. The main channel is fed from the DAW input - depress the **DAW** input flip (8). This leaves the line input free to feed the alternative cue input - depress **ALT I/P** (14) - and return the processed signal, via the cue bus, onto the main mix.



Cue Send

Each channel features a stereo cue send with independent source selection, level and pan control.

When the **ALT I/P** switch (14) is not engaged (UP position), the cue send is taken from the main channel path - either pre or post-fader according to the cue **POST** switch (12). Remember that the main channel may be fed from mic, line or DAW inputs according to the status of the **LINE** input select (4) and **DAW** input flip (8).

If the **ALT I/P** switch (14) is engaged (DOWN position), then the stereo cue bus is fed from the alternative cue input. This can be used to return different inputs to the main channel and alternative cue paths, see page [22](#).

Cue Pan (11)



This control pans the source onto the stereo cue bus.

The control provides a smooth taper with an approximate pan law of -4dB. This provides a generally-accepted compromise between “constant power” and “constant voltage” pan law designs.

The control has a centre detent for easy mono positioning.

If the alternative cue input is used to add extra inputs to the mix, then this control acts as the “channel pan” for the input.

Cue Post Fader Assign (12)



In the DOWN position, the cue bus is fed with a post-fader signal from the main channel path - ideal if the cue bus is used as an effects send, as you will want changes in your fader level to affect the send.

In the UP position, the cue bus is fed pre-fader - great for generating a foldback send for artists headphones. In this case, you do not want your control room fader positions to affect what the artists hear.

Note that this switch has NO effect if the cue **ALT I/P** (14) is engaged, as the alternative cue input overrides the cue send.

Cue Level (13)



This control sets the level of the source feeding the cue bus.

If the alternative cue input is used to add extra inputs to the main mix, then this control acts as the “channel fader” for the input.

Cue Send Applications

The cue section has three main applications:

Foldback - when recording, the cue bus usually feeds the artists headphones.

A stereo cue mix is created from the 8 main channel paths, with sends usually taken pre-fader (or from the alternative cue input) so that your control room balance does not affect what the artists hear.

What feeds the cue output is then determined by the **Cue/FB** controls in the master section. This can combine the cue bus with other inputs such as the stereo DAW input, auxes, etc. You can also talk to the cue output using the **TALKBACK** controls. See page [35](#) for details.

Effects Send - the stereo cue bus can be used as an extra effects send, to complement aux 1 & aux 2. This is ideal if you are feeding a stereo in/stereo out FX device.

Extra Mix Inputs - by activating the alternative cue inputs (14), and combining the cue bus with the main mix, you can mix an extra 8 inputs with individual level and pan control. This turns the **Control 2802** into a 16-input mixer: 8 main channel plus 8 alternative cue input paths.

Monitoring the Cue Bus

You can monitor the cue bus on your control room headphones by selecting **CUE BUS** as the **CUE/FB** source and routing this to your headphones (**CUE/FB TO PHONES**). See page [35](#) for details.

Building Headphone Mixes in your DAW

If you use your DAW software to build headphone mixes, then you will be able to apply effects, and save and recall settings with your session. Although this may introduce latency, newer computers and low latency DSP devices make this feasible for certain circumstances.

The stereo DAW FB input on the rear connector panel is intended for returning headphone mixes from the DAW. This input can be easily monitored, or added to the cue output using the **CUE/FB** panel. See page [35](#) for details.

*** TIP!** Use the HUI™ Control Surface Functionality of the **Control 2802** to flip the Fader the Fader Strips to the DAW layer to setup headphone mixes in superfast time.

Cue Alternative Input Select (14)



This switch activates the alternative cue input. When active, the cue source is determined by the **DAW** input flip (8) as shown in the diagram below.

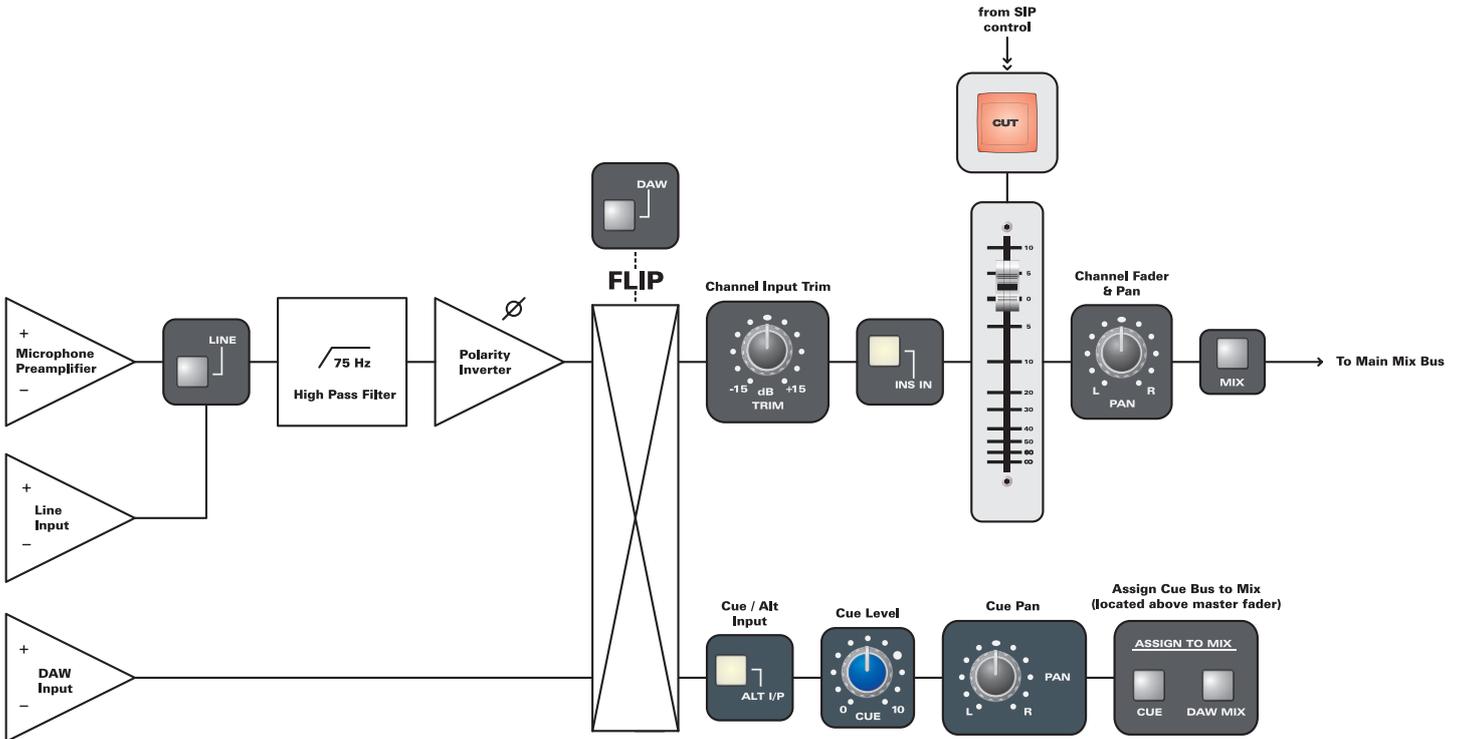
- **ALT I/P DOWN + DAW UP** - the mic/line input feeds the main channel while the DAW return feeds the cue section. By adding the stereo cue bus to the main mix in the master section (**ASSIGN TO MIX - CUE**), the DAW return can be used as an extra mix input. The DAW input level and pan are controlled from the cue “send” (11 & 13).
- **ALT I/P DOWN + DAW DOWN** - the mic/line input feeds the cue bus while the DAW return feeds the main channel. This can be used in the same manner as above (with the inputs reversed). Or, it can be used to generate a zero latency cue send for your artist’s headphones. See page 19 for details.

The **ALT I/P** can be selected on a channel-by-channel basis, producing some very flexible routing combinations.

ASSIGN TO MIX



The **ASSIGN TO MIX - CUE** switch, in the master section, routes the cue bus onto the main mix. Note that ALL cue sends feed the main mix when this switch is active.



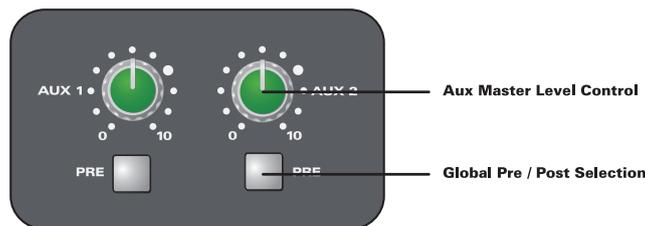
Aux Sends

Each channel features two mono auxiliary sends - ideal for sending to outboard effects devices, such as reverb, or for creating additional cue mixes.

*** TIP!** Use either the stereo FX Returns or the Summing Input to return effects devices to the mix.

The two mono auxes are ideal for mono in/stereo out devices. If you require a stereo send, either use the stereo cue bus, or feed aux 1 to the left and aux 2 to the right input to the device.

By default, the aux sends are taken from the main channel path, post-fader. Use the **AUX 1 PRE** (or **AUX 2 PRE**) switch, in the master section, to switch to pre-fader - note that sends for ALL 8 channels are affected:



Aux Level Controls 15 16



This control sets the level feeding the AUX 1 (or AUX 2) bus. Send levels can be adjusted individually, channel-by-channel.

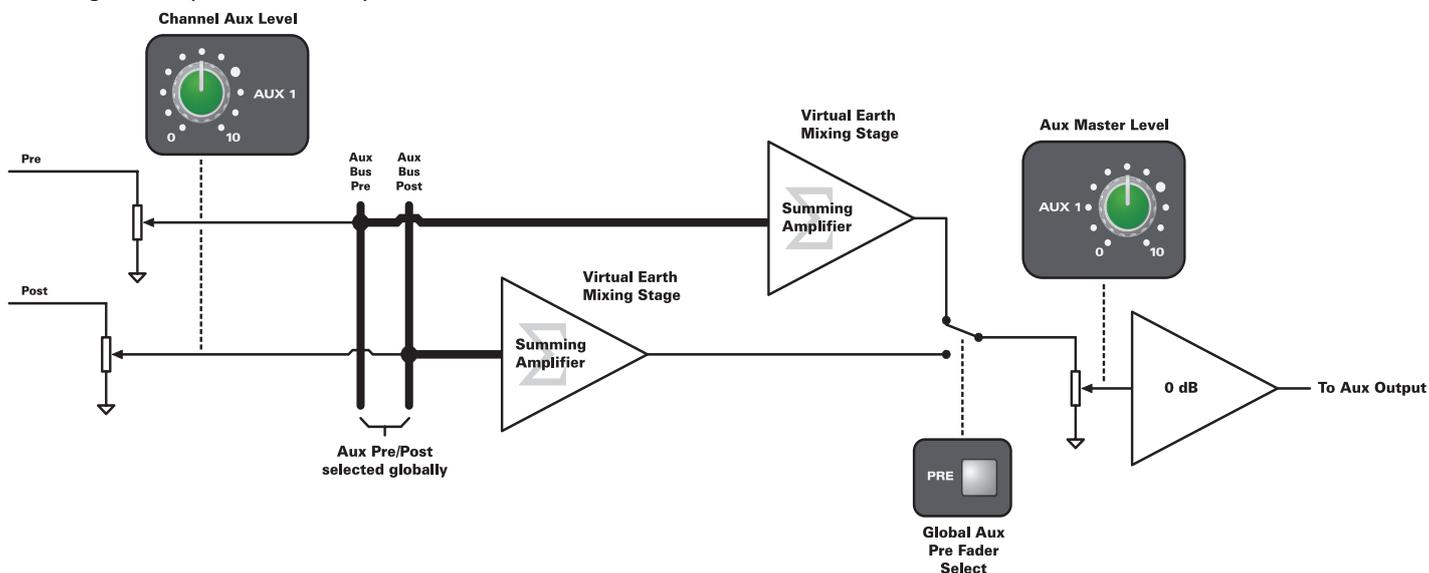


Please note that if you change from pre to post-fader, or vice versa, the same level is applied.

Monitoring the Aux Busses

Each of the aux busses may be selected as a control room or **Cue/FB** monitor source. This makes it easy to listen while building an aux mix, and to use auxes to feed artists headphones. See page [34](#) for aux masters, page [35](#) for the cue output, and page [37](#) for monitoring.

Aux 1 Signal Flow (Aux 2 is identical):



MIX Bus Assign 17



This switch routes the main channel path (post fader and pan) onto the main stereo mix bus.

Channel Pan 18



This control pans the main channel path onto the stereo main mix bus.

The control provides a smooth taper with an approximate pan law of -4dB. This provides a generally-accepted compromise between “constant power” and “constant voltage” pan law designs.

The control has a centre detent for easy mono positioning.

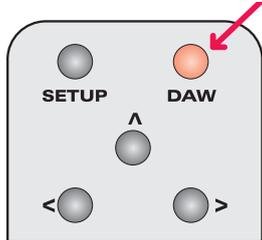
Dual-Layer Fader Strips

Layer Switching

The 8 channel faders, **CUTs** & **SOLOs** can control either the:

- Analogue level, of the main channel paths.
- 8 DAW fader tracks, within the DAW software.

The layer switching is handled from the Control Surface Panel:



Select **DAW** (it turns red) to flip all 8 Fader Strips to the DAW layer.

Deselect **DAW** (not illuminated) to control the analogue layer.

When you change layer, all 8 channel faders (plus **SEL**, **CUT** & **SOLO** switches) update to their corresponding positions.

*** NOTE** Both layers are ALWAYS active, even if the DAW is not connected. This means that if you set an analogue fader level, and then switch to **DAW**, you will still hear the analogue channels (providing audio is present).

Therefore, take care NOT to change layer by accident - if nothing is happening to your analogue channel level when you move a fader, check that you are controlling the correct layer!

Channel Fader (22)

The channel fader is a high-quality ALPS 100mm linear fader that can control either the analogue main channel level or a DAW fader. The fader is motorised and can automate both the analogue level and DAW fader layers.

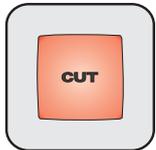
Fader Scale & Resolution

Please bear in mind that the scale beside the channel fader is for analogue level indication only.

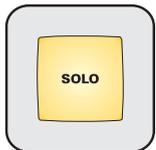
When controlling the DAW layer, the fader scale is dictated by your software. For example, in Logic, faders can operate to +6dBFS above unity, yet in Pro Tools it is +12dBFS. This means that the resultant fader scale, and the resolution at the top of the fader differs depending on the DAW.

This is a limitation of the HUI™ protocol.

Channel SOLO & CUT (20) (21)



The **CUT** switch mutes either the analogue main channel or a DAW fader. It can be automated on both analogue and DAW layers. The **CUT** switch is also activated when using solo-in-place (see below).



The **SOLO** switch places the channel into one of three solo modes: AFL (after-fade listen), PFL (pre-fade listen) or SIP (solo-in-place). The mode is selected globally from the master section, see page 40

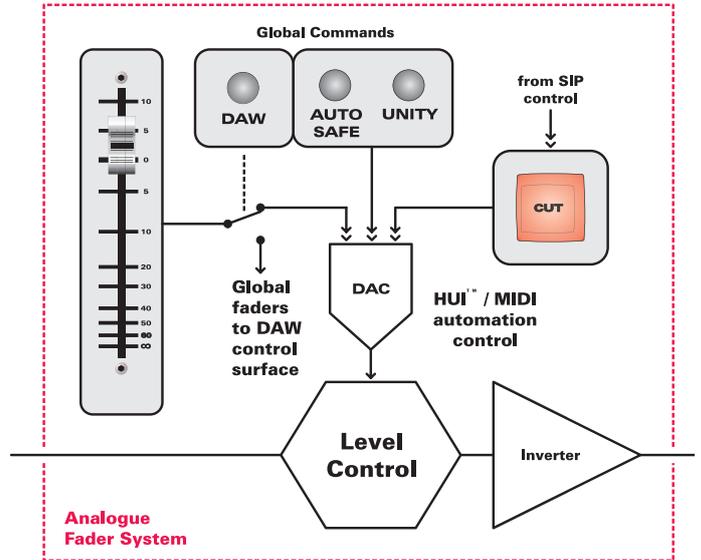
You can select multiple **SOLOs** in all modes.

When using solo-in-place, all channels not in solo are **CUT**. You can isolate a channel, so that it does not **CUT** in response to other **SOLOs**, using the **SIP SAFE** function, see page 25.

When using either **AFL** or **PFL**, some of the main mix can be “blended in” behind the solo’d channels using the **SOLO IN FRONT** master control. See page 40 for details.

Analogue Level Control

The diagram below illustrates what occurs when the fader strips are switched to the analogue channel layer:



The fader (and **CUT**) are not in the audio path but act as control signals, sending D.C. control voltage changes to the analogue “Level Control”. This applies attenuation, mute and up to +10dB of gain to the analogue main channel path.

In addition to responding to physical fader (and **CUT**) moves, the analogue level control is adjusted by:

- HUI™ / MIDI automation - this allows you to automate the faders and **CUTs** of the 8 analogue main channel paths.
- Global commands - for example, press **UNITY** (on the Control Surface Panel) and then press a fader strip **SEL** switch - the analogue level (and fader) are set to 0dB.

The components used in **Control 2802** ensure a clean, accurate and repeatable automation platform - control resolution is very fine, and when channels are **CUT**, click-free silent muting is applied.

For more on analogue fader automation, please see the “Automation & Control Surface Manual”. For more on **UNITY**, see page 25.

DAW Fader Control

When fader strips are switched to the DAW layer, their D.C. control voltages are processed by on-board microprocessors to output HUI™ control surface functionality and automation.

The microprocessors also respond to incoming mute, solo-in-place commands, fader positions and MIDI automation.

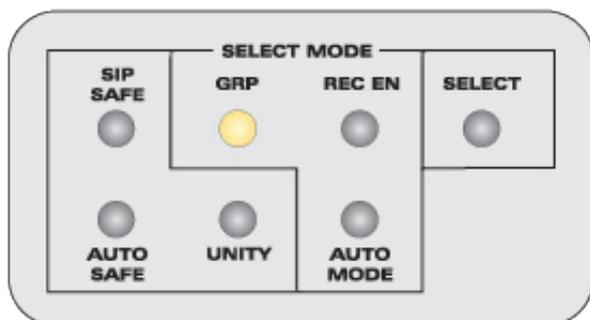
This enables you to control DAW parameters, and write and replay automation.

The setup and operation of the DAW layer is covered fully in the “Automation & Control Surface Manual” which can be downloaded from our website at www.focusrite.com.

SElect Switch & SELECT MODES 19



The **SEL** switch selects the fader strip. It is used in conjunction with the SELECT MODEs on the Control Surface Panel to perform a number of functions.



For all analogue layer functions, the operation is as follows:

Press the SELECT MODE - e.g. **UNITY** - and then press the **SEL** switches (19) on each and every channel you wish to affect - in our example, press **SEL** on fader strips 1 and 2 to position faders 1 and 2 at unity gain.

* NOTE

- The status of channel selections is stored for all SELECT MODEs. This means that when you re-select a function, the **SEL** switches are how you last left them.
- This applies even after a power down. Therefore, you should clear or reset channel **SEL** switches if the next session requires a different set-up.
- One of the SELECT MODE switches is always active, so it is a good idea to return to a benign mode, such as channel **SELECT**, once you have finished making changes.

Analogue Layer SELECT MODEs

The following SELECT MODEs are available when controlling the analogue layer:



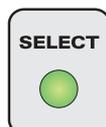
SIP Safe (Solo-In-Place Safe) – used in the analogue layer to isolate channels that you do not want **CUT** when a **SOLO** on another channel is active - very useful for reverb returns.



Automation Safe – used in the analogue layer to isolate channels from automation. This prevents HUI™ controller / MIDI automation from adjusting the analogue channel level. Please see the “Automation & Control Surface Manual”.



Unity – used in the analogue layer to provide a quick way to position the channel fader at unity gain (0dB). The channel remains locked at 0dB until you deselect it. This function is ideal for returning stem mixes, or any application where you want the faders at 0dB.



Channel Select – this mode is intended for use in the DAW layer to select the channel. The switch can be selected in the analogue layer, but it has no function.

DAW Layer SELECT MODEs

The following modes are available only when controlling the DAW layer:

- **GRP** – used to group tracks.
- **REC EN** – used to record enable tracks.
- **AUTO MODE** – used to select host-specific automation modes such as read, touch and latch.

Please see the “Automation & Control Surface Manual” for full details on these modes.

Main Mix Bus

The main stereo mix bus, and all signal combining stages on **Control 2802**, are optimised for clean summing with high headroom.

Signal Flow - up to 32 Mix Inputs!

The mix bus summing and its output signal flow are illustrated in the diagram at the bottom of this page.

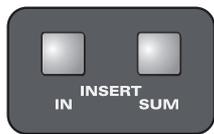
Signal leaves the console via the balanced male XLR outputs (MIX O/P L/R) on the rear panel.

The following inputs may be assigned to the main mix:

- Main channel paths 1 to 8 (via channel strip **MIX** switches).
- Stereo cue bus (via **ASSIGN TO MIX - CUE**). Use this option to add the 8 alternative cue input paths to the mix.
- Stereo DAW MIX input (via **ASSIGN TO MIX - DAW**). This input is usually fed from the main mix of your DAW, but could come from any stereo input. There is no input level control so adjust the output from the connecting device.
- 8-channel SUMMING INPUT (via **SUMMING INPUT - MIX**). This can return four stereo, or 8 mono inputs. There is a single level and balance/pan control for all inputs.
- 2 x stereo FX Returns (via **FX RETURN 1 & 2 - MIX**). These can return two stereo inputs. There are level and balance/pan controls for each stereo input.
- Stereo Mix Insert Return (switched to “parallel sum” mode as described opposite). There is no input level control so adjust the output from the connecting device.

Use all these options to mix up to 32 inputs simultaneously!

Mix INSERT



The mix output features a switchable insert point before the master fader. The insert send is always active. Thus it is possible to take a pre-master fader feed from the insert send.

Engage the **IN** switch - DOWN - to activate the insert return. You can now insert outboard devices to apply processing.

The insert send and return are available as balanced 1/4" TRS jack connectors on the rear panel. If nothing is connected, then the TRS jacks are normalled. This prevents signal loss if **INSERT IN** is accidentally turned on.

Mix INSERT - Parallel Sum Mode

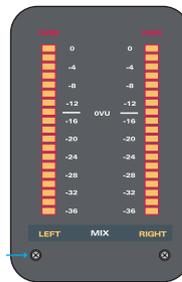
By default, the insert return interrupts the main mix path. However, if you enable the parallel sum mode, then the insert return and main mix are combined.

Engage both the **IN** and **SUM** switches - DOWN position - the insert return is now summed with the original mix.

This can be used to subtly enhance or colour the mix via an outboard EQ or compressor in the insert loop - a technique known as parallel processing.

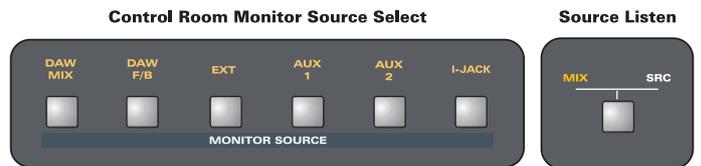
Alternatively, the mode can return an extra stereo input to the mix - connect an input to the mix insert return, and it will be summed with the main mix bus. (Control the input level by varying the output from the connecting device).

MIX Output Metering



The LEFT and RIGHT MIX meters are actually fed from the monitor output (after the MONITOR SOURCE selector but before **VOLUME**, etc). This is often known as “follow monitor” metering, which allows you to meter any source, not just the main mix.

As a result, the meters are switched by the MONITOR SOURCE selector. To meter the main mix, make sure that the MIX/SRC switch is assigned to the **MIX** position:



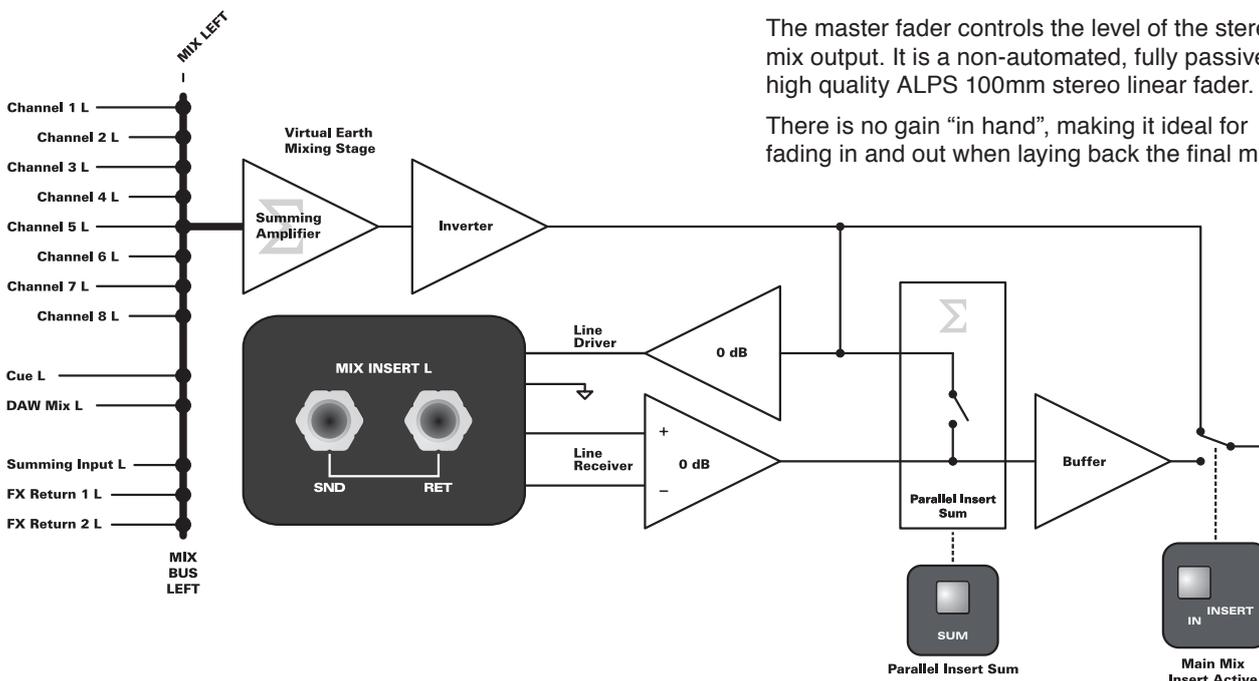
To gauge your headroom, these are peak bargraph meters with 20 large LED segments. They are calibrated to 0VU = +4dBu = -14dBFS, with 36dB of range. The recessed meter trims can be used to calibrate the 0VU point.

Master Fader

The master fader controls the level of the stereo mix output. It is a non-automated, fully passive, high quality ALPS 100mm stereo linear fader.

There is no gain “in hand”, making it ideal for fading in and out when laying back the final mix.

MASTER FADER



Bus Compressor

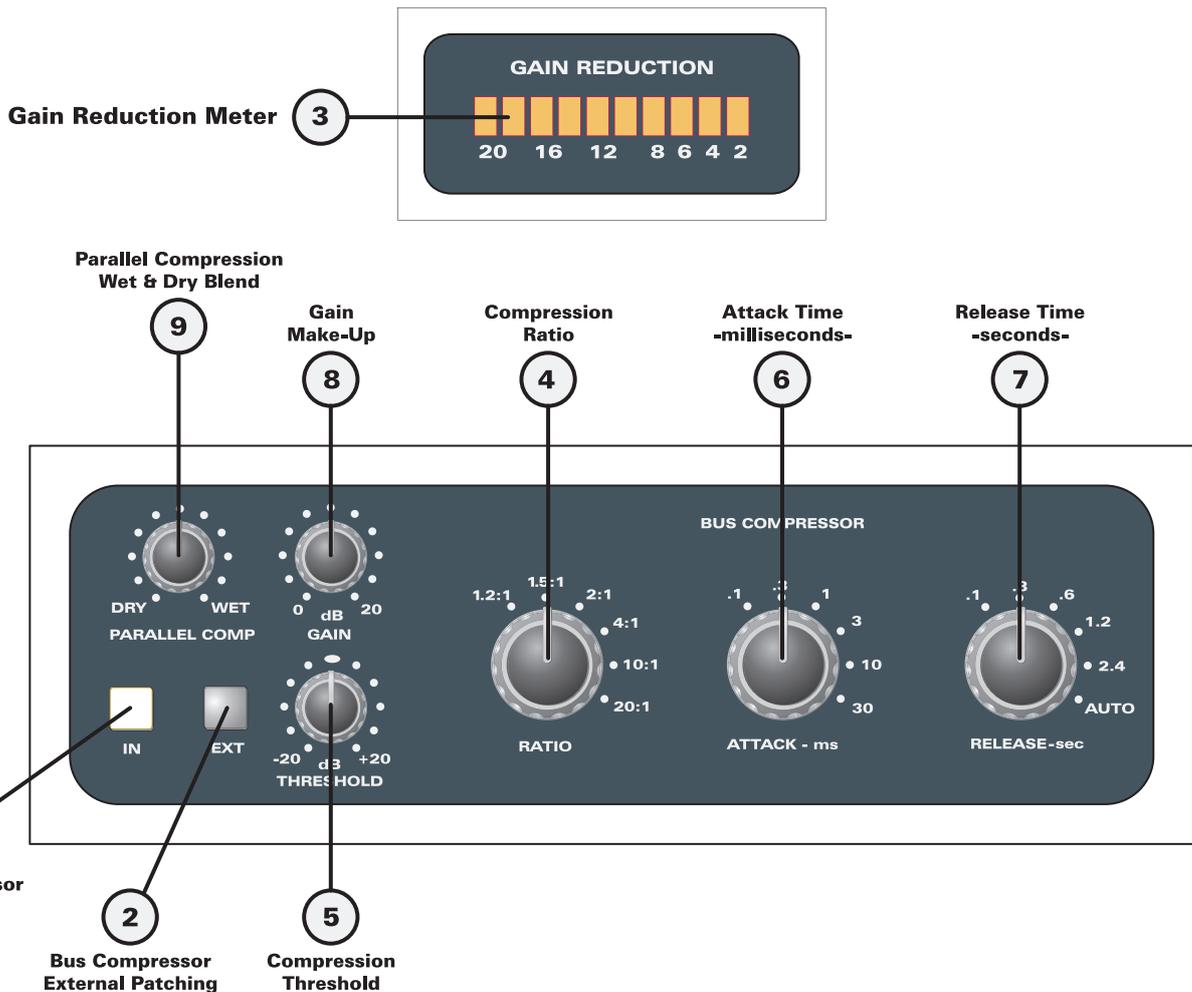
The master section of **Control 2802** features an integrated stereo mix bus compressor.

The compressor uses a soft-knee VCA design that is optimised for mix bus use.

By default, the compressor sits within the main mix output path, after the mix bus insert and before the master fader. However, this can be rearranged, or the compressor can be removed from the main mix and patched elsewhere using its own external IO.



*** TIP!** You may find it useful to print this page so that you can refer to the control numbering while reading this section.



*** TIP!** **Tips and Suggestions**

The bus compressor is a VCA (Voltage Controlled Amplifier) gain cell design. It is a flexible tool suitable for a variety of sources - for example, to gently “glue” together your main mix, or add impact and energy to drums.

Perhaps you dislike mix bus compression or have your own favourite outboard mix compressor...

... then use the **EXT** switch (2) and COMPRESSOR IN/OUT connections (on the rear panel) to patch the compressor to another path - for example, to the insert send and return of a pair of **Control 2802** channels. You may then apply the bus compressor to any signal feeding the channels - individual instruments, subgroups, or final mixes.

Use the **DRY-WET** control (9) to “crossfade” between the uncompressed and compressed signal - a technique known as parallel compression. Try this on your drum bus. It is fantastic at preserving transient information, while adding body and punch.

When using the **DRY-WET** control (9), there is a slight amount of gain in the sweep of the pot. Therefore, you may not need as much **GAIN** make-up (8). This is a great way to increase level post compression, and achieve a nice sense of punch!

Try starting the **DRY-WET** control (9) somewhere between 1 and 4 o’clock, and adjust the **GAIN** make-up (8) to taste.

Please see “Bus Compressor Settings” on page 31 for some example settings.

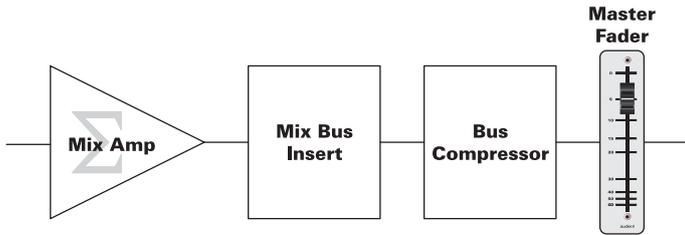
Compressor IN ①



This illuminated push switch engages the compressor and also the GAIN REDUCTION meter (3).

Signal Flow

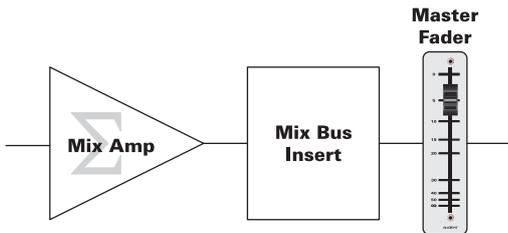
By default, the compressor sits within the main mix output path, after the mix bus insert and before the master fader:



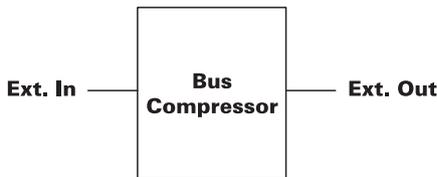
Note that for the compressor to be active, the IN switch must be depressed.

You can remove the bus compressor from the main mix and patch it elsewhere, by activating the EXT switch (2):

Mix Output (with EXT active):



Bus Compressor (with EXT active):



EXT (External) Input ②



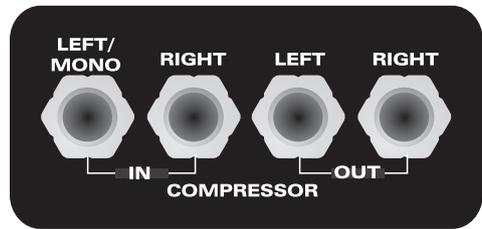
Press this switch to remove the bus compressor from the main mix, so that it can be patched elsewhere.

Connections

The compressor inputs and outputs appear on balanced 1/4" TRS jack connectors on the rear panel:



External Compressor I/O



To use the compressor on a mono source, connect only the left IN and OUT.

The ability to patch the compressor opens up many possibilities:

Example 1

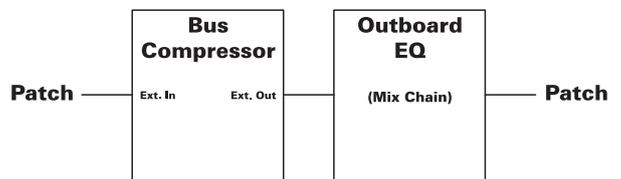
To use the bus compressor on a pair of channels:

- On the rear panel, patch the channel's insert SND to the compressor IN, and the compressor OUT to the channel's insert RET for left and right channels.
- Now engage the EXT switch and the channel strip INS IN switches.
- By selecting the channel's DAW input flip, you can now feed any signal to the main channels from your DAW. This could be individual tracks, subgroups or final mixes.
- Print back the processed signal via the channel's direct out.

Example 2

To change the order of processing within the main mix path - for example, to apply EQ after the bus compressor:

- Engage the EXT switch and the main MIX INSERT IN.
- On the rear panel, patch the MIX INSERT SND to the COMPRESSOR IN.
- Patch the COMPRESSOR OUT to the outboard EQ IN.
- Patch the outboard EQ OUT to the MIX INSERT RET.



GAIN REDUCTION Meter ③



This is a 10-segment LED bargraph meter, indicating the amount of gain reduction applied when compression is active. The scale shows 2dB to 20dB of compression.

*** TIP!** To keep the amount of bus compression at “musical” levels, only the first few LEDs should be flickering.

Compression RATIO ④



This control sets the amount of compression. The options are:

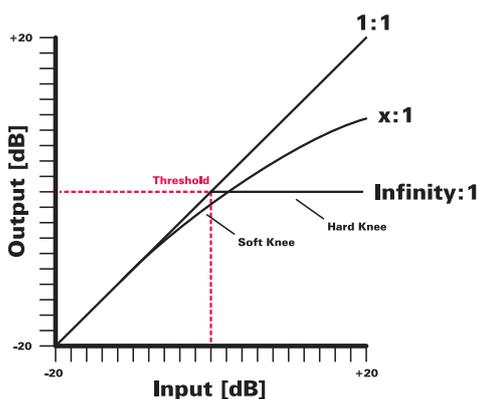
- 1.2 : 1 (very gentle)
- 1.5 : 1
- 2 : 1
- 4 : 1
- 10 : 1
- 20 : 1 (close to limiting)

The ratio is the amount of compression applied once input signal exceeds the threshold level. The options are specified as a ratio of input vs output - for example, 2:1 indicates that once the input signal crosses the threshold by 2dB, an increase of only 1dB is presented at the output (i.e. gain reduction of 1dB is applied).

*** TIP!** Generally, low ratios such as 1.5:1 or 2:1 are good for the main mix bus.

*** TIP!** As the ratio increases, the compressor acts more like a limiter (10:1 or 20:1). This can be very powerful on dynamic sources such as drums or vocals.

The compressor uses a soft knee design to produce smooth results. The compressor begins to act slightly below the threshold, and this point is similar for each ratio. This makes it possible to set the threshold, and then audition ratios without re-adjusting the threshold control.



Compression THRESHOLD ⑤



This control sets the threshold level, and is variable from -20 to +20dB.

The threshold level is the point at which compression begins.

ATTACK Time ⑥



This control sets the attack time in milliseconds (ms). The options are:

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

The attack is the time taken for the compressor to react to the input signal, once it reaches the threshold level, and start compressing (apply gain reduction).

*** TIP!** Slower attack times (10 or 30ms) are often preferred for mix bus or program material, as they will not affect transients or distort low frequency content as much as a faster attack time.

RELEASE Time ⑦



This control sets the release time in seconds (sec). The options are:

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

The release is the time taken for the compressor to react to the input signal, once it passes below the threshold level, and stop compressing (release the gain reduction).

RELEASE Time (7) (Continued)



*** TIP!** Use faster release times if you want to get the compressor out of the way quickly, and produce very obvious compression which can lead to “pumping”. For your drum bus, try starting at **0.1 sec** and then work backwards until you get a suitable balanced effect.

*** TIP!** Longer release times and **Auto Release** are good for a vocal bus, piano or the main mix where you want a more subtle effect.

Auto release alters the release time in a program-dependent manner. It produces a smooth result, ideal for the main mix bus or vocal subgroup.

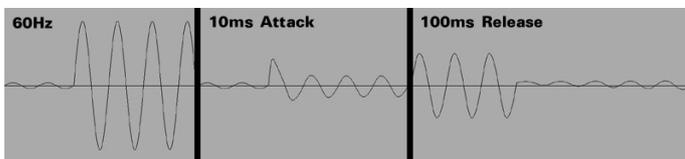
Waveshaping with Fast Attack/Release

If the attack, or release time, is faster than one cycle of the lowest frequency to be reproduced, then distortion will occur. This is because the compressor alters the waveshape. You can calculate this as follows:

$$\text{One cycle, T (sec)} = 1/f \text{ (Hz).}$$

So, at 60Hz, one cycle takes approximately 17ms.

The following diagram illustrates typical waveshaping at 60Hz when a relatively fast attack time is applied (**10ms** attack; **0.1s** release; ratio of **4:1**):



Note that aside from some attack waveshaping, the bus compressor is very clean!

GAIN Make-Up (8)



This control adjusts the gain after compression, and is variable from 0 to 20dB. It is very useful for restoring the output level, which will have been reduced by active compression.

*** TIP!** When using the bus compressor on the main mix, set the **GAIN** make-up so that there is no change in level when you press the compressor **IN** switch (1). You can now easily compare the compressed and uncompressed mix.

DRY-WET Parallel Compression (8)



This control “crossfades” between the uncompressed and compressed signal, a technique known as parallel compression:

- fully anticlockwise = 100% **DRY** (only uncompressed).
- fully clockwise = 100% **WET** (only compressed).
- 12 o'clock = 50% **DRY** + 50% **WET** (equally balanced).

This popular mix technique allegedly started in New York, and is fantastic at preserving transient information while adding body and punch.

*** TIP!** Use parallel compression to add **energy** to your mix!

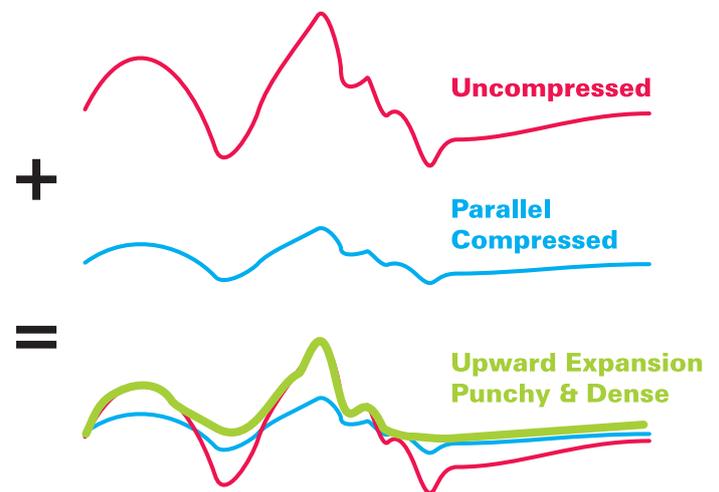
*** TIP!** You can use this technique with an outboard mix compressor, by using the **MIX INSERT** parallel **SUM** mode, as described on “Main Mix Bus” on page 26.

Parallel Compression Techniques

There are two useful approaches to parallel compression:

One is to set some gentle compression (low ratio), and turn up the control to 100% **WET**. Then edge back to add a small amount of **DRY** signal. This will add back some punch.

The other involves setting up a more aggressive compressor (high ratio, faster attack), and then adjusting the control close to 100% **DRY**. Then add a small amount of the **WET** signal to add energy and dynamic colour to the mix:



Bus Compressor Settings

Below are some useful suggested settings. Please bear in mind that these are only examples and will require adjustment for the best results - remember, your ears are the best monitors!

Mix Bus



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

Drum Bus



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

Drum Squash



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

Vocal Bus



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

Percussion Bus



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

Parallel Lift



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

Brass or Electric Guitar Bus



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

Crazy Smash - Electronica



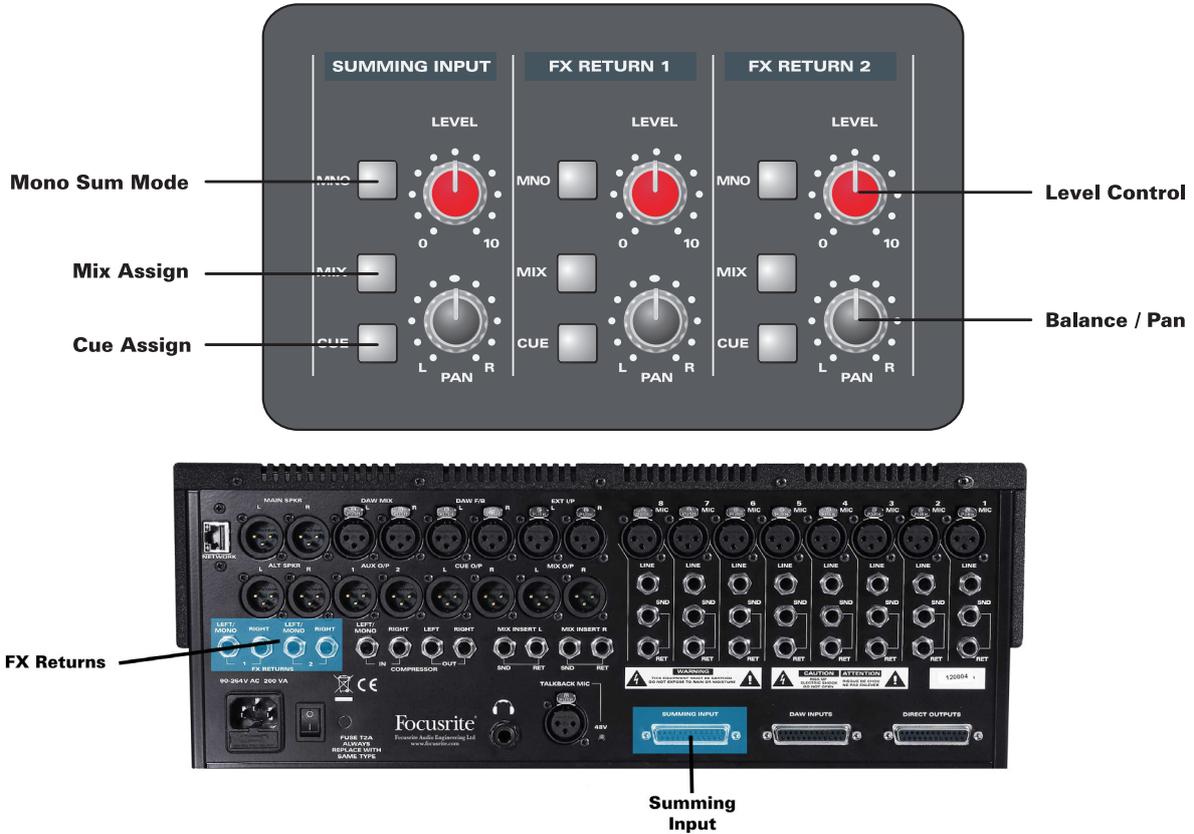
Adjust threshold for >10dB G.R
Adjust make-up gain accordingly

Summing Input & FX Returns

The **Control 2802** master section provides two ways of returning processing devices, such as reverbs and delays:

- An eight-channel Summing Input
- Two Stereo FX Returns

This provides a total of 12 additional inputs from this section of the console.

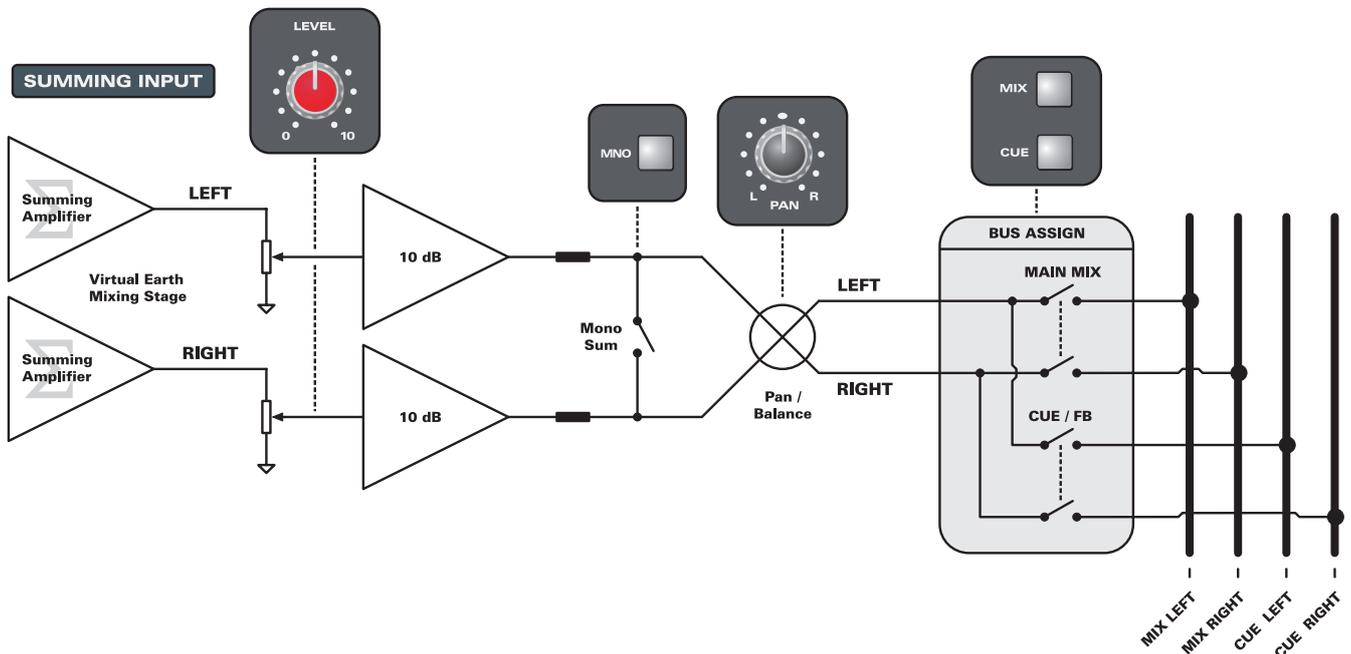


Summing Input

The Summing Input is fed from a DB25 connector on the rear panel, providing easy connection to a D-Sub patchbay system or any DB25 capable device (including many D/A converters). The input conforms to the Tascam DA-88 wiring specification for an eight-channel balanced analogue 25-pin D-Sub. See page 46 for pin-outs and wiring information.

The eight channel input is configured as four stereo pairs - Sum 1 L/R, Sum 2 L/R, Sum 3 L/R and Sum 4 L/R - which may be routed to the stereo main mix or stereo cue bus.

Use the **MNO** switch to mono the Summing Input mix bus - it can then be treated as 8 summed mono inputs (1 to 8).



Summing Input (Continued)

The SUMMING INPUT master section provides:

- **LEVEL** - overall level control for the combined eight channels.
- **PAN** - stereo balance or pan (if **MNO** is active).
- **MNO** - sums the Summing Input pairs to mono.
- **MIX** - routes the Summing Input to the stereo main mix bus (as four stereo pairs, or 8 summed mono inputs).
- **CUE** - routes the Summing Input to the stereo cue bus (as four stereo pairs, or 8 summed mono inputs).

Summing Input Applications

The Summing Input is ideal for returning sources to the main mix or stereo cue bus. Typical applications include:

- Bringing in stereo submixes from your DAW.
- * **TIP!**
- Flip the Fader Strips to the DAW layer in order to setup the submix in superfast time. Then record each submix while applying the fader automation.
- Bringing in stereo inputs from external synthesisers and samplers. For example, to feed the cue bus for artists foldback during practice sessions.
 - Returning stereo or dual mono FX processors - use the output controls on each outboard device to balance their levels.

Stereo FX Returns

FX Return 1 and 2 are fed from 1/4" TRS balanced jack connectors on the rear panel.

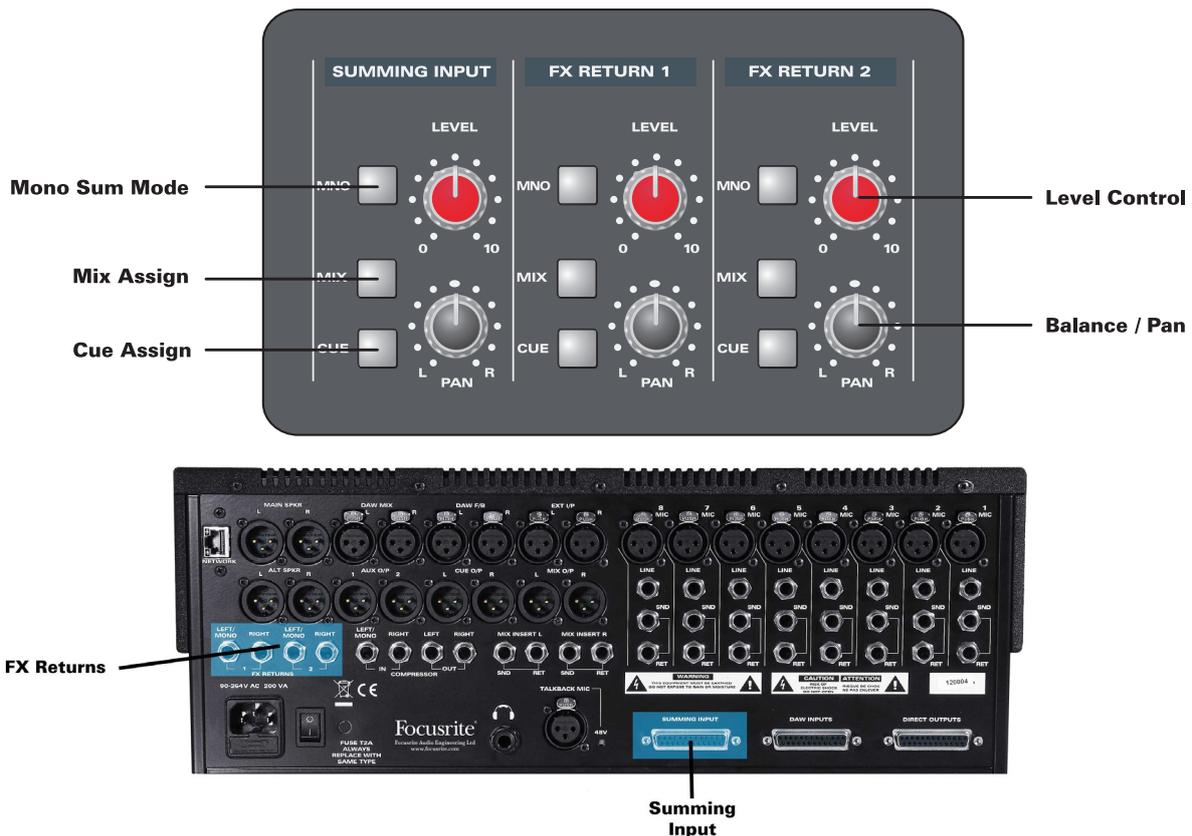
Each stereo FX RETURN section provides:

- **LEVEL** - overall level control for the stereo input.
- **PAN** - stereo balance or pan (if **MNO** is active).
- **MNO** - sums the stereo FX Return to mono.
- **MIX** - routes the FX Return to the stereo main mix bus.
- **CUE** - routes the FX Return to the stereo cue bus.

FX Return Applications

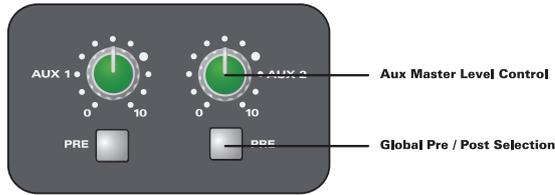
The FX Returns are ideal for:

- Returning stereo FX processors to the main mix - for example, those fed from the mono aux sends (aux 1 and 2).
- Adding effect to an artists headphone mix - by routing onto the cue bus. This ensures that the artist feels comfortable during their performance.
- Bringing in stereo submixes from your DAW or any other external stereo input.



Aux Masters

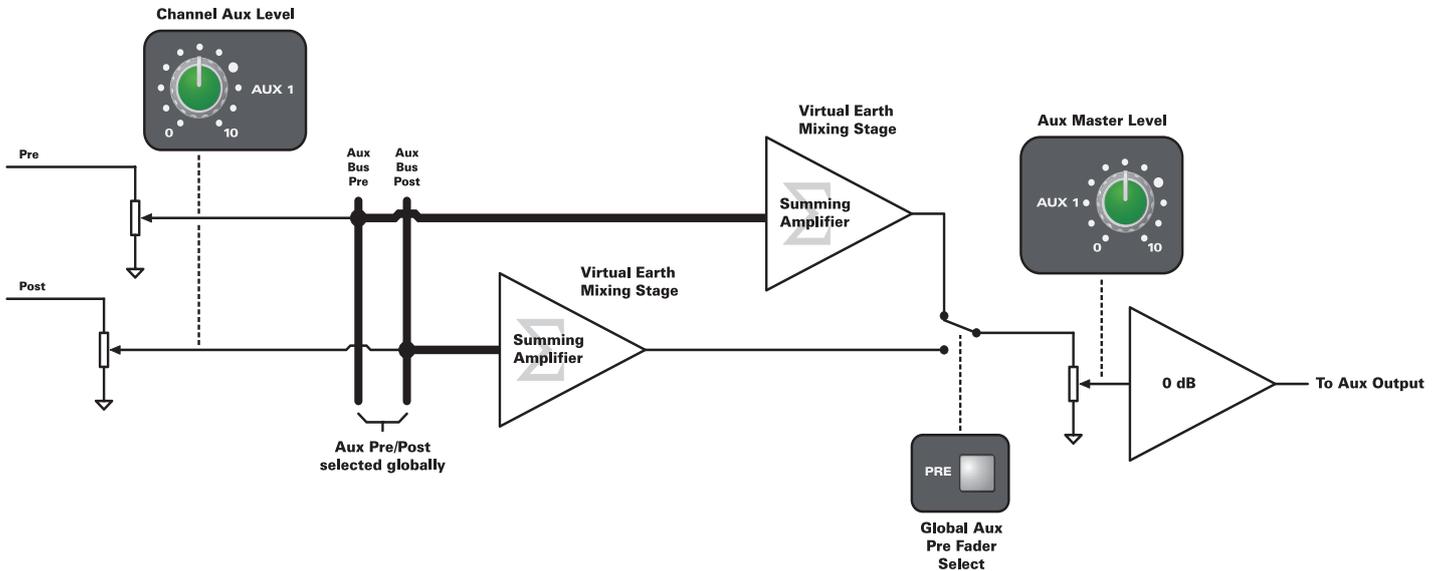
Control 2802 has two mono aux buses with a simple set of master controls:



- **LEVEL** - master level of the aux output.
- **PRE** - switches channel aux sends from post-fader to pre-fader. Note that the sends from ALL 8 channels are affected.

Signal leaves the console via the balanced male XLR outputs (AUX O/P 1 & 2) on the rear panel.

Aux 1 Signal Flow (Aux 2 is identical):

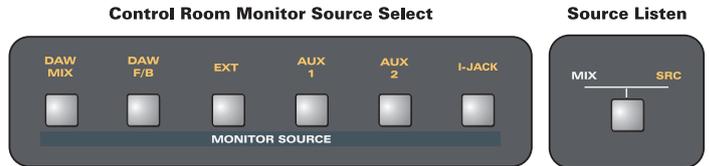


Monitoring the Aux Buses

Each of the aux busses may be selected as a control room or **Cue/FB** monitor source. This makes it easy to listen in the control room while building an aux mix, and/or to use auxes to feed artists headphones (via the cue output).

To monitor an aux bus in the control room:

Select **AUX 1** or **AUX 2** from the MONITOR SOURCE section. Then press MIX/SRC so that **SRC** is selected:

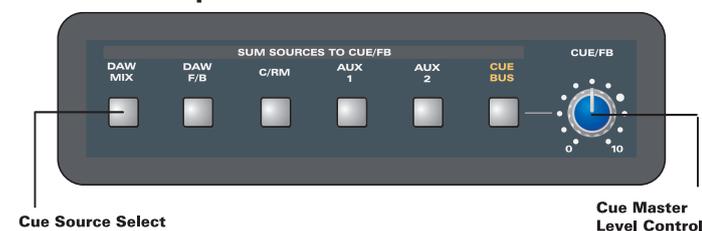


Please note that monitor sources sum if you make more than one selection. So, make sure that you deselect the other options if you wish to monitor an aux bus in isolation.

Cue Master

Control 2802 features a very flexible cue output. The main application is to send foldback to artists headphones; or the cue output may be used as a stereo effects send.

CUE/FB Output



The **CUE/FB** panel selects which sources feed the cue output - signal leaves the console via the balanced male XLR outputs (CUE O/P L/R) on the rear panel.

Use the **CUE/FB** level control to adjust the master output level.

Sources include the stereo cue bus, but also other inputs such as auxes, a stereo DAW input, and so on. Please note that sources sum if you make more than one selection. This allows you to combine the different options.

Six sources are available for the cue output:

- **DAW MIX** - a stereo line level input fed from the balanced female XLR connectors (DAW MIX L/R) on the rear panel.

The **DAW MIX** input is usually fed from the main mix of your DAW. Select this option if you want the artist to hear what is being summed "in the box".

- **DAW FB** - a stereo line level input fed from the balanced female XLR connectors (DAW FB L/R) on the rear panel.

The **DAW FB** input is intended for returning foldback mixes from your DAW. Select this option if you want the artist to hear a headphone mix generated within your DAW.

*** TIP!** You can use either of stereo line level inputs - **DAW MIX** or **DAW FB** - to feed an external two-track device to the cue output.

- **C/RM** - the control room monitor output (after the MONITOR SOURCE selector but before **VOLUME**, etc.)

Select **C/RM** if you want the artist to hear the control room balance. This is a great way of instantly getting a mix to the cue output, especially if you are using the alternative cue path for extra mix inputs.

Source Listen The monitor output is picked up after the MONITOR SOURCE selector. Therefore, if you change the control room monitor source, this will also affect the cue output. If you want the artist to receive the main mix, make sure that the MONITOR SOURCE MIX/SRC switch is assigned to the **MIX** position.



- **AUX 1 or AUX 2** - the mono aux 1 or aux 2 outputs.

Select either of these options to send an aux bus to the cue output - for example, if you are using the alternative cue input for extra mix inputs, then you may wish to generate a cue mix from an aux.

*** TIP!** Combine **C/RM** with either **AUX 1** or **AUX 2** to send the artist the control room balance plus more of "themselves". Turn up the aux send on the artist's channel, and switch the aux to pre-fader - this ensures that the artist hears themselves regardless of the control room balance.

CUE/FB Sources (Continued)

- **CUE BUS** - the stereo cue bus.

Select **CUE BUS** to send the stereo cue mix to the **CUE F/B** output.

Make sure that this is the only source selected, if you want to use the cue bus in isolation. Or, combine this with say the **DAW MIX** to send the artist both a cue mix from the recording channels plus pre-mixed tracks from "in the box".

*** TIP!** You would select this option to use the cue output as a stereo effects send. Make sure that only the **CUE BUS** is selected as the **CUE/FB** source. Then set up the channel sends, and adjust the master **CUE/FB** level control.

Monitoring the Cue Output

The best way to monitor the cue output is on your control room headphones. This will allow you to make judgements while listening in an identical manner to the artist, and therefore produce a more comfortable foldback mix.

Monitoring on headphones is also useful if you are the artist! Or, if you wish to record say a singer or guitar player in the control room, away from the studio and the rest of the band.

To monitor the cue output on the control room headphones:



Turn on the **CUE/FB TO PHONES** switch in the monitor section - this routes the cue output to the headphone connector on the rear panel.

Make sure that the headphones are **ON** and adjust the **PHONES** level.

Alternatively, you can monitor the cue bus on the control room speakers as follows:



Route the cue bus to the main mix via the **ASSIGN TO MIX - CUE** switch.

Then deassign the channels from the main mix to avoid summing the same signal twice - turn off the **MIX** bus assign switch on each of the 8 channel strips.

Then make sure that the MONITOR SOURCE MIX/SRC switch is assigned to the **MIX** position to monitor the main mix.

Note that this routes the cue bus, rather than the cue output, to the control room monitors.

Talking to the Cue Output

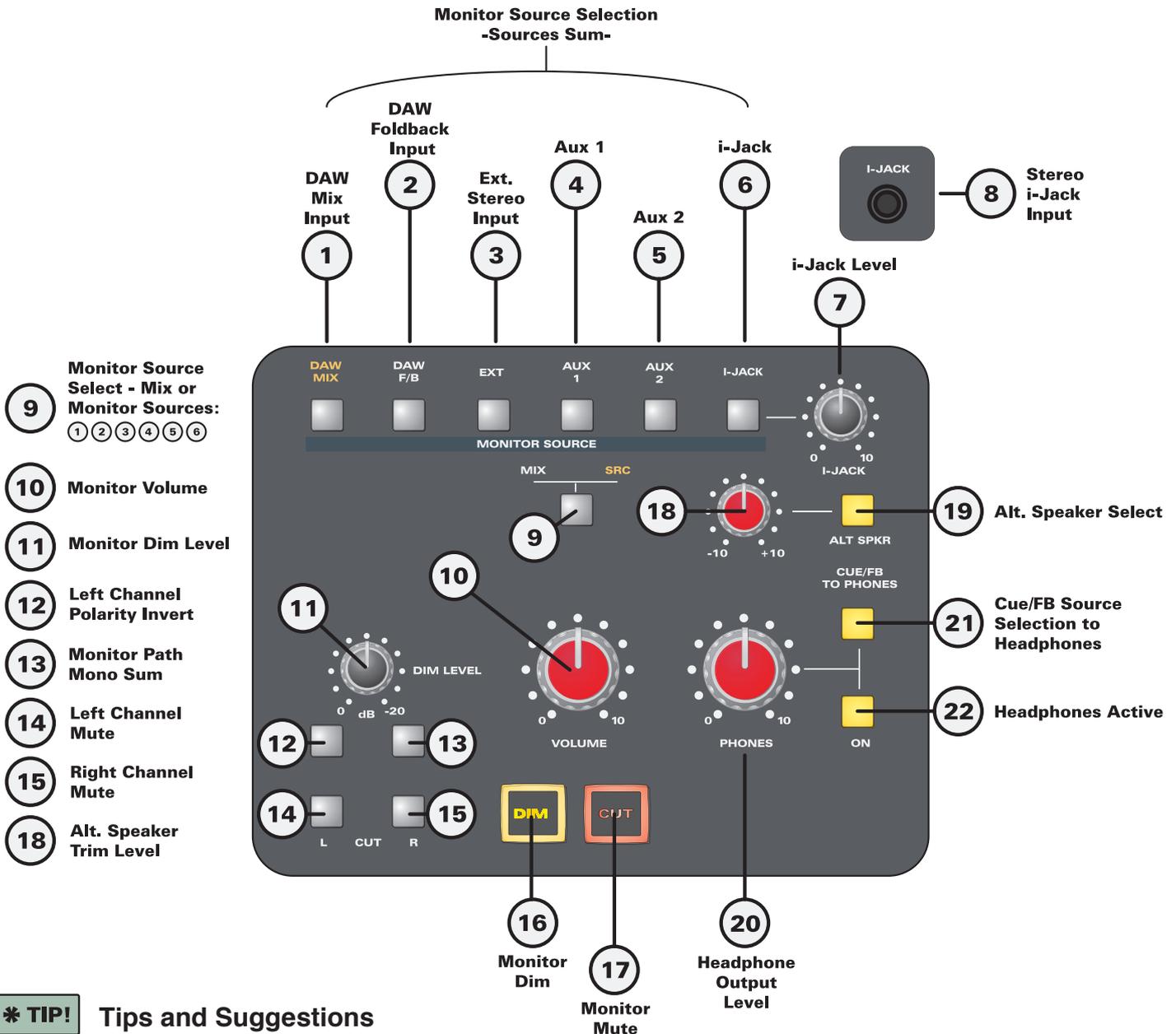
The **TALKBACK** controls in the master section can be used to talk to the cue output. See "Talkback" on page [39](#).

Monitor Section

The monitor section of **Control 2802** provides comprehensive control room monitoring. Outputs are provided for main and alternate speakers plus headphones. See page 8 for more details on the rear panel connections.

The circuitry is mostly passive, with only one buffer stage before the balanced loudspeaker outputs. Thus the console provides a tonally neutral and accurate monitor path.

*** TIP!** You may find it useful to print this page so that you can refer to the control numbering while reading this section.



* TIP! Tips and Suggestions

Select **SRC** (9) + **DAW MIX** (1) to quickly audition the mix from your DAW - great when summing “in the box”.

Select **SRC** (9) + **DAW FB** (2) to monitor headphone mixes generated by your DAW. Using this workflow, you can save and recall cue mixes from the DAW session and apply DAW effects. Beware of latency though.

Select **SRC** (9) + **EXT** (3) to monitor a two-track device such as a mastering tape machine, CD player, DAT, etc.

Use the handy mini jack socket (8) to connect a portable music player (or similar). Then select **SRC** (9) + **I-JACK** (6) and adjust the level (7) - this is very useful for checking rough mixes, MP3 mix translations or reference material (we recommend high resolution, lossless audio for use as reference material).

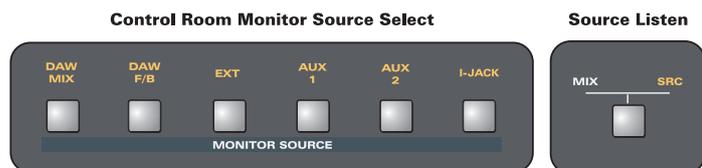
Use the polarity invert (12) to check for phase issues, and **MONO** sum (13) to check mono compatibility. Select both switches (12 & 13) to create “sum & difference” monitoring - great for checking stereo content.

Use **CUT L** (14), or **CUT R** (15), to monitor mono sources correctly on a single speaker. Mono is more accurate from a single source as it minimises phase cancellation in the listening environment.

Select **ALT SPKR** (19) and adjust the level trim (18) to check mixes on a second playback system, connected to the balanced male XLR outputs (ALT SPKR L/R).

Select **CUE/FB TO PHONES** (21) and adjust the **PHONES** volume to audition an artist’s cue output on your control room headphones (connection on the rear panel).

MONITOR SOURCE Selection ① – ⑥



Six sources are available for the control room monitor output (**DAW MIX**, **DAW FB**, **AUX 1** and **AUX 2** are identical to the options on the **CUE/FB** panel).

Please note that monitor sources sum if you make more than one selection. Therefore, be sure to deselect all other options if you wish to monitor one source in isolation.

Also note that the **MIX/SRC** switch (9) must be assigned to the **SRC** position in order to monitor the output of the selector.

The six source options are:

- **DAW MIX** (1) - a stereo line level input fed from the balanced female XLR connectors (DAW MIX L/R) on the rear panel.

The **DAW MIX** input is usually fed from the main mix of your DAW. Select this option to quickly audition what is being summed “in the box”.

- **DAW FB** (2) - a stereo line level input fed from the balanced female XLR connectors (DAW FB L/R) on the rear panel.

The **DAW FB** input is intended for returning foldback mixes from your DAW. Select this option to audition headphone mixes generated within your DAW.

- **EXT** (3) - a stereo line level input fed from the balanced female XLR connectors (EXT IP L/R) on the rear panel.

The **EXT** input is intended for two-track mastering devices such as a half-inch or quarter-inch tape machine, CD player, DAT or digital recorder. Select this option to monitor the playback.

- **AUX 1** or **AUX 2** (4 & 5) - the mono aux 1 or 2 outputs.

Select either of these options to monitor the aux output.

- **I-JACK** (6) - a stereo unbalanced mini jack connector located on the front panel (see below).

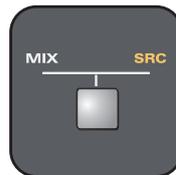
I-JACK Level Control & Input ⑦ – ⑧



This stereo mini jack input (8) is designed for connecting portable music players. This is a stereo unbalanced input with approximately 10k Ohms input impedance.

Select **I-JACK** (6) as your **MONITOR SOURCE** and adjust the level control (7) to monitor playback.

MIX / SRC Monitor Select ⑨



This switch determines whether you are monitoring the main mix or the output of the **MONITOR SOURCE** selector. One of the options must be active:

- **MIX** - the main mix bus output.
- **SRC** - the output of the **MONITOR SOURCE** selector (1-6).

Monitor VOLUME ⑩



Use this level control to set the main listening level.

Monitor Calibration

You should calibrate your listening level if possible.

By calibrating your monitor amplifiers to produce approximately 83 to 85dB SPL (for a known and typical source level), it is possible to achieve a listening level that ensures the ears operate at their flattest frequency response (according to aural perception and equal loudness contours).

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

Use a calibrated SPL meter, set to read with a slow response time, RMS detection and C-weighted filter.

C-weighting is the most wideband weighting filter and most similar to the human ear at 85dB SPL.

Place the microphone from the SPL meter in the listening position, and set the monitor **VOLUME** control to a memorable position (this will become your normal listening level).

Then adjust your monitor amplifiers until you read 85dB SPL from each loudspeaker.

By using this **VOLUME** control position, when listening to program material, you should have a good chance of obtaining a constant and accurate reference level.

Monitor DIM LEVEL ⑪



This control sets the amount of “dim” which will be applied when the **DIM** switch (17) is active.

The dim level can be set anywhere from 0 to -20dB.

Left Channel Polarity Invert (12)



Press and hold this switch to invert the polarity (phase) of the left monitor channel. The switch is momentary to ensure that it is not left on by accident!

It can be used together with the **MONO** switch (13) to check for out of phase stereo content (see below).

Monitor MONO Sum (13)

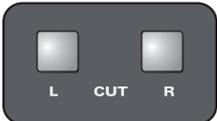


This switch sums the left and right monitor channels to produce a mono output.

This feature is very useful to check for out of phase stereo content (especially at low frequencies) and ensure the mono playback compatibility of your mix. Select **MONO**, and if your mix sounds thin, or certain frequencies “disappear”, then press and hold the polarity invert switch (12). This creates a L-R sum instead of L+R. If the mix problems are corrected, then some of your stereo sources, or the mix itself, may be out of phase.

The **MONO** switch is also useful when positioning stereo microphones. With the two microphones connected and panned L/R to the mix, select **MONO**. Now adjust the position of each microphone to get the best tonal balance. This should minimise phase cancellation once you switch back into stereo.

CUT L and CUT R (14) – (15)



These switches mute the individual left and right monitor channels - perfect for checking a mono sum on a single speaker.

This is often the most accurate method of checking a mono source, as it minimises phase cancellation in the listening environment.

Monitor DIM (16)



Press this switch to dim the monitor output. The amount of dim is set by the **DIM LEVEL** control (11).

Note that **DIM** is engaged automatically by the **TALKBACK** switch. This is to prevent “howl-round” when using the talkback communication system, see page [39](#).

Monitor CUT (17)



Press this switch to mute the monitor output. This is a global cut that disconnects both the main and alternate loudspeaker outputs.

Alt. Speaker Select & Output Trim (18) – (19)



A second set of loudspeakers may be connected to the alternate outputs (ALT SPKR L/R). Applications include nearfield or mini “grot-box” monitors.

Use the **ALT SPKR** switch (19) to mute the main speaker outputs (MAIN SPKR L/R) and feed the monitor output to the alternate outputs (ALT SPKR L/R).

Use the level control (18) to trim the alternate speaker level. This can be used to match levels between the speaker sets.

Overall volume is still determined by the monitor **VOLUME** control (10).

Headphone Output (20) – (22)

A high quality stereo headphone output is available on the rear connector panel:



On the front panel, use the **ON** switch (22) to activate the headphone output.

Use the **PHONES** level control (20) to adjust the listening level. You will find plenty of level available, enough to drive most studio headphones to a sufficiently loud SPL level.

By default, the headphone output is fed from the monitor section. Thus, the source is selected by the **MIX/SRC** switch (9) and **MONITOR SOURCE** selector panel (1 to 6).

Alternatively, you can monitor the cue output, by engaging the the **CUE/FB TO PHONES** switch (21) - great for auditioning what the artist is hearing. Note that this selects the cue output (after the **CUE/FB** source selector), and therefore may be the sum of several cue sources. See page [35](#) for details.

Talkback

The master section of **Control 2802** provides communication functions for use during tracking and overdub sessions.

You may use either the integrated talkback microphone, or an external microphone, to talk to the cue section. When active, talkback feeds the balanced male XLR outputs (CUE O/P L/R) on the rear panel.

Connections

Internal Talkback Mic

An integrated electret talkback microphone is located in the master section above the power indicators:



External Talkback Mic

Alternatively, connect an external talkback microphone to the female XLR input on the rear panel (TALKBACK MIC):



External Talkback Microphone Input

To engage 48V dc phantom power for the external microphone, engage the small recessed switch found next to the XLR on the rear panel:



Talkback Source

The internal mic is used as the default talkback source.

To override this and use the external mic, engage the small recessed switch located above the TALKBACK LEVEL control - you will need to use a long thin non-conductive object to depress the switch:



Talkback Controls

Press the **TALKBACK** switch to talk to the artist. When active, talkback feeds the cue output - the balanced male XLR outputs (CUE O/P L/R) on the rear panel.

Use the **LEVEL** control to adjust the talkback microphone preamplifier. Gain may be adjusted from +20 to +55dB.

Note that the **TALKBACK** switch is momentary. This is to avoid leaving it on by accident - for example, during delicate performance-related discussions!

Also note that the **TALKBACK** switch automatically dims the monitor output. This is to prevent "howl-round" loops between the talkback mic and the loudspeakers.

Solo Functionality

Control 2802 has three solo modes, plus “solo-in-front” capability, controlled globally from the master section:



Note that when a channel is placed into solo, the master section **SOLO** indicator lights.

AFL – After Fade Listen

This is the default solo mode - both the **PFL** and **SIP** switches are in the UP position.

When a channel **SOLO** is active, signal is taken from the main channel path (after the fader, cut and pan), and sent to the stereo AFL solo bus. The monitor section automatically switches to listen to this bus.

Use the **SOLO LEVEL** control to trim the level of the solo bus.

Use the **SOLO IN FRONT** control to “blend in” some of the main mix (see opposite).

PFL – Pre Fader Listen

This mode requires the **PFL** switch to be in the DOWN position; **SIP** remains in the UP position.

When a channel **SOLO** is active, signal is taken from the main channel path (before the fader, cut and pan), and sent to the mono PFL solo bus. The monitor section automatically switches to listen to this bus.

Use the **SOLO LEVEL** control to trim the level of the solo bus.

Use the **SOLO IN FRONT** control to “blend in” some of the main mix.

SIP – Solo In Place

This mode requires the **SIP** switch to be in the DOWN position; **SIP** overrides both AFL and PFL and therefore the **PFL** switch can be either UP or DOWN.

This mode does not use the solo bus. Instead when a channel **SOLO** is active, all other channels are **CUT** from the main mix. This mode is often known as “destructive” solo, as it interrupts signals going to the main mix.

Solo-in-place is great for mixdown as it allows you to hear a channel complete with all its effects, such as reverb or delay processing connected to FX Returns 1 & 2.

You can also use the **SIP Safe** function to isolate channels that you do not want **CUT** when a **SOLO** is active. This is very useful if you are returning reverb, delay or other processing through the main channel paths. See page [25](#) for more on **SIP Safe**.



SOLO LEVEL Trim



In either **AFL** or **PFL** mode, use this control to trim the level of the solo bus. Level may be trimmed by +/-10dB.

This enables you to control any level jumps that may occur when changing between the full mix and solo'd channels.

Please note that SOLO LEVEL does not work in SIP mode.

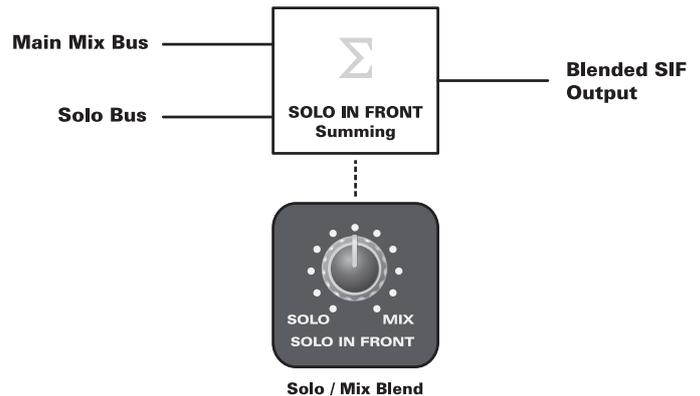
SIF – Solo In Front

In either **AFL** or **PFL** mode, the main mix can be “blended in” behind the solo'd channels using solo-in-front. This is very useful as it allows you to isolate a channel while listening in context.

Use the **SOLO IN FRONT** control to adjust the level of the main mix:

- fully anticlockwise = listen only to the solo bus.
- fully clockwise = listen to the main mix and solo bus at equal levels.

The control is a continuously variable potentiometer. Position it somewhere in the middle to isolate a channel while listening in context:



Please note that SOLO IN FRONT does not work in SIP mode.

Technical Information

Control 2802 Specifications

The following are general performance figures taken with Prism DScope test equipment:

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

Firmware Updates

When updates are released 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.focusrite.com to obtain the latest firmware from the **Control 2802** webpage. Then follow the simple instructions provided with the firmware download to flash the upgrade to the console memory.

If you experience any issues please read the **Control 2802** FAQ found online at www.focusrite.com.

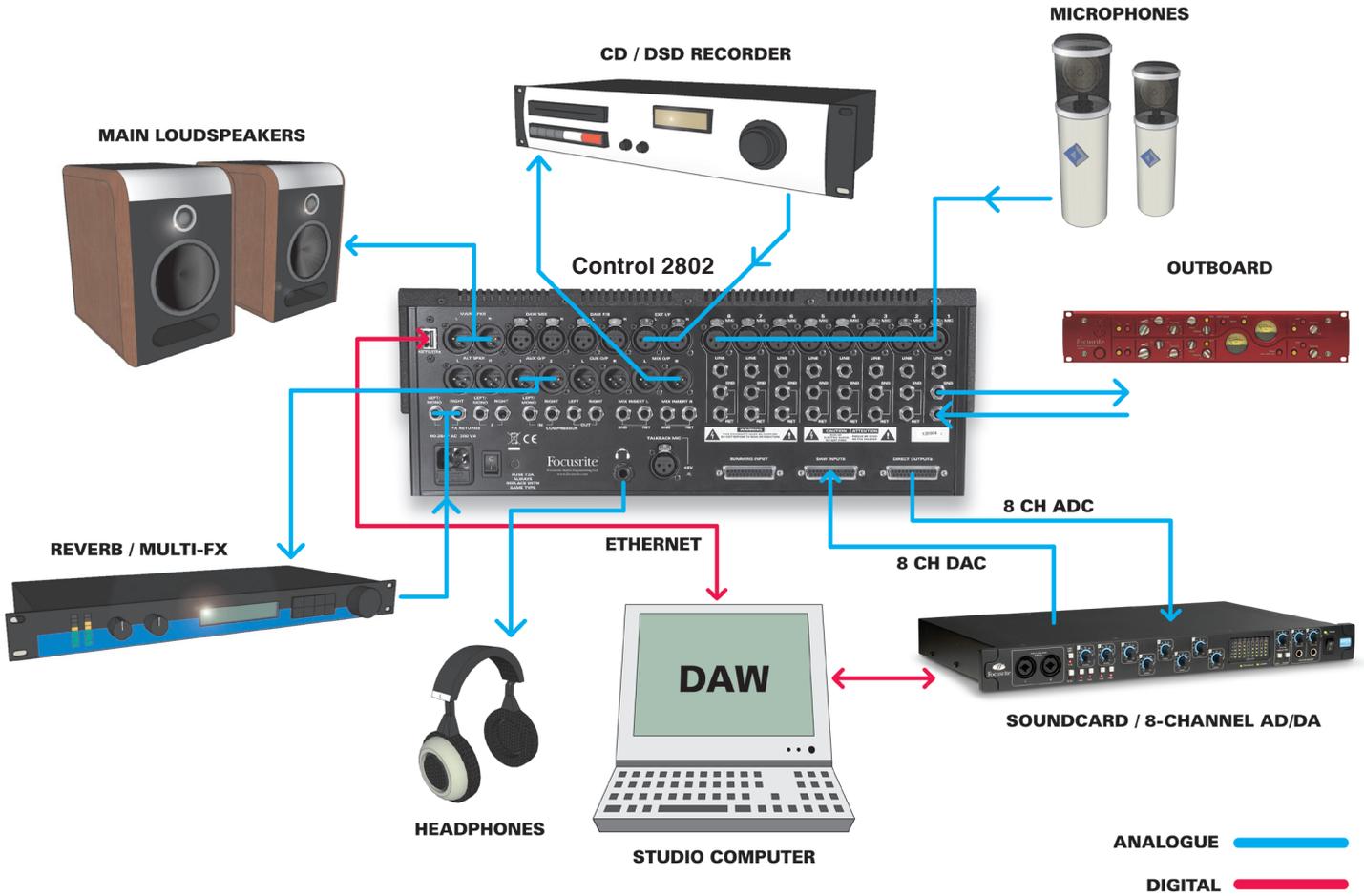
Example Systems

Example System 1: 8-Channel DAW System

This example illustrates a basic system with a typical 8-in and 8-out soundcard or AD/DA interface. There are eight 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.

8-CHANNEL DAW SYSTEM

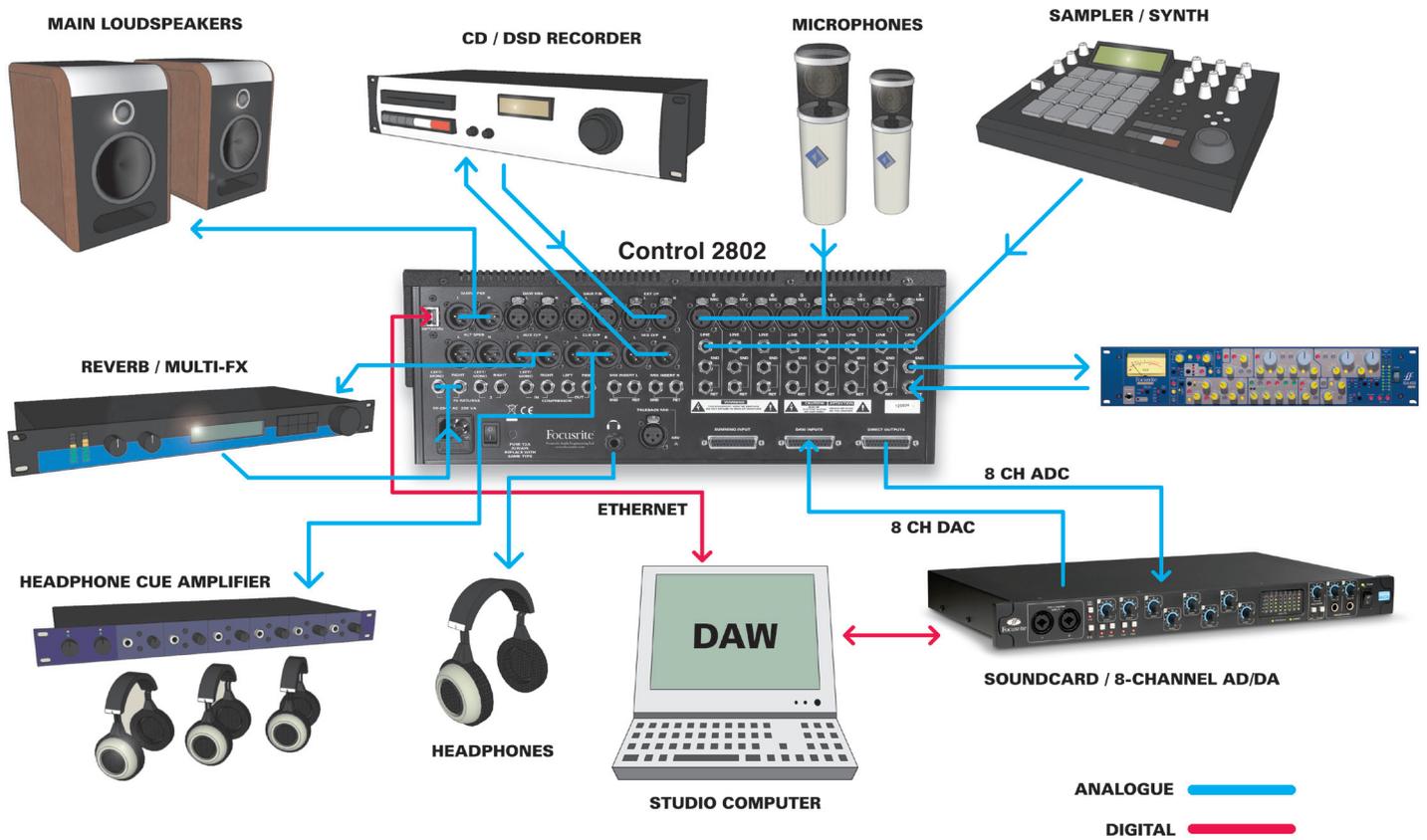


Example System 2: Hybrid Production Platform

As example 1, except 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.

HYBRID 8-CHANNEL PRODUCTION SYSTEM



Example System 3: 16 Channel Tracking Platform

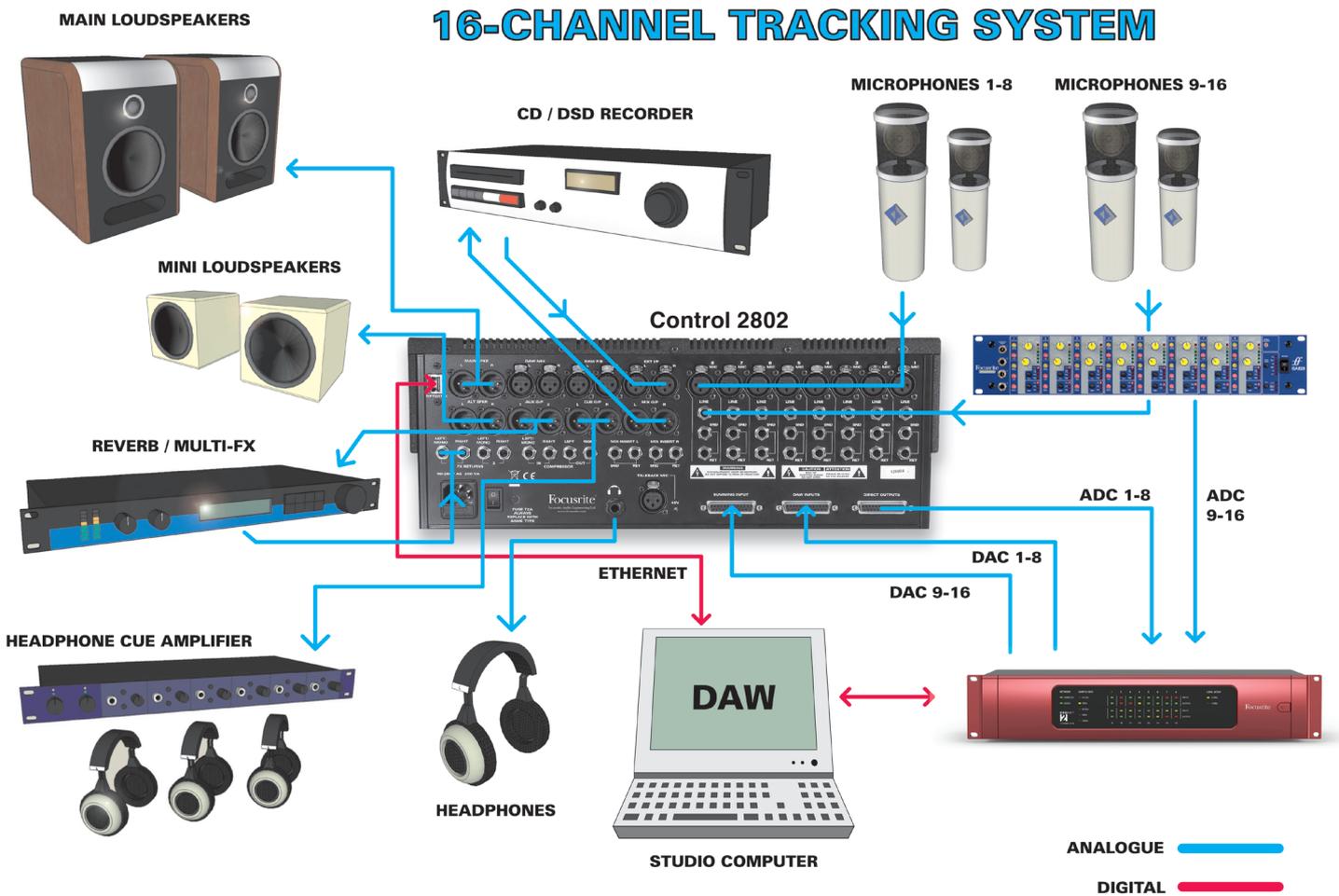
In this example a second pair of monitors are added to provide a useful reference for judging midrange and balances.

In order to provide 16 microphone inputs for tracking, an extra ISA 828 preamplifier is used. This is routed to a 16 channel capable soundcard / interface.

These eight feeds are sent back to the eight-channel summing input on **Control 2802**, allowing monitoring of these channels and access to the cue section.

In this example, the analogue line outputs of the pre-amplifier are also routed to the **Control 2802** line inputs, allowing optional access to the direct output system here (note that they would override the console microphone inputs at that point).

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



Example System 4: 24 Channel Mixing Platform

The final example illustrates a heavily connected **Control 2802** 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.

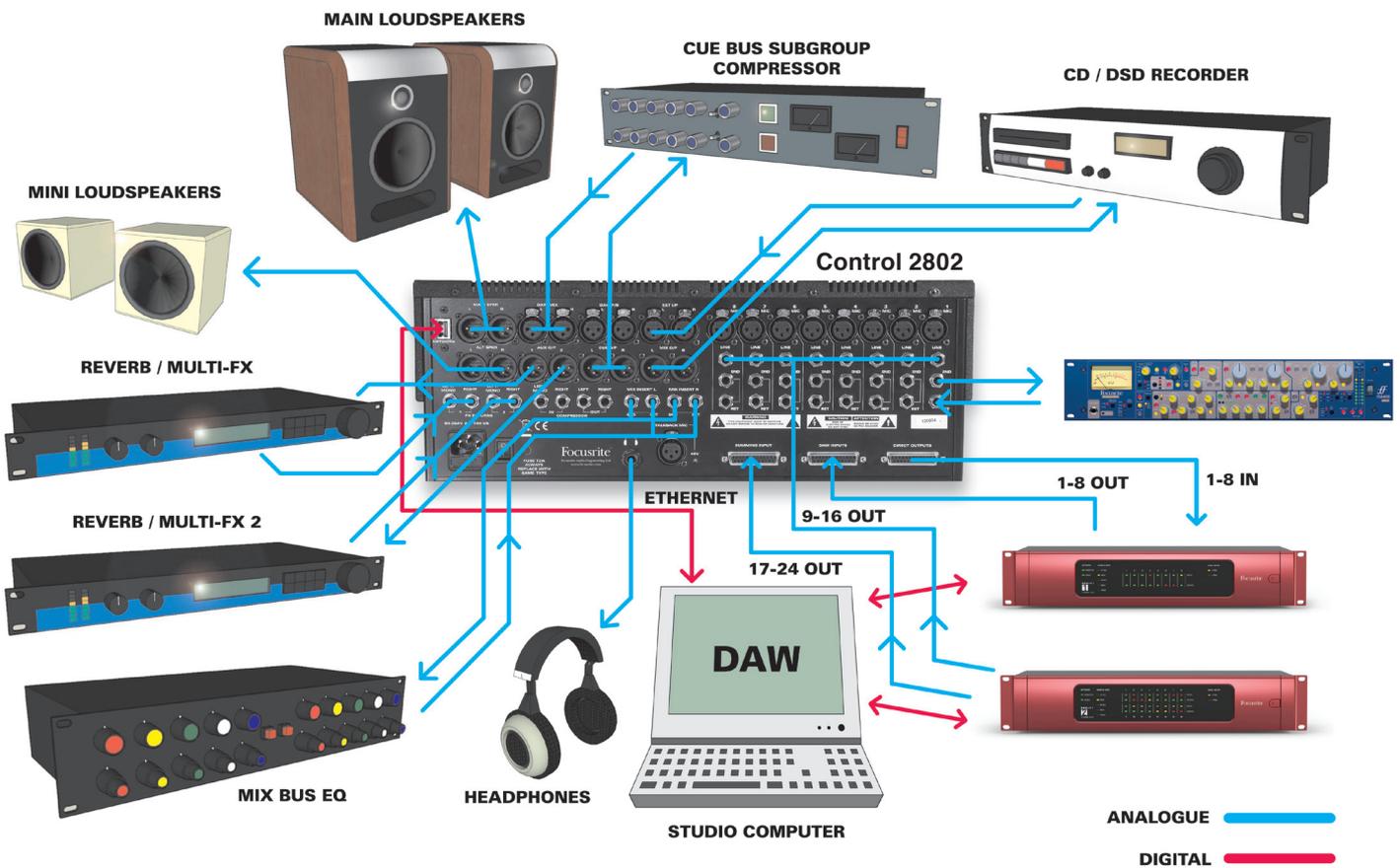
The first eight channels feed the standard DAW Inputs and can be used to access the main automation channels on **Control 2802** when selecting **DAW** input mode. These eight channels can also access the outboard selection via the switched insert points.

The following eight outputs (9–16) feed the **Control 2802** 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.

24 CHANNEL MIXING SYSTEM WITH CUE SUBGROUP



Connector Wiring & Specifications

For an overview of the rear panel connections, please see page 8. This section provides technical data including wiring and pin-out diagrams.

XLR Connections

All XLR balanced input and outputs are wired as:

- Pin 1 shield (chassis/shield ground)
- Pin 2 hot (+ve)
- Pin 3 cold (-ve)

TRS Jack Connections

All 1/4" TRS jacks are wired as:

- Tip (+ve)
- Ring (-ve)
- Sleeve (shield)

Channel Strip Inputs and Outputs

Each of the 8 channel strips provide:

MICROPHONE INPUT (balanced, female XLR):

- Input impedance is >2.5k Ohms.
- Maximum input level is approx. +16dBu (where 0dBu is referenced to 0.775Vrms).

LINE INPUT (balanced, 1/4" TRS jack):

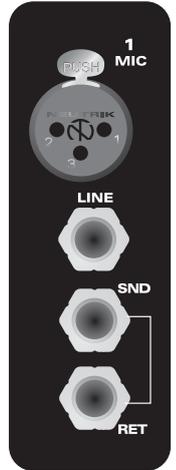
- Input impedance is >10k Ohms.
- Maximum operating level is +27dBu.

INSERT SEND (balanced, 1/4" TRS jack):

- Output impedance is 150 Ohms.
- Drive capability is +21dBu.

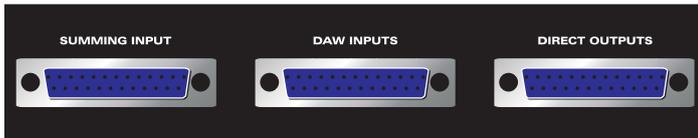
INSERT RETURN (balanced, 1/4" TRS jack):

- Input impedance is >10k Ohms.
- Maximum input level is +21dBu.



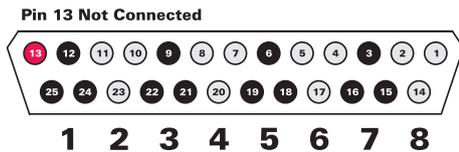
25-pin D-Sub Connections

The remaining channel strip connections and Summing Input are found on three 8-way female DB25 connectors at the bottom of the rear panel:



DB25 Tascam DA-88 Wiring Specification

Each conforms to the Tascam DA-88 wiring specification for an eight-channel balanced analogue 25-pin D-Sub as follows:



DAW INPUTS 1-8 (balanced, 8-way female DB25)

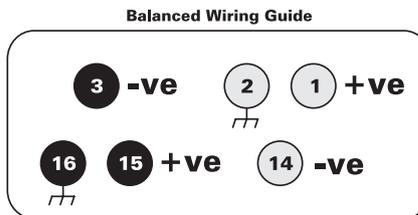
- Input impedance is >10k Ohms.
- Maximum input level is +27dBu.

DIRECT OUTPUTS 1-8 (balanced, 8-way female DB25)

- Output impedance is <75 Ohms.
- Maximum output level is +27dBu.

SUMMING INPUT 1-8 (balanced, 8-way female DB25)

- Input impedance is >10k Ohms.
- Maximum input level is +27dBu.



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

Channel Connections

Microphone Input

XLR Female - Balanced
 Pin 1: Chassis Ground
 Pin 2: Hot +ve
 Pin 3: Cold -ve
 Input Impedance: >2k5 Ohms
 Maximum Input Level: +20 dBu

Line Input

1/4" TRS Jack - Balanced
 Tip: Hot +ve
 Ring: Cold -ve
 Shield: Chassis Ground
 Input Impedance: >10k Ohms
 Maximum Input Level: +27 dBu

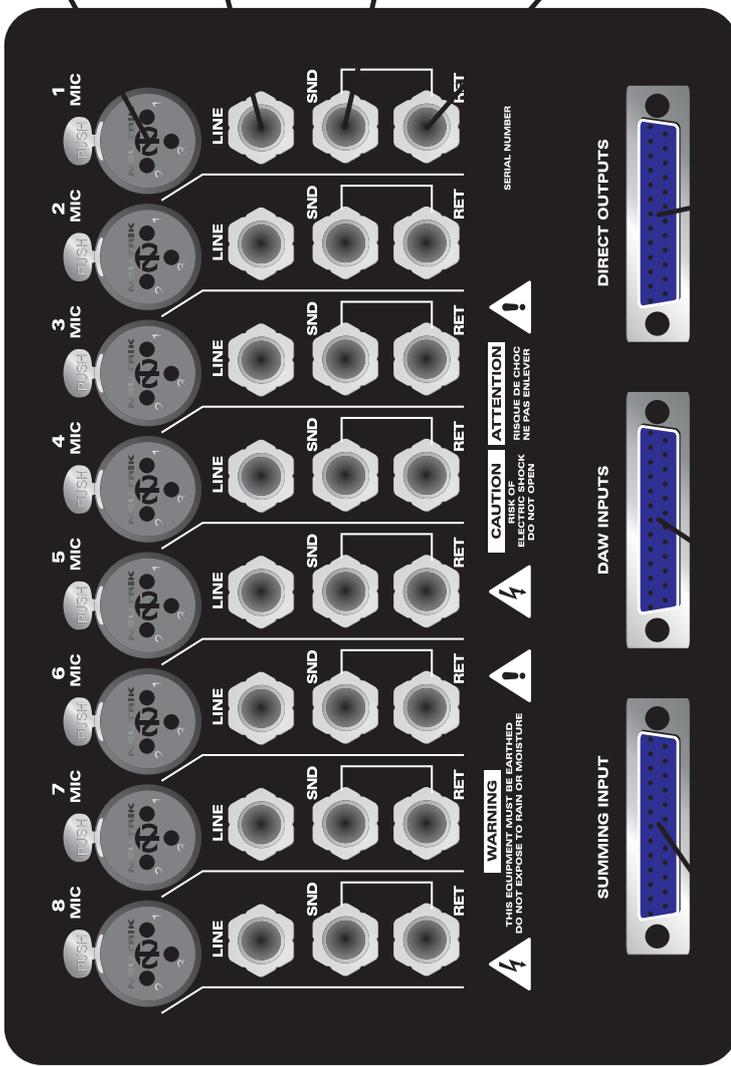
Insert Send

1/4" TRS Jack - Impedance Balanced
 Tip: Hot +ve
 Ring: Cold -ve
 Shield: Chassis Ground
 Output Impedance: <75 Ohms
 Maximum Output Level: +21 dBu

Insert Return

1/4" TRS Jack - Balanced
 Tip: Hot +ve
 Ring: Cold -ve
 Shield: Chassis Ground
 Input Impedance: >10k Ohms
 Maximum Input Level: +21 dBu

NOTE* The send jack is normalled to the return jack via switching contacts when no jacks are inserted.



Summing Inputs - Channels 1-8

DB25 - Balanced
TASCAM DA-88 Pinout
 (see manual for pinout specification)
 Input Impedance: >10k Ohms
 Maximum Input Level: +27 dBu

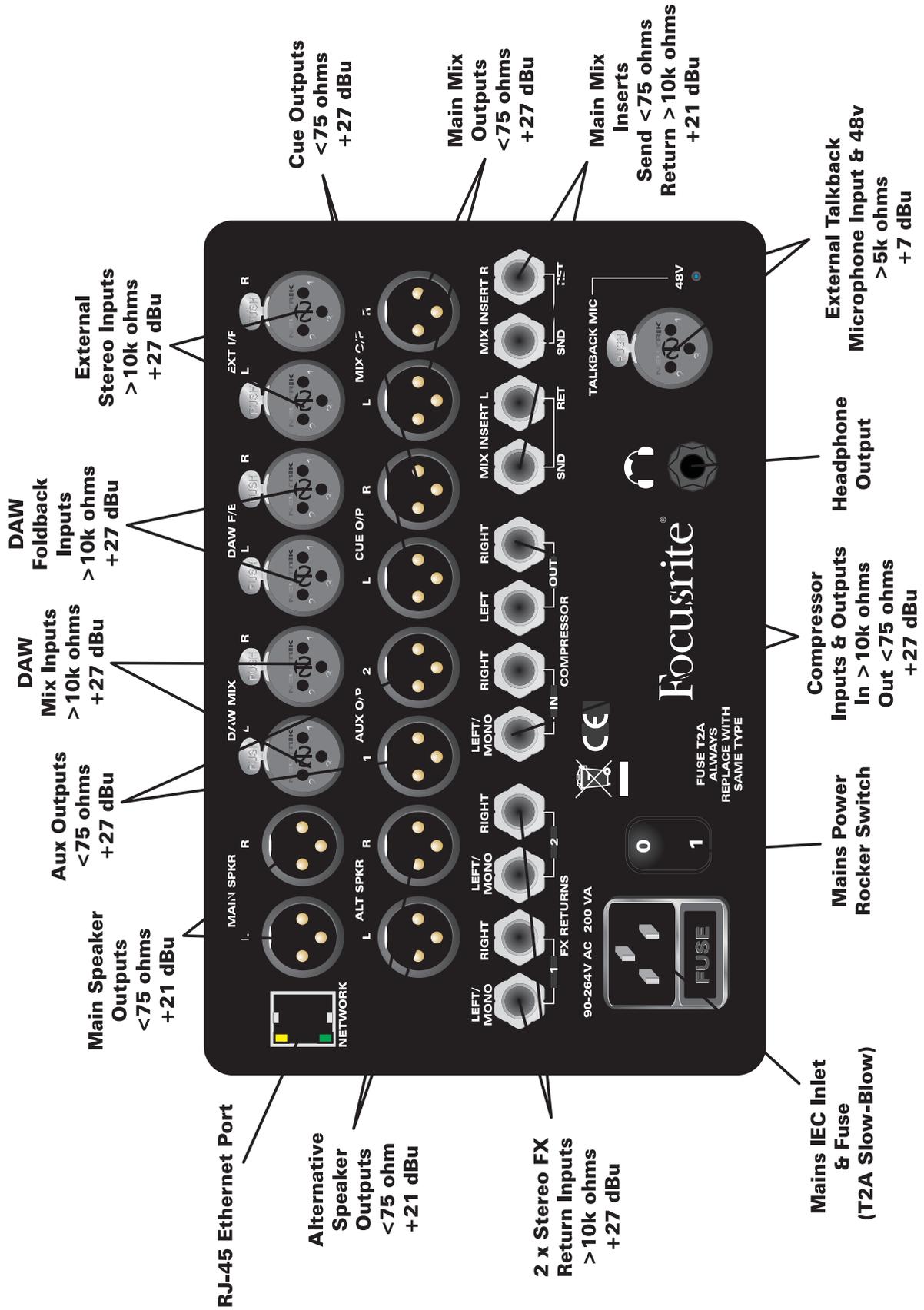
DAW Inputs - Channels 1-8

DB25 - Balanced
TASCAM DA-88 Pinout
 (see manual for pinout specification)
 Input Impedance: >10k Ohms
 Maximum Input Level: +27 dBu

Direct Outputs - Channels 1-8

DB25 - Balanced
TASCAM DA-88 Pinout
 (see manual for pinout specification)
 Output Impedance: <75 Ohms
 Maximum Output Level: +27 dBu

Master Section Connections



Master Section Connections

Working from top left to right:

Control 2802 connects to your DAW via IP-based networking protocols, using the standard **RJ-45 CAT-5e Ethernet** port. This 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 +27dBu.

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 +27dBu.

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 +27dBu.

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 +27dBu.

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 +21dBu, while the return jacks offer >10k Ohms input impedance with +21dBu of maximum input level.

The **Mains Power** input is a stand 3-pin IEC connector with integrated fuse, it will accept line voltages from 100-240V ±10%. 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!

The **External Talkback** microphone input is a balanced female XLR connector with >5k Ohms input impedance and a maximum input level of +7dBu. Please note 48V phantom power can be provided to this input.

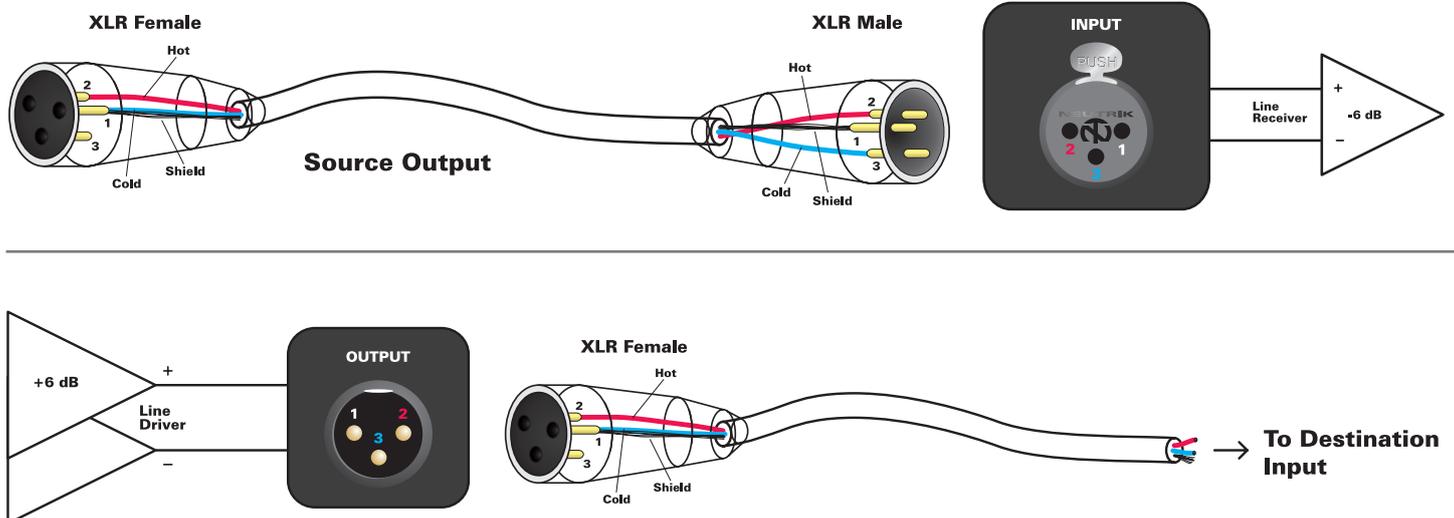
Unbalancing Control 2802

To unbalance a **Control 2802** 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 **Control 2802** line input must be left wired as balanced to preserve some of the common-mode rejection performance (CMRR) of the input stage. Please refer to the first diagram at the bottom of the page.

To unbalance an output from **Control 2802**, 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 refer to the second diagram at the bottom of the page.

Please note that you should 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 helpful.

Connection	Quantity	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 Wiring

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

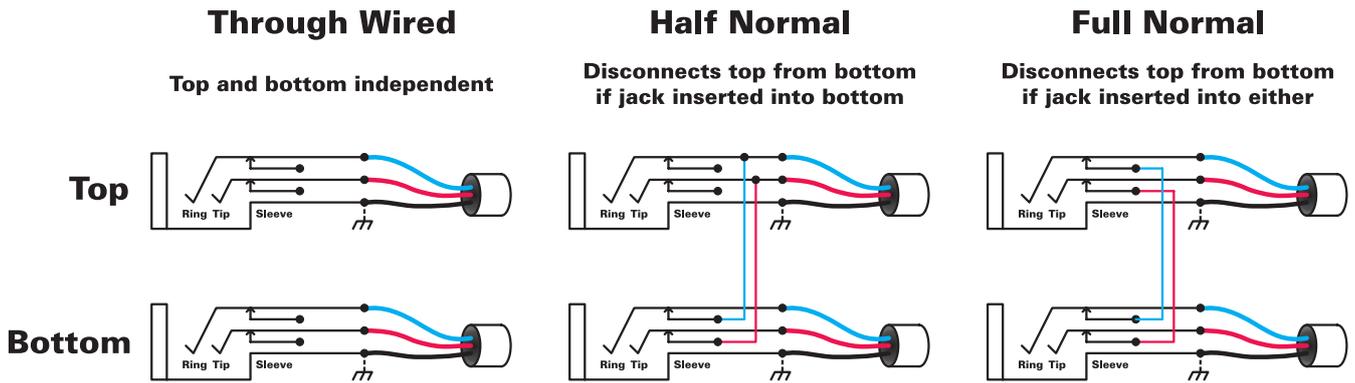
It also features a 16 channel DAW interface that provides 16 outputs for mixdown using the summing input and eight main channels. Obviously you can juggle this around and adapt the ideas for your optimal workflow and exact I/O count.

Although two bantam bays may look excessive, with a few quality pieces of outboard gear and a set of hardware synthesisers and samplers 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 **Control 2802** 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 and provide ultimate flexibility.



Panels



