TRIAD

# Fleximix System
# Owners Manual

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# INTRODUCTION

The Trident Audio Developments Fleximix system has been designed to fulfill the needs of people requiring a console that would be equally at home from the point of view of both technical performance and facilities in either a live concert environment or a multi-track recording studio.

Because the Fleximix system has achieved this objective, applications become possible where before it would have been necessary to use a different mixer for each specific purpose. An artist can for example, use the Fleximix for live stage mixing whilst making a simoultaneous multi-track recording. The same console can then be used later on to perform a sophisticated master mix-down quite comparable with any obtainable in a major recording studio.

By simple interconnection of mainframes, input and output configurations can be expanded or contracted in a very short period of time. This means that for live stage mixing or mobile location recordings, input or output modules surplus to requirements are not carried around unnecessarily.

There are two standard mainframe sizes in the Fleximix system, one capable of housing up to 15 modules of any type and the other capable of housing up to 8 modules of any type. Each of these mainframes contain a fully regulated and short circuit proof power supply unit. In the event of a power supply failure, the mainframe containing the faulty supply can be 'slaved' from an adjacent mainframe in a very quick and simple manner. In the case of smaller systems where only one mainframe is used, screw connectors are provided at the top of the unit to facilitate the connection of either an external supply or batteries.

For portable applications where the console is to be used as a mobile unit, the Fleximix system readily adapts to a flight-case because of it's compact nature. By using a multi-way connector cable it is even possible to electricaly join mainframes together whilst they are housed in flight cases.

Because of the highly versatile nature of the system, new types of modules can be added to the system at any time during it's lifespan thereby ensuring that the console never becomes obsolete. Limiter-compressors, electronic cross-overs and graphic equalisers are just a few of the modules that the owner will be able to add to the system at any convenient time.

Servicing and maintenance of a console as sophisticated as the Fleximix system has been in the past a major problem for non-technical operators such as musicians. This has been overcome by providing simple step by step fault finding procedures and by mounting all the important signal carrying circuitry on plug-in holders which are in turn soldered to the module printed circuit board. By doing this, most normaly encountered electrical faults can be rectified extremely quickly and without using a soldering iron.

Constant use of the Fleximix system will provide the operator with an increasing awareness and understanding of the creative possibilities that the console can provide.

## A1 Voltage adjustments:

Each Fleximix mainframe contains a dual voltage power supply to provide 45 volts D.C. for the signal electronics and 10 volts D.C. for the L.E.D. and relay circuits. The mains transformer for this supply has a dual voltage primary winding that can be switched to operate on mains voltages of either 110 or 220 volts A.C 50/60Hz.

In order to change operation of the console to suit either voltage, all that is necessary is to remove the cover plate at the top right hand side of each mainframe (next to the power inlet socket) and slide the switch over to the required voltage, taking care to replace the cover after adjustment. It is important to note that the fuses specified on the top panel are for 220 volt operation. Therefore when connecting the unit to a 110 volt supply it is essential to double the fuse ratings. Fig.13 section F (Mainframe and Power Supply) shows the voltage selector panel with the cover removed.

## A2 System interface:

Because of to-day's low impedance techniques, the Fleximix system is unbalanced except for the microphone inputs. Whilst an unbalanced system has a great many advantages such as low phase shift, greater headroom at low frequencies etc., care must be taken to avoid hum loops by making sure that all the equipment earths are connected together correctly.

The best way to achieve this is to make the mixing console the central piece of equipment from an earthing point of view and return all other equipment earths to a 'technical earth' at the console. In order to do this, the procedure is as follows:

Connect the Fleximix power lead(s) earth to the mains earth but disconnect the mains earth from all other equipment that will be connected to the console. Re-connect a piece of heavy duty cable (preferably copper) to the chassis of each piece of equipment and connect these to the Fleximix earth point.

On signal leads to and from equipment connected to the console, such as tape recorders etc., disconnect the earth or screen of the cable at one end so that the input lead to external equipment has an earth at the console end only and the output lead from external equipment has an earth connection at the equipment end only.

By using the above methods all equipment remains safely earthed and the probability of hum and clicks etc., will have been greatly lessened.

## B1 General description:

The Fleximix input module is designed to accept the signal from either a low impedance 200ohm condenser or dynamic microphone and a high level source of up to 15kohms impedance. To facilitate the use of condenser microphones which require 'phantom powering' a 45 volt supply is made available at the microphone input X-L-R connector. By using condenser microphones of this type, they can be connected to the console in exactly the same manner as a conventional dynamic microphone, thereby eliminating bulky power supplies.

The amplification of the microphone amplifier can be continually adjusted from 0 to 60db making it possible to accomodate the wide range of signal levels available from all types of condenser and dynamic microphones.

A toggle switch allows instant selection of either the microphone or line input and since the two sources have separate input connectors (microphone: 3 pin X-L-R, line: standard ¼" jack), this facility can be used for a variety of purposes. For theatre or live concert work where it is often necessary to playback sound effect tapes for short durations, an input module amplifying a microphone that is not used at the time the effect is required can be switched over to the line position for the time necessary for the effect and then instantly switched back to the microphone. Because the microphone level control does not affect the line mode, no re-calibration of controls is necessary.

After the microphone and line signal processing the signal passes through an 'equaliser' circuit which can be used to modify the tonal quality by either amplifying or attenuating chosen portions of the frequency spectrum. In the Fleximix input module these portions are split into the low (bass), mid (presence) and high (treble) frequencies. The low range can be selected by a switch to either 60 or 150hz, the mid range can be continuously varied from 300hz to 10khz and the high range can be selected by a switch to either 8 or 12khz. Both the low and high ranges have 'shelving' characteristics whilst the mid range is of the 'peaking' type. The amount of amplification of the chosen frequencies can be varied continuously from 15db of boost to 15db of cut. The entire equaliser section can be switched in or out of circuit by means of a toggle switch. This has the effect when switched to the 'out' position of electricaly placing all boost and cut controls of the equaliser to their midway position. This is very useful when an instant comparison is required between the original and 'equalised' sound.

After the equaliser section the signal appears at a jack socket marked 'channel send' so that at this point in the programme chain, the signal can be connected if required to the input of an external processing device such as a limiter-compressor, phaser etc. The output of a device such as this is connected to another jack on the input module marked 'channel return'. For ease of connection to external devices, these two jacks are located next to each other. If no external piece of equipment is connected to the signal chain, programme continuity is maintained by the use of 'back contact' connections on the jack sockets.

From the 'channel return' jack, the signal passes through a "buffer" amplifier which provides a constant impedance to the input module slide fader and auxiliary send circuits 1 and 2. These auxiliary send circuits are derived before the fader and each has it's own level control. They are taken from a point before the slide fader so that the signal in these circuits will be unaffected by any level changes made to the fader. This means that auxiliary sends 1 and 2 would be idealy suited for use as 'foldback' or headphone feeds to musicians in the studio or on stage.

At exactly the same point as the auxiliary circuits are taken from, another independent circuit called 'cue' or 'pre-fade listen' is provided. This system is activated by a toggle switch and is indicated by a lamp (L.E.D) when in operation. The 'cue' system provides an output from the module for quality checking purposes and has the ability to 'solo' the module selected on the monitoring system. The 'cue' circuitry allows for the signal from a number of modules to be mixed together so that a group of instruments can be 'cued' at the same time. For this reason the 'cue' switches are of the locking type for ease of operation. The 'cue' system can be used for a variety of purposes such as instantly checking the equalisation of one instrument amongst a group of others, isolating a distorted microphone or instrument for example and checking microphones before a recording without adjusting the fader level. Because the 'cue' system only affects the monitoring system of the console, it is quite in order to use the facility whilst recording. In live concert situations where the control room monitoring facility is not required, a pair of 600ohm headphones can be connected to the monitor output jacks (situated on the Auxiliary module) to make use of the system.

After the signal has passed through the input module fader which can be either a high quality carbon or conductive plastic type according to customer choice, the signal passes through a line amplifier which gives the signal an additional amplification of 5db. This gain is factory set by a resistor on the module printed circuit board (R43) but can be raised to a maximum of 15db or reduced to zero gain if required.

From the line amplifier the signal passes through a 'channel on/off' toggle switch, which when activated completely mutes the output of that particular input module. Because operation of this switch means a total loss of input module signal, an L.E.D indicator lamp is provided to show when the module is switched off. Thereby warning the operator that no signal is present.

Following the 'channel on/off' switch, the signal is fed to a jack socket designated 'direct output' so that programme can be taken from each module output before any main signal routing or panning occurs. When the direct output jack is in use, continuity through the rest of the module is broken isolating the programme from all routing and panning systems. The direct output facility can be particularly useful in multi-track recording when the number of output groups available from the console does not equal the number of tracks available on the tape-recorder. If instruments using only one microphone are required on separate tracks, the direct output facility can be utilised and those instruments requiring a number of microphones can be routed through the normal panning and grouping systems.

In situations where it may be necessary to record multi-track at the same time as providing a fully mixed stereo composite signal, such as live concert performances, the direct output system can be very easily modified so that it can be used as a parallel of the input module output without disconnection of the normal routing and panning.

At the same time as feeding the 'direct output' jack, the signal is routed to a V.U meter so that a visual indication is given of the signal level from each input module. As the meter is situated electronicaly at the module output, adjustment of the microphone level control fader and equaliser settings will all register on the meter. This means that the meter can be used to accurately optimise the level, distortion and noise from each module.

Since the V.U meter is taken at a point after the 'channel on/off' switch, when the module is muted, the meter will not indicate signal. This is so that the operator will not be mislead into thinking that the module is operative if for example the 'channel on/off' switch is activated but the L.E.D has failed for some reason. In some applications however it may be preferable to forego this safety factor for the advantage of being able to set up levels using the meter only whilst the channel is switched off. A typical example could be live concerts where it may be necessary to switch a microphone into circuit very quickly due to a cable failure for example. Under these circumstances it is essential that when the new microphone is switched into circuit the level is correct. It is therefore very easy to modify the metering if required to indicate before the on/off switch. This modification can either be specified when the console is ordered or can be carried out by the customer in accordance with the instructions given later on in this section of the handbook.

As well as feeding the V.U meter and 'direct output' jack, the signal is also routed to two rotary level controls designated 'auxiliary sends 3 and 4'. These post fader sends can be used for sending a portion of the signal to an echo device etc. and because they are derived after the fader the echo signal will diminish as the fader is attenuated thereby giving the operator control of the main and echo signal on a single element.

If the operator requires a different combination of auxiliary sends i.e three pre fade sends (foldback) and one post fade (echo), this again can be accomplished by simple modification either at the factory or again by reference to the 'Customer modifications' section given later on.

After the 'direct output' jack socket the signal passes through a panoramic potentiometer (pan pot) and on to the module routing push-buttons. The pan pot has been designed to maintain the maximum possible mono stereo compatibility so that when a stereo signal is reduced to mono such as in radio transmission, the change in balance or perspective between the instruments is as minimal as possible.

The ten routing push-buttons fed by the pan pot are split into two rows of five odd and even numbers. The odd numbered buttons (green indication) are fed from the left hand section of the pan pot and the even numbered buttons (orange indication) are fed from the right hand section of the pan pot. Any number or combination of push-buttons can be used simoultaneously but when more than one combination of odd and even numbered groups is depressed, loading effects will come into play on the pan pot and will effectively reduce the level being sent to each output group.

Figure 1 shows the input module signal flow in schematic form and reference to this will help to give a better understanding of the way in which the signal is routed from the microphone input to the module output.

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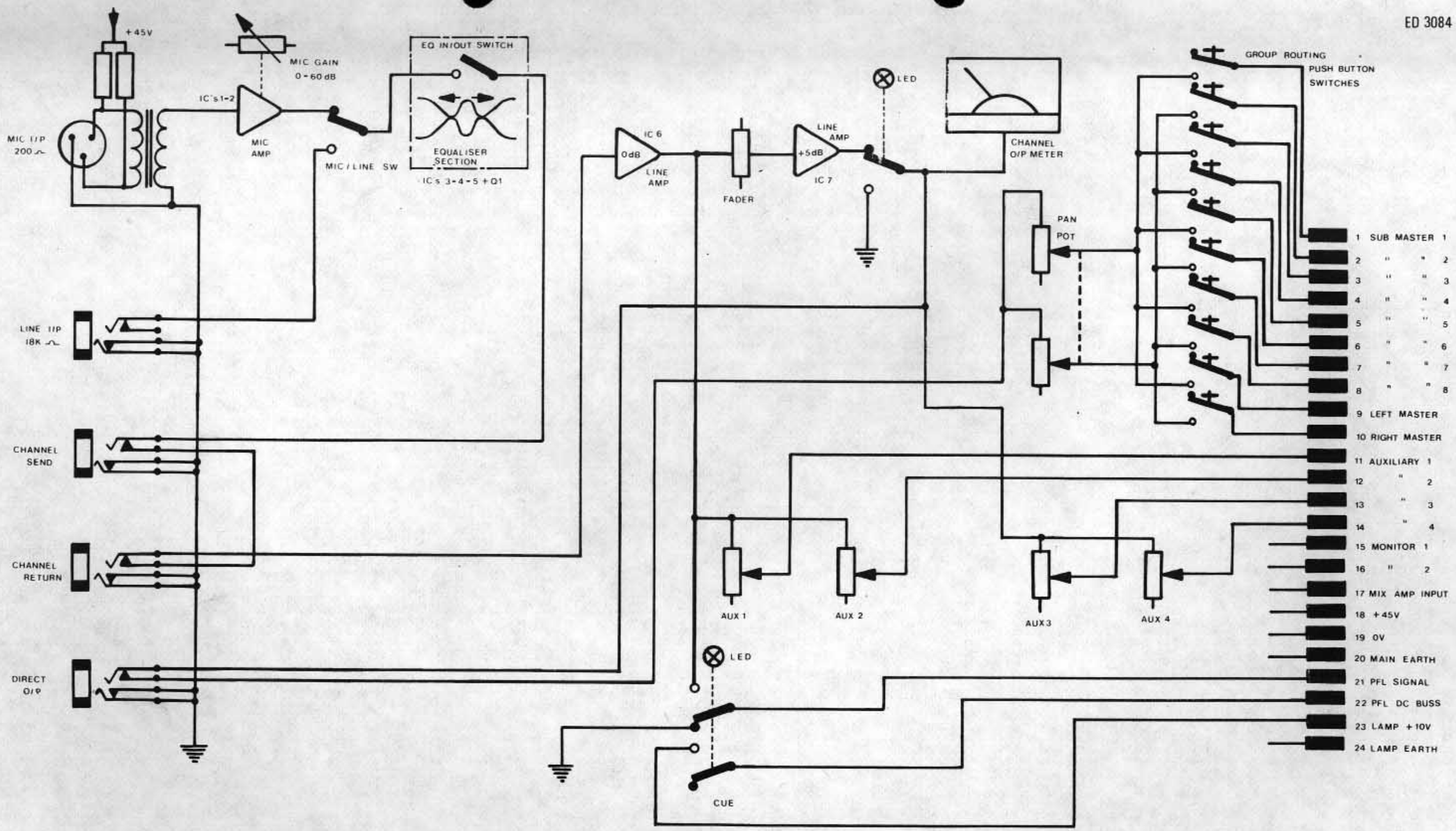



TRIAD **Fleximix System** INPUT MODULE SCHEMATIC CCT DIAGRAM

FIG 1

## B2 Operational procedure:

With reference to Fig.2 which shows the input module front panel and controls, connect the output from either a balanced microphone of 200 to 600 ohms impedance to the female X-L-R socket at the top of the module or the line output of a tape recorder to the line input jack socket also situated at the top of the module. The microphone connections should be wired so that pin 1 of the X-L-R is earth or screen, pin 2 is negative phase (blue wire) and pin 3 is positive phase (red or white wire). For unbalanced line inputs the earth or screen should be connected to the outer terminal of the jack plug (usually incorporating a cable clamp) and the signal lead should be connected to the inner terminal or tip of the plug. In the case of a balanced signal being connected to the line input it is essential that the negative phase signal (blue wire) is connected to earth otherwise there will either be no signal or a 'thin' sound.

Because the input module provides separate microphone and line inputs, it is quite in order to plug both sources into the module at the same time so that the 'mic/line' switch selects either input instantly.

Having therefore selected the input source by means of the 'mic/line' selector switch, the next step is to adjust the module amplification to obtain a suitable level indication on the module V.U meter. Before adjusting either the microphone level control or the slide fader, the equalisation circuit (black knobs) should either be switched out of circuit by means of the equaliser 'in/out' toggle switch or the equaliser level controls (black knobs on left hand side) should be adjusted to their mid point of travel. After checking this it is also important to make sure that the 'channel on/off' switch is selected to the 'on' position. For microphone level setting, make sure that the microphone level control is at minimum anti-clockwise travel ('0' position). With the slide fader at the top of it's travel ('0' position) the microphone level control should be advanced until the V.U meter deflects approximately threequarters of it's travel or just into the red area on peaks. Many operators prefer to have extra gain available on the fader so that if necessary the signal can be raised during quiet passages etc. In this case the microphone level control should be advanced clockwise until the required extra gain is obtained. Care should be exercised not to bring the fader too far from the top as this will make the microphone amplifier work harder until eventually distortion will occur. Operation of the fader a around the -10 position will ensure enough extra gain for most occurancies whilst still maintaining a good overload margin.

If equalisation is to be used, the microphone input control should be attenuated proportionately with any boost introduced in the equaliser circuit. Similarly if the equaliser is used to attenuate frequencies, the microphone level control may have to be advanced to obtain a suitable reading on the module V.U meter.

The setting up of input levels can be accomplished either inaudibly using the method just described or audibly by using the consoles monitoring system. By far the quickest and easiest method is to use the 'cue' system as this does not require any main signal panning or routing. To use this system an amplifier and speaker (or headphones) will have to be connected to the monitor outputs situated on the 'Auxiliary module'. As explained previously the 'cue' system is derived before the module fader and therefore adjustment of the fader will have no effect on the monitored signal level. However by using this system in conjunction with the V.U meter, the sound quality can be monitored and the level set by the meter.

Since the 'cue' system is taken at a point after the equaliser section, any equalisation introduced can be monitored when using the 'cue' system as a method of adjusting amplification levels through the module.

If it is required to introduce a further signal processing device such as a limiter-compressor or graphic equaliser etc. into the programme chain, this can be accomplished by connecting the input of the device to the jack socket marked 'channel send' and the output of the device to the jack socket marked 'channel return'. Both of these jacks are located at the top of the input module and are designed to accept unbalanced signals. This means that as with line inputs previously described, a balanced signal must be made unbalanced by connecting the negative signal phase to earth before connection to the system.

If the external processing device contains level controls, these should be adjusted so that when the device is connected the V.U meter on the input module reads approximately the same level as before the external device was connected.

If the auxiliary sends are to be used either for 'foldback' or echo signals, the procedure is as follows: for 'foldback' connect an amplifier and speaker or 600 ohm headphones to the jack sockets marked 'aux. 1' or 'aux 2' on the Auxiliary module front panel. The 'auxiliary master' level controls (also situated on the Auxiliary module front panel) should be turned fully clockwise and the appropriate auxiliary send level control on the input module should be advanced clockwise until the desired level is achieved in the speaker or headphones. When using a speaker for 'foldback' purposes great care should be taken not to allow 'feedback' to occur. This is when the signal from the speaker is picked up by a microphone, amplified and sent back through the speaker again in a continuous cycle.

For echo, the same procedure is followed except that the input to an echo device should be connected to the jack sockets marked 'aux. 3' or 'aux. 4' on the Auxiliary module front panel. The output from the echo device should be connected to either an input module 'line input' jack so that the signal can be routed, panned and equalised if necessary or to the 'line input' jack of a Sub-master module or Left/Right master module if it is desired to hear the echo effect on the monitoring system only. Returning the output of an echo device for monitor purposes only will be dealt with more fully in the sections describing the Sub-master and Left/Right master module.

## B3 Input module fault finding procedure:

Since nearly all of the signal processing operational amplifiers are connected by plug in sockets to the module printed circuit board, fault finding and rectification can be accomplished without test equipment provided that logical steps are taken to locate the faulty part of the circuit. To help locate the various operational amplifiers used in the input module Fig.1 gives their function and printed circuit board locations for easy identification. Listed below are the possible faults and their rectification that can occur during the modules life. If by following the procedures the fault cannot be rectified, the module should be returned either to an authorised dealer or to the factory so that a more thorough examination can be carried out.

No microphone signal:

Check whether the module is operational in the 'line' mode by connecting either the output from a tape recorder or the console oscillator (situated in the Auxiliary module) to the 'line input' jack socket. If the module functions normally in the 'line' mode, change either or both of the 1741SC integrated circuits mounted on the small printed circuit board behind the microphone level control. If this fails to cure the problem, check that the connections to the microphone input X-L-R socket are not broken and that the screened lead connecting the microphone transformer to the small printed circuit board is also intact. Check also that the microphone level control is not malfunctioning due to dirt etc.

No module output from either microphone or line:

Check first of all whether there is an output from the 'channel send' jackpoint by connecting a programme source or oscillator to the 'line input' socket and an amplifier or other monitoring device, to the 'channel send' socket. If signal appears at this point, change either or both the fader buffer I.C6 and the output amplifier I.C7. If there is no signal at the 'channel send' jack, change sequentially all of the equaliser integrated circuits I.C's 3,4 and 5. The discrete transistor Q1 is an input buffer to the parametric (sweep) mid-range equaliser circuitry and can be changed provided that an equivalent replacement is available (BC109C etc.) If these measures do not restore the signal, check for a broken wire on the fader, channel on/off switch etc.

## B4 Customer modifications:

Direct output not disconnecting signal path:

Connect a wire link accross the back (switch) contacts of the 'direct output' jack terminals on the track side of the printed circuit board. The back contacts of the socket are the rear terminals, furthest from the front panel.

Metering before the mute switch:

Disconnect the yellow wire from both the meter and printed circuit board. Re-solder a wire from the meter to the blue wire connected to the 'on/off' switch.

Adjusting output line amplifier gain:

By changing the value of R43 the post fader line amplifier gain can be altered. The 1K resistor already fitted gives a gain of 5db. Increasing this value will increase the gain, i.e 3K = 10db. Similarly by reducing the value of R43 the gain can be decreased. For unity gain i.e 0db a wire link should be inserted in place of the resistor.

Changing auxiliary sends 1 and 2 to post fader operation:

Disconnect the violet and grey wires connected to auxiliary level controls 1 and 2 where they are soldered to the printed circuit board and re-solder to the orange and red wires attached to auxiliary level controls 3 and 4.

Changing auxiliary sends 3 and 4 to pre-fader operation:

Disconnect the orange and red wires connected to auxiliary level controls 3 and 4 where they are soldered to the printed circuit board and re-solder to the violet and grey wires attached to auxiliary level controls 1 and 2.

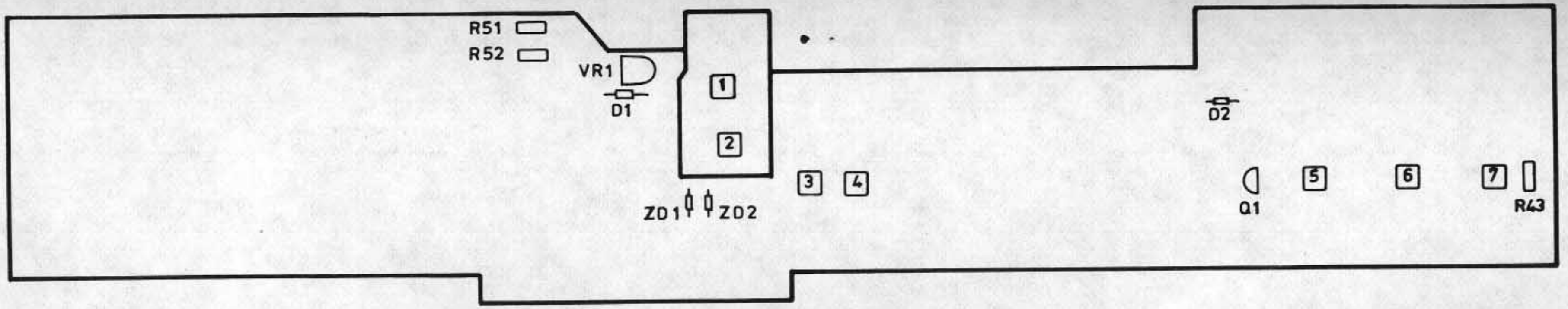
Adjusting pan-pot centre loss:

The centre loss of the pan-pot is set for 3db by means of 'slugging' resistors mounted on the back of each section of the dual gang pan control. To achieve a 6db centre loss the 3K9 resistors fitted should be replaced by two 2K2 resistors.

Changing faders from carbon to conductive plastic:

Unscrew the aluminium mounting block which secures the printed circuit board to the front panel assembly and unscrew the fader bracket at the other end of the fader. It will now be possible to remove the carbon fader with the brackets attached. The conductive plastic fader will now re-locate using the same holes previously used to mount the carbon fader brackets. It is a good idea to leave the carbon fader attached to the mounting bracket so that it can be used again at any time.

Figure 3 shows the component side of the input module with the pan-pot centre loss resistors and the location of R43 for gain adjustments outlined.




1 MC 1741S CP1 MIC AMP 1ST STAGE

2 " " " " " 2ND "

3 " " " INPUT BUFFER

4 " " " EQ SHELF SECTION

5 " " " EQ PARAMETRIC SECTION

6 " " " CHANNEL RETURN BUFFER

7 " " " OUTPUT LINE AMP

D1 0A47 METER RECTIFIER

D2 1N4002 CUE LAMP SUPPLY BLOCKING DIODE

ZD1 16V 1W 5% ZENER DIODE PCB SUPPLY REGULATOR

ZD2 " " " " " " " "

Q1 BC 413C PARAMETRIC BUFFER TRANSISTOR

R43 OUTPUT LINE AMP GAIN RESISTOR

R51 PANPOT CENTRE LOSS RESISTOR

R52 " " " "

VR1 METER SENSITIVITY ADJUST

# INPUT MODULE COMPONENT LOCATION

Fig 3

## C1 General description:

The sub-master module contains all the necessary mixing and monitoring systems that are required for use with multi-track tape machines etc. After the signal has been sent to a sub-master mixing bus by means of the input module routing push-buttons and pan-pot, the signal is combined with that of other input modules by means of resistors which isolate each signal so that the faders do not interact with each other. After combining the signals together these resistors attenuate the signal greatly so that an amplifier is required to restore the programme to normal operating levels. The amplifier used to achieve this in the Fleximix system is of the'virtual earth'type so that whether one input module or all input modules are routed to a sub-master output, the programme level remains very constant (within 0.5db).

After the signal has been restored to normal level by the mixing amplifier, i.e the same level that was present prior to the mixing resistors, the signal is connected to a jack socket marked 'pre-fade send'. This is so that an external signal processing device such as a limiter/compressor or graphic equaliser etc. can be used to affect the entire mixed group of instruments. The output of the device is connected to the 'pre-fade return' socket situated next to the'send'jack at the top of the module. As on the input module, when no external device is connected, switch contacts on the jack sockets ensure signal continuity. From the'return'jack the signal connects to the sub-master fader which can be of high quality carbon or conductive plastic. After the fader the programme passes through the sub-master output line amp which provides an extra 5db of amplification to the sub-master group. This means that any input routed to a sub-master group will be raised by this amount thereby increasing the overall gain of the system. After the output line amplifier the signal passes through an 'on/off' switch so that an entire sub group can be muted instantly. An L.E.D. indicator is provided so that it illuminates when the signal is muted. From the 'on/off' switch the signal is connected to a jack socket designated 'sub-group output' for connection to a tape recorder input etc.

At the same point that the signal feeds this socket, it is connected to a pan-pot so that the sub-master output signal can also be panned into either the left or right master output groups. This facility has many useful applications such as in live concert use where a number of inputs can be routed to a sub-master module and this in turn can be re-routed by means of the sub-master pan-pot to the Left/Right master module. This latter module would then be used to feed the main amplifier/speaker system. By using this method, a pre-determined group of microphones can also be treated by an external signal processing device and faded up or down together. By using the sub-master 'on/off' switch it is also possible to simoultaneously mute the entire sub group. It is from this method of being able to 'sub-group' the outputs that the module derives it's name.

It might not of course be desirable to pan the sub-master output permanently into the Left/Right master module and for this reason a switch is connected in the signal path before the pan-pot designated 'group' and 'external'. In the 'group' mode the panning system will function as previously described with the sub-group being panned into the Left/Right master module. In the 'external' mode the input to the pan-pot is disconnected from the sub-master output and is connected instead to a jack socket designated 'ext. pan I/P'. In this mode an external piece of equipment such as an echo unit, can have it's output connected to the 'ext. pan I/P' socket and in this manner it can be panned into the Left/Right master module. This in effect gives an 'echo return' facility so that if all input modules are being used, an echo device can be brought back into the system in this manner.

If no external source is connected to the 'ext. pan I/P' jack the switch will of course act as a means of disconnecting the sub-group output from the panning facility. Also, at the point where the sub-group output appears at the 'sub-group output' jack socket, the signal is fed to a control room speaker monitoring system. Firstly the signal is routed to a switch marked 'source-group/line'. In the 'group' position the sub-master output is fed to the monitoring system. In the 'line' position, the output from an external line level source such as a tape recorder output or echo device is connected via a jack socket to the monitoring circuitry. The jack used for this purpose is designated 'line' and connects to a buffer amplifier before routing into the monitor system. As in the connection of a line input to the input module, care should be taken to unbalance a balanced source correctly if it is to be connected into the system. When the monitor source switch is selected to the 'line' position an L.E.D. indicator will illuminate to show the operational mode.

By connecting each of the multi-track recorder outputs to the corresponding sub-master module line inputs and each of the multi-track recorder inputs to the corresponding sub-master module outputs, instant monitoring comparisons can be made between the 'live' and recorded programme. Also by connecting the 'sel-sync' outputs of the tape recorder (if separate) to the sub-master line inputs, over-dubbing of multi-track can be accomplished very easily. Tracks previously recorded can be selected on the monitor source switch to 'line' whilst those about to be recorded are selected to the 'group' mode thereby enabling the operator to monitor a mixture of live and recorded tracks. When playback of a newly recorded track is required, it is only necessary to switch the multi-track recorder to playback and select the sub-master module that was used to monitor the recording to 'line'. The system can be very easily changed back in a matter of seconds if the over-dub proves unsatisfactory.

After the monitor source switch the signal is metered on a 10 L.E.D. column display. This type of metering was chosen for the Fleximix system for a number of reasons. Firstly, owing to the narrow width of the modules used as a standard in the Fleximix system for compactness, an ordinary type of meter would not have been precise enough as that required to fulfill the important task of metering the sub-master output levels. A small meter of the type used on the input modules, whilst quite satisfactory as a general level indicator for the input signals would not be accurate enough for the output groups. Secondly, in low ambient lighting conditions such as are found in concert halls, the L.E.D. column offers a very easily read and visual indication of programme levels, even at some distance from the console. Thirdly, because green L.E.D. indicators are used up to the '0' V.U point and red L.E.D's are used above this level a number of outputs can be 'scanned' by the eye with much greater perception and without the parallax errors associated with pointer meters.

The ballistics of the column indicators conform to B.B.C Peak Programme Meter specification ED1477 having a very fast response to transients and a slow decay time. Access to the L.E.D meter input can be gained by connecting the output of any source it is desired to meter to the 'ext. mtr.' socket at the top of the sub-master module. When this facility is used, switch contacts on the socket remove the signal previously routed to the column indicator. This system can very easily be modified so that instead of gaining access to the circuit via the jack socket it becomes a parallel feed of whatever signal is feeding the column indicator thereby making it possible to connect a standard discrete V.U or other type of meter in tandem with it.

At the same point as the meter signal is taken from, the sub-master cue system is derived. By being at this point in the signal chain, the cue signal follows the monitoring mode selected by the monitor source switch i.e group or line. An L.E.D. indicator shows when the cue facility is operational.

From this point the signal is sent simoultaneously to auxiliary sends 1 and 2 by means of rotary potentiometers. Because these two level controls are derived at this point, they operate independently from the monitor level control and will also follow the monitor mode selector switch. This is particularly useful when the console is used in studio applications for multi-track recording and over-dubbing work. Since for example when recording a drum track, several microphones will be mixed and equalised there is very little point in creating a headphone mix from each of the appropriate input modules when a composite mix is available from the sub-master modules that are being used to send and monitor the feed to a multi-track recorder. It makes sense therefore to send the foldback feed from the auxiliary sends on the output groups, which also has another advantage not possible when sending from the input modules. Because the output group auxiliary sends follow the monitoring mode, when over-dubbing, there is no need to re-balance the foldback levels once the monitor source switches have been selected to 'line' and the tape-recorder to sel-sync playback. The musicians who were previously hearing themselves as they recorded will instantly receive a playback of their recording together with any new track they are about to record as a composite mix. Multi-track work is therefore much simplified and far less time consuming as far as the problem of headphone balancing is concerned.

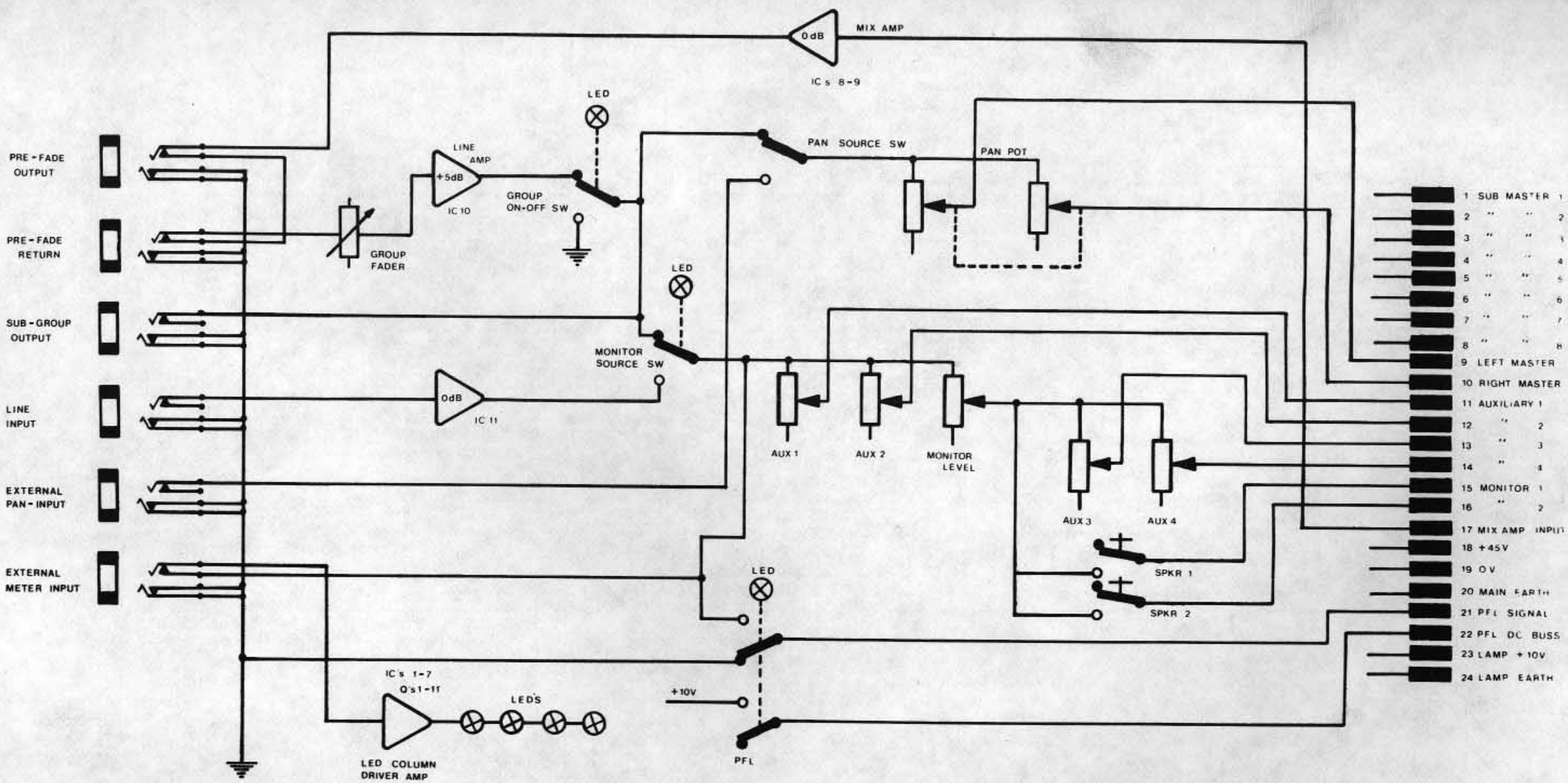
In parallel to the auxiliary send level controls 1 and 2 the signal passes through a monitor level control and after this feeds auxiliary send level controls 3 and 4. These auxiliary sends can be used for sending echo from a sub-master group and because they are derived after the monitor level control the echo will automatically be dependent on the monitor level control setting. Sending echo from mixed output groups can be advantageous if for example a 'blanket' of echo is required over a number of instruments such as violins. The echo can also be monitored only, depending on how the output of the echo device is routed back into the system. This means that the effect of the echo on instruments can be judged while the instrument itself is being recorded 'dry' or without any echo.

Since the four auxiliary send systems on the output sub-master modules follow the same path and mix together with the four auxiliary send systems on the input modules, a mixture of auxiliary sends can be derived from both types of module simoultaneously if required.

At the same time as the signal feeds auxiliary sends 3 and 4 it is also connected to the two'speaker select'push-buttons. These buttons allow the operator to choose between either or both of two control room speaker systems for stereo monitoring. Any adjustments made to the console monitoring controls will be completely independent of and will not affect the outputs to the tape-recorder or any other device that the console may feed. If the console is being used in a live concert situation and the output groups themselves are feeding an amplifier system, the operator can plug a pair of headphones into the monitor outputs of the console so that use can be made of the 'cue' system which is purely a monitor function.

Figure 4 shows the sub-master module in flow diagram form and reference to this will assist in further understanding the facilities available on the module.

ED 3087




TRIAD **Fleximix System** SUB MASTER MODULE SCHEMATIC CCT DIAGRAM FIG 4

## C2 Operational procedure:

With reference to Fig. 5 which shows the sub-master module front panel controls, the operation of the module for multi-track recording in a studio environment is as follows: Assuming that an input module has been routed to a sub-master group via the eight sub-master routing buttons and pan-pot, make sure that the 'group on/off' switch is selected to 'on' (no L.E.D. illumination) and that the monitor 'source' switch is selected to 'group' (no L.E.D. illumination). By pushing the slide fader towards the top of the module, the L.E.D. column indicator will illuminate in accordance with the programme level. When the desired output level has been obtained by reference to the column, (approximately -2 to +4 on the scale) the fader can be set at that point. If however in order to achieve the required output level the sub-master fader has to be set to a position lower than -10 on the fader scale, the input module faders should be attenuated equally until the sub-master fader can be set nearer the top. All input modules routed to the sub-master module will now be controlled simoultaneously by the sub-master fader. To monitor the programme on control room speakers in stereo, select which monitor speaker is required via the 'speaker select' push-buttons i.e button 1 left hand speaker, button 2 right hand speaker and both buttons if a central image is desired. Advance the 'monitor level' control clockwise until the desired amount of monitor signal from that particular instrument is obtained. If after turning the monitor level control clockwise no signal is heard, check that the master monitor level control situated on the Auxiliary module is advanced fully clockwise and that the 'monitor mute' switch is not activated (an L.E.D next to the mute switch will illuminate if this is the case.) Adjustment of the monitor level control will have no effect whatsoever on the signal feeding the tape-recorder etc. from the sub-master output jack.

To send a headphone feed via auxiliary systems 1 or 2 from the sub-master modules, the approriate send level control for each of the two systems should be advanced clockwise until the required balance is achieved in the musicians headphones. For echo, auxiliary sends 3 and 4 should be used in the same manner as auxiliary systems 1 and 2 from an operational standpoint but instead of being connected to an amplifier and headphone system as in the case of foldback, the output from either send 3 or 4 should be connected to the input of an echo device. The output from the device should be connected either to an input module and routed like any other line level signal to an output group if it is required to record the signal or it can be connected to the 'line' input jack of an output module if echo on the monitor speakers only is required. In this latter mode, the output module source selector switch should be selected to 'line' and the echo signal can then be routed accross the monitor speakers via the monitor level control and speaker select push-buttons. When sending echo from a sub-master module so that it can be recorded, care must be taken not to send echo from the same group that the output of the echo device is being routed to as this will cause an oscillation to occur.

If it is required to route the output of a sub-master module into the Left/Right master module for sub-grouping purposes, the 'pan input' switch located above the pan-pot should be selected to 'group' and the pan-pot should then be adjusted so that the signal appears accross the appropriate Left/Right master output group. If sub-grouping is not required, the 'pan input' switch should be selected to 'ext'. thereby disconnecting the sub-master output from the pan-pot.

In order to limit or compress the sub-master output, the 'pre-fade output' and 'pre-fade return' jacks are used. The input to the device should be connected to the 'pre-fade output' jack and the output of the device should be connected to the 'pre-fade return' jack. These jackpoints operate before the sub-master slide fader so that the limit or compression ratio is unaffected by alteration of the fader.

For replay purposes, the output lines of the tape-recorder (replay) should be connected to the 'line input' jacks at the top of each sub-master module. By selecting the monitor source switch to 'line' it is possible to hear exactly what has been recorded at the same level as when it was recorded without making any adjustments. In this mode, the metering and auxiliary sends 1 to 4 will also be fed with the tape-recorder replay signal since they follow the mode of the monitor source switch. For over-dubbing in multi-track work, the tape-recorder output should be switched to the 'sync' position and by using the monitor source selector switch a combination of monitoring can be obtained between recorded (sync) tracks and group outputs (tracks about to be recorded). The foldback and echo will therefore be switched automatically with the change of monitor mode.

To 'quality check' or 'solo' one sub-master module either in the 'group' or 'line' monitor mode, activation of the 'cue' switch will accomplish this. When the 'cue' switch is operated (shown by an L.E.D. indicator) all other modules will effectively be muted on the monitor speakers only so that recording will continue uninterrupted. If any other modules are switched to 'solo' at the same time, a mixture of the signals will be heard. All 'cue' or 'solo' signals appear as a central image accross both speakers.

For use in live concert work, the sub-master or Left/Right master output groups will normally feed an amplifier and speaker system so that the control room monitor system will not be used. If however the operator wishes to listen to an individual instrument or output group, this can be accomplished by connecting a pair of 600 ohm headphones to the monitor output sockets situated on the Auxiliary module. By this method full advantage can be taken of the 'cue' system and the operator can also use the monitor system to obtain a headphone feed of the signal being sent from the output groups unaffected by any hall colouration that may be present. Auxiliary sends 1 and 2 can also be used in live concert work as stage monitor feeds for the musicians sent either from input modules or output groups.

## C3 Sub-master module fault finding procedure:

Like all other modules in the Fleximix system, the sub-master module contains all operational amplifier integrated circuits on plug-in sockets to facilitate ease of servicing. Fault finding therefore becomes a simple matter of tracing through the system logically to isolate the defective I.C and replacement without using a soldering iron.

No signal from sub-group output jack:

Check first of all whether there is any output from the pre-fade output jack. If there is, the sub-master output line amplifier is at fault which is I.C 10 and should therefore be replaced. If no signal appears at this point, it is likely to be the sub-master mix amp at fault which comprises I.C 8 and I.C 9. Either or both of these should therefore be replaced. If none of the above steps rectifies the fault, then check for a broken fader wire or faulty jack socket etc.

No line return from tape-recorder to to monitoring circuits:

Since there is only one buffer amplifier consisting of I.C 11 between the tape recorder and the monitoring system, this should be checked first of all and replaced if necessary. If this does not cure the problem check again that the tape-recorder output itself is not at fault and also check for a faulty jack socket or broken wire to the monitor level control etc.

No metering:

In order to achieve the L.E.D column indicator a fairly complex circuit has to be used consisting of a number of integrated circuits and discrete transistors. As a consequence it is not possible to to cover every type of fault that can occur in this handbook. The following faults are therefore given as being the most probable that could occur if a fault should develop; If two L.E.D's illuminate at the same time, it is possible that one of the dual op-amp driver integrated circuits is faulty (I.C's 3 to 7). By swapping the I.C that drives the two L.E.D's that are illuminating together with one further up the chain, the faulty device can be isolated and changed if necessary. Since each I.C drives two L.E.D's and is positioned behind the L.E.D column, the appropriate device can be found by counting in two's from either end of the column. If one L.E.D. fails to illuminate entirely, a faulty transistor could be the cause. These are Q1 to Q10 which are situated right behind their associated L.E.D. If the column fails to work at all it could be the transistor array I.C.1 or the current drive transistor Q11. Since the transistor array I.C.1 (CA 3046) is a plug in device this should be changed first and if this does not remedy the fault Q11 (40362) should be changed.

## C4 Customer modifications:

Adjusting group output gain:

By changing the value of R73 the sub-master output line amp gain can either be increased or decreased. The 1K resistor already fitted gives 5db of amplification. To increase the gain, the resistor value should be increased i.e 3K= 10db. Similarly, by reducing the value of R73 the gain can be reduced. For unity gain i.e 0db a wire link should be inserted in place of the resistor.

Changing auxiliary sends 1 and 2 to post fader operation:

Disconnect the red and orange wires connected to auxiliary level controls 1 and 2 where they are soldered to the printed circuit board and re-solder to the violet and red wires attached to auxiliary level controls 3 and 4.

Changing auxiliary sends 3 and 4 to pre-fader operation:

Disconnect the violet and red wires connected to auxiliary level controls 3 and 4 where they are soldered to the printed circuit board and re-solder to the red and orange wires attached to auxiliary level controls 1 and 2.

Adjusting pan-pot centre loss:

The centre loss of the pan-pot is set for 3db by means of 'slugging' resistors mounted on the back of each section of the dual gang pan control. To achieve a 6db centre loss the 3K9 resistors fitted should be replaced by two 2K2 resistors.

Changing faders from carbon to conductive plastic:

Unscrew the aluminium printed circuit mounting block from one end of the fader and the fader bracket with the fader attached, from the other end. It will now be possible to remove the carbon fader with both brackets attached. The conductive plastic fader will re-locate under the aluminium block at one end and using the same screws, will re-locate directly at the other end into the original tapped hole.

Meter re-calibration or alignment:

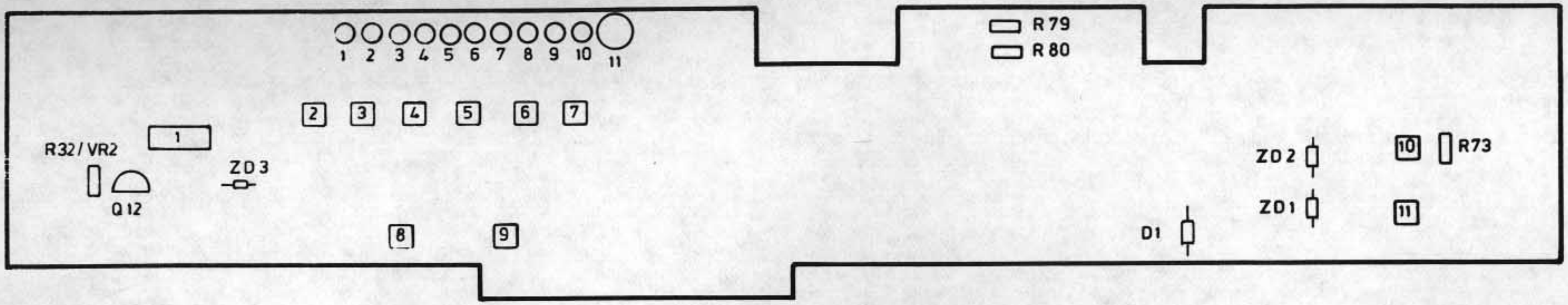
The column L.E.D indicator is factory calibrated so that the '0' scale reading equals +4dbm (0 V.U.) on a steady state tone. If it is rquired to re-calibrate the meter to a different standard of operation, the input sensitivity resistor R32 or VR2 (whichever is fitted) should be adjusted to suit the required operating level. The meter should then be re-aligned as follows:

1. Set VR1 to mid travel.
2. Apply a 1KHZ signal to the 'ext mtr.' socket so that '0' L.E.D (highest green) just illuminates.
3. Attenuate the signal by 20db.
4. Adjust VR1 so that the -20 L.E.D (lowest green) just illuminates.
5. If neccessary, repeat from 2 above.

Connecting 'ext. mtr.' socket as a parallel meter feed:

Connect a wire link accross the two rear terminals of the 'ext. mtr'. jack socket on the track side of the printed circuit board. The rear terminals are the two furthest from the front panel.

To assist in fault finding and customer modifications, Figure 6 shows the component side of the module with the neccessary resistors etc outlined.




| | | | | |
|---|---|---|---|---|
| 1 | IC CA3046 | P.P.M. DIFFERENTIAL AMP | Q12 TRN BC 413 C PPM PHASE SPLITTER | |
| 2 | " 741CP " | BUFFER AMP | ZD1 16V 1W 5% ZENER DIODE PCB SUPPLY REGULATOR | |
| 3 | " MC 1458CP1 | DUAL COMPARATOR | ZD2 " " " " " " " " | |
| 4 | " " | " " | ZD3 BZY 8V2 5% P.P.M. REF VOLTAGE GENERATOR | |
| 5 | " " | " " | D1 1N 4002 CUE SUPPLY BLOCKING DIODE | |
| 6 | " " | " " | R32 / VR2 LED COLUMN I/P SENSITIVITY ADJUST | |
| 7 | " " | " " | R73 OUTPUT LINE AMP GAIN RESISTOR | |
| 8 | " MC 1741'S CP1 | MIX AMP 2ND SECTION | R79 PAN POT CENTRE LOSS " | |
| 9 | " " | " " 1ST " | R 80 " " " " " | |
| 10 | " " | OUTPUT LINE AMP | | |
| 11 | " " | LINE INPUT " | | |

○ 1-10 TRN BC 107 OR BC167A LED DRIVERS

◯ 11 TRN 40362 CONSTANT CURRENT SOURCE

## SUB-MASTER MODULE COMPONENT LOCATION

Fig 6

## D1 General description:

The Left/Right master module carries out an almost identical electronics function to the sub-master module and differs mainly in the fact that since it is designed to accept the output from the sub-master modules for sub-grouping purposes, the module is not fitted with a pan control.

Owing to the fact that a double width module is used to accomodate both master output groups, standard V.U. metering is possible on large scale illuminated meters. This therefore makes it possible to make a comparison between the sub-master column peak programme L.E.D meters and the conventional V.U. ballistic meters fitted to the master output groups. By this method the advantages of both metering systems can be utilised to the fullest extent.

Since the sub-master and Left/Right master modules are identical apart from the small differences outlined above, it follows that the general description and operational procedure of the controls will be virtually the same. Reference to Figure 7 (flow diagram) and Figure 8 (front panel controls) should therefore be all that is needed to fully understand the modules function.

## D2 Left/Right master module fault finding procedure:

Again, fault finding procedures for the Left/Right master module follow very closely those for the sub-master module. The only difference is in the location of the various integrated circuits and their designation numbering. The Left master mixing amplifier comprises I.C's 4 and 5 whilst the Right master mixing amplifier comprises I.C's 2 and 3. The Left master line output amplifier is I.C 7 and the right master line output amplifier is I.C 8. The Left master line input buffer for tape replay monitoring is I.C 1 and the Right master line input buffer for tape replay monitoring is I.C 6. Figure 7 (flow diagram) also shows the I.C numbering of the various amplifiers which are contained in the Left/Right master module.

## D3 Customer modifications:

Adjusting group output gain:

By changing the value of R45 (Left master output) and R48 (Right master output) the Left or Right master output groups gain can either be increased or decreased. The 1K resistors already fitted give 5db of amplification. To increase the gain, the resistor values should be increased i.e 3K=10db. Similarly by reducing the values of R45 and R48 the gain can be reduced. For unity gain i.e 0db a wire link should be inserted in place of the resistors.

Changing auxiliary sends 1 and 2 to post fader operation:

Left master output: disconnect the red and white wires connected to auxiliary level controls 1 and 2 where they are soldered to the printed circuit board and re-solder to the red and white wires attached to auxiliary level controls 3 and 4.
Right master output: disconnect the orange and yellow wires connected to auxiliary level controls 1 and 2 where they are soldered to the printed circuit board and re-solder to the orange and yellow wires attached to auxiliary level controls 3 and 4.

Changing auxiliary sends 3 and 4 to pre-fader operation:

Left master output: disconnect the red and white wires connected to auxiliary

level controls 3 and 4 where they are soldered to the printed circuit board and re-solder to the red and white wires attached to auxiliary level controls 1 and 2.
Right master output: disconnect the orange and yellow wires connected to auxiliary level controls 3 and 4 where they are soldered to the printed circuit board and re-solder to the orange and red wires attached to auxiliary level controls 1 and 2.

Modifying mix amplifier inputs to connect to sub-master busses:

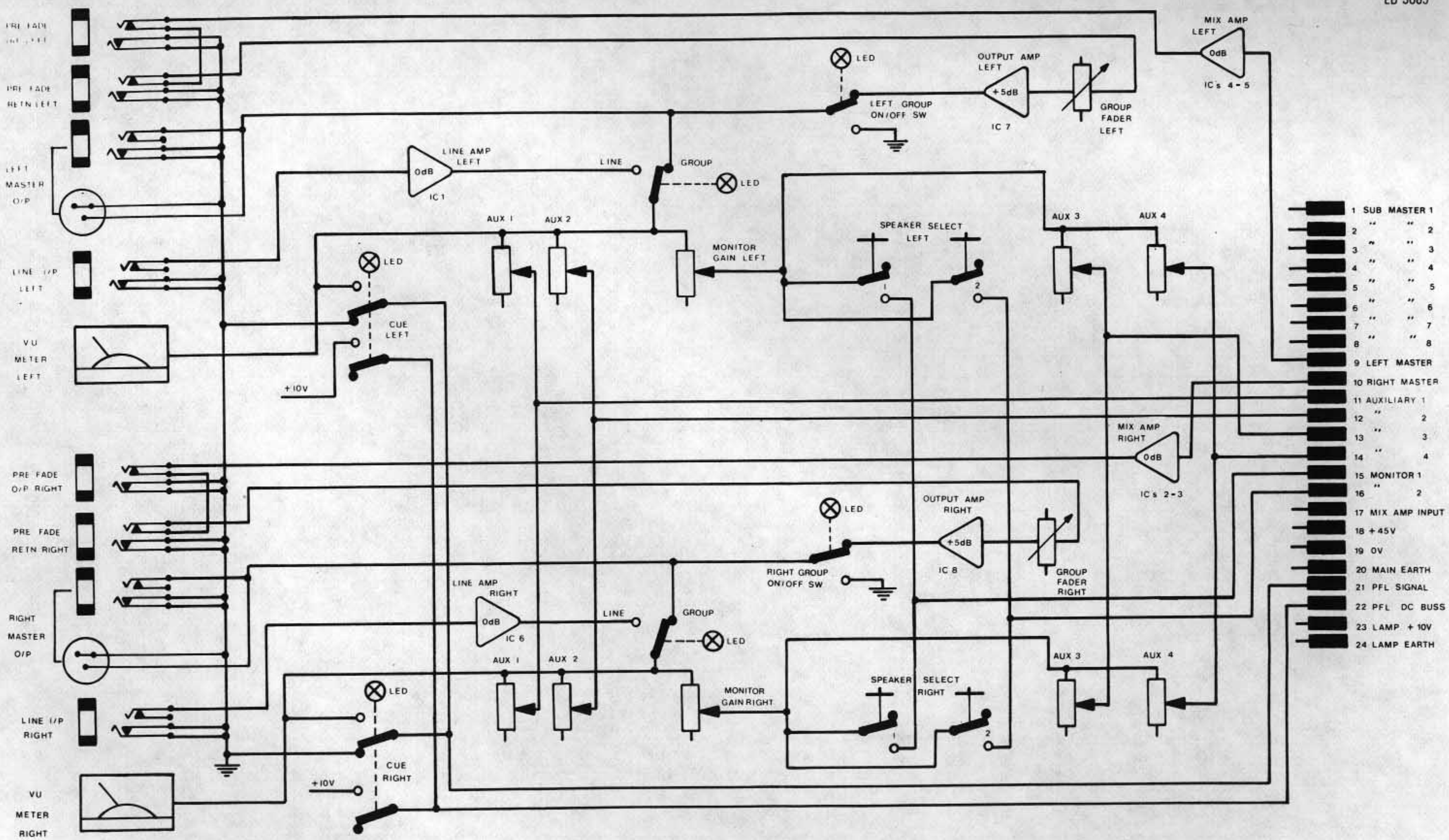
This modification is so that if required, Left/Right master modules can be used in place of sub-master modules. This does not allow sub-grouping but means that conventional V.U. metering can be utilised. Disconnect the two short wire links situated at the end of capacitors C16 (left mix amp input) and C17 (right mix amp input). Re-solder new wire links from the hole nearest the + end (positive) of each of the two capacitors into the appropriate hole (already provided) above the edge connector terminals for the sub-master busses.

Adjusting V.U. meter sensitivity:

The meter sensitivity can be changed from the standard calibration of'O' V.U (1.23 volts) by changing the value of R15 for the left master output group and R16 for the right master output group.

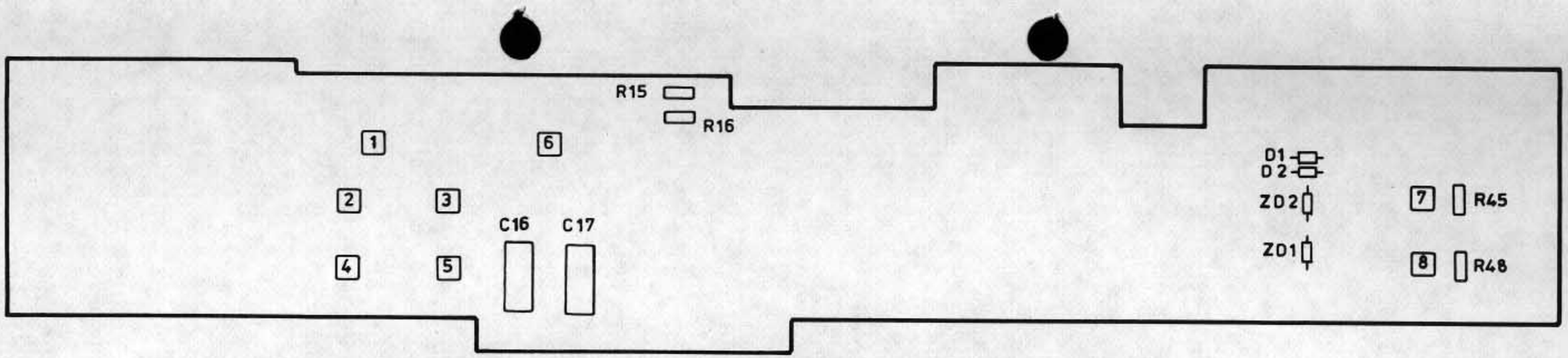
Figure 9 shows the component side of the Left/Right master module with the resistors outlined that are relevant to customer modifications.

ED 3085




TRIAD **Fleximix System** LEFT/RIGHT MASTER MODULE SCHEMATIC CCT DIAGRAM

FIG 7




1 IC MC1741S CP1 LEFT MON LINE INPUT AMP

2 " " " RIGHT GRP PRE FADE O/P MIX AMP 2ND STAGE

3 " " " " " " " " " " 1ST "

4 " " " LEFT " " " " " " 2ND "

5 " " " " " " " " " " 1ST "

6 " " " RIGHT MON LINE INPUT AMP

7 " " " LEFT GRP MASTER OUTPUT AMP

8 " " " RIGHT " " " "

D1 LEFT HAND CUE SUPPLY BLOCKING DIODE

D2 RIGHT " " " " "

ZD1 16V 1W 5% ZENER DIODE PCB SUPPLY REGULATOR

ZD2 " " " " " " " "

R15 LEFT VU METER 1 SENSITIVITY RESISTOR

R16 RIGHT VU METER 2 SENSITIVITY RESISTOR

R45 LEFT MASTER OUTPUT LINE AMP GAIN RESISTOR

R48 RIGHT " " " " " "

C16 LEFT MASTER MIX AMP INPUT CAPACITOR

C17 RIGHT " " " " "

## DUAL OUTPUT MODULE COMPONENT LOCATION

Fig 9

## E1 General description and operation:

As it's name implies, the function of the auxiliary module is to house and control all the facilities that are considered to be of an auxiliary nature to the main functions of the console. These 'auxiliary' functions include such necessary items as a talkback system, an alignment oscillator and master level controls for the studio monitoring, cue and auxiliary send systems. Because therefore the auxiliary module contains many seperate and not neccesarily related functions, each section will be dealt with individually starting at the top of the module and working downwards through the facilities. Figure 11 shows the location of all front panel controls.

Oscillator section:

The oscillator section provides switched sine wave outputs at the following six frequencies: 50HZ, 100HZ, 500HZ, 1KHZ, 5KHZ and 10KHZ. There is also a continuously variable level control allowing adjustment from infinity, (fully anti-clockwise) to +10dbm (fully clockwise). The oscillator also has an on/off switch and another switch to route the output either to a jack socket or accross all output groups. This latter facility is called 'slate' and is intended for use when it is necessary to send a tone from the console to align or calibrate an external piece of equipment such as a tape-recorder etc.

Auxiliary meter section:

The auxiliary module contains an illuminated V.U. meter similar to the one fitted to the input module, so that it can be switched accross various output lines as a visual level indication. The associated eight position switch allows the following signals to be read: auxiliary sends 1,2,3 and 4, monitor sends 1 and 2, cue and oscillator.

Auxiliary send 1 to 4 master level controls:

The auxiliary send master level controls are equivalent to master faders and will adjust the total output level from each of the four auxiliary send systems. This means that when for example auxiliary sends 1 and 2 are used for headphone feeds, the output level from the console can be accurately matched to the input sensitivity of the amplifier used to drive the headphones. Each master level control also has an associated switch marked 'cue'. This enables the operator to switch any individual or combination of auxiliary sends into the console cue system so that the balance or quality can be monitored on the studio control room speakers. This will mute all other signals feeding the monitor speakers (unless another 'cue' switch is depressed elswhere in the console) so that the appropriate auxiliary send(s) will be heard in isolation or 'solo'.

This facility is extremely useful when a musicians headphone balance is being created as it allows the operator to monitor exactly what mix is being fed into the headphones thereby eliminating possible confusion that can arise during initial headphone balancing. Since 'cueing' of the auxiliary sends does not affect the main signal in any way, the operator can 'cue' the appropriate auxiliary send whilst adjusting the headphone balance and can therefore make sure that it is exactly as the musicians require.

Cue master control:

This consists of a rotary level control (or fader) which will adjust the overall level at which the 'cue' signal is fed into the control room monitor system. This is a particularly useful facility when under normal monitoring conditions a complex mixture of signals are being fed to the control room speakers, as for example in the case of a multi-track remix. Under these conditions, if an individual input module is 'cued' it is very likely to appear on the monitor speakers in an attenuated condition in comparison to the rest of the signal. By using the cue master fader the level of the individual signal can be amplified from a monitor point of view so that it sounds equally as loud. Since the cue master fader is completely independent from and is not affected in any way by the master level control, the original control room speaker level will remain unaltered when the normal monitoring mode is resumed.

Monitor master controls:

A dual gang potentiometer is used to simoultaneously fade the two control room speaker feeds. Coupled with this control is a master monitor mute switch and L.E.D. indicator so that if required, the entire monitor speaker system can be muted instantly. The L.E.D will illuminate when the muting switch is activated so that clear visual indication is given that the control room monitor system is disabled. The L.E.D. will also illuminate if any 'cue' switch on the console is activated so that again a visual warning is given that the monitor system is not operating in a normal mode.

Talkback system:

An omni-directional high quality talkback microphone is mounted in the auxiliary module together with a level control and three talkback routing push-buttons. The level control allows the talkback microphone amplifier gain to be varied from 0db to a maximum of 60db continuously.

The talkback routing push-buttons allow the output of the talkback microphone to be routed to 1)the console Sub-master and Master Left/Right output groups (slate), 2)Auxiliary outputs 1-4, 3) a jack socket for connection to an amplifier and speaker for use as a studio talkback and communication system. When any of the three push-buttons are operated, the control room monitoring system is automatically attenuated so as to avoid the possibility of 'howl-round' occuring. The amount of attenuation for each of the three push-button functions is pre-set by fixed resistors mounted next to each push-button. For the 'talkback to auxiliary sends' and 'talkback to studio' functions the amount of attenuation is 20db whereas in the 'slate' mode the resistors have been omitted so that maximum attenuation takes place. This is because when the talkback output is routed accross all main output groups, the level accumulation which takes place could conceivably defeat an attenuation of 20db thereby creating 'howl-round'. If it becomes desirable to change any of the attenuation factors for the three push-buttons, full details are given in the 'customer modification' section for this module.

## E2 Auxiliary module fault finding procedure:

Like all other modules in the Fleximix system, the auxiliary module contains all operational amplifier integrated circuits on plug-in sockets to facilitate ease of servicing. Fault finding therefore becomes a simple matter of tracing through the system logically to isolate the defective I.C and replacement without the use of a soldering iron.

Oscillator:

The oscillator circuitry consists of I.C 6 and associated components. In the event of no oscillator output, I.C 6 should therefore be suspected first. If this fails to remedy the problem, the thermistor TH 1 should be checked and replaced if neccessary. A faulty or broken wire to the level potentiometer or frequency selector switch should also be looked for if neither of the preceding steps cures the fault.

Auxiliary meter:

If there is no level indication from any of the eight auxiliary meter switch positions, the most likely cause is a broken wire from the switch to the meter. Since there are no active elements involved in the auxiliary meter system, a faulty control or possibly meter is more than likely to be the cause.

Auxiliary sends:

Each auxiliary mixing system consists of three integrated circuits. Two of these comprise the virtual earth mixing amplifier whilst the third is an output line amplifier. Auxiliary system one consists of I.C 1 (output line amplifier) and I.C's 2 and 3 (mixing amplifier). Auxiliary system 2 consists of I.C 7 (output line amplifier) and I.C's 4 and 5 (mixing amplifier). Auxiliary system 3 consists of I.C 9 (output line amplifier) and I.C's 10 and 11 (mixing amplifier). Auxiliary system 4 consists of I.C 8 (output line amplifier) and I.C's 12 and 13 (mixing amplifier). To check a non working system an I.C can be 'borrowed' from a working auxiliary send and the three appropriate I.C's can be changed systematically to isolate and locate the faulty device. If replacement of I.C's does not remedy the problem, check for a faulty potentiometer or a broken wire etc.

Cue system:

The cue mixing amplifier consists of IC's 14 and 15. These should therefore be changed first of all if the cue system fails to work. There is no output line amplifier in the cue system since it utilises the monitor output line amplifiers when in operation. Relays RL 1 and RL 2 switch the normal monitoring path onto the cue system and should therefore also be checked if I.C replacement does not cure the fault.

Monitor speaker system:

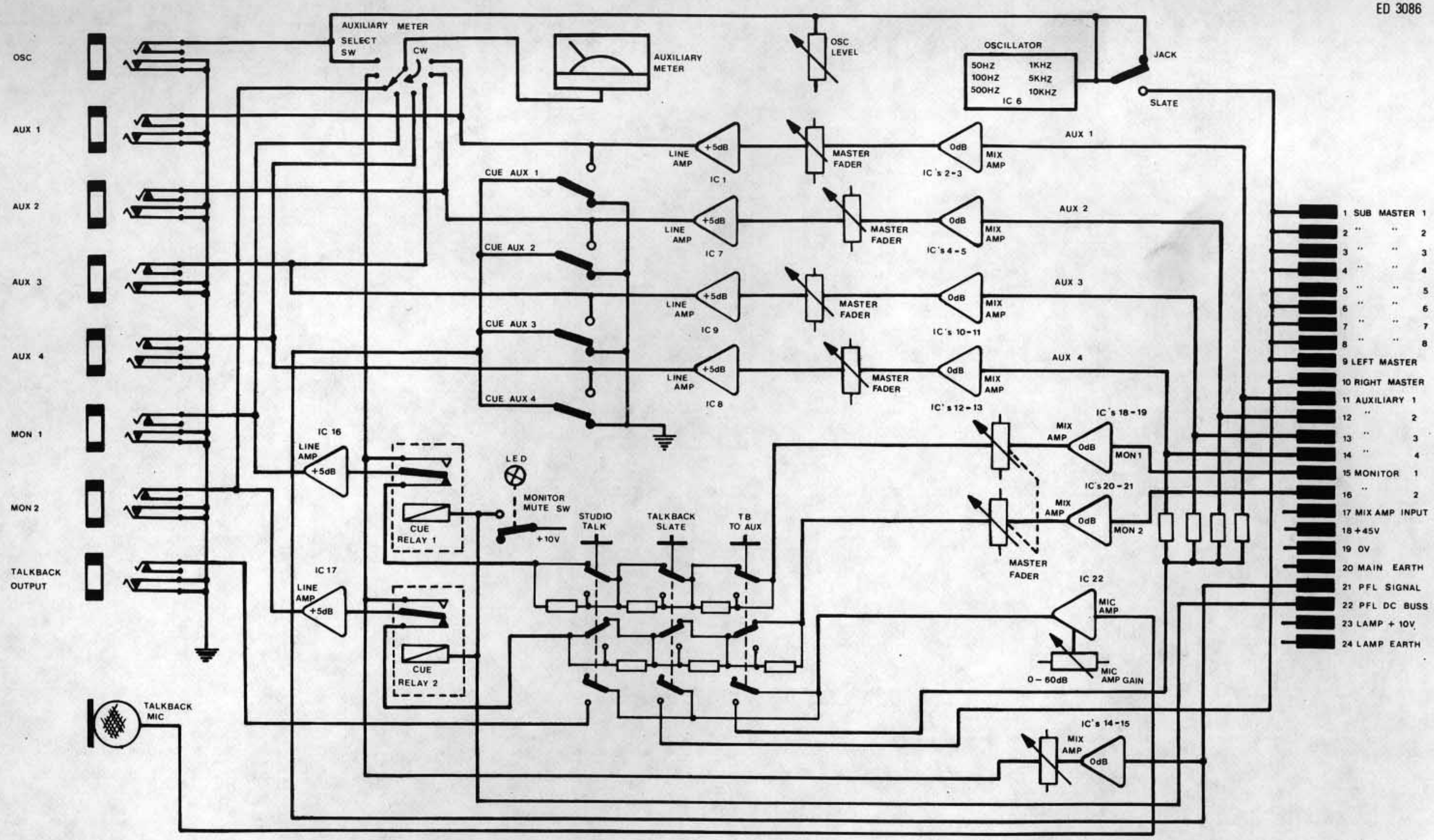
Like the auxiliary send mixing systems, the monitor mixing circuits comprise three integrated circuits each. Two of these are for the mixing amplifier and one is for the output line amplifier. Monitor system one therefore consists of I.C 16 (output amplifier) and I.C's 18 and 19 (mixing amplifier). Monitor system two consists of I.C 17 (output amplifier) and I.C's 20 and 21 (mixing amplifier). Since the monitor speaker signal also passes through relays RL 1 and RL 2, these should also be checked if I.C replacement fails to remedy the problem. If none of these measures have any effect, check for broken wires etc. on the master monitor level control or mute switch.

Talkback microphone:

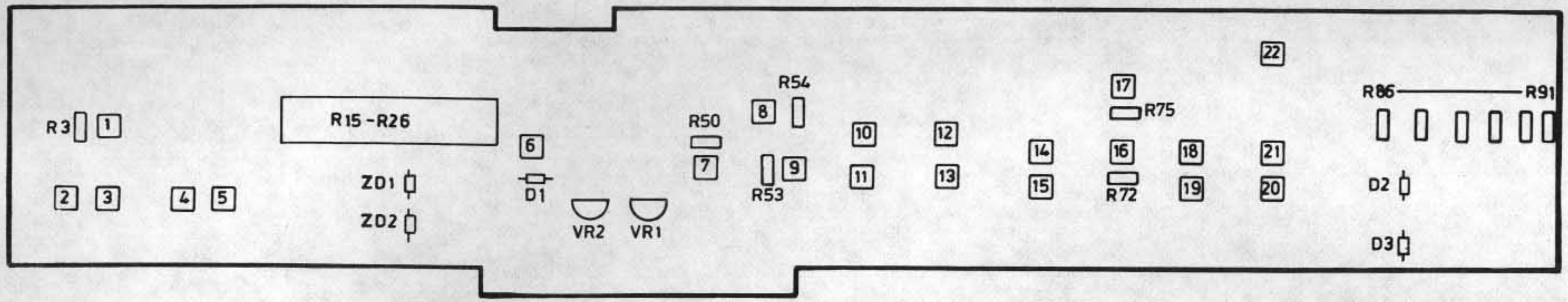
The talkback microphone amplifier consists of one integrated circuit, I.C 22. If replacement of this component fails to cure the fault, check the microphone level potentiometer for either faulty operation or a broken wire. Check also that the talkback microphone itself is functioning correctly and that the wiring from the microphone to the printed circuit board is not broken. It is also a good idea to check that all three talkback push-buttons are working correctly and that none of their terminals have become short-circuited.

Figure 10 shows the auxiliary module in flow diagram form with all integrated circuits identified.

ED 3086




TRIAD **Fleximix System** AUXILIARY MODULE SCHEMATIC CCT DIAGRAM FIG .10.




[1] IC MC1741S CP1 AUX1 OUTPUT LINE AMP

[2] " " " MIX AMP 2ND STAGE

[3] " " " " " 1ST "

[4] " " AUX 2 MIX AMP 2ND STAGE

[5] " " " " " 1ST "

[6] " " OSCILLATOR

[7] " " AUX 2 OUTPUT LINE AMP

[8] " " AUX 4 " " "

[9] " " AUX 3 " " "

[10] " " AUX 3 MIX AMP 2ND STAGE

[11] " " " " " 1ST "

[12] " " AUX 4 MIX AMP 2ND STAGE

[13] " " " " " 1ST "

[14] IC MC 1741S CP1 CUE MIX AMP 2ND STAGE

[15] " " " " " 1ST "

[16] " " MON 1 OUTPUT LINE AMP

[17] " " MON 2 " " "

[18] " " MON 1 MIX AMP 2ND STAGE

[19] " " " " " 1ST "

[20] " " MON 2 " " " "

[21] " " " " " 2ND "

[22] " " TALKBACK AMP

D1 0A47GE DIODE METER RECTIFIER

D2 1N 4002 SI DIODE RELAY CLICK SUPPRESION

D3 " " " " " "

ZD1 16V 1W 5% ZENER DIODE PCB SUPPLY REGULATOR

ZD2 16V 1W 5% ZENER DIODE PCB SUPPLY REGULATOR

R3 AUX 1 OUTPUT AMP GAIN RESISTOR

R15-26 OSC FREQ' SELECTION " 'S

R50 AUX 2 OUTPUT AMP GAIN "

R53 " 4 " " " "

R54 " 3 " " " "

R72 MON 1 " " " "

R75 " 2 " " " "

R86-91 TALKBACK DIMMING " 'S

VR1 AUX METER SENSITIVITY ADJUST

VR2 OSC OUTPUT LEVEL ADJUST

# AUXILIARY MODULE COMPONENT LOCATION

Fig 12

## E3 Customer modifications:

Adjusting output level of auxiliary sends 1-4:

The auxiliary send output line amplifiers have been factory set for an additional 5db gain so that the drive requirements of most power amplifiers or echo devices can be catered for. Since auxiliary sends 1 and 2 on the input modules are derived pre-fader and prior to the input module output line amplifier which has an extra 5db gain, the gain of the final auxiliary send line amplifiers have been increased 10db so that all four auxiliary send outputs become equal. The resistor values already fitted to the printed circuit board will therefore be different for auxiliary sends 1 and 2 as opposed to 3 and 4. Resistors R3 and R50 (3K ohm) determine the gain of auxiliary sends 1 and 2 respectively, whilst R53 and R54 (1K ohm) determine the gain of auxiliary sends 3 and 4 respectively. Increasing the value of these resistors will increase the gain and decreasing the value will decrease the gain. For 0db gain a wire link should be inserted in place of the appropriate resistor.

Adjusting output level of monitor sends 1 and 2:

Like the auxiliary send output amplifiers, the monitor send output amplifiers have also been given an extra 5db gain so that the sensitivity requirements for all makes of power amplifiers can be easily met. To increase or decrease the gain of monitor send 1 R72 should be changed and similarly for monitor send 2 R75 should be changed. Increasing the value will increase the gain whilst decreasing the value will decrease the gain. A 3K ohm resistor will therefore give 10db gain whilst a wire link in place of the resistor will give 0db gain. Since the cue system is part of the control room monitoring system, any adjustment to the monitor output line amplifiers will also affect the cue signal level.

Changing oscillator frequencies:

Resistors in close tolerance pairs are used to determine the oscillator sine wave frequencies and as a consequence it is a relatively simple procedure to change any of the six frequencies if required. Listed below are the resistors fitted as standard, their values and the frequencies they generate:

| | | |
|---|---|---|
| R15 and R21 | 680K ohm | 50HZ |
| R16 and R22 | 330K ohm | 100HZ |
| R17 and R23 | 68K ohm | 500HZ |
| R18 and R24 | 33K ohm | 1KHZ |
| R19 and R25 | 6.8K ohm | 5KHZ |
| R20 and R26 | 3.3K ohm | 10KHZ |

As can be seen from the above table, increasing the resistor value will lower the frequency and vice versa.

Adjusting talkback dimming levels:

The amount of control room speaker attenuation when the 'talk to auxiliary' or 'talk to studio' push-buttons are depressed is determined by resistors R88-R89 and R90-R91 respectively. Lowering the value will decrease the attenuation whilst raising the value will increase it. Since the 'slate' resistors R86-R87 are not fitted, maximum attenuation will be introduced when this button is depressed.

Figure 12 shows the component locations for the above gain changes etc.

## F1 General description:

The Fleximix 8 and 15 way mainframe systems have been designed as rugged, self contained modular 'blocks' each with it's own power supply and interconnection sockets ready for expansion at any time.

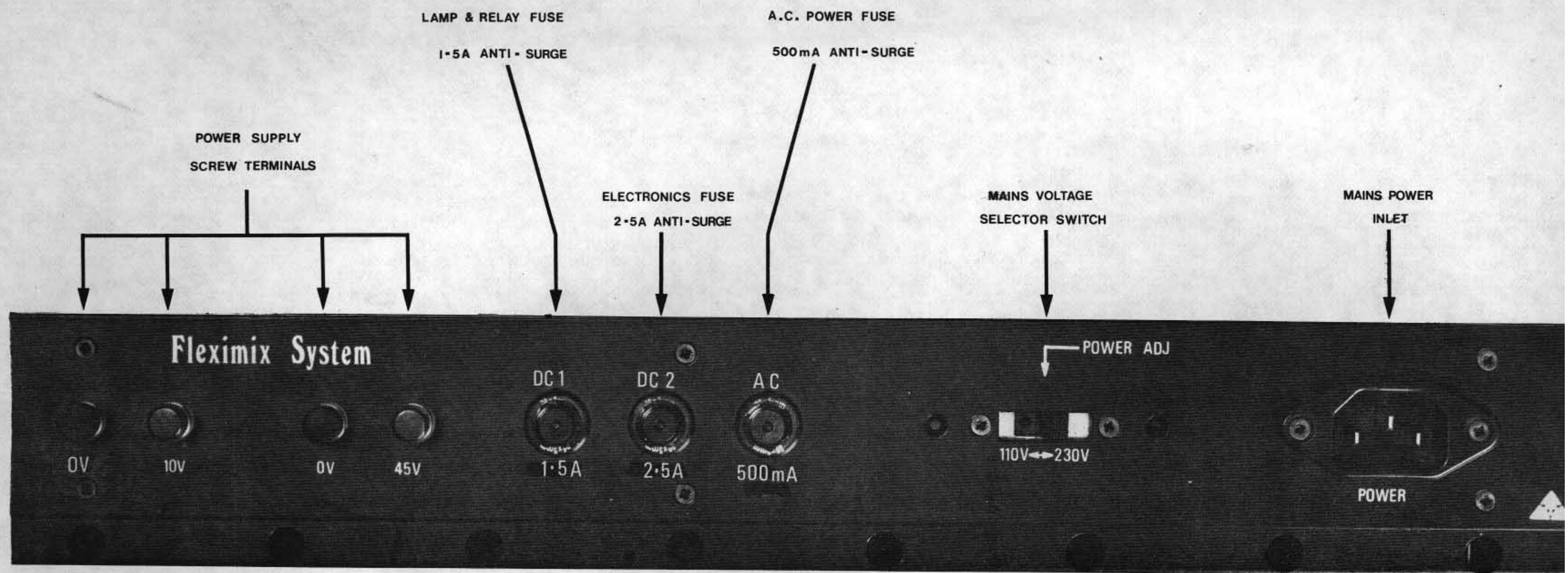
By using a 'motherboard' printed circuit board upon which all module edge connector sockets are mounted, wiring has been reduced to a minimum thereby eliminating the problem of bulky cable harnesses, strain and possible human error. By using a standardised system of edge connections whereby every Fleximix module has identical pin connections, conventional 'locked-in' placement of modules in the mainframe can be completely ignored. This means that with an absolute minimum of time and effort, the operator can determine exactly where the various modules should be placed for maximum operational ease. Only in the case of one module is it necessary to solder a simple link to the 'motherboard' in order to utilise this facility fully. The rest of the modules can be placed anywhere in the mainframe with no modification whatsoever and will function perfectly.

The physical and electrical connection of one mainframe to another is likewise a very quick and simple affair using only a minimum of tools. A series of bolts at strategic places along the joining faces of the mainframes ensure a rigid structure whilst a simple interconnection card slots into an edge connector aperture on each of the adjoining end panels.

Because of the compact nature of the system it readily adapts to portable operation and to this end can easily be mounted into a 'flight-case' for maximum protection whilst travelling. If required, mainframes can still be electrically joined together whilst mounted in flight cases by the use of a special interconnection cable. This means that for mobile recording or sound reinforcement applications only the required compliment of modules have to be carried thereby eliminating the neccessity of transporting a larger console, often the case when a conventional mixer is used for a mobile application requiring only a small number of input or output modules.

Both the 8 and 15 way Fleximix mainframes contain identical power supply units, fully short circuit protected and stabilised, capable of operation on both 110 and 220 volt A.C power. Since any console power supply can, in the event of failure, render the console unusable, the Fleximix power supply system has been designed to provide as many 'fail-safe' facilities as possible. Under normal conditions each supply runs at approximately one third of it's maximum current capability thereby making it possible for the supply in one mainframe to 'slave' the power requirements of another in the event of a supply failure. For this purpose therefore, screw terminals are provided at the top of each mainframe so that wires can be 'jumpered' accross very quickly should this situation occur.

If however, the 'Fleximix' system consists of a single mainframe only, the screw terminals are then able to provide a means of connecting either an external power supply system or batteries. When either 'slaving' one power supply to another or using an external source, the two D.C fuses on the mainframe voltage selector panel should be removed in order to isolate the internal supply from the 'motherboard'. Figure 13 shows the mainframe voltage selector panel and fuses in detail.




MAINFRAME VOLTAGE PANEL. Fig.13.

## F4 Power supply 'slaving':

In the unlikely event of a power supply failure, the following procedure should be adopted; remove the mains power lead from the faulty mainframe and from the mainframe that is to be used to 'slave' accross. Remove the two D.C fuses from the voltage selector panel which will then isolate the faulty supply output from the motherboard. Connect accross four pieces of approximately 5 amp single wire from the 'slave' mainframe screw terminals to the 'faulty' mainframe screw terminals. The console will now operate as before with one supply driving two mainframes. Because identical power supplies are used in both 8 and 15 way frames, it is quite in order to use an 8 way frame to slave a 15 way frame. Obviously the faulty power supply should be repaired as soon as possible in order to maintain the high reliability margins obtainable with this system. When wiring 'jumper' leads from one mainframe to another it is very important to make sure that the terminals are screwed down tightly and that there is no possibility of the wires shorting together or against the mainframe chassis.

## F5 Mainframe interconnection procedure:

To expand the Fleximix system, the method of joining mainframes together is very quick and simple requiring only a few tools. The procedure is as follows: Remove the black end cladding panels from the appropriate ends of the two mainframes to be joined together by means of the black 'mushroom'head screws. Remove enough modules from the left hand mainframe so that a screwdriver can reach the end panel from inside. Remove two or three modules from the right hand frame so that a spanner can be manoeuvered next to the end panel. The mainframe joining card should now be inserted into the edge connector slots on each mainframe. This is achieved by inserting the card first into one frame and then sliding the the other frame onto the card so that the two mainframes butt together. The countersunk screws should now be passed through the appropriate holes in the left hand mainframe end panel and secured with the locking nuts. The two mainframes will now assume a rigid structure that can be treated as a single unit. Figure 17 gives all mainframe dimensional data whilst Figure 18 details pictorally the mainframe interconnection procedure.

## F6 Motherboard sub-group linking:

In order to make the entire Fleximix system as versatile as possible so that modules can be placed anywhere in the mainframe and still function correctly, it was decided to make the linkwiring neccessary for each sub-master module to connect to it's appropriate sub-group buss, external to the module. This is achieved by a simple wire link that is connected accross pins located next to each edge connector socket on the motherboard. Because this linking is only relevant to the sub-master module, any other type of module inserted into a previous sub-master edge connector socket will function as normal with no modification being neccessary. It also follows on from this therefore that if the linking arrangement is changed, previous links can be left in situ on the motherboard so that the original combination can be returned to at any time. In practical terms, this means that a console can be arranged so that merely by swapping modules, the complete output group system can be changed say from the right hand side of the mixer to the left hand side at will.

The advantages of this kind of versatility are obvious for example in mobile recording where according to location it may be neccessary to change the output group placement owing to space restrictions.

MODULE UNIT WEIGHTS

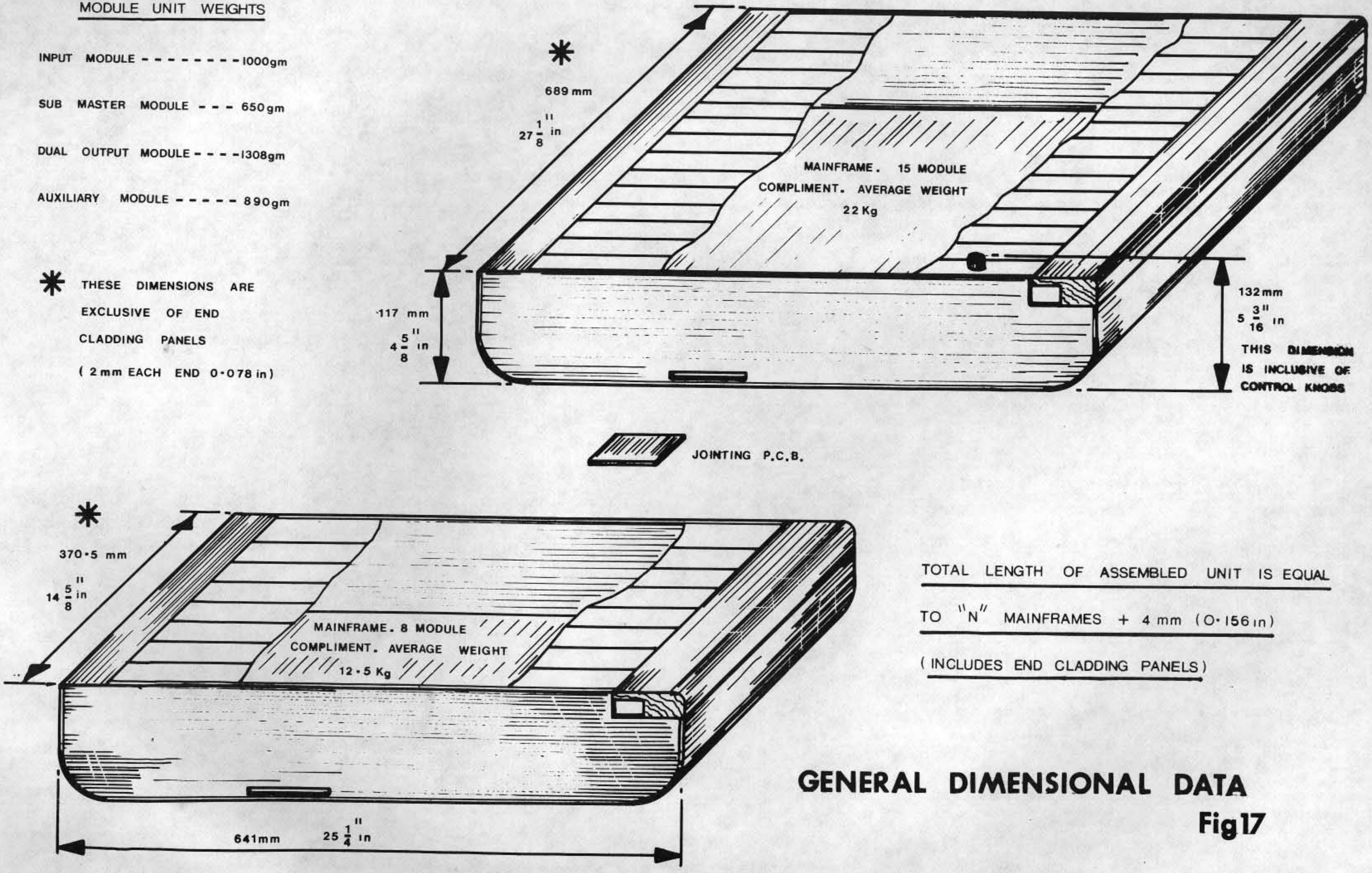
INPUT MODULE - - - - - - - - - 1000gm

SUB MASTER MODULE - - - 650gm

DUAL OUTPUT MODULE - - - - 1308gm

AUXILIARY MODULE - - - - - 890gm

* THESE DIMENSIONS ARE EXCLUSIVE OF END CLADDING PANELS ( 2 mm EACH END 0·078 in)




TOTAL LENGTH OF ASSEMBLED UNIT IS EQUAL TO "N" MAINFRAMES + 4 mm (0·156 in)

( INCLUDES END CLADDING PANELS )

# GENERAL DIMENSIONAL DATA

**Fig 17**

FOLLOWING INSTRUCTIONS TO BE EXECUTED ON COMPLE[illegible] FLAT SURFACE. (AFTER REMOVAL OF END CLADDING PANELS IF APPLICABLE)

① INSERT JOINTING P.C.B. (track side up) INTO 1st MAINFRAME SLOT.

② POSITION 2nd MAINFRAME TO 1st MAINFRAME.

③ INSERT SCREWS IN POSITIONS 2, 3, 4 & 5.

④ INSERT SCREWS IN POSITIONS 1 & 6.

⑤ INSERT SCREWS IN POSITIONS 7 & 8.

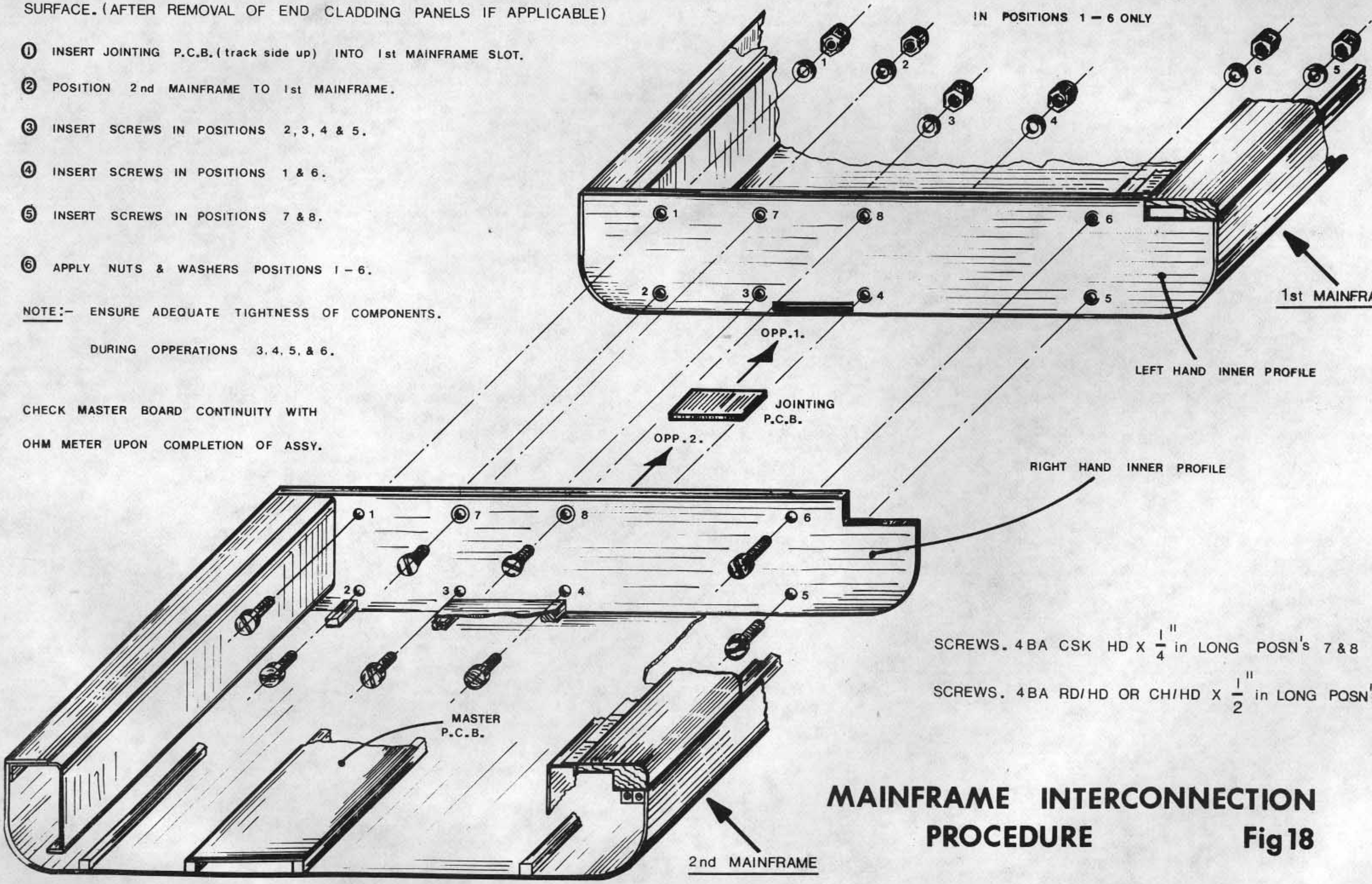
⑥ APPLY NUTS & WASHERS POSITIONS 1 – 6.

NOTE:– ENSURE ADEQUATE TIGHTNESS OF COMPONENTS.

DURING OPPERATIONS 3, 4, 5, & 6.

CHECK MASTER BOARD CONTINUITY WITH

OHM METER UPON COMPLETION OF ASSY.



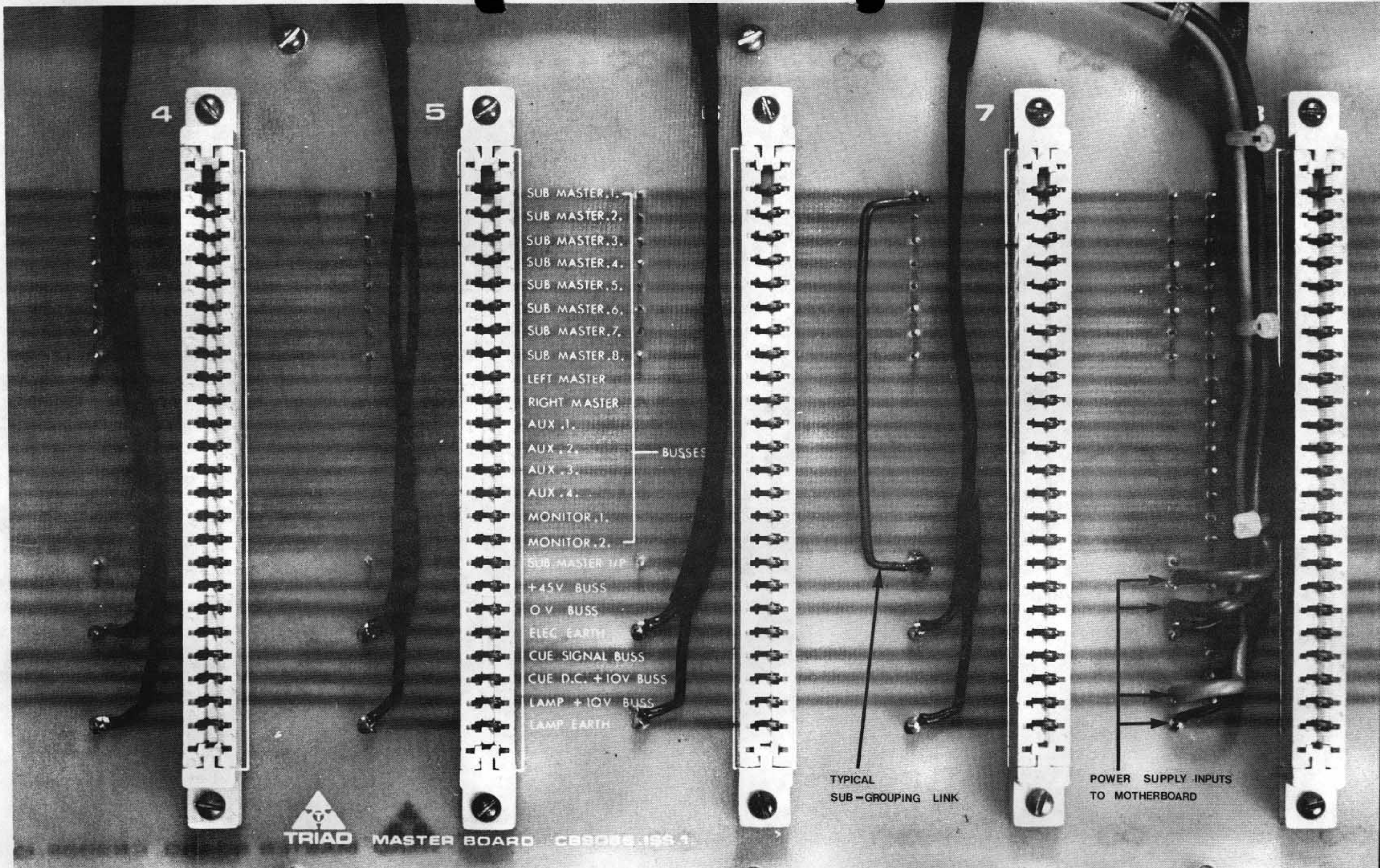

SCREWS. 4 BA CSK HD X $\frac{1}{4}''$ in LONG POSN's 7 & 8

SCREWS. 4 BA RD/HD OR CH/HD X $\frac{1}{2}''$ in LONG POSN's 1 — 6

# MAINFRAME INTERCONNECTION PROCEDURE

Fig 18

Figure 19 shows a close-up of the 'motherboard' detailing a typical sub-grouping link. At the top and to the left of each edge connector socket is a vertical row of eight pins. Each of these pins is for each of the sub-group busses, starting with buss 1 at the top. By the side of terminal 17 on each edge connector is another pin which is the input to the sub-group mixing amplifier for the appropriate sub-master module plugged into that edge connector. Selection of the desired sub-master module position is therefore a simple matter of soldering a wire link between the appropriate sub-group buss pin and mix-amp input pin. To assist in wiring these links to the 'motherboard' it is useful to note that beside edge connector sockets 5 and 12 there is a printed list of all the module pin connections.




MOTHERBOARD LINKING DETAIL

Fig. 19.

# TRIDENT AUDIO DEVELOPMENTS LTD.

Your Frame is fitted with a TRIDENT FLEXIMIX POWER SUPPLY MK 3.
This power supply is one complete unit and is connected to the frame with a Molex 15 Pin Connector. The power supply has a crowbar built with an Audiable Warning Device on it. The Audiable Warning Device "buzzes" whenever there is a breakdown on the Power Supply (40 Volt, 10 Volt, or Phantom 45 Volts).

NOTE that when the frame is switched on it might crowbar the Power Supply, switching on the Audiable Warning Device. So it is recommended that the frame should be switched off and after 15 secs. switched on again which will make the frame ready for operation.

NOTE if you have two or more frames together it is necessary to switch all frames together, a failure to do so will leave your frames with 4 Volt Lamp supply instead of 10 Volts.

RECOMMENDED FUSE FOR MK 3 POWER SUPPLY.

| | 110 VOLT OPERATION SINGLE FRAME | | 230 VOLT OPERATION SINGLE FRAME |
|---|---|---|---|
| AC:- | 500 M.A. | 20mm. | 315 M.A. |
| 40 Volts:- | 1·5 A | 20mm. | 1·5A |
| 10 Volts:- | 1·5 A | 20mm. | 1·5A |

WHEN SLAVING TWO FRAMES

AC:- 500 M.A. or 800 M.A. (if available). 315 or 500 M.A

ALL FUSES ANTI-SURGE

Factory: Shepperton Studios, Squires Bridge Road, Shepperton, Middlesex.
Registered in England No. 1063678