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Soundcraft Series 6000

1.1 12/1/89

# THE SERIES 6000 MIXING CONSOLE.

The Series 6000 is the first of a new generation of consoles designed to meet new standards of performance that will outstrip any digital equipment available now or in the foreseeable future. We believe that the specifications attained are unique for this class of console.

Input facilities include individual phantom power, phase reverse, low-cut filter, 4-band sweep EQ, and 6 independent aux sends. Also provided are programmable muting, and solo-in-place as well as PFL.

The output modules include monitor sections with 2-band EQ, two aux sends, and PFL and ON switches. The group summing amplifiers use an advanced hybrid discrete/integrated configuration that reduces mixing noise significantly. The vital mix summing amps use similar but even more sophisticated low- noise technology.

The Series 6000 attains its performance by utilising innovative concepts and circuit configurations (some of which we have patented) rather than by using expensive components, making it highly affordable despite its sophistication. Almost all of the electronic parts are industry standards and widely obtainable; it is their application and arrangement which is special.

A newly developed input preamplifier covers the complete mic gain range of 2 to 70 dB with a single control, eliminating the compromises inherent in using input attenuators. The common-mode rejection sets new standards, remaining extremely good down to low frequencies where it is most necessary. The Line gain range is from  $\cdot 10$  to + 20 dB and accepts all the usual operating levels without internal adjustment.

The Soundcraft Active Panpot is a unique system that renders panpot operation virtually perfect. The panning law is generated by law synthesis techniques that give a much closer approach to the theoretical sine/cosine characteristic than the conventional 'resistor-bending' method, giving smooth panning without level shift. The left-right isolation when panned hard over is also improved by 25 dB or more, an important consideration for any console that routes in pairs.

The Series 6000 is available in both 16 and 24 bus formats; these differ only in the number of output modules fitted and the number of routing switches on the input modules, and this manual applies to both versions.

The Series 6000 is primarily intended for recording applications, but also performs superbly in live/PA work. Its high CMRR, proprietary grounding system, and ground cancel mixing techniques make it highly resistant to hostile electrical environments. The frame is built to the normal Soundcraft standards of extreme robustness for live consoles and can easily cope with the demands of life on the road. In this type of use the output module monitor sections are conveniently configured for use as effects returns either to subgroups or the main stereo outputs.

# PERFORMANCE SPECIFICATIONS.

FREQUEN Any Input to	CY RESPONSE. any output.	+ <b>0.0,</b> ·	0.5 dB 20H2	20kHz
THD.	Groups:		Less than .004%	(1kHz)
			Less than .01%	(10kHz)
	Any other outpu	ıt	Less than .008%	(1kHz)
			Less than .02%	(10kHz)
All measure	id at 20dB above	nominai level.		
<b>NOISE.</b> (150 Ohm s	Mic equivalent source res, 20kH	•	Less than -128dBu. velghted)	
	Group noise	(16 ch routed):	-82 dBu	(S/N -86 dB)
	Mix noise	(16 ch routed):	·87 dBu	(S/N ⋅91 dB)
	Aux nolse (32	ch console):	-84 dBu	(S/N ·88 dB)
CROSST	LK Mic to Line:		Better than -90 dB	(1 kHz)
	Line to mic:		Better than -110 dB	(1 kHz)
	Adjacent chan	net:	Better than -110 dB	(1 kHz)
	•	n/line out)	Better than -95 dB	(10 kHz)
	Channel mutin	g:	Better than -110 dB	(1 kHz)
	Fader kill;		Better than -85 dB	(1 kHz)
			Better than .78 dB	(10 kHz)
	Panpot Isolatio	on:	Better than -90 dB	(1 kHz)
	Aux send kill:		Better than -90 dB	(1 kHz)
	Routing:		Better than -110 dB	(1 kHz)
	-	to group)	Better than 100 dB	(10 kHz)
INPLIT IM	PEDANCES.	Mic inputs:	2 KOhm balanced.	
		Line inputs:	20 KOhm balanced	
		insert returns:	10 KOhm unbal.	
OUTPUT	IMPEDANCES.		Less than 75 Ohms.	
ουτρυτ	CAPABILITY. Balanced outp	outs:	Not less than + 26 o	dBu into 600 Ohms.

# **MODULE DESCRIPTIONS**

## **INPUT MODULE**

A) + 48V Phantom Power Switch. Pressing this switch applies + 48V phantom power to a suitable capacitor microphone plugged into the XLR mic input socket. This voltage is supplied through two 6K8 resistors in the standard way. Accidental application of phantom power to low-impedance microphones is most unlikely to cause any damage, but care should be taken not to use it when internal-battery electret microphones or electronic DI boxes are connected.

B) Mic/Line Switch. The high-impedance, high level line input is selected instead of the mic input when this switch is depressed. The isolation between the two inputs is exceptionally good on this console; for example, when the MiC position is selected, crosstalk from the line input is undetectable by the most sophisticated instruments, even when the mic input is unterminated.

The first 16 or 24 input modules are fitted with 'tape- normalling' on the line input. When no jack is plugged into the line input it is fed internally from the same-numbered tape- replay input. This allows all the facilities of the input module to be used at mixdown with a minimum of repatching.

The mic mode input impedance has been carefully chosen so that it allows hard-wired mic-splitting (sometimes called Y-splitting) without excessively loading a typical microphone.

C) input gain control. This knob controls the input gain in both mic and line modes. The Series 6000 input amplifier is unique in that it provides the correct gain for inputs ranging from 2dBu to 70dBu in one turn of the control, without the need for additional pad or attenuator switches. This saves valuable panel space as well as eliminating the awkward juggling of two interdependent controls.

This advanced technology also provides lower noise at intermediate gain settings, (at high gain settings it is determined almost entirely by the microphone source impedance) and an exceptionally high common-mode rejection-ratio that allows the input to ignore electrical interference. Unlike conventional systems, the CMRR remains very high down to low frequencies such as 100Hz, where it is most needed.

The gain range in line mode provides for nominal input levels between +10 and -20 dBu, catering for almost every conceivable type of equipment.

SOUNDCRAFT MICROPHONE PREAMP TECHNOLOGY IS COVERED BY PROVISIONAL PATENT No. 8719014.

D) Low-cut filter switch. Pressing this switch brings in a filter that cleanly and effectively removes low-frequency rumble and mic-proximity disturbances. The filter has a Butterworth maximally-flat characteristic with a roll-off commencing at 100Hz, and an ultimate slope of 12dB per octave.

E) Equalisation section. Usually shortened to 'EQ' this section incorporates HF and LF controls plus two sweep middles, giving precise control over four sections of the audio spectrum, either for the correction of deficiencies in the material, or to create individual effects. All boost/cut controls are centredetented for rapid re-zeroing.

HF. The high frequency control provides 15dB of boost or cut at 10 kHz with a sheMing characteristic; in other words having reached maximum boost or cut the EQ curve stays at that level.

HI-MID. A sweepable mid equaliser variable between 600Hz and

10kHz. The response is of the bell-curve or peaking type, with up to 15db of boost or cut available. The Q at maximum boost is 1.5.

LO-MID. As for hi-mid, except that the frequency range is 150Hz to 2.4kHz. The two mid controls overlap in the centre of the audio band to allow subtle and complex effects to be created.

LF. The low frequency control gives up to 15dB of boost/cut at 60Hz, with a shelving characteristic.

EQ switch. This switches the the entire EQ section in and out, but does not affect the low-cut filter. This allows instant comparison with the unmodified signal, when the EQ is switched out, it is not only entirely removed from the signal path, but the drive to it is also removed.

F) Auxiliary Sends. These have two main functions; providing foldback so that musicians can monitor their own performance, and giving a feed to external effects units. In both cases the fed may be taken from one module, or it may be a mix of any number of sources. Foldback sends are normally prefade (not affected by fader setting) while effects sends, and in particular echo/reverb sends, are almost always postfade so that the effect level fades down with the main signal.

AUX 1 & 2. These are normally set postfade, but the associated PRE switch changes this to prefade. An internal link allows the PRE source to be changed to be pre-EQ as well as pre-fader.

To change Aux 1 & 2 to pre-EQ, move the push on jumper near the Aux 2 pot from the J3 to the J2 position.

Aux 3 & 4. These are normally postfade, but may be set prefade by use of a PRE switch.

Aux 5 & 6. These are set permanently postfade, as an echo/reverb send is almost always required, no matter what the current operation of the desk.

**G)** Panpot. This is a centre detented control that places the module signal left or right in the stereo field. The Soundcraft Active panpot uses unique circuitry (patent pending) that allows the panning law to be much closer to the theoretically correct sine/cosine characteristic than conventional consoles, and also provides much greater left/right isolation; it yields an improvement over the usual techniques of at least 25 dB. The centre drop is set to 4.5dB as a compromise between stereo and mono operation.

SOUNDCRAFT ACTIVE PANPOT TECHNOLOGY IS COVERED BY PROVISIONAL PATENT No. 8800168.

H) ON Switch. This gates on and off signals passing through the module, using a silent FET switch positioned just before the fader in the signal path. The signal is in fact rapidly faded up and down (over about 10msec) and this prevents the generation of clicks when the channel is turned on and off with signal passing through it. ON status is shown by a green LED illuminating. Channel ON/OFF status is controlled not only by the action of the

ON switch, but also by the programmable mute bus and the Solo-In- Place facility. In all cases the true channel status is shown by the LED.

i) Mute Bus Switch. When this pressed the channel is connected to the muting bus. When the master mute switch is operated all channels thus connected are muted, regardless of the ON/OFF switch status. Releasing the master mute switch leaves the channel in its original state.

J) SOLO Switch. This activates either PFL or Solo-in-Place (SIP) operation, depending on the position of a mode switch on the master module. A red LED provides indication.

PFL mode routes the prefade module signal directly to the control-room outputs, summed with the prefade signals of any other modules with their SOLO buttons pressed. The only change in signal flow is the switching of the control-room outputs, and so this mode may be used during recording or live operation for quality checking without affecting the main console outputs.

SIP mode operates by muting all channels that do not have their SOLO buttons pressed. The channel contributions therefore appear in the correct stereo positions and at the correct relative level, with echo etc, providing the effects return facilities have been arranged not to respond to SIP muting.

When input modules are being used as effect returns an internal link (marked SIP SAFE, & J1, situated near the panpot) allows this to be done. To safe the module the link should be removed.

SIP is sometimes called a 'destructive solo' in the sense that normal signal flow to the main console outputs is disrupted when it is used.

**K)** Peak LED. This illuminates when the channel signal exceeds + 14 dBu, and is approx 8dB below clipping. The signal is monitored at the insert send, is after the EQ but before the FET muting and fader. There is a fast-attack/slow-decay action to give clear indication of short peaks.

L) MIX switch. This routes the channel signal directly to the stereo mix bus, to allow simple mixdown of many inputs to stereo. M) Routing switches. These route the channel signal to the desired groups, either for assignment to the desired tape tracks, or to create subgroups. The Series 6000 routing matrix uses unique technology that gives unprecedentedly low inter-group crosstalk.

Routing is in pairs; the switch selects the pair and the signal can then be panned between them. Odd group numbers are treated as Left, and even as Right.

N) Channel fader. This a long-throw unit with carefully optimised control law and feel. The total travel is 100 millimetres.

**O)** Channel insert. An unbalanced insert point operating at a nominal level of -2 dBu is positioned just before the electronic mute. The PFL switch accesses the insert return signal before muting.

NB. When input metering is fitted to the console, the metering takeoff point is at the insert return, the same as for the PFL system.

## STEREO INPUT MODULE

a) input Gain Control. This knob controls the input gain of the module for both left and right channels. The input sensitivity covers the range from +10dBu to -20dBu, covering all standard line level signals including -10dBV ("TASCAM" level) and +4dBu. The input stage is electronically balanced.

**b) Phase Reverse.** This button reverses the phase of the left hand channel only, to allow correction of any phasing errors in stereo signals.

c) High Pass Filter. This switch brings in a low cut filter affecting both left and right channels, to remove any low frequency noise or rumble.

The filter has a Butterworth characteristic with roll off commencing at 100Hz and an ultimate slope of 12dB per octave.

d) Equalisation Section. The EQ section provides HF and LF controls plus a sweep mid section. All cut/boost controls are centre detented to allow easy return to their "flat" positions.

HF The high frequency control provides 15dB of boost or cut at 10kHz, with a "shelving" characteristic.

MID The mid frequency section provides 15dB of cut or boost over a limited band of frequencies, which may be swept between 300Hz and 5kHz.

LF The low frequency section provides 15dB of cut or boost at 60Hz, also with a shelving characteristic.

EQ Switch: this switches the entire EQ section into the signal path when pressed. The High Pass filter is not affected by the position of the EQ switch.

e) Width Control. By varying the separation between left and right channels, the width control allows continuously variable control of the width of the stereo image of the signal passing through the module. This may vary between mono (both left and right channels of the module are fed with a sum of the left and right inputs) to full stereo (the left channel of the module is fed only with the left input, the right channel only with the right input.)

This continuous control of width allows stereo signals with a very wide stereo image (for instance stereo electronic instruments) to be integrated with the existing stereo mix without reducing them entirely to mono.

f) Auxiliary Sends. The auxiliary sends are the mono sum of the stereo signal in the module. Both pre- and post-fade signals are cut when the input is muted.

AUX 1 and 2 These are normally taken post-fader, but may be switched pre-fader pressing the associated PRE button.

AUX 3 and 4 These are normally taken post-fader by may be switched pre-fade by pressing the associated PRE button.

AUX 5 and 6 These are always post-fader.

g) Pan Control. This centre detented pot controls the relative levels of left and right channels sent to the mix or groups selected using the routing switches.

When turned fully anti-clockwise, signal is sent only to odd groups and mix left (if selected with the routing switches.) When turned fully clockwise, signal is sent only to even groups and mix right (if selected.) When used in conjunction with the width control, the PAN allows the stereo signal to be "trimmed" to any desired width, and steered to any position in the mix.

**h) ON Switch.** This button turns the module signal on and off, using an electronic FET muting circuit to kill all signals from the channel except PFL (if selected) and insert send.

Operation is indicated by the associated green LED which is illuminated when the module is on. The module may also be muted by SOLO-IN-PLACE and the MUTE BUS.

i) Mute Bus. When pressed, this button connects the module to the console master mute system. When the master mute button on the master module is pressed, any mono or stereo inputs on which the MUTE BUS is selected will be muted, over-riding the ON switch. This allows many modules to be conveniently muted from a single control.

When master mute is released, the module will revert to its previous state.

**J)** SOLO Switch. Pressing the solo button will cause either a PFL (pre fade listen) or SIP (Solo In place) operation, depending on the mode selected on the master module. Operation of the switch is indicated by a red warning LED.

PFL mode routes a mono sum of the pre-fade, post-insert signal from the module to the control room speakers (or head-phones) without affecting the signal to the groups or main mix outputs.

SIP mode causes all non-soloed input channels to be muted, while the control room speakers continue to monitor the main mix output. This allows the selected input to be heard in isolation (all other inputs are muted) but in its correct position and level in the mix.

The SIP muting of a module may be disabled by removing internal jumper J1 ("SAFE") on that module. The module will now not be muted when an SIP is triggered from another input.

If the stereo input is used as an effects return, this allows the soloed input to be heard both in its correct position, at the correct level, and with any added effects.

**k)** Peak indicator LED. The peak indicator illuminates when the signal level in either left or right channels exceeds + 14dBu, corresponding to a level approximately 7dB below clipping. The LED will remain illuminated for about 1 second after the peak which triggered it, so even short peaks are indicated clearly.

The signal is monitored at the insert sends.

1) Routing Switches. These switches allow the signal from the module to be routed to the main stereo mix bus, and to the group busses in pairs.

By using the PAN control set fully clockwise or anti-clockwise, it is possible to route the signal from the module to only one group (although this signal will of course then only be mono.)

**m)** Fader. The fader is a long throw device with a total travel of 100mm, giving a cut-off of better than 80dB from the unity position, while still leaving 10dB of extra gain in hand.

**n) inserts.** There are unbalanced insert points for left and right channels of the module, operating at the console internal level of < 2dBu.

The insert is situated after the EQ section and width control, but before the mute circuitry.

# **OUTPUT MODULE**

A Series 6000 output module is a dual unit containing two separate groups, each with their own tape/group monitor section. One half of the module only is described, the two halves being identical except for the physical positioning of the controls.

Each monitor section can access either the associated group output or the corresponding tape return. Since the monitor section gives a path to the stereo mix bus it may also be used at mixdown as an effects return facility; to make this possible without moving all the tape return cables over to the line inputs, separate monitor input sockets are provided. These are normalled so that the tape return signal usually flows to the monitor section, but a jack plugged into a monitor input overrides this. When this is done the tape return signals are still connected to the first 16 or 24 input modules by being normalled through the line input sockets.

A) Group/Tape switch. This is labelled with the number of the group/tape track to which it refers. Even numbers are at the top of the module and odd ones in the lower section. When the switch is in the monitor section is accessing the tape replay signal.

**B) Equalisation section**. This is a two band system with HF (treble) and LF (bass) shelving controls. Both are centre- detented for rapid resetting. The HF control gives up to of boost/cut at 10 kHz, and the LF control up to boost/cut at 50 Hz.

C) Aux 1 and Aux 2 sends. These are normally post of the monitor volume control; in other words the send level is also controlled by monitor volume. This is the usual requirement when the monitor sections are being used as effect returns at mixdown. When recording or overdubbing both sends may be switched prefade, to provide foldback and headphone mixes. Both sends are always controlled by the monitor section ON switch.

D) PRE switch. This switches Aux sends 1 and 2 prefade of the monitor volume control

E) Panpot. Controls the positioning of the monitor section signal by panning it across the stereo mix bus. The centre-drop value is 4.5 dB to give optimal mono/stereo compatibility.

F) SUB switch. Depressing this alters signal flow in both a group and its associated monitor section, instantly configuring a subgroup. The monitor section becomes a dedicated effects return to that group, and the group is fed directly to the mix bus via its fader and the monitor panpot only.

Its operation is dealt with in detail in a separate section below.

**G) Monitor volume.** This controls the amount of the monitor signal that is added in to the stereo mix. Unity gain is at the 7 mark on the knob calibration, and 10 dB of extra gain is therefore available.

H) ON switch. When this is pressed in the monitor section operates normally, adding its contribution to the total mix. When not depressed the feed to the mix is muted, and both aux sends are also killed, even if they are set PRE of the volume control. A green LED illuminates when the monitor section is ON.

I) PFL switch. Activates the prefade listen mode, adding the monitor section signal to the PFL bus. The PFL switches on the output modules do not activate the Solo-In-Place mode even if this is selected for the input modules, but instead give an over- riding PFL condition. The PFL takeoff point is after the EQ but before the ON switch.

J) Group fader. A high-quality long-travel unit that controls the group signal level, being placed between the summing amplifier and the output stage.

K) Group output. This is electronically balanced, with a self- compensating function so that the output simulates the action of a centre-tapped transformer winding. In other words, when one leg of the output is grounded, to give an unbalanced feed, the other leg provides a level increased by 6 dB so that the overall level is unchanged. The output will drive up to +26 dBu into a 600 Ohm load.

L) Group insert. This is an unbalanced insert point, operating at a nominal level of  $\cdot 2$  dBu. It is positioned immediately before the group fader in the signal path.

# **OPERATION OF THE SUB SWITCH.**

This facility may require a little time for familiarisation as it is a powerful control that changes the signal flow of the group and monitor section in several ways, eliminating the need for extensive repatching when using subgrouping.

When SUB is pressed, the monitor section signal, which will now probably be an effect return destined for that subgroup only, is removed from the mix bus and instead summed into the group in the same way as a channel signal. Similarly the group signal is summed directly into the mix bus, its level being controlled by the group fader as usual and its stereo positioning being set by the monitor panpot. The group output operates in the normal way but is of course affected only by the fader and not the panpot. The function of the numbered group/tape switch is now confined to switching the metering between the group and the monitor section input; it does not affect the signal flow.

# +4/-10 SWITCHING.

The group outputs and monitor inputs may be switched together to change their operating level from +4 dBu (normal professional level) to -10 dBv (sometimes called Tascam level) for optimal matching with a given tape machine. The switching changes both outputs in a module simultaneously.

This is done by means of a push-button switch that is mounted internally to the module, with its actuating shaft pointing downwards. The shaft is OUT for +4 operation and IN (is pushed towards the module front panel) for  $\cdot 10$ . The circuitry used actually afters stage gains rather than simply attenuating, and noise and headroom performance is therefore preserved. The inputs and outputs naturally remain fully balanced.

# OUTPUT MODULE METERING.

Metering is by LED bargraph, Internally selectable between peak and VU (average) mode. The meter source is a function of the monitor section and switches with the group/tape switch.

Peak/VU mode change is by moving push on links on the rear of the meter PCB. For each meter two must be moved; both are placed to the LEFT for peak mode, and both RIGHT for VU. (Looking from the rear of the console) The meter calibration presets are also accessible from here.

## MASTER MODULE.

The master module performs the following functions:

1) Stereo mix bus summing and master fader control.

- 2) Control room speaker signal selection and metering.
- 3) Studio speaker feed control.
- 4) Auxiliary send masters and monitoring.
- 5) Line-up oscillator and talkback facilities.
- 6) Solo mode selection.
- 7) Mute bus master control.

These controls and features are described in detail below.

A) Mix master fader. This is a dual long-travel unit that controls the overall mix output level. It is placed between the summing ampliflers and the output stage in the signal path.

B) Mix insert points. These are unbalanced and operate at the standard level of  $\cdot 10 \text{ dBv}$  ( $\cdot 7.8 \text{ dBu}$ ). They are placed immediately before the fader in the signal path.

C) Mix outputs. These are electronically balanced, with a self- compensating function so that the output simulates the action of a centre-tapped transformer winding. In other words, when one leg of the output is grounded, to give an unbalanced feed, the other leg provides a level increased by 6 dB so that the overall level is unchanged. The output will drive up to +26 dBu into a 600 Ohm load.

The mix outputs may be configured for -10 dBv operation by changing two push-on link positions on the master RH PCB. Move LK9 to LK10 position, and LK11 to LK12 position. This has implications for the 2-track return sensitivity; see below.

D) Control-Room Select. These switches route either the stereo mix bus or one of three two-track returns to the control-room speakers. The two-track inputs are switchable in sensitivity between +4 and -10 levels by altering push-on links on the master RH PCB; alternatively two-track C only may be configured for cassette replay levels in the same way.

To change all 2-track returns to -10 dBv mode remove links LK3, LK4, LK13 & LK14. Ensure the mix output level has been similarly changed or there will be a jump in level on switching between mix and 2-track replay.

To change 2-track C only to cassette level (-10dBv) add links LK1, LK2, LK7 & LK8.

E) Studio Level. Adjusts the feed from the Control-Room select switches described above to the Studio speak output. This output is intended for direct connection to a power amplifier driving studio loudspeakers, so that takes can be replayed to musicians at once without repatching. The studio output is ground- cancelling to prevent hum-loop formation, and is not affected by PFL operations. It is also unaffected by the MONO and DIM switches. (See below) F) Studio ON. A switch that allows the studio feed to be muted at once to prevent acoustic feedback, without losing the desired level set up on the Studio volume control.

**G)** Control Room Level. Sets the level fed from the control room select switches (or PFL bus in PFL mode) to the monitoring power amplifiers. The CRM outputs are ground cancelling to eliminate ground loops and have the capability to drive loads down to 600 Ohms.

NB. These outputs are automatically disconnected when headphones are plugged into the front-panel jack socket.

H) MONO switch. Sums together the L and R control-room signals for mono compatibility checking, eg to allow for AM radio broadcasts in mono. Signals recorded with incorrect phase on one channel will tend to disappear in mono mode. No other output is affected.

i) DIM switch. Reduces the control-room level by 20 dB to allow telephone answering, etc, without altering the level set on the main volume control. No other output is affected.

J) PFL/AFL LED. Illuminates when a PFL or AFL button is pressed anywhere on the console, indicating that the PFL signal has been switched to the CRM outputs and L-R metering. Does not come when Solo-In-Place is activated.

K) SOLO mode switch. When this is depressed, the input module Solo buttons activate the Solo-in-Place mode rather than PFL. Buttons marked PFL (on the output modules) and AFL (on the aux masters) do not change in function.

L) SOLO-IN-PLACE LED. Lights in Solo-In-Place mode only.

M) Mute bus master switch. Pressing this mutes all inputs that have their mute bus buttons depressed.

N) Mute Bus LED. Illuminates when mute bus is active, le muting taking place.

O) Oscillator ON switch. Turns on the internal line-up oscillator. When set to off, the oscillator is completely disabled to eliminate any possibility of signal leakage. The oscillator is normally routed to all groups and all six auxes, but this may be altered by means of push-on links on the master RH PCB. These links also control the routing of the talkback signal.

To remove talkback/osc from Auxes 1 & 2, remove LK5.

To remove talkback/osc from Auxes 3 & 4, remove LK6.

The talkback/osc signal is not directly routed to the stereo mix bus. The oscillator signal is not routed to the studio outputs.

NB. When the talkback facility is in use, the oscillator is automatically disabled.

**P) 10K button.** Switches the oscillator frequency from 700 Hz, the normal line up frequency, to 10 kHz, which is useful for tape machine alignment. The oscillator is specially optimised for minimal level change with frequency and fast settling.

Q) OSC LEVEL. Controls oscillator output level.

**R)** Talkback ON switch. Enables the Internal talkback microphone. When not enabled the mic gain is set to minimum to prevent any trace of crosstalk. The control-room outputs are automatically dimmed to prevent acoustic feedback, and the line-up oscillator is disabled.

Talkback is normally routed to all groups and all auxes, but not the stereo mix bus. It is also routed to the studio outputs, after the studio ON switch and level control. The aux routing may be altered as described for the oscillator. (See above)

S) Talkback level control. Sets the gain of the talkback mic stage.

T) Aux master controls. The six aux master sections are all identical and consist of a level control and an AFL button that gives access to the combined aux signal after the level control.

When only one source is routed, the unity gain position is at the 7 mark.

When aux metering is fitted, the metering point is the same as the AFL takeoff point, le after the level control.

U) Phones jack socket. This supplies the same signal as the control-room outputs, but at a level and impedance suitable for headphones down to and including 8 Ohms. A useful controlled impedance feature automatically matches the output to the type of headphones plugged in, so that 600 Ohm types give approximately the same loudness as 8 Ohm versions without adjustment.

NB. Plugging in headphones disconnects the control room outputs on the rear of the console.

**MASTER METERING** is by two VU meters with associated peak LEDs. The metering point is just after the PFL switching and before the CRM volume control. In PFL mode these meters both read the PFL bus level, so that levels almost anywhere in the console may be checked.

# 24 Track Monitor/Effects Return Module

Each optional module contains four identical monitor/effects return sections. Up to two modules may be fitted in a Series 6000.

### 1) Effects Return Input Sockets

On16-bus consoles, the inputs to the four monitor sections are normally connected to four tape returns (either tracks 17, 18, 19 and 20 or tracks 21, 22, 23 and 24.)

However, plugging a patch cord into the patch socket at the top of the module over-rides this signal, and allows the monitors to be used as line level effects returns to the stereo mix.

Note: on 16 channel consoles, there is no connection for tape tracks  $17 \cdot 24$ , and so these modules can only be used as effects returns.

On 24-bus consoles, all tape monitoring is carried out using the monitor sections on the group modules. The 24 Track Monitor/Effects Return modules are used only as effects returns.

The inputs are electronically balanced, and the sockets are wired as follows:

Tlp:	нот	(in phase signal)
Ring:	COLD	(Out of phase signal)
Sleeve:	GROUND	

The sensitivity of the inputs may be either + 4dBu or -10dBu ("TASCAM" level), and can be set individually for each section by means of internal jumpers, as follows:

Effects Return 1: for +4dBu fit J 1: for -10dBu fit J 2. Effects Return 2: for +4dBu fit J 5: for -10dBu fit J 6. Effects Return 3: for +4dBu fit J 9: for -10dBu fit J10. Effects Return 4: for +4dBu fit J13: for -10dBu fit J14.

### 2) Auxiliary Sends

Two auxillary sends are available. These are normally token after the VOL pot, but may be switched prefade by pressing PRE.

In the pre-fade position, the sends may be taken before or after the ON switch, depending on internal jumper selection.

Effects Return 1:	before ON switch, fit J 4
	after ON switch, flt J 3
Effects Return 2:	before ON switch, fit J 8
	after ON switch, fit J 7
Effects Return 3:	before ON switch, fit J12
	after ON switch, fit J11
Effects Return 4:	before ON switch, fit J16
	after ON switch, fit J15

## 3) Pan

The PAN pot is a centre detented control with a loss of 4.5dB at its centre position. It allows the monitor/effects return signal to be panned to the required position in the stereo mix.

## 4) Vol

The volume pot enables the level of the monitor/effects return signal in the stereo mix to be adjusted.

## 5) On

The monitor/effects return section and its associated auxiliary sends are activated by pressing the ON button. Operation is indicated by an adjacent green LED.

## 6) PFL

The PRE FADE LISTEN button allows the monitor/effects return signal to be soloed, independent of the ON or VOL controls. Operation is indicated by an adjacent red LED.

# **REAR CONNECTOR PANELS (STANDARD)**

# **INPUT CONNECTOR PANEL**

## 1) Line Inputs

These stereo standard jacks carry the electronically balanced Line Inputs. They are wired as follows:-

Tlp:	НОТ	(in phase signal)
Ring:	COLD	(Out of phase signal)
Sleeve:	GROU	ND

Line inputs 1-24, (1-16 on 16 bus consoles), are normalied from the tape returns. This allows the Tape Returns to be sent directly to the input Channels without having to re-patch.

#### 2) Mic Inputs

These carry the electronically balanced Microphone inputs and are wired as follows:-

Pin 1:	GROUI	ND
Pin 2:	нот	(in phase signal)
Pin 3:	COLD	(Out of phase signal)

The Microphone inputs can be fed with a +48v Phantom Power for Capacitor microphones. This is controlled by the Phantom Power switch on the individual input Channels.

#### 3) Channel Inserts

The Channel inserts are standard, unbalanced, stereo jacks used to carry both insert send and insert return signals at the internal level of -2dBu.

Under normal conditions with nothing inserted the signal is normalied through the jack socket and thus inserting a jack will automatically break the link. Channel inserts are wired as follows:-

Tip:	Insert return - unbalanced
Ring:	Insert send - unbalanced
Sleeve:	Ground

### 4) Line Outputs

The Line Outputs are standard, unbalanced, stereo jacks used for sending signals to peripheral equipment at the internal level of ·2dBu. Minimum loading is 2kOhm. They are wired as follows:-

Tip:	HOT (in phase signal)- unbalanced
Ring:	Ground
Sleeve:	Ground

# OUTPUT CONNECTOR PANEL 1-8, 9-16, 17-24

#### 1) Group Inserts

This is a stereo, standard break point jack socket which contains both insert send and return at the internal level of -2dBu. The socket is connected as follows:-

Tip:	Insert return	(unbalanced)
Ring:	Insert send	(unbalanced)
Sleeve:	Common ground	

Under normal conditions i.e. with socket not in use, the send is connected to return so that inserting a jack automatically breaks the link.

## 2) Group Outputs

Fed from the eight individual groups these electronically balanced outputs are wired as follows:-

Pin1;	GROUND	
Pin2:	HOT	(in phase signal)
Pin3:	COLD	(Out of phase signal)

### 3) Monitor Inputs

The Monitor Inputs carry FX returns via the Monitor section or the SUB button without having to un-plug the Tape Returns. They are wired as follows:-

Tip:	нот	(In phase signal)
Ring:	COLD	(Out of phase signal)
Sleeve:	GROUND.	

#### 4) Tape Returns

These stereo standard jack sockets, which are normalled to the Monitor input socket and the corresponding Line input, carry the balanced tape returns and are wired as follows:

•		•
Tip:	HOT	(in phase signal)
Ring:	COLD	(Out of phase signal)
Sleeve:	GROUND	

## 5) Auxiliary Outputs (1-8 only)

These are standard jack sockets for the ground compensated Auxiliary outputs, and are wired as follows.

Tip:	Signal
Ring:	Ground Sense
Sleeve:	Ground

# MASTER CONNECTOR PANEL

The master connector panel contains the Control room and studio monitor outputs, the 2 track returns, the main mix outputs, the mix inserts and the PSU connector.

#### 1. C/Room and Studio

These standard jacks carry the outputs to the control room and studio monitors. These are ground compensated and are wired as follows:-

Tlp:	Signal
Ring:	Ground Sense
Sleeve:	Ground

#### 2.2-track Returns B and C

These carry the 2-track returns and are wired as follows:

Tip:	нот	(in phase signal)
Ring:	COLD	(Out of phase signal)
Sleeve:	GROUND.	

#### 3. Mix Outputs and 2-track A Returns

The main mix outputs and 2-track A returns are electronically balanced and are wired as follows:-

Pin 1:	GROUND	
Pin 2:	НОТ	(In phase signal)
Pin 3:	COLD	(Out of phase signal)

#### 4. Mix Inserts-

The mix inserts allow access to the main mix immediately before the main stereo fader, they are unbalanced at the internal console level of 2dBu and are wired as follows:

Tip:	insert return
Ring:	insert send
Sleeve:	Common Ground

#### 5. PSU

Power to the console is provided via the PSU connector located on the master rear connector panel.

#### 6. Ground

The main console ground is available on a binding post.

# STEREO INPUT CONNECTOR PANEL (OPTIONAL)

## 1) Left and Right Inputs

The left and right inputs to each module are electronically balanced, and are carried on XLR-type connectors wired as follows:

- Pin 1: GROUND
- Pin 2: HOT (In phase signal)
- Pin 3: COLD (Out of phase signal)

## 2) Left and Right Inserts

The inserts are stereo 1/4" jacks, used unbalanced to carry both send and return signals at the console internal level of -2dBu. The sockets are wired as follows:

Tip:Insert return (unbalanced)Ring:Insert send (unbalanced)Sleeve:Common ground

# **REAR CONNECTOR PANELS (PATCHBAY)**

## **Input Connector Panel**

#### Mic Inputs

These male XLR connectors carry electronically balanced microphone inputs, wired as follows:

Pin 1:	Ground	
Pin 2:	нот	(in phase signal)
Pin 3:	COLD	(Out of phase signal)

The microphone inputs can be fed with +48V for phantom powering capacitor microphones. This is controlled by the phantom power switches on the individual input channels.

## **Master Connector Panel**

The master connector panel contains sends and returns for three 2-track mastering machines, as well as the PSU connector.

The sends are connected internally to the main Mix output; however, other signals may be sent to the 2track outputs using the sockets on the patchbay.

#### 1) 2-Track A Send

This stereo electronically balanced signal is carried on two male XLR connectors, wired as follows:

Pin 1:	Ground	
Pin 2:	HOT	(In phase signal)
Pin 3:	COLD	(Out of phase signal)

#### 2) 2-Track A Return

This stereo, electronically balanced signal is carried on two female XLRs, wired as above.

### 3) 2-Track B and C Sends

These stereo, electronically balanced signals are carried on 1/4" stereo jacks, wired as follows:

Tip:	HOT	(in phase signal)
Ring:	COLD	(Out of phase signal)
Sleeve:	Ground	

#### 4) 2-Track B and C Returns

These stereo, balanced signals are carried on 1/4" stereo jacks, wired as above.

#### 5) PSU Connector

Power to the console is provided via the PSU connector.

#### 6) Ground Connector

The main console ground is available on a binding post.

# **Output Connector Panel**

#### 1) Auxiliary Outputs

These are ground compensated outputs on male XLR connectors, wired as follows:

Pin 1:	Ground
Pin 2:	Signal
Pin 3:	Ground Sense

#### 2) Mix Output

This stereo, electronically balanced output is carried on two male XLR connectors, wired as above. The main desk Mix output is normally connected to these sockets; however, other signals may be sent to the output using the sockets on the patchbay.

### 3) Control Room and Studio Outputs

The outputs to the control room and studio monitors are ground compensated, and are carried on male XLR connectors, wired as follows:

Pin 1:	Ground
Pin 2:	Signal
Pin 3:	Ground Sense

#### 4) Multi-Track Tape Machine Connectors

The signals to and from the multi-track tape machine are carried on three, 56-way VARELCO connectors.

Each connector carries 8 sends and 8 returns, and is wired as shown in the drawing MI2589.

All signals are electronically balanced.

The tape sends are connected internally to the group outputs; however, other signals may be sent to the tape machine using the sockets on the patchbay.

# **Tie Line Connector Panel**

64 balanced tie-line connections are provided to the patch-bay. These are carried on four, 56-way VA-RELCO connectors, wired as shown in the drawing MI2579.

# **STEREO INPUT CONNECTOR PANEL (OPTIONAL)**

Each panel carries the input connections for four Stereo input Modules.

Inputs Left and Right

The inputs to the Stereo Modules are at line level, and are electronically balanced.

The inputs to each channel are carried on both 1/4" stereo jack sockets and XLR-type connectors, wired in parallel.

The jack sockets are wired as follows:

Tip:HOT (In phase signal)Ring:COLD (Out of phase signal)Sleeve:Ground

The XLRs are wired as follows:

Pin 1:GroundPin 2:HOT (In phase signal)Pin 3:COLD (Out of phase signal)

Soundcraft Series 6000

# PATCHBAY

The patchbay system installed on the Series 6000 greatly enhances the console's versatility and ease of use by providing convenient, standardised connection points to many sections of the console, and to numerous pieces of external equipment. This allows rapid insertion of peripheral equipment into the system by means of patch-cords.

The jacks used are of the small "Bantam" type. All balanced connections are wired thus:

> Tip: Ring: Sleeve:

HOT COLD GROUND (in phase signal) (Out of phase signal)

All unbalanced connections are wired thus:

Tip:	нот	(in phase signal)
Ring:	GROUND	
Sleeve:	GROUND	

All ground compensated outputs are wired as follows:

Tip:SignalRing:Ground SenseSleeveGround

## **Master Patch Points**

### 1) Parallels

A set of four balanced jacks together connected in parallel with each other. These may be used to connect one output to a number of inputs. The must not be used to connect outputs together.

#### 2) 2-Track Sends

The three 2-track sends are normally connected to the main MIX output. These balanced jacks allow alternate signals to be sent to the 2-track machines, for example when making copies of a master tape.

#### 3) 2-Track Returns

These balanced jacks allow access to the outputs of the 2-track machines. Inserting a jack accesses to 2-track output signal, but does not affect the signal to monitor switching on the master module.

#### 4) Mix Inserts

These unbalanced jacks carry the main mix insert points, at the console's internal level of -2dBu. The insert points are situated before the main MIX fader. Inserting a patch cord into the SEND socket gives the pre-insert signal, but does not break the signal flow through the insert point.

inserting a patch cord into the RETURN socket breaks the signal path through the insert point, and replaces the signal with that on the patch cord.

### 5) Mix Outputs

These balanced jacks allow access to both the main Mix output from the desk, and to the MIX OUT sockets on the rear connector panel.

Inserting a patch cord into the left socket of each pair (the "tail" of the arrow ) gives the console mix output, but does not affect signal to the output sockets.

Inserting a patch cord into the right socket of each pair (the "head" of the arrow) enables an alternate signal to be sent to the mix output sockets.

Note: these jacks do not affect the signal to the 2track sends.

#### 6) Control Room Outputs

These jacks give a balanced insert point for the control room outputs, and act in a similar way to the MIX outputs described above.

#### 7) Studio Outputs

These jacks give an insert point for the STUDIO outputs, and act in a similar way to the MIX outputs described above.

#### 8) Aux Outputs

These jacks give a balanced insert point for the auxiliary outputs, and act in a similar way to the MIX outputs described above.

#### 9) Phase Reverse

Two sets of jack sockets for phase reversal. Each pair of jacks is wired tip to ring and ring to lip. These may be used to correct phasing errors in balanced signals.

# **Channel Patch Points**

#### 1) Line In

These jacks carry the electronically balanced line inputs.

Line inputs  $1 \cdot 24$  ( $1 \cdot 16$  on 16 channel desks) are normalied from Tape Returns  $1 \cdot 24$ . This allows the tape returns to be sent directly to the input channels for mixing without the need for re patching. Inserting a patch cord into the Line in socket over-rides the Tape return signal.

## 2) Insert

The SEND and RETURN jacks give an insert immediately before the input channel fader, and operate similarly to to those on the mix.

The jacks are unbalanced and operate at the console Internal level of -2dBu.

#### Line Out

These jacks carry the post fade signal from the input channels, unbalanced, at the console internal level of -2dBu.

# **Group and Monitor Patch Points**

#### 1) Insert

The SEND and RETURN jacks give an unbalanced insert point immediately before the group faders, at the console internal level of -2dBu.

### 2) Group Out

Each group output is normalled to a multi-track tape send (Group 1 to send 1; group 2 to send 2 etc.) These balanced jacks also carry the group output, to allow the signal to be patched to other destinations.

(Note: on 16 bus consoles, groups 1 - 8 are also normalled to sends 17 - 24 respectively.)

#### 3) Mon

The outputs from each track of the multi-track tape machine are connected to the relevant monitor sections. These balanced jacks allow other signals to be fed to the tape return inputs of the monitors to allow their use as additional line inputs during re-mix, for example an effects returns. The inputs are electronically balanced.

# **Tape Machine Patch Points**

### 1) Send

Normally, the tape sends are connected to the group outputs as described previously. However, insertion of a jack into the SEND jack will interrupt the normal signal path from the group and replace it with the signal on the patch cord, to allow other signals to be sent to the tape machine.

#### 2) Ret

The RETURN jack allows access to the output from the multi-track tape machine. Inserting a jack will not affect the signal-from the tape machine to input channels or monitors.

# **Tie Line Patch Points**

64 balanced tie-lines are available for the connection of external equipment.

## Stereo Input Patch Points (optional)

1) Insert Left and Right

The send and return jacks give a pre-fade insert point for the left and right channels of each input. The jacks are unbalanced and operate at the console internal level of -2dBu.

# INSTALLATION

#### **Applying Power**

Before switching on the Series 6000 console check that the mains voltage selector in the power unit is set at the correct mains voltage for your area, and that the fuse is of the correct rating.

For operation on voltages between 220 and 240V ac the fuse should be rated at 3.15A, 20mm, antisurge.

For operation on voltages between 100 and 120V ac the fuse should be rated at 6.30A, 20mm, antisurge.

#### **Interface** Levels

The Soundcraft Series 6000 is designed to work with 2 different nominal input/output levels. These are the usual +4dBu, (1.228Vrms), and -10dBV (= -7.8dBu), often called the Tascam level. These levels refer only to matching with the tape machine being used; mic sensitivity is not affected.

Note that whichever gain mode is in use, the nominal internal level in most parts of the console, including the insert points, is -2dBu and this does not change.

#### a)OUTPUT GROUPS

Two things need to be changed on a group; the nominal output level and the tape-return sensitivity. This is done by using S1, the push switch on the rear of the module PCB pointing backwards away from the facia.

+4 MODE: \$1 OUT -10 MODE: \$1 IN

b)MASTER MODULE RIGHT PCB - MIX, CONTROL ROOM SELECT etc Refer to the Master Module circuit description.

c)EFFECTS RETURN/24 TRACK MONITOR MODULES For details of input sensitivity selection, refer to the module description.

#### **Connector Conventions**

All XLR type connections are normally wired to the following standard:-

- Pin 1 GROUND
- Pin 2 HOT (in phase signal)
- Pin 3 COLD (Out of phase signal)

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## **General Wiring Procedures**

To take full advantage of the excellent signal to noise ratio and low distortion of Soundcraft consoles care must be taken to ensure that incorrect installation and wiring does not degrade the performance of the desk. Hum, buzz, instability and Radio Frequency Interference can usually be traced to earth loops and inferior earthing systems. In some areas, especially heavy industrial areas, the incoming mains earth will not be adequate and a separate technical earth for all the audio equipment must be supplied. However, check with your local electricity supply company to ensure that safety regulations are not infringed or negated.

The successful, hum free, installation of a system requires forethought and the establishment of a set of ground rules which must be consistently adhered to at all stages of installation.

#### 1) Initial Wiring Considerations.

a) For optimum performance, it is essential for the earthing system to be clean and noise free, as all signals are referenced to this earth. A central point should be decided on for the main earth point sys tem and all earths should be "star fed" from this point. It is common electrical practice to "daisy chain" the earths to all electrical outlets but this method is unsultable for audio installations. The preferred method is to run an individual earth wire from each outlet back to the system star point to provide a safety earth of screen reference for each piece of equipment.

A separate earth wire should also be run from each equipment rack and area to the star point. This may or may not be used depending on circumstances, but it is easier to install in the first place than later when problems arise.

The location of the star point should be a convenient, easily accessible place preferably at the rear of the console or in the main equipment rack.

b) Install separate "clean" and "dirty" mains outlets, wired individually back to the incoming mains dis tribution box. Use the "clean" supply for all audio equipment and the "dirty" supply for all lighting, vending machines etc. Never mix the two systems.

c) If necessary, to provide sufficient isolation from mains borne interference, install an isolating transformer for the "clean" supply. The isolation transformer should be provided with a Faraday Shield which must be connected to earth.

d) Never locate the incoming mains distribution box near audio equipment, especially tape recorders, which are very sensitive to electro-magnetic fields.

e) Ensure that all equipment racks are connected to earth, via a separate wire back to the star point.

f) Equipment which has unbalanced inputs and outputs may need to be isolated from the rack to prevent earth loops.

## 2) Audio Wiring

Having provided all equipment with power and earthing connections, consideration must be given to the method of providing audio interconnection and adequate screening of those interconnections. This must be done in a logical sequence to avoid problems and assist in the localisation of problem equipment.

a) Connect Control Room Monitor system to the console and check for any hum, buzz, or RFI. Only when you are satisfied with the quietness of the console and the monitor system should you proceed with the next step.

b) Connect multitrack tape recorder, via noise reduction system if applicable and again check that the system is still clean.

c) Connect stereo tape recorders, studio monitors, echo and foldback sends one at a time, checking and isolating any connection which degrades performance.

d) Connect all peripheral devices.

e) Connect all microphone lines.

By following this sequence much time and future trouble will be saved and the result will be a quiet, stable system.

### 3) Shielding

Audio equipment is supplied with a variety of input and output configurations, which must be taken into consideration when deciding where the screen connections should be made. There are three sources of unwanted signal being impressed on the screen, which are as follows:

- i Extraneous electrostatic or electromagnetic fields.
- ii Noise and interference on the earth line.
- iii Capacitive coupling between the screen and signal wires.

To minimise the adverse affects of the unwanted coupling to the signal wires, it is important that the screen is connected at one end only, i.e. the screen must not carry any signal current. Any signal on the wires within the screen will be capacitively coupled to the screen and this current will ultimately be re turned to the source of the signal, either directly, if the screen is connected at the signal source end, or indirectly via the earthing system, if the signal is connected at the signal destination end. The indirect connection will cause an increase in high frequency cross-talk and should be avoided wherever possible. Therefore, in general, always connect the shield only at the signal source end. In high RF areas, the screen can also be connected to earth via a 0.01 micro Farad capacitor. This will present a short circuit at RF frequencies, thus lowering the effective shield impedance to ground. However, at low audio frequencies the reactance of the capacitor will be sufficiently high not to cause an earth loop problem.

Combinations of unbalanced, balanced and electronically balanced, (differential), systems mean that there are nine interconnection permutations. The optimum of the screen in each case is shown in Table 1.

# **SERIES 6000 OPERATION**

## Introduction

The Soundcraft Series 6000 console has been designed to provide the varied facilities required in a small modern recording studio.

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

1. Record mode: Recording direct from microphone or line input onto the multitrack tape recorder.

2. Multitrack playback: Listening to what has been recorded.

3. Overdubbing: Recording more tracks whilst listening to what has already been recorded.

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

A detailed explanation of all the controls and switches has been given and it is assumed that the user is familiar with basic multitrack recording methods.

## **Record Mode**

### 1.Record Mode

This is the basic starting point in making a recording. Input channels are placed in the microphone mode by placing the LINE button in the "up" position. The signal is routed to the desired console group output by pressing the relevant group routing button on each channel. The signal can be panned between odd and even numbered groups by using the channel Pan Pot.

The signal may be metered at pre-fade, post EQ, by pressing Solo and using the mix meters and at the relevant Group Output to the tape machine on the Group Output meter, if the tape return button is in the "up" position. A monitor balance is achieved using the appropriate level control. The monitor Pan Pot will pan the signal between the left and right speakers.

The various gain and level controls throughout the signal chain may now be adjusted to set the operating levels for optimum signal to noise ratio and headroom conditions. 2.Channel and Group Faders

In general the channel and group faders should be set to approximately the zero position. This will en able the engineer to increase the level by 10dB, or fade out completely, while normally operating in the fine resolution area of the fader travel. The absolute position of the fader is not too critical, but situations where, for example, the channel fader is operated at -30dB will usually mean that the pre fader headroom has been reduced to a point where an amplifier could overload and cause distortion. Operating with the channel fader below the zero point and the group fader above zero also means that the noise contribution from the group bus could degrade the normally excellent signal to noise ratio.
#### 3. Microphone Gain

Having set both channel and group faders as above, the microphone gain is set to give the required level at the group output. The amount of gain required depends on the type of microphone, the sound pressure level developed by the instrument or voice and the distance between the sound source and microphone.

#### 4.Insert 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 points or the Group Insert points. Which insert point is used will depend on the type of effect required. If only an individual instrument is to be controlled, the channel insert point would be used. However, if a number of microphone channels have been mixed to a single group, for instance, backing vocals, then it will be necessary to patch in an overall limiter, using the group insert point.

#### 5. Headphone Mix

During recording, it is of course essential for the musicians to hear what they and everybody else are playing. Headphone mixes are derived using any or all of the auxiliary sends, either direct from the input channels, or alternatively from the group monitor channels. Deriving the headphone mix from the monitor channels has the advantage of giving the musicians a mix when replaying the recording just made, which is useful when overdubbing to enable them to hear their performance. However, a mix derived from the monitor channels during recording will be affected by any gain riding of the channel or group faders, which may cause problems.

This can be avoided by using the input channel auxiliary send 1 and 2, switched to PRE. The head phone signal is thus derived from before the channel fader, and is unaffected by any change in the post-tion 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 sig nal to echo remains constant, regardless of fader position. This condition is met on auxiliary sends 1-6. If PRE is NOT selected.

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 echo send will be derived from the input channels. However, if it is a mix for monitoring purposes only, then it would be more relevant to use the monitor channel sends to drive the echo send mix. 7.Echo Returns

The output of the echo or reverberation device, i.e. the echo return signal, is patched into either an unused input channel if this signal is required to go to tape, or into an unused monitor channel 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 LINE. 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.

### **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 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 the next section.

### 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 but tons. 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.

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 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 re corded 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 play back 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.

### **Remix Mode**

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

Remixing is the process of combining all the previously recorded tracks together with any special effects devices such as harmonizers, flangers and delay lines.

The remix mode is entered by selecting Line Input, (LINE) on the input channels 1-16 or 1-24 depend ing on the number of groups on the console. This connects the outputs of the multitrack machine to the inputs of each channel, via the gain trim control. As when laying the initial tracks, the gain trim can be adjusted to allow the fader to work near its nominal zero position.

By selecting MIX, the channel can be routed directly to the stereo mix, via the channel Pan Pot, to allow positioning of the signal within the stereo perspective.

Sub-groups may be formed by routing a number of signals to any one group and pressing the SUB button on that group. This signal is summed into the relevant group bus AND that group is routed to the stereo mix bus.

NOTE that pressing SUB routes the Monitor Input to the relevant group and ALSO routes that group to the stereo mix bus.

On the Series 6000 each Group/Monitor module can be used as a Line level input, say for FX returns, and directly accesses the stereo mix bus.

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

All 6 auxiliary sends can be used to feed echo or effects devices, as headphone mixes are not required during remixing.

The composite stereo mix will be controlled in level by the stereo master fader and the level metered on the stereo mix meters. These meters will also indicate the level of any solo signal, or any of the stereo tape returns selected on the monitor source selection buttons. These monitor source selection but tons allows quality and level checking of the stereo recording in the same way as described for multitrack playback.

### MAINTENANCE

Every console that leaves Soundcraft undergoes a thorough testing at all stages of manufacture. These tests include a thorough testing of all the functions of the completed mixer, which consist 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, with the minimum of test equipment needed to isolate and rectify faults.

### **General Fault Finding**

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

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

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

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

If in doubt about the module operation, set up an adjacent module in exactly the same way, which will allow a direct comparison between a working and possible non-working module. Route the channel directly to MIX, so that the channel may be monitored in the normal way. Using an osciliator 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 circultry can be by passed by switching out the Hi pass filter and the equalizer.

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

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

### **Removing Modules**

Before removing modules, always switch off the console.

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. A module should NOT be unplugged or plugged in with the power ON.

### **Meter** Alignment

Each VU meter has its own individual drive card attached to the rear of the meter.

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

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

### Lamp Replacement

Illumination of the VU meters is provided by 2 wire ended lamps in each meter. These are 12 volt lamps wired in series. The 100 Ohm series resistor provides turn on surge current limiting to prolong lamp life.

## **TECHNICAL DESCRIPTION.**

The technical details below are provided to assist with fault. finding, and to give a general appreciation of the philosophy behind a console that provides a significantly improved level of audio performance. We believe that this will be of help in obtaining the best possible results from the equipment. We also hope that even if you have no intention of servicing the console yourself you will find parts of it worth reading. Sadly, in a concise description of this sort it is not possible to describe all of the many detailed design decisions that have been made.

The table below gives a quick guide to PCB numbers and their associated circuit diagrams:

	PCB No.	Diagram No.
Input PCB.	SC2721	ED2721
Master RH PCB	SC2722	ED2722
Master LH PCB	SC2726	ED2726
Stereo I/p PCB	SC2727	ED2727
PSU interface	SC2723	
Patchbay o/p rear conn.	SC2724	
Master rear conn.	SC2752	
Output rear conn.	SC2753	
Standard i/p rear conn.	SC2754	
Patchbay i/p rear conn.	SC2755	
Output PCB.	SC2762	ED2762
Effects Return/24-Track Mo	on. SC2787	ED2787

### INTERNAL LEVEL.

The internal operating level of the console is a nominal -2 dBu.

## **INPUT MODULE. (SC2721)**

#### INPUT AMPLIFIER.

The input stage is built around TR1,TR2 and IC1,IC2. In line mode only IC2 is used and only one section of the dual gain-control pot. In mic mode switch sections S2a & S2d short the line input feeds (from the line input resistors on the rear connector PCBs) together and to ground to prevent mic/line crosstalk in the input cableform. This has no significant effect on the impedance seen by external equipment. S2d also changes the gain range of the second amplifier stage IC2a. Line gain setting, like that of mic mode, is done by active gain control rather than a combination of attenuation and amplification. This keeps noise to the minimum while preserving maximum headroom. The noise performance is distinctly superior to the conventional approach of attenuating the line input and applying it to the mic amp.

S1 switches the +48V phantom power to the mic input via 6K8 resistors R1,R2. VR1·A (the first section of the gain control pot) controls the gain of the first stage, varying it between approx 0 and +50dB. The two discrete transistors, of a special low-noise type, and the two op-amp sections of IC1, make up a high-gain stage that combines the low noise of discrete devices with the high linearity and drive capability of integrated op-amps.

ASPECTS OF THIS MICROPHONE PREAMPLIFIER ARE COVERED BY SOUNDCRAFT PROVISIONAL PATENT No. 8719014.

C8, C31, C40 optimise stability at low gains and faulty components here may result in HF oscillation at some gain settings. Diodes D1, D2 prevent latch-up of IC1a on overdrive; diodes D3, D4 protect the opamp against phantom-power transients. R1 sets the loading on the microphone, and may be re duced in value if special circumstances demand it without affecting other input parameters.

#### HIGHPASS FILTER.

This is conventional, with a Butterworth maximally-flat characteristic and a -3 dB frequency of 100 Hz. EQUALISER.

The EQ sections are apparently quite standard, using Wien-type networks, but in fact incorporate careful optimisations of impedance and loading to give the lowest possible distortion. When the EQ is switched out it is completely removed from the audio path and the input to it is also removed, minimising the generation of redundant ground currents.

#### PEAK DETECTOR.

This consists of TR9,TR10 & LED3. When the peak signal level at the insert send exceeds  $\rightarrow$  14 dBu (set by R106,107) TR9 turns on, turning off TR10 and removing the short across LED3. C36 provides a fast-attack, slow-decay action. The current passing through R109, LED3 & R110 is also used to drive the ON indicator, LED1, which is controlled by D11. The action of these two LEDs is however completely independent.

#### PFL.

The pre-fade listen takeoff point is immediately after the insert return, R40 being the mix resistor summing the channel signal into the PFL virtual earth bus. R41 is the PFL detect resistor, which similarly sums a DC current into the PFL detect bus, which also operates at virtual earth. This is a proprietary Soundcraft feature that prevents any possibility of clicks being induced into the audio mix buses. When fault-finding it is important to remember that the PFL detect bus does not in normal operation change its DC level when a PFL button is pressed. It does however go briefly low at power-up to reset the ON flip-flops. (See below)

ELECTRONIC MUTING & SOLO LOGIC.

This is performed by the series shunt pair of FETs TR3, TR4. As in many circuits in the S6000, the sophistication of this circuit is not immediately obvious, but need not concern the maintenance engin eer as it is a property of the configuration employed rather than adjustment or selection of components. IC5-B reverses the phase-inversion produced by IC6-A and provides drive for the input metering when fitted. The control voltages for TR3, TR4 are carefully tailored and are provided by components R47-R50 & R105, C25,26, and D5,6. The voltage on TR3 gate swings between 0V and 12V, and that on TR4 between + 0.4V and -12V. It is important that the gates do not swing more positive than this. The antiphase drive to R49,R50 is provided by TR7, TR8; note that TR7 also controls the ON LED via D11.

IC6-B is a flip-flop toggled by the ON switch; it is kept in a stable state by positive feedback through R52,R54. The output is low (-16V) when the channel is unmuted. Initial reset at power-up is applied via D7 and R55, which are connected to the PFL-detect bus. This is normally at 0V and does not affect the flip-flop, but at power-up the PFL-detect line is briefly pulled low by the master module, jamming the flip-flop output low (channel on). This bi-directional logic saves a bus conductor for other more important uses. The flip-flop output is applied to FET driver TR7 via level-shifting components R56,57,58 & TR6. Mute signals from the programmable mute bus are OR- ed with this signal by D8 and TR11; S9 controls whether the module responds to the mute bus. This mute bus is at +17V when inactive and +15V when muting.

Solo-In-Place mutes are applied separately via safeing jumper J1

and are OR-ed with other mutes by D9. The input module(s) originating the SIP state must not mute, and therefore Solo switch S10 turns on TR5 via R111 and defeats the incoming SIP mute signal. CN4 is the MIDI computer port; note that the flip-flop is computer-controlled on an edge-triggered basis. This means that in the unlikely event of computer failure the console will continue to operate manually without any problems.

#### FADER POSTAMP.

A conventional stage with high drive capability and iow noise.

#### ACTIVE PANPOT.

This is built around VR14 and IC8. The buffer stages provide a modest amount of gain, set by R117,118 and R113,116, that is modified by impedance-controlled feedback through R75,76. Once again the exact mode of operation is not entirely obvious but faultfinding is straightforward.

SOUNDCRAFT ACTIVE PANPOT TECHNOLOGY IS COVERED BY PROVISIONAL PATENT No. 8800168.

#### ROUTING MATRIX.

This is straightforward in all except its physical construction; if you are unable to find the summing resistors they are located under the front of the routing switches. If resistor replacement is necessary 1/8 Watt types must be used, with care taken that they do not foul the switch- cap when installed. The Soundcraft proprietary routing system is used which combines the minimal bus noise of conventional switching with the low-crosstalk capabilities of back-grounding. This Third Method has been in use since 1980 and in our opinion offers overwhelming advantages over standard techniques.

#### POWER SUPPLY.

Rail decoupling and fusing is by R103,104. If a serious fault such as a short between the rails occurs, it is localised to the module. If these resistors are replaced, they should be 1/4W 10 Ohm types spaced off the PCB.

# **OUTPUT MODULE. (SC2762)**

This consists of two identical sections of group + monitor, and only one (the odd-numbered group) is described.

#### GROUP CIRCUITRY.

The group summing amp is is built around IC1 and TR1, TR2. As with the mic preamp, the requirement is to achieve the low noise of discrete devices combined with the high linearity and economy of industry-standard ICs. A unique differential-mode summing action is produced by TR1,2 that uses commonmode rejection to prevent supply-rall perturbations from degrading the crosstalk figures. If this is not taken into account it can easily provide the ultimate limitation on a console signal isolation. DC conditions are set by R2,3 and negative shunt feedback applied to generate the bus virtual earth through R1,4,5. IC1-B is a unity-gain inverter that provides a balanced feedback path to the transistor pair, maintaining symmetry and supply-rall rejection. D1 prevents opamp latchup and R11 ensures stability in the face of external capacitative loads.

IC2-B is a low-noise fader post-amp giving +10 dB of gain, with its feedback network (R13,14,15) also doing duty as a potential divider giving the  $\cdot 10$  dBv output, selected by S1-A, the global level-change switch.

IC3 is the EBOS (Electronically Balanced Output Stage), that simulates a floating output transformer. This operates by virtue of complex negative/positive feedback and current sensing through RiC1,2 & R17,18. This need not concern the engineer as maintenance of this stage is normally limited to IC replacement if the opamp encounters something unpleasant in the outside world. The preset PR1 is used to set the two outputs of the EBOS to equal signal level at manufacture, and will not normally need ad justment unless RIC1 or RiC2 are replaced. It is not necessary after opamp replacement.

If this adjustment is required, the simplest method is to connect two 22K resistors in series to the hot & cold o/p pins, and set PR1 for a null at the resistor junction.

If the EBOS stage is giving inexplicably poor linearity, check C13. Local decoupling is essential for HF stability.

#### MONITOR SECTION.

The monitor input (either tape replay or effects return) is via IC9-A, a differential amplifier with its gain switched for  $+4/\cdot10$  by S1-B. This is followed by a conventional HF/LF 2-band EQ and the PFL takeoff point. The monitor ON switch, S10, shorts the audio path to ground both before and after the +10 dB post-amp IC2-A, so that providing unused monitors are switched off, the unwanted noise added to the mbx bus is a minimum.

From here the signal passes to VR6, a conventional panpot. The mix resistors R42,44 have the low value of 4K7 so that crosstalk and noise are minimised.

The SUB switch S9 operates as follows to create a sub-group with dedicated effects return:

1) S9-A & S9-B connect the return amplifier output to the monitor EQ regardless of the position of group/tape switch S7.

2) The monitor section signal is routed directly into the group bus via S9-C and R40.

3) The monitor panpot is connected to the group output by  $S9 \cdot D$ , so that the subgroup signal can be panned between L & R mix buses.

#### POWER SUPPLY.

The two halves of the output module have separate supply fusing/decoupling in R45,46 and R47,48. Hence if one gp/mon section is disabled by a serious fault, the other continues unaffected. Resistor replacement must be with 1/4W 10 Ohm.

## MASTER RH PCB. SC2722

### (Mix, Control-Room Monitor, etc)

Where this PCB is in stereo, only the Left channel is described.

#### MIX SUM AMPS.

These use similar technology to the group summing amps. In this most vital part of the console the requirements are for minimal noise (aided by the low value of mix summing resistors at 4K7) and excellent supply-rall rejection. This is particularly important as the mix bus usually has a large number of summing resistors connected to it and therefore operates at a higher noise-gain.

The mix sum amp is built around IC1 and TR12,13. The mix fader has 0 dB at the top of its travel, allowing the sum amp to operate with  $\cdot 10$  dBv ( $\cdot 7.8$  dBu) at its output to give an enhanced headroom of 28 dB. Negative feedback is via R4,5,6. D10,11 prevent opamp latchup. C3 adjusts HF phase margins for optimal stability.

POST-AMP & EBOS.

IC3-A is the mix fader post-amp, with link-selectable output level by LK9,10. The gain in the  $\pm 4$  position is  $\pm 10$  dB. IC4 is the mix output EBOS (see output module description)

#### CONTROL-ROOM SELECT

is performed by switches S1,2,3,4, which select the input resistors for differential amp IC6-A. Unused resistors are grounded on the inward end to prevent crosstalk. R29,32 can be switched out by LK3,4 to increase the sensitivity for -10 operation.

#### CRM OUTPUTS.

From the CRM select amp IC6 the signal passes through PFL switching FET TR8, which normally presents only a few Ohms of resistance, and CRM volume control VR4. Stereo metering is taken off just before VR4. The MNO switch S5 sums left and right together via R38,138. DIM is performed by grounding the bottom of R37,137.

The CRM output stage is built around IC7-A and TR10,11. The IC provides the loop gain while the discrete transistors provide the muscle to drive loads down to 100 Ohm. The output is ground-cancelling; in other words the cold leg is used as an INPUT to sample the ground potential of the external equipment. This is then added to the console output, so that the external equipment has its ground and input moving together, and experiences no input-ground signal that would allow the entry of interference. R46 defines the cold leg impedance as very low, and R41 sums the signal from the external ground with the desired signal. If simple unbalanced operation is necessary then the cold leg should ideally be grounded at the console, but the low value of R46 means that this is not vital. The output transistors operate in class AB when driving a significant load such as headphones, biased

by D2,3. R47 compensates for varying headphone impedances and ensures the stage is short-circuit proof.

When headphones are plugged in, the jack socket contacts break the CRM feed to the rear sockets

#### STUDIO OUTPUTS.

These are ground cancelling as for the CRM outputs, but without the high current capability; the minimum load impedance is 2 kOhm. The studio feed is taken off before FET TR8 and is therefore unaffected by PFL operations. Talkback is injected into the output via R49.

#### TALKBACK SYSTEM.

An Internal electret mic (powered via R76 & C30) is amplified by IC11-A, VR2 setting the gain. S7 enables the system; S7-C routes the talkback and defeats the oscillator output, while S7 A and S7-B dim the CRM signal. S7-D reduces the electret preamp gain to minimum when talkback is not in use, preventing unsuspected clipping from affecting other circuitry.

#### OSCILLATOR.

The line-up oscillator is a modified Wien bridge with 700 Hz/10 kHz frequency switching by S9. Amplitude control is by FET TR5, its nonlinearity being reduced by the usual Vds/2 feedback via R83,84. Excessive amplitude makes zener D8 conduct, turning on TR7 and thus TR6. This tends to turn off TR5 by taking its gate negative and thus increases the negative feedback via R85,87. The control time-constant is set by C36 & R91. R89 improves the level-control law.

When the oscillator is not switched on, S8 B applies 100% negative feedback to the opamp and oscillation is impossible.

#### PFL SUMMING & SWITCHING.

IC9-A is the PFL-bus summing amplifier; It is followed by IC9- B which re-inverts the PFL signal to keep the phase correct. The combined signal is applied to the PFL FETS (TR9,19) through DC blocker C25, while R56 bootstraps the FET gates for low THD when they are on. Normally D4 pulls these gates to -17V, and the FETs are firmly off.

IC10-A is the virtual earth PFL detector; its detect bus is normally held at 0V by feedback through R69. This detect bus signals PFL only, and not SIP, and is therefore used by output and aux master modules only. The equivalent detect bus for the input modules is separate as it may signal SIP instead. The opamp output is offset to +10V by R67 Pressing any PFL/AFL button connects + 17V to the bus via a 100K resistor, and IC10-A output swings negative to -0.6V, clamped by D6. This turns off FET driver TR1, which turns on TR2 and the PFL indicator LED1. The normal CRM FETs are turned off via D1,D101, while TR9,19 turn on as D4 ceases to conduct.

As described for the input PCB, the PFL detect bus is also used to reset the channel ON flip flops at power-up. This is done by network R65,66 & C28,D5, which pulls the IC10-A reference briefly low. This is reflected in the DC level of the detect bus.

#### SOLO-IN-PLACE SWITCHING.

The input module detect bus is applied to IC10-B, which is a virtual earth stage like the PFL detector. With no Solo condition, its output sits at  $\pm 10V$ , but fails to  $\pm 0.6V$  when an input module Solo button is pressed, and is clamped by D7. This turns off TR3, and its collector goes high to  $\pm 17V$ . When Solo mode switch S10 is in the 'PFL' position, this voltage rise is coupled to the PFL detector through R68, and causes PFL action exactly as described above.

When S10 is in SIP mode, the voltage rise is applied to emitter follower TR4, which pulls the SIP-mute bus up to  $\pm 1.3V$ , set by R73,74. (Its normal level is 0V) LED2 illuminates to indicate that an SIP is actually occurring. The SIP-mute bus now mutes all inputs that have not been safed, and are not originating a Solo.

#### PROGRAMMABLE MUTE BUS.

This line normally sits at +17V, but when S11 is pressed the short across LED3 (MUTE ACTIVE) is removed and it illuminates. The bus voltage fails to +15V and any input with its mute programming switch in is muted.

### **METERING SYSTEMS.**

#### LED BARGRAPHS.

Eight-way meter PCBs are used for output (group/tape) metering, and 6-way for aux metering, when fitted. Both consist of identical repeated sections. Preset PR1 sets the calibration, and IC1-A & D1,2 form a precision rectifier. In VU mode J1 & R1 increase the rectifier gain, while J2 selects either fast-attack/slow-decay for peak mode (R5 charging C2), or integration by R4,C2 in VU mode. IC1-B is a unity-gain buffer driving the internal comparators of the 3914 bargraph ICs IC2,3. R6,R7 generate a quasi-logarithmic law by adding the drive voltage to the 3914 comparator reference ladder. R8 & D3 drop the + 24V supply to + 18V for the 3914s, decoupled by C3. LED current is defined by R9,R10.

## MASTER LH PCB (SC2726)

The Series 6000 has a unique auxiliary summing system that employs Negative Impedance Converters to enhance summing amp headroom without compromising noise performance. The headroom is 30dB, as opposed to the 22dB that would be possible with conventional technology. We believe this to be an industry first.

The technique is protected by Soundcraft Provisional Patent No. 8800168.

The Left-Hand Master PCB contains six auxiliary master sections, each with a master gain control and AFL switch. The six sections are identical and only Aux 1 is described.

The aux summing amplifier is built around TR1, TR2 and IC1. The use of a differential input stage gives good supply- rail rejection. IC1-B performs differential to single-ended conversion (phase-summing) and passes the signal to the feedback resistor R4 via inverter/Negative Impedance Converter IC1 A. C4 gives DC blocking and R7 defines the maximum excursion of the transistor collectors to prevent common-mode problems with IC1-B.

When VR1 is at a low setting, for example when a large number of aux signals are being summed, the voltage across R10 is small and IC1. A acts solely as an inverter. When VR1 is set high, positive feed-back is applied through R11 to enhance the gain of the stage. This is equivalent to concatenating a variable negative impedance with R4, and the circuit acts as though R4 has increased in value. The effective summing amp gain is therefore varied by VR1 so that it is only increased when more overall gain is required, and under normal circumstances it is low enough to give 30dB of headroom.

IC2-B acts as a unity-gain buffer, and IC2-A as a ground- cancelling output stage with 600 Ohm drive capability. The external ground voltage is sampled via R19 and added to the output such that it cancels out on reaching the external equipment; this gives most of the benefits of a balanced output without the complexity and cost. The negative feedback for this stage is split between R16 and R17 to maintain linearity down to 10 Hz when driving heavy loads. The aux metering feed (aux metering is not always fitted) is taken via R14, and the AFL feed via R12.

The AFL LEDs are arranged in a chain for current economy, the chain current being set by R121. When ever an AFL switch is pressed the short across its LED is removed, and it illuminates.

# 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 hand ling, 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.









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