# SOUNDCRAFT 3200 MIXING CONSOLE MANUAL.

**NOTE: THIS MANUAL IS PROVISIONAL.**

ISSUE: P3.

DATE: 02 AUGUST 1990.

EDITOR: GARETH CONNOR.



**NOTE:** While all efforts have been made to ensure accuracy within this manual, it is provisional, and there may be minor errors and/or omissions present within the current content.

As soon as the production issue of the 3200 Manual is completed, console users who are in posession of this issue will receive a Production Issue Manual.

Soundcraft Electronics Limited apologise for any inconvenience this may cause.

## THE SOUNDCRAFT 3200 MIXING CONSOLE.

### ABOUT THIS MANUAL.

This manual is divided into several sections. The first major part is the console description, which gives detailed information on the facilities and controls of the console from the point of view of the user. The installation and connections section gives all the information required to make the console ready for use, including lists of pin assignments for the EDAC multiway connectors. The maintenance section gives information required for routine procedures such as meter bulb replacement. The technical description gives a detailed account of the operation of every piece of circuitry in the console, and is essential reading in the event of fault-finding.

CONTENTS

## 1.00 CONSOLE FACILITIES & GENERAL DESCRIPTION.

The Soundcraft 3200 is the second of a new generation of consoles designed to meet new standards of performance. We believe that the specifications attained are unique for this class of console.

Input facilities include a full-range microphone preamplifier of advanced design (patent pending), a versatile noise gate, 4-band parametric EQ, and 12 independent aux sends. If all 32 group buses are not in use, a further 8 mono or 4 stereo aux sends may be generated on buses 25-32 by use of the **RTG** facility.
4 programmable mute buses, true solo-in-place as well as PFL and AFL are provided.

The output modules include monitor sections that have almost identical features to the inputs, including noise-gate and 4-band EQ, 12 aux sends, (as 2 stereo and 8 mono) and a 100mm fader. The group summing amplifiers use an advanced hybrid discrete/integrated configuration that reduces mixing noise significantly. The vital mix summing amps use similar but even more sophisticated low-noise technology.

The master section consists of the following modules:

1) Aux master and oscillator.
2) Effects-return and communications.
3) Control-room / Studio monitor and Phones-mix (CSP).

When only 24 outputs are installed a further module is fitted to accept the extra sends that use group buses 25-32 via the **RTG** facility. This high-group module has similiar facilities to the aux master module.

The Soundcraft 3200 has been designed for rapid but flexible operation. Master controls are provided for mic/line, tape/group, and Line A/Line B input switching, supplemented by local over-ride of each function on every module.

Global pre/post aux switching is included; the console may be changed from having 12 prefade sends to 12 postfade sends in three seconds.

The Soundcraft 3200 attains its performance by utilising innovative concepts and circuit configurations (some of which Soundcraft has patented) rather than by using expensive components, making it highly affordable despite its sophistication. Almost all of the electronic parts are industry standard and widely obtainable; it is their application and arrangement which is special.

A newly developed input preamplifier covers the complete mic gain range of 0 to 70 dB with a single control, eliminating the compromises inherent in using input attenuators. The common-mode rejection sets new standards, remaining extremely good down to low frequencies where it is most necessary. The Line gain range is from -10dB to +20dB and accepts all the usual operating levels without internal adjustment.

The Soundcraft Active Panpot is a unique system that renders panpot operation virtually perfect. The panning law is generated by law-synthesis techniques that give a much closer approach to the theoretical sine/cosine characteristic than the conventional ´law-bending´ method, giving smooth panning without level shift. The left-right isolation when panned hard over is also improved by 25dB or more.

The Soundcraft 3200 is available in both 24 and 32 bus formats; these differ mainly in the number of output modules fitted, and this manual applies to both versions.

## 2.00 PERFORMANCE SPECIFICATIONS.

This specification is provisional and subject to change without notice.

FREQUENCY RESPONSE. +0.0, -0.5 dB 20Hz- 20kHz
Any input to any output.

THD.

| | | | |
|---|---|---|---|
| Line I/P to Group O/P: | Less than .004% | ( 1kHz) |
| | Less than .02% | (10kHz) |
| Line I/P to mix O/P: | Less than .004% | ( 1kHz) |
| | Less than .02% | (10kHz) |

All measured at 20dB above nominal level.

NOISE. Mic equivalent input noise: Less than -128dBu.
(150 Ohm source res, 20kHz bandwidth, unweighted)

| | | |
|---|---|---|
| Group noise (16 ch routed): | -85dBu | (S/N -89dB) |
| Mix noise (16 ch routed): | -80dBu | (S/N -84dB) |
| Aux noise (32 ch console): | -85dBu | (S/N -89dB) |

CROSSTALK.

| | | |
|---|---|---|
| Mic to Line: | Better than -100dB | ( 1 kHz) |
| Line A to mic: | Better than -100dB | ( 1 kHz) |
| Line A to Line B: | Better than -105dB | ( 1 kHz) |
| Channel muting: | Better than -100dB | ( 1 kHz) |
| Fader kill: | Better than - 98dB | ( 1 kHz) |
| | Better than - 95dB | (10 kHz) |
| Panpot isolation: | Better than - 90dB | ( 1 kHz) |
| Aux off (pot): | Better than - 85dB | ( 1 kHz) |
| Aux off (switch): | Better than -120dB | ( 1 kHz) |
| Routing: | Better than -120dB | ( 1 kHz) |
| | -110dB | (10 kHz) |

INPUT IMPEDANCES.

| | |
|---|---|
| Mic inputs: | 2 KOhm balanced. |
| Line inputs: | 15 KOhm balanced. |
| Insert returns: | 10 KOhm balanced |

OUTPUT IMPEDANCES. Less than 75 Ohms.

OUTPUT CAPABILITY.
Balanced outputs: Not less than +26dBu into 600 Ohms.
Gnd-cancel outputs: Not less than +20dBu into 2K Ohms.

HEADROOM.
Not less than 22dB at any point in console.
Not less than 28dB at mix summing amps.
Not less than 30dB at aux summing amps.

NOISE-GATE.

| | | |
|---|---|---|
| Max atten: | Not less than -90dB | ( 1 kHz) |
| Min attack time: | Less than 40 microSeconds. | |

## 3.00 CONSOLE DESCRIPTION BY MODULE.

### 3.01 INPUT MODULE.

**ROUTING SWITCHES.**
These 32 switches route the channel signal to the desired groups, either for assignment to the desired tape tracks, or to create subgroups. In stereo operation odd group numbers are treated as Left, and even as Right. Intelligent signal-handling only drives switches 1 to 24 when the matrix is actually in use, giving significant advantages in isolation and power consumption.

In 24-output consoles, switches 25-32 are dedicated to providing extra aux sends via the **RTG** facility. (see below). The routing matrix uses the Soundcraft routing configuration combined with fully-balanced operation that gives unprecedentedly low inter-group crosstalk.

**PWR**
+48V Phantom Power switch. Pressing this switch applies +48V phantom power to a suitable capacitor microphone connected to the mic input. This voltage is supplied through two 6K8 resistors in the standard way. Accidental application of phantom power to low-impedance microphones is most unlikely to cause any damage, but care should be taken not to use it when internal-battery electret microphones or electronic DI boxes are connected.

**PHI**
Phase switch. This reverses the phase of the selected input only, to cope with incorrect wiring or microphone placement problems.

**B**
Line B switch. Switches between the two line inputs A and B. If the noise gate is being keyed from a separate external signal then this is taken from the Line B input, and this path is not affected by the line gain control, Line B or mic/line switches.

**LI**
Mic/Line Switch (non-latching). A high-impedance, high level line input is selected instead of the mic input when this switch is illuminated. This switch locally flips the mic/line status.

Mic/line switching on the Soundcraft 3200 can also be changed globally by use of the master-status pushbuttons on the console master section. Use of these operates the mic/line switching on all input modules simultaneously, for rapid mixdown setup, but the local switch on each module allows the mic/line selection to be individually toggled. All modules are pulled back into synchronisation whenever the master-status is changed. Tape-normalling to the line input means it is fed internally from the same-numbered tape-return input. This allows all the facilities of the input module to be used at mixdown with a minimum of repatching. (Note that all global switching from the Master Control Module is disabled when **RED LIGHT** is on). The mic mode input impedance has been carefully chosen so that it allows hard-wired mic-splitting (sometimes called Y-splitting) without excessively loading a typical microphone.

**GAIN**
Input gain control. A dual concentric knob gives separate control of the input gain for mic and line modes.

The mic input sensitivity is variable between 0 and -70 dBu in one turn of the control, without the need for additional pad or attenuator switches. This saves valuable panel space as well as eliminating the awkward juggling of two interdependent controls. The advanced technology used provides lower noise at intermediate gain settings, while at high gain settings noise is determined almost entirely by the microphone source impedance. An exceptionally high common-mode rejection-ratio is achieved which allows the input to ignore electrical interference, and unlike conventional systems, the CMRR remains very high down to low frequencies such as 100Hz, where it is most needed.

The gain range in line mode provides for nominal input levels between +10 and -20dBu, catering for almost every conceivable type of equipment.

**SOUNDCRAFT MICROPHONE PREAMP TECHNOLOGY IS COVERED BY PROVISIONAL PATENT No. 8719014.**

FIL
Pressing the FIL switch brings in high-cut and low-cut filter sections. Both have variable turnover frequency, and an ultimate slope of 12 dB per octave; if only one of the filter sections is required the other is removed from operation by setting it to the extreme of its frequency range. The high-cut filter efficently controls problems of HF noise or excess sibilance, while the low filter cleanly and effectively removes low-frequency rumble and mic-proximity disturbances. Both filters have a Butterworth maximally-flat characteristic with a roll-off commencing at 100Hz, and an ultimate slope of 12dB per octave.

When the filter section has been switched into the noise- gate sidechain, it is not available and this button has no effect. To underline this condition its LED is held off despite the mechanical position of the switch.

**NOISE GATE**
This gives comprehensive gating facilities. An external keying signal may be selected to drive the sidechain, and the high and low filters may be switched in to modify this control signal and not the main audio path. There is a fixed Hold time of 10mS. This stops gate dither when using fast attack and decay times.

**THRESHOLD.** Varies the gate on/off switching point between -60 and +10 dB (ref nominal internal level).

**DEPTH** control. Sets the amount of attenuation when the gate is off. (ie muted) Variable between 0 and -90 dB.

**ATTACK.** Sets the rate at which gain increases when the threshold is exceeded. Variable between 50 microseconds and 200 milliseconds.

**DECAY**. Sets the rate at which gain falls when the input falls below the threshold. Variable between 0.5 mS and 1 second.

**GATE**
Switches in the noise-gate. Gate circuitry is switched out of circuit and its drive removed when not in use.

**EXT**
Switches an external keying signal into the gate sidechain, instead of using the main signal. Input is via Line B, regardless of the state of input selection. The line gain control does not operate in this case, as the external signal is taken off before it. The threshold control provides all the gain control that is necessary.

**FIL** (to sidechain). Transfers the filter section out of the audio path and into the gate sidechain. The filter section is automatically made active, and it is not necessary to press in the FIL button next to the frequency controls. The filter section does not transfer if the gate is not switched into use.

**MUTE** This LED lights when the gate is muted (closed). It is always off when the gate section is not in use.

**CHANNEL INSERT.**
An insert point operating at a nominal level of -2dBu is normally placed after the EQ, and just before the electronic mute in the signal path. It is switched in/out by use of the **IN** button. The **PRE** button just above moves the insert to before the EQ section when in.

The insert has a ground-cancelling send amplifier and a fully electronically-balanced return. The ground-cancelling output improves on an unbalanced output by sensing the ground potential of the remote equipment, and adding it to the console output. Thus the remote equipment does not 'see' any unwanted signals on its ground, as its input is moving in exactly the same way.

**EQ SECTION.**
This section incorporates sweepable HF and LF controls plus two fully parametric middle sections, giving precise control over four sections of the audio spectrum, either for the correction of deficiencies in the material, or to create individual effects. All boost/cut controls are centre-detented for rapid re-zeroing.

**HF.** The high frequency control provides 15dB of boost or cut with a shelving characteristic; in other words having reached maximum boost or cut the EQ curve stays at that level. Frequency of operation is variable between 1.2 and 20 kHz.

**HI-MID.** A parametric mid equaliser variable between 700Hz and 11kHz. The response is of the bell-curve or peaking type, with up to 15db of boost or cut available. The Q is fully variable between 0.7 and 5.

**LO-MID.** As for hi-mid, except that the frequency range is 70Hz to 1100Hz. The two mid controls overlap in the centre of the audio band to allow subtle and complex effects to be created.

**LF.** The low frequency control gives up to 15dB of boost/cut. Frequency of operation is variable between 20Hz and 320Hz. The characteristic is switchable between peaking and shelving, using the **PK** switch.

**EQ** switch.
This switches the the entire EQ section in and out, but does not affect the low-cut filter. This allows instant comparison with the unmodified signal. When the EQ is switched out, it is not only entirely removed from the signal path, but the drive to it is also removed, to reduce ground currents and power dissipation to a minimum.

**AUXILIARY SENDS.**
These have two main functions; providing foldback so that musicians can monitor their own performance, and giving a feed to external effects units. In both cases the feed may be taken from one module, or it may be a mix of any number of sources. Foldback sends are normally prefade (not affected by fader setting) while effects sends, and in particular echo/reverb sends, are almost always postfade so that the effect level fades down with the main signal.

All auxes on the Soundcraft 3200 have global prefade/postfade switching controlled from the master section, so the status of a send may be changed without pushing 60 or more buttons spread across the console. For unusual applications requiring a mixture of pre and post feeds to one aux bus, the 5-6 pair have a local status reverse switch, **REV**.

**STEREO A.**
The inner knob controls level, and the outer panning. The centre-drop is 4.5dB. The associated **ON** switch removes the mix resistors from the buses when set to off, to minimise noise and crosstalk.

**STEREO B.**
As Stereo-A.

**AUX 1 & 2.**
Two mono sends switched on/off as a pair by the associated **ON** button. They are also switched pre/post as a pair, from the master control section.

**AUX 3 & 4.**
As Auxes 1 & 2.

**AUX 5 & 6.**
As Auxes 1 & 2, except that a **REV** button reverses the pre/post status of the pair on that module only.

**AUX 7 & 8.**
As Auxes 1 & 2, except for the provision of the **RTG** switch. This allows the top 8 group buses to be used as extra postfade sends when not otherwise required.

When **RTG** is pressed in, this removes the feed from aux buses 7 & 8 and transfers it to routing matrix switches 25 to 32. The feed to the control pair is now taken from after the panpot, so that it is possible to make up extra stereo aux sends, using say Group 27 (L) and Gp 28 (R). Alternatively the high group buses may be used as 8 extra mono sends. The **ON** switch operates as before.

Sends created using **RTG** are always postfade. Auxes 7 & 8 are controlled by the global pre/postfade switching as before, and are accessed as usual by other modules not in RTG mode.

**PEAK LED.**
This illuminates when the channel signal exceeds +14dBu, and is approx 8dB below clipping. The signal is monitored at three positions: pre-EQ, post-EQ, and at the output of the fader post-amp. Both positive and negative peaks are detected, and there is a fast-attack/slow-decay action to give clear indication of short peaks.

**PANPOT.**
This is a centre-detented control that places the module signal left or right in the stereo field. The Soundcraft Active panpot uses unique circuitry (patent pending) that allows the panning law to be much closer to the theoretically correct sine/cosine characteristic than conventional panpots, and also provides much greater left/right isolation; it yields an improvement over the usual techniques of at least 25dB. The centre drop is set to 4.5dB as a compromise between stereo and mono operation.

**SOUNDCRAFT ACTIVE PANPOT TECHNOLOGY IS COVERED BY PROVISIONAL PATENT No. 8800168.**

**PAN** switch.
Brings the panpot into circuit. When not pressed, full level is sent to both left and right regardless of the panpot setting.

**MIX** switch.
Routes the input module directly to the stereo mix bus.

**ON/CUT** button.
A large illuminated switch, carrying the number of the input module. This switches on and off signals passing through the module, using a silent FET switch positioned just before the fader in the signal path. The signal is in fact rapidly faded up and down (over about 10 milliseconds) and this prevents the generation of clicks when the channel is turned on and off with signal passing through it. The switch may be individually configured for **ON** (illuminated when not muted) or **CUT** (illuminated when muted) operation on each module. The on/off state adopted on console power-up can also be individually set. For details see the Technical Description (Section 7.01) or Configuration (Section 4.04). Channel on/off status is controlled not only by the action of the **ON** switch, but also by the programmable mute buses and the Solo-In-Place facility. In all cases the true channel status is shown by the ON/CUT button.

**MUTE BUS SWITCHES. (A,B,C,D)**
When one of these is pressed the input is connected to the appropriate muting bus. When the master mute switch is operated all inputs and outputs thus connected are muted, regardless of the ON/CUT switch status. Releasing the master mute switch leaves the input in its original state. If more than one mute bus is assigned to the input, all will mute it; in other words the switches are OR-ed together.

**SOLO** button.
This large illuminated switch activates PFL, AFL, or Solo-In-Place (SIP) operation, depending on the position of a mode switch on the CSP module.

**PFL** mode routes the prefade module signal directly to the control-room outputs, summed with the prefade signals of any other modules with their **SOLO** buttons pressed. The only change in signal flow is the switching of the control-room outputs, and so this mode may be used during recording or live operation for quality checking without affecting the main console outputs.

**AFL** operates in the same way, but uses the postfade signal, allowing a greater sense of relative signal levels to be maintained.

SIP mode operates by muting all channels that do not have their SOLO buttons pressed. The channel contributions therefore appear in the correct stereo positions and at the correct relative level, with echo etc, providing the effects return facilities have been arranged not to respond to SIP muting.

SIP is sometimes called a 'destructive solo' in the sense that normal signal flow to the main console outputs is disrupted when it is used. SIP mutes are disabled when **RED LIGHT** is in force. (See CSP Module, Section 3.05).

**SFE** switch.
When input modules are being used as effect returns etc, this switch prevents the module from responding to SIP mutes.

**INPUT FADER.**
This is a long-throw unit with a carefully optimised control law and feel. The total travel is 100 millimetres.

**INPUT METERING.**
If input metering is fitted to the console, the metering take-off point is pre-fader, and post-EQ. It may be configured to be either pre or post the input mute. For details see the Technical Description section of this manual.

**3.02 OUTPUT MODULE.**
This consists of two sections that can operate either together (when recording) or independently (at mixdown).

The monitor section is used either to simulate the mix that will later be made from the tape track, or as another input in its own right. It has all the facilities of the input module, (arranged in the exactly the same way so that the console is as easy to use as possible) with the exception of the mic input, the insert, and the routing matrix. Only the differences are described below; if in doubt consult the input module description in Section 3.01.

The group section combines all the signals routed to its bus, controls the level of the combined signal with the group fader, provides the low-impedance balanced output, and allows routing to the stereo mix bus.

**MONITOR SECTION.**

There are three possible inputs to the monitor section; the group signal, Line A, and Line B. Line A is normalled to the main multitrack tape return, and Line B will be used for either a second multitrack, or electronic keyboards etc. The large buttons at the bottom of the module switch between **GROUP** and **TAPE**, the latter meaning either Line A or Line B, as selected. Both GROUP/TAPE and Line A/Line B switching are performed by silent electronic switching (in the form of a 10 millisecond crossfade) and all outputs may be switched at once by use of the master-control buttons. Local reversals of status are cancelled by a master change, so that all modules are ´pulled into line´.

The desired power-up states can be programmed on the master-control module. See Section 4.04.

**B**

The non-latching Line A/Line B switch provides the local line input selection. The switch LED is ON for Line B.

**PHI**

The phase-reverse switch for Line A/Line B inputs. It does not affect the feed from the group to the monitor section.

**LINE INPUT GAIN**

This dual-concentric control sets input gain independently for Line A (inner knob) and Line B (outer). The input sensitivity range is from +10 to -20dBu, as for input module line gain.

**FILTERS**

Exactly as for input module.

**NOISE GATE**

Exactly as for input module. The EXT input is always taken from Line B, regardless of the monitor path input selection.

**EQ SECTION**

Exactly as for input module.

**STEREO-A, STEREO-B, AUXES 1 to 6**

Exactly as for input module.

**AUXES 7 & 8**

These differ from those on the inputs because there is no RTG facility- there is, after all, no routing matrix to allow its use. These controls always feed aux buses 7 & 8.

**PAN** switch.
The aux 7-8 panpot switch. Normally Auxes 7-8 are fed from the usual pre/post feeds, but when this switch is pressed in the two level controls are fed by the monitor panpot, giving a postfade stereo send to aux buses 7 & 8. Aux 7 is Left, and Aux 8 is Right. Unless the two concentric knobs are set to the same level, the Aux 7-8 stereo positioning will be offset relative to the monitor panpot.

**PEAK LED**
This illuminates when the monitor signal exceeds +14dBu, and is approx 8dB below clipping. The signal is monitored at three positions: pre-EQ, post-EQ, and at the output of the monitor fader post-amp. Both positive and negative peaks are detected, and there is a fast-attack/slow-decay action to give clear indication of short peaks.

**MONITOR PAN**
This is a centre-detented panpot that normally places the monitor signal left or right on the stereo mix bus. The Soundcraft Active panpot uses unique circuitry (patent pending) that allows the panning law to be much closer to the theoretically correct sine/cosine characteristic than conventional panpots, and also provides much greater left/right isolation; it yields an improvement over the usual techniques of at least 25dB. The centre drop is set to 4.5dB as a compromise between stereo and mono operation.

**SOUNDCRAFT ACTIVE PANPOT TECHNOLOGY IS COVERED BY PROVISIONAL PATENT No. 8800168.**

**GRP**
This switch removes the monitor pan signal from the mix bus and routes it to the associated pair of groups, ie the group on the same module plus the adjacent one. Odd-numbered groups are left, and even groups right. Therefore if Output 5 has GRP pressed, its monitor section will pan between Group 5 (L) and Group 6 (R). Likewise, the monitor on Output 6 will pan between groups 5 and 6. The monitor signal is injected into the group path just after the group insert and just before the group fader. It is mixed with the existing group signal, which is unaffected in level.

When this switch is active, there is a danger that a feedback loop may be created in the module if the monitor source is selected to GROUP. To prevent this, the **GRP** switch forces the monitor input select into TAPE mode only. The module will not respond to an attempt to switch it into either GROUP or TAPE+GROUP mode, either locally or globally.

**MUTE BUS SWITCHES. (A,B,C,D)**
Control the monitor ON/CUT status, as for the input module. If more than one is pressed they are OR-ed together.

**SFE**
Prevents the monitor mute from responding to Solo-In- Place mutes. Useful if the monitor section is being used as an effects return at mixdown.

**TAPE**
A large illuminated button that locally selects the multitrack tape return to be the monitor path input. This is Line A or Line B, as selected by the **'B'** switch. (see above) This may be set globally by using the master-control module.

**GROUP**
A large illuminated button that locally selects the group signal as the monitor input. This may be set globally by using the master-control module.

**TAPE+GROUP** modes.
When performing tape drop-ins, it is useful to be able to hear both TAPE (replay of existing tape signal) and GROUP (artist's run-up to the actual material to be dropped in). To enter this mode press both **TAPE** and **GROUP** buttons simultaneously. Both will light to indicate that you are hearing a mix of the two signals. The meter will read GROUP only, as this is the signal of interest. To leave **TAPE+GROUP** mode, press either button by itself.

When the tape machine is dropped into record, its own internal switching will replace the off-tape signal with its input, and so the monitor will be receiving the GROUP signal twice.

**MODE CONTROL FROM TAPE MACHINE.**
If a master-record logic line is available from the multitrack, this may be connected to the Soundcraft 3200 via an internal opto-isolator. On entering record, **TAPE+GROUP** will switch automatically to monitor **TAPE** only. The console metering will still be reading group, and the **GROUP** button will flash to indicate this. When the master-record line signals that recording has ended, the monitor will switch back to the normal **TAPE** mode.

**ON/CUT** button.
Operates on the monitor path, as for the input module.

**SOLO** button.
A large illuminated button that initiates a solo condition for the monitor path. This may be PFL, AFL, or Solo-In- Place, as selected on the CSP module.

**MONITOR FADER.**
A high-quality long-travel (100mm) unit that controls the monitor signal level. It is situated on the upper module of the group/fader module pair.

## GROUP SECTION.

The group summing amplifier is a proprietary Soundcraft design. It uses discrete transistors to achieve the minimum possible noise levels, and its inherent symmetry exploits to the full the noise and crosstalk benefits possible with balanced mixing.

**MIX**
This switch routes the postfade group signal directly to the stereo mix bus, without passing through the monitor section.

**GROUP PAN**
This control pans the group signal left/right on the stereo mix bus when **MIX** (above) is pressed in.

**PFL**
This switch activates the prefade listen mode, adding the group signal to the PFL/AFL bus. The PFL switches on the group do not activate the Solo-In-Place mode even if this is selected for the input modules, but instead give an over-riding PFL or AFL condition, depending on the setting of the master PFL/AFL switch on the CSP module. The PFL takeoff point is before the **ON** switch, and the AFL takeoff immediately post the monitor fader.

**FDRS SWAP**
The fader-swap switch. Exchanges the function of monitor and group faders, for operational convenience. All other controls are unaffected. The Monitor fader is on the upper module (Monitor), and the group fader is on the lower (Fader) module.

**A,B,C,D, & SFE**
These mute-bus switches operate on the monitor section only, control the monitor ON/CUT status, as for the input module. If more than one is pressed they are OR-ed together.

**GROUP FADER**
A high-quality long-travel (100mm) unit that controls the group signal level, being placed between the summing amplifier and the output stage.

**GROUP OUTPUT**
This is electronically balanced, with a self- compensating function so that the output simulates the action of a centre-tapped transformer winding. In other words, when one leg of the output is grounded, to give an unbalanced feed, the other leg provides a level increased by 6dB so that the overall level is unchanged. The output will drive up to +26dBu into a 600 Ohms.

**GROUP INSERT**
This insert has a ground-cancelling send and an electronically balanced return, operating at a nominal level of -2dBu. It is positioned before the group fader in the signal path.

**OUTPUT METERING**
The meter source is normally the same as the monitor section, whether this is set to **GROUP** or **TAPE**. However, in **TAPE+GROUP** mode the group path only is metered directly.

## 3.03 AUX MASTER MODULE.

This consists of the aligment oscillator and the aux master sections for Stereo-A, Stereo-B, and Auxes 1 to 8. If more aux sends have been created by use of RTG then these are dealt with either by the relevant groups (32-output consoles) or the high-groups module. (24-output consoles).

### OSCILLATOR SECTION.

This is a sophisticated sinewave oscillator for alignment purposes. Sixteen fixed frequencies are provided, covering the whole audio spectrum, and including all popular line-up frequencies.

The oscillator may be routed to all groups and all auxes, plus the stereo mix bus and a pair of dedicated patchbay outputs. The oscillator is specially optimised for minimal level change with frequency and fast settling.

**NB.** Both LISTEN and TALKBACK modes also access some of the same destinations, and both are given higher routing priority than the oscillator; see below for details. The oscillator is also completely disabled by a **RED LIGHT** condition.

**2TRK**
This switch routes the oscillator to the stereo mix bus. If talkback is being routed to this bus then it takes priority, and the oscillator routing is suppressed. **LISTEN** does not route to mix, and therefore has no effect.

**MTRK**
This switch routes the oscillator to all group buses. This includes group buses being used as auxes (via **RTG**) in a 24-output console.

If **LISTEN** is being routed to the group buses (by **LISTEN->TAPE**) then it takes priority over the oscillator, which is suppressed. Talkback takes priority over both.

**AUX**
Routes the oscillator to: Stereo-A, Stereo-B, Auxes 1 to 8.

**ON**
Switches on the oscillator. Unless one of the above routing switches is pressed, then the oscillator will only be sent to the patchbay output. This output is electronically balanced and 600 Ohm-capable, and simulates a floating transformer winding. When set to off, the oscillator is completely disabled to eliminate any possibility of signal leakage.

**CAL**
The 'calibrated' switch. It disables the normal oscillator level knob, and gives an output level determined only by the multi-turn preset which is accessible just below the **CAL** switch. The preset turns clockwise to increase level, in the usual way.

**LEVEL**
Oscillator output level control. Affects all oscillator outputs.

**FREQUENCY-SELECT** buttons.
Eight mechanically-interlocked switches that select the basic oscillator frequency. If more than one button is pressed in accidentally the output frequency will be subject to errors.

**X10** button.
When pressed the oscillator frequency is multiplied by ten, allowing the whole range 20Hz-20kHz to be covered.

**OSC LEVEL**
Controls oscillator output level.

## AUX MASTER SECTIONS.

All of the Soundcraft 3200 aux masters make use of advanced Negative-Impedance Converter (NIC) technology to provide enhanced headroom without any compromise on noise levels. The summing amplifiers use discrete low-noise devices to reach the lowest noise levels possible. The inherent circuit symmetry enhances the rejection of crosstalk and other interfering signals.

All aux outputs are electronically balanced and 600 Ohm- capable, and simulate a floating transformer winding.

When aux metering is fitted, the metering point is the output of the auxes, ie after the **ON** switch and level control.

STEREO-A & STEREO-B.
Both sections are identical.

**LEVEL**
A ganged rotary control that sets the stereo aux master gain. Both summing-amp gain and postamp gain are varied to maintain optimum noise/headroom conditions, using NIC technology. This means that if the aux master is receiving excessive level, it is normally possible to simply turn down the aux master, without fear of the aux summing amp clipping. On conventional consoles it may sometimes be necessary to turn down all the contributing sends, while attempting to maintain the relative balance between them.

When only one source is routed, the unity-gain position is at the 7 mark.

**ON**
Enables the aux master, allowing the signal to reach the output.

**AFL**
This button gives access to the combined aux signal after the level control, and routes it to the solo system. The mode of operation is AFL regardless of the position of the master PFL/AFL control. AFL is not disabled by the **ON** switch, and is presented as a mono mix of the stereo aux.

**PRE**
This button sets the pre/post-fade status of its associated aux send for the entire console. This operation is not guaranteed to be totally silent; it is not advisable to make such drastic changes while recording.

**AUXES 1 to 8**
These are all identical, and very similiar to the stereo aux masters described above. The only difference is that the master pre/post-fade switching operates in pairs.

It should be remembered that the local pre/post reverse on auxes 5-6 (the **REV** button) is not cancelled by operation of the master control, unlike mic/line master switching, for example.

## 3.04 EFFECTS RETURN/COMMUNICATIONS MODULE.
This module consists of four identical stereo effects returns, and controls for the listen mic and talkback mic routing.

### EFFECTS RETURN SECTION.
Each effects return has two-band EQ of the Baxandall type, a prefade aux send selectable to either of the stereo aux buses, on/off and PFL switches, and a rotary fader to control the return level to the mix bus.

**HF**
Shelving characteristic.
Provides up to 15dB of boost or cut at 10kHz.

**LF**
Shelving characteristic.
Provides up to 15dB of boost or cut at 50Hz

**ST-A**
Prefade aux send control, normally to the Stereo-A aux bus.

**ST-B**
This switch when pressed in routes the aux send to Stereo-B instead.

**ON**
Enables the effects return signal path to mix. It does not affect the prefade aux send.

**PFL**
Routes the prefade signal to the solo system. The signal is always prefade, and is not affected by the PFL/AFL master switch on the CSP module.

**FADER**
A ganged rotary control setting the level of effect returned to the stereo mix bus. Advanced active-gain-control technology is used to maximise headroom, minimise noise, and enhance stereo balance.

## LISTEN SYSTEM.

The LISTEN system provides an extra mic input that can be routed either to the control-room monitors, for communication with the studio etc, or routed to the group buses for recording count-ins and so on. The LISTEN mic is often hung from the ceiling to give coverage over the whole studio.

**LISTEN**

This rotary control sets the listen mic amp gain, over the range +20 to +70dB.

**LISTEN->CRM**

Routes the LISTEN signal to the monitor speakers. Any feed already going to the monitors is dimmed by a fixed amount of 20dB. Note that this is the only fixed dim ratio, and all other dim conditions are controlled by the **DIM RATIO** knob on the crm/phones module. (See section 3.05 below).

Use of any talkback switch disables **LISTEN->CRM** to prevent acoustic feedback. (howlround)

**LISTEN->TAPE**

Routes the LISTEN signal to all 32 group buses. This is disabled by **RED LIGHT** mode.

## TALKBACK SYSTEM.

The talkback system provides communication from the console operator to all other parts of the installation. The talkback mic is installed in the meter bridge, and feeds a low-impedance balanced preamp suitable for 200 Ohm mics, providing the same high quality as the mic preamps on the input modules, though with a more restricted gain-range. The talkback routing switches are all momentary in action; any number may be pressed at once.

All except those routing to the phones system are disabled in **RED LIGHT** mode. Whenever **LISTEN** and talkback are routed to the same destination, talkback takes priority. Both **LISTEN** and talkback take priority over the oscillator.

Use of any talkback switch dims the control-room monitors and disables the **LISTEN->CRM** facility.

**TALKBACK**

The rotary talkback level control. Sets the gain of the talkback mic stage, over a range of +20dB to +70dB.

**M/TRK**

Routes the talkback signal to all 32 buses. This includes consoles where only 24 outputs are fitted.

**2TRK**

Routes talkback to the stereo mix bus.

**EXT 1,2**

Routes talkback to the relevant external output. These outputs are intended for connection to intercom systems, etc, and are ground-cancelling to prevent the formation of hum-loops.

**PHNS 1,2,3,4**
Routes talkback to the relevant phones mixer, where it is injected just before the phones level control.

**STUDIO**
Routes talkback to the studio speaker feed. It is injected after the studio level control, and before the studio **CUT** switch.

**BOOTH**
Routes talkback to an external output. This is intended for an isolation booth, but can be used for any purpose. The output is ground-cancelling.

**ALL**
Pressing this is equivalent to pressing all talkback buttons, and talkback is routed to all destinations.

**RED LIGHT**
Pressing this puts the console into the **RED LIGHT** safety mode. This state disables certain of the console facilities to minimise the chance of accidental disruption of recording. Its functions are as follows:

*Disable master status switches for Mic/Line, Group/Tape, and Line-A/Line-B. Master mutes and master aux pre/post switching are not disabled.

*Disable the alignment oscillator.

*Prevent the **LISTEN** mic being routed to the group buses by **LISTEN->TAPE**.

*Disable all talkback except that to phones mixers.

*Prevents Solo-In-Place from muting any signal path.

*Provides an opto-isolated output that can be used, via an external relay, to switch the studio "RECORDING" red light.

## 3.05 CONTROL-ROOM / STUDIO / PHONES (CSP) MODULE.

This consists of:

*Stereo mix bus summing system.

*Four independent mixers for generating headphone feeds.

*Studio loudspeaker source selection and control

*Control-room monitor speaker source selection and control.

*Solo mode control.

### STEREO MIX SYSTEM.

**MIX MASTER FADER**
This is a dual long-travel unit that controls the stereo mix output level. It is placed between the summing amplifiers and the output stage in the signal path. It is mounted on the fader panel directly below the control-room/phones module.

**MONO MASTER FADER**
A high-quality long-travel unit that controls the output level of the mono sum of the stereo mix outputs. It is mounted on the fader panel directly below the effects return/comms module.

**MIX INSERT POINTS**
These have a ground-cancelling send, a balanced return, and operate at the standard level of -10 dBv (-7.8 dBu). They are placed immediately after the summing amps and just before the mix master fader in the signal path.

**MIX OUTPUTS**
These are electronically balanced, with a self- compensating function so that the output simulates the action of a centre-tapped transformer winding. In other words, when one leg of the output is grounded, to give an unbalanced feed, the other leg provides a level increase of 6dB so that the overall level is unchanged. The output will drive up to +26dBu into a 600 Ohms.

### HEADPHONE MIXERS.

Four separate headphone mixes can be generated. These will normally consist of the source selected to the studio feed (see below) plus an auxiliary send. The four mixers are identical.

**A**
This rotary control normally sets the level of Stereo-A Left added to the headphone mix left channel. The feed to this control is routed via the patchbay so that other signals may be substituted.

**B**
This rotary control normally sets the level of Stereo-A Right added to the right channel of the headphone mix. The feed to this control is routed via the patchbay so that other signals may be substituted.

**MNO**
The mono mode switch. When pressed **A** and **B** above feed both channels of the headphone mix. Therefore separate signals can be added to the mix by suitable patching.

**STUDIO**
A rotary control setting the amount of studio-feed signal added to the headphone mix. The signal is that selected by the studio source-select switches. The feed is taken off before the studio speaker level control, and so is not affected by it.

**LEVEL**
A rotary control that sets the final output level of the headphone mix.

**PFL**
This switch routes the pre-level-control headphone mix to the solo system. (in mono).

## STUDIO SPEAKER CONTROL.

**STUDIO SOURCE SELECT**
These switches select the required stereo feed to the studio speaker system. The source may be the stereo mix bus, one of three two-track returns, cassette replay, compact-disc player, or one of two uncommitted external inputs. Except for the mix bus, all switch positions simply access external inputs; they are named for convenience but can of course be used for any purpose.

All of these external inputs are balanced. The sensitivity of all except the cassette input is +4dBu nominal. The cassette input sensitivity is -10dBv (-7.8dBu) nominal.

**STUDIO**
This rotary level control adjusts the feed from the Studio Source-Select switches described above to the Studio speaker output. This output is intended for direct connection to a power amplifier driving studio loudspeakers, so that 'takes' can be replayed to musicians at once without repatching. The studio output is ground-cancelling to prevent hum-loop formation, and is not affected by solo operations. It is also unaffected by the **MONO** and **DIM** switches. (See below).

**CUT**
This switch allows the studio feed to be muted at once to prevent acoustic feedback, or for recording, without losing the desired level set up on the Studio volume control.

## CONTROL-ROOM MONITOR SPEAKER SYSTEM.

**CONTROL-ROOM SOURCE SELECT**
These switches select the required stereo feed to the control-room monitor system. The source may be selected from the same options as the Studio feed, ie the stereo mix bus, one of three two-track returns, cassette replay, compact-disc player, or one of two uncommitted external inputs. Except for the mix bus, all switch positions simply access external inputs; they are named for convenience but can of course be used for any purpose.

All of these external inputs are balanced. The sensitivity of all except the cassette input is +4dBu nominal. The cassette input sensitivity is -10dBv (-7.8dBu) nominal.

**DIM** switch.
Reduces the control-room level by between 3dB and 20dB (as set by the **DIM RATIO** control) to allow telephone answering, etc, without altering the level set on the main volume control. **DIM** condition is automatically initiated when any talkback button is pressed.

The **LISTEN->CRM** facility introduces a fixed 20dB pad into the main monitor signal when it is used.

**DIM RATIO**
Controls the amount by which the monitors are attenuated when in **DIM** mode. Variable from 3dB to 20dB.

**MNO**
The mono switch. Sums together the L and R control-room signals for mono compatibility checking, (eg for AM radio broadcasts in mono). Signals recorded with incorrect phase on one channel will tend to disappear in mono mode. No other output is affected.

**MAIN, ALT, NFS**
These three interlocking switches direct the monitor signal to the required set of loudspeakers. These may be the MAIN monitors, the second (ALTernate) speaker outputs, or the third NFS outputs. (Near Field Speakers, usually mounted on the meter bridge).

**ALT VOL**
A rotary control that sets the level going to the Alternate speakers, or to the Near Field Speakers when **NFS** is pressed.

**MAIN VOL**
This rotary control sets the level fed from the control-room select switches (or PFL bus in PFL mode) to the monitoring power amplifiers. The CRM (control room) outputs are fully balanced, as for the main mix outputs, and have the capability to drive loads down to 600 Ohms.

**CUT**
This switch mutes all monitor outputs.

## SOLO SYSTEM CONTROL.

**AFL**
This master switch selects PFL or AFL mode when the solo system is not set up for Solo-In-Place. Note that some modules have solo switches marked 'PFL' or 'AFL' and these always supply that signal regardless of the master switch setting.

SIP
SOLO *mode* switch. When this is depressed, the input module Solo buttons activate the Solo-In-Place mode rather than PFL. Buttons marked PFL (eg, on the output modules) and AFL (on the aux masters) do not change in function.

When Solo-In-Place mode is active, all modules are muted except those that have their solo buttons pressed, and those which are 'safed' (by use of the **SFE** button) and so will not mute. This is sometimes called 'destructive solo' as signals are actually removed from the mix. This mode is not normally used when recording, and is locked out by the **RED LIGHT** condition.

In the normal PFL/AFL mode, pressing a solo button causes the selected module to send either its prefade or after-fade (postfade) signal to a separate summing system. The combined signal, in mono, of as many modules are as soloed replaces the main monitor signal, and the master meters are also switched to meter the PFL/AFL signal.

**PFL/AFL TO NFS**
When pressed in this switch alters solo operation such that a PFL/AFL condition causes the monitor signal, if selected to MAIN or ALT speakers, to dim by whatever amount is preset, while the PFL/AFL signal is presented on the near-field (NFS) speakers. If the monitor signal is already selected to NFS, then the PFL/AFL completely replaces it.

**SOLO ON LED**
This LED flashes when a SOLO, PFL or AFL button is pressed anywhere on the console, indicating that a solo mode exists.

**MASTER METERING**
This is by two large VU meters. The metering point is just after the PFL switching and before the MAIN volume control. In PFL/AFL mode these meters both read the PFL/AFL bus level, so that levels almost anywhere in the console may be checked. Meter operation is not affected by speaker selection or use of **PFL/AFL TO NFS**.

## 3.06 MASTER-CONTROL MODULE.

**MIC/LINE MASTER SWITCH**
Switches the mic/line input status of all input modules. Any local status reversals are cancelled. This control is disabled by **RED LIGHT** mode.

**GROUP/TAPE MASTER SWITCH**
Switches the group/tape monitoring status of all output modules. Any local status reversals are cancelled. This control is disabled by **RED LIGHT**.

**LINE-A/B MASTER SWITCH**
Switches the Line-A/Line-B input status of all output modules. Any local status reversals are cancelled. This control is disabled by **RED LIGHT**.

**MUTE SWITCHES A,B,C,D**
The mute master switches. Pressing one of these activates the relevant mute bus and mutes all inputs and outputs that have the same-lettered mute-bus buttons depressed.

## 3.07 HIGH-GROUPS MODULE.
When only 24 output modules are fitted, the high-groups module is required to deal with group buses 25-32, so that they can be used as extra aux sends via the **RTG** button. The module carries eight identical sections that duplicate the functions of the aux masters, except that there is no pre/post status switching. Aux sends created by **RTG** are always postfade, in the same way that routed groups are normally postfade.

**LEVEL**
This rotary control sets the overall level of the combined group signal.

**ON**
Enables the group send.

**AFL**
Routes the group signal to the solo system. It is always taken post the level control, regardless of the position of the master **PFL/AFL** switch. The AFL feed is not disabled by the **ON** button.

**OUTPUTS**
The high-group outputs are fully balanced, simulating a floating transformer winding, and have the capability to drive loads down to 600 Ohms.

**METERING**
The metering take-off point is at the module output, after the level control and ON switch.

## 3.08 PATCHBAY.

The patchbay allows cross-connection of the permanent wiring (both internal and external to the console) at selected points in the system.

There are 3 seperate module sections to the patchbay:

| | |
|---|---|
| TOP: | Channels & Groups 1-24, |
| CENTRE: | Channels & Groups 25-32, Channels 33-48, Masters. NOTE: Facilities differ between Group (32 groups) and High-Group (24 groups + 8 Hi-Groups) consoles. |
| BOTTOM: | Tie Lines 1-160. |

**TOP MODULE: CHANNELS & GROUPS 1-24.**
24 identical vertical positions, each with 14 bantam jack sockets make up the top section of the patchbay. For ease of socket location, use is made of coloured zones, arranged as 3 blocks of 8 channel/group positions; 1-8, 9-16, 17-24. From top to bottom, the jack functions are:

1 **CHANNEL LINE INPUT A.**
Normalled from Tape Return A.
Configured as a break jack:
When a plug is inserted into it, the signal on the plug overrides the normal signal from the tape return.

| | |
|---|---|
| Input type: | Electronically balanced line. |
| Input level is: | +4dBu. |
| Input impedance is: | 15k ohms. |
| Input headroom is: | +32dBu. |

2 **CHANNEL LINE INPUT B.**
As 1 above, but connections are to and from B Inputs and Tape Returns.

3 **CHANNEL INSERT SEND.**
Normalled to the Insert Return.
Non-break jack: The signal is always available.

| | |
|---|---|
| Output type: | Ground-cancelling line. |
| Output level is: | -2dBu. |
| Output impedance is: | <75 ohms. |
| Output headroom is: | +20dBu into 2k ohms. |

4 **CHANNEL INSERT RETURN.**
Normalled from the Insert Send.
Configured as a break jack.

| | |
|---|---|
| Input type: | Electronically balanced line. |
| Input level is: | -2dBu. |
| Input impedance is: | 10k ohms. |
| Input headroom is: | +26dBu. |

5 **CHANNEL LINE OUTPUT.**
Not available on EDAC connectors.

| | |
|---|---|
| Output type: | Unbalanced line. |
| Output level is: | -2dBu. |
| Output impedance is: | <75 ohms. |
| Output headroom is: | +20dBu into 2k ohms. |

6 **TAPE RERTURN A.**
External interface via EDAC connectors.
Balanced line level input connected directly to:
Output of tape machine A.
Channel Line Input A.
Normalled to Monitor Input A.
Non-break jack.

7 **MONITOR INPUT A.**
Normalled from Tape Return A.
Configured as a break jack.

| | |
|---|---|
| Input type: | Electronically balanced line. |
| Input level is: | +4dBu. |
| Input impedance is: | 15k ohms. |
| Input headroom is: | +32dBu. |

8 **TAPE RETURN B.**
See 6 above.

9 **MONITOR INPUT B.**
See 7 above.

10 **GROUP INSERT SEND.**
See 3 above.

11 **GROUP INSERT RETURN.**
See 4 above.

12 **GROUP OUTPUT.**
Non-break jack.
Normalled to:
Tape Send A.
Tape Send B.

| | |
|---|---|
| Output type: | Electronically Balanced line. |
| Output level is: | +4dBu. |
| Output impedance is: | <75 ohms. |
| Output headroom is: | +26dBu into 600 ohms. |

13 **TAPE SEND A.**
External interface via EDAC connectors.
Balanced line output normalled from the Group Output.
Break jack.

14 **TAPE SEND B.**
See 13 above.

**CENTRE MODULE:**

24 vertical strips of 15 bantam jack sockets make up the centre section of the patchbay. For ease of socket location, use is made of coloured zones, arranged as blocks.

**32-GROUP CONSOLES.**
**CHANNELS & GROUPS 25-32.**

1-14 The same functions as for Channels & Groups 1-24.

15 **PARALLEL.**
Strip positions 25-28, and 29-32 are paralelled for signal distribution.

HIGH-GROUPS CONSOLES.

CHANNELS 25-32.

1-5 Exactly the same functions as 1-5 on the top module.

6 GROUP OUT.
As 12 above.

7 TAPE SEND A.
As 13 above.

8 TAPE SEND B.
As 14 above.

CHANNELS 33-48.

The patchbay has been designed to cater for up to 48 channels. 33-40 and 41-48 are arranged as two blocks for ease of jack location. Jacks 1-5 perform exactly the same functions as 1-5 on the top module.

MASTERS.

The Master section comprises sockets 6-15 of each vertical strip 33-48. Like the rest of the patchbay, the Master area is zoned by colour blocks for easy location of jack sockets.

AUX OUTPUTS.

These take up 3 positions in the patchbay:

33 Aux outputs 1-4, jacks 6-13,
34 Aux outputs 5-8, jacks 6-13,
35 Stereo A & B, L & aux outputs, jacks 6-13.

Each of these 12 patch-points consists of 2 sockets configured as an output break pair:

The upper socket is the non-break output from the 3200 and is normalled to the lower socket, which breaks when a jack plug is inserted into it. The signal on the jack plug overrides the normal signal. The lower socket connects to the appropriate multipin EDAC for interfacing to external equipment.

| | |
|---|---|
| Output type: | Electronically Balanced line. |
| Output level is: | +4dBu |
| Output impedance is: | <75 ohms. |
| Output headroom is: | +26dBu into 600 ohms. |

OSCILLATOR OUTPUT.

Takes up 2 positions in the patchbay:

33 & 34 jacks 14 & 15.

The oscillator output is parallelled accross jacks 14 on each position. Both positions are configured as an output break pairs.

| | |
|---|---|
| Output type: | Electronically Balanced line. |
| Output level is: | +4dBu. |
| Output impedance is: | 600 ohms. |
| Output headroom is: | +26dBu into 600 ohms. |

## FX RETURNS.

4 stereo FX Returns occupy 2 positions:

36 FX Returns 1-4, left channel, jacks 6-13,
37 FX Returns 1-4, right channel, jacks 6-13,

Each of these 8 patch-points is configured as an input break pair: The upper socket is the non-break signal from the external equipment (via the FX Return EDAC). It is normalled to the lower socket, which breaks when a jack plug is inserted into it. The signal on the jack plug overrides the normal signal. The lower socket connects to the appropriate FX Return in the FX Return /Comms module of the 3200.

| | |
|---|---|
| Input type: | Electronically balanced line. |
| Input level is: | +4dBu. |
| Input impedance is: | 15k ohms. |
| Input headroom is: | +26dBu. |

## COMMS (COMMUNICATIONS).

Communications to and from the 3200 console route via 4 pairs of jack sockets. 1 position is occupied:
Position 38: EXT 1, EXT 2, BOOTH. Jacks 6-11.
Configured as output break pairs.

| | |
|---|---|
| Output type: | Ground-cancelling line. |
| Output level is: | +4dBu. |
| Output impedance is: | <75 ohms. |
| Output headroom is: | +20dBu into 2k ohms. |

Position 38: LISTEN. Jacks 12 & 13.
Configured as an input break pair.

| | |
|---|---|
| Input type: | Balanced dynamic microphone. |

## PHONES.

3 positions are occupied by the Phones patching facilities:
39 Phones Inputs 1-4, Left & Right, jacks 6-13.
The L&R signals are normally sourced from Stereo Aux A L&R, however, inserting a jack plug into a Phones input will override the feed to the selected phones input.
40 Phones Outputs 1-4, Left Channel, jacks 6-13.
41 Phones Outputs 1-4, Left Channel, jacks 6-13.
Each of these 8 patch-points consists of 2 sockets configured as an output break pair:
The upper socket is the non-break output from the 3200 and is normalled to the lower socket, which breaks when a jack plug is inserted into it. The signal on the jack plug overrides the normal signal. The lower socket connects to the appropriate multipin EDAC for interfacing to external equipment.

| | |
|---|---|
| Output type: | Ground-cancelling line. |
| Output level is: | +4dBu. |
| Output impedance is: | <75 ohms. |
| Output headroom is: | +20dBu into 2k ohms. |

## MIX PATCHING.

3 positions are occupied by the Mix patching facilities:

**Mix inserts:**

42, 43, 44 Insert Send; Left, Right, Mono; jack 6.
42, 43, 44 Insert Return; Left, Right, Mono; jack 7.

MIX INSERT SEND.
Normalled to the Insert Return.
Non-break jack: The signal is always available.

| | |
|---|---|
| Output type: | Ground-cancelling line. |
| Output level is: | -10dBv (-7.8dBu). |
| Output impedance is: | <75 ohms. |
| Output headroom is: | +20dBu into 2k ohms. |

MIX INSERT RETURN.
Normalled from the Insert Send.
Configured as a break jack.

| | |
|---|---|
| Input type: | Electronically balanced line. |
| Input level is: | -10dBv (-7.8dBu). |
| Input impedance is: | 10k ohms. |
| Input headroom is: | +26dBu. |

**Mix Outputs:**

42, 43, 44 Left, Right, Mono; jacks 8 & 9.
Configured as output break pairs.

| | |
|---|---|
| Output type: | Electronically Balanced line. |
| Output level is: | +4dBu. |
| Output impedance is: | <75 ohms. |
| Output headroom is: | +26dBu into 600 ohms. |

Jack 9 of each position connects to the appropriate multipin EDAC for interfacing to external equipment.

**2-TRACK SENDS A & B, LEFT & RIGHT.**

Position 45:
2-Track A Left & Right, jacks 6 & 7.
2-Track B Left & Right, jacks 6 & 7.
Normalled from Mix Outputs Left & Right.
Configured as break jacks.
These jacks connect to the appropriate multipin EDAC for external interfacing.

**STUDIO MONITORS.**

Position 40 & 41, jacks 14 & 15.
Configured as output break pairs.

| | |
|---|---|
| Output type: | Ground-cancelling line. |
| Output level is: | +4dBu. |
| Output impedance is: | <75 ohms. |
| Output headroom is: | +20dBu into 2k ohms. |

Jack 15 of each position connects to the appropriate multipin EDAC for interfacing to external equipment.

**CONTROL ROOM MONITORS.**

Position 42 & 43, jacks 10 - 15.
Configured as output break pairs.

| | |
|---|---|
| Output type: | Electronically balanced line. |
| Output level is: | +4dBu. |
| Output impedance is: | <75 ohms. |
| Output headroom is: | +26dBu into 600 ohms. |

Jacks 11, 13 & 15 of each position connect to the appropriate multipin EDAC for interfacing to external equipment.

**2-TRACK RETURNS.**

44 2-Track Returns A, B, C, left channel, jacks 10-15,
45 2-Track Returns A, B, C, right channel, jacks 10-15,

Each of these 6 patch-points is configured as an input break pair: The upper socket is the non-break signal from the external equipment (via the EDAC). It is normalled to the lower socket, which breaks when a jack plug is inserted into it. The signal on the jack plug overrides the normal signal.

| | |
|---|---|
| Input type: | Electronically balanced line. |
| Input level is: | +4dBu. |
| Input impedance is: | 15k ohms. |
| Input headroom is: | +26dBu. |

**MONITOR SOURCES.**

46 Ext 1 & 2, Cassette, Disc, left channel, jacks 6-13,
47 Ext 1 & 2, Cassette, Disc, right channel, jacks 6-13,

Functionally and electrically as 2-Track Returns above.

**BOTTOM MODULE.**

TIE LINES 1-160.
160 jacks arranged as 5 rows of 32. They are uncommitted balanced connections intended for interfacing external equipment.
The jacks:
are not normalled,
do not utilise break contacts,
connect directly to EDAC connectors.
There are 16 jacks per EDAC, configured as 1-16 to the first Tie EDAC, 17-32 to the second Tie EDAC, etc.

## 4.00 INSTALLATION AND COMMISSIONING.

This section covers all aspects of installation of the 3200:
- Audio inputs and outputs,
- Power supply wiring,
- Installation testing and commissioning.

**Audio inputs and outputs.**
The standard signal interface for the 3200 console is via 56 pin EDAC 516 series connectors.

**NOTE:**
Line inputs and Tape returns on the Soundcraft 3200 are via EDAC connectors, and are therefore **not automatically terminated** when not in use, as jack socket inputs would be.

**Power supply interface** is via SRC Connectors:

| | |
|---|---|
| 2 x 16 pin | for Large PSUs. |
| 4 x 8 pin | for Booster PSUs (36/32 consoles).<br>(3 only on 32/24 consoles). |

**System Grounding.**
1 x Grounding Post for system earthing.

## 4.01 PHYSICAL SIZES

5 versions of the console are available in 2 frame sizes:

| | |
|---|---|
| 48 input, 32 output, remote patchbay. | Large frame. |
| 36 input, 32 output. | Large frame. |
| 44 input, 24 output. | Large frame. |
| 36 input, 24 output. | Small frame. |
| 32 input, 24 output. | Small frame. |

Sizes of the 2 frames are:

Large:

| | |
|---|---|
| Max height of overbridge | (m) |
| Max width (end to end) | (m) |
| Max depth (front to back) | (m) |

Small:

| | |
|---|---|
| Max height of overbridge | (m) |
| Max width (end to end) | (m) |
| Max depth (front to back) | (m) |

## 4.02 AUDIO INPUT / OUTPUT CONNECTIONS

### EDAC CONNECTOR LISTING.

| | | |
|---|---|---|
| EDAC | 1 | Mic inputs 1-16. |
| EDAC | 2 | Mic inputs 17-32. |
| EDAC | 3 | Mic inputs 33-48. |
| | | |
| EDAC | 4 | Tie Lines 1- 16. |
| EDAC | 5 | Tie Lines 17- 32. |
| EDAC | 6 | Tie Lines 33- 48. |
| EDAC | 7 | Tie Lines 49- 64. |
| EDAC | 8 | Tie Lines 65- 80. |
| EDAC | 9 | Tie Lines 81- 96. |
| EDAC | 10 | Tie Lines 97-112. |
| EDAC | 11 | Tie Lines 113-128. |
| EDAC | 12 | Tie Lines 129-144. |
| EDAC | 13 | Tie Lines 145-160. |
| | | |
| EDAC | 28 | Not fitted. |
| EDAC | 29 | Line Input A 33-40.* |
| EDAC | 30 | Line Input B 33-40.* |
| EDAC | 31 | Not fitted. |
| EDAC | 32 | Line Input A 41-48.* |
| EDAC | 33 | Line Input B 41-48.* |
| | | |
| EDAC | 14 | Aux Outputs 1-8,<br>Stereo A & Stereo B outputs. |
| EDAC | 15 | FX Return Inputs,<br>Monitor external inputs. |
| EDAC | 16 | Comms inputs and outputs,<br>Red Light. |
| EDAC | 17 | Phones outputs,<br>Monitor speaker outputs. |
| EDAC | 18 | Mix outputs,<br>2-Track Tape Send & Return. |
| EDAC | 19 | Not fitted. |
| | | |
| EDAC | 20 | Tape machine A<br>Tracks 1- 8 Send & return. |
| EDAC | 22 | Tape machine A<br>Tracks 9-16 Send & return. |
| EDAC | 24 | Tape machine A<br>Tracks 17-24 Send & return. |
| EDAC | 26 | Tape machine A<br>Tracks 25-32 Send & return. |
| | | |
| EDAC | 21 | Tape machine B<br>Tracks 1- 8 Send & return. |
| EDAC | 23 | Tape machine B<br>Tracks 9-16 Send & return. |
| EDAC | 25 | Tape machine B<br>Tracks 17-24 Send & return. |
| EDAC | 27 | Tape machine B<br>Tracks 25-32 Send & return. |

**NB: EDACs 29, 30, 32, 33** are for Line Inputs only. They use the same pin configuration as Tape I/O EDACs.