

Soundcraft

SPIRIT

Live

LEFT

RIGHT

MONO

USERS

GUIDE

SPIRIT

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Issue 1
Part No. ZZ2655

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User Guide

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SPiRiT *Live*

INTRODUCTION

Congratulations on your purchase of a **SPiRiT Live** mixer. Owning a Soundcraft console brings you the expertise and support of one of the industry's leading manufacturers and the results of over 17 years experience supporting some of the biggest names in the business.

Designed by engineers who understand the individual needs of musicians, **SPiRiT Live** has been built to the highest standards using quality Japanese components and employing automated assembly techniques beyond the reach of most manufacturers of compact mixers.

A rugged steel chassis is combined with moulded side trims to give protection and distinctive appearance. Custom-moulded controls, designed for the best 'feel' and visual clarity complement the styling, resulting in a truly professional product which is ideal for both touring and fixed PA installations.

SPiRiT Live is available in 8, 16 and 24 channel frame sizes, and the 8 and 16 may be extended by adding an 8 channel Expander. By releasing the right hand side end trim the Expander can be attached quickly and securely to the console, requiring only a screwdriver to complete the installation.

SPiRiT Live incorporates circuit technology identical to that used on some of the most sophisticated Soundcraft consoles. The input channels are able to accept a wide range of Microphone and Line level signals from separate input sockets. Every channel features wide range gain control and Line input pad, 3-band Equalisation with swept Mid and LF range, plus a Hi-Pass Filter, 4 Auxiliary Sends, PFL(Pre Fade Listen), Peak LED, Panning to a Stereo Bus and separate routing to a Mono Bus. The additional bus makes **SPiRiT Live** unique at this price level and is ideal for driving a centre loudspeaker cluster, or with the Left and Right busses used as subgroups and mixed down to the Mono bus as the main output. Each channel is controlled by a high-quality long throw fader.

The Master section provides master level control for the Left, Right, Mono and Auxiliary Send busses, with separate AFL monitoring on each Auxiliary Send. Four Stereo Effects Returns are provided, arranged in pairs, each of which can be monitored and assigned to the stereo bus. Two of these Effects Returns have a SLOPE control which provide a basic bass/treble equalisation.

A switch is included which enables the L/R outputs to be added to the Mono mix if required. The three main outputs all have insert points for the connection of external signal processing.

The Monitor section allows the engineer to meter and listen to either, or both, the stereo(L/R) or Mono outputs on headphones. Two 16-segment 3-colour LED bargraph meters provide clear display of the selected signals. Pressing any PFL or AFL switch puts the selected signal onto both sides of the headphones, and the right meter. The meters are factory set to PEAK characteristic, but may be changed internally to VU characteristic if required. Refer to the Selectable Options section on Page 26 for details.

SPiRiT Live is designed to be as user-friendly as possible, but a few minutes spent reading through this manual will help you become familiar with the product away from the pressure of a live session, and allow you to gain full benefit from the superb performance offered by your new mixer.

Above all, remember that your **SPiRiT** mixer is designed to extend your creativity. The more you explore the controls and the effect they have on the sound output, the more you will appreciate how you can influence and enhance the final sound.

BASIC PRINCIPLES OF PA MIXING

There was a time when the P.A. system and the operator existed only to increase the overall volume of the performers, so that they could be heard in a large room or above high ambient noise levels. This just isn't true any more. The sound system and the sound engineer have become an integral part of the performance, and the artists are heavily dependent on the operator's skill and the quality of the equipment.

The following introduction to the basics of mixing are included for the benefit of those users who may not have any significant familiarity with sound equipment, and who are baffled by the endless jargon used by engineers and artists alike.

The Mixer

As one would expect, the main purpose of the mixer is to combine sounds, but under precise and smooth control. This is why long-throw faders are essential on any professional product. The faders provide you with total control of the final sound at your finger tips and like an artist playing an instrument you should listen to your fader movements, not look at your hands.

SPIRIT *Live*

Your **SPIRIT Live** mixer accepts a wide range of input signals via a microphone input, for very low level signals, or a line input, for higher level signals from, for instance, tape machines, effects processors, etc.

The mixer is split into two sections. The **Inputs** receive, match and process individual source signals, and distributes them at precise mix levels to a choice of outputs. The **Master** section allows overall level control of all outputs, and provides monitoring of the audio signal at many points in the mixer, either on headphones or meters.

The **Equaliser** controls are the most flexible and potentially destructive feature of the mixer. They have a similar effect on the frequency response of the input channel as the tone controls on a hi-fi system, but with much greater precision, and allow particular characteristics of the input signal to be emphasised or reduced. It is very important that you become familiar with the effect each control has on the sound and this is best achieved by spending time *listening* to the effect of each control on a well-known track played through the mixer.

The **Auxiliary Sends** provide a way of routing the input signals to a number of secondary outputs, for artists foldback, echo units or additional speaker outputs.

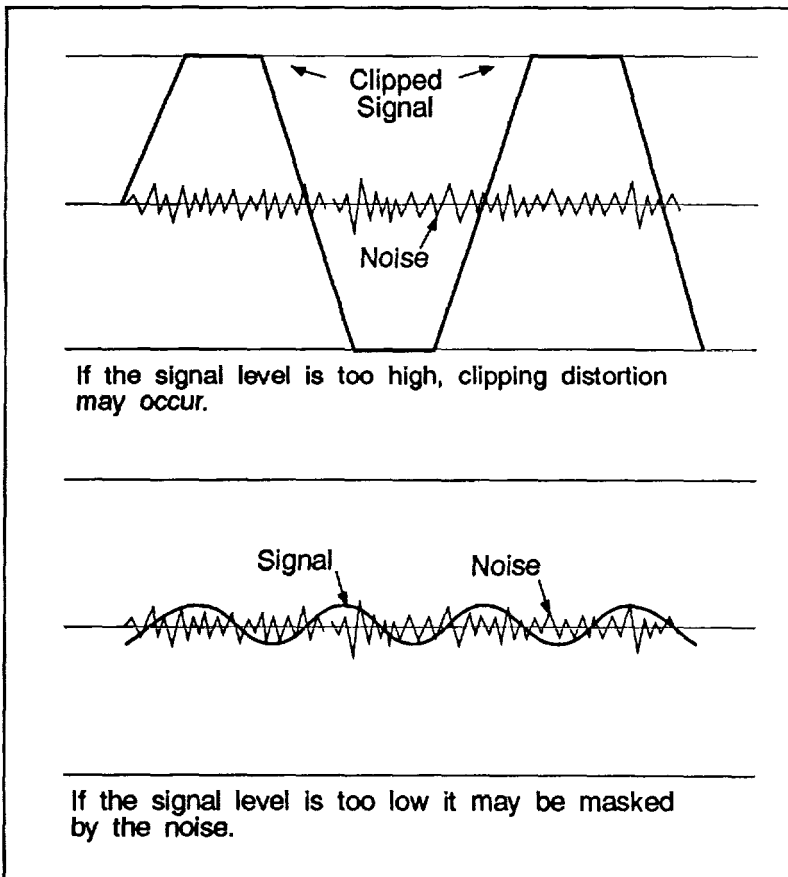
The **Pan** control adjusts the position of the input signal within the stereo mix, and can be swept from full left, through to full right. This allows particular artists to retain their correct spatial position within the mix, and can be valuable for live effects.

Pre-Fade-Listen(PFL) allows you to monitor the signal at many points in the mixer. Pressing any PFL switch places the signal at that particular point onto the headphones and the right meter, to check the quality of the signal or to pin-point problems. Using PFL will not affect the signals on the L/R or Mono outputs.

Each input channel and the three main outputs have an **Insert 'A'** gauge jack socket, which is a break point in the signal path. It allows the signal to be taken out of the mixer, through an external piece of equipment and then back into the mixer directly after its original exit point. The Insert point is normally bypassed by the 'A' gauge jack socket contacts, and is only brought into operation when a plug is inserted. Typical uses would include Effects Processors, Limiters, additional Equalisers or Delay units.

The terms **PRE** and **POST** are often used in the context of Inserts, Equalisers and Auxiliary Sends, and describe whether that facility is placed before (Pre) or after (Post) another particular section. This is explained further in the detailed description of facilities.

A mixer is often judged, amongst other factors, by the amount of Headroom available. This is a measure of the reserve available to cope with sudden peaks in the input signal, without distortion caused by Clipping, when the signal becomes so high that it would exceed the power supply rail voltages and is as a result limited. This commonly occurs where gain settings are incorrectly set or where sources are improperly matched to the mixer input. If the source signal is too high, clipping and distortion results. If the signal is too low it becomes masked by the background noise which is present to some degree in all mixers. The diagram below illustrates this point.

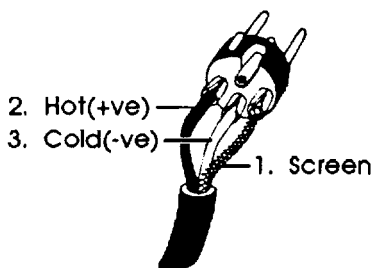


GETTING STARTED

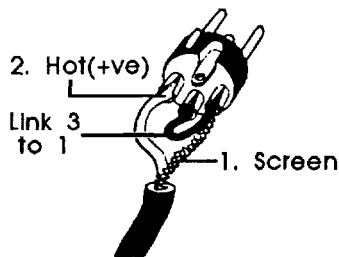
CONNECTIONS AND CONNECTORS

Although this may seem a simple subject, *faulty connectors and cabling are the source of most sound system problems*. Correctly-made cables of the proper type, with the right connectors for the job will ensure peak performance from your system with minimum noise pick-up. The following section will help you to connect **SPIRIT Live** correctly.

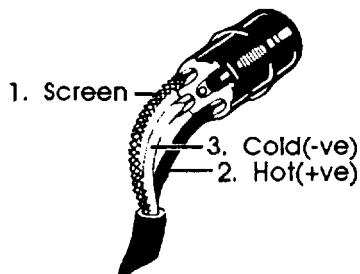
Two different types of audio connectors are used, 3-pin XLR and 1/4" three pole ('A' gauge) jacks. These are used in several configurations as shown in the diagrams below.



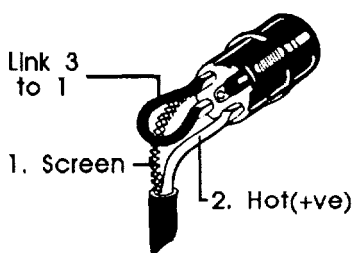
Balanced
Input



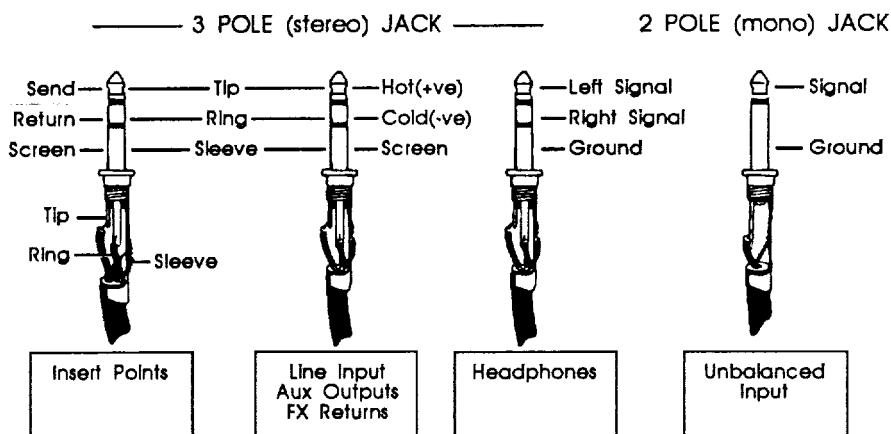
Unbalanced
Input



Balanced
Output



Unbalanced
Output



Balanced and Unbalanced

All channel inputs are balanced, i.e. there are separate +ve(hot) and -ve(cold) wires for each signal plus a ground. The design of the differential input amplifiers is such that interference picked up on these wires is cancelled out. This is because, since both wires are in close proximity, the *same* interference will be picked up on each wire and balanced input amplifiers will only amplify the *difference* between +ve(hot) and -ve(cold). Any signal on both hot and cold (i.e. noise) will not be amplified - this is known as common mode rejection (CMR). Balanced inputs should always have both +ve and -ve connected or if only an unbalanced source, the -ve pin shorted to ground.

Note: many modern audio/musical instruments have electronically balanced outputs which should not be unbalanced by shorting one wire to ground. Always use your inputs balanced where possible.

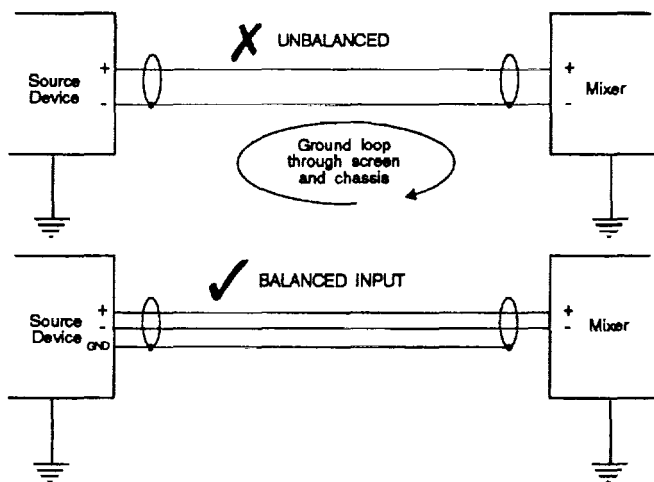
The three main outputs and the auxiliary outputs are ground compensated and provide a very effective way of optimising noise immunity, without the cost and complexity of balanced outputs. These outputs employ ground compensation techniques to cancel out the effects of variation in ground potential between the mixer and other equipment which would otherwise show up as hum. If the output is driving a device or amplifier that has an unbalanced input, connect the -ve(cold) signal to the ground at the destination, not at the output of your SPIRIT *Live* console.

Polarity

You will probably be familiar with the concept of polarity in electrical signals and this is of particular importance to balanced audio signals. Just as a balanced signal is highly effective at cancelling out unwanted interference, so two microphones picking up the same signal can cancel out, or cause serious degradation of the signal if one of the cables has the +ve and -ve wires reversed. This phase reversal can be a real problem when microphones are close together and you should therefore take care always to connect pins correctly when wiring audio cables.

Grounding and Shielding

For optimum performance it is vital that all signals are referenced to a solid, noise-free earthing point and that all signal cables have their screens connected to ground. To avoid earth 'loops', use balanced connections where possible and ensure that all cable screens and other signal earths are connected to ground only at their source and not at both ends.



Avoid running audio cables or placing audio equipment, close to thyristor dimmer units or power cables.

Noise immunity is improved significantly by the use of low impedance sources, such as good quality professional microphones or the outputs from most modern audio equipment. Avoid cheaper high impedance microphones, which may suffer from interference over long cable runs, even with well-made cables.

Fault Finding Guide

Repairing a sound mixing console requires specialist skills, but basic *Fault Finding* is within the scope of any user if a few basic rules are followed.

- Get to know the Block Diagram of your console (see rear cover)
- Get to know what each component in the system is supposed to do.
- Learn where to look for common trouble spots.

The **Block Diagram** (see inside rear cover) is a representative sketch of all the components of the console, showing how they connect together and how the signal flows through the system. Once you have become familiar with the various component blocks you will find the Block Diagram quite easy to follow and you will have gained a valuable understanding of the internal structure of the console.

Each **Component** has a specific function and only by getting to know what each part is supposed to do will you be able to tell if there is a genuine fault! Many 'faults' are the result of incorrect connection or control settings which may have been overlooked.

Basic Troubleshooting is a process of applying logical thought to the signal path through the console and tracking down the problem by elimination.

- Swap input connections to check that the source is really present. Check both Mic and Line inputs.
- Eliminate sections of the channel by using the insert point to re-route the signal to other inputs that are known to be working.
- Route channels to different outputs or to auxiliary sends to identify problems on the Master section.
- Compare a suspect channel with an adjacent channel which has been set up identically. Use PFL and AFL to monitor the signal in each section.

GETTING TO KNOW YOUR CONSOLE

FACILITIES

Refer to the fold-out front panel diagram which shows the control functions on the **SPiRiT Live**. Each facility is described below, and is identified by the reference number.

INPUT CHANNEL

1. MICROPHONE INPUT

The Microphone input is via a standard female XLR-3 connector and is available when the LINE switch is released. It is designed to accept a wide range of balanced or unbalanced low impedance input signals.

+48V Phantom Power is available on each input microphone socket. Should you wish to configure inputs without +48V Power, see Selectable Options on Page 24 for details of this modification.

Transformer-coupled dynamic microphones may be used without causing damage, even when the +48V power is connected, but care must be taken when using unbalanced sources, because of the voltage present on pins 2 and 3 of the XLR connector. Microphone input level is set by the GAIN control(5).

NOTE: Phantom powered mics should not be plugged in with the +48V switched on. Also you should be aware that some microphones draw an unusually large current which may overload the power supply, resulting in distortion. Consult your microphone supplier for guidance if necessary.

2. INSERT

The INSERT is a break point in the input channel signal path. It allows the signal to be taken out of the mixer, through an external piece of equipment and then back into the mixer to continue through to the final output. The Insert is a 3-pole ¼" 'A' gauge Jack Socket, which is normally by-passed. When a jack plug is inserted, the signal path is broken at a point just after the Hi-Pass Filter, but before the EQ section. The signal from the channel appears on the TIP of the plug and is returned on the RING. The insert point allows limiters, compressors and other signal processing units to be added as required to particular input channels and because it is located PRE EQ, noise generated by the external equipment may be reduced by a small amount of H.F. cut in the Equaliser.

3. LINE INPUT

The LINE Input is a 3-pole ¼" A' gauge jack socket, to accept balanced or unbalanced line level sources when the LINE switch(4) is pressed. Unlike the low impedance Microphone input, this stage presents a high impedance(>10kΩ) to the input signal, enabling many types of instruments to be plugged straight in without D.I. boxes or external preamplifiers.

Line inputs will be found useful as extra Effects Returns, where additional post-effect equalisation is required.

4. LINE SELECT

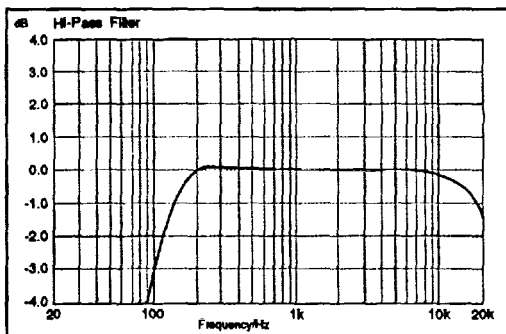
The LINE switch selects Line input when pressed, and Microphone input when released. When Line is selected the Gain range is reduced by 20dB(see 5 below).

5. GAIN CONTROL

When the Microphone input is selected this control acts as a SENSITIVITY control covering a 50dB range. Channel signal level increases as the control is turned clockwise. When the Line input is selected it serves as a GAIN control, with the scaling reduced by -20dB from the printed scale. A nominal 0dB input signal will pass through at unity gain, with the knob set at the 20dB position. Some audio equipment, particularly that intended for domestic use, operates at a nominal -10dBV level and an increased Gain setting will be required.

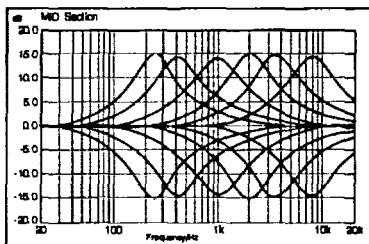
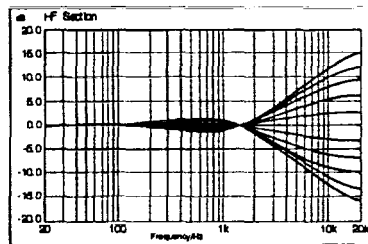
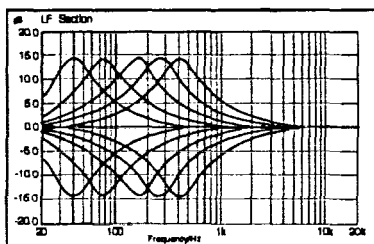
6. HI-PASS FILTER

Pressing this switch inserts a 12dB per octave 100Hz Hi-Pass Filter in the signal path, immediately after the input amplifier. This is particularly useful on live vocals, and its use is strongly recommended, even on male vocals. It can also be used for filtering out low frequency hum.



7. EQUALISER

The Equaliser(EQ) comprises three sections. The upper control provides H.F.(treble) boost and cut of $\pm 15\text{dB}$. The lower two pairs of knobs are arranged as a cut/boost control (lower knob) of $\pm 15\text{dB}$, and a SWEEP(frequency) control which determines at which frequency the boost/cut action will be centered. The LF sections are essentially Swept Low Frequency controls, covering a range from 40 to 400Hz and providing much greater flexibility than a conventional LF(bass) control. The MID section, with a frequency range from 250Hz to 8kHz is particularly versatile for vocals, enabling particular characteristics of the singer to be lifted or suppressed very precisely.



8. AUXILIARY SENDS

These controls route the input channel signal to any one or more Auxiliary busses. These are separate from the main outputs and can therefore provide additional outputs for foldback, echo units or extra loudspeaker 'fills'.

AUX 1 & 2 are normally derived before the channel fader(PRE FADE), and are therefore unaffected by the fader position. This makes them particularly suitable for foldback or monitor feeds, which need to be controlled separately from the main P.A. mix. These feeds are muted if neither of the routing switches (see 11/12) are selected.

AUX 3 & 4 are derived after the channel fader(POST FADE), and therefore follow any changes in fader level. They are normally used to drive effects processing units which are fed back into the mixer and which must fade out with the input channel.

AUX 2 may be altered internally to be POST FADE. Refer to the Selectable Options section (Page 27) for a description of this modification.

9. PAN

The Pan control determines the position of the signal within the stereo image. Rotation fully anticlockwise feeds the signal solely to the Left mix buss, while rotation clockwise sweeps the image to the right.

10. PFL/PEAK LED

When the PFL switch is pressed, the Pre-Fade signal is fed to the headphones, where it replaces the selected source. The PFL/AFL Led on the master section illuminates to warn that the monitor and the meters are now responding to the PFL/AFL selection and the PFL Led on the input channel lights to identify the active channel. This is a useful way of listening to any required input signal without interrupting the main mix, so that adjustments can be made or problems traced.

When the PFL switch is released the Led on the channel serves as a PEAK indicator, to warn when an excessively high signal level is present in the channel. The signal is sampled at two points in the channel, PRE INSERT, and POST EQ. The Peak Led will illuminate approximately 4dB before clipping and therefore give warning of a possible overload even if the peaks are removed by external equipment plugged into the Insert.

11/12. MONO & L-R

The input channel signal is routed to the main STEREO mix (L-R) or a separate MONO mix buss, by pressing the respective switches. If either or both switches are pressed, the ON Led illuminates to show that the channel is contributing to the output and that any Auxiliary Send is potentially active.

13. CHANNEL FADER

This long-throw fader determines the proportion of the channel in the mix and provides a clear visual indication of channel level. Normal operating position is at the '0' mark, providing 10dB of gain above that point if required.

MASTER SECTION

1. MAIN OUTPUTS

The LEFT, RIGHT and MONO outputs are standard male XLR-3 connectors, driven by ground-compensated output amplifiers.

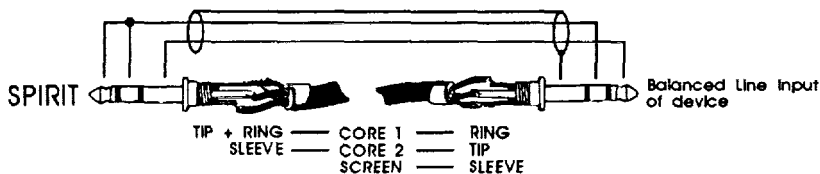
2. D.C. POWER SOCKET

The cable from the power supply connects to the console via this 6-pin socket.

3. INSERTS

These are similar to the Input Channel Inserts and allow external processing equipment to be 'inserted' into the output signal path. The 1/4" 3 pole 'A' gauge jack sockets are by-passed except when a plug is inserted. It should be noted that the signal at these sockets is the reverse phase to the inputs. If you need to use this signal to feed another device in phase, a special lead will be required as shown below.

NOTE: This does not apply when *Inserting* a processing device using these sockets.



4. EFFECTS RETURNS

Four Stereo Effects Returns are provided on pairs of 1/4" 3 pole 'A' gauge jack sockets, to allow external equipment to be returned to the mixer and routed to the stereo mix, without using up valuable input channels. They are used in conjunction with the SLOPE (9) and LEVEL (10) controls described below. A mono return signal may be plugged into the upper socket only of each pair to be fed equally to left and right busses. The Effects Returns are unbalanced - the sockets automatically unbalancing any source connected to them.

5. AUXILIARY OUTPUTS

The auxiliary outputs are standard 1/4" 3 pole 'A' gauge jack sockets, driven by ground-compensated output stages.

6. TAPE RECORD

Separate Left and Right Tape Record feeds are provided from the main outputs at an nominal -10dBV level, suitable for most Hi-Fi or semi-professional tape machines.

Internal links allow the Mono output to be added to these feeds if required. These links are not fitted at the factory, but this modification is described in the Selectable Options section on page 25.

7. BARGRAPH METERS

Two sixteen segment, three colour bargraph meters provide visual monitoring of output levels as selected on the Mono and L-R switches (13, 14). These are factory set to a PEAK characteristic, but may be changed internally to a VU characteristic. Please refer to the Selectable Options section (Page 26) for details.

Normally the meters display Left and Right or Mono signals. If any PFL or AFL switch is activated the left meter is turned off and the right meter displays the level of the selected PFL or AFL signal. Note that if under normal operation L-R and Mono are selected together, although the mixer output will be true, the additive effect on the combined meter signals may produce an inaccuracy of up to 3dB on the meter reading.

The bargraphs may be calibrated by trimmers fitted on the edge of the PCBs and accessed via holes in the panel above each meter. Adjustments may be made using a small screwdriver, taking care not to damage the trimmers.

8. AUXILIARY MASTERS

Each of the four Auxiliary Send busses is provided with a rotary master level control and an AFL switch with indicating LED which monitors the final output *after the fader*.

9. SLOPE

The Slope control provides a method of basic equalisation for Effects Returns 1 and 3. Rotation anticlockwise provides 6dB of bass boost plus 4dB of Treble cut. Clockwise rotation gives 6dB of treble boost and 4dB of bass cut.

10. EFFECTS RETURN LEVEL

Each Effects Return has a master level control, to accomodate a wide range of input signal levels.

11. AFL/L-R

These switches route each pair of Effects Return signals to the stereo mix (L-R) or to the monitor (AFL) after the level control. Leds show those switches that are active.

12. L-R TO MONO

This switch feeds the final Left and Right outputs onto the MONO bus, allowing the stereo bus to be used as two mono subgroups mixed down to a single output when stereo is not required.

This function is also a quick method of utilising the mono bus to restore the output if a fault occurs on external equipment (perhaps the power amplifiers or speakers) fed from the stereo output of the mixer, or for reinforcing the stereo mix in the centre cluster of large sound systems where the distance between the left and right loudspeaker is considerable.

13, 14. MONITOR SELECTION

The headphones and the meters will normally be driven as determined by these switches. Note the comment regarding level addition in (7) above.

15. HEADPHONE VOLUME

The master volume control for the monitor, PFL and AFL output signals to the headphone socket (17).

Note that due to the way that the ground-compensated outputs operate, it is possible that a hum might be audible under AFL listening, which is in fact cancelled on the actual output. This would indicate that there is a ground loop in your system which you may wish to correct.

16. OUTPUT MASTER FADERS

Master faders for Left, Right and Mono outputs. Unity gain is at the top of their travel.

17. HEADPHONE SOCKET

A standard ¼" 3 pole 'A' gauge jack socket for monitoring headphones. See the diagram on Page 7 for connection details.

This output is suitable for headphones with an impedance of 400Ω or greater.

USING YOUR SPIRIT *Live* CONSOLE

The final sound from your P.A. system can only ever be as good as the quality of the source signal. Just as you need to become familiar with the control functions of your mixer, so you must recognise the importance of correct choice of inputs, microphone placement and input channel settings. However, no amount of careful setting up can take account of the spontaneity and unpredictability of live performance and the mixer must be set up to provide 'spare' control range to compensate for changing microphone position and the absorption effect of a large audience (different acoustic characteristics from sound check to show). Suitable initial control positions are shown on the front panel drawing inside the rear cover.

Initial Set Up

The diagram on page 5 demonstrated how the matching of input gain to the signal source was crucial to avoid distortion at one extreme and excessive noise at the other. Set up individual input channel as follows:

- Connect the input required (microphone, keyboard etc.)
Note: Phantom powered mics should be connected before the +48V is switched on.
- Set Master faders at 0, input faders at 0, and set power amplifier levels to about 70%.
- Provide a typical performance level signal and press the PFL button on the particular channel, monitoring the level on the right-hand meter.
- Adjust the input gain until the meter is just reaching the amber section (0dB) at a typical maximum source level. This allows sufficient head-room to accommodate peaks and establishes the maximum level for normal operation (but see note below).
- Repeat this procedure on other channels as required. As more channels are added to the mix, the meters may move into the red section. Adjust the overall level using the Master Faders if necessary.
- If you cannot obtain a satisfactory setting, e.g. the gain control is right at the extreme low end of the scale on Microphone Input, try using the Line Input instead.
- Listen carefully for the characteristic sound of 'feedback'. If you cannot

achieve satisfactory input level setting without feedback, check microphone and speaker placement and repeat the exercise.

You are now ready to start building the mix and this should be done progressively, listening carefully for each component in the mix and watching the meters for any hint of overload. If this occurs, back off the appropriate Channel Fader slightly until the level is out of the red segments, or adjust the Master Fader.

Note: The level of any source signal in the final output is affected by many factors, principally the Gain control, Channel Fader and Output Fader. You should try to use only as much microphone gain as required to achieve a good balance between signals, with the faders set as described above. If the input gain is set too high, the channel fader will need to be pulled down too far in compensation to leave enough travel for successful mixing and there is a greater risk of feedback because small fader movements will have a very significant effect on output level. If the gain is set too low, you will not find enough gain on the faders to bring the signal up to an adequate level.

Microphone Placement

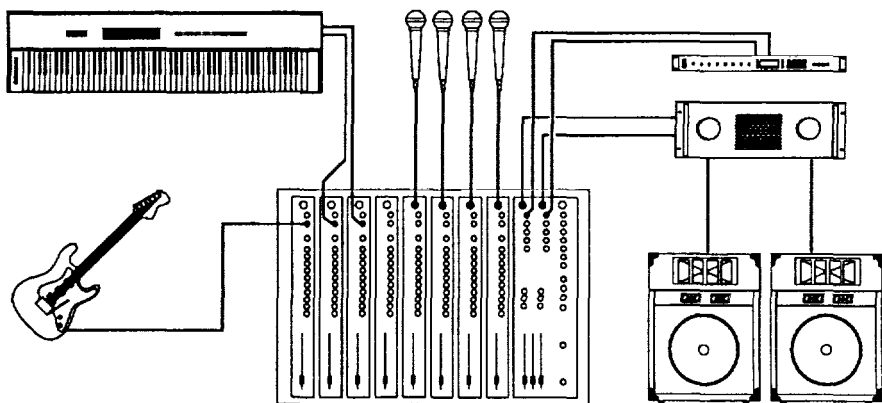
Careful microphone placement and the choice of a suitable type of microphone for the job is one of the essentials of successful sound reinforcement. The aim should be to place the microphone as close as physically possible to the source, to cut out unwanted surrounding sounds, allow a lower gain setting on the mixer and avoid feedback. Also a well-chosen and well-placed microphone should not need any appreciable equalisation.

APPLICATIONS

SPiRiT *Live* is designed primarily as a sound reinforcement mixer, but may also be used for basic multitrack recording. The following diagrams show typical configurations which will illustrate how the mixer is connected to other equipment.

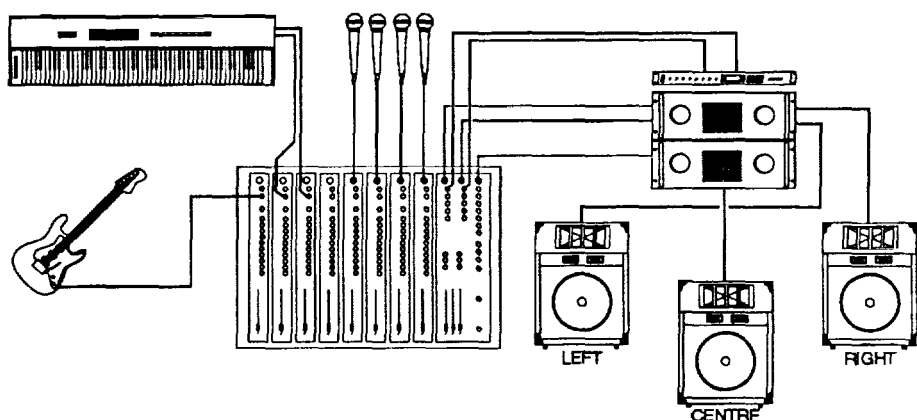
Example 1 - Public Address - Stereo Output

In this basic set-up, an assortment of sources are connected to the input channels, microphone to mic. inputs, and a keyboard and guitar to line inputs. Note that some guitars would not produce sufficient level for a direct connection, and would require a D.I. box connected via the microphone input. The main stereo outputs are connected to the power amplifier and speakers, and an effects processor is included in the output signal path via the Left and Right insert points.



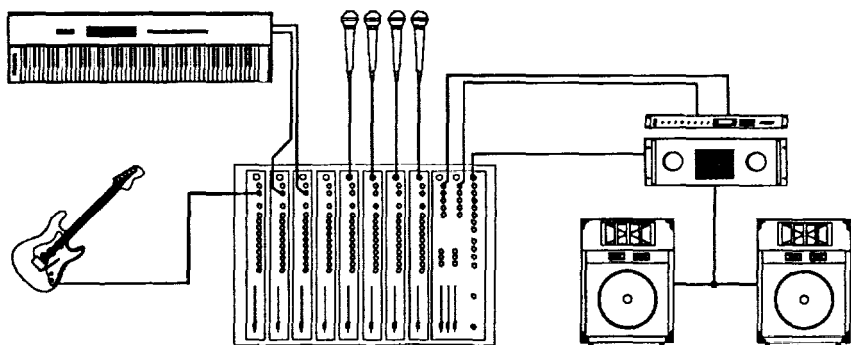
Example 2: Public Address - Stereo + Centre Feed

In this second example the Mono Output is used to drive an additional centre speaker, with individual fader control. Note that the input channels provide separate routing assignment for this Mono bus.



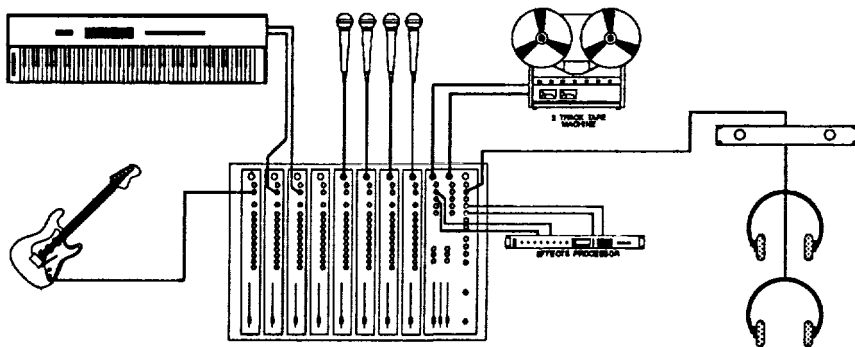
Example 3: Public Address - Mono Output

In this example the console is used to feed loudspeakers in Mono only. In this configuration the Stereo mix busses may be used as subgroups, mixing to the Mono Output by selecting L-R TO MONO. The effects processor remains in the Left and Right Inserts for use on these subgroups if required.



Example 4: Stereo Recording

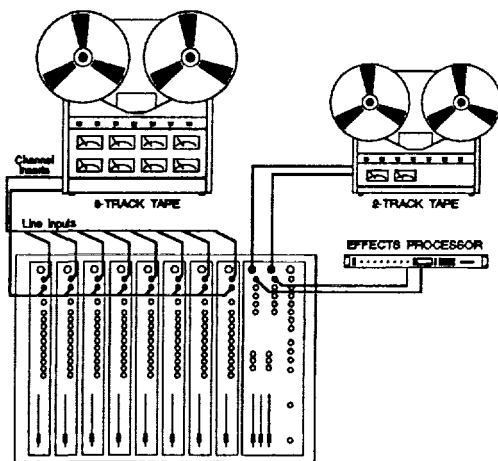
The console may also be used for 2-Track or basic multitrack recording live or in the studio. The set up is similar to the previous examples but with the Stereo Output feeding a 2-track Tape Machine. Aux 1 is used to provide artists foldback on headphones, and Aux 3 and 4 feed an effects processor which is returned to the console on FX1 and FX2.



Example 5: Multitrack Recording

In this last example the mixer is connected to a multitrack tape machines, and the main outputs to a 2-track machine for mix-down. Once again an effects processor has been included in the insert points.

Feeds to the multitrack are tapped off the insert sends by shorting tip and ring at the plug, thereby leaving the channel signal path intact. Outputs from tape are fed back to the channel line inputs and can be mixed down to the stereo output as required.



CARE OF YOUR MIXER

General Precautions

Avoid storing or using the mixer in conditions of excessive heat or cold, or in positions where it is likely to be subject to vibration, dust or moisture.

Keep the mixer clean using a soft dry brush, and an occasional wipe with a damp cloth or ethyl alcohol. Do not use any other solvents which may cause damage to paint or plastic parts.

Avoid placing drinks or smoking materials on or near the mixer. Sticky drinks and cigarette ash are frequent causes of damage to faders and switches.

Regular care and inspection will be rewarded by a long life and maximum reliability.

Glossary

auxiliary send	an output from the console comprising a mix of signals from channels and groups derived independently of the main stereo/group mixes. Typically the feeds to the mix are implemented on rotary level controls.
balance	the relative levels of the left and right channels of a stereo signal.
clipping	the onset of severe distortion in the signal path, usually caused by the peak signal voltage being limited by the circuit's power supply voltage.
dB (decibel)	a ratio of two voltages or signal levels, expressed by the equation $dB=20\text{Log}_{10}(V1/V2)$. Adding the suffix 'u' denotes the ratio is relative to 0.775V RMS.
DI(direct injection)	the practice of connecting an electric musical instrument directly to the input of the mixing console, rather than to an amplifier and loudspeaker which is covered by a microphone feeding the console.
equaliser	a device that allows the boosting or cutting of selected bands of frequencies in the signal path.

feedback	the 'howling' sound caused by bringing a microphone too close to a loudspeaker driven from its amplified signal.
foldback	a feed sent back to the artistes via loudspeakers or headphones to enable them to monitor the sounds they are producing.
frequency response	the variation in gain of a device with frequency.
(sub) group	an output into which a group of signals can be mixed.
headroom	the available signal range above the nominal level before clipping occurs.
highpass filter	a filter that rejects low frequencies.
line level signals	at a nominal level of -10 to +6dBu, usually coming from a low impedance source.
pan (pot)	abbreviation of 'panorama': controls levels sent to left and right outputs.
peaking	an equaliser response curve affecting only a band of frequencies i.e. based on a bandpass response.
PFL (pre-fade listen)	a function that allows the operator to monitor the pre-fade signal in a channel independently of the main mix.
rolloff	a fall in gain at the extremes of the frequency response.
shelving	an equaliser response affecting all frequencies above or below the break frequency i.e. a highpass or lowpass derived response.
spill	acoustic interference from other sources.
talkback	the operator speaking to the artistes or to tape via the auxiliary or group outputs.
transient	a momentary rise in the signal level.

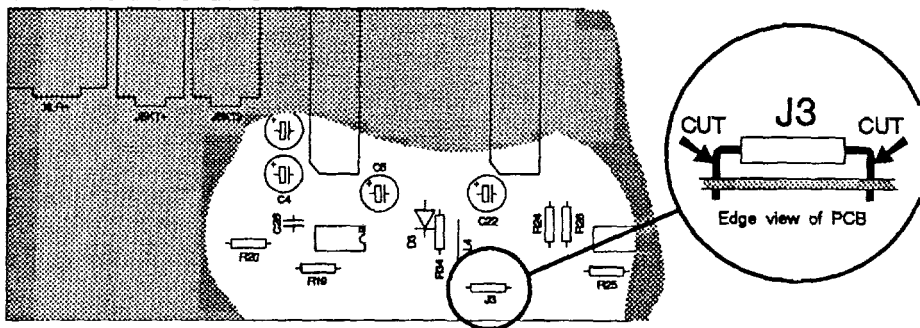
Selectable Options

REMOVAL OF +48V ON MIC INPUTS

To disable the +48V phantom powering on the microphone inputs, remove link J3 from Input PCB SC2952. This can be done without removing the PCB as shown below, by carefully cutting the leads of the link above at the points marked.

To reinstate the +48V, fit a replacement link, Part No. AZ2222, which is available from your dealer.

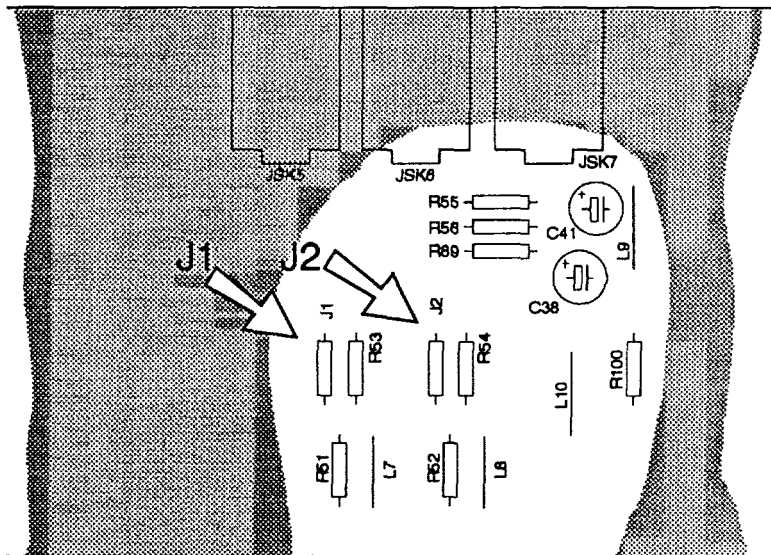
INPUT BOARD SC2952



FEEDING MONO OUTPUT TO TAPE RECORD SOCKETS

The Tape Record sockets are factory set as a stereo feed. Provision is made on the Master PCB SC2953 to add the Mono Output to either or both of the Tape Record outputs by the insertion of links J1 and J2 as shown below after removing the PCB from the console.

MASTER PCB SC2953



- To enable the Mono feed to Tape Left J1 must be fitted.
- To enable the Mono feed to Tape Right J2 must be fitted.

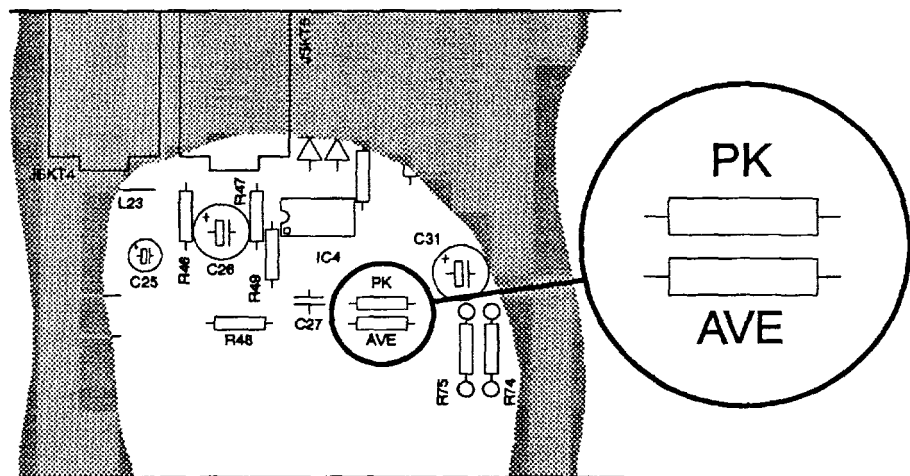
This operation requires care and skill in soldering to avoid damage to the PCB and should only be attempted by competent technicians.

Replacement links, Part No. AZ2222 are available from your dealer.

SELECTING AVERAGE RESPONSE ON BARGRAPHS

The Bargraph Meters are factory set for PEAK characteristic, but may be modified for AVERAGE response by repositioning a link on the Output PCB SC2954 as shown below, after removing the PCB from the console.

OUTPUT PCB SC2954



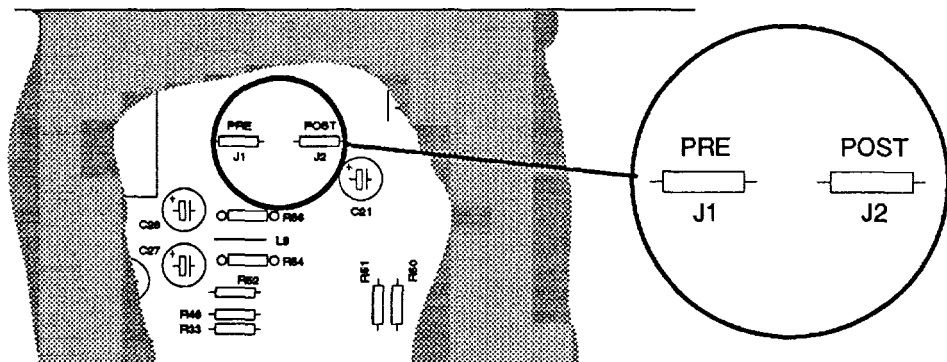
To select AVERAGE response, carefully unsolder the link in the PK position using the minimum of heat to avoid the possibility of tracks lifting on the PCB. Replace the link in the AVE position.

This operation should only be carried out by competent technicians who possess the necessary soldering skills.

RECONFIGURING AUX 2 AS A POST FADE SEND

AUX 2 is configured as a PRE FADE send at the factory, but may be altered to be POST FADE if required by repositioning a link on the Input PCB SC2952 as shown below, after removing the PCB from the console.

INPUT BOARD SC2952



To change AUX 2 to POST FADE, carefully unsolder link J1 using the minimum of heat to avoid the possibility of tracks lifting on the PCB. Replace the link in position J2.

This operation should only be carried out by competent technicians who possess the necessary soldering skills.

SPECIFICATIONS

Typical Figures for a 16 Channel Console

NOISE

Measures RMS, 22Hz to 22kHz Bandwidth

Line inputs selected at unity gain and terminated 150R

BUSS NOISE:

	<i>Masters down</i>	<i>Masters up, nothing routed</i>	
Mix Left	-99dBu	-97dBu	
Mix Right	-99dBu	-97dBu	
Mono	-99dBu	-98dBu	
Aux 1	-100dBu	-90dBu	
Aux 4	-100dBu	-91dBu	
<i>Masters up</i>	<i>1 Ch. routed</i>	<i>8 Ch. routed</i>	<i>16 Ch. routed</i>
Mix Left	-93dBu	-85dBu	-81dBu
Mix Right	-93dBu	-85dBu	-81dBu
Mono	-93dBu	-85dBu	-82dBu

E.I.N.

Microphone Input, Maximum Gain, terminated 150R	-129dBu
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C.M.R.R.

Measured at 1kHz

Microphone Input at maximum gain	-90dB
Line Input at unity gain	-55dB

DISTORTION

THD measured 1kHz at +20dBu, 20Hz to 20kHz Bandwidth

Line in to Mix Out	> 0.006%
Line in to Mono Out	> 0.006%
Line In to Aux Out	> 0.006%

CROSSTALK

Measures 1kHz Sine Wave

Routing Isolation > 100dB Mix L/R & Mono

Max. Fader Attenuation > 90dB typical

Max Aux Send Attenuation > 90dB typical

FREQUENCY RESPONSE

Measured 20Hz to 20kHz Bandwidth, relative to 1kHz

Mix Left/ Right Outputs +/- 0.5dB

Mono Outputs +/- 1dB

Aux Outputs +/- 0.5dB

INPUT & OUTPUT IMPEDANCES

Microphone Input > 2k Ω

Line Input > 10k Ω

Insert Sends 75 Ω

Insert Returns 10k Ω

Outputs 75 Ω

INPUT & OUTPUT LEVELS

Mic Input Maximum Level +10dBu

Line Input Maximum Level +30dBu

Mix Out Maximum Level +21dBu

Mono Out Maximum Level +21dBu

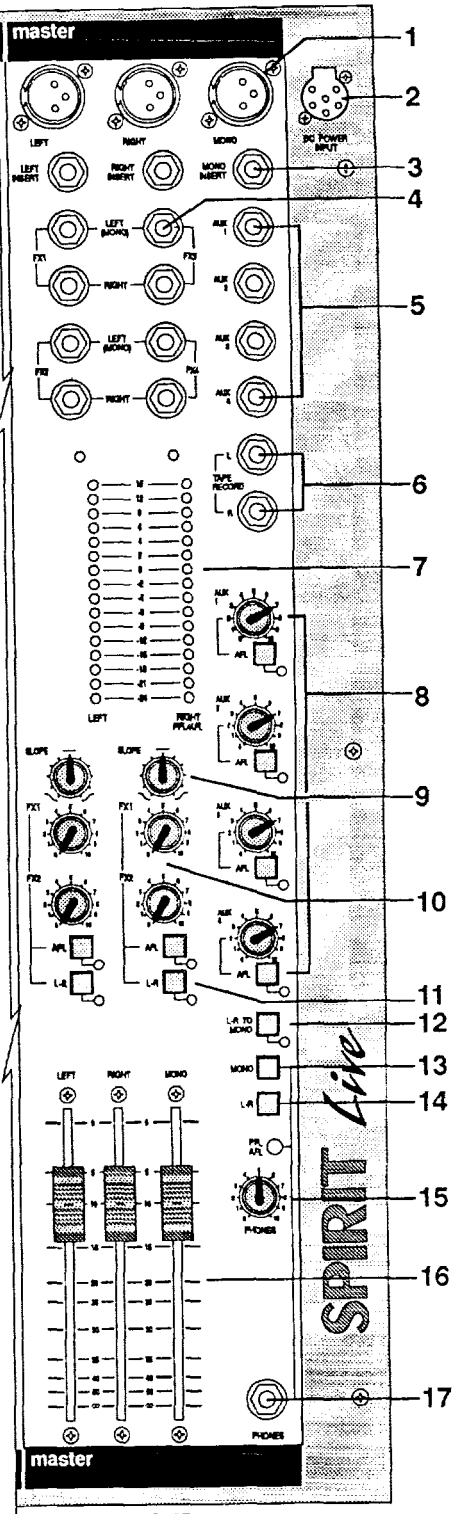
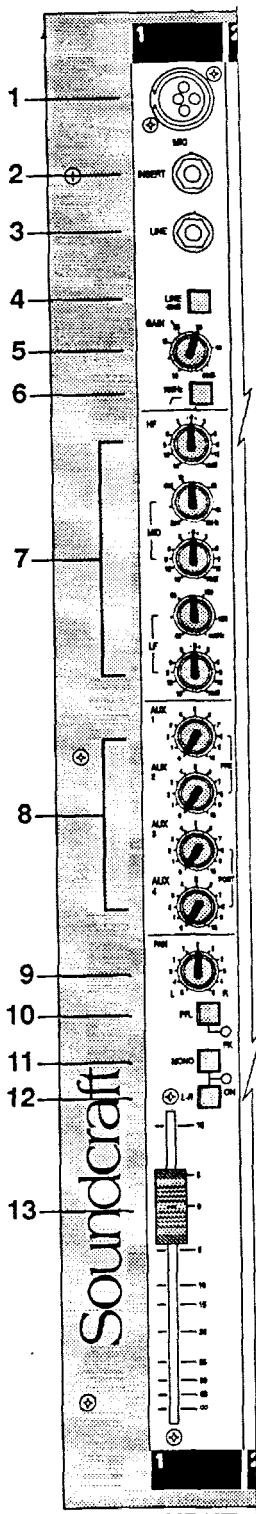
Aux Out Maximum Level +21dBu

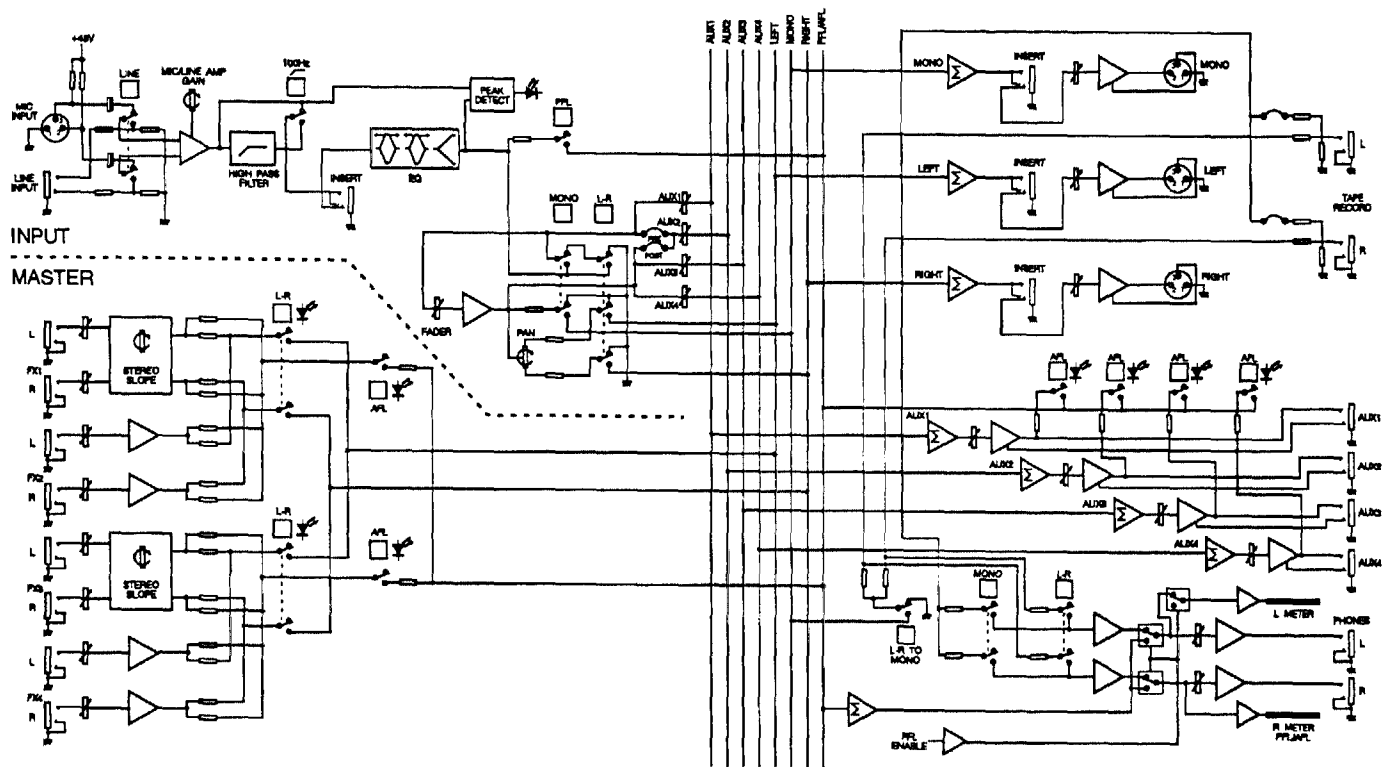
METERING

16 Segment LED Bargraph

Selectable PEAK or AVERAGE Reading

Accuracy Relative to 0dB +/- 1dB





SYSTEM BLOCK DIAGRAM