

SPIRIT *Monitor*

Soundcraft

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User Guide

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INTRODUCTION

Congratulations on your purchase of a **SPIRIT Monitor** 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 Monitor** has been built to the highest standards using quality 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 stage monitor mixer which is ideal for both touring and fixed installations.

SPIRIT Monitor is available in 16 and 24 channel frame sizes, and is designed to complement the **SPIRIT Live** range of front- of-house consoles.

SPIRIT Monitor 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, phase reverse switch, 3-band Equalisation with swept Mid and LF range, plus a Hi-Pass Filter, 8 Monitor Sends, PFL(Pre Fade Listen), Peak LED and a high-quality linear fader.

Each of the eight Monitor Output sections is provided with a variable high-pass filter, insert point, LED bargraph meter, Talkback and Dim switches, a variable send to the engineer's wedge and a high quality linear fader.

The master section comprises controls for PFL monitoring, engineer's wedge, LED bargraph meter for Wedge/PFL and master Talkback and Dim switches. The wedge output is provided with an insert point and high quality linear fader.

SPIRIT Monitor 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 the flexibility offered by your **SPIRIT Monitor** mixer.

BASIC PRINCIPLES OF MONITOR 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. While the quality of the front-of-house PA mix is of prime importance, the ability of the artists to deliver the best performance may be directly influenced by the quality of the stage monitor mix. Indeed, the monitor engineer may be required to provide a number of quite different individual monitor mixes, often under the most adverse conditions.

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. 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.

Your **SPIRIT Monitor** 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 **Monitor Outputs**. The **Master** section provides monitoring of the audio signal at many points in the mixer, either on headphones or meters and provides additional master control of talkback functions.

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.

Phase Reversal enables the operator to reverse the polarity of the signal entering the mixer, as a convenient way of checking for incorrect wiring or problems with microphone placement.

The **Monitor Sends** provide a way of routing the input signals to a number of outputs, to create individual mixes for artists foldback or additional speaker outputs.

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 PFL meter, to check the quality of the signal or to pin-point problems. Using PFL will not affect the signals on the monitor outputs.

Each input channel and all 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 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 Channel 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.

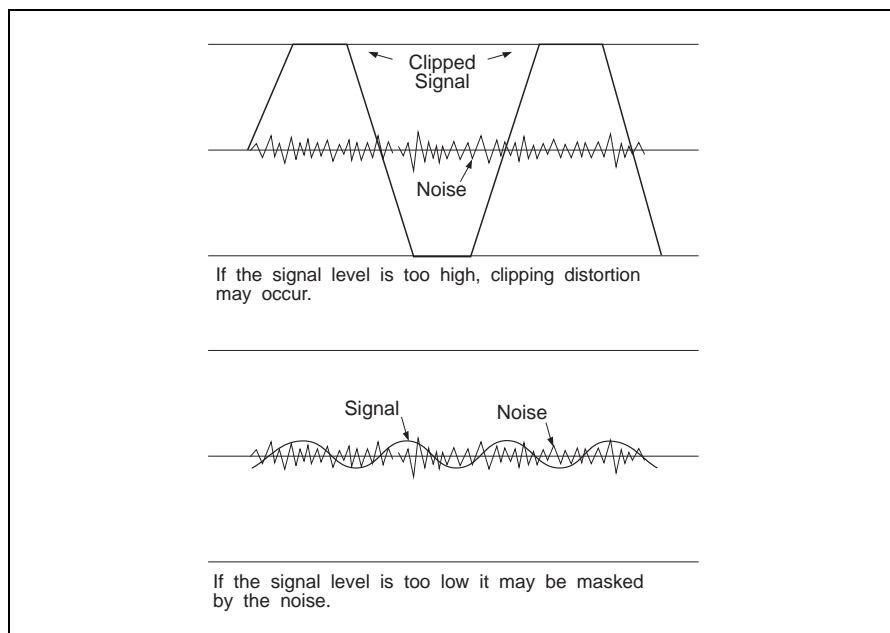
SPIRIT *Monitor*

The **Wedge** output refers to the engineer's own monitoring output, which will typically feed a floor-mounted 'wedge-shaped' speaker via a suitable power amplifier.

The **Talkback** system allows the monitor engineer to talk back to artists, either individually via selected monitor outputs, or together using a switch on the master section.

Creating monitor mixes on stage in close proximity to microphones demands constant awareness of the problem of acoustic feedback. The **Dim** buttons enable the engineer to instantly lower the output level on any outputs to avoid this problem while more precise level adjustments are made.

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.

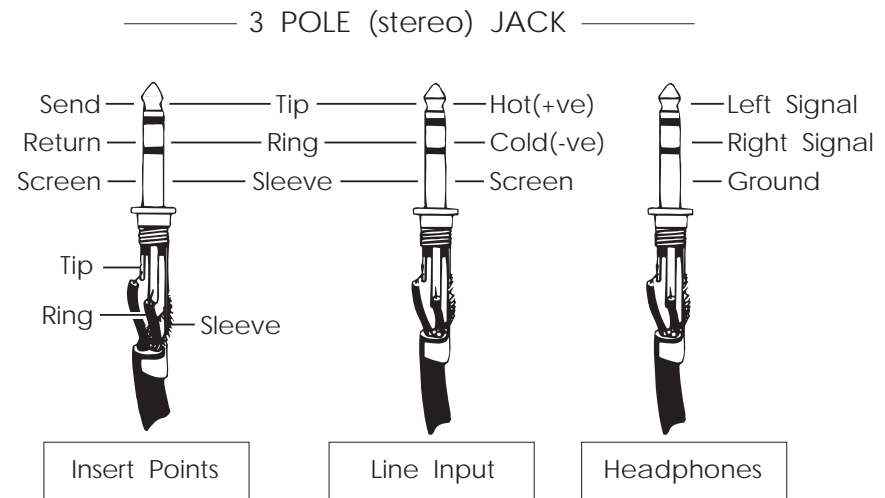
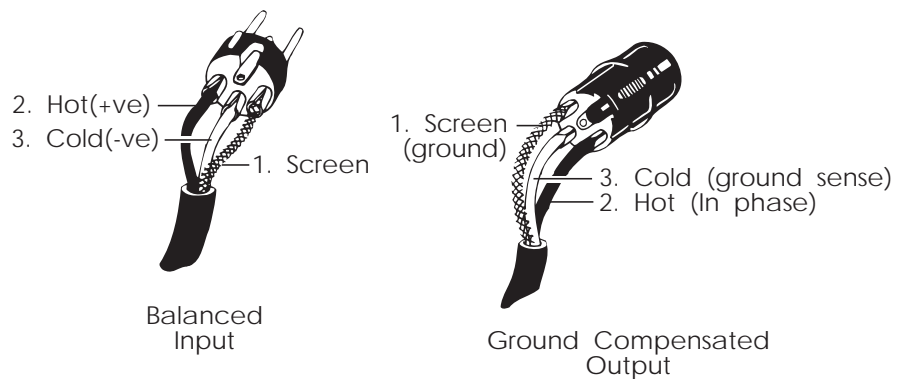


GETTING STARTED

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 Monitor** 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 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 the source is unbalanced source, the signal should connect to +ve and the -ve pin of the input should be 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.

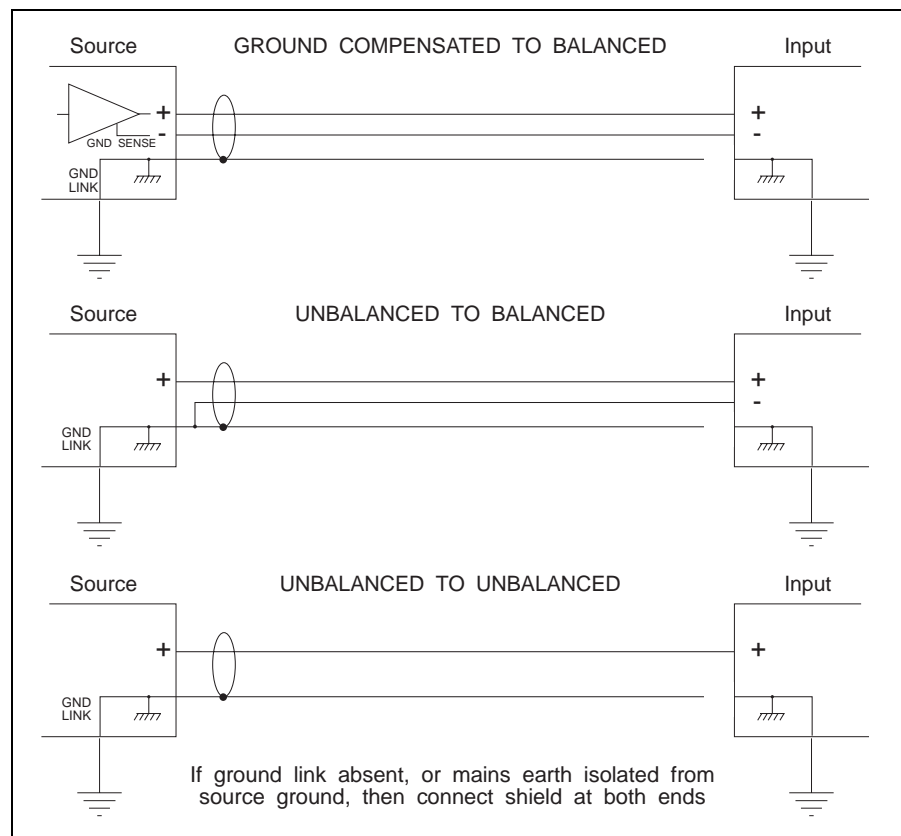
All of the outputs are **ground compensated** a technique which provides a very effective way of optimising noise immunity, without the cost and complexity of balanced outputs. These outputs employ ground compensation to cancel out the effects of variation in ground potential between the mixer and other 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.

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 back of this manual)
- Get to know what each component in the system is supposed to do.
- Learn where to look for common trouble spots.

The **Block Diagram** 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 other outputs to identify problems on particular outputs or the Master section.
- Compare a suspect channel with an adjacent channel which has been set up identically. Use PFL to monitor the signal in each section.

GETTING TO KNOW YOUR CONSOLE

FACILITIES

Refer to the fold-out front panel diagram at the rear of this manual, which shows the control functions on the **SPIRIT Monitor**. Each facility is described below, and is identified by a 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, with an overall ON/OFF switch at the right hand side of the console, beside the power input socket.

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.

NOTE: The microphone should always be plugged in before switching the +48V on or off. 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, 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(5) 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.

4. **PHASE REVERSE(Ø)**

The PHASE REVERSE switch reverses the polarity of the selected input, providing a convenient method of compensating for incorrect wiring or microphone placement. In some circumstances it may be found that pressing PHASE REVERSE will help with feedback avoidance. The switch should normally be released when not required.

5. **LINE SELECT**

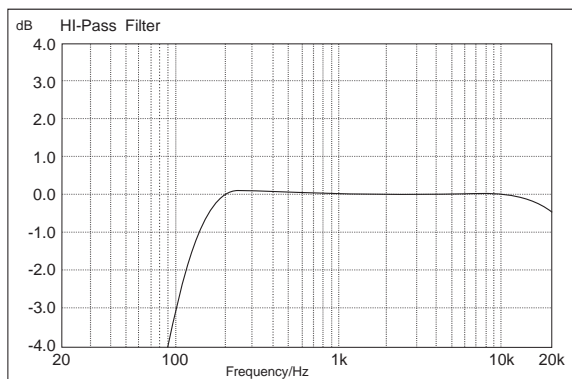
The LINE switch selects Line input when pressed, and Microphone input when released. When Line is selected the Gain range is reduced by 20dB(see 6 below).

6. **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.

7. **HI-PASS FILTER**

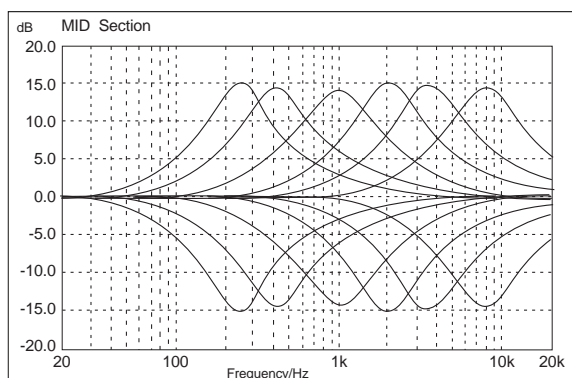
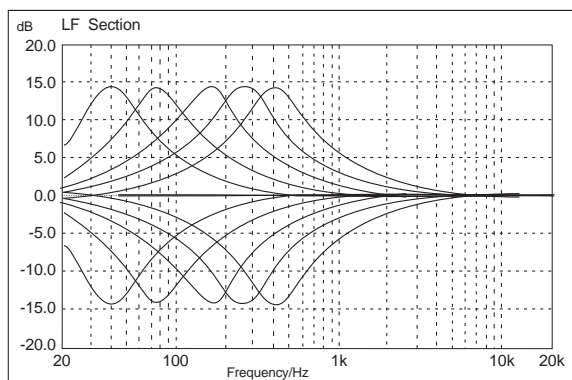
Pressing this switch inserts a 12dB per octave 100Hz Hi-Pass Filter in the signal path, immediately after the input amplifier. This is particularly useful on live vocals, and its use is strongly recommended, even on male vocals. It can also be used for filtering out noise such as stage rumble.

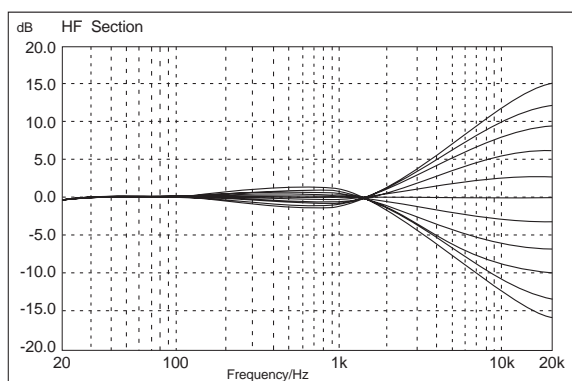


8.

EQUALISER

The Equaliser(EQ) comprises three sections. The upper control provides H.F.(treble) boost and cut of ± 15 dB. The lower two pairs of knobs are arranged as a cut/boost control (lower knob) of ± 15 dB, and a SWEEP(frequency) control which determines at which frequency the boost/cut action will be centred. 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.

9. MONITOR SENDS

These controls route the input channel signal to any one or more Monitor busses and the associated Monitor Outputs, allowing a number of completely different monitor mixes to be created. The sends are arranged as two groups of four (1-4 and 5-8) and both groups are normally fed after the channel fader (POST FADE) and therefore follow any changes in fader level. Either group may be reconfigured with the feed before the channel fader (PRE FADE) by changing internal links, and details of this modification will be found under Selectable Options on page 25. Any Monitor Sends configured as PRE FADE will be unaffected by the input fader position. All sends are muted when the ON switch (see 10 below) is released, regardless of the Pre/Post setting.

10. ON

The ON switch routes the channel signal to the monitor sends, and is positioned PRE FADER to mute all outputs from the respective channel. The associated LED is illuminated when the switch is pressed, showing that the channel is active.

11. PFL/PEAK LED

When the PFL switch is pressed, the Pre-Fade signal is fed to the headphones and wedge output, where it replaces the selected source. The PFL ACTIVE LED on the master section illuminates to warn that the wedge/phones outputs and the meter are now responding to the PFL 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 monitor mixes, 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 (PRE Hi-PASS FILTER), 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.

12. CHANNEL FADER

This linear fader determines the proportion of the channel in the mixes 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.

OUTPUT SECTIONS

There are eight Monitor Outputs, arranged in pairs. Each Output offers identical facilities.

1. MONITOR OUTPUTS

Each Monitor Output is driven by a ground-compensated amplifier and fed to standard male XLR-3 connectors.

2. MONITOR INSERTS

These allow external processing equipment to be '*inserted*' into the monitor signal path. The ¼" 3 pole 'A' gauge jacks are bypassed except when a plug is inserted.

3. SUB

The SUB control and associated switch and LED feed the monitor output to the engineer's submix, which is the normal source for the headphones and wedge outputs. The signal is derived before the Talkback and Dim circuitry, and the submix is therefore unaffected by either of those functions in the Output section. The submix feed is enabled when the ON

switch is pressed (LED illuminated).

4. HI-PASS FILTER

A variable frequency filter is provided covering a range of 20Hz to 200Hz., and this is always in circuit. This may be found useful to reduce the level of stage-driven low frequency feedback, or particularly to tailor the output frequency to match more closely the frequency range of smaller wedge monitor speakers which may not be able to tolerate high levels of LF signal. Rotate the control fully anticlockwise when the filter is not required.

5. TALKBACK

Pressing the TALKBACK switch routes the engineer's talkback microphone signal to the monitor output, at the level set on the T/B gain control on the Master section, and independent of the output fader level. At the same time the local monitor mix is dimmed by approximately 6dB to allow the engineer's voice to be heard over the mix, and the Wedge output also dims to prevent feedback through the talkback mic. Note that Talkback is disabled when DIM is pressed.

6. DIM

Pressing the large DIM switch attenuates the monitor output by 6dB as an immediate way of eliminating feedback while the source of the problem is identified.

7. PFL

When the PFL switch is pressed, the Pre-Fade signal is fed to the headphones, replacing the normal submix signal. The PFL ACTIVE LED on the master section illuminates to warn that the wedge/phones outputs and the meter are now responding to the PFL 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 signal so that adjustments can be made or problems traced.

8. BARGRAPH METER

A sixteen segment, three colour bargraph meter monitors the output signal. The meter has a peak response, and since it reads the final output signal it will also be affected by the DIM switch and include any talkback signal. The bargraph may be calibrated by a trimmer accessed via a hole in the panel above the meter. Adjustments may be made using a

small screwdriver, taking care not to damage the trimmer.

9. **FADER**

A linear fader controls the overall level of each output. Unity gain is at the top of the travel.

MASTER SECTION

1. **D.C. POWER SOCKET**

The cable from the power supply connects to the console via this 5-pin SRC connector. To the right of the power connector is the master switch for +48V Phantom Power to all inputs.

2. **WEDGE OUTPUT**

The WEDGE output normally drives the engineer's monitor speaker via a power amplifier, and when no PFL switches are pressed this is fed by the submix signal. The WEDGE output automatically switches to the PFL signal when any PFL switch is active.

The WEDGE output always listens to the same source signal as the headphones, but is also affected by DIM status and any external connections to the Insert socket.

3. **WEDGE INSERT**

This is 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 jacks are by-passed except when a plug is inserted.

4. **PHONES**

A standard 1/4" 3 pole 'A' gauge jack for monitoring headphones. See the diagram on Page 8 for connection details.

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

5. **TALKBACK MICROPHONE SOCKET**

A standard female XLR-3 connector is provided for a low impedance dynamic talkback microphone on a gooseneck mount

or plugged in via a cable.

6. BARGRAPH METER

A sixteen segment, three colour bargraph meter provides visual monitoring of WEDGE/PFL level. Normally the post-fade Wedge signal is displayed, unless any PFL switch is active, in which case the PFL signal is shown. The level of the displayed Wedge signal follows the actual Wedge output, including the effect of the Dim circuit if active. The bargraph may be calibrated by a trimmer accessed via a hole in the panel above the meter. Adjustments may be made using a small screwdriver, taking care not to damage the trimmer.

7. TALKBACK GAIN

Sets the level of Talkback microphone signal which may be routed to any Monitor Output.

8. ALL TALKBACK

Pressing the ALL T/B switch dims all monitor outputs and adds the Talkback mic signal, also dimming the wedge output to prevent feedback. Note that ALL T/B is disabled when ALL DIM is pressed.

9. ALL DIM

Pressing ALL DIM attenuates all outputs, except the wedge output. This provides an instant remedy for serious feedback, while the source of the problem is identified.

10. PFL ACTIVE LED

The LED illuminates whenever a PFL switch is pressed to show that the Wedge and Phones outputs and the Wedge(PFL) bargraph are now receiving the PFL signal.

11. PFL TRIM

The rotary PFL TRIM control provides level adjustment of the PFL signal to allow for differences in operating levels. The TRIM control has no effect on the level of PFL signal fed to the Wedge(PFL) meter.

12. SUB TRIM

The rotary SUB TRIM control provides level adjustment of the engineer's Submix. The TRIM is pre-insert and pre-fader, and will therefore affect both the level at the wedge output and

the bargraph reading.

13.

HEADPHONE VOLUME

The master volume control for the wedge or PFL signal being fed to the headphone socket (4).

14.

WEDGE FADER

The master fader for the Submix or PFL signals feeding the

USING YOUR SPIRIT *Monitor* CONSOLE

Your choice of a **SPIRIT *Monitor*** console has provided you with a professional product capable of top quality sound mixing. Good results will however only come through experience and time spent understanding the facilities on your console. Rehearsal sessions and live performances must focus on the skill and creativity of the artists, which must not be hindered by unfamiliarity and difficulty with the operation of the console. It is important to recognise, and learn by experiment, the importance of correct choice of inputs, microphone placement and control settings, particularly with respect to feedback avoidance.

Suitable initial control positions are shown on the front panel drawing inside the rear cover.

Initial Set Up

The diagram on page 7 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 Output faders at 0, input faders at 0, channel monitor sends fully anticlockwise (off) and set power amplifier levels to about 70%.
- With a typical performance level signal present, press the PFL button on the particular channel, monitoring the level on the Wedge(PFL) meter.

- Adjust the input gain until the meter is just reaching the amber section (0dB) at a typical maximum source level. This allows sufficient headroom to accommodate peaks and establishes the maximum level for normal operation (but see note below).
- Repeat this procedure on other channels as required.
- 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, and +48V powering is not required, try using the Line Input instead to give an extra 20dB range.
- 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.
- Build up the required individual monitor mixes by feeding the selected channels to the monitor outputs using the channel monitor send controls. The corresponding output faders should be initially set at the '0' mark. Listen carefully for each component in the mix and watch the respective output meter for any hint of overload. As more channels are added to the mix, the meters may move into the red section. Adjust the overall level using the Output Faders if necessary, or back off the appropriate channel faders until the level is out of the red segments.

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 suc-

SPIRIT *Monitor*

successful sound mixing.

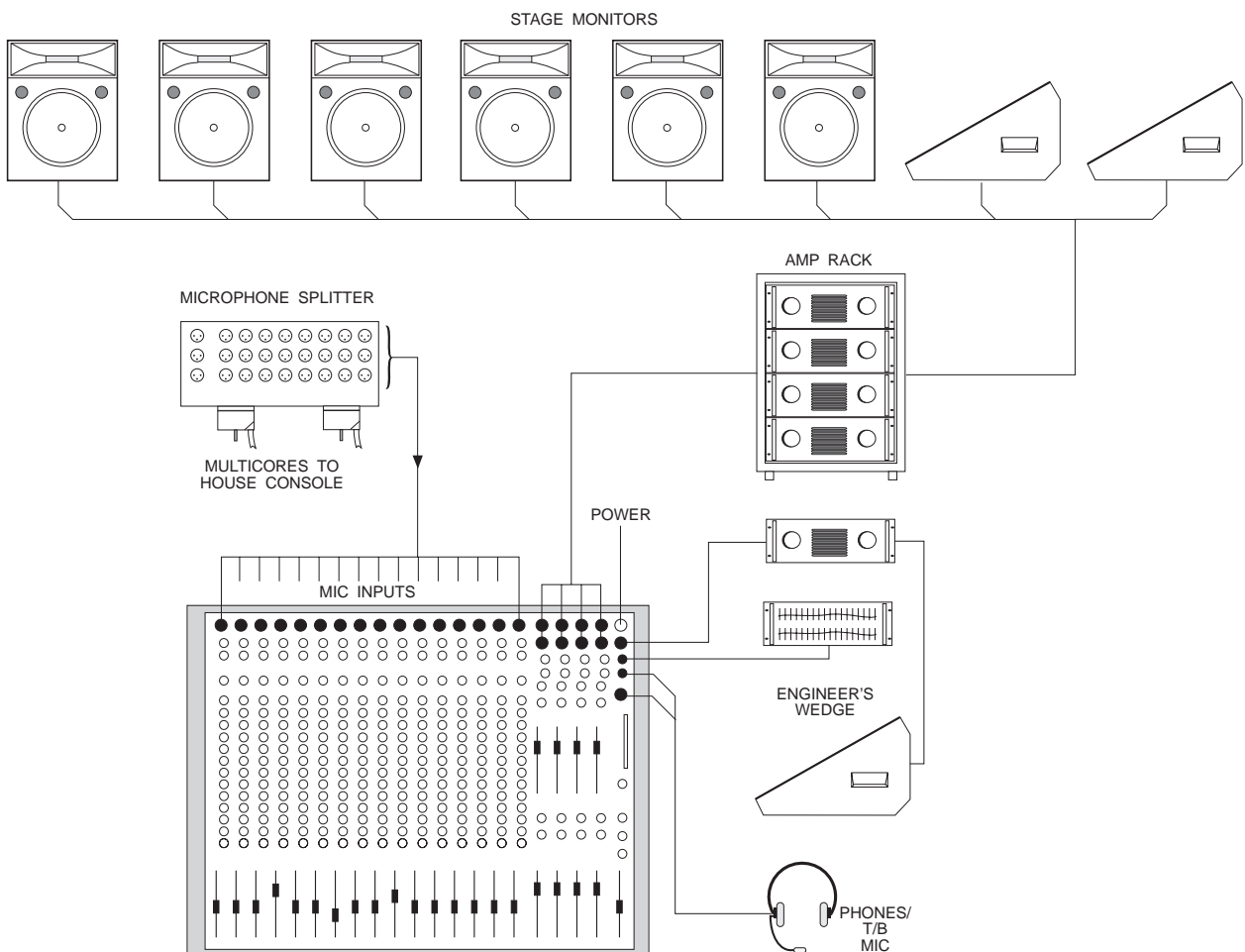
The aim should be to place the microphone as close as physically possible to the source, to cut out unwanted surrounding sounds, allow a lower gain setting on the mixer and avoid feedback. Also a well-chosen and well-placed microphone should not need any appreciable equalisation.

Input Connections

The monitor mixer will often share inputs with a front-of-house mixer and it is particularly important to ensure that the input connections provide complete isolation between the two consoles to avoid hum problems and possible degradation of the signal. The provision of balanced splitter transformers or isolating transformers will ensure a clean signal to both con-

APPLICATIONS

The diagram below shows a typical application of the **SPIRIT Monitor** as a stage monitor console, with inputs shared via a splitter box with the front-of-house console



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

Glossary

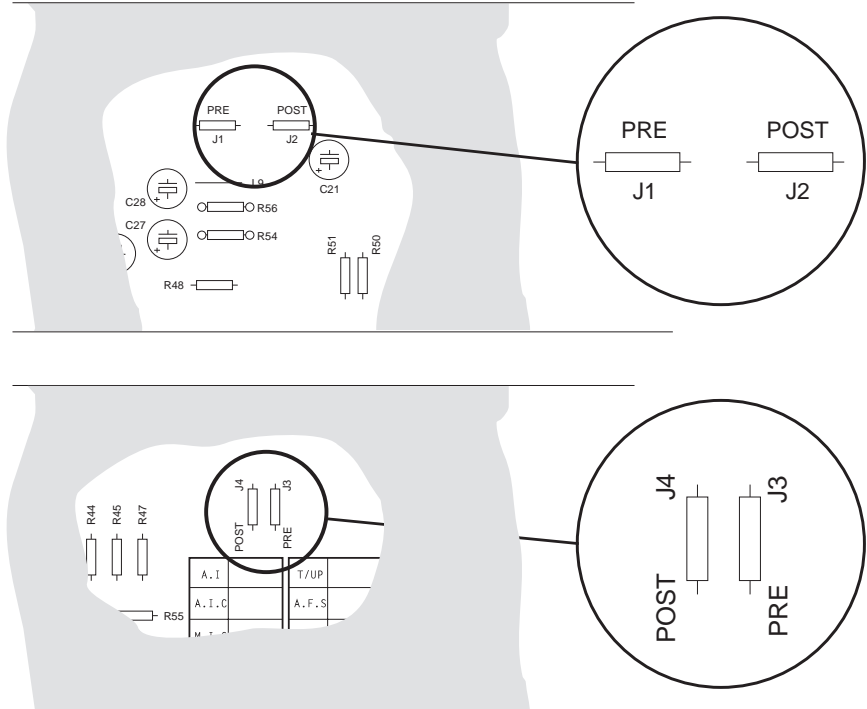
balance	the relative levels of the left and right channels of a stereo signal.
clipping	the onset of severe distortion in the signal path, usually caused by the peak signal voltage being limited by the circuit's power supply voltage.
dB (decibel)	a ratio of two voltages or signal levels, expressed by the equation $dB=20\log_{10}(V1/V2)$. 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.
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

The eight channel Monitor Sends are factory set as POST FADE, but may be altered to PRE FADE in blocks of four (1-4 and 5-8) by repositioning a link on the Input PCB SC3005 as shown below, after removing the PCB from the console.

INPUT BOARD SC3005



To change Sends 1-4 to PRE FADE carefully unsolder link J4 using the minimum of heat to avoid the possibility of tracks lifting on the PCB. Replace the link in position J3.

To change Sends 5-8 to PRE FADE carefully unsolder link J2 using the minimum of heat to avoid the possibility of tracks lifting on the PCB. Replace the link in position J1.

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

SPECIFICATIONS

Typical Figures for a 24 Channel Console
Measured RMS, 22Hz to 22kHz Bandwidth

E.I.N.

Microphone Input, max gain, 150R	-128.5dBu
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OUTPUT NOISE

24 channel console, channels sends turned down	-80dBu
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C.M.R.R.

Measured at 1kHz	
Microphone Input at maximum gain	80dB
Line Input at unity gain	50dB

DISTORTION

THD measured 1kHz at +20dBu, 20Hz to 20kHz Bandwidth	
Line in to Output	< 0.006%

CROSSTALK

Measured 1kHz Sine Wave	
Channel Send attenuation	90dB
Channel Fader Attenuation	85dB
Channel 'ON' switch isolation	100dB
Between any outputs	80dB

FREQUENCY RESPONSE

Any Input to Output (Output hi-pass filter, which is always in circuit, turned fully anticlockwise)	-3dB @20Hz, -1dB @ 20kHz
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INPUT & OUTPUT IMPEDANCES

Microphone Input	> 2k Ω
Line Input	> 10k Ω
Insert Sends	75 Ω
Insert Returns	10k Ω

SPIRIT *Monitor*

Outputs

75Ω

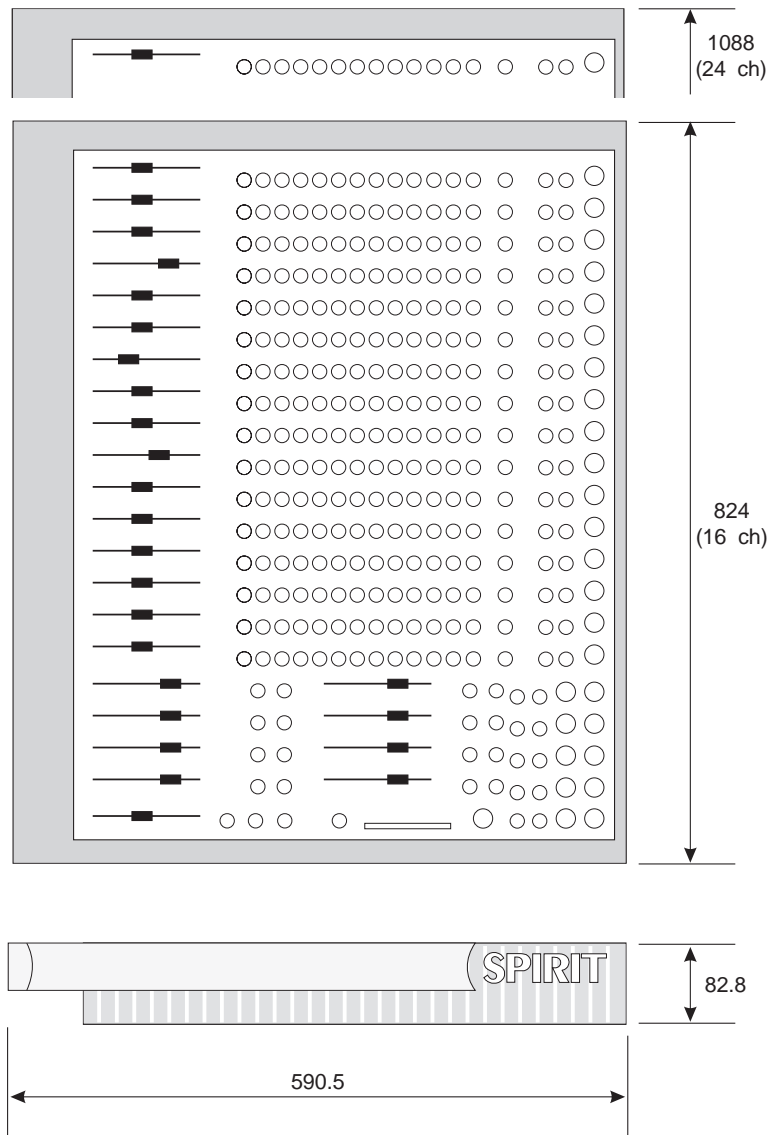
INPUT & OUTPUT LEVELS

Mic Input Maximum Level	+10dBu
Line Input Maximum Level	+30dBu
Any Output Maximum Level	+21dBu

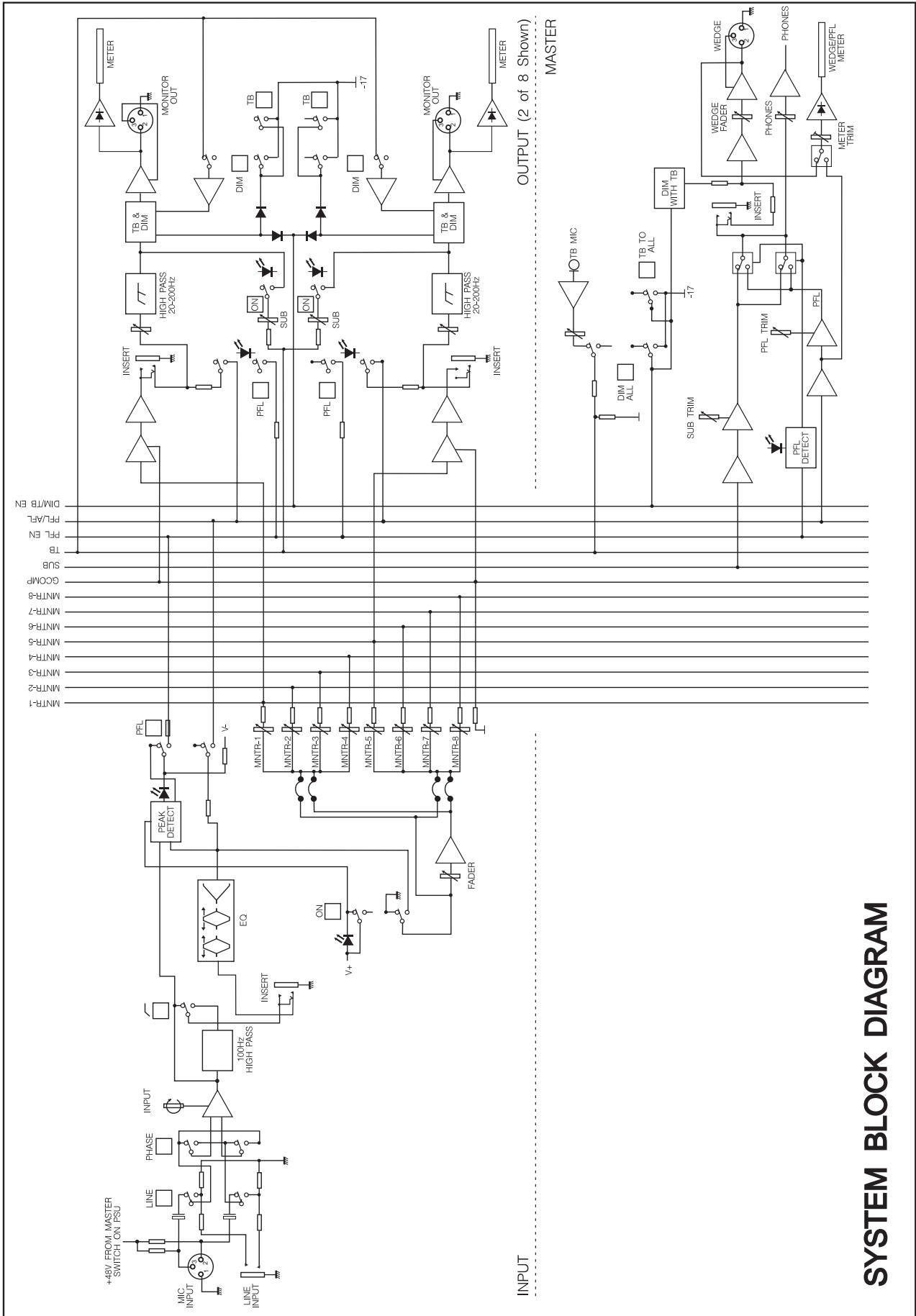
METERING

16 Segment LED Bargraphs	
Accuracy Relative to 0dB	+/- 1dB

Dimensions



SPIRIT Monitor



SYSTEM BLOCK DIAGRAM

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