

- 2-TRACK RETURNS Tape returns allow access to the outputs of the 2-track tape machines. (eg. A jack inserted into the return socket takes the signal from the tape machine). The 2-track returns are routed from the patchbay to the Control Room select switches.
NB: 2-track A return is normalled to 2-track B and C sends. Inserting a jack into 2-track A return does not break the signal flow to 2-track B and C sends. This allows simple tape to tape transfer without the need to reconnect the tape machines.
- Ext 1 and 2 These are external inputs to the control room monitor. The signal is fed the the CRM select switches located on the Command Module.
- MONO These are insert send and return points, the insert send is normalled to the return. Inserting a jack into the return socket breaks the signal flow to the mono output. Inserting a jack in the send socket does not break the signal flow..
- PARALLEL There are also Parallel facilities, sets of 4 jacks which are connected in parallel with each other, to enable an output to be connected to a number of inputs. They must NOT be used to connect a number of outputs together.

Also on the patchbay there are facilities for Data from the automation system to be accessed.

Balanced tielines are available for the connection of external peripheral equipment. All are connected identically and can be used to suit the needs of the studio.

b) Command Patch Points for later TS24 models.

PHASE REVERSE	Two jack sockets are available for phase reversal. (eg If the connectors on external equipment have been miswired.) These are connected Tip to Ring and Ring to Tip.
TALKBACK	This is an output for the feed from the Talkback and may be used to route the Talkback signal to other sections of the console not covered by the Communications section of the command module. The talkback signal is routed to the patchbay via pin 4 on the 40-way connector from the Command Module to the patchbay. This line is live at all times.
OSCILLATOR	Used as an output point from the internal oscillator, and may be used for tape alignment.
PHONES	An output from and an input to Phones 1 and 2, Left and Right is available. This allows the operator to route the Headphone signal to parts of the console not covered by the Phones selection on the Command Module or signals may be sent to the Headphone circuit from sources which are not covered on the Phones select matrix.
AUX	These are Auxiliary listen jacks and can be used to route the auxiliaries to other sections of the console or to peripheral equipment.
ALT	This is an input to the Alternative speakers to allow a signal to be routed to the speaker.
MIX INS	Mix insert sends and returns are available. Inserting a jack into the send socket does not break the signal flow. Inserting a jack into the return socket breaks the signal flow from the insert send. The Insert send and return jacks, access the stereo mix immediately before the stereo fader. The insert send is Ground Compensated to enable the operator to interface to a variety of external equipment.
MIX O/P	An alternative output from the main Mix is available to route the signal from the main Mix to peripheral equipment if required. The Mix outputs are normalised to 2-track A send.

2-TRACK SENDS	The tape sends allow access to the inputs of the 2-track tape machines. (eg. A jack inserted into the send socket sends the signal to the input of the tape-machine).
2-TRACK RETURNS	Tape returns allow access to the outputs of the 2-track tape machines. (eg. A jack inserted into the return socket takes the signal from the tape machine). The 2-track returns are routed from the patchbay to the Control Room Select switching. NB: 2-track A return is normalled to 2-track B and C sends. Inserting a jack into 2-track A return breaks the signal flow to 2-track B and C sends. This allows simple tape to tape transfer without the need to reconnect the tape machines.
MONO	These are insert send and return points, from and to the mono feed, the insert send is normalled to the return. Inserting a jack into either the send or the return socket breaks the signal flow.
Ext 1 and 2	These are external inputs to the control room monitor. The signal is fed the the CRM select switches located on the Command Module.
CRM INSERTS	These are insert send and return points, from and to the Control Room Monitors, the insert send is normalled to the return. Inserting a jack into either the send or the return socket breaks the signal flow.
STUDIO INSERTS	These are insert send and return points, from and to the Studio Speakers, the insert send is normalled to the return. Inserting a jack into either the send or the return socket breaks the signal flow.
LISTEN MIC INS	FROM STUDIO - This allows the output from the Listen Microphone to be routed to sections of the console which are not available on the Listen Mic select switches. TO MASTER - This enables signal from the Listen Mic to be sent to the 2-track master. The From Studio point is normalled to the To Master point, inserting a jack into either socket breaks the signal flow between the two.

PARALLEL

There are Parallel facilities, sets of 4 jacks which are connected in parallel with each other, to enable an output to be connected to a number of inputs. They must not be used to connect a number of outputs together.

Also on the patchbay there are facilities for Data from the automation system to be accessed.

Balanced tielines are available for the connection of external peripheral equipment. All are connected identically and can be used to suit the needs of the studio.

Optional Stereo Sub-Group Patchcard

Each stereo sub-group is provided with 8 patch points. All patch points are balanced and use small Bantam type jacks wired as follows:-

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

INPUT A LEFT	Enables a signal from an external device to be routed to the Left Input of Line Input A.
INPUT A RIGHT	Enables a signal from an external device to be routed to the Right input of Line Input A.
INPUT B LEFT	Enables a Signal from an external device to be routed to the Left Input of Line Input B.
INPUT B RIGHT	Enables a signal from an external device to be routed to the Right input of Line Input B.
LEFT INSERT	These are insert send and return points to the Left-hand signal path of the stereo sub-group. The send is normalled to the return. Inserting a jack into the send socket does not break the signal path, inserting a jack into the return socket breaks the signal path from the send socket.
RIGHT INSERT	These are insert send and return points to the Right-hand signal path of the stereo sub-group. The send is normalled to the return. Inserting a jack into the send socket does not break the signal path, inserting a jack into the return socket breaks the signal path from the send socket.

NB: The Insert points are located after the Equaliser and is "flipped" between the Channel and Mix paths when the Equaliser is moved.

Optional Effects Return Patchcard.

The FX return module is provided with eight jack points, two for each of the returns. Return 1 is located at the top of the patchcard and return 4 at the bottom.

RETURN 1	The top input jack is normalled through to the lower input jack. Inserting a connector into the lower input jack will break this normalling, eg. the signal flow from the above input socket will be broken.
RETURNS 2 - 4	These are identical to Return 1.

2.07 OPTIONAL VCA SUB-GROUPING SYSTEM

The TS24 VCA sub-grouping system has been designed to provide the operator with an additional sub-grouping facility that is both easy to use and install. The VCA sub-grouping system can be retrofitted and involves the addition of control panels on both the Input/Output modules and the Command Module. These control panels may be fitted into the space beside the fader which can otherwise be used for the Soundcraft Automation system.

1. General Facilities

Each channel VCA may be assigned to one of eight sub-groups, which are controlled from eight sub-group master faders on the master module. Each sub-group also has its own CUT button, its status being indicated by an LED.

2. Channel VCA Module

Each channel VCA module is mounted beside the normal linear fader occupying the space which is otherwise used for the automation option. The VCA action is superimposed on the existing linear fader, and can either be used for fine trimming or complete fades.

The only control is the push-button switch used for assigning that particular module to a given sub-group. A seven segment LED display indicates which sub-group is currently controlling that module; each time the button is pressed the internal logic advances the display by one, up to 8. When the button is held down, it advances automatically at the rate of about one digit per second.

The initial state of the display on power-up is blank, indicating that the VCA module is not controlled by any sub-group. In this state the VCA is switched out of the circuit completely, and the channel fader functions as if the VCA system was not there. This state is also returned to after the displayed number exceeds 8.

3. Master Module

This consists of eight linear faders and eight illuminated CUT push-buttons. The latter have an electronic toggle action similar to the CUT buttons on the TS24 I/O module. When pressed it is illuminated and sets that sub-group to maximum attenuation. (In excess of 80dB). A second press of the a CUT button will return the signal.

Two master reset switches are also provided so that the sub-group assignment of all channels can be quickly returned to the blank (non-VCA) state. Both buttons must be pressed for this to happen; this is a safeguard against resetting the whole desk accidentally.

2.08 OPTIONAL AUTOMATION

The Soundcraft automation system records onto tape, in real time any changes in fader level and mute conditions. The system can be regarded as a data acquisition system and central processing unit. Data from the console is sent to the CPU which processes the data and sends the result back to the console and down to tape.

The system is easy to install and has been designed with the operator in mind. There are only 8 master controls, and one button for each of the 3 functions per module; fader automation, mute control, and VCA grouping.

Using the automation system to remember and perform all fader movements means that complicated mix-downs become extremely easy. If, having "written" a first mix, you wish to alter the fader level in some sections, simply place the system in up-date and the system will allow you to alter the level up or down from where the fader happens to be. A section may be re-written if the operator requires.

Null LEDs are provided so that, when leaving up-date or re-write, fader levels can be matched to any previous settings to avoid any jump in level.

The mute routine is handled independently of the fader level control, and is designed for the fastest possible response time, giving accurate repeats of the original mute. Once written, a mute can be re-written at any time by pressing the CUT button near the fader. To return to the previously recorded information press the CUT button again.

Any channel can be a VCA group master. Merely hold down the nominated master SET button and set the faders required to that group by pressing the appropriate SET buttons once. There can be any number of group masters and an intelligent interrogation routine allows easy recognition of which fader has been assigned to which group. Each master is recognised by its LED and, by holding down any channel SET button, the interrogation routine will be initiated and masters and their associated faders will be displayed.

By the use of priority encoding and a fast microprocessor, Soundcraft have developed a rapid system, providing quick response time and reducing cumulative time delay. Although the system is priority based, the steady state of the desk is also recorded some of the time, this allows the operator to start mixing half-way through a particular track.

The system requires two tracks on the multi-track machine, information is read from one and new information recorded on the other. This routing is controlled from the master module.
NB: The two tracks should not be adjacent.

2.09 OPTIONAL DISK BASED AUTOMATED MIXING

Greater accuracy and flexibility on mix down may be achieved by recording the changes made to a console throughout the mix. These changes may then be automatically reproduced so that further updates can be recorded into the mix, thus building progressively towards a standard of accuracy which is hard to achieve manually. The Master Mix automation system records, or WRITES, fader movements and channel mutes onto floppy disk. The system may be supplied factory fitted from Soundcraft as an integral part of the console, or retro-fitted in the studio to expand and enhance an existing installation.

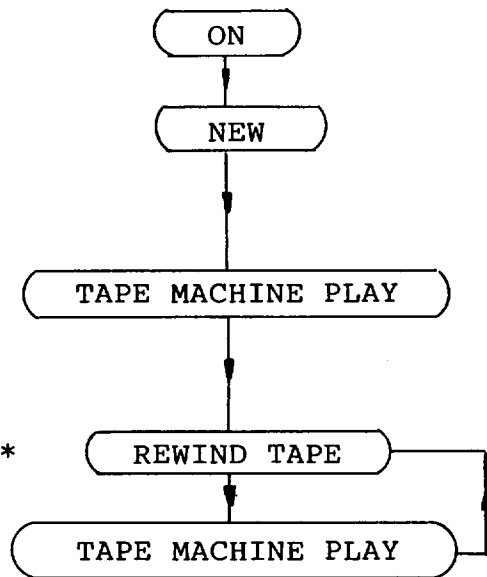
For the TS24 Soundcraft have developed a unique interface to the Master Mix VCA fader which, unlike other less integrated system, interfaces the full flexibility of the console muting facilities to control the automation. By using the console mutes, a full channel mute which follows the console logic gives more than ergonomic enhancement. In MIX status, when the aux's are normally used for FX sends, the cuts mute both Pre- and Post-fade sends, without such an interface automation mutes cut only Post-fader.

The mix data is synchronised to a single track of standard SMPTE/EBU time code recorded onto tape, the same standard code is used for most synchronising applications. The built-in multi-standard SMPTE/EBU generator also features a facility for generating code backwards for extending an inadequate pre-roll at the start of a mix.

The Master Mix system is logical in operation and enables the engineer to perform quite complex mix downs swiftly and easily. The system is designed so that the operator does not have to continually select and deselect various functions thus allowing the operator to concentrate on the mix without having to worry about what the computer is doing!

In a mix down situation the following procedure would normally be adopted:-

OPERATING PROCEDURE



Whilst in Play select
UPDATE

Repeat from * to fill memories C and D if required.

Fifth pass

TAPE MACHINE PLAY

COMPUTER OPERATION

Brings the computer On-line.

Computer is aware that this is a new mix and selects WRITE A, All Write/Mute Write.

Any changes in fader level or mute conditions are recorded onto floppy disk and stored in memory A.

Computer selects Read A, Write B

Changes may be made to the previous mix. At the end of the mix the data stored will be a combination of data from the previous mix and any new data. Data is now stored in memory B. Data in memory A is retained.

Memories A,B,C and D full.

Computer selects Read D, checks to see if SAFE is selected on Memory A.

If No - Computer selects Write A.

If Yes - Computer checks to see if SAFE is selected on Memory B.

If No - Computer selects Write B.

If Yes - Computer selects Write C, unless SAFE is selected.

If SAFE is selected on A,B and C then data is assembled in a scratch memory and written back into Memory D at the end of the pass.

As well as offering all the benefits of an automated mix down Master Mix also has an integrated VCA subgrouping system. This allows any channel to be assigned as a Group Master. A Grand Master can also be created to control the overall level of all the Groups. The Grand Master and the Group Masters are LED indicated at all times and a simple interrogation of any channel will indicate which other channels form part of that group.

Since some engineers prefer to record tracks without the VCA, a bypass facility is provided which re-routes the audio through the fader, thus allowing the original recording to be made conventionally. However, the VCA circuitry has been designed to give extremely good noise and distortion figures so that on recording the VCA is virtually invisible.

The Master Mix automation system complements the Soundcraft design philosophy to produce a console that is logical in operation and meets the demands of today's recording studios.

3.00 Installation

3.01. Connector Conventions

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

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

All inputs and outputs are electronically balanced.

3.02. 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 heavily 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 system, 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 unsuitable 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 TS24 "star point" is located on the PSU distribution board, under the master module on the main framework. Earths can be taken from here and taken with the main wiring loom to external equipment.

- b) Install separate "clean" and "dirty" mains outlets, wired individually back to the incoming mains distribution 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 born 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 returned 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.

TABLE 1

	OUTPUT	INPUT	SCREEN
1	Unbalanced	Unbalanced	Source
2	Unbalanced	Balanced	Source
3	Unbalanced	Differential	Source
4	Balanced (Note 1)	Unbalanced	Destination
5	Balanced	Balanced	Source
6	Balanced (Note 2)	Differential	Destination
7	Differential (Note 3)	Unbalanced	Source
8	Differential	Balanced	Source
9	Differential	Differential	Source

- Note 1 - The shield is connected to the destination earth point, which is opposite to normal practice, because the signal wires being shielded are referenced to the input earth, not the output earth.
- Note 2 - If the output transformer is centre tapped to earth, the screen should be connected at the source.
- Note 3 - When an active differential output is operated in unbalanced mode, it is very important that the output current returns to earth via the shortest, least reactive route. Check for instability at the output.

N.B.

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

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

3.03 PATCH BAY INTERFACE

To interface the TS24 console to incoming mic-lines, tape machines etc, there is a number of Molex connectors on the Patchbay PCBs which are for this purpose. (Female connectors are provided).

The 4-pin Molex connectors used on the patchbay interface cards are 0.156" FML housing. The crimp terminals used are Comway 2578 TL.

The gold 3-pin Methode connectors used for the Mic inputs are also 0.156" FML housing. The crimp terminals used are Comway 2578 TL.

Both connectors take 7 x 0.2mm wire.

NB: Early TS24 models.

On the base of the console there is a series of XLR sockets for the following:-

Output to Headphone circuits 1 and 2.

Output from the 6 Auxiliary sends.

Output to Control Room Monitor speakers

Output to the Alternative speakers.

Output to the Studio speakers.

Input from the Listen Microphone.

1. I/O Modules

Each I/O Module is provided with 8 patch points. Each set of 8 patch points is located on patch cards, constructed with PCB mounted jack sockets.

On the base of the patch cards there is a series of connectors which connect the various patch points to the wiring loom and hence the rest of the console and any mic inputs, external equipment etc.

The two Molex connectors located at either end of the PCB enable the operator to connect up to any external source to suit the needs of the studio, these would normally be the Mic inputs, the gold three pin connector, (gold pins are used to ensure consistent noise free contact on low level signals), and the Tape send and return, the four pin connector.

Female connectors are supplied and should be wired as follows:-

Three pin gold connector

Pin 1: Earth

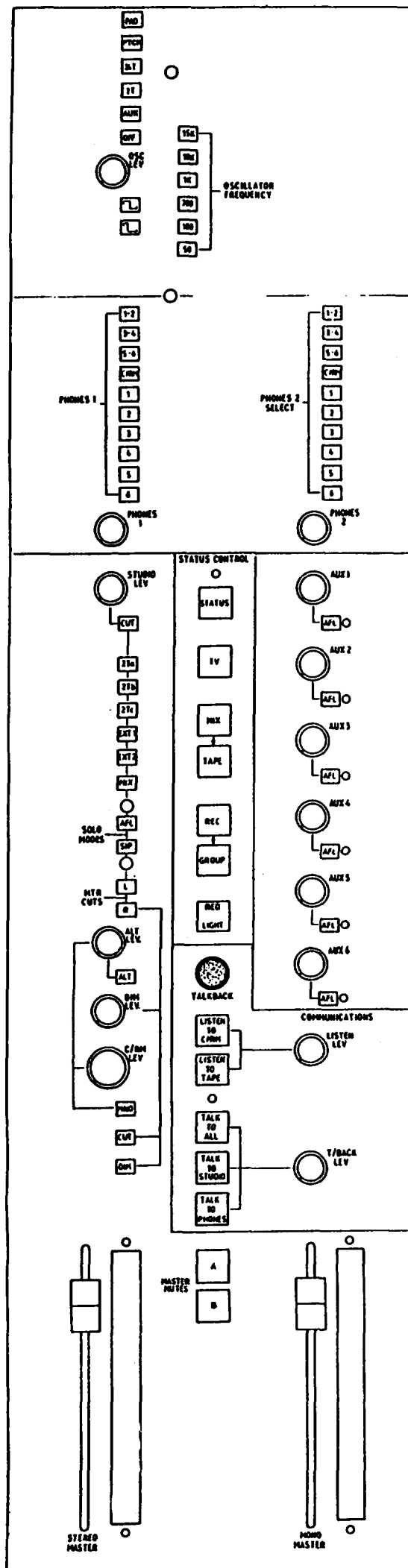
Pin 2: -ve

Pin 3: +ve

Four pin connector

Pin 1: -ve Tape send signal.

Pin 2: +ve



Pin 3: -ve Tape return signal.
Pin 4: +ve

(See MI2394 - Pin numbering may not conform to pin 1 ident on connectors.)

2. Master Patch Cards

a) Early models.

Molex connectors are located on the Master Patch Cards to enable the operator to connect to any peripheral equipment.

These should be wired as follows:-

OSC	Pin 1	-ve
	Pin 2	+ve
TB	Pin 3	-ve
	Pin 4	+ve

MIX O/P	Pin 1	-ve	Mix o/p left.
	Pin 2	+ve	
	Pin 3	-ve	Mix o/p right.
	Pin 4	+ve	

TAPE SENDS/RETURNS, EXT 1 AND 2

Pin 1	-ve	Left.
Pin 2	+ve	
Pin 3	-ve	Right.
Pin 4	+ve	

MONO O/P	Pin 1	-ve
	Pin 2	+ve
	Pin 3	-ve
	Pin 4	+ve

(The Mono signal actually appears twice, eg. on Pins 1 & 2 AND Pins 3 & 4.)

(See MI2391/2392/2393 - Pin numbering may not conform to pin 1 ident on connectors.)

NB: On early TS24 consoles the Headphone sends, the Auxiliary sends, output to the Control Room monitor speakers, the Alternative speakers, the Studio speakers and the input from the Listen Microphone appear at the XLR patchbay, located underneath the console.

All XLRs are wired as follows:-

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

b) Later models.

Molex connectors are located on the Master Patch Cards to enable the operator to connect to any peripheral equipment.

These should be wired as follows:-

TB	Pin 1	-ve	
	Pin 2	+ve	
OSC	Pin 3	-ve	
	Pin 4	+ve	
PHONES 1 AND 2	Pin 1	-ve	Phones left.
	Pin 2	+ve	
	Pin 3	-ve	Phones right.
	Pin 4	+ve	
AUX	Pin 1	-ve	Odd numbered Auxes.
	Pin 2	+ve	
	Pin 3	-ve	Even numbered Auxes.
	Pin 4	+ve	
MIX O/P	Pin 1	-ve	Mix o/p left.
	Pin 2	+ve	
	Pin 3	-ve	Mix o/p right.
	Pin 4	+ve	

TAPE SENDS/RETURNS, EXT 1 AND 2

Pin 1	-ve	Left.
Pin 2	+ve	
Pin 3	-ve	Right.
Pin 4	+ve	

MONO O/P

Pin 1	-ve
Pin 2	+ve
Pin 3	-ve
Pin 4	+ve

(The Mono signal actually appears twice, eg. on Pins 1 & 2 AND Pins 3 & 4.)

CONTROL ROOM AND STUDIO SPEAKER INSERTS

Send	Pin 1	-ve	Send left.
	Pin 2	+ve	
	Pin 3	-ve	Send right.
	Pin 4	+ve	
Return	Pin 1	-ve	Return left.
	Pin 2	+ve	
	Pin 3	-ve	Return right.
	Pin 4	+ve	
FROM STUDIO	Pin 1	Earth	Output from Listen Mic.
	Pin 2	-ve	
	Pin 3	+ve	
TO MASTER	Pin 1	Earth	Input to 2-track master.
	Pin 2	-ve	
	Pin 3	+ve	

(See MI2391/2392/2393 - Pin numbering may not conform to pin 1 ident on connectors.)

3. Stereo Sub-group Patch Card

Each stereo sub-group is provided with 8 patch points. Each set of 8 patch points is located on patch cards, constructed with PCB mounted jack sockets.

On the base of the patch cards there is a series of connectors which connect the various patch points to any external equipment and to the rest of the console.

The two four pin molex connectors located on the left-hand end of the patch card enable the operator to connect up to any external source, these are normally Line inputs A and B. (Female connectors are provided).

These should be wired as follows:-

Pin 1:	-ve	Left input.
Pin 2:	+ve	
Pin 3:	-ve	Right input.
Pin 4:	+ve	

4. Optional Effects Return patchcard

Each return is provided with two input patch points. The top socket is normalled through to the lower socket, inserting a jack in the lower socket breaks the signal from the top socket.

Molex connectors are located at the bottom of the patchcard and any equipment that is to be permanently wired in should be connected to the female connectors provided. These should be wired as follows:-

Pin 1:	-ve
Pin 2:	+ve
Pin 3:	-ve
Pin 4:	+ve

(See ED2247 - Pin numbering may not conform to pin 1 ident on connectors.)

4.00 OPERATION

4.01 INPUT/OUTPUT MODULE

CHANNEL

The Channel section with its rotary fader is usually the section that feeds an individual Mic or Line level to the routing matrix.

The Channel input section is located near the top of the module, with its associated controls; Input Gain, phantom power on/off, mic - D/I select, pad and phase reverse.

The clearly defined "Equalisation" block, containing the 4-band parametric EQ, and the HP filter is normally located in the CHANNEL path when the Master Status is RECORD and in the MIX path when the Master Status is MIX, although this may be changed locally by pressing the FLIP button. Any change in the Master Status disregards any previous "FLIPS". Both the HP Filter and the Equaliser may be switched in or out of the signal path independently.

For recording "dry" sounds the High Pass filter and the Equaliser can be switched out of the signal path. The High Pass filter is independent of the EQ.

INS places the insert patch point into the same signal path as the Equaliser, post both the High Pass filter and the Equaliser. When the Insert point on the patch bay is positioned in the CHANNEL path this allows access to the Channel Signal Path immediately before the Channel Fader. (For instance, you may wish to compress a sound before it goes to tape, a compressor can be placed in the signal path via the channel insert send and return.)

When positioned in the MIX path the insert point allows access to the Mix signal path immediately before the Mix faders.

The Auxiliary sends, which are arranged in pairs, are usually fed from the Mix path but by pressing the CH button the Auxiliary send is taken from the Channel path either Pre or Post Channel fader. This may be required, for example, when headphone mixes need to be derived from individual inputs, rather than from the Group output monitor, or when recording an instrument with a specific effect.

During RECORD the CHANNEL fader level would be set to about 0dB to allow the operator to increase the level by 10dB or to fade out completely, the Input Gain may be adjusted to give the required 0dB. (It may be necessary to switch in the 15dB PAD to avoid over-loading the Input.)

From the Channel fader the signal in the CHANNEL path may be sent to the routing matrix, via the Pan Pot, to any of the Groups or the Main Mix, the signal may be routed directly to the Group below by selecting DIR.

In certain situations it may be necessary to "ride" a level onto tape. This is usually much easier with a sliding fader, and for these occasions a fader reverse switch, REV, is provided. Only the roles of the faders change, the pan pots retain their normal functions. (The REV button is located below the Mix Pan Pot.)

Group Output level is controlled by the Group Trim, located at the top of the Module, (Group Output 1 at the top of Module 1 etc). This is the skirt of a dual concentric potentiometer with a reference mark at 0dB. (Normally during RECORD this would be set to about 0dB). The signals from the routing matrix are summed by an active gain control and then sent to the multitrack machine.

During remix, the Channel section need not lie idle, but may be used to provide extra inputs to the console or extra outputs; this choice may be individually determined on each module:-

- i. For use as an extra input (eg. for an FX return) the signal in the Channel Path may be routed to the Mix bus via the channel pan pot by pressing the button marked MIX on the routing matrix.
- ii. To use the Channel section on a module as an extra send, the button marked FDR disconnects the normal Channel input and feeds the Channel fader with a signal from the Mix Path below. This signal may be either PRE or POST the Mix fader, determined by the associated PRE button. Thus by using the Channel fader and pan pot it is possible to send to any or all of the 24 groups on the routing matrix. These can be used as extra sends during remix.

To prevent an accidental feedback loop, the operation of the FDR button prevents the Routing Matrix from being fed with the Group Output, ie. the Routing Matrix can only be fed with Tape Return when FDR is pressed.

The Channel path can be CUT independently of the Mix path by selecting CUT (located above the Channel fader). Also the Channel path has its own PFL (Pre-Fade Listen) facility. When PFL is selected a mono feed is sent to the main monitors, this facility is independent of the CUT button.

MIX

The MIX signal path normally goes to the stereo Mix bus. This means, for example, the monitor mix in "Record" mode, (if you have been monitoring on the long throw faders), becomes your main mix in "Mix" mode, and therefore a rough master mix is already set up on the faders ready for remix.

Input to the Mix path may be either Tape return or Group Output, or Tape and Group together. If the input is tape and group together there is a compensating drop in level to accommodate the two signals, (so that although two signals are being summed there is no audible difference). The source of the Mix Input is usually determined by the Master Status, Tape or Group, but may be flipped locally by pressing the SRC, Source, button. The LED located next to the Source button provides a visual indication of the Mix input source. (The LED illuminates if the source is Group.)

Monitoring when "dropping in" during an overdub session is made easier by the T and G (Tape and Group) facility. This allows both the Group Output monitor and the Tape Return on a particular module to be heard, without any appreciable audible difference in level. (Both signals are attenuated by 3dB). This means that both the musician warming up and the existing track can be heard, thus assisting both the performer AND the engineer to time the "drop-in", particularly when the "drop-in" point is tight. When the tape machine goes into record the output automatically reverts to the Line i/p.

Should a tape have been recorded at a higher level than the normal console working level of +4dBu then on early models there is a 6dB pad provided immediately after the Tape Return input. On later models a Line Trim is provided. This is the top of the dual concentric potentiometer located at the top of the module.

The Bounce (BNC) button disconnects the Mix fader and pan pot from the main Left and Right buses and sends the output to the routing matrix, OVERRIDING the normal Channel Input. This enables simple, instant track bouncing.

For example; a good balance of say 6 backing vocals has been achieved and the operator wishes to bounce this down into stereo, whilst maintaining the same balance.

1. Select BNC on the 6 modules which have been used for the individual vocals.
2. Select a pair of Groups (say 21 and 22) on the routing matrix of the 6 modules.
3. Set the Mix faders on modules 21 and 22 for the level required. Adjust the Group trims if necessary.

This has the overall effect of sending the balance being monitored directly to tape without adjustment, but INCLUDING any equalisation added.

The Mix path SOLO is dependent on the Command Module set up, which may be one of three modes, solo in place, PFL or AFL. If SIP (Solo in place) is selected then the soloed signal will appear in its normal stereo position. If PFL/AFL is selected then a mono feed will be heard but the main outputs will not be cut. PFL or AFL selection overrides any Solo-In-Place condition.

Pressing the SAFE button prevents the signal in a module from being cut by another solo. For example, normally any effects returns would be kept ON whilst soloing.

The programmable electronic muting is extremely useful when a number of signals are required to be cut at the same time. A mute may be programmed by selecting either A or B on the I/O module, pressing Master Mute A or B on the Command Module mutes all the I/O modules with that particular selection. (Master Mute A cuts all those channels with A selected and Master Mute B cuts those with B selected.) Both A and B may be selected at once on any module.

The Peak Led inspects BOTH signal paths (Channel and Mix) in the module at the following points:-

Post Channel input amplifier, Post tape/group amplifier and Post equaliser. When any of the points exceeds +15dB (5dB below clipping) the LED will light. Even short duration transients will register since the characteristics are such that reaction time is very fast, but the decay is emphasised. Both positive and negative peaks are detected.

Metering for each module is provided by a 30 segment Bar-graph display in the overbridge. The characteristics may be either PEAK or VU, this being determined by a master switch in the overbridge.

The Meter above each module always relates to the Mix path signal. When the Mix path on the module is switched to Group, then the meter will read the Group Output; when the Mix path is switched to Tape then the meter will monitor the level coming into the console from the tape machine. However, when T and G is pressed the meter will monitor the Group Output, even though both Tape and Group may be heard.

Many of the operations on the Input/Output module are ultimately defined by the MASTER STATUS buttons on the COMMAND MODULE, these determine if the console is in RECORD mode or MIX mode and determine if the Input to the Mix path is TAPE or GROUP.

4.02 COMMAND MODULE

STATUS SELECT

The Master Status buttons RECORD, MIX, MASTER TAPE and MASTER GROUP, automatically configure the whole console, and disregard any "FLIPS" or local "SOURCE" selects.

As a safety feature, the status currently in force cannot be changed until the STATUS button has been pressed. When STATUS has been pressed there is a period of 8 seconds during which the status of the console may be changed.

When the console is first powered up the initial status will be "MIX" and MASTER TAPE will be selected.

In "MIX" mode the Equaliser block and Insert point are automatically placed in the Mix Path. In this mode all cut operations (including soloing) will CUT both Pre and Post fader sends without exception, ensuring that no Effects Returns are sent to tape when the operator wishes to cut all signals to tape.

If MASTER TAPE is selected then the Input to the Mix path will be the Tape Returns, if MASTER GROUP is selected then the Input to the Mix path will be the Group Outputs.

In "RECORD" mode the Equaliser and Insert point are automatically placed in the Channel path. In this mode cut operations only cut Post fader sends. Headphone mixes may be set up without individual channels being switched ON.

When RED LIGHT is selected this activates an opto-isolated output which may be used for switching on the Studio Red Light. Red Light selection is also a safety feature.

The Red Light disables the Oscillator, prevents Listen Mic to tape, and prevents Talkback to anywhere other than in the headphones. Red light also locks the current console status, including Master Tape or Master Group. The switch is latching.

MUTE SELECT

The programmable electronic muting system is a noiseless design with defined attack and decay times, this allows the operator to perform fast mutes without fear of 'clicks' occurring.

The Master Mute controls are located between the main faders. A mute can be programmed by selecting either A or B on individual modules. If A has been selected then pressing Master Mute A on the Command Module will cut that Mix path, if B has been selected then pressing Master Mute B will cut that Mix path.

Programmable mutes can be very convenient when a number of signals need to be cut at the same time. On the TS24 console the electronic muting allows a mute to be primed prior to when the cut is to take place and you need only press one button when the cut is to occur, no more running out of hands!

HEADPHONE CIRCUITS

A headphone mix may be set up by using any pair of Aux sends or any individual aux send or the control room monitor feed or a combination of all three. The two headphone circuits allow for two different headphone mixes to be set up. For instance; you have a stereo cue mix being sent via Auxiliaries 1 and 2 to headphone circuit 1. One of the artists requests more of himself.

- 1 Tell him to plug into headphone circuit no 2
- 2 Select AUX 1 and 2 AND press say AUX 3 on the select buttons on Phones 2
- 3 Turn up Auxiliary send no 3 on the appropriate module to suit.

The artist has the same mix as before, but with the addition of more of himself.

OSCILLATOR SELECT

Before any recording session commences it is essential that the tape machines, both multitrack and 2-track master are correctly aligned.

The internal oscillator on the TS24 may be routed to the 24 Group Outputs and hence the multitrack machine by pressing the button marked 24T on the Oscillator section. The oscillator may also be routed to a 2-track machine via the main mix outputs by selecting 2T. The alignment of the tape machine may be checked at selected frequencies between 15kHz and 50Hz.

The output of the Oscillator may be also be routed to other parts of the console using the Oscillator output jack socket on the patch bay, this is useful for faultfinding purposes.

TALKBACK FACILITIES

The Talkback signal can be routed to the Auxiliaries, Headphone circuits and the Studio monitors by selecting the appropriate button. A jack socket is also available on the patchbay to allow the Talkback signal to be routed to other sections of the console if required.

An input is provided for a LISTEN microphone. This is an external microphone which would normally be located in the Studio to enable 2-way conversation. The LISTEN microphone signal can be routed to the Control Room Monitors or to Tape by pressing the appropriate button. Routing the LISTEN mic signal to tape, via the Group Outputs, allows floor counts etc. to go directly onto tape.

On later models there is also a Listen Mic output jack socket available on the patchbay so the Listen Mic signal can be routed elsewhere in the console.

MONITORING FACILITIES

Outputs are provided for three sets of monitor speakers, these would normally be the main Control Room monitors, outputs for an Alternative pair of monitor speakers and Studio monitor speakers. Each of the monitors are provided with a variable gain control and CUT facilities.

An Alternative set of speakers is useful if the operator wishes to have a set of small speakers to check balance etc.

The main Control Room monitors are provided with a DIM level control to allow output from the Control Room monitor to be attenuated. Pressing DIM introduces the attenuator into the signal path. This is useful for checking balance during Mix down. Also, conversation and telephone calls etc in the Control Room can be heard without having to compete with the band!

The MONO button sums the left and right outputs for a mono compatibility check, as well as being useful for checking for out of phase microphone pick-ups, etc.

The source for the monitors can be the main mix left and right or one of five external sources, (2-tracks A, B, or C , Ext 1 or 2).

SOLO SELECT

Housed in the same section as the monitor select switches are the Solo select buttons. These determine the solo mode for the console, which may be either PFL, AFL or Solo-in-place.

When Solo-in-place is enabled its associated LED will be on to give a visual warning. This flashes when SIP is actually happening. A flashing LED is also provided to give a visual indication of PFL/AFL select.

MAIN FADERS

The Main stereo fader, located on the left, is a long throw linear Penny and Giles fader with 0dB at the top of its travel.

A separate mono output is provided via another Penny and Giles long throw fader, located on the right of the Command Module, this also has 0dB at the top of its travel.

METERING

Metering for each module is provided by a 30 segment LED bargraph display in the overbridge. The characteristic may be either Peak or VU and is determined by a master switch located in the overbridge.

The meter above each module always relates to the Mix path signal. When the Mix path on the module is switched to Group then the meter will read the Group Output, when the Mix path is switched to Tape the the meter will read Tape Return.

If Tape and Group has been selected the meter will monitor the Group Output, even though you are listening to both tape and group.

Bargraph meters are also provided for the six auxiliary sends.

The main Mix outputs are provided with wide scale VU meters for both left and right outputs. The Phase correlation meter displays the phase relationship between the left and right outputs giving the operator a visual indication that an in-phase signal is present at the main mix output. By selecting tape returns on the Control Room Monitor switches the operator can also check that no phase errors have been introduced between the console and the tape machine.

In-phase signals are indicated by the illuminated segment moving to the extreme right-hand side of the display, for out-of-phase signals the segment will move to the extreme left. For signals which do not have a clearly defined phase relationship the segment could move to an indeterminant position. With no signal present the segment should be positioned at the centre of the display.

4.03 OPTIONAL STEREO SUB-GROUP

CHANNEL

The Channel section with its rotary fader is usually the section that feeds any of the stereo line level signals to the routing matrix.

The Channel input section is located at the top of the Module with its associated Gain controls for Line inputs A and B.

With all the buttons up on the Module the signal from Line Input A is routed in stereo to both the Channel and the Mix paths, the same as the mono Input/Output Module, with the exception of the Pan Pots that now become Balance controls.

Selecting ALT allows Line Input B to be sent instead of Line Input A.

Pressing the SPLIT button, located near the Channel fader allows Input A to be routed, in stereo, to the Mix path and Input B to be routed, in stereo to the Channel path.

When SPLIT and ALT are both selected then Line Input B is now routed, in stereo to the Mix path and Line Input A is routed, in stereo, to the Channel path.

When the SUB button is pressed signals from any pair of Groups 1-16, are sent, in stereo, to both the routing matrix AND the Mix bus. If used in conjunction with the SPLIT button then this allows Line Input A to be routed to the matrix AND the signal from the Group is routed to the Mix bus.

When ALT is used in conjunction with SPLIT and SUB the signal flow is not changed, Input A still feeds the routing matrix AND the Group feeds the Mix path.

See Table 1

TABLE 1

SWITCH			SIGNAL PATH	
Alt	Split	Sub	Channel	Mix
0	0	0	A	A
1	0	0	B	B
0	1	0	B	A
1	1	0	A	B
0	0	1	G	G
0	1	1	A	G
1	1	1	A	G
1	0	1	A	G

NB: 1 = Switch pressed
 0 = Switch NOT pressed

A = Input A
 B = Input B
 G = Input from Group

The L and R switches allow the Module to be used in one of three modes.

- i With both switches out then the module operates in the normal stereo mode.
- ii With either L or R selected both paths of the module are fed by either the Left input or the Right input.
- iii With both switches selected, both paths of the module are fed by a mono sum of the left and right inputs.

The button marked ST1 cuts the signal from the main stereo bus only, the number corresponds to the Module number.

It should be noted that all Auxiliary sends are the mono sum of the Left and Right signals.

The balance controls correct or deliberately create any errors in stereo balance.

The rest of the Module functions as a normal Input/Output module except all signal paths are now in stereo, (except the AUX sends, which are mono).

4.04 OPTIONAL EFFECTS RETURN MODULE

The optional Effects Return module further enhances the versatility of the Soundcraft TS24 in-line console. The Effects Return module provides 4 additional inputs to the console which feed the main mix bus, the signal may also be sent to any of six auxiliaries, in three pairs.

Input to the module is via the associated patchcard. By using molex connectors, located at the bottom of the FX return patchcard, equipment which is often used can be permanently patched into the console. Alternatively, input to the module can be via the associated input jack socket on the patchcard. The FX return has balanced inputs.

The return signal from 4 separate peripheral devices may be processed and routed to up to 6 auxiliary sends and/or the main mix bus. Eg. Signal from an I/O module is routed to an effects device, this is then required to be returned to the console without having to use another I/O module. The return signal from the effects device can be routed back into the console via the FX Return where the signal may be further processed.

From here the signal may be routed to the main mix bus and hence to a tape machine, also the signal can be routed to any of 6 auxiliaries.

5.00 MAINTENANCE

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

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

5.01 General Fault Finding

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 channel input, in both microphone and D/I 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 the Mix bus by selecting MIX on the routing section and ensuring that the console status is MIX. Using the internal oscillator set it to 1kHz and patch the oscillator signal into the channel input at the patchbay, NB: Ensure that the +48V phantom power is off. If all is well, an undistorted signal should now be heard when the fader is pushed up. More likely, because of the fault it won't.

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

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

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