AD 031
MICROmixer

Handbook

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MICROmixer

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1.3 *ELECTRONIC FEATURES*

As mentioned above, the MICROmixer is a modular, battery operated, studio performance mixer squeezed into a very small case. The basic requirement was seen to be for an eight input, two output, mixer with as many other features as were possible within the size limitations. It was decided therefore to partition the system into eight identical input modules, each capable of operating from microphones or high level inputs, and a single output module to service all inputs. The MICROmixer is intended to operate from internal batteries and the popular U2, HP2 or international 'D' cell was chosen as likely to be available world wide. The supply voltage was set at 24 volts because this is suitable for giving the desired maximum signal levels, while not being too extravagant with batteries. The necessary 16 cells are contained in a battery box which is integral with the case construction, adding to the rigidity of the mixer, and they are replaceable by removing the back cover of the unit. This is done by unscrewing the four plastic feet which secure the cover, which when removed also reveals the printed wiring of the motherboard, making fault location much easier. Since it was expected that the MICROmixer might be used where mains power was available, provision has been made for the necessary operating voltages to be fed in from an external power unit and a switch has been fitted to effect the changeover. Finally, it was foreseen that the comprehensive facilities contained within the MICROmixer would, on occasions, be insufficient and that arrangements would have to be made to allow other apparatus to be 'patched' into the system. This was provided for in two ways. Firstly, insertion points to enable compressors or external equalisers to be used were added to the main output groups A and B. Secondly, an Extender socket was fitted which could enable more input channels to be used and it is AUDIO DEVELOPMENTS intention to introduce a matching unit which could house the additional facilities, such as compressors, talkback and monitor amplifiers, together with the necessary power supplies. This Extender Unit would interconnect with the MICROmixer by means to the Extender socket.

1.4 *ELECTRONIC SPECIFICATION*

MICROPHONE — Sensitivity -84dbm in MIC and -62dbm in ATT, referred to 600R terminating impedance, for 0dbm out. Input impedance is 1000 ohms

NOISE CHART

TYPICAL FIGURES 10Hz TO 100KHz 600 ohm TERMINATION	Noise dBm	Gain dBm	Total dBm
a) All Fader down.	−70	0	−
b) Group Fader up .	−66	0	−
c) Channel Fader up + (b)	−63	0	−
d) Channel Fader up with EQ selected	−62	0	−
e) Channel Gain Control up+(c) on Line	−63	+12	− 75
f) (e) on ATT MIC.	−54	+62	− 116
g) (e) on MIC.	−48	+84	− 132

2.1 *GENERAL*

Fig. 2A shows a diagram of the signal paths within the mixer and the text below follows the route taken by an input signal, explaining where appropriate, the reasons why the system is designed as it is. The aim is to give the user an understanding of the possibilities and the limitations of the MICROmixer, so that the best results can be obtained.

In fig. 2A, principal signal paths are shown by heavy lines and subsidiary functions are light lines. The dotted boxes enclose the parts contained within a module and the numbered connections represent the printed circuit terminations of each module. Pin 1 is nearest the input/output panel in all cases. Amplifiers are shown as a triangle and the gain is marked within the triangle in decibels. Switches are shown either in the unoperated or OUT position for push buttons or in the anti clockwise position (1 position) for rotaries. The faders marked with an asterisk * are slide faders, the rest being rotary.

2.2 *MIXING BUSSES*

The diagram shows the mixing bus lines right in the centre of the picture and this is logical because they are the heart of the system. They are the very low impedance summing points of the mixing amplifiers and they extend across the full width of the mixer to pick up the inputs from the various points. Each input module has its own mixing resistors mounted within it and so the sum currents are fed from each channel onto the bus rails via the printed circuit connectors. To minimise hum and noise pickup on these very sensitive lines, they are shielded on both sides by earthed conductors and, when the back cover is in place, by the close proximity of the casework which is also earthed. In addition to the inputs from the channel amplifier, the mixing busses are also taken to an Extender socket which enables signals to be added to the mix from external sources. The technique proved effective in the AD007 and so has been retained for the MICROmixer. In the diagram, only one input channel feed has been drawn in, the other seven channels being identical.

2.3 *THE MICROPHONE/LINE AMPLIFIER MODULE*

This is shown on the left of Fig. 2A and it accepts low level signals from Microphones

or high level signals from lines, amplifying and equalising them before distributing them to the mixing busses as described above.

2.3.1 Inputs

To select between the different types of input likely to be encountered, the module is fitted with an input selector switch marked MIC, ATT and LINE. The diagram shows how this switch enables a single socket to be used for both microphone and line inputs, this being necessary because of space limitations. It would have been possible to retain the facility of separate sockets by using smaller types (such as DIN) but it was considered essential for compatibility with existing equipment and to maintain professional standards that a CANNON type of connector be used. The chassis mounting socket is the XLR-3-31 which mates with an XLR-3-12c fitted to the incoming cable. In line with the convention used by AKG, the connector is wired as follows:—

Pin 1 Earth

Pin 2 Inphase

Pin 3 Return

The selector switch has three positions, each one being intended for a particular input condition and each is considered separately below.

(i) MIC.

In this position, the feed from the input socket is routed to the input microphone transformer and the pre-amplifier. Since, for reasons of maintaining the highest possible signal to noise ratios, there is no attenuation possible prior to the pre-amplifier, the pre-amplifier has been designed to accept the highest input level likely to be encountered in a practical situation, without overloading. This has been done by calculating the likely maximum signal level from a dynamic microphone and then restricting the amplifier gain so that the maximum output level is not exceeded. On this basis, the microphone pre-amplifier has a gain of 35 dB, giving it a maximum input level capability of −20dBm, while offering a load to the microphone of 1000 ohms. This makes it suitable for microphones of from 200 to 600 ohms self impedance. If much lower impedance microphones are to be used

impedance is > 100 kilohms.This is made deliberately high so that users are not restricted to 600 ohm lines, and many other equipments, such as tape recorders and so forth, need a higher impedance load than the 10 kilohms normally offered. If a 10 kilohm load is needed, then a fixed resistor of this valve can be wired between pins 2 and 3 of the input connector. The module will accept unbalanced inputs and it is recommended that in this case, pins 1 and 3 of the connector be strapped and pin 2 used as the live input.

2.3.2 Phantom Powering

The MICROmixer is fully adapted for phantom powering by the artificial centre-tap method recommended by AKG. The powering resistors used are fitted to each Channel Amplifier module, and as the diagram (Fig. 2A) shows, the socket is powered in either the MIC or ATT position. The resistor value (10 kilohm) is suitable for supply voltages between 24 and 52 volts and the internal 24 volts supply can be used if desired by connecting together pins 3 and 4 on the external power socket. Alternatively, power from an external source can be fed in via pin 4 and the phantom power voltage is also available at the Extender Socket for feed to or from an Extender unit.

N.B. There is no reason to switch off the power to dynamic microphones if they are used in conjunction with condensers because if powered in this way, there will be no current flow through the dynamic microphone coil, and so no damage can possibly result. A warning is necessary in case it is required to employ un-balanced dynamic microphones while condensers are in use. This must not be attempted.

2.3.3. Channel Gain Control & Power Switch

The gain control for the channel is positioned in between the pre-amplifier and the main amplifier so that the best compromise between noise level and dynamic range is achieved. The main amplifier has a gain of 37dB giving a maximum overall gain on microphones of 72dB. In practice, the channel gain is set to avoid overloading the main amplifier, whatever the input signal level from the microphone. The Channel ON/OFF switch is incorporated with the Gain Control and this allows the user to prolong battery life by switching off the channels not in use and so reducing the current drain. Since the current

requirement of each channel is about 16 milliamps, this is a worthwhile procedure.

2.3.4. *Equaliser*

Comprehensive equalising facilities are fitted consisting of three controls, each capable of boosting or cutting a given range of frequencies and Fig. 2B shows a graph of the measured performance of a typical equaliser. A bypass switch is fitted, which, when unoperated, allows the equaliser to be removed from circuit. This enables a quick and easy comparison to be made between the equalised and 'flat' performance of the channel when the mixer is in use.

The BASS and TREBLE controls are of the common Baxendall type and the MIDDLE range control is a peaking equaliser, centred on 2.4 KHz as standard. This frequency was chosen to be the same as the presence frequency used by broadcast engineers. Other frequencies could be provided to special order but switchable frequencies are not possible due to lack of space.

2.3.5. *Channel Overload Indicator*

Because it was not considered possible to arrange for metering of each channel, a means of warning the user of approaching channel overload has been built into the Module. A Light Emitting Diode with drive amplifier is arranged to flash when the channel output exceeds +15dBm. This indication is given prior to the channel fader because no re-positioning of this will correct for overload. The channel gain control must be used to reduce the amplification to the point where overload fails to occur (see 2.3.4)

2.3.6. *Insertion Points*

These are not provided on the standard MICROmixer but provision has been made on the channel to enable an insertion to be effected to special order. It is possible to bring out these insertion points to a multiway socket, but for most applications these are not needed. In the standard MICROmixer, they are linked across internally.

2.3.7. *Signal Routing*

a) AUXILIARY

This function has been included due to customer demand, and to increase its usefulness.

at the same time as the stereo groups A and B are being employed. The gain of the mixer is 0 since overloading is likely if more were provided.

2.4.3. Insertion Points (A and B)

Although not available on the channels, pre-fader group insertion points are provided. P.O. Office jacks, (gauge B) are used, the "OUT" jack being wired as a bridging point and the "IN" jack as a break point. This means that unbalanced outputs can be extracted from the groups without disturbing the internal signals. The minimum load resistance that may be imposed at this point is 10 Kilohms. When the IN jack is used, the feed from the internal circuits is broken and so an unbalanced input from an external equipment can be made. It is most important that the correct size jacks be used because those found on domestic equipment are usually gauge A which have a much larger tip than the gauge B. Use of these larger jacks can distort the contact leaves in the jack and cause permanent damage.

The insertion points are provided so that compressors or extra equalisers etc, can be 'patched' into groups A and B. The signals are unbalanced, the tip of the jack being the live line with the ring and sleeve earthed.

2.4.4. Group Faders

Groups A, B and C are fitted with slider fader, calibrated in decibels in the same way as the channels. The Auxiliary output control is a rotary fader uncalibrated.

2.4.5. Output Amplifiers

Because of size and cost limitations, it was decided to offer electronic balancing in preference to the more normal transformer balancing. In this method an unbalanced output signal is developed in the normal way, and in addition, a signal of exactly the same amplitude but inverted in phase is generated by a self balancing amplifier which uses the normal output as its control signal. For practical purposes then we have two signals, exactly equal in amplitude but opposite in phase, available at pins 2 and 3 of the output Cannon Connector. The only difference is that there is no D.C. connection between 2 and 3

pins 2 and 3 of the output connector, then the meter reads 4. This means that since each amplifier is only responsible for half the output voltage, then actual level at the input terminals of the meter is –6dBm. This level difference should be remembered when unbalanced outputs are taken.

2.6 MONITOR

Provision is made for a pair of headphones to be used either in stereo from groups A and B or in mono from group C. A toggle switch is provided to make the change-over and output is via a P.O. jack. Buffering resistors are included to avoid click and an extra pair of outputs is made available at the Extender Socket. The intention here is that they could be fed to external amplifiers for loudspeaker monitoring.

2.7 POWER SUPPLIES

The normal 24 volt supply for the MICROmixer is taken from a pack of 16 'D' cells (U2, HP2 or similar.) The high power versions are preferred for their longer life and the fact that they are sealed. This prevents corrosive fluids from leaking into the battery compartment and damaging the metal work, and it is highly recommended that batteries should not be left in the mixer while not in use. The power is turned on and off by a slide switch on the top panel marked INTernal/EXTernal. When set to Internal use, the batteries are under load and at all other times, the switch should be set to EXT.

The warning to replace exhausted batteries is given by the LOVOLT MONITOR which illuminates when the supply voltage falls below 20. Below this voltage, the mixer will still give good performance for a time but for complete assurance, the cells should be replaced. This LOVOLT circuit is mounted on the Output Module.

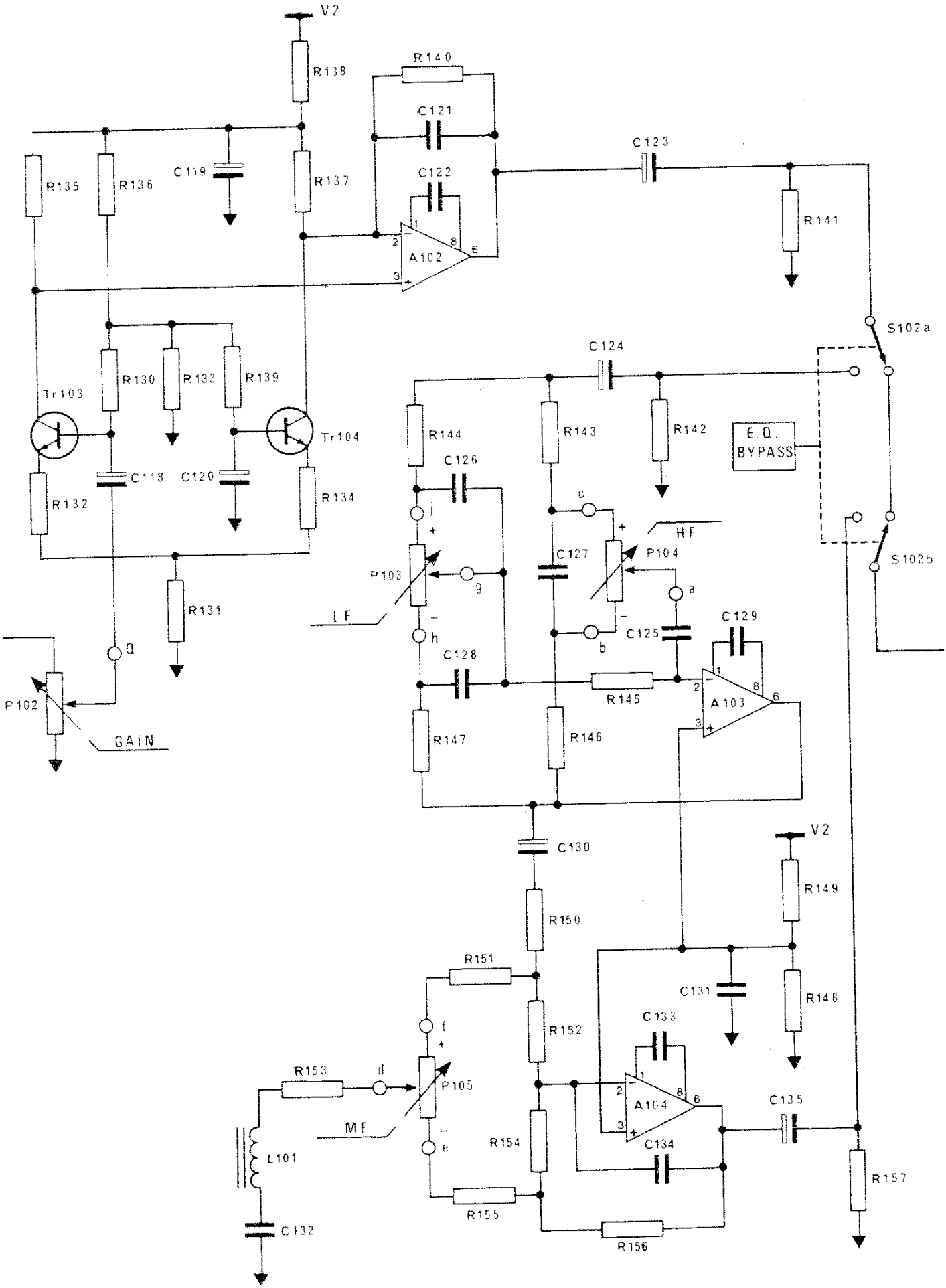
External power supplies are normally provided by a mains operated power unit and access to the mixer is by way of an XLR-4-31 chassis mounting plug. The diode in the circuit is to avoid damage to the mixer if the incoming supplies are accidentally reversed in polarity. This is more likely if it became necessary to use a pair of 12 volt car batteries as the power source. An additional problem in this case is that it is possible to 'blow' the 1 Amp fuse due

ADO31 DRAWING LIST

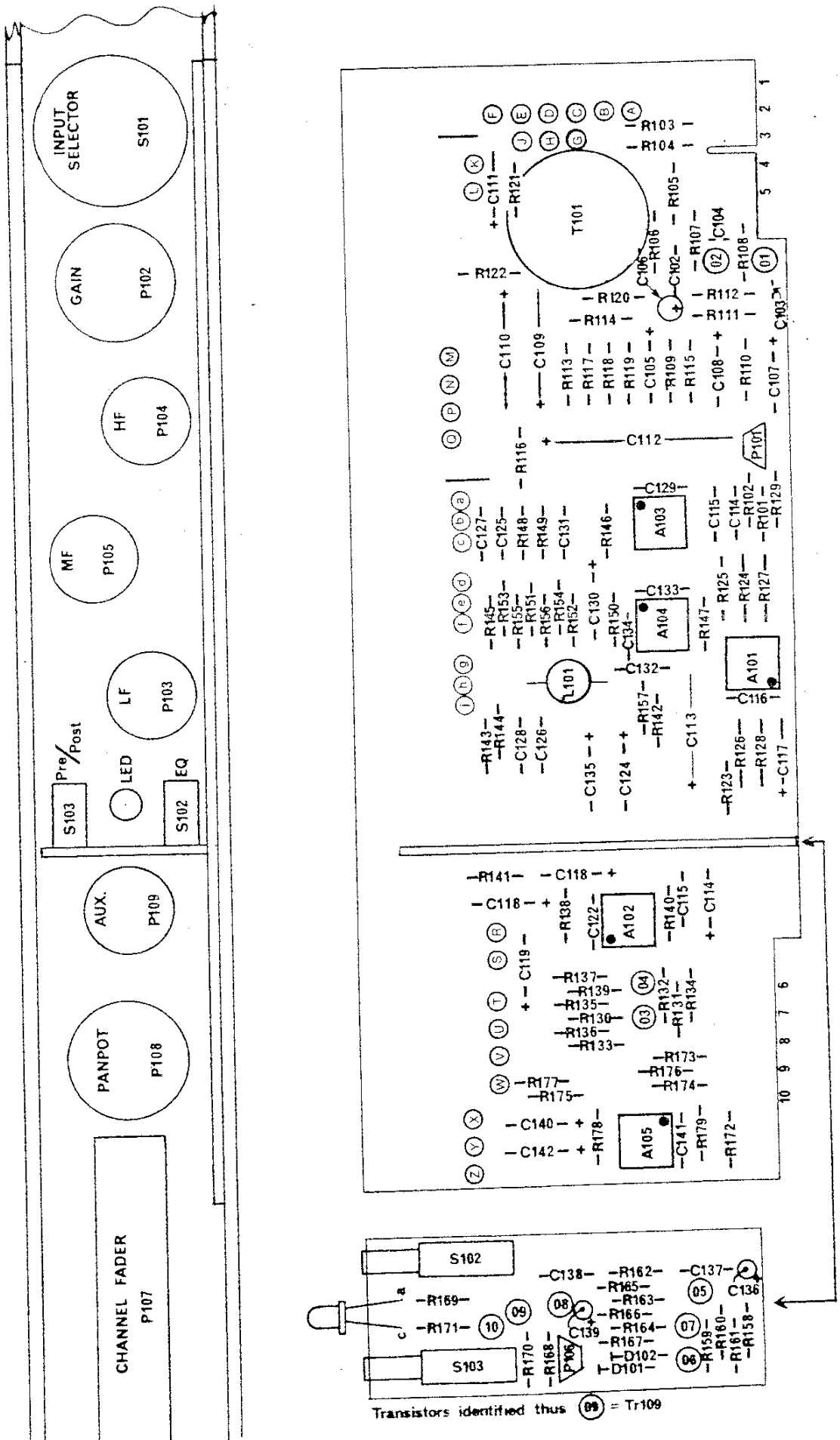
The ADO31 has many options. Only those drawings relevant to the Mixer SERIAL NO. ..... are included in this Service Manual.

Page	Module	Drawing Title	Drawings Included in this manual
G.1.	General	Module Interconnections	
G.2.	"	Motherboard Layout	
M/L 1.	Mic/Line	Circuit Diagram Part 1.	
M/L 2.	"	" " " 2.	
M/L 3.	"	" " " 3.	
M/L 4.	"	Parts List	
M/L 5.	"	Component Location	
O/P 1.	Output	Schematic Part 1 - Versions a or b	
O/P 2.	"	" " 2	
O/P 3.	"	Sub-section details	
O/P 4.	"	Monitor Arrangements " a,b,c,d,e,	
O/P 5.	"	Parts List	
O/P 6.	"	Component Location " a,b,c.	
PPM 1.	Peak Program Meter	Parts List	
PPM 2.	" " "	Circuit & Layout	
PPM 3.	" " "	Setting up & Calibration	
VU 1.	Volume Unit Meter	Parts List	
VU 2.	" " "	Circuit & Layout	
C/L 1.	Compressor/Limiter	General Arrangement	
C/L 2.	" "	General Circuit Diagram	
C/L 3.	" "	Sub Module Circuits	
C/L 4.	" "	Parts List	
C/L 5.	" "	Sub-Module Component Location	
C/L 6.	" "	Setting up & Calibration	
O/B 1.	Output Buffer	Circuit & Layout	
CON 1.	DC-DC Converter	Parts List	
CON 2.	" "	Circuit & Layout	
M/B	Monitor Booster	Circuit & Layout	
PS 1.	Power Supply	Circuit diagram	
PS 2.	" "	Component Location	

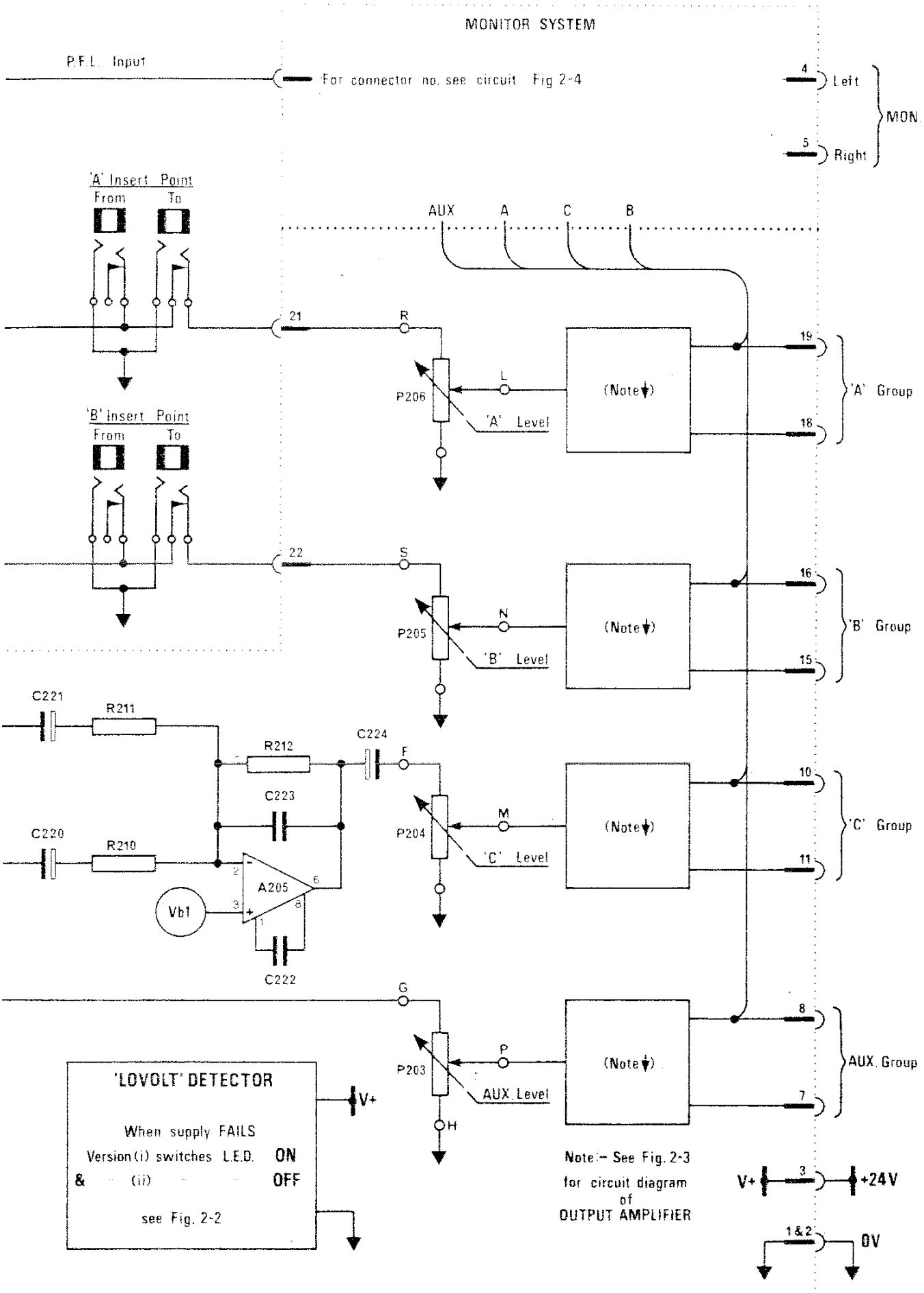
MIC/LINE MODULE (Part 2)

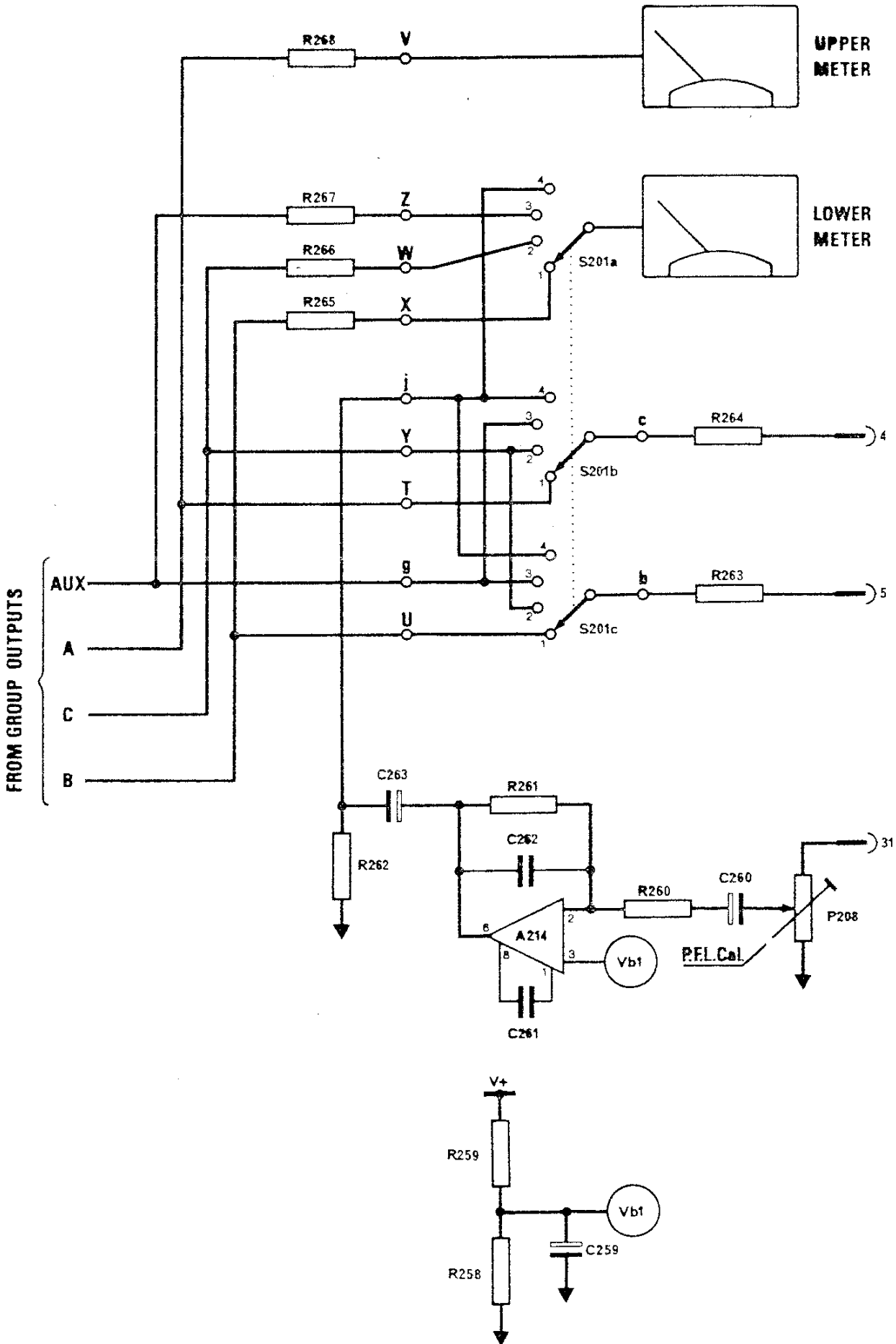


MIC./LINE MODULE
COMPONENT LOCATION



OUTPUT SCHEMATIC Part 2



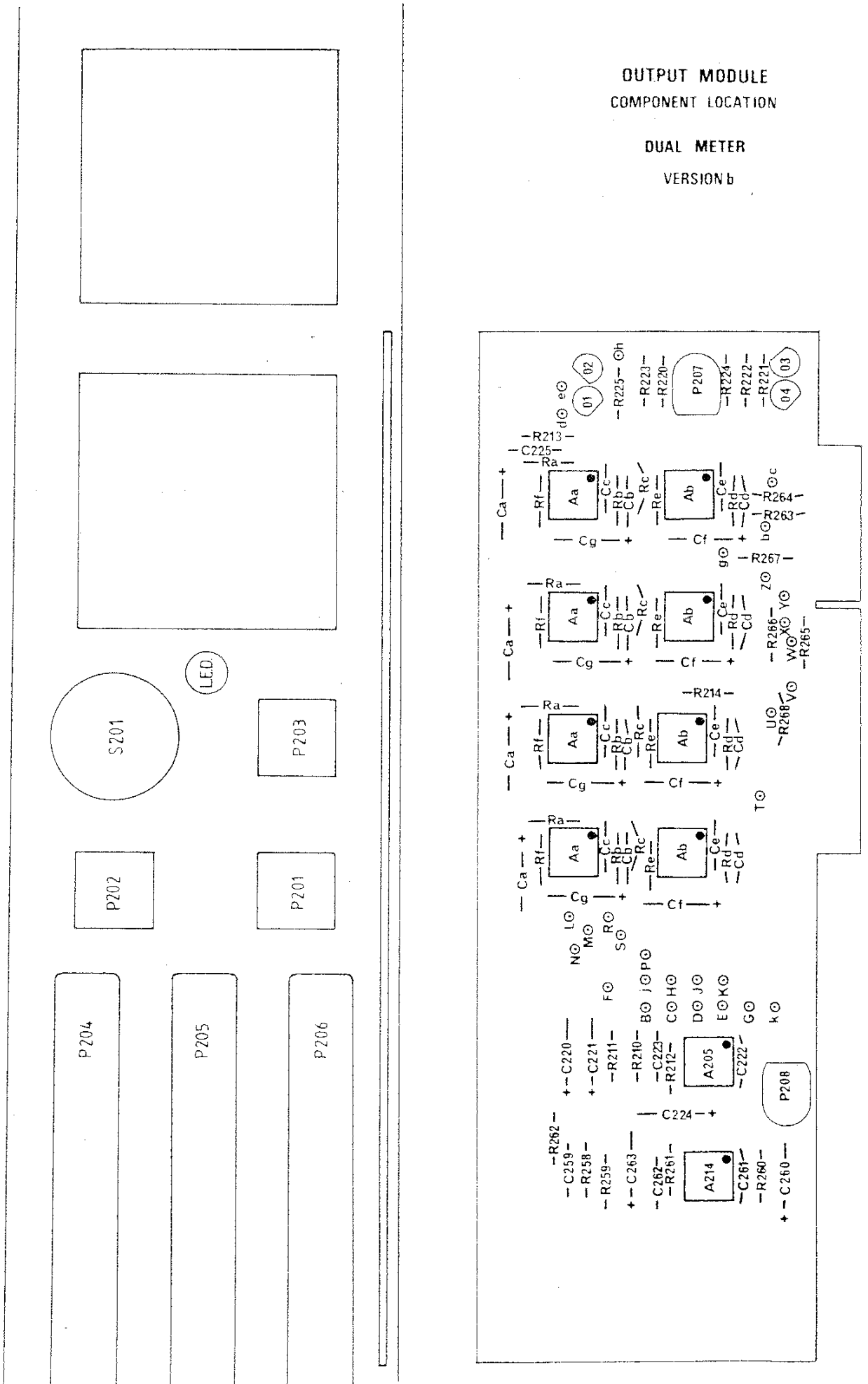

R268
V
UPPER METER
R267
Z
R266
W
R265
X
LOWER METER
S201a
j
Y
T
g
U
S201b
S201c
c
R264
b
R263
AUX
A
C
B
FROM GROUP OUTPUTS
C263
R261
R262
C262
R260
C260
P208
A214
Vb1
P.F.L.Cal.
C261
V+
R259
R258
C259

OUTPUT MODULE

COMPONENT LOCATION

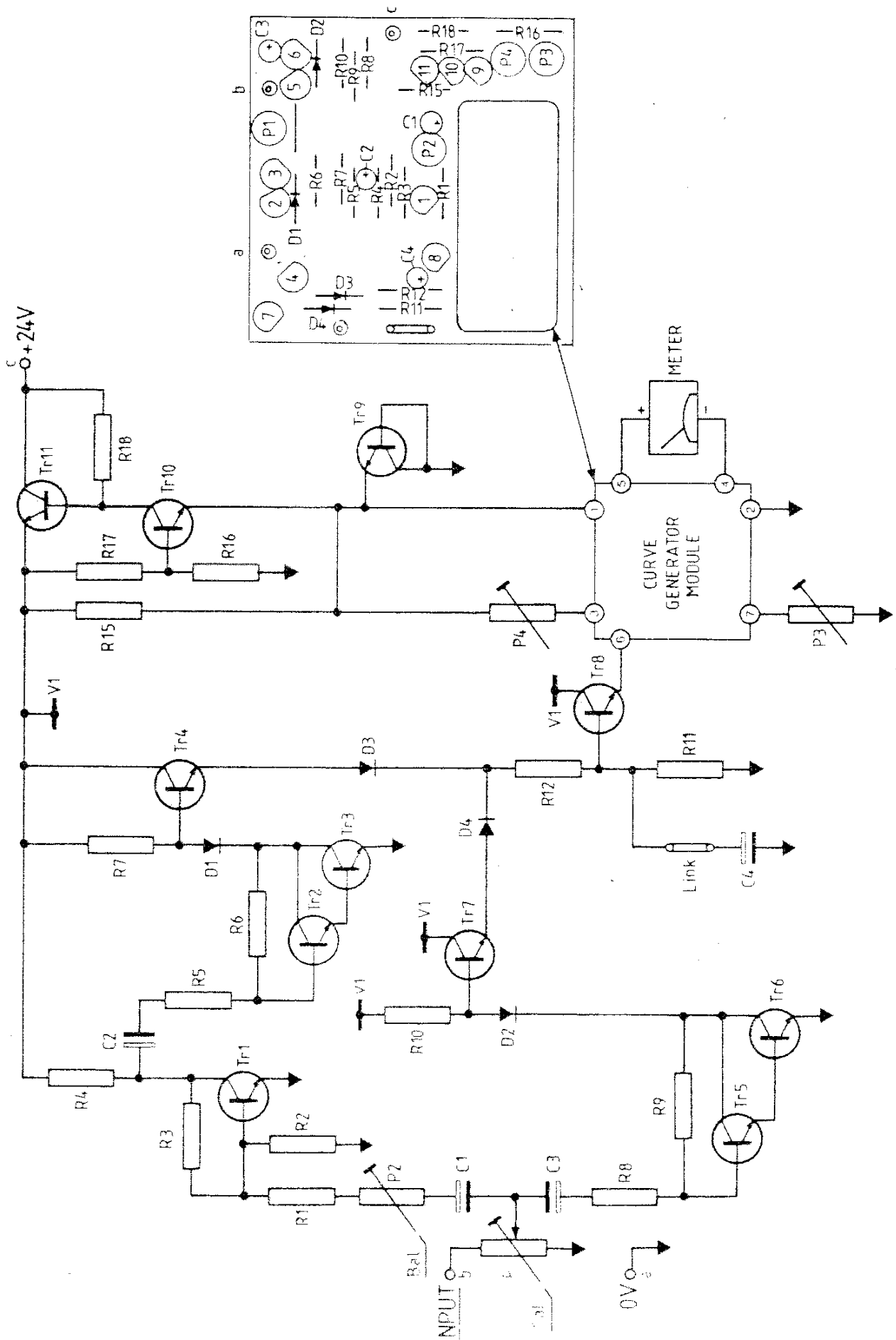
DUAL METER

VERSION b



PEAK PROGRAM METER DRIVER

CIRCUIT & LAYOUT



PEAK PROGRAMME METER

TEST AND CALIBRATION ROUTINE.

D.C. Checks:

Supply	24v	at	18 mA.
V1			20v
TR9	Emitter		7.5v
TR2/3	Collector		1.2v
TR5/6	Collector		1.2v

CALIBRATION:

Drive meter with a signal of 1 KHz OdB level.

Symmetry check - remove link.

At the junction of R11 and R12 adjust P2 until the peaks of the rectified pulses are equal.

Reconnect link.

a) Adjust P1 until meter reads Mark 4 (OdB)

b) Increase signal level to +8dB.

c) Adjust P4 until meter reads Mark 6.

d) Lower signal level to OdB. Adjust P1 if necessary.

e) Lower signal level to -8dB.

f) Adjust P3 until meter reads Mark 2.

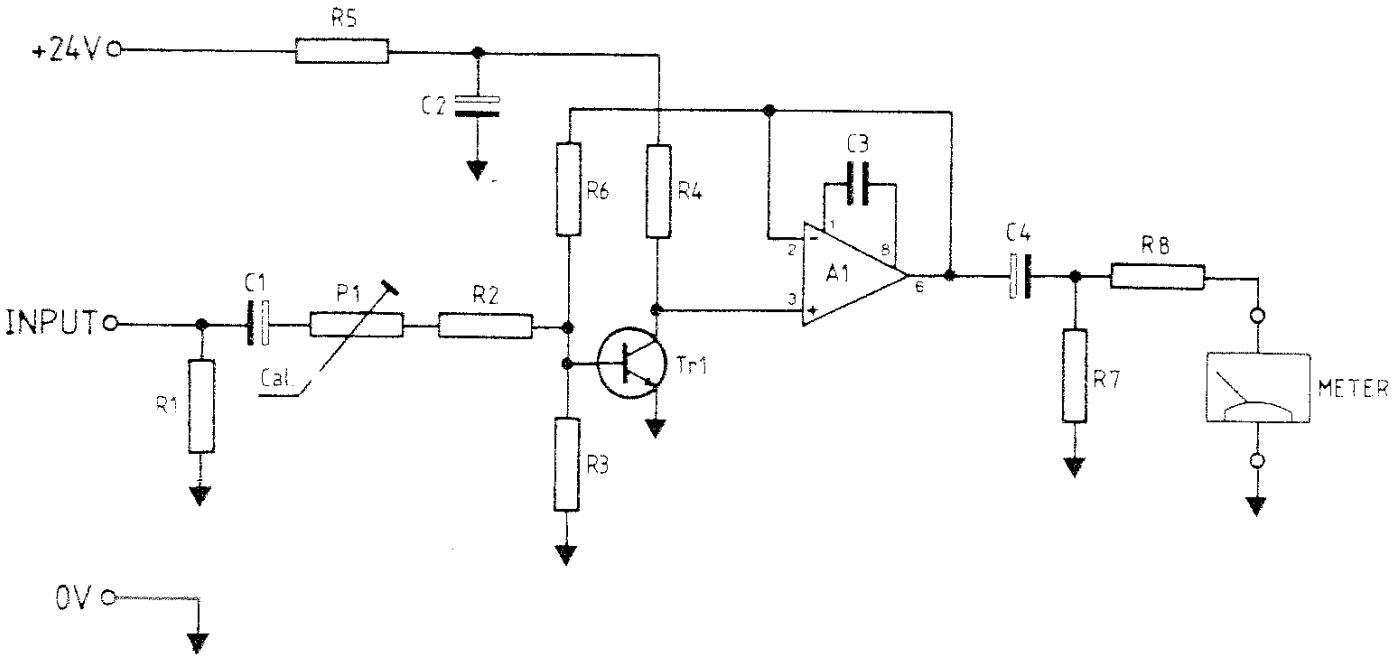
g) Repeat (d)

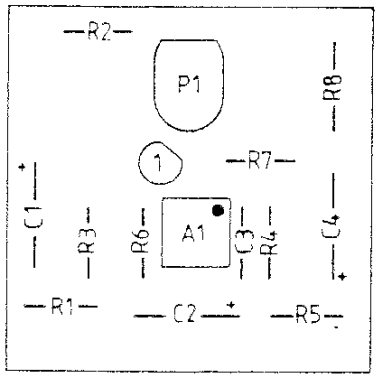
h) Repeat (b) & (c)

Repeating the above should be carried out until satisfactory readings are obtained.

Check overall scale calibration as final check.

V.U. METER DRIVER
CIRCUIT & LAYOUT

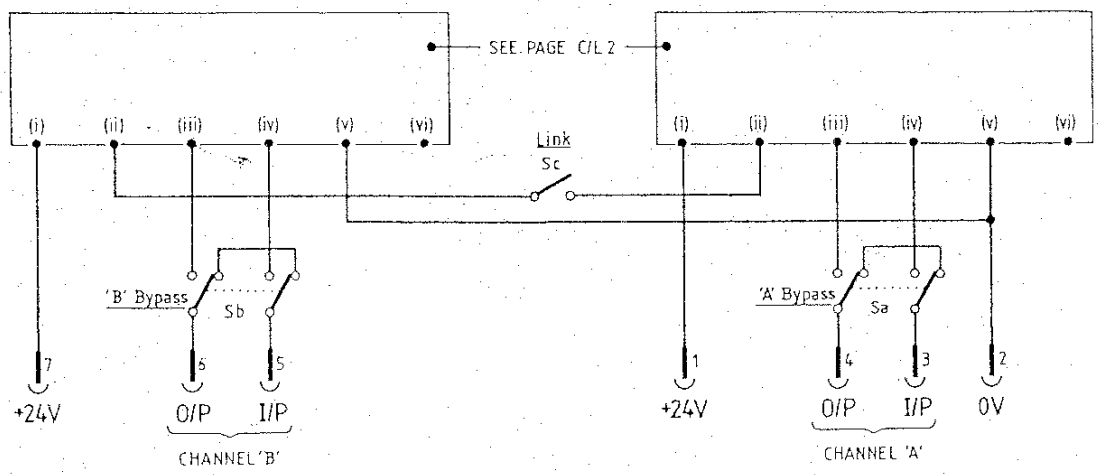


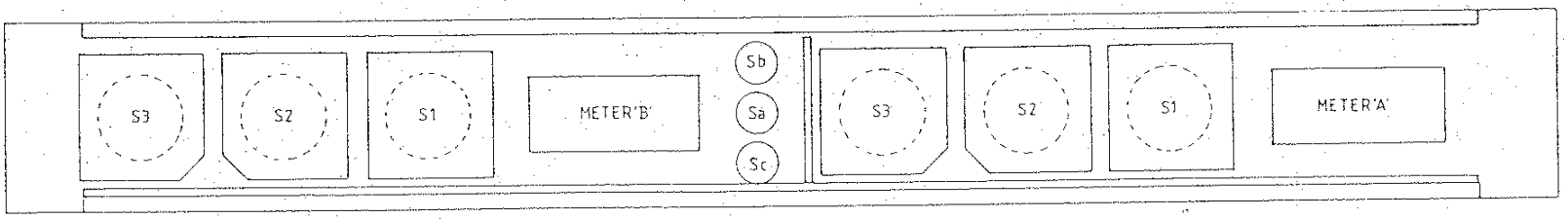


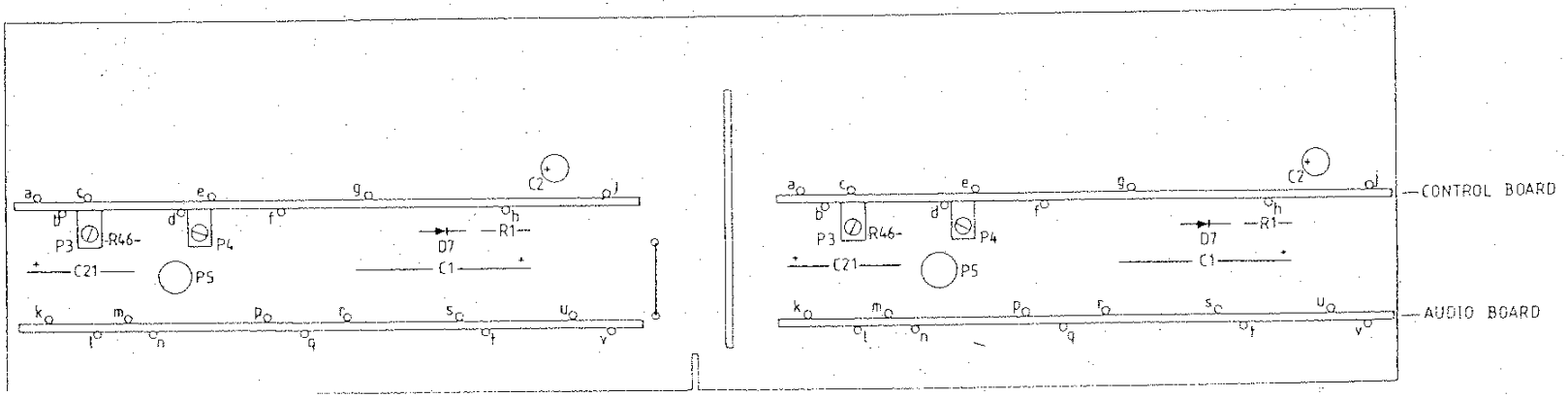
AD031

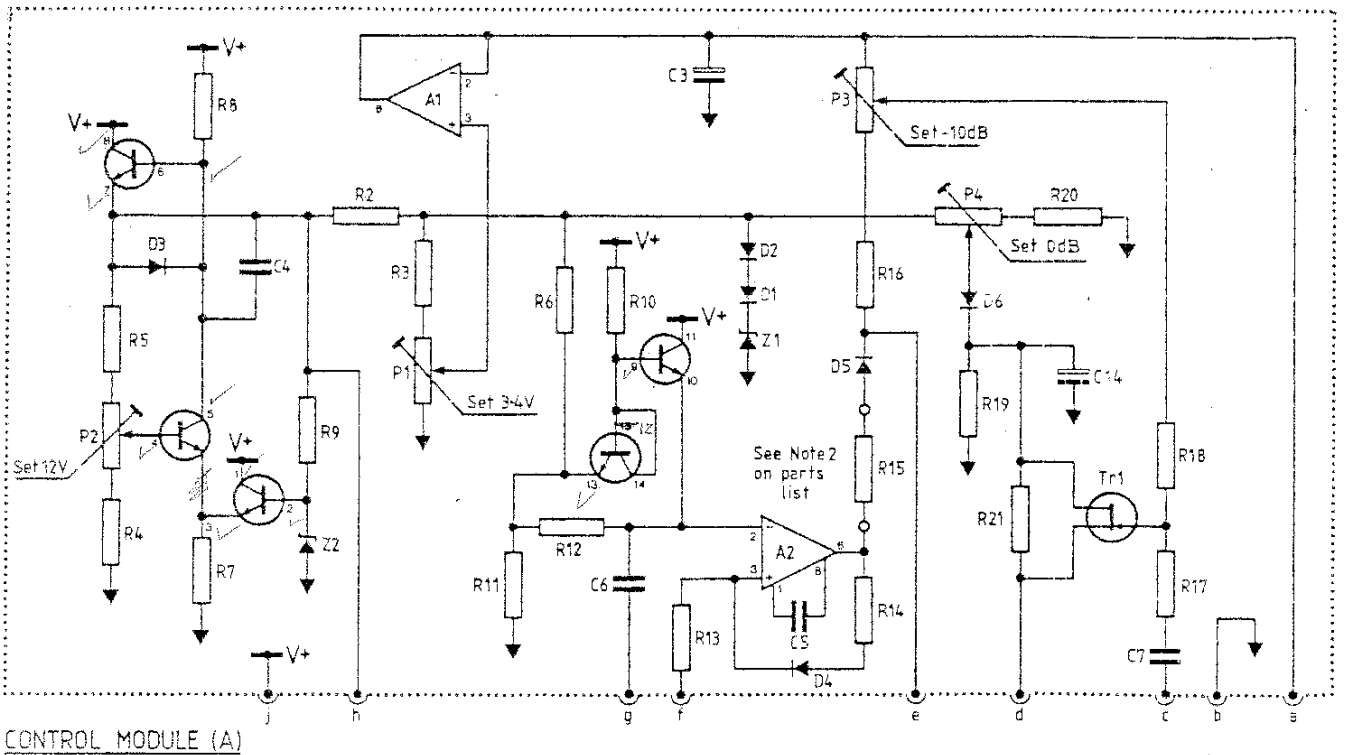
DUAL COMPRESSOR/LIMITER

GENERAL ARRANGEMENT

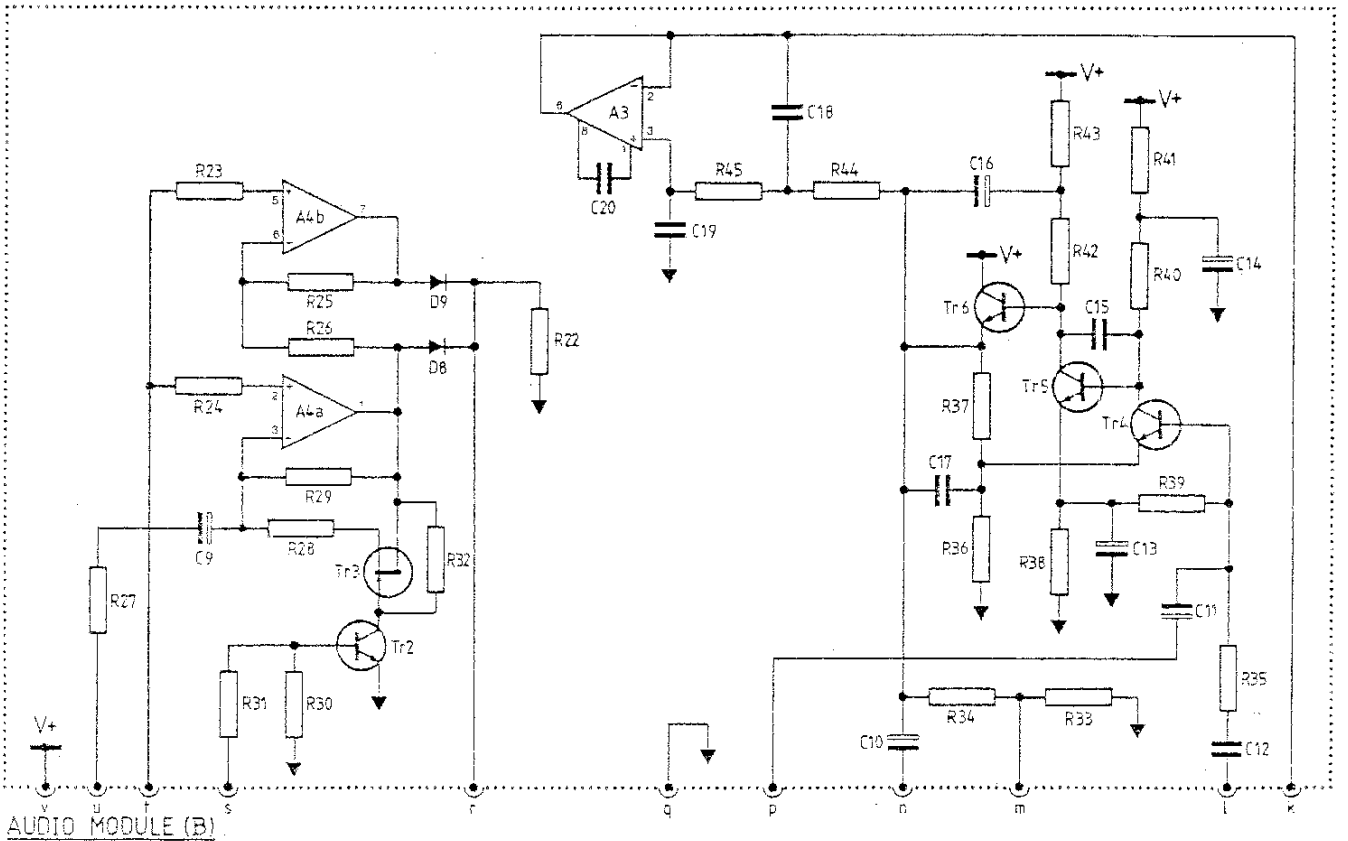




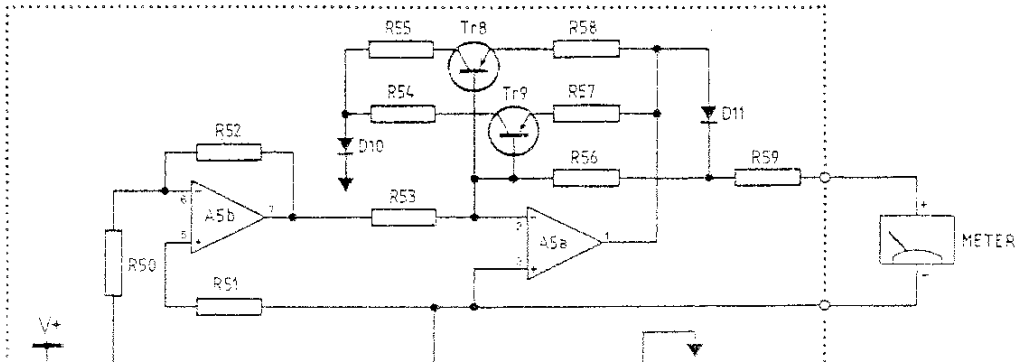




CONTROL MODULE (A)



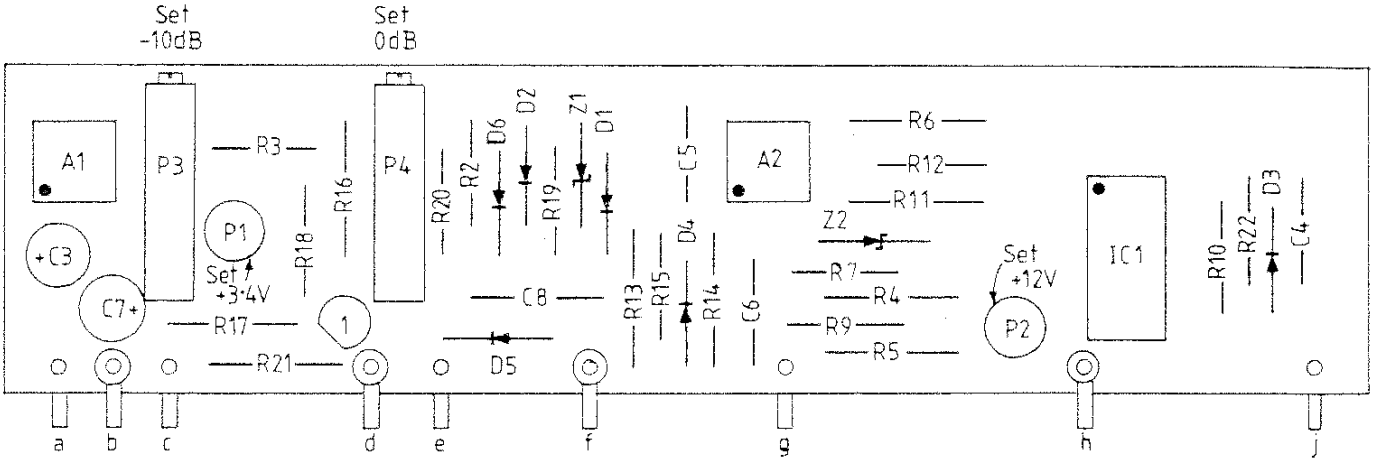
AUDIO MODULE (B)



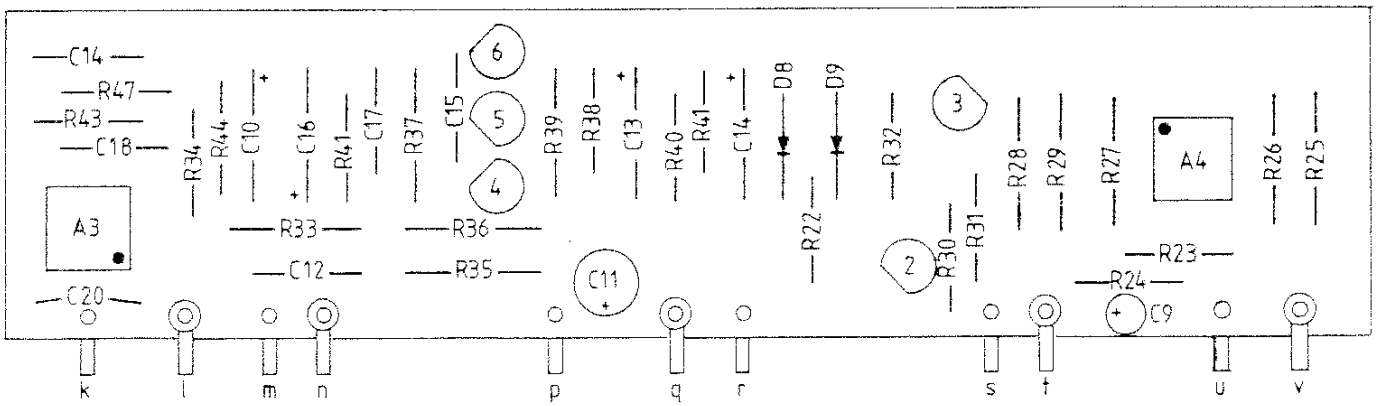
COMPRESSOR/LIMITER

COMPONENT LOCATION

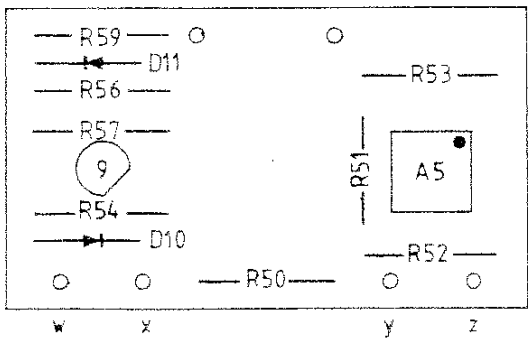
CONTROL MODULE (A)

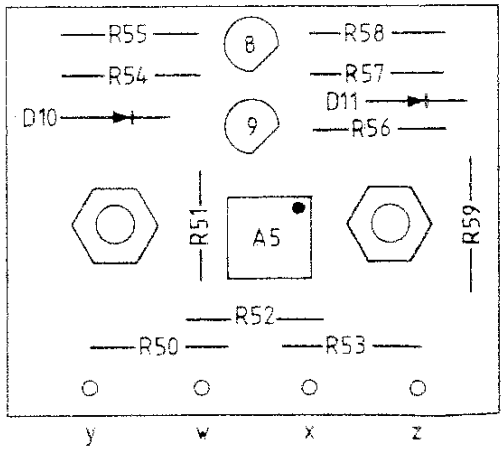


AUDIO MODULE (B)



METER DRIVERS





COMPRESSOR/LIMITER MODULE

SETTING UP & CALIBRATION PROCEDURE

EQUIPMENT NEEDED:

Digital Voltmeter
Distortion Factor Meter
Audio Oscillator (10Hz - 100KHz. <0.01% Distortion, + 20 dBm)
Audio Millivoltmeter.
Oscilloscope.

D.C. VOLTAGE CHECKS:

1. The main d.c. supply is 24v.
2. Bias voltage at test point 1; 12v ± 0.1v adjusted by P2.
3. Bias voltage at point (a); 3.4v ±0.5v adjusted by P1.
   (If two channels are involved, it is important that voltage is set accurately, to ensure correct stereo matching).

A.C. CALIBRATION:

1. Set OdB.
   (a) Set channel controls:- Threshold = +20dBm; Ratio = Limit
   Release = 300mS.
   (b) Feed in OdBm at 1KHz. Adjust P4 (Set OdB) so that output is OdBm.

2. Set -10dB.
   (a) Set channel controls:- Threshold = OdBm; Ratio = Limit;
   Release = 300mS.
   (b) Feed in +10.5dBm at at 1KHz.
   (c) Using the digital Voltmeter set the d.c. voltage between points (a) and (e) to 2.5v ±0.05v using P3 (Set -10dB), point (e) is the most positive.

3. DISTORTION.

   NOTE:- The oscillator used to drive the compressor/limiter must be of low enough distortion to be insignificant with respect to the distortion of the compressor.

   (a) Set channel controls:- Threshold = +20dBm; Ratio = Limit;
   Release = 300mS.

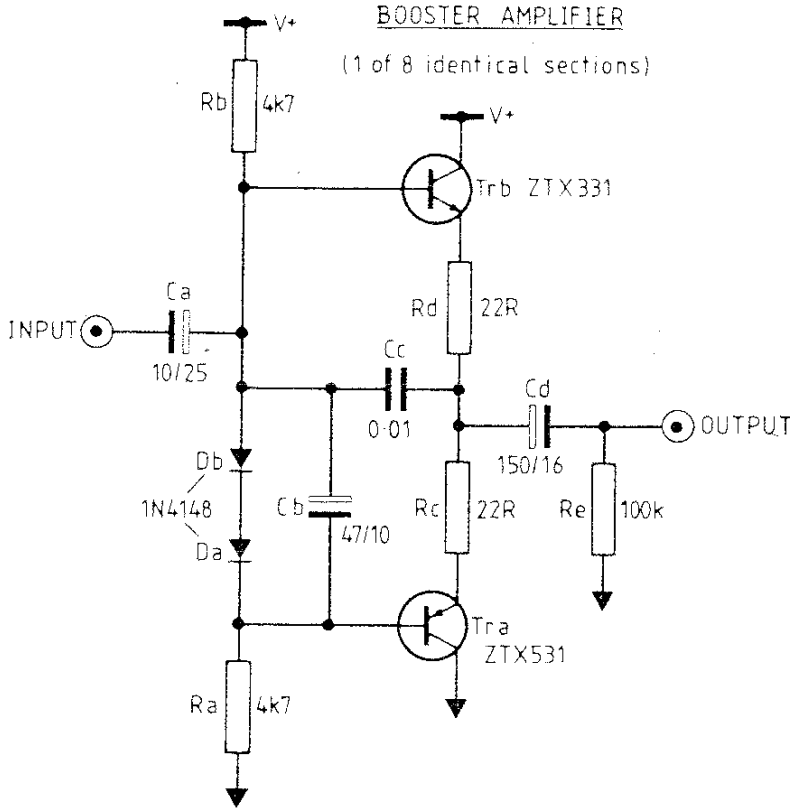
   (b) Drive compressor with Od m at 1KHz. Measure the distortion and set to a minimum with P5 control.

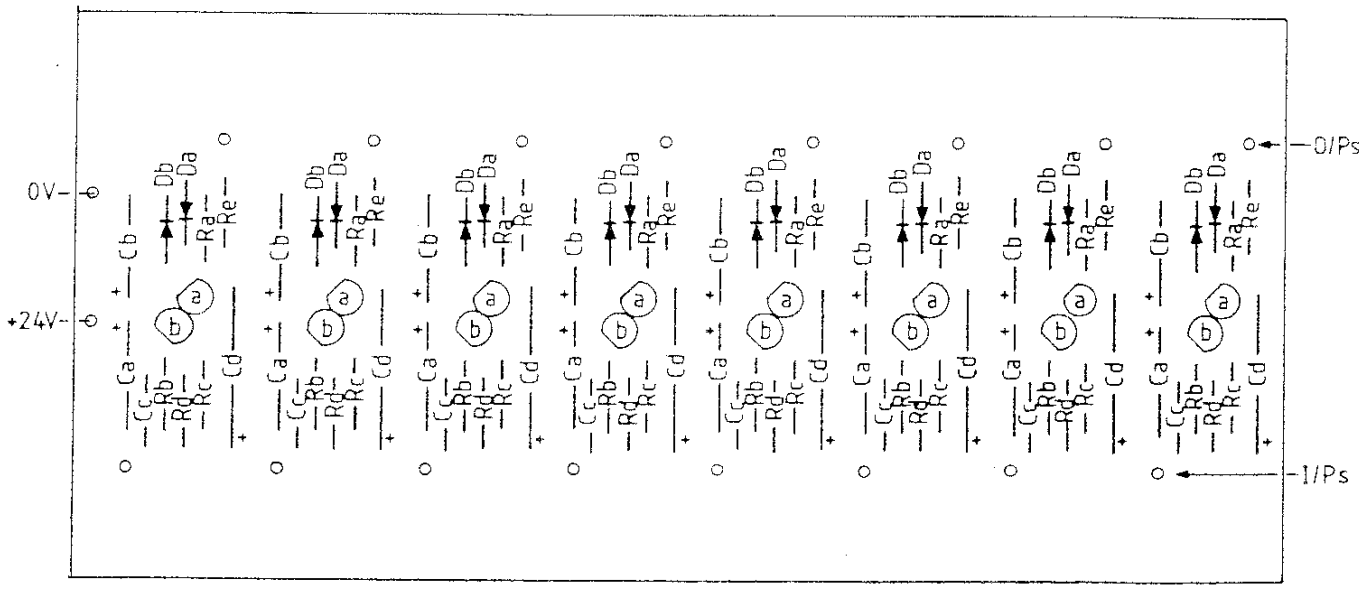
   NOTE:- If the distortion was very great, say 2%, this would alter the a.c. calibration, so this should be re-checked by repeating tests 1 and 2.

AD031 MicroMixer

BOOSTER AMPLIFIER

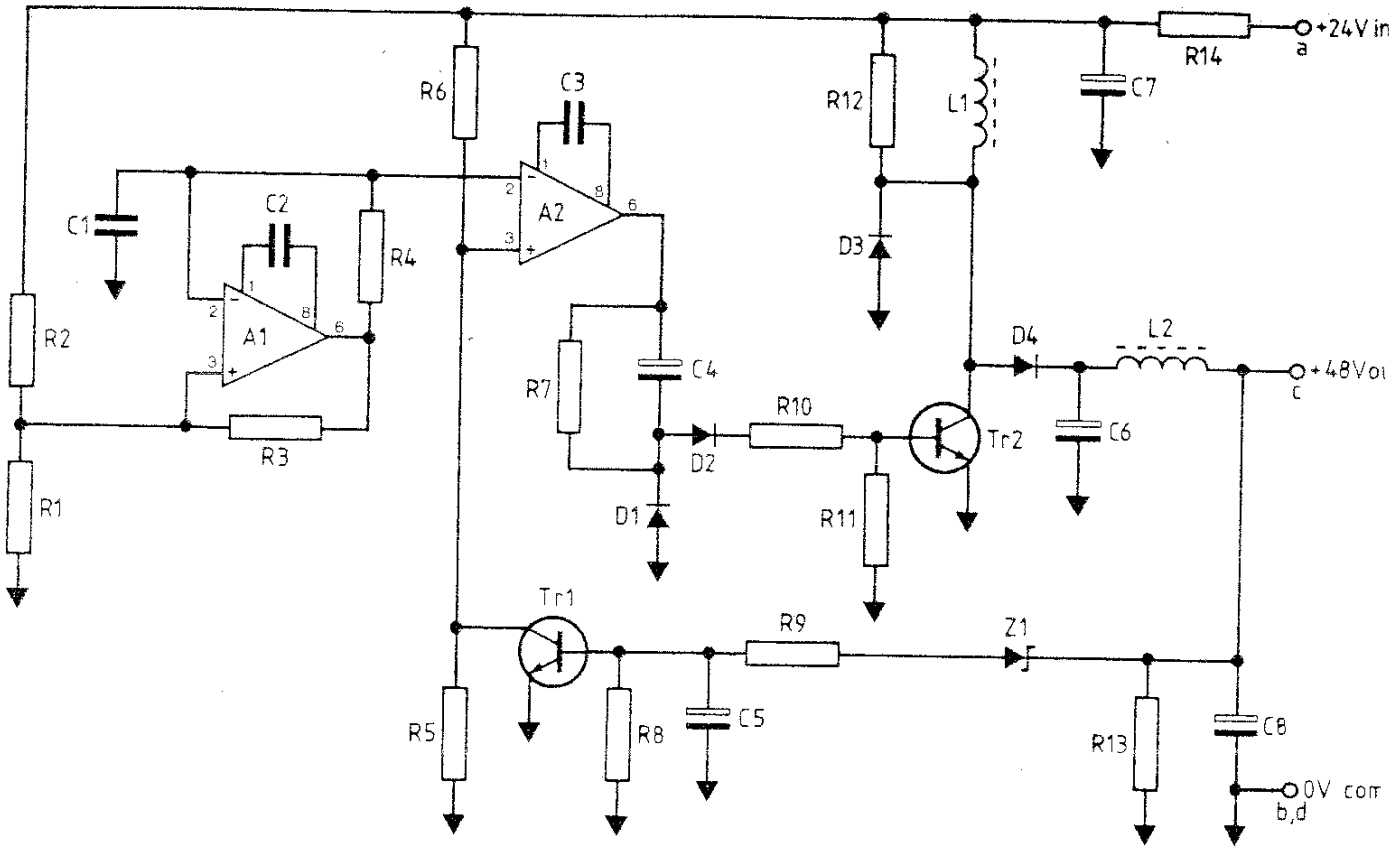
(1 of 8 identical sections)

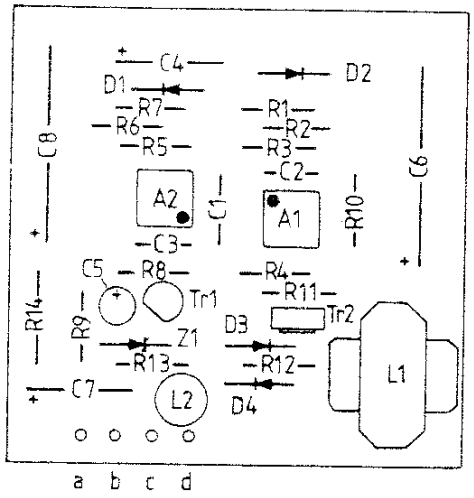




D.C./D.C. Converter

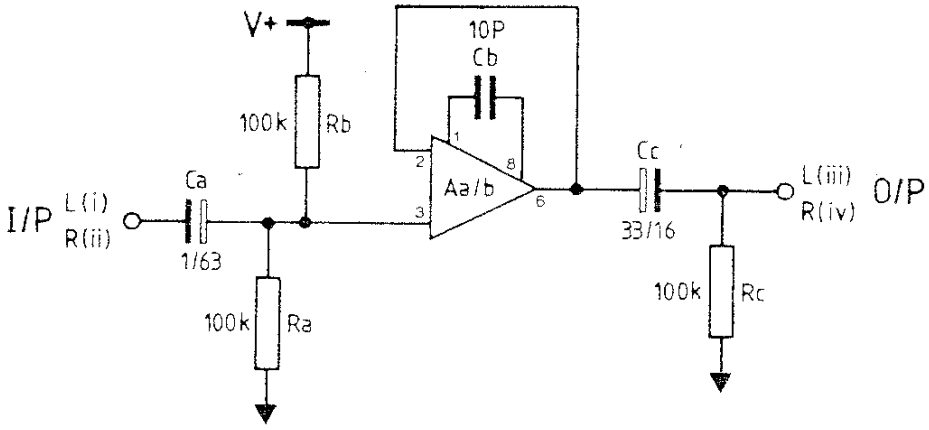
CON



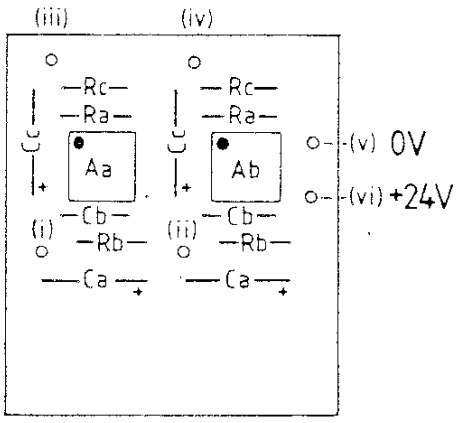


MONITOR BUFFER

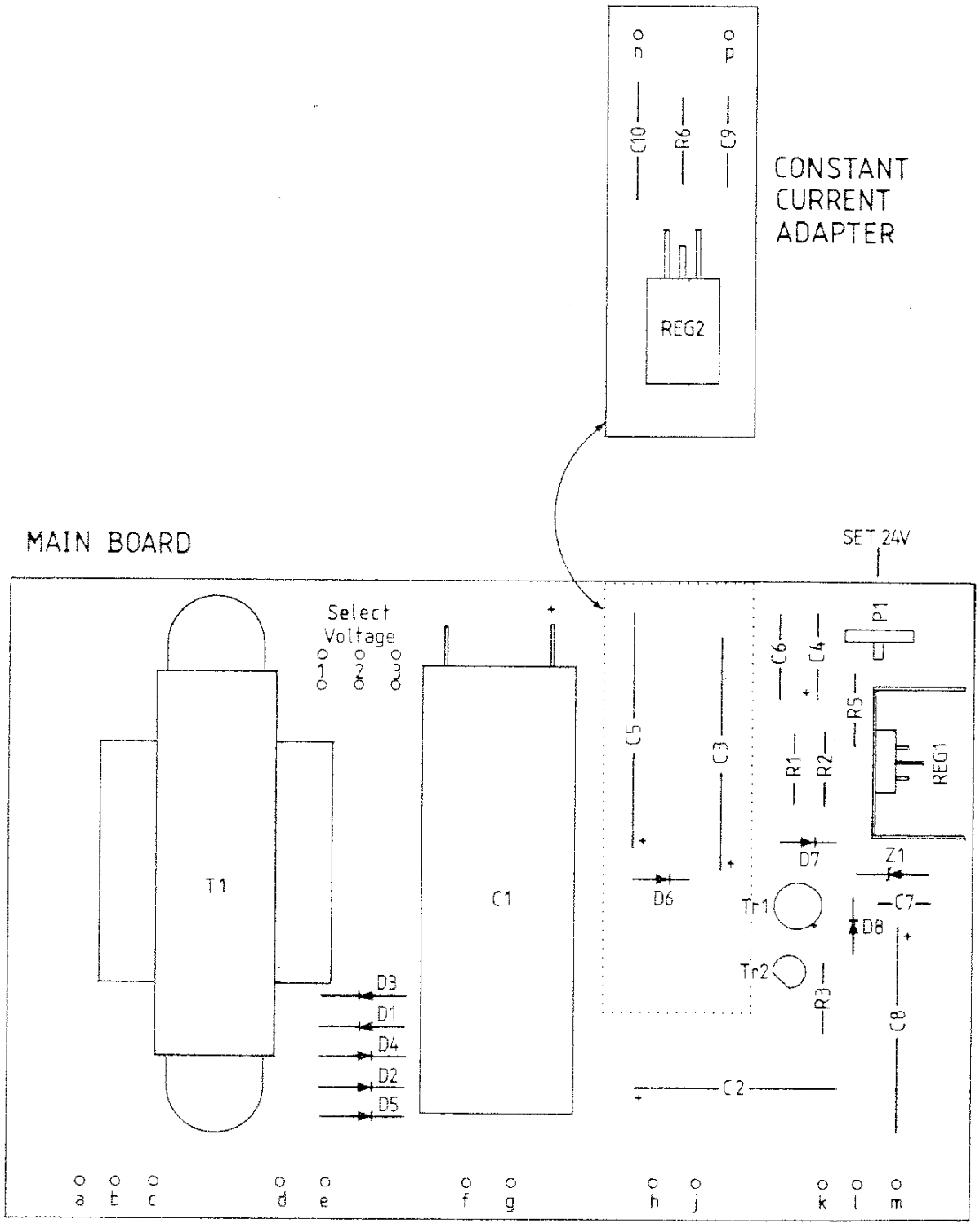
CIRCUIT & LAYOUT



[One of two identical sections]



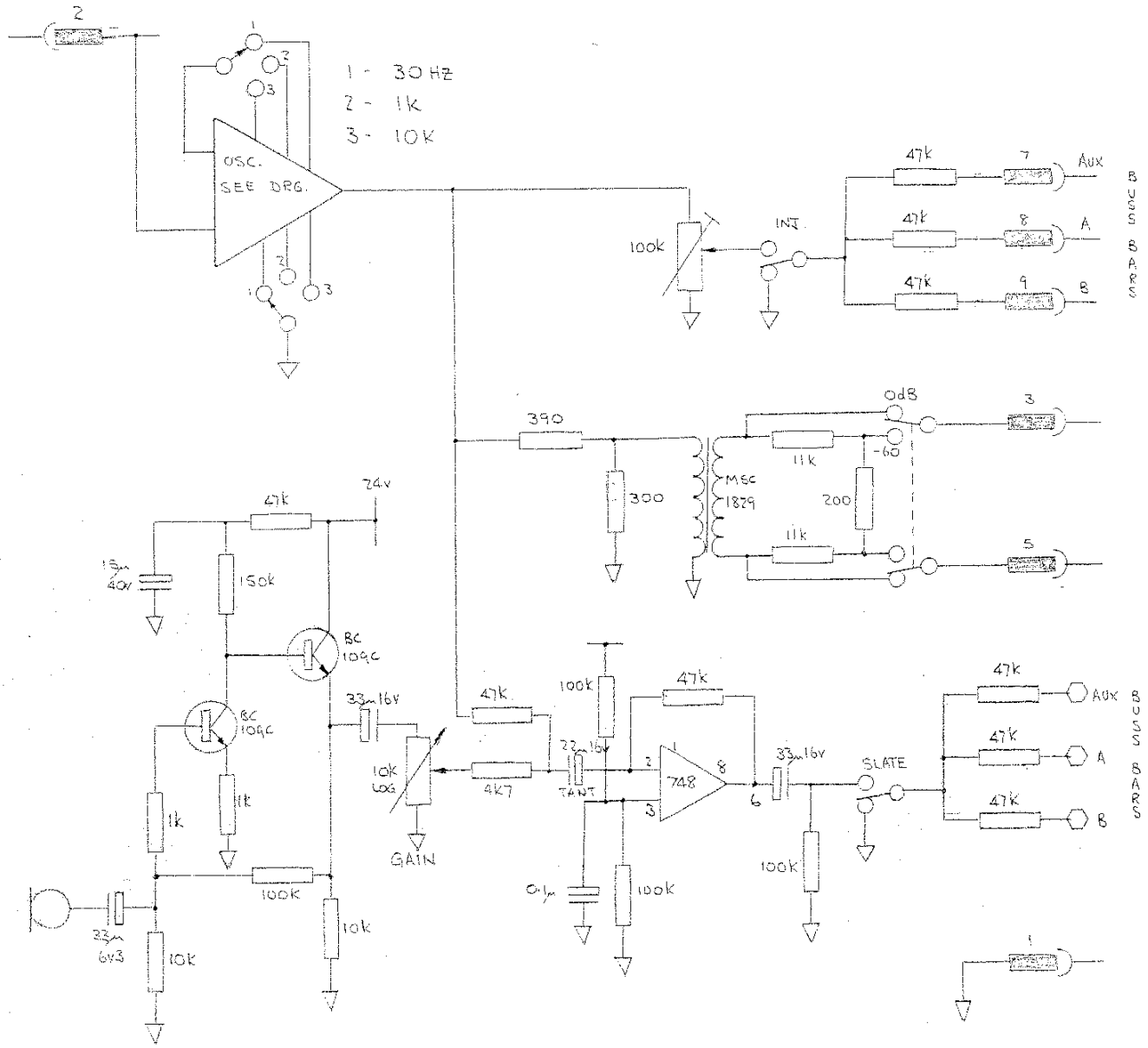
POWER SUPPLY
COMPONENT LOCATION



DRG. No.-
AD 031 D 71

THIRD ANGLE PROJECTION

DO NOT SCALE



5 PIN XLR	
PIN NO	FUNCTION
1	0 v.
2	TONE O/P LIVE
3	TONE O/P RETURN
4	TALKBACK O/P
5	0v

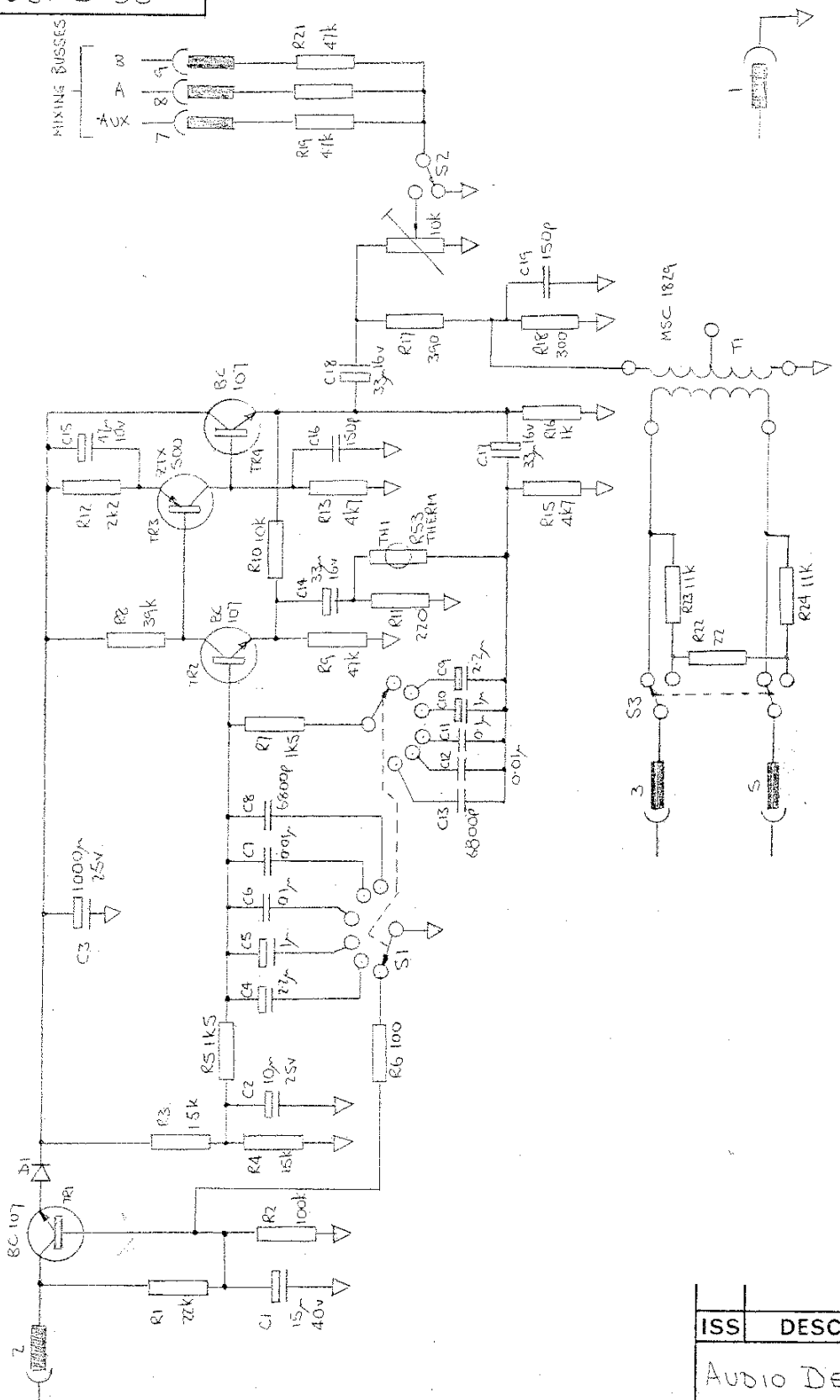
ISS	DESCRIPTION	SIG.

AUDIO DEVELOPMENTS,
Hall Lane,
Walsall Wood, Staffs

Drawn R.F.T.	Scale	Unstated Tolerances	AD 031 MICRO MIXER.
Checked	Material	2 Dec. Pl. ±.010	CIRCUIT DIAGRAM TONE & TALKBACK MODULE.
Approved	Finish	3 Dec. Pl. ±.005	
		Fractions ± 1/64	
		Metric ±2mm	

DRG. No.
AD 031 D 71

DRG. No.	THIRD ANGLE PROJECTION	DO NOT SCALE
AD 031 D 66		



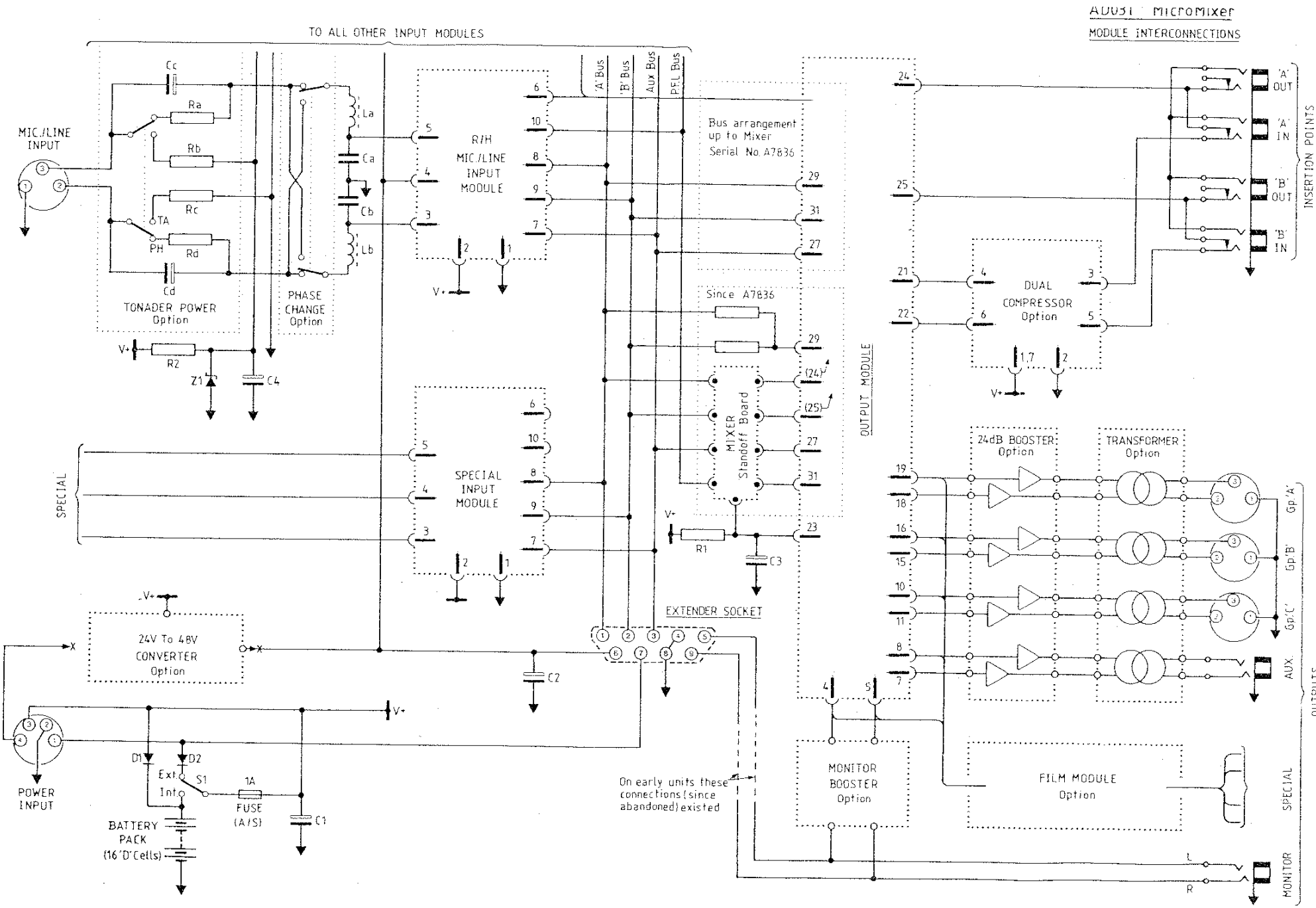
ISS	DESCRIPTION	SIG.

AUDIO DEVELOPMENTS.

Hall Lane,
Walsall Wood, Staffs

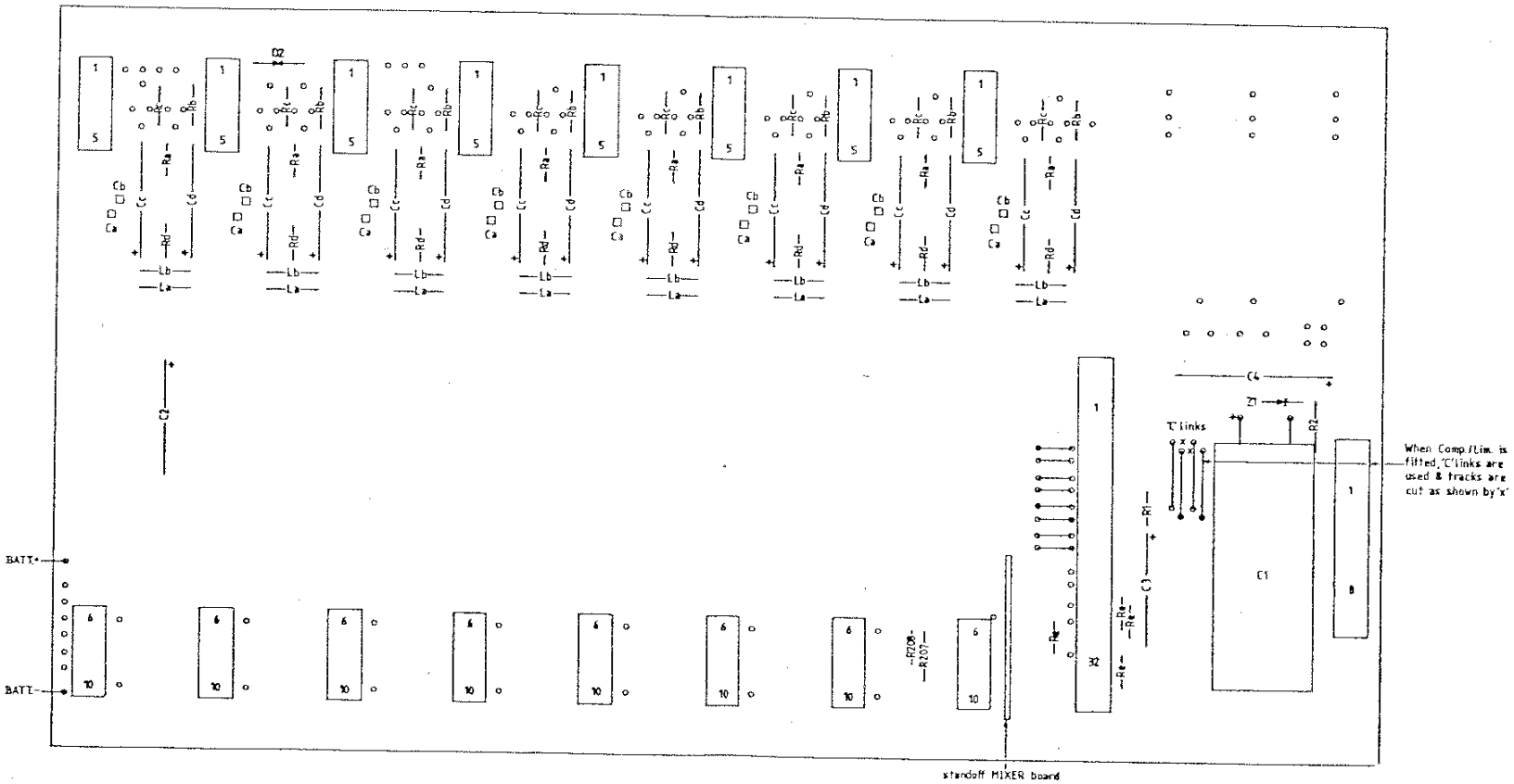
Drawn	Scale	Unstated Tolerances	AD031 MICRO MIXER
RFT		2 Dec. Pl. ±·010	TONE MODULE
Checked	Material	3 Dec. Pl. ±·005	CIRCUIT DIAGRAM
Approved	Finish	Fractions ± 1/64	
		Metric ±·2mm	

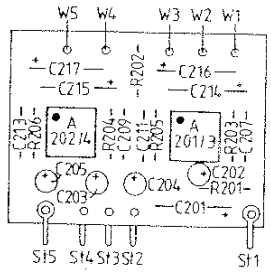
DRG. No.
AD 031 D 66


AUDIX MicroMixer
MODULE INTERCONNECTIONS
TO ALL OTHER INPUT MODULES
MIC./LINE INPUT
Cc
Ra
Rb
Rc
Rd
TA
PH
Cd
TONADER POWER Option
PHASE CHANGE Option
La
Ca
Cb
Lb
R/H MIC./LINE INPUT MODULE
V+
R2
Z1
C4
'A' Bus
'B' Bus
Aux Bus
PFL Bus
Bus arrangement up to Mixer Serial No. A7836
Since A7836
MIXER Standoff Board
R1
C3
SPECIAL
SPECIAL INPUT MODULE
OUTPUT MODULE
EXTENDER SOCKET
24V To 48V CONVERTER Option
C2
POWER INPUT
D1
D2
Ext.
Int.
S1
1A
FUSE (A/S)
BATTERY PACK (16 'D' Cells)
C1
On early units these connections (since abandoned) existed
MONITOR BOOSTER Option
DUAL COMPRESSOR Option
24dB BOOSTER Option
TRANSFORMER Option
FILM MODULE Option
'A' OUT
'A' IN
'B' OUT
'B' IN
INSERTION POINTS
Gp.'A'
Gp.'B'
Gp.'C'
AUX
SPECIAL
MONITOR
L
R
OUTPUTS

AD031

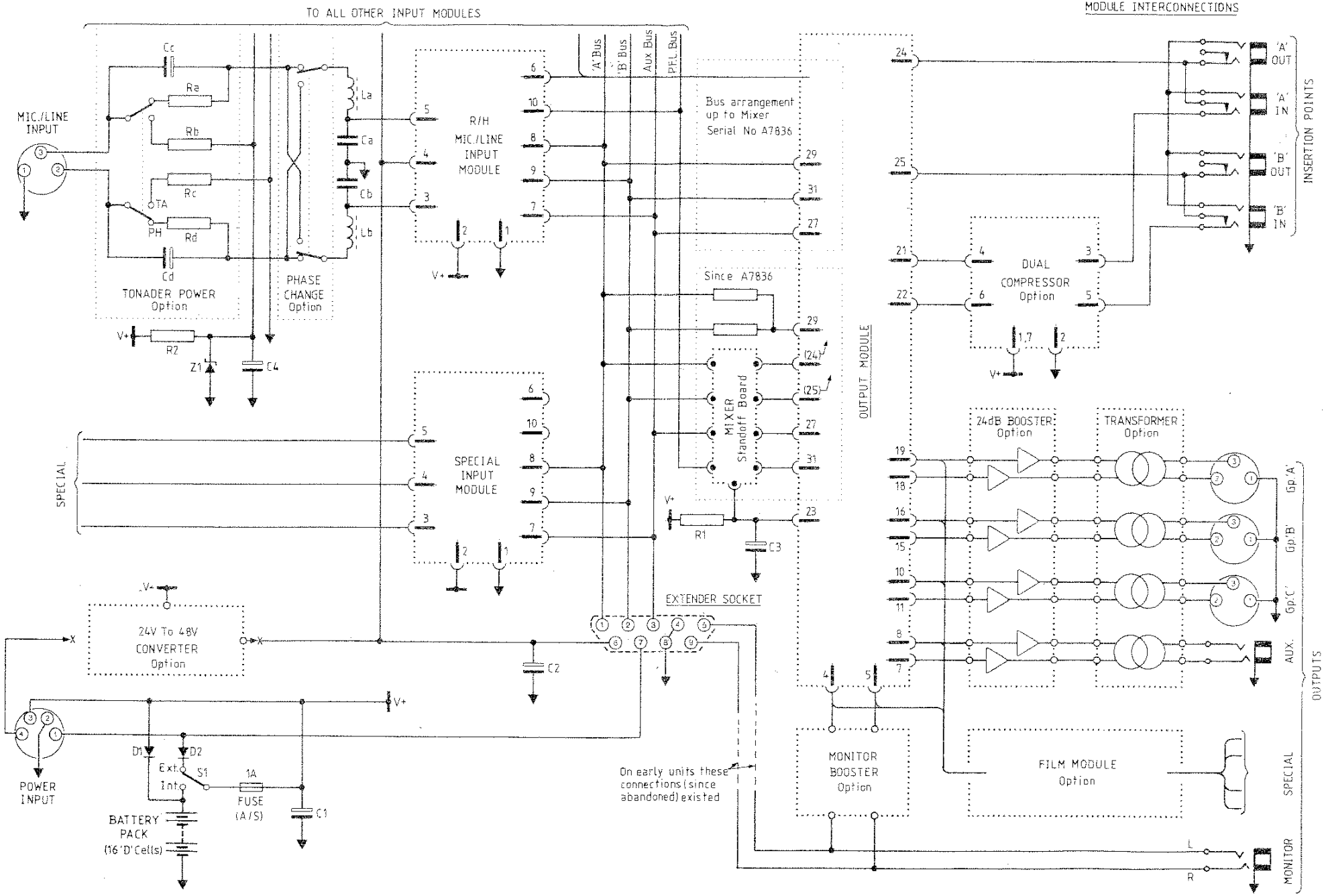
MOTHER-BOARD LA



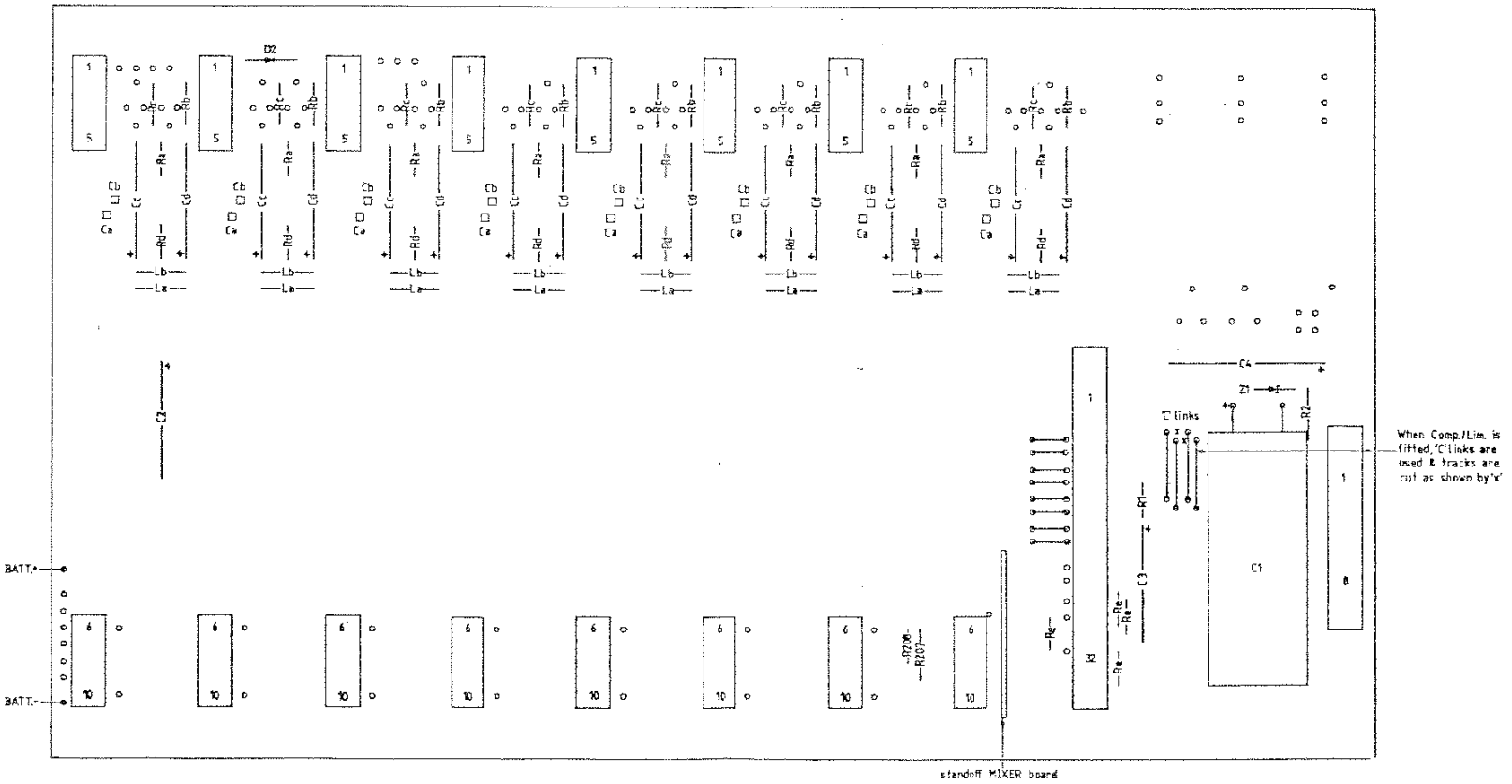


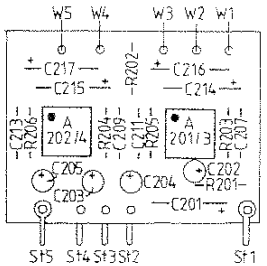
AD031 MicroMixer

MODULE INTERCONNECTIONS



MOTHER-BOARD LAYOUT





DUAL COMPRESSOR/LIMITER

GENERAL ARRANGEMENT

