Series 400B
User Manual

Soundcraft Electronics Ltd.
Unit 2, Borehamwood Industrial Park,
Rowley Lane,
Borehamwood, Herts. WD6 5PZ England.
Tel: 01-207-5050.
Telex: 21198
Facsimile: 01-207-0194

Soundcraft Electronics USA.
1517, 20th Street,
Santa Monica,
California. 90404.
Tel: (213) 4534591
Telex: 664923.
Facsimile: (213) 453 5634.

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

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

SERIES 400B

CONTENT | PAGE

1.00 SERIES 400B

Model No. .... FRAME SIZE ....

Serial No. .... PSU Serial No. .... as used in final test

Original Customer .... Works Order No. ....

PROGRESS	NAME	DATE	SUPERVISORS INSPECTION
Frame Fitted by			
Assembled/Wired by			
First Test By			
Final Test By			
Despatch Inspection By			

EQUIPPED WITH	TYPE	QUANTITY ISSUE
Input Modules		
Output Modules		
Other Modules		
P.S.U.		

OPTIONS - SPECIFY

ALTERATIONS TO SPEC.

SPECIAL INSTRUCTIONS

DESPATCH REQUIRED

2.00 (SERIES 400B CONSOLE DESCRIPTION)

2.01 General Description

The series 400B is a fully modular 4 - bus mixing console and is available in two specifications both available in 24/4/2 and 16/4/2 formats.

Standard

Module types:

4001 Standard Input
4002 Standard Output
4005 Standard Master
4006 Effects Return

Meter Display:
4 VU meters reading outputs 1-4 individually switchable to read 5-8 plus 2VU meters for monitor source (mix/2 track replay/solos)

N.B.: An option is available for 4 extra VU meters.

Monitor

Module types:

4003 Monitor Input
4004 Monitor Output
4008 Monitor Master
4006 Monitor return

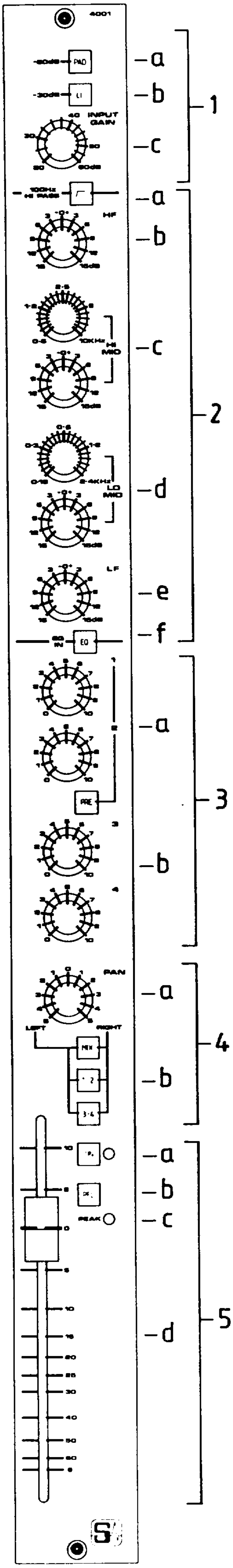
Meter Display:

8VU meters for monitor outputs, 2 VU meters for monitor source (mix/2 track reply/solos)

Power Supply

For both specifications power is supplied from external unit supplying +17V/-17V/+48V.

4001
STANDARD INPUT MODULE



2.02 4001 STANDARD INPUT MODULE

1. CHANNEL INPUT SECTION

The channel can be operated in either the microphone or line input modes.

The microphone input is an electronically balanced, transformerless design, configured for optimum low noise performance. Electronic balancing reduces the degradation of signal quality which is introduced by the more normal transformer coupled designs, and endures superior transient response, minimal phase shift, and excellent common mode rejection, even with high frequence interference. The input impedance is greater than 2kOhms, which will not cause any loading effects on any normally used studio microphone. The high level line input is electronically balanced, with an input-impedance of greater than 10kOhms which is high enough to interface to any normal professional peripheral equipmment, without loading the source.

a) **Pad**

Depressing the PAD button inserts a 20dB attenuator into the input of the microphone amplifier, and 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. Direct injection boxes are also capable of providing high signal levels.

b) **LI (Line Input)**

The high level input from a stereo type jack socket, is selected by depressing the LI button. Tape returns should be replugged into the line input sockets for mix down purposes.

c) **Input Gain Trim**

The microphone input can be varied between 20dB and 60dB of gain. Used in conjunction with 20dB PAD, a 60dB control range is available. The line level input can be varied between -10dB and 30dB

2. EQUALISER SECTION

The equaliser is an exceptionally 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, i.e., 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. 15dB of boost or cut is available. The response is of the "bell" type, i.e., 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 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 variable between 150Hz and 2.4kHz.

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 or out of the signal path, independent of the high pass filter.

3. AUXILIARY SECTION

There are 4 auxiliary send controls available 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. In both cases, they are post equaliser, insert point, and channel on/off switch.

b) **Sends 3 and 4**

Auxiliary sends 3 and 4 are permanently post-fader.

4. ROUTING SECTION

The channel input signal can be routed to any or all of the 4 group Outputs and the stereo Mix, by selecting the relevant routing button.

a) 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) Selection of any routing button assigns the channel signal to a pair of output groups, or to the stero mix, via the pan pot.

5. CHANNEL STATUS SECTION

a) 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 soloes the pre-fader, post insert jack signal, independently of the ON switch.

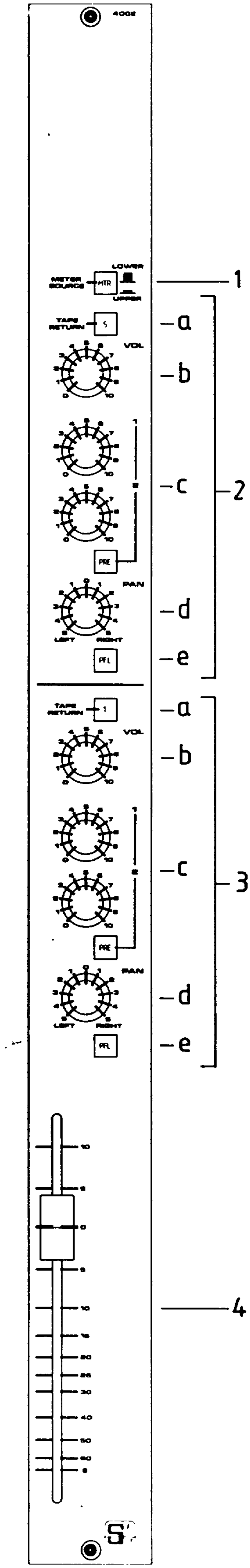
c) **Peak**

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

d) **Channel Fader**

The channel fader has a slide length of 100mm and an exceptionally smooth feel. Infinity cut off is greater than 90dB.

4002
STANDARD OUTPUT MODULE



2.03 4002 STANDARD OUTPUT MODULE

The 4002 Recording Output contains a group section and 2 tape return monitor channels. The lower monitor section is used to monitor either its associated group output, or one of the tape return tracks 1 to 4 depending on its position in the console. The upper monitor section is used to monitor one of the tape return tracks 5 to 8. Additionally, this section can also be used as a line level input during remixing, with its signal panned into the main stereo mix.

1. MTR (METER SOURCE)

The signal that is fed to the meters is selected by this switch. In the out position the meter is fed from the lower section and when depressed is fed from the upper section.

NB An option is available to provide 4 extra meters enabling all 8 monitor sections to be shown simultaneously.

2. UPPER MONITOR SECTION (Tracks 5 - 8)

a) Tape Return

The tape return switch selects the monitor source between either tape send (group output) or tape return (monitor input).

b) Vol

The monitor volume control enables the monitor contribution to the stereo mix to be adjusted in level, to allow a satisfactory monitor balance to be achieved during recording or playback of the multitrack tape.

c) Auxiliary Sends 1 and 2

Auxiliary sends 1 and 2 are normally post-fader, but can be switched pre-fader by selecting PRE. In both cases, the auxiliary signal is derived after the monitor ON switch. These are used to provide headphone mixes during recording and overdubbing.

d) Pan

The pan control allows the monitor signal to be panned to the required position within the stereo mix.

e) PFL (Prefade Listen)

PFL the monitor channel signal independently of the Vol control.

3. LOWER MONITOR SECTION (Tracks 1 to 4)

The lower monitor section is similar in operation to the upper section, and provides monitoring of tracks 1 to 4.

4. GROUP OUTPUT

Each group output controls the signal level to 2 tracks of the multitrack recorder during the recording process i.e., group 1 will feed tracks 1 to 5, group 2 will feed tracks 2 to 6 etc. the group output signal is then normally monitored by the relevant upper or lower monitor section.

2.04 4005 STANDARD MASTER MODULE

This module contains the stereo output as well as the four unbalanced auxiliary outputs.

It also contains the controls for the monitoring facilities and the talkback system.

1. PHANTOM POWER

Capacitor microphones can be powered by the internal 48 Volt phantom power supply by depressing the 'ON' button. This supplies power to all mic inputs. Operation is indicated by a red LED.

N.B. When using direct injection boxes or unbalanced sources, the phantom power supply should not be switched on.

Individual channel phantom power switching is available as an option.

2. AUXILIARY MASTERS 1-4

Each of the 4 auxiliary busses has an overall master level control. An associated AFL (after fade listen) button allows the signal at the auxiliary output to be monitored and metered.

3. OSCILLATOR

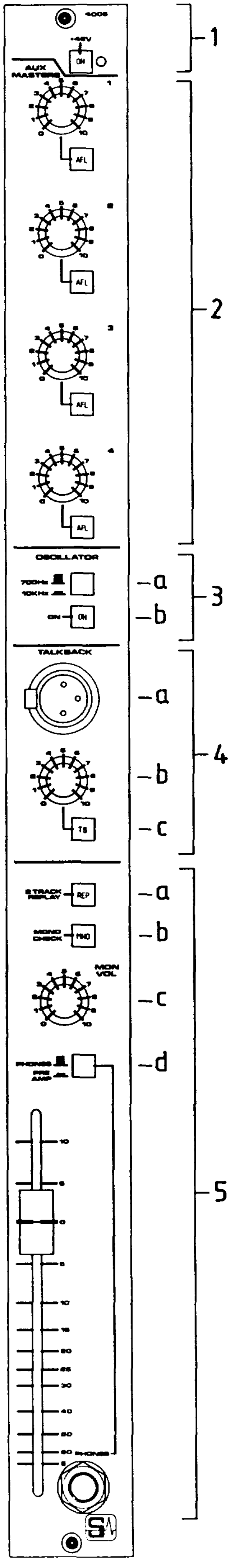
a) Frequency

The oscillator frequency can be changed between two set values of 700HZ when the button is 'out' and 10KHZ when the button is depressed.

b) ON

Operation of this routes the oscillator signal to all the auxiliary and group busses.

4005
STANDARD MASTER MODULE



4. **TALKBACK**

a) **Input XLR**

This provides the input point for a dynamic microphone.

b) **Mic Level**

The level of the talkback mic signal can be fully adjusted.

c) **TB**

Depressing this button routes the talkback signal to all aux and group busses.

5. **Monitor Section**

a) **REP**

Depressing REP will monitor the electronically balanced 2 track return.

b) **MNO**

To provide a compatibility check of the stereo signal, the left and right monitor channels can be summed together by the MNO switch.

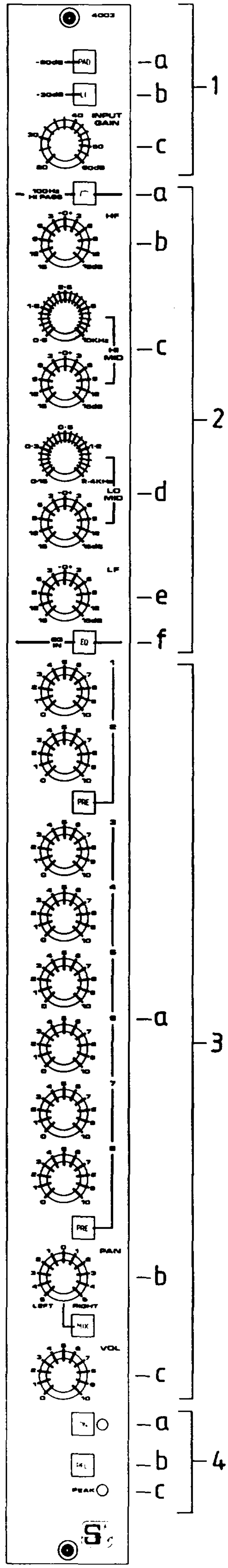
c) **Monitor Level**

The signal level to the headphones socket can be adjusted.

d) **Phones/Pre Amp**

The headphones output allows monitoring of the main stereo mix output, also of any soloed signal. Depressing the button decreases the signal by 8dB, to allow the use of an external power amplifier to drive monitor loudspeakers.

4003
MONITOR INPUT MODULE



2.05 4003 MONITOR INPUT

1. CHANNEL INPUT SECTION

The Channel can be operated in either the microphone or line input modes.

The microphone input is an electronically balanced, transformerless design, configured for optimum low noise performance. Electronic balancing reduces the degradation of signal quality which is introduced by the more normal transformer coupled designs, and ensures superior transient response, minimal phase shift, and excellent common mode rejection, even at high frequencies. This results in excellent immunity to Radio Frequency interference. The input impedance is greater than 2KOhms which will not cause any loading effects on any normally used studio microphone. The high level line input is electronically balanced, with an input impedance of greater than 10K ohms, which is high enough to interface to any normal professional peripheral equipment, without loading the source.

a) PAD

Depressing the PAD button inserts a 20dB attenuator into the input of the microphone amplifier, and 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. Direct injection boxes are also capable of providing high signal levels.

b) LI (Line Input)

The high level line input is selected by depressing the LI button, with access via al/4 inch stereo jack on the rear panel.

c) Input Gain Trim

The microphone input can be varied between 20dB and 60dB of gain. Used in conjunction with the 20dB PAD, a 60dB control range is available.

The line input can be varied between -10dB and 30dB.

2. EQUALISER SECTION

The equaliser is a versatile unit, allowing 5 areas of control to be exercised 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 effectively removes low frequency stage rumble, and other extraneous signals.

b) **HF (High Frequency)**

15dB of boost or cut is available at 100KHz, with a "shelving" characteristic, i.e., the slope of the 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. 15dB of boost cut is available. The response is of the "bell" type, i.e., having reached maximum amplitude (or minimum in the case of cut) at the selected 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 variable between 150Hz and 2.4KHz.

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 or out of the signal path, independent of the high pass filter.

3. MONITOR SECTION

a) The 8 monitor sends are normally post the channel level control, they can however be selected pre-fader. It is possible to select sends 1-2 independently of 3-8 or vice versa. It is of course possible to select all 8 as pre-fader.

b) The channel signal can be indpendently routed to the stereo mix and panned between left and right.

c) The channel signal can be varied by the rotary channel fader.

4. CHANNEL STATUS SECTION

a) **ON**

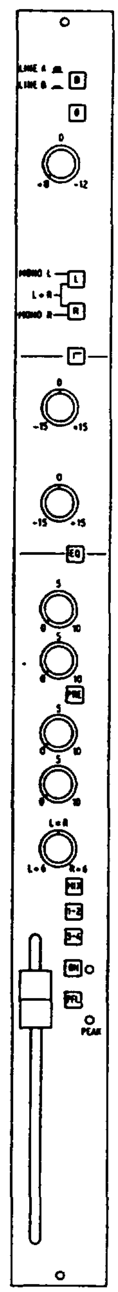
The channel "ON" mode is indicated by a green LED. When a channel is switched off, all auxiliary sends are also switched off, with the exception of the signal to the insert send jack.

b) **PFL (Pre Fade Listen)**

PFL soloes the pre-fader, post insert jack signal, independently of the ON switch.

c) **Peak**

A red LED indicates the peak signal level at the output of the equaliser. It illuminates at a level of 4dB below clipping.



S400B Stereo Input Module

DRG No MI 2289

NOTES.

HOLE INDEX

TOLERANCE.
All imperial dimensions ±0·010
All metric dimensions ±0·25mm
All angles ±0·50°
Unless otherwise stated.

THIRD ANGLE PROJECTION

MATL.	—
FINISH.	—
SCALE.	Full size
DRG.	Barry
TRCD.	
CHKD.	

SOUNDCRAFT ELECTRONICS LTD
5-8 GREAT SUTTON STREET
LONDON EC1V 0BL
TELEPHONE 01-251-3631/2/3
TELEGRAMS SOUNDCRAFT LDN EC1
TELEX UK No 21198 USA No 01-2303

TITLE S400B STEREO INPUT MODULE

DRG No MI 2289

Issue 1 31-1-83
Issue 2 start button deleted 19-4-83

2.05 (b) 4009 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) **Input Gain**

The Input Gain can be varied between -12dB and +8dB 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 depressed, both channels of the module are fed by a mono sum of the left and right input.

2. EQUALISER SECTION

The Equaliser allows 3 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 12kHz, with a "shelving" characteristic, i.e. 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) **LF (Low Frequency)**

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

d) **EQ**

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

3. AUXILIARY SECTION

There are 4 Auxiliary send controls available 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 swtched Pre-fader by pressing the appropriate PRE button. In both cases they are post equaliser. Auxiliaries 1 and 2 can be selected mono or stereo and dependant/independant of the channel mute by push-on links located on the PCB.

b) **PRE**

Auxiliary sends 1-2 may be routed Pre-fader by pressing the PRE button.

c) **Sends 3 and 4**

Auxiliary sends 3 and 4 are permanently post-fader. Sends 3 and 4 can also be selected mono or stereo by push-on links.

4. ROUTING SECTION

The channel input signal may be routed to any of the pairs of Group Outputs (1-2, 3-4) or 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 the auxiliary sends 1 and 2 are not necessarily cut and depend on the set up of the push-on links (jumpers).

b) PFL

Pre-fade Listen Soloes the Pre-fader 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.

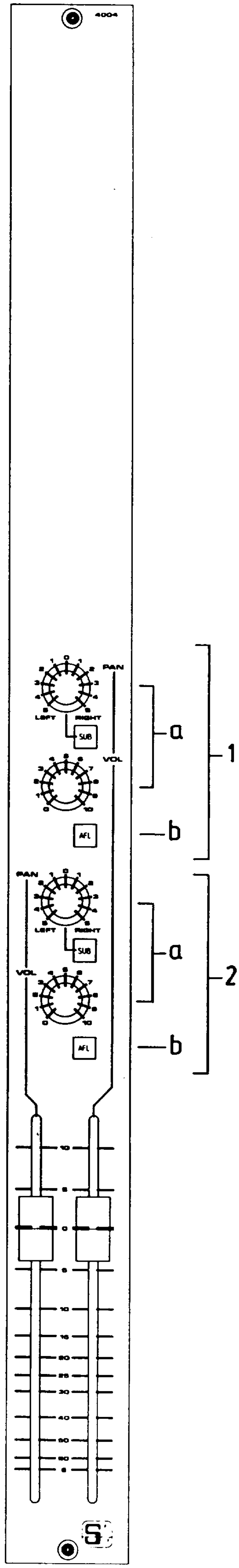
c) Peak

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 stereo fader of slide length 100mm and an exceptionally smooth feel. Infinity cut off is greater than 90dB.

4004
MONITOR OUTPUT MODULE



2.06 4004 MONITOR OUTPUT

The 4004 Monitor output is a double module, although physically the same width as a single module, providing two totally independent outputs. The modules are configured such that the lower sections provide outputs 1,3 5 and 7 while the upper sections provide outputs 2,4,6 and 8. Each output is provided with its own meter.

1. UPPER SECTION

a) SUB

This places the output of the volume pot into the stereo mix via the pan pot.

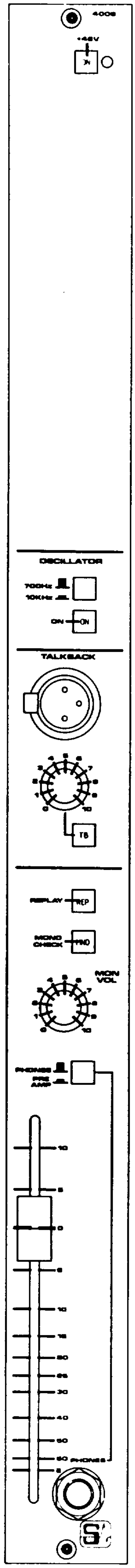
b) AFL (After Fade Listen)

Group output signal can be soloed.

2. LOWER SECTION

The lower section is similar in operation to the upper.

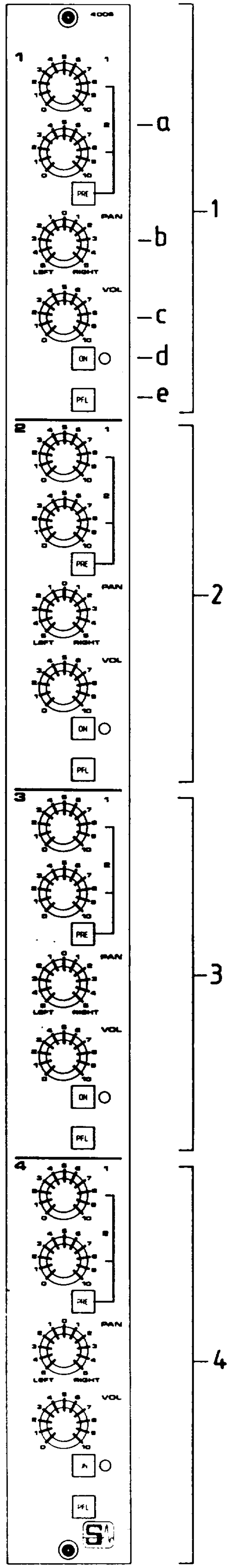
4008
MONITOR MASTER
MODULE



2.07 4008 MONITOR MASTER

This is similar in all operation with the exception that the auxiliary's are not present on a monitor desk.

4006
EFFECTS RETURN
MODULE



2.08 4006 EFFECTS RETURN MODULE

This module contains 4 identical sections for controlling signals from external effects devices.

1. SECTION 1

a) Auxiliary Sends 1 and 2

Auxiliary Sends land 2 are normally post-fader, but can be switched pre-fader by selectign PRE. In both cases, the auxiliary signal is derived after the ON switch.

b) PAN

The pan control allows the signal to be panned to the required position within the stereo mix.

c) VOL

The volume control enables the contribution to the stereo mix to be adjusted in level.

d) ON

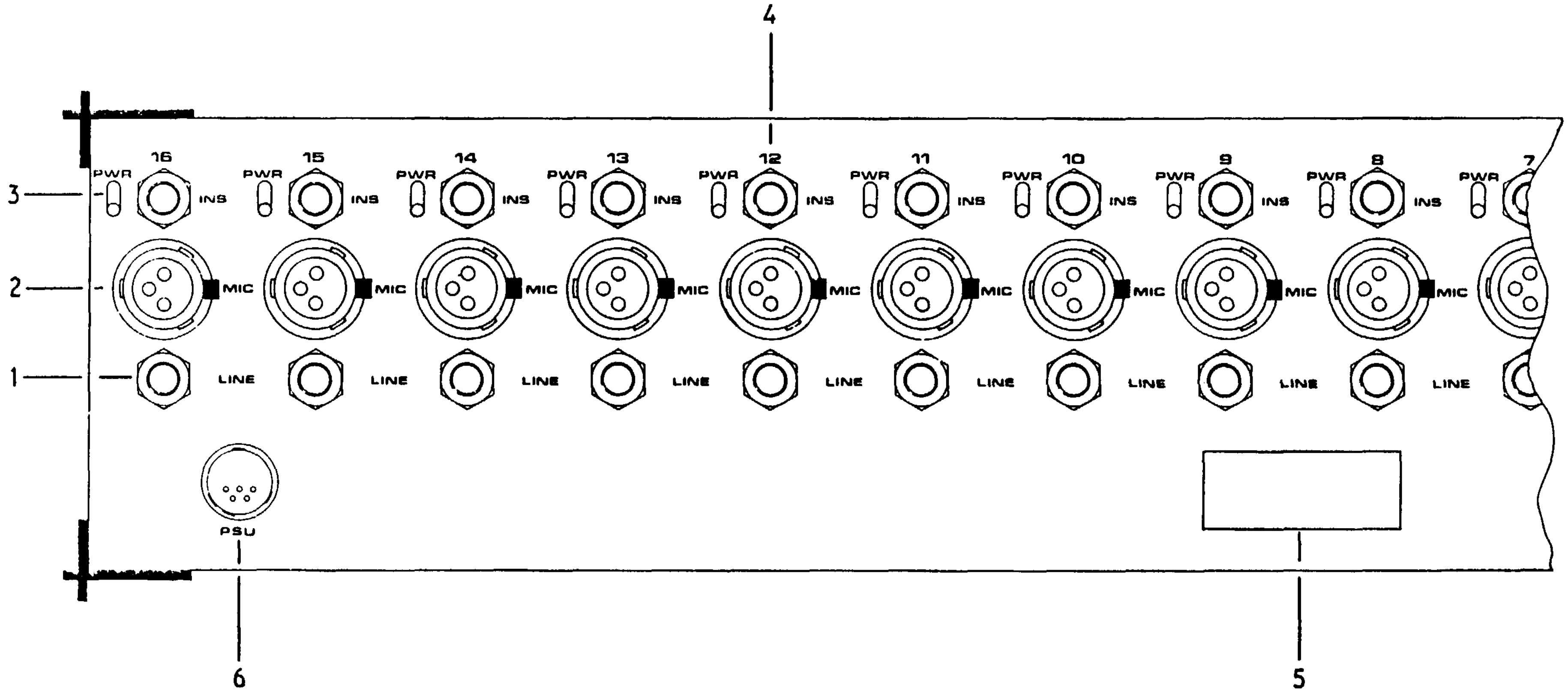
The return channel and associated auxiliary sends are activated by depressing the "ON" button. Operation is indicated by an adjacent green LED.

e) PFL

The return signal can be soloed independent of the "ON" switch.

SECTIONS 2, 3 AND 4

These sections are similar in operation to section 1.


4
3
2
1
16 15 14 13 12 11 10 9 8 7
PWR
INS
MIC
LINE
PSU
6
5

2.09 INPUT CONNECTOR PANEL

This panel is common on both standard and monitor consoles.

1. **Line Inputs**

   These stereo standard jacks carry the balanced line inputs. They are wired with the tip as signal high, the ring as signal low and the sleeve as earth.

   These connections are not normalled from the tape returns.

2. **Mic Inputs**

   These carry the balanced microphone inputs. They are wired with pin 3 as signal high, pin 2 as signal low and pin 1 as earth.

   They can be fed with +48V phantom power for capacitor microphones. This is controlled by a master phantom switch on the master module.

   N.B. The phantom power should not be turned on when an unbalanced microphone or direct inject box is used.

3. **Phantom Power Switching**

   In addition to the master phantom switch individual channel phantom switches can be fitted as an option.

4. **Channel Inserts**

   These stereo standard jacks carry both the insert send and insert return signals. They are wired with the tip as insert return, the ring as insert send and the sleeve as earth.

   Under normal conditions with nothing inserted the signal is normalled through by the jack socket and thus inserting a jack will automatically break this link.

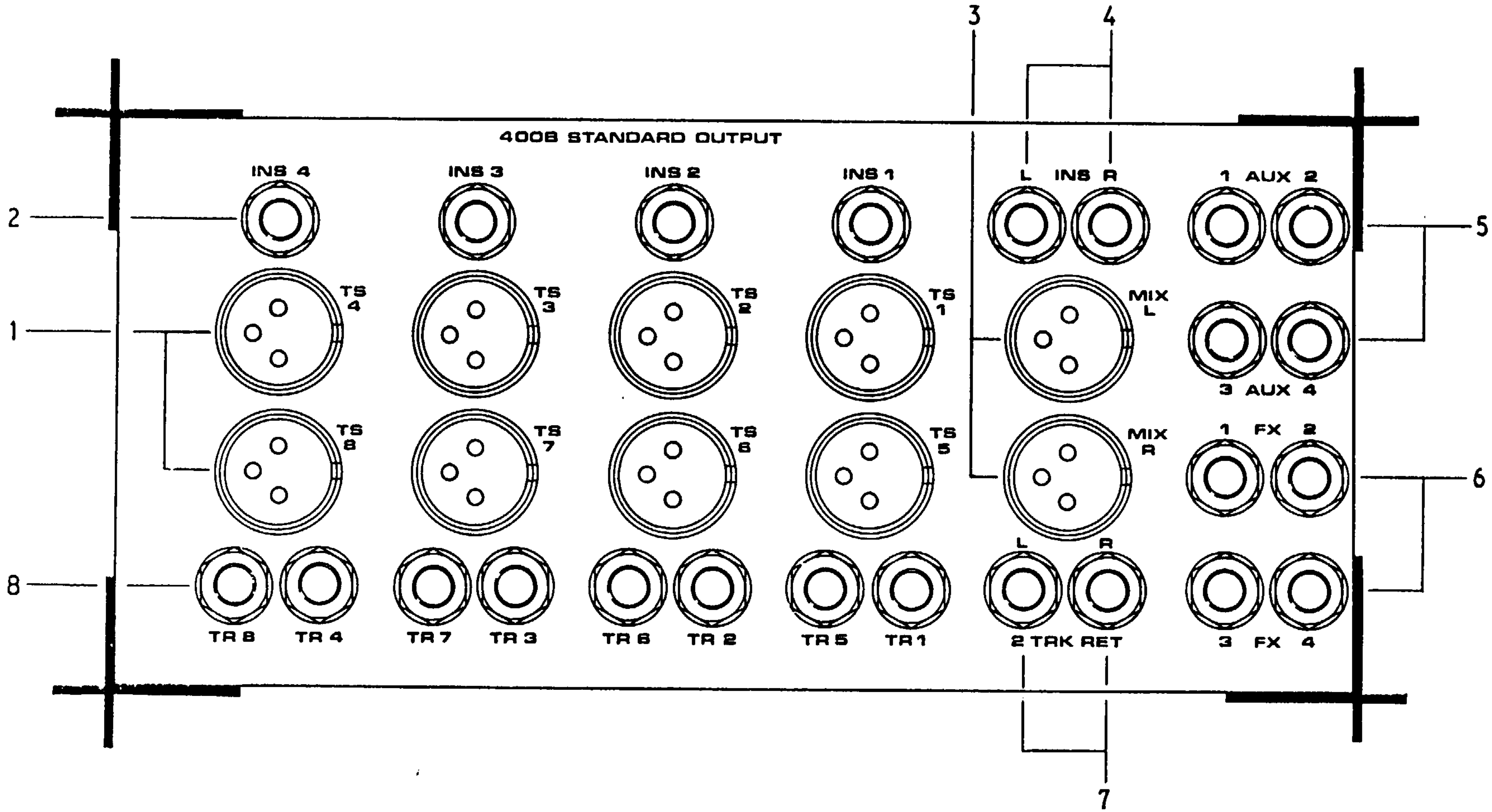
   The Insert is post eq. but is before the channel fader and is independent of the channel on switch.

5. **Multicore Fixing**

   This provides the facility to mount a 56 way Varelco connector which would normally be wired to carry the microphone inputs.

6. **PSU Connector**

   This provides the d.c. power to the console from the external power supply. Please ensure cable connector is screwed fully in to alleviate any chance of intermittent contact.


3
4
4008 STANDARD OUTPUT
INS 4
INS 3
INS 2
INS 1
L INS R
1 AUX 2
2
5
TS 4
TS 3
TS 2
TS 1
MIX L
1
3 AUX 4
TS 8
TS 7
TS 6
TS 5
MIX R
1 FX 2
6
L
R
8
TR 8
TR 4
TR 7
TR 3
TR 6
TR 2
TR 5
TR 1
2 TRK RET
3 FX 4
7

2.10 STANDARD OUTPUT CONNECTOR PANEL

1. **Tape Sends**

   These are fed direct from the group outputs. They are connected in pairs, i.e. group 1 feeds tape sends 1 and 5 etc. These outputs are unbalanced with pin 3 as signal.

2. **Group Inserts**

   This is a stereo, standard break point jack socket which contains both insert send and return. The socket is connected with the tip as insert return and the ring as insert send. The sleeve is, of course, ground. Under normal conditions i.e. with socket not in use the send is connected to return by the socket such that inserting a jack automatically breaks this link.

   The insert point is before the group fader.

3. **Mix Outputs**

   These sockets carry the main stereo outputs. These outputs are unbalanced with pin 3 as the signal.

4. **Mix Inserts**

   These are identical in form and operation to the group insert points.

5. **Auxiliary Outputs**

   These are standard jack socket outputs for the auxiliary masters. These outputs are unbalanced.

6. **Effect Returns**

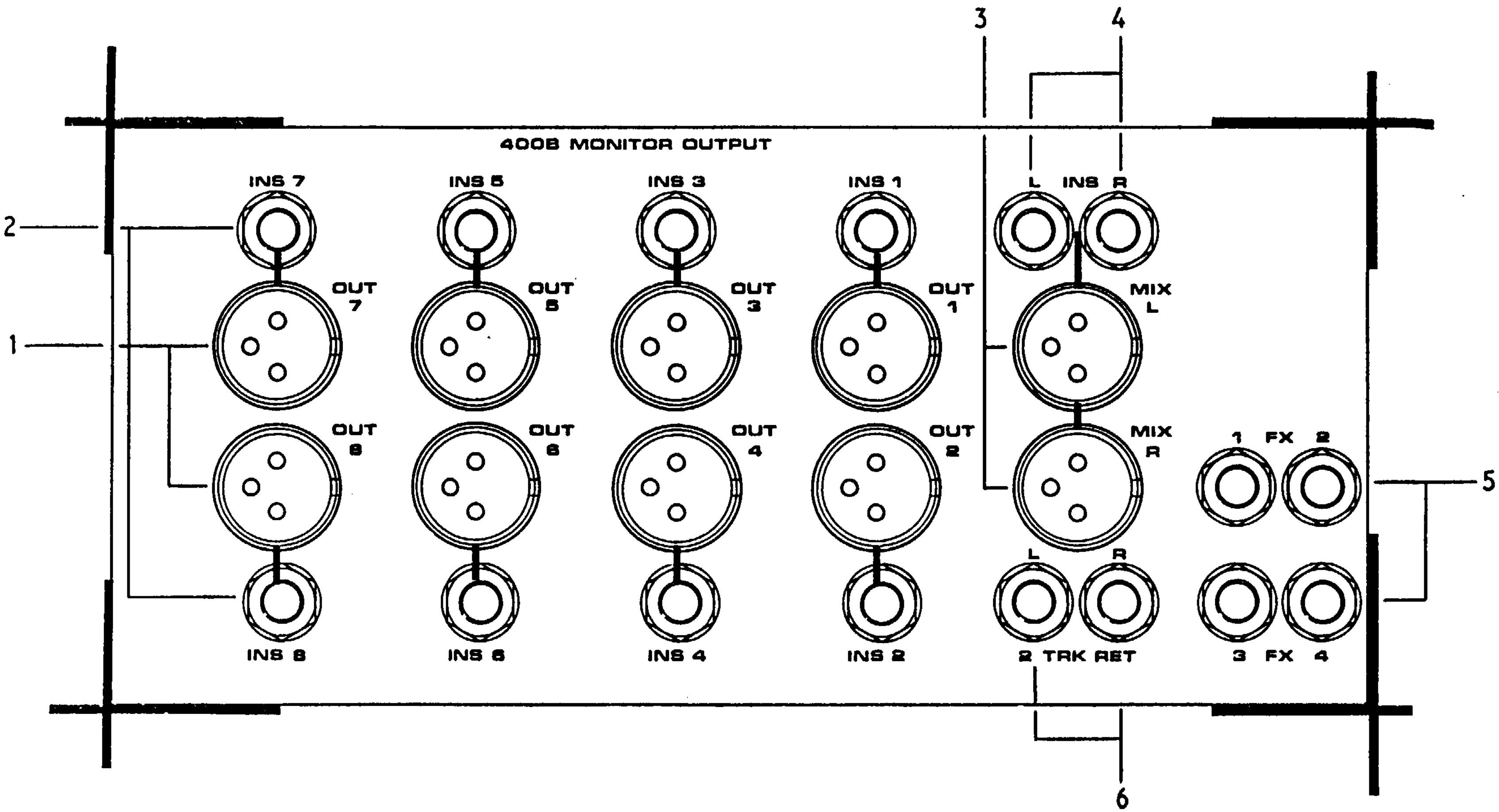
   These stereo standard jack sockets carry the balanced inputs for the optional 4006 Effects Returns Module. These are wired with the tip as the signal high, the ring as the signal low and the sleeve as earth.

7. **2-Track Return**

   These stereo standard jacks carry the balanced 2-track returns. These are wired with the tip as signal high, the ring as signal low and sleeve as earth.

8. **Tape Returns**

   These stereo standard jack sockets carry the balanced tape returns and are wired with the tip as signal high, the ring as signal low and the sleeve as earth.


4008 MONITOR OUTPUT
INS 7
INS 5
INS 3
INS 1
L INS R
OUT 7
OUT 5
OUT 3
OUT 1
MIX L
OUT 8
OUT 6
OUT 4
OUT 2
MIX R
1 FX 2
L
R
INS 8
INS 6
INS 4
INS 2
2 TRK RET
3 FX 4
1
2
3
4
5
6

2.11 MONITOR OUTPUT CONNECTOR PANEL

1. **Group Sends**

   These are fed from the eight individual group outputs. The outputs are unbalanced with pin 3 as the signal.

2. **Group Inserts**

   This is a stereo, standard break point jack socket which contains both insert send and return. The socket is connected with the tip as insert return and the ring as insert send. The sleeve is, of course, ground. Under normal conditions i.e. with the socket not in use the send is connected to return by the socket such that inserting a jack automatically breaks this link.

   The insert point is before the group fader.

3. **Mix Outputs**

   These sockets carry the main stereo outputs. These outputs are unbalanced with pin 3 as the signal.

4. **Mix Insert**

   These are identical in form and operation to the group insert points.

5. **Effect Returns**

   These stereo standard jack sockets carry the balanced inputs for the optional 4006 Effects Returns Module. These are wired with the tip as the signal high, the ring as the signal low and the sleeve as earth.

6. **2-Track Return**

   These stereo standard jacks carry the balanced 2-track returns. These are wired with the tip as signal high, the ring as signal low and sleeve as earth.

3.00 INSTALLATION

3.01 APPLYING POWER

Before switching on the series 400B check that the mains voltage selector on the power unit is set at 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 1.60 amps. 10mm Anti surge.

For operation on voltage between 100 and 120VAC, the fuse should be rated at 3.15amp, 20mm Anti surge.

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

3.02 INTERFACE LEVELS (STANDARD CONSOLE)

The Series 400B is normally supplied to provide compatible level interfacing with standard professionsal equipment i.e., +4dBr (ref 0.775v volt). However, provision has been made to allow the user to modify the tape interface levels for use with semi-professional equipment, operating at -10dBV (ref 1.00 Volt), or other levels.

The change in tape interface operating level is accomplished by reducing the console group output level by the required amount, and increasing the console monitor return gain by an identical amount, and can be done by the addition of 1 resistor, the removal of a link and changing 4 resistors on each standard output circuit board.

Group Output Level

1. Cut track across rear of R11.

2. Add resistor in position RX1. Recommended value is 750 ohms.

Monitor Return Gain

For interfacing to the normal Teac/Tascam operation level of -10dBV R14, 15, 31 and 32 should be changed to 22kOhms in value.

Auxiliary Outputs

Power amplifiers are often rated at 300mV sensitivity, for full output. In such cases, an attenuator should be installed at the input of the power amplifier, to attenuate the +4dBr (1.228V) signal from the console, by approximately 10 to 15 dB. This can be achieved by using a 2.2kOhm series resistor, and a 680 ohm, shunt resistor across the amplifier input.

3.03 CONNECTOR CONVENTIONS

All XLR type connections are normally wired to the following standard

Pin 1	Signal Earth	(Screen)
Pin 2	Signal Low	
Pin 3	Signal High	

On Series 400B all main outputs are unbalanced.

3.04 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 its performance. Hum, buzz, instability and Radio Frequency Interference (RFI) can usually be traced to earth loops and inferior earthing systems. In some areas, especially heavy 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 rgulations are not infringed or negated.

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

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

b) Connect multitrack tape recorder, via noise reduction system if applicable, 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 effects devices.

e) Connect all microphone lines.

By following this sequence much time annd 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 the 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, and 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 entire earthing system, if the screen 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 R.F. areas, the screen can also be connected to earth via a 0.01 micro farad capacitor. This will present a short circuit at R.F. frequencies, thus lowering the effective shield impedance to ground. However,, at low audio frequencies the reactance of the capacitor will be sufficiently high to not 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 on Table 1.

TABLE 1

	OUTPUT	INPUT	SCREEN
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 designation 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) Install 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.

d) Equipment which has unbalanced 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 R.F. 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.

4.00 OPERATION: RECORDING CONSOLE

4.01 INTRODUCTION

The Series 400B console, when fitted with standard input and standard output modules has been designed to provide the varied facilities required in a modern 4 or 8 track recording studio.

The recording process can be broken down into 4 basic sequences:-

1. Record mode: Recording direct from microphone or line input on to the multitrack tape recorder.

2. Multitrack Playback: listening to what has been recorded.

3. Overdubbing: Building up the track complement while listening to what has already been recorded. Basically, this is a combination of one and two.

4. Mixing: Combining all the recorded tracks, various effects, echo etc. to form the final stereo mix.

   A detailed explanation of all controls and switches has been given in section 2, and it is assumed that the user is familiar with basic multitrack recording methods.

4.02 RECORD MODE

1. **Record Mode**

   This is the basic starting point in making a recording. Input channels are placed in the microphone mode by placing the mic/line switch in the "up" position. The signal is routed to the desired console group output by depressing the relevant group routing button on each channel. The signal can be routed to more than one group if desired and panned between any odd and even numbered groups by using the channel pan pot.

   The signal appearing at the console group output to the tape recorder can now be metered on the relevant VU meter, if the tape return button is in the "up" position. A monitor balance is achieved using the appropriate monitor level control. The monitor pan pot will pan the signal between left and right speakers.

   The various gain and level controls throughout the signal chain may now be adjusted to set the operating levels for optimum signal to noise ratio and headroom conditions.

2. **Channel and Group Faders**

In general the channel and group faders should be set to approximately the zero position. This will enable the engineer to increase the level by 10dB, or fade out completely, while normally operating in the fine resolution area of the fader travel. The absolute position of the fader is not too critical, but situations where for example, the channel fader is operated at -30dB, with the associated group fader at +10dB should be avoided, as this will degrade the normally excellent signal to noise ratio of the console. Similarly, the opposite case runs the risk of distortion.

3. **Microphone Gain**

Having set both channel and group faders as above, the microphone gain is set to give the required level at the group output. The amount of gain required depends on the type of microphone, the sound pressure level developed by the instrument or voice and the distance between the sound source and microphone. In the case of high output and high sound pressure levels, it may be necessary to switch in the -20dB attenuator pad, to prevent overloading the input stage.

4. **Insert Patch Points**

In situations where the dynamics of the input signal are very wide or unpredictable, or where a particular effect is required, it is often necessary to patch in a limiter/compressor, equaliser or effects device into the signal chain. This can be done either via the channel insert jack points or the group insert jack points. Which insert point is used will depend on the actual effect required. If only an individual instrument is to be controlled, the channel insert point would be used. However, if a number of microphone channels have been mixed to a single group, such as backing vocals for instance, then it will be necessary to patch in an overall limiter, using the group insert point.

5. **Headphone Mix**

During recording, it is of course essential for the musicians to hear what they and everybody else are playing. Headphones mixes are derived using any or all of auxiliary sends, either direct from the input channels, or alternatively from the group monitor channels. Deriving the headphone mix from the monitor channels has the advantage of giving the musicians a mix when replaying the recording just made, which is useful when overdubbing to enable them to hear their performance. However, a mix derived from the monitor channels during recording will be affected by any gain riding of the channel or group faders, which may cause problems. This can be avoided by using the input channel auxiliary send 1 and 2, switched to Pre. The headphone signal is thus derived from before the channel fader, and is unaffected by any change in the position of that fader.

6. **Echo Sends**

Similarly to headphone mixes, echo sends may also be generated from either the input channels or the monitor channels. However, unlike headphone mixes the echo send is usually required to be derived after the fader, so that it is proportional to the fader level. This ensures that the percentage of direct signal to echo signal remains constant, regardless of fader position. This condition is met on auxiliary sends 1 and 2, if Pre is not selected, and also on sends 3 and 4.

The individual situation determines whether the echo send is derived from the input channel or the monitor channel. Normally if the echo return signal is to be recorded on the multitrack tape recorder, then the echo send will be derived from the input channels. However, if it is an echo mix for monitoring purposes only,, then it would be more relevant to use the monitor channel sends to drive the echo send mix.

7. **Echo Returns**

The output of the echo or reverberation device, i.e., the echo return signal, patched into either unused input channels if this signal is required to go to tape, or alternatively patched into unused monitor channels if it is only monitor echo and no input channels are available.

If an input channel is used, the return signal is patched into the line input jack of the relevant channel. The channel must then be placed in the line mode by depressing LI. Line gain and fader position can now be set up as described for microphone inputs.

The echo return can now be routed to the stereo mix, or to the group outputs if it is required to record the echo signal, by depressing mix or any of the group routing buttons.

Naturally when an input channel is used as an echo return, the echo return signal may also be sent to any of the auxiliary sends. This allows echo to be added to the headphone mix if desired. It is also possible of course, to send the echo return to the echo send and create a loop in the echo system. When used with a reverbaration device the effect will be to increase the reverberation time. When used with a tape delay, it will create multiple delays, decaying in amplitude in a manner controlled by the send and return levels. This effect must be handled with care. If the echo send level from the echo return channel is set too high, the entire system will start to feed back and go into oscillation, instead of decreasing in amplitude with each successive loop.

If it is not required to record the echo signal, but merely to route it direct to monitors select mix. This will route the signal directly to the stereo mix buss, which is also the monitor signal in this mode.

4.03 MULTITRACK PLAYBACK

Having actually recorded something on the multi-track recorder, it is necessary to be able to listen to the recording. This is achieved by simply selecting Tape Return. The effect of this is to connect the individual group monitor to the output of the relevant track of the multitrack recorder. Therefore the signal from the recorder now follows the same monitor signal path as the group output signal had been using. This means that all levels and panning, and therefore the monitor mix, will remain the same as they were when the recording was being mmade. This is assuming, of course, that the tape machines are correctly aligned.

The facility of monitoring the playback of the multitrack in this way enables the operator to perform a quality check even while the recocrding is being made, providing that the following precautions are observed.

1. Isolation between control room and studio is good. If not, the musicians will hear a delayed version of what they are playing.

2. Headphone mix is derived from the channel faders, otherwise the musicians will again hear a delayed version.

3. Noise reduction system is either a simultaneous encode-decode system, or not used. If it is not a simultaneous encode-decode system, then while recording it will be in the encode mode, and the signal presented to the console monitor return will probably be the tape recorder line input signal i.e., the signal from the console group output. In this case, depressing Tape Return will appear to have no effect, as it will be the same signal in both cases. This particular effect can in some cases be put to good use, however, when overdubbing as it allows the monitor switching to be achieved automatically, depending on the mode of the tape recorder. This will be dealt with more thoroughly in Section 4.04 which describes the overdubbing process.

4.04 OVERDUBBING

Overdubbing is the process of building up a recording track by track, while listening to all the previously recorded trcks. As such it will be clear that this is a combination of the previous two modes, with some channels in the microphone (Recording) mode, and some monitors in the Tape Return (Multi-track playback) mode.

Any tracks which have already been recorded are monitored by selecting the relevant Tape Return buttons. A headphone mix may be set up using the auxiliary sends on these monitor channels. Monitor echo can also be similarly set up, as already described in Section 4.02.

At this stage, a decision must be made as to the source of the headphone mix signal from the overdub channels. It can be derived from either the input channel or the group monitor channel or both.

1. **Input Channel**

   In this case, the musian always hears himself. However he will not be able to hear his previously recorded signal off the tape recorder, if heneeds to "drop in" in the middle of a take.

2. **Monitor Channel**

   If the headphone mix is derived from the monitor channels, the engineer can decide whether the musician hears himself live or the previously recorded signal off tape (syncplayback(, by depressing the Tape Return button. However he will not be able to hear both, and a vocalist for instance may find it difficult to match levels, and intonation.

   This switching can sometimes be achieved automatically if a noise reduction unit is in use, and with some tape recorders. If the tape recorder or noise reduction unit is arranged to switch its sync output to line input whenever the tape recorder is in stop, fast forward, rewind, or record and only switch to sync playback off tape when the tape recorder is in the play mode, the automatic switching condition will be met.

   All that is now usually necessary when overdubbing is to select Tape Return on the relevant monitor channel, and let the tape reccorder and/or noise reduction system perform the work of monitor switching.

   The musician now hears himself live at all times except when the tape recorder is actually in the sync play mode, when he will hear his previous signal off tape. This method saves the engineer from continually switching monitor sources, but can only be used with certain tape recorders.

3. **Both**

   If derived from both, then the musician is able to hear himself live, via the input channel, and his previous recording via the monitor channel, if it is switched to monitor Tape Return, until the moment of entering the record mode. At that point, most tape recorders switch fromm sync playback to line input. The effect of this will be a slight increase in the level of that signal in the headphone mix, due to the addition of the input channel signal and tape recorder line input signal, which is effectively the same signal.

4.05 REMIX MODE

When all recording and overdubbing has been completed, the console is placed in the remix mode.

Remixing is the process of combining all the previously recorded tracks together with any special effects devices such as harmonisers, flangers and delay lines.

It is now necessary to replug the tape machine outputs into the input module line input sockets. Any combination of inputs may be used, not necessarily arranging track 1 to input 1, track 2 to input 2 etc. Once there have been replugged, the monitor channels may now be used at effects return channel by plugging the effects device outputs into the tape return juck sockets. It is best to use monitors 5 to 8 for this purpose, leaving monitors 1-4 to act as subgroups.

The remix mode is entered by selecting line input on each input channel (LI), which connects the output of the multitrack tape recorder to the input of each channel, via the line trim control. This is normally set approximately to the "0" position if the recorder is operating at a nominal +4dB line level. However, like the mike trim, it can be adjusted to allow the channel fader to work near its nominal "0" position.

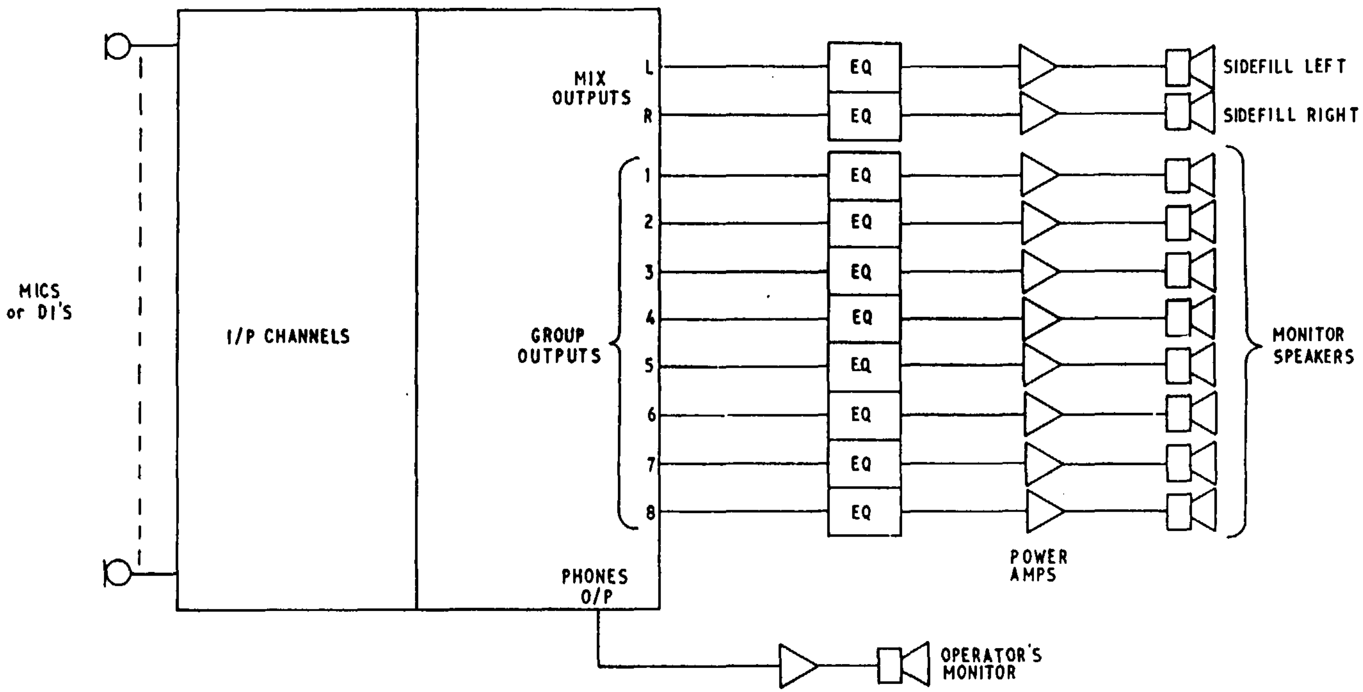
By selecting MIX, the channel can be routed directly to the stereo mix, via the channel pan pot, to allow positioning of the signal within the stereo perspective.

Alternatively the signal can be routed to any of the output groups to enable the formation of sub groups, which can then be re-routed back into the stereo mix.

Limiters or other effects devices can be connected to any channel or output group by patch into the relevant insert points.

All 4 auxiliary sends can be used to feed echo or effects devices, as headphones mixes are not required during remixing.

The composite stereo mix will be controlled in level by the stereo master fader and the level metered on the stereo mix meters. These meters will also indicate the level of any solo signal, or of the stereo tape return selected on the 2 track replay (REP) button. This allows quality and level checking of the stereo recording in the same way as described for multitrack playback.



MICS
or DI'S
I/P CHANNELS
MIX
OUTPUTS
L
R
GROUP
OUTPUTS
1
2
3
4
5
6
7
8
PHONES
O/P
EQ
EQ
EQ
EQ
EQ
EQ
EQ
EQ
EQ
EQ
POWER
AMPS
SIDEFILL LEFT
SIDEFILL RIGHT
MONITOR
SPEAKERS
OPERATOR'S
MONITOR

5.00 OPERATION: MONITOR CONSOLE

Monitor mixers are used as a convenient way of obtaining a number of independently controllable mixes from one set of sound sources.

Typical applications for monitor mixers being:

a) providing on stage foldback for musicians

b) audio distribution in theatres, conferences, etc.

Taking case a) as a practical example, and referring to the diagram, all instruments are connected to the mixer via microphones or direct injection (D.I.) boxes, in the usual way. Each output would then be assigned to one or more monitor loudspeaker on the stage.

e.g. Mix L + R of outputs might feed the sidefills, Groups 1 - 8, the musicians' individual monitors.

Phones output feeds the mixer operator's monitor.

Sends 1 - 2 on the input module are PRE/POST fade selectable, independently of sends 3 - 8, thus giving the system a high degree of flexibility. If, for example, the mix outputs were to be used to make a stereo tape recording of a performance, then sends 1 - 2 could feed the sidefills in the same way as MIX, by being selected to POST fade.

Sends 3 - 8, set to pre fade, could then be used for the musicians.

The console could, in fact, be used as a main P.A. mixer, if necessary, using MIX to drive the P.A. amps, and the groups for auxiliary and foldback functions.

6.00 MAINTENANCE

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 to carry out, with the minimumm of test equipment needed to isolate and rectify faults.

6.01 GENERAL FAULT FINDING

With the exception of the electronically balanced microphone amplifier all signal electronics are configured around high slew rate, low noise integrated circuits. The microphone amplifier is a proprietory 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 the audio faults can be repaired by simply replacing the I.C., having first isolated the fault to be a particular stage in the signal chain. The isolation can often be done without even removing the module from the console, by judicious use of the 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 the 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 a first appears to be an extremely complicated piece of equipment can be simpliefied into a series of simple sequential stages. This is the basic first move in all types of fault finding, and usually requires nothing more than a certain amount of logical thought. Servicing a mixing 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 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 to approximately 1kHz, 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 bypassed by switching out the Hi-pas filter, and the Equaliser.

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 then be checked and if necessary replaced. Integrated circuits, due to their internal complexity, are the most likely causes of problems followed by mechanical components such as switches and faders, which are susceptible to physical contamination from oxidisation, dust, and liquids.

6.02 REMOVING MODULES

Remove the two 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. Although damage to the module will not be caused by plugging and unplugging the module with the power still switched on, this is not recommended for the inexperienced, as it is possible to bend the connector pins if care is not taken.

6.03 METER ALIGNMENT

Each V.U. meter has its own individual drive circuit on the circuit board of the meters. This card also contains the detection and drive circuitry for the peak LED which is pre-set to indicate a peak level of 8dB above OVU.

OVU is normally adjusted to indicate a line level of +4dBv, i.e., a level of 1.228 volts. However, it can be re-adjusted to indicate a different line level if required by the pre-set potentimeter on the drive card.

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

Note that the peak LED will always indicate a level of 8dB above whatever the OVU level has been set at.

6.04 LAMP REPLACEMENT

Illumination of the VU meters is provided by 2 lamps in each meter, these are 6 volt lamps wired in series. The lamps in each pair of meters are wired in series and powered by the ± 17 volt audio supply. A series resistor in each meter pair provides turn on current limiting to prolong the lamp life.

To replace the lamp remove the VU meter from the panel. This is done by the following method:-

1. Remove the two plastic clamps
2. Unscrew the two screws through the P.C.B.
3. Desolder the illumination wires from the tags on the meter.

The meter can now be slipped free and the front removed to change the bulbs.

6.05 POWER SUPPLY SERVICING

The series 400B power supply provides the following regulated supply rails;

i) ± 17 volts Audio
ii) + 48 volts Phantom Power

If a power supply fault is suspected, first ensure that it really is the P.S.U. which is at fault, and not a short circuit in the console. This can be checked by disconnecting the P.S.U. from the console, and measuring the voltage at the connector. A load across the supply should be provided, to simulate the normal load conditions imposed by the console.

A 10ohm, 20 watt resistor should be connected across each of the audio rails. The Phantom power supply can be loaded with a 2.2kohm, 1 watt resistor.

The ripple and noise value of the various supply rails can now be measured, using a millivoltmeter or an oscilloscope.

If a fault is found to exist in the P.S.U. disconnect the mains supply and remove the cover. Check visually for any obvious problems, such as blown fuse, burnt components etc. If nothing obvious is observed, reconnect the mains, and measure the voltages across the various electrolytic smoothing capacitors, which should be as follows:

Audio Supply	C1	=	+26 volts
	C2	=	-26 volts
Phantom Supply	C12	=	+59 volts

Difference of +10% are acceptable, due to variations in the incoming mains voltage. If satisfactory, the problem lies in the regulator section. If not however, check the bridge rectifier, smoothing capacitor and transformer for failure.

7.00 RECOMMENDED SPARES KIT

SEMICONDUCTORS

Transistor 842GR	Soundcraft Part No.	BD0301 (2)
Transistor 1681BL	Soundcraft Part No.	BD0302 (2)
Transistor PN4355	Soundcraft Part No.	BD0312 (2)
Single Op Amp IC TL071	Soundcraft Part No.	BE0404 (2)
Dual Op Amp IC TL072	Soundcraft Part No.	BE0413 (2)
Single Op Amp IC 5534	Soundcraft Part No.	BE0407 (1)
Dual Op Amp IC 5532	Soundcraft Part No.	BE0428 (1)
Quad Analogue Gate DG308	Soundcraft Part No.	BF0517 (1)

POTENTIOMETERS

Input Gain and TB vol (10KRD)	Soundcraft Part No.	DM1104 (1)
Eq. Lift and Cut (10KB)	Soundcraft Part No.	DM1102 (1)
Eq. Frequency (100KRDx2)	Soundcraft Part No.	DM1106 (1)
Aux sends/masters, monitor sends and monitor vol (10KA)	Soundcraft Part No.	DM1103 (2)
Phones volume (10KAx2)	Soundcraft Part No.	DM1105 (1)
Pan (10KBx2)	Soundcraft Part No.	DM1101 (1)
All faders	Soundcraft Part No.	DD0316 (1)

MISCELLANEOUS

Green Led.	Soundcraft Part No.	JA0001 (1)
Red Led.	Soundcraft Part No.	JA0002 (1)
V.U. meter	Soundcraft Part No.	JD0315 (1)
V.U. meter bulb	Soundcraft Part No.	JB0122 (5)
Fader knob - white	Soundcraft Part No.	KA0027 (2)
red	Soundcraft Part No.	KA0028 (1)
yellow	Soundcraft Part No.	KA0029 (1)
Brown control knob	Soundcraft Part No.	KA0030 (5)
Cap for above red	Soundcraft Part No.	KC0235 (1)
black	Soundcraft Part No.	KC0230 (1)
orange	Soundcraft Part No.	KC0234 (1)
blue	Soundcraft Part No.	KC0231 (1)
green	Soundcraft Part No.	KC0232 (1)
yellow	Soundcraft Part No.	KC0236 (1)
All switches	Soundcraft Part No.	DF0516 (1)
Module retaining screws	Soundcraft Part No.	NA0084 (15)

N.B. The above spares are available as a kit in the quantities noted in brackets. This kit is suitable for End Users not wanting to keep large stock levels. To order please quote the Soundcraft Part No. RZ2240.

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 to which they may be entitled against the seller of the Equipment.