Series 800
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 800.

CONTENTS.

SERIES 800

Issue Date ....

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			
Master Module			
Effects Ret. Module			
Other Modules			
P.S.U.			

OPTIONS - SPECIFY

ALTERATIONS TO SPEC.

SPECIAL INSTRUCTIONS

DESPATCH KIT REQUIRED

Series 800 System Measurements.

1. **Channel Line Input to Mix Output.**

Channel	THD (Ref 1KHz, +4dBv)		Frequency Response (ref 1kHz) 25Hz	20kHz
1	0.0	0.0	-0.	-0.
2	0.0	0.0	-0.	-0.
3	0.0	0.0	-0.	-0.
4	0.0	0.0	-0.	-0.
5	0.0	0.0	-0.	-0.
6	0.0	0.0	-0.	-0.
7	0.0	0.0	-0.	-0.
8	0.0	0.0	-0.	-0.
9	0.0	0.0	-0.	-0.
10	0.0	0.0	-0.	-0.
11	0.0	0.0	-0.	-0.
12	0.0	0.0	-0.	-0.
13	0.0	0.0	-0.	-0.
14	0.0	0.0	-0.	-0.
15	0.0	0.0	-0.	-0.
16	0.0	0.0	-0.	-0.
17	0.0	0.0	-0.	-0.
18	0.0	0.0	-0.	-0.
19	0.0	0.0	-0.	-0.
20	0.0	0.0	-0.	-0.
21	0.0	0.0	-0.	-0.
22	0.0	0.0	-0.	-0.
23	0.0	0.0	-0.	-0.
24	0.0	0.0	-0.	-0.
25	0.0	0.0	-0.	-0.
26	0.0	0.0	-0.	-0.
27	0.0	0.0	-0.	-0.
28	0.0	0.0	-0.	-0.
29	0.0	0.0	-0.	-0.
30	0.0	0.0	-0.	-0.
31	0.0	0.0	-0.	-0.
32	0.0	0.0	-0.	-0.

2. Channel Line Input to Auxiliary Outputs.

Auxiliary Output	THD (ref 1kHz, +4dBv)	
1	0.0	%
2	0.0	%
3	0.0	%
4	0.0	%

3. Channel Line Input to Control Room Outputs.

Control Room Output	THD (Ref 1kHz, +4dBv).	
Left (Via PFL)	0.0	%
Right (Via PFL)	0.0	%

4. Channel Line Input to Studio Output (via all Groups).

Studio Output	THD (Ref 1kHz, +4dBv).	
Left	0.0	%
Right	0.0	%

5. Crosstalk (at 10 kHz).

Stereo Mix	-	dBv
Channel to Channel	-	dBv
Mute	-	dBv

6. Mix Noise. (DIN Audio, all channels and groups at unity gain).

Mix Left	-	dBv
Mix Right	-	dBv

7. Power Supply.

Output	Ripple and Noise		Voltage
+17V Audio	-	dBv	Volts
-17V Audio	-	dBv	Volts
+24V Logic	-	dBv	Volts
+48V Phantom	-	dBv	Volts

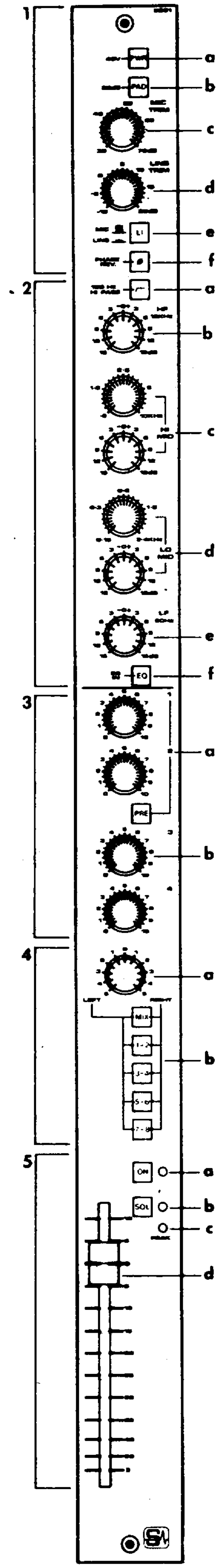
Notes

0dBv = 0.775 Volt.
0dBV = 1.000 Volt.

The constraints and conditions under which the above performance figures have been measured are configured so as to ensure that all signal paths are within specification, with a minimum of separate tests. Many are therefore recorded via very long signal paths or under other worst case conditions.

The results should not, therefore, be taken as representative of typical published specifications, which would normally be conducted under standard operative conditions.

Standard Input Module



2.01 Standard 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 unbalanced, 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) PWR.

Capacitor microphones may be powered by the internal 48 Volt phantom power supply by depressing the PWR button. When using direct injection boxes, or unbalanced sources, the phantom power supply should not be switched on.

b) PAD.

Depressing the PAD button inserts a 30dB 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.

c) MIC TRIM.

The microphone input can be varied between 30dB and 70dB of gain, using the MIC TRIM control,a 41 position detented potentiometer, allowing resettability with essentially continuously variable gain control. Used in conjunction with the 30dB PAD, a 70dB control range is available.

d) LINE TRIM.

The line input can be varied between - 10dB and +20dB of gain using a detented potentiometer.

e) L1 (Line Input).

The high level line input is selected by depressing the L1 button. This will normally connect the relevant tape return to the input channel for remixing or overdubbing purposes. i.e. Tape Return 1 will connect to channel input 1. However, an insert jack for each channel allows access to the line input for other signal sources.

2. Equaliser Section.

The Series 800 equaliser is a versatile unit, allowing 5 areas of control to be excercised over the audio spectrum. All amplitude pots are centre detented for easy zeroing, and the mid frequency select controls are 41 detented position types.

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 sections 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 can be switched in or out of the signal path, independent of the high pass filter.

3. Auxiliary Section.

There are 4 auxiliary sends available for use as echo, foldback or other auxiliary effects units. All controls are 41 position, detented potentiometers.

a) Sends 1 and 2.

Auxiliary sends 1 and 2 are normally post fader, but may 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 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.

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 can be switched on and 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) SOL (Solo).

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

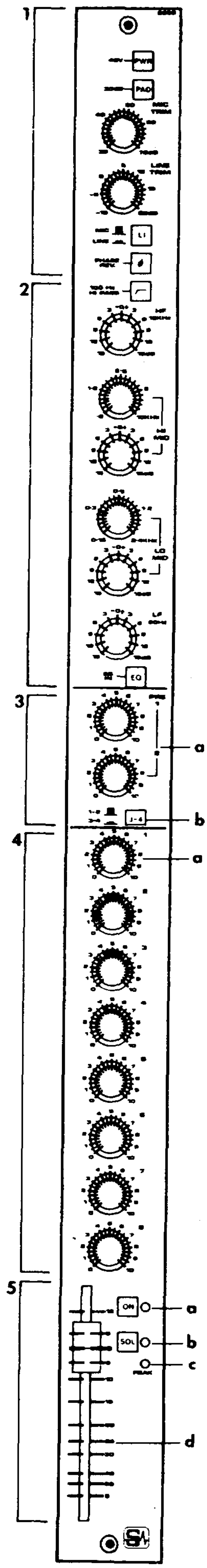
c) PEAK.

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

d) Channel Fader.

The channel fader is high quality Penny & Giles conductive plastic unit, for low noise operation, and long life.

Monitor Input Module



2.02 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 2k Ohms which will not cause any loading effects on any normally used studio microphone. The high level line input is unbalanced, 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) PWR.

Capacitor microphones may be powered by the internal 48 volt phantom power supply by depressing the PWR button. When using direct injection boxes, or unbalanced sources, the phantom power supply should not be switched on.

b) PAD.

Depressing the PAD button inserts a 30dB 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.

c) MIC TRIM.

The microphone input can be varied between 30dB and 70dB of gain, using the MIC TRIM control, a 41 position detented potentiometer, allowing resettability with essentially continuously variable gain control. Used in conjunction with the 30dB PAD, a 70dB control range is available.

d) LINE TRIM.

The line input can be varied between - 10dB and +20dB of gain using a detented potentiometer.

e) LI (Line Input).

The high level line input is selected by depressing the LI button, with access via a ¼ inch mono jack on the rear panel.

2. Equaliser Section.

The Series 800 equaliser is a versatile unit, allowing 5 areas of control to be excercised over the audio spectrum. All amplitude pots are centre detented for easy zeroing, and the mid frequency select controls are 41 detented position types.

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 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. 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 charecteristic "bell" shape. The Q of the network (a measure of bandwidth) is 1.5.

d) LO MID.

The Lo Mid sections is identical to the Hi Mid section, with the exception that the frequency is variable between 15Hz and 2.4kHz.

e) LF (Low Frequency).

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

f) EQ.

The equaliser can be switched in or out of the signal path, independent of the high pass filter.

3. Auxiliary Section.

There are 4 auxiliary send busses available for use as echo, foldback or other auxiliary effects units, although only 2 (1 and 2 or 3 and 4) can be used at any one time. Both level controls are 41 position detented types.

a) Both auxiliary controls are permantly pre-fader, but after the ON switch.

b) 3-4.

The auxiliary controls are normally assigned to auxiliary busses 1 and 2, but can also be assigned to busses 3 and 4.

4. Monitor Section.

a) 8 independent post-fader monitor outputs, using 41 position detented controls.

5. Channel Status Section.

a) ON.

The channel can be switched on and 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) SOL (Solo).

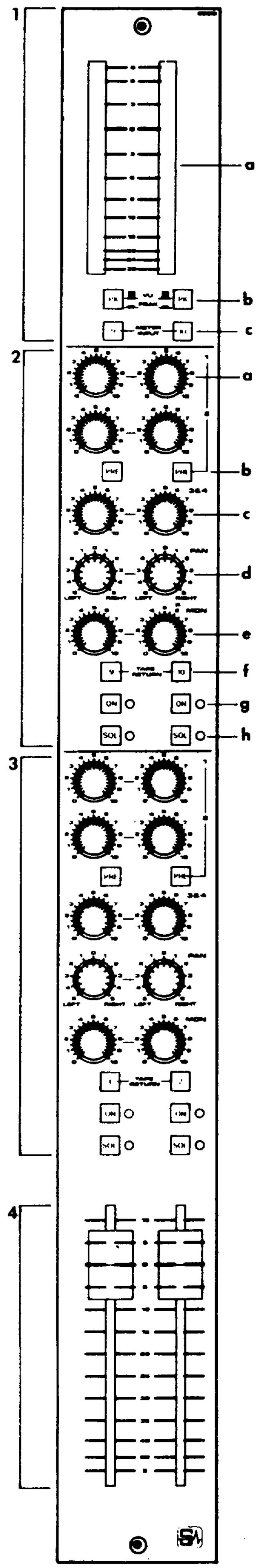
SOL Soloes the pre-fader, post insert jack signal, independently of the ON switch. Solo operation is indicated by a red LED on the channel, and a master solo warning lamp on the Master module.

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.

d) Channel Fader.

Recording Output Module



2.03 Recording Output.

The double width group/monitor module handles the functions of 2 group outputs, and 4 monitor return inputs, during the recording process. However, when in the remix or overdub mode, the module can be used to provide other functions.

The sub group section can be used to form audio sub groups, which can then be compressed, limited or otherwise processed, before being combined with the rest of the stereo mix. The upper monitor section can provide additional line level inputs into the stereo mix.

1. Metering.

Signal level metering is provided by a 30 segment LED bar graph display, covering a signal level range from -30dB to +8dB.

b) PK (Peak).

Each meter can be individually selected to either Peak or V.U. characteristics.

c) Meter Input.

Each meter source is selectable between the lower or upper monitor source signal, by depressing the relevant Meter Input button.

2. Lower Monitor Section (Tracks 9-16).

a) Auxiliary Sends 1 and 2.

Auxiliary sends 1 and 2 are normally pre-fader, but can be switched post-fader. In both cases, the auxiliary signal is post the monitor ON switch.

b) PRE.

Pre converts the auxiliary sends from post-fader to pre-fader operation.

c) Auxiliary Sends 3 and 4.

Auxiliary sends 3 and 4 are controlled by a single stereo control to provide a stereo effects send or foldback signal. This signal is post-fader and post monitor panpot.

d) PAN.

The monitor signal can be panned between left and right of the stereo mix.

e) MON.

The monitor signal level to the stereo mix is controlled by the monitor level potentiometer.

f) TAPE RET.

The tape return switch selects the monitor and meter source signal between tape send (group output) and tape return (monitor input).

g) ON.

On activates the monitor channel and auxiliary sends, and is indicated by a green LED.

h) SOL.

Sol soloes the monitor pre-fader signal independent of the ON switch and is indicated by an adjacent red LED, and a master solo warning lamp on the Master Module.

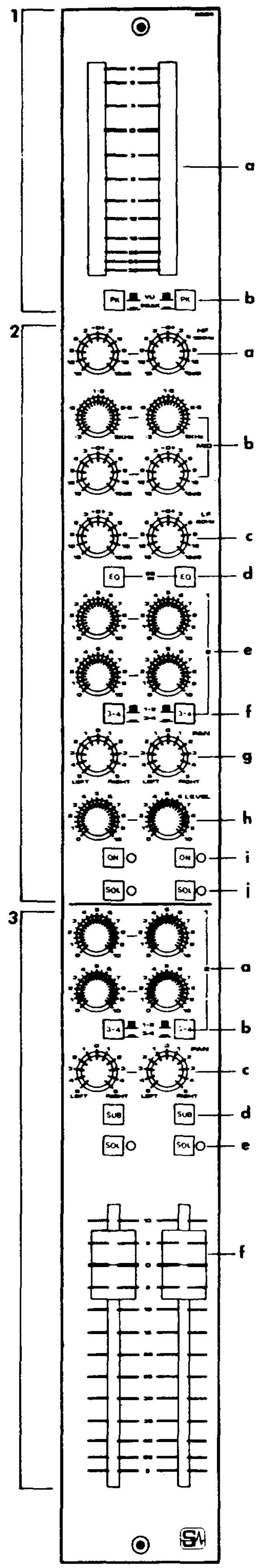
3. Lower Monitor Section (Tracks 1-8).

The lower monitor section is identical to the upper section, except that it provides monitoring of tracks 1 to 8.

4. Group Fader.

Each group outout is controlled by a high quality Penny and Giles conductive plastic fader.

PA Output Module



f) 3-4.

When selected the auxiliary send signals are derived post-fader and post equaliser, and routed to auxiliary busses 3 and 4.

g) PAN.

The effects return signal is panned between left and right of the main stereo output by the centre-detented pan-pot.

h) Level.

The level of the effects return signal is controlled by a 41 position rotary potentiometer.

i) ON.

The effects return channel can be switched on or off. Operation of the channel is indicated by a green LED.

j) SOL (Solo).

SOL Soloes the pre-fader signal, independently of the ON switch. Solo operation is indicated by an adjacent red LED and a master solo warning lamp on the Master Module.

3. Group Output Section.

a) Auxiliary Sends.

Access to auxiliary send busses 1 and 2 or 3 and 4 is provided by the 2 auxiliary send controls. The signals are derived post group fader.

b) 3-4.

When selected, the auxiliary send signals are routed to auxiliary busses 3 and 4.

c) PAN.

If the SUB button is selected, the sub-group output can be panned across the main stereo output.

d) SUB.

The group output can be used to form a sub-group which is returned into the main stereo output via the pan-pot and the SUB button.

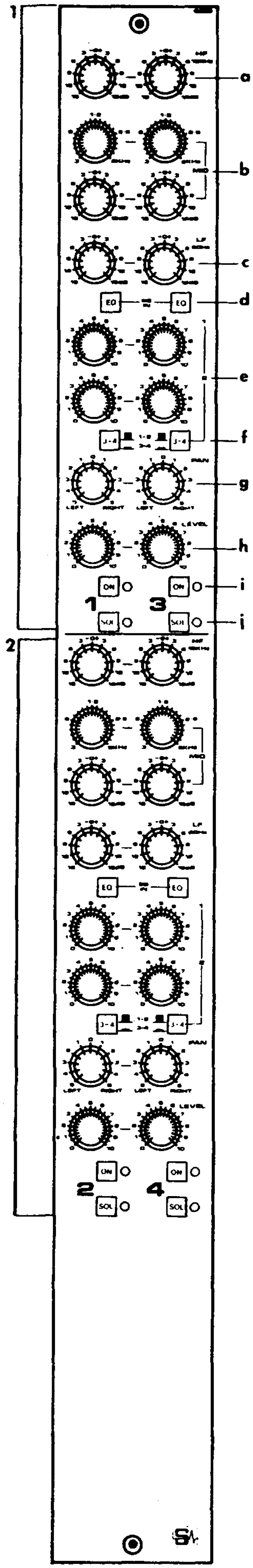
e) SOL (Solo).

The group output can be soloed and this condition is indicated by an adjacent red LED and the master solo warning lamp on the Master module.

f) Fader.

The overall level of the group output is controlled by a high quality Penny and Giles conductive plastic fader.

Effects Returns Module



2.05 Master Module.

The Master module is common to all versions of the Series 800 console. It contains metering and monitoring facilities for the main stereo output, and tape recorder returns, in addition to master auxiliary send controls, talkback facilities and the alignment oscillator.

1. Monitoring Section.

a) Metering.

Metering of the main stereo output, 2 track tape returns, or any soloed signal is provided by a pair of 30 segment LED bar graph meters. A signal range from -30dB to +8dB is covered.

b) PK (Peak).

Each meter can be individually selected to either Peak or V.U. characteristics.

c) Auxiliary Masters.

Overall level control of each of the 4 auxiliary busses is provided by a 41 position level control. The auxiliary outputs are all electronically balanced, without the use of transformers. Each output can be soloed, with its output level indicated on the stereo output meters. Auxiliaries 1 and 3 are indicated on the left meter, and 2 and 4 on the right.

d) Monitor Source.

Four interlocking push buttons, 2Ta, 2Tb, 2Tc and MIX, route the outputs from 3 external stereo sources or the main stereo mix output to the control room, studio, and headphone monitoring systems. The monitor inputs are electronically balanced.

d) Studio Level.

The volume of the studio monitoring system is controlled by the Studio level potentiometer. The source signal is selected by the Monitor Source switches.

f) Control Room Level.

Control Room Level adjusts the volume of the control monitoring system, as selected by the Monitor Source selection. The source selection is overidden by the operation of any solo function. Because the meter drive signal is derived from the signal to the control room level potentiometer, the level of any soloed signal will also be indicated on the main stereo meters. This enables the auxiliary send signal level or any channel pre fader level to be measured.

g) Headphones Level.

The volume of the headphone monitoring system can be varied by the headphone level control. The source selection is identical to the control room.

h) MNO (Mono).

To provide a mono compatibility check of the stereo monitor signal the left and right channels can be summed together by the MNO switch, operative on both control room and headphone outputs.

i) DIM.

Partial muting (20dB) of the control room and headphone signals is achieved by the DIM switch. This allows a conversation to be held without disturbing the setting of the monitor volume.

2. Communications Section.

a) Mic Level.

The level of the talkback signal to all group outputs auxiliary sends and studio monitor systems can be adjusted.

b) Slate.

Depressing Slate routes the talkback signal to all group busses, with the addition of a 30Hz tone, to identify tape sections. The 30Hz tone is normally audible during the fast wind mode of the tape recorder, even though the tape is not in contact with the heads, and allows the rapid location of tape sections.

c) COMM (Communicate).

Depressing COMM routes the talkback signal to the auxiliary busses, to allow communication with musicians who are wearing headphones.

d) Talkback.

Depressing Talkback routes the talkback signal to the auxiliary busses and also the studio monitor system.

e) Solo.

Operation of any solo function on the console will illuminate the master solo warning lamp. A red LED adjacent to the selected solo switch will also be illuminated.

f) Headphones.

The internal headphone amplifier will drive headphones of medium to high impedance, and allows the monitoring of any signal selected by the source selection switch, or any soloed signal.

3. Oscillator Section.

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

b) ON.

Switches the oscillator on.

c) Oscillator Level.

The level of the oscillator can be varied by the oscillator level control.

d) GRP (Group).

The oscillator can be routed to all group mix busses by depressing GRP. This allows alignment tones to be recorded and facilitates recorder alignment.

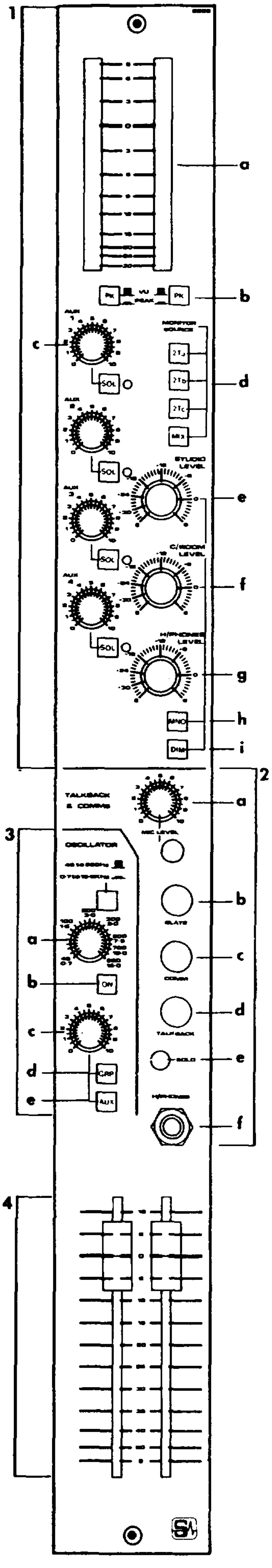
e) AUX.

The oscillator can also be routed to all 4 auxiliary busses.

4. Fader.

The overall level of the main stereo mix is controlled by 2 independent high quality Penny and Giles conductive plastic faders.

Master Module



2.06 Effects Return Module.

The optional Effects Return module contains 4 identical line level inputs to the main stereo mix busses. All Series 800 consoles are prewired to accept this module, which can therefore be added at any time.

a) H.F. (High Frequency).

15dB of boost or cut is available at a frequency of 12kHz, with a "shelving" characteristic.

b) MID.

The mid frequency is continuously variable between 300 Hz and 5kHz. 15dB of boost or cut is availalbe with a "bell" characteristic, and a "Q" of 1.5.

c) L.F. (Low Frequency).

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

d) E.Q.

The equaliser can be switched in or out of the effects return signal path.

e) Auxiliary Sends.

Access to auxiliary busses 1 and 2 or 3 and 4 is provided by the 2 auxiliary send controls. When selected to 1 and 2, the signals are pre-fader, but post-fader and post equaliser when selected to 3 and 4.

f) 3-4.

When selected, the auxiliary send signals are derived post-fader, and post equaliser, and routed to auxiliary busses 3 and 4.

g) PAN.

The effects return signal is panned between left and right of the main stereo output using the centre detented pan-pot.

h) Level.

The effects return level is adjusted by a rotary potentiometer.

i) ON.

The effects return channel can be switched on, by depressing the ON switch. Operation of the channel is indicated by an adjacent green LED.

j) SOL (Solo).

SOL soloes the pre-fader signal, independently of the ON switch. Solo operation is indicated by an adjacent red LED and a master solo warning lamp on the Master Module.

3.00 OPERATION: Recording Console.

3.01 INTRODUCTION.

The Series 800 console, equipped with standard input and recording output modules has been designed to provide the facilities required in a modern 8 or 16 track recording studio, with a minimum of operator effort and to allow free and ininhibited use of these facilities, without unnecessary patching, or redundant switching.

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

1. Record Mode: Recording direct from microphone 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 1) and 2).

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

An understanding of the signal flow through the mixer is best gained by a study of the block diagram. 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.

3.02

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 to 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 any even numbered groups by using the channel pan pot.

The signal now 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, and depressing the monitor "ON" button. 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 "0" 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 microphones and high sound pressure levels, it may be necessary to switch in the -30dB 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 on the rear panel of the console. 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. Headphone 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 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 sends 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 group 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 derive the echo send mix.

7. Echo Returns.

The output of the echo or reverberation device, i.e. the echo return signal, are patched into either unused input channels if this signal is required to go to tape or alternatively patched into unused monitor channels (9 to 16 only) 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 signal 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 reverberation 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 return signal, but merely to route it direct to monitors, depress mix. This will route the signal directly to the stereo mix buss, which is the monitor signal.

3.03 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 tape 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 made. This is assuming, of course, that the tape machines is correctly aligned.

The facility of monitoring the playback of the multitrack in this way enables the operator to perform a quality control check even while the recording 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, otherwide 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 3.04 which describes the overdubbing process.

3.04 Overdubbing.

Overdubbing is the process of building up a recording track by track, while listening to all the previously recorded tracks. As such it will be clear that this is a combination of the previous 2 modes, with some channels in the microphone (Recording) mode, and some monitors in the Tape Return (Multitrack 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 3.02.

Input channels in the microphone mode, which will form the overdub signal can now be routed to the output group corresponding to the track to be recorded, in exactly the same way as described in Section 3.02.

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 musician always hears himself. However he will not be able to hear his previously recorded signal off the tape recorder, if he needs 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 (sync playback), 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 to 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 recorder 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. <u>Both.</u>

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 will switch from 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.

3.05 Remix Mode.

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

This is done by selecting line input of each input channel (LI), which connects the output of the multi track 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 +4dBv line level. However, like the mike trim, it can be adjusted to allow the channel fader to work near its nominal "0" position.

By depressing 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.

Limiters or other devices can be connected to any input channel or output group by patching into the insert points on the rear panel.

All 4 auxiliary sends can be used to feed echo or effects devices, as headphone 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 LED Bargraph meters.

These meters can also indicate the level of any solo signal, or any of the stereo tape returns selected on the monitor source buttons. The monitor source selection buttons allows quality and level checking of the stereo recording in the same way as described for multitrack playback.

3.06 Sub Groups.

In the remix mode of operation, the Series 800 console becomes an extremely flexible mixer, due to the use of the normally redundant monitor and group output section to provide effects returns and sub groups.

Tape Return Inputs 9 to 16 can be used as extra line inputs to the stereo mix busses, via the pan pot. Access to the monitor return is obtained by patching into the Monitor Return jack on the rear panel, and selecting Tape Return on that monitor channel.

In addition to this, the group section can be used to form 8 mono or stereo sub groups from the input channels, to feed external equipment or even to feed the stereo mix busses, via the lower monitor section, totally independant of the upper monitor section.

4.02 Interface Levels.

The Series 800 is normally supplied to provide compatible level interfacing with standard professional equipment. i.e. +4dBv (ref 0.775 Volt). However, provision has been made during the design stage, 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 2 resistors and the removal of a link on each group output circuit board.

Group Output Level.

1) Remove L2 (link across R14).
2) Add resistor in position RX1.

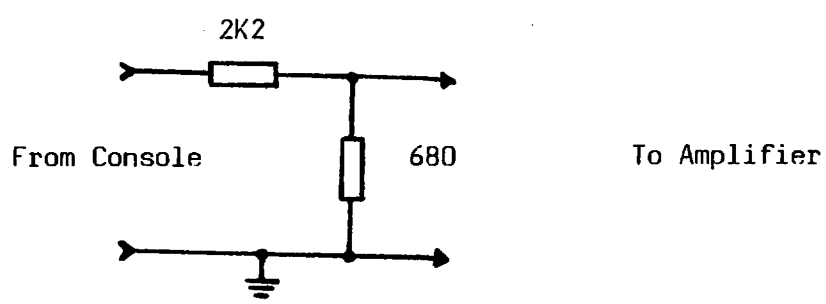
Monitor Return Gain.

1) Add resistor in position RX2 and RX3.

The value of the resistor must be the same for both RX1 and RX2 for correct operation. For interface to the normal Teac/Tascam operating level of -10dBV, a value of 820 ohms is suitable, and will provide a nominal 600 ohm source impedance to feed the tape recorder.

Control Room, Sudio and Auxiliary Outputs.

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



4.03 General Wiring Procedures.

To take full advantage of the excellent signal to noise ratio and low distortion of the Series 800, care must be taken to ensure that incorrect installation and wiring does not degrade its performance. Hum, buzzes, instability and Radio Frequency Interference (RFl) 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 seperate 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.

The successful, hum free, installation of a system requires careful forethought, and the establishment of a set of ground rules, which must be consistantly 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 for the system, and all earths should be "star fed" from this point. It is common electrical practice to "daisy chain" the earths to all electrical outletes buth 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 and 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 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 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.

a) Extraneous electrostatic or electromagnetic fields.
b) Noise and interference on the earth line.
c) 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 crosstalk, 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 connection of the screen in each case is shown in 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 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. Do not use single screened cable.

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 and noise levels will be futile. In extreme cases there will be no alternative but to provide a completely seperate and independant "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.

5.00 Servicing.

Although the Series 800 console has been designed for reliability, 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 minimum of test equipment.

5.01 General Fault Finding.

With the exception of the electronically balanced microphone amplifier, all signal electronics are configured around high slew rate, low noise integrated ciruits. 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 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 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 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 simple sequential stages. This is the basic first move in all types of faultfinding, 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 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 interupting 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 a possibly non-working module.
Route the channel directly to mix, so that the channel may be monitored in the normal way. Using the internal 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 the Phase Reverse, or switching out the Hi-Pass 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 cause of problems, followed by mechanical components such as switches and faders, which are susceptible to physical contamination from oxidisation, dust, and liquids.

5.02 Removing Modules.

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. Although damage to the module will not be caused by plugging and unplugging the module with the power still switched on, this is no recommended for the inexperienced, as it is possible to bend the connector pins if care is not taken.

5.03 Meter Alignment.

Each meter has its own individual drive electronics which contains the detection and drive circuitry for either V.U. characteristics or Peak characteristics which is preset to indicate a peak level of 8dB above 0.V.U.

0.V.U. is normally adjusted to indicate a line level of +4dBv, i.e. 1.228V. However, it may be adjusted to indicate a different line level, if required, by adjustment of the meter calibration pre-set potentiometer.

To align the meter drive, connect a millivoltmeter to a group output of the console. Route the oscillator, set to approximately 1kHz, to groups and adjust the level to the millivoltmeter for the required console output level. (Normally, this would be +4dBv). Adjust the meter calibration preset to indicate 0.V.U. and repeat for all other groups, and the Stereo Mix outputs.

Note that the console output level for 0dB Peak will always be 8dB above whatever the 0.V.U. level has been set to.

5.04 Power Supply Servicing.

The Series 800 power supply provides the following regulated supply rails:

a) $\pm$ 17 volts, Audio

b) + 24 volts, Logic

c) + 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 20ohm, 20 watt resistor across each of the audio supply rails and a 47ohm 5 watt resistor across the logic supply rail is suitable. The phantom power supply can be loaded with a 2.2 kohm, 1 watt resistor.

The ripple and noise value of the various supply rails can now be measured, using a millivoltmeter or an oscilloscope, and a value of at least -80dB (ref 0.775v, DIN Audio) should be obtained on the audio and phantom supply rails.

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 = 23 Volts

C2 = 23 Volts

Logic Supply C7 = 38 Volts

Phantom Supply C10 = 56 Volts

Differences of $\pm$ 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.

Audio Supply.

The Audio supply is extremely simple, comprising of 2 Monolithic regulator integrated circuits. A 15 Volt positive regulator, type LM318, is configured to provide +17 Volts, by means of the potential divider R1 and R3, raising the potential at the common terminal.

The negative supply is similar, except for the use of a LM337 regulator. In this case the potential divider is formed by R2 and R4.

Logic Supply.

The Logic supply is similar to the +VE audio supply with the exception of the ratio of the potential divider R7 and R8, to give +24 Volts.

Phantom Supply.

The Phantom supply allows up to 100 m.a. at 48 Volts to be drawn.

TR3 compares the Zener diode voltage with the output voltage, via the potential divider formed by R17 and R18, and attempts to keep them equal. If the output voltage starts to increase, TR3 will begin to turn on, and will therefore progressively starve the pass transistor TR2 of base current, from the constant current source TR1. TR2 will then begin to turn off, and therefore reduce the output voltage.

Overcurrent protection is provided by TR4. Current drawn is sensed by TR4 by the voltage dropped across R14 and R15. As the voltage at the Base-Emitter junction approaches 0.6V, TR4 will begin to turn on, and divert base current from the series pass transistor, TR2, which will begin to turn off, thereby preventing overdissipation and consequent damage.