

A PRECISION RHYTHM SYNTHESIZER

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George K. Schattschneider

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GEORGE K. SCHATTSCHNEIDER

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ABSTRACT

Rhythm is one of the most important factors that influence the structure of music. However, the precise scientific study of musical rhythm perception thresholds and the hierarchy of rhythm-influencing factors is at present still in its early stages.

One of the reasons for the lack of quantitative knowledge in this field is the fact that until recent years it would have been economically prohibitive to design and build an electronic device with sufficient precision to produce meaningful rhythm experiment results.

This thesis describes an electronic instrument which synthesizes a variable sequence of notes. The frequency, amplitude and duration of these notes can be operator adjusted to produce a wide variety of simple rhythm patterns with great precision. The discussion includes the theory, design and testing of the synthesizing circuitry.

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I INTRODUCTION

1.1 Nature of the Problem

Musical rhythm, an essential element of music, has been with mankind for thousands of years. It is surprising, therefore, to discover that there is very little quantitative knowledge of musical rhythm available.

The following paragraphs are excerpts from a paper entitled "Experiments on the Perception of Rhythm" by C. A. Haenselman of the School of Music of the University of Manitoba. The paper briefly describes the meaning of rhythm, some of the factors which affect rhythm perception, and some of the experimental work which has already been done in an effort to understand rhythm. The excerpts presented below have been edited slightly for direct applicability to the subject matter of this thesis.

"Experiments done in the area of perception of musical rhythm indicate that there are discrepancies between the objective patterns and the way in which they are perceived. Even where an objective grouping does not occur, the perceiver tends to impose a subjective grouping on a series of regularly recurring pulses. This mode of rhythmic organization and perception is classified as metric rhythm. Metric rhythm is defined as the subjective or objective grouping of regularly recurring events such that an accented event is perceived with one or more unaccented events.

Although it is generally recognized that changes in intensity, duration or pitch can create a perceived metric rhythm, the hierarchy of importance of these factors in perception has not been established. Concerning thresholds at which differences in intensity and duration are perceived, only that of duration has been studied. Another aspect which has not been clarified is the threshold at which sound events are perceived as being separated by silence.

In order to compare the objective sound patterns with the perceived patterns, electronic equipment is being constructed which precisely produces the duration, amplitude and pitch of sound events. With this equipment, a series of experiments can be done which will yield data concerning the perception of rhythmic grouping, the thresholds at which differences in amplitude and duration are detected, and the hierarchical importance of intensity, duration and pitch as determinants in rhythmic grouping."

If it is desired to pursue the subjects of rhythm and rhythm perception to a greater depth than what has been presented here, Appendix A contains a fairly substantial list of pertinent references which were collected by Professor Haenselman.

1.2 Project Description

The purpose of this thesis is to design, construct and test the electronic equipment (rhythm synthesizer) mentioned in section 1.1. The following paragraphs detail the specific requirements of the School of Music with respect to the necessary capabilities of the rhythm synthesizer.

The electronic circuitry which will be used by the School of Music in their rhythm perception experiments is required to generate a series of audible notes. To eliminate as many variables as possible, a "note" is defined as a time-limited block of signal of constant frequency and amplitude. Figure 1.1 illustrates what is meant by a "note".

The circuitry must be designed in such a way that with the appropriate operator manipulation of the frequency, amplitude and duration of the notes, a single stressed beat can be produced on every second, third or fourth note in the sequence. To illustrate this, Figure 1.2 shows three sequences of notes with stressed beats on every

third note. The rhythm influencing factors in the three sequences are, respectively, frequency, amplitude and duration.

All of the variables (frequency, amplitude, duration and notes per minute) should be continuously variable except for the sequence length which should be adjustable to any desired discrete whole number value.

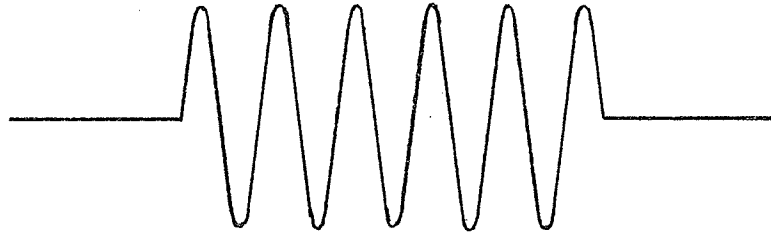
To determine the hierarchy of importance of the rhythm-influencing factors, the circuitry must also be able to generate two simultaneous rhythms. For example, the amplitude of every third note could be different from the preceding two and, at the same time, the duration of every fourth note could be different from the preceding three.

1.3 Basic Design Approach

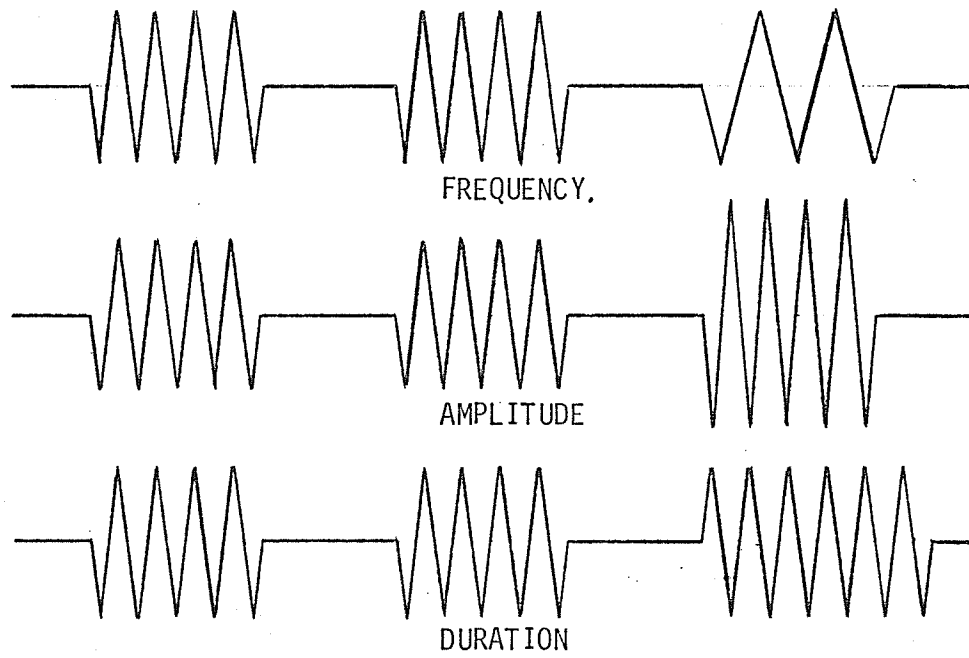
If one examines the basic requirements of the electronic circuit as given in section 1.2, it soon becomes obvious that the only way to achieve these requirements with reasonable precision and for a reasonable cost is to design an analog synthesizing section which is controlled by a digital logic section.

The minimum number of items needed by the synthesizing section which will still satisfy all of the requirements is six: two variable oscillators (frequency), one fixed and one variable amplifier (amplitude), and two variable signal gates (duration).

The minimum requirements for the digital control section are a variable clock (notes per minute control), some means of accurately controlling the starting point and the stopping point of



1.1 A NOTE



1.2 RHYTHMS

the sequence of notes (sequence control), and some means of determining the arrival of every second, third and fourth note in the sequence (rhythm control).

It was decided that the most logical approach to the design of the circuitry would be to have a main synthesizing section consisting of one variable oscillator, a fixed amplifier, and one variable duration producer. When used in conjunction with the clock and the sequence control, the main synthesizing section would produce a predictable and controllable series of identical notes. The rhythm control would switch in the second oscillator, the variable amplifier, or the second duration producer at the appropriate time to complete the desired rhythm synthesis procedure. A block diagram which illustrates the preceding text is given in Figure 1.3.

It should also be noted at this point that the prime objective in the design of the synthesizer was one of high precision and high accuracy in terms of controllability, observability and repeatability. The absolute purity of the notes in terms of harmonic distortion and signal-to-noise ratio was not considered to be of prime importance because the experiments in which the equipment will be used will be ones which require the perception of small relative differences in the note sequences.

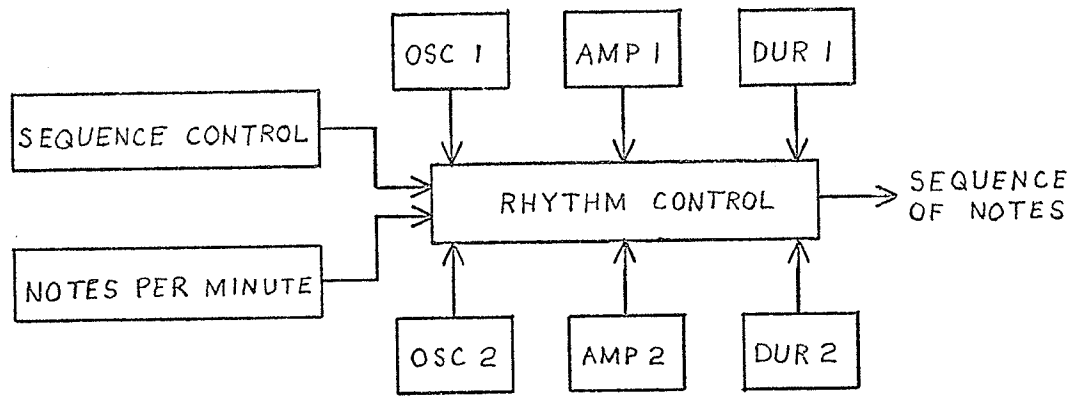
1.4 Schematic Diagram Information

The following four chapters detail the design of the electronic circuitry. The chapters are divided into a number of sections, each of which deals with a specific sub-assembly of the

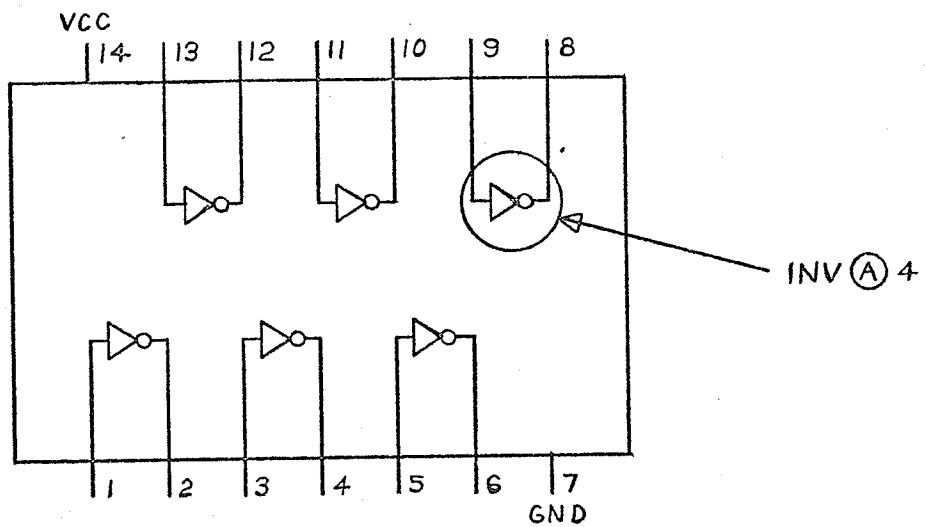
circuitry being examined in that particular chapter. In general, each section contains a schematic diagram which illustrates the material in the corresponding text. The position of the individual schematic diagrams relative to the entire synthesizer circuit is illustrated in Appendix B.

The integrated circuits (IC's) in all of the schematic diagrams are labelled. The labels begin with a two or three letter prefix which indicates the electronic function of the IC. Appendix C gives the corresponding full name and National Semiconductor part number. Following the prefix is a single encircled letter which when correlated to the diagram in Appendix D determines the exact physical location of the IC on the main circuit board.

Finally, if a single IC contains more than one separate circuit component (for example: a CD4081 contains four individual two-input AND gates), the encircled letter in the label will be followed by a number. The number corresponds to the component number defined by National Semiconductor in their data books. If undefined, the number corresponds to the position of the component in the IC data book connection diagram with number one being the component closest to pin one and then proceeding around the IC to the highest pin number. Figure 1.4 illustrates the labelling procedure.



1.3 BLOCK DIAGRAM OF SYNTHESIZER



1.4 IC LABELLING PROCEDURE

II ANALOG SYNTHESIZING CIRCUITRY

The following chapter examines the design of the analog section of the synthesizing circuitry. Specifically, it describes:

- (a) the oscillators which produce the synthesizer frequencies,
- (b) the amplifiers which are used to control the amplitudes of the signals,
- (c) the timers which are the duration producers, and
- (d) the output circuitry which provides the final signal processing before the power amplifier.

2.1 Frequency

The oscillators used to synthesize the desired frequencies are two LM566 voltage controlled oscillators (VCO). They are relatively inexpensive, are extremely stable, and have a triangle output which can easily be diode shaped to approximate a sinusoid.

The LM566 is frequency programmable by means of a control voltage, a timing resistor and a timing capacitor. It was decided that for maximum stability and tuning accuracy and a minimum parts count, the voltages and capacitors would be fixed and the resistors would be made variable. The outputs of the oscillators are followed by buffered single-pole high-pass filters which remove the DC components of the triangle waves. Finally, a simple diode shaper, which is shared by both oscillators, is connected between the buffered outputs and ground to convert the triangle signals to waveshapes that more closely approximate sinusoids. One complete oscillator (the other one is identical) is shown in Figure 2.1.

The timing resistor for a 566 is required to be between $2k\Omega$ and $20k\Omega$ for optimum performance of the oscillator. This allows for an output frequency range of approximately 10 to 1.

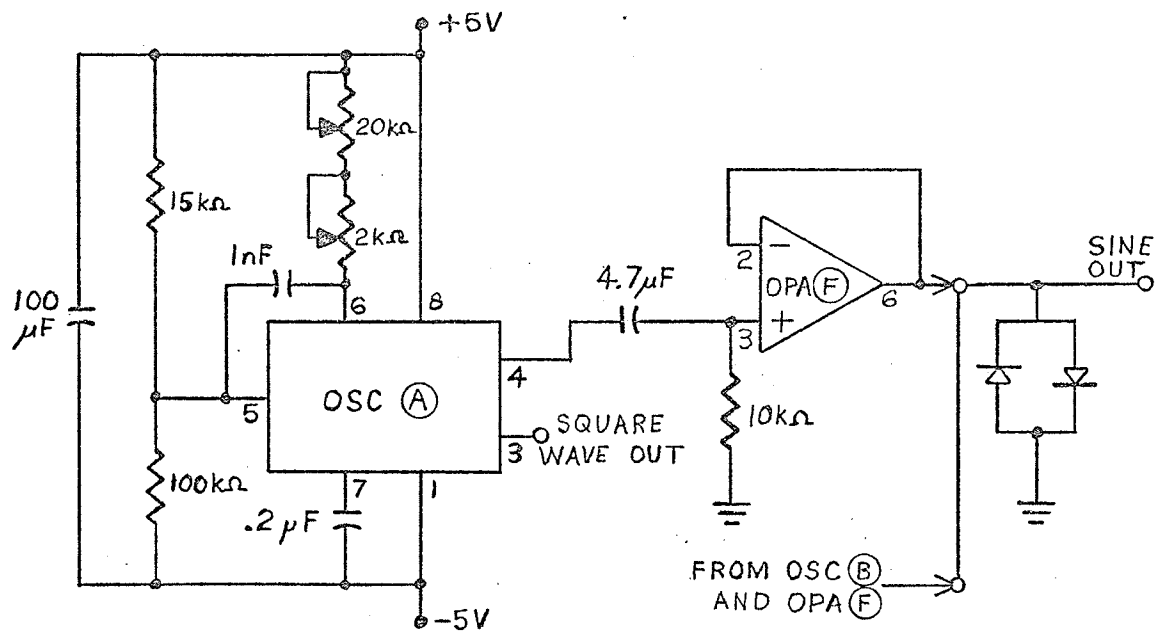
Since most of the musical fundamentals are located between 100Hz and 1kHz, it was decided to choose the timing capacitor and the control voltage so that this particular range would be achieved with resistor values of between $2k\Omega$ and $20k\Omega$.

2.2 Amplitude

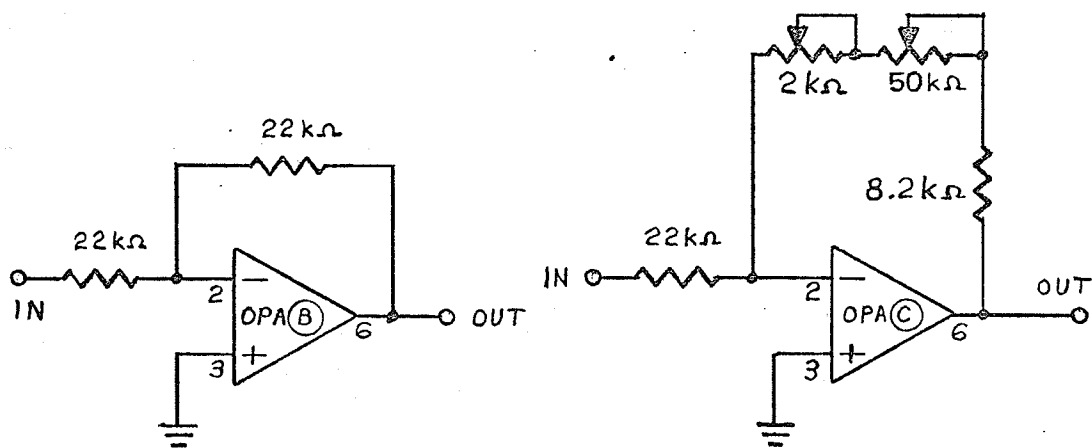
The amplifiers used to control the amplitude of the signal are two 741 op-amps employed in the inverting mode to minimize the component count. The main amplifier has a fixed gain of 1 and the secondary amplifier has an operator controlled variable gain of about .4 to 2.4. Strictly speaking, the gains of the amplifiers are negative values which means that the output is inverted around ground potential with respect to the input. However, since the synthesizer signals are always either at ground potential or are symmetric around ground potential, the minus signs are essentially of no practical consequence and are therefore omitted for the sake of clarity.

The gain of the fixed amplifier was chosen to be 1 because the 1.2 volt (peak-to-peak) outputs of the oscillators are already large enough to ensure a good signal-to-noise ratio and yet small enough to allow the variable amplifier to amplify the signal by at least two times and still be comfortably within the linear operating range of the 741 op-amp. The complete amplifiers are shown in Figure 2.2.

It could be argued that if a minimum parts count is desired, then the main amplifier should be left out completely and the secondary one should be a non-inverting amplifier. One of the reasons for not using this approach goes back to the basic design approach stated in section 1.3. That is, if the amplifiers do add noise or distortion to the signal, then the same amount must be added to the signal for both choices of amplitude. Furthermore, a basic non-inverting amplifier



2.1 OSCILLATOR



2.2 AMPLITUDE AMPLIFIERS

has a minimum gain of 1 and it is desired to produce gains of less than 1. Certainly, a divider network could be placed before the input of the op-amp but then the parts count would be increased to the point where very little or nothing is gained over the chosen method. Finally, the fixed amplifier provides signal buffering which decreases the loading effect that the succeeding stages would otherwise have on the preceding ones.

The variable amplifying ratio limits of .4 and 2.4 enable the operator to produce a secondary signal that is at least 6 decibels (db) larger or smaller than the main signal. This figure was chosen because it is well accepted that almost anyone can detect an amplitude difference of 3db. Therefore, the amplitude control gives the operator more than twice as much latitude (in terms of decibels) as he is likely to need.

2.3 Duration

The durations of the notes are controlled by two LM3905 precision timers. The timers are entirely self-contained and require only a resistor and a capacitor to set the timing period and a load resistor to limit the current through the output transistor of the IC.

The timing period of the LM3905 is equal to the product of the values of the timing resistor in ohms and the timing capacitor in farads. Furthermore, it was desired to produce durations varying from 0 to 2 seconds. This is to allow the operator to adjust the note durations from 0% (no note) to 100% of even the longest period between notes. The necessity for this will become clear in sections 3.1 and 4.5.

With the aforementioned requirements in mind, the Suggested Timing Components graph for the LM3905 was referred to and it was determined that the optimum timing components were a $.22\mu\text{f}$ capacitor and a $10\text{M}\Omega$ variable resistor. One complete duration timer (the other one is identical) is shown in Figure 2.3.

2.4 Output Circuitry

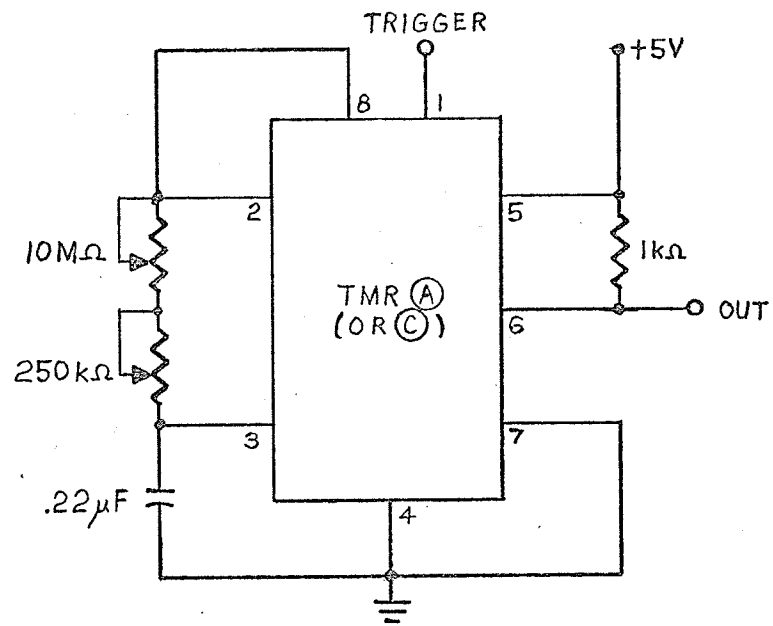
At the output of the synthesizing circuitry, just before the power amplifier, it was necessary to insert an adder and a buffered low-pass filter. The adder is an inverting variable gain amplifier which takes the outputs of the main and secondary synthesizing circuits, adds them together, and allows the operator to control the output amplitude of the synthesizer without affecting the relative amplitudes of the notes in the sequence.

To ensure that the relative amplitudes of the main and secondary signals remain constant, one of the adder input resistors is fixed and the other one is a ten-turn trimpot which is adjusted to exactly equal the value of the fixed resistor. The gain of the amplifier was chosen to be from 0 to .6 which allows the maximum use of the linear range of the power amplifier.

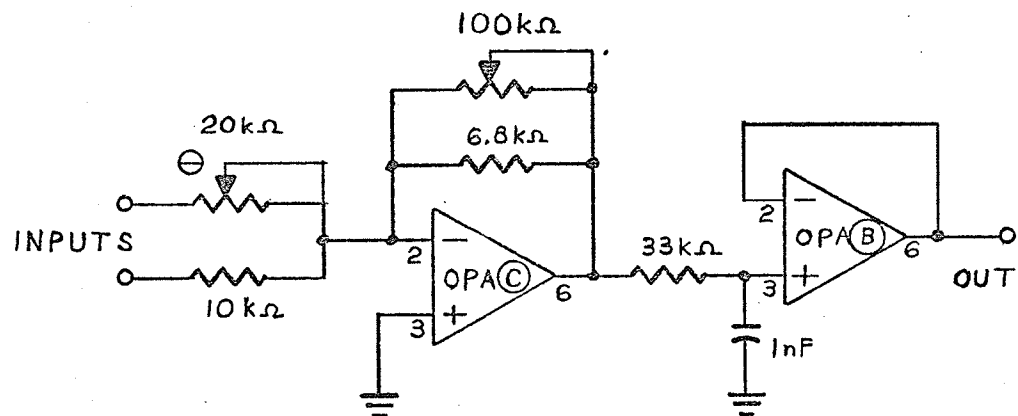
The one-pole low-pass filter removes a great deal of the undesirable high frequency noise that has crept into the signal from the digital control section. The 3db-down point of the filter is approximately 5kHz which passes all the desired fundamental frequencies (up to 1kHz) and at least the next four harmonics. The reason for

allowing some of the harmonics through is that the signals are diode shaped triangle waves and not pure sinusoids. Therefore, to maintain a constant amplitude throughout the entire chosen range of frequencies, it is necessary to allow some of the harmonics to pass through unattenuated.

The output of the buffer amplifier is connected directly to the power amplifier inputs as well as an externally accessible phone jack at the back of the synthesizer case. This is to allow the operator to use a power amplifier and speaker system different from the one provided. The entire output circuit is shown in Figure 2.4.



2.3 TIMER



2.4 MISCELLANEOUS ANALOG CIRCUITRY

III DIGITAL CONTROL CIRCUITRY

The following chapter examines the design of the digital control section of the synthesizer circuitry. Specifically, it describes:

- (a) the main system clock which regulates the digital control circuitry,
- (b) the sequence control which governs the length of the sequence of notes,
- (c) the rhythm control which provides the digital signals which permit the production of rhythms,
- (d) the duration control which ensures that spurious noise pulses at the beginnings and ends of the notes are reduced to a minimum, and
- (e) the analog switches which interface the digital control circuitry and the analog synthesizing circuitry.

3.1 Clock

The main system clock is generally the heart of any digital circuit. This one is no exception. The clock in this circuit not only regulates every aspect of the digital circuitry but also directly controls the number of notes being produced per minute.

The heart of the clock is the VCO section of an LM565 phase locked loop (PLL). This IC was used instead of the almost identical LM566 VCO because it is less expensive as well as being somewhat more temperature stable. The advantages of the LM566 VCO are a buffered triangle wave output and a maximum frequency of 1MHz as opposed to 500kHz for the LM565. Since neither of these factors are of any benefit in this particular application, the LM566 was not used.

To enable the clock frequency and the duration readout interfacing to track each other accurately (the reason for this is explained in section 4.5), it is necessary to use a fixed timing capacitor, a ten-turn trimpot for the timing resistor, and an operator adjusted voltage to control the clock frequency. The linear range of the VCO is controlled by an input potential of between approximately 2.2 and 4.85 volts which provides a frequency range of approximately 10 to 1.

To provide this control capability, a 741 op-amp is used in non-inverting amplifier form as a voltage controlled voltage source. The non-inverting input of the op-amp is connected to the midpoint of a resistor divider network at approximately 2 volts. A ten-turn trimpot is

connected between the inverting input and ground is adjusted so that the op-amp produces a maximum output of 4.85 volts.

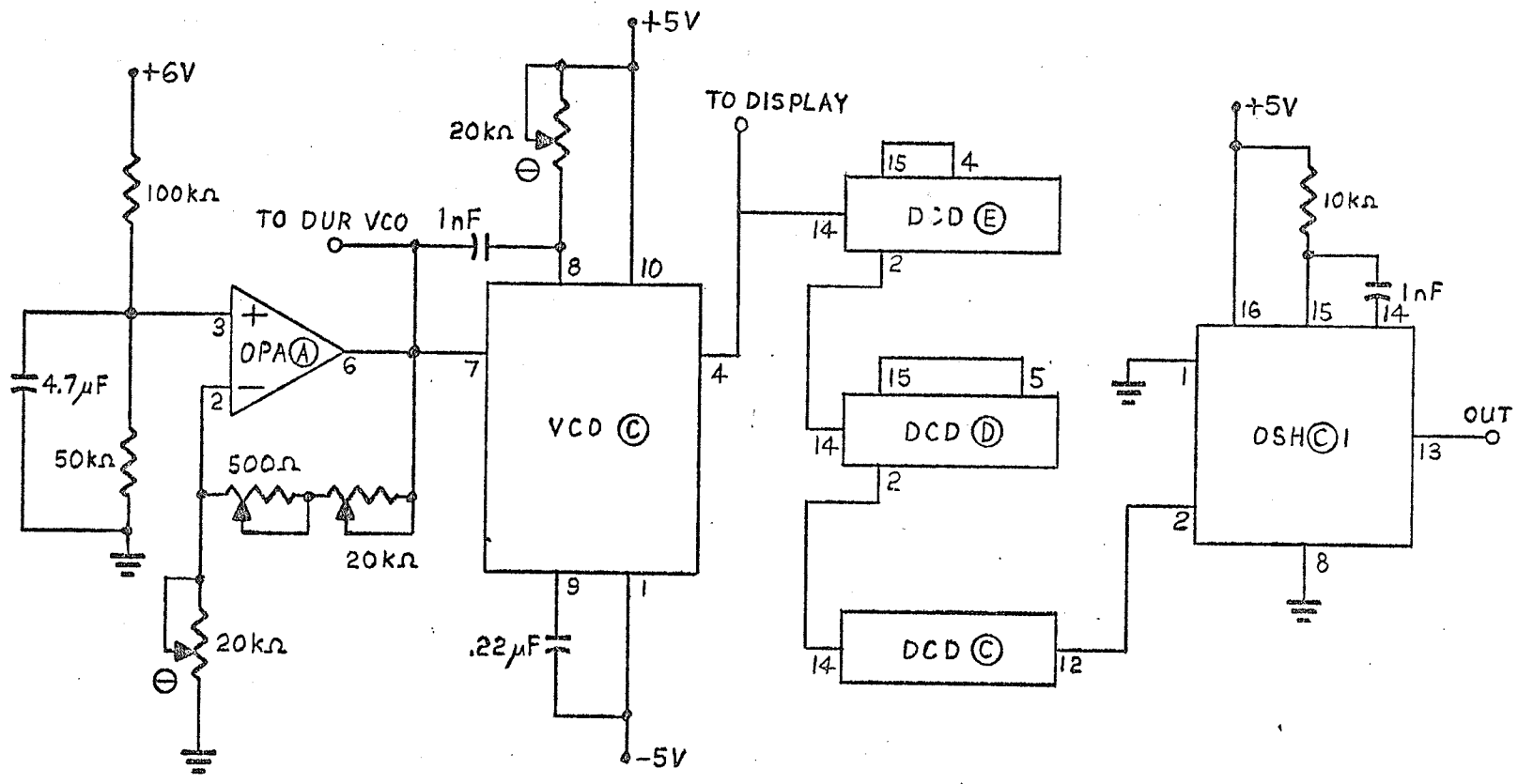
To enable simple and accurate interfacing to the digital readouts for the clock (this is explained in section 4.2) it is necessary to adjust the trimpot on the VCO to produce a frequency output range of 60Hz to 600Hz. The VCO is followed by a digital division section which divides the VCO output by 120. This provides clock frequencies of .5Hz to 5Hz or, equivalently, 30 to 300 beats per minute. This range was decided upon as being more than adequate to cover the rhythm experiment requirements.

Finally, the divider section is followed by a one shot which provides a 10 μ s pulse at the rising edge of every divider output cycle. The reason for this is explained in section 3.4. Figure 3.1 shows the entire clock circuitry.

3.2 Sequence Control

The sequence control is the section of the digital circuitry which starts the sequence of notes after a command from the operator, stops the notes after a prescribed number of them have been produced, and then resets all of the digital circuitry in preparation for the next note sequence.

When the power switch of the synthesizer is turned on, a .5 second one shot fires which immediately resets all of the digital circuitry and thereby avoids the possibility of a series of notes being produced without intentional operator triggering. When the start button is pushed, the output of a D-type flip-flop (FF), which



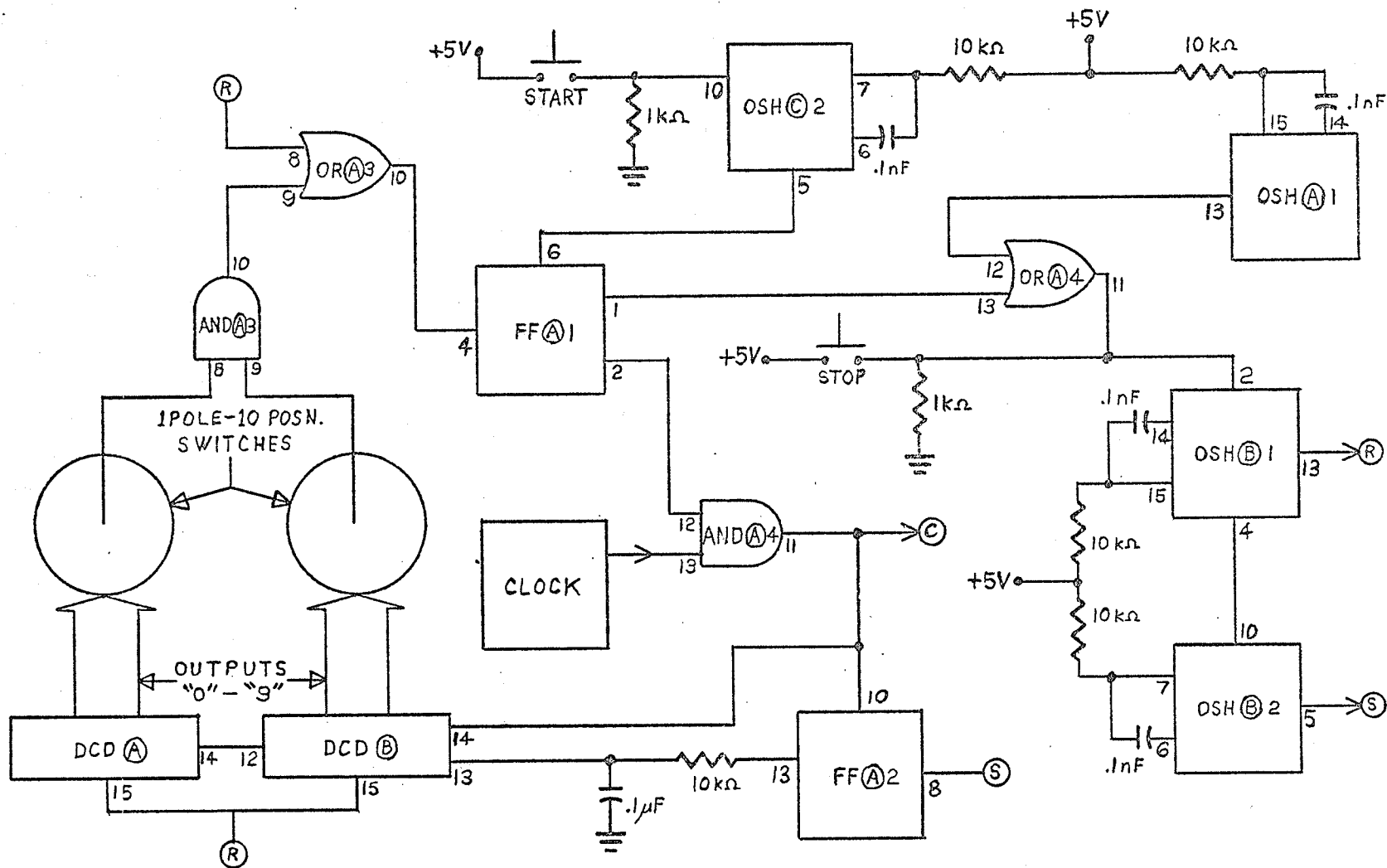
3.1 CLOCK

is being used as a bistable multivibrator, goes high and allows the clock pulses to pass through an AND gate to the rest of the circuitry.

The clock pulses go to the input of the first of two cascaded decade counters which provide a maximum sequence count capability of 99. When the outputs of the decade counters match the sequence length chosen by the operator, an AND gate forces the previously mentioned FF output to zero which removes the clock pulses from the rest of the circuitry. Finally, two oneshots provide consecutive pulses which once again reset and set the control circuitry in preparation for the next note sequence.

Since the decade counters stop the sequence immediately at the rising edge of the n th note, only $n-1$ notes will be heard. To avoid this, a FF followed by a low-pass network was inserted between the clock output and the clock enable terminal of the first decade counter. In the set position, the FF keeps the clock enable terminal high which disables the clock terminal of the decade counter. The FF output goes low at the arrival of the first operator triggered clock pulse. The lowpass network is chosen with a time constant of approximately 1ms which keeps the clock enable high during the entire first clock pulse but allows it to go low before the second pulse arrives. This timing procedure applies to the entire available range of system clock rates.

An operator controlled stop button which immediately halts the note sequence when depressed is also provided. The entire sequence control is shown in Figure 3.2.



3.2 SEQUENCE CONTROL

3.3 Rhythm Control

The rhythm control provides the capability for the digital control circuitry to switch in the second VCO, the variable amplifier, or the second duration timer at the appropriate position in the note sequence.

Since the School of Music only requires the capability to tailor every second, third, or fourth note, it was decided to solve the rhythm control problem by using groups of shift registers.

The first group of four D-type flip-flops is hooked up in a continuous loop such that the output of each one goes to the input of the next. With the shift registers arranged in this fashion, if the output of only one of them is high at any point in time, then that same one will not go high again until four clock pulses later. In the "ready" position of the synthesizer, the output of one FF is set to high and the other three are reset to low. The control signal for a beat on every fourth note is taken from the output of the FF which was initially set to high.

The second group of shift registers is set up similarly to the first group except that there are only three flip-flops. This group obviously provides the control signal for a beat on every third note. The third group, which provides a beat on every second note is merely a single FF with the output tied back to the input through an inverter.

The various terminals of the flip-flops are hooked up such that at the end of a note sequence, when the reset and set pulses are

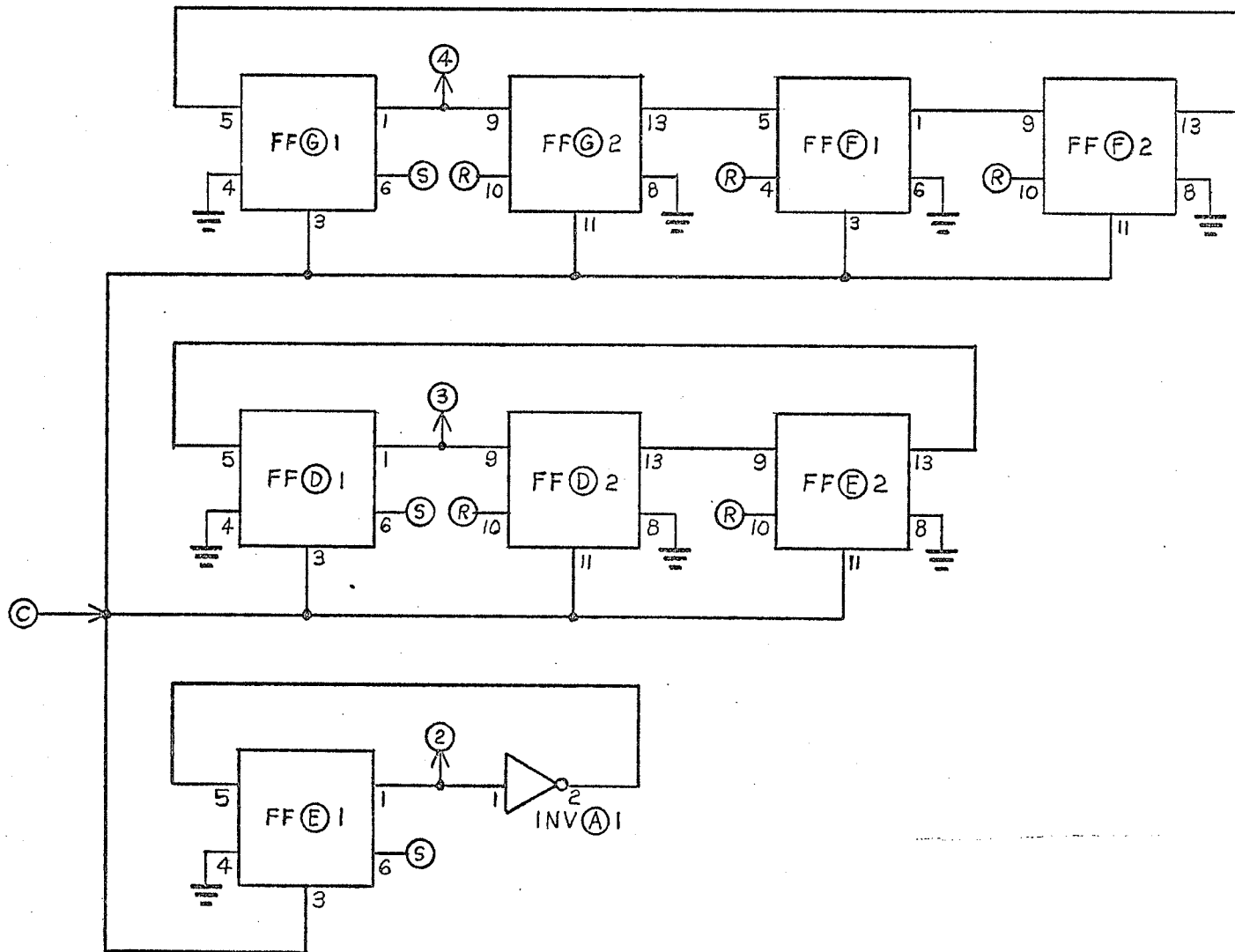
generated, all the flip-flops will be reset to low except for the designated control signal FF in each group which is set to high. The entire rhythm control circuit is shown in Figure 3.3.

3.4 Duration Control

In section 2.3, a timer or "duration producer" was described. Obviously the timer must be triggered by the system clock so that the time between the beginnings of any two consecutive notes in a sequence is the same as the time of one cycle of the desired notes per minute rate.

The problem is that the clock must also perform all of the digital control tasks. If the duration timers are triggered at exactly the same time as the flip-flops in the rhythm control are clocked, the beginnings of the notes are plagued by spurious noise pulses injected into the analog circuitry from the settling digital circuitry. To alleviate this problem it is necessary to delay the beginning of the duration pulses $10\mu\text{s}$ beyond the start of the notes per minute clock pulse. This explains the need for the one shot in Figure 2.1. The inverting output of the $10\mu\text{s}$ one shot is tied to the input of a $1\mu\text{s}$ one shot which in turn triggers the duration timers.

Another problem arises when the timers "open" and "close" and allow a certain portion of the desired oscillator frequency to pass. Looking at Figure 3.4.1 we see how, in general, the timer triggered directly by the clock might pass a certain "block" of signal to produce a note. In the example shown, rather than hearing a single pure tone, one would hear a small "tick" followed immediately by the note which would end in an even louder "tick".



3.3 RHYTHM CONTROL