

N.B.

- a) In all cases, use good quality twin screened audio cable. Check for instability at the output.
- b) Always connect both conductors at both ends, and ensure that the screen is only connected at one end.
- c) Do not disconnect the mains earth from each piece of equipment. This is needed to provide both safety and screen returns to the system star point.
- d) Equipment which has balanced inputs and outputs may need to be electrically isolated from the equipment rack and/or other equipment, to avoid earth loops.

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

4.00 OPERATION SERIES 1600

4.01 Introduction

The Soundcraft Series 1600 console, when fitted with standard input and recording output has been designed to provide the various facilities required in a modern 8 or 16 track recording studio.

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

1. Record mode: Recording direct from microphone or line input onto the multitrack tape recorder.
2. Multitrack playback: Listening to what has been recorded.
3. Overdubbing: Building up the track complement whilst listening to what has already been recorded.
4. Mixing: Combining all the recorded tracks, various effects, echo etc to form the final stereo mix.

A detailed explanation of all the controls and switches has been given in section 2.00 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 pressing 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 may be metered at both the input, by pressing PFL and using the mix meters and at the relevant Group Output to the tape machine on the Group Output meter, if the tape return button is in the "up" position. A monitor balance is achieved using the appropriate level control and pressing the monitor ON button. The monitor Pan Pot will pan the signal between the 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 will usually mean that the pre-fader headroom has been reduced to a point where an amplifier could overload and cause distortion. Operating with the channel fader below the zero point and the group fader above zero also means that the noise contribution from the group bus could degrade the normally excellent signal to noise ratio.

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 where a particular effect is required, it is often necessary to patch in a limiter/compressor, equalizer or effects device into the signal chain. This can be done by using either the Channel Insert jack points or the Group Insert jack points. Which insert point is used will depend on the type of 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, for instance, backing vocals, 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 the 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

Similar to headphone mixes, echo sends may also be generated from either the input channels or 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 the direct signal to echo remains constant, regardless of fader position. This condition is met on auxiliary sends 1-2, 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 machine, the the echo send will be derived from the input channels. However, if it is a 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, is patched into either an unused input channel if this signal is required to go to tape, or into an unused monitor channel (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 pressing LI. Line gain and fader positions can now be set up as described for microphones.

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 pressing 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 tape delay, it will create multiple delays, decaying in amplitude in a manner controlled by the echo send and return levels. This effect must be handled with care.

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If the echo send level from the echo return 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 the monitors select MIX. This will route the signal directly to the stereo mix bus, which is also the monitor signal in this mode.

4.03 Multitrack Playback

Having actually recorded something on the multitrack 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 machine. Therefore, the signal from the tape machine 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 were 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 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, otherwise the musicians will again hear a delayed version.
3. Any noise reduction 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, pressing the 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 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 tracks. As such it will be clear that this is a combination of the previous two modes, with some of the 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 the 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 musician always hears himself. However, he will not be able to hear his previously recorded signal off the tape machine, 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 pressing 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 machines. If the tape machine or noise reduction unit is arranged to switch its sync output to line input whenever the machine is in stop, fast forward, rewind, or record and only switch to sync playback off tape when the machine 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 machine and/or noise reduction perform the work of monitor switching.

The musician now hears himself live at all times except when the tape machine is actually in the sync play mode, then 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 machines.

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 record mode. At that point, most tape machines 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 machine 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 harmonizers, flangers and delay lines. It is in this mode that the outstanding flexibility of the Series 1600 becomes most apparent.

The remix mode is entered by selecting line input on each input channel (LI), which connects the output of the multitrack tape machine 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 +4dBu 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. Up to 8 mono or 4 stereo sub groups can be used, which can then be re-routed back into the stereo mix.

Additionally, the upper monitor section on each output module can be used as a line level input which can be routed either into the stereo mix by selecting SUB, or into its associated sub group, by selecting LOC (Local).

If the 8018 24 track monitor module is fitted (24 input patch bay consoles only), the outputs from 8 effects devices can be plugged into the 8 jack sockets at the top of the double width module from any of the 32 tie lines on the patchbay.

All 8 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 meters. These meters will 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 allow quality level checking of the stereo recording in the same way as described for multitrack playback.

5.00 MAINTENANCE

Every console that leaves Soundcraft undergoes a thorough testing at all stages of manufacture. These tests include individual testing of every function on all the PCB's, a thorough testing of all the functions of the completed mixer, a soak test of 48 hours before the final test, which consists of listening, measuring and mechanical function checks prior to packaging and shipment. In this way we try to ensure that any faulty components or manufacture show up long before the console leaves the company. Thus a long and trouble-free life can be expected.

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

5.01 General Fault Finding

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

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

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

The first step is to ensure that a fault really does exist! Check that the module is in the correct mode of operation, and that no jacks are inserted in the insert points, which may be interrupting the signal flow.

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

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

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

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

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

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 not recommended for the inexperienced, as it is possible to bend the connector pins if care is not taken.

5.03 METER ALIGNMENT

Each VU meter has its own individual drive card attached to the rear of the meter. This card also contains the detection and the drive circuitry for the peak LED which is pre-set to indicate a peak level of 8dB above 0VU.

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

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

Note that the peak LED will always indicate a level of 8dB above whatever the 0VU level has been set to.

5.04 LAMP REPLACEMENT

Illumination of the VU meters is provided by 2 wire ended lamps in each meter. These are 9 volt lamps wired in parallel. The lamps in each group of 4 meters are wired in series and powered by the $\pm$ 17volt audio supply. A series resistor provides turn on surge current limiting to prolong lamp life.

The stereo mix meters have an additional series resistor to simulate the voltage drop of the missing pair of meters.

To replace the lamps, first remove the VU meter front cover. This is best done by applying upward pressure to the underside of the meter front cover and then pulling the top of the cover forward. The 2 lamps will now be visible and can be unsoldered and replaced.

It is recommended that both lamps be replaced even if only one has failed, as the remaining lamp will have been overstressed and its life substantially reduced.

5.05 Power Supply Servicing

The Series 1600 Power Supply Unit provides the following regulated supply rails are provided;

- i) +/- 17 volts, Audio
- ii) + 24 volts, (not used)
- iii) + 48 volts, Phantom Power
- iv) +/- 7.5 volts, logic

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 10 Ohm, 20 Watt resistor across each of the audio supply rails and a 20 Ohm, 5 Watt resistor across the +24 volt rail is suitable. 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, and a value of at least -80dB, (ref 0.775V, DIN audio should be obtained on the audio), on the 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	C12,C14 = +26volts
	C13,C15 = -26volts
+24volt Supply	C1,C2 = +36volts
Phantom Supply	C8 = +59volts
Logic Supply	C32 = +13volts
	C33 = -13volts

Differences 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.

AUDIO SUPPLY

Theory of operation

(Numbers in brackets refer to the negative regulator)

The bipolar audio supply is a dual tracking regulator, with overvoltage protection, capable of supplying up to 5 Amps at +17volts.

The amount of current drawn is sensed by the voltage drop across the parallel resistors, R33, R34, R15 for the positive rail, and R35, R36, for the negative rail. As the current drawn increases to cause a voltage drop approaching 0.6 volt, TR17, (TR18), will begin to turn on, which in turn will turn on TR 15, (TR 16). This will starve TR 5, (TR 7), of base current, from the current source, TR 6, (TR 8), and therefore start to turn TR 19, (TR 20), (the main pass transistors), off, thereby reducing the output voltage.

Output voltage regulation is by means of TR 9, and TR 10, (TR 11, TR 12). TR 9, (TR 11), compares the voltage across zener diode ZD 2, (ZD 3), with the voltage across the output, via the potential divider R 41, R 39, (R 42, R 40), and attempts to keep them equal. For example, if the output voltage starts to rise, the voltage at the base of TR 9, (TR 11), will start to exceed the emitter voltage set by the zener diode and the transistor will start to turn on. This will turn TR 10, (TR 12), on which will starve TR 5, (TR 7), of current, turning TR 19, (TR 20), off and, therefore, reducing the output voltage accordingly.

The positive and negative regulators are made to track each other by means of the cross connection diode D 7, and TR 13, (TR 14). Under normal circumstances, when both output voltages are equal, the base of TR 13, (TR 14), will be approximately at the mid point of the supply rails, ie. at 0volts plus the diode voltage drop.

However, if for example the negative supply should start to fall, the mid point voltage will now move positive with respect to the 0V rail, and will start to turn on TR 13. This will starve TR 5, (TR 7), of current and therefore reduce the positive supply voltage to regain equilibrium between the supplies.

Servicing

As the bipolar audio supply is arranged as a tracking regulator, a fault on one half of the supply will usually affect the other half in the same manner.

The tracking facility can be disabled for test purposes by connecting each end of Diode D 7 to ground.

All transistors and diodes should be checked for normal operation, and replaced if necessary. Bear in mind that a failure of one semiconductor will often cause the failure of other associated components.

NB: During the currency of this manual, the PSU circuit diagram may change. However, the theory of operation and the component numbers referred to above will remain the same.

5.06 SOUNDCRAFT MEDIUM POWER SUPPLY UNIT

The new power supply for the Series 800B and Series 1600 provides the following regulated supply rails;

- i) +/- 17 volts, Audio
- ii) + 24 volts, (not used)
- iii) + 48 volts, Phantom Power
- iv) +/- 7.5 volts, (not used)

POWER SUPPLY SERVICING

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 10 Ohm, 20 Watt resistor across each of the audio supply rails and a 20 Ohm, 5 Watt resistor across the +24 volt rail is suitable. 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, and a value of at least -80dB, (ref 0.775V, DIN audio should be obtained on the audio), on the 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	= +26volts
	C5	= -26volts

+24volt Supply	C9	= +36volts
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Phantom Supply	C13,C14	= +59volts
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Differences 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.

MEDIUM POWER SUPPLY TECHNICAL DESCRIPTION

The operation of this power-supply is relatively conventional, and so only a brief description will be given.

Mains input is via a standard IEC mains connector, the live connection being fused by F1. This should be a 3.15 Amp anti-surge type for 220-240V operation, or a 6 Amp anti-surge for 110-120V. Mains voltage selection is by a slide-switch S1, which must be operated by the blade of a small screwdriver.

The mains transformer is a toroidal type for high efficiency and low external hum-field, with an electrostatic shield to prevent radio-frequency interference entering via the mains. There are four secondary windings; two 19V rms for the +/-17V regulators, 26V rms for the +24V regulator, and 48V rms for the +48V phantom supply.

The +17V regulator is built around an LM338K high-quality regulator IC, fed by a standard full-wave rectifier circuit. (REC1 and C1). The third (reference) terminal maintains a constant 1.2V between it and the output terminal, so that R1 and R2 form a potential divider that sets the output voltage; the voltage across R1 is constant, so increasing R2 raises the output voltage. C2 increases ripple-rejection by bypassing AC to ground. C3, C4 reduce the HF output impedance and ensure the regulator's stability.

The +24V regulator is similar to the +17V, except that it has two pre-regulator transistors, TR 6 and TR 7. These act as an emitter-follower driven by R 15, R 16, so as to divide by two the voltage drop across the regulator circuit. This reduces device temperature and ensures that REG 3 will survive a short circuit.

The +48V regulator, however, is a more complex discrete type. The reference for this supply is a Zener diode ZD1, which is decoupled by C15 to reduce internal noise. This Zener is supplied by current from the regulated output by R11. A variable proportion of the output is tapped off by R9, PR1, R10, and applied to the base of comparison transistor TR3. When this base voltage tends to rise compared with the emitter volts set by the Zener, TR3 turns on more, turning on amplifier stage TR2. This diverts current away from series-pass transistor TR5, and reduces the output voltage to the correct value. The converse happens if the output volts should start to fall.

Drive current for TR5 is provided by a constant current source TR1. This improves ripple rejection; TR1 is biased by D1,D2 so that there is a single diode-drop (0.6V) across R8.

TR4 provides overload protection. If the voltage drop across R12 becomes enough to turn on TR4 via R13, then base-drive current is shunted away from TR5 to provide current limiting.

6.00 GLOSSARY OF TERMS USED

AFL	After fade Listen: This button will "solo" the signal (or ALL with their AFL buttons down) on the monitors, and the feed for this solo is taken AFTER the fader.
Attenuate	To reduce the electrical level or amount of gain.
Auxiliary Send	Extra output from the console, usually used for echo sends and foldback.
Bus	Wire carrying a signal or sum of a group of signals.
Cold	The negative going current of a signal. With 2 signal wires, one is positive going (hot), and the other is negative going.(cold)
Cut	To cut a channel means to turn it OFF.
dB (decibel)	A logarithmic ratio used to represent voltage or power gain. The reference about which the ratio is made is usually stated.
Ground	Earth or screen of a cable when refering to connecting leads.
Group Output	The output of a group bus which is carrying a sum of all the signals assigned to that group number.
Hot	Positive going current of a signal. With 2 signal wires, one is positive going (hot), and the other is negative going.(cold)
Hz	Measurement of frequency (Hertz) 1Hz = 1 cycle per second.
Insert	An insert point allows peripheral equipment to be introduced into the signal path.
Khz	Measurement of frequency expressed to the power of 1000.i.e. 1Khz = 1000 cycles per second.
KOhm	Measurement of electrical resistance expressed to the power of 1000.i.e. 1 KOhm = 1000 Ohms.
Mains	Local Electrical Supply.
Multitrack Logic	Either the multitrack machine's monitor switching or its safe/record switching.

Ohm	Measurement of electrical resistance.
Overdubbing	The process of recording new tracks on a multitrack tape recorder whilst listening back in synchronisation with previously recorded tracks.
Pan Pot	A pan pot places a signal across two stereo lines (left & right) turning it to the left will send all the signal to the left line, and to the right, all of the signal will be sent to the right side. If the pan pot is left at its centre detent, an equal amount of signal will be fed to both sides and the image in the stereo picture will be central.
PFL	Pre fade Listen: This button will "solo" the signal (or ALL with their PFL buttons down) on the monitors, and the feed for this solo is taken BEFORE the fader.
Phantom Power	A voltage (usually +48 Volts) across the microphone input to power capacitor microphones.
POST	Post means after the fader.
PRE	Pre means before the fader.
Ring	The connecting part in the middle of a stereo jack, (it mates second).
Signal to Noise Ratio	The ratio between the level of signal and the level of unwanted noise.
Sleeve	The connecting part of a stereo jack which mates last and is always earth.
Star Point	A single point to which ALL earths are separately connected.
Sync	To be operating the multitrack machine from the synchronised head. i.e. recording and playing back onto and from the same tape head.
Tip	The connecting part at the end of a stereo jack, (it mates first).
Track Bouncing	Taking a group of previously recorded tracks and recording them as a group onto another track. e.g. bouncing down 4 vocals from 4 tracks to just one track "frees" 3 tracks for fresh recording.

7.00 SOUNDCRAFT RECOMMENDED WARRANTY

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

1. `Soundcraft' means Soundcraft Electronics Ltd.
 `End User' means the person who first puts the equipment into regular operation.
 `Dealer' means the person other than Soundcraft (if any) from whom the End User purchased the Equipment, provided such a person is authorised for this purpose by Soundcraft or its accredited Distributor.
 `Equipment' means the equipment supplied with this manual.
2. If within the period of twelve months from the date of delivery of the Equipment to the End User it shall prove defective by reason only of faulty materials and/or workmanship (but not faulty design) to such an extent that the effectiveness and/or usability thereof is materially affected the Equipment or the defective component should be returned to the Dealer or to Soundcraft and subject to the following conditions the Dealer or Soundcraft will repair or at its option replace the defective components. Any components replaced will become the property of Soundcraft.
3. Any Equipment or component returned will be at the risk of the End User whilst in transit (both to and from the Dealer or Soundcraft) and postage must be prepaid.
4. This warranty shall only be available if:-
 - a) the Equipment has been properly installed in accordance with instructions contained in Soundcraft's manual; and
 - b) the End User has notified Soundcraft or the Dealer within 14 days of the defect appearing; and
 - c) no persons other than authorised representatives of Soundcraft or the Dealer have effected any replacement of parts maintenance adjustments or repairs to the Equipment; and
 - d) the End User has used the Equipment only for such purposes as Soundcraft recommends, with only such operating supplies as meet Soundcraft's specifications and otherwise in all respects in accordance with Soundcraft's recommendations.

5. Defects arising as a result of the following are not covered by this Warranty: faulty or negligent handling, chemical or electro-chemical or electrical influences, accidental damage, Acts of God, neglect, deficiency in electrical power, air-conditioning or humidity control.
6. The benefit of this Warranty may not be assigned by the End User.
7. End Users who are consumers should note their rights under this Warranty are in addition to and do not affect any other rights which they may be entitled against the seller of the Equipment.

SERIES 800B/1600 CUSTOMER SPARES KIT

Complete Kit Part No: RZ2235

AZ2201	THERMISTOR RA53	1
AZ2202	RES NETWORK 10kx6	1
AZ2204	8P SIL 4x4K7 RES NETWORK	2
AZ2206	8P SIL 4x7K5 RES NETWORK	2
BA0001	DIODE IN4148	3
BB0106	ZENER DIODE 400MW 11V	1
BC0204	BDG RECT KBP02 200V 1.5A	1
BC0207	BDG RECT KBL02	1
BD0301	PNP TRANS 2SA842GR/2SA970GR	1
BD0302	NPN TRANS 2SC1681BL/2SC2240BL	5
BD0306	NPN TRANS BC300	1
BD0311	PNP TRANS BD136	1
BD0312	PNP TRANS PN4 355	2
BD0315	PNP TRANS TIP2955	1
BD0316	NPN TRANS TIP3055	1
BDO317	NPN TRANS BD135	1
BE0403	QUAD OP AMP IC TL074	1
BE0404	SGL OP AMP IC TL071	2
BE0407	SGL OP AMP IC TDA1034/NE5534	1
BE0413	DUAL OP AMP IC TLO72	5
BE0428	DUAL OP AMP IC NE5532	3
BE0430	V.REG +1.2/37V 1.5A (T0220)	1
BE0431	V.REG -1.2/37V 1.5A (T0220)	1
BE0438	V.REG LM338K +1.2/32V 5A (T03)	1
BFO502	QUAD 21/P NAND 4011	1
BF0505	QUAD ANGL SWITCH 4016	1
BF0517	QUAD ANALOGUE GATE DG308/CJ	1
DA0001	PREH S/SP C/D + BRAKT 4K7LIN x2	1
DA0002	PREH S/SP 41P/D 10K + LOG	3
DA0003	PREH S/SP 41P/D 10K + LOG x2	1
DA0004	PREH S/SP 41P/D + BRKT 10K - LOG	1
DA0005	PREH S/SP C/D 47K LIN	1
DA0014	PREH S/SP 10K LIN	1
DD0316	ALPS 90MM MONO FADER SA012C	1
DE0401	CERMET TRIMMER HORIZ 90H 2K2	1
DE0409	CERMET TRIMMER 90V 1K	1
DF0512	C&K PCB MNT MICRO SWITCH	1
DF0515	RAFI ILLUM PUSH SWT LATCHING	1
DF0516	ALPS (SUN TYPE) 2 POLE PUSH SWT	4
DF0517	ALPS (SUN TYPE) 4 POLE PUSH SWT	2
DL1001	NON-ILLUM ROCKER SWT DPST BLK	1
DJ0801	MARQUARDT TAP CHANGE	1
HB0124	S2400/S3B MNS TX TR7228	1
JA0001	TOSHIBA MINI LED GREEN TLG102	1
JA0002	TOSHIBA MINI LED RED TLR102	1
JA0003	TOSHIBA LED RED TLR104	1
JB0110	RS WEDGE BULB 12V T5.5	1
KA0027	ALPS SA012C FDR KNOB WHITE	1
KA0028	ALPS SA012C FDR KNOB RED	1
KA0029	ALPS SA012C FDR KNOB YELLOW	1

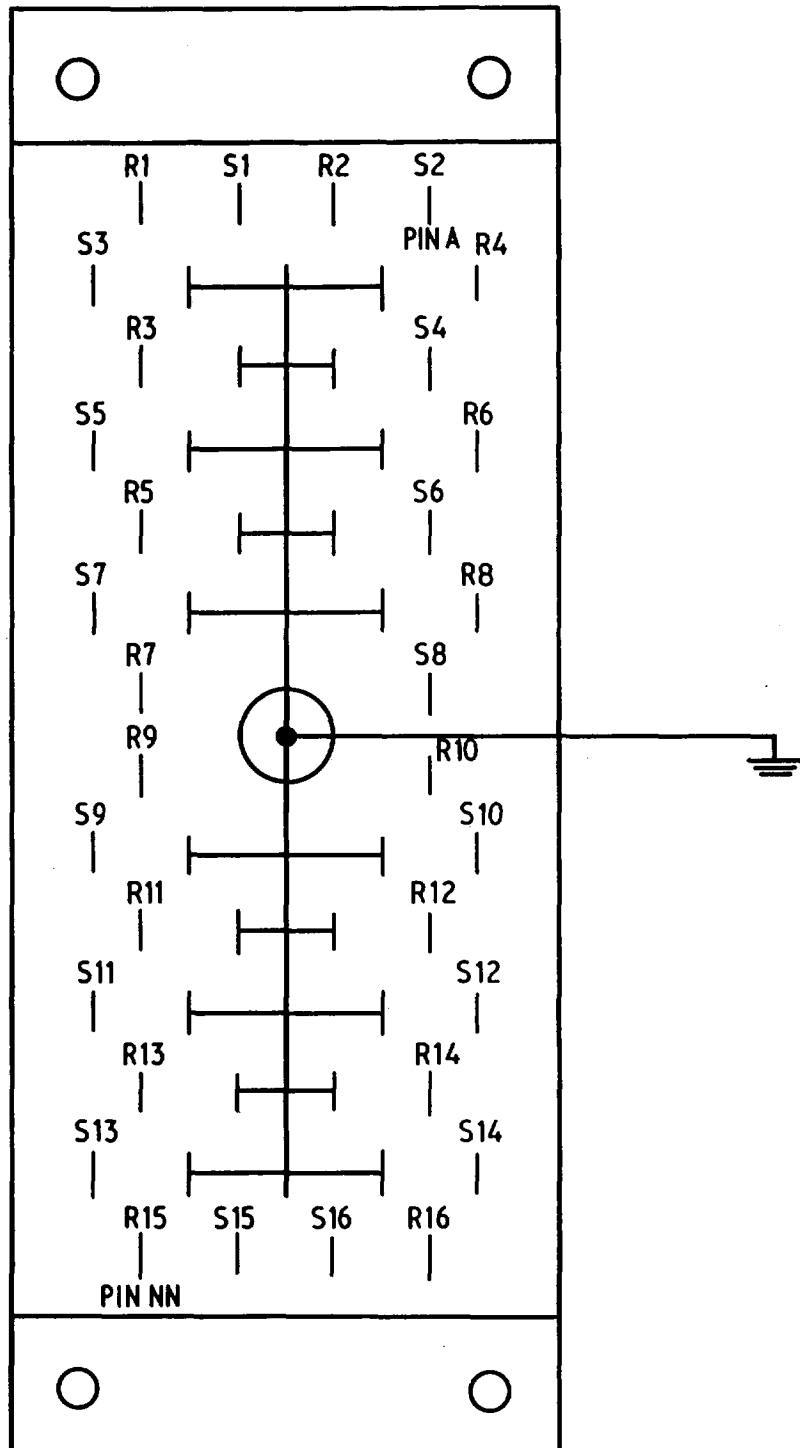
KA0030	11MM/LINE 4MMBRN KNOB RE-AN	2
KCO227	C&K 0.45"SQ SWT CAP RED	1
KCO228	C&K 0.45"SQ SWT CAP GREY	1
KCO229	C&K 0.45"SQ SWT CAP GREEN	1
KCO230	RE-AN 11MM KNOB CAP BLACK	2
KCO231	RE-AN 11MM KNOB CAP BLUE	1
KCO232	RE-AN 11MM KNOB CAP GREEN	1
KCO233	RE-AN 11MM KNOB CAP GREY	1
KCO234	RE-AN 11MM KNOB CAP ORANGE	1
KCO235	RE-AN 11MM KNOB CAP RED	1
KCO236	RE-AN 11MM KNOB CAP YELLOW	1
ZD0305	3.15A ANTI-SURGE FUSE	1
ZD0309	FUSEHOLDER (SCHURTER)	1

9.00 SCHEMATICS FOR SOUNDCRAFT SERIES 1600 - Iss 1

MI2250 STANDARD INPUT BLOCK DIAGRAM
ED2243 STEREO LINE INPUT SIGNAL FLOW DIAGRAM
MI2251 RECORDING OUTPUT BLOCK DIAGRAM
MI2253 AUX/MASTER BLOCK DIAGRAM
MI2254 MASTER MODULE BLOCK DIAGRAM

ED2102 STANDARD INPUT MODULE CCT DRG
ED2146 STEREO INPUT MODULE CCT DRG
ED2100 RECORDING OUTPUT CCT DRG
ED2109 24-TRACK MONITOR CCT DRG
ED2105 AUX MASTER CCT DRG
ED2104 MASTER CCT DRG
ED2093 STANDARD PSU CCT DRG
ED2208 MEDIUM PSU CCT DRG

S1600/S2400 EFFECTS MULTICORE
VIEWED FROM SOLDER PINS OF FREE CONNECTOR



SIMILAR FOR 17-32

SERIES 1600

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1.00

SERIES 1600

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

Auxiliary Module

Master Module

24 Track Monitor Module

Other Modules

P.S.U.

OPTIONS - SPECIFY

ALTERATIONS TO SPEC.

SPECIAL INSTRUCTIONS

DESPATCH KIT REQUIRED

1. Channel Line Input To Mix Output

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

2. Channel Line Input To Auxiliary Outputs

Auxiliary Output	THD (Ref 1kHz, +12dBv)	
1	0.0	%
2	0.0	%
3	0.0	%
4	0.0	%
5	0.0	%
6	0.0	%
7	0.0	%
8	0.0	%

3. Channel Line Input To Control Room Outputs

Control Room Output	THD (Ref 1kHz, +12dBv)	
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, +12dBv)	
Left	0.0	%
Right	0.0	%

5. Crosstalk (at 10kHz)

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
+7.5V Logic	-	dBv	Volts
-7.5V Logic	-	dBv	Volts

NOTES

OdBv = 0.775 Volt
OdBV = 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.

Formats:-

16/8/2, 24/8/2, 32/8/2, 24/8/2 with patchbay. All these have 16 track monitoring and the patchbay version has 24 track tape recorder interface.

Module types:-

8011	Standard Input
8012	Recording Output
8014	Auxiliary Master
8015	Master
8018	24 Track Monitor

Power Supply:-

External +17v/-17v/+24v/+7.5v/-7.5v/+48v
(Common to Series 2400 and Series 800B)

Meter Display:-

16/8/2 Format:	8 large VU meters switchable 1-8/9-16 plus 2 large VU meters for monitor source (mix/2trk returns/solo)
Other Formats:	16 small VU meters plus 2 large VU meters for monitor source

ALL METERS HAVE A PEAK LED SET 8dB ABOVE 0 VU

Important Features:-

- 8 sub-groups
- 16 track monitoring
- Monitors 9-16 available as effects returns
- 8 auxiliary sends from inputs and 16 monitors
- 24 input patchbay format interfaces to 24 track recorder, and up to 32 external devices

Applications:-

8/16/24 track recording

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 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 10kOhms, which is high enough to interface to any normal professional peripheral equipment, without loading the source.

a) PWR

Capacitor microphones can 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 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.

c) MIC TRIM

The microphone input can be varied between 20dB and 55dB 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 20dB PAD, a 55dB control range is available.

d) LINE TRIM

The line input gain can be varied between -10dB and +20dB 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. ie. 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 equaliser is an exceptionally versatile unit, allowing 5 areas of control over the audio spectrum. All amplitude pots are centre detented for easy zeroing, and the mid band 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, ie. the slope of the EQ curve does not keep rising with frequency, but having reached the desired amount, flattens out or "shelves" from that frequency on.

c) HI MID

The Hi Mid Frequency is continuously variable between 600Hz and 10kHz. 15dB of boost or cut is available. The response is of the "bell" type, ie. having reached maximum amplitude (or minimum in the case of cut) at the selected frequency, the amplitude response returns to zero on either side of that frequency. The shape of the curve, when plotted shows a characteristic "bell" shape. The Q of the network (a measure of 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, independant of the high pass filter.

3. AUXILIARY SECTION

There are 4 auxiliary send controls available. Each control can be routed to 2 auxiliary mix busses, to give a total of 8 auxiliary sends 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 can be switched pre-fader. Additionally, they can be routed to auxiliary busses 3 and 4. In both cases, they are post equaliser, insert point, and channel on/off switch.

b) Sends 5 and 6

Auxiliary sends 5 and 6 are permanently post-fader, and can route to busses 5 and 6, or 7 and 8.

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 "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 send jack.

b) (Prefade listen)

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