ELECTRONOTES 90

NEWSLETTER OF THE MUSICAL ENGINEERING GROUP

1 PHEASANT LANE

ITHACA, NEW YORK 14850

VOLUME 10, NUMBER 90 JUNE 1978

GROUP ANNOUNCEMENTS:

CONTENTS OF EN#90

We have made a couple of changes in our parts business which you may want to note. The most important change is that we have grouped our parts into different categories. Category 1 includes the parts that are difficult to get anywhere else, and we will be making every effort to have these parts in stock at all times so that you can depend on them. Category 2 includes parts we will generally carry, but which may not always be in stock. These are things like op-amps which are available many other places, but which we can give you a little better deal on when we do have them. Category 3 includes parts that are either "surplus" lot one-time deals, or parts (like plugs and jacks) that we will have only occasionally when we feel there is enough demand built up so that we can move a large lot rapidly. This should aid the person who is buying parts for a project since he can better judge whether he can get them from us or not. You should get parts from Category 3 for stock, since they may not be around later. Parts from Category 2 may be ordered from us, but if you are in a hurry, you may want to get a few from another source as well. We hope that you will be able to depend on getting parts from Category 1 without delay.

The second 1978 Preferred Circuits Collection will be mailed about August 20. Orders for it should reach us no later than August 15. The third 1978 Preferred Circuits Collection will be mailed about October 1, and orders for that should reach us by September 20. The cost of either edition is $6.50, available to current subscribers only at the present time. A general review of available circuits with discussion of current options is tentatively scheduled to be published in the next issue of this newsletter.

In this issue, we have a discussion of simple ways to make circuit boards, and have the second part of the waveform animator series, with two more parts planned at the moment.

News And Notes:

Craig Anderton has written a new book, Home Recording for Musicians, which has been published by Guitar Player Books, Box 615, Saratoga, CA 95070 and is available at a cost of $9.95 plus 50¢ handling and shipping. The book is in much the same style as Craig's popular Electronic Projects for Musicians, and Craig's work is probably well enough known to our readers that we need say little more about it.

A staff opening for "Electronic Music Technician" has been announced by Oberlin College. The position involves maintenance and design of equipment, as well as providing advice and teaching of a course on analog/digital circuit design when so requested. Interested persons should contact Dean David Boe, Oberlin College Conservatory. Oberlin, OH 44074.

Chicago area readers will be interested in the Omega Intermedia Center, 3433 N. Halsted, Chicago, IL 60657, phone (312)-477-9863. The center sponsors concerts and offers study in progressive art including electronics for artists, microcomputers and art, laser art, multi-media, and electronic music. For more information, contact the center.

Congratulations go to Mike Matthews, president of Electro-Harmonix who was named N.Y. State Small Business Person of the Year. [We note that this award was previously won by Bob Moog, so this is probably the first repeat for an industry as small as electronic music.]

A few papers of interest to electronic music as published in J. Aud. Eng. Soc. recently are "Applications Considerations for IC Data Converters Useful in Audio Signal Processing" by Walter Jung appearing in Vol. 25, No. 12, Dec. 1977; "The Use of the Phase Vocoder in Computer Music Applications" by James Moorer in Vol. 26, No. 1/2, Jan/Feb 1978; and "VOSIM - A New Sound Synthesis System" by Werner Kaegi and Stan Tempelaars in Vol. 26, No. 6, June 1978. The paper on the VOSIM system should be helpful because previous information on this was available in the hard to get Interface (Holland) publication, so this should answer many questions on the system which people have asked us.

We have not seen the following two books published by Tab Books, Blue Ridge Summit, PA 17214, which sound interesting. The first is the Master Op-Amp Applications Handbook by H. Fox, Tab Book 856, $9.95 and the second is the CMOS Databook by W. L. Hunter, Tab Book 984, $6.95.

Reader's Questions:

▶ Q: I have a question about the ENS-76 envelope generator Option 3 (voltage controlled).

A: The answer to any question on this circuit is to not build it. Thanks to the SSM 2050 IC envelope generator, we can do most of the same things the Option 3 envelope generator did, and do it cheaper and much much simpler. See EN#87 (16).

▶ Q: I see in the application note series that you are printing calculator programs for the TI series calculators. What about those of us with HP calculators?

A: I have a personal preference for the TI programmable calculators, but I have a feeling I am in the minority. Certainly we will be glad to publish reader submitted application notes giving HP programs. While it would seem to be possible, although not what you would call easy, to convert a TI program to HP, I think there is still a use for TI programs for HP users. This is that they give the HP programmer an idea about a problem that yields well to a calculator program. For example, the "graphical" calculation of frequency response by programmable calculator. We give the method in the notes and the actual TI program is an example, a bonus if you have the calculator.

How I Make My P.C. Boards In A Matter Of A Few Hours:

-by Bernie Hutchins

Over the years I have developed a simple and reliable method of producing P.C. boards which I find so logical that I really don't see how anyone else would want to try anything else for small quantities. Although I have described this in the MEH and in the Application Notes (AN-14 to AN-18), it seems to be useful to go over it briefly again here in the newsletter. We get a lot of questions about boards, and those builders who I have converted to my system will vouch for it. Typically you should be able to prepare a board (a VCO say), etch it, and begin soldering on the parts within a couple of hours of the time you first decide on which schematic to use.

Some essential points about the method: (1) You are going to be working on only one side of the board - the copper traces and all components will be on the same side. You will only drill holes in the corner to mount the board. (2) You will apply the resist pattern using a fine artist paint brush and a thinned down (with lacquer thinner - add about 50% to the original volume) lacquer paint. (3) You will be making up your layout as you paint, and painting freehand with a minimum of guidelines. You will be working directly from the schematic to the final board with no sketch in between. While all this may seem like it requires immense concentration and an artists hand, you should not have too many problems if you just accept the fact that it is not going to look 100% professional, and you are going to have a few more jumper wires than you might need if you worked on the design for a few days!

The first step is to select a board of the proper size. This is a matter of placing the IC's and the larger components in approximate positions using the schematic layout as a basic guide. Next clean the board with steel wool pads or scouring powder using a circular motion to assure that scratches are randomly placed making it easy to see pencil lines later. Next I make a border around the edge either with a ruler (to be painted later) or with black electrical tape if I can stand a wider edge. This serves as a ground area. Then lay out the IC's and other large components again. Try to line IC's lengthwise so that power supply lines can be run underneath. Make a pencil trace using a ruler for all supply lines to all IC's. Then accurately position the IC's relative to the supply lines and mark the positions where their "feet" contact the copper with a little dashed pencil line.

The next step is the most important one - you have to start painting. Plunge right in! If you think about it, you have already arranged to mount all the IC's at this point and have arranged for them to get power. All that remains is to connect up all the smaller resistors, capacitors, and other components, which after all is the really interesting part of the circuit anyway. Happily, most of these components are "free jumpers" - that is, they naturally will jump over supply lines and other lines which you add. Somewhere you will probably run into trouble and leave out some connection. Just leave a little "tab" of copper at each end, and remember to run a wire later when you solder on the components. After you have made a board or two, you will find this painting to be much easier. You will find that certain structures (around op-amps for example) are so common that when you see an op-amp on the schematic you will not even have to think about how you are going to paint them. It will always be the same. One especially useful result of this sort of board preparation is that everything is on the same side of the board and is easy to see. Op-amps are mounted as they are usually drawn. You can almost read the board just as you would a schematic, making it easy to check and troubleshoot as well.

Once the board is painted and dry (with the thinned down paint, this is just a matter of a few minutes), you will be ready to etch it. I use ordinary ferric chloride etchant (available at Radio Shack stores or at similar dealers) and dump the whole bottle into one of those plastic freezer dishes (rectangular, about 6" by 8" with a cover) which is used for etching,and storage of the solution for later boards. I find the best way to etch is to just float the board and come back later. I don't heat the etchant or agitate the sulution during etching. It may come as a surprise to some readers that you can float a board on top of the etching solution but all this wonder can be attributed to surface tension of the liquid, and the relatively dense solution (which I find decreases as the etchant is used). By floating the board, I find there is no trouble achieving a good etch within 15 minutes to 1 hour depending on how far you want to push the useful life of the solution. There are two things to watch out for. Be sure when you float the board that you do not get an air bubble under it or you have a spot that does not get etched the first time. One way to prevent this is to tilt the board ever so slightly as it makes contact with the etchant, and then lower it slowly. The other thing is that you should make sure the board does not sink or else it just sits on the bottom with a thin layer of spent etchant against the copper and you have to etch again. Once the board goes down, it will continue to do so unless you remove it and dry it thoroughly before trying to float it again. Why not let it set face up in the bottom of the dish? Well, you can do this with a fresh solution, but with a used solution a sort of "mud" will settle on the copper surface and tend to inhibit the etching process. Also, it is harder to remove a board from the bottom of the dish than from the top. You will probably get some etchant on your hands as you work and this will not hurt you, although you should wash it off. The biggest problem from the ferric chloride sulution is that it tends to stain any porous surface (wood, fingers, cloth) so take proper precautions.

It will be useful to go through a brief example. We will show some sketches of the entire process, but for a very small circuit. Fig. 1 shows the schematic of a simple two op-amp triangle-square oscillator. Normally, this circuit would be part of a larger circuit and would not get its own circuit board, but we want to keep things simple here so that the example remains clear. In Fig. 2, we have selected a circuit board, painted a border around it (for mounting and ground) and have lined in and painted the power supply lines (the positions the op-amps will take are shown with dotted lines). The pins other than the power supply lines are just marked with pencil dots at this point. Fig. 3 shows the remaining paint lines which are put in freehand. The board is ready to etch at this point. Fig. 4 shows the board after etching, cleaning, and mounting of components. Note that two jumper wires are used, and that the components are shown with long leads while in practice you would keep them as short as possible.

The example should make the general ideas clear. In addition to the idea that this is really a simple thing to do, I wish especially to point out the striking similarity between Fig. 1 and Fig. 4, the original schematic and the final circuit board. In fact, you might think of Fig. 4 as a "conducting schematic." I beleive you will find that many of our schematics will easily take on a corresponding circuit board form. Readers who may have been sold on this general idea may want to take a look at more of the suggestions in the MEH and in the application notes listed above (these application notes are part of the "preferred circuit collection" as it is currently being offered).

Other readers may be thinking,"Yes, it's fast and logical, but not neat." If it is necessary to be especially neat, it is possible to improve the technique so that things look better. However, we insist that only one side of the board be used, so if your idea of a neat professional look is a board that looks like one in a commercially made unit, this will not do the trick. Assuming you just want your own equipment to look a little better when you open it up, but in general want to follow this method, we suggest two improvements. First, after painting on all the etch lines, but before etching, take a few moments to neaten things up. This can be done with the paint brush when it is desirable to increase the width of some lines or to neaten up a corner, or with a sharp pointy object for removing minute amounts of resist paint. A good tool here is made from a 1/4" to 3/8" wooden dowel with a small wire brad nail driven in the end. You then file or grind off the head to a sharp point, effectively making

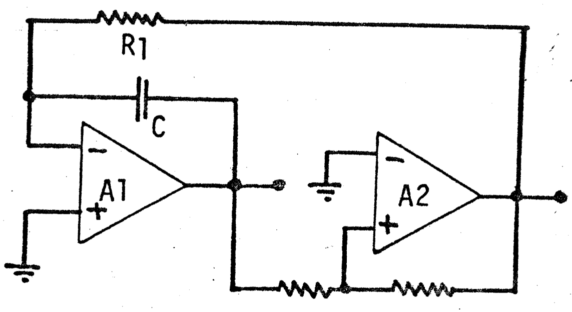


Fig. 1

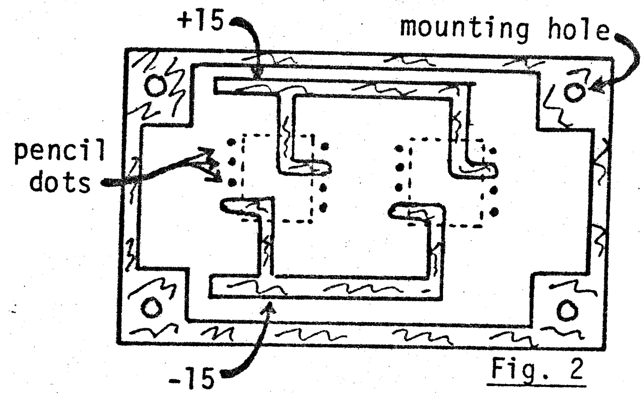


Fig. 2

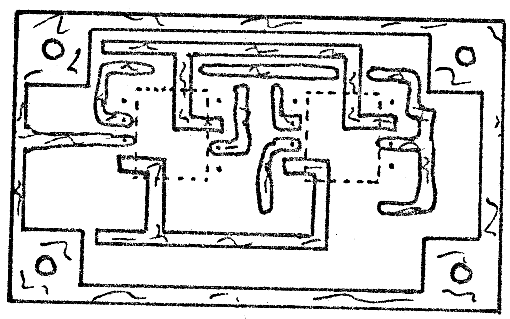


Fig. 3

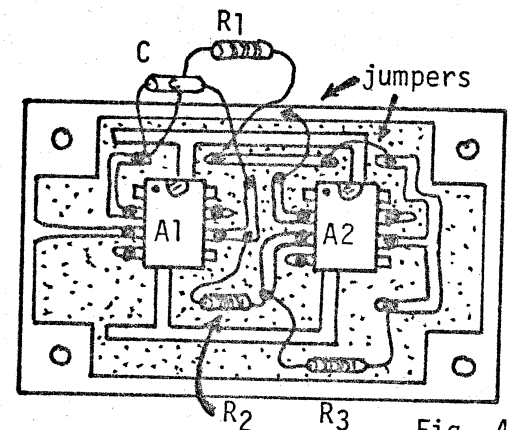
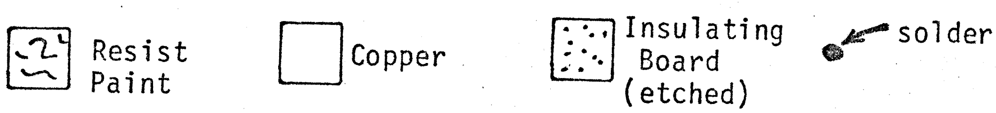


Fig. 4



yourself a "pencil" with iron lead. You can use this freehand to scratch off bits of resist paint (and to "erase" mistakes during the original painting), or you can lay a ruler along a painted line and run the point up and down the edges removing any slight imperfections. Five minutes work here will greatly improve the appearance in many cases. The second improvement is to tin the board. This is often a good idea in any case since it lowers the resistance of the lines and covers any hair-line cracks, as well as making the actual mounting of components easier. To do this, you first clean the etched board thoroughly, and possibly even buff it slightly with toothpaste. This makes it easy to apply a light coat of solder to all copper surfaces. The board is now tinned, in a satisfactory but obvious way. For a more professional look, you might want to try for a "plated" appearance. To do this, I turn the board upside down and remove excess solder by letting it run down on to the hot soldering iron. This removes much of the excess solder and leaves a shiny appearance with possibly grainy looking areas and bits of burned rosin flux. The final trick now is to use steel wool on this board to even the surface, remove the excess shine, and to remove the burned flux. The result is an evenly tinned appearance and a board appearing clean of any excess solder or flux. The components are now mounted in the normal manner, just as they would have been on the bare copper if we had chosen not to tin the board.

I hope this all convinces you that you can, if you wish, do your own boards yourself, and that obtaining boards need not be a major problem. Keep in mind that as long as you are at least partly in the "build it yourself" business, making your own boards is consistent with the rest of what you are doing. In addition to the advantages that have been described above, we should also point out that the preparation of a board is one additional step that helps you to become familiar with the circuit you are building. Since, as we pointed out, the method above is very similar to a redraw of the schematic, preparing your own boards in this way will force you to become aware of all the components in the system and their interconnection.

A High-Ripple VCF For Waveform Animation:

The Synthesis Of "Animated" Sounds - Part 2:

-by Bernie Hutchins

INTRODUCTION TO THIS SECOND PART: Since voltage-controlled filters (VCF's) can be and are used for waveform animation, what remains to be discussed has to do with the exact parameters of the filter to be used and the method of application. Let's review a bit. What can a (linear) filter do to a waveform? It can only change the relative strengths of the harmonics in the waveform, and change their relative phases. In a static condition, the filter only converts one waveform to another. It is only when a dynamic condition is achieved that we get a truly significant improvement from a musical point of view. To do this, either the input waveform frequency must move relative to the filter response, or the response must move relative to the input frequency. Two filter structures are commonly used for these purposes: the standard VCF and the filter bank. Fig. 1 illustrates the use of the standard VCF in what is probably the most common use of a VCF in a synthesizer system. Here the waveform frequency is fixed while the filter's response moves relative to it. Fig. 2 shows the common application of the filter bank where an input waveform's frequency is varied across the features of the filter bank response. In this case, we can think of the input frequency as varying under keyboard control (for example) with the filter bank providing a note-by-note change of harmonic content (a "formant" synthesis process), or the change of input frequency can be periodic and relatively small (e.g., vibrato) in which case the filter bank adds complex structure to the periodic changes. The reader will note that the VCF has a single feature in its response while the filter bank has many features (many peaks), It becomes obvious to inquire about the possibility of using a "voltage-controlled filter bank" so that the filter bank peaks can be made to move relative to the input frequency. Due to the relative difficulty of implementing control structures and associated filters, as compared to the implementation of fixed filters, the voltage-controlled filter bank is a major job, and has only been approached in the past by paralleling available VCF's, tying up available filters and at best giving a poorly defined resultant filter response. The goal here is to achieve a limited version of the voltage-controlled filter bank. This will be a three peak version which will move relative to an input frequency as indicated in Fig. 3.

CHOICE OF FILTER RESPONSE:

It's always nice if you can say exactly why you made the choice you did, but this is not often possible in the investigation of new methods of electronic music. You just have to try something you feel will be representative, and go with that since it may be difficult or time consuming to investigate alternative, but similar methods. Such is the case here. The filter response chosen was 6db ripple, 6th Order Chebyshev (equal ripple). This was chosen for engineering (not musical engineering) reasons. In the first place, the 6th order gives three peaks in the passband, approximating a filter bank. In the second place, the 6db ripple (a 2:1 ratio) gives a reasonable peak-to-valley ratio. The combination of 6th order and 6 db ripple is about as far as we can go without running into possible problems with excessively high Q in at least one section of the filter. Chebyshev response was selected because it is well known and easily described (by the name Chebyshev). Also, the peaks do not fall at simple ratio intervals, and this more complex spacing is something we desire in a filter bank to avoid substantial enhancement of some frequencies as compared to others. So for circuit considerations we choose 6th order 6db Chebyshev and hope we will obtain useful musical results from the resulting device. The 6db ripple Chebyshev filters are not common (this is too much ripple for many applications, but it is exactly this ripple that we want here), but the necessary data is easily derived (see AN-76, 81, 82).

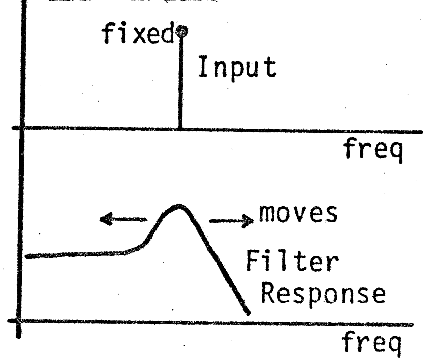


Fig. 1 VCF

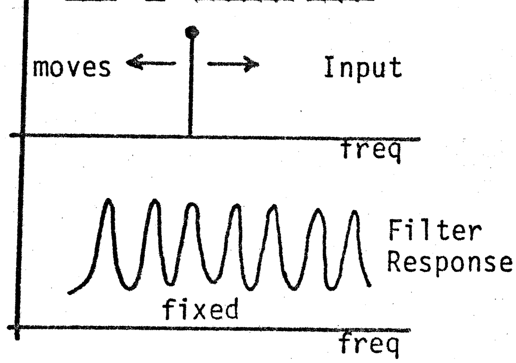


Fig. 2 Filter Bank

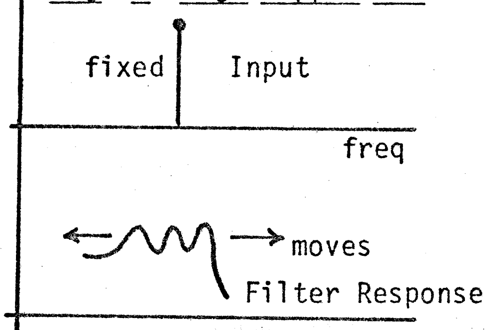


Fig. 3 High-Ripple VCF

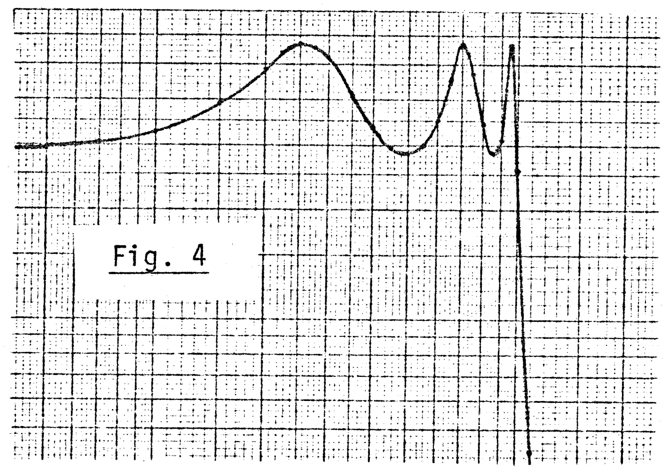


Fig. 4

The 6th Order 6db Ripple Chebyshev response is shown as a log-log plot in Fig. 4. This response is a calculated response based on the pole data of AN-82, using the calculator program of AN-77.

OBTAINING DESIGN DATA

According to the data from AN-82, the poles of the 6th order 6db Chebyshev are:

$$\begin{array}{l} -0.0884 + 0.2588\,j \\ -0.0884 - 0.2588\,j \\ -0.0647 + 0.7071\,j \\ -0.0647 - 0.7071\,j \\ -0.0237 + 0.9659\,j \\ -0.0237 - 0.9659\,j \end{array} \qquad (1a,b,c,d,e,f)$$

These six poles correspond to three second order denominators. Obtaining these denominators is a simple matter of multiplying the pairs of complex conjugate poles together. For a pole at $-\alpha + \beta j$, with corresponding complex conjugate pole at $-\alpha - \beta j$, the quadratic factor is:

$$s^2 + 2\alpha s + (\alpha^2 + \beta^2) \qquad (2)$$

A typical or "standard form" for a second-order denominator would be:

$$s^2 + (D/\tau)\,s + 1/\tau^2 \qquad (3)$$

where D is the damping (=1/Q) and τ is a characteristic time constant. We will be interested in obtaining from the pole data the relative spacing of pole frequencies of three second-order sections, and the required damping values for the three sections. This is not at all difficult and is a matter of using equations (1), (2), and (3) above and doing the required bookkeeping. Plugging the values from the pole expressions (1) into equation (2) we get the three denominators:

$$s^2 + 0.1768\,s + 0.074792 \quad (4a)$$

$$s^2 + 0.1294\,s + 0.504176 \quad (4b)$$

$$s^2 + 0.0474\,s + 0.933524 \quad (4c)$$

By comparing equations (2) and (3) we can obtain the damping D (= 1/Q) as:

$$D = 1/Q = 2\tau\alpha = 2\alpha/\sqrt{\alpha^2+\beta^2} \quad (5)$$

and the damping values are:

$$D_1 = 0.6465 \quad (6a)$$
$$D_2 = 0.1822 \quad (6b)$$
$$D_3 = 0.0491 \quad (6c)$$

The pole frequencies are related to $1/\tau$ and are thus to appear at the ratios:

$$F_1 = 0.27348 \quad (7a)$$
$$F_2 = 0.71005 \quad (7b)$$
$$F_3 = 0.96619 \quad (7c)$$

We are thus able to realize a network for the 6th order 6db Chebyshev by realizing three second order sections and cascading them; the frequencies are given by (7a), (7b), and (7c) while the corresponding dampings are given by (6a), (6b), and (6c) respectively. All this is quite standard and can be found in many texts on active filters. The purpose in giving it here is to provide a numerical example which a reader could use as a model should he wish to try a different filter from among the ones listed in AN-82.

SELECTING A FILTER CONFIGURATION:

It will come as no surprise to our readers that we will be using the standard state-variable filter section as the basis for each of our three sections of the sixth order filter. This is chosen for the ease of implementing voltage control, and because it is easy to achieve the relatively high Q's we need with this configuration. The control elements will be the CA3080 OTA, and our standard VCF methods will be followed, for the most part. One place where we shall have to make a revision is that each of the three sections has a different pole frequency, and thus each pair of CA3080's has to have a different equivalent resistance. This could be achieved by supplying different control currents to each pair, but since we would like to use only one exponential current source (with mirrored currents for the three sections), it is convenient if we leave all currents the same. The equivalent resistance is then changed by altering the effective input voltage to the CA3080. This is a simple matter of changing the input attenuator. Our standard input attenuator is 100k to 220 ohms, limiting our 5 volt signal levels to about 11 mV at the actual input terminal of the CA3080, assuring essentially linear response of the input stage. We know however that we don't generally get an unacceptable level of non-linearity if we allow this input level to reach 20 mV, and the signal-to-noise ratio does not become unacceptable if we lower this level to 5 mV, so the approximately 4:1 ratio of pole frequencies in equations (7a), (7b), and (7c) can be accomidated by changing the attenuation to an appropriate value (we will change the 220 ohm resistor here). We could also have done the tuning of the sections by changing the integrating capacitors, but for 6th order, we would like to use 1% or 2% capacitors, and getting these in the proper values would be a problem. Here we can use all six capacitors in the same 1% value, and this particular value could take any value in about a 10:1 range.

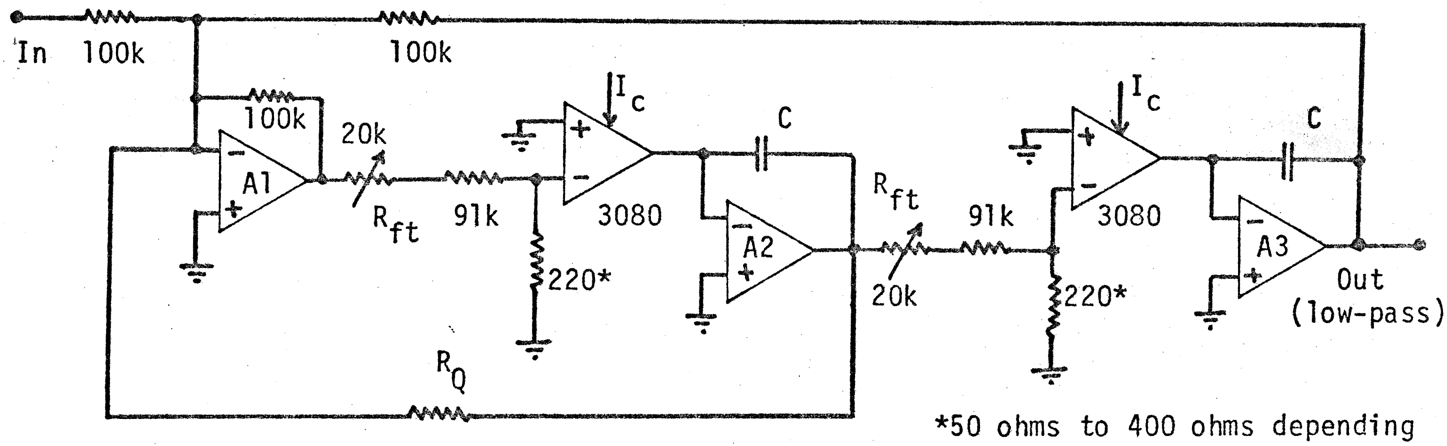


Fig. 5

A typical state-variable voltage-controlled section is shown in Fig. 5 where the frequency control measures we suggested have been implemented. The first of these steps was to select an appropriate target value for the two resistors marked 220 ohms. Once this value is selected, a final trim of the frequency can be made experimentally using the two R_{ft} (frequency trim) trim pots which vary the nominal 100k input leg of the attenuator by ±10k approximately. I suggest that these be simple one turn trimmers as these give enough resolution and have the advantage that the two trimmers can be adjusted visually so that they always have approximately the same resistance. That is, if we want to raise the experimentally measured pole frequency, we would decrease the resistance of both Rft trimmers but at the same time try to keep both at approximately the same setting.

To finish the design of the filter sections, we have only to specify the value for R_Q, which is very simple in this case due to the fact that by using the (-) input of the CA3080 control elements, we achieve a positive integrator instead of the negative integrators of the fixed designs. The value of R_Q is related to the damping (see AN-11) as 100k/D, where the appropriate D is chosen from (6a), (6b), or (6c).

APPLICATIONS:

This surely seems like a strange place to discuss applications, since we have not yet even concluded the design stage, but it is necessary here so that we can justify certain additional circuitry that we want to add to the total schematic. We have two main applications in mind: a more or less standard extension of the standard VCF technique, and a moving filter bank application (roughly, Fig. 1 and Fig. 3). The additional circuitry involves the exponential control stage, which the reader will find familiar and expected, and a sort of multiple low-frequency control oscillator.

For a moment, let us talk about low frequency control. When we use a single oscillator (let's say a 1 Hz triangle to be specific) to control a synthesis parameter, the result is a variation that is easy to follow. If we mix two different triangles and use this as a control, it is less easy to follow, but still some patterns of variation are evident in the controlled output. If we use a great number of low-frequency oscillators and sum their outputs for a single control, we expect that all perceivable patterns will be absent (and some other strange effects will also be present). What about three oscillators, or eight, or any other number in this general range? Experience shows that three oscillators can be enough to give a sum that is lacking in obvious patterns, especially when the controlled effect is somewhat subtle (for example, the control of upper harmonic content is subtle compared with the control of fundamental pitch). Yet, in the first part of this series (EN#87) we described a Multi-Phase Waveform Animator that used a mixture of eight phase shifted sawtooth waves all under the control of a separate low-frequency oscillator. I don't believe this works anywhere near as well with only three sections. So why is three enough in one case while eight seems to be required in another case? The difference appears to be that in one case (the multi-phase mix) the eight sections are used in a tone enrichment process while in the other (subtle low-frequency control) three sections are enough to confuse the listener about the patterns that are actually present. Thus we can mix low-frequency control effects either to enrich a tone, or to confuse the listener. By confuse, we mean just that the pattern should not seem too regular, and yet it should not be random either. We seek a middle ground that seems to be satisfied by three, or possible a little better with four low-frequency oscillators. Keep in mind that the mixture of a regular pattern (such as a oscillator waveform) with a random pattern (such as white noise) gives a resulting sound that sounds like a mixture, not like a relatively pure tone with interesting variations. The use of random effects is not a means of turning regular electronic sounds into musical ones. This comment applies even when the random effect is introduced through a control means and not as a direct addition to the original tone. Thus, we look on the mixture of low-frequency control signals as a means of avoiding pure (detectable) regularity and yet, of avoiding obvious random effects. Some readers may find it interesting to compare the effects achieved with a mixture of three or four low-frequency oscillators with the effects achieved with a low-frequency random voltage.

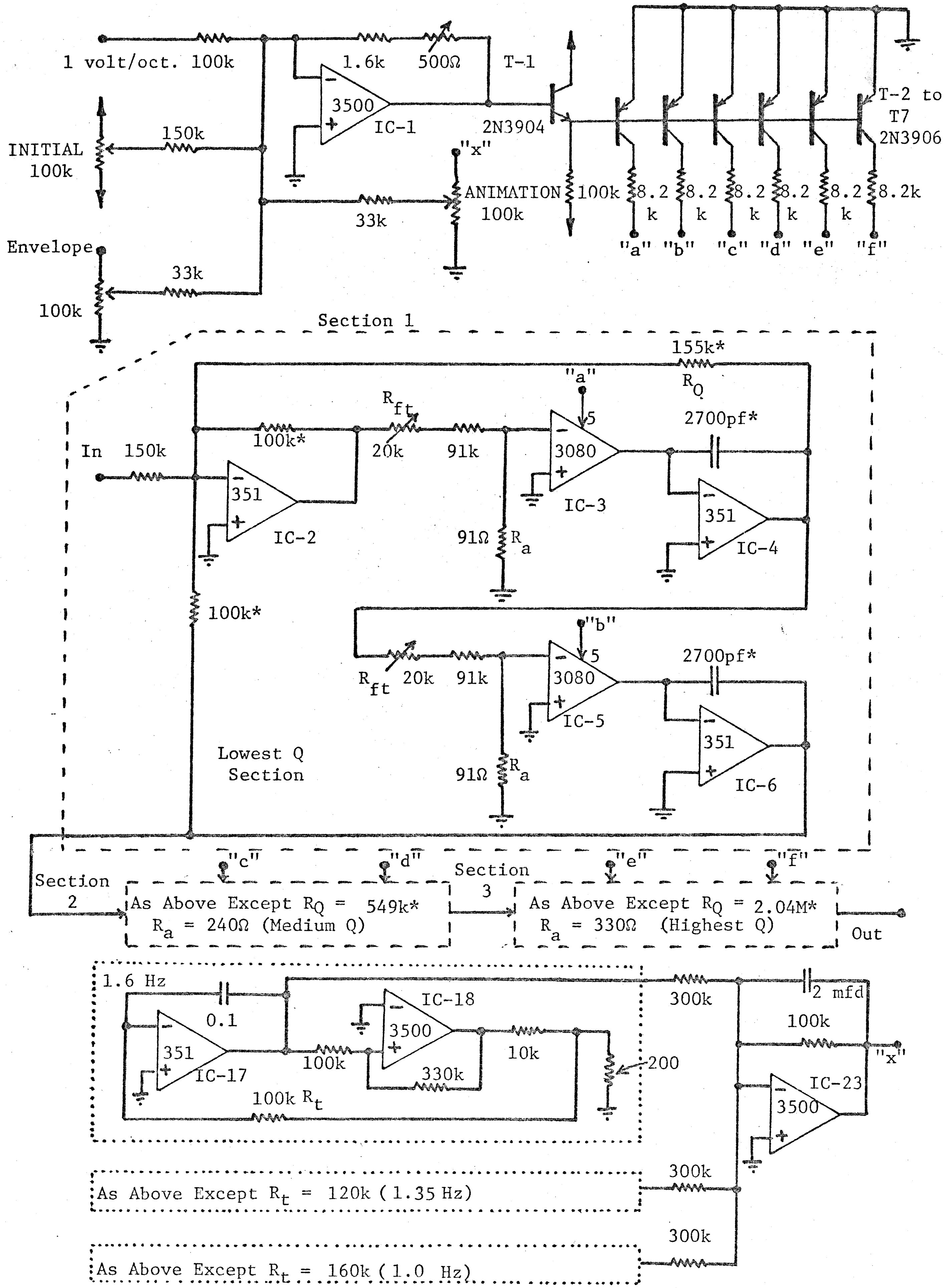
To get back now to the present device, we use the discussion above to justify the inclusion of a three section low-frequency oscillator bank so that a control signal is internally available (and can be made available for external use as well) to control the frequency of the 6th order 6db filter. This will be used as an animation scheme. For example, the filter is initially set so that the ripples are in the region of the harmonics of a sawtooth wave. The filter tracks the sawtooth at the preset interval. In addition, the three component low-frequency signal is appled to vary the filter frequency relative to the sawtooth, providing a timbre modulation without obvious periodicity. This is what we have in mind when we speak of animation of a waveform.

<u>THE COMPLETE CIRCUIT</u>:

The full circuit diagram of the High-Ripple VCF Animator is shown in Fig. 6. As a result of previous discussions, nothing here should be much of a surprise. We show the circuit a little rough, and have avoided some standard refinements such as matched monolithic transistors, temperature compensating resistors, fine initial frequency controls, control voltage reject pots on the CA3080's, and other such "frills" that we find on our better VCF's, but which we feel are less justified in this "animator" device. Of course, if you prefer, you can add these in. The one thing that is critical here is the setting of the pole frequencies of the sections and of the Q-determining resistors (R_Q). For this reason, certain resistors are shown with a (*) indicating that they should be 1% resistors. I used 5% resistors selected to 1%. The capacitors in the VCF sections (as on IC-4 and IC-6) should be 1% tolerance, but any value in the range of 400 pf to 4000 pf should work out. All six capacitors of this type should be the same.

IC-1 along with T-1 forms the exponential converter, with transistors T-2 to T-7 performing the major part of the temperature compensating while at the same time providing six nominally equal currents through nodes "a" through "f" to control the CA3080's. The 500 ohm trim pot should be sufficient for adjusting the filter to a standard 1 volt/octave response. If you have a good supply of 2N3906's and have what you consider a reasonable means of matching them, go ahead and match T-2 through T-7. You can't do much worse than you will by scooping them up. The six 8.2k resistors are just current limiting (protection) resistors which perform no real function except limiting the control currents to the CA3080's in the event of an unusual failure.

Fig. 6 The High-Ripple VCF



The actual filter consists of three sections of the type shown in Fig. 5. Section 1 realizes the lowest Q pole pair (1a and 1b, or 4a), while section 2 realizes the medium Q pair (1c and 1d, or 4b) and section 3 realizes the highest Q pair (1e and 1f, or 4c). The equivalent resistance of the CA3080 is determined by the input attenuation and the control current. Since the control current is going to be the same for all six CA3080's, and all capacitors in the integrators (as in IC-4 and IC-6) are the same, we will be achieving different pole frequencies by changing the attenuation. The equivalent resistance of the CA3080 is inversely proportional to the attenuation. The characteristic frequency of the integrator is also inversely proportional to the equivalent resistance. Thus we can increase the characteristic frequency of the voltage-controlled integrators (and thereby change the pole frequency of the state-variable section) by increasing R_a, or by decreasing the total resistance of the R_{ft}-91k combination. We will use the R_a adjustment as the coarse setting of frequency which can then be trimmed by R_{ft}. Thus, the three values of the R_a resistors for the three sections are set at 91 ohms, 240 ohms, and 330 ohms, approximately in the ratios as given by equations (7a), (7b), and (7c). Since these can be trimmed, 5% resistors can be used for the R_a type resistors. The values for R_Q are easily found from equations (6a), (6b), and (6c) and are given by 100k/D as 155k, 549k, and 2.04M. I suggest that you select these resistors carefully, or otherwise arrange that the proper summing ratio at IC-2 (and in the two sections below) is properly set. The reason you should give this some care is that once done, you can forget it and will be able to concentrate on the trimming of the pole frequencies knowing that the damping is set. Trimming will be a matter of setting three numbers, and not six as you would have if the proper damping was left to a trimmer rather than to a predetermined fixed ratio. In general, it is much more difficult to experimentally set a damping factor (set the Q) than it is to set a frequency. Not shown are the full details of section 2 (IC-7 through IC-11) and section 3 (IC-12 - IC-16) which are similar to section 1, with the exceptions noted in the dashed boxes.

IC-17 and IC-18 form one of three low-frequency oscillators which are summed by IC-23. This is the standard Integrator-Schmitt-Trigger triangle-square oscillator (with output attenuator on the square wave to make it possible to use smaller capacitors at low-frequencies - see the discussion of this oscillator in EN#87 on the MPWA). The frequencies are set at 1.6 Hz, 1.35 Hz, and 1.0 Hz and were determined experimentally. IC-23 is a summer and low-pass filter combination. The cutoff frequency of this low-pass is a little below 1 Hz. You can try the summer with and without this low-pass 2mfd capacitor. I felt it had a smoother and more natural response with the filter capacitor in since this took off the sharp points of the triangles. If you should be unlucky, you might hit on a set of resistors (R_t) that happen to give undesirable patterns. If things seem a little too regular, try adding a series resistor of about 3k to one of the R_t resistors. Not shown in full detail are the second two oscillators (1.35 Hz, IC-19 and IC-20) and (1.0 Hz, IC-21 and IC-22), which are similar to the first oscillator with the exceptions noted in the dotted boxes.

Before we go on to discuss the actual tuning and checking of the filter, we should say a few additional things about the construction. First note that the op-amp listed as 3500 is the Burr-Brown 3500 which we happen to have on hand as a surplus item at the present time. Other op-amps that will work here in place of the 3500 include the 307 and the 351. Also, instead of the 351 you can use the CA3140 in places specifying the 351 in Fig. 6. The layout of the circuit is not critical, but there is one fairly high-Q section (section 3) which may require some care. This is particularly true because of the Schmitt trigger switching op-amps (IC-18, etc.) which may glitch the supply lines. This can cause a ringing of section 3, which is evident as a patternless ringing at the output of the filter with no input attached. I had this problem and got rid of it with two 0.02 mfd capacitors bypassing the power supply lines between the filter section and the oscillators. This cured the problem, but I put in four more 0.02 mfd caps as well just to be sure. This sort of precaution is standard even when not mentioned, but here is a real case where you can see that it is actually necessary!

TUNING UP AND CHECKING OUT THE FILTER-ANIMATOR:

Let's assume you have built the filter and applied power and a test signal with trim pots originally centered. You have also found that the filter does function as a low-pass and that the voltage-control section does tune the low-frequency cutoff. If you carefully check the response you will probably find two ripples and possibly a third weak one. This is to be expected - a filter of this low damping and high order is not going to come up perfect the first time. In checking out the fine tuning, make sure the animation control is off (wiper at ground). There are now two possible ways of tuning up the required response: the trial and error method, and the careful step by step verification approach. Trial and error may give you a satisfactory approximation to equi-ripple within a few minutes, but if you want to know that you have things tuned right, a systematic approach is required. For this reason, we will be giving here a good deal of calculated and experimentally verified data. This sort of filter can be a real nightmare if you are not sure what the component parts of the response are supposed to look like, and only know the final desired response.

First we give the response of the three sections individually. To test these, wire the three inputs in parallel, all to the same test sine wave, and allow for a means of controlling the input amplitude of this sine wave. Fig. 7 shows the three response functions. Fig. 7a shows the response of section 1 of Fig. 6, while figures 7b and 7c show the response of sections 2 and 3. Pay the most attention to the solid lines as these are the calculated results while the dots represent experimental points that were measured in my particular circuit. If you have any problems with your setup, you should compare the individual section responses with these. Note that the DC gain of all three sections is just 0.667, as determined by the 150k to 100k feedback of the input stage. When comparing your response with the calculated ones given here, you can check the whole curve, but it will be almost as good just to verify that the DC gain is 0.667, and then find the peak gain. These peak gains should be 1.09, 3.67, and 13.6 for sections 1, 2, and 3 respectively. (It is because the third stage has a gain of 13.6 that you will have to cut the input amplitude back a little when measuring this stage separately.) If the peak gains are off from what they should be, you might suppose that you should adjust the resistor R_Q for the section. However, we suggest that you leave that alone, and try to get the peak gain right by changing one of the controlling transistors T2 through T7, making sure that the one you are changing is appropriate to the section being tested. For example, if section 2 is off, try changing T4 or T5 with an extra 2N3906 to see if you get a better response. With a few tries, it should come out pretty well. This procedure is useful because it not only gives the correct damping, but also gives us some assurance that the two transistors for the section are reasonably well matched.

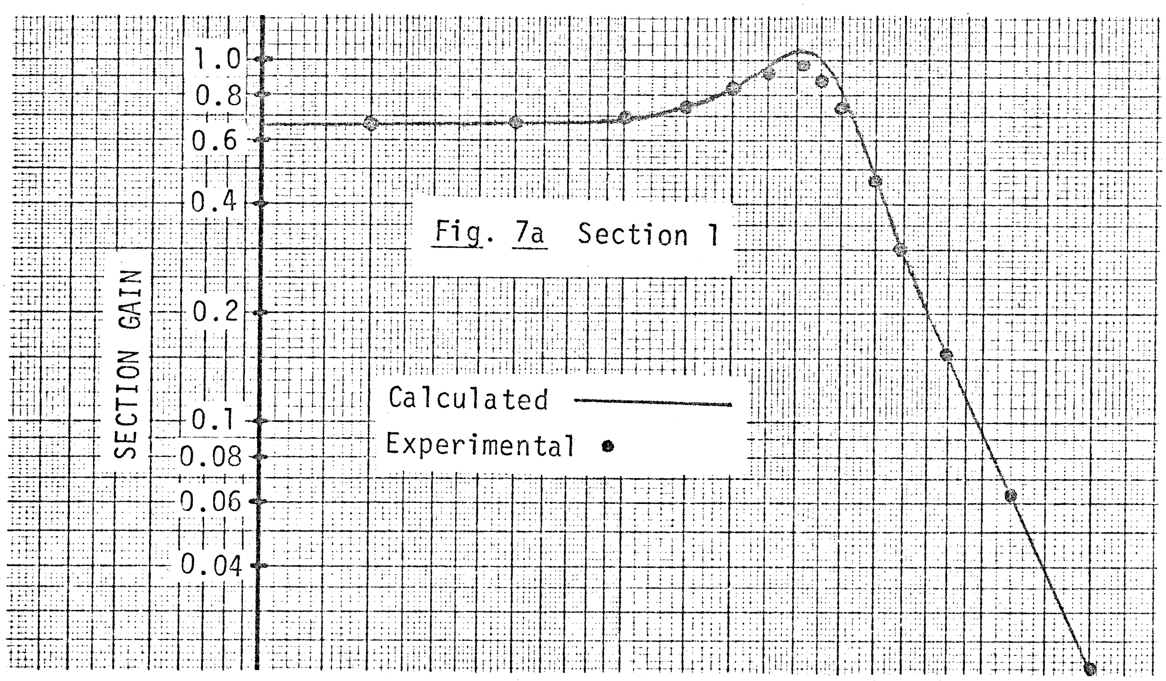
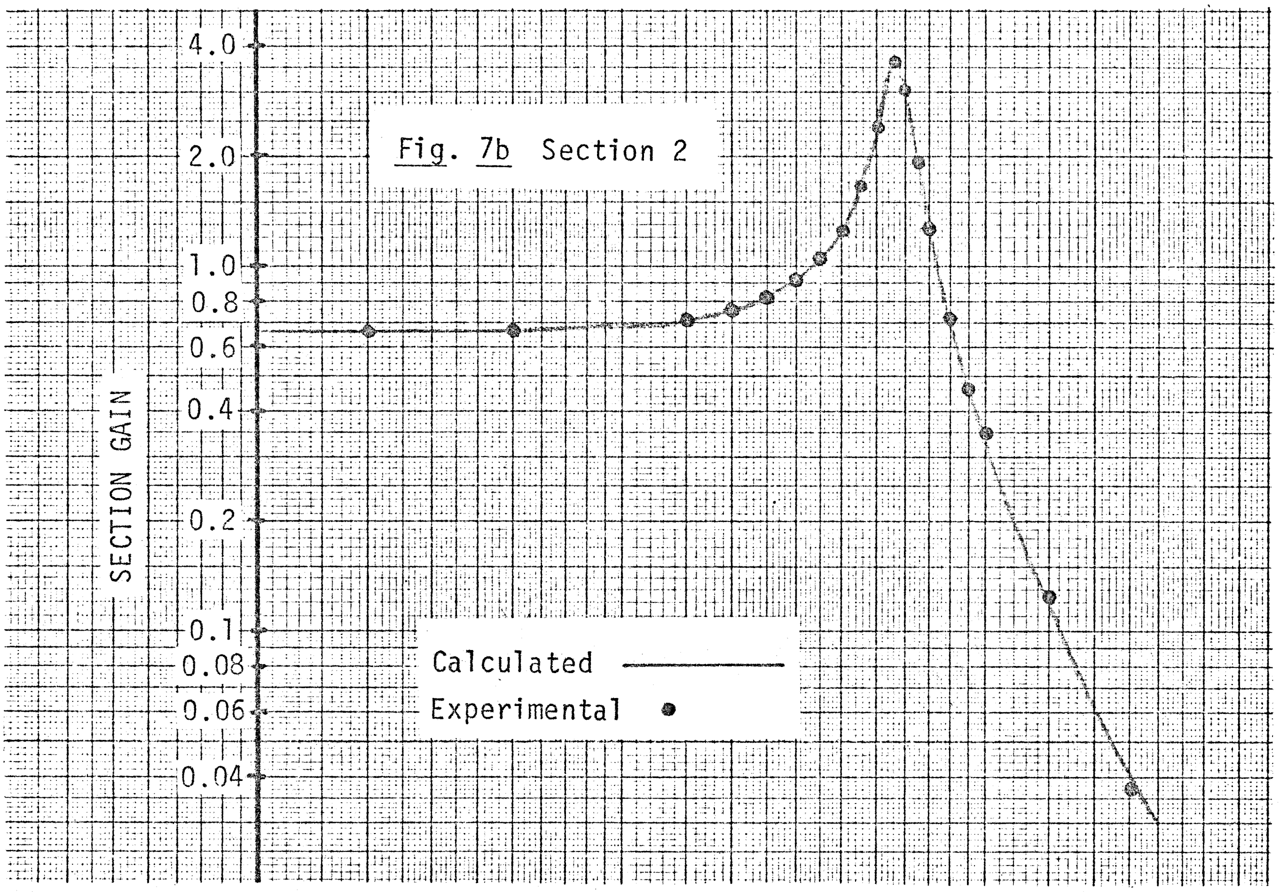
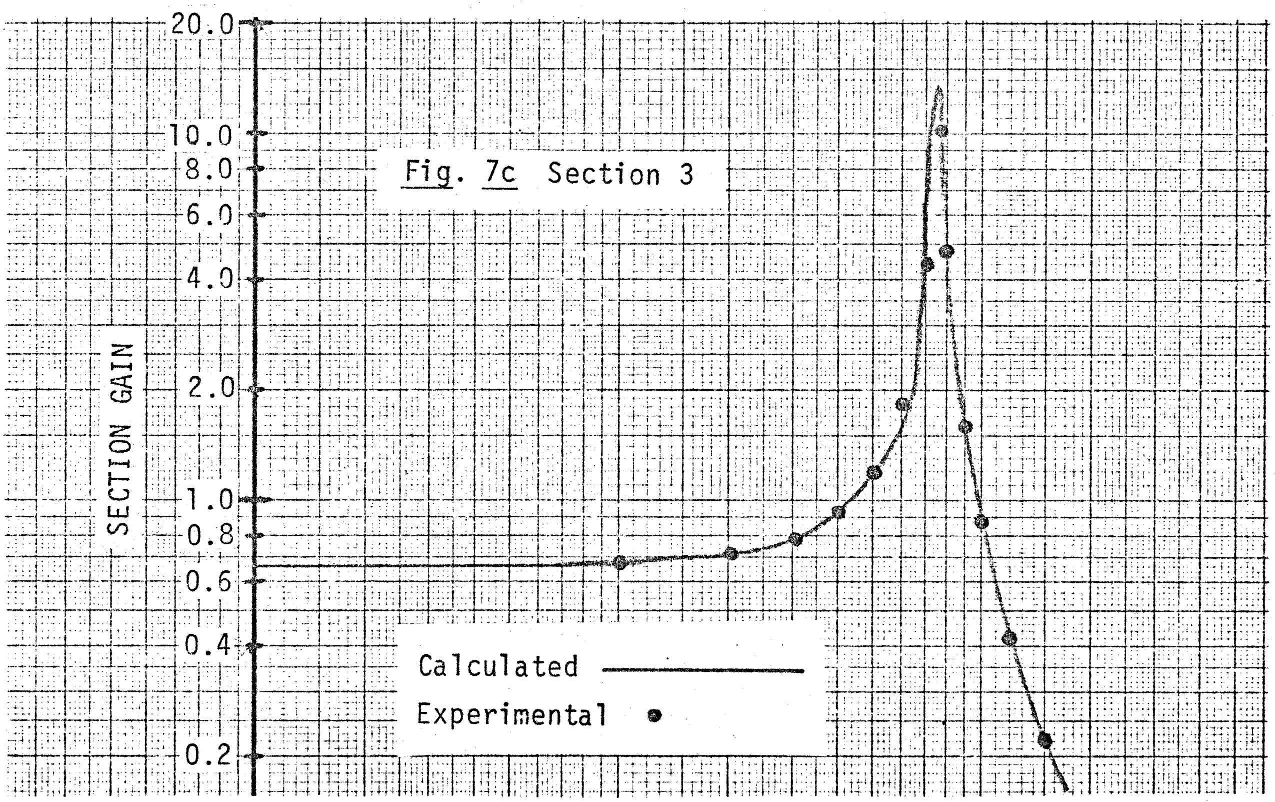


Fig. 7a Section 1





Once the damping is set correctly by achieving the proper DC gain to peak gain ratio, it is necessary to adjust the frequency at which the peak occurs to a proper value. This is best done on each section individually since the interaction that is present if you connect all three sections in series at this point is likely to be confusing. Thus you should leave the three sections in parallel, and adjust the peaks to the values of the pole frequencies. Now, note that we have said nothing about the actual frequencies that we are using to this point (no frequencies are given in Fig. 7). This is because all frequencies are changed by the voltage-control. It is good practice to set up this filter using frequencies in the range of 100 Hz to 1000 Hz, but in the end, it all depends on the control voltage, so we are interested in setting the frequencies of the peaks at a certain ratio, and not at any specific values. The ratios are given by equations (7a), (7b), and (7c) which we can normalize to the highest pole frequency as:

$$0.283 \qquad 0.735 \qquad 1.000 \tag{8}$$

where these numbers are obtained by dividing those in equations (7) by the largest value present, which is 0.96619. Thus, for example, if we set the peak frequency of section 3 to 1000 Hz, then section 2 should be set to 735 Hz and section 1 to 283 Hz. [Note that because of the high Q of the poles, this ratio is essentially the same as the peaks in the total response, which is essentially the same as the imaginary parts of the pole positions.] The final part of the tuning job reduces to the problem of setting the peak frequencies of the three sections to a ratio as given in (8) above.

With this job done, you connect the three sections in series, and you should observe at least some approximation to a 6th order 6db ripple response. That is, there should be three peaks with a peak to valley ratio of 2:1. If you don't have a respons that satisfies you, recheck the tuning procedure. You may well find that the process of tuning the frequencies has upset the damping of the individual sections slightly. These are in fact interactive due to the fact that we can not set both trimmers to the exact same rotation as we change the frequency. Here is what to do now. First, suppose the damping is a little too low, which you will observe as too large a value of peak gain to DC gain. You want to increase the damping, and this damping is proportional to the inverse of the feedback through R_Q. We don't want to mess with R_Q, but we can increase the damping by increasing the gain of the first integrator slightly, and the gain of any integrator can be increased by lowering its characteristi frequency. Thus, if you increase the value of R_{ft} slightly (the R_{ft} associated with the first integrator, IC-3 and IC-4 for the first section, etc.) you can restore the proper damping. This lowers the characteristic frequency of the total section, so to compensate, decrease slightly the resistance of the R_{ft} trimmer associated with the integrator IC-5 and IC-6. All this explanation above is for the benefit of those who really care, and to give some insight into why the switching around of different transistors can result in the proper Q. [In rare cases, it may also be necessary to increase or decrease the value of the resistors marked 91k in one section or another.] For the reader just interested in final tuning of the filter, the above paragraph can be reduced to:

Peak gain to DC gain too high:	Increase the left R_{ft} trimmer a hair Decrease the right R_{ft} trimmer a hair Recheck damping and peak frequency.
Peak gain to DC gain too low:	Decrease the left R_{ft} trimmer a hair Increase the right R_{ft} trimmer a hair Recheck damping and peak frequency

As you can see, the tuning procedure is a little complicated, but is straight-forward if you keep your wits about you. Resist at all cost the temptation to adjust at random, or you will loose everything and have to start over. Doing this right is just a matter of knowing what you are supposed to do, and doing each step with a lot of care and forethought.

As a final check, the total response of the system can be checked against the calculated curve, and such a curve is shown in Fig. 8. The calculated and experimental fit is quite good here, but is probably about optimum. The response curve can be expected to vary some as the frequency of the VCF is changed. In any case, since we had no real reason to demand the ideal response in the first place, your best effort will probably serve well.

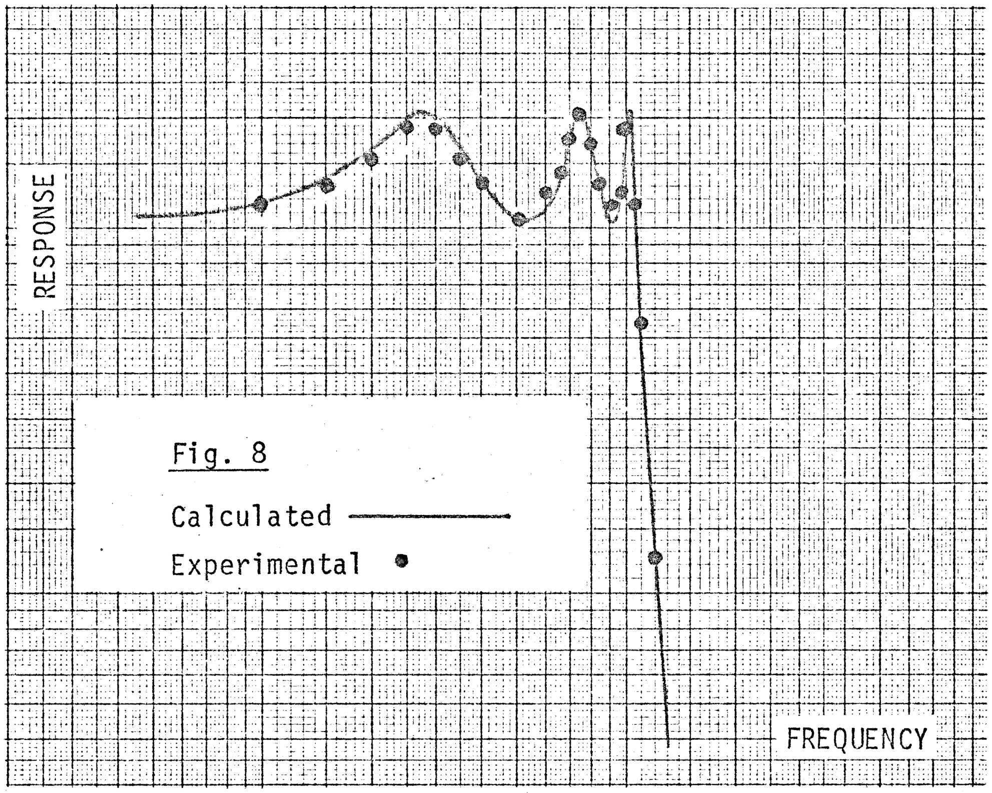


Fig. 8

Before ending this discussion of the tune up, it might be a good idea to list in a table the parameters that you need for the process as this will save going through the text as you work at your test bench. The table is given below:

SECTION	DAMPING	PEAK GAIN TO DC GAIN	PEAK FREQUENCY REL. TO TOP	PEAK IN TOTAL RESPONSE DUE TO THIS SECTION
1	0.6465	1.63:1	0.283	0.268
2	0.1822	5.51:1	0.735	0.732
3	0.0491	20.39:1	1.000	1.000

EVALUATION:

There are several areas of discussion that will be appropriate to our evaluation of this filter as a viable electronic music module. These will include the application as a standard VCF (taking advantage of the sharp 6th order cutoff), the use as an animator of upper harmonics, and some special uses for vocal effects. We can begin by saying that it is felt that all the results so far indicate that this is a good choice when you decide to add another VCF to your system.

In the first case, the 6th order 6db ripple VCF is suitable for standard VCF applications (Fig. 3). This is the first time we have had an opportunity to test a 6th order VCF that has a sharp cutoff (the variable slope VCF of EN#72 was 6th order, but composed of all first order sections and had a very weak initial cutoff). VCF effects, like must electronic music effects, are difficult to describe in words, but it can be said that the dynamic processing of the harmonics of the input waveform, as achieved with this filter, is at least as great as that achieved with a corner peaked 4th order filter, and certainly greater than that of a second-order state variable filter. In this sort of application, it is difficult to tell if the effect is achieved due to the ripple in the passband, the very sharp initial rolloff, or to the 6th order 36db/octave final rolloff. Of course, we suspect that it is a combination of these factors, but we are not sure which of these makes the system different from the corner peaked 4th order filter, and there is a clear (but hard to describe) difference between the two, although the overall dynamic effect might be judged to be of the same order.

In the case the VCF is used an an animator, the lowest ripple is placed relative to the input frequency so that it falls on the second or third harmonic of the input. Such a situation is represented in Fig. 9. We can imagine that the input spectrum is stationary (as during a single tone) and that the VCF response, under the control of the mixture of three low-frequency oscillators moves over the input spectrum thus shaping it in a time dependent fashion. Actually, the effect is not as dramatic as it might appear from Fig. 9, or rather we should perhaps say that the effect is dramatic, but of two components. The first component of the effect is the strongest and is due simply to the motion of the filter, as a low-pass, allowing more harmonics or fewer harmonics through. This would result with any low-pass VCF, although the 6th order response here makes it more dramatic. The second component of the effect, the more subtle one, is the effect of the ripple. Suppose for example that the highest ripple, shown centered on the 10th harmonic, varies so that it comes down no lower than the third harmonic. In this case, the third harmonic ripples up and down in amplitude as the filter moves, but it never is cut completely out. This is an interesting effect, but one which is hard to hear because of the upper harmonics jumping in and out. One way to hear it is to add a series low-Q VCF (like state variable) that cuts off at the third harmonic. Thus, even though the upper harmonics are "pinging" in and out, they are blocked by the added VCF, and the more subtle ripple effect is heard. This setup is indicated in Fig. 10. Of course, there is no accounting for what a musician may actually end up using and either animation effect (with or without the second VCF of Fig. 10) may be used.

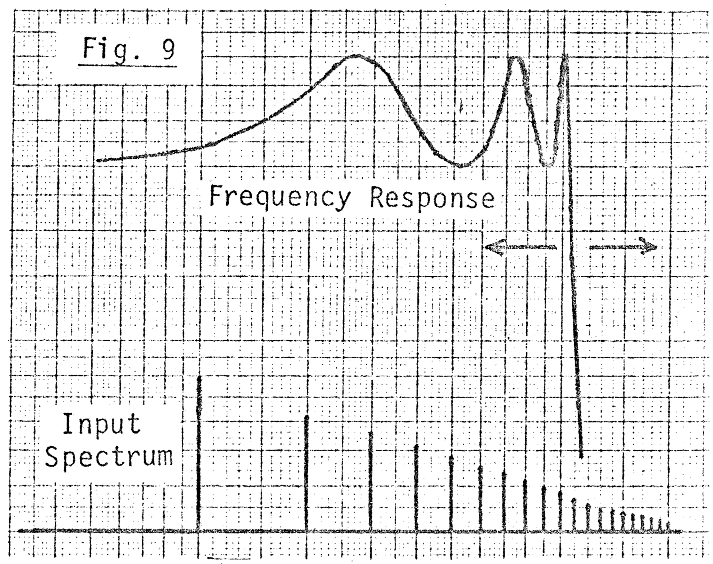
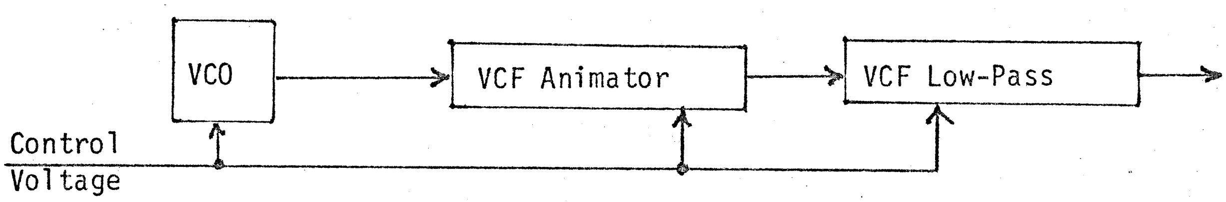


Fig. 10



For the final part of the evaluation, we want to observe that we have three peaks in the VCF response, and that commonly we think of vocal-like sounds as having three peaks, or "formants" as they are called. For an excellent discussion of the synthesis of vocal like sounds, see the series of three columns by Bob Moog in Contemporary Keyboard for April, May, and June 1978. We might wonder if the spacing of the three peaks in the VCF corresponds to any typical vocal sound. In the first of Moog's columns, he gives typical formant frequencies for some sounds. In the table at the right, we have normalized these so that the highest is always 1.00, and we compare these with the VCF peaks. As you can see, the spacing is similar for the VCF and the "eh" sound. Naturally, discovering this it is a reflex action to put in a pulse waveform and see if the output is an "eh" vocal sound. Well it

	"FORMANTS"		
SOURCE	F1	F2	F3
VCF	0.27	0.73	1.00
"ah"	0.31	0.45	1.00
"eh"	0.22	0.74	1.00
"ee"	0.09	0.76	1.00
"aw"	0.23	0.35	1.00
"ooh"	0.13	0.38	1.00

sort of is, but is not too strong. Probably this is because the peaks are not of a high enough Q for a good vocal sound. However, you do get a reasonable hint of a vocal sound, and this, as much as anything, should give you some idea about the general sound of the filter. It might be interesting to think of multi-ripple VCF responses in terms of the equivalent vocal sound, as this would probably be more useful for musicians. Of course most musicians would probably get more use out of a description of a VCF as an "eh"-sound VCF than to say that it is 6th Order 6db Ripple Chebyshev.

While this VCF module is not as dynamic an animator as the MPWA described in EN#87, the fact that it can double as a more or less standard VCF at the same time, and has a sort of "brick wall" initial cutoff, makes the whole module a valuable addition to your collection.

* * * * * * * * * * * * * * * *

RECORD REVIEWS:

-by Craig Anderton

From Here to Eternity (Giorgio Moroder; Casablanca) Here is an example of disco electronic music. The sound is spare and sparse; the rhythm track churns like a machine in the background, as somewhat ineffectual vocal and instrumental lead parts fly in and out from time to time. While Kraftwerk-like in sound on the surface, there are some great dissimilarities...what Kraftwerk glorifies, Moroder tends to trivialize. He has the ability to put together hypnotic and intricate rhythm tracks (listen to the production on Donna Summer's hit, "I Feel Love"), but an album of just rhythm tracks gets to be a little wearing after a while. Nice enough to have around, but if you can only afford one record this month I suggest this not be it.

The Man-Machine (Kraftwerk; Capitol) More Kraftwerk for the faithful. While many people consider their music tedious, I think the non-electronically oriented listener may fail to see the constantly changing subtleties occuring throughout a Kraftwerk album. The phase changes, filtering changes, mixing level changes...all of these contribute to a texture that is dynamically in motion, and far from monotonous to my ears. Probably the weakest moment is when Kraftwerk strays into the "pop song" territory ("The Model"); while their intentions are good, when the focus shifts from electronics to vocals something gets lost in the process. Overall, I wish they would use more imaginative processing with the voice parts, but since vocals are not the main feature of the album, it's not too serious a problem.

While somewhat more sombre than "Trans-Europe Express", this album still retains the basic Kraftwerk recipe of drama, whimsey, repetition, electronic percussion, and deceptively simple (yet harmonious) keyboard work. They have the ability to create a sound that is truly original; a sound with a certain power and majesty. As they continue to perfect their style, I suspect they will gain the respect of more and more listeners.

Encore (Tangerine Dream; Virgin) This band built up an incredibly loyal and intense following, but I never really saw the reason for all the adulation. T.D. played good enough music, yes, but it never sounded fully developed or unified to me. As time went on, personal frictions within the group caused enough tensions so that the people on this album are no longer playing together.

However, they went out in style. This live album spotlights the best of their last American concert tour; considering that T.D. was an improvisational band, a live context really is the best way to listen to them. You can hear the parts where the pieces "catch", and all three musicians sound as if they were all plugged in to the same creative source; just as quickly, you can hear a piece decelerate. Overall, the effect is not so much listening to a piece of music, but rather a piece of drama that uses notes instead of dialogue. My major complaint is the noise level of the audience (although mercifully, no one yells "boogie on down" during the quiet parts), but that's about it. With this album I have a much greater insight into the quality that made T.D. unique---I hope future albums under the same name build, and expand, on the best qualities evident in "Encore".

Kosmos (Isao Tomita; RCA) I really admire the sounds Tomita gets, but question the musical applications of those sounds. "Kosmos" is a collection of pieces, some well known, some not so well known, many of them modified extensively by Tomita. We start off with the theme from "Star Wars" (yes, another one) as interpreted by the happy robots choir. From there it gets more interesting, but by this point, it hardly matters; the sonic overload is so heavy my ears just kind of boggled at the strain. There is none of the careful editing and pacing of someone like Jean-Michel Jarre, just effect after effect after effect. I think Tomita is clearly a genius, but I'm not all that sure he's a musician.

...and Then There Were Three (Genesis; Atlantic) While not an electronic music album per se, keyboard players might well enjoy Tony Banks' inventive use of synthesizer and other keyboards. The work is always done in an ensemble context, with a maximum of taste and class. If you liked previous Genesis albums you may have a hard time adjusting to their latest sound, but they continue to make fine, keyboard-based progressive rock music.

Before and After Science (Brian Eno; Island) This album contains some of my favorite qualities of a good album---innovative use of electronic effects, technically correct and tasteful playing, humor, and taking of chances. Side 1 opens with "No One Receiving", with excellent drumming by Phil Collins and preverb on the voice. "Kurt's Rejoinder" has an amazing analog delay line part applied to bass guitar; "King's Lead Hat" is Eno à la New Wave, and is the first song I've ever heard that includes both the words "kilocycles" and "kiloHertz" (how can you dislike a mind like that?). Side 1 is up-tempo, side 2 is more subdued; but all songs are of high quality and sincerity. All in all, a very satisfying album that may break Eno through to a somewhat broader audience.

Heroes (David Bowie; RCA) This second Bowie/Eno collaboration is starkly emotional, densely electronic, and hypnotic in character. The mix emphasizes a rhythmic kind of 1984-style rock and roll in the background, with much electronic processing of instruments and subliminal effects. Bowie's voice is still more that of an actor than a singer, but the clever use of vocal processing helps give his voice an other-worldly quality that makes the record as a whole more successful. In some ways, I enjoyed "Low" (his previous album) more because it was a lighter album; but both albums are thought-provoking, spontaneous, and well-done. The second side of "Heroes" is some of the most successful "free form" music I've heard; I hope Bowie keeps pursuing his current tack for at least another album or two.

* * * * * * * * * * * * * * * * *

READER'S QUESTIONS, Contunued from Page 2

Q: I can understand that the Nyquist criterion is necessary, but how can it be a sufficient condition for high fidelity in digital audio transmission? For example, if one samples a 19 kHz sine wave at 40 kHz, doesn't one introduce a 1 kHz modulation frequency as in Figure A? Can you explain why this doesn't happen?

A: This is a very interesting question. The answer to a question of this type must of necessity involve considerations that are both mathematical and psychoacoustical in nature. In such cases, a feeling about what we think we see in a sketch of the waveform is often misleading. We can look at this question in two ways.

First, let's extend the diagram in Fig. A. The bottom line of Fig. A, extended so that it is about five times as long is shown in Fig. B. This waveform should look a little familiar to you - it looks like the result of a balanced "ring" modulation. It is a 20 kHz square wave with amplitude determined by the sampled value of a sine wave. Note that the dotted line is the "envelope" of the square wave, and is a 1 kHz sine wave. Note also that the amplitude bursts occur at a rate of 2000 per second. We will not be totally rigorous here, and some may call the explanations we give here

Fig. A

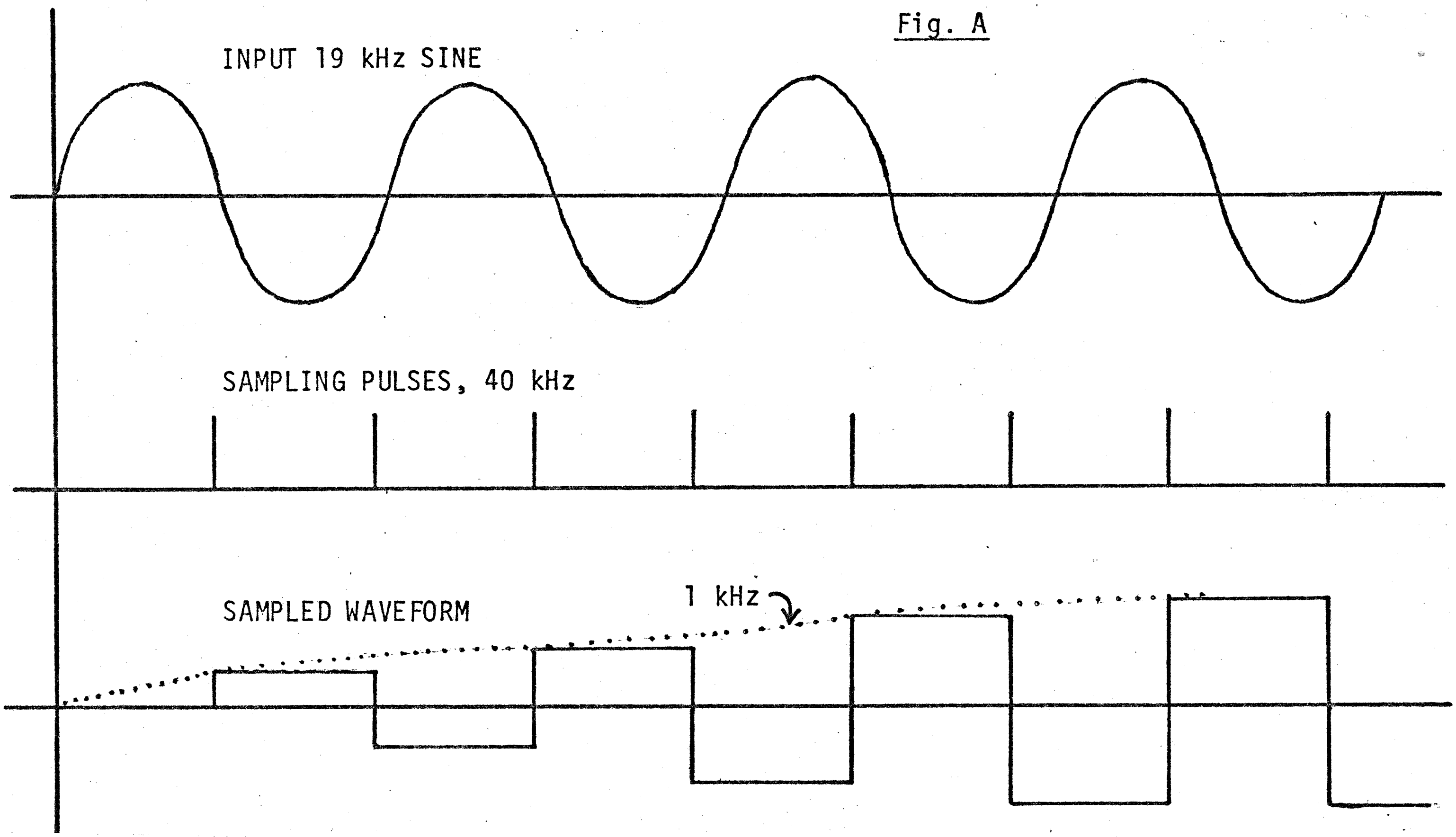
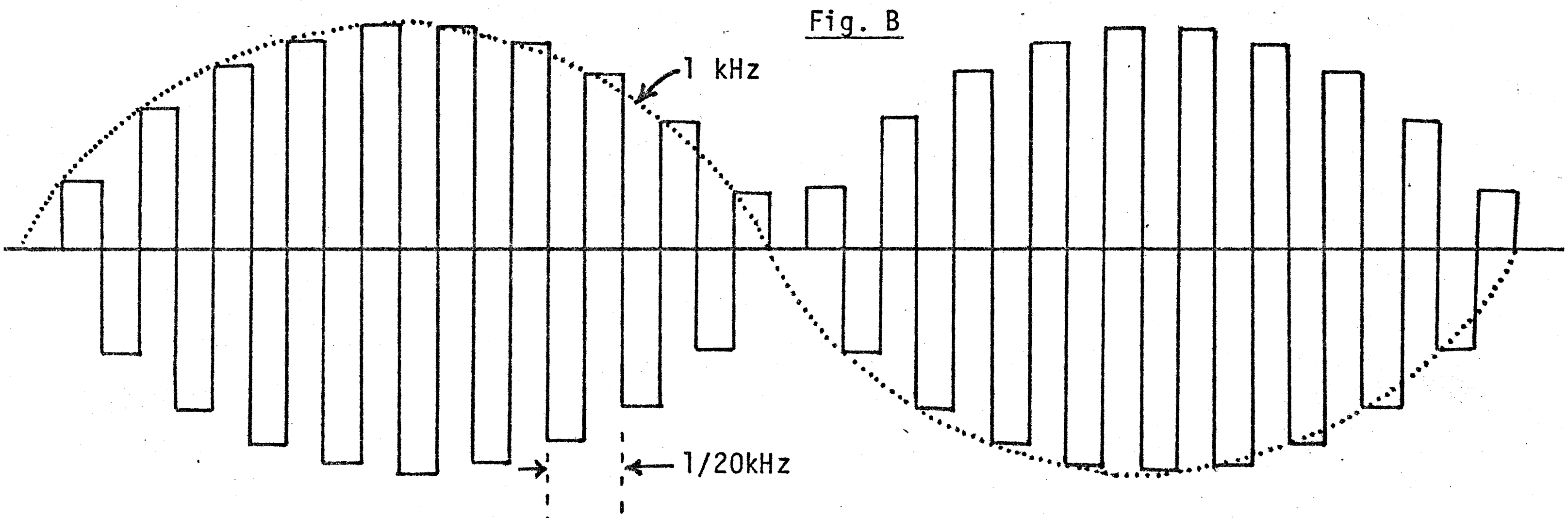


Fig. B



to be "hand waving," but the purpose is just to give a reasonable feeling for why things are the way they are. [First, there is a difference between sampling and sampling-and-holding which we ignore. Secondly, Fig. B is not a balanced modulation between a square wave and a sine wave because if it were, the tops of the resulting waveform would not be flat, but curved. Finally, our analysis will ignore all but the first few spectral components.] We now ask what frequency components would be present if we had a balanced modulation between a 20 kHz <u>sine</u> wave and a 1 Khz sine wave. As you know, or can easily verify using trigonometry, this balanced modulation (which is the multiplication of two waveforms) results in sum and difference frequencies, 20 kHz + 1 kHz = 21 kHz, and 20 kHz - 1 kHz = 19 kHz.

Fig. C

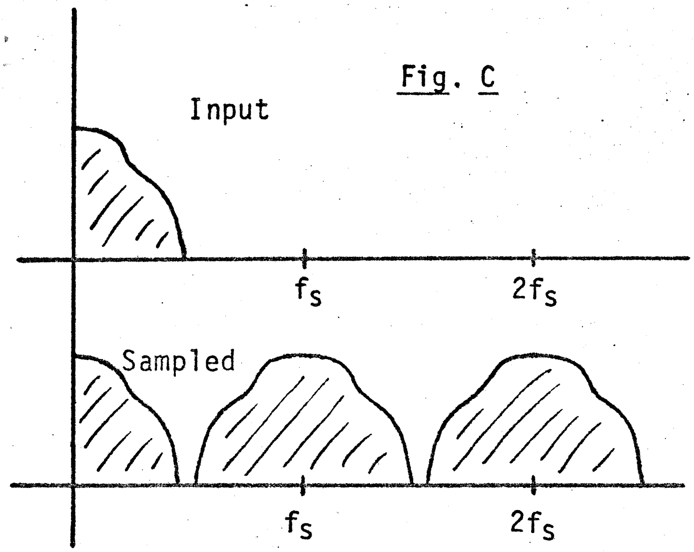
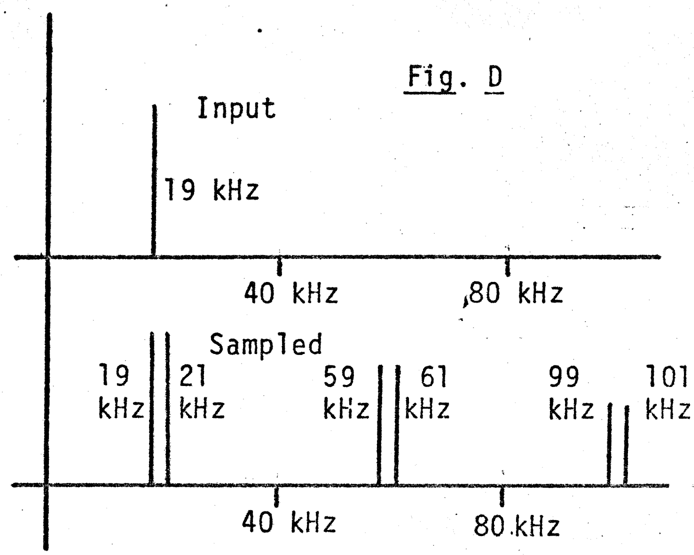


Fig. D



In the second view, we will look at the sampling process from a spectral point of view. It is well established that the sampling process causes the spectrum to spread out and reflect up and down about the sampling frequency, and about multiples of the sampling frequency, as indicated in Figure C, where a contunuous spectrum is shown. In the case of the 19 kHz input sine wave, there is only one component (19 kHz) in the input spectrum. With a 40 kHz sampling frequency, this spectrum is reflected about 40 kHz (down 40 kHz - 19 kHz = 21 kHz; up 40 kHz + 19 kHz = 59 kHz) and reflected about 80 kHz (producing 61 kHz and 99 kHz), and so on. This gives us a better idea about the components in the waveform of Fig. B. It is not difficult to suppose that the same higher components we see in Fig. D will also be present if we extend the balanced modulator ideas of our first view.

We now have two views of the sampling process, both of which indicate that the 1 kHz signal (or the 2 kHz envelope repetition rate) is not present in the sampled spectrum. Since these signals are not present mathematically, they can not be heard by the ear unless generated by some psychoacoustical process from components actually present. Whether or not such a process exists becomes academic however because a digital system will always include a smoothening filter when the digital signal is converted back to analog form, which is necessary if we are going to listen to it. This filter (ideally) has a cutoff of 1/2 the sampling frequency, or 20 kHz in this case. The filter is mathematical, not psychoacoustical, and thus cuts everything our of the spectrum except the 19 kHz component, which is the signal we started out with. [I tried the experiment of mixing 19 kHz and 21 kHz sine waves and listening to the result. Played through an amplifier and speaker I heard nothing. Perhaps the amplifier's frequency response blocks both tones from ever reaching the speaker. I next drove a set of earphones directly from an op-amp with 1000k series resistance and did hear the 2 kHz beat tone very weakly, but perhaps this was due to a magnetic non-linearity in the earphones. Comments?]

▶ Q: What sort of equipment do I need for actually building and testing out your circuit designs?

A: You of course need a soldering iron and small tools. Some of the small tools need not be of high quality, and in fact I find a standard fingernail clipper to be superior to a diagonal cutting pliers for cutting wire leads, and I find a small piece of sheet metal, folded over and bent until it breaks, to have an edge that is excellent for stripping insulation off a wire without cutting the wire inside.

As far as test equipment goes, I feel that a scope is essential, but it need not be very elaborate. I just use my scope to see what is going on qualitatively. When I want to make a quantitative measurement, I use either my frequency counter (home made -

see EN#40, or you can probably make one even easier with today's special IC's) or my digital meter. You can probably get a satisfactory digital meter for under $100. You can probably use your synthesizer to provide signal sources, so you don't need a signal generator.

If you are just starting out, I think money is better spent building up a stock of parts and on other convenience measures, rather than on, for example, a $1000 scope that has features you may never use.

▶ Q: Which basic components (op-amps, etc.) would you suggest to have on hand for a university course in which we do a limited amount of introductory building and breadboarding?

A: It is probably not too important which op-amp you use for instructional purposes. Possibly something like the 307 is simple and rugged enough (and cheap enough) for use by students with little prior experience. Newer FET op-amps can also be used, but due to stray capacitance problems, these may not be as suitable for breadboarding. I think it is more important that you have available a good supply of 5% carbon resistors in the 1/4 watt size that fits the white breadboard sockets (save a few dollars making do with the 1/2 watt size, and you can ruin your $17 breadboard sockets in one year!). Also, have some capacitors on hand that are better than the ceramic units that are intended only for bypass use and may have a tolerance of -20%+80%. The student has the right to know that a capacitor is close to the correct value. Capacitors should be at least 10% tolerance and 5% and a few 1% capacitors are desirable. As for other components, they should be of good quality and fit the breadboard well. Also, you may want to make some sort of setup for switches and pots. This can be done by mounting these controls on a panel in front of the breadboard, or you may use things like single turn trim pots which have about 3/4" of hookup wire soldered on their terminals so that they fit into the breadboard.

▶ Q: How come you are only offering the preferred circuit collection to regular subscribers and not to everyone?

A: There are several good reasons for this, which I could give you here. Perhaps it will be easier for me to just ask you the question: "Why doesn't Playboy offer an edition with only the pictures?" After thinking about that for a few moments you should have at least six answers, all of which are pretty much the same as our reasons. If you really must have a copy, ask a regular subscriber (the one who told you about the collection?) to order it for you. This is fine with us.

* * * * * * * * * * * * * * * * *

CLASSIFIEDS:

FOR SALE: One Muse Synthesizer 3-4 years old - good condition. Please call evenings (216)-781-3886 or write "TAV", 797 Starkweather Ave., Cleveland, OH 44113
WANTED: PC Boards, photo positives, or art work for ENS-76 VCO option 4, delay module option 1, VCF variable slope option 3, envelope generator option 3, random noise source option 3. Also Jan Hall's analog delay (EN#61), pitch-to-voltage converter by Robert Iodice (EN#84). Jay H. Jones, 2 Anchor Drive #481, Oakland, CA 94608.

ELECTRONOTES, Vol. 10, No. 90 [June 1978](published July 1978)
Published by B.A. Hutchins, 1 Pheasant Lane, Ithaca, NY 14850
Phone: (607)-273-8030
Please use above address and phone for subscription information, technical questions, and for all other business. Best phone hours - Wednesday evenings, 8PM to 2AM EDT. Avoid calling mornings, and avoid Tuesday and Thursday afternoons.