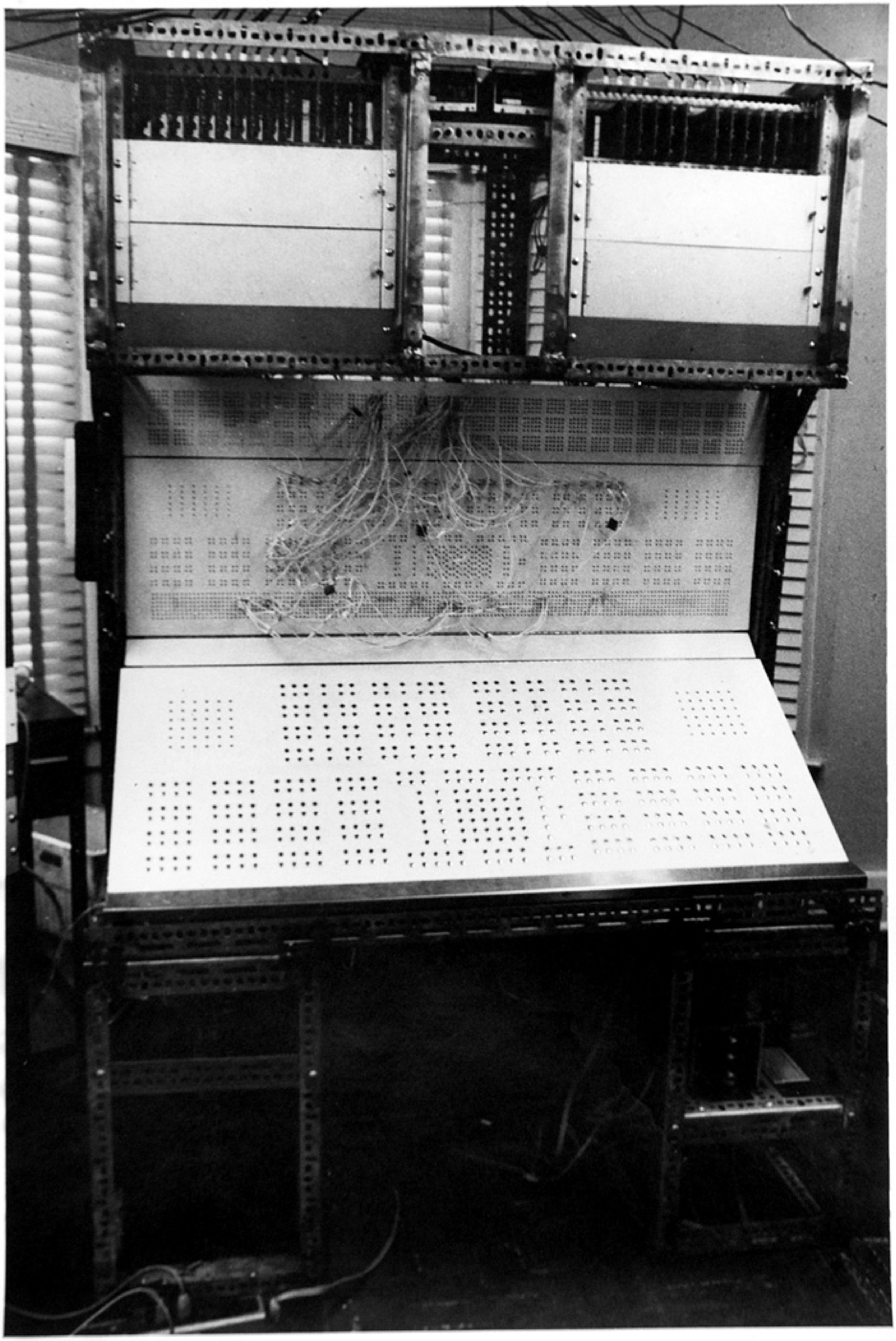


PROGRESS REPORT #1

AN ELECTRONIC MUSIC INSTRUMENT WHICH COMBINES

THE COMPOSING PROCESS WITH PERFORMANCE IN REAL TIME

April 1971

SALVATORE MARTIRANO

The report is divided into the following sections:

Introduction

"Can you play 'Melancholy Baby' in Eb?" Could Beethoven or Bachrach compose 'Mary had a little lamb'? "Yes, why not," say I. Do I and did he? Custom dictates that a discussion about music making shall begin with a consideration of its motivation, message and significance. Does anyone know or ask why a shoemaker makes shoes? Music is as necessary as a pair of shoes: the process for the crafting of each is similar though not necessarily identical.

Part of the difference between music composed for electronic and automated circuitry and music composed for instrumentalists and vocalists is this: the emphasis, in the first case, is placed only on a study of the limits of human perception whereas in the second case, a study of the performer's capability is the subject which de facto sets the limits of perception. I'd like to soften the intention, though not the meaning, of the last statement. My intention in this comparison is not to negate traditional music. Composers always work within limits and many, including myself, only claim that music which involves concepts of automation is a logical development.

A composer of music for musicians works alone at the creation of tasks for others to perform. This is not the case here. A group, consisting of specialists, students and faculty have come together in the past 2½ years to create, develop and share responsibilities and ideas.

A statement about faculty, students and specialists

The following is a brief introduction to those who have contributed ideas, talent and work in the past, and, to members of the group presently involved.

In 1969 I audited Dr. Malmstadt's summer course, "Digital Electronics for Scientists". Subsequently, Dr. Malmstadt was kind enough to lend me two Heathkit Analog-Digital Designers. I used these devices, drawing guidance from the Malmstadt-Enke book, "Digital Electronics for Scientists" and I was able to make a little progress toward an understanding of the basic principles governing logic design.

In November, 1969, Mr. Dominic Skaperdas of the Coordinated Science Laboratory here at the University designed the digital circuitry for the sound distribution system which was later expanded to become the present system.

Four concerts were presented: 1) at the Krannert Center for the Performing Arts on February 25, 1970; 2) at the Art Institute of Chicago on February 27, 1970; and 3) at the University of Wisconsin-Madison on March 1, 1970; at the Temple University Festival of Music, Philadelphia, Penn. on July 20, 1970.

In September, 1970, I first met Dr. James Divilbiss who is presently a faculty member at CSL and in the Graduate School of Library Science. The MAR-VIL CONSTRUCTION (VIL for Divilbiss) was the result of six months of intensive effort. His logic designs formed a basis for the present system. He also designed the documentation systems that are presently in use. Further he designed and built most of the frame that houses the instrument. Still further, he contributed enormous stores of components. With his knowledge and experience he was able to design circuits using components that are economically feasible.

Two concerts were presented with the MAR-VIL CONSTRUCTION: 1) at Automation House, N.Y.C., N.Y. on March 5, 1971; (see N.Y. Times review, page 24a) and 2) at the State University of New York at Albany on March 12, 1971.

When it became impossible for Dr. Divilbiss to continue to give such a large block of time to the project he introduced me to Mr. Sergio Franco, Doctoral Candidate in the Computer Science Department, who accepted the half-time assistantship granted to me by the Research Board here at the University. His involvement in the project is complete. Not only has he created many of the designs presently in use but he has had the patience to set up the difficult experiments which resulted in the parameter specifications for particular circuits. Certain specifications, as the report will show, have turned out to be exacting, wide-rangeing and consequently difficult to realize. Nevertheless, he has accomplished a prodigious amount of work since June, 1971 in an exponential progression that promises much for the coming year. It is also expected that his work will form the basis for a thesis that will satisfy a requirement for his PHD in Computer Science. The full-time assistantship requested for the coming year is for Mr. Franco.

Mr. Thomas Noggle, an electrical engineering graduate student here at the University, inaddition to designing the expanded digital circuit for sound distribution, has acted as a general consultant and advisor for the past six months. For obvious reasons it has not been possible for me to pay him. Notwithstanding, he has given time despite the fact that he has a job and a formidable course of study that demands most of his time. His contribution is valued and appreciated. In my request for support I have included a one half-time assistantship for him.

A point that I would like to make is that though the project is exciting work for all of us, paid and unpaid, it is not amiss to give a helping hand to unlegislated interaction, whose lines cross a variety of disciplines.

Mr. Jay Barr, presently employed by the Bell Telephone System in Champaign, Illinois as a maintenance technician for their computer system, has worked with me during his off-hours and on weekends since 1970. His work is excellent, as an examination of the instrument will show. As much as to the design, it has been due to his work that there has been zero "downtime" caused by defective wiring.

Much of what has been accomplished is owed to the cooperation and advice received from many specialists connected with operations at the Digital Computer Lab and at the Coordinated Science Lab. Special thanks to Mr. R. C. Amendola and to Mr. Rich Borovec of the Illiac III Project. Incidently, Mr. Amendola took the photographs used in this report. Much is owed to a large number of people here at the University and it is important to stress that a project such as this could only be seriously considered at the University of Illinois. Proof that I'm convinced of this is evident if one considers that I passed my sabbatical leave, 1970-71, here in town rather than attempting to find another place to work and study.

It is unfortunate that occasionally within the University, cooperative efforts are inhibited just at the edge of an extraordinary achievement because of financial need. It's the nature of the beast and I do not doubt the existence of priorities. But, as the understanding and enlightened sage first said, "Though I live to know you, I live too, you know."

General description of the instrument

With technical projects which are goal-oriented and based on a model, team-work is not uncommon. Though there is no clear model here, this is not to say that we have not adopted as much as seemed useful from what others have done, or that a few basic ideas have not emerged and developed.

If there is a model it could only be described in the vaguest of terms. Would you believe -------- a musical instrument, in which varying quantities of information are controlled in a scale of perceivable steps that range from a pseudo-random set, through permutations that are predictable to a significant degree, to a totally predictable set. If a proper range is selected for a pseudo-random set, an analogy to traditional instruments exists. For example, consider the non-involvement of the pianist and piano-maker in the specifics that define the individual decay rate of each string on each piano when a tone is held.

An idea for the system that I'll describe in musical terms involves a total of 73 instruments (individual sound sources) that are divided into four orchestras. I'm inclined to form orchestras that have a specific and limited set of mixed characteristics rather than grouping the instruments according to type: (e.g. --1) voices; 2) winds; 3) strings; 4) percussion.)

Each orchestra has a soloist and an accompanying group of instrument

Orchestra I: solo voice; a chorus of 18 voice sounds consisting of vowels, diphthongs, syllables, barks, etc.; 13 strings; 1 tunable timpany; 1 harp; totaling 34.

Orchestra II: solo sax; 20 percussion consisting of shake instruments, tunable skins, dead and live metal, wood; totaling 21.

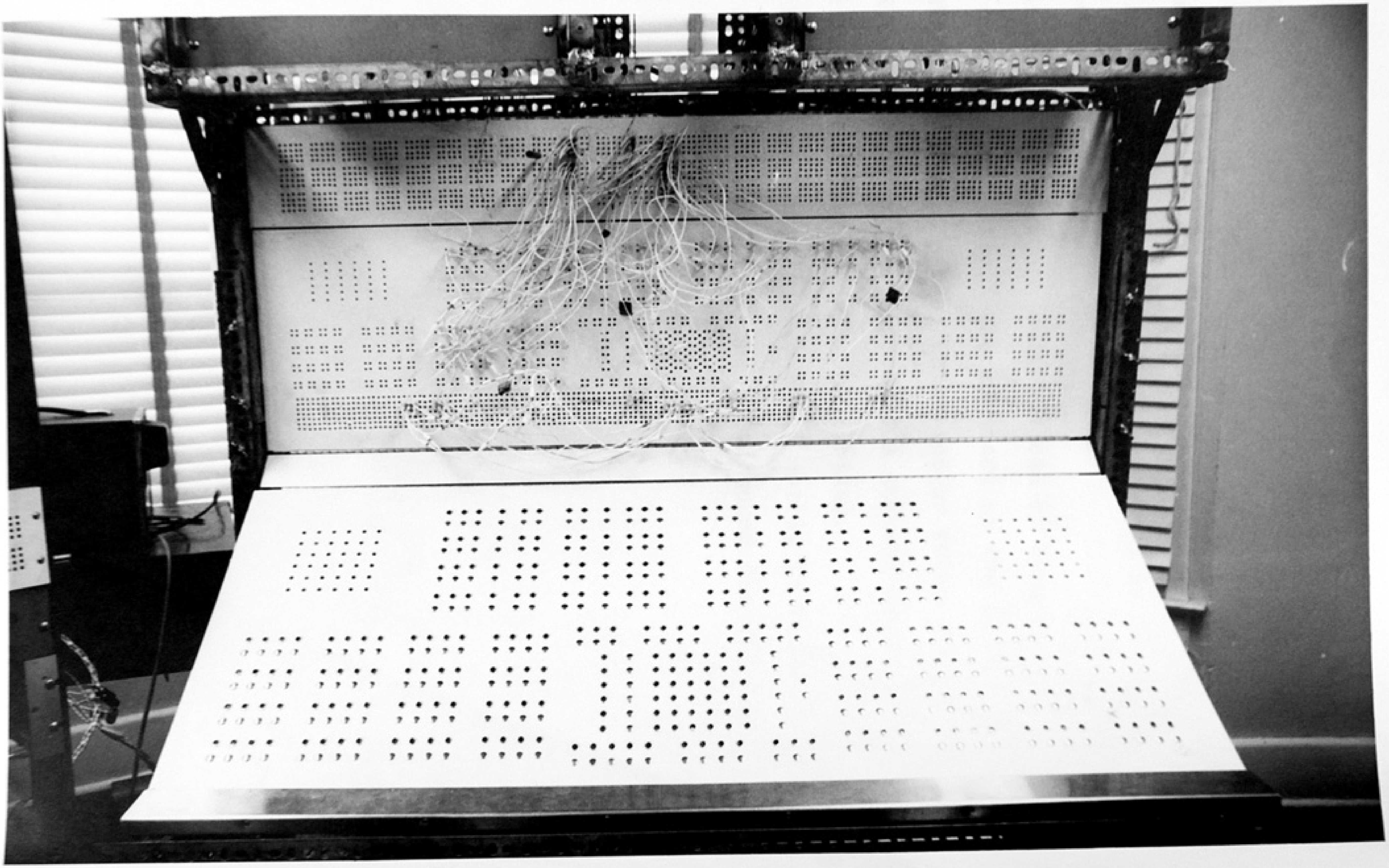
Orchestra III: solo trumpet-trombone; 12 winds; totaling 13.

Orchestra IV: solo violin; 3 flutes; 1 double bass; totaling 5.

The composer will directly control the 4 soloists and will allow the same information to be used by the accompaniment after being modified by a function that can also be specified in real time according to a number of options.

A sample list of options that an accompaniment will have are: 1) rest; 2) play without soloist; 3) subtract collection "X" from information; 4) add collection "Y" to dynamic level; 5) inhibit evolution of the program; etc.

The above description should be understood as an outline for a system of sound differences and is not to be literally interpreted. We are not interested in developing instruments which will imitate those presently existing. (see pp. 36, figure 1 Overall view of control system)



A portion of the system is complete. However, intrinsic in the concept and allowed for in the package, are possibilities for change as study and feedback stimulate the development of new ideas and techniques.

An array of 291, 2-state switches can be controlled by logic circuits or played manually. The keyboard of the instrument features TOUCH CONTROL (no moving parts) which set or reset a flip-flop when the TOUCH CONTROL circuit is closed to a current source. (see pg.39 fig. 3, circuit designed by James Divilbiss).

At the center of the panel a 6 X 6 matrix is built into the physical design to accomodate control of sound distribution in space.

The patch panel is best described as 3 sections, each with a separate function.

a. set and reset inputs and outputs to the TOUCH CONTROL.

b. inputs and outputs to logical circuits such as shift registers, memories, counters, gates, etc.

c. inputs to analog controls, such as D/A converters and analog switches.

As well as an analogy to traditional instruments, an analogy to traditional composition exists, considering that particular patching patterns are developed over a period of time. In performance a composer can improvise, better said, compose in real time, within a large set of musical possibilities.

A system is presently in use for duration or clock rate which allows independent control of six clocks. The rate of each clock is continuously controllable from 60 seconds per cycle to 8,000 cycles per second (cps). The 6 clocks can be "ANDed" and "EXCLUSIVE ORed" in a variety of ways. For example, 6 groups of 4 clock rates can be ANDed in the following manner: 1,2,3,4; 2,3,4,5; 3,4,5,6; 4,5,6,1; 5,6,1,2; and 6,1,2,3. A common denominator frequency which is the product of 4 clock rates lends a coherence to the durations produced. This idea was among the earliest implemented and though striking effects are obtained it now seems inadequate.

A block diagram for another system as yet undeveloped is included on page 37 figure 2.

A circuit designed by James Divilbiss illustrates one mode of interaction possible between the performer and automatic circuitry. TOUCH CONTROLS function as the upper rank of a shift register. The performer has access to a parallel input on each bit of the register and can program a "1" or a "0". (see pages 40 & 41 figure 4 or 5)

Circuits which allow the performer to decide whether a particular function is under manual or automatic control were developed. An important feature allowed independent control of pitch and register (octave), however, I will not describe these because a circuit for a Digitally-Controlled Oscillator (DCO) recently completed by Mr. Franco has suggested a richer direction that includes control of waveshape.

In commercially available music systems sound is synthesized by oscillators with five waveshapes, which are then added together to produce composite waveshapes. A common complaint of many composers regarding the possibilities of timbral variety available with such a system is partially due to the fact that waveshapes are produced in this manner.

Our method is different. In conjunction with a counter and 2-64 bit Random Access Memories (RAM's), the DCO can assume a variety of arbitrary waveshapes in real time. The DCO generates a train of pulses whose frequency can be digitally specified to lie in the audio x 16 range, 320 cps to 320 kcps. The period of a sound is divided into 16 equal time slots, and within each slot one may specify one of 16 possible voltage levels that the waveshape can assume. These levels are obtained by decoding the output of a 64 bit RAM (16 words each 4 bits wide) by means of a Digital-to-analog converter (DAC). In this way the input pulse train causes each of the 16 words to appear at the decoder input in a sequence which repeats itself every 16 input pulses. At the output of the decoder a waveshape in the audio range is generated in the form of a staircase due to the discrete time and level quantization. The undesireable effect of this quantization is eliminated by sending the waveshape through a voltage-controlled filter which smoothes the sharp edges of the staircase. (see figure 11, 12, 13, 14, page 54).

By changing the data contained in the words of the memory it is possible to obtain a rich class of waveshapes which cannot be achieved by other methods of waveshape generation. (see figure 6, 7, 8, page 47,48 & 49)

Since two 64 bit RAM's are being driven by one DCO and modulo 16 counter, two independently controllable waveshapes that are a function of the same frequency are produced at the output of the RAM's. A method for continuous change of waveshape is implemented in the following ways.

1) The output of each memory is sent to signal in 1 and signal in 2 of the waveshape mixer. The waveshape mixer, or SEE-SAW as we call it, gets its name from the following characteristics. When the voltage applied at the control input of the SEE-SAW rises from 0vdc to +10vdc, the amplitude of signal in 1 rises from minimum to maximum at output of the SEE-SAW at the same time that signal in 2 falls from maximum to minimum. Blending or mixing the two waveshapes is achieved by varying the voltage at the control input.

2) A pair of comparators sense the control voltage. When it is below +1vdc a gate triggers appropriate logic signals for the WRITE mode control on RAM 1, which is near maximum attenuation at the output of the SEE-SAW, and thus allows the RAM 1 DATA inputs to the 16 words to receive information for a new waveshape. The same process is repeated for RAM 2 when the comparators sense a control voltage that is above +9vdc.

Designs for the SEE-SAW and COMPARATORS are completed and prototypes are in operation. (see page 50 & 51 figure 9 and 10)

This method for changing the contents of the RAM's performing waveshape was adopted because of the esthetic necessity to update the memory when its waveshape was <u>below</u> audibility. When we attempted to change a waveshape in the audible range redundant "clicks" were heard. This is attributable to the high speed analog gates within the IC. The method chosen is simple and inexpensive.

We are working to develop a central processor of information (WAVESHAPE UPDATE) which will supply waveshapes to all of the oscillators The plan is to replace the 5 basic shapes with 4,095 (2^{12}-1) basic shapes so that subtle contrast is intrinsic. In addition, because of the fact that 2 RAM's are driven by one oscillator, any two waveshapes can be mixed to produce more than 8,000,000 combinations. With variable amplitude and phase relationships the availability of the entire spectrum of waveshapes is approached.

We want to maintain a reasonably constant amplitude at the output of the SEE-SAW so that overall signal envelope may be independently controlled at another place in the system. We have developed a partial list of the specifications necessary for a sequence for supplying waveshapes that will avoid changes in amplitude that are more than 10%. For example, attention must be given so as to provide similar RMS characteristics in adjacent or combined waveshapes and also to the avoidance of cancellation due to phase that might result if random control of waveshape were applied. The large number of waveshapes desired rules out use of a READ ONLY memory for economic reasons.

An experiment with a fixed most positive voltage and a moveable most negative voltage is shown and described on pages 61 and 62, fig. 18. Time slots between are random. It's interesting but we'll have to do better.

A possible answer to the problem of maintaining a constant amplitude for the waveshape at the output of the SEE-SAW could be to update the contents of a memory by performing a sequence of elementary DATA swaps. A swap consists of an exchange of the contents of different words in the memory, selected at random. Whether a swap of WORDS or a swap of DATA will prove to be simpler is still to be decided. A sequence of several swaps can be performed in a short time compared to the time that the SEE-SAW will remain in the enabling mode. When con trol arrives at a given memory, the corresponding comparator is interrogated. If the comparator is in the disable state, no action is performed, and control is passed on to the next memory in a circulating pattern. If the comparator is in the enable state, the processor performs a sequence of DATA swaps, and then continues to the next memory in the sequence. The result, hopefully, is that the loudness of a sound remains unaffected by timbral changes.

At this point it is appropriate to discuss the pseudo-random number system which may have musical application generally. I am presently studying sequences whose values are produced by a circulating, programmable length shift register whose output is routed to a programmable number of inputs of an EXCLUSIVE OR gate. The output of the gate is fed back to the input of the register. (see page 63 and 64 for a few examples of the vari-sized segments within total collections that are thus produced)

Though my experience with these possibilities is still limited to information which is not yet adequate for music, this is an area open to exploration which might result in a large enough system of collections that could then be used to drive the WORD SELECT and DATA inputs on x number of local memories which would feed DAC's that specify analog parameters to such an extent as to be musically viable. A composer would steer and combine collections according to his musical prerogatives in real time.

And, those options which can be empirically defined as being equally desirable could be controlled by pseudo-random sequences An analogy to the tonal music system used in the 18th and 19th centuries is inferable if one considers that the C major triad is I in the key of C, V in the key of F, lV in the key of G, etc. Or, with the use made by contemporary composers of sets and subsets in music written today. The instrument is, here, congruent with the development of compositional systems.

A multi-dimensional matrix that I composed for UNDERWORLD and later used in my composition, BALLAD, illustrates the same idea in amother way. The tonal elements are structured around a single 12-tone row which partitions evenly into 4 primary triad cells and three primary tetrachord cells. These cells were chosen in such a way that a large variety of transposition and inversion combinations give 12-tone rows.

A discussion of the row in terms of its triadic partitions is sufficient to make the point.

Let the four triads be denoted A_1, A_2, A_3, A_4. The structure (module 12) of the triads is:*

A1 = (1,2,3) A2 = (0,4,5) A3 = (10,7,~~8~~ 11) A4 = (6,9,8)

This structure has the following properties:

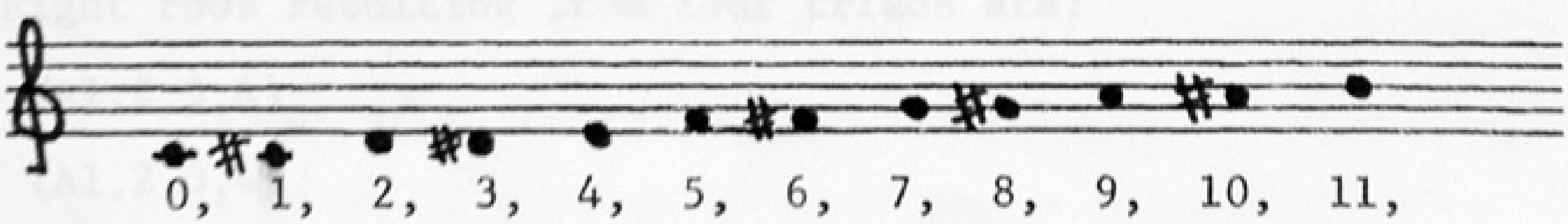
(1) The important intervals represented by each cell (permuting the notes to their natural (numerical) orders) are:**

A1: (1,1) A2: (4,1) A3: (3,1) A4: (2,1)

Thus each cell is characterized by a unique interval: 1,2,3, or 4, and the interval 1. The unique interval defines the cell's character.

(2) By the all-combinatorial property of each cell, a 4 X 4 matrix can be constructed which will contain 12-tone rows in its rows and columns, its diagonals, and various blocks of four cells forming obvious patterns in the matrix.

* I'm using pitch class notation whereby 0 = C, 1 = C#, ...11=B.



0, 1, 2, 3, 4, 5, 6, 7, 8, 9, 10, 11,

** Here, 1 = a ½ step, 2 = a whole step, 3 = a minor third, etc.

Partitioned row in musical notation:

(1,2,3) (0,4,5) (10,7,11) (6,9,8)

The Matrix

	Po	T6	I2	I8
A1	(1,2,3)	(7,8,9)	(6,5,4)	(0,11,10)
A2	(0,4,5)	(6,10,11)	(7,3,2)	(1,9,8)
A3	(10,7,11)	(4,1,5)	(9,0,8)	(3,6,2)
A4	(6,9,8)	(0,3,2)	(1,10,11)	(7,4,5)

P = Prime

T = Transposition

I = Inversion

There are twenty-four 12-tone rows represented in this matrix. They can be catagorized into three basic types: those based on the characteristic interval of a single triad; those based on the characteristic intervals of two triads; and those based on the characteristic intervals of all four triads. The <u>character</u> of a six or twelve note set is defined by the intervals within three-note cells.

The following is a list of the rows.

Four rows resulting from one triad are:

1) A1 (PO,T6,I2,I8)

2) A2 (PO,T6,I2,I8)

3) A3 (PO,T6,I2,I8)

4) A4 (PO,T6,I2,I8)

Eight rows resulting from four triads are:

5) PO (A1,2,3,4)

6) T6 (A1,2,3,4)

7) I2 (A1,2,3,4)

8) I8 (A1,2,3,4)

9) A1(PO), A2(T6), A3(I2), A4(I8)

10) A1(T6), A2(PO), A3(I8), A4(I2)

11) A1(I2), A2(I8), A3(PO), A4(T6)

12) A1(I8), A2(I2), A3(T6), A4(PO)

Twelve rows resulting from 2 triads are:

12) A1,2(PO,T6)

14) A1,2,(I2,I8)

15) A3,4,(PO,T6)

16) A3,4(I2I8)

17) A1,3(PO,T2)

18) A1,3(T6,I8)

19) A2,4(PO,I2)

20) A2,4(T6,I8)

21) A1,4(PO,I8)

22) A1,4(T6,I2)

23) A2,3(PO,I8)

24) A2,3(T6,I2)

There are eleven transpositions and twelve inversions of PO that can be derived within the twelve-tone system. The matrix above utilizes four of the total twenty-four possible twelve-tone rows and 16 additional twelve-tone subsets can be derived.

The twenty remaining rows can be mapped into five additional and similar matrices. Thus, one hundred and forty-four row combinations within the six matrices can be derived from the original 24. Further, twelve-tone sets can be derived by combining a segment of one matrix with a segment of another. For example, A1,2(PO) -- (1,2,3; 0,4,5) can be combined with its inversion A1,2(I0) -- (10,9,8; 11,7,6) to form twelve.

It can also be combined with its inversion at another transposition level (4,3,2; 5,1,0) and maintain a pitch identity.

The important consideration here in regard to music is that if the gamut of change from identity to novelty can be defined, a composer can control the gradations between. In music, Newton's first law of dynamics is only partially true. What goes up only occasionally comes down. Set combinations can be biased to stress certain pitches by repeating them. And as well by withholding certain other pitches for a time, provide intrinsic contrast which can be used at will.

At any rate the analogy to tonal music of the past holds true if one considers that the character of A1(PO) changes depending on the context that is created by the particular segment with which it is associated.

Such a method for the control of pitches is economical and efficient in as much as it could be programmed with eighteen 4 X 4 random-access register files, or with two-hundred and eighty-eight bits of READ ONLY MEMORY.

Registral control can be provided independently.

MEMOGRAM

To: Page of expletive

From: Monolithic Speakers

one-shot RAM's ROM STOP EX OR's

pair, NANDS ODD STOP SHIFT SCHOTTKY'S pot STOP

(TUBE)

(GO input AND multiply)

ADD ADDADDAD DAD DAD DAD DA D-DA D-DA D-DA D-DA D-DA D-DA

(RE, MI, FA, SOL, LA, TI)

(DOUGH)

cc: L.S.I.

M.S.I.

S.S.I.

(RATIONAL CONTINUATION)

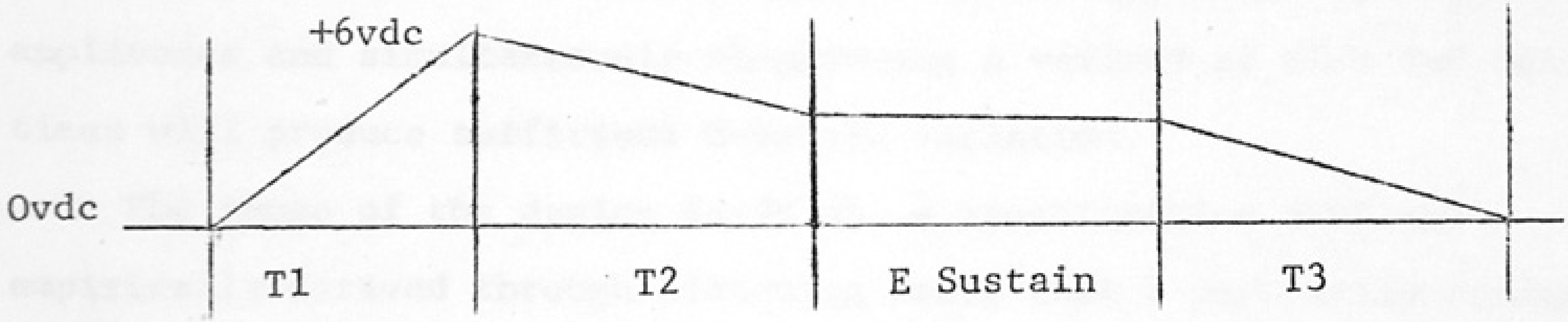
In our system loudness control or SOUND ENVELOPE is achieved by a novel and efficient method. As another example of the limitations of commercially available systems, in the Moog System envelope is programmed by setting 4 potentiometers.

Pot 1) T1 - Rise Time (range; 500ms to 10 secs)

Pot 2) T2 - Initial Fall time (range; 500ms to 10 secs)

Pot 3) E Sustain (range; 0 - to 10 secs)

Pot 4) T3 - Final Fall Time (range; 500ms to 10 secs)



Four knob settings are necessary to achieve a paucity of possibilities. Though it is possible to control the rise time, it is not possible to alter the +6vdc peak amplitude. Further, it is impossible to program the dimples and pimples in a shape that custom has conditioned our ears to expect in a musical sound.

I used a system similar to this to compose the electronic part of Underworld. Though the part is only 12 minutes long, it took over 3000 hours to produce because each note had to be separately shaped, spliced in a chain, and finally mixed with 8 other channels to form a 9 channel whole.

In our system (see fig. 15, 16 page 57 & 58) 12 bits of control allow one to produce an enormous number of envelopes ranging from .02 seconds to 20 seconds. The main principle in the circuit that allows flexibility and variety is simple. If the amplitude increases the program for rise time is operative, if the amplitude decreases the program for fall time is operative.

A pseudo-random set can be used to program the least significant bits in order to produce dimples and pimples to one's heart's content.

If a long fast string of pizzicati-like envelopes are desired: the rise time is held at .02"; the fall time is held at .15"; and the amplitude is toggled between 0 and a specified peak at a fast rate. Add a smidge of reverb to the package. Obviously, toggling adjacent amplitudes and simultaneously programming a variety of rise and fall times will produce sufficient envelope variation.

The range of the device is 96 db, a specification that was empirically derived through listening tests that I personally conducted with Mr. Franco and two students presently studying composition here.

Performance in real time suggests that a composer will need many options for interfering with the program. On this analog circuit alone there are three possibilities for interference.

A REST control is provided that overrides the program. In addition a HOLD control that enables the user to inhibit evolution of the program. (see page 58 fig. 16) Also, a means of implementing amplitude modulation.

The amplitude of the waveshape can be modulated at other points in the system as well. Several of the circuits already designed are inherently multipliers. As an example, the 4-bit DAC that accepts the memory output can be easily modified to become a multiplying converter. A choice will be made fixing the best place to apply balanced or unbalanced modulation based on experimentation. However, in the event that it is convenient to apply modulation with a separate circuit, a four quadrant analog multiplier for balanced modulation is shown on page 59 fig. 17.

The first stage of the SOUND ENVELOPE control can be used in other applications as well.

The 0 to 10 vdc ramp at the output is used as a control voltage for the SHIFT input to the DCO.

It is also used to control the amplitude ratio between waveshape 1 and 2 in the SEE-SAW.

The printed circuit card for the SOUND ENVELOPE control includes an option that bypasses the second stage exponential converter. The circuit is then useful as a linear rather than exponential attenuator. As a digitally-controlled multiplier in this configuration, it is used to implement the continuous variation of the number of equally-tempered tones per octave from 12 to 20. The output of the 7-bit DAC is routed to the audio input of the multiplier. The multiplier output is fed-back to the SHIFT input. The 7-bit DAC is now a multiplying converter thereby allowing continuous control of the number of steps per octave.

Though there are exciting and quite fantastic feasts for the ears in sound moving through the air at the same time that the source is itself moving at controllable rates, the inovation in this system is with respect to its capability for distributing 4 music programs independantly of each other.

The first experiment that I know of was realized at the Bruxelles World's fair in 1958 in a composition by Edgar Varese. This used a single program. In 1964 I constructed a manually operated sound-rotator consisting of a 4-pole switch on which 92 extra contacts had been soldered between the poles. I scored for it in UNDERWORLD much as one would write an instrumental part.

In 1970 Akiro Akiyama engineered a digitally controlled sound rotator for Roger Reynolds which was used at the Cross Talk Festival in Tokyo. It controlled the clockwise movement of sound around the room at a variable clock rate.

As far as I know, my experiment is the first with a system for the distribution of sound involving multiple programs that allows the use of rhythmic and contrapuntal concepts ~~as for instance in orchestral composition but realizable in real time,~~ to music composition. The performer literally plays the space.

In order to present a clear description of the system I'll divide the following according to its digital and analog functions.

SOUND DISTRIBUTION

DIGITAL

A 24 x 4 wide shift register supplies control bits for 96 channels. Each 24 bit register is divided into 6 sections, 2-3 bit, 2-4 bit and 2-5 bit registers. The registers are bi-directional and can be independently controlled. A 6 x 6 matrix by 4 wide allows control such that the output of any register can be routed to the input of any other register and to itself within a 24 bit register. In addition, each register has a parallel as well as serial input. The parallel input allows the performer to control the density of the circulating bits in real time. By manually holding the SET input to the touch control to a "1", "1's" will propagate through the path chosen and fill the registers of a program at the clock rate specified. Hold a "0" and "0's" will propagate and empty the register. A 4 channel multiplexer provides a means for independently controlling the clock rate, the density, and the path through the matrices in each of the 24 bit registers. The system utilizes 4-64 bit RAMS and peripheral logic circuitry for the matrices. The circuit provides 2 way communication so that a dialed program is automatically read-out on the panel by one set of twenty-four lamps for the registers and one set of 36 lamps for the matrices. Visual corroboration of the path and density is indispensable during the learning process. See figs. 19 through 25 on pages 68-74 .

ANALOG

The system described above is used to control the on and off times of 4 analog switches on each of 24 power amplifiers. Each of the 24 speakers is powered by a dedicated amplifier. There are certain features inherent to the very idea of sound switching that impose a definite set of constraints upon the analog switch performance. Switching a sound on or off results in the generation of "extra" undesireable harmonics due to the transients that occur during the attack or decay. If the rise or fall time is too fast for the frequency being switched, a contour other than the waveshape is generated before settling. This redundant effect is perceived by the ear as a "click". It was first thought that a simple zero detector would solve the problem. It is true that the elimination of a D.C. component considerably improved matters but the redundant "click" though less anoying was still there. We concluded that "click" resulted from the source's abrupt change of position in the space. Intrinsic to the concept is the necessity for a smooth transference of sound from place to place so that envelope control can be independantly programmed.

With experimentation we discovered that a 30ms overlapping rise or fall time was the ideal trade-off value to both eliminate "click" and still allow the possibility of a fast switching rate. In addition, a requirement that the switch have a linear behavior during the attack or decay had to be satisfied.

Unfortunately, we were not able to find a commercially available switch which satisfied the specifications. Analog switches of the photo-resistive type suffer from the fact that switching times are pre-determined and have highly asymetrical rise and fall times. Switches that rely on photo-transistors or on variable resistance FET's are either too fast and without a simple method for slowing them down, or have a large parameter spread due to temperature instability, or are non-linear in the intermediate region of rise and fall time.

Therefore, a new scheme was needed which Mr. Franco designed and which is presently operative. See fig. 26 and page 81 for detailed discussion of switch-power amplifier.

The "polyplanar" speakers are a recent inovation of the Magitran Corp., Moonachie, New Jersey. They are made of what looks to be a kind of styrofoam. The monolithic structure of the material eliminates the need for a cone shaped structure for strength that is associated with conventional speakers. The measurements are H. 14½" x W. 11½" x 1" thick and the frequency response is flat from 80 cps to 15 kcps. They are inexpensive, at $6.55 a piece, and are adequate at this stage of development. Directionality in sound is only accurately perceived at frequencies above 100 to 150 cps in any case. We plan to add 4 woofers to boost the bass. Sound above 80 cps will be filtered with a low pass filter designed to provide a slope of 12 db per octave, so as not to confuse the directionality of the higher sounds. We shall have to experiment to fix the parameters, as the above is only an educated guess. See photo of speakers page 28 and 85.

N.Y. TIMES MARCH 7, 1971

Martisano Explores Electronic Music With a What Is It

By RAYMOND ERICSON

Whether following, leading or in the midst of avant-garde compositional styles, Salvatore Martirano has been one of music's more successful practitioners. His "O, O, O, O. That Shakespeherean Rag," Ballad for Amplified Singer and Instrumental Ensemble, "L's G. A." and other works made a distinct impression when they appeared during the last decade.

Friday night at Automation House, he demonstrated a new direction he has taken. He had brought with him an "instrument" called the Mar-Vil Construction, which he had conceived and built with James De-Vilbis, a computer engineer. He has been working on it for the last five months at the University of Illinois, where he teaches, and it is a variation on synthesizers.

It is beautiful to look at. An upright panel decked out with hundreds of wires, resembled a white-on-white Jackson Pollack painting. Below it, a slanting panel held a symmetrical array of blinking lights and what seemed to be thumb-tack heads. By tapping or whisking his hand across these lights and heads he could improvise an electronic composition without effort. The sounds came out of 21 speakers hanging at various levels around the room.

Mr. Martirano is still exploring his new instrument, and what he created for his audience was not yet drastically new. He has such a good ear, however, that the sounds he used—almost all having pitch—were shaped into complex, quasi - contrapuntal textures with more freshness and structure than many electronic works. He obviously was in control of what he was composing.

With further development of the instrument and knowledge of its use, Mr. Martirano should open a new and, hopefully, brilliant sonic world to work in.

DETAILS OF CIRCUITRY, SCHEMATICS AND BLOCK DIAGRAMS

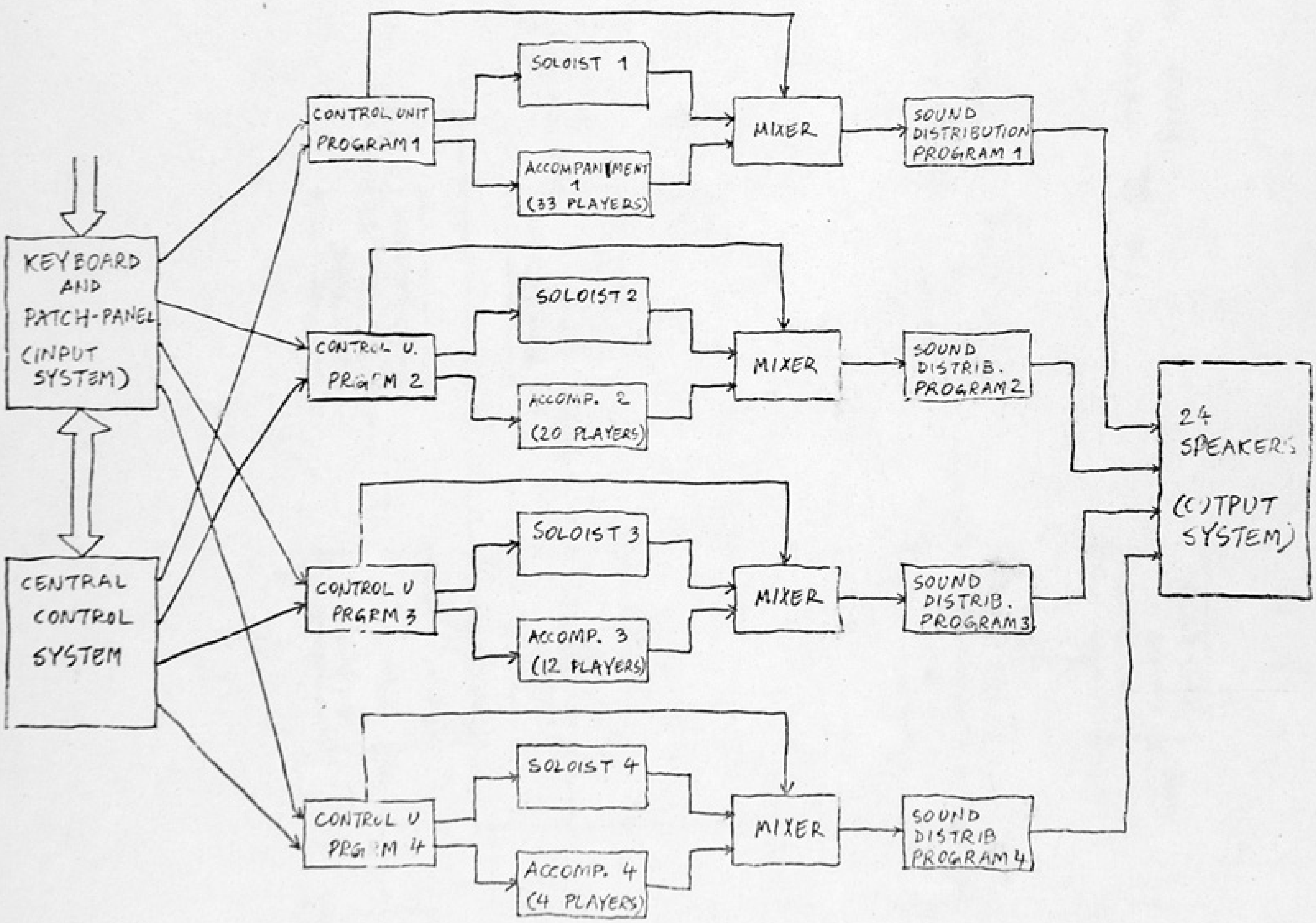


FIGURE 1 : OVERALL VIEW OF CONTROL SYSTEM.

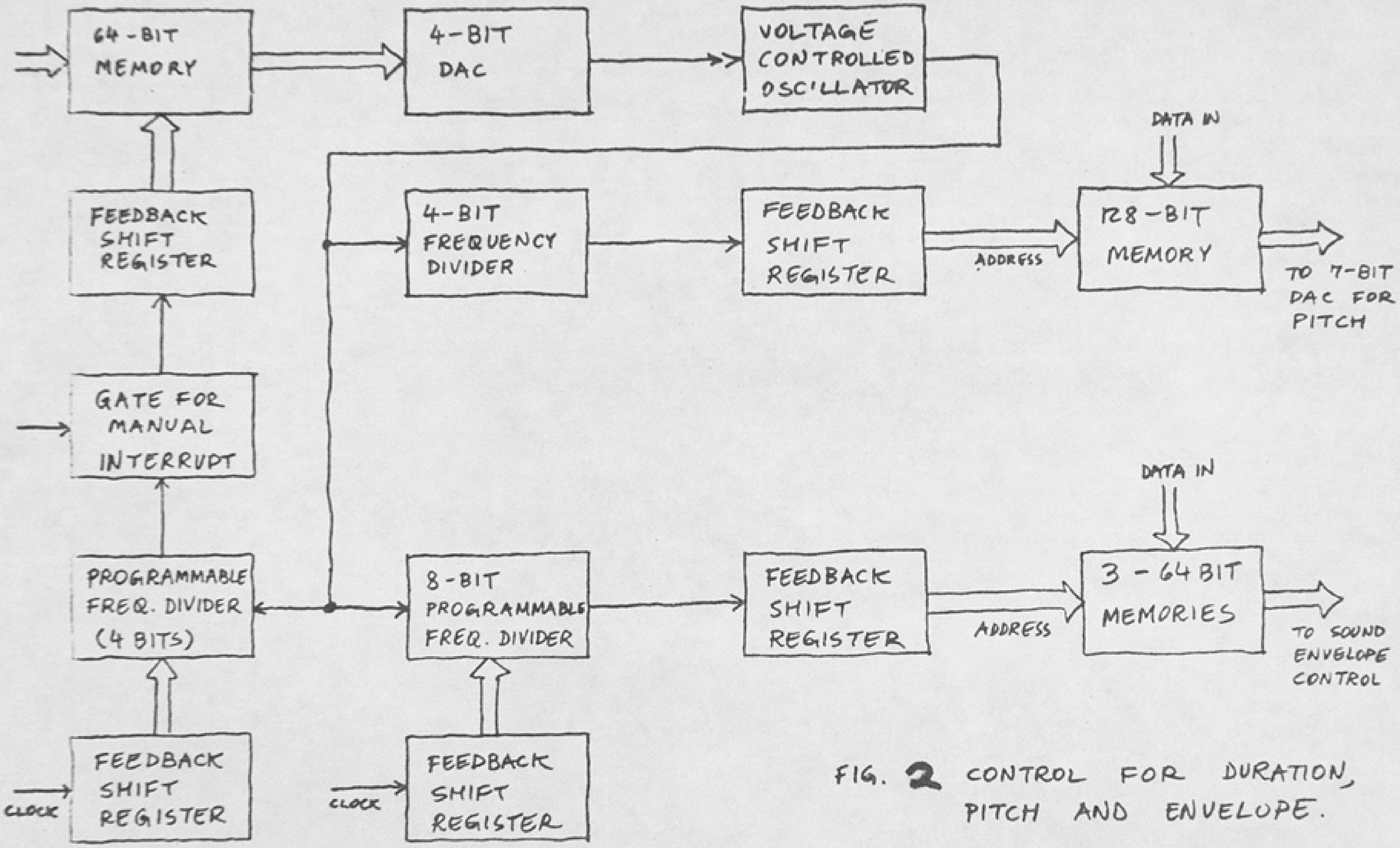
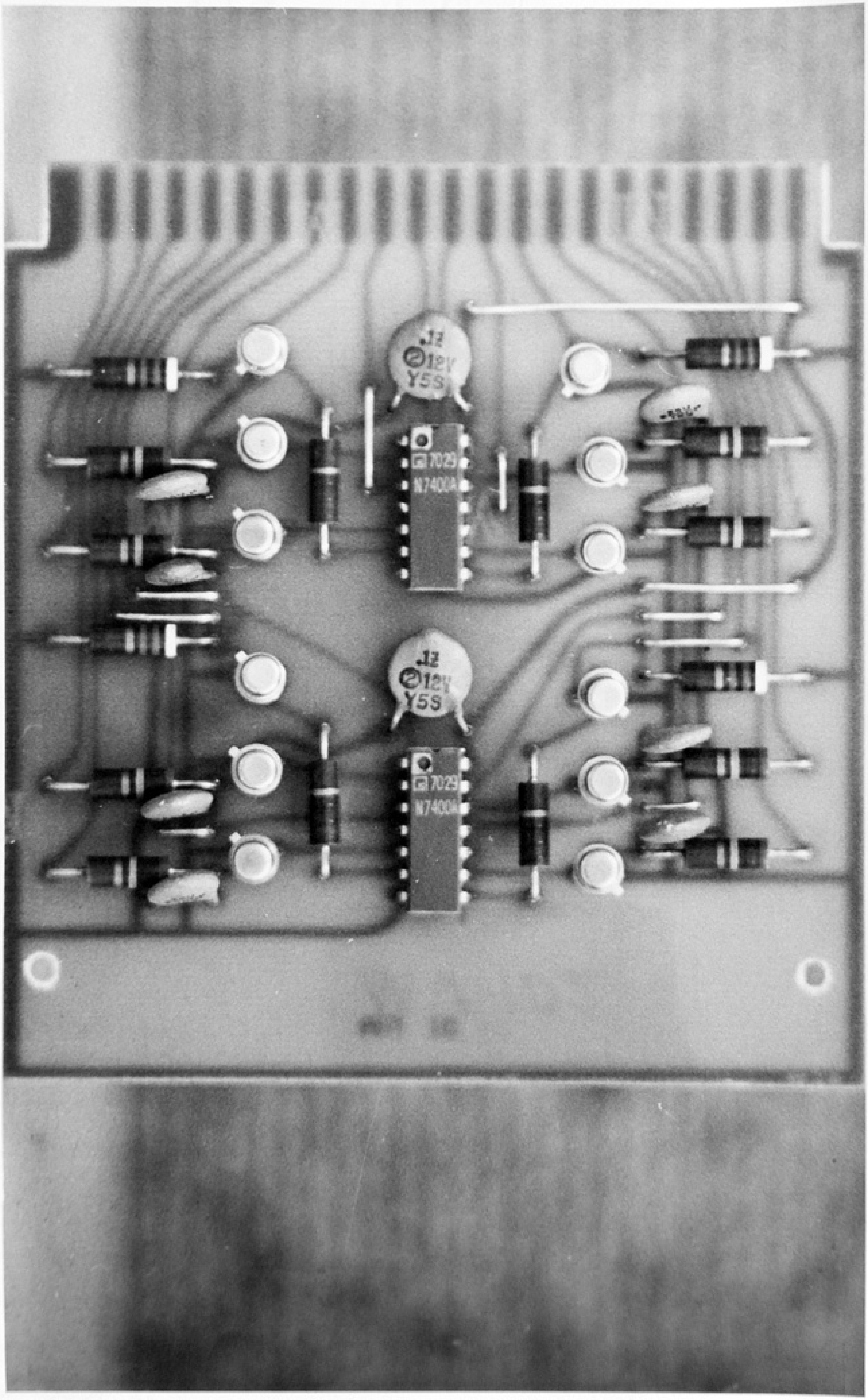
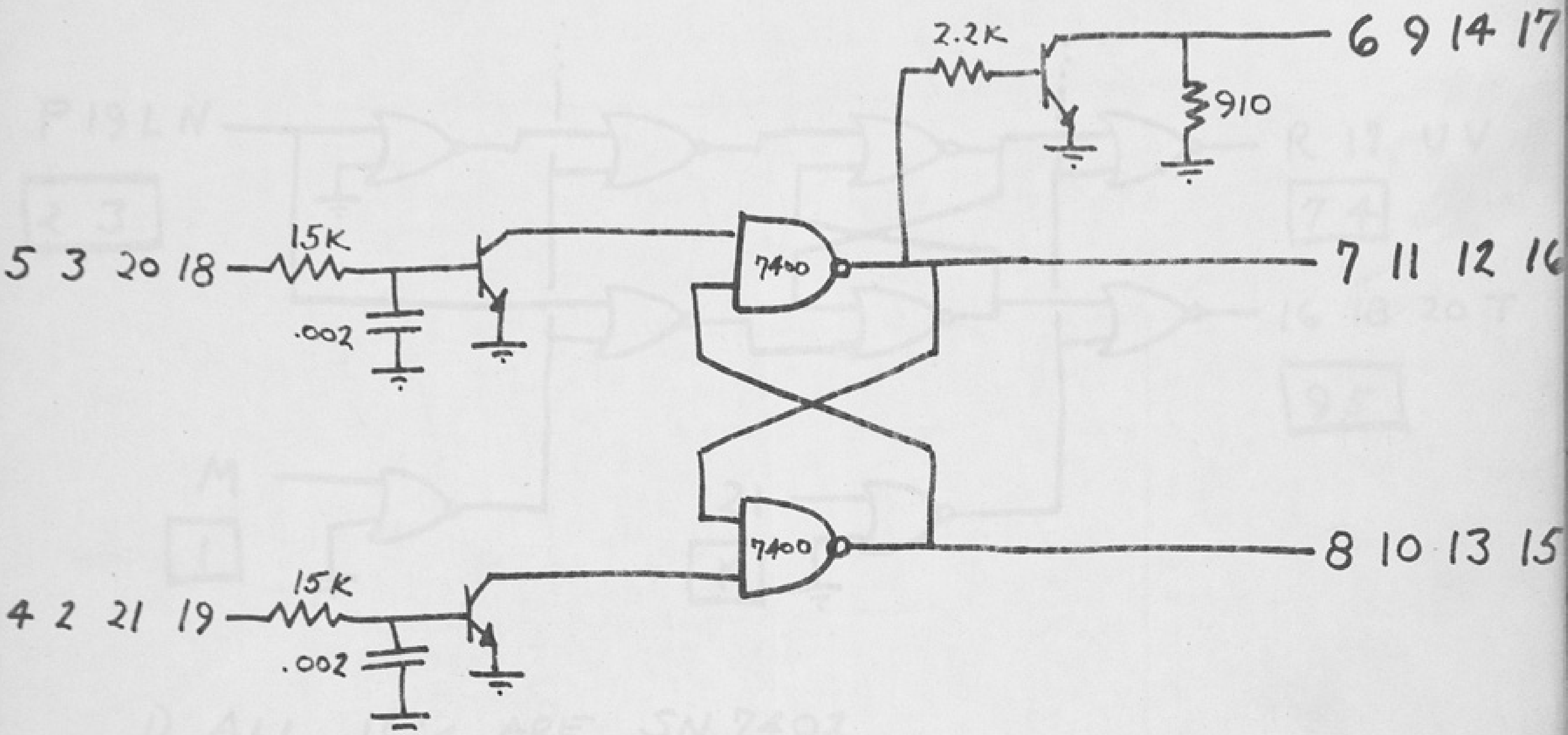


FIG. 2 CONTROL FOR DURATION, PITCH AND ENVELOPE.


7029
N7400A
7029
N7400A

MU-100



FOUR CKTS PER BOARD

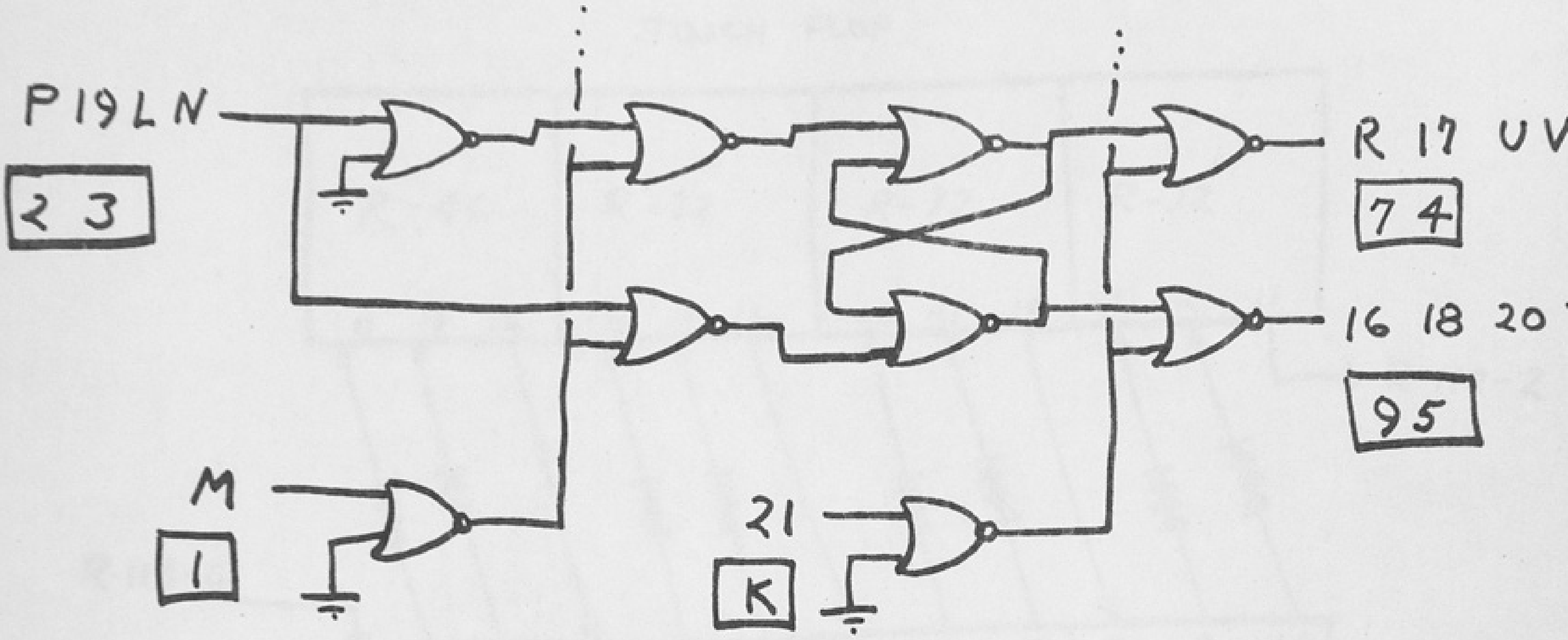
POWER = PIN 22

GROUND = PIN 1

E — 2
F — 3
H — 4
J — 5
S — 18
T — 19
U — 20
V — 21

MU-100
TOUCH FLOP
1/9/71 JD

RE-100

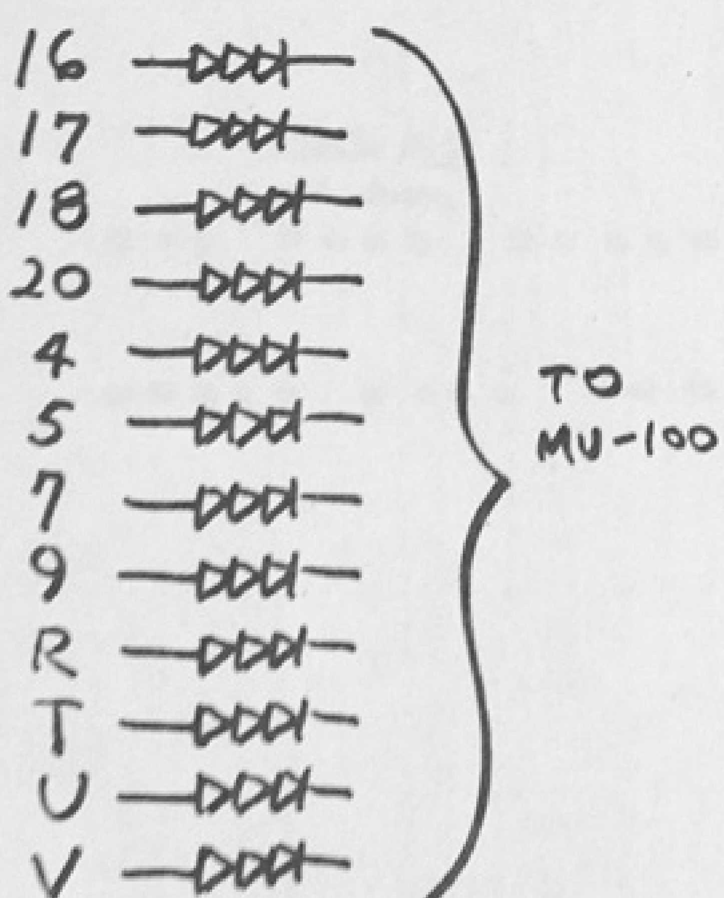


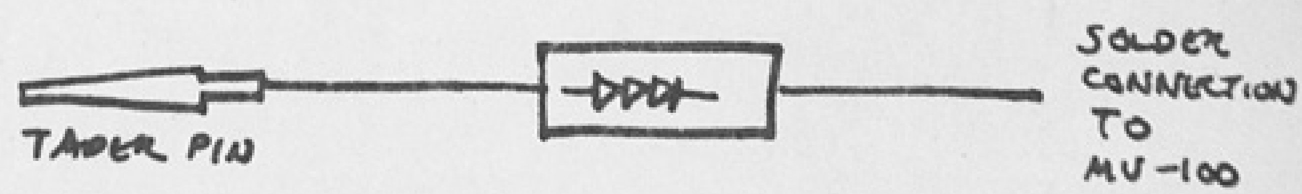
1) ALL IC's ARE SN 7402

2) SIX REGISTOR POSITIONS/CARD GROUPED FOUR AND TWO

3) M AND 1 = GATE IN 21 AND K = GATE OUT

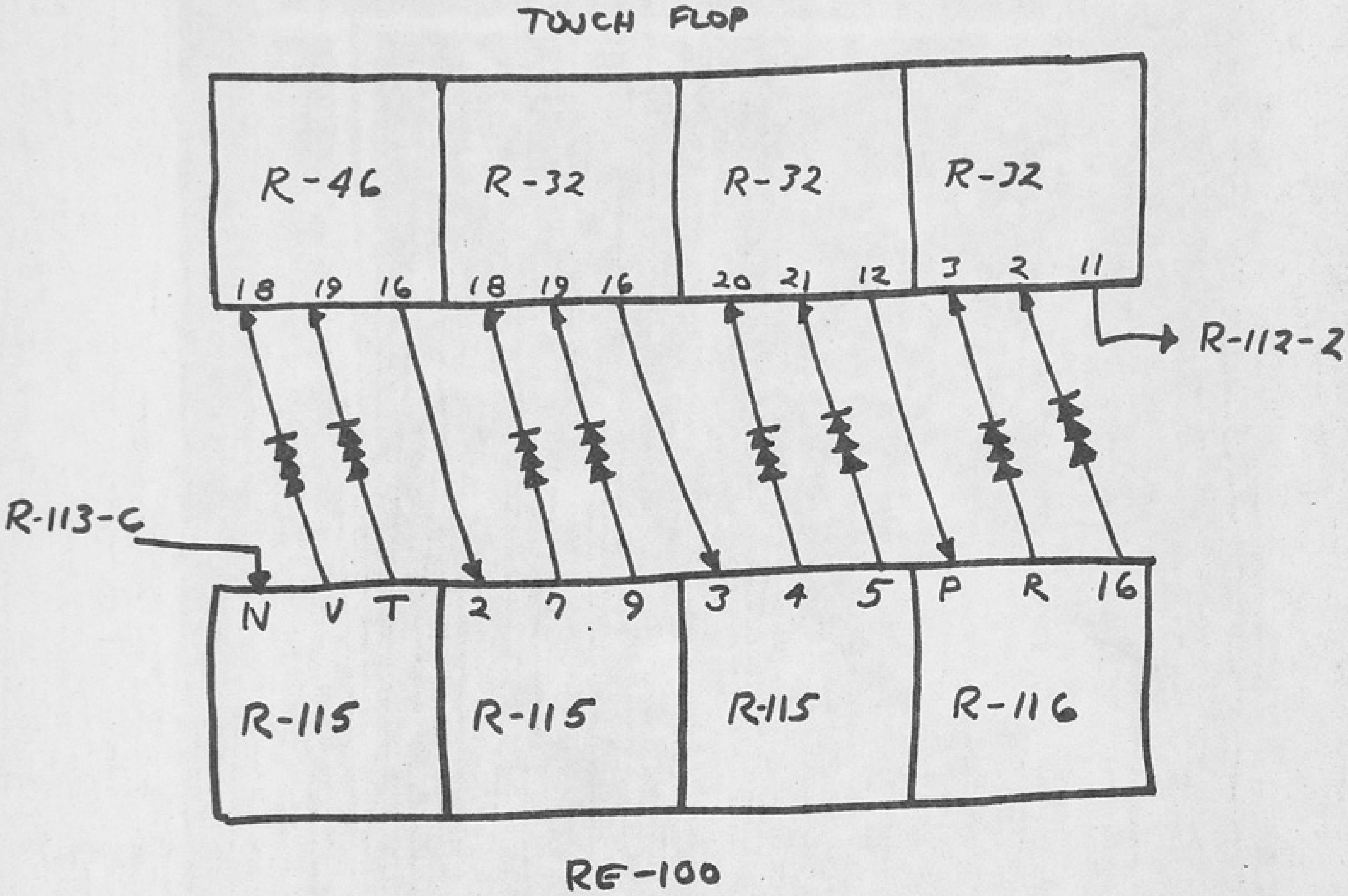
4) GROUND = PINS A, Z +5 = PIN 22

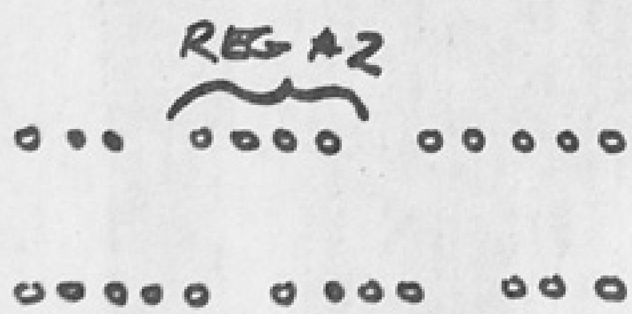




RE 100 REGISTER CARD

1/27/71 JD

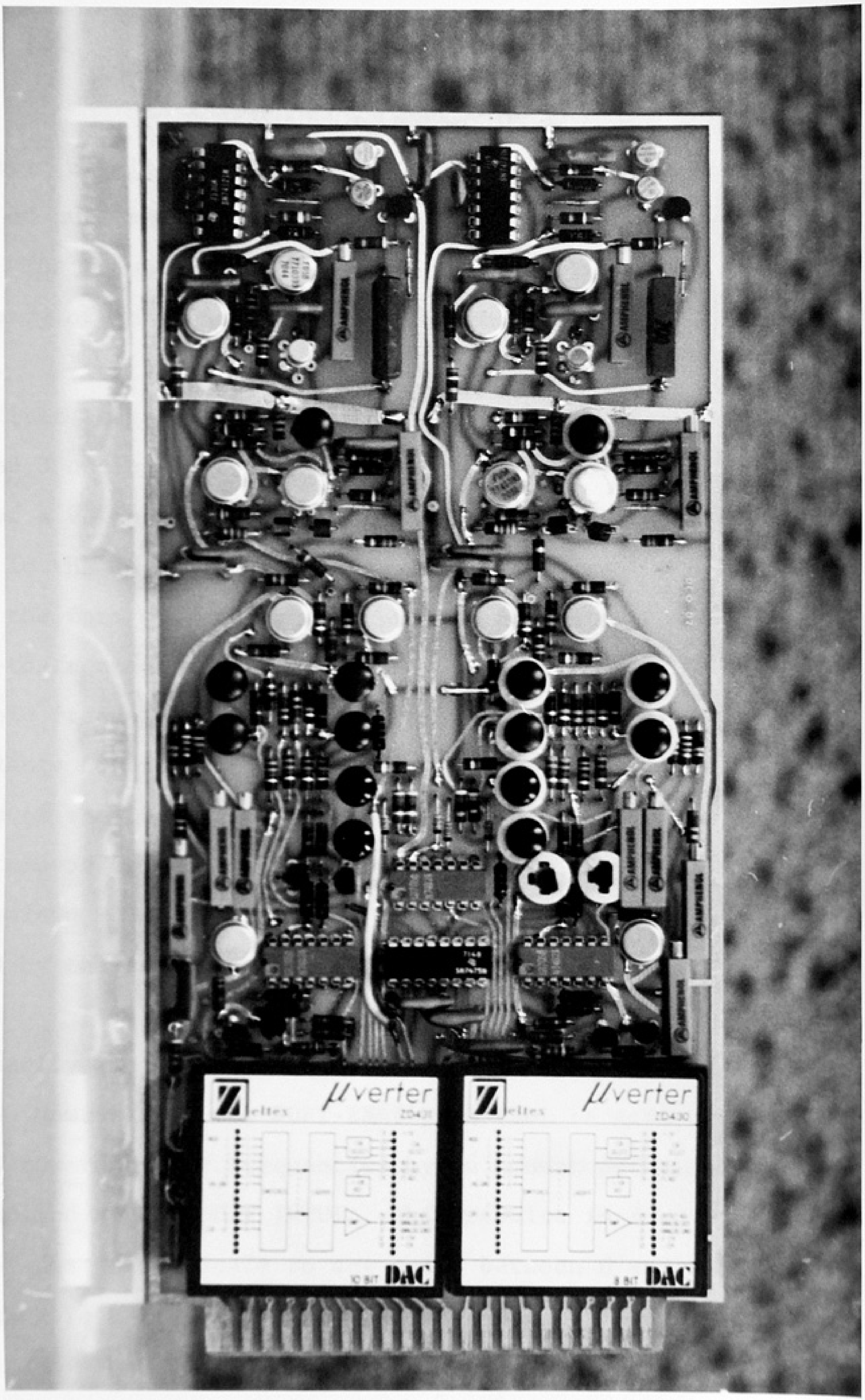




REGISTER #2

1/23/71

JD


μverter
DAC
μverter
DAC

FREQUENCY GENERATION

A number of electronic circuits are used to generate and control the train of pulses which cause the words in memory to be read out in sequence. The frequency of this pulse train divided by 16 produces the frequency of the fundamental tone of the wave shape which is to be generated.

The first stage of the circuit (fig. 13) accepts a set of 12 digital and 3 analog control inputs and generates an analog signal, in the form of a voltage, which is fed to the input of an exponential converter in the second stage. (fig.14) The output of the second stage, in the form of a current, is fed to the third and last stage, a current-to-frequency converter whose output is a pulse train with programmable frequency.

Base and Range Control Stage 1

Seven of the 12 digital control inputs are decoded by a digital-to-analog converter and are used to specify any frequency value in a scale containing 128 equal steps. The base and range of the scale are determined by the remaining 5 bits which are used for gating and mode control.

Ordinarily the base of the scale is 20 x 16 cycles per second (320 cps). However, if the SHFT-ENABLE CONTROL is activated, the base is shifted toward higher frequencies by an amount determined by the voltage applied at the SHFT INPUT. This results in an overall transposition of the entire scale to any point between 0 and 4 octaves corresponding to a control voltage 0 and +10 vdc. A similar function is performed by the MOD-ENABLE and MOD inputs respectively, the difference being that the SHFT input accepts a varying control voltage from 0 to 10vdc from a dedicated circuit which includes digital control of a voltage-controlled amplifier while the MOD input will accept an audio signal which varies between plus and minus 10 v. This will come from one or more oscillators elsewhere in the system such that an interactive net of outputs controlling inputs is possible when desireable.

Though there are 128 equally spaced pitch intervals possible, there are four options that allow control of the number intervals per octave. Each corresponds to one of four possible input states of two latches whose outputs perform control. A transition between two modes is enabled when the clock input to the latches is high. The following truth table defines the output in each state.

A	B	output
0	0	12 equally tempered tones per octave
0	1	16 equally tempered tones per octave
1	0	20 equally tempered tones per octave
1	1	continuously variable number of tones per octave ranging from 12 to 20.

The circuitry necessary to implement the first three modes consists simply of an operational amplifier whose gain can be digitally controlled.

By simply changing 2 resister valves it is possible to change both the SHFT and MOD range from 0 to 4 octaves to a larger range, say 0 to 10 octaves and the continuously variable number of tones per octave from 12 to 20 to a larger range, for example 5 to 128 tones per octave.

Obviously, wide ranges would require sophisticated digital logic combined with sensing circuits that would inhibit information that would cause the oscillator to operate without aural effect in the super-audio range and to likewise inhibit information from causing the oscillator to sound an enormous number of subtle changes of pitch, say 128 tones per octave, for too long a time. In other words the options available at this time are a reasonable beginning but can be quite easily changed if an opinion resulting from feedback dictates that the rules of the game be changed.

Exponential Conversion Stage 2

This circuit accepts the voltage generated by the circuit of the previous stage and converts it to a current which is an exponential function of the input. This function is obtained by making use of the law relating emitter current to the base-to-emitter voltage drop of an npn transistor. In order to obtain high temperature stability and repeatibility, a matched pair of temperature controlled transistors is ~~are~~ used. The law relating emitter current to the base -emitter junction voltage is exponential only if the ohmic voltage drop resulting from the bulk resistance of the semi-conductor is negligible. Unfortunately this component cannot be neglected even at currents as small as 1 mA since it causes the law to depart from being exponential by as much as 20%. The circuit presently in operation incorporates a novel and very simple scheme for the compensation of this effect such t that an exponential conversion is obtained with an accuracy and repeatibility of better than 0.2%. A buffer at the current output insures a proper signal between stage 2 and 3.

Current-to-Frequency Conversion   Stage 3

When considering musical parameters it is generally acknowledged that the ear is most sensitive to differences in frequency. In a 12 tone per octave system the ear can readily distinquish an 1/8 tone difference between 2 pitches. Particular care was devoted throughout the design to the production of scales that would be accurate and repeatible, simultaneously without excessive complication and avoiding use of expensive components. The main difficulty presented by the specification was to satisfy the requirement of a 1000 to 1 range at audio x 16. Current to frequency conversion is achieved by the usual method of integrating the input signal by means of a good quality capacitor, which is discharged after the voltage at it's terminal reaches a certain threshold value, thereby repeating the cycle again. Unfortunately the finite amount of time taken by the comparator circuitry to react and by the switch to discharge the capacitor introduces a non-linearity which becomes relevant at high frequencies. This problem was solved by means of a novel and simple circuit in which the threshold is made to vary with the input current in such a manner that the original inherent non-linearity is compensated for, ~~and~~ thus generating a frequency that is accurately proportional to the input current.

The overall accuracy of the pulse train generator has been measured to be better than 0.3% over the 10 octave range from 320 cps to 320kcps.

NOTES:

1) DIGITAL SI NALS ARE REPRESENTED BY → AND ⇉.

2) ANALOG SIGNALS ARE REPRESENTED BY →>>

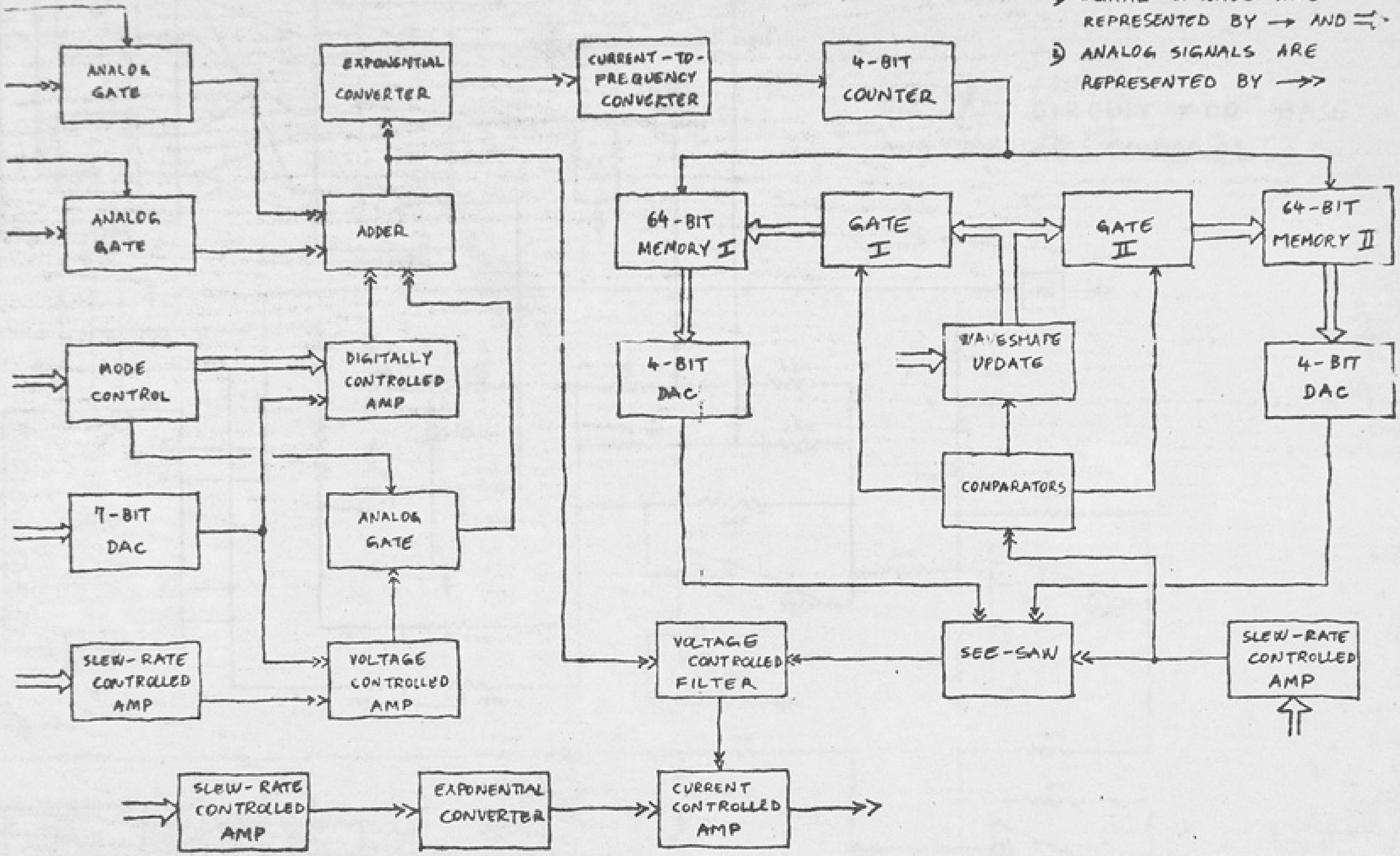


FIG. 6 BLOCK DIAGRAM FOR DIGITALLY CONTROLLED OSCILLATOR - WAVESHAPER.

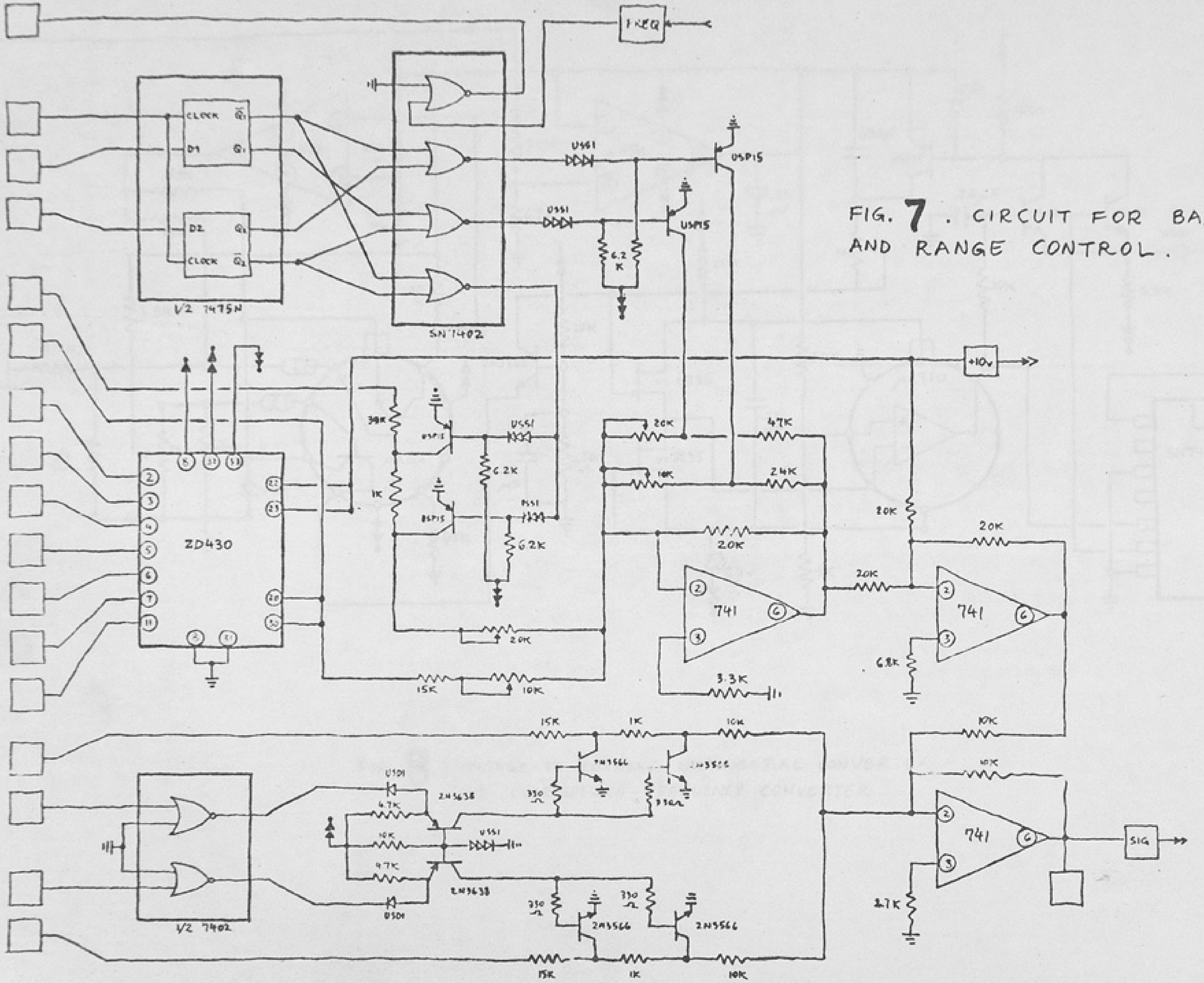


FIG. 7. CIRCUIT FOR BASE AND RANGE CONTROL.

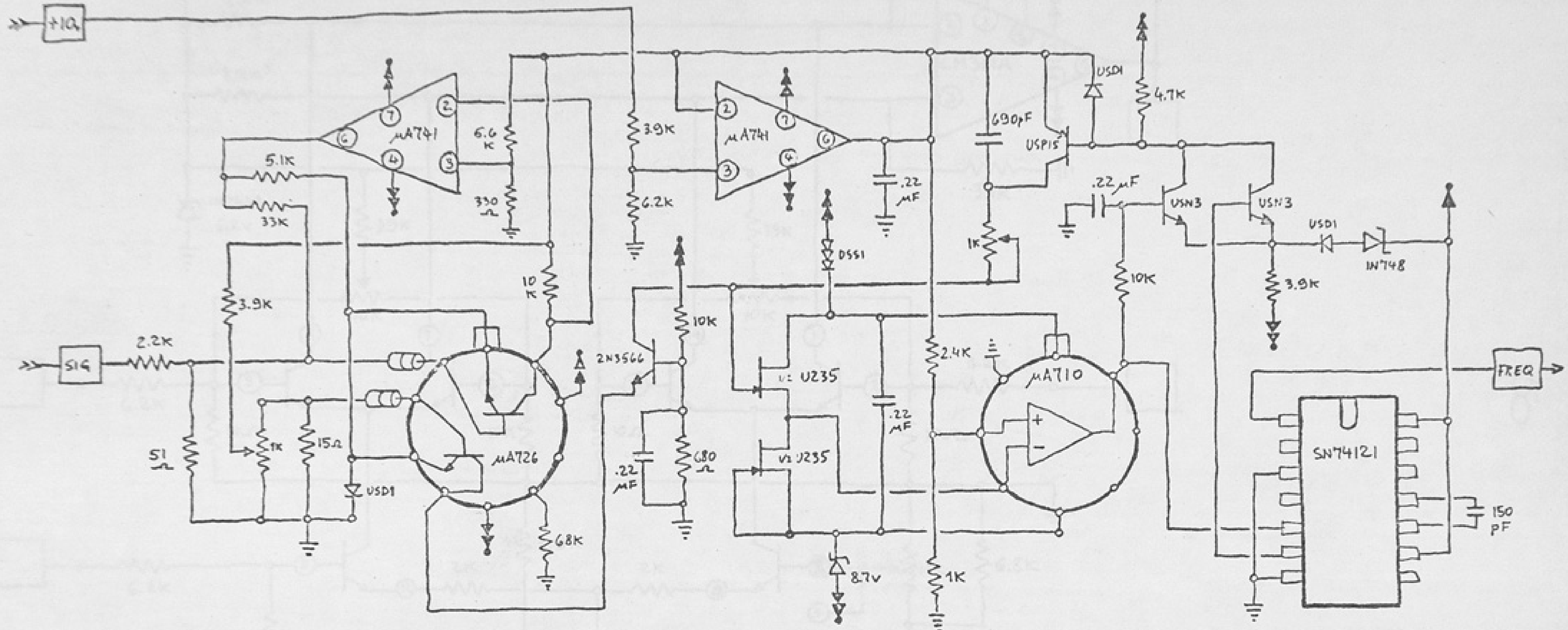
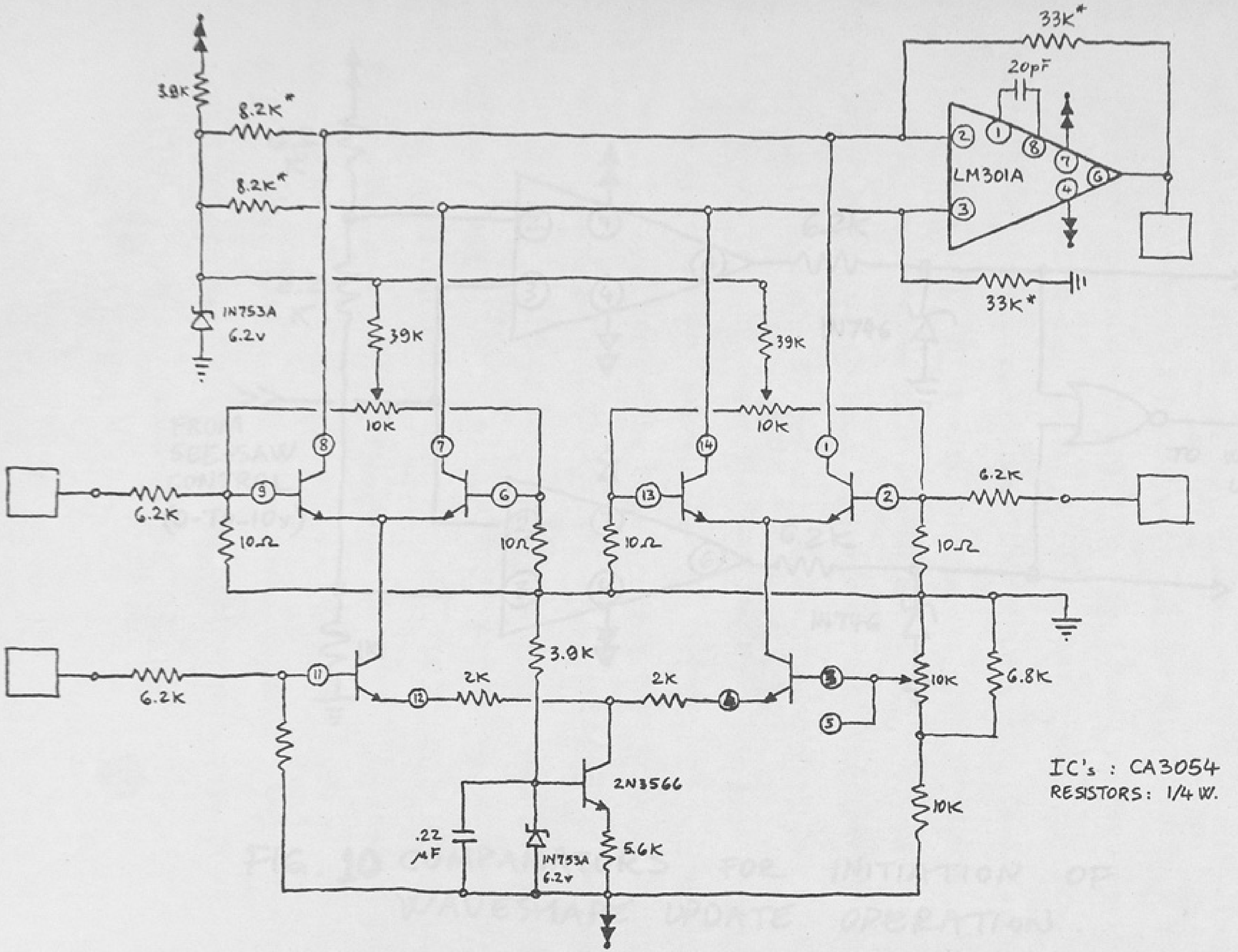


FIG. 8. VOLTAGE-TO-CURRENT EXPONENTIAL CONVERTER AND CURRENT-TO-FREQUENCY CONVERTER



SEE-SAW

FIG. 9. [redacted] CIRCUIT.

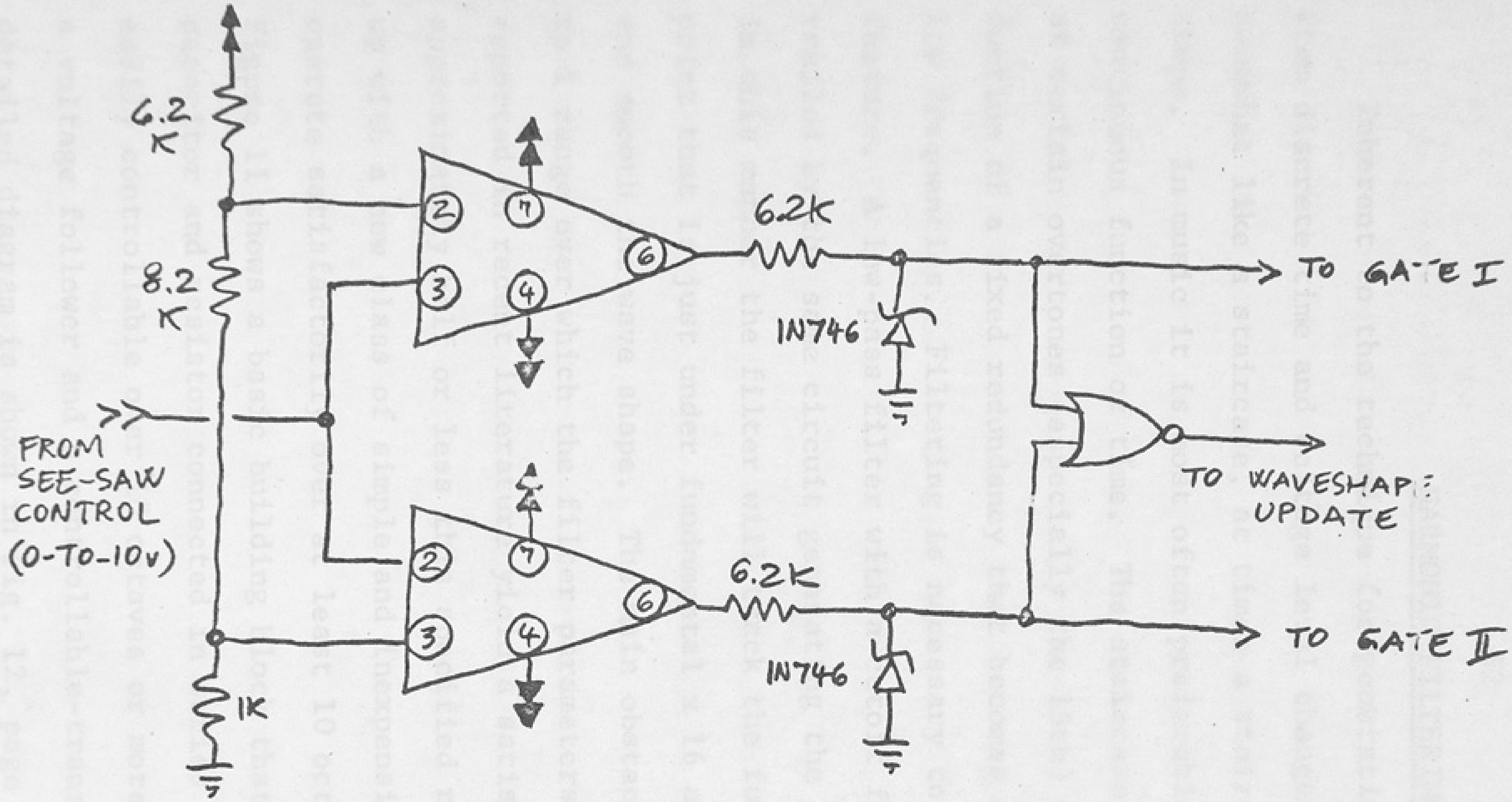


FIG. 10 COMPARATORS FOR INITIATION OF WAVESHAPE UPDATE OPERATION.

HARMONIC FILTERING

Inherent to the technique for generating a waveshape that results from discrete time and voltage level changes is an output that looks somewhat like a staircase, at times a staircase with diverse height steps. In music it is most often preferable to have a shape that is a continuous function of time. The staircase results in the enhancement of certain overtones (especially the 15th) with the consequent introduction of a fixed redundancy that becomes particularly noticeable at low frequencies. Filtering is necessary to eliminate this static feature. A low-pass filter with a cutoff frequency that can be controlled by the same circuit generating the fundamental is necessary. In this manner the filter will track the fundamental with a cut-off point that is just under fundamental x 16 and filter out the overtones and smooth the wave shape. The main obstacle is presented by the 1000 to 1 range over which the filter parameters must be controlled. Designs reported in recent literature yield a satisfactory behaviour over approximately half or less this specified range. Mr. Franco has come up with a new class of simple and inexpensive configurations that operate satisfactorily over at least 10 octaves of dynamic range. Figure 11 shows a basic building block that is equivalent to a capacitor and resistor connected in series with a time constant that is easily controllable over 10 octaves or more. The circuit consists of a voltage follower and a controllable-transconductance amplifier whose detailed diagram is shown in Fig. 12, page 54.

The control voltage is converted into an exponential current by a complimentary transistor pair and as this current drives the transconductance amplifier directly there is no signal degradation.

If the circuit of fig. 11, page 54 is driven from port 1 with port 2 connected to the common line, it will behave like a CR network with fixed C and variable R. If the roles of the ports are interchanged, it will behave like an RC network with a fixed R and variable C. A variation of the same circuit is shown in fig. 13. The operational amplifiers in addition to performing the roles required by the configurations lower the output impedance and (for CR configuration of the circuit shown in fig. 13) increase the input impedance. Greater flexibility with respect to circuit decoupling is offered for use in a variety of designs utilizing these principles. We expect to use these circuits as basic building blocks for voltage-controlled networks of greater complexity. As an example, fig. 14 is a design of a second-order, maximally-flat, low-pass voltage-controlled filter which we may use provided it passes all the tests.

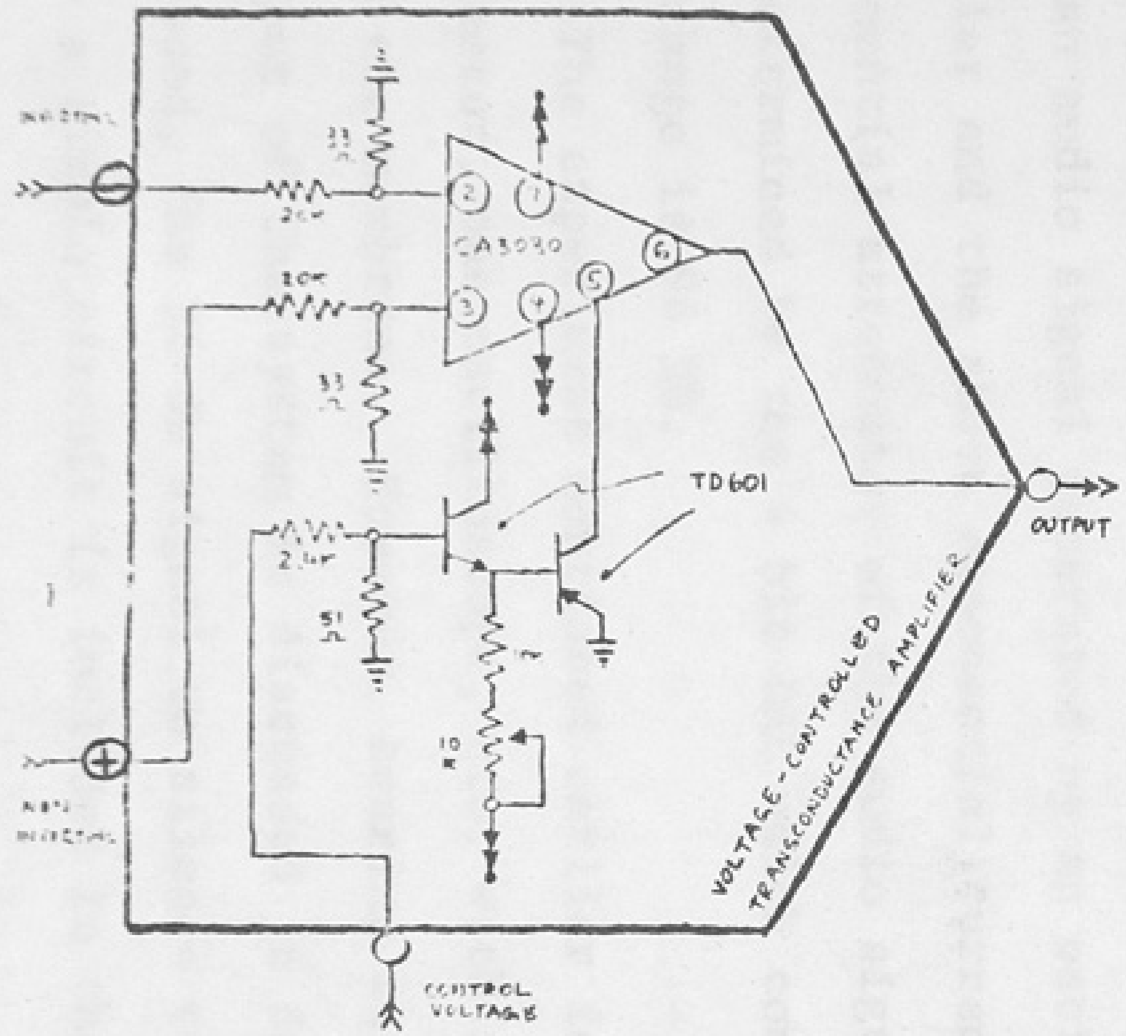


FIG. 12. VOLTAGE-CONTROLLED TRANSCONDUCTANCE AMPLIFIER.

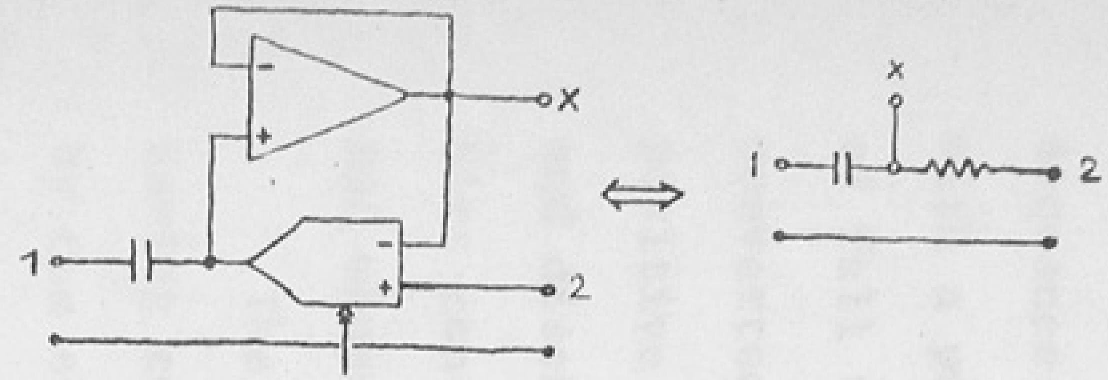


FIG. 11. VOLTAGE-CONTROLLED FILTER BLOCK.

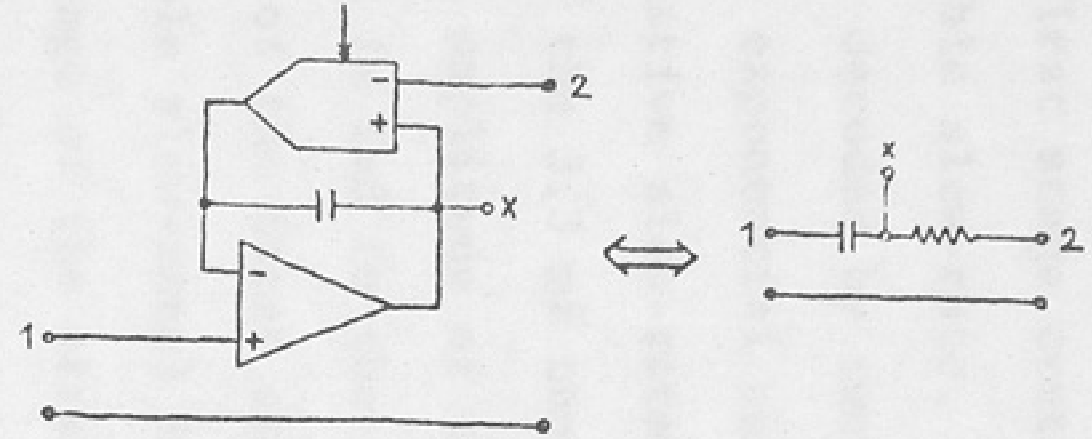


FIG. 13. VOLTAGE-CONTROLLED FILTER BLOCK

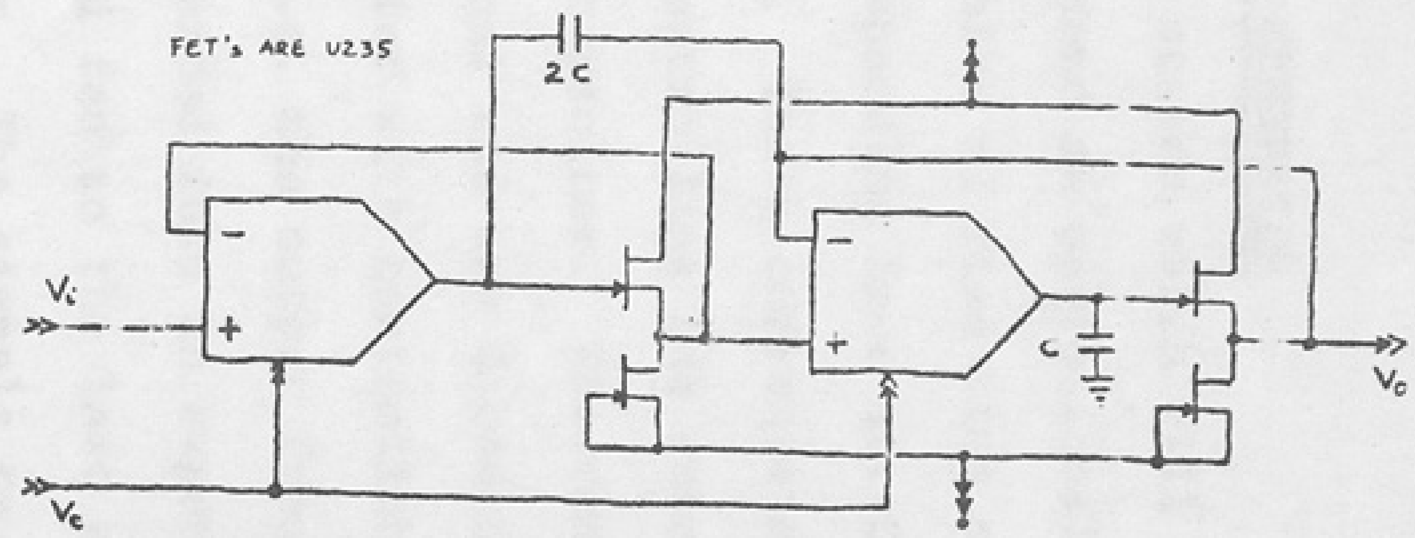


FIG. 14. SECOND-ORDER LOW-PASS BUTTERWORTH FILTER.

SOUND ENVELOPE GENERATOR

This circuit consists of various stages which will be described in sequence. The first stage centers around an operational amplifier with a programmable slew-rate. Four bits of rise time and four bits of fall time are decoded by two corresponding four bit DACS ~~AND~~ and then converted into 2 exponential currents. These currents determine the positive and negative slew-rates by controlling the rate of charge and discharge of the 3.3 mf tantalum capacitor. The remaining four bits control the amplitude of the signal and are decoded by a third DAC whose output is fed to the amplifier with controllable slew-rate.

The output of the first stage (i.e. the output from the amplifier having controllable slew-rate) is converted into an exponential current by the second stage of the circuit and fed to the last stage which is a variable-transconductance multiplier. The signals to be multiplied are an audio signal generated by an oscillator-waveshaper described earlier and the above exponential current. The multiplier acts as an exponential attenuator of the audio signal. The degree of attenuation is determined by the 4 bit DAC which control the amplitude circuit. The range is 96 DB.

The experiment mentioned earlier involved tests with one speaker, one meter, one oscilloscope, etc. with our ears about 6 inches away from the membrane. However, bearing in mind that there are 24 speakers (output of the system is discussed in detail later) and many sources of sound, the 96 db signal to silence range may be inadequate. Therefore a simple circuit is included in the design that monitors the output of the first stage and whenever it falls below 10 to 20 millivolts corresponding to an attenuation of 96db, it shuts off the multiplier circuit in the last stage thereby providing an effective silence of about 140db.

Provision is made so that a user may HOLD an amplitude during the evolution of a rise or fall time independently of the program controlling the amplitude circuit. Blocking the information alone will not always do it because the specific moment chosen to inhibit the rise or fall may be before it has reached the peak of the amplitude programmed. A control bit is utilized which blocks and holds by disabling the positive or negative current. The output of the first stage will remain at a voltage value which is determined by the charge stored in the tantalum capacitor at the moment that current is disabled.

REST control

Another latch which allows the performer to override the program in real time is the REST control. With a "1" at its output, events proceed according to program. When it is changed to "0", the four amplitude bits are forced to "0". The amplitude will decay with whatever information is operative in the fall time program. By making use of the probability that different fall times will be operative at different times according to the moment chosen by the performer, redundancy when a rest is introduced is avoided. Silence will continue as long as the latch remains at "0".

A striking musical effect is achieved if several music signals proceed from "1" to "0" or from "0" to "1" at varying rates.

Simple counterpoint results from group of music signals whose rest controls are independently programmed.

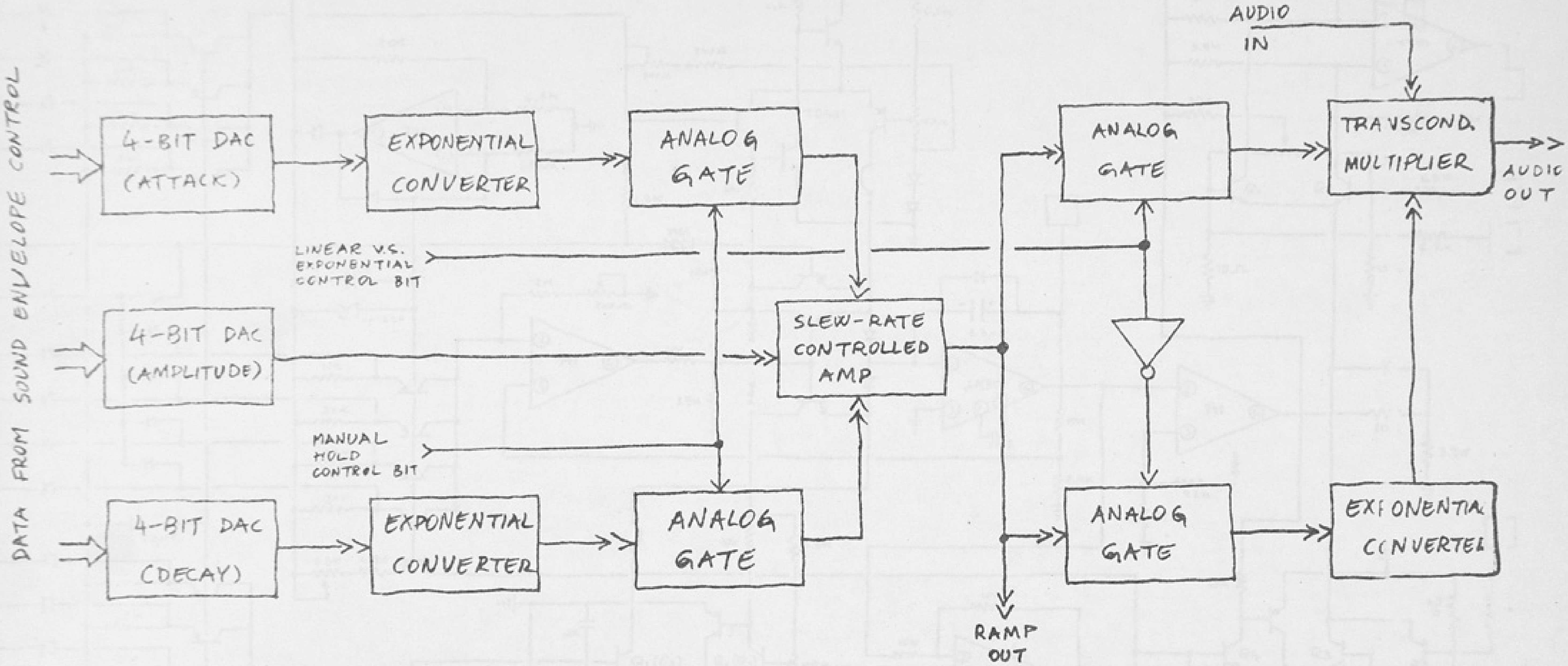


FIG. 15: ENVELOPE GENERATION BLOCK DIAGRAM

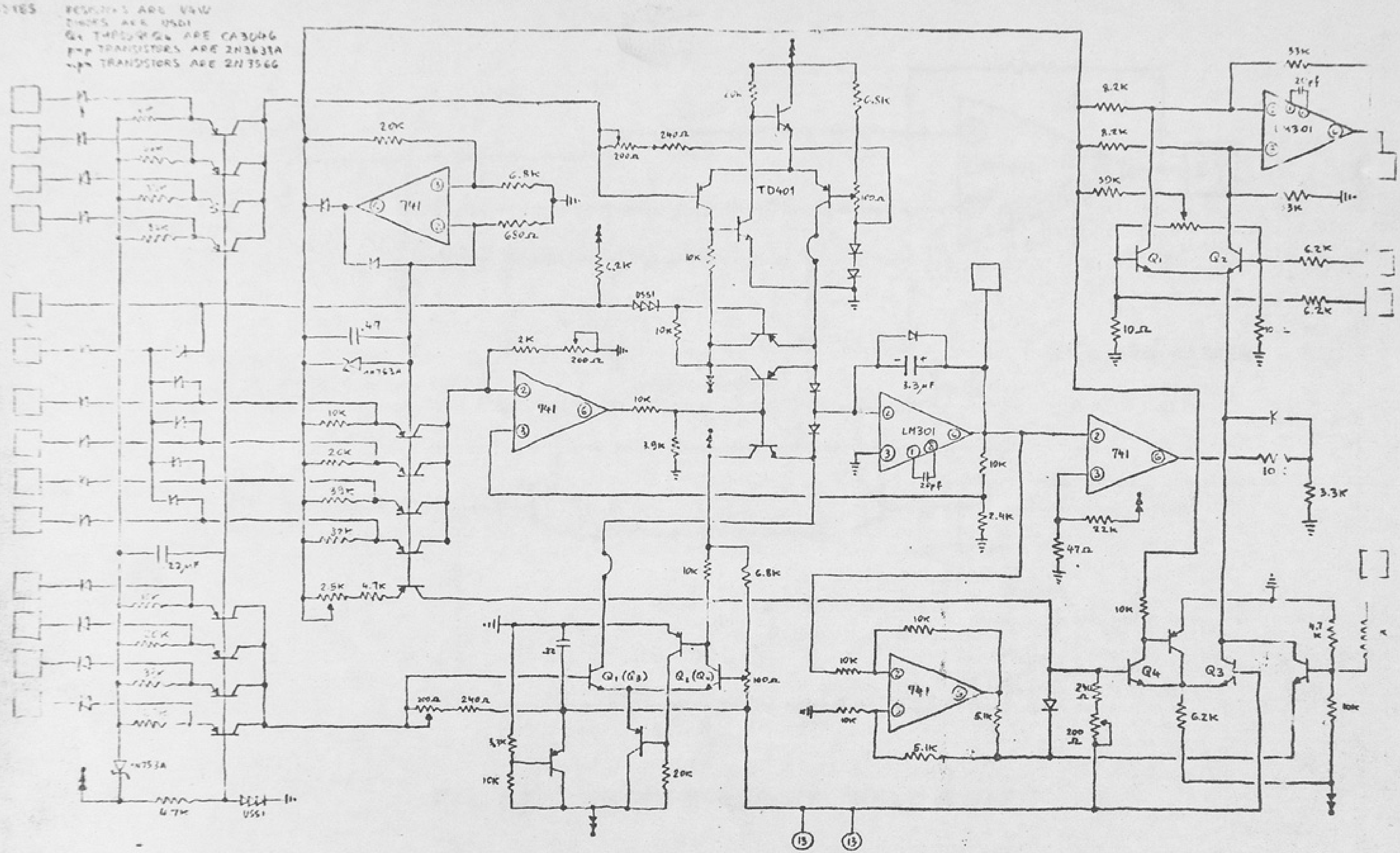


Fig.16: Envelope Generator

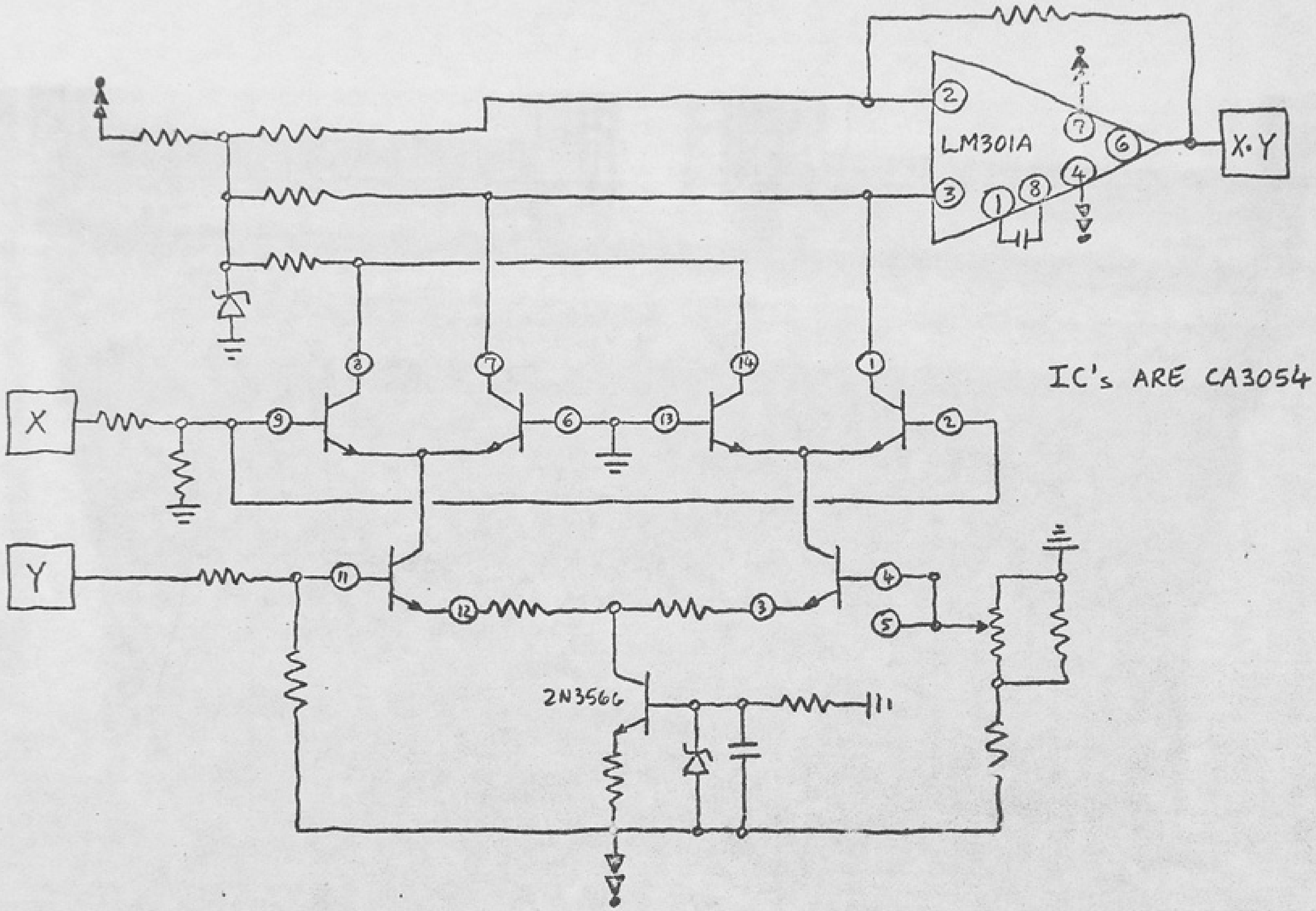
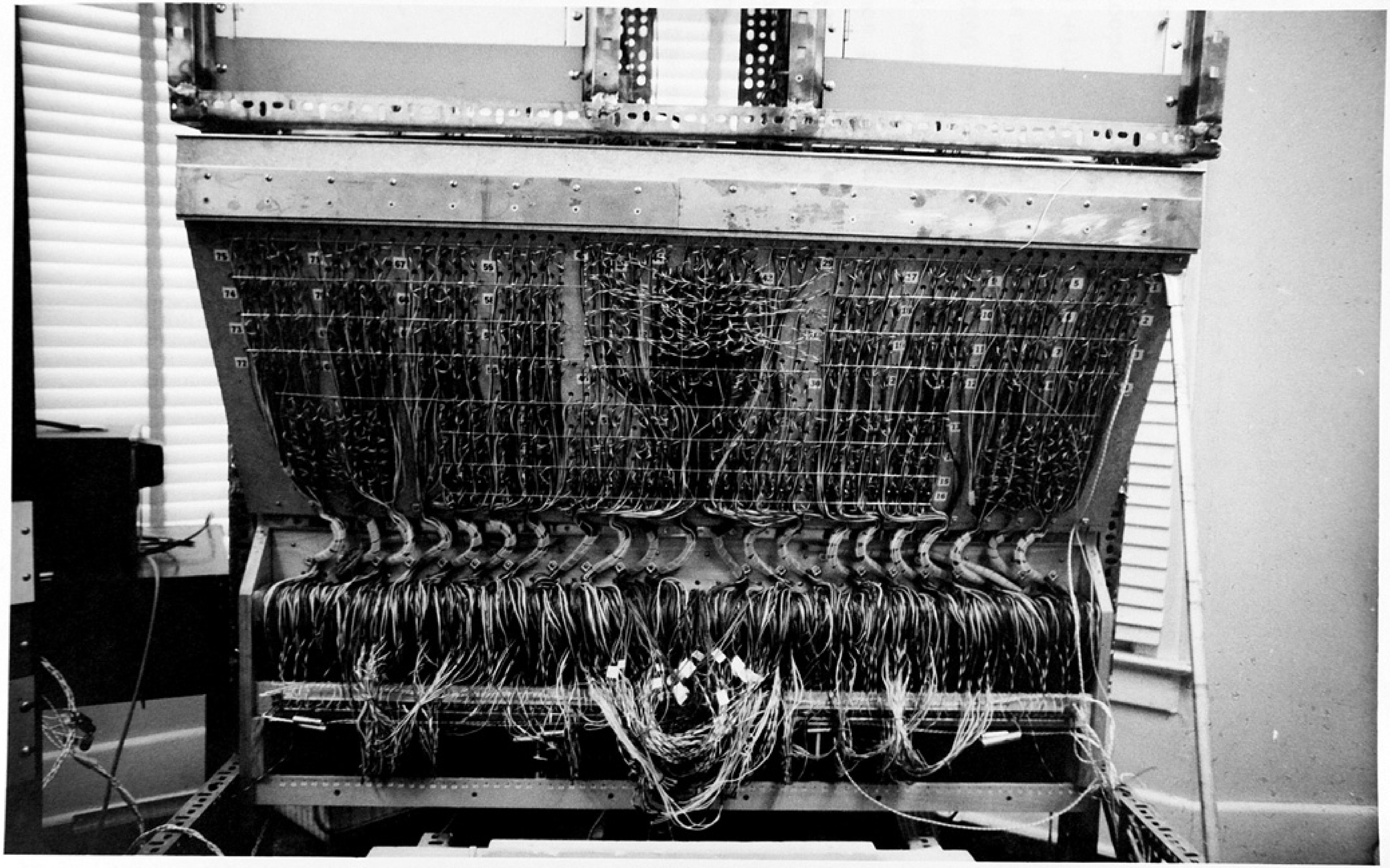


FIG. 17. FOUR-QUADRANT MULTIPLIER.



WAVESHAPE EXPERIMENT

Fig. 18, page 62, is a circuit which allowed experimentation with waveshapes. Though not adequate, it's a beginning. Whenever the counter loops to 0000, "1's" are sent to all 4 inputs on both of the DAC's and +10vdc is at the output of each. The most negative voltage, -10vdc, is movable to any of the 15 out of 16 time slots. The EX-OR and the ZERO detector used with 4 bit shift register (7495) never allow 4 zeros to be programmed in the latches (7475). The output of the latches pair with the output of the counter when it arrives through the EX-OR's and "0's" are sent to all 4 inputs on both of the DAC's. Random information is derived from non-related sources in the system for the DATA inputs on the RAM's (7489). The modified D-type flip-flop uses the outputs of the COMPARATORS to clock the latches.

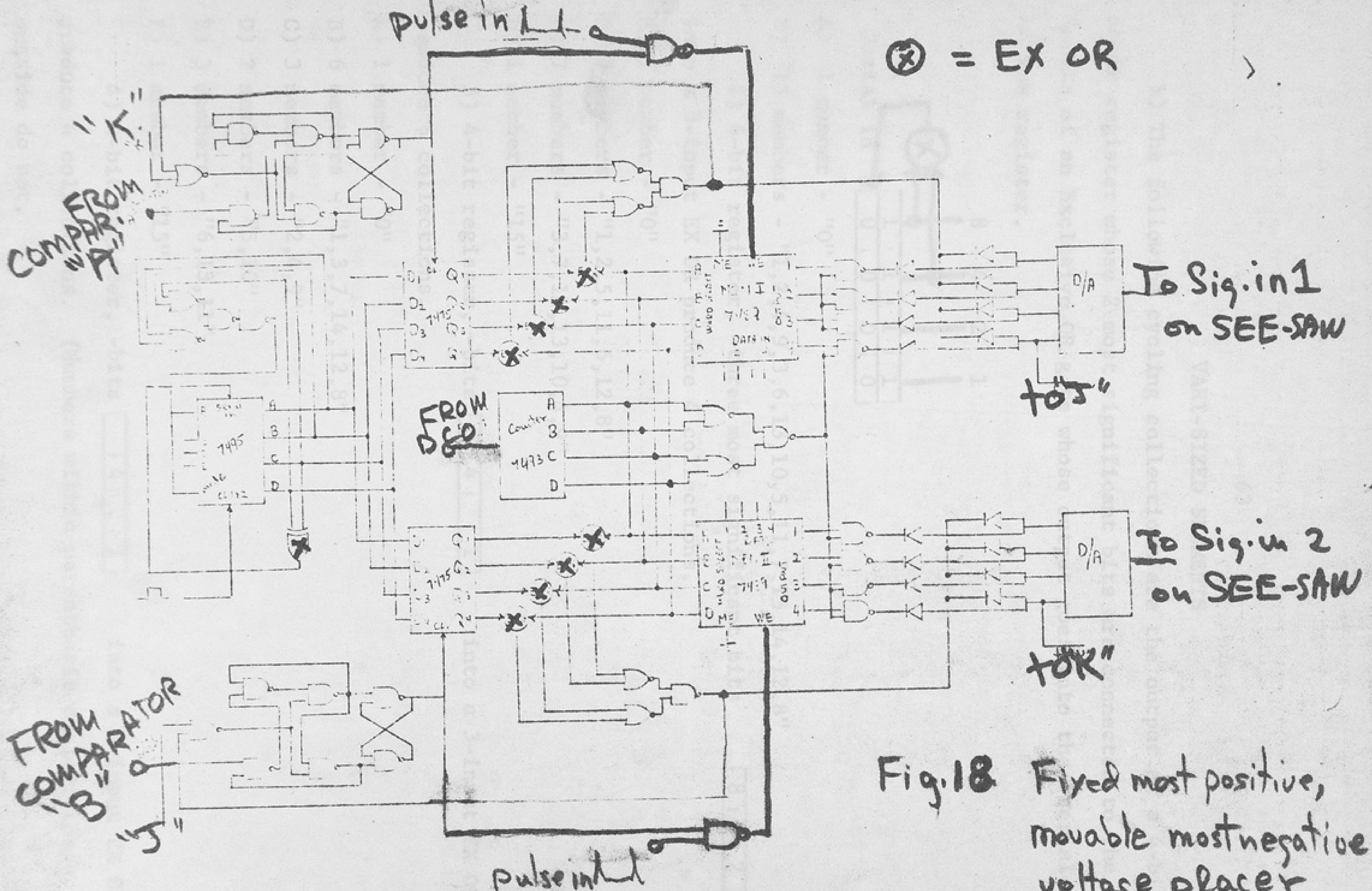
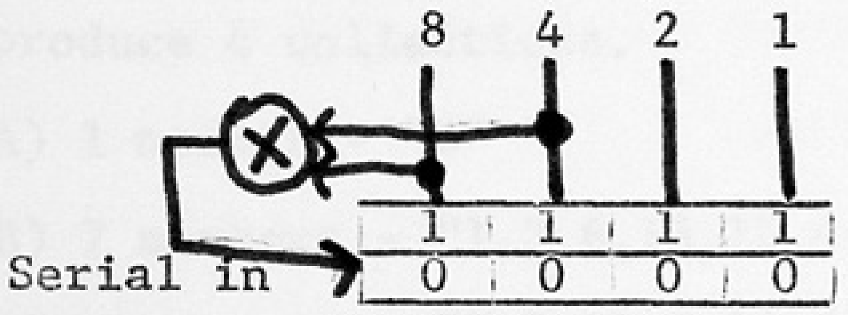


Fig. 18 Fixed most positive, movable most negative voltage placer S.M.

VARI-SIZED SEGMENTS

1) The following cycling collections are the output of a 4-bit shift register whose 2 most significant bits are connected to the inputs of an Exclusive OR gate whose output goes into the "serial in" on the register.



A) 1 member - "0"

B) 15 members - "1,2,4,9,3,6,13,10,5,11,7,15,14,12,8"

2) 4-bit register, -three most significant bits $\boxed{8 \mid 4 \mid 2 \mid \ }$ into a 3-input EX OR produce 4 collections.

A) 1 member - "0"

B) 7 members - "1,2,5,11,6,12,8"

C) 7 members - "3,7,14,13,10,4,9"

D) 1 member - "15"

3) 4-bit register, -bits $\boxed{8 \mid 4 \mid \ \mid 1}$ into a 3-input EX OR produce 6 collections.

A) 1 member - "0"

B) 6 members - "1,3,7,14,12,8"

C) 3 members - "2,4,9"

D) 2 members - "5,10"

E) 3 members - "6,13,11"

F) 1 member - "15"

4) 4-bit register, -bits $\boxed{\ \mid 4 \mid 2 \mid 1}$ into a 3-input EX OR produce 4 collections. (Numbers within parenthesis cycle), those outside do not.

A) 1 member - "0"

B) 5 plus 4 members - "1,4,8,11,14,(6,12,9,3)"

C) 2 plus 2 members - "2,13,(5,10)"

D) 1 plus 1 member - "7,(15)"

5) 4-bit register, bits [8 | | 2 | 1] into a 3-input EX OR produce 4 collections.

A) 1 member - "0"

B) 7 members - "1,3,6,13,10,4,8"

C) 7 members - "2,5,11,7,14,12,9"

D) 1 member - "15"

6) 4-bit register, bits [8 | 4 | 2 | 1] into a 4-input EX OR produce 4 collections.

A) 1 member - "0"

B) 5 members - "7,15,14,13,11"

C) 5 members - "2,5,10,4,9"

D) 5 members - "1,3,6,12,8"

7) 5-bit register, bits [16 | 8 | | |] into a 2-input EX OR produce 4 collections.

A) 1 member

B) 21 members

C) 7 members

D) 3 members

8) 6-bit registers, bits 32 | 16 | | | | into a 2-input EX OR produce 2 collections.

A) 1 member

B) 63 members

The next step is to design a computer program and a set of intelligent questions in order to discover if there are properties in this system that would be useful for controlling large quantities. For example, a 16=bit shift register in which 4 of the bits give the value, in conjunction with an EX OR and other gates, and 12 of the bits are used for address.

MEMORY DESCRIPTION FOR SOUND DISTRIBUTION CONTROL

The memory unit is designed to provide the storage of the matrix for the four programs present in the shift registers. The matrix for each individual program provides the routing of any output of the six shift registers into the input of any of the other five or back into itself. In addition, the memory unit must be able to store the contents of the matrix under the program desired and to display the matrix of any program on command. Finally, it appeared that the memory unit must be easily expandable beyond the four programs presently used, but without sacrificing anything in terms of redesign or slowing down the memory unit.

With these ideas in mind, the following memory unit was designed. A brief explanation of each of the various subsections will be given now, and a specific description in a later report.

The control logic (fig. 19 page 68) has as its function the generation of various clock and storing signals, and the generation of properly coded addresses for the decoding logic. The store button generates a pulse on the LOAD input of the 74161. When the LOAD pulse goes away, the counter begins to count from either 0000 or 1000; a 0000 indicates the information is loaded in the first eight words of memory, corresponding to programs I or III, while a 1000 stores the information in the second eight words of memory, corresponding to programs II or IV. After the counter reaches 0111 or 1111, indicating the storage cycle has been completed, a signal is generated which clears an R-S flip-flop and allows generation of another LOAD pulse, if needed. The action of the STORE signal is inhibited by the presence of any of the pre-clock (P.C.) signals, but a flip-flop is set which will generate the STORE signal when the P.C. signals have gone away. Finally, generation of properly phased clock pulses and properly coded signals are also generated on the control logic board.

The eight-line to one-line multiplexer board (fig. 20 page 69) is fairly straightforward. The multiplexers are only operative during receipt of a STORE signal, and decode the three least significant bits of the counter to scan the matrix and present this information to the memory.

The memory board (fig. 21 page 70) is probably the most complex board of the whole memory unit. Its operation is as follows: during receipt of a STORE signal, the information received by multiplexers is steered to either board A or B by the most significant bit of the program select (P.S.MSB). Thus board A stores programs I and II and board B stores programs III and IV. After the store signal goes away, the counter returns to the READ mode under the command of the ME and WE signals. Since the counter is now free-running, the memory is being continuously read out. The outputs of the shift registers (i.e. I out 1, III out 4, etc.) are ANDed with the information in the memory, and the AND outputs are then ORed and inverted to form the signals Mem I and II out and Mem III and IV out.

The memory output distribution board (fig. 22 page 71) then accepts the information from the memory and stores it in the appropriate flip-flop under the command of the address generator. This provides the system for generating the inputs to the shift registers.

Because the direction is stored as the zeroeth and eighth word in the memory, the direction storage registers (fig. 23 page 72) are strobed when the zeroeth and eighth words are present in order to store the direction data. These registers then present the direction (i.e., right shift or left shift) to the appropriate shift registers.

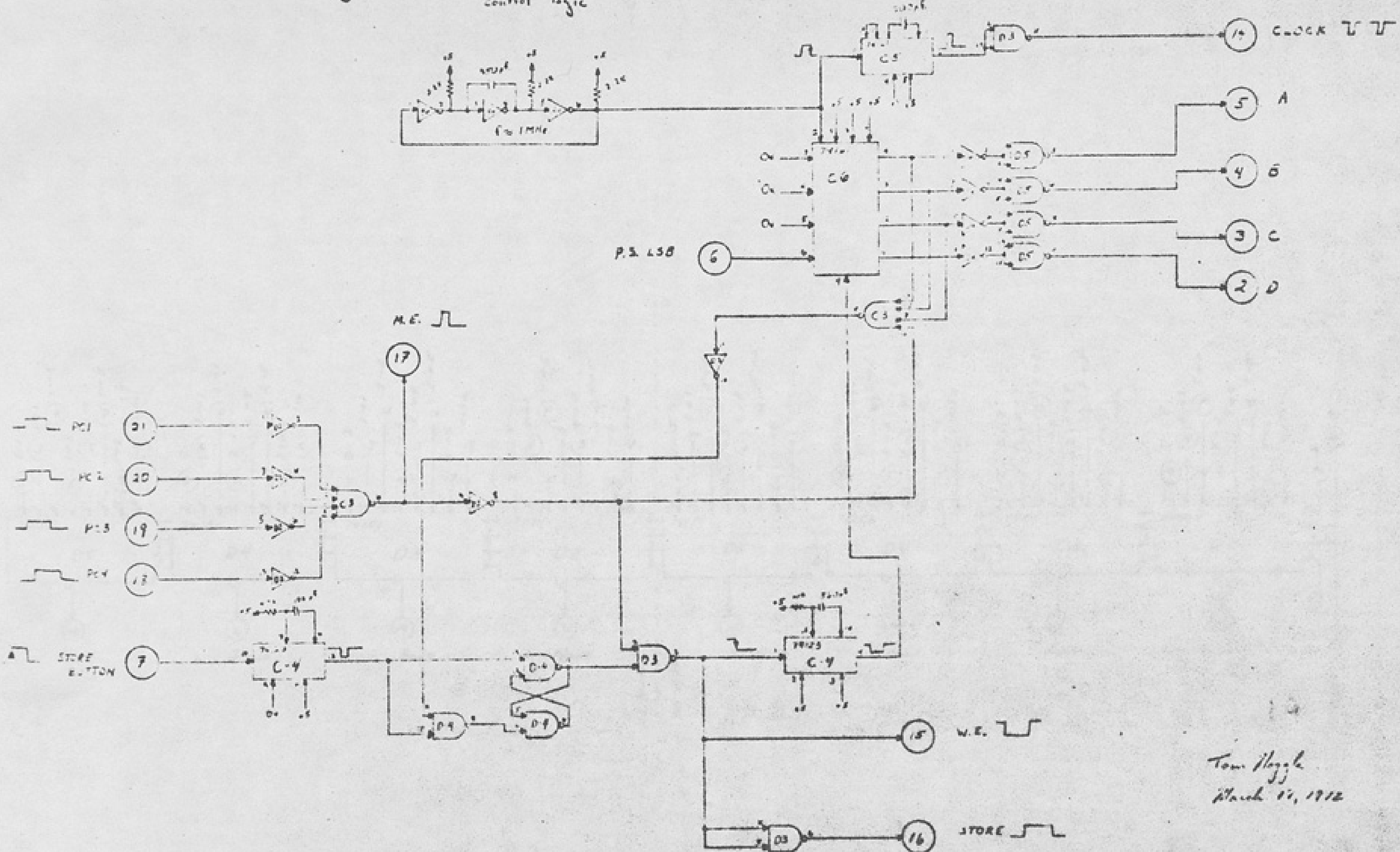
The ME-106 (fig. 24 page 73) board contains a 16-line demultiplexer which does the address generation for the memory output latches. It also contains part of the display control section, for which there was not room on another board.

The display control (fig. 25 page 74) decodes the two program select bits, the memory output words, and the address bits to allow storage of the proper bits for the program desired. The outputs of the display control are stored in a series of latches (not shown) and are then displayed on the control panel.

Fig. 19

ME-101

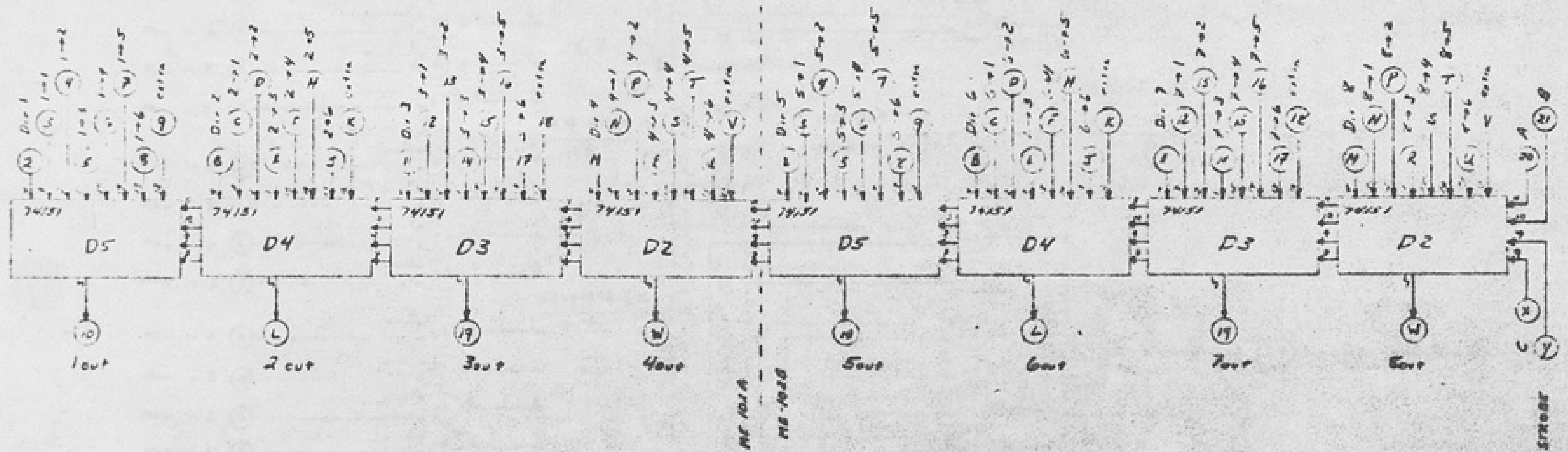
control logic



March 11, 1972

Fig. 20 ME-102

8-1 multiplexer



Tom Hoggle
March 18, 1972

70

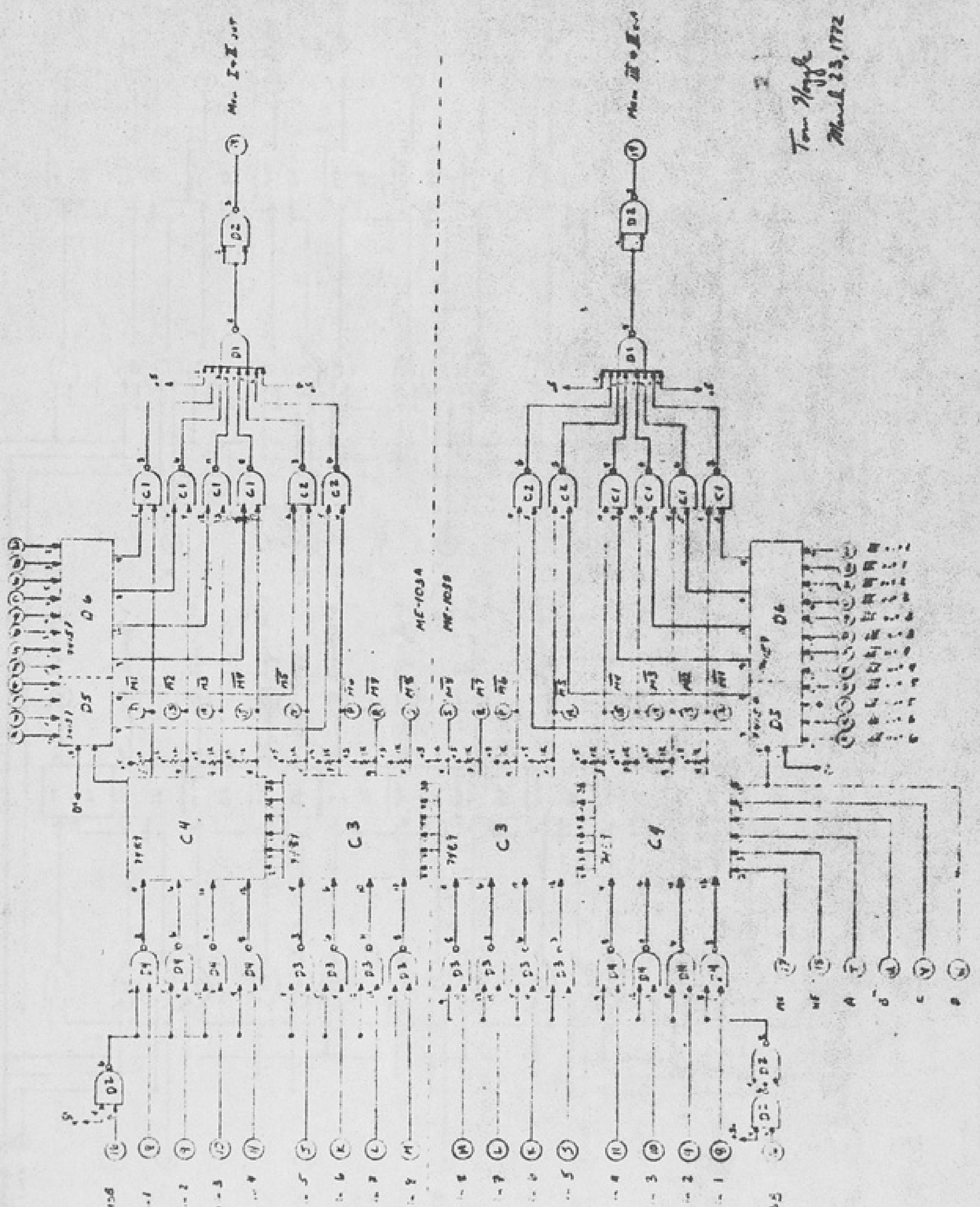
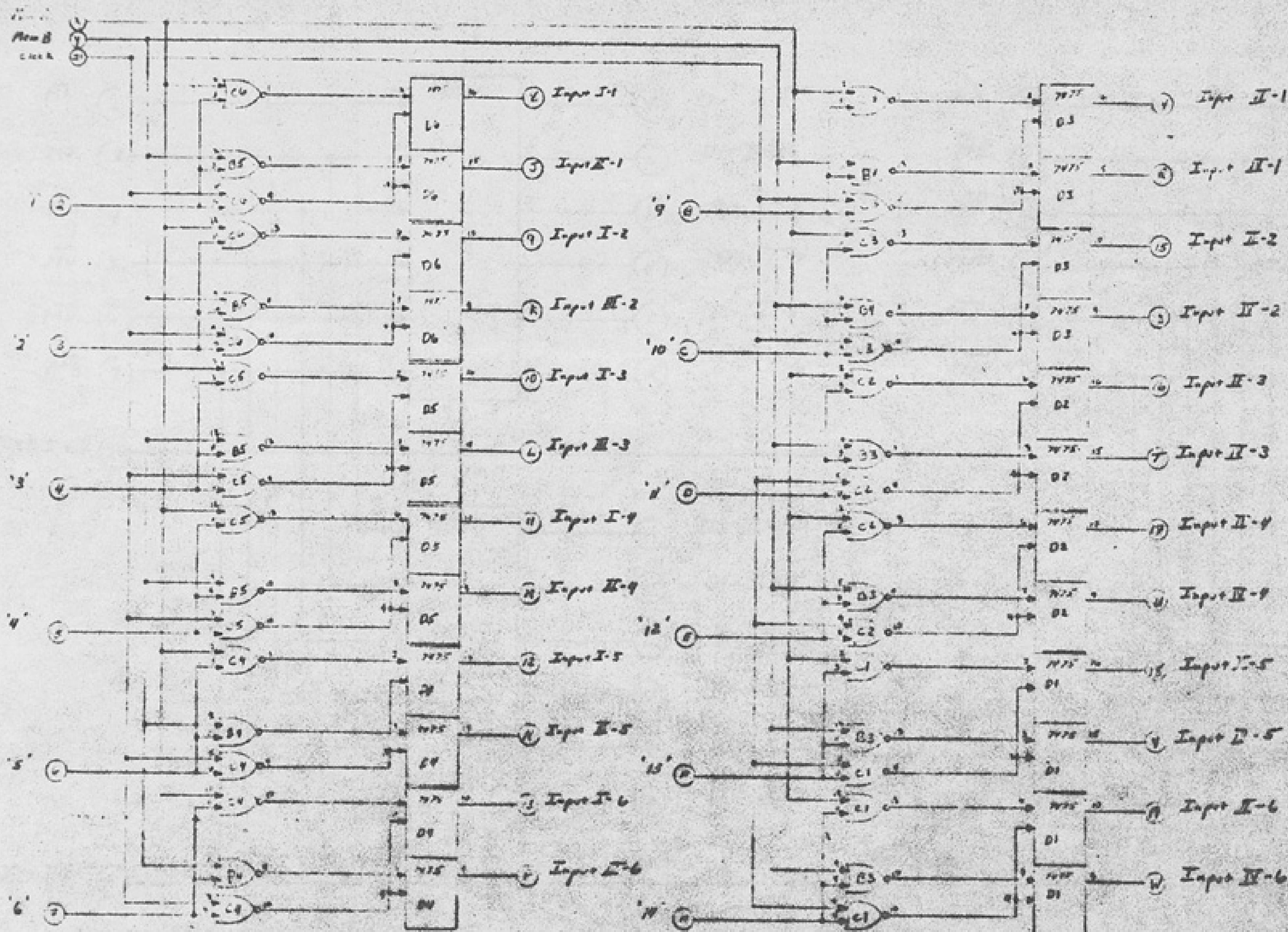


Fig. 21

ME-103

Fig. 22 ME-104



Tom [illegible]
March 25, '72

Fig 23

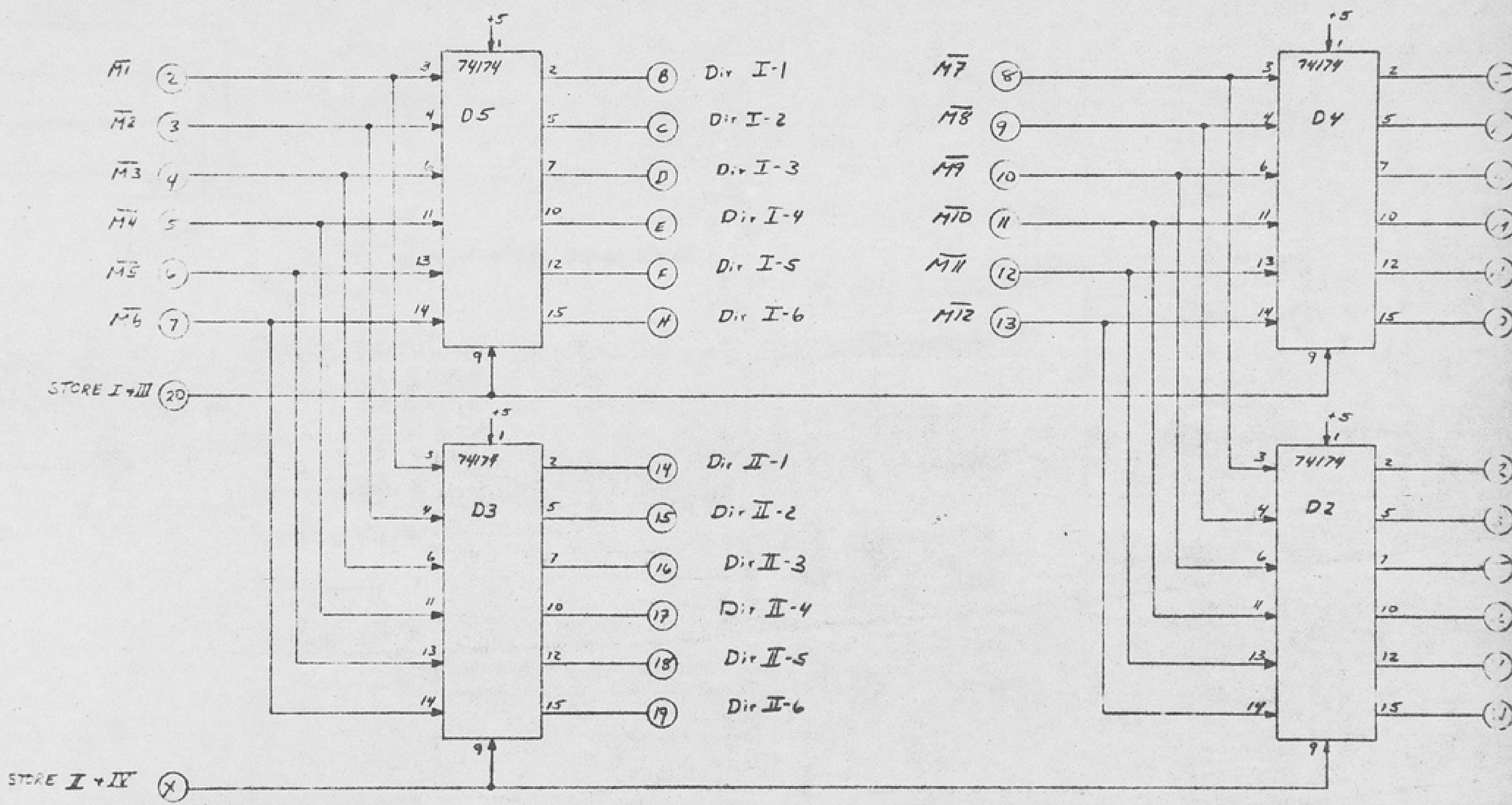

74174
D5
D3
D4
D2
+5
Dir I-1
Dir I-2
Dir I-3
Dir I-4
Dir I-5
Dir I-6
Dir II-1
Dir II-2
Dir II-3
Dir II-4
Dir II-5
Dir II-6
STORE I + III
STORE II + IV

Fig. 24 ME-106

16 line demultiplexer
and partial display control

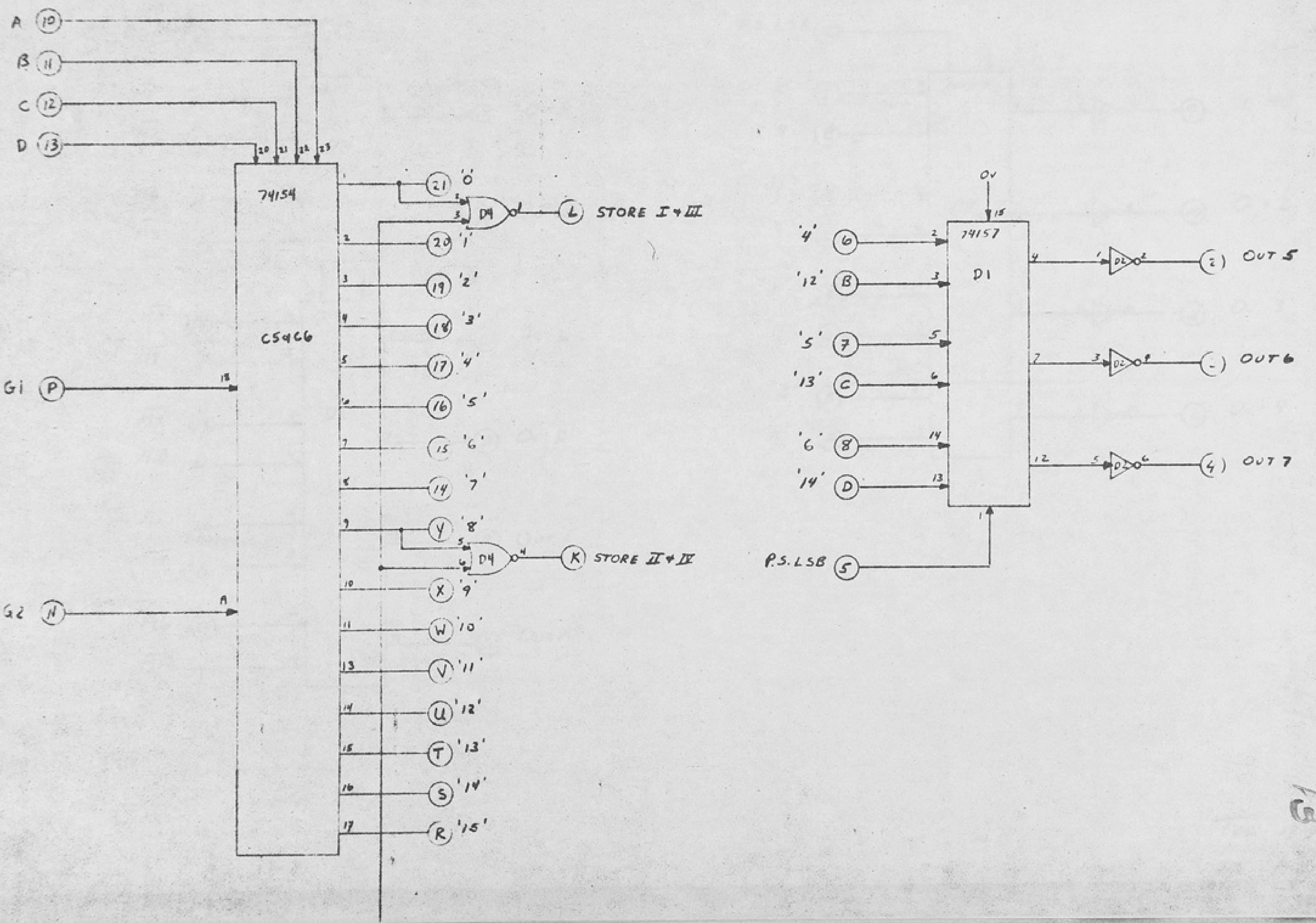
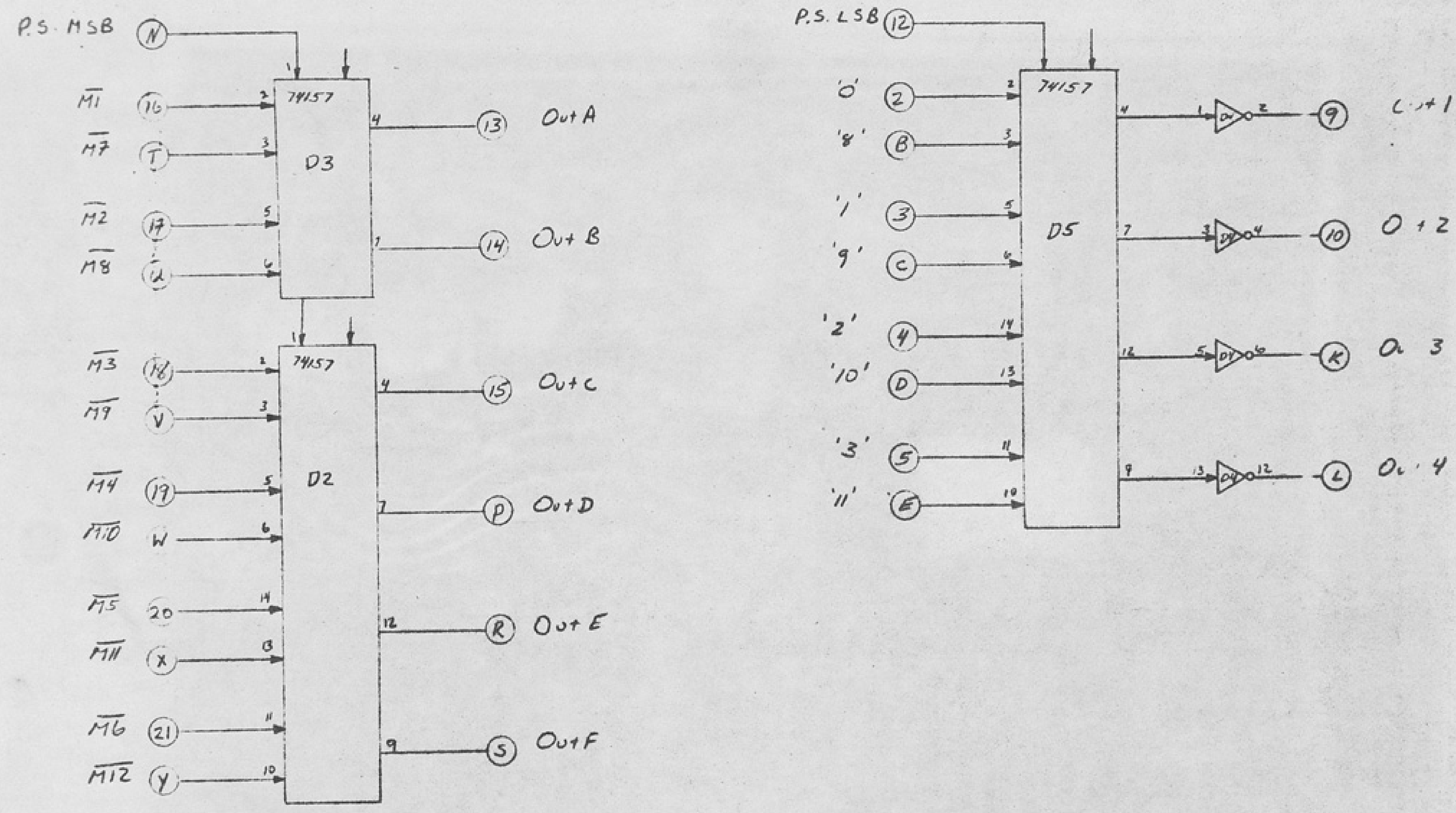
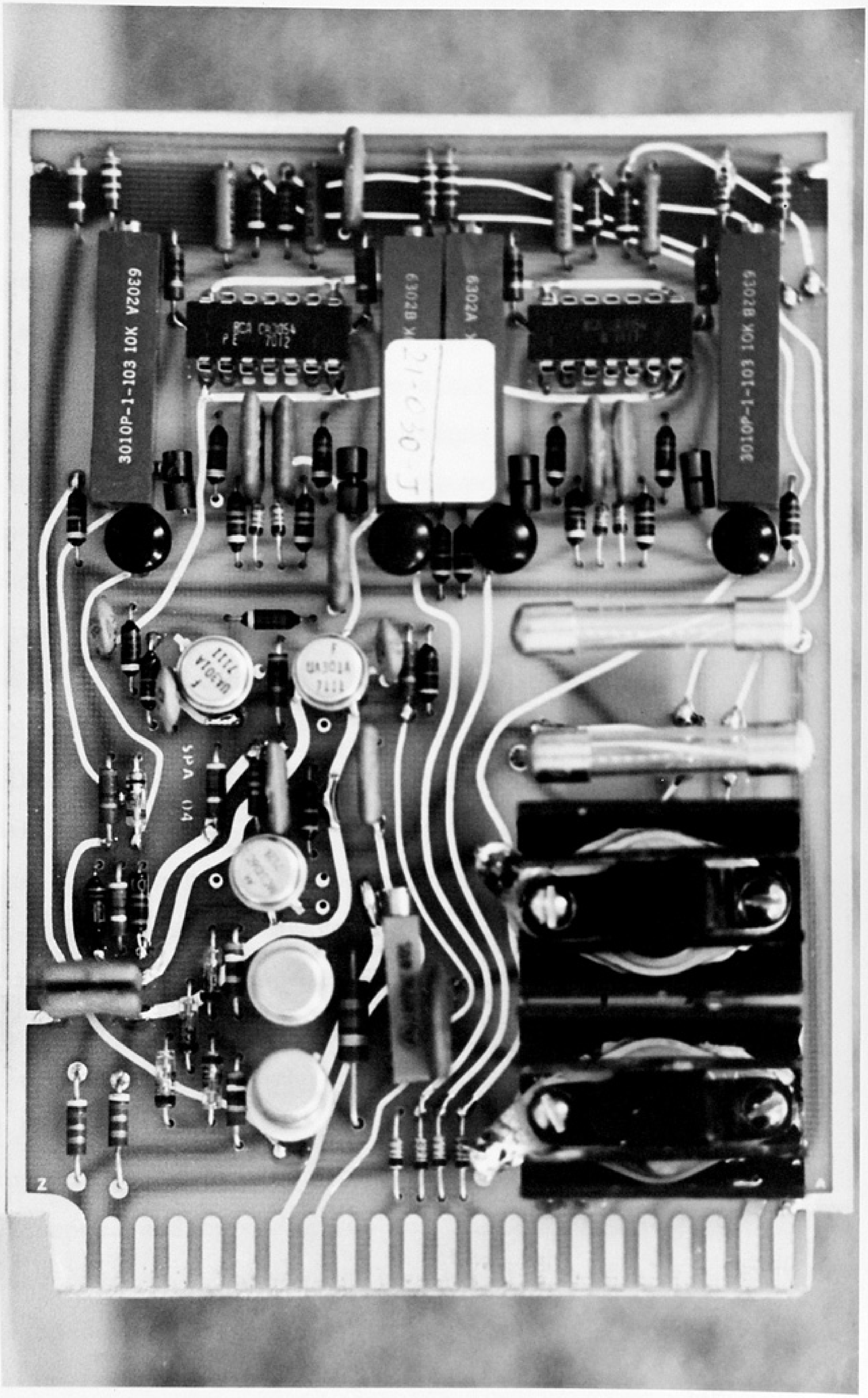


FIG. 25



74

Tom
Mar 1.26


3010P-1-103 10K
SPA 04

(SHIFT REGISTERS)

Each program contains two -3 bit, two -4 bit, and two -5 bit shift registers of which a typical 4 bit shift register is shown. These registers are designed to allow parallel loading of all positions except during a short interval of time immediately before and after a clock pulse. A signal designated pre-clock (P.C.) is this 'envelope' about the clock pulse. When the P.C. signal arrives, the parallel load mode is inhibited and the logic steers the input to the appropriate point and selects the proper output signal. After the main clock pulse has appeared, the P.C. signal remains up until, after selection of the proper program through twelve 4 line to 1 line multiplexers, the display strobe allows the contents of the selected program to be displayed for a short period of time (about 100 micro-seconds). Since the touch switch is itself a flip-flop, the touch switch will assume the state of the information displayed to it, unless, of course, the operator is touching the touch switch. If the operator is intervening, the touch switch may assume a useless state for the period of time during the display strobe. However, the display strobe disappears before the P.C. disappears, thus, if there is input of information from the outside, the touch switch can assume the state desired before the shift register returns to the parallel load mode.

The control logic for the shift registers generates a series of properly sequenced and timed pulses for the operation of the shift registers and display. The normal sequence of operation of the control logic is as follows: a clock request is received by an R.-S flip-flop which acts to store the request. The actual generation of the clock pulse is initiated by 'NOR'ing the clock request and address change lines. The request for P.C. is stored in another R-S flip-flop, which generates the P.C. signal, but the request for a P.C. may be delayed by the presence of a STORE signal. The P.C. signal is delayed by about 20 micro-seconds (to insure one cycle of information through the memory) and then, if the P.C. clock was initialized by a clock request signal, a main clock pulse is generated. On the other hand, if a P.C. signal was generated by the address change, no main clock signal is produced. The reason for this organization is that the P.C. signal also generates the display strobe, thus, when one changes to another program, it would be desirable to have everything displayed but without clocking the shift registers. After the display strobe has occurred, the P.C. end signal is generated, which returns all the R-S flip-flops to a state in which they can again be triggered. One half of the 4 line to 1 line demultiplexer (74153) selects the proper display strobe pulse for the display, and the other half selects the proper P.C. end signal to clear the flip-flop for the address change, thus returning all the circuitry to a state in which it can again generate the proper sequence of pulses.

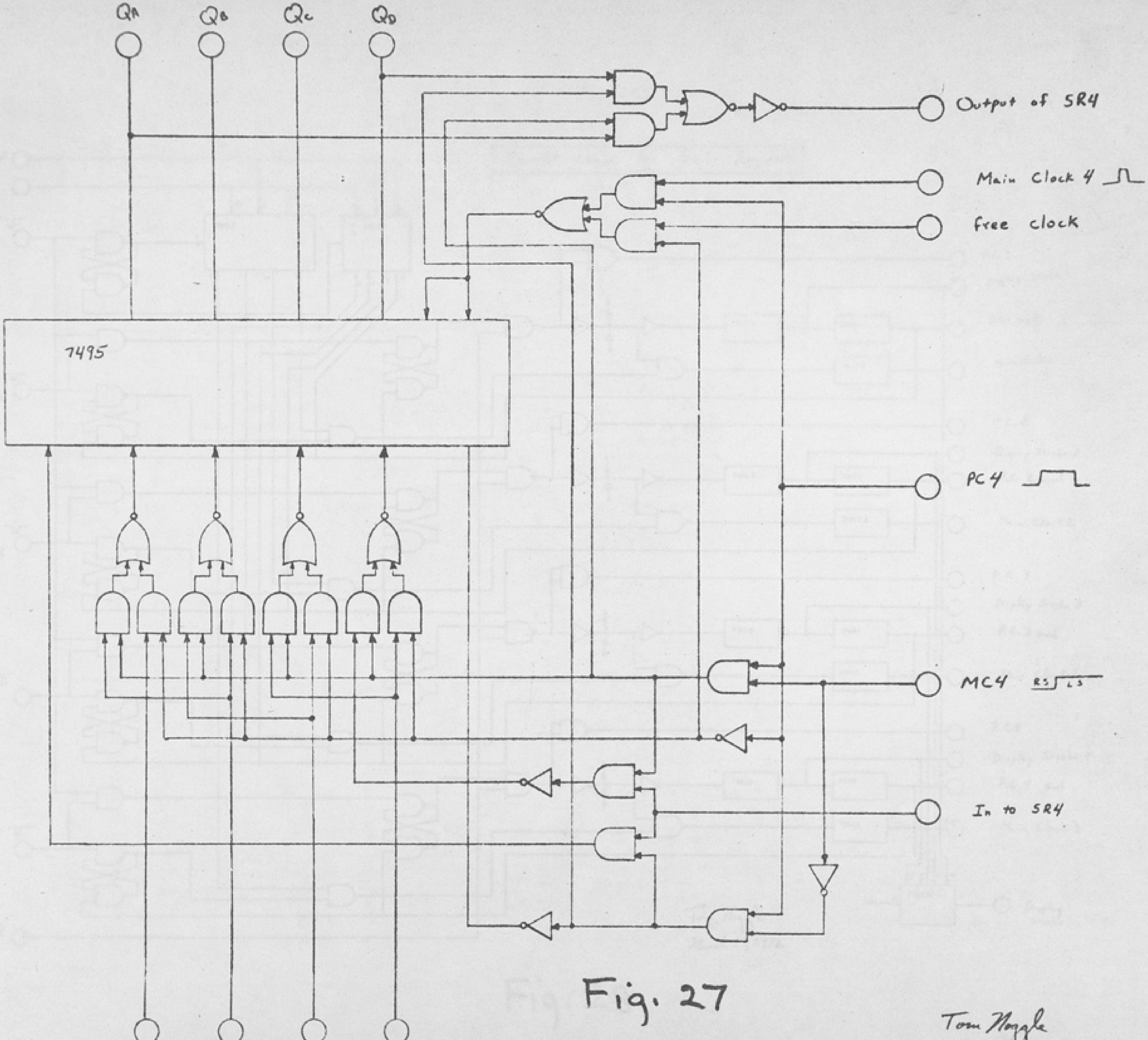


Fig. 27

Tom Noggle
March 27, 1972

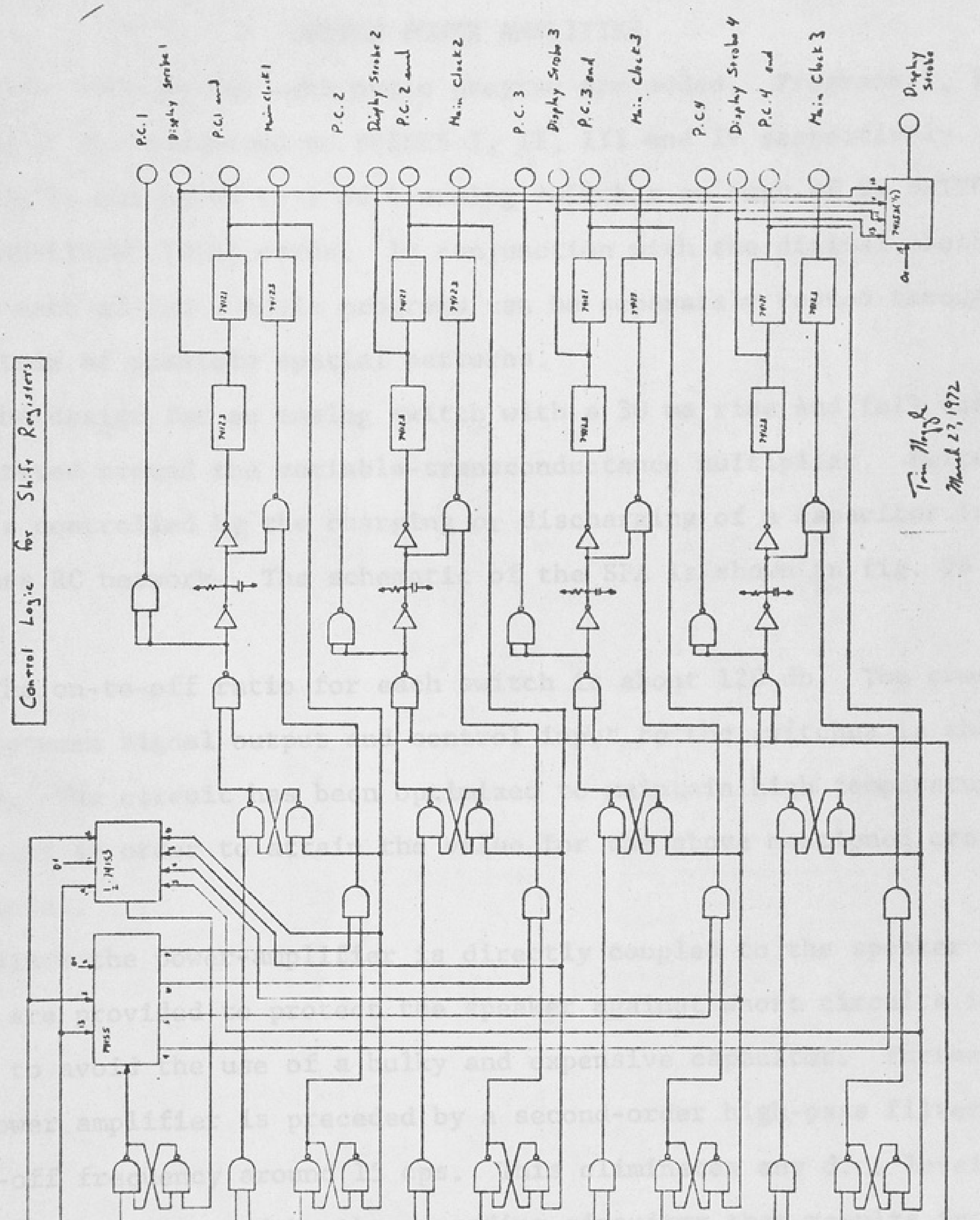

Control Logic for Shift Registers
P.C.1
Display Strobe 1
P.C.1 and
Main Clock 1
P.C.2
Display Strobe 2
P.C.2 and
Main Clock 2
P.C.3
Display Strobe 3
P.C.3 and
Main Clock 3
P.C.4
Display Strobe 4
P.C.4 and
Main Clock 3
Display Strobe
P.S. MSB
P.S. LSB
Address change
Clock request 1
Clock request 2
Clock request 3
Clock request 4
STORE
March 27, 1972

Fig. 28

SWITCH POWER AMPLIFIER

Sound sources for each music program are added. Programs I, II, III and IV are connected to BUSSES I, II, III and IV respectively. Each bus is connected to 1 of 4 analog switches on each of 24 SWITCH POWER AMPLIFIER (SPA) cards. In conjunction with the digital control system each of the 4 music programs can be separately routed through the matrix of possible spatial patterns.

The design for an analog switch with a 30 ms rise and fall time is centered around the variable-transconductance multiplier. Switching time is controlled by the charging or discharging of a capacitor in a low-pass RC network. The schematic of the SPA is shown in fig. 26 page

The on-to-off ratio for each switch is about 120 db. The cross-talk between signal output and control input to the switches is about -80 db. The circuit has been optimized to maintain high temperature stability in order to attain the value for the above mentioned cross-talk level.

Since the power-amplifier is directly coupled to the speaker fuses are provided to protect the speaker against short circuits in order to avoid the use of a bulky and expensive capacitor. Further, the power amplifier is preceded by a second-order high-pass filter with a cut-off frequency around 15 cps. This eliminates any d.c. level which may be generated by the preceding circuitry that results from faulty components or errors of operation. The filter also boosts the lowest range of the scale and thus improves the performance of the speakers at low frequencies.

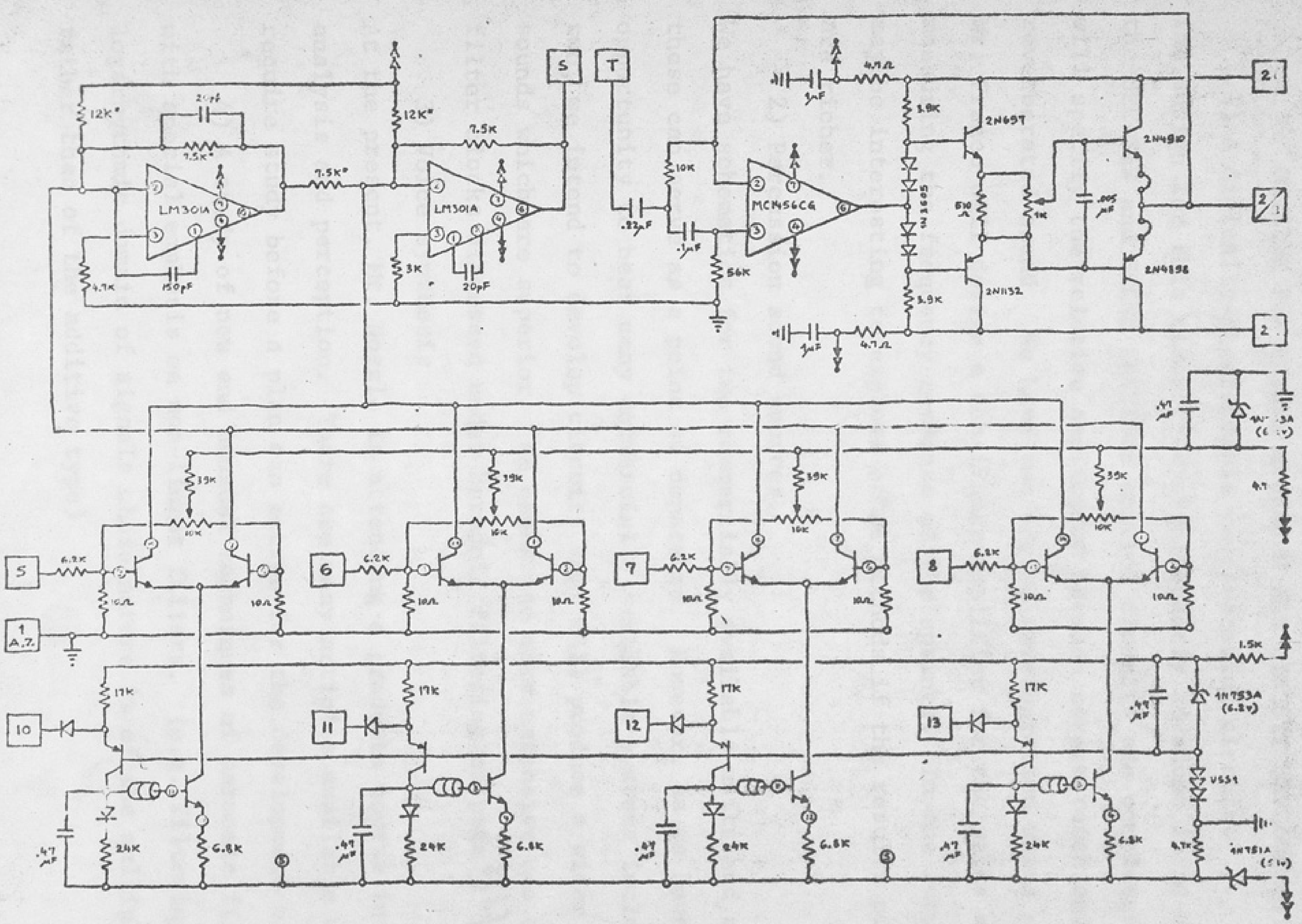


FIG. 26 SWITCHED POWER AMPLIFIER.

NOTES: pnp TRANSISTORS ARE 2N3138A
DIODES ARE USD1
INTEGRATED CIRCUITS ARE C 3054
RESISTORS ARE 1/4 WATT, 5%

OUTLINE FOR FUTURE RESEARCH AND CIRCUIT DEVELOPMENT

1) A digitally-programmable reverberation circuit.

The design for this circuit is significantly advanced in as much as the SEE-SAW and SOUND ENVELOPE control circuits are completed. These will specify the relative amplitudes between reverberated and un-reverberated sound. We have two spring reverberation units on hand. Mr. Franco will design a non-linear amplifier for the units after measuring the frequency response of the springs. In the future it may be interesting to explore other methods if the results promised are richer.

2) Percussion sound sources.

We have schematics for two commercially available units and a study of these can serve as a point of departure. However, as we have had an opportunity to hear many commercially available systems including these two, we intend to develop circuits that will produce a wider range of sounds which are superior. We expert to make extensive use of the filter blocks discussed under harmonic filtering on page 54

3) Voice synthesis

At the present, Mr. Noggle is attending a graduate course in speech analysis and perception. There are many articles available which will require study before a plan can emerge for the development of a system.

4) A study of new and unusual techniques of harmonic filtering, with special emphasis on non-linear filters. (e.g. filtering in the logarhythmic domain of signals whose nature is of the multiplicative rather than of the additive type)

5) Digital systems for the creation of information and information control in real time, e.g. a study of shift register sequences; a search for ways to modify information such that the musical significance of the change is congruous.

6) Analog-to-digital conversion and a system of range controls so that performing instrumentalists may join the composer to cooperatively control the instrument.

7) Experiments that result from preceeding developments.

8) Concerts

9) Experiments with color video.

10) Music for the OUT-OF-DOORS.

WHAT IS REAL TIME?

Those two four letter words have been used in this proposed report ? times.

Does real time only exist when you think of it? Have you, who have skimmed through, thought of a better way to say it? Are you aware that the process that allows a real musical time to happen is a real musical? Where's the trance? Can you sing and dance? Where's the reflex? Is Wagner's idea to put all the melodies together at the end of the overture less of an inspiration than the melodies themselves?

The best is A HEAD.