

# Soundcraft

## 8000

### USER MANUAL

---

Soundcraft Electronics Ltd.  
Cranborne House  
Cranborne Industrial Estate  
Cranborne Road  
Potters Bar  
Hertfordshire EN6 3JN  
England  
Tel: 0707 665000  
Fax: 0707 660482  
Telex: 21198

*Soundcraft Canada Inc.,  
1444 Hymus Boulevard,  
Dorval,  
Quebec, Canada, H9P 1J6.  
Tel: (514) 685 1610  
Telex: 05 822582  
Fax: (514) 685 2094*

*JBL,  
PO Box 2200,  
8500 Balboa Boulevard,  
Northridge,  
CA 91329.  
Tel: 818 893 4351  
Telex: 23 66 4923  
Fax: 818 893 3639*

*Soundcraft Japan,  
4F Yoyogi Living,  
12-21 Sendagaya 5,  
Shibuyaku, Tokyo, 151 Japan.  
Tel: (03) 341 6201  
Fax: (03) 341 5260*

The information in this document has been carefully checked and is believed to be entirely reliable. However, no responsibility is assumed for inaccuracies. Furthermore, Soundcraft reserves the right to make changes to any product to improve reliability, function or design.

Soundcraft does not assume any liability arising out of the application or use of any product or circuit described herein.

This document is copyrighted and all rights are reserved.  
This document may not in whole or in part be copied,  
photocopied, reproduced or reduced to any electronic me-  
dium or machine readable form without prior consent in  
writing from Soundcraft Electronics Limited.

**Copyright © Soundcraft Electronics Limited 1987**  
All rights reserved.

Manual design: Dave Pallant 1987

Edition:-		
One	9 October 1987	All Chapters
TWO	17th JULY 1989	Update.

---

# TABLE OF CONTENTS

---

## CONSOLE DESCRIPTION 1-1

---

## PERFORMANCE SPECIFICATIONS 1-2

---

## 8206 HOUSE INPUT MODULE 2-1

---

- |    |                         |     |
|----|-------------------------|-----|
| 1. | Channel input section   | 2-1 |
| 2. | Equaliser section       | 2-2 |
| 3. | Auxiliary section       | 2-3 |
| 4. | Routing section         | 2-4 |
| 5. | Channel Status section- | 2-5 |
- 

## 8200 STEREO INPUT MODULE 2-6

---

- |    |                        |      |
|----|------------------------|------|
| 1. | Channel Input Section  | 2-6  |
| 2. | Equaliser section      | 2-7  |
| 3. | Auxiliary section      | 2-8  |
| 4. | Routing Section        | 2-9  |
| 5. | Channel status section | 2-10 |
- 

## 8207 PA INPUT MODULE 2-11

---

- |    |                        |      |
|----|------------------------|------|
| 1. | Channel Input Section  | 2-11 |
| 2. | Equaliser Section      | 2-12 |
| 3. | Auxiliary Section      | 2-13 |
| 4. | Routing Section        | 2-14 |
| 5. | Channel Status Section | 2-15 |
- 

## 8203 PA OUTPUT (Effects Return) 3-1

---

- |    |                   |     |
|----|-------------------|-----|
| 1. | Effects Return    | 3-1 |
| 2. | Equaliser Section | 3-2 |
| 3. | Sub-group         | 3-3 |
- 

## 8204 PA OUTPUT (Matrix) 3-4

---

- |    |                   |     |
|----|-------------------|-----|
| 1. | Matrix Output     | 3-4 |
| 2. | Equaliser         | 3-5 |
| 3. | Sub-group section | 3-6 |
- 

## 8202 AUXILIARY MASTER MODULE 3-7

---

- |    |                      |     |
|----|----------------------|-----|
| 1. | Auxiliary master 1-8 | 3-7 |
| 2. | Oscillator           | 3-7 |
| 3. | Headphone Output     | 3-7 |
-

## Table of Contents

### 8205 MASTER MODULE

3-8

1.	Talkback Section	3-8
2.	Cassette	3-8
3.	Phones	3-9
4.	Auxiliary mix output	3-9
5.	Main Mix	3-9

### THE 8000 VCA SUB-GROUPING SYSTEM 3-10

### CONNECTOR PANELS

4-1

Input connector panel	4-1
Stereo input connector panel	4-2
Output connector panel (FX Return)	4-3
Output connector panel (Matrix)	4-4

### INSTALLATION

5-1

Applying Power	5-1
Interface levels	5-1
Connector Conventions	5-1
General Wiring Procedures	5-1

### MAINTENANCE

6-1

General Fault Finding	6-1
Removing Modules	6-2
Meter Alignment	6-2
Lamp Replacement	6-2

### SOUNDCRAFT RECOMMENDED WARRANTY 7-1

---

## CONSOLE DESCRIPTION

---

The Soundcraft Series 8000 console has been designed to cater for public address applications. Three input module types are available to cater for the various needs of Public Address work. Two output module types are also available, allowing the user to specify a console precisely for his needs.

Key features on the desk include parametric equalisation, up to eight auxiliary sends, dedicated mix outputs (left and right), and auxiliary outputs (left and right). There is an 8-way output mixing matrix available to order. All balanced inputs and outputs use an electronic transformerless design to ensure low inherent noise.

The use of electronic balancing reduces the degradation of signal quality which is introduced by more conventional transformer coupled designs, ensuring superior transient response, minimal phase shift and excellent common mode rejection even at high frequencies.

Metering on the Series 8000 is in the form of 8 VU meters which may read group or matrix output or effects input, depending on the output module used. 2 VU meters read the Mix output or solo signal.

Being modular in construction the Series 8000 is easy to dismantle making any necessary maintenance extremely straight-forward, even when on the road.

The power supply is a 19" rack mounted unit supplying the console with  $\pm 17\text{v}$  rails, a +48volt rail for phantom powered microphones and a +24volt logic rail.

---

# PERFORMANCE SPECIFICATIONS

---

These figures are typical of the 8000 series but may vary slightly between options and desk sizes.

### DISTORTION

Measured at unity gain with +20dBu at output.	1kHz	10kHz
Line input to group output	.006%	.012%
Line input to mix output	.007%	.008%

### CROSSTALK (measured with sine wave)

Between left and right mix outputs	-64dB	-60dB
Any input to any output	-72dB	-66dB
Channels on switch isolation	-88dB	-86dB

### NOISE

Measured with 20Hz - 20kHz bandwidth	
Mic input EIN, 200 Ohm source	-128.5dBu
Group output noise (1 channel routed)	-86dBu
Typical mix output noise (24 channels routed)	-80dBu
Typical aux output noise (24 channel console)	-80dBu

### FREQUENCY RESPONSE

Measured at +10dBu, equaliser bypassed, ref 1kHz	20Hz:-0.8dB	20kHz:-0.5dB
--	-------------	--------------

### GAIN

Max gain from mic input to group output	90dB
Max gain from line input to group output	30dB

### OPERATING LEVELS

External interfaces	+4dBu
Internal level	-2dBu

### INPUT/OUTPUT IMPEDANCES

Mic Input	2 kohm
Line level inputs	10 kohm
Any output	< 75 ohm

### MAX OUTPUT LEVELS

Unbalanced outputs	+21dBu into 5kOhm
Balanced outputs	+26dBu into 600Ohm

### DIMENSIONS

16 channel frame	L:42.4" D:29.5" H:12.5"
24 channel frame	L:54.2" D:29.5" H:12.5"
32 channel frame	L:66" D:29.5" H:12.5"
40 channel frame	L:77.8" D:29.5" H:12.5"

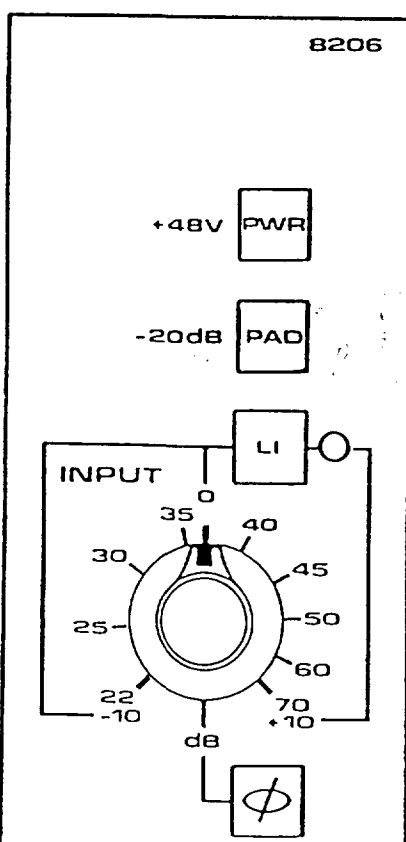
# 8206 HOUSE INPUT MODULE

## 1. Channel input section

Each channel is individually switchable between the Microphone and Line input by pressing the Line Input, LI, switch.

The microphone input is electronically balanced using a transformerless design, configured for optimum low noise operation. The input impedance of the mic input is greater than 2kOhms. This gives a good compromise for flat response and maximum power transfer from the microphone.

The high level line input is balanced with an input impedance greater than 10kOhms. This is high enough to interface to any normal professional peripheral equipment without loading the source.



### a)PWR

Capacitor microphones can be powered by the consoles internal +48volt phantom power supply by pressing the PWR button. When using Direct Injection boxes, or unbalanced sources, the phantom power should NOT be switched on.

### b)PAD

Pressing the PAD button inserts a 20dB attenuator into the input of the microphone amplifier. This allows extremely high level input signals to be catered for without overloading the input stage. Such high level signals can easily occur from high output capacitor microphones used in close proximity to musical instruments.

### c)MIC TRIM (INPUT gain control)

The microphone input can be varied between 22dB and 70dB of gain using the MIC TRIM control, a 41 position detented potentiometer, allowing resettability with essentially continuously variable gain control. When used in conjunction with the 20dB PAD, a 65dB control range is available.

### d)LI

The high level line input is selected by pressing the LI button.

### e)LINE TRIM (INPUT gain control)

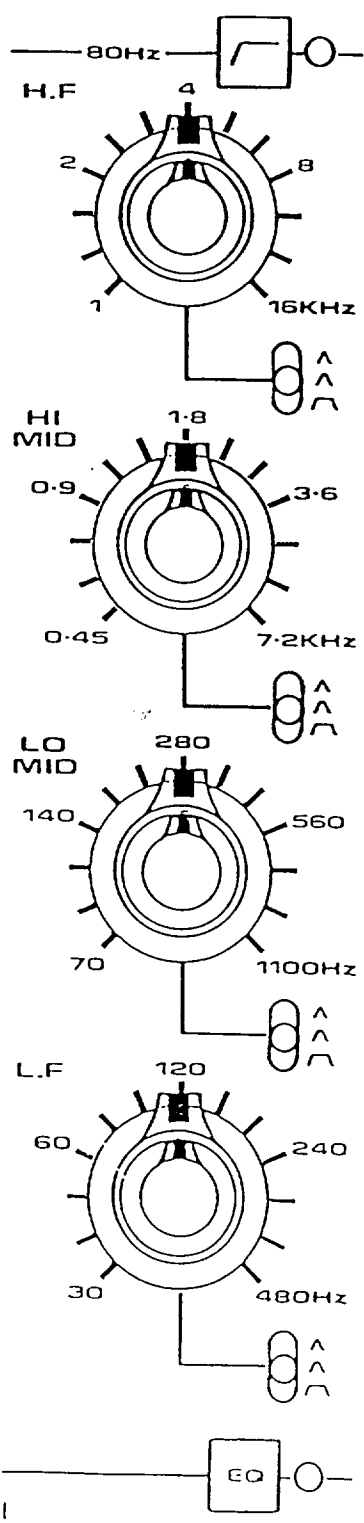
The line input can be varied between -10dB and +10dB of gain using a 41 position detented potentiometer.

### f)PHASE $\phi$

Pressing the phase button reverses the phase of the input signal on both Mic and Line to correct for mis-wired microphones or out of phase mic pick-up in multi-mic situations.

## 2. Equaliser section

The Equaliser unit is an exceptionally versatile device, allowing 5 areas of control over the audio spectrum. All boost/cut pots are centre detented for easy zeroing and the frequency select controls are 41 detented position types.



### a) HIGH PASS FILTER

The High Pass filter operates at 80Hz with an ultimate slope of 24dB/octave. This will effectively remove any low frequency stage rumble and other extraneous signals.

### b) HF (High Frequency)

The HF is continuously variable between 1.0kHz and 16kHz. 15dB of boost or cut is available. The response is of the "bell" type, ie. Having reached maximum amplitude (or minimum in the case of cut), at the selected frequency, the amplitude response returns to zero on either side of that frequency. The shape of the curve, when plotted, shows a characteristic "bell" shape. The Q of the network (a measure of band-width), is switchable between  $Q = 2.4$  for narrow band control,  $Q = 1.3$  and  $Q = 0.65$  for broad band control.

### c) HI MID

The Hi Mid frequency is continuously variable between 450Hz and 7.2kHz. 15dB of boost or cut is available. The response is of the "bell" type. The Q of the network (a measure of band-width), is switchable between  $Q = 2.3$  for narrow band control,  $Q = 0.96$  and  $Q = 0.4$  for broad band control.

### d) LO MID

The Lo Mid section is identical to the Hi Mid section, with the exception that the frequency is variable between 70Hz and 1.1kHz. The Q of the network is switchable between  $Q = 2.3$ , for narrow band control, and  $Q = 1.04$  &  $Q = 0.45$  for broad band control.

### e) LF (Low Frequency)

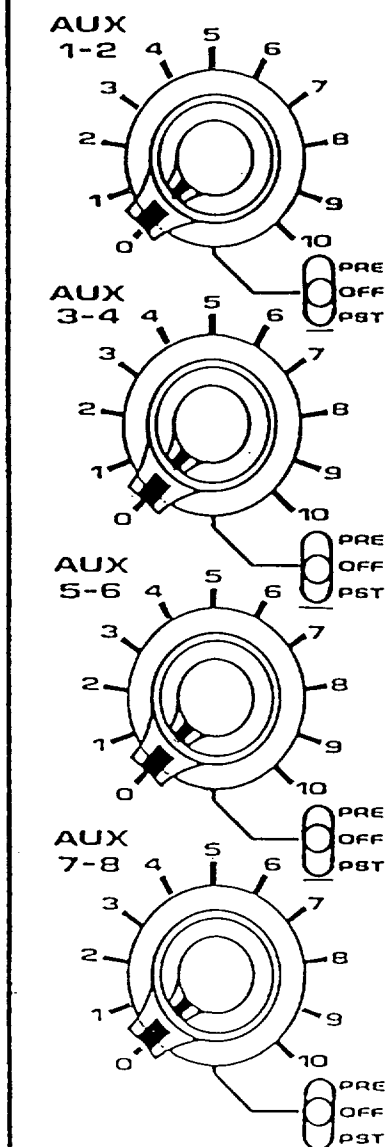
Identical to the HF with the exception that the frequency is variable between 30Hz and 480Hz. The Q of the network is switchable between  $Q = 2.5$ , for narrow band control, and  $Q = 1.04$ , &  $Q = 0.44$  for broad band control.

### f) EQ

The Equaliser circuitry can be switched in and out of the signal path independently of the high pass filter.



### 3. Auxiliary section

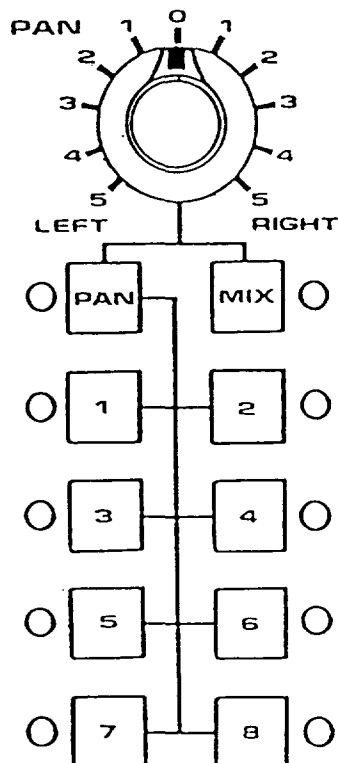


There are 4 dual concentric auxiliary send controls available for use as echo, foldback or other auxiliary effects units. All level controls are 41 position, detented potentiometers. The top feeds odd numbered sends and the skirt feeds even numbered sends.

All auxiliary sends are switchable in pairs pre/post fader or off. In both cases they are post equaliser, insert point and channel on/off switch.

## 4. Routing section

The channel input can be routed to any or all of the 8 Group Outputs and the Stereo Mix, by selecting the relevant routing button.



### a) PAN POT

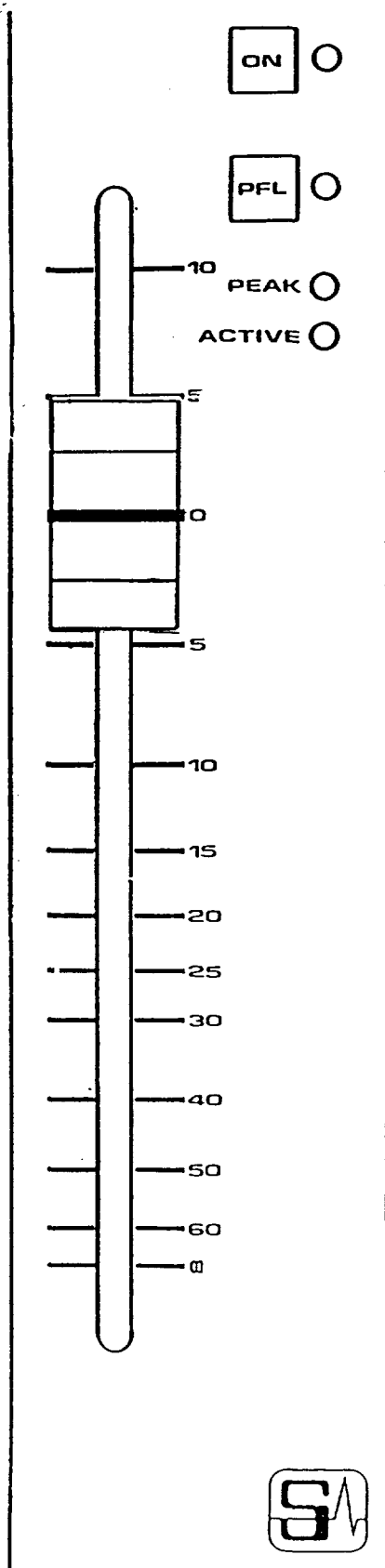
The Pan Pot is a centre detented control, with a loss of 4.5dB at its centre point. This is a compromise between the 3dB loss required for constant power panning, and 6dB loss required for constant voltage panning.

Pressing the PAN button allows you to pan between even and odd numbered groups. However Pan is always in circuit for the stereo mix.

### b) ROUTING

Selection of any routing button assigns the channel signal to an output group or to the stereo mix via the Pan Pot.

## 5. Channel Status section-



### a) ON

The channel "ON" status is indicated by a green LED. When a channel is switched off, all auxiliary sends are switched off with the exception of the signal to the insert send jack.

### b) PFL (Pre Fade Listen)

PFL solos the pre-fader, post insert jack signal, independently of the ON switch. PFL operation is indicated by a red LED on the channel, and a master warning LED on the Master module.

### c) PEAK

A red LED indicates the peak signal level at the insert send point, and illuminates at a level of approximately 4dB below clipping.

### d) ACTIVE

The active LED illuminates when the signal in the module has reached its nominal output level of +4dBu.

### e) CHANNEL FADER

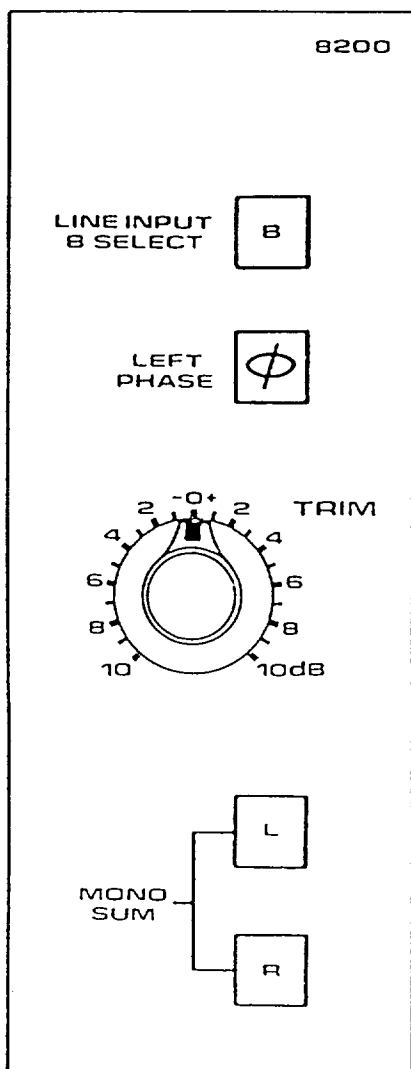
The Channel Fader is a long throw linear fader.

### f) Optional Fader meter.

There is an option available to fit a 20 led bargraph meter into the blank panel next to the fader slot. There are internal links to select the meter to be pre or post fader and Peak or VU response. The system is factory fitted to VU response. If the response of the meter is changed to Peak, the meter will have to be recalibrated using an external, accurate meter by adjusting the preset on the fader bargraph PCB. Alternatively the meter can be recalibrated to suit the users particular levels and operation.

## 8200 STEREO INPUT MODULE

### 1. Channel Input Section



The Channel can be operated using either Line Input A or Line Input B. Both Line Inputs are balanced, with an input impedance of greater than 10kOhms, which is high enough to interface to any normal professional peripheral equipment without loading the source.

**a)B**

Line Input B may be selected by pressing the button marked B.

**b)Phase**

Pressing the Phase button will invert the phase on the left-hand input only to correct for any input mismatch.

**c)TRIM (INPUT GAIN)**

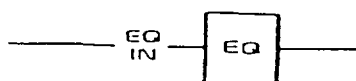
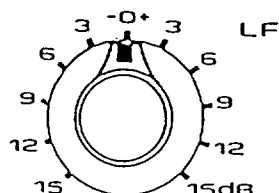
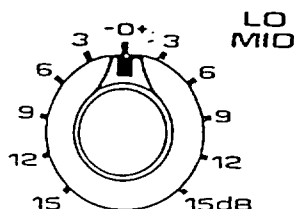
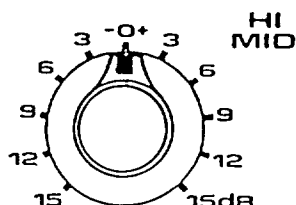
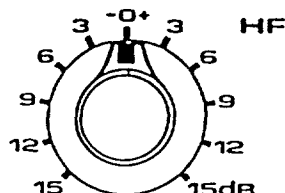
The Input Gain can be varied between -10dB and +10dB of gain using the Gain Trim control.

**d)L & R.**

- i. With both these switches out the module works in stereo mode.
- ii. With either L or R switched in, both channels of the module are fed by either the left or right input.
- iii. With both switches pressed, both channels of the module are fed by a mono sum of the left and right input.

## 2. Equaliser Section

HI PASS  
FILTER



The Equaliser is a versatile unit, allowing 5 areas of control over the audio spectrum. All amplitude pots are centre detented for easy zeroing.

### a) High Pass Filter

The High Pass Filter operates at 100Hz with an ultimate slope of 12dB/Octave. This will effectively remove low frequency stage rumble and other extraneous signals.

### b) HF (High Frequency)

15dB of boost or cut is available at 10kHz, with a "shelving" characteristic, ie. the slope of the EQ curve does not keep rising with frequency, but having reached the desired amount, flattens out or "shelves" from that frequency on.

### c) HI MID

15dB of boost or cut is available at 2kHz, with a "bell" characteristic, ie. having reached maximum amplitude, (or minimum in the case of cut) the amplitude response returns to zero on either side of the frequency. The shape of the curve, when plotted shows a characteristic "bell" shape. The Q of the network (a measure of bandwidth) is 1.5.

### d) LO MID

The Lo Mid section is identical to the Hi Mid section, with the exception that the frequency is 300Hz.

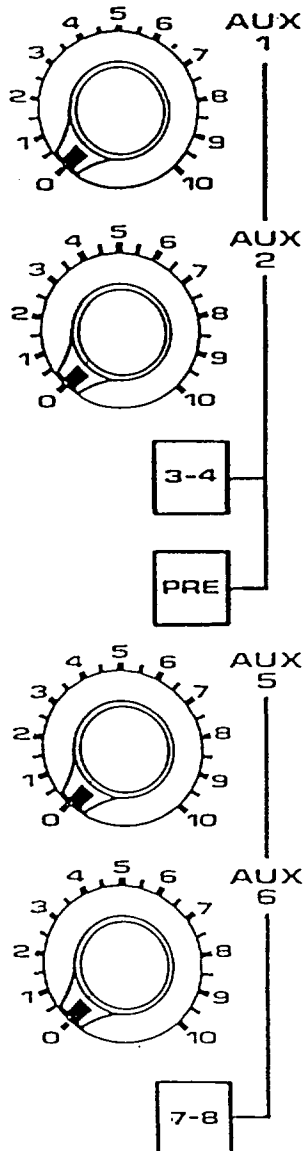
### e) LF (Low Frequency)

15dB of boost or cut is available at 60Hz, with a "shelving" characteristic.

### f) EQ

The Equaliser circuitry can be switched in and out of the signal path, independently of the high pass filter.

### 3. Auxiliary Section



There are 4 Auxiliary send controls available. Each control can be routed to 2 auxiliary buses, to give a total of 8 Auxiliary sends for use as echo, foldback or other auxiliary effects units.

#### a) SENDS 1 and 2

Auxiliary sends 1 and 2 are normally post-fader, but can be switched Pre-fader by pressing the appropriate PRE button. In both cases they are post equaliser. Auxiliaries 1 and 2 can be either mono or stereo and dependent or independent of the channel mute by push-on links.

#### b) SENDS 3 and 4

Auxiliary sends 1 and 2 may be routed to auxiliary buses 3 and 4 by pressing the 3-4 button. Auxiliaries 3 and 4 can be selected mono or stereo by push-on links.

#### c) PRE

Auxiliary sends 1-4 may be routed Pre-fader by pressing the PRE button.

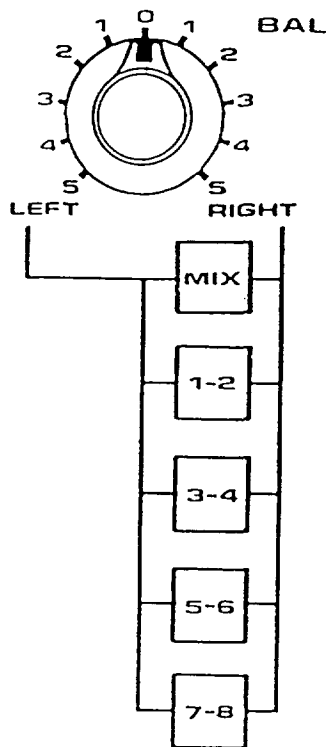
#### d) SENDS 5 and 6

Auxiliary sends 5 and 6 are permanently post-fader.

#### e) SENDS 7 and 8

Auxiliary sends 5 and 6 may be routed to auxiliary buses 7 and 8 by pressing the 7-8 button.

## 4. Routing Section



The channel input signal may be routed to any of the pairs of Group Outputs (1-2, 3-4, 5-6, 7-8) and the Stereo Mix, by selecting the relevant routing button.

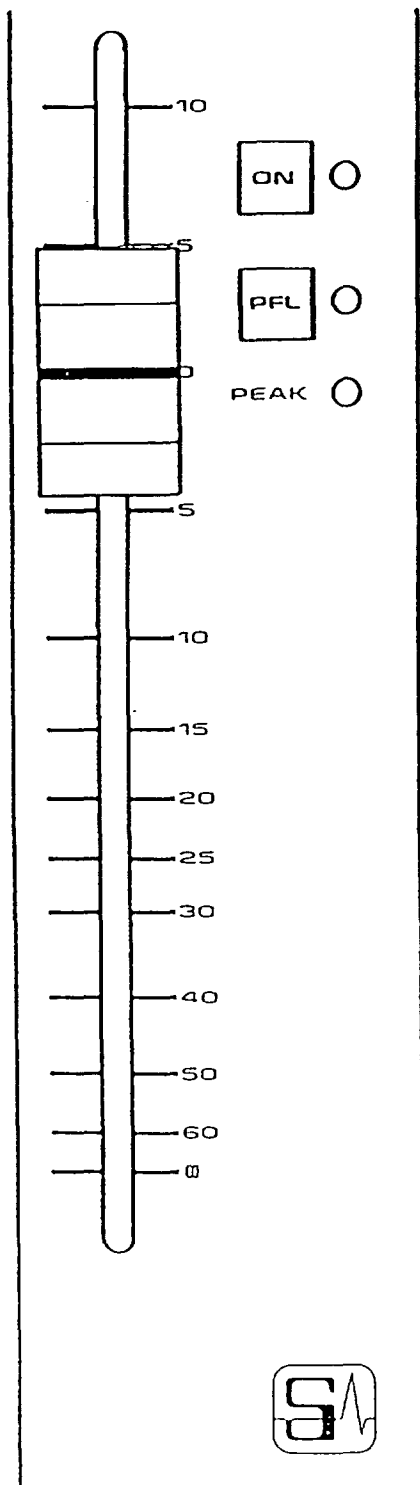
### a) BALANCE

The balance control corrects or deliberately creates any errors in stereo matching before the fader and the auxiliary sends.

### b) ROUTING

Selection of any routing button assigns the channel signal to a pair of output groups, or the stereo mix, via the balance correction.

## 5. Channel Status Section



### a) ON

The channel "ON" status is indicated by a green LED. When a channel is switched off, all auxiliary sends are also switched off.

### b) PFL

Pre-fade Listen solos the Pre-fader signal independently of the ON switch. This gives a mono check of signal before the on/off switch. PFL operation is indicated by a red LED on the channel and a master warning LED on the master module.

### c) PEAK

A red LED indicates the peak signal level. This gives visual warning that the higher signal Right or Left is within 5dB of clipping.

### d) CHANNEL FADER

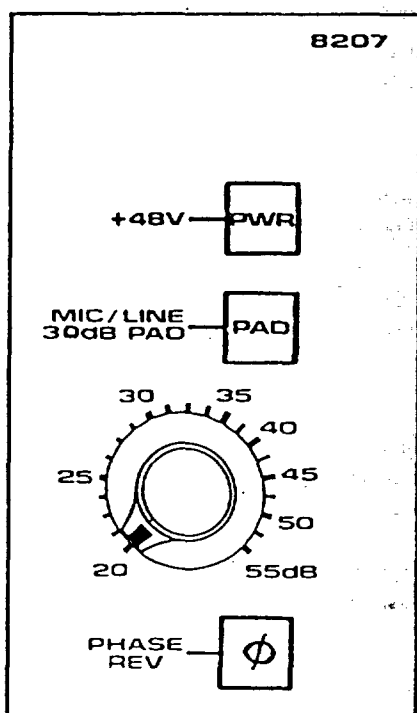
The channel fader is an accurately matched long throw stereo fader. Infinity cut off is greater than 90dB.



## 8207 PA INPUT MODULE

The 8207 PA input module has been specifically designed for use in "front of house" PA applications. The input is electronically balanced, and can be used with either microphone or line level input signals if the 30dB input attenuator is switched into the circuit. Interface to the channel is via the microphone input XLR connector on the rear panel. The associated line input jack is not connected with this model.

### 1) Channel Input Section



The microphone/line input is electronically balanced, using a transformerless design, configured for optimum low noise operation.

The input impedance is greater than 2kOhms, which will not cause any loading effects on any normally used microphone. When the 30dB pad is inserted, the input impedance is greater than 5kOhms, which is high enough to interface to any normal professional peripheral equipment without loading the source.

#### a) PWR

Pressing the PWR button enables capacitor microphones to be powered by the console's internal 48Volt Phantom Power supply. CAUTION: It is not advisable to use a Direct Injection box when the Phantom Power is on.

#### b) PAD (MIC/LINE SELECT)

Pressing the PAD button inserts a 30dB attenuator into the input of the microphone amplifier and allows line level input signals to be catered for, without overloading the input stage. High level signals can also occur from high output capacitor microphones used in close proximity to musical instruments. Direct injection boxes are also capable of providing high signal levels.

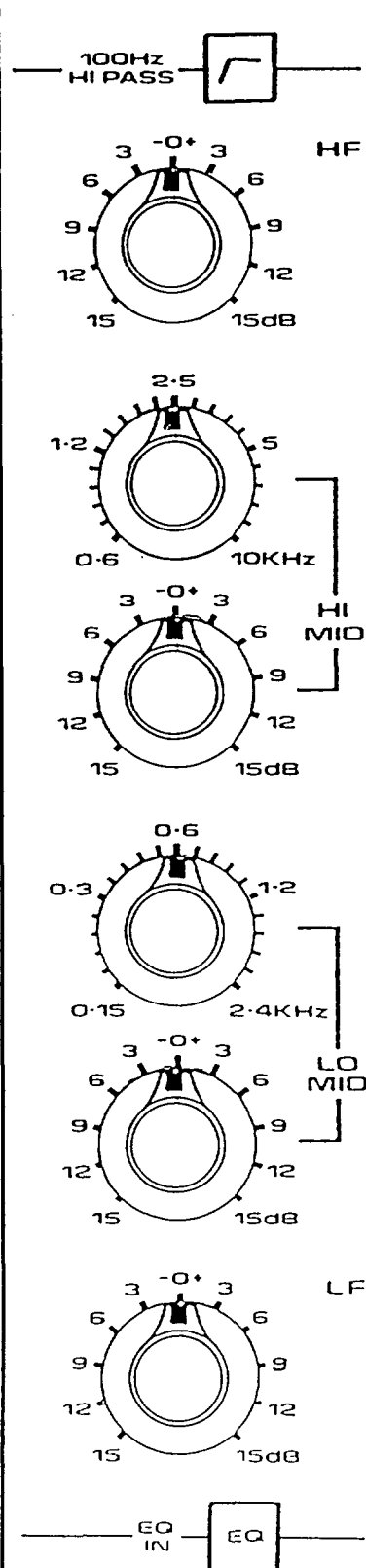
#### c) MIC/LINE TRIM

The microphone/line input can be varied between 20dB and 55dB of gain using the input TRIM control a 41 position detented potentiometer with essentially continuously variable gain control. When used in conjunction with the 30dB PAD, a 65dB control range is available.

#### d) PHASE

Pressing the phase button reverses the phase of the input signal to correct for mis-wired microphones or out of phase mic pick-up in multi-mic situations.

## 2) Equaliser Section



The equalizer is a flexible device allowing five areas of control to be exercised. All amplitude pots are centre detented for easy zeroing, and the frequency select controls are 41 detented position types. The equaliser may be switched in or out of circuit, independently of the high pass filter.

### a) THE HIGH PASS FILTER

The High Pass Filter operates at 100Hz with an ultimate slope of 12dB/octave. This will effectively remove low frequency stage rumble and other extraneous signals.

### b) HIGH FREQUENCY

15dB of boost or cut is available at 10kHz, with a "shelving" characteristic, ie. the slope of the EQ curve does not keep rising with frequency, but having reached the desired amount, flattens out or "shelves" from that frequency on.

### c) HI MID

The Hi Mid Frequency is continuously variable between 600Hz and 10kHz, with 15dB of boost or cut available. The response is of the "bell" type, ie. having reached maximum amplitude (or minimum in the case of cut) at the selected frequency, the amplitude response returns to zero on either side of that frequency. The "Q" (a measure of the bandwidth) of the network is 1.5.

### d) LO MID

The Low Mid section is identical to the Hi Mid section with the exception that the frequency is variable between 150Hz and 2.4kHz.

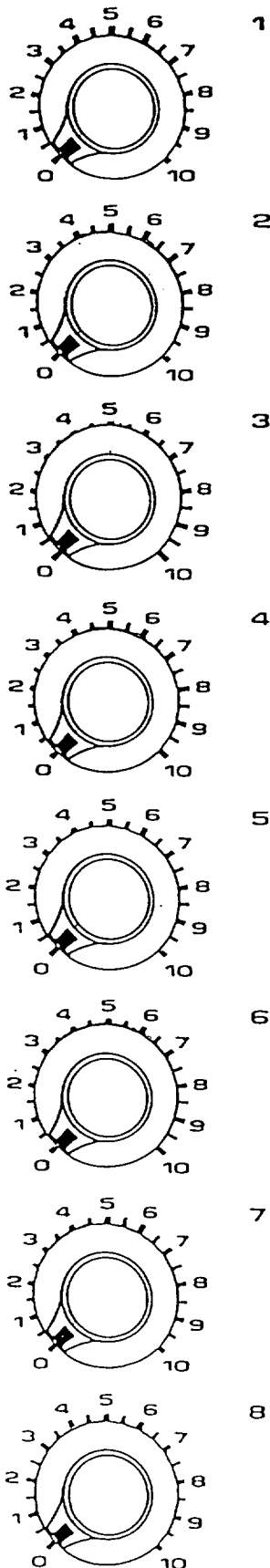
### e) LOW FREQUENCY

15dB of boost or cut is available at 60Hz, with a "shelving" characteristic.

### f) EQ BUTTON

The equaliser circuitry can be switched in or out of the signal path, independently of the High Pass filter.

### 3. Auxiliary Section



There are 8 auxiliary send controls available which can be used for echo, foldback or other auxiliary effects units. All controls are 41 position, detented potentiometers. Each pair of sends can be internally selected by a link on the PC Board to be either pre or post the channel fader.

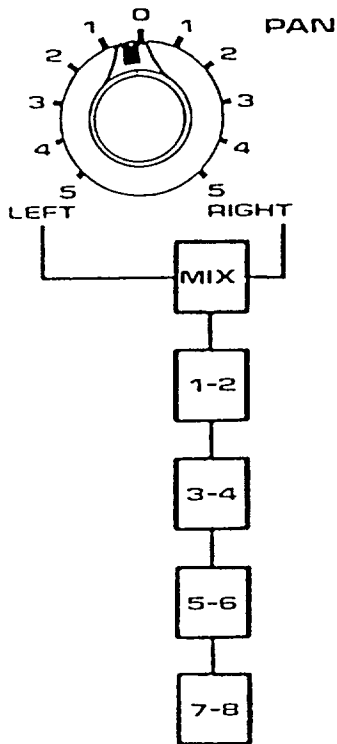
For Pre fader sends odd numbered links should be ON. ie. L1, L3, L5 and L7.

For Post fader sends even numbered links should be ON. ie. L2, L4, L6 and L8.

Links 1 and 2 are used for Aux 1 and 2

Links 3 and 4 are used for Aux 3 and 4 etc.

## 4. Routing Section



The channel input signal can be routed to any or all of the 8 Group Outputs and the stereo Mix by selecting the relevant routing button.

To route the input signal to one particular Group press the relevant routing button. Groups 1,3,5 and 7 correspond to the left-hand side of the pan-pot and Groups 2,4,6 and 8 the right-hand side.

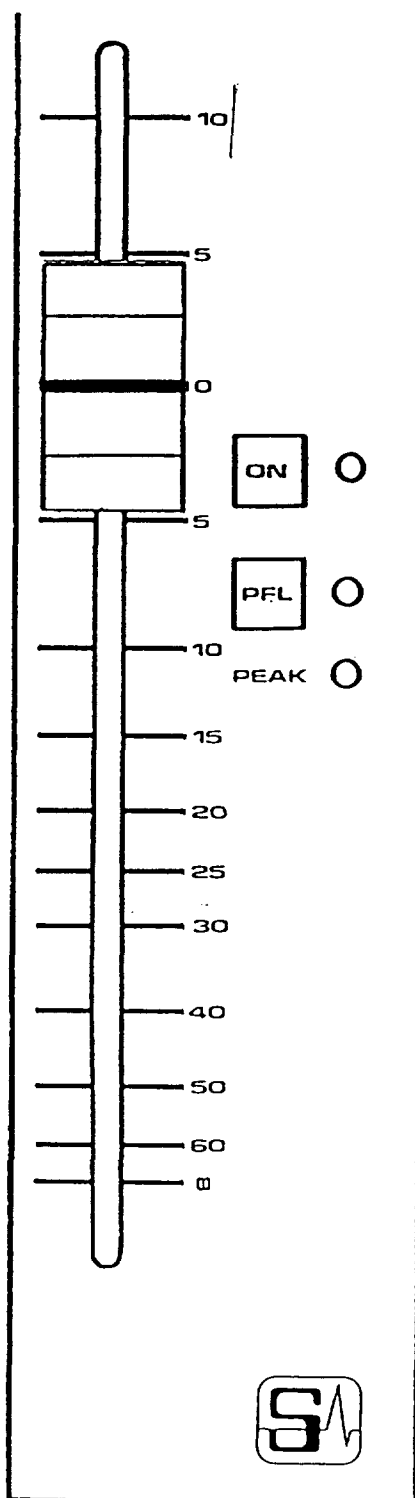
### a) PAN POT

The Pan Pot is a centre detented control, with a loss of 4.5dB at its centre point. This is a compromise between the 3dB loss required for constant power panning and 6dB loss required for constant voltage panning.

### b) ROUTING

Selection of any routing button assigns the channel signal to a pair of output groups, or to the stereo mix via the Pan Pot.

## 5) Channel Status Section



### a) ON

The channel "ON" status is indicated by a green LED. When a channel is switched off, all auxiliary sends are also switched off, with the exception of the signal to the insert jack.

### b) PFL (Pre Fade Listen)

PFL solos the pre-fader, post insert jack signal, independently of the "ON" switch. PFL operation is indicated by a red LED on the Master Module.

### c) PEAK

A red LED indicates the peak signal level at the insert send point, illuminating at a level of approximately 4dB below clipping.

### d) CHANNEL FADER

The channel fader is a long throw linear fader. Infinity cut off is greater than 90dB.

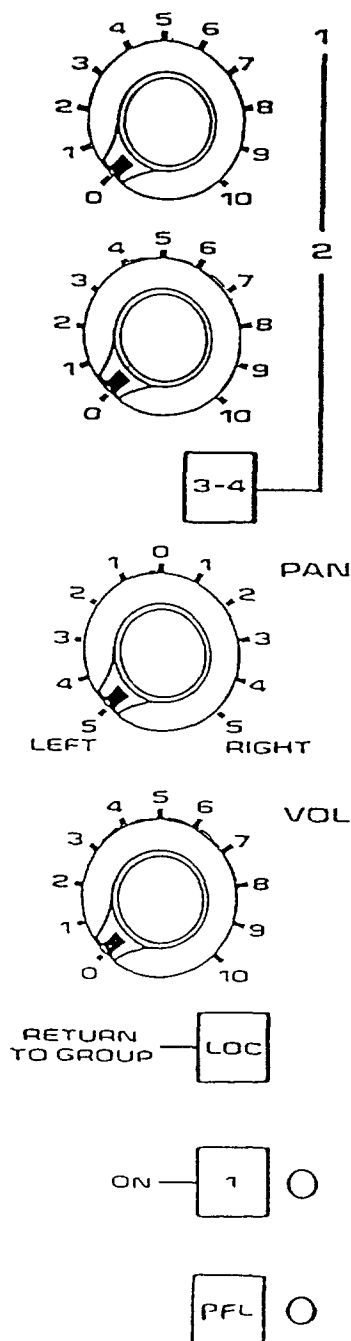
**Input Sections**

A large, empty rectangular box with a thin black border, occupying the majority of the page. It is intended for input sections, likely for a technical drawing or diagram.

## 8203 PA OUTPUT (Effects Return)

The 8203 PA Output module is designed for "front of house" PA applications. It is divided into 3 sections and contains a sub-group, an effects return channel which can be routed back into either the stereo mix or the sub-group and a 3 band equaliser which can be inserted into either the sub-group or the effects return signal paths. The effects return input and the sub-group output are electronically balanced.

### 1. Effects Return



#### a) AUXILIARY SENDS 1-2

The auxiliary send controls are located after the effects return level control and also post the equaliser (if inserted into the effects return).

#### b) 3-4

The auxiliary sends can be alternatively routed to auxiliary outputs 3 and 4 by pressing the 3-4 button.

#### c) PAN

The effects return signal can be panned between left and right of the stereo mix. The pan pot is a centre detented control, with a loss of 4.5dB at its centre point. This is a compromise between the 3dB loss required for constant power panning, and 6dB loss required for constant voltage panning.

#### d) VOL

The relative level of the effects return channel is controlled by the VOL pot.

#### e) LOC (Local)

Normally the effects return signal is routed into the stereo mix, via the pan pot. However, if LOC is selected, the signal is diverted from the pan pot, and returned into the summing bus of its associated sub-group.

#### f) ON

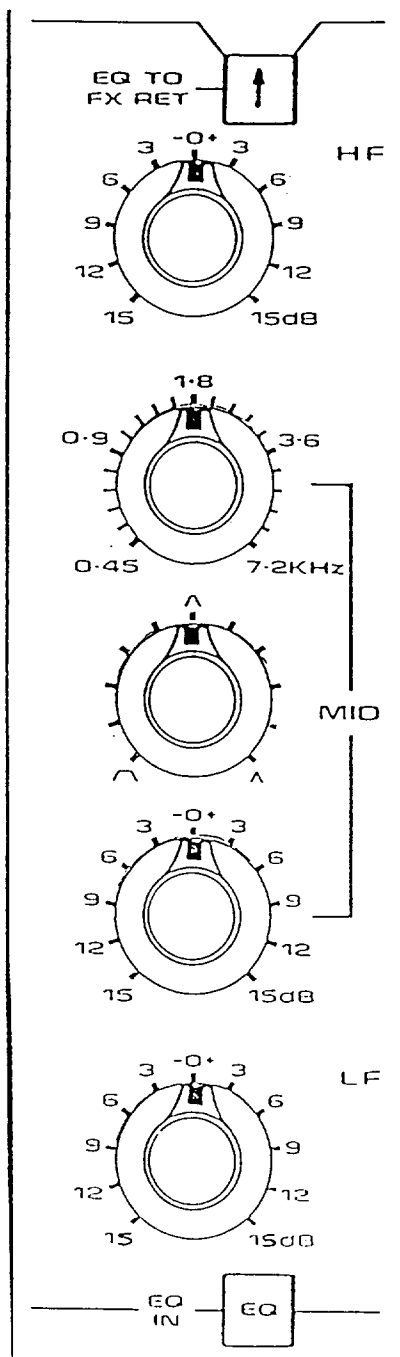
The effects return channel, and auxiliary sends are switched into operation by pressing the ON button. Operation is indicated by an adjacent green LED.

#### g) PFL (Pre-fade listen)

Pressing the PFL button solos the pre-fader signal and functions independently of the ON switch. A warning LED adjacent to the button and a master warning LED on the master module indicates a PFL condition.

## 2. Equaliser Section

The equaliser is a 3 section device comprising of "shelving" type HF and LF controls and a fully parametric mid frequency section.



### a) (EQ to Effects Return)

The equaliser is normally positioned in the sub-group, but may, alternatively, be inserted into the effects return signal path.

### b) HF (High Frequency)

15dB of boost or cut is available at 12kHz with a "shelving" characteristic, ie. the slope of the EQ curve does not keep rising with frequency, but having reached the desired amount, flattens out, or "shelves" from that frequency on.

### c) MID FREQUENCY

15dB of boost or cut is available, the mid frequency is continuously variable between 450Hz and 7.2kHz. The bandwidth may also be varied from broad-band control to very narrow band, almost spot frequency, effects type processing.

### d) LF (Low Frequency)

15dB of boost or cut is available at 60Hz, with a "shelving" characteristic.

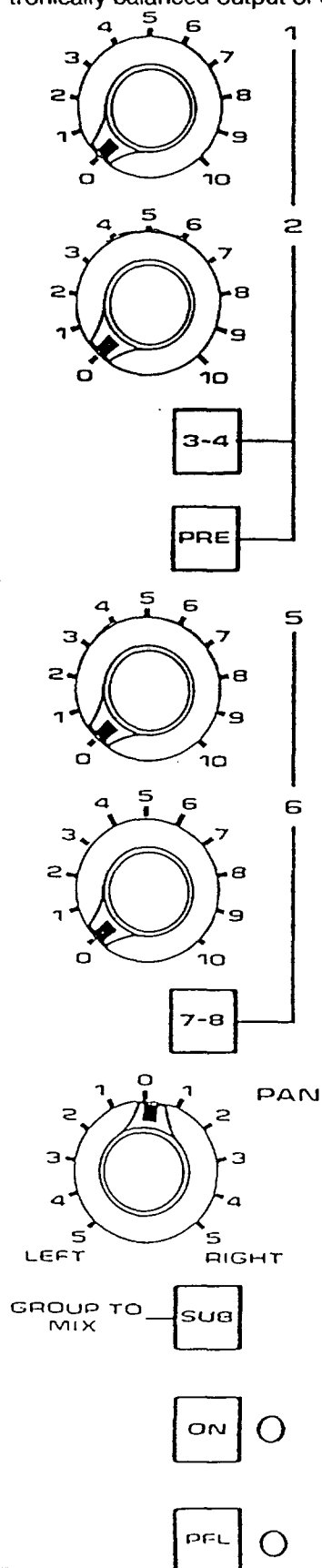
### e) EQ

The equaliser may be switched into the circuit by pressing the EQ button.



### 3. Sub-group

The sub-group provides a mix of any required number of input channels, either for direct use, via an electronically balanced output or as a sub-group for re-routing into the stereo mix.



#### a) AUXILIARY SENDS 1-2

Auxiliary sends 1 and 2 are normally post fader, but can be switched to pre-fader by selecting PRE. Pressing the 3-4 button routes sends 1 & 2 to auxiliary buses 3 and 4.

#### b) AUXILIARY SENDS 5-6

Auxiliary sends 5-6 are permanently post-fader. Pressing the 5-6 button routes sends 5 & 6 to aux buses 7 and 8.

#### c) PAN

The pan control adjusts the relative balance of the sub-group signal into the stereo mix, if SUB has been selected.

The pan pot is a centre detented control, with a loss of 4.5dB at its centre point. This is a compromise between the 3dB loss required for constant power panning and 6dB loss required for constant voltage panning.

#### d) SUB (Group to mix)

Selecting SUB routes the sub-group signal directly into the stereo mix, via the pan pot, without affecting the signal to the group output.

#### e) ON

The sub-group and auxiliary sends are switched into operation by selecting the ON switch. A green LED indicates sub-group operation.

#### f) PFL (Pre-fade Listen)

The pre-fader group signal can be soloed, independently of the ON switch.

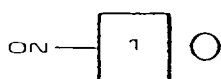
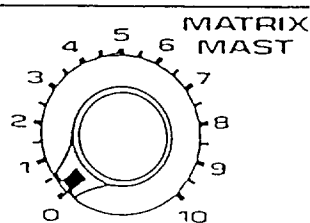
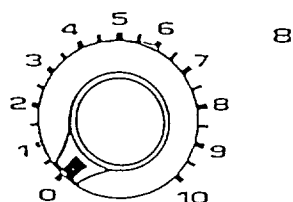
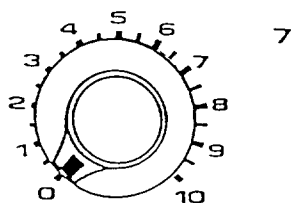
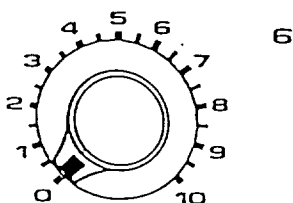
#### g) FADER

The Group fader is a long throw unit with a cut-off of greater than 90dB.

## 8204 PA OUTPUT (Matrix)

The 8204 PA Output (Matrix) has been designed primarily for theatre and stage monitor applications, where an 8-way matrix output provides a large degree of flexibility of operation. Both the sub-group and matrix outputs are electronically balanced. A 3 band equaliser section is switchable to either the sub-group or matrix outputs.

### 1. Matrix Output



#### a) MATRIX SENDS

A signal derived either pre or post each sub-group fader can be mixed into one of the 8 matrix outputs. These are specifically designed for speaker matrix, either for foldback applications where each musician can receive an individual mix of the group outputs or for arrays of speakers around a hall where spacial effects can be created with different mixes of groups to the different arrays.

#### b) PRE

The matrix sends can be taken from either pre or post the sub-group fader. For pre-fader sends the PRE button should be pressed.

#### c) MATRIX MASTER

A matrix master level control is provided to adjust the overall level of the sum of all matrix sends to that matrix output.

#### d) ON

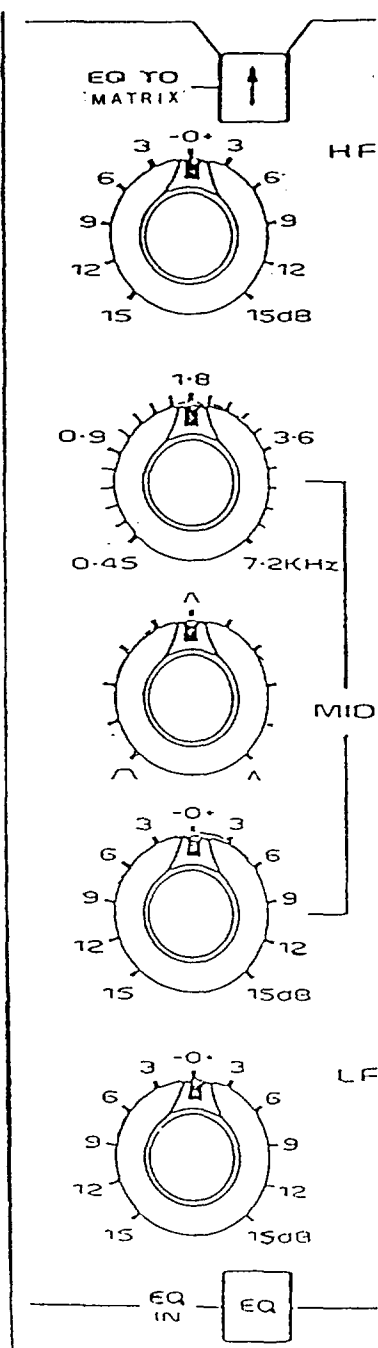
The matrix output is switched into operation by selecting ON. Operation is indicated by a green LED.

#### e) AFL (After-Fade Listen)

The matrix master signal can be soloed.

## 2. Equaliser

The Equaliser is a 3 section device, comprising of shelving type HF and LF sections and a fully parametric mid frequency section.



### a) (EQ to Matrix)

The equaliser is normally positioned in the sub-group, but may, alternatively, be inserted into the matrix signal path.

### b) HF (High Frequency)

15dB of boost or cut is available at 12kHz with a "shelving" characteristic, ie. the slope of the EQ curve does not keep rising with frequency, but having reached the desired amount, flattens out, or "shelves" from that frequency on.

### c) MID FREQUENCY

15dB of boost or cut is available, the mid frequency is continuously variable between 450Hz and 7.2kHz. The bandwidth may also be varied from broad-band control to very narrow band, almost spot frequency, effects type processing.

### d) LF (Low Frequency)

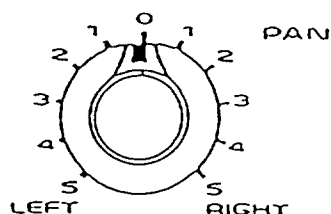
15dB of boost or cut is available at 60Hz, with a "shelving" characteristic.

### e) EQ

The equaliser may be switched into the circuit by pressing the EQ button.

## 3. Sub-group section

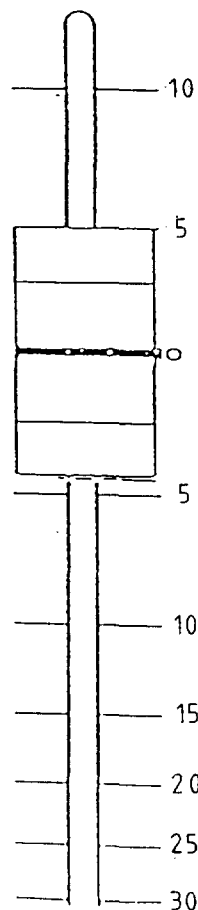
The sub-group provides a mix of any required number of input channels. The sub-group output is electronically balanced. Additionally, the sub-group can be used as a sub-mix and re-routed back into the stereo mix.



SUB

ON

PFL



### a) PAN

The pan control adjusts the relative balance of the sub-group signal into the stereo mix, if SUB has been selected.

### b) SUB (Group to Mix)

Selecting SUB routes the sub-group signal directly into the stereo mix, via the pan pot, without affecting the signal to the sub-group output.

### c) ON

The sub-group is switched into operation by pressing the ON button. A green LED indicates sub-group operation.

### d) AFL

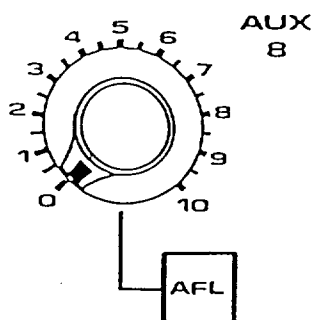
The group output signal can be soloed.

### e) FADER

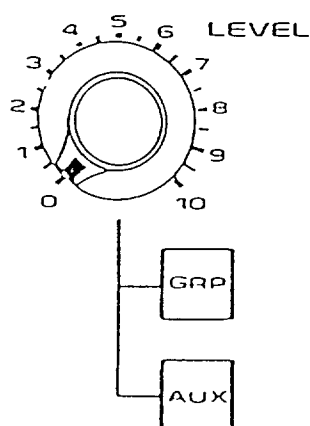
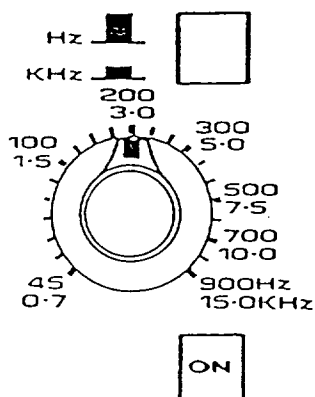
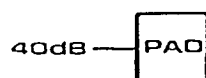
The Group fader is a long throw unit with a cut-off of greater than 90dB.

## 8202 AUXILIARY MASTER MODULE

The 8202 Auxiliary Master module contains the master level controls for the 8 auxiliary outputs, a sweep frequency oscillator and a stereo headphone drive amplifier. The auxiliary and oscillator outputs are electronically balanced.



### OSCILLATOR



## 1. Auxiliary master 1-8

Each of the 8 auxiliary buses has an overall master level control. An associated AFL (After Fade Listen) button allows the signal at the auxiliary to be monitored.

## 2. Oscillator

### a) PAD

The oscillator output can be reduced by a 40dB attenuator. This allows the signal at the oscillator jack to be used for testing microphone channels.

### b) FREQUENCY

The oscillator frequency is continuously variable between 45Hz and 15kHz in 2 ranges; 45Hz to 900Hz and 700Hz to 15kHz.

### c) ON

Enables the oscillator.

### d) LEVEL

Adjusts the level of the oscillator. When used in conjunction with the 40dB Pad, a wide range of output levels can be obtained, suitable for testing microphone and line level inputs.

### e) GRP (Group)

The oscillator can be routed to all output groups for alignment and test purposes.

### f) AUX

Selection of Aux routes the oscillator to all 8 auxiliary buses.

## 3. Headphone Output

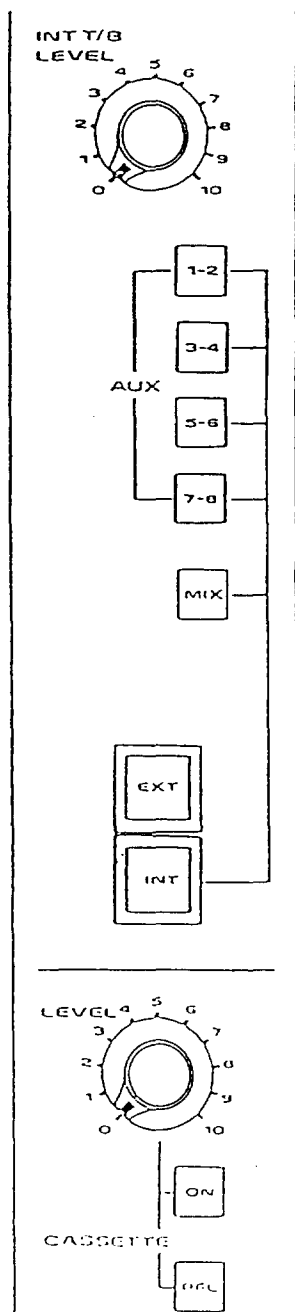
The headphone output is at the bottom of the Aux Master module.

The headphone output allows monitoring of the main stereo mix output and of any PFL/AFL signal. It will drive headphones of all impedances, though use of 8 Ohm phones at high listening levels may cause slight headphone crosstalk into the stereo mix.

## 8205 MASTER MODULE

The 8205 Master Module contains the main electronically balanced stereo output and auxiliary output, cassette input and the talkback system.

### 1. Talkback Section



An electronically balanced XLR input is available with a gain range of 20dB to 60dB for the talkback microphone, phantom power is not provided.

#### a) INTERNAL TALKBACK MIC LEVEL

Adjusts the level of the talkback signal.

#### b) ROUTING

When INT is selected the talkback signal can be routed to any pair of auxiliaries and the stereo mix bus.

#### c) EXT

Selecting EXT routes the talkback signal to the Talkback output on the rear panel for communications between front of house and side stage consoles. If both consoles are Series 8000, or, if the other console is a Soundcraft Series 500 Monitor desk the EXT on the receiving console will illuminate. Similarly, if the other desk has EXT selected, then the EXT switch on the local 8000 will illuminate and the input on the rear panel Talkback input socket will be routed to the headphones.

Any signal can be sent to the EXT input of an 8000 by supplying a "phantom" D.C. voltage on the EXT input to switch the internal selection circuitry of the 8000. Please refer to the circuit diagrams or to Soundcraft for further information.

### 2. Cassette

An input is provided for an external cassette to be mixed into the main mix bus.

#### a) LEVEL

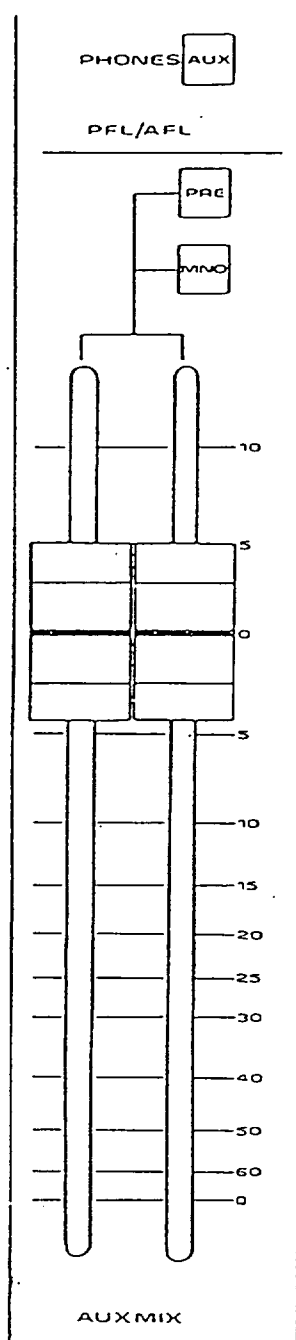
A level control is provided calibrated from 0-10.

#### b) ON

Pressing the ON button enables the cassette input.

#### c) PFL (Pre Fade Listen)

PFL solos the cassette input before the level control.



### 3. Phones

The headphone mix normally follows the main Mix, selecting AUX allows you to listen to the Auxiliary mix output. Phones will always listen to any PFL or AFL signal.

### 4. Auxiliary mix output

An auxiliary mix output is provided for sending a stereo mix to a tape machine or for transmission. The signal always follows the main stereo mix.

#### a) PRE

Selecting PRE takes the auxiliary output signal from before the main Mix faders for independent level control.

#### b) MNO

When MNO is selected the left and right auxiliary output are summed producing a mono signal. The signal is summed pre-fade thereby making the two large faders independent mono outputs.

#### c) FADERS

The Aux faders are long throw units with a cut-off of greater than 90dB.

### 5. MAIN MIX

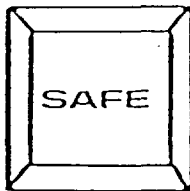
The main mix faders are long throw linear devices with 10dB in hand at the top of travel.

# THE 8000 VCA SUB-GROUPING SYSTEM

The VCA Sub-grouping system for Series 8000 console will allow any channel to be controlled by one or more of the four VCA Group master faders. Because the faders are electronically grouped pre-pan, the four VCA groups are each, effectively, a stereo pair.

The sub-groups each have a group mute and the design has enabled a true Solo-in-Place system to be implemented allowing individual channels to be soloed in their stereo position on the main mix.

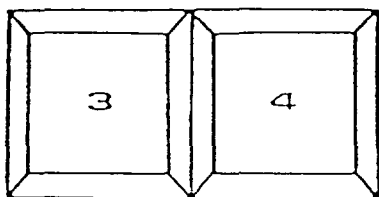
## VCA Group Master Module



There are four identical Group fader sections on the VCA Group Master Module.

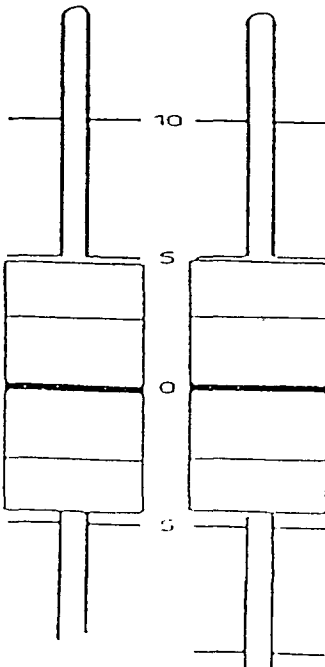
### a) Solo Safe

Because the Solo-in-Place system actively mutes all channels other than those selected, an accidental SIP in the middle of a concert could be disastrous. During a performance, the SIP system can be disabled with the Solo Safe switch. With the Safe switch lit, the system is off.



### b) Group Mute switch

Each group master fader has a group mute switch which will mute all the channels associated with that group. The group is muted when the switch is lit. If a channel is assigned to more than one group, the channel will be muted if **any** of its group master mutes are alight. On power up, all groups are muted.



### c) Group Master Fader

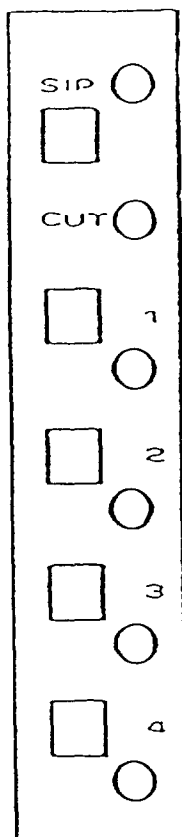
The group fader is marked off in dB from the zero point up to +10dB and down to -infinity, or off. Any channels assigned to one group will be unaffected while the relevant group master fader is at zero. Raising the master to the top of its travel will increase the level of the channel up to a maximum of 10dB. Moving the group master downwards will decrease the level of the channels assigned.

When a channel is assigned to more than one group master, the levels of the different masters are additive. If two masters for example, are both at zero, the effect on the channel will be zero. If both masters are at their -10dB points, the channels assigned to the two masters will be attenuated by 20dB. If **any** of the masters that a channel is assigned to is at its -infinity point, then the channel will be off.

With a channel or group master fader at the lower limit of its travel the channel or channels assigned will have a cut-off as good as that of a standard fader.



## VCA Channel Module



### a) Solo-in-Place

The Solo-in-Place will allow any selected channels to be listened to in their stereo position. It does this by muting all channels not selected to SIP. The yellow led illuminated indicates the channel has been selected to SIP.

### b) Cut Led

The Cut led will light when the individual channel is muted, either by another channel being soloed, or by a group master mute.

### c) Group Select Switches

There are four group select switches each with its own yellow led. The led will light to show that the channel has been assigned to that individual group. With a group selected on these switches, the channel can be controlled by the relevant master fader and mute.

Any one or more of the groups can control a single channel at the same time. This allows VCA sub-groups to overlap. For example; sub-group 1 might consist of all the channels of a drum kit whilst sub-group 2 has all the percussion including a second section of drums. All the channels of the main kit would be selected to both groups 1 and 2.

The other percussion channels would just be selected to group 2. The two group master faders used could then be labelled KIT and PERCUSSION respectively. Similarly a third, overlapping group, could be set up by assigning all the channels except vocals to group 3. Group 4 might then have all the Vocal channels assigned to it. (Diag 1).

## Diagram 1

Channel No.	Channel Name				
1	Kick Drum				
2	Snare				
3	Hi-hat				
4	High toms				
5	Low toms				
6	Kit overhead left				
7	Kit overhead right				
8	Bass D.I.				
9	Bass cab				
10	Guitar 1				
11	Guitar 2				
12	Conga				
13	Toms				
14	Sax				
15	Trumpet				
16	Trombone				
17	Backing Vox 1				
18	Backing Vox 2				
19	Lead Vox 1				
20	Lead Vox 2				
21	Keyboards left				
22	Keyboards right				
23	Effects left				
24	Effects right				
		Group 1	Group 2	Group 3	Group 4

---

# CONNECTOR PANELS

---

---

## INPUT CONNECTOR PANEL

---

### 1. Line Inputs

These standard stereo jacks carry the balanced Line Inputs. They are wired as follows:-

Tip: HOT  
Ring: COLD  
Sleeve: Signal common

### 2. Line Outputs

These standard jacks carry the unbalanced Line outputs and are wired in the following way.

Tip: HOT  
Sleeve: Signal common

### 3. Channel insert sends

These standard jacks carry the Channel insert sends and are normalised to the channel insert returns and are unbalanced, they are wired in the same way as the line outputs.

### 4. Channel Insert Returns

These standard jacks carry the Channel insert returns and are unbalanced. Inserting a jack into the insert return breaks the signal flow from the insert send and replaces it with the signal on the jack plug. They are wired the same as the line outputs.

### 5. Mic Inputs

These carry the electronically balanced Microphone Inputs and are wired as follows:-

Pin 1: Ground  
Pin 2: COLD (Out of phase signal)  
Pin 3: HOT (In phase signal)

## STEREO INPUT CONNECTOR PANEL

---

Inputs are provided for channels A and B Left and Right.  
These XLRs should be wired as follows:-

Pin 1:	Ground
Pin 2:	COLD (Out of phase signal)
Pin 3:	HOT (In phase signal)

## OUTPUT CONNECTOR PANEL (FX Return)

### 1. Aux mix inserts (L & R)

These standard stereo jacks carry the insert sends and returns. The insert send is normalled to the insert return, inserting a jack into the insert return breaks the signal flow from the insert send jack and replaces it with the signal on the jack plug. They are wired as follows:

Tip: HOT  
 Ring: Common ground  
 Sleeve: Common ground.

### 2. Mix Inserts (L&R)

These standard stereo jacks carry the insert sends and returns. The insert send is normalled to the insert return, inserting a jack into the insert return breaks the signal flow from the insert send jack and replaces it with the signal on the jack plug. They are wired the same as the Aux mix insert (L&R) sockets.

### 3. Mix outputs

These are electronically balanced outputs and are wired as follows:-

Pin 1: Ground  
 Pin 2: COLD (Out of phase signal)  
 Pin 3: HOT (In phase signal)

### 4. FX Returns

These are electronically balanced inputs and are wired as follows:-

Pin 1: Ground  
 Pin 2: COLD (Out of phase signal)  
 Pin 3: HOT (In phase signal)

### 5. Group inserts

These standard stereo jacks carry the insert sends and returns. The insert send is normalled to the insert return, inserting a jack into the insert return breaks the signal flow from the insert send jack and replaces it with the signal on the jack plug. They are wired the same as the Aux mix insert (L&R) sockets.

### 6. Group Outputs

The Group Outputs are electronically balanced and are wired in the same way as the Mix outputs.

### 7. Auxiliary mix outputs

These are electronically balanced and are wired the same as the Mix outputs.

### 8. T/B out

This is an electronically balanced output wired the same as the Mix outputs.

### 9. T/B in

This is an electronically balanced input wired the same as the FX returns.

### 10. Cassette inputs (L & R)

These electronically balanced inputs are wired the same as the FX returns.

### 11. Auxiliary outputs

These are electronically balanced outputs and are wired the same as the Mix Outputs.

# OUTPUT CONNECTOR PANEL (Matrix)

### 1. Aux mix inserts (L & R)

These standard stereo jacks carry the insert sends and returns. The insert send is normalled to the insert return, inserting a jack into the insert return breaks the signal flow from the insert send jack and replaces it with the signal on the jack plug. They are wired as follows:

Tip: HOT  
Ring: Common ground  
Sleeve: Common ground.

### 2. Mix Inserts (L&R)

These standard stereo jacks carry the insert sends and returns. The insert send is normalled to the insert return, inserting a jack into the insert return breaks the signal flow from the insert send jack and replaces it with the signal on the jack plug. They are wired the same as the Aux mix insert (L&R) sockets.

### 3. Mix outputs

These are electronically balanced outputs and are wired as follows:-

Pin 1: Ground  
Pin 2: COLD (out of phase signal)  
Pin 3: Hot (In phase signal)

### 4. Matrix outputs

These are electronically balanced outputs and are wired in the same way as the Mix outputs.

### 5. Group inserts

These standard stereo jacks carry the insert sends and returns. The insert send is normalled to the insert return, inserting a jack into the insert return breaks the signal flow from the insert send jack and replaces it with the signal on the jack plug. They are wired the same as the Aux mix insert (L&R) sockets.

### 6. Group Outputs

The Group Outputs are electronically balanced and are wired in the same way as the Mix outputs

### 7. Auxiliary mix outputs

These are electronically balanced and are wired the same as the Mix outputs.

### 8. T/B out

This is an electronically balanced output wired the same as the Mix outputs.

### 9. T/B in

This is an electronically balanced input wired the same as the FX returns.

### 10. Cassette inputs (L & R)

These electronically balanced inputs are wired the same as the FX returns.

### 11. Auxiliary outputs

These are electronically balanced outputs and are wired the same as the Mix outputs.

---

# INSTALLATION

---

## Applying Power

---

Before switching on the Series 8000 check that the mains voltage selector on the power supply unit is set to the correct mains voltage for your area and that the fuse is of the correct rating.

For operation on voltages between 220 and 240Vac, the fuse should be rated at 3.15 amps, 20mm anti-surge.

For operation on voltages between 100 and 120 Vac, the fuse should be rated at 6.3 amp, 20mm.

Do not replace the fuse with any other type, as this could become a safety hazard and will void the warranty.

## Interface levels

---

The Series 8000 is normally supplied to provide compatible level interfacing with standard professional equipment, ie. +4dBu (ref 0.775V).

## Connector Conventions

---

All XLR type connections are normally wired to the following standard:-

Pin 1	GROUND
Pin 2	COLD (Out of phase signal)
Pin 3	HOT (In phase signal)

All main outputs are electronically balanced. The Soundcraft output stage allows either the Hot or Cold to be grounded, without any change in output level or other adverse effects, in the same way as a transformer coupled output. This allows much greater freedom when interfacing to a variety of unbalanced and balanced equipment and a minimisation of earth loop problems.

In most cases, the output can be viewed in the same way as a transformer coupled output, but without the signal degradation inherent in transformer design. However, if the output is driving a long cable run into an unbalanced input, it is usually beneficial to unbalance the output directly at the console to avoid instability.

## General Wiring Procedures

---

To take full advantage of the excellent signal to noise ratio and low distortion of Soundcraft consoles care must be taken to ensure that incorrect installation and wiring does not degrade the performance of the desk. Hum, buzz, instability and Radio Frequency Interference can usually be traced to earth loops and inferior earthing systems. In some areas, especially heavily industrial areas, the incoming mains earth will not be adequate and a separate technical earth for all the audio equipment must be supplied. However, check with your local electricity supply company to ensure that safety regulations are not infringed or negated.

## Installation Notes

The successful, hum free, installation of a system requires forethought, and the establishment of a set of ground rules, which must be consistently adhered to at all stages of installation.

### 1) Initial Wiring Considerations.

a) For optimum performance, it is essential for the earthing system to be clean and noise free, as all signals are referenced to this earth. A central point should be decided on for the main earth point system, and all earths should be "star fed" from this point. It is common electrical practice to "daisy chain" the earths to all electrical outlets but this method is unsuitable for audio installations. The preferred method is to run an individual earth wire from each outlet, back to the system star point to provide a safety earth of screen reference for each piece of equipment.

A separate earth wire should also be run from each equipment rack and area, to the star point. This may or may not be used depending on circumstances, but it is easier to install in the first place, than later when problems arise.

The location of the star point should be a convenient, easily accessible place preferably at the rear of the console, or in the main equipment rack.

b) Install separate "clean" and "dirty" mains outlets, wired individually back to the incoming mains distribution box. Use the "clean" supply for all audio equipment and the "dirty" supply for all lighting, vending machines etc. Never mix the two systems.

c) If necessary, to provide sufficient isolation from mains borne interference, install an isolating transformer for the "clean" supply. The isolation transformer should be provided with a Faraday Shield which must be connected to earth.

d) Never locate the incoming mains distribution box near audio equipment, especially tape recorders, which are very sensitive to electro-magnetic fields.

e) Ensure that all equipment racks are connected to earth, via a separate wire back to the star point.

f) Equipment which has unbalanced inputs and outputs may need to be isolated from the rack to prevent earth loops.

### 2) Audio Wiring

Having provided all equipment with power and earthing connections, consideration must be given to the method of providing audio interconnection and adequate screening of those interconnections. This must be done in a logical sequence to avoid problems and assist in the localisation of problem equipment.

a) Connect Control Room Monitor system to the console and check for any hum, buzz, or RFI. Only when you are satisfied with the quietness of the console and the monitor system should you proceed with the next step.

b) Connect multitrack tape recorder, via the noise reduction system, if in use, and again check that the system is still clean.

c) Connect stereo tape recorders, studio monitors, echo and foldback sends one at a time, checking and isolating any connection which degrades performance.

d) Connect all peripheral devices.

e) Connect all microphone lines.

By following this sequence much time and future trouble will be saved, and the result will be a quiet, stable system.

### 3) Shielding



Audio equipment is supplied with a variety of input and output configurations, which must be taken into consideration when deciding where the screen connections should be made. There are three sources of unwanted signal being impressed on the screen, which are as follows:-

- i Extraneous electrostatic or electromagnetic fields.
- ii Noise and interference on the earth line.
- iii Capacitive coupling between the screen and signal wires.

To minimise the adverse affects of the unwanted coupling to the signal wires, it is important that the screen is connected at one end only, i.e. the screen must not carry any signal current. Any signal on the wires within the screen will be capacitively coupled to the screen. This current will ultimately be returned to the source of the signal, either directly, if the screen is connected at the signal source end, or indirectly via the earthing system, if the signal is connected at the signal destination end. The indirect connection will cause an increase in high frequency cross-talk, and should be avoided wherever possible.

Therefore, in general, always connect the shield only at the signal source end. In high RF areas, the screen can also be connected to earth via a 0.01 micro Farad capacitor. This will present a short circuit at RF frequencies, thus lowering the effective shield impedance to ground. However, at low audio frequencies the reactance of the capacitor will be sufficiently high not to cause an earth loop problem.

Combinations of unbalanced, balanced and electronically balanced, (differential), systems mean that there are nine interconnection permutations. The optimum of the screen in each case is shown in Table 1.

**TABLE 1**

1	Unbalanced	Unbalanced	Source
2	Unbalanced	Balanced	Source
3	Unbalanced	Differential	Source
4	Balanced (Note 1)	Unbalanced	Destination
5	Balanced	Balanced	Source
6	Balanced (Note 2)	Differential	Destination
7	Differential (Note 3)	Unbalanced	Source
8	Differential	Balanced	Source
9	Differential	Differential	Source

Note 1 -The shield is connected to the destination earth point, which is opposite to normal practice, because the signal wires being shielded are referenced to the input earth, not the output earth.

Note 2 -If the output transformer is centre tapped to earth, the screen should be connected at the source.

Note 3 -When an active differential output is operated in unbalanced mode, it is very important that the output current returns to earth via the shortest, least reactive route. Check for instability at the output.

## N.B.

a)In all cases, use good quality twin screened audio cable. Check for instability at the output.

b)Always connect both conductors at both ends, and ensure that the screen is only connected at one end.

c)Do not disconnect the mains earth from each piece of equipment. This is needed to provide both safety and screen returns to the system star point.

## Installation Notes

d) Equipment which has balanced inputs and outputs may need to be electrically isolated from the equipment rack and/or other equipment, to avoid earth loops.

It is important to remember that all equipment which is connected to the mains is a potential source of hum and interference and may radiate both electrostatic or electromagnetic radiation. In addition, the mains will also act as a carrier for many forms of RF interference generated by electric motors, air-conditioning units, thyristor light dimmers etc. Unless the earth system is clean, all attempts to improve hum noise levels will be futile. In extreme cases there will be no alternative but to provide a completely separate and independent "technical earth" to replace the incoming "noisy earth". However, always consult your local electricity supply authority to ensure that safety regulations are not being infringed.

---

## MAINTENANCE

---

Every console that leaves Soundcraft undergoes a thorough testing at all stages of manufacture. These tests include a thorough testing of all the functions of the completed mixer, which consist of listening, measuring and mechanical function checks prior to packaging and shipment. In this way we try to ensure that any faulty components or manufacture show up long before the console leaves the company. Thus a long and trouble-free life can be expected.

Although all Soundcraft Consoles have been designed with long term reliability in mind, it is inevitable that occasional maintenance will be required. However, due to the amount of attention given to the problems of maintenance during the design stages of this console and the modular construction, servicing tends to be extremely simple, with the minimum of test equipment needed to isolate and rectify faults.

### General Fault Finding

---

With the exception of the electronically balanced microphone amplifier and the hybrid discrete/op amp summing amps, all signal electronics are configured around high slew rate, low noise integrated circuits. The microphone amplifier is a proprietary design, utilizing a discrete transistor, noise cancelling front end, differentially summed via a low noise integrated circuit.

The use of integrated circuits means that the majority of audio faults can be repaired by simply replacing the I.C., having first isolated the fault to a particular stage in the signal chain. The isolation can often be done without even having to remove the module from the console, by judicious use of insert points and/or switching the module to various modes. As with all servicing a good knowledge of the basic signal flow is necessary for best results. Each module should be viewed as a number of signal blocks, through which the signal must flow. If the signal appears at the input to a block, but not at the output, then the fault lies within that block. By dividing a module into individual sections, what at first appears to be an extremely complicated piece of equipment can be simplified into a series of sequential stages. This is the basic first move in all types of fault finding and usually requires no more than a certain amount of logical thought. Servicing a console is more a matter of clear thinking and having an understanding of what should be happening, than having a highly developed technical knowledge.

To illustrate the method of logical fault finding, let us assume that we have a non functioning input module, in both microphone and line modes.

The first step is to ensure that a fault really does exist!

Check that the module is in the correct mode of operation and that no jacks are inserted in the insert points, which may be interrupting the signal flow.

If in doubt about the module operation, set up an adjacent module in exactly the same way, which will allow a direct comparison between a working and possible non-working module.

Route the channel directly to MIX, so that the channel may be monitored in the normal way. Using an oscillator set it to approximately 1kHz and patch the oscillator signal into the channel Line Input. If all is well, an undistorted signal should now be heard. More likely, because of the fault it won't.

Large sections of the module circuitry can be by-passed by switching out the Hi-pass filter and the equalizer.

If switching out a section causes the signal to re-appear, then the fault is located in that section, which can then be traced at component level, by removing the module from the console frame and reconnecting it via extender cables.

With the module installed on extender cables, access is now available to all parts of the module and the signal may be traced through the various stages using an oscilloscope, millivoltmeter or even high impedance headphones. Refer to the Block Schematic which shows the signal flow through the modules. When a point is reached where the signal is not present or is distorted, the probable faulty components can be checked out and if necessary replaced. Integrated circuits, due to their internal complexity, are the most likely cause of problems, followed by mechanical components such as switches and faders, which are susceptible to physical contamination from oxidation, dust and liquids.

## Removing Modules

Before removing modules, always switch off the console.

Remove the 2 module retaining screws, which will allow the module to be carefully withdrawn from the console. The ribbon cable will now be exposed and may be detached from the module. The module will still have some cables attached, but these are sufficiently long to allow the module to be completely withdrawn from the console. Extender cables can now be plugged into the main ribbon cable and the module, taking care not to twist the extender cable. A module should NOT be unplugged or plugged in with the power ON.

## Meter Alignment

Each VU meter has its own individual drive card attached to the rear of the meter.

0VU is normally adjusted to indicate a line level of +4dBu ie. a level of 1.228 volts. However, it can be re-adjusted to indicate a different line level if required by the pre-set potentiometer on the drive card.

Connect a millivoltmeter to the group output. Route the oscillator set to 1kHz to the group and adjust the group output level to read the required level on the millivoltmeter. (Normally this would be +4dBu). Adjust the VU drive pre-set to indicate 0VU on the VU meter and repeat for all other groups and the stereo mix meters.

## Lamp Replacement

Illumination of the VU meters is provided by 2 wire ended lamps in each meter. These are 12 volt lamps wired in series. The 100 Ohm series resistor provides turn on surge current limiting to prolong lamp life.

To replace the lamps first remove the meter bridge. (The retaining screws are located on the front of the meter bridge.) The lamps are located on top of the meter drive card. New bulbs can now be soldered in place.

# SOUNDCRAFT RECOMMENDED WARRANTY

**This warranty applies to sales within the UK and should form the basis of the warranty offered by the overseas vendor of Soundcraft products.**

▣ **1.**

**'Soundcraft'** means Soundcraft Electronics Ltd.

**'End User'** means the person who first puts the equipment into regular operation.

**'Dealer'** means the person other than Soundcraft (if any) from whom the End User purchased the Equipment, provided such a person is authorised for this purpose by Soundcraft or its accredited Distributor.

**'Equipment'** means the equipment supplied with this manual.

▣ **2.**

If within the period of twelve months from the date of delivery of the Equipment to the End User it shall prove defective by reason only of faulty materials and/or workmanship (but not faulty design) to such an extent that the effectiveness and/or usability thereof is materially affected the Equipment or the defective component should be returned to the Dealer or to Soundcraft and subject to the following conditions the Dealer or Soundcraft will repair or at its option replace the defective components. Any components replaced will become the property of Soundcraft.

▣ **3.**

Any Equipment or component returned will be at the risk of the End User whilst in transit (both to and from the Dealer or Soundcraft) and postage must be prepaid.

▣ **4.**

This warranty shall only be available if:-

- a) the Equipment has been properly installed in accordance with instructions contained in Soundcraft's manual; and
- b) the End User has notified Soundcraft or the Dealer within 14 days of the defect appearing; and
- c) no persons other than authorised representatives of Soundcraft or the Dealer have effected any replacement of parts maintenance adjustments or repairs to the Equipment; and
- d) the End User has used the Equipment only for such purposes as Soundcraft recommends, with only such operating supplies as meet Soundcraft's specifications and otherwise in all respects in accordance with Soundcraft's recommendations.

▣ **5.**

Defects arising as a result of the following are not covered by this Warranty: faulty or negligent handling, chemical or electro-chemical or electrical influences, accidental damage, Acts of God, neglect, deficiency in electrical power, air-conditioning or humidity control.

▣ **6.**

The benefit of this Warranty may not be assigned by the End User.

▣ **7.**

End Users who are consumers should note their rights under this Warranty are in addition to and do not affect any other rights which they may be entitled against the seller of the Equipment.