2. Channel and Group Faders

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

3. Microphone Gain

Having set both channel and group faders as above, the microphone gain is set to give the required level at the group output. The amount of gain required depends on the type of microphone, the sound pressure level developed by the instrument or voice and the distance between the sound source and microphone. In the case of high output and high sound pressure levels, it may be necessary to switch in the -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 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, 2, 3, and 4, 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 1-4, if PRE is NOT selected and also on sends 5 and 6.

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 an 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 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.

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.

This is done by selecting line input on each input channel (LI). This connects the output of the multitrack tape machine to the input of each channel, via the line trim control. This will normally be set approximately to the "0" position if the recorder is operating at a nominal +4dBu line level. However, like the mic trim, it will usually be adjusted to allow the channel fader to work near its nominal "0" position.

The channel may be routed directly to the stereo mix, by pressing Mix, or via the channel pan pot by also pressing pan to allow positioning of the signal within the stereo perspective.

Alternatively, the signal may be routed to any of the output groups, to enable the formation of sub groups. (See section 4.06)

Limiters or other devices may be connected to any input channel by patching into the channel insert points on the jackfield.

All 6 auxiliary sends may be used, to feed echo or effects devices, as headphone mixes will not be 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 VU meters.

These VU meters 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 also allows quality checking of the stereo recording in the same way as described for multitrack playback. See section 4.03.

4.06 SUB GROUPS

In the remix mode of operation, in the Series 2400 console becomes an extremely flexible mixer, due to the innovative use of the normally redundant monitor and group output section to provide effects returns and sub-groups.

All monitor inputs may be used as extra line inputs to the stereo mix buses, complete with equalisation, 6 auxiliary sends, and panning. Access to the monitor return inputs is obtained by patching into the monitor return jack on the patchbay, and selecting Tape Return on that monitor channel.

In addition to this, the group section may be used to form mono or stereo sub-groups from the input channels, to feed external equipment or even to feed the stereo mix buses, totally independent of the monitor section. If required, the auxiliary sends, and the pan pot may also be allocated to the group section, instead of being assigned to the monitor section.

This flexibility is achieved by the operation of 2 buttons, on the group and monitor module; CPG, and SUB.

Operation of these 2 buttons in various combinations will give 4 modes of operation, which will best be understood by refering to the simplified block diagram, in conjunction with the following description.

1. This is the normal recording mode, as described in section 4.02. In this mode, the group output is normalled to the tape send from where it will feed the multitrack tape machine. The multitrack tape return is normalled to the monitor return.

2. SUB (Group to mix)
   Pressing the SUB button allows the formation of sub-groups from the input channels, which are directly assigned to the stereo buses. All odd numbered groups are assigned to the left stereo bus, while even numbered groups are assigned to the right stereo bus. In this way, a number of input channels can be routed to a pair of groups, a limiter or other device patched into the group insert points to provide overall control, and by pressing SUB, the groups will be directly assigned to the left and right of the stereo mix.

   Independent of this, the monitor return section can also be used to provide a line level input to the stereo mix, via the pan pot, with the monitor equaliser, and auxiliary sends operative on this input. This is done by patching into the monitor return jack and selecting Tape Return.

3. SUB and CPG (Cues and Pan to Group)
   In this mode, the basic operation is the same as the previous mode, ie. the formation of a sub-group. However, pressing CPG transfers the auxiliary sends (Cues) and pan

pot to the sub-group, and away from the monitor return. This enables the sub-group to be panned between the left and right stereo buses, and the auxiliary sends to be used to provide signals for echo, or other devices from the sub group.

The monitor return section is still available for use, but now is assigned directly to left and right of the stero mix depending on whether it is an odd or an even numbered monitor. This is useful as a direct stereo return into the stereo mix complete with equalisation.

4. CPG
When CPG is selected the auxiliary sends and pan pot are disconnected from both the monitor and group signal chain. This allows the formation of a stereo sub group with EQ, by setting the monitor faders to the unity gain position, and not selecting tape return. All odd numbered groups will route to stereo mix right. This signal path is also available in mode 1, with the addition of auxiliary sends and panning to the stereo mix.

In modes 1 and 4, if tape return is selected, the group output has no direct route to the stereo mix, but is available only at the jackfield. In this way, it can be used a an additional effects send group, formed from any number of input channels.

The monitor return input, ofcourse, is now available as an effects return, directly routed to the stereo mix. Used in this way, the group output becomes the send to an effects device, eg. delay line, harmoniser etc. while the monitor return becomes an equalised effects return to the stereo mix, conveniently located close to the effects send.

5. Fader Reverse (FDR)
In any mode, the functions of the long group fader and the short monitor fader may be interchanged. This allows the engineer to have all monitor returns controlled by the long faders, while group outputs are controlled by the short faders. During remix, this means that all line inputs into the stereo mix are controlled by the high resolution conductive plastic faders.

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 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 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 cicuits, 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.

NB: A module should not be removed or plugged in with the power still switched ON. Care should be taken that the connector pins are not bent when removing or replacing a module.

5.03 REMOVING PATCH CARDS

The entire patchbay is made up from rows of plug in patch cards, constructed with printed circuit ounting jack sockets. Each row can be easily removed for cleaning and replacement if this should prove necessary.

Remove the 4 patch bay retaining screws, located at the corners of the patchbay. This will allow the entire assembly to be lifted up and tilted to one side. Take care to avoid scratching the paint.

Locate the card which is to be removed and unplug the ribbon cables connected to it, and if necessary, to the adjacent cards. Remove the 2 screws holding the card to the panel and withdraw the card from the assembly.

Each individual jack socket has a dust cover which can be unclipped for cleaning or adjusting the contacts.

Reassembly of the patchbay is the reverse of the above procedure.

5.04 VU 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 drive circuitry for the Peak LED which is preset to indicate a peak level of 8dB above 0VU.

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

Connect a millivoltmeter to the output of the console. 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 preset to indicate 0VU on the VU meter and repeat for all other groups and the stereo mix meters.

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

5.05 LAMP REPLACEMENT

Illumination of the VU meters is provded 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 ±17volt audio supply. A thermistor and a series resistor provide turn on surge current limiting to prolong the 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 front cover of the VU meter. This is best done by applying upward pressure to the underside of the meter front cover and then pulling the top of the front 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.

5.06 LED METERS

The LED metering option consists of an array of 40 segment bargraph displays. In response to various commands from the Meter Control PCB (See ED2092) the display will function in either peak or VU modes. In addition the group meter section can also be set to operate as a 1/3 octave spectrum analyser of the signal present in the main stero mix meters. This allows spectrum monitoring of the entire mix as well as any individual soloed channel. When operating in the spectrum mode an extra slow response can be selected which will better indicate the average frequency content of the signal.

Each meter display is controlled by a Rectifier PCB (See ED2091) which performs full wave rectification of the audio signal as well as the integration time function for Peak, VU and Slow modes. Control voltages A,B, and C from the Meter Control PCB enable CMOS switching gates (IC4) on the Rectifier PCB which alter the RC time constant of the integration stage. (IC3).

The 3 modes respond to the following truth table:

VU ....A $\overline{B}$ C

Peak....X B $\overline{C}$

Slow....$\overline{A}$ $\overline{B}$ C

Note...X = Dont Care

The signal input to each Rectifier PCB is also controlled by command voltages from the Meter Control PCB. These commands are Group Enable, Spectrum Enable, and Slate, which connect the Rectifier PCB input to the Group Output, the Spectrum Analyser board, or to the Slate bus for alignment purposes. This switching is performed by another CMOS switch, IC1.

METER ALIGNMENT

If IC3 has been replaced for any reason it may be necessary to readjust the DC offset from the Rectifier PCB. Connect a DC. meter or an oscilloscope to the output of the Rectifier PCB. Adjust the 22kOhm preset connected to IC3 for 0volts offset or until the last segment of the display is extinguished. This preset can also be used to correct for level tracking inaccuracies at low signal levels.

Signal level adjustment is performed in the same way as described for VU meter alignment, using the preset connected to IC2.

SPECTRUM ANALYSER BOARD

The Spectrum Analyser filtering function is achieved by using 9 Reticon R5604 switched capacitor filter integrated circuits. Each IC covers an octave and contains 3 1/3 octave bandpass filters. The centre frequency of each filter is directly proportional to the clock frequency applied to the chip. As each chip covers an octave all that is needed is to divide the clock by a factor of 2 for each subsequent lower octave. The clock frequency is divided by IC5, which is a 4069 CMOS divider chip. The main system clock is generated by IC4. For correct standard filter frequencies the clock frequency measured at pin 7 (Reticon clock input) of IC8 should be 1395.2kHz.

The incoming signal to the Spectrum Analyser Board passes through a series of anti-aliasing low pass filters, IC1, IC2, and IC3, which remove any high frequency signals which could cause interaction problems with the clock frequency.

The small sub PCB (RSC190) sums the left and right signals from the main stereo mix meters to provide the signal to be analysed. The signl to the Slate input of each Rectifier Board is also derived from this board.

5.05 Power Supply Servicing

The Series 2400 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 positiove 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 negataive 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.

5.06 SOUNDCRAFT MEDIUM POWER SUPPLY UNIT

The new power supply for the Series 2400 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
Phantom Supply	C13,C14	= +59volts

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 sheild to prevent radio-frequency interferance 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 based 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 vie 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 logarthmic 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. 1kOhm = 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	Used whilst overdubbing; previously recorded tracks are played back through the record head whilst you record on other tracks.
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.

8.00 SERIES 2400 CUSTOMER SPARES KIT LIST

COMPONENT ITEM NO.	DESCRIPTION	QUANTITY
BB0106	ZENER DIODE 400MW 11V	4
BC0204	BDG RECT KBP02 200V 1.5A	4
BC0207	BDG RECT KBL02 200V 4A	4
BC0208	BDG RECT RS262-315 200V	4
BD0301	PNP TRANS 2SA842GR/2SA97	10
BD0302	NPN TRANS 2SC1681BL/2SC2	4
BD0303	PNP TRANS BC143	4
BD0304	PNP TRANS BC214L	4
BD0306	NPN TRANS BC300	4
BD0311	PNP TRANS BD136	4
BD0312	PNP TRANS PN4355	24
BD0315	PNP TRANS TIP2955	4
BD0316	NPN TRANS TIP3055	4
BD0317	NPN TRANS BD135	4
BE0403	QUAD OP AMP IC TL074	6
BE0404	SGL OP AMP IC TL071	6
BE0407	SGL OP AMP IC TDA1034/NE	6
BE0423	DUAL OP AMP IC UAF772	6
BE0427	LED BAR DRIVER IC LM3914	4
BE0428	DUAL OP AMP IC NE5532	6
BE0430	V.REG +1.2/37V 1.5A (TO2	4
BE0431	V.REG -1.2/37V 1.5A (TO2	4
BE0433	RETICON 1/3 OCT FILTER R	2
BF0502	QUAD 21/P NAND 4011	20
BF0505	QUAD ANLG SWITCH 4016	6
BF0514	12WY DIVIDER 4040	4
BF0515	HEX INVERTER 4069	4
DA0001	PREH S/SP C/D +BRKT 4K7LI	8
DA0002	PREH S/SP 41P/D 10K+LOG	8
DA0003	PREH S/SP 41P/D 10K+LQG	4
DA0004	PREH S/SP 41P/D +BRKT 10K	4
DA0005	PREH S/SP C/D 47K LIN	8
DA0006	PREH S/SP 41P/D 100K- x2	8
DD0301	RUWIDO MONO FDR 10K LOG	4
DD0309	P&G 105MM MONO 1120/D D2	2
DD0310	P&G 105MM STER 1122/D D2	2
DF0503	SCHADOW 2 POLE PUSH SWIT	8
DF0504	SCHADOW 4 POLE PUSH SWIT	4
DF0512	C&K PCB MNT MICRO SWITCH	2
DJ0801	MARQUARDT TAP CHANGE	2
DL1001	NON-ILLUM ROCKER SWT DPS	2
HB0124	S2400/S3B MNS TX TR7228	2
JA0001	TOSHIBA MINI LED GREEN T	8
JA0002	TOSHIBA MINI LED RED TLR	8
JA0003	TOSHIBA LED RED TLR104	8
JA0016	LED ARRAYS 10WY BAR D.I.	4
JA0017	LED ARRAYS 10WY BAR D.I.	4
JB0102	INDICATOR 14V MA/68 RED	4
JB0114	BULB FOR COMPONEX R55 MT	40
JB0120	9V BULB FOR COMPNX R45 M	40

COMPONENT ITEM NO.	DESCRIPTION	QUANTITY
JD0310	COMPONEX VU MTR R55	4
JD0312	S1624 VU MTR R45	4
KA0002	11MM/LINE 4MM PUSH ON BR	10
KA0013	P&G 1120 FDR KNOB 15MM Y	2
KA0015	P&G 1120 FDR KNOB 15MM R	4
KA0018	RUWIDO 1120 FDR KNOB GRE	4
KA0020	P&G 1120 FDR KNOB 15MM W	4
KC0206	CAP C110 BLACK	4
KC0207	CAP C110 BLUE	4
KC0208	CAP C110 GREEN	4
KC0210	CAP C110 ORANGE	4
KC0211	CAP C110 RED	4
KC0212	CAP C110 YELLOW	4
KC0223	D6 SNAP CAP GREY	4
KC0224	C&K PCB MNT MICRO SWT CA	2
KC0225	C&K PCB MNT MICRO SWT CA	2
KC0226	C&K PCB MNT MICRO SWT CA	2
ZB0107	EAGLE MIC CAPSULE C1200	2
ZD0305	3.15A 20MM ANTI-SURGE FUSE	4
ZD0307	6.3A 20MM FUSE	4
ZD0309	FUSEHOLDER (SCHURTER)	4