



# assembly manual

## Technical Data DX 400/500

AM 334 1st Edition



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## A PURPOSE OF THE MANUAL

This manual, part of the DX 400/500 manual series, complements the other manuals in the series by presenting the technical principles behind the operation of the DX. 400/500 organs. It is not necessary to know the information in this manual to assemble the organ! However, it is possible to gain a better understanding of the musical possibilities of the organ by learning the technical aspects of its operation. Also, knowing such information, can be a big help if you ever---hopefully never---have to troubleshoot the organ.

The technical data is presented in a manner that is aimed more at the interested layman than the electronic specialist. Rather than going into fine detail about how each circuit functions, the text gives more of an overview of how the various subassemblies work. Some acquaintance with microcomputer terminology will help in understanding the principles presented. The manual also presents practical procedures for addressing the question, "What to do if. -

## B. TECHNICAL CONCEPT

### I. Traditional Organ Systems

To put the new **DX** digital organ concept in proper perspective, we will first briefly look at three approaches to electronic organ technology.

#### 1. Analog Organs

In this type---and this is the type most commonly used---sine wave tones are produced by a tone generator, selected by mechanical or electronic switches, processed in filters to produce the desired voicing, amplified and applied to a speaker.

Of interest here is that it is the hardware (the system components) alone that determines the capabilities of such an instrument.

#### 2. Quasi Digital Organs

These organs are touted as being 'digital;' especially in the promotional literature, but a closer look shows that they do not merit this tag. To be sure, they employ digital support techniques---serial data transmission, for example---along with complex IC circuitry. But the tones are still developed and processed mainly by analog means.

#### 3. Digital Systems Using Original Sounds in Memory

Committing original musical sounds to a digital memory is a widely known technique used in digital synthesizers and rhythm units. In moderately priced instruments, reflecting moderate technical sophistication, the sounds of instruments such as trumpets, violins and pianos are stored exactly as those of drums, cymbals or cowbells---in short, single tones.

These original tones are read out when the instrument is played and are assigned to the desired instrument range according to a clock frequency controlled from the keyboard. A digital-to-analog converter changes the digital data back into audible analog tones. The tones sound rather genuine when played in the tonal range of the original instrument; however, the tones become less and less true the further the player deviates from this range. This is because the tone formants cannot be held constant over the entire range of the keyboard.

### II. The DX System With Full Digital Tone Processing

In this new Wersi technology, a microprocessor system computes all the sounds and digital-to-analog conversion

- produces audio tones from the resulting data.
- The processor system consists of a master processor and up to
- five co-processors, known as slaves.

The slaves are responsible for the correct processing of the tones, receiving essential data such as pitch, volume (envelope amplitude), frequency, vibrato, formants, etc. from the master processor. The master processor sends new data to the slave anytime there is a change in input status, whether it is a registration change or a new key stroke.

The outstanding advantage of this system is that a wide range of musical forms can be achieved, even though the hardware never changes. It is the software---data programs read into the organ from a cassette recorder or a computer---that creates so many musical possibilities. The data in the organ's memories can be added to or changed completely, giving the organ a whole new repertoire, if desired. Sounds from liturgical organ to

synthesizers to conventional musical instruments are possible through software.

A further advantage of this system is that a given sound quality, once achieved and filed as software, will be the same when reproduced by an organ in the series, since hardware variations will have virtually no effect on the sound properties.

Additional special effects can be achieved through further processing of the digitally developed voices via a voltage-controlled filter (VCF) and a phase vibrato (Wersivoice) circuit.

Apart from its contrast to other organ systems, the DX concept offers the most possibilities in terms of sound and functions, and is the most flexible concept for the electronic production of individual musical voices yet presented.

### III. The DX System Block Diagrams

#### 1. Block Diagram of the Complete Organ

Fig. 1 is a block diagram of the complete DX series organ; the following is a brief explanation of the organ's functions:

The master central processing unit (CPU), located on the master processor board, is the control center of the organ. The CPU obtains, via the peripheral bus, all the information it needs to control the immediate production of any sound the player selects; this is because the CPU continuously polls the instantaneous status of all selector switches, drawbars, controls, and keyboard and pedal contacts. And while it's doing that, it also sends back the necessary signals to illuminate the light-emitting diodes (LED's) in the activated switches.

The CPU transforms the incoming data into the appropriate instructions to send via the master bus to circuits such as the VCF, slaves, etc., which generate analog (audio) signals based on these instructions. From there on out, it's all analog; the audio bus sums all the audio signals, which then are amplified and applied to the speaker to produce the desired sounds.

#### Block Diagram of the Slave Processor

Fig. 2 is a block diagram of a slave processor. The DX organ uses up to five slave processors to produce sounds.

Each slave generates as many as four free-programmable complex voices, each with two simultaneous sound components. The coordination of these, via the master processor, can arbitrarily stem from the upper manual, lower manual, auto-accompaniment or pedals.

Since every voice consists of two envelope-shaped tone components each, eight audio channels are required to produce four voices. By this means, all voices can be constructed from two components, for example, draw-bars + percussion, piano + strings, even smooth string tones + the plucked tones of a guitar.

At the moment a keyboard or pedal key is depressed, the master CPU loads a "slave program," as well as data which will set the sound parameters, into the slave processor's 2-port random access memory (RAM) via the master bus. The bus switch then transfers the RAM data lines to the slave bus during the slave's internal processing sequence.

To enable a maximum transfer of data, the 2 MHz clock frequencies of the master CPU and the slave CPU are shifted a half cycle out of phase with each other. Accordingly, large blocks of data-for complex voices, for example-can be transferred between the processors at a simulated 4 MHz rate. The bus switch is constantly switching the RAM from the master bus to the slave bus and , consequently, both processors continuously receive RAM control data.

After the master CPU loads the RAM, the slave CPU carries out the internal slave program. Five timers set up the readout speed according to the program. Four timers determine the pitch (according to the readout of the RAM sound parameter tables) and the fifth determines the repeat frequency for the envelope control count.

The direct memory access (DMAC), under control of the timeout information and the sequence logic, enables a direct readout of the RAM sound parameter table apart from CPU influence.

The sequence logic, via a write (WR) signal, loads the direct-read digital data for four 2-component voices, one after the other, into eight signal digital-to-analog (DAC) converters. On a data transfer (XFER) signal, the signal DAC's convert the digital data into analog signals. The interval memory loading and the direct timer-controlled readout of the digital data result in a jitter-free analog signal of outstanding purity, not affected by the internal cycling.

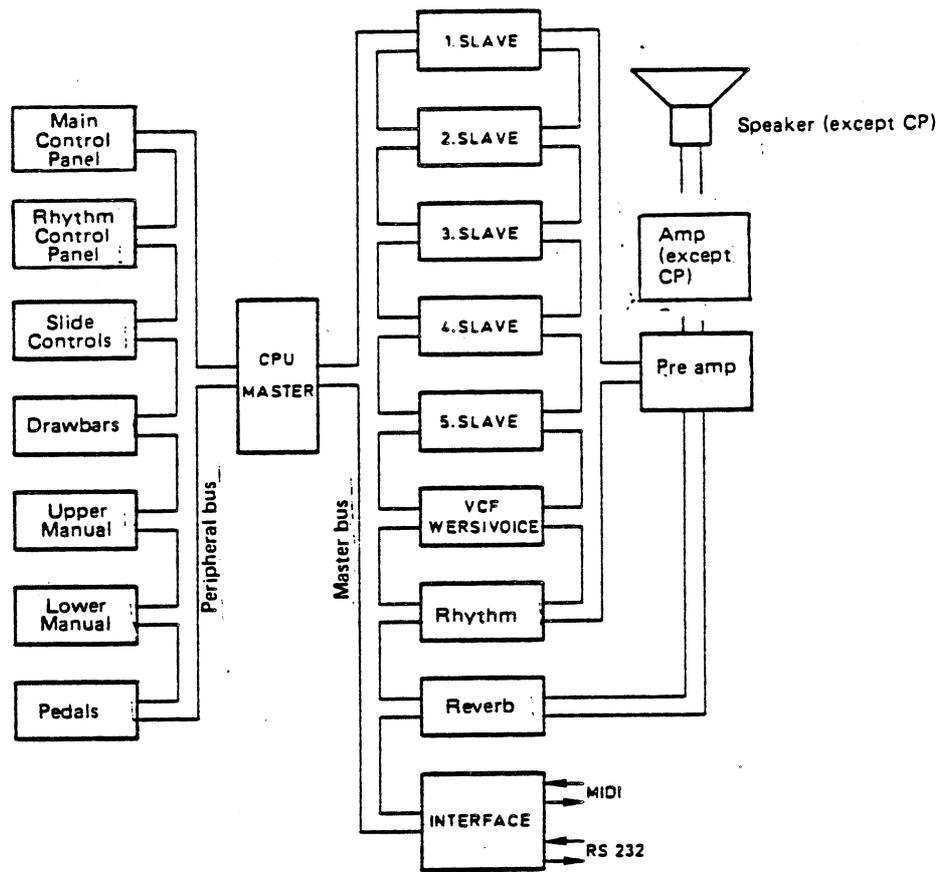


Fig. 1: Block diagram of the DX 400/500 organ series

At the same time, the envelope control DAC (left, Fig. 2) converts digital data to envelope control voltages for the eight voice components. The 8-channel multiplex sample and hold circuitry routes these voltages to the signal DAC's, which, as multiplying DAC's, directly set the volumes of the instantaneous analog signals.

The crosspoint matrix routes the analog (audio) signals to the desired audio bus channels.

### 3. Audio Block Diagram

The audio signals from the slaves are routed to five audio channels, Fig. 3, where they undergo various types of processing:

- a) Superdeemphasis (superlowpass) - For bass voices,
- b) Deemphasis (lowpass) channel - For "round" tone voices, such as drawbars
- c) Direct (bright) channel - Slave audio passes unchanged to the amplifiers
- d) VCF channel - Processing for VCF effects such as wah wah
- e) Wersivoice channel - Complex multiple voice processing; string choir, for example

The functional sequencing of the VCF and Wersivoice is controlled by the master CPU.

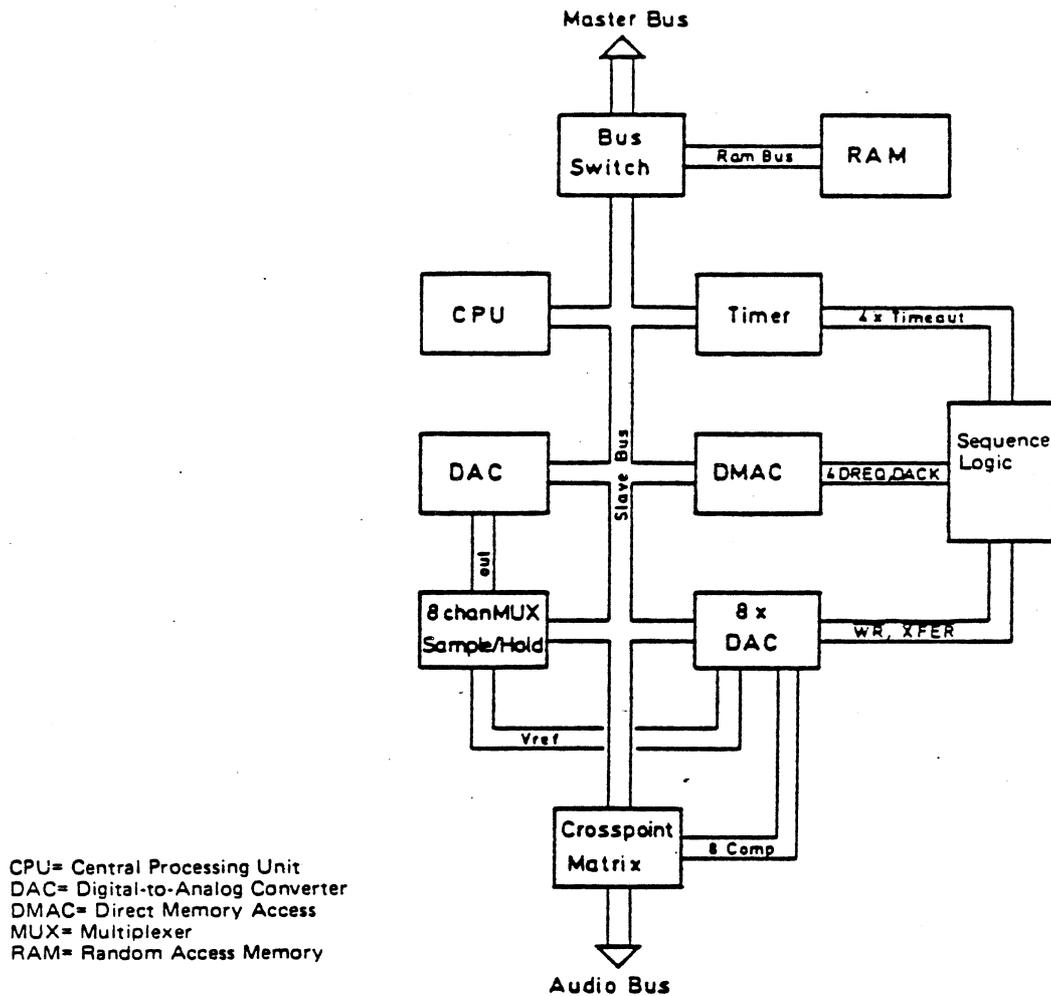


Fig. 2: Block diagram of a slave processor

#### 4. Digital Rhythm Instruments

Although the functional sequencing of the rhythm and accompaniment is completely controlled by the master CPU, the actual sound generation takes place on the instrument board, which carries the digitally memorized instruments. The instrument board has its own CPU to control the internal processing. This CPU, in accordance with its program, periodically polls its input registers to determine whether and which instruments should sound. The resulting digital information then goes to a digital-to-analog converter (DAC), which builds it into an instrument sound analog signal. This signal then goes through a lowpass/bandpass filter and on to an instantaneous volume control circuit.

#### 5. Digital Reverb Unit

The master CPU controls the reverb modes short/long, fast echo/slow echo.

The digital reverb functions as a processor-supported "real time" system, whose CPU controls the analog-to-digital (ADC) conversion, the digital display and the digital-to-analog (DAC) conversion.

#### 6. Interface

The interface enables the organ to exchange data with:

- Similar instruments via the musical instrument digital interface (MIDI)
- Home computers via the RS232 interface
- A cassette recorder via the tape interface

To enable such transfers, the parallel digital data from the master bus is converted to serial data, under control of the CPU. Incoming serial data, conversely, is converted to parallel data. For cassette loading or storage, a modem converts the serial data to audible signals and vice-versa.

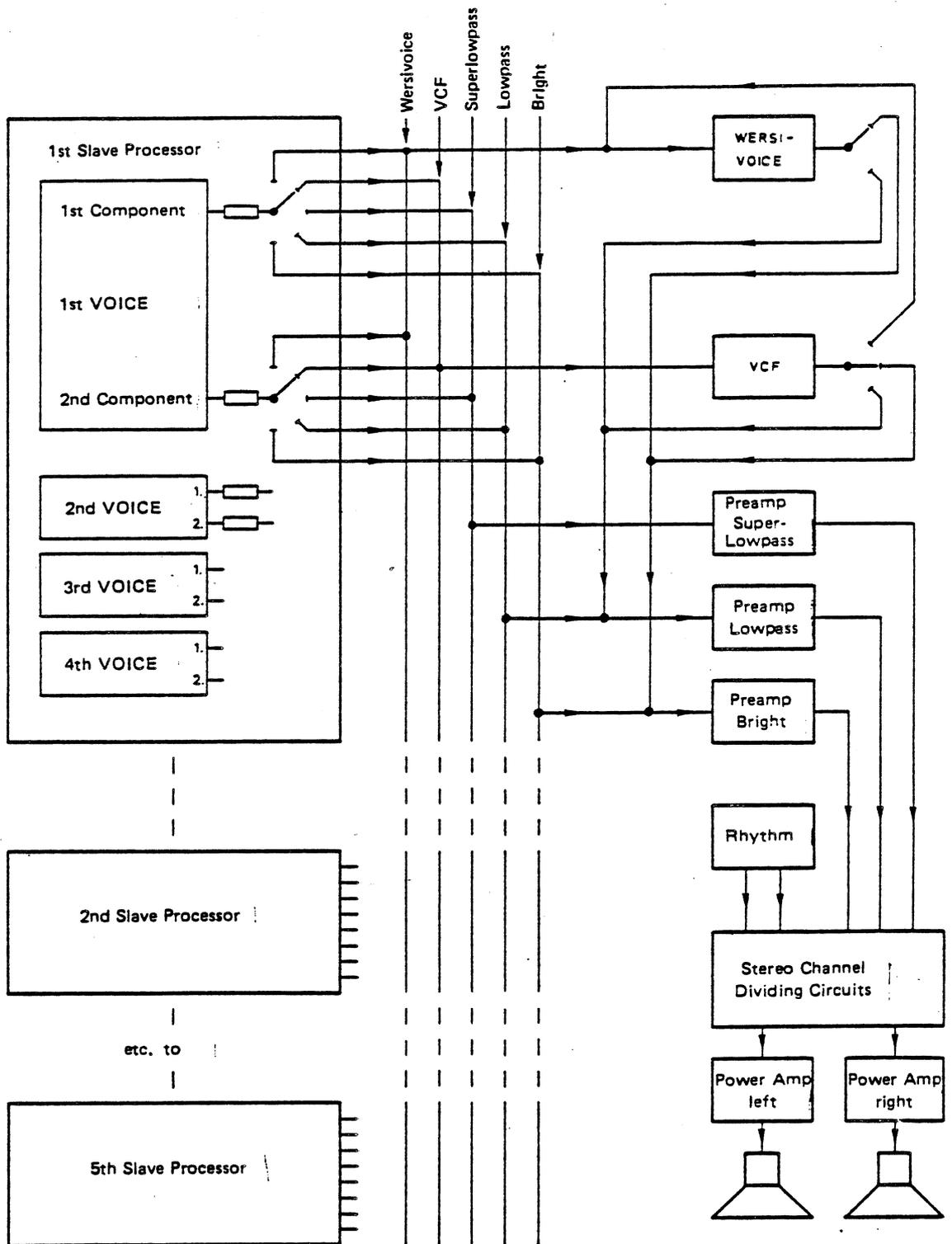


Fig. 3: Block diagram of the audio section

## C. SCHEMATIC DIAGRAM

In this chapter you will find detailed schematic diagrams, principles of operation and useful notes on adjustments and tests. To help you find the desired information easier, the schematics are grouped as follows:

- I. Central Electronics
- II. Control Panels
- III. Peripheral Boards

### I. Central Electronics

This section includes the pin layout of the backplane board MB 30, detailed schematic diagrams of all the plug-in boards which mate with MB 30 and component layout diagrams with functional cal louts.

#### 1. Backplane Board MB 30

This PC board has three basic functions:

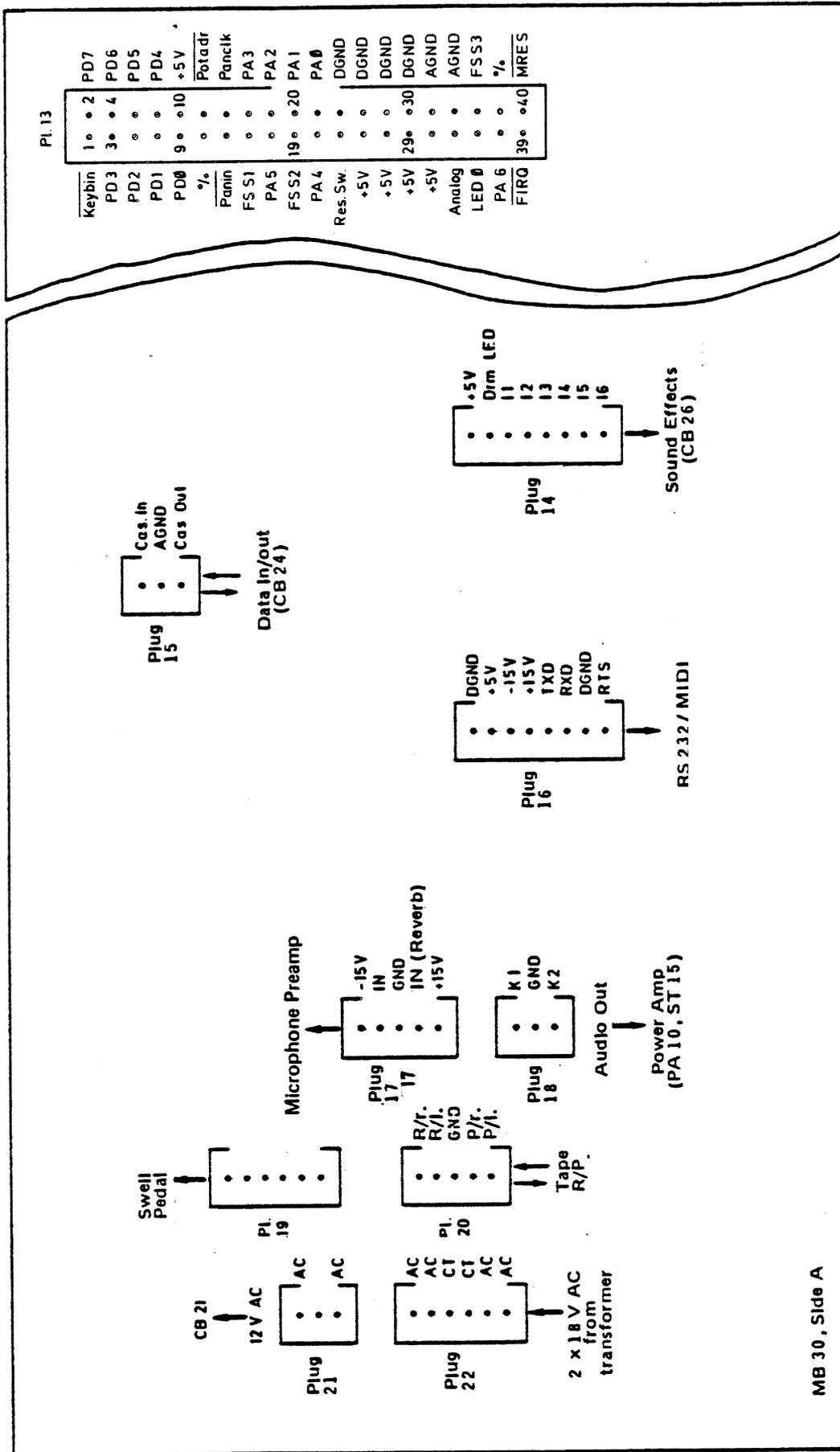
- It is a mechanical support for all the plugin boards
- It connects the boards together electrically
- It carries all the connectors for wiring the peripheral boards via ribbon cables

The accompanying table shows the pin layout of the edge connectors on the MB 30; these are the access points to the plug-in boards. The diagram shows the layout of the ribbon cable edge connectors on MB 30; these are the access points to the "outside world."

Testing or troubleshooting a plug-in board is made easier by plugging the board into the extender board EXT 1 and - important - then inserting the EXT 1 in the correct slot for the board being tested. Except for the five slave boards, which can be interchanged in their designated slots, no plug-in board must ever be plugged into any slot other than that designated for it.

PS 12	AF 12		DH 10		EF 20		EF 19		DDB3	SL 3 (15)		SL 3 (14)		SL 3 (13)		SL 3 (12)		SL 3 (11)		MST 3		
	b	a	b	a	b	a	b	a		b	a	b	a	b	a	b	a	b	a	b	a	b
1 AGND	1 AGND	1 AGND	1 AGND	1 AGND	1 AGND	1 AGND	1 AGND	1 AGND	1 AGND	1 AGND	1 AGND	1 AGND	1 AGND	1 AGND	1 AGND	1 AGND	1 AGND	1 AGND	1 AGND	1 AGND	Q3	1 PD7
2 AGND	2 AGND	2 AGND	2 AGND	2 SDE	2 AGND	2 AGND	2 AGND	2 AGND	2 AGND	2 AGND	2 AGND	2 AGND	2 AGND	2 AGND	2 AGND	2 AGND	2 AGND	2 AGND	2 AGND	2 AGND	PD3	2 PD6
3 DE	3 DE	3 DE	3 DE	3 PA.5 E	3 J1	3 DE	3 DE	PD2	3 PD5													
4 Reset	4 NDE	4 Inp. Hall	4 WVNDE	4 TXD	4 J2	4 NDE	4 NDE	PD1	4 PD4													
5 Reset Sw	5 BASS	5 Out Hall 1	5 WV	5 O.2	5 AFRH1	5 WV	5 WV	Reset	5 PD0													
6	6 WVNDE	6 AGND	6 VCF	6 NON	6 AFRH2	6 VDF	6 VCF	PanelK	6 Keybin													
7 AGND	7 WV DE	7 AGND	7 AGND	7 AGND	7 AGND	7 AGND	7 AGND	7 AGND	7 AGND	7 AGND	7 AGND	7 AGND	7 AGND	7 AGND	7 AGND	7 AGND	7 AGND	7 AGND	7 AGND	7 BASS	PA0	7 Panin
8 AGND	8 VCF NDE	8 AGND	8 AGND	8 Inh.	8 AGND	8 AGND	8 AGND	8 AGND	8 AGND	8 AGND	8 AGND	8 AGND	8 AGND	8 AGND	8 AGND	8 AGND	8 AGND	8 AGND	8 AGND	8 AGND	PA1	8 PA4
9	9 VCF DE	9 Out Hall 2	9 PA.5 E	9 PC.6	9 DCS	9 VCF	9	PA2	9 PA5													
10	10 AFRH1	10 Hall Ein 2	10 DR Vol.	10 DAC	10 MA.6	10 DAC	10	PA3	10 PA6													
11	11 In Hall	11 Hall Ein 1	11 PA.4 E	11 PA.4 E	11 J3	11 PA.4 E	11	MRD	11 ANIN													
12	12 O. Hall 1	12 Aux. Hall	12 O.1	12 MWR	12 J4	12 O.1	12	MWR	12 MRES													
13	13 O. Hall 2		13 O.2	13 MD0	13 MD0	13 O.2	13	MD0	13 MRES													
14 AGND	14 In Hall		14 O.3	14 MD1	14 MD1	14 O.3	14	MD1	14 MA0													
15 AGND	15 Micro		15 NON	15 MD2	15 MD2	15 NON	15	MD2	14 MA1													
16	16 Mic. Hall		16 Inh.	16 MD3	16 MD3	16 Inh.	16	MD3	15 MA2													
17 AC1	17 TB W li		17 PA.6 E	17 MD4	17 MD4	17 PA.6 E	17	MD4	15 MA2													
18 AC1	18 TB W re		19 PC.6	18 MD5	18 MD5	19 PC.6	18	MD5	16 MA3													
19 CT1	19 TB A li		20 PC.7	19 MD6	19 MD6	20 PC.7	19	MD6	17 MA4													
20 CT2	20 TB A re		21 PB.1	20 MD7	20 MD7	21 PB.1	20	MD7	17 MA4													
21 AC2	21 PAK1		22 PB.2	21 MA.8	21 J5	22 PB.2	21	ME	18 MA5													
22 AC2	22 PAK2		23 PB.3	22 Aux.H.	22 J6	23 PB.3	22	ME	18 MA5													
23	23 FS		24 PB.4	23 PA.6 E	23 DRUM LED	24 PB.4	23	MQ	19 MA6													
24	24 DR Vol.		25 PB.5	24 SE	24 MA.7	25 PB.5	24	MQ	20 MA7													
25 +5V	25 RE Vol.		26 PB.6	24 FIR0	24 MA.8	26 PB.6	25	SE	20 MA7													
26 +5V	26 Reset		27 PB.7	25 Q.3	25 MA.8	27 PB.7	26	SQ	21 MA8													
28 DGND	28 +15V		28 +15V	26 RXD	26 SE	28 +15V	27	SQ	21 MA8													
29 DGND	29 +5V		29 +5V	27 /	27 RESET	29 +5V	27	SQ	22 MA9													
30 -10V	30 SDE		30 -10V	28 +15V	28 +15V	30 -10V	28	SOE	22 MA9													
31 -15V	31 -15V		31 -15V	+5V	29 +5V	31 -15V	29	SOE	23 MA10													
32 DGND	32 DGND		32 DGND	-10V	30 -10V	32 DGND	30	SOE	23 MA10													
				31 -15V	31 -15V															31	SOE	24 MRes
				32 DGND	32 DGND															32	SOE	24 MRes

BACKPLANE BOARD MB 30, EDGE CONNECTOR PIN LAYOUT, VIEWED FROM SIDE A



Backplane board MB 30, plug layout

## 2. Plug-in Board PS 12 - Power Supply

The plug-in board PS 12 develops all the voltages required to run the DX series organs:

- + 5 V - switched and regulated, pins 25 & 26
- 10 V - electronically regulated, pin 30
- 15 V - electronically regulated, pin 31

All voltages are referenced to GND (=0 V), present at pins 1,2,7,8, 14, 15,28,29 and 32 of PS 12.

**Note: The boards in the DX series strongly differentiate between A GND (= analog ground) and D GND (= digital ground), even though both initially occur together on PS 12.**

### Principles of Operation

Fig. 5 is a schematic diagram of PS 12. The AC voltage from the power transformer (2 x 18 V AC), which enters the board via pins 17 thru 22, is rectified by bridge rectifier BR 1 - Unregulated DC voltages PWR + and PWR - are present across filter capacitors C14/15 and C1 2, respectively. (These are used as supply voltages for the integrated power amplifier on PC board AF 11 or AF 110 when the PS 12 is used with the Alpha series).

IC3 and IC4 are voltage regulators, developing highly regulated, short circuit-protected -15 V DC and +15 V DC supply voltages, respectively. These voltages are indicated by the green and yellow light-emitting diodes (LEDs), respectively. Zener diode ZD4 and resistor R23, wired across the -15 V line, develop an additional supply voltage of -10 V DC.

Most of the circuitry on PS 12 (over half of the Fig. 5 schematic) is devoted to developing the +5 V DC supply voltage. The source voltage is the PWR + line (25-30 V DC). The circuit is a switched power supply, keeping power losses to a minimum.

Q4 functions as a switch transistor. It is driven by Q2 and Q3, which are in turn driven by IC1. When Q4 is switched through, memory choke Li builds up a field and electrolytic capacitor C16 charges up until the output voltage (pins 25/26) reaches +5 V DC. The red LED indicates this.

During Q4's shutdown period, the charge on C16 maintains the supply current, while diode D4 serves to maintain the current flow resulting from the collapsing field around Li.

The regulation of the output voltage is achieved by controlling the duty cycle of the switching pulses, which IC1 sends to switching transistors Q2 and Q3. IC1 receives the reference voltage for this from IC2 (pin 2). If the supply voltage at pin 1 of IC 1 varies from the reference voltage at pin 2, the internal operational amplifier (IC1, right) produces a corresponding correction voltage to restore the correct duty cycle.

Further regulation is achieved by current sensing. The left operational amplifier in IC1 senses the voltage drop across R22 and compares it to a reference voltage at pin 16. This results in corresponding correction voltages.

The remaining circuit components mainly serve to protect the power supply and the circuits it is supplying. For example, as and zener diode ZD3 work together to prevent an overvoltage condition on the +5 V DC line. If the voltage rises above 5 volts, ZD3 conducts, turning on thyristor Thi This places a short circuit-across the +5 V DC line, which will remain until the power supply is turned off momentarily.

Zener diode ZD1 stabilizes the supply voltage for IC2, which, besides producing the reference voltage mentioned above, also produces a reset pulse during organ turn-on. This pulse momentarily mutes the audio amplifiers on PC board AF 12 until the digital organ system has stabilized.

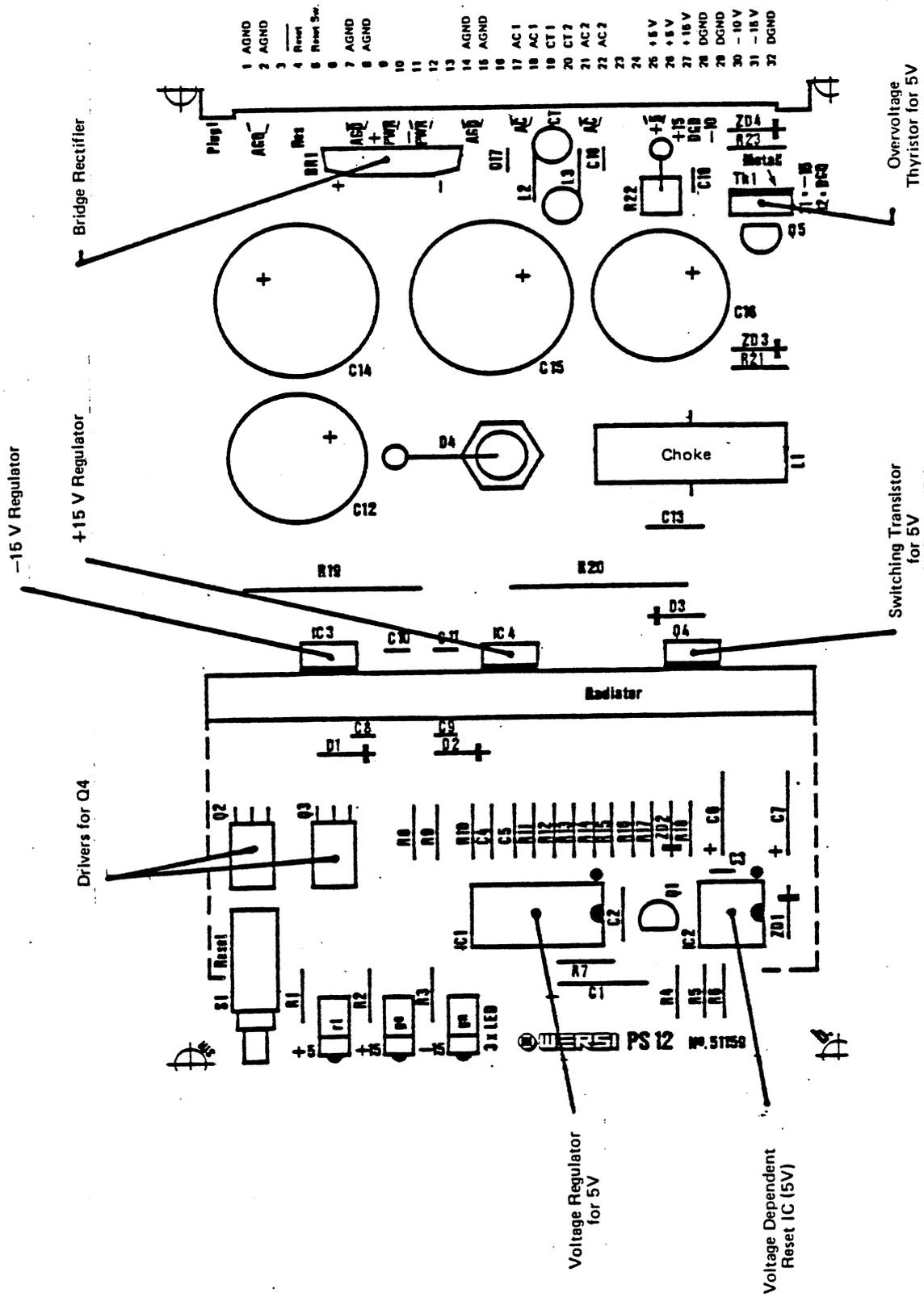


Fig. 4. Plug-in board PS 12, component layout and functions



### 3. Plug-in Board AF 12 - Preamplifier

Plug-in board AF 12 contains analog processing circuitry for all the organ's audio tone signals as well as tone signals from external sources. AF 12 apportions the signals between the two audio channels as shown in the following table:

Signal	From Input (pin no.)	Via IC	To Channel
Slave DE <sup>1)</sup>	3	3	A+B
Slave NDE <sup>2)</sup>	4	3	A
Slave SDE <sup>3)</sup>	5	3	A+B
W-Voice NDE	6	2	A+B
W-Voice DE	7	2	A+B
VCF NDE	B	2	B
VCF DE	9	2	B
Rhythm A	10	4	A
Rhythm B	11	4	B
Reverb A	12	6	A
Reverb B	13	6	B
Microphone	15		A+B
Mic. Reverb	16	5	A+B
Tape Playb. I.	17	1	A
Tape Playb. r.	18	1	B
Tape Record	19	-	fmA
Tape Record r.	20		fm.B

- 1) D E - Deemphasis = Lowpass
- 2) NDE - Nondeemphasis Practically linear
- 3) DE - Superdeemphasis - Super lowpass

#### Principles of Operation

Fig. 6 shows the component layout of plug-in board AF12 and Fig. 7 is its schematic diagram. All the audio tone signals developed in the organ enter AF 12 via pins 3 thru 9 and go to the two sections of control amplifier 1C2. The three slave signals (pins 3, 4, 5) are initially

amplified by the four sections of 1C3. The gains of 1C2's amplifiers are dependent on a DC control current from Q3 applied to pins 1 and 16. This control current, in turn, is dependent upon the position of the swell pedal (pin 23); the phototransistor in the reflex coupler of PC board OS 2 produces an output current whose level is determined by the pedal position. Trimpot PS determines the point of maximum volume (visually indicated by the LED fed by IC5b). The minimum volume level with a fully retracted pedal can be set via P6.

The signals from 1C2 pins 8 (channel A) and 9 (channel B) go via summing buses A and B to summing amplifiers 1C4c and d. (Microphone audio is applied to the summing buses via R101 and R103, rhythm via R98 and R105, and reverb via R99 and R104). The summing amplifiers are quieted when field effect transistors (FETs) Q1 and Q2 are conducting. This occurs when, for example, the organ is turned on and (as explained in PS 12) a reset pulse via D4 turns off the normally conducting Q6; FETs Q1 and Q2 then "see" a negative (less positive) potential via D1 and R134; this drives them in-to conduction and they effectively become low resistance shunts across their respective amplifiers. The processor system also quiets the summing amplifiers, via D3, during processing runs which may cause noise in the audio system.

The outputs of the summing amplifiers go via R4 and R6 to the tone control network built around IC1 - Audio to be recorded is taken off before the tone control network via R51 and R52. Tape playback audio is applied to the network via R31 and R5. Trimpots P1 and P3 control the bass, P2 and P4 control the treble. The outputs on IC1's pins 8 and 14 are the processed audio signals to be sent to the power amplifiers (cf. Fig. 36).

The rhythm signals (cf. Fig. 15) are amplified by IC4a and b, and go via control amplifiers 1C7 as well as R98 and R105 to summing buses A and B. This portion of the signal is independent of the swell pedal. In order to achieve a certain amount of swell pedal control over the rhythm, however, portions of the rhythm signals go to control amplifiers 1C2 via R13 and R60. Rhythm control amplifiers 1C7 receive their control current from QS, as is driven by ICS, which processes control voltage impulses sent via pin 24 from the rhythm volume control as well as the processor system.

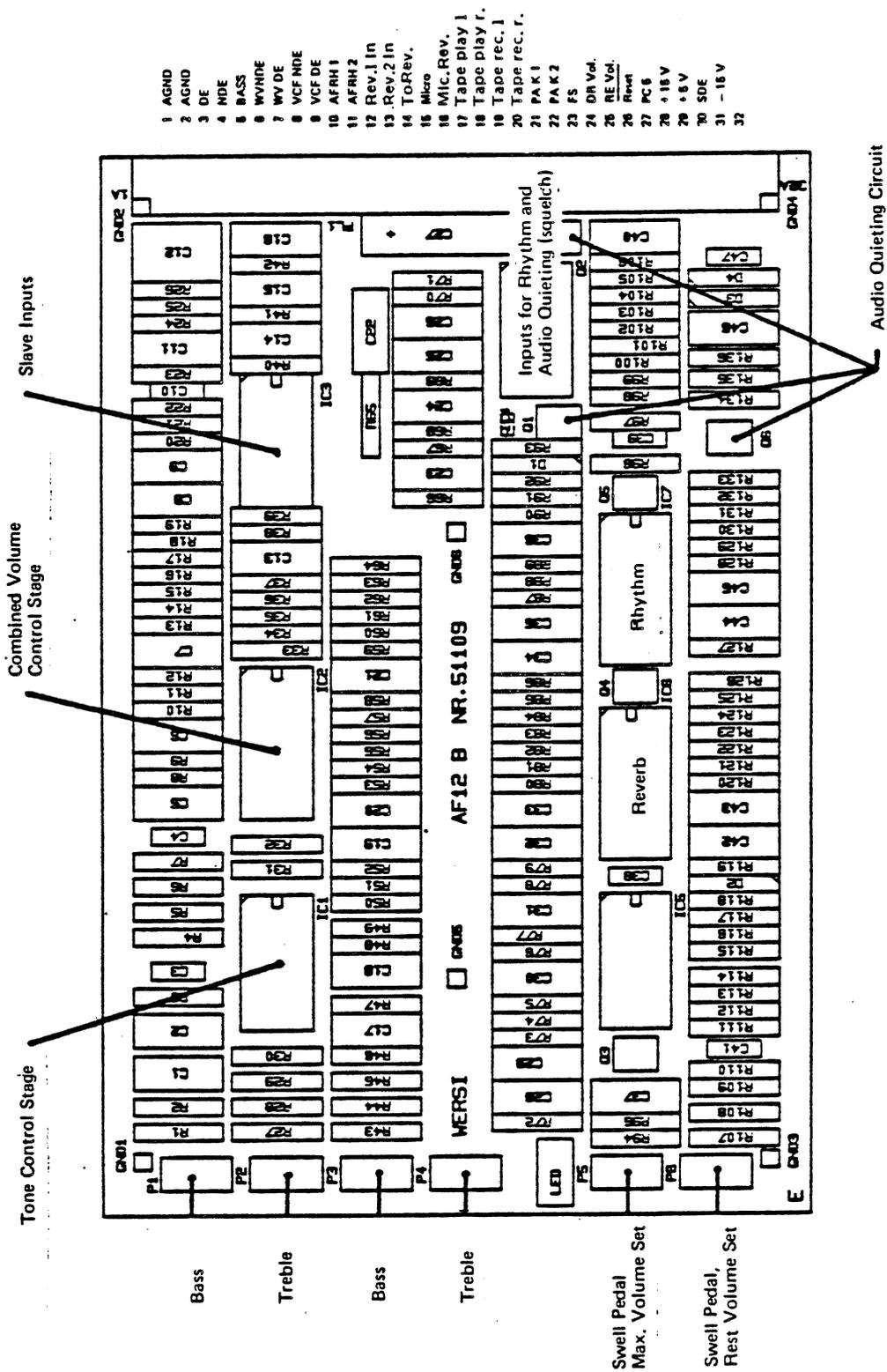


Fig. 6: Plug-in board AF 12, component layout and functions

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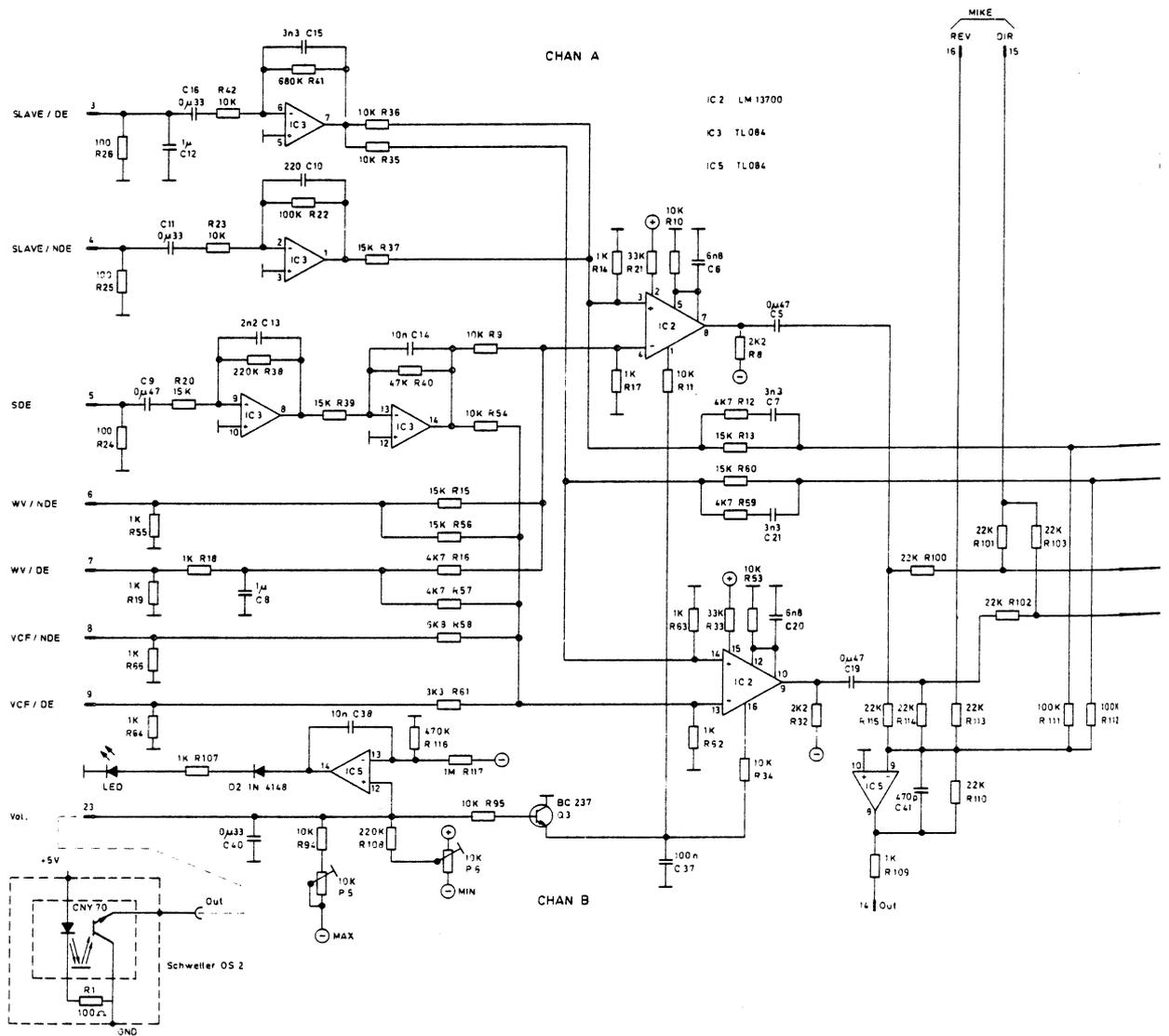
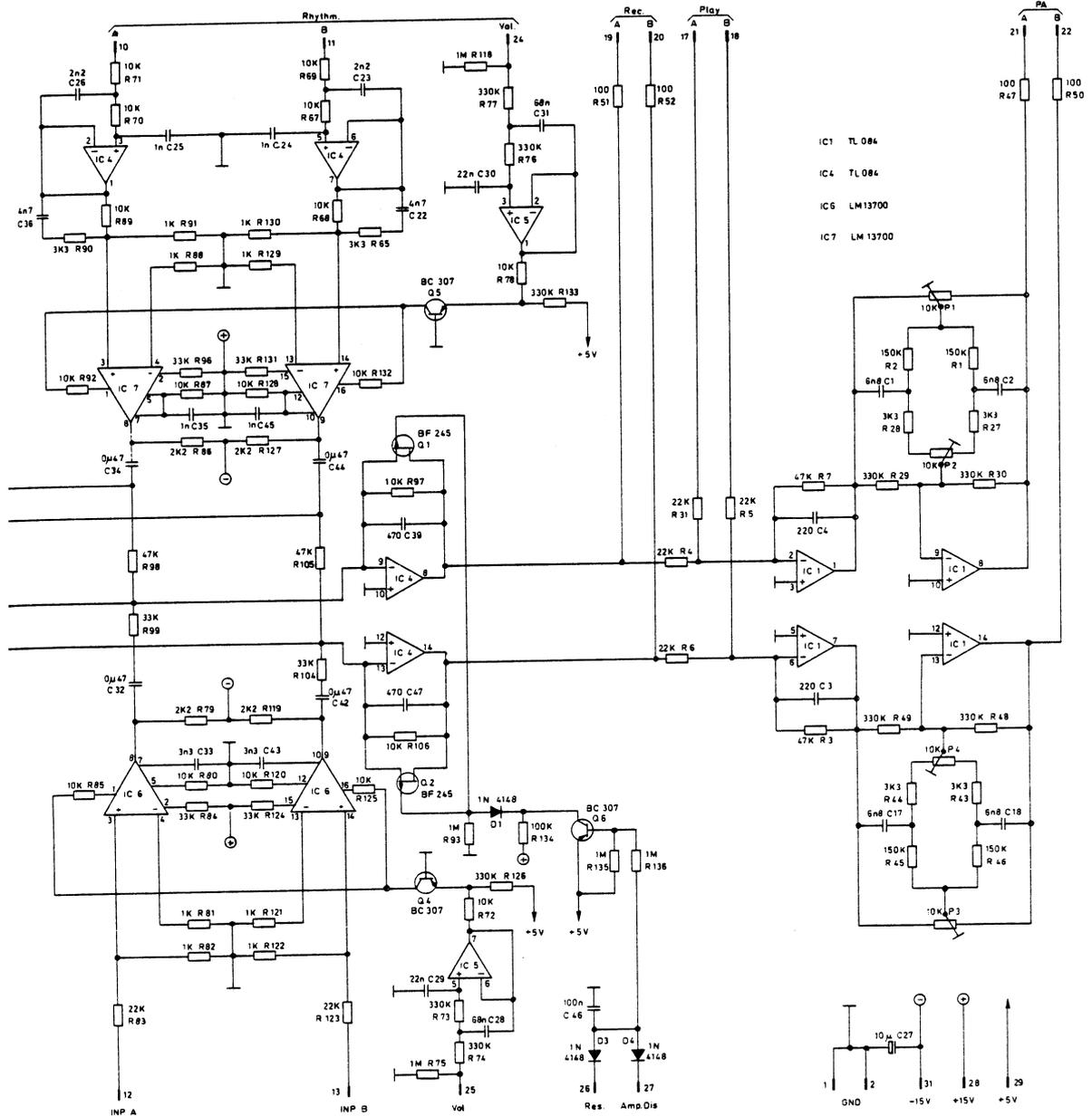


Fig. 7: Plug-in board AF 12, schematic diagram



- IC 1 TL 084
- IC 4 TL 084
- IC 5 LM13700
- IC 7 LM 13700

2/783 B

The processing circuits for the reverb work similarly. All the signals which are to be reverberated are sent via IC5c to the reverb unit DH 10 (Fig. 9). The processed signals-divided into two channels-return via pins 12 and 13 and are applied to control amplifiers IC6. This controls the reverb volume via a control current from QS, driven by IC5b. The reverberation signals go to the summing buses via R99 and R104 to be mixed with the "regular" signals.

#### 4. Plug-in Board DH 10 - Digital Reverb

##### Principles of Operation

Fig. B shows the component layout of the DH 10 digital reverberation board; Fig. 9 is its schematic diagram. Refer to Fig. 9 for the following description.

The input signal on pin 5 of the edge connector goes to IC14a via C31; the latter serves to emphasize the higher frequencies; IC14a and IC16a and their associated components form a dynamic compression and limiting circuit. Zener diodes D7 and D8 limit the signal to a maximum 5-Volt level. IC16b, c and d form a lowpass filter, limiting the audio signal bandwidth (4 kHz: appr. -60 dB).

Sample-and-hold circuit IC13 holds the amplitude of the signal constant for the analog-to-digital conversion cycle in IC 11. During this cycle, IC11 converts the analog input signal to a 1 2-bit digital signal. This signal is divided into two bytes which, via IC5 and IC6 are sequentially switched to IC1, the central processing unit (CPU). The CPU is controlled by programs (short and long reverb, fast and slow echo) contained in IC2, a 2-kilobyte erasable programmable memory (EPROM) chip.

The digital information (data) from the CPU feeds via a 16-bit bus into IC7, a 2K x 8 random-access memory (RAM), from which it is periodically read out; the RAM then receives additional data from the CPU, reads it out when required, and so on. This continuous feed-and readout cycle achieves a delayed feedback of the digital signal; i.e., the delayed signal is added to the 12-bit output of the analog-to-digital converter IC 1: In this way, reverb delays of up to 4 seconds are achieved.

The processed (reverberated) signal, again in 12-bit form, is again divided into 2 bytes and fed in sequential steps to digital-to-analog converter IC12; the latter produces an analog signal corresponding to the digital data input. This signal is further processed in IC15c; however, the output signal here contains amplitude spikes and undesired frequency components resulting from the digitalization. IC15a, b and d form a lowpass filter which removes the undesired. components from the signal. C40 and IC14b serve as deemphasis and bandspreading circuits respectively, reversing the high frequency emphasis and band limiting described previously.

The reverberated and delayed output signal goes via Jul to pin 6 in single-channel instruments. In stereo instruments, IC17 develops a second-channel output. It achieves this via defined phase shifting in certain frequency ranges and applies the second output to pin 9. Further stereophonic audio processing results in an impressive room-filling reverberation effect.

The switching of the various operational modes (short and long reverb, fast and slow echo) is accomplished by applying voltage impulses to the NMI input of the CPU via pin 12. Applying a reset to the CPU via pins 10 and 11 starts the CPU's program run.

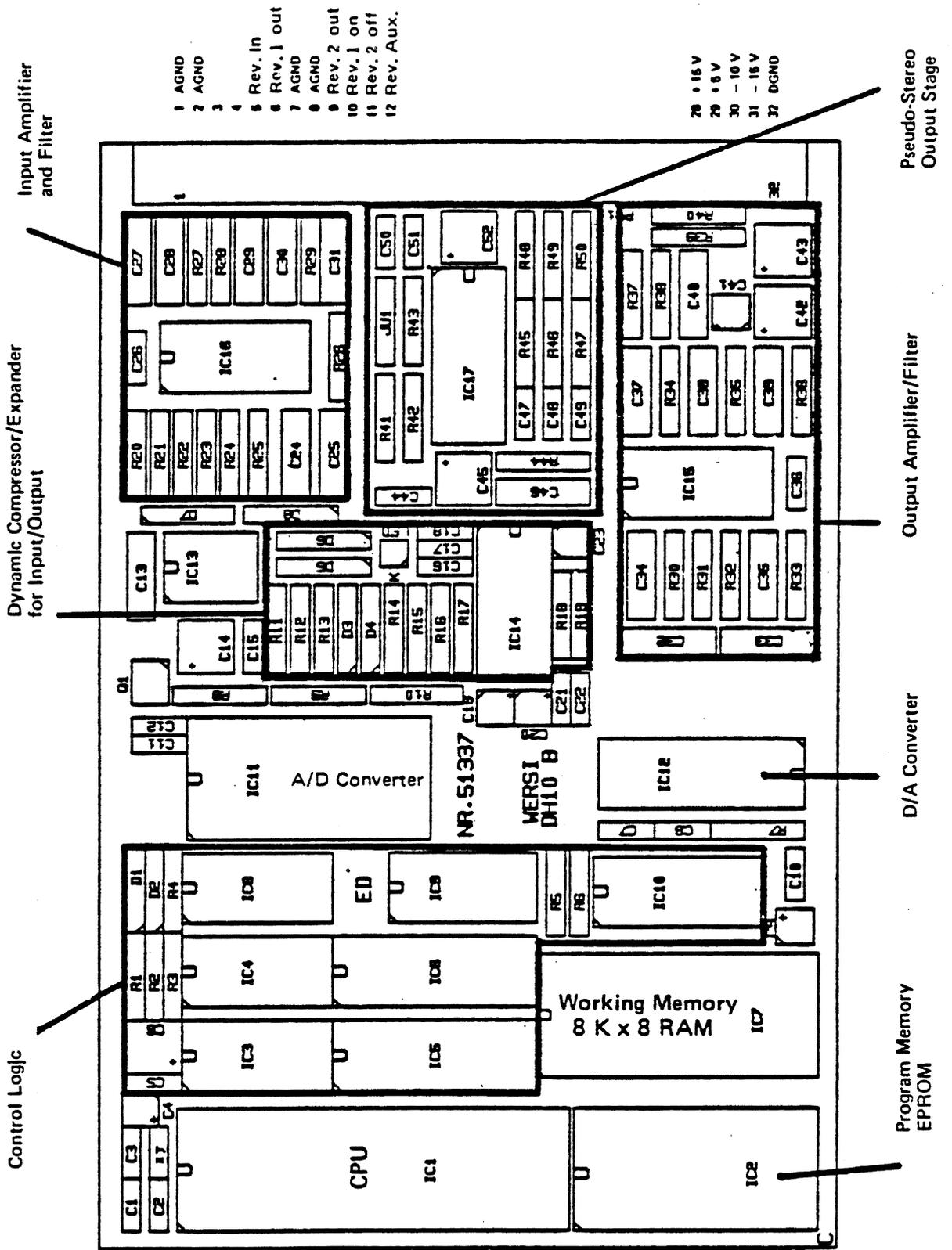
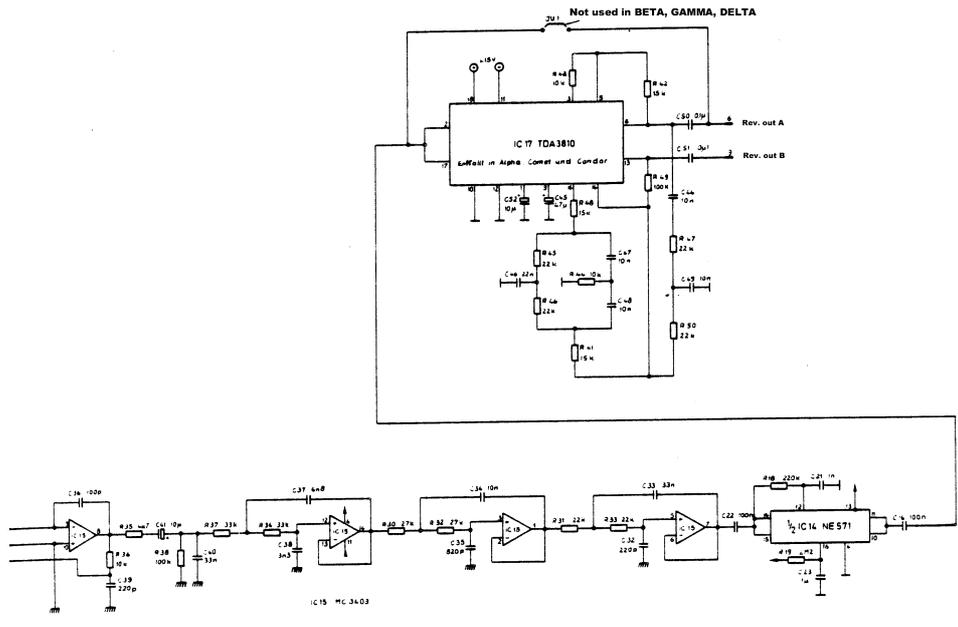


Fig.8: Plug.in board DH 10 component layout and functions.





## 5. Plug-in Board EF 20 ---Special Effects, Analog Portion

### Principles of Operation

Fig. 10 shows the component layout of the EF 20 special effects board; Fig. 11 is its schematic diagram Refer to Fig. 11 for the following description.

#### a) Wersivoice

The Wersivoice consists chiefly of the three bucket brigade chips, 1C2, 4 and 6, which receive their clock frequencies from three voltage-controlled oscillators (VCOs) 1C1, 3, 5. The control voltages for the VCOs are generated by the serial analog signal "DAC" via demultiplexer 1C13.

The audio signal to be processed goes via 1C11 to low-pass filter 1C12, which suppresses the higher frequency signal components. The output signal from the bucket brigades is filtered anew by 1C11 and fed to analog switch 1C14, which handles the "deep" and "flat" functions of the Wersivoice.

The processed signal appears at pin 14 of 1C14. The signal is switched to either pins 3 or pins 4 (edge connector) by 1C15. From there it goes to amplifier AF 12.

#### b) VCF

The voltage-controlled filter (VCF) receives its input from pin 6a of the edge connector. 1C15 determines whether the signal intended for VCF processing is to be distorted (for E-Guitar, 1C8a does the distorting) or go directly to VCF 1C16. The quality of the VCF is determined by 1C17.

Just as for the Wersivoice, the control voltages for the VCF are generated by 1C13 from the signal "DAC." 1C19 determines whether the VCF is to function as a bandpass or lowpass filter and whether the VCF works directly or in conjunction with the Wersivoice.

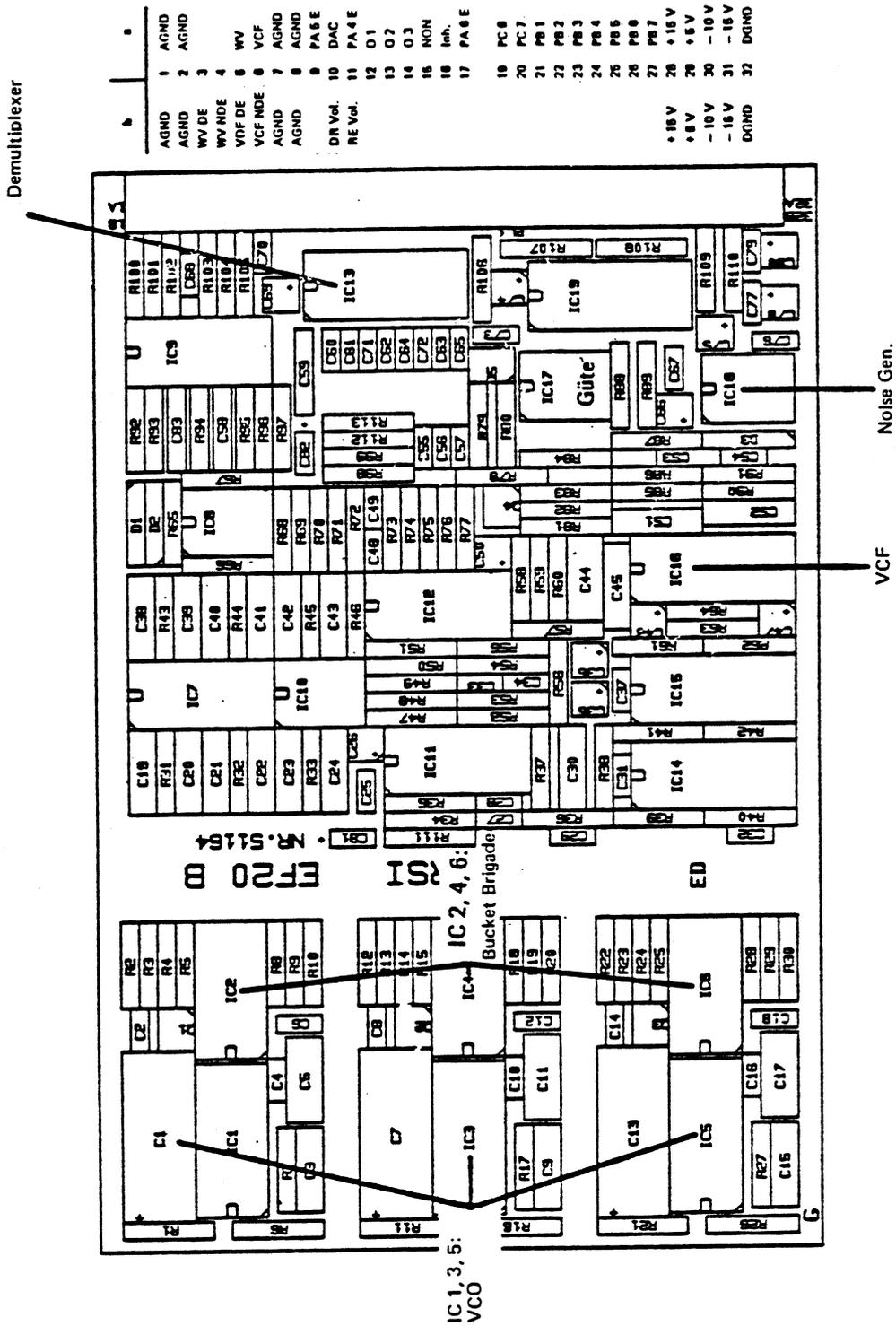


Fig. 10: Plug-in board EF 20, component layout and functions

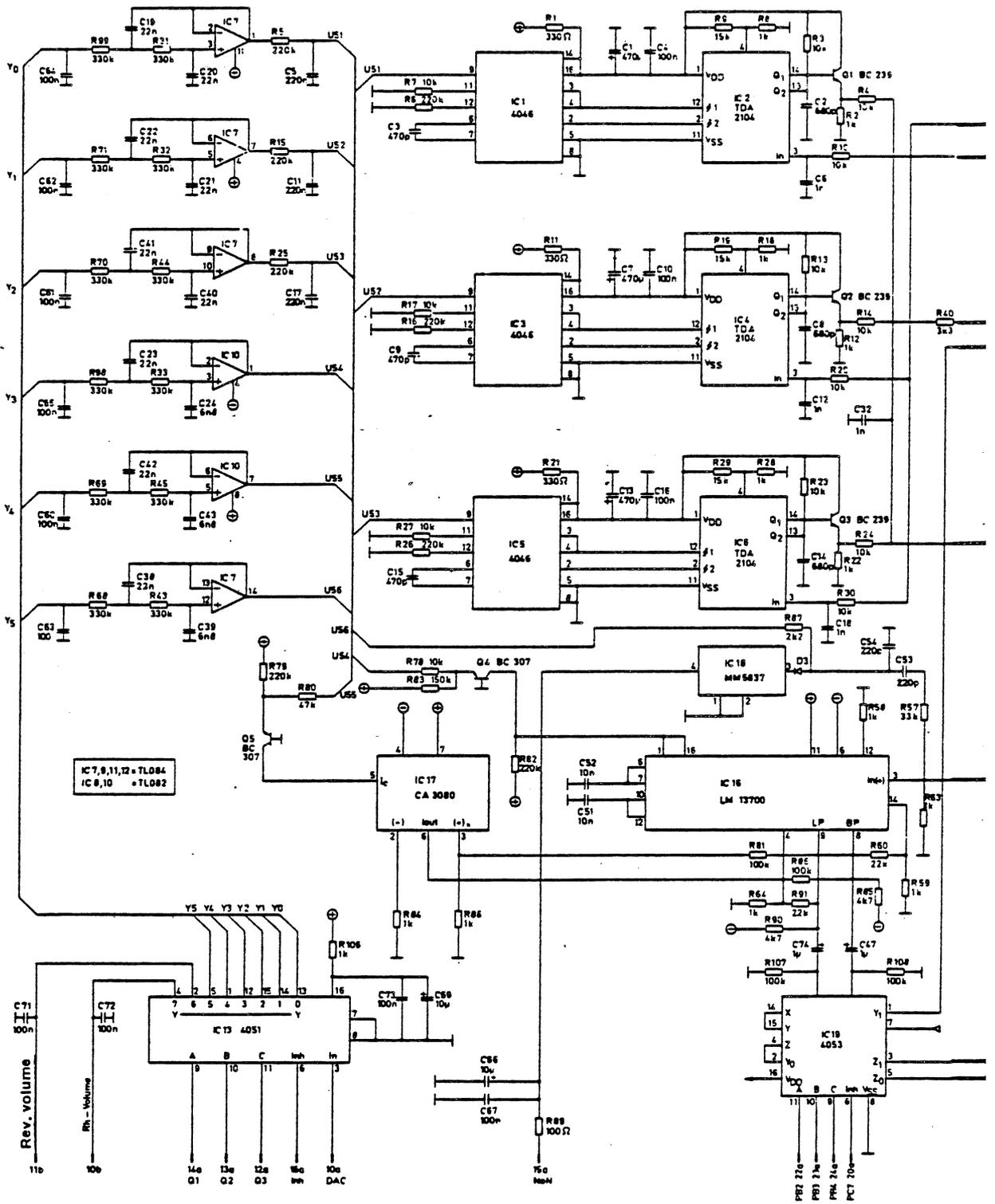
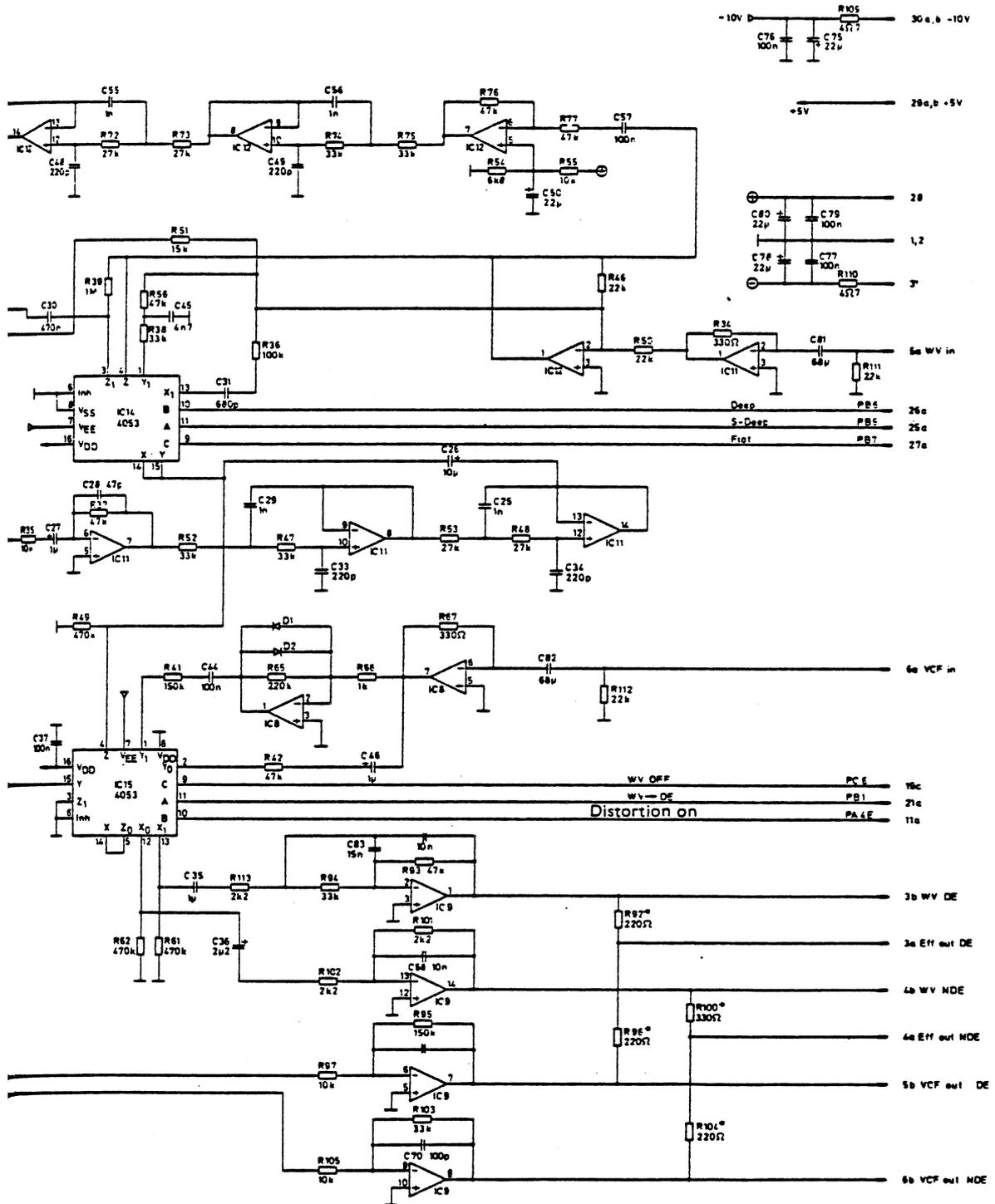


Fig. 11: Plug-in board EF 20, schematic diagram



\* Omitted In Beta, Gamma and Delta

## 6. Plug-in Board EF 19 ---Special Effects, Digital Portion

### Principles of Operation

Plug-in board EF 19 contains three interface circuits: master/special effects, organ/computer (RS232) and organ/cassette. Fig. 12 is the component layout of EF19; refer to Fig. 13, EF 19's schematic diagram, for this description -

#### a) Master/Special Effects Interface

Via address decoder 1C6, the master address bus selects 1C10, 11, 12 and 13, which are then addressed by the master data bus. Control signals PA0 thru PA7 and PB0 thru P87 exist statically at the outputs of 1C11 and 1C12 respectively. PA4 thru PA7 and PBO thru PB7 control the audio signal routing on the second special effects board EF20.

The 8-bit digital word at the output of 1C10 is converted into an analog signal in 1C3. By this means, all the control voltages required on EF20 are transferred serially.

Control signals for the digital reverb are available statically at output pins 15 and 17 of 1C13. The 6ut-put from pin 14 of 1C13 is used to quiet (squellch) the amplifier on board AF 12.

#### b) RS232 Interface

1C9, a special RS232 interface IC, converts 8-bit parallel data words into serial data and vice-versa, depending upon the mode of operation. This serial signal (TXD) is switchable in multiplexer 1C8 to cassette or to RS232 operation. When RS232 is selected, the signal goes to TTL-to-RS232 driver 1C19, where it is inverted and level-shifted to +/- 15 Volts.

Conversely, a serial RS232 signal from an external computer enters RS232-to-TTL receiver 1C18 and is level-shifted to +5 Volts. It then passes through multiplexer 1C8 to 1C9, where it is converted into 8-bit parallel data.

#### c) Cassette Interface

1C9 converts an 8-bit data word from the master bus into serial data, just as in RS232 operation. This serial signal is then applied to 1C14, pin 12, and from 1C14 is applied to the cassette recorder as data to be recorded -

An incoming serial signal from a cassette recorder is amplified by 1C1, conditioned and shaped in 1C7 and 1C5 and applied to 1C9, where it is reconverted into parallel data.

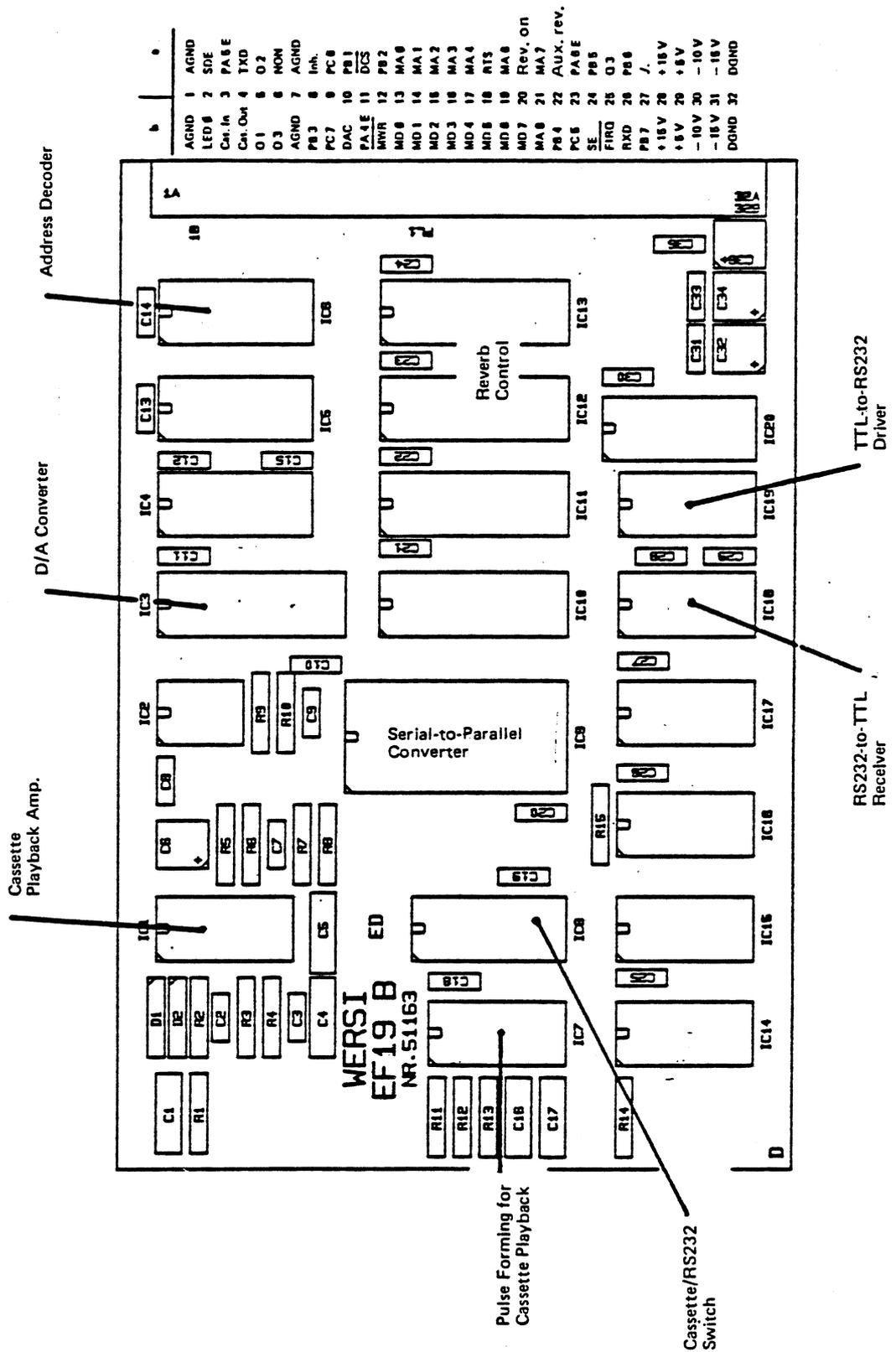


Fig. 12: Plug-in board EF 19, component layout and functions

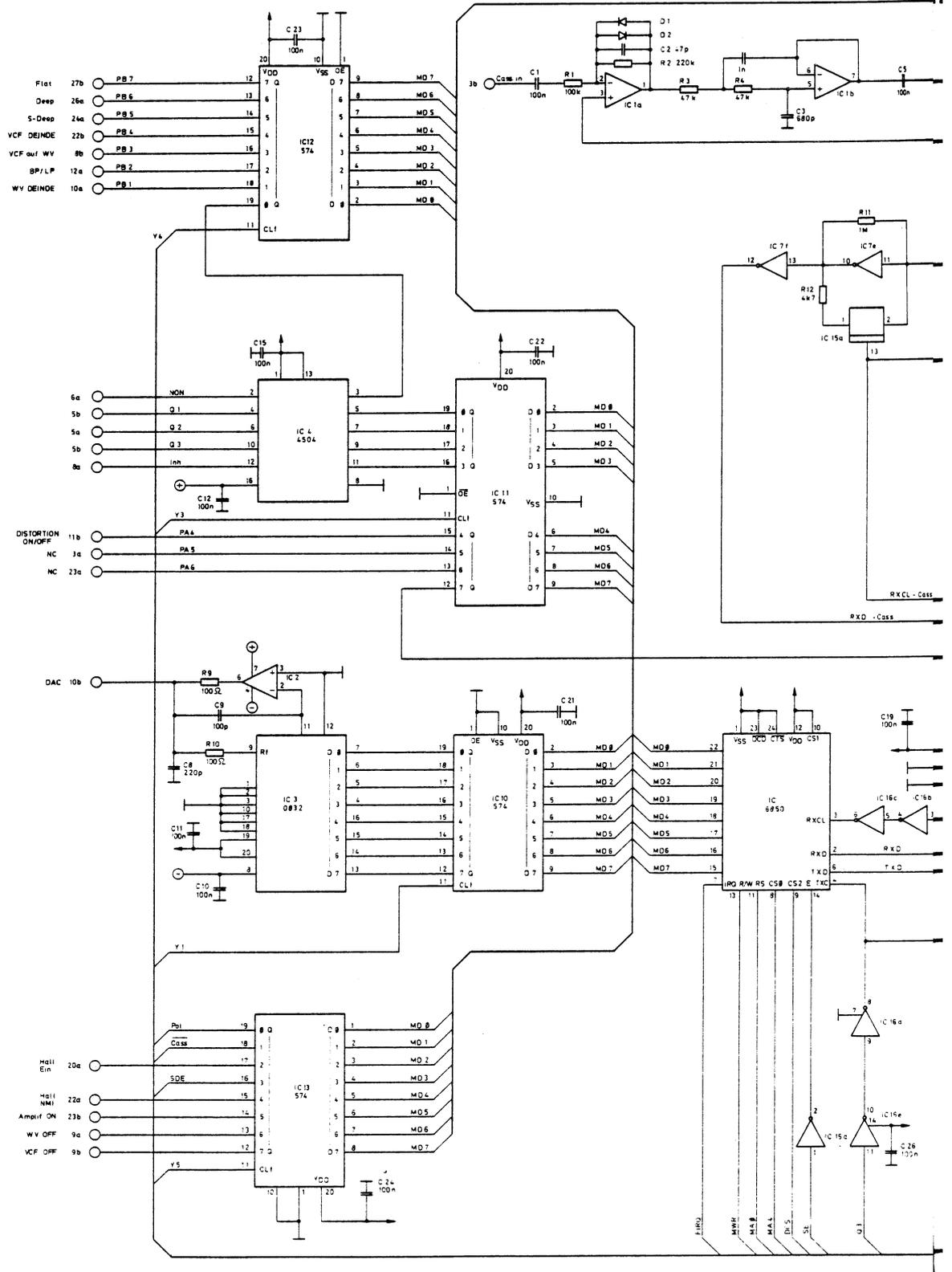
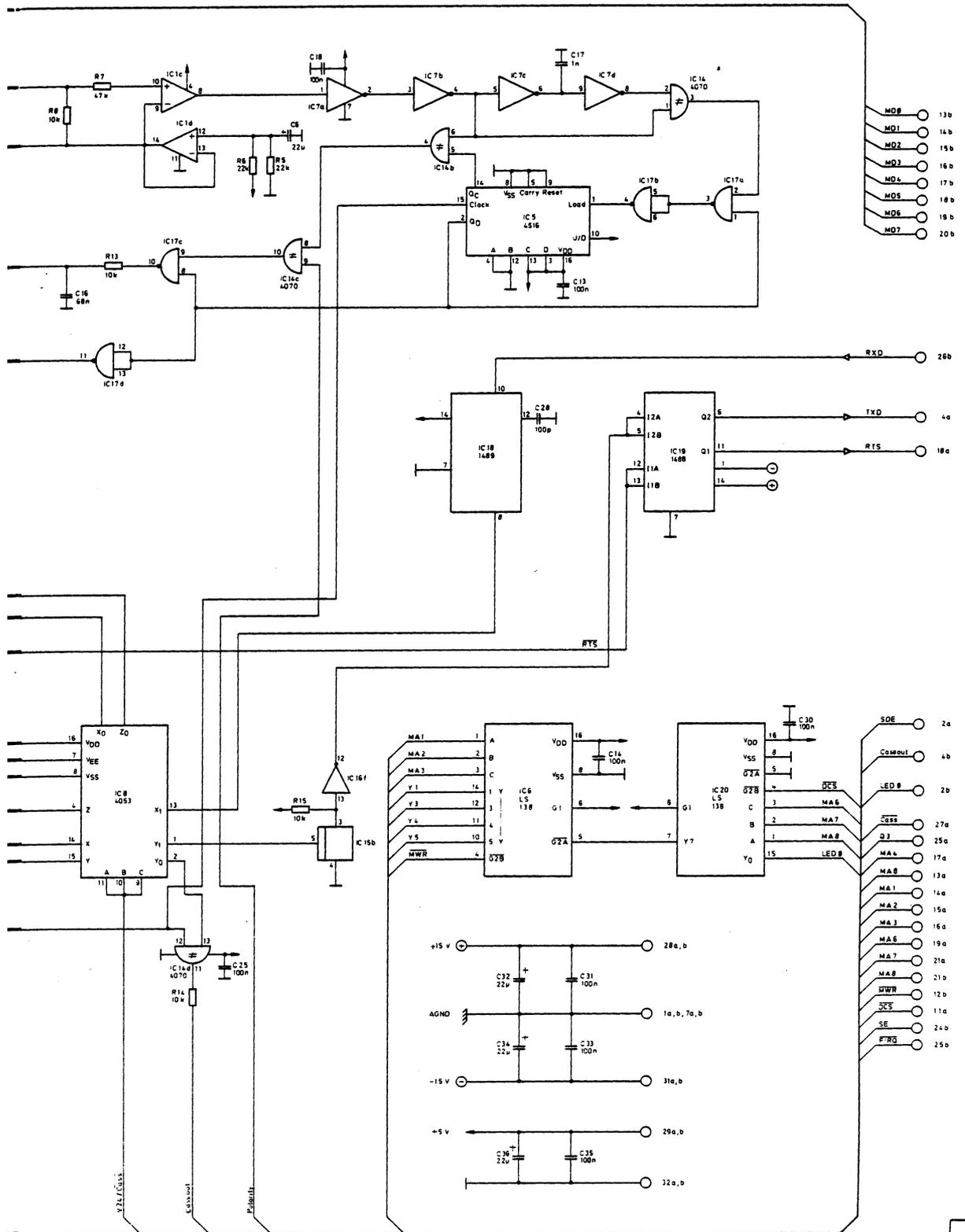


Fig. 13: Plug-in board EF19, schematic diagram



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## 7. Plug-in Board DDS 3 - Rhythm

Fig. 14 shows the component layout of plug-in board DDS 3; refer to Fig. 15, the schematic diagram, for the following description.

Depending upon the instrument trigger pulses it receives from the master data bus via IC20, microprocessor IC14 reads out percussion instrument sounds digitally stored in digital sound modules IC23 thru IC26.

These sounds appear at output pins 12 thru 19 of IC9 as momentary digital data in multiplexed form. The data is converted into an analog signal in DAC IC1. The signal then goes to IC4, where its volume can be controlled. The percussion sounds appear at the output of IC2 in the form of a multiplexed analog signal.

Demultiplexer IC5 routes each sounding instrument to its own audio channel. The composite percussion sounds are present at the outputs of the two sections of IC8 for further stereo processing.

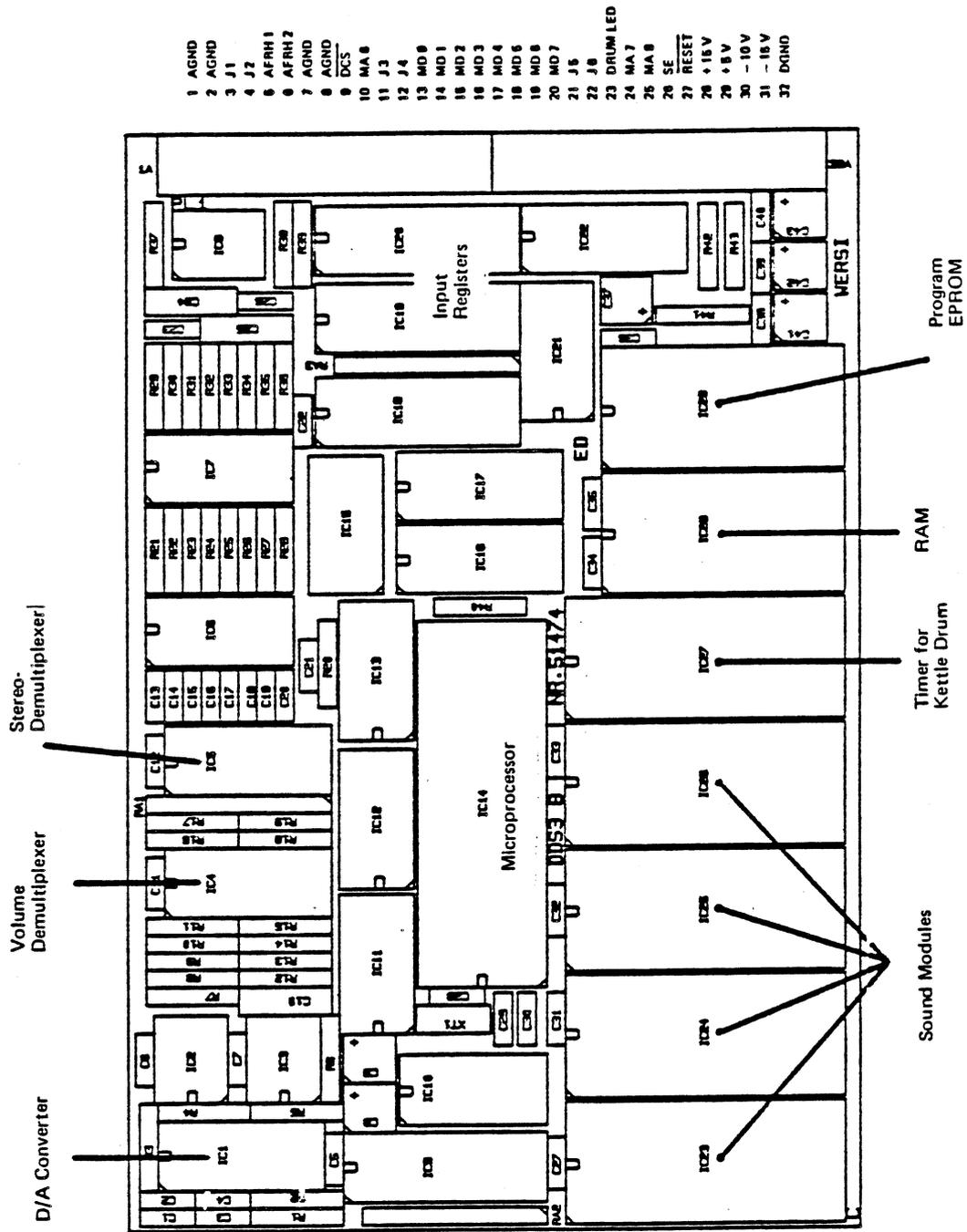


Fig. 14: Plug-in board DDS 3, component layout and functions

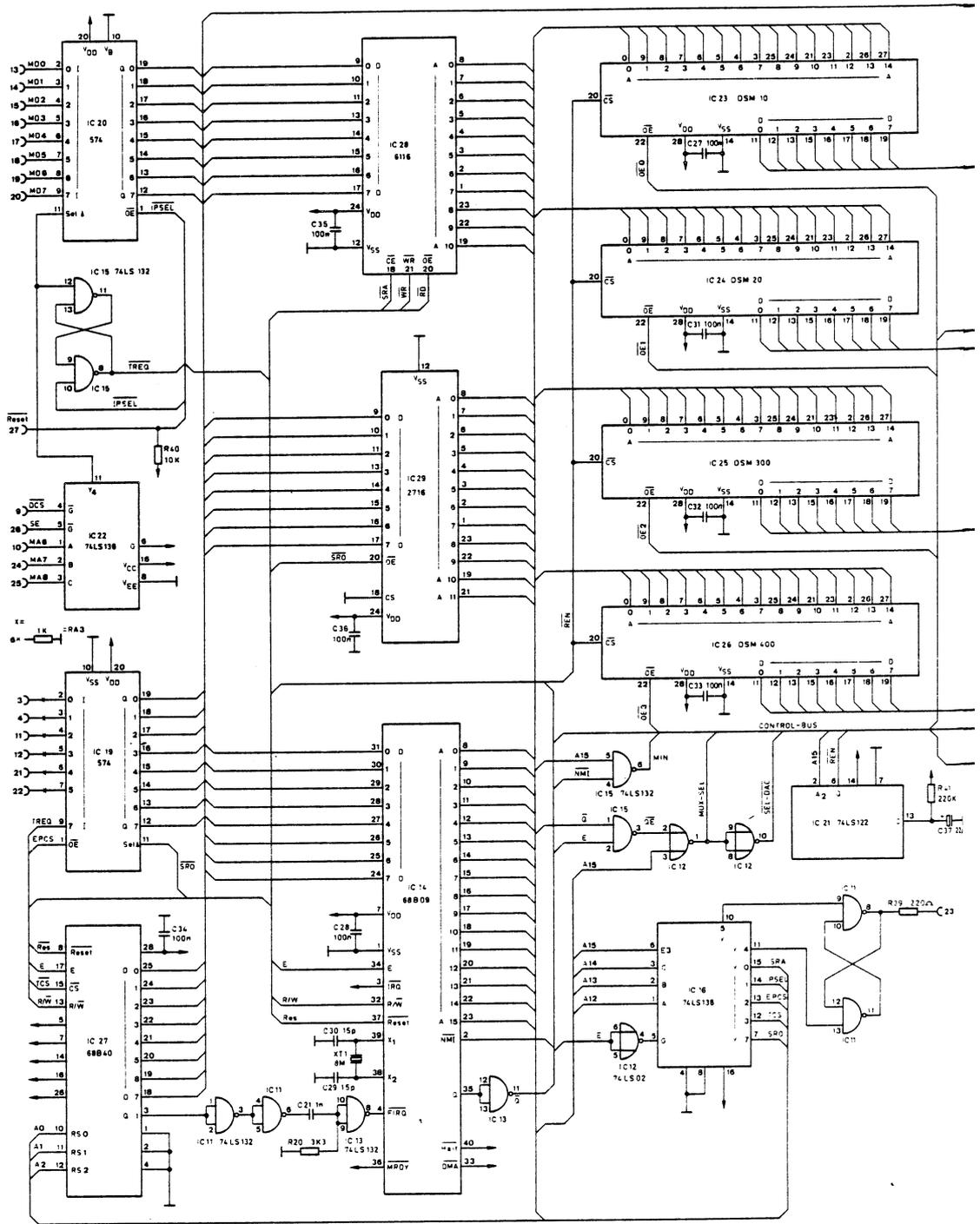
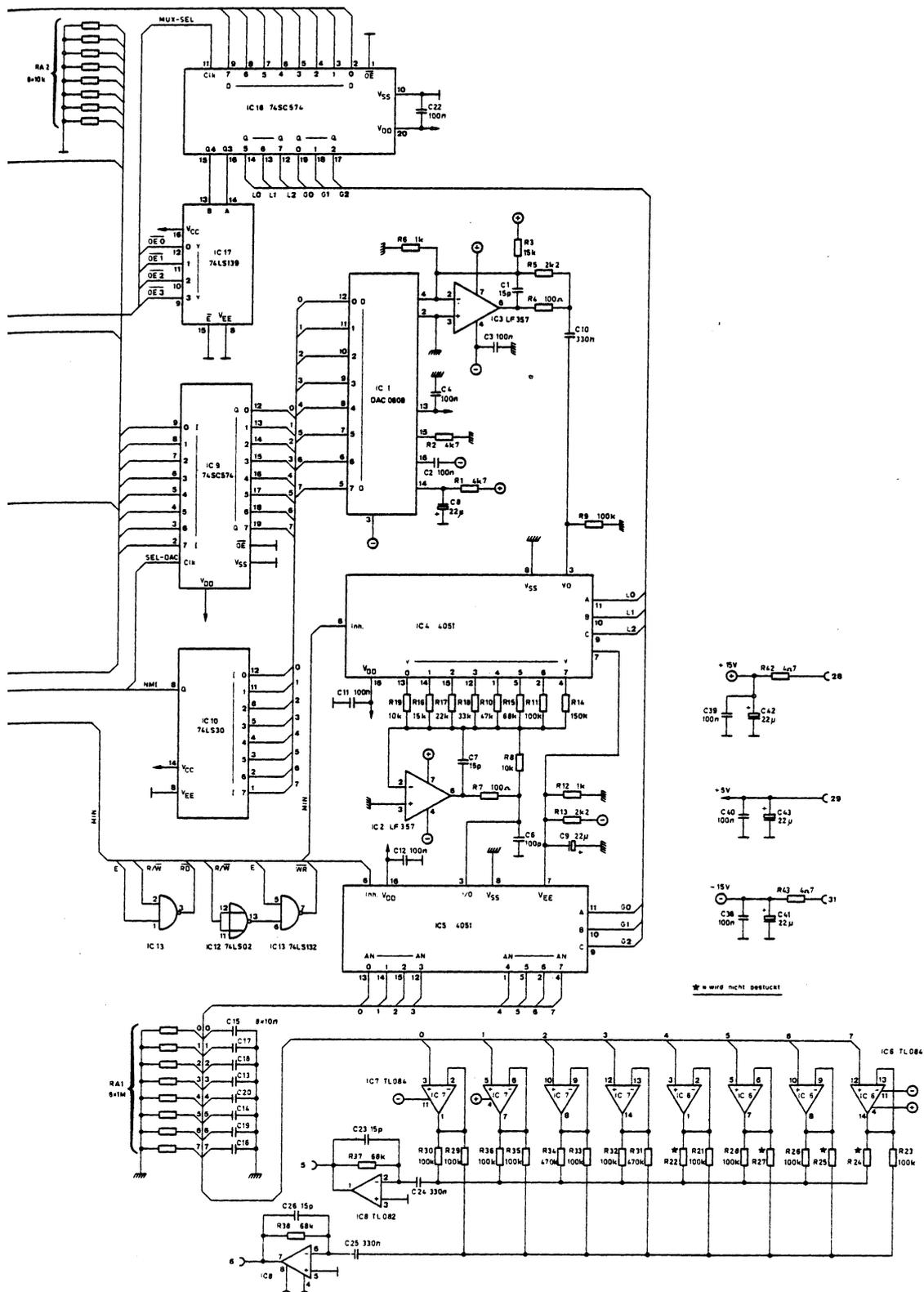


Fig. 15: Plug-in board DDS3, schematic diagram



## 8. Plug-in Board SL 3 - Slave Processor

Fig. 16 shows the component layout of plug-in board SL 3; Fig. 15 is the schematic diagram.

The following test enlarges upon the initial test of the slave processor in the assembly manual, enabling you to completely check every voice on every slave. Proceed as follows:

- a) Turn on the organ; if already on -
- b) Press "Reset."
- c) Press "Compute."
- d) Select the slave:

Press one of the five switches W, E, R, S, I, depending upon the slave you wish to test:

W	for Slave 1 (immediately beside the master processor board)
E	for Slave 2
R	for Slave 3
S	for Slave 4
I	for Slave 5

If, for example, you wish to test Slave 3, press R.

- e) Select the voice:

Again, press one of the five switches:

W for 1st voice  
E for 2nd voice  
R for 3rd voice  
S for 4th voice  
I for 5th voice

Again, by way of example, if you have selected Slave 3 and wish to test the 1st voice, press W.

- f) Select any DMS instrument (the Horn, for example) and play on the selected manual. The organ will now sound only monophonically, that is, only one tone is produced at a time and both of its components' become slightly apparent and can be evaluated by moving the slide controls Chan. 1 and Chan. 2 up or down. Listen for possible problems such as noise, distortion or a missing component.
- g) Test other slaves by pressing "Reset," "Compute" and proceeding as above.

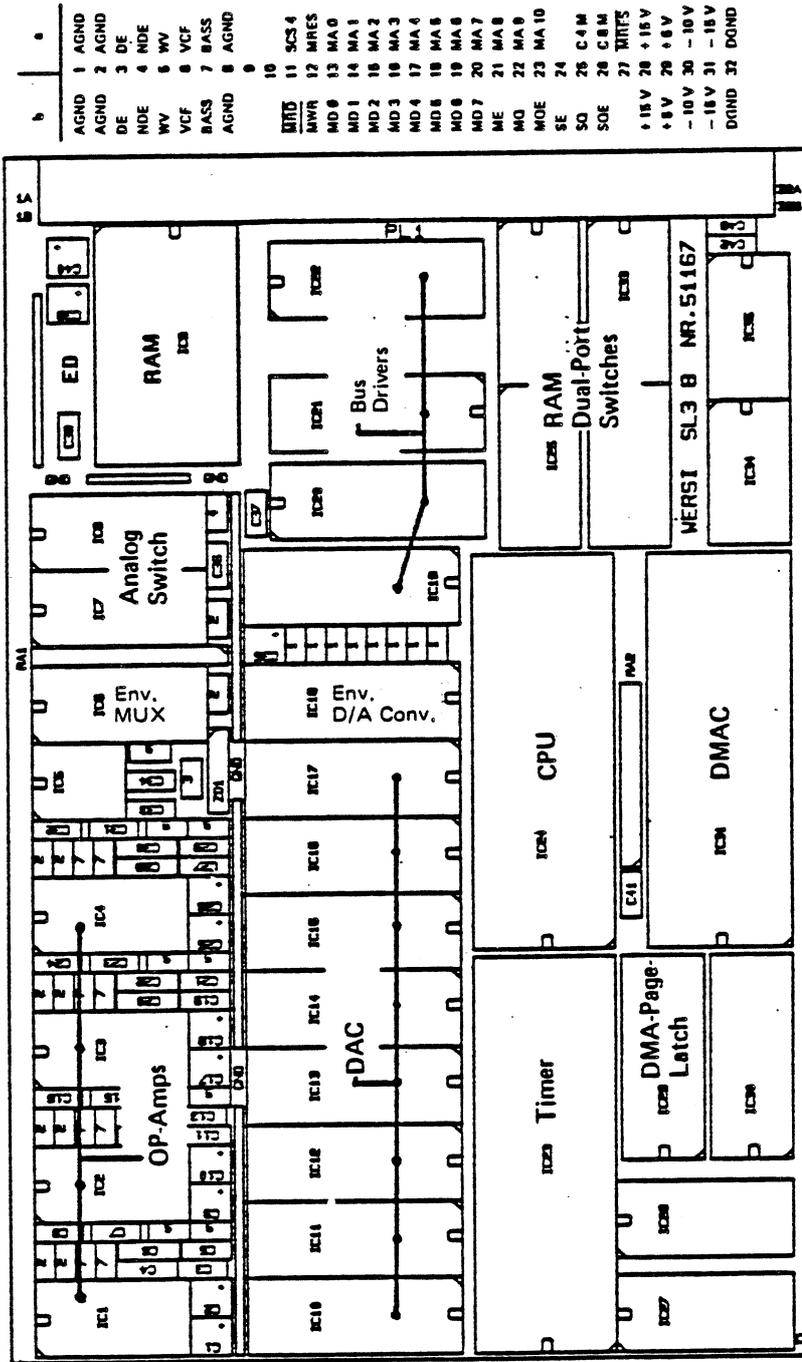


Fig. 16: Plug-in board SL 3, component layout and functions

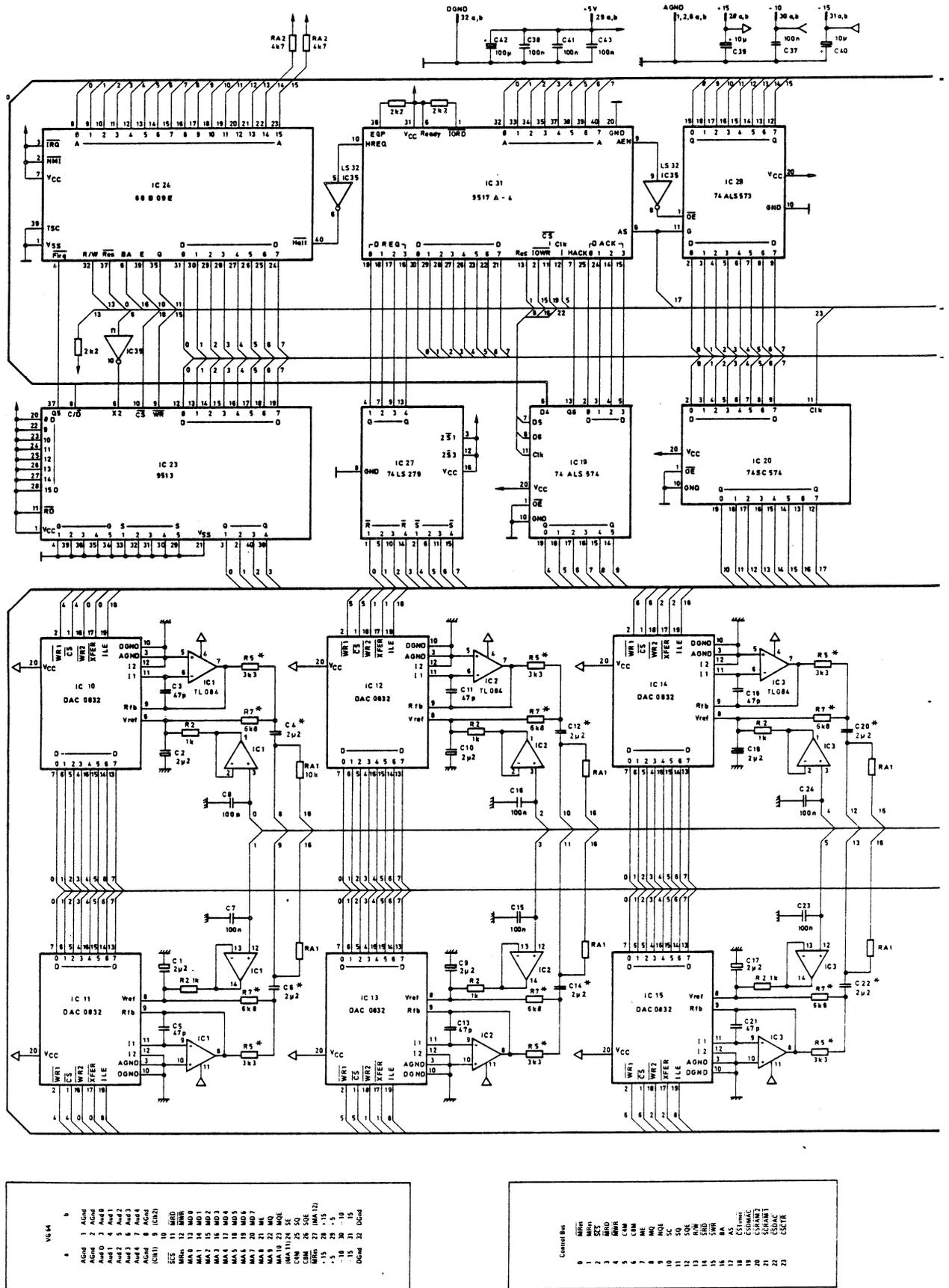
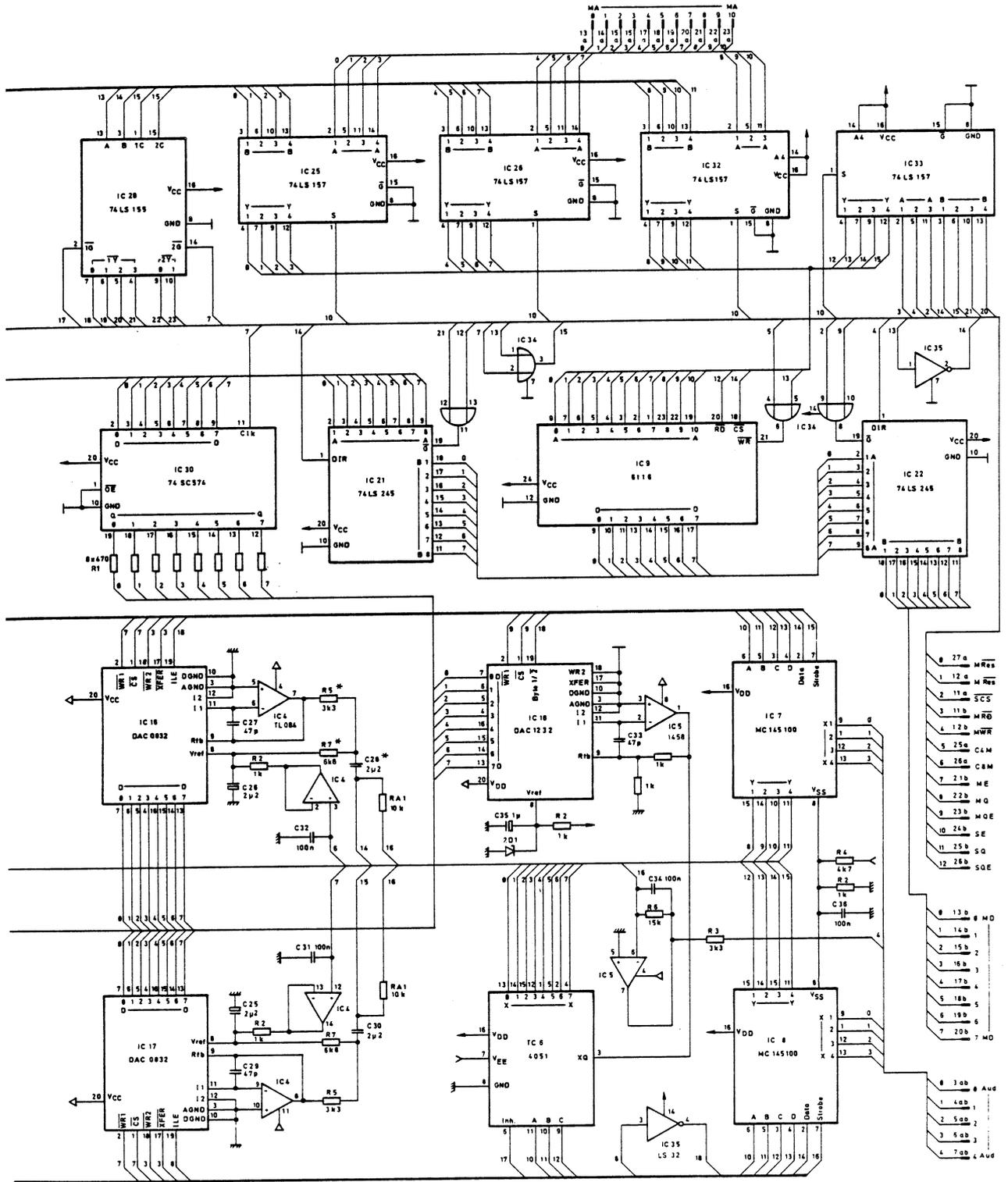


Fig. 17: Plug-in board SL3, schematic diagram  
AM 334-01-502



Abkürzung	Bezeichnung
0	0
1	1
2	2
3	3
4	4
5	5
6	6
7	7
8	8
9	9
10	10
11	11
12	12
13	13
14	14
15	15
16	16
17	17
18	18

Abkürzung	Bezeichnung
0	0
1	1
2	2
3	3
4	4
5	5
6	6
7	7
8	8
9	9
10	10
11	11
12	12
13	13
14	14
15	15
16	16
17	17
18	18

\* Nur bei SL 30: R5 = 75k  
 R7 = 15k  
 C4, 6, 12, 14, 20, 22, 28 und 30 = 0,33µF

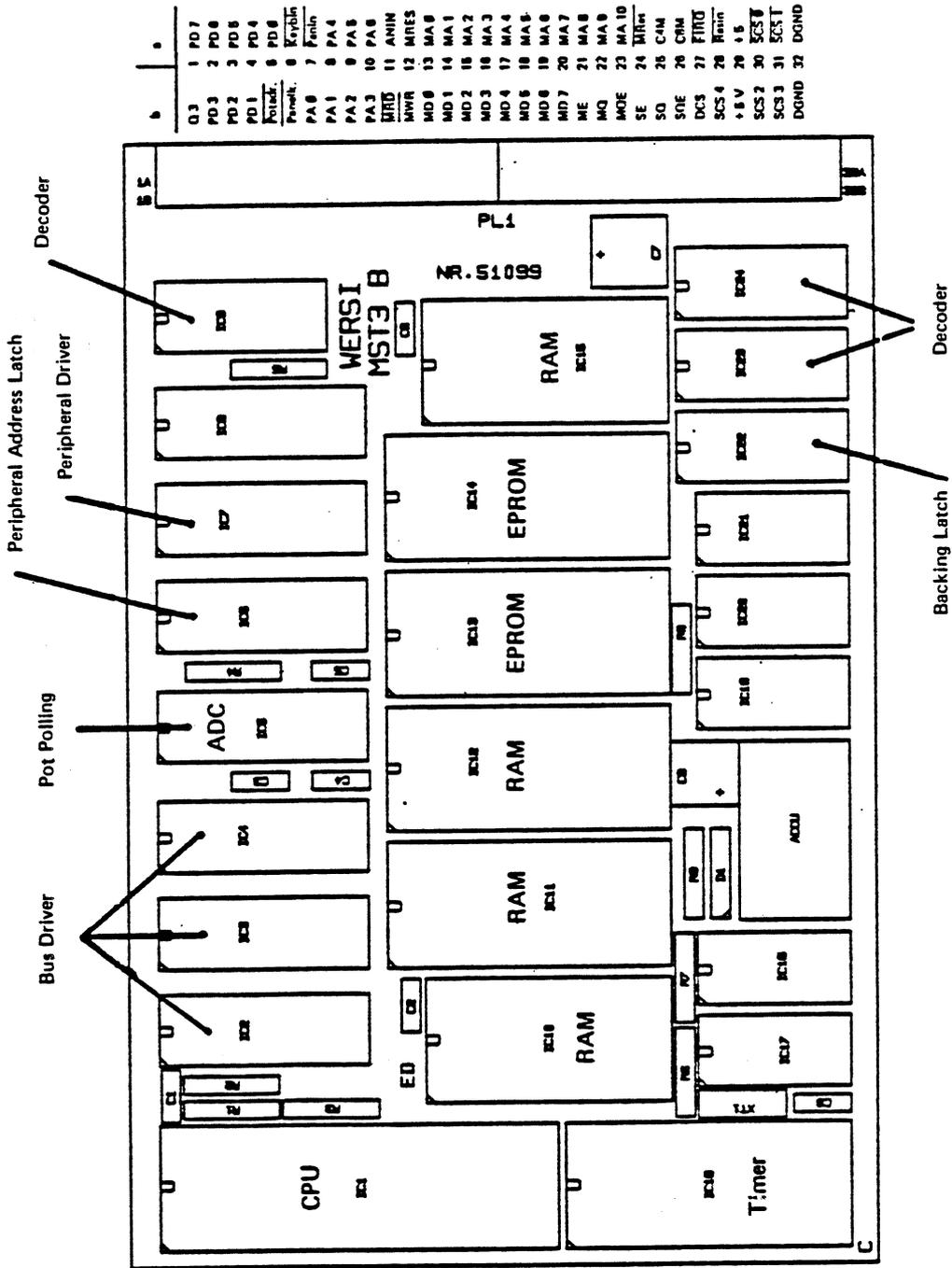


Fig. 18: Plug-in board MST 3, component layout and functions

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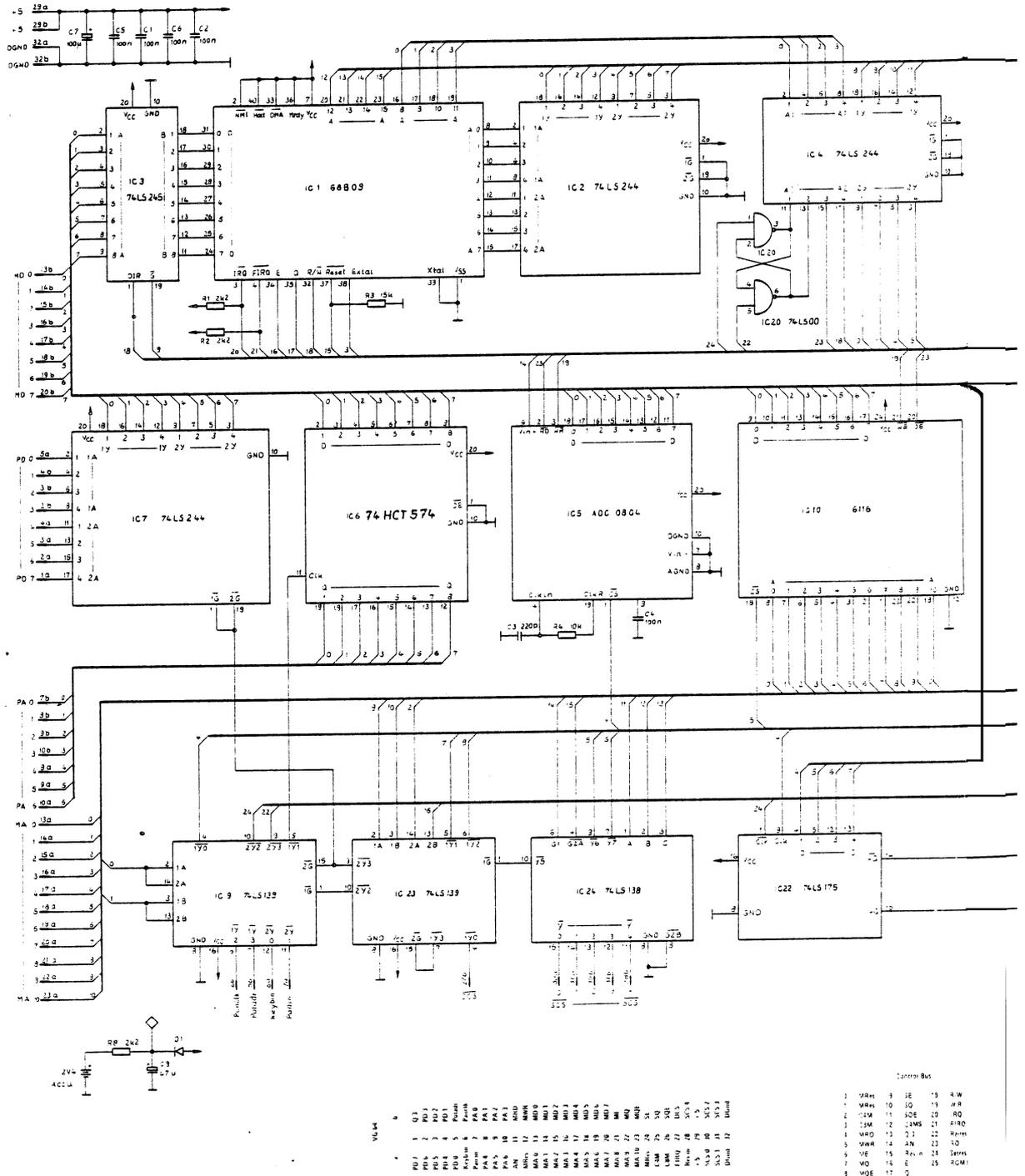
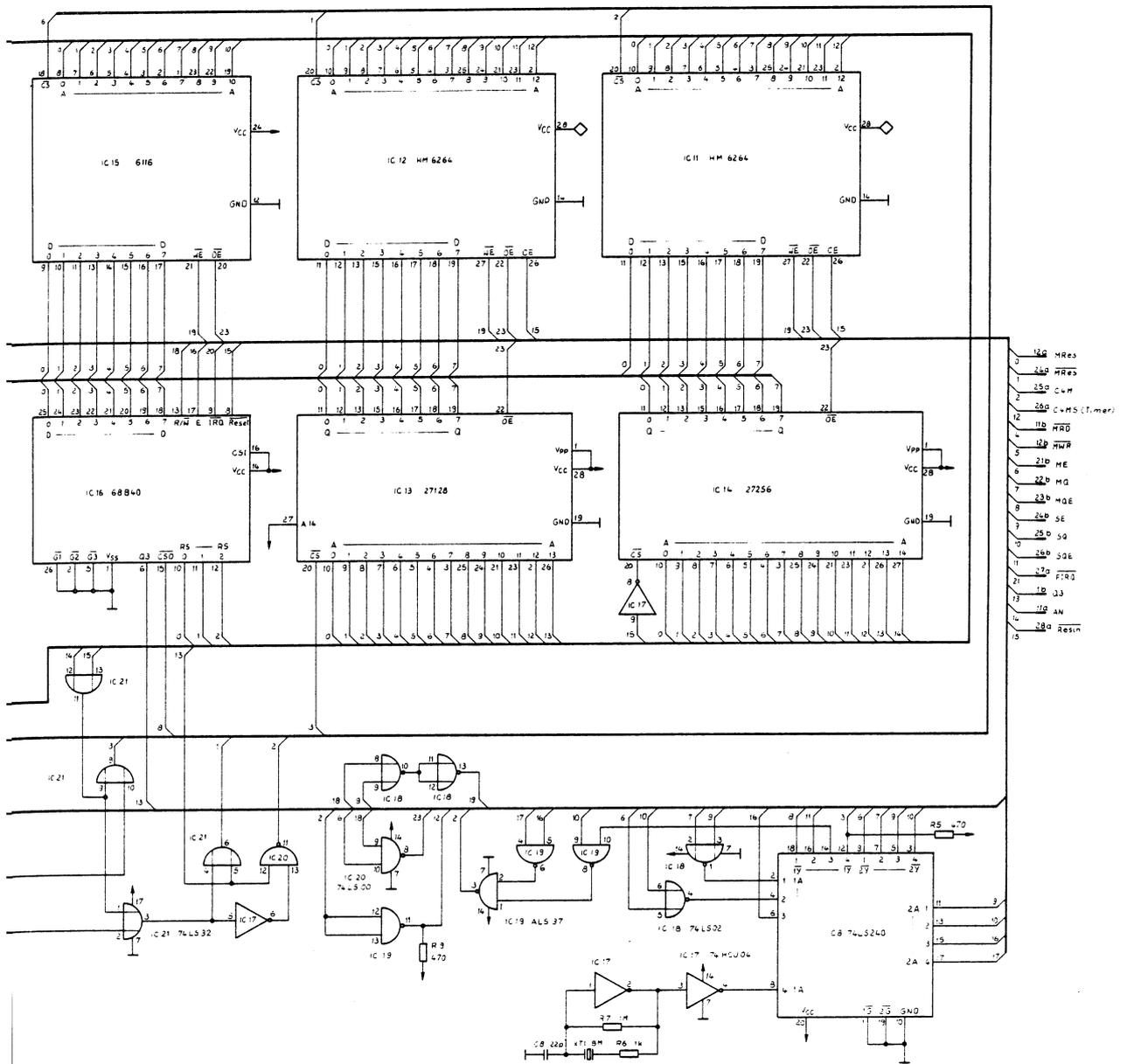


Fig. 19: Plug-in board MST 3, schematic diagram



Memory Map

0000	Bank: 00	RAM 3 - 4	5800	59FF	DCS	6E00	Bank
	Bank: 80	ROM 2 L	5A00	5BFF	ADC	6E01	Prasdr
3FFF	Bank: CD	ROM 2 H	5C00	5DFF	Timer	6E02	Prasdr
4000	37FF	SCS 0	5E00	5FFF	: 0	6E03	Prasdr
3800	3FFF	SCS 1	7000	77FF	RAM 1	6E04	Keyo.in
5000	57FF	SCS 2	7800	7FFF	RAM 1	6E05	Keyo.in
5800	5FFF	SCS 3	8000	FFFF	ROM 1	6E06	Prasdr
5000	67FF	SCS 4				6E07	Prasdr

## II. Control Panels

The following pages contain schematic diagrams of the control panel PC boards CB. . . which include slide controls, switches and drawbars, and key contact boards KD. . . which decode and route inputs from the key. boards and pedals.

### 1. CB Boards

The control panel PC boards CB. . . work on a principle of continuous polling of the control switches by the CPU on master processor board MST 3, On, for example, CB 21 (Fig. 21), the parallel-out serial shift registers IC2, 7, 8, 9, 1 and 10 are clocked by peripheral output line pulses from the CPU. The register outputs continuously and rapidly poll the status of the control panel function switches.

A momentary switch closure results in a sense pulse, which is applied to the CPU's data bus. This sense pulse occupies a specific time slot in the CPU's sequence program and instructs the CPU to initiate the function for that time slot. The CPU, in turn, senses a serial data code to the appropriate register set, designating the LED(s) to illuminate on the panel switches.

### 2. Key Contact Boards

#### KD 1a (Fig. 25a)

The dynamic perception of keyboard action is possible through the principle of time measurement. To accomplish this, the spring contact for each key performs a normally closed-to-normally open switching function. The time from the opening of the normally closed (rest) state of the contact to the closing of the normally open (working) state is measured electronically.

Every eight contacts are assigned to a matrix address. For five octaves (61 keys), eight matrix addresses are necessary. These matrix addresses are sequentially switched to ground so that the statuses of the 61 contacts can be polled. The speed at which the polling takes place is determined by the setting of the code switch S1. (Note: "Reset" must be pressed after any change in the code switch). If the polling rate is very short, a key must be pressed quickly to achieve maximum dynamic value.

The dynamic value is transferred to one of two 2-byte parallel ports. The first port, IC5, transfers data that determines pitch values and whether a note should be turned on or off. The second port transfers data determining the dynamic value. When the dynamic value is entered, IC8a, a flip-flop, toggles, releasing an FIRQ (interrupt request) shot to the CPU.

During the port read cycle, the key (IC5) is read first and the FIRQ appears in its highest bit (on IC4/pin 13). This tells the CPU which manual the FIRQ is for. When this is determined, the dynamic value for that manual is read and the flip-flop resets; In addition, the FIRQ can be cancelled by the other flip-flop, IC8b. IC7 handles the address decoding.

#### KD 1b (Fig. 25b)

The circuitry of this board polls the pedal switching matrices to determine pedal switch closures. Specific matrix schemes are used according to organ type and these are selected by jumpers Ju7 thru 14.

#### KD2/KD4

These extend the key contact matrices according to the number of octaves (KD4 for 4 octaves, KD2 for 5 octaves).

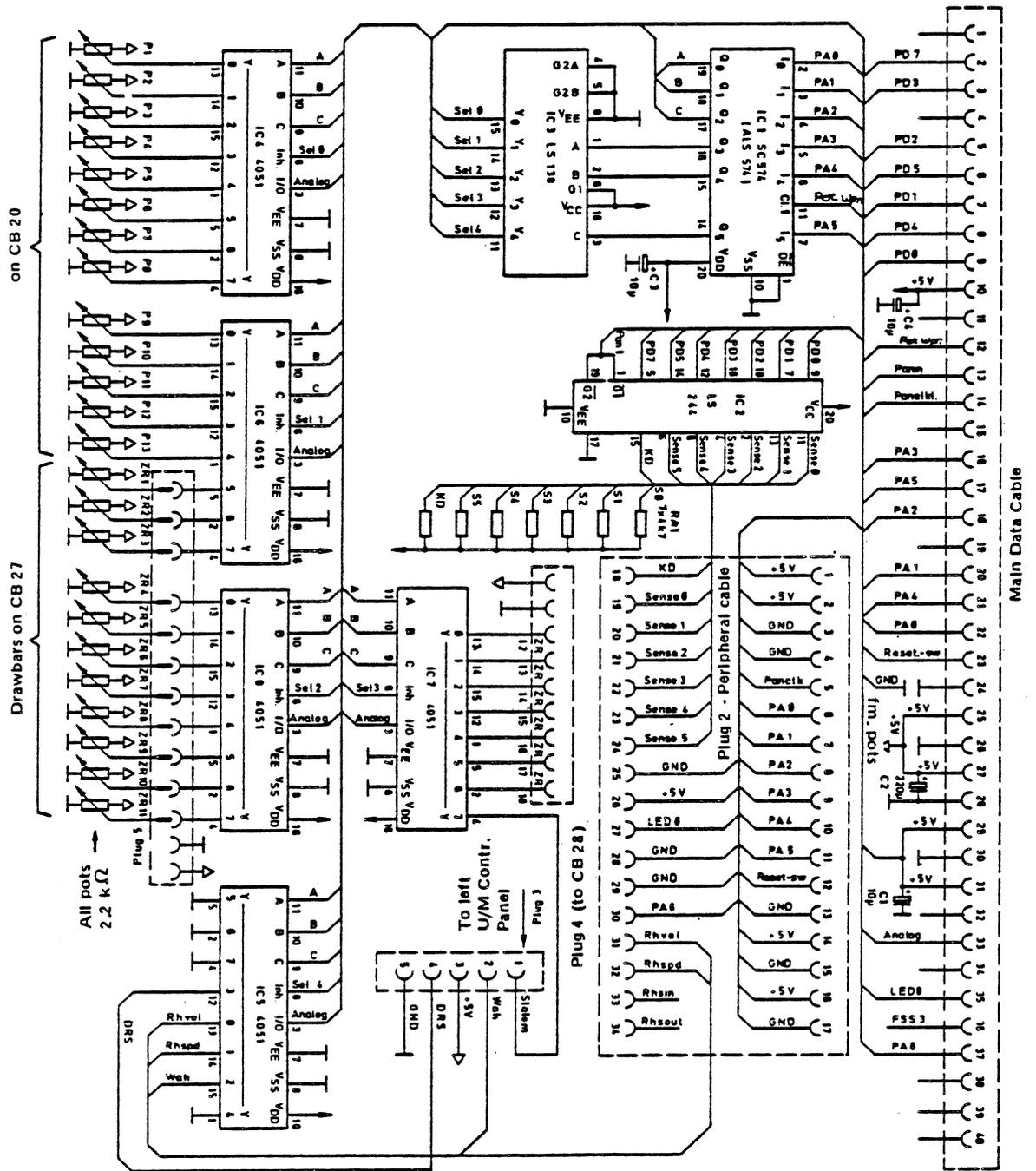


Fig. 20: Control boards CB 20, 27, 28 (slide controls, drawbars), schematic diagram

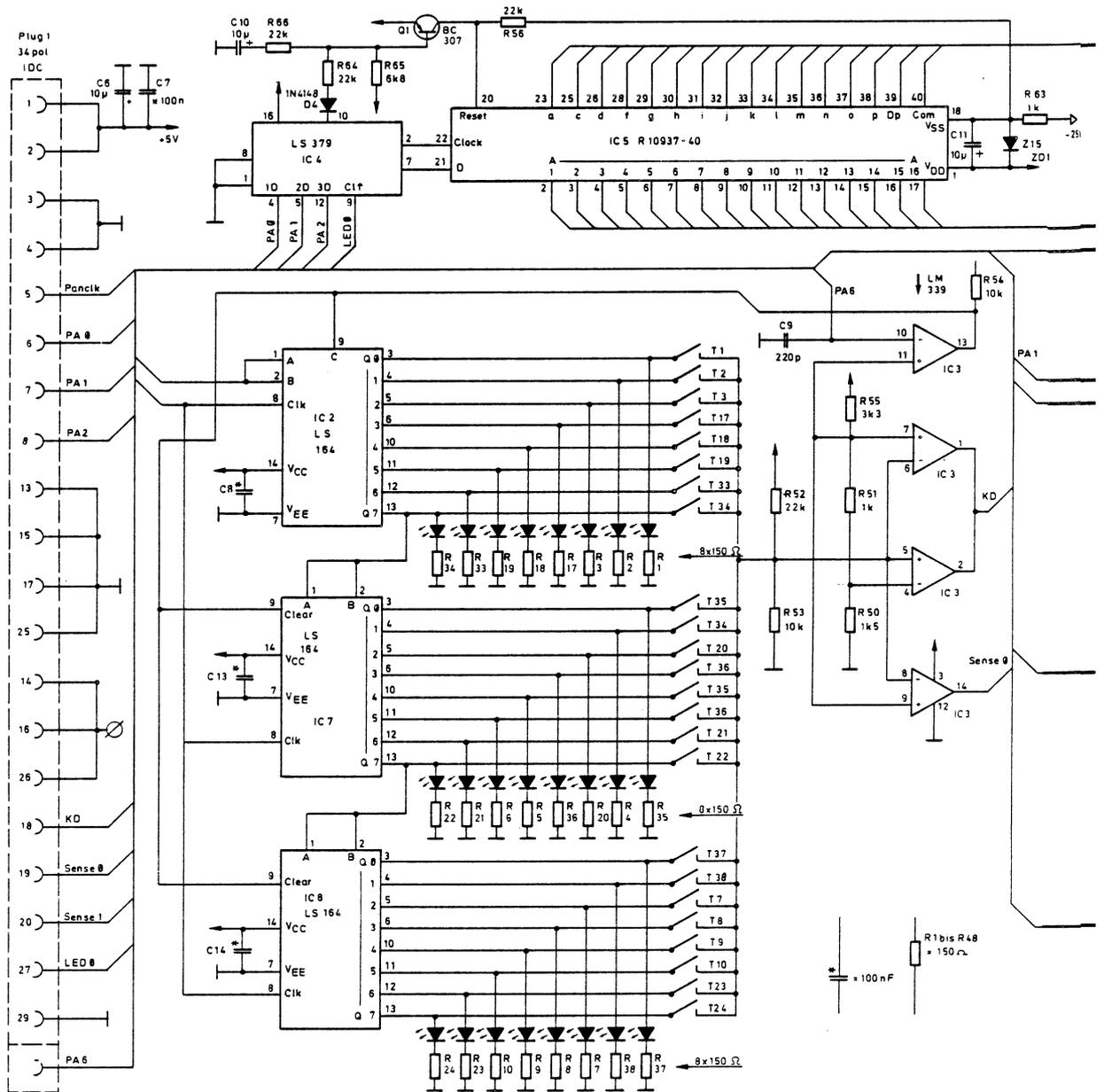
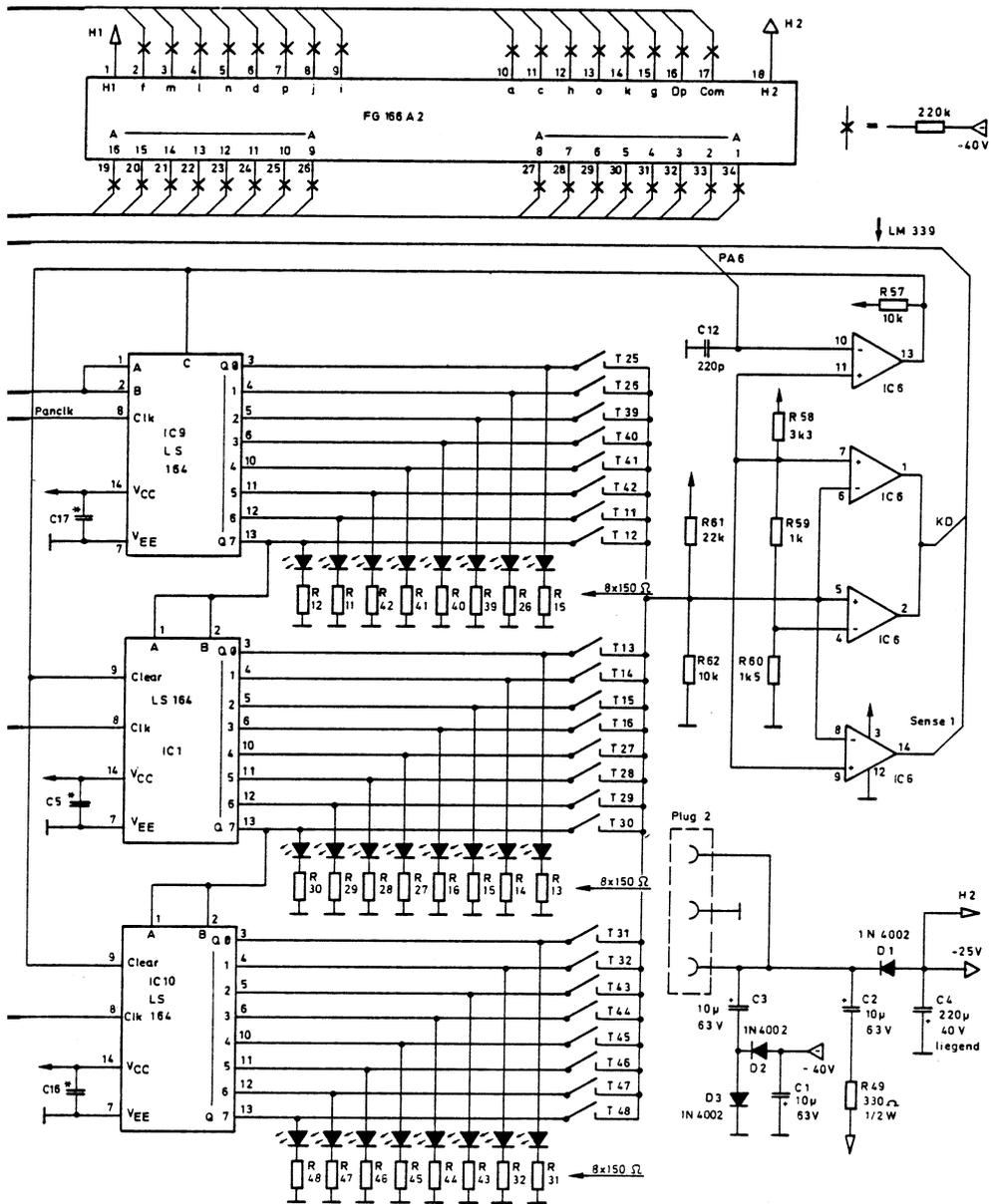


Fig. 21: Control board CB21 (switches, display), schematic diagram



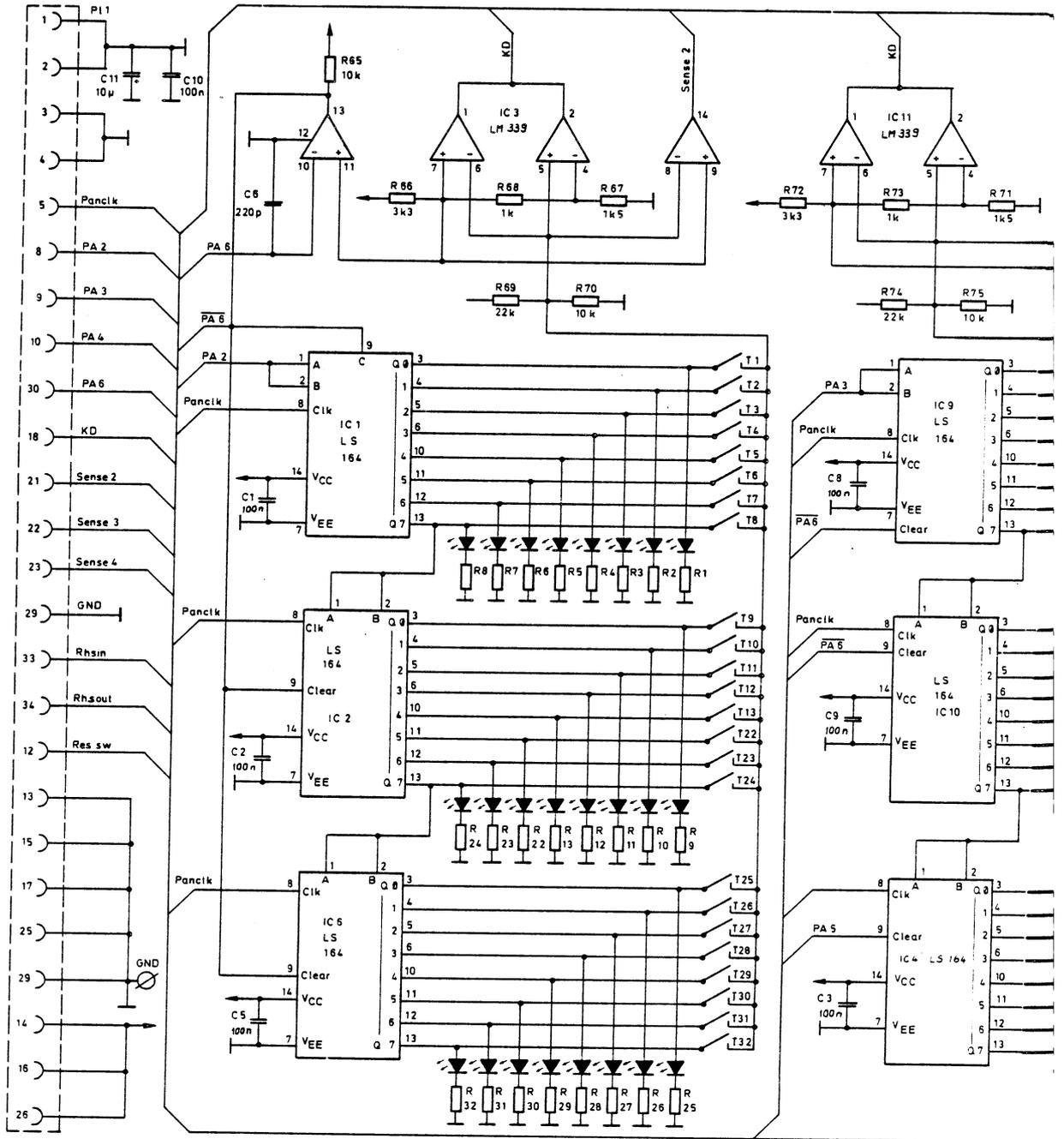
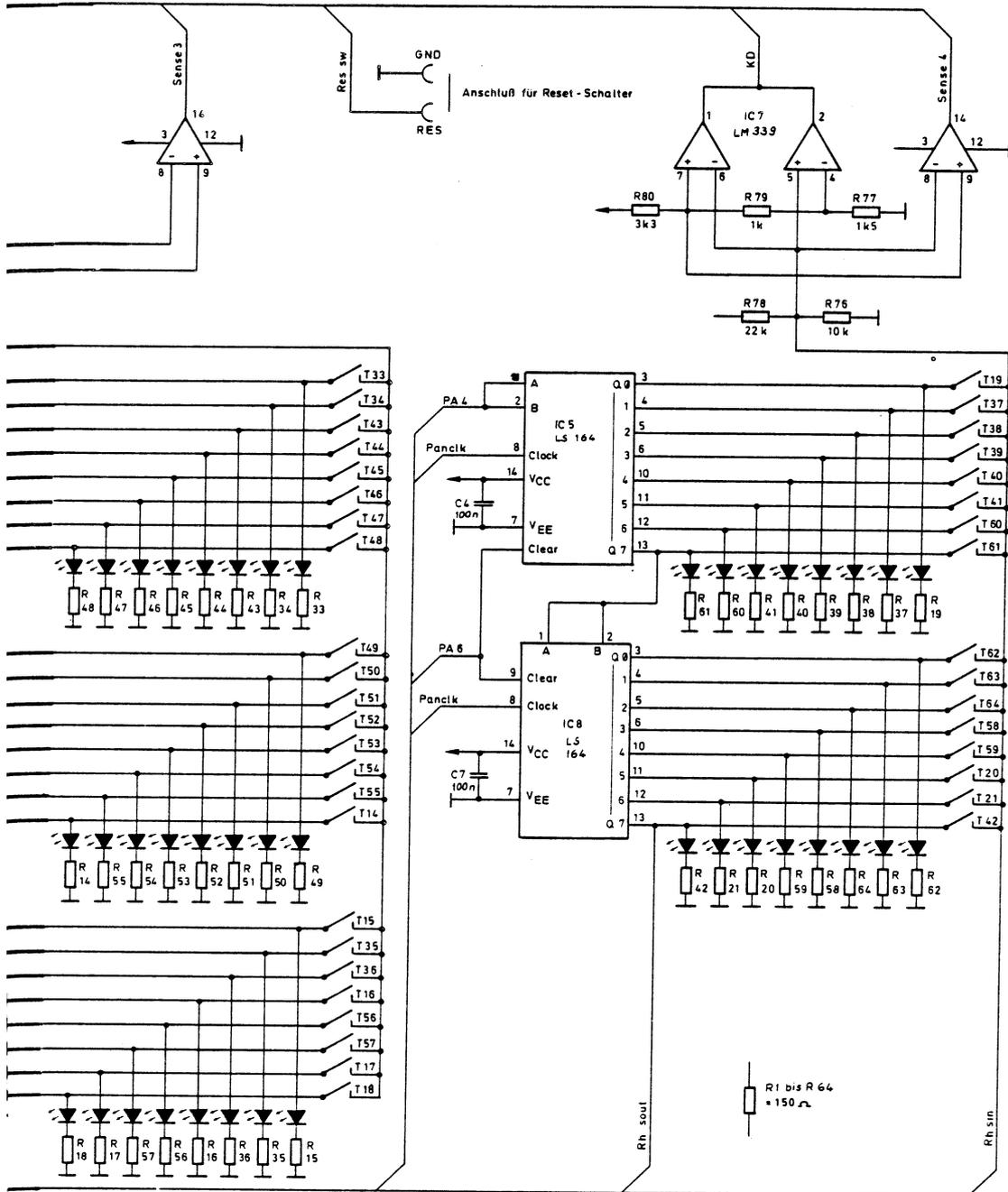


Fig. 22: Control board (CB22 switches), schematic diagram



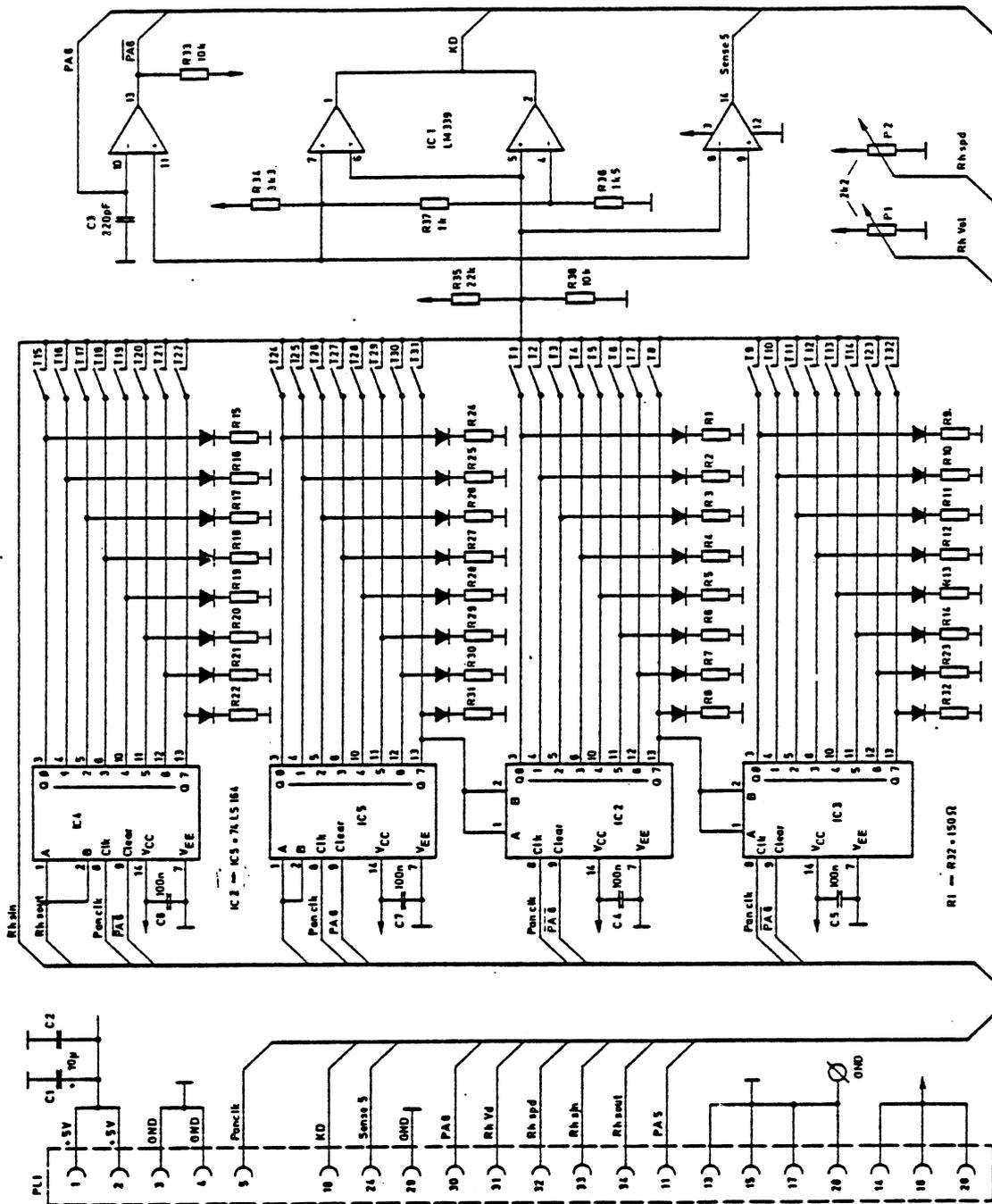


Fig. 23: Control board CB 23 (rhythm CX 4), schematic diagram

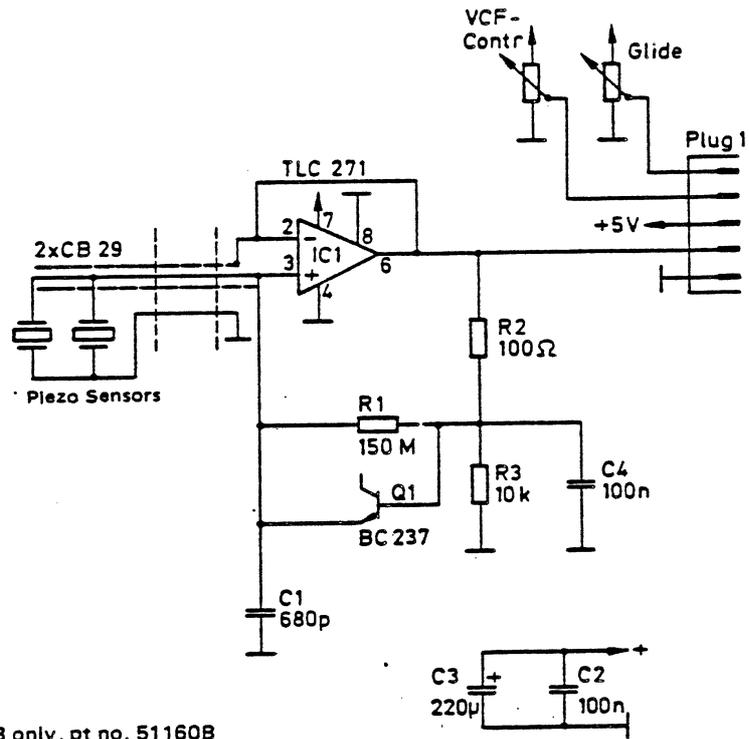


Fig. 24: Control board CB 25 (VCF, Glide)

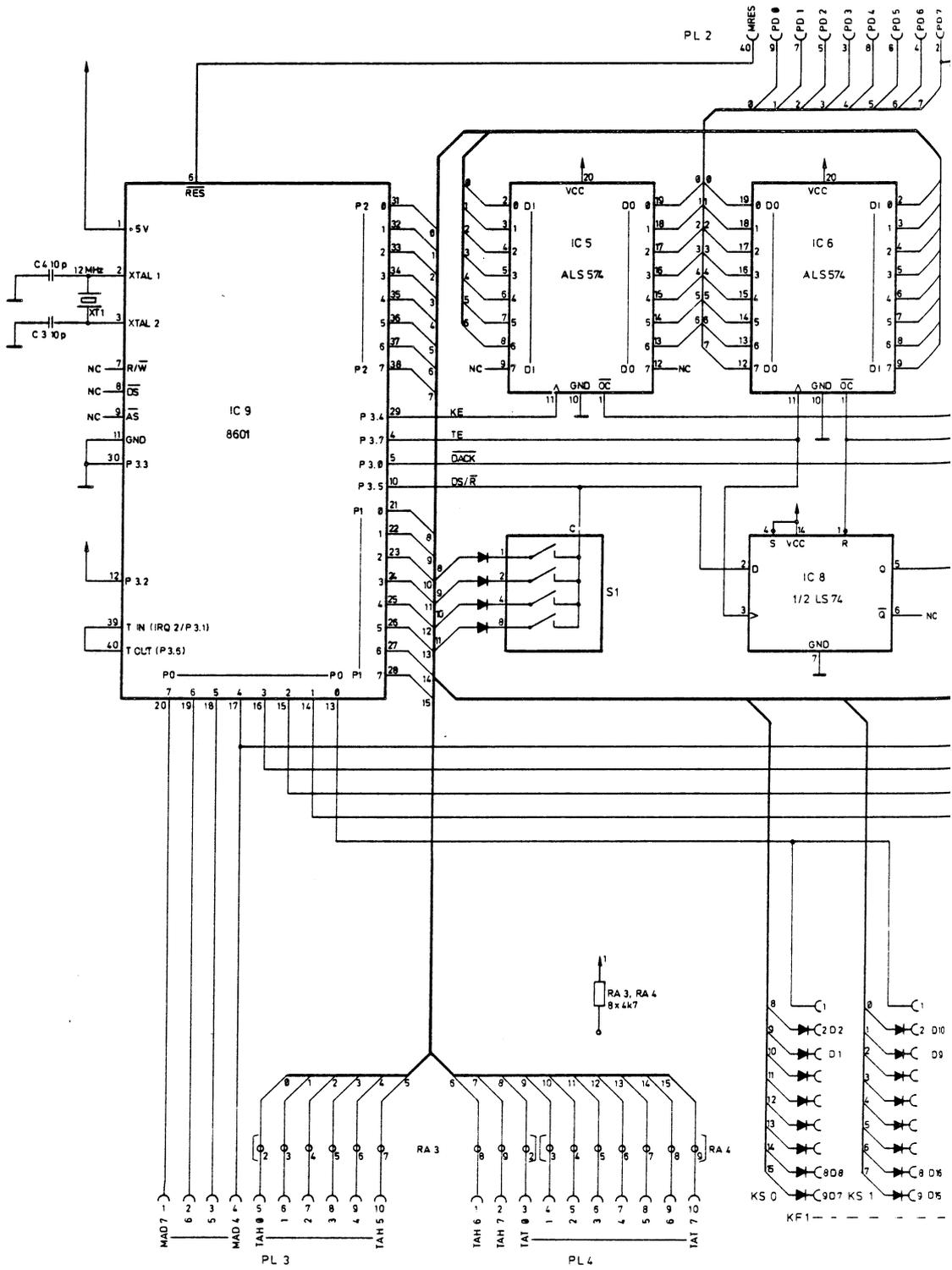
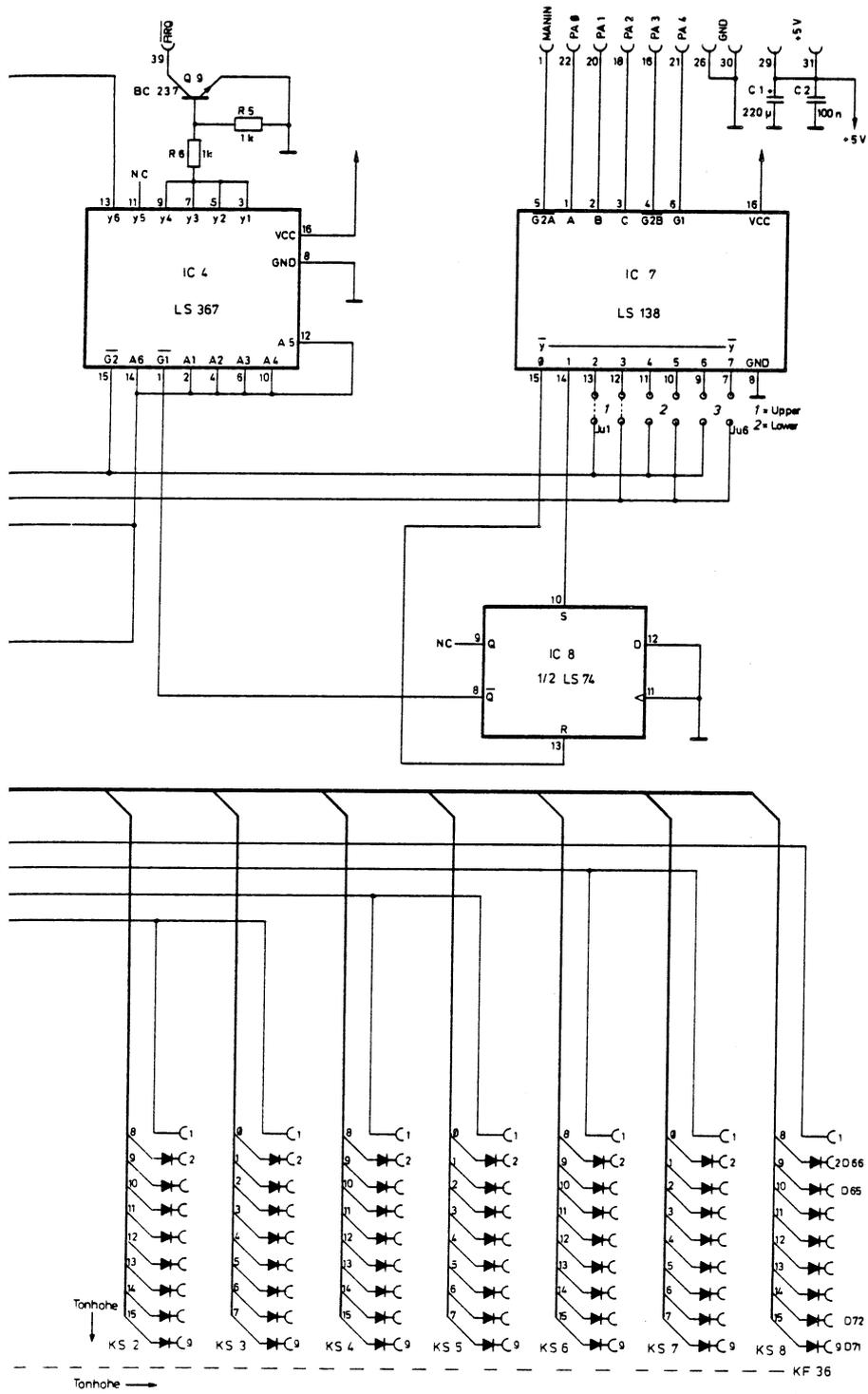


Fig 25a: Key contact board KD1, schematic diagram



2/796 b

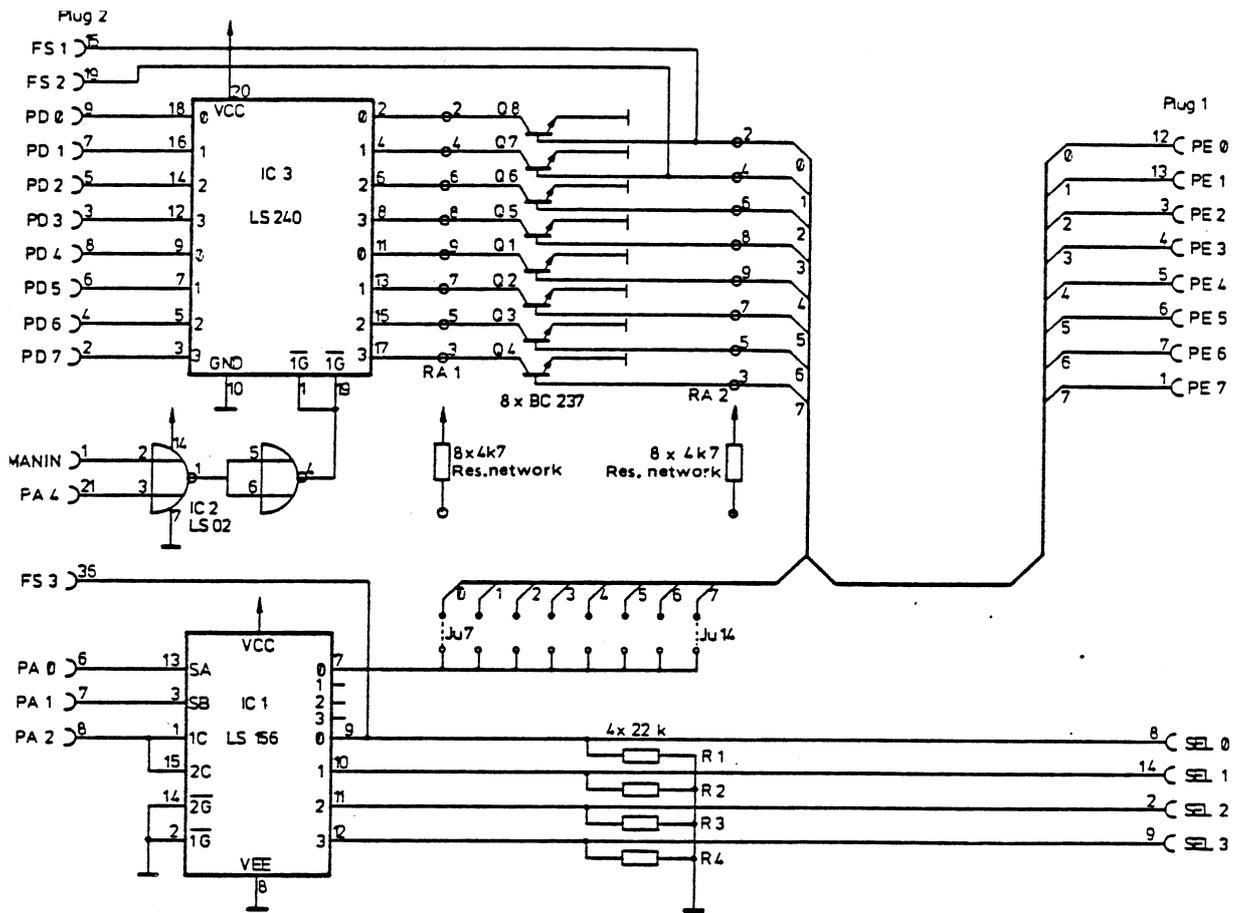


Fig. 25b: Key contact board KD1, schematic diagram b



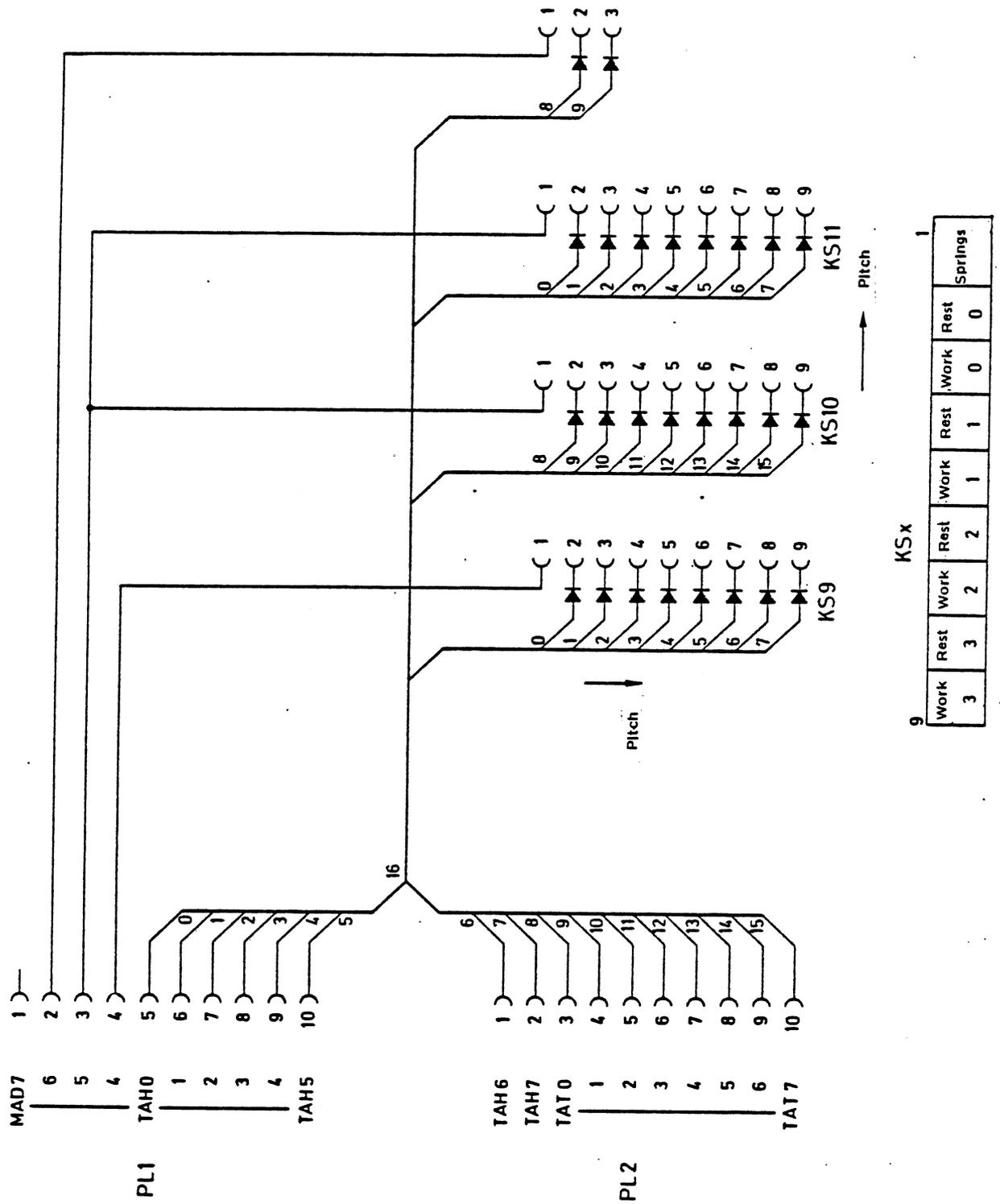


Fig. 27: Key contact board KD4, schematic diagram (DX400 only)

### III. Peripheral PC Boards

In this chapter, we will consider PC boards which are not directly involved in tone generation or processing but rather serve as connection or interface points to the "outside world" (peripheral) devices such as the swell pedal, a microphone, a tape recorder, headphones, M.I.D.I. or a personal computer.

#### 1. PCBoards CB24 and ST15

PC board CB 24 (Fig. 28) contains the interface circuitry for the Musical Instrument Digital Interface (M.I.D.I.), RS232 (home computer) and the audio output. (The BETA CP has the ST 15 instead of the CB 24). The M.I.D.I. function allows 'the organ to exchange data with other organ models or with other musical instruments which have M.I.D.I., a standard interface.

The organ transmits keying and registration information via "M.I.D.I. Out" jack, pin 5. The voltage which exists on pin 4 drives the LED in the optocoupler of the receiving device (cf., "M.I.D.I. In" jack). The optocoupler assures complete electrical isolation between the instruments. The received data goes to the organ electronics via IC1 - The data from the home computer passes to and from the organ electronics via the RS232 connector -

On ST 15 (Fig. 29), the BETA CP's swell pedal connects to the "Volume" jack. As explained under AF 12, the overall volume of all the organ signals is controlled by a varying control current from the swell pedal. Zero current means the volume is down completely; increasing current (up to 0.6 mA) means increasing volume. The BETA CP is transportable and the swell pedal is a separate unit. What do you do if you have forgotten to bring the swell pedal? Well, the ST 15 contains a "missing swell pedal detection switch" circuit that will save you from total embarrassment.

If nothing is connected to the "Volume" jack, QS is nonconductive because its base is held virtually at emitter potential by R22 and R24; this makes QS look like a high resistance. In this case, R23 and D4 supply an auxiliary current to the 'abc line; this current goes to the AF 12 board, maintaining the amplifiers at full volume.

If the swell pedal (PC board OS 1) is plugged in, the LED current of the optocoupler (on O51) flows through P1, R3, pin 4 of the jack and R24 on ST 15, causing a voltage drop across R24 sufficient to turn on QS. This biases D4 off, stopping the auxiliary current. The swell pedal now has sole control over the volume.

#### 2. PC Boards MP 2 thru MP 6

These boards (Fig's 30 thru 34) contain the interfaces for microphone, headphones and tape recorder. They differ by organ model:

MP 2: DX 400	-	Microphone amplifier
MP 3: DX 400	-	Microphone connector panel
MP 4: DX 500	-	Microphone amplifier and connector panel
MP5:DX40OCP	-	Microphone, tape volume
MP6:DX40OCP	-	Headphone amplifier, interface panel for tape recorder and microphone

#### 3. PC Boards TS Sand PA 10

The TS 5 triac switch and the PA 10 power amplifier are both in the power chassis LE 30. The LE 30 also contains the power transformer, which, along with the line voltage components and wiring, is enclosed in a protective metal cage.

##### a) Triac Switch (Fig. 35)

The line voltage comes in via the AC panel plug and goes first through a noise/static suppressor. It then goes through one of the two fuses, depending upon the line voltage being used, through the transformer primary winding and through the triac switch.

The triac switch TS 5 is comparable to a relay: a low-level, non-dangerous voltage is used to activate a relay, which then switches through the high line voltage. In this way, the operator is not exposed to the line voltage. In this case, triac BT139 functions as an electronic relay. Its trigger voltage comes from winding W2 of transformer NT1 -

Since NT1 is permanently connected across the AC line via R1, the triac is triggered on each half cycle of line voltage, so that the circuit between pins 3 and 8(7) of PC board TS5 is continuously switched through. However, if winding W3 of NT1 (only a minute voltage is present here) is shorted out by a closure across pins 9 and 10 of TS 5, there is no longer any voltage on any of NT1's windings and the triac cannot be triggered on. It can only be switched through again when the short across W3--present when the organ's AC line switch is closed--is removed. Hence the AC line switch works just the reverse of the way a power switch normally works: when it's closed, the organ is off; when it's open, the organ is on!

**Note:** Even with the triac off, there is a small residual voltage on the transformer primary through the RIC unit (100 Ohms/0.1 uF). This voltage is stepped up by the transformer and goes to the organ electronics, causing some LEDs to glow dimly. This phenomenon is completely harmless and consumes an insignificant amount of current. The master switch on the AC panel plug can completely switch off any current to the organ, but this is normally left on.

**b) Power Amplifier PC Board PA 10 (Fig. 36)**

PC board PA 10 contains the power supply for the hybrid power amplifiers, the speaker switch relay and the discrete components for the hybrid amplifiers. The two hybrid amplifiers are mounted separately on the LE 30 heat sink and are connected to the PA 10 board via ribbon cables. Fig. 36 shows the circuit for only one hybrid amplifier since the two circuits are identical. The corn-

ponent designations of each circuit's discrete components are shown together (ex., R7/14).

When the LE 30 is first turned on, capacitor C4 charges slowly through R1 until the collector voltage of Q1 becomes high enough to turn on Q1. This energizes the relay; the relay's contacts close, completing the connection from the amplifiers to the speakers. This delayed speaker turn-on permits the electronic circuits to stabilize before the speakers are connected, preventing a loud "Pop" or "blupp" at turn-on.

The audio signals coming in via input jack Plug 6 pass through the stereo volume control pot to the hybrid amplifiers, where they undergo voltage as well as power amplification. The discrete components associated with each amplifier determine its sensitivity and bandwidth. The output signals go through the relay contacts and the switching phone jacks to the external speakers.

Model	Cabinet kg/lb	Assembled kg/lb	Bench kg/lb	Pedals kg/lb	Leg Assm. kg/lb	Dimensions H x W x D cm/in
BETA S	62.0 / 136	85.0 / 187				102.0 x 116.5 x 67/ 40 x 45.8 x 26.4
BETA T	59.5 / 131	82.5 / 182			9.5 / 21	101.5 x 116.5 x 67/ 40 x 45.8 x 26.4
BETA CP		38.0 / 84		5.0 / 11	9.2 / 20	95.5 x 117.0 x 66.5/ 37.6 x 46 x 26
BETA CP with cover		45.0 / 99				
DELTA	57.5 / 127	83.0 / 183	14.5 / 32		10.0 / 22	106.0 x 123.0 x 70/ 41.7 x 48.4 x 27.5
GAMMA	59.0 / 130	87.0 / 191	16.0 / 35	22.0 / 48		107.5 x 123.0 x 70/ 42.3 x 48.4 x 27.5

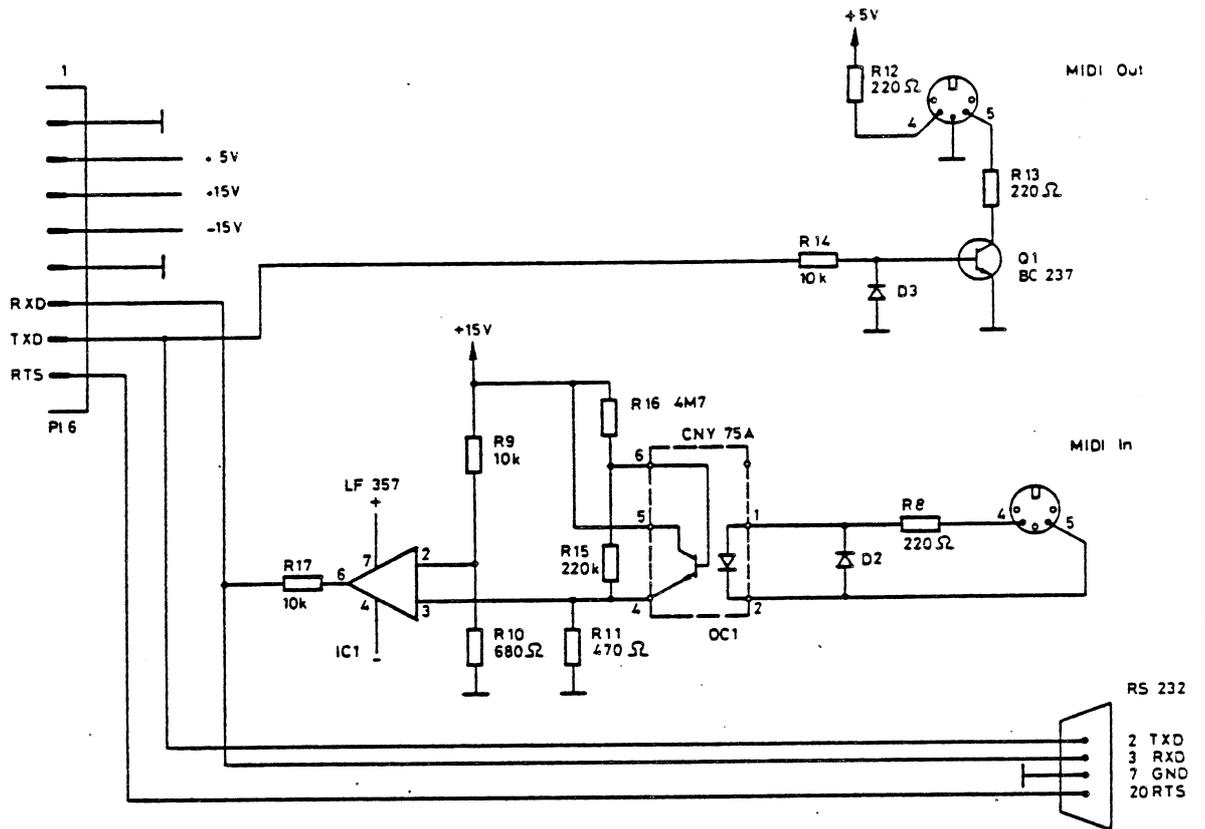


Fig. 28: Interface panel CB 24, MIDI circuits

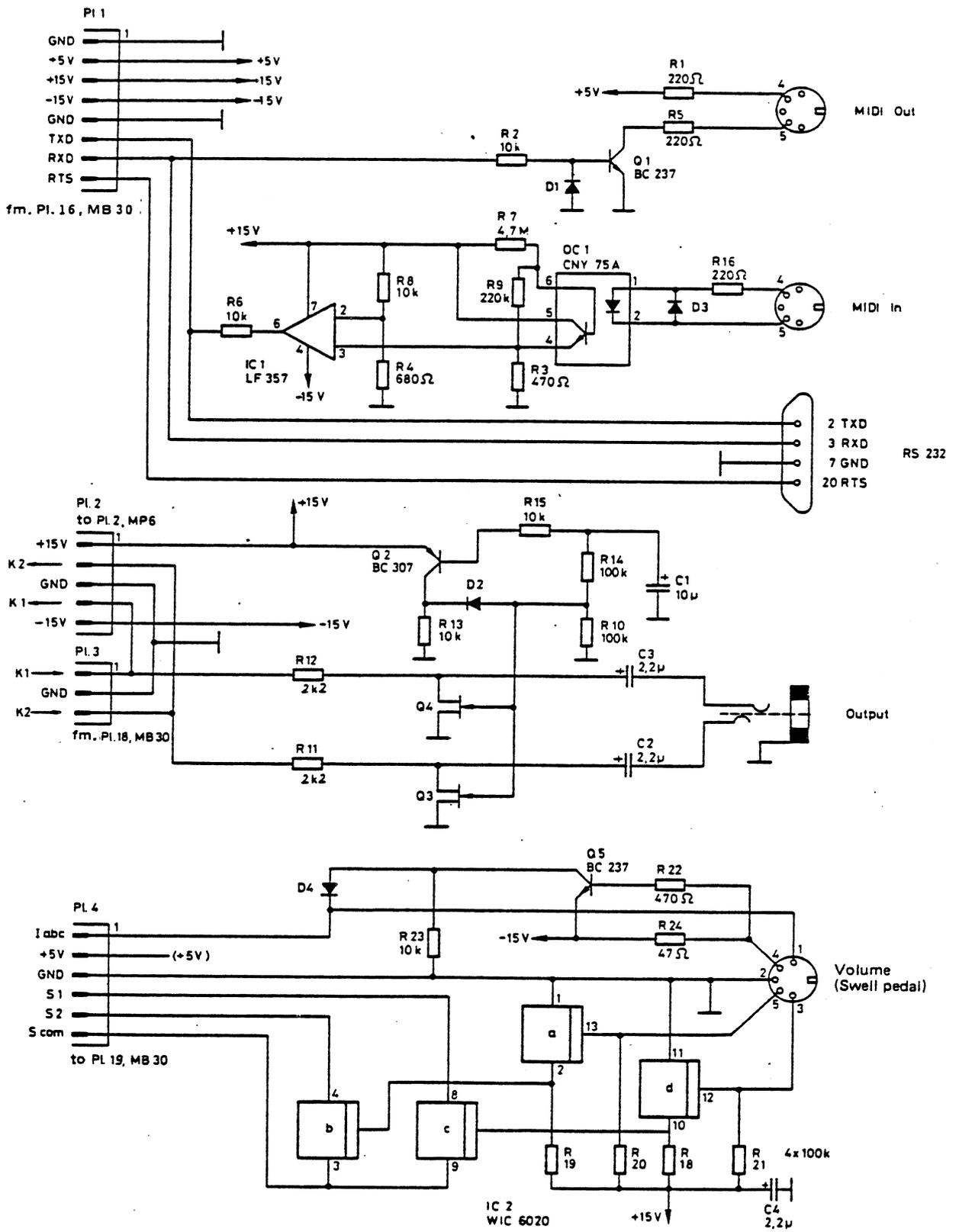


Fig. 29: Interface panel ST 15, schematic diagram (BETA CP only)

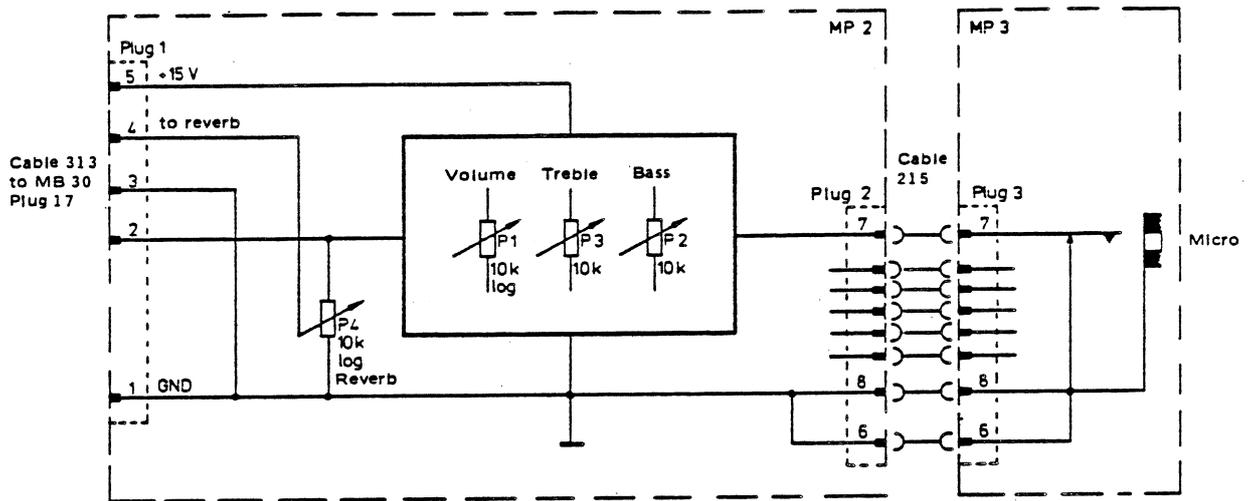


Fig. 30: Microphone preamp MP 2/MP 3 (BETA S and T only)

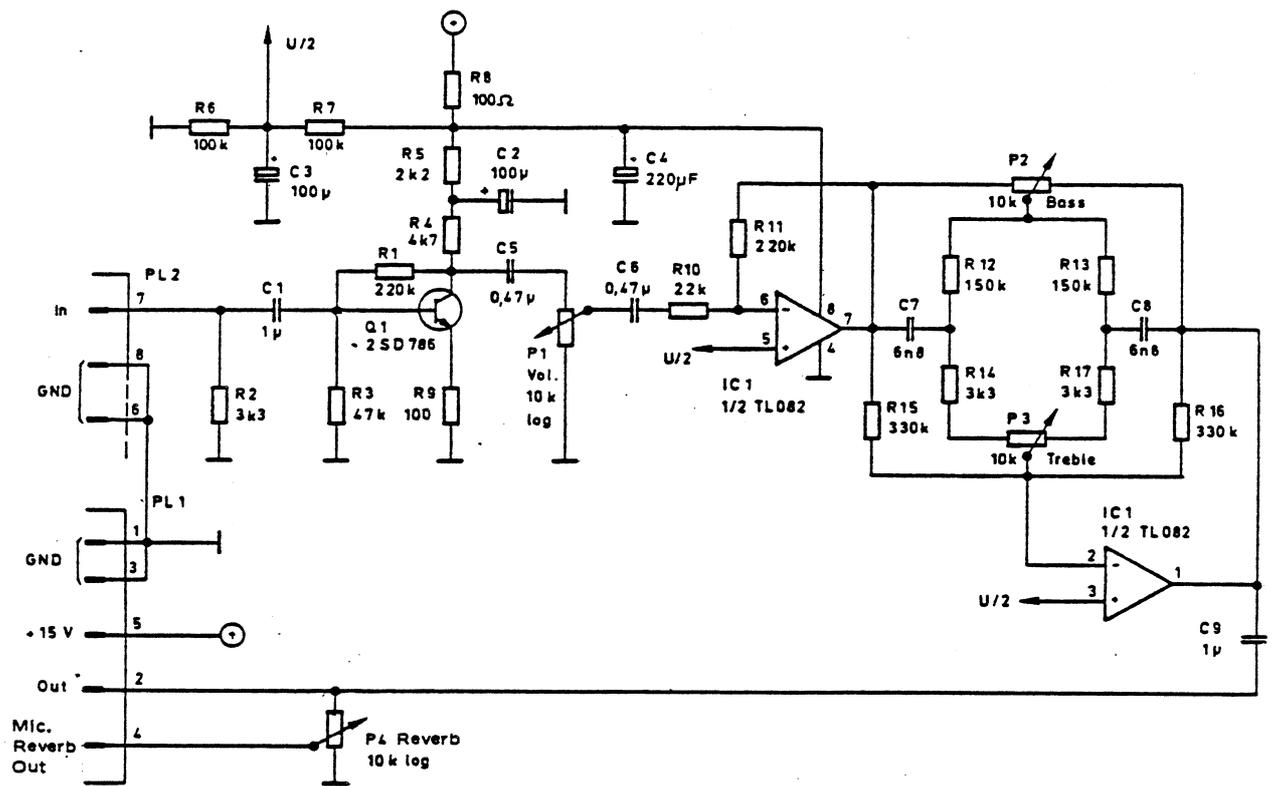


Fig. 31: Microphone preamp MP 2 (BETA S and T only)

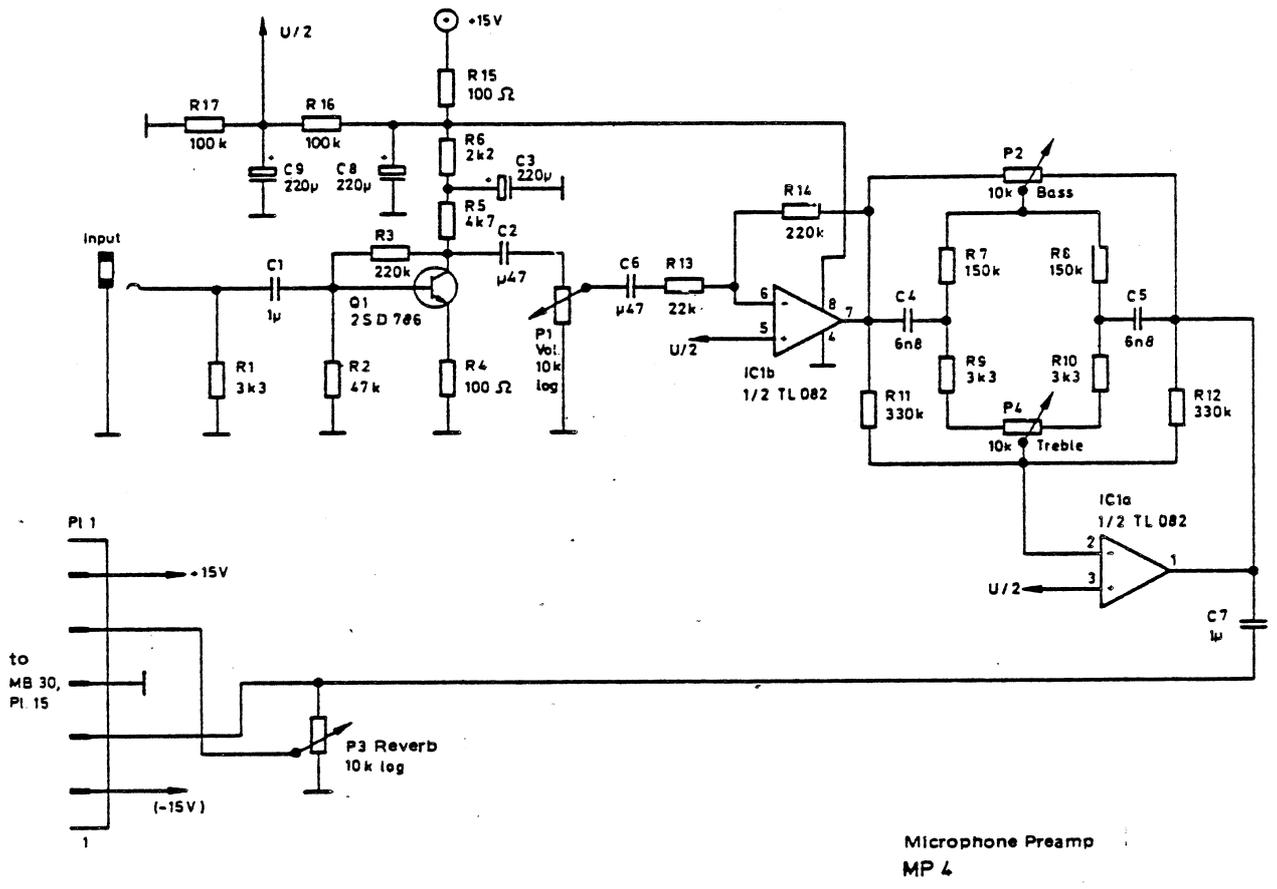


Fig. 32: Microphone preamp MP 4 (DX 500 only)

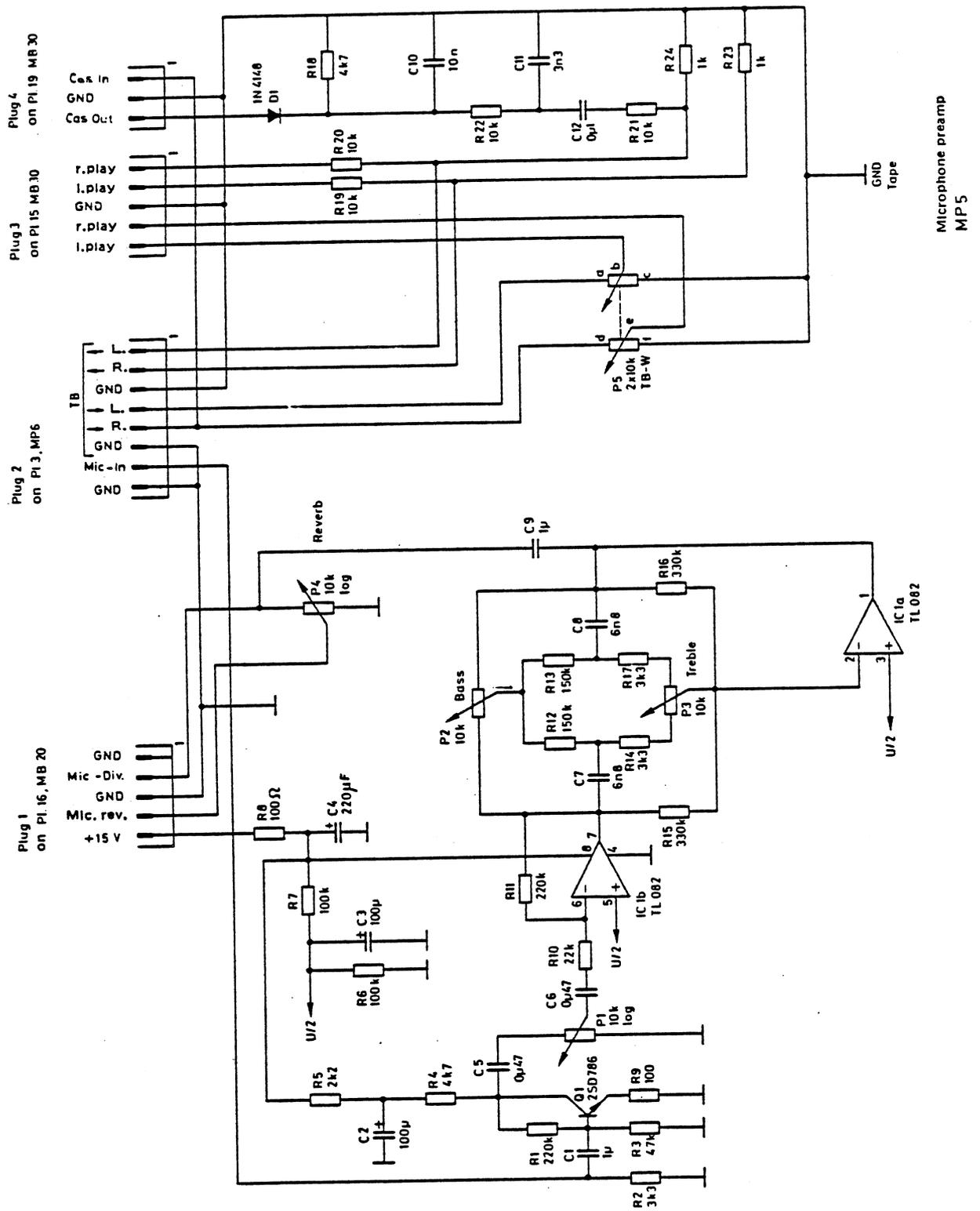


Fig. 33: Microphone preamp MP 5 (BETA CP only)

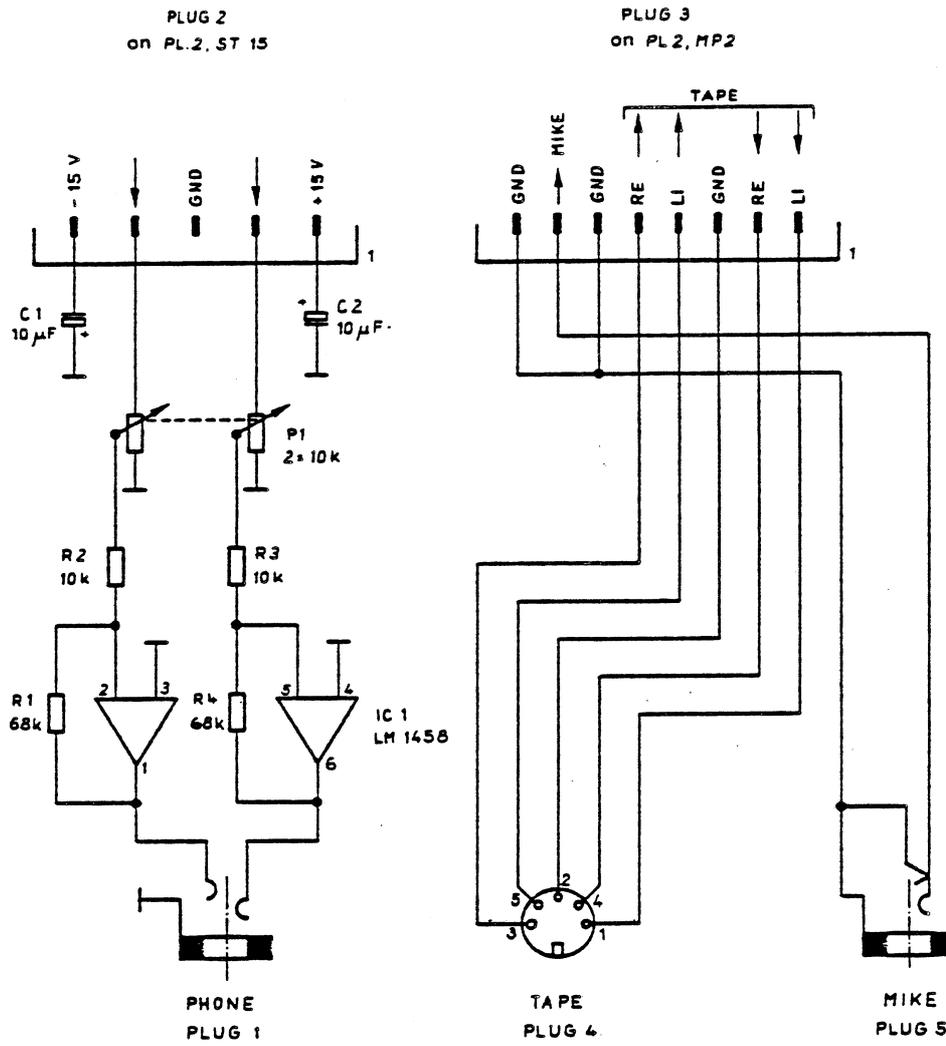


Fig. 34: Headphone amplifier NIP 6 (BETA CP only)

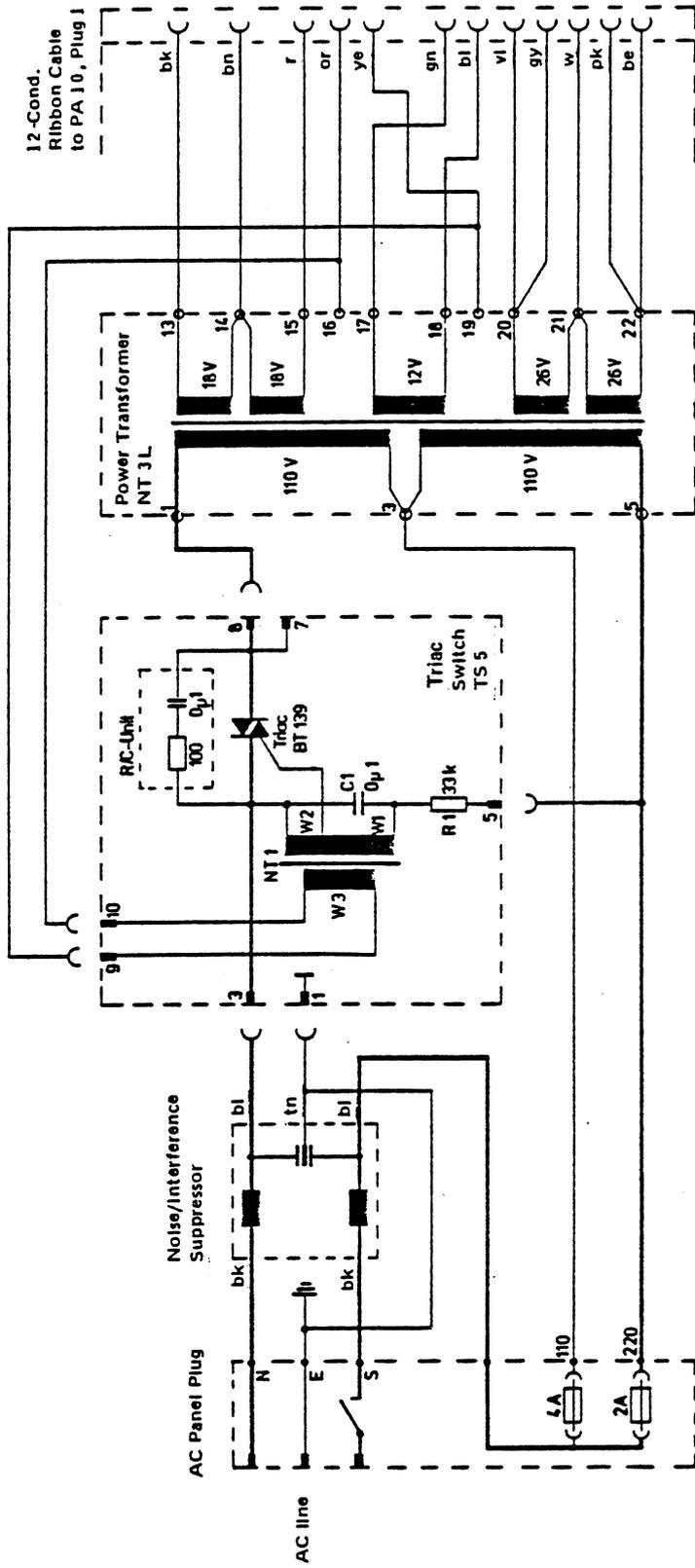


Fig. 35: Triac switch TS 5, schematic diagram (not in BETA CP)

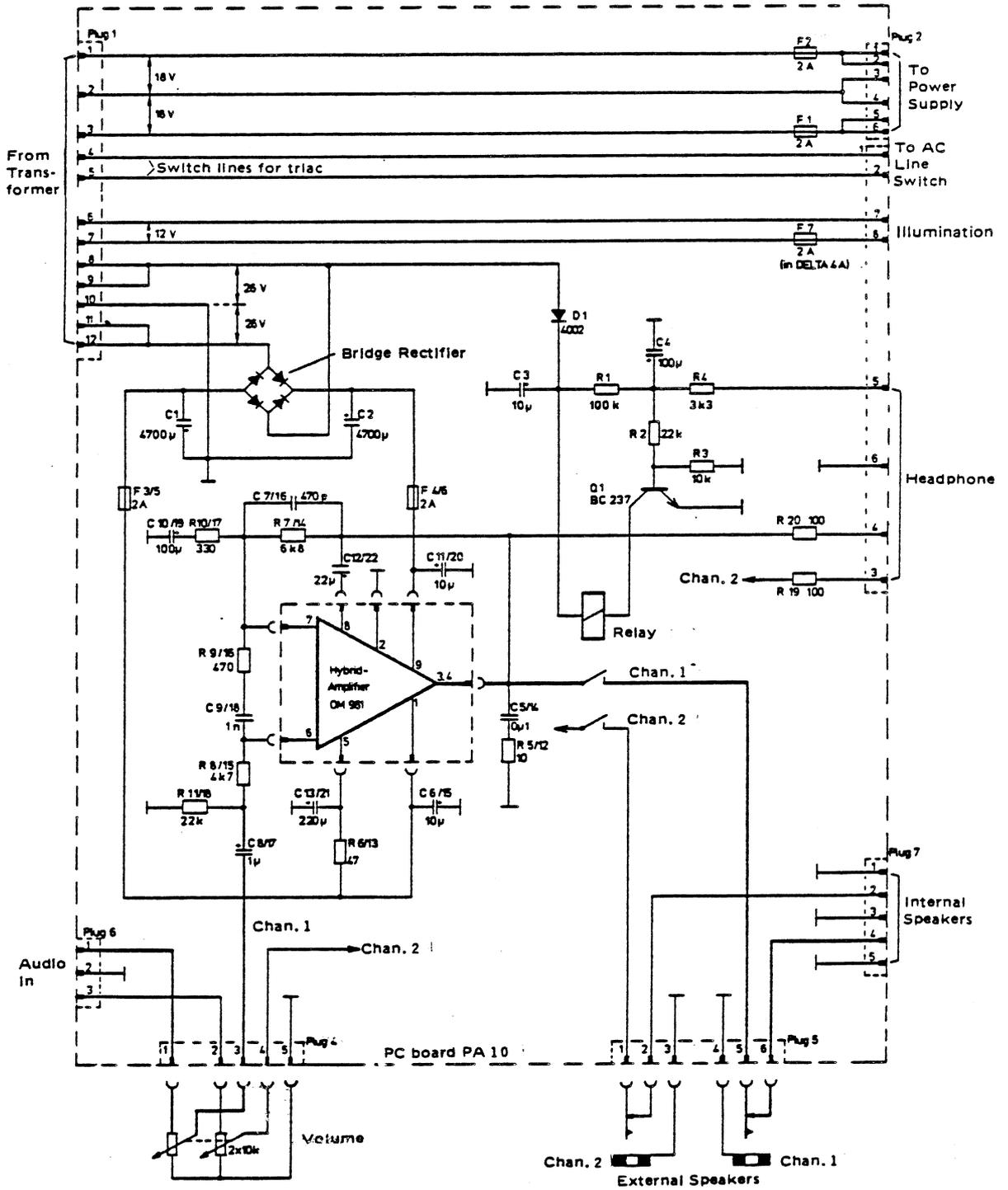


Fig. 36: Power amplifier PA 10, schematic diagram (not in BETA CP)

## D. GLOSSARY OF MICROCOMPUTER TERMS

If the technical data presented in this manual is your first contact with microcomputer technology, you have doubtless encountered some concepts which are new to you and will need some explanation. Some of the more frequently used digital "buzzwords" are presented here in alphabetical order::

**Address** - A unique and selectable position in a memory device (see RAM, ROM, EPROM) where data can be stored and retrieved.

**Address Bus** - A collective term for the lines by which address data is transferred from the central processing unit (CPU) to the memory devices.

**A/D Converter** - Analog- to-Digital converter; a device that converts analog data, i.e. data that is continuously variable, such as audio, into digital data for processing and storage in a digital system.

**Bit** - Short for "binary digit," or the smallest unit of information in the binary number system. A bit represents one of two fixed states, a "1" or a "0".

**Bus** - Collectively, the lines on which data is transferred; ex.: data bus, address bus, control bus.

**Byte** - A sequence of 8 adjacent bits, considered as a digital unit.

**Chip** - A small piece of semiconductor material on which integrated circuits are fabricated; the actual circuit of an IC.

**Clock** - The timing signals used in a microprocessor system; the circuit used to generate these signals.

**CPU** - Central processing unit; the unit in a computer that performs and controls the execution of instructions, performs arithmetic functions, and generates clock signals.

**D/A Converter** - Digital-to-analog converter; converts digital data into analog signals such as control voltages or audio.

**Display** - A visual indicator used as a means to read out digital data.

**EPROM** -- Erasable programmable read-only memory (see ROM); a ROM capable of storing new data; existing data can be erased and new data can be read into the ROM.

**Hardware** - The components which make up the computer proper: the electronics, control devices, readouts, etc. The hardware carries out the computer functions in accordance with the software entered into it. See Software.

**Interface** - A takeoff point for connecting a computer function to an "outside world", or peripheral, device.

**LED** - Light-emitting diode; a semiconductor device that emits light when voltage is applied to it. Used as an indicator.

**Master Processor** - In a multi-processor system, the processor that is responsible for the central control and execution of instructions for the rest of the system -

**Microprocessor** A single-chip (IC) central processing unit.

**Program** A series of instructions which define the steps a computer must follow to perform a given function.

**RAM** - Random access memory; a memory into which data may be stored or read out in any order at random. A RAM loses its stored data if its supply voltage is removed, and therefore must be protected against memory loss by a battery supply.

**ROM** -Read-only memory; contains permanent data entered at manufacture. The data cannot be altered and remains even if the supply voltage is removed.

**Slave Processor** - In a multi-processor system, a processor which is subservient to the master proces; or, from which it receives its instructions and control.

**Software** -- The computer programs, procedures and documents necessary for the operation of a computer system.

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