ELECTRONOTES 88

NEWSLETTER OF THE MUSICAL ENGINEERING GROUP

1 PHEASANT LANE

ITHACA, NEW YORK 14850

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GROUP ANNOUNCEMENTS

CONTENTS OF EN#88

For a number of reasons, we find it useful to make this issue oversized, and will have a short issue for May. This issue has three major articles, two of which are on new controllers, and also a number of interesting shorter ideas from our readers.

NEWS AND NOTES:

Soleil Laser Productions is looking for laser performers. Performing experience on synthesizers, some technical know-how, and lots of interest are primary qualifications. Also looking for engineer interested in perfecting the next generation of laser performance hardware. Send resume or call: Bruce Rogers, Soleil, 89 Rocky Brook, Cranbury, NJ 08512 (609)-443-1061

Rivera Music Service is looking for people to work as a team in both the service of electronic music hardware and the development of custom electronic music products. The skills they require include: experience in servicing or designing both analog and digital circuitry including microprocessors, independent product design from conception to finished product, circuit design, prototyping, testing, PC layout, enclosure design, negotiating, purchasing parts for production, supervising, scheduling, production, and writing instructions. Both full-time and part-time positions are available. Experience in electronic music is desirable. Send resume to Rivera Music Services, 48 Brighton Ave. #11, Boston, MA 02134. All responses will be confidential.

The offering from Octave Electronics now includes the Kitten Synthesizer and the CAT SRM synthesizer. For more information, write Octave Electronics Inc., 32-73 Steinway St., Long Island City, NY 11103.

A new book on electronic music by J. Bermúdez, Nueva Generación de Instrumentos Musicales Electrónicos, written in Spanish, has been published by Marcomba S.A., Av. José Antonio 594, Barcelona 7, Spain. The circuitry seems up to date, and interested readers can probably have a local book store or college book store order it for them.

A new type of recording instrument, a sort of "smart chart recorder" is available in the Bascom-Turner series 8000 recorders with data processing and storage. The instrument has eight input channels, flexible disk storage, data acquisition up to 10,000 points per second for single channel acquiring 6000 points before dumping, data processing (integrate, differentiate, smoothen, averaging, etc., as well as disk entry of new programs), and readout onto a strip chart recorder. Not for the home workshop, but probably of interest to researchers who can afford it and can us it as a sort of hard copy storage scope with processing. Write Bascom-Turner, 111 Chapel St., Newton, MA 02158.

Eμ Systems has announced price changes on their modular equipment effective April 1, 1978. Their address is 3046 Scott Blvd., Santa Clara, CA 95050.

Dondisound Studios, Inc., 12 St. John St., Red Hook, NY 12571 has recently added some new equipment and offer recording studio facilities on a rental basis. Their phone is (914)-758-5167.

Computer music studies at MIT are scheduled. These include "Techniques of Computer Sound Synthesis" to be held June 12-23 or June 26-July 7, 1978, and "Workshop in Computer Music Composition" to be held July 10-28, 1978. This is a summer session program, and information on registration, tuition, facilities, and so on may be obtained from the Office of Summer Sessions, Rm. E19-356, MIT, Cambridge, MA 02139. The program is open to teachers, composers, and other interested persons, and more information can be obtained from the director, Barry Vercoe, in the music department at MIT.

A new course for musicians is being offered at the Philadelphia College of Performing Arts. Titled "Learn How to Build Electronic Modules," this six week session is designed to give musicians the necessary basic training in electronic kit building to enable them to build their own synthesis and signal processing modules. Course begins June 19 and ends July 27, 1978. For additional information drop a post-card to Lee Silvan, 708 64th Ave., Philadelphia, PA 19126.

Dennis Electronics, 2130 Metcalf St., Honolulu, HI 96822 is offering a "Control Voltage-Processor," a device that performs a variety of functions for adjusting control voltages to individual needs. It is useful for microtonal scales, portamento, transposing, pitch blending, inversion, as a control oscillator, as an envelope generator, as a slew limiter, and for many other purposes. The processor is available at a price of $159.99.

READER'S QUESTIONS:

Q: What is the difference between a "Phasor" and a "Flanger" and which is better?

A: Either of these divices is a special effects processing device, both of which

(continued on page 32)

WOODWIND-TYPE CONTROLLER FOR SYNTHESIZER II. - KEYSWITCH FINGERING SYSTEM

-by Ian Fritz

I. INTRODUCTION

In part 1 of this double article[1] an "electronic mouthpiece" wind controller was described. In this second part, a keyswitch controller with a standard 1v/octave output is described. This controller has a fingering system similar to traditional woodwind instruments, and people who can play fulte, sax, clarinet, etc. should have little difficulty learning to play it. Other woodwind-type controllers have been developed, e.g. the Lyricon instrument, whose controller can now be purchased separately, and the Bionic Sax of Greg Leslie[2]. One approach that has been used for designing keyswitch systems is to design logic circuitry to decode switch closures according to a given desired fingering system. This approach may be like balancing your checkbook by using binary arithmetic - you can do it if you're in love with digita techniques, but there is an easier way.

II. BASICS

To illustrate the fundamental design idea, a system that produces a fingering system similar to flutophone will be given. The flutophone is a toy instrument, often taught as an introduction to woodwinds, that has a one octave range. A descending C scale is produced by putting down one finger at a time on the instrument. This system is the basis for fingering of woodwind instruments. The controller I am presently using is a modification of this system, and it will be discussed in Sections III and IV.

The circuit for the flutophone controller is given in Fig. 1a and consists of an op-amp summer (with resistors as indicated), a set of seven switches, and a reference voltage V_R. It is assumed that the reader understands how an op-amp summer works.

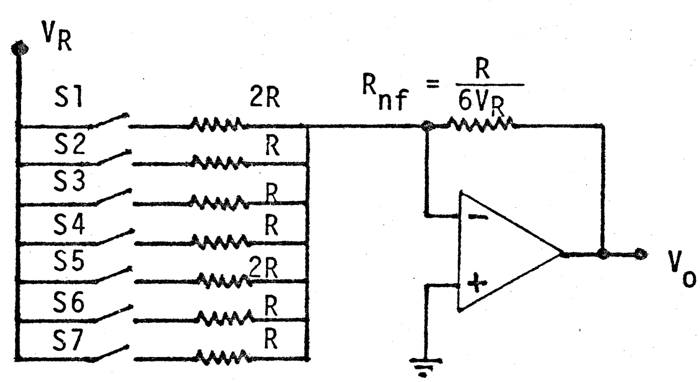


Fig. 1a Circuit for "Flutophone" Fingering

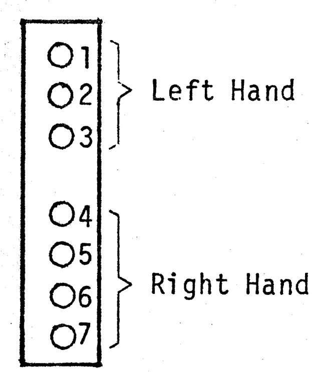


Fig. 1b Arrangement of Keyswitches

Suppose, to begin, that all switches are open. The output voltage is then $V_o = 0$. Suppose also that V_o controls a VCO which is tuned to C when $V_o = 0$. If S_1 is closed, V_o changes to $V_0 = -1/12$ volt, as can easily be verified, so that the VCO pitch changes down a semitone to B. Now if S_2 is closed, with S_1 still held closed, V_o must decrease an additional 1/6 volt to $V_o = -3/12$ v. This is because the resistor connected to S_2 is half the value of that connected to S_1. So the VCO pitch goes down to A. Now, closing S_3, with S_1 and S_2 still held closed, lowers V_o another 1/6 volt so the VCO goes to G. Continuation of the process closing one switch at a time produces a descending C scale, and if the switches are arranged as shown in Fig. 1b, that scale is fingered exactly as on a flutophone.

Notice from Fig. 1a that closing (or opening) a given switch always lowers (or raises) the pitch by the same interval, independent of how many or which of the other switches are closed. Also notice that any combination of switch closures produces some note of the tempered 12 tone chromatic scale. There are two consequences of these facts. First, there are several fingerings for most notes. Second, it is possible to play a complete 1 octave chromatic scale. The fingering chart of Fig. 2 is a set of fingerings for the chromatic scale, but is not complete; for example there are six ways to finger A#.

1	●	○	●	○	●	●	●	●	○	●	○	●	○
2	●	●	●	●	●	●	●	●	●	●	●	○	○
3	●	●	●	●	●	●	●	●	●	○	○	○	○
4	●	●	●	●	●	●	○	○	○	○	○	○	○
5	●	●	●	●	●	○	●	○	○	○	○	○	○
6	●	●	●	●	○	○	○	○	○	○	○	○	○
7	●	●	○	○	○	○	○	○	○	○	○	○	○
	C	C#	D	D#	E	F	F#	G	G#	A	A#	B	C

Fig. 2 Fingering Chart (Partial) for "flutophone" controller

● = Closed Switch ○ = Open Switch

III. ACTUAL SYSTEM

The keyswitch controller's circuit diagram is shown in Fig. 3a. As in the previous section, the basic circuit is an op-amp summer. The input resistors have weights of 2R, R, and R/6 corresponding to pitch changes of a semitone, a whole tone, and an octave, respectively. The pair of octave keys S_{11} and S_{12} are normally closed switches and either one or both can be opened with the thumb of the left hand. These switches allow for a range of over three octaves. To avoid the many "cross-fingerings" that would be necessary to play on the controller of Fig. 1, three other switches, S_8, S_9, and S_{10} have been added. S_8 and S_9 are normally closed, so that pushing on either switch raises the pitch (rather than lowering as for the case of the normally open switches) by a semitone. S_{10} allows low B to be played: it is easiest to play this kind of controller if the registers have some overlap; in fact it would be useful to also have a B^b key. This could easily be added. Fig. 3b is an illustration of how the keys are arranged. S_{11} and S_{12} are underneath the instrument, below S_1.

The device is set up so that the output voltage is always positive; this is accomplished by holding the + input of the op-amp at 3-1/3 v. A good, low input bias op-amp should be used, and a 3140 is indicated in the diagram. [My unit uses a 3130, but there is some problem of linearity with the device for input voltages less than about 0.2 v, so some of the resistance values need to be changed to move the output range up slightly if a 3130 is used.] For proper operation of the controller, careful matching of the input resistors is necessary. Since all but five or these resistors appear in combinations it is possible to combine slightly high and low values to get a resultant that is the correct value. I was able to easily select sets of resistors for two units from a batch of 100 1% resistors (R = 20k), with input resistors matched to $\leq$ 0.05%. There are no noticable differences in pitch for different fingerings of the same note.

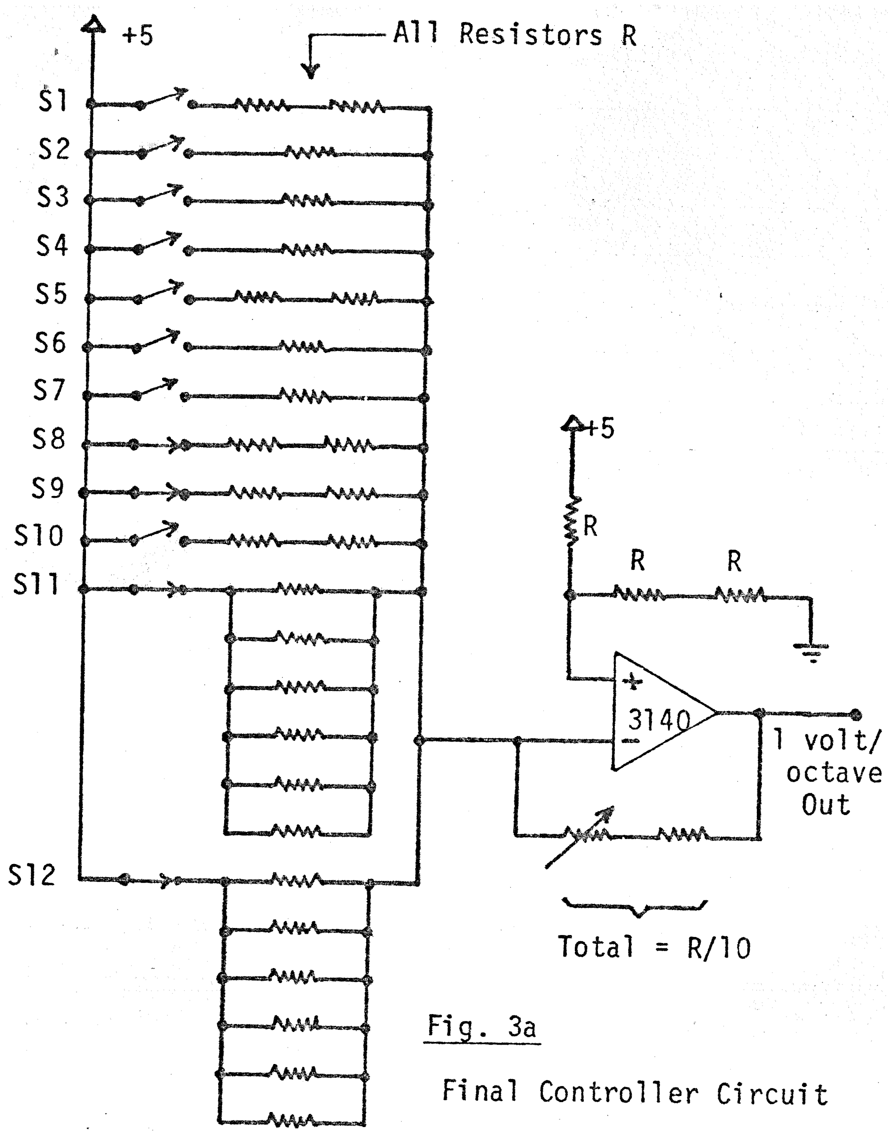


Fig. 3a

Final Controller Circuit

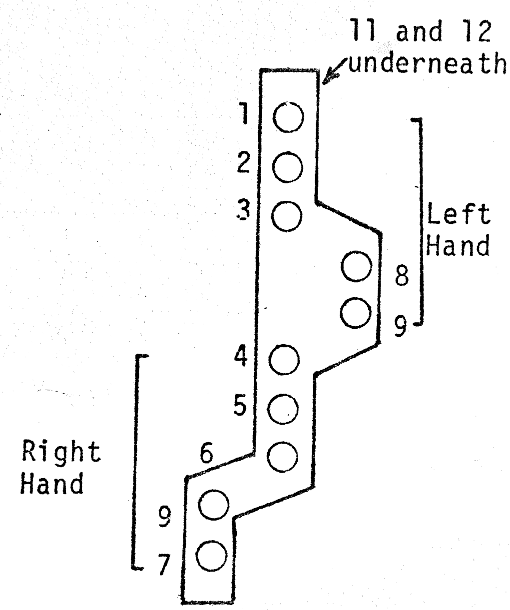


Fig. 3b

Arrangement of Switches

As with the flutophone controller, the final design allows several fingerings for most notes. At first glance, this complexity might seem to make the controller difficult to learn to play. This is not true, however, as it is not necessary to learn and practice all the fingerings for all the notes. In Fig. 4, a fingering chart is given that has only one or two fingerings for each note. With this set of fingerings you can get around the instrument well enough for most playing situations. For trills it works out best to learn some special fingerings, but this is also the case for traditional instruments. In fact, trills are quite easy: for example, you can do any half note trill by using key 1.

The use of a large number of "alternative" fingerings can be looked upon as development of a kind of advanced technique that you can work on or not as you choose. I have found many instances where a tricky technical passage can be gotten around by finding a comfortable set of "alternative" fingerings. Such a passage usually has to be practiced until the fingering pattern is memorized, but the procedure is not as difficult or unusual as it might appear. For instance, string players, who generally have a choice of several fingerings, use a similar approach to playing. As an extreme example of a situation where it is easy to learn alternate fingerings, consider the playing of a C# major scale. One way would be to use fingerings from Fig. 4. The "alternative fingering" approach, which is actually much easier, is to play a C scale (flutophone style) except with key 8 held down - key 8 effectively transposes the scale from C to C#. These advanced fingering techniques are easier to understand by actually experimenting with them than reading about them. They give the controller a nice versitality and flexibility not available on traditional woodwinds or on controllers using digital decoding techniques.

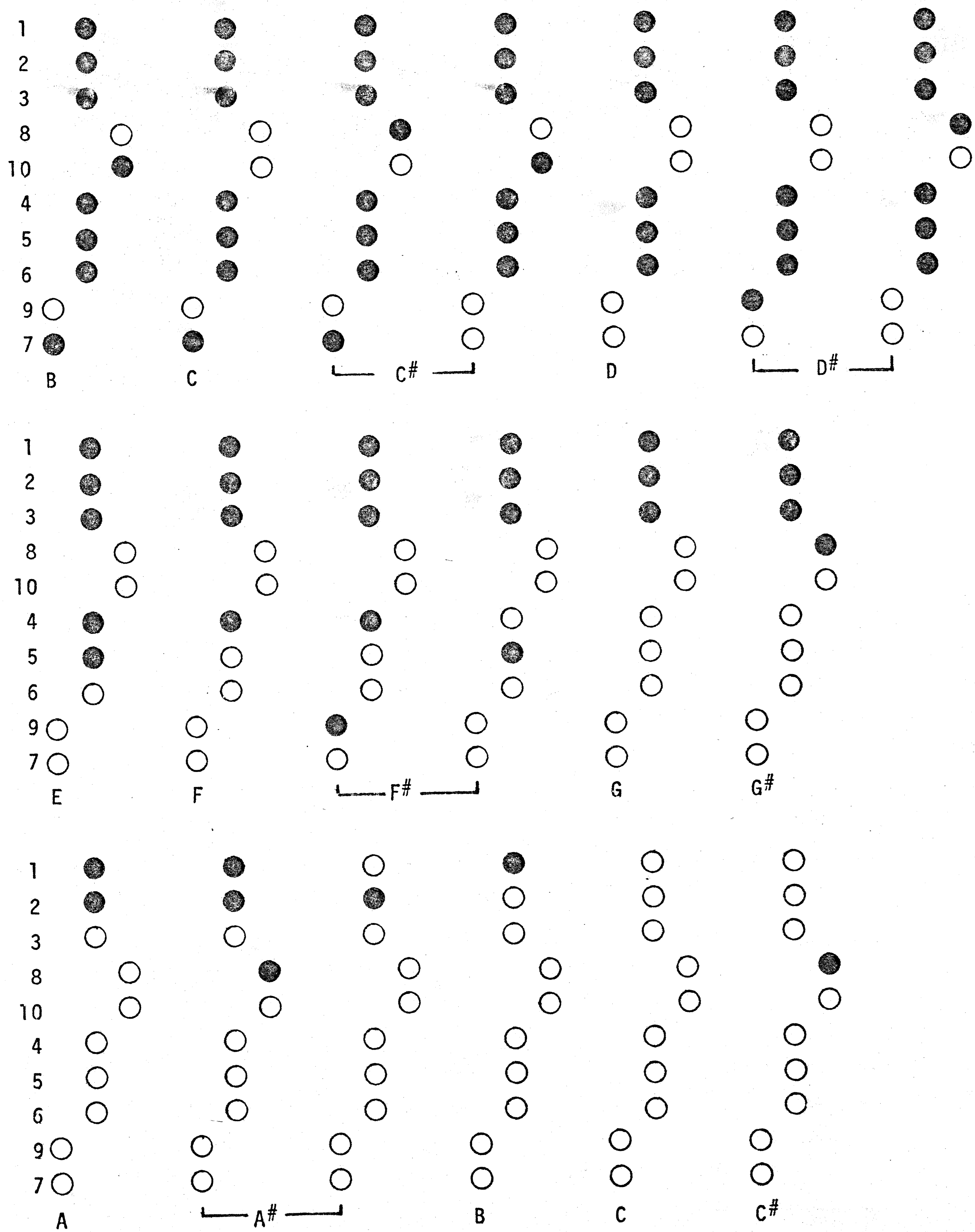


Fig. 4 Fingering Chart (partial) for One Octave

IV. DELAY CIRCUITRY

A problem arises in playing the keyswitch system, having to do with the (essentially) instantaneous nature of the switch closings. Suppose you want to slur from note A to note B and this involves changing more than one switch. Unless your fingers are exactly coordinated you will not be able to click all the switches at exactly the same time, and extra, intermediate notes will result. This is also a problem on traditional woodwinds; however the problem seems to be much more severe on the keyswitch system. The reason for this is not obvious, but the following plausable mechanism might be a cause. On an acoustic woodwind, closure of the holes is not instantaneous, as it takes a finite time to move the finger or the pad from the raised position to the position where the hole is stopped. Thus there is probably some "transition region" in time where the tone is somehow disturbed. Changing notes also requires a finite time for the new vibrational pattern of the air column to be established. This transition time may be large enough that exact synchronization of the various fingers is not required. With the keyswitch system the voltage changes are (except for switch contact bounce) instantaneous, so there is no grace period in which to get all your fingers changed, and extra notes easily result. To alleviate this coordination problem, the delay circuit of Fig. 5 was developed.

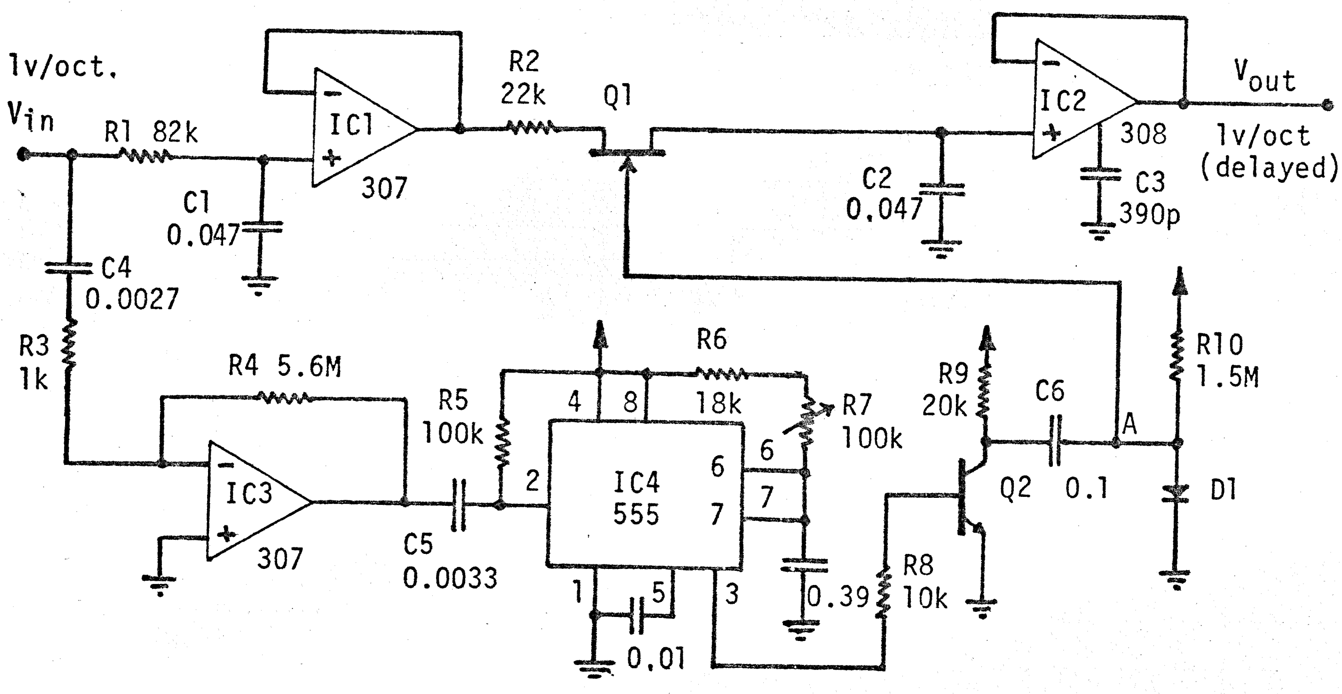


Fig. 5 Delay Circuit

Operation of the dalay circuit is as follows. The 1v/octave control voltage is fed through the lag network R1C1 and subsequently buffered by IC1. The output of IC1 then glides from note to note with a time constant of ∿4ms. Every time a new note is produced, the high-gain differentiator built around IC3 triggers the one-shot multivibrator circuit IC4, whose output pulse width can be varied between about 5 and 50 ms. The switch Q1 is normally closed, as its gate is biased at +0.6v by R10 and D1. When IC4 fires, its pin 3 goes high, so the collector of Q2 goes from near the positive supply voltage down to a voltage near zero. The point marked A, which is connected to the gate of Q1, is therefore "pushed" down to a large negative voltage, and since C6 can only charge slowly through R10, point A stays near that negative voltage until IC4 turns off, at which time it returns to 0.6 v. The result of the just described events is that as soon as you change fingerings from note A to note B, Q1 opens and the voltage stored on C2 corresponds very nearly to note A (because of the lag produced by R1C1 and R2C2). At the end of the delay period, Q1 closes and C2 charges quickly through R2 to the voltage corresponding to note B. If intermediate notes are fingered between notes A and B, and if their duration is less than the delay period, they do not appear at the output. Thus, every time a new note is fingered, the previous note is held for a delay period, which is a grace period for getting all the switches for the new note closed. The timing sequence is illustrated in Fig. 6. It is necessary that the delay period be quite short, at least less than the time between notes of a rapid passage. Otherwise notes will be lost. A delay time of 15-30 ms seems about right.

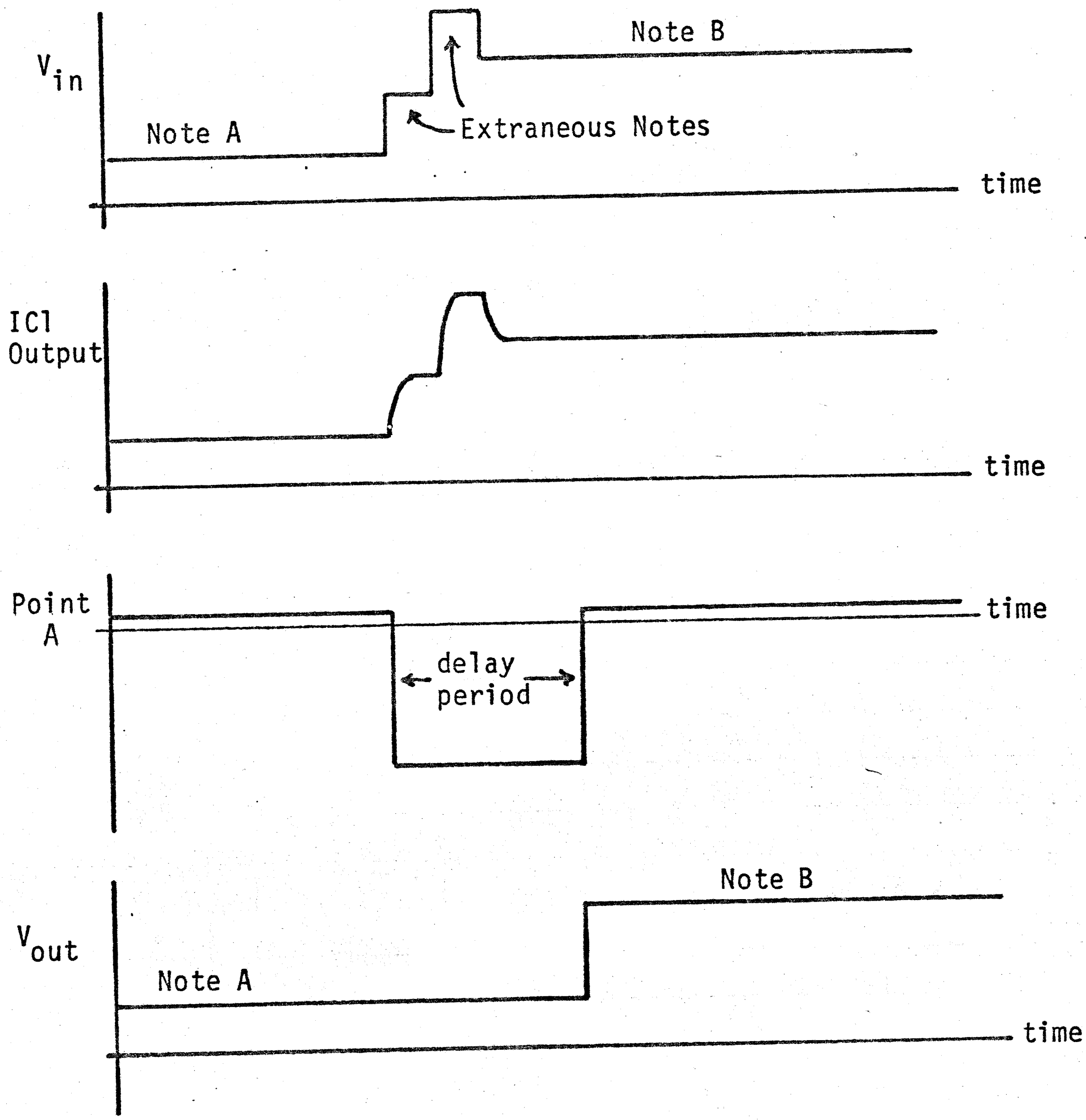


Fig. 6 Timing of Delay Circuit

V. CONSTRUCTION

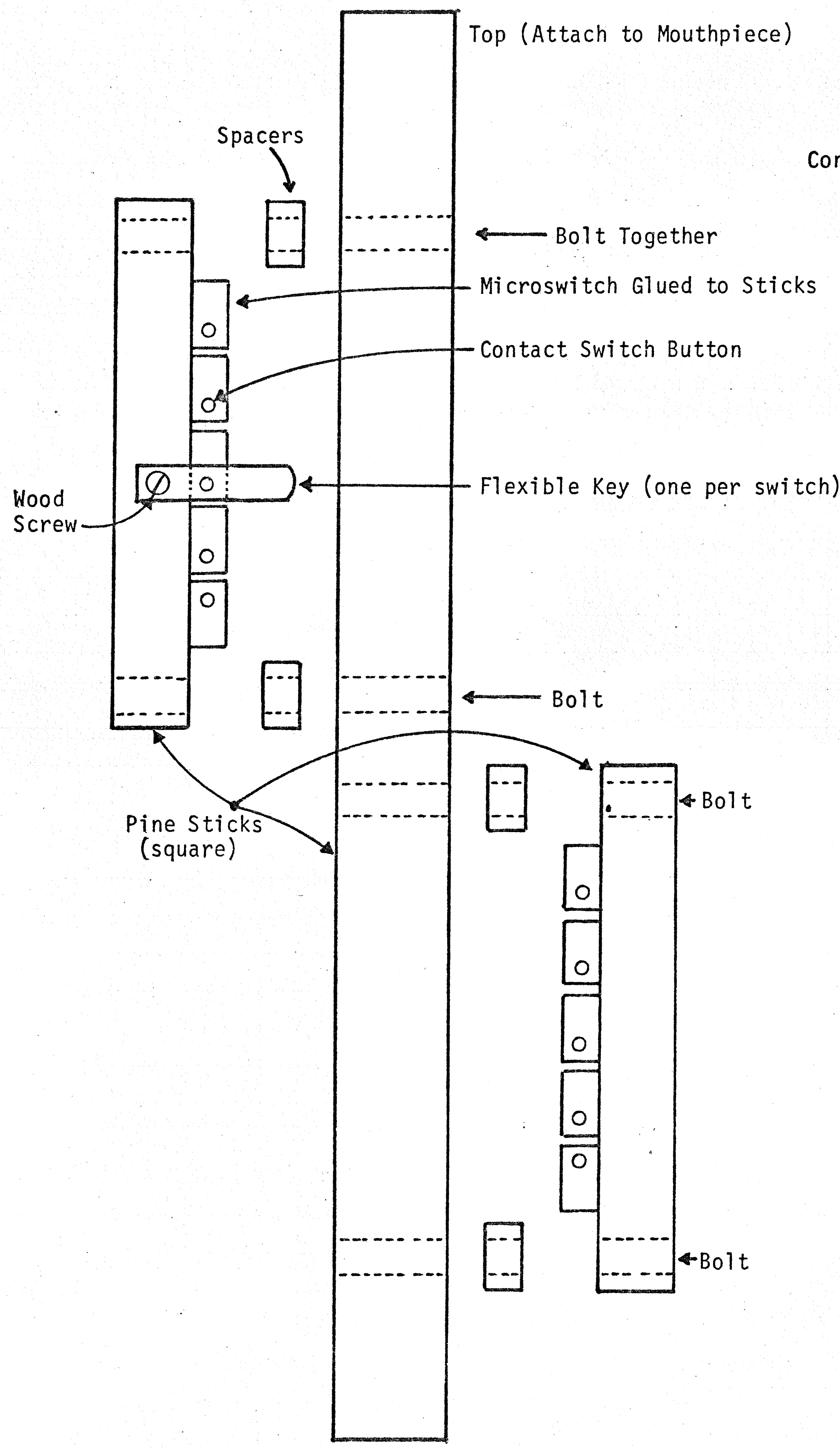
Various construction techniques are possible, and the best mechanical layout is a matter of personal taste. Leslie[2] suggests splitting a plastic tube, mounting computer switches on one half, and gluing the tube back together. My system uses microswitches, with flexible, cantilevered keys to actuate them. It takes a bit of experimenting to get the keys so that they have a good feel and a fast action. Weight and balance of the overall unit must also be adjusted to give the controller a comfortable feel. This turns out to be harder than one might think. My unit isn't optimized mechanically, so I won't give detailed construction plans, but the basic technique is shown in Fig. 7.

VI. CONCLUSION

This section will be used to present some miscellaneous ideas and to tie up some

Fig. 7

Construction



loose ends.

People interested in building new controllers should look at the article by Greg Leslie[2] in Polyphony. Leslie's technique for breath control is particularly simple. The player blows across or into a small microphone element. The noise output from the microphone is fed into a standard envelope follower to produce a voltage proportional to blowing force.

Other transduction schemes for breath control are possible. A variation of Leslie's idea would be to remove the element from an inexpensive ceramic phonograph cartridge and blow against it. A row of these could be mounted and used for the pressure transducers in Jeff Nobel's proposed Oral Joystick[3]. National Semiconductor has a new publication, the Transducer Handbook, which discusses their line of IC pressure transducers. These new devices probably warrant investigation for electronic music control applications.

The most difficult part of developing the woodwind controller has been the mechanical construction of the keyswitch section. The unit is reasonably comfortable to use now, and I am developing a fair amount of technique with it, but getting the switch lever positions, tensions, and motions optimized has been a long frustrating project. Coordination is still difficult and it is not clear whether this is due to variations of the actions of the different keys or whether the device just needs to be practiced on more. It would be interesting to try touch-sensitive pads instead of movable keys.

As a final note it should be emphasized that the controller described in this article is certainly not considered to be the ultimate synthesizer controller. The results of this project have been gratifying in that the simple, inexpensive device developed provides a great improvement in the amount of musical expression possible using a monophonic synthesizer. Probably many other simple, effective controllers could be devised and easily constructed. Perhaps other Electronotes readers have ideas for possible new controllers or actual working devices that they would like to tell us about.

REFERENCES:

1. Part 1 of this article appeared in EN#86, 14 (1978)
2. G. Leslie, Polyphony Nov. 1977, pg. 21, and Feb. 1978 pp 15, 20.
3. J. Noble, EN#84, 15 (1977).

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LITERATURE REVIEW: COMMENTS ON ERNST'S "THE EVOLUTION OF ELECTRONIC MUSIC"

-by R. Bruce Elder

A commonplace in discussions of electronic music has it that the development of electronic methods of sound generation and processing has resulted in the expansion of the range of sonic materials available to the contemporary composer. Unfortunately, even though what this statement asserts is substantially true, the statement itself is often used in such a way as to obscure more than it enlightens. For what it seems to imply is that technological developments afforded certain heretofore unavailable sonic materials to the composer who then was faced with the task of making creative use of these materials. Thus it seems to suggest that electronic music is a kind of artistic Artemis which sprang fully grown from the head of a technological Zeus. This claim not only attributes altogether too much importance to the role of technological developments in the shaping of this music, but also acts to obscure the historical origins of electronic music, thus clouding its aesthetic background. In fact, the major musical developments of the three centuries preceeding our own lead directly to the formation of that aesthetic upon which electronic music is based. It is therefore only by understanding the significance of these developments that one can really come to terms with this art form.

The structures of traditional composition were to no small degree founded upon traditional harmonic and melodic patterns which were in turn based upon certain basic modes and scales. The modes of early music provided the basis for a musical system in which all the pitches used within a musical composition were related in simple fashion to a tonal centre. The tonal centre constituted a strong centre of attraction for the other pitches. Tension could be created through arresting the movement toward that centre or resolved by allowing this natural movement to occur. With the development of diatonic scales, pitches which had less firm relations to the tonal center and which were, in consequence, less powerfully attracted toward it, began to be used. The development of equally-tempered tunings carried this a great deal further for in this system the tuning relationship of any pitch to the tonic is not at all simple or absolute; consequently the tonic's force of attraction on other pitches is not as great The increased use of chromatic pitches, i.e., pitches lying outside the diatonic scale upon which the composition is based, the use of higher and higher intervals of the chord in which the upper intervals are related to the root in an increasingly complex fashion, the extensive use of modulation involving pitches and melodic patterns not related to the tonic, and finally, the development of pantonality, atonality and serialism, led in the end to the breakdown of the tonal system.

These developments were paralleled by changes in the structure of melody itself. The voice-based melodic structures in early music, with their step-wise pitch progressions were transformed into structures whose progressions were based first on the intervals of triad, then on the chromatic scale, and finally, on the atonal tone-row. Pitch progressions became increasingly broad; thus they became decreasingly imitative of the natural movements of the human voice, and increasingly instrumental in character. This developed into the tendency to have the melodic structures reflect the unique features of the particular instruments for which they are written. A parallel tendency was to have a work's compositional organization reflect the unique timberal features of the instruments for which it was scored. Rhythmic structure similarly evolved away from those steady insistent rhythms which were tied to extra-musical functions such as the dance toward a greater flexibility in the placement of accents and greater complexity in their groupings.

All these changes had far-reaching effects on musical structure. From the classical period onward, an increasingly strongly codified system had developed according to which variety and tension could be created by modulating entire compositional blocks out of the tonic and resolved by re-modulating into the tonic. Such common forms as the sonata are, of course, based on this simple pattern. Such compositional edifices, could be created only as long as music's harmonic substructure could be relied upon to hold firm. With the breakdown of this substructure, new compositional structures had to be developed. In the process of searching for new compositional forms, composers conceived of the possibility that the colour of sound, as much as its pitch or the harmonic grid into which it fitted could act as the primary structural determanant of a composition. Thus, by the twenties, Schoenberg could propose the possibility of compositions whose structure was based upon timberal variations (Harmonielehre, 1922).*

*[Editor's Note: To understand what a natural extension of pitch-based structures this was for Schoenberg, it is interesting to know that he wrote in Harmonielehre: "I cannot accept the distinction between tone color and pitch as it is generally stated. I find that tone makes itself noticed through color, one dimension of which is pitch. Tone color is therefore the large area, of which pitch is one division."]

All of this, moreover, led to a shattering of the homogeneity of texture which had characterized music until a century and a half ago. The simple scales and modes of early music were essentially a method of ranking notes which were in any case already related naturally into a hierarchy. Thus the dominant note in any scale was accorded a greater strength than, say, the leading tone. This kind of hierarchilization provided the basis for establishing a kind of unity based on melodic movement, for the less strong tones seemed to gravitate toward the stronger. And this flow itself determined other sorts of unity and homogeneity, for example, textural, rhythmic and harmonic. In fact the harmonic substructure of music itself had much the same effect; for its laws not only guaranteed, if only by the dominance of theory over sound itself, a kind of harmonic homogeneity, for through its figures of choradal resolution it operated to assure unbroken "natural" flow.

The breakdown of the harmonic system and the resultant widening of intervals in pitch progressions led to a shattering of these unities. Perhaps the most important consequences of these is that individual pitches were no longer arranged in a hierarchy which determined that some notes served only to lead toward others; each note could now be considered as a kind of distinct entity. This not only led to the extremely fragmentary textures of pointillistic music but also led to an interest in the sonic characteristics of the individual notes, e.g. their envelope, their harmonic composition and so on.

The importance of the development of sound synthesis equipment can best be understood against this background of growing engagement with the materiality of the artwork. Synthesis equipment provided composers with tools to control precisely the various characteristics of their sonic materials. More importantly, such equipment allowed composers to vary any characteristic (e.g. timbre) independently of any other characteristic (e.g. pitch) something which cannot be done with orthodox musical instruments. As a result, synthesizers allow composers to create compositional structures in which any or all of these characteristics are controlled by independent (though usually interacting) systems.

Even though the foregoing remarks are for the most part commonplaces of advanced musical criticism, a history of electronic music which understands the place of electronic music within the framework I have just sketched has still to be written. David Ernst's new book, The Evolution of Electronic Music (Shirmer Books, 1977), does little to meet this need. The most cursory glance at almost any page of Ernst's book reveals what he conceives the importance of electronic sound synthesis methods to be. He repeatedly states that composers use electronically generated or processed sounds to increase timberal variety in their work.

At best, this is a half-truth. It simply asserts that composers are searching for greater timberal variety without identifying the changes in the substructure of a musical composition which have brought composers to engage in this quest. Ernst never discusses how the search for timberal variety is an outgrowth of historical forces which led to rupturing of the traditional compositional unity nor does he ever relate this "quest" to the materialism inquiry which preoccupies modernist composition. Furthermore, the claim suggests that composers are searching for added types of variation within the traditional framework while in fact this framework itself has been rejected by many composers.

Considering the importance which Ernst attributes to the timberal aspects of recent music, it is surprising that no discussion into the structural possibility of timbre is ever presented and that no individual composition is ever shown to have a structure which depends on timbre. Needless to say, such compositions exist. Perhaps this is due in part to the fact that no comprehensive theoretical understanding of timbre seems to underlie the work.

Ernst's repeated claim that timberal variety is employed in a certain composition in order to avoid literal repetition also indicates a real lack of understanding of a common strategy of modernist composition and modernist art in general. Many compositions are based on structures designed to hold the majority of the variables constant, while a single variable is subjected to gradual, progressive alteration. Thus a materialist based inquiry into the effect of altering a certain sonic characteristic can be conducted and in complete detail and with precision. Ernst's discussion of Stockhausen's Telemusik (pp 62-64) shows a similar lack. His treatment of that composition consists primarily of pointing out that the procedures which Stockhausen employed for producing that work enabled the composer to generate a continum of timberal variables. Although he acknowledges that the technique of intermodulation used in that work consists in causing two variables to interact with one another, he fails to discuss the materialist problemtic posed by this procedure.

Because he fails to come to terms with these profound changes in musical structure Ernst cannot distinguish between those more adventuresome compositions which challenge the very fundamental principles of composition and those pieces in which synthesis equipment is used for such less radical purposes as the creation of timberal variety. Thus Ernst can lump together in a single chapter the electronic compositions of Karlheinz Stockhausen with those of Keith Emerson and can even pass from a discussion of one to a discussion of the other without so much as a shift of tone. Most probably it is this lack of ability to grasp the profoundly particular character of "serious" advanced composition that leads Ernst to advance the rather silly theory that currently a rapproachment between various types of composition is being effected. The truth of the matter is the gulf between popular and "serious" composition has never been wider.

This book is beset with difficulties right from the very beginning. In the introductory section, Ernst presents a rather cursory summary of the prehistory of electronic music which focuses primarily on two topics: the growth of understanding of the physics of sound and the development of musica futuristo. The first topic Ernst presents as historical background to the technological developments in the field of audio. The form of presentation adopted reveals a radically limited understanding of the musical importance of these studies. The studies of acoustics (and more particularly psycho-acoustics, a field of study which Ernst neglects entirely) have helped to deepen the composers' understanding of the nature of sonic structures. It is, in fact, this understanding upon which the formal structures of modernist composition depend. Ernst entirely neglects this aspect of the field, preferring it seems, technologically-oriented discussion to aesthetics.

No less revealing is the importance which Ernst attributes to musica futuristo. Admittedly, the futurists were among the first (though certainly not the first) to use industrially generated sounds in their works. They were also among the first to conceive of a composition as a constructed sonic object. These facts alone undoubtedly assures them of a place in the pre-history of electronic music. This notwithstanding, the futurists' desire to burden sound with extra-musical import has placed them quite outside the aesthetic of the mainstream of twentieth century music. It is therefore curious that Ernst should give them such a large role in his introduction. It seems that Ernst is intrigued primarily by the radical nature of sonic sources they employ. This surely represents a kind of technological fetishism.

Even more significant than what he includes in his presentation of electronic music pre-history is what he excludes. Nowhere does he discuss changes in musical language and musical structure that preceeded the development of electronic music. The search for new compositional forms can largely be understood in terms of the search to discover new bases for compositional unity now that the unifying system of harmony has disintegrated. Cage's work, for example, can be considered as post-harmonic in this sense. He continually poses the question of how one can structure a composition under something other than a conventional harmonic formula. The answer he most frequently uses is that one can create a compositional structure that is analogous with structure of natural processes. Ernst's book fails to pose these questions and in this way presents a far less adequate treatment of the pre-history of electronic music than does Elliot Schwarz's now lamentably outdated Electronic Music: A Listener's Guide.

The book, too, lacks that sort of organization which an understanding of the aesthetic ideals of new music provides. It's grab bag assortment of superficial comments on an inordinately large number of compositions are filed under chapters according to the nature of their sound sources. Elevating the manner by which the sounds are generated to the level of the book's organizing principles transforms the text into a celebration of technological resources.

Had Ernst probed this organizing idea a little more deeply, he might have been led to an understanding of one of the more important structural basis of new music. A moment's reflection might have revealed that an enormous number of electronic compositions utilize sounds from sources that are radically different in nature, for example, synthesizers and human voices. As Ernst has observed, the reason for this is that sounds possess antithetical characteristics. Thus, a composer can create tension by juxtaposing sounds possessing such antithetical characteristics and resolving that tension by resolving the antithesis.

This is undoubtably a very common principle governing the structure of a goodly number of recent compositions. Wörner points out in his work, Stockhausen: The Man and His Music, that it is the basis of many of Stockhausen's compositions, both electronic and non-electronic. The bulk of Eliot Carter's work, too, is structured in this way. Ernst should therefore be expected to be able to come to terms with it. Unfortunately, the book is riddled with evidence that he failed to do so. On page 85, for example, when discussing Jacob Druckman's Animus I for trombone and tape recorded electronic sounds, he suggests that "timberal unification (of trombone and electronic sounds) is achieved by equating trombone effects with electronic modifications and sources": flutter tonguing = amplitude modulation, mute = filter, blowing through mouthpiece = noise. Even overlooking the error of categorizing flutter tonguing, amplitude modulation and use of noise sources as simple ways of achieving timberal modifications, one is surprised as the lack of rigour which characterizes this remark. Surely it is evident that the relation between trombone and electronic sounds is far more complex than Ernst describes it as being. Druckman, is in fact, using sounds with antithetical characteristics and using the devices just enumerated to resolve the opposition between these antithetical sounds. The relationship between these could, therefore, be described as a dialectical one involving moments of antithesis and snythesis. The dialectical nature of this relationship is important for it established the basis for the creation of resolution and tension. Similarly, Ernst's discussion of Stockhausen's Gesang Der Jünglinge fails to take into account that in this work there are interacting dialectical structures: speech as meaning and speech as sound are contrasted as well as speech and non-speech sounds. These oppositions are resolved by breaking the sematic units into meaningless phonomes and by integrating electronic sounds with the phonemes. Thus he creates a continuum between purely formal sounds and semantically significant speech sounds. Thus paralleling the dialectical relation between the sound sources is a dialectic based on the opposing processes of fragmentation and recombination. Ernst's flimsy analytic apparatus does not furnish him with the means to trace such a homology of structures.

Ernst's failure to resolve the fundamental aesthetic problems posed by the new music is also evidenced by the contradictory statements with which the book is riddled. One example of such a pair of contradictory assertations concerns the importance of the unique character of synthesizer music. On the one hand, Ernst criticizes the standard keyboard-like configuration of the voltage-controller on the grounds that it encourages musicians to think of the synthesizer as a glorified keyboard instrument and not as an unique instrument in its own rights (p 213). On the other hand, he approvingly quotes Patrik Moraz' statement "The ultimate goal is to make a Moog or an ARP or a string synthesizer not sound like a Moog or an ARP or a string synthesizer" (p 197).

By far, the most satisfactory chapter of the work is the final one which is devoted to a discussion of compositional techniques. Of particular interest in this section is Ernst's demonstration of a variety of techniques for obtaining a spectrum of sounds of graded difference. His favoured idea seems to be to subject a single sound source (or antithetically related sound sources) to a wide variety of processing methods which operate simultaneously and to provide a means of allowing the results to interact with one another. The idea seems a good one for it insures a certain unity will subtend that immense range of sonic characteristics which the synthesizer is capable of producing. In fact, this whole section, though lamentably short, is remarkably suggestive. Ernst's efforts might have been better spent by developing this section into a book-length treatise on electronic music production. A good text in this field is still much needed.

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Pitch Extractor For Guitar And Microphone:* -by Dale Wills

Pitch extraction is a problem that has been haunting many engineers for quite some time now. Recently, a number of methods have appeared [1, 2, 3] and here a method intended mainly for guitar will be described. First, I would like to thank Bernie Hutchins, Robert Iodice, Bill Rix, and Gregory Hockman for their help in one form or another, mostly another.

Number one, we must probe the heart of the problem itself, the excitation source, where the complications first appear. On a guitar these complications are byproducts of things done intentionally, as on many instruments (for example, a piano is intentionally mistuned). On a guitar, when the string is struck, the bridge (usually brass) and sounding board (the face of the guitar) resonate overtones into the strings (a constant feedback loop). The amount of feedback is determined by the type of guitar, type of physical composition, and type of string being used. Engineers call the results of this feedback harmonics, musicians call it tone.

*Editor's Notes: It seems appropriate here for me to say a few words about Dale Wills with regard to his role as a musician turned engineer turned writer as I feel this will be of interest, and perhaps inspirational to some readers. I first heard from Dale a year or so ago when he called about a pitch-to-voltage converter. After a brief discussion, I found that he did not know a resistor from an op-amp, so I tried to dissuade him from trying to build my pitch extractor, which is probably our most difficult project. As a rule, I try to keep people from taking on a project that is over their heads, and this is something I feel is still a good policy even though Dale has proven to be the exception to the rule. As you will see from Dale's report, he did manage to learn by doing, and in the course of a year he learned the electronics he needed, and was even able to design an entirely new system. So it can be done, but this probably is an effort we will not see often duplicated. In his writing, the reader may find nonconventional usage of some engineering terms. We have refrained from substantial editing because this does not seem to change the factual information Dale is giving, and this information is of considerable interest and is clear as written. Perhaps it is also of interest to engineers to see how musicians describe their instruments in engineering terms without the engineers jumping in and telling the musicians what they "really mean" to say.

On the neck of the guitar are the frets which are not placed perfectly, but are placed as a compromise, because each string is a different diameter. Guitar manufacturers use many different fret scales. As the strings go dead through deterioration with use, they become very hard to keep on pitch, because the harmonics do not recycle as easily, leaving just the fundamental note. For the pitch extractor, we want that dead note, even from new strings, and this is achieved using highly damped filters.

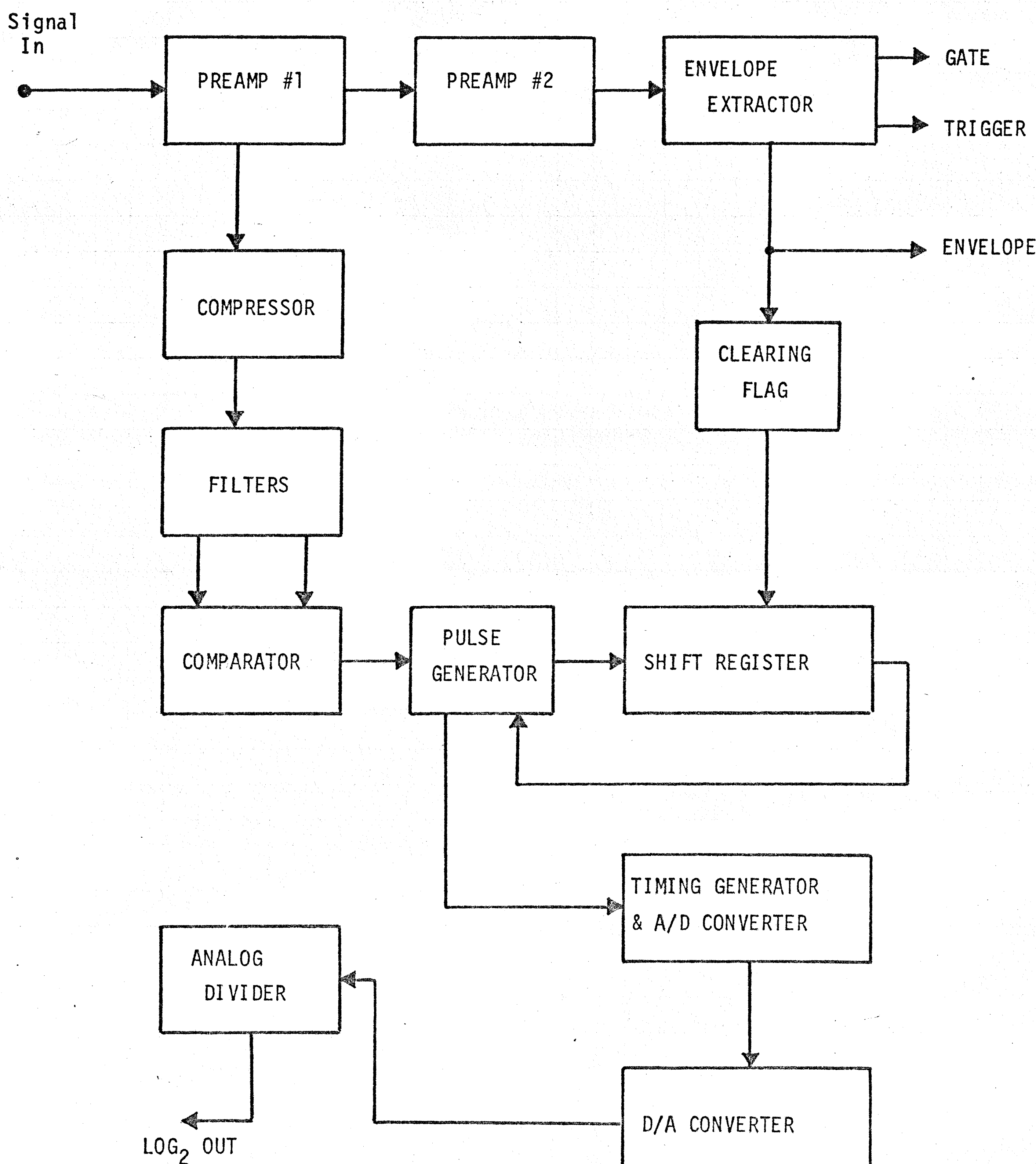
When the string is attacked by the plectrum (see Fig. 1), it goes flat and then rises to pitch as the amplitude decreases. This is because the string is streatched for a moment. The ear and the brain ignore this and put the note in pitch due to the speed with which this occurs. Also, the moment the string is struck, an enormous amount of physical energy is applied and this then decays rapidly. This causes problems for pitch extraction. A down stroke dissipates instant energy and an upstroke delays the energy dissipation. This is caused by the rotation of the plane of vibration of the string and its phasing across the magnetic pickup. This can be observed by hooking a strobe light, triggering every cycle, to the instrument. Solid body guitars decay much faster than semi-acoustics due to the dense matter, giving the fundamental much quicker. Both types have different harmonic qualities, like hitting a glass full of water and a glass that's half full.

CIRCUIT DESCRIPTION:

A block diagram of the pitch extractor is seen on page 17. The circuitry of the preamps and the compressor is shown in Fig. 12 on page 19. The preamps are the same ones used in an earlier design [1]. IC-1 drives the compressor, and IC-2, with adjustable gain, drives Bob Iodice's envelope extractor [4]. IC-3 is a negative going inverting peak detector. The high impedance resistor and capacitor network and IC-4 buffer and prevent a ramp. The harder the attack, the more negative IC-4 goes, and with no signal, IC-4's output will standby at -2 volts to prevent noise problems. IC-5 and IC-6 are a VCA controlled by IC-4; as a unit, forming a compressor. This prevents preamp and filter saturation upon attack, increases sustain, and balances amplitude, reducing the gain of the harmonics (see Fig. 2). The compressor has then conditioned the signal for the filters.

Fig. 13 on page 20 shows the circuitry used for filtering and providing the clearing (flag). IC-7 and IC-8 form the first filter which is highly damped. IC-9 is the second filter which is damped less than the first filter. The scheme here is to keep the first filter completely out of phase from the second one, as can be seen from Fig. 3 and Fig. 4. Then they are fed into a comparator. What happens is the harmonics are always higher in frequency than the fundamental, and are between fundamental cycles with the filters out of phase; filter #1 produces a near perfect sine wave while harmonics appear slightly on filter #2. The harmonics appear between DC level crossings and the comparator does not pass them until the signal amplitude decays to an unusable form. The output of IC-10 should be a raw square wave (see Fig. 5) corresponding to the fundamental. As you go up in frequency, the filters decay in amplitude in equal amounts so IC-10 still compares them. If damping is to be increased or reduced in order to tune the filters to a particular instrument, they must be kept proportionally out of phase. This may have to be done in some cases. IC-11 produces a flag to the clear pin of the shift register, using the signal envelope [4]. The 301 is used with a Schmitt trigger snap action. The inverting input should have about 1.5 volts with higher voltages relating to harder triggers and quicker clearing. The output of IC-10 will carry on sporadically when not in use, but this is all ignored by the shift register due to the option of IC-11.

The pulse generator and shift register section is shown in Fig. 14 on page 21. IC-12 is a CD4093 with hysteresis gates. IC-12(A) is an inverting risetime improver (see Fig. 6). IC-12(B) is a pulse generator using negative going edges as a trigger. The pulse out will be 100 ms. IC-12(C) is a free-running clock (370.554 KHz) clocking the 18 stage shift register formed from D flip-flops. Every positive going edge of



BLOCK DIAGRAM OF THE PITCH EXTRACTOR

Fig. 1

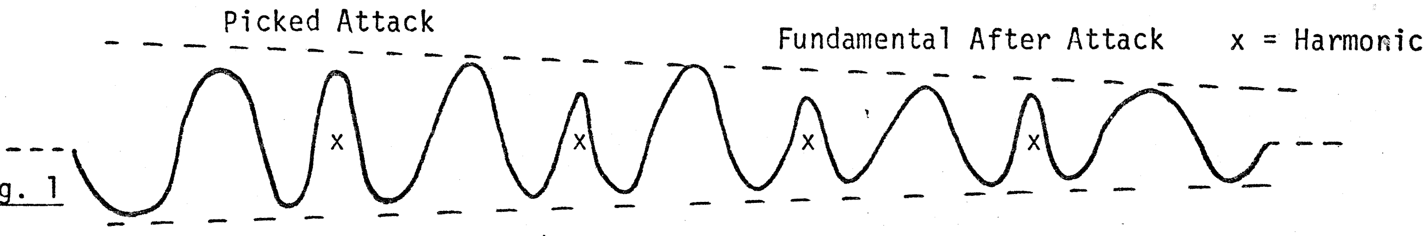


Fig. 2

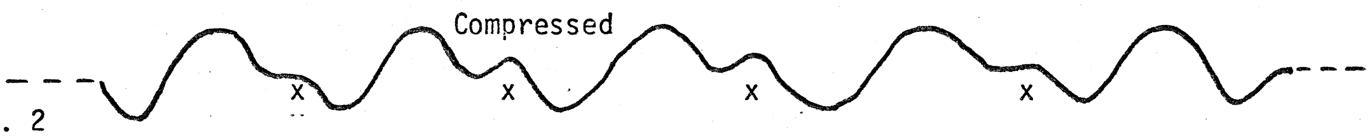


Fig. 3

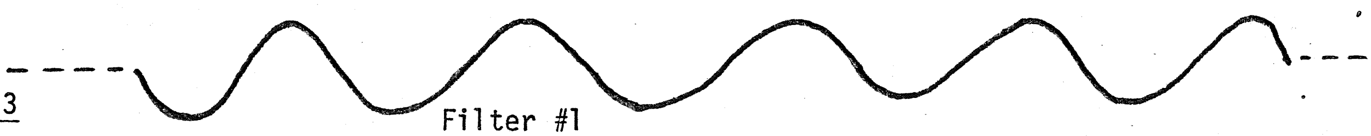


Fig. 4

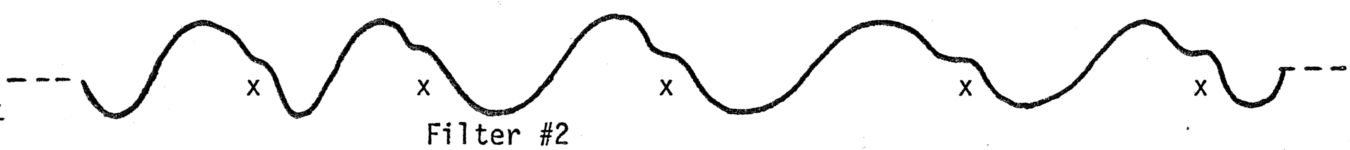


Fig. 5

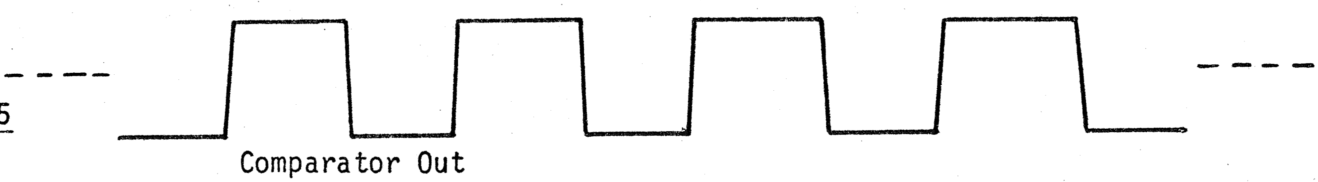


Fig. 6

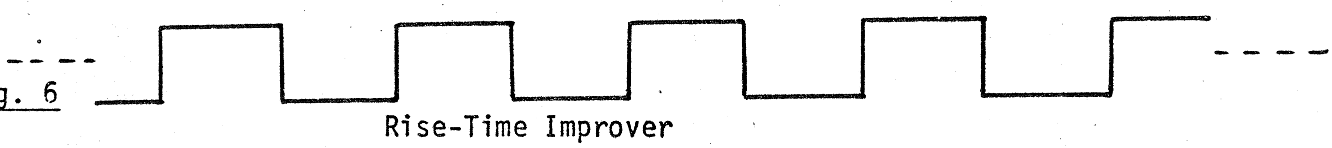


Fig. 7

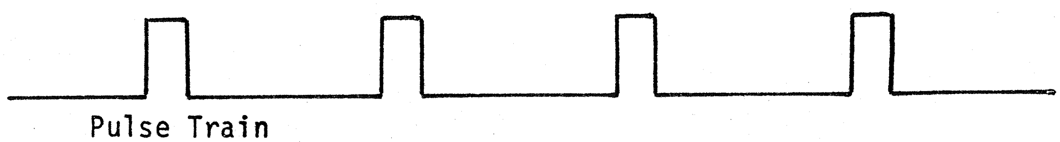


Fig. 8

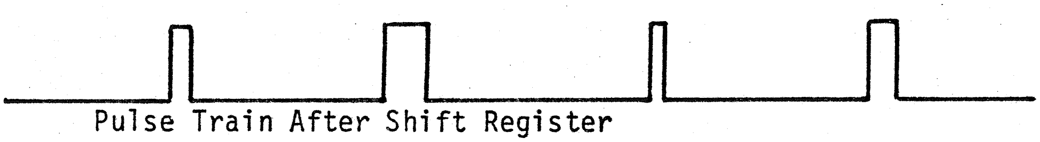


Fig. 9

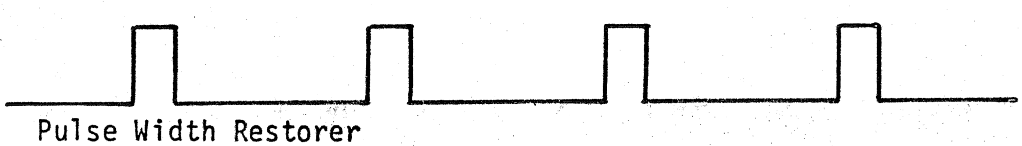
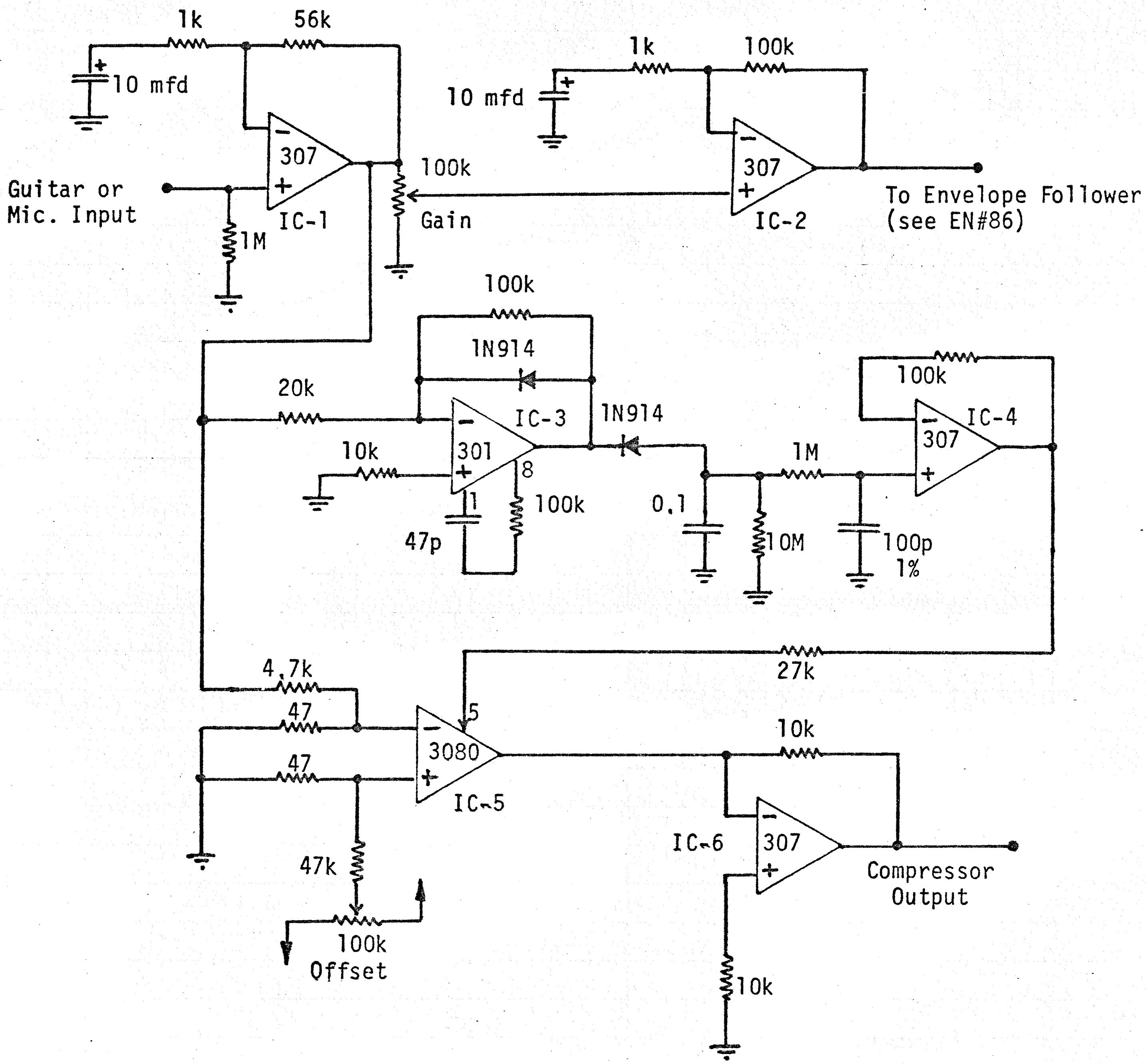
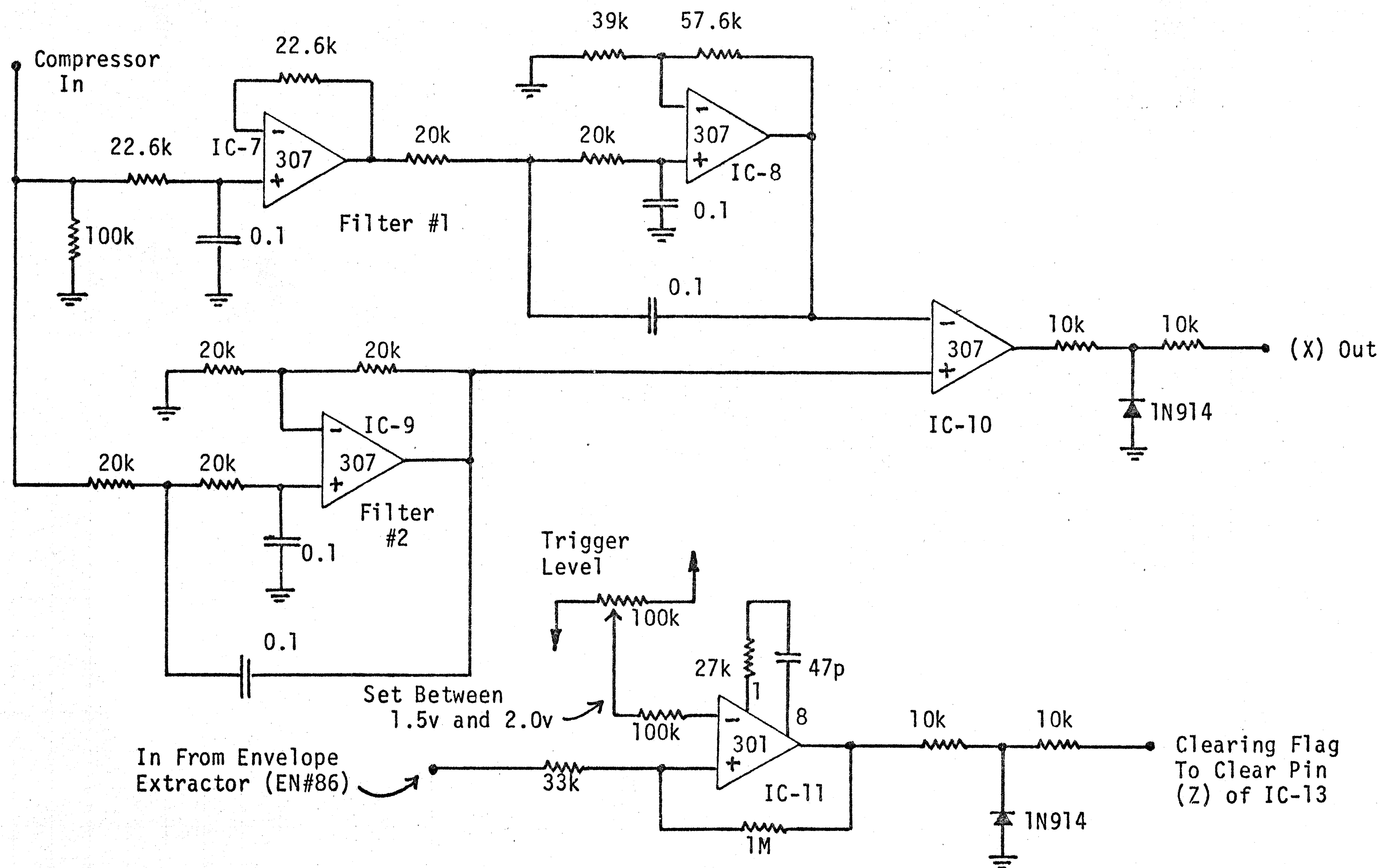


Fig. 12   PREAMP AND COMPRESSOR



the clock passes information stored on the first stage to the second stage and so on sending it down the chain delaying the pulse train (Fig. 7); so the clearing flag clears the first six stages before anything funny starts to happen. IC-12(D) is a pulse-width restorer. Running a pulse train through a shift register with a clock causes problems in that the clock chops up the pulse width (see Fig. 8), outputting variable frequency on the Log_2 output. This pulse width modulation is corrected (see Fig. 9) by IC-12(D) reproducing the same constant pulse width as was input to the shift register. The output of IC-12(D) is then fed to the timing generator as designed by Robert Iodice [5].

Fig. 13 FILTERS AND CLEARING (FLAG)



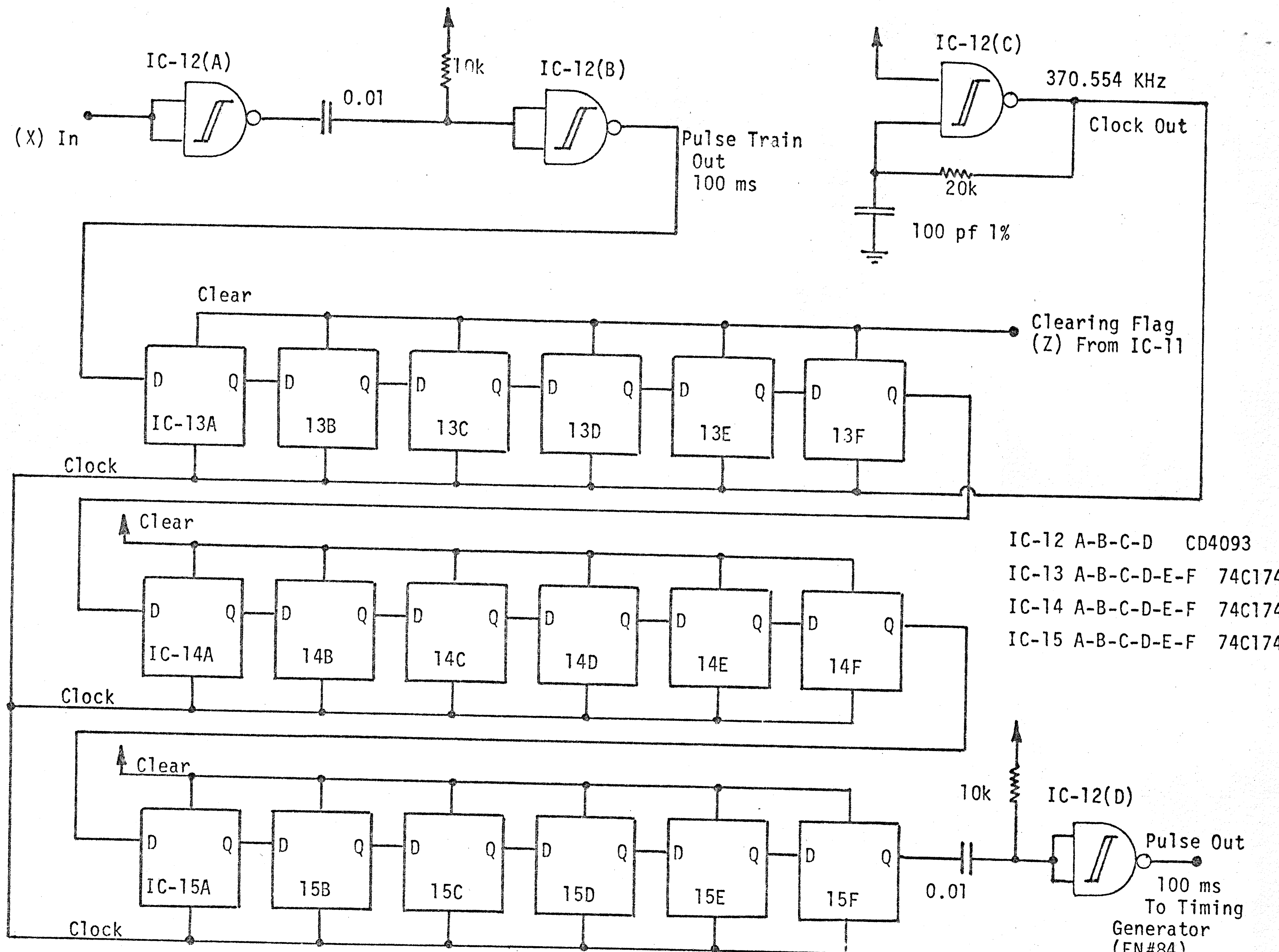


Fig. 14 PULSE GENERATOR, SHIFT REGISTER, AND CLOCK

CIRCUIT PERFORMANCE

Overall, the circuit with Bob Iodice's Frequency-to-Voltage Converter [5] and Envelope-Gate-Trigger Extractor [4] performs very well, beyond what I expected, between 95 and 100% with guitar, and 100% with single line instruments (horn, etc.), as long as the frequency range is not exceeded. Vocally, it worked well also, but I didn't like the sound I got with the synthesizer following vocal lines - it sounded like a kazoo following along.

The setup I used was a Gibson ES-355 with hot extra clean pickups and 3 lbs of brass in the body for extra sustain, a minimoog model D, and the pitch extractor. The only problem I encountered is on my unwound G string which many guitar players use. With this string, harmonic overtones come through two or three seconds after I strike the open string and the four notes after that. However, the VCA on the synthesizer is usually shut down before this causes problems. Other than taking a little care in not striking two notes at the same time (you can bump and brush without problems), the other 133 notes play perfectly at any speed. Other guitars (solid body) were used with weaker pickups. Harder preamping was needed and slight filter modifications to improve the performance were needed. The construction should be tailored on an individual basis.

The shift register could be extended to many more stages and the clock run at a much higher frequency, and then fed to a divider circuit (see Fig. 10). Then if both frequencies 1 and 2 were fed simultaneously to the clock inputs, and then frequency 1 was dropped, leaving frequency 2 in the absense of an input, slowing down the shifting, then indefinite sustain could be achieved. However, in the present design, 18 stages of delay are enough to not be noticed, with much less hardware, and eliminate the damping problems (see Fig. 11).

The addition of a good blanking pulse [1] could well bring performance up to 100% with any type of excitation source.

For polyphonic purposes, a hexpohonic pickup [6] could be used to drive six two octave filters and then into their own A/D converters and to six synthesizers. This would be much easier to design, but the board layout and parts involved would look like an N.Y. City roadmap and cost more money and time. The hexophonic pickup and six two octave filters could be used in the monophonic type for greater string separation except the addition of a data selector or some type of flip-flop arrangement would be necessary to select the string being played.

In conclusion, I would like to say that after 2000 hours and 80 bottles of asprin that I'm enjoying what I learned and made and intend to use it for live performance.

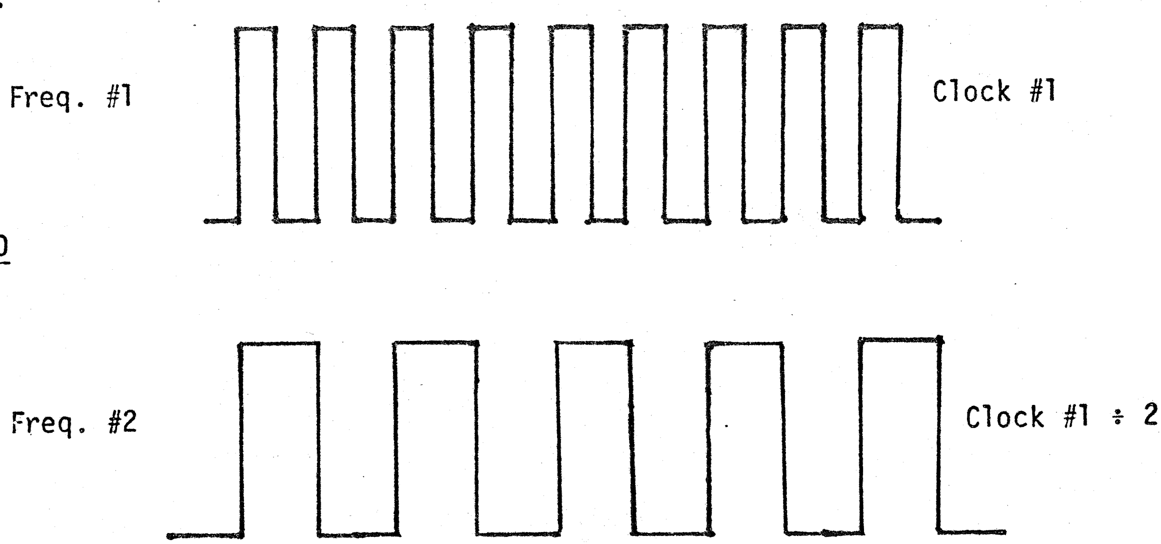
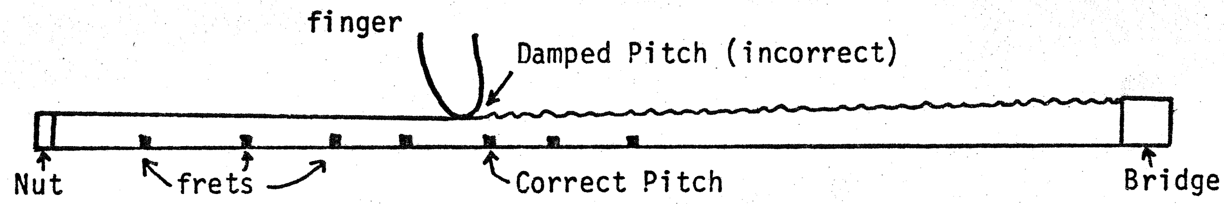


Fig. 10

Fig. 11 Damping



Damping occurs when the finger is lifted so that the string loses contact with the fret while it is still vibrating. The finger then takes on the function of the fret, outputting the wrong pitch.

REFERENCES:

[1] B. A. Hutchins, "Pitch Extraction, Part 3, The Complete Experimental Device" Electronotes, Vol. 7, No. 55, July 1975

[2] B. A. Hutchins, "Comb Filters Applied to the Pitch Extraction Problem" Electronotes, Vol. 9, No. 77, May 1977

[3] I. J. Fritz, "Simple Pitch Extractor for Clarinet" Electronotes, Vol. 9, No. 81, September 1977

[4] R. Iodice, "One Chip Envelope, Gate, and Trigger Extractor" Electronotes Vol. 10, No. 86, Feburary 1978

[5] R. Iodice, "A Frequency-to-Voltage Converter for Use With Pitch-to-Voltage Conversion Devices" Electronotes , Vol. 9, No. 84, December 1977

[6] Dale Wills, "Guitar Pickup" Electronotes , Vol. 10, No. 87, March 1978

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READER'S EQUIPMENT AND IDEAS:

A. ENVELOPE FOLLOWER WITH IMPROVED RIPPLE AND RESPONSE TIME CHARACTERISTICS:

-by Denny Genovese, Dennis Electronics, 2130 Metcalf St, Honolulu, HI 9682

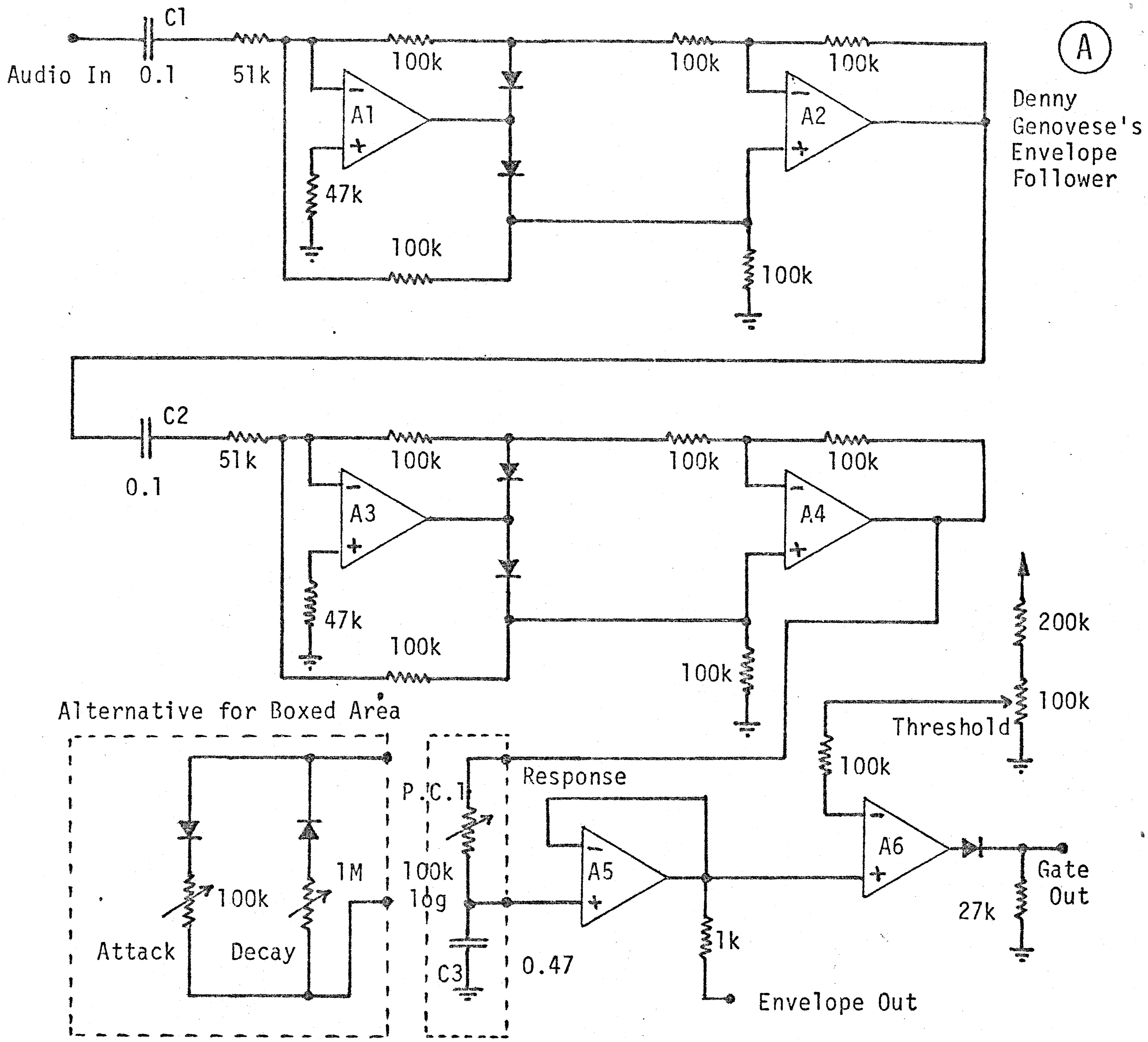
Here is a tested envelope follower design that effectively reduces the ripple vs: response time dilemma. The trick is that two full wave rectifiers are used in series, thus quadrupling the input signal frequency before filtering rather than only doubling it as in other designs. This allows the filter to be set much higher, allowing for faster response. The full circuit appears on the top of page 24.

A1 and A2 comprise the first rectifier which is essentially the same circuit submitted by Dave Rossum and used in the Musical Engineer's Handbook, but with the gain raised to 2, by halving the value of the input resistor. This is necessary, since the output of the filter will be the average voltage present at its input.

A3 and A4 comprise the second rectifier, which is identical to the first. The coupling capacitor (C2) makes the second rectification possible, by returning the double frequency signal from A2 to a zero average.

C1 was added to keep the input waveform as symmetrical as possible, taking another crack at possible causes and components of ripple.

P.C. 1 and C3 form a simple but effective filter, followed by A5. An alternative treatment is also shown for the boxed area which would allow separate response time characteristics for attack and decay.



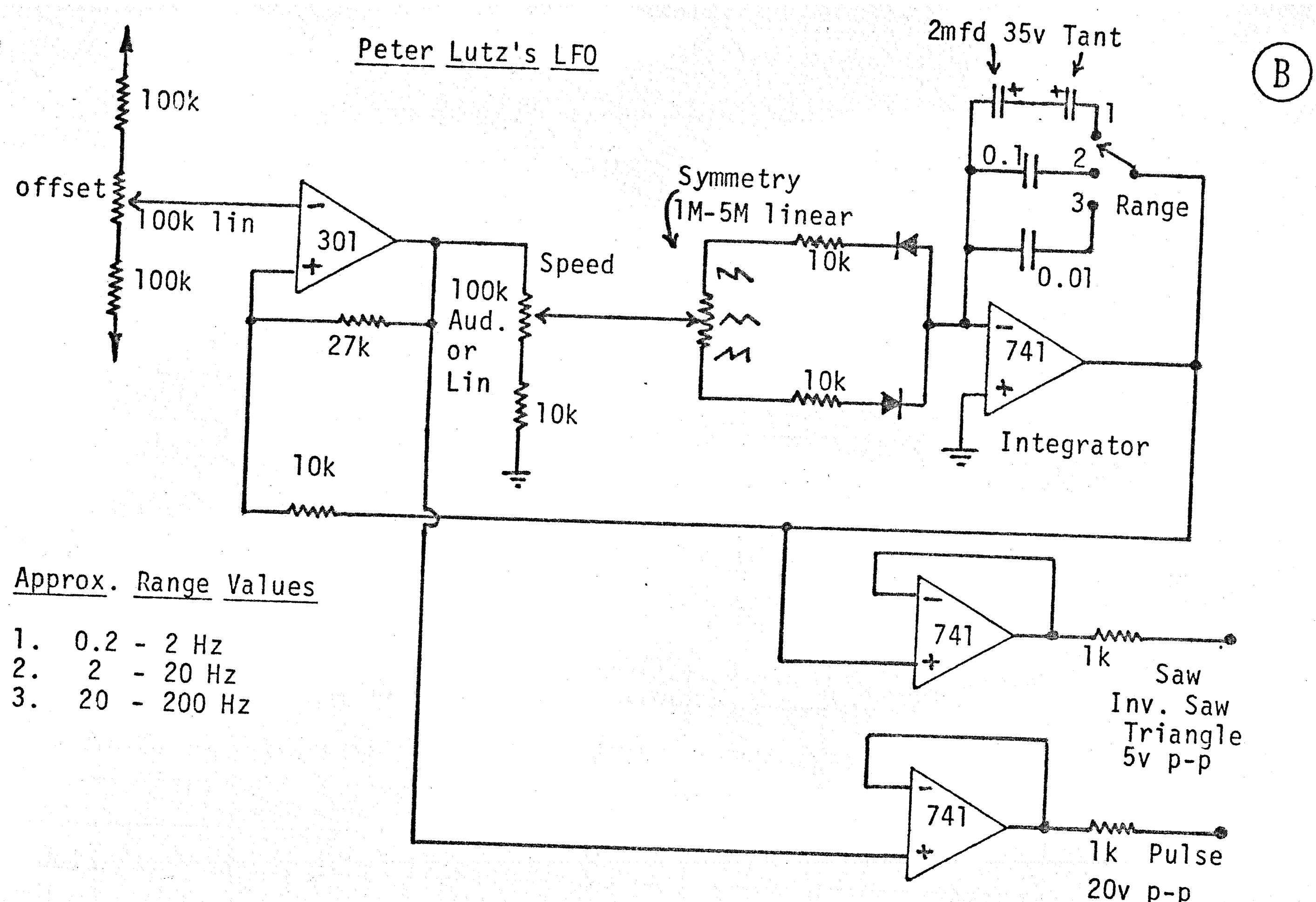
The final stage, A6 is a comparator for obtaining a gate whenever the envelope voltage exceeds the threshold set by P.C. 2. This threshold is variable between 0 and 5 volts. More definition could be obtained with a lower voltage range by increasing the value of the 200k resistor.

B. LFO WITH RAMP, INVERTED RAMP, TRIANGLE, PULSE-WIDTH FROM 10% TO 90%:

-by Peter A. Lutz, Wasatch Music Systems, Salt Lake City, UT 84109

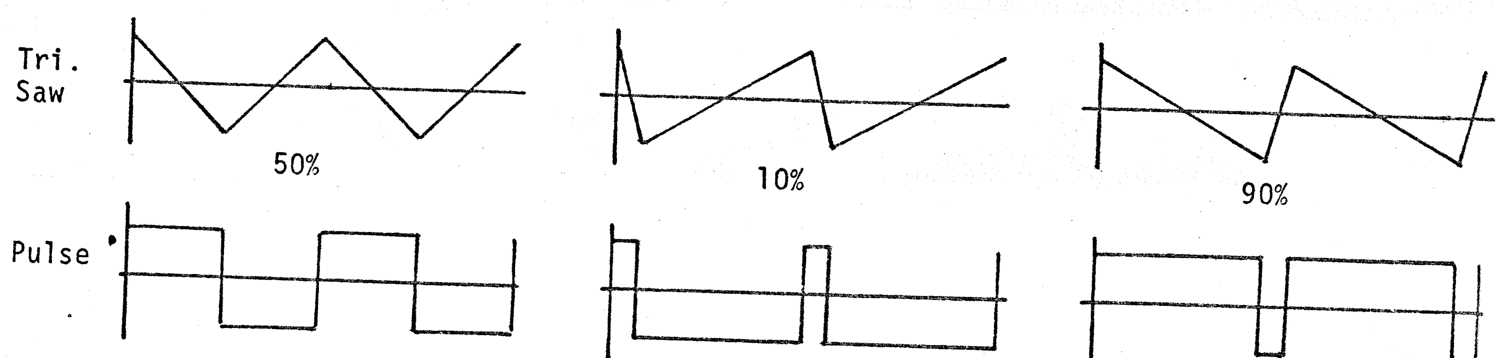
Pete Lutz has submitted the low-frequency oscillator circuit shown on the top of page 25. The basic circuit is the triangle-square oscillator formed from an integrator and a Schmitt trigger, but separate charge and discharge paths are used so that the rise and fall times can differ, resulting in variable symmetry in the triangle wave, and thus ramps as well as the symmetric triangle can be obtained. The circuit also features a DC offset control for the triangle (ramp and inverted ramp as well) output, and this allows the output to shift by about 5 volts in either direction.

Peter Lutz's LFO

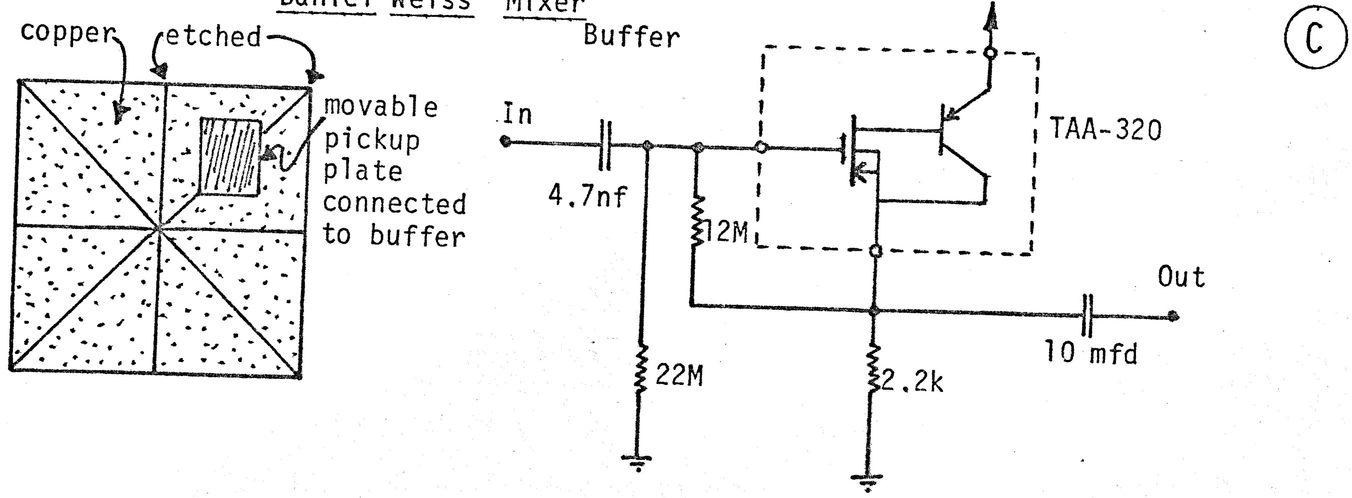


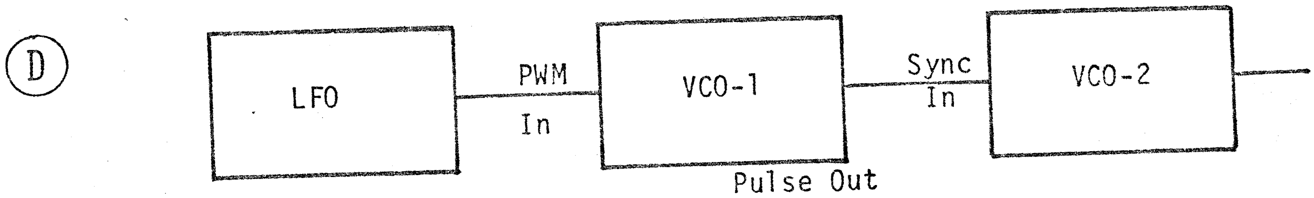
Approx. Range Values

1. 0.2 - 2 Hz
2. 2 - 20 Hz
3. 20 - 200 Hz

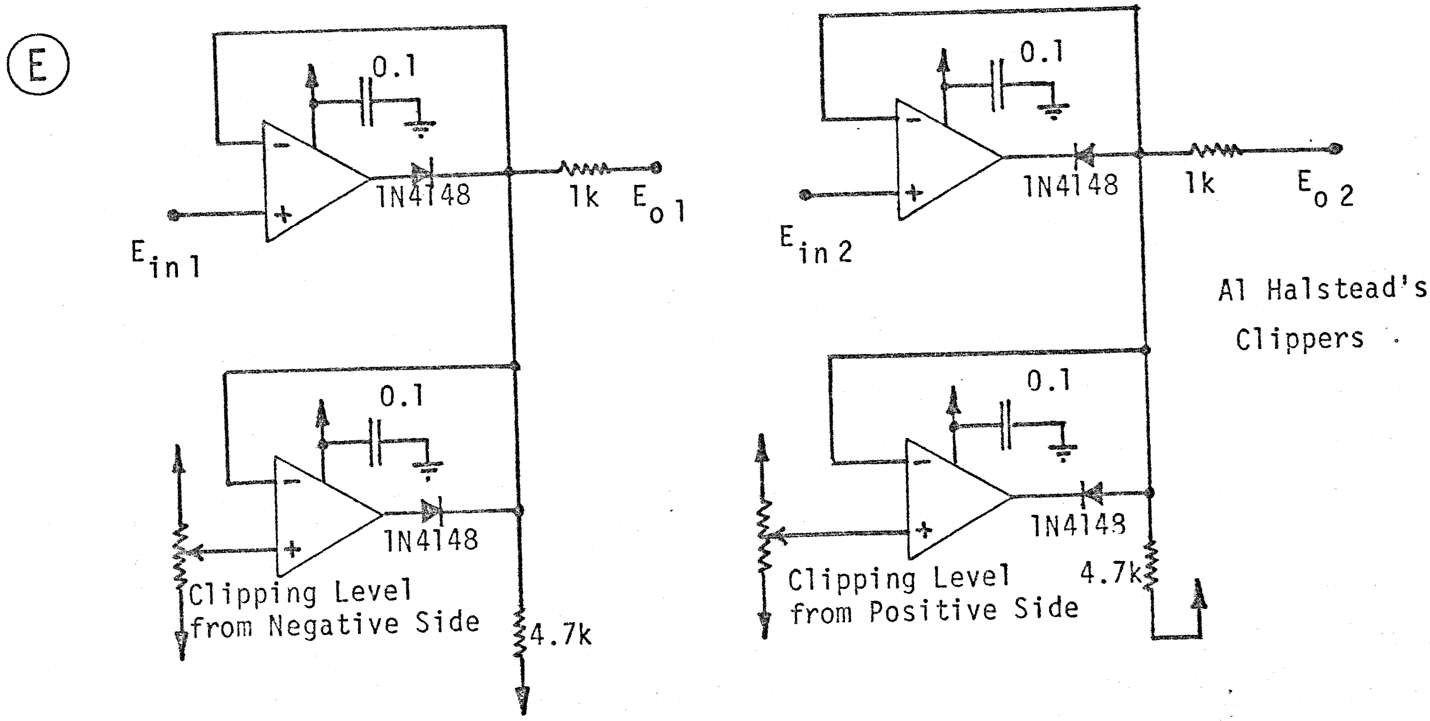


Daniel Weiss' Mixer





Daniel Weiss' Patch for Controlled Beating of Oscillators



C. <u>MIXING PLATE BASED ON CAPACITIVE COUPLING:</u> -by Daniel Weiss

Daniel Weiss' capacitor coupled mixer plate is shown at the bottom of page 25. It is constructed from a PC board plate etched to provide eight electrically isolated triangular "islands" which are the source electrodes. Signals are fed to these source electrodes with shielded cable. The board is then sprayed with plastic spray to provide electrical isolation between the electrodes and the pickup plate, which receives signals by capacitive coupling. The pickup buffer is also shown, and Daniel indicates that a CA3140 op-amp might be used for this buffer instead of the circuit shown. With this mixer, it is possible to move the pickup plate very rapidly, and with different combinations of sourcing to the source electrodes, mixing patterns can be altered much more rapidly than is possible with knob turning.

D. <u>PATCH FOR CONTROLLED BEATING WITH TWO VCO'S</u> -by Daniel Weiss

The patch at the top of this page was submitted by Daniel Weiss and is intended to offer the option of controlled beating between VCO's. Normally, two VCO's which are intended to track each other will either beat in an uncontrolled manner (this is often musically interesting however), or are locked into a no-beat pattern by synching. In Daniel's scheme, the beating is controlled by the LFO. The LFO controls the PWM of the first oscillator, and thus the edges of the pulse output change relative to the phase of the other waveforms of the oscillator. The pulse edges of the first oscillator then are applied to the sync terminal of the second oscillator, and control the phase of the waveforms of the second oscillator relative to the first. In this way, the phasing of the two oscillators is controlled, and the effective beating between them is controlled.

E. <u>VARIABLE POSITIVE AND NEGATIVE WAVEFORM CLIPPER</u>:

-by Al Halstead, Halstead Sound Laboratories

Al Halstead has submitted a pair of complementary clipping circuits similar to those described by Jung in the <u>Op-Amp Cookbook</u>, pg. 188. The circuits are shown in the lower illustration on page. 26. The 0.1 mfd capacitors on the +15 supply are used to remove any instability that might appear on the waveform.

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BASIC APPLICATIONS OF COMPONENTS AVAILABLE THROUGH ELECTRONOTES:

-by Bernie Hutchins

Here we want to give a complete run through of the parts we are currently offering for sale to our subscribers. This will serve to provide this basic information in a unified form, and to give readers a better idea why we offer certain parts and how they fit into our designs and the reader's overall parts gathering efforts.

OP - AMPS

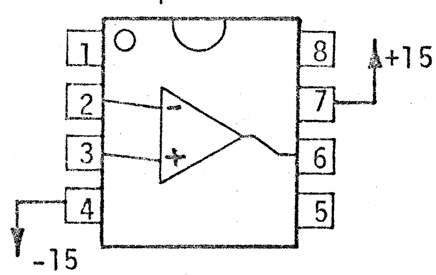
<u>OP-AMPS</u>: We are currently offering three op-amps to our subscribers: the LF351, the CA3140, and the BB3500. The table below gives the basic parameters of interest at a glance. For comparison, the types "307" and "556" are also shown:

Op-Amp Type	Input Bias	Slew Rate	Power Bandwidth ±5 volts	Power Bandwidth ±10 volts	Handling and Application Precautions	Approx Cost
BB3500	30 na	0.9v/μs	28 kHz	14 kHz	None	20¢
LF351	50 pa	13v/μs	414 kHz	207 kHz	May need 10pf in feedback	45¢
CA3140	10 pa	9v/μs	286 kHz	143 kHz	Protect from static, may need 10 pf in feedback and 0.1 mfd on -15 supply	45¢
"307"	70 na	0.5v/μs	16 kHz	8 kHz	None	---
"556"	8 na	2.5v/μs	80 kHz	40 kHz	None	---

Of the three op-amps available, there is a clear difference between the BB3500 and the FET types LF351 and CA3140. The FET types cost about twice as much, provide about 10 times the power bandwidth, and are about 1000 times lower in input bias current. The FET types may require a little care in handling and application however. Note that the BB3500 is about twice as fast as the "307" (or the "741") type, and will easly reach an audio power bandwidth (say 15 kHz) at a ±5 volt signal level, and gets very close (to 14 kHz) at ±10 volt signal levels. This means that in systems where it is necessary to keep costs to an absolute minimum, and where ±5 volt signals are used, it is quite reasonable to use the BB3500 for all applications not requiring the lower input bias currents of the FET types. This includes probably 75% of applications, so for a typical module, using the BB3500 could save a dollar or so. Choosing between the two FET types is a little more difficult. While the LF351 has a slightly faster slew rate (13v/microsecond as compared to 9v/microsecond for the CA3140), the CA3140 has a lower input bias current (10 pa as compared to 50 pa for the LF351). Since most of our circuits will get along fine on something like 5 v/microsecond, the CA3140 might seem like a logical choice for all applications. However, there are two drawbacks to the CA3140. First, it has a fancy output stage that allows the output to reach the negative supply rail, and while this is a good idea in general, there are some problems with this when we try to drive our standard 1k output resistances (see EN#69, pg. 13). Secondly the CA3140 is much more static sensitive than the LF351, and requires much more careful handling. When installing a CA3140, leave it in the protective foil (or in the anti-static plastic rail if shipped that way) until you are ready to install it, install all other components first, and touch circuit ground with your finger before touching the IC to its circuit position. In view of these factors, the LF351 is the logical choice for most applications, but we should not neglect to use the CA3140 in internal structures where its lower bias current is an advantage.

When using the FET types (LF351 and CA3140), depending on the particular layout, you may need a few extra capacitors to stabilize the circuits. It is a good idea to attempt to avoid trouble by installing capacitors in the range of 0.01 to 0.1 mfd between +15 and ground and -15 and ground every few inches throughout the circuit. These are "power supply bypass capacitors" and serve to provide a local "emergency" supply of charge to keep the op-amp happy and unaware of its neighbors. With a CA3140, pay particular attention to the -15 bypass, particularly if you see any sort of distortion on the negative portion of output waveforms. If a general sort of high frequency oscillation is seen on top of the desired waveforms, add a few pf (say 10 pf) of capacitance between the (-) input and the output. This prevents instability due to the very high input impedance and stray capacitance on the (-) input traces. In circuits with many op-amps, you may find the oscillation on all op-amps and placing a capacitor in the feedback loop of one may cure all. This may be because the op-amps are in cascade, or because the oscillation propogates through the circuit by means of some "unauthorized" path such as the power supply lines. Thus it is a good idea to locate the original source, or to put in a few extra capacitors even when the problem is apparently sloved.

Top View



All the op-amps we have to offer, and most others you will find have the same pinout as shown at the right. These are all internally compensated, and no critical offsetting is generally needed, so you need only be concerned with the (-) input, pin 2; the (+) input, pin (3); the negative supply, usually -15 volts, pin 4; the output, pin 6; and the positive supply, usually +15, pin 7.

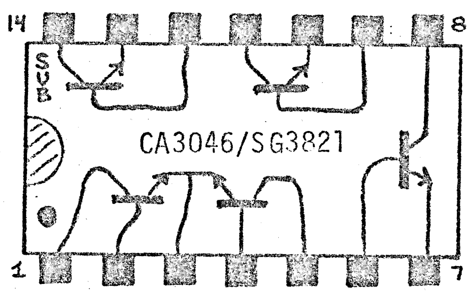
When looking over previous issues of the newsletter and selecting circuits from these, you will often find the "741", the "307", and the "556" suggested. The BB3500 is the suggested improved replacement for a 741 or a 307, while the LF351 should be used to replace a 556.

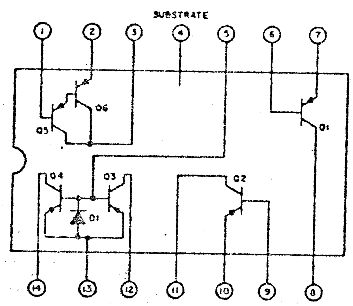
▷ TRANSISTOR ARRAYS

TRANSISTOR ARRAYS: We are currently offering the following transistor pairs and arrays: the AD818, the AD821, the CA3046, the SG3821, and the CA3084. The table below will show the important features of these components at a glance:

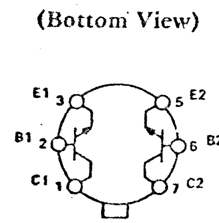
Type Number	Transistors Number	Type	Pins and Package	Features and Applications	Approx. Cost
CA3046	5	NPN	14 pin DIP	Exponential Converter	$0.75
SG3821	5	NPN	14 pin DIP	Premium type 3046	1.60
CA3084	4	PNP	14 pin DIP	Suggested for Current mirrors	3.00
AD818	2	NPN	6 wires	High-grade log. performance	4.20
AD821	2	PNP	6 wires	well matched PNP pair	2.35

Let's begin with a discussion of the matched pairs, the AD818 and the AD821, which are the best of the parts we offer. The AD818 is probably the best log conformance pair available, featuring a bulk emitter resistance below one ohm. It is thus used for the best exponential converters. It is an NPN matched pair. We would prefer to use NPN devices because they can generally be made better than the PNP devices, but the current control of the CA3080 OTA often causes us to use PNP pairs in our exponential converters, and the AD820, 821, 822 series is one of the few PNP pairs available. We choose the 821 because its specs are quite a bit better than the AD820 and only costs a little more, while the AD822, while significantly better than the 821, costs quite a bit more. Of the arrays, the CA3046 is a basic type featuring five transistors, inculding a matched pair on pins 1, 2, 3, 4, and 5 with the emitters connected together and to pin 3. The CA3046 can be used instead of the AD818 with some degrading of performance by using the matched pair on the 3046 chip. Additional compensation tricks with the CA3046 bring its performance up so that it can be compared with the AD818, although the 818 is still superior. A somewhat better grade of the CA3046 is available in the Silicon General type SG3821, and is suggested instead of the CA3046 for exponential converter applications. The CA3046 is still fine for VCA applications, and similar. The CA3084 is a rare array of PNP transistors, and is of interest here because it can be used for a 4 way current (source) mirror for driving a number of CA3080 control currents in parallel (using a diode also on the chip). It is not well suited as an exponential converter however, and should not replace the AD821 in such applications. Pin diagrams for the arrays are given below:

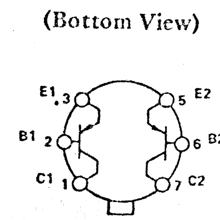




CA3084



AD818



AD821

G A I N   B L O C K S   (M U L T I P L I E R S)

GAIN BLOCKS, MULTIPLERS, AND THE OTA: We are currently carrying the CA3080 OTA (Operational Transconductance Amplifier), and the Analog Devices AD533 four-quadrant multiplier. In simple terms, the CA3080 is basically a two-quadrant multiplier costing about 75¢ while the AD533 is a full four-quadrant multiplier costing about $7.20. There are so many applications of the CA3080 in electronic music that we can only briefly outline them here. In addition to many examples in past issues of the newsletter, Application Notes AN-21 through AN-24 list most of the major applications of the CA3080. The applications range from the linear ones, where the CA3080 acts as a two quadrant multiplier as:

$$I_{out} = 19.2 \cdot I_c \cdot V_{in}$$

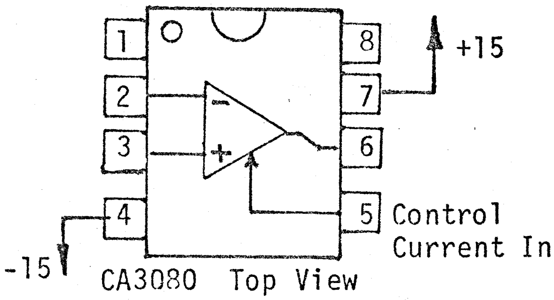
where I_c is the control current into pin 5 and V_{in} is ±10 mV or less, to the saturated ones where:

$$I_{out} = I_c \cdot (\text{Sign } V_{in})$$

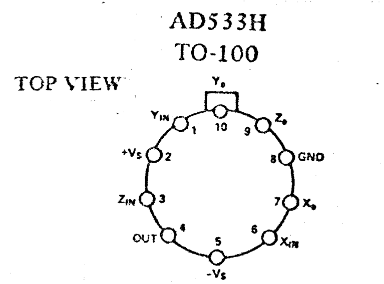
in which case the CA3080 is a current switch. In between, there is also a region of non-linearity where the response characteristic is suitable for triangle-to-sine conversion.

For most gain control operations, the OTA works as well as a four-quadrant multiplier would, or is even better. In cases where a true four-quadrant multiplier is needed, the AD533 is suggested as being simple to use, although somewhat expensive. It was used in our ENS-76 frequency shifter design (EN#83), and will also stand well by itself as a balanced modulator (ring modulator).

There are few if any cases where one must decide between the CA3080 and the AD533 that are not easily determined on some obvious performance advantage, and this failing, there is the large difference in price to help with the decision. The base diagrams of these two components are shown below:



CA3080 Top View



TEMPERATURE COMPENSATING RESISTOR Q81

The Tel Labs type Q81 resistor is available from us in the value of 2k at a cost of about $2.20. This resistor is used for a single purpose - to compensate for the temperature change of exponential and logarithmic circuits. These resistors are wire wound from a special alloy giving a resistance that changes by about 1 part in 300 for each °C change in temperature. Exponential converters have a temperature term that goes as 1/T, where T is "Absolute" or "Kelvin" temperature, relative to "absolute zero." Relative to absolute zero (-273 °C, or 0°K), we live and work at about 25°C, which comes out to about 300°K. Thus, a change of 1°C is a change of about 1 part in 300. To get a feel for what this means, consider what would happen if you were using a synthesizer in a room at 70°F and the temperature then changed to 60°F, a change of 10°F. The degrees on the centigrade scale are the same size as those on the Kelvin (absolute) scale and are 9/5 the size of the degrees on the Fahrenheit scale, so the change in room temperature is about 5.5°K, or about 5.5 parts in 300, or nearly 2%, which is about 1/3 of a semitone drift on the synthesizer pitch. Certainly this is significant for all but the most basic "playing around" type applications of synthesizers. Thus, the Q81 resistor is used so that its 1 part in 300 change (3300 ppm or 3500 ppm in the specs) balances the natural 1 part in 300 behavior of the exponential converter.

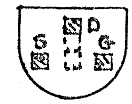
One assumption about what we have said above is that the exponential converter (in particular, the transistor pair in the converter) are at the same temperature as the compensating resistor. To assure this, it is desirable that we do more than just locate these components close together on the same curcuit board. It is thus a good idea to have these parts actually touching each other, and to put some heat sink compound (a greasy, usually white colored compound with good heat conductivity, often used to improve the thermal contact of regulators and power transistors with a surface to remove excess heat), in the cracks between them.

THE KE4859 SWITCHING FET

The KE4859 switching FET (also as 2N4859) is used in our designs as a reset switch to rapidly discharge an integrating capacitor in a VCO. There is not much that needs to be said about this part, except that it does work very much better than an ordinary type FET such as the 2N3819, MPF-102, or TIS-75, which should not be substituted. These other types give a very sluggish discharge relative to the KE4859.

The base diagram of the KE4859 is shown at the right. Some configurations of the leads have all three wires in line at the very base of the package, but the center lead bends away further out. You should have no trouble getting this in properly, and if worse comes for worse, you can always try all possible combinations (of which there are six, two of which work) without causing any damage to the FET or the rest of the VCO circuit.

Bottom View

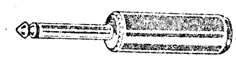


KE4859/2N4859

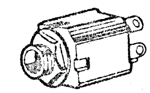
PLUGS AND JACKS

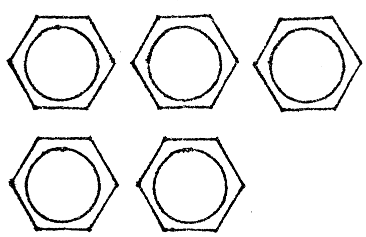
PLUGS AND JACKS: We are currently offering standard 1/4" phone jacks and plugs. Thes are standard Switchcraft parts. The plug is Switchcraft type 250, has solder terminals and a black plastic handle that is 1/2" in diameter. The jacks are either type 111 or type 112A, and are of the "Hi-D" (high density) type. That is, they mount in a 3/8 hole and can be placed as close together as 5/8", which is as tight as you would want them. The general appearance of these parts is as shown in the figure at the right. The mounting hole positioning is indicated full size below, to give you a better feel for how close these can be mounted. Finally, we should tell you the difference between the 111 and the 112A. Both the 111 and the 112A have a center metal bushing, supplying the necessary grounding connection, assuming a metal panel, or you can run a separate wire. There is also, on both jacks a terminal that represents the tip of the plug. Now, the 112A has a third terminal not present with the 111. This terminal, in the absense of an inserted plug, will supply a signal to the other non-grounded terminal. This means that you can set up a standard signal path through the jack, and when a plug is inserted, it defeats the standard path, and supplies its own signal. This is what is called in the synthesizer business "patch over hard wire." The 112A can be used in the same way a 111 is if you don't need the switching arrangement, but the 112A is slightly more expensive as you would expect. The switching schematics for the jacks are shown below:

250

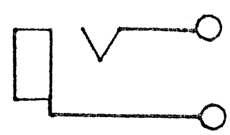


111
112A

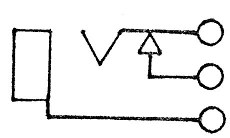




Typical Hi-D Mounting



111



112A

CAPACITORS

We are offering a couple of values of polystyrene capacitors, simply as a matter of making them available to readers in small quantities. The 1000pf value is used in several VCO designs, and the 2500pf value will work in the VCO Option 1 of the ENS-76 series. These poly caps are intended mainly for use in the VCO circuits, and other types will suffice in most other applications, or as specified.

provide sounds that are somewhat similar. There is really no way to say that one is better than the other, and neither is it possible to say that one phasor sounds the same as another, although most flangers sound much the same (except for noise). The main difference in circuitry is that a phasor uses a cascade of phase shift networks while a flanger uses a signal delay that is the same time at all frequencies. This delay may be either an analog delay from a charge-transfer device, or a digital delay. The analog delay line is much less expensive.

Note: The following comments and questions might have gone into a "Forum" feature, but it seems to fit in well here. We are printing the whole thing, because there are some interesting ideas here.

Q: I am highly disappointed in your response to a question in the June 1977 issue (EN#78) concerning a digital delay project. If you really support the attitude you expressed there, then why does Electronotes exist? If you, or anyone else, actually believes that once a circuit is "solved" then the case is closed, we would all still be using 1960 vintage Moog and Buchla filters. Fortunately, with the help of people like yourself, electronic music continues to develope and improve through advancement in technology and creative thinking. Thanks to these new developments, we can now build a high quality 24 db/octave VCF for $15 (using the Eμ Chip) which, two years ago would have cost twenty times as much and provided lower performance. The point is, if technology continues to advance, no project should be dismissed, and if the project appears too complicated and expensive, then obviously your looking at it wrong. Certainly, a delay line will be expensive and complicated if you ignore recent developments and concentrate on "traditional" (if that's the proper term) approaches. The price of memory chips is falling faster than Soviet satellites--a 4k by 8 memory, with quality PC board and address decoding can be had for $70. There is currently a 32k board for less than $500. 4k should be enough for delays of around 500 ms. at a 2.5 kHz bandwidth. If we follow the traditional approach of hardware conversion (specifically A/D), there are a number of companies (National, for one) producing low cost A/D and D/A hybrid converters which require only a few passive components external to the chip. But what about the newly available technologies, such as companding D/As which will give the same dynamic range from fewer bits? How about including one of these in a microprocessor-based software A/D and forget about the hardware A/D? The microporcessor would yield a whole new versatility to the system allowing you to achieve virtually all time domain effects with a single unit. Of course software development might be a bit complex, but I'm sure a number of your readers must have a processor by now and there are various companies and local computer stores who will burn PROMs for a reasonable price.

I haven't tried any of these techniques and don't guarantee that any will work, but surely they deserve consideration rather than simply ignoring them. They would certainly be cheaper than the $1000+ digital delays are currently selling for.

Your statement "besides, we have... simpler and cheaper (analog delay lines)..." is a cop-out...Analog delay lines can't begin to approach the quality of a good digital delay. One would assume that a person buys (or builds) a delay line to get delay, and the more the better. Digital delay time can be increased with additional memory, with no degradation of audio quality, to whatever extent the user can afford. Can the same be said for analog delays?

A: First, when I go back and read my original comments in EN#78 I find that nothing said above changes my first thoughts on this matter. I still don't see this as a good type of "hobby project."

Now, for some of the things I disagree with in the above comments. It is true that the Solid State Music chips, available through Eμ do make possible a good quality 24db VCF for around $15 for parts, but the main improvement is in space and convenience. The earlier designs in EN using the CA3080 probably costs about the same, $15, and I'm not sure where the figure "twenty times as much" came from. I do not agree that "if technology continues to advance, no project should be dismissed." I feel it is precisel because technology advances that certain projects should be dismissed. We should not tak on a project that involves a great effort if we can reasonably expect a better technological solution in the not to distant future. A good example of what I am tryin to say is found by reviewing the trade journals several years back where the "design ideas" columns were full of "how to do ______ on a four-function calculator." These gave methods of doing logs, sines, etc. with a moderately involved procedure on a four-function calculator. While these were clever, they were not very useful because they were long and hard to remember, and now of course, the functions are available on scientific calculators at a price that is possibly less than the four-function jobs at the time these ideas were published. We want to avoid the same sort of thing. To be sure, we have flirted with the same folly with things like the Hadamard transform network (not a practical synthesis tool at the present time), and even with the ENS-76 envelope generators where a fully voltage-controlled design was given, now obsolete beyond belief thanks to (and I do mean thanks to) the SSM chip.

As to the statement "if a project appears too complicated or expensive, then obviously your looking at it wrong," this seems totally wrong to me. Perhaps it would be handy if it were true, but unfortunately, certain things just really are complicated and expensive, and looking at it in different ways will not help much, except in that eventually technology may provide a better solution for us.

As to the specific ideas on digital delay lines suggested above, a couple of comments should be made, keeping in mind that the person asking the question has warned us that some of the ideas might not work. From what I understand of microprocessors, a software A/D is probably an order of magnitude too slow for serious audio work, and even if it were made fast enough (involving some compromise we might otherwise wish to avoid?), the hardware A/D might still be cost effective. The same general limitations seem to apply to time-domain processing of the digitally encoded signals - while the microprocessor can change processing parameters fast enough, it cannot do all the manipulations necessary to actually do the processing.

It seems that the contest gets down to hardware digital delay lines vs. analog delay lines of the charge transfer type (e.g., bucket brigade). Digital purists will not likely accept the analog delay lines, although from an audio engineering viewpoint their reluctance is not justified, either from performance or cost viewpoints. It is true that you can always seem to build a better digital delay line by using more bits, higher clock rates, etc., while analog delay lines have certain limitations. However, the charge-transfer technology is advancing rapidly, and even today, delay lines meeting practical needs for delay time can be realized either with digital or charge-transfer methods, and they can be compared. For most applications, it is not true that the more delay the better, and a relatively small delay is all that is needed. At the present time, it seems that charge-transfer devices will be used for these applications (for example, flangers, artificial reverb).

Certainly digital delay lines have their place, and certain processing devices (such as pitch changers) may work out best with digital methods. As suggested, things like companding A/D's and D/A's will make things easier.

Finally, it is a good idea here to say something about the type of things that do find their way into these pages. Many things are considered, and perhaps it is not evident because we don't feel that too many comments of the "why we don't do ____" are appropriate. To appear in print, an idea must be created and developed, and it must somehow take on the form of an article or a report. Someone must do some work - in other words. When this does not get done, things don't get into print. If someone were to send us an article on a digital delay line that is in good shape, we would print it. We have never had such an article submitted in any form. I don't even know if any individual has built one, but I suspect that even if such a project has been completed, the builder would be reluctant to write it up, simply because of the work involved. Many digital projects are fairly easy to build using just the data sheets, just knowing the general idea of how the thing works. For example, wiring a set of latches to a D/A is meaningful if you know the significance of the latched bits and the inputs to the D/A - you just do it. The corresponding wiring list (pin 8 to pin 12, etc.) is not meaningful, and subject to errors which are ignored by the knowledgable builder, but which trap the unwary. For this reason, block diagram descriptions of major digital projects may be the most useful in terms of providing the necessary construction information to a builder who will be able to carry it out. We are open to both block descriptions and full wiring diagrams however.

It will be interesting to see how applications of different types of delay lines develope. There is perhaps an irony in that analog delay lines have encouraged new applications, previously too expensive, that may find their way back to digital.

Q: I built your frequency shifter from EN#83 and it worked very well, and I have been very pleased with it. Recently it has been giving problems however. It works when first turned on, but later it fades out, except when the control frequency is changing rapidly. Any ideas where the trouble is?

A: I'll bet you have the Wien bridge oscillator adjusted too fine. That's the circuitry associated with IC-1, and you should adjust TP-1 so that there is a little more distortion at the output of IC-1. Remember, the low-pass filter (IC-2) was put in just so that you could tolerate a little more distortion in the oscillator. If you adjust TP-1 too fine, you do get a near perfect sine wave, but oscillation is marginal in such a case. Component drift with age or temperature may cause the oscillator to stop, except when it is excited by an external source such as having the power turned on, or perhaps, when the VCO control oscillator nearby passes over the natural frequency of the Wein bridge oscillator. Perhaps you built the circuit in the colder winter weather and the warmer summer temperatures are causing the problem. Just let the oscillator "bump its head" a little more and it should keep going under all conditions.

Q: In regard to your "Multi-Phase Waveform Animator," I don't see how you can say that the spectrum changes during the animation. It seems to me that all that changes is the phase, and this should not change the power spectrum. Please explain.

A: Let's take a closer look at this. If we took several different sawtooth phases, examined the spectra, added these spectra, and looked at the total, then the animation process would do nothing. However, what is happening here is that we are adding the waveforms, not adding the spectra. Your ear then listens to the spectrum of the waveforms summed in the time domain. Look again at Fig. 5 on page 9 of EN#87 and it should be clear from the example given there. It is often forgotten that the spectrum of the sum is not necessarily the sum of the spectra.

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