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1. GROUP ANNOUNCEMENTS:

The feature article in this issue is quite long. It presents a 4-pole network that can be used as a 4-pole Low-Pass VCF, and as a quadrature VCO so it should provide the answer to the needs of many persons. A great deal of analysis and design has been included, both because some comments on filter theory are long overdue, and as an experiment. We would be interested to hear from readers as to whether or not they want this much explanation, and if it is presented on the proper level for them. Section 8a includes a simple VCA and the suggestion that a whole set of simple circuits could be made into a non-keyboard synthesizer for beginners if there is enough interest.

NEW MEMBERS:

Edward G. Abinsay	198 Alvarado Ave., Pittsburg, CA 94565
Al Hauser	2904 So. Estelle, Seattle, WA 98144
Steven Jenkins	24 Deepwood Dr., Madison, CT 06443
J. Mailen Kootsey	Rt. 2, Box 431 D, Hillsborough, NC 27278
Richard Nelson	PO Box 432, Evergreen, Colo. 80439
Richard H. Rae	P.O. Box 791, Emporia, VA 23847
Richard Robinson	Rt 2, Box 98, Vilas, NC 28692
Jim Soya	217 Oleckna St., Throop, PA 18519
David Williams	Apt #21-F, Seymour Village, E. Granby, CT
Christopher Yewell	8277 Valley Dr., Chagrin Falls, Ohio 44022

2a. NEWS FROM OUR MEMBERS: David Friend has asked us to announce that he is now accepting papers for the electronic music section of the Fall AES Convention. Submit papers and ideas to Dave at ARP Instruments, Inc., 320 Needham St., Newton, Mass. 02164.

3. A FOUR POLE VOLTAGE-CONTROLLED NETWORK; ANALYSIS, DESIGN, AND APPLICATION AS A LOW-PASS VCF AND A QUADRATURE VCO: -by Bernie Hutchins, ELECTRONOTES

The op-amp inverter circuit shown in Fig. 1 should be a familiar device by now. It inverts the polarity of the input, and scales it by a factor of R_f/R_1. With the input grounded, the output remains very close to ground as well, but what happens when we replace the resistor R_f with a capacitor C as in Fig. 2? Initially, current V_1/R_1 which flows to the summing node (the - input junction point) would normally go through R_f. Since R_f is replaced by C, the output current must flow into C, thus charging it. The output of the op-amp will change as is necessary to maintain the differential input voltage at zero (i.e., the summing node at virtual ground). The output thus goes from zero toward -15, but since this is the supply voltage, it can go no further than -15. At this point, the feedback quits, and the differential input voltage starts to go positive giving the op-amp a second good reason for remaining at -15 volts output. Finally, the - input will reach the input voltage $+V_1$, the capacitor is charged to a total voltage of $15 + V_1$, and the input current stops.

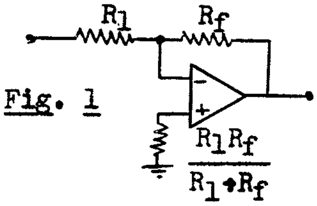


Fig. 1

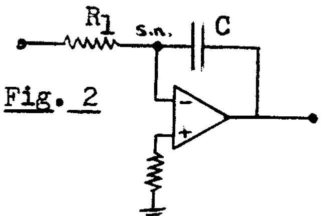


Fig. 2

You might expect that if you connected the input to ground, the output would remain at ground. In the case of a real op-amp, this is not true because the differential input voltage and/or the bias current to the - input will cause the capacitor to slowly charge, as these values will never be zero. The circuit in Fig. 3 will illustrate this nicely.

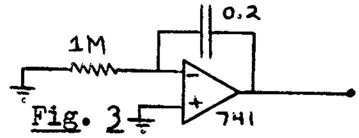
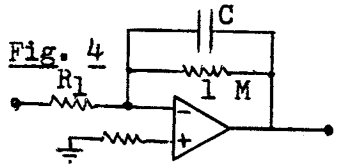


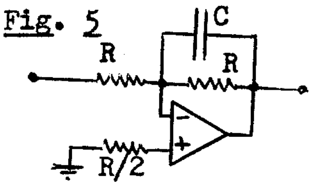
Fig. 3

The circuit with the capacitor in place of R_f is of course the usual op-amp integrator circuit, and we have been using this as a filter, so we are most interested in its AC characteristics. So why all the worry about the DC characteristics above? Simply because even small DC errors may cause these integrators to charge up against the supply or close to it, causing clipping and/or complete cutoff.

The traditional means of stabilizing the DC characteristics of the integrator is to effectively make a leaky capacitor to drain off any charge leaking on to it. This can be done with the circuit in Fig. 4.

Fig. 4


We can next consider the circuit of Fig. 5 where we use both a capacitor and a resistor in the feedback circuit. At DC and very low frequencies, the capacitor's reactance is much more than the resistance R, and the magnitude of the gain at low frequencies is thus just 1, as the capacitor is effectively out of the circuit. At very high frequencies on the other hand, the capacitor's reactance will be much less than R, and the resistor is effectively out of the circuit. The gain is therefore $X_c/R = 1/2\pi fRC$, and the thing starts to look like a filter as the gain (transfer function) depends on frequency. Since the response falls off as 1/f, we can see that for a one octave change, the gain drops by $\frac{1}{2}$. This is, in decibels, $20 \log_{10} \frac{1}{2} = -6$ db/octave. Since gain decreases with frequency, it is a low-pass filter. For reasons which we will consider later, it is called a single pole filter.

Fig. 5


We have seen very often in EN that integrators have been used to form filters. This is the two-pole design using two integrators known as the state variable, the universal active filter, the biquad, and by a variety of other names. In this circuit, DC stability is achieved through the use of the inverter. Since we had good results making voltage controlled filters by this means using voltage controlled integrator sections (with multipliers, or with CA3080 OTA's), we want to consider using these voltage controlled integrator sections for higher order filters. First, we have to take a look at some filter theory as it applies to the integrator.

We must first become familiar with the "s-plane", which is the "map" on which filters live. The quantity "s" is a complex frequency. We normally think of frequency as existing on a scale from zero to ∞ along one axis, On the s-plane, frequency has both a real, and an imaginary (in the mathematical sense) aspect. In fact, the useful physical properties of the filter will be derived from a frequency response above the s-plane while following a path defined by the imaginary axis. This is a good example of the fact that there is really nothing very imaginary about the so called imaginary numbers. Their main use in filter theory is to keep track of phase information automatically. In order to appreciate why we want to look at filter response above a plane, rather than just along the one line of interest, consider the following: Suppose you are to travel along a straight line on a map, and there are no features marked along this line. To get an estimate of what your journey may be like, you would look for features near your line of travel. For example, a mountain peak near your path will most likely indicate some high ground will be encountered along your path. On the s-plane, the straight line path you inspect is the positive imaginary axis, the elevation of the path is the filter's frequency response, and the mountains are the "poles" of the filter.

Consider for example the simple integrator in Fig. 2 as a filter. The gain of the integrator is $-1/2\pi fCR$. In filter terminology, this gain is called a "Transfer Function" which we will denote by T, and we will write this in terms of the complex frequency s as $T(s) = -1/sCR$. Note that s is generally written as $\sigma + j\omega$ where σ (sigma) is the real part, and ω (omega) is the imaginary part, and $j = \sqrt{-1}$.

The most important question to ask about $T(s)$ is: where does it blow up. That is, where does it become infinite, or equivalently, where does the denomenator become zero. For the integrator, $T(s)$ becomes infinite when $s \rightarrow 0$. Thus, if we plot $T(s)$ over the s-plane, we get a steep mountain around $s = 0$ as shown in Fig. 6. The mountain is called a pole. The positions of the poles of a filter are important in the sense that they greatly influence the contour of $T(s)$ over the entire s-plane. In particular, we examine the "height" of $T(s)$ above the $j\omega$ axis but pole position is also important in determining the stability of a filter. If the poles are located on the $j\omega$ axis or if they move into the right half of the s-plane (positive values of σ), instability (oscillations) may occur. This is the case of a filter looking so hard for a certain signal that it makes its own! Active filters are those which use an amplifier stage to position the poles in the desired positions on the left half of the s-plane. Books on active filters often indicate the pole positions by a cross mark ($\times$) rather than by the mountains we have used.

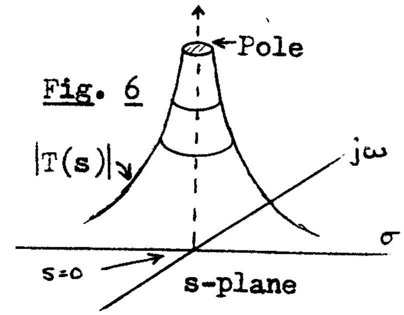


Fig. 6

Note that the mountain gets steeper and steeper as you get closer and closer to $s = 0$. It becomes a thin rod (which may be the origin of the name "pole"). In our practical circuit however, the pole has a flat plateau as shown due to the finite voltage of the power supply. It is also to be observed that this pole might be a deep hole if we are looking at $T(s)$, but we generally look at $|T(s)|$, which is in the case of $T(j\omega)$: $[T(j\omega)\cdot T(-j\omega)]^{\frac{1}{2}}$. At $s = 0$, the response becomes infinite, or we might say that the circuit "oscillates" at zero frequency (DC), and this is what we observed when we saw that the circuit of Fig. 2 responded all the way to the power supply voltage.

Traveling along the $j\omega$ axis, we see the response falling off as $1/f$ which we have identified with a 6 db/octave drop. This can be drawn as in Fig. 7, but usually is drawn on a log-log plot as in Fig. 8, where the 6 db/octave response becomes a slope of -1. The log-log plot is usually used for filter frequency response.

Fig. 7

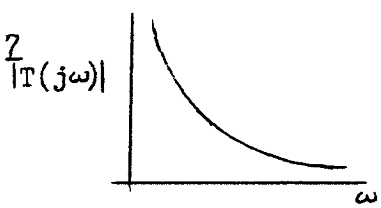
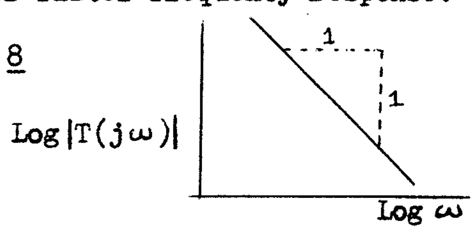


Fig. 8



Suppose for the moment that we have somehow avoided the DC stability problems, how close is the op-amp circuit of Fig. 2 to an ideal integrator, and does it look anything like what we would consider a useful filter? These characteristics are displayed in Fig. 9. The interested reader is directed to a separate reference[1] for a discussion of the exact features of the op-amp integrator curve. Note the similarity of this curve to the "open-loop gain curves" of certain internally compensated op-amps such as the 741. This is no coincidence, as such op-amps are actually compensated with a single pole.

The op-amp integrator doesn't look much like the ideal low-pass filter as you can see: $T(s)$ is way too high at the low frequency end, and doesn't fall off vertically as the ideal low-pass does, but with a slope of -1. Let's look at a means of improving the fall off first.

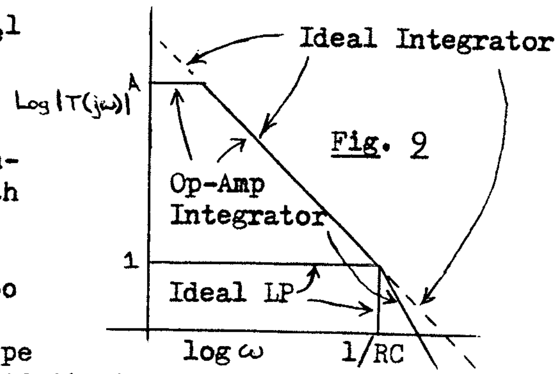


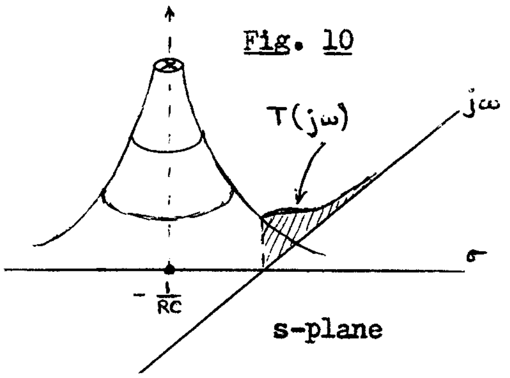
Fig. 9

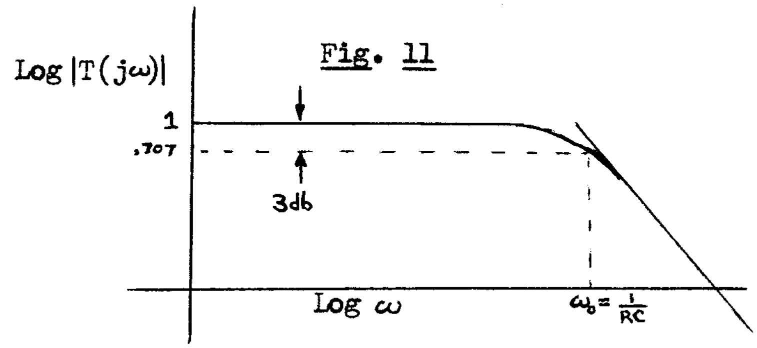
If we cascade two or more stages like Fig. 2, we get the product of the transfer functions of the individual stages. For two stages this becomes $T(s) = 1/s^2R^2C^2$ and $T(s)$ falls off as $1/s^2$ (12 db/octave). There are now two poles of $T(s)$, both of which are at $s = 0$, making this what is called a "second-order pole". Likewise, we can cascade three integrators for 18 db/octave (3-pole), and four integrators for 24 db/octave (4-pole).

We have already seen how to bring the low frequency gain down - use the circuit of Fig. 5. The transfer function for this is:

$$T(s) = \frac{-\text{Feedback}}{\text{Input } R} = \frac{-\text{Parallel } R \text{ and } C}{R} = \frac{\frac{-R/sC}{R + 1/sC}}{R} = \frac{-1}{1 + sRC}$$

This has a pole where the denominator of $T(s)$ becomes zero: when $s = -1/RC$. Thus, we have moved the pole from $s = 0$ to $s = -1/RC$, thus pushing it back down the negative sigma axis. This is equivalent to moving the mountain back from the $j\omega$ axis. Fig. 10 shows this, and Fig. 11 shows a log-log plot of the response along the $j\omega$ axis.





At $s = 0$, $|T(s)| = 1$, and from that point on, it begins to fall, slowly at first, and finally at 6 db/octave. We can now sharpen this drop off as we suggested before by cascading stages, adding more poles. Fig. 12 shows experimental data taken on 1, 2, 3, and 4 stages (1, 2, 3, and 4 poles at $s = -1/RC$).

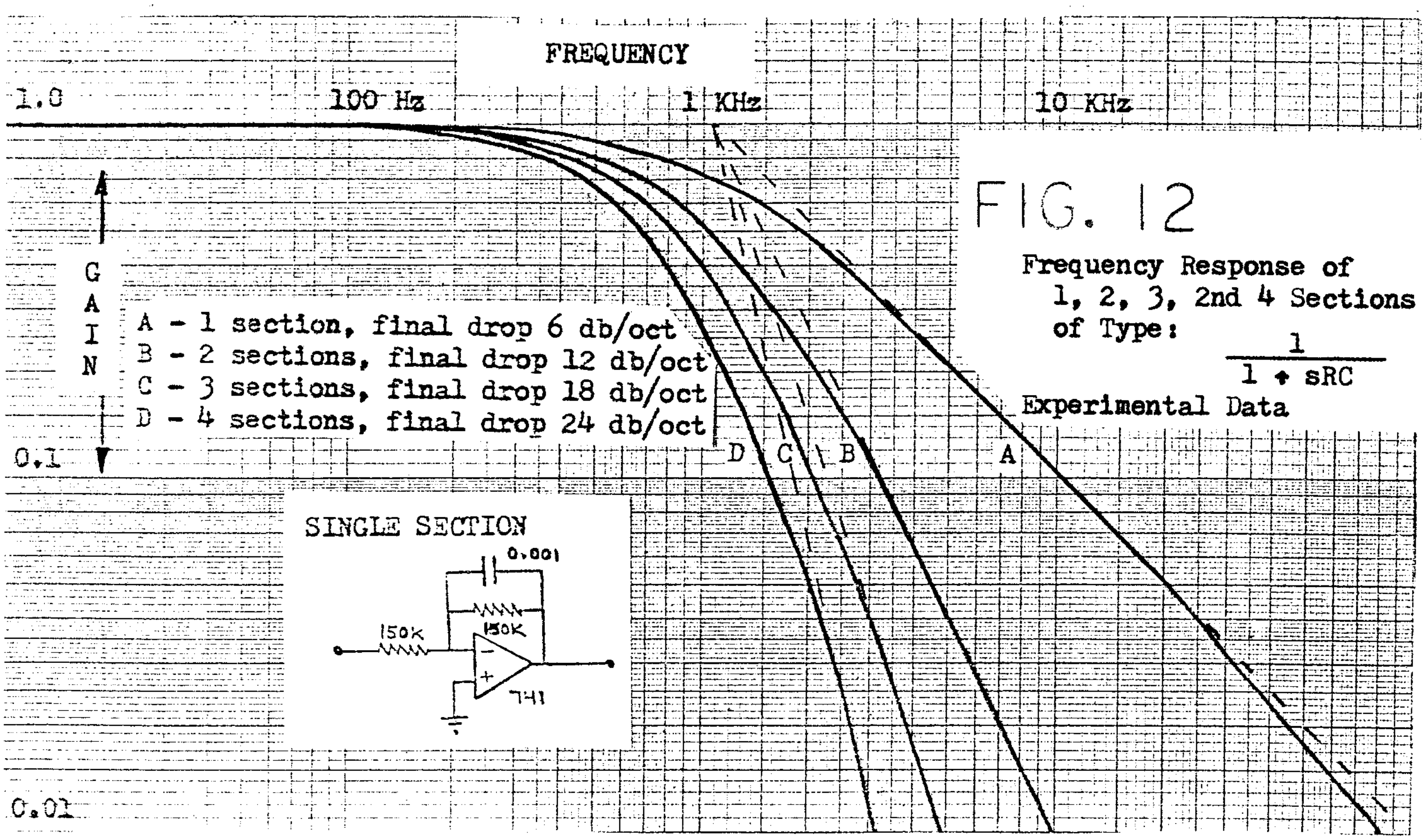


FIG. 12

Frequency Response of 1, 2, 3, 2nd 4 Sections of Type: $\frac{1}{1 + sRC}$

Experimental Data

These are fairly good approximations to the ideal low-pass filter as we go to higher and higher order, but it should be pointed out that this is not the best that can be done. The best approximation (while keeping the pass band flat) is the so called Butterworth response or maximally flat response. The single stage used in Fig. 12 happens to be a single pole Butterworth filter. Cascading two first-order stages increases the number of poles, and makes the filter second-order, but <u>does not</u>

preserve the Butterworth characteristics in the overall response. Two first order stages give:

$$T(s) = (1/1+sRC)(1/1+sRC) = 1/(1 + 2sRC + s^2R^2C^2)$$

$$= \frac{1}{\frac{s^2}{\omega_o^2} + 2\frac{s}{\omega_o} + 1} \quad \text{where } \omega_o = 1/RC$$

which can be compared with the second order Butterworth response of:

$$T(s) = \frac{1}{\frac{s^2}{\omega_o^2} + \sqrt{2}\frac{s}{\omega_o} + 1}$$

To get a true Butterworth response, the stages must be designed to work together to give the correct transfer function.[2] A comparison of a 4-pole Butterworth and the corresponding 4-pole, 4-sections damped integrator response is shown in Fig. 13. The Butterworth has a sharper corner. Fig. 14 shows the poles for the cascaded sections (all four poles piled up) while Fig. 15 shows the poles for the Butterworth.

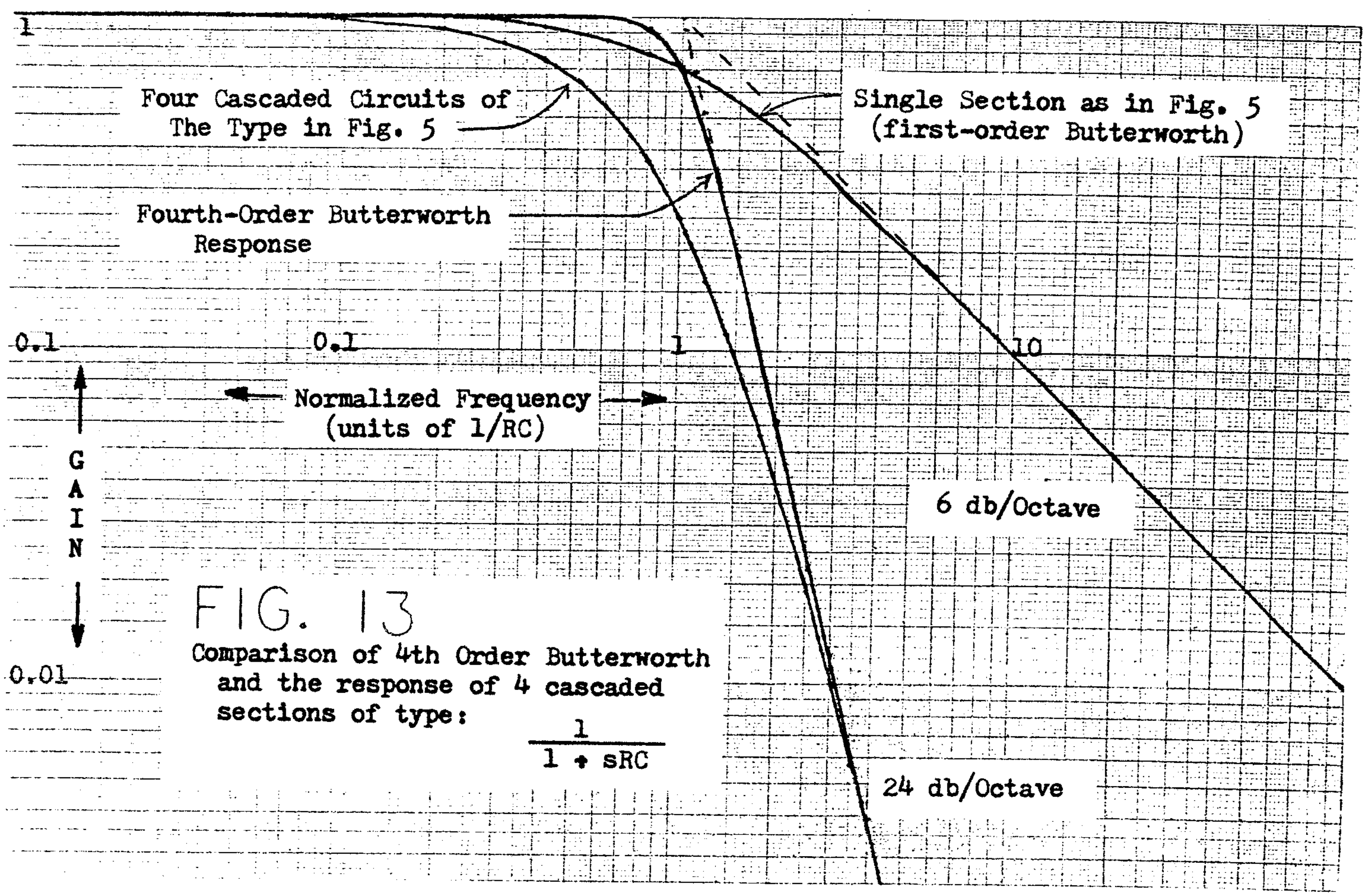


FIG. 13

Comparison of 4th Order Butterworth and the response of 4 cascaded sections of type: $\frac{1}{1 + sRC}$

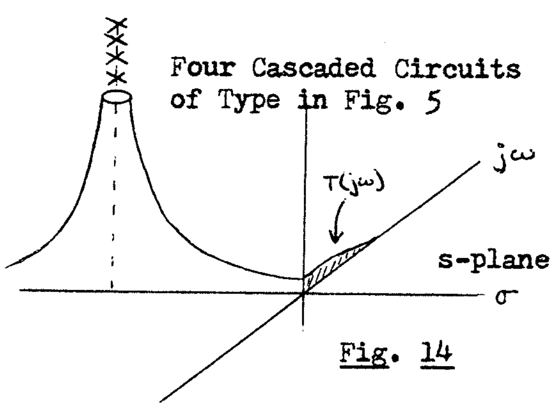


Fig. 14

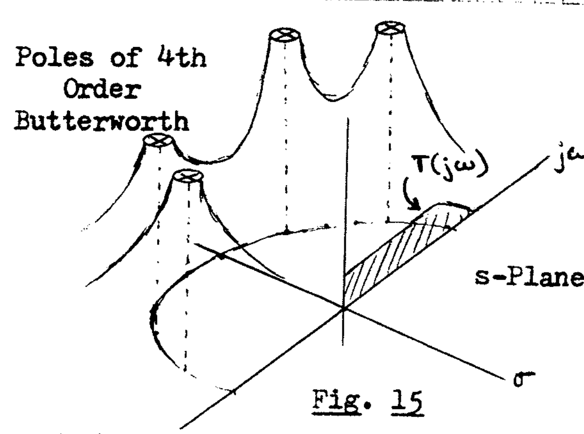
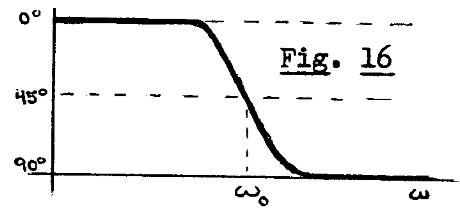


Fig. 15

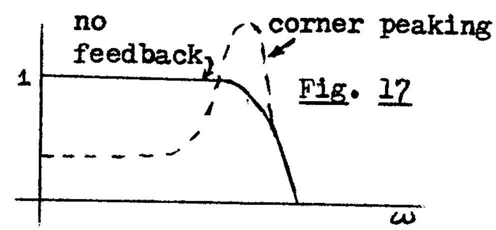
Moving the poles around toward the $j\omega$ axis and away from the σ axis causes higher ground to appear at the middle and corner of the response curve, thus making the pass band flatter, and the corner sharper. In any event, we will select the 4-pole cascaded sections, as this is easier to do in a practical circuit. The difference between using four cascaded sections of type $1/(1 + sRC)$ and the Butterworth may not be important musically, especially when we consider that the filter is often used with feedback to produce corner peaking. A successful electronic music VCF[3] does in fact use the cascaded section response.

<u>Adding</u> <u>Feedback</u>:

It is a usual practice in filter design (called the leapfrog approach) to consider feedback from different stages. In the present case, we want to feed the output of the last stage back to the input. Here, we must consider not only the magnitude of signal fed back, but the phase as well. The phase response of the single section is shown in Fig. 16. At low frequency, the phase of four sections adds up to 0°. At very high frequencies, the phase adds up to 90° + 90° + 90° + 90° = 360° = 0°. At $\omega_o = 1/RC$, the phase is 45° + 45° + 45° + 45° = 180°. So by adding an inverter to the output, and feeding it back to the input, we can enhance frequencies around 1/RC while further rejecting lower and higher frequencies. The result is corner peaking and is illustrated in Fig. 17. A control pot can be added to the feedback to control the sharpness of this peak. This control then serves as the "Q" control for the filter.



<u>Fig</u>. <u>16</u>



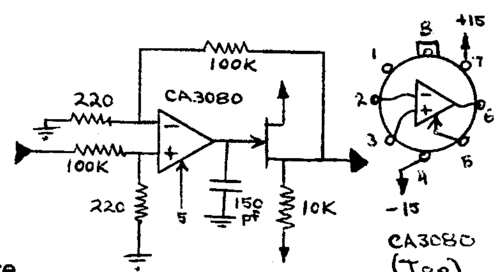
<u>Fig</u>. <u>17</u>

<u>Making</u> <u>it</u> <u>Oscillate</u>:

When we put a sine wave into the filter and adjust the frequency to 1/RC, each stage decreases the amplitude by $1/\sqrt{2} = .707$. After four stages, this will be down to $\frac{1}{4}$ the input. If we now replace the input sine wave with the output amplified by a factor of four, the filter will oscillate. In practice, we are inverting the signal first, either by a separate op-amp or using the opposite input if available, and changing the feedback resistor to $\frac{1}{4}$ the normal input value. For the experimental network of Fig. 12, an inverter is added to the output, and a resistor of about 37k replaces the 150k input resistor. The frequency of oscillation is 1/RC, and the successive stages produce 45° phase shifts, so we can take signals from the second and fourth stages for example to get a quadrature oscillater. The fourth stage is at $\frac{1}{2}$ the amplitude of the second, but this overall filtering is useful for the following reason. If the feedback is too small, oscillation will die out. Making the feedback larger will cause the first stage to clip at the power supply slightly. We want to run this first stage at a <u>slight</u> clip to keep oscillation stable, and thus the second stage will filter out this clip and give a signal level of about 10.6 volts. The fourth stage will have a very pure sine wave at a level of about 5.3 volts. In practice, all that we need to do to make the filter oscillate is to increase the Q control to the point where oscillation just starts.

<u>The</u> <u>Complete</u> <u>Voltage-Controlled</u> <u>Network</u>:

Voltage controlling a four stage network is now a simple matter of using voltage controlled integrator sections and feeding back each stage for unity gain at DC. This stage was first introduced by Terry Mikulic in EN#33 (5) and the modification for unity DC gain is shown in Fig. 18. We will use the exponential converter and summing stage from the VCF in EN#37 (6). The complete filter is shown in Fig. 19. The total range as drawn is 1 Hz to 22 KHz as either a filter or an oscillator. If you need more low range and are using the network mainly as a quadrature oscillator, you may want to change the capacitors from 150 pf to 1000 pf.



<u>Fig</u>. <u>18</u>

In Fig. 19, the filter output can serve to take out one of the quadrature signals when used as an oscillator, as the 5.3 volt level should be close enough to the 5 volt levels of the ENS-73. The other quadrature signal from the second stage can be reduced to a 5 volt level with 1000 ohms output impedance by using a voltage divider of two 2k resistors as shown. If you desire to use the quadrature oscillator as a circular location modulator, you may want to obtain the inversions of the quadrature signals as well. A set of four op-amps can be used for this as shown in Fig. 20. This could be simplified a little, but this circuit with four op-amps allows accurate settings of signal levels. For more information on location modulation, see EN#33 and section 8a below.

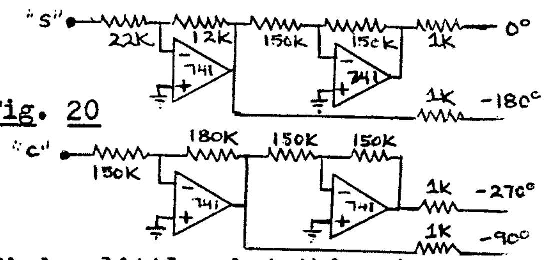


Fig. 20

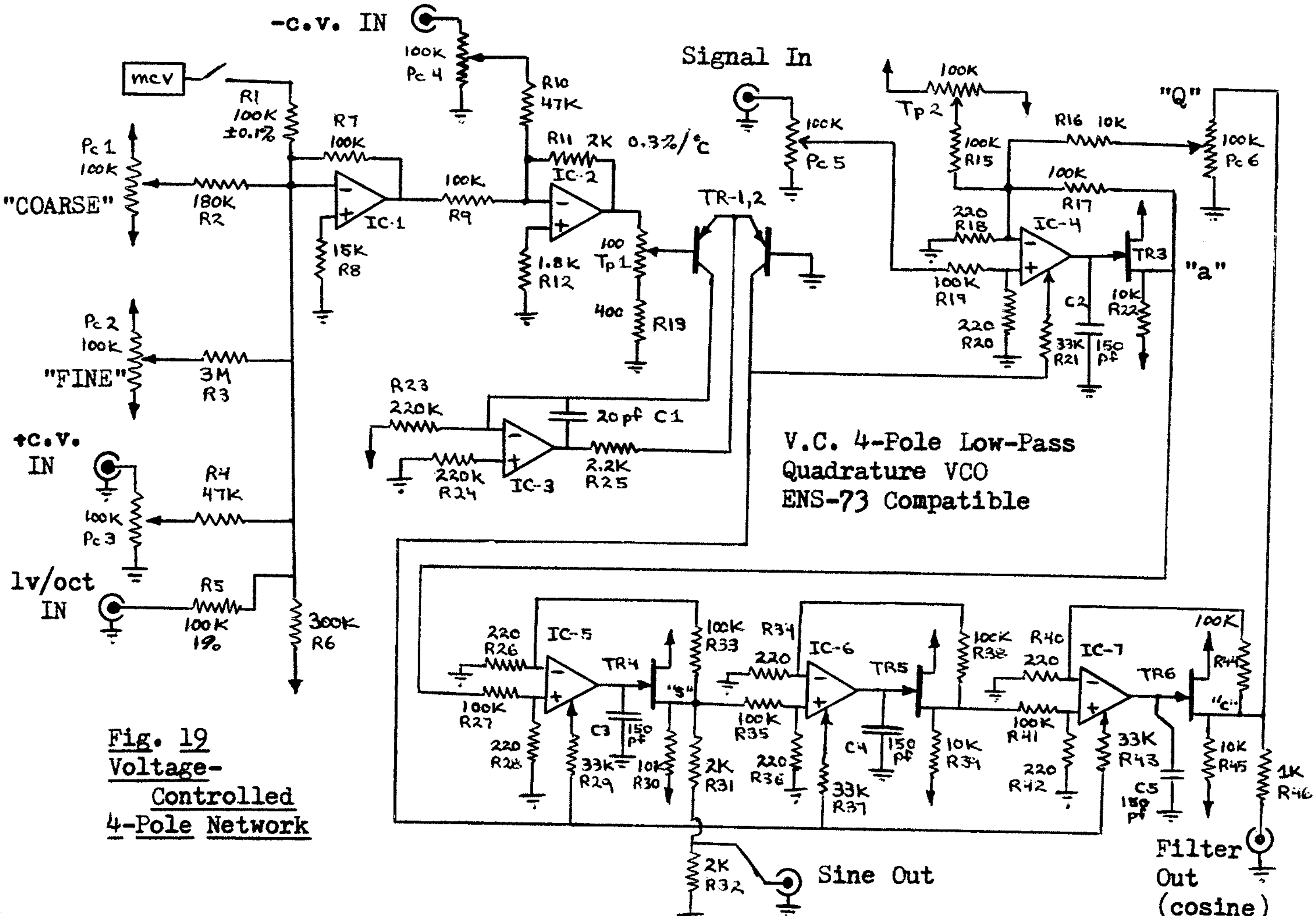


Fig. 19 Voltage-Controlled 4-Pole Network

IC-1, 2, 3: 307 or 741
IC-4, 5, 6, 7: CA3080
TR-1,2: Matched Pair AD821
TR3, 4, 5, 6: 2N3819 FET's

Adjust Tp1 for 1 volt/octave
Adjust Tp2 for symmetric clip at "a" during oscillation at low frequency.

The 2k 0.3% temp. comp. resistor and TR-1,2 should be in close thermal contact.

REFERENCES:

1) J. Graeme, G. Tobey & L. Huelsman, Operational Amplifiers Design and Applications Mc-Graw Hill (Burr Brown) 1971 (page 213)
2) Any good book on active filters, or use the short overview in Millman & Halkias, Integrated Electronics, Section 16-6 (Mc-Graw Hill 1972)
3) R.A. Moog, A Voltage-Controlled Low-Pass High-Pass Filter for Audio Signal Processing, AES Preprint 413 (Oct. 1965)

5c. <u>ESTHETICS OF NEW MUSIC</u>:

Thoughts On Electronic Music

II

Music - Electronic vs. Electrified*

-by Jim Wilson, 509 Cherry St., Winnetka, Ill 60093

The present use of the term "electronic music" covers a wide range of musical phenomenon from certain rock groups and the Hammond organ to the most sophisticated computer techniques. It seems that the only requirement for music to be given the adjective electronic is for electricity to be present. Evidently, this is considered necessary and sufficient.

What I would like to suggest is that the above use of the term "electronic music" is a striking example of definition by non-essentials. It confuses and blurs the characteristic of electronic music which makes it unique and revolutionary. That characteristic is that electronic music is <u>not</u> a performing art. This immediately distinguishes electronic music from all other types of music. It also radically alters the relations between composer - audience, composer - performer, composer - sound. In addition the status of the recording is drastically changed.

But what about live electronic music? I propose to call this <u>electrified</u> music. What I want to stress here is that electrified music does not change the composers relations with audience, performer or sound, and that electrified music represents a part of a general evolutionary trend we have seen in the history of music. That trend being to create instruments which expand the timbre and range available to the composer.

In short - electrified music is evolutionary, electronic music, revolutionary. We can see this by examining certain relations one by one.

The Relation Between Composer and Audience:

In performed music, electrified or acoustic, the composer addresses his audience only indirectly, through the medium of the performer, or twice removed through the performer and a distorted recording. The only exception to this is if the composer himself performs a work he has written for an unaccompanied instrument, to a live audience.

The composer of electronic music has a more direct, more intimate communication with his audience <u>because</u> electronic music is not a performing art. No performers means no distortion of the composer's intentions. Also, the recording of an electronic music opus does not represent a distortion. (More on this later.)

Paradoxical as it may sound, even tho the electronic music composer communicates more directly to his audience than the composer of performed music, the electronic music composer also has less physical contact with his audience. This is because the electronic music composer's audience is dispersed, rarely coming to concerts because electronic music sounds as good on one's own stereo (or quad) as it does in the concert hall. The electronic music composer speaks to his audience in the same manner as a novelist. The novelist never meets the vast majority of his readers and may even deliberately cut himself off from contact with his audience. For the first time the composer, if he writes electronic music, can do the same.

The Relation Between Composer and Performer:

In performed music, both electrified and acoustic, the composer is at the performer's mercy. He must first convince the performer that his music is worth playing. But even should he convince the performer that his music is good the performer must also consider his reputation and whether or not the audience will want to listen to a new piece. The situation is immensely frustrating since a composer's work may be rejected on grounds that have nothing to do with esthetics. And even if the performer finally deigns to play the composer's opus, the performer may not play the piece the way the composer would like.

The electronic music composer's relation to performers is very simple. Since there are no performers in electronic music no relation exists. The performer becomes irrelevant, and, thank goodness, the attendant problems

 *See EN#40 (9)- Do not copy this paper

The Relation Between Composer and Sound:

The material a composer uses is sound and this, of course, is true for both electronic music and performed music. But how the composer relates to sound undergoes a severe change in non-performed electronic music.

In performed music, either acoustic or electrified, the composer's relation to sound is distant and the composer has to rely heavily on his inner ear. This was the result of the fact that it was extremely difficult to gather the proper musicians together just to hear if a certain passage sounded correct. A positive cult grew up labelling any composer who composed at the piano inferior.

With electronic music the composer deals directly with sound. He therefore does not have to wait to hear certain passages, he can try them out, over and over again if need be. Here the composer resembles a sculptor and it might be appropriate to call electronic music sound sculpture. An electronic music composer can take a bloc of sound or passage and whittle it away, add to it, or change it and hear the results very soon.

The Status of the Recording:

In performed music, electrified or acoustic, the recording represents a reproduction in the same sense as a reproduction of a painting. The recording of a piece of music is generally considered an esthetically inferior product to hearing a "live" performance. (I must confess that I have never understood this prejudice for I have never heard a piece of music in a concert hall with the amazing clarity with which I hear the same piece on a recording. Besides, if I want to stand up while listening to a recording no one will mind and I won't be bothered by anyone else'e coughing, shuffling, whispering, etc., etc.)

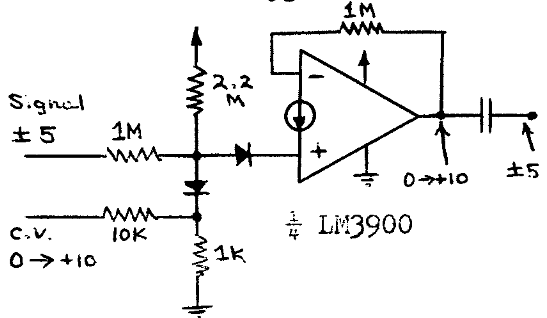
In electronic music the recording becomes the actual art work itself. It is not a reproduction. Just as each copy of a novel is the novel, so also each copy of an electronic music opus is the art work.

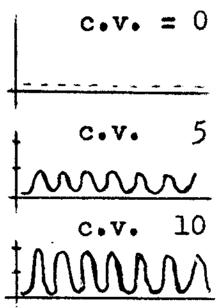
This leads directly to the chamber like quality of electronic music for electronic music is perfectly suited for home listening either from records, tape, or radio.

I think the above case speaks for itself. In no significant way does performed electronic music change the relations between the composer and performer or sound or the recording or audience. Electronic music, as a non-performing art, substantially changes all of these relations. Therefore electronic music and electrified music should be conceptually differentiated.

8a. A SIMPLE VCA WITH APPLICATIONS: -by Bernie Hutchins, ELECTRONOTES

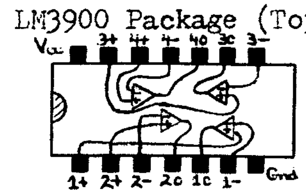
We often get requests for simple circuits; those which are both inexpensive and easy to build. While some of these would probably prove useless in the long run, there are several good reasons for considering some of these. They will prove useful for persons for whom the ENS-73 is too involved to start with. Furthermore, many new instrument designs are becoming practical which use whole sets or banks of devices that we have been calling modules. In many cases, relaxed specifications can be applied, and the expense of more sophisticated devices can't be justitied. The VCA below uses the new type of "Norton" or "current differencing" amplifier.





Off (c.v. = 0) is greater than 54 db down from full on.

LM3900 Package (Top)



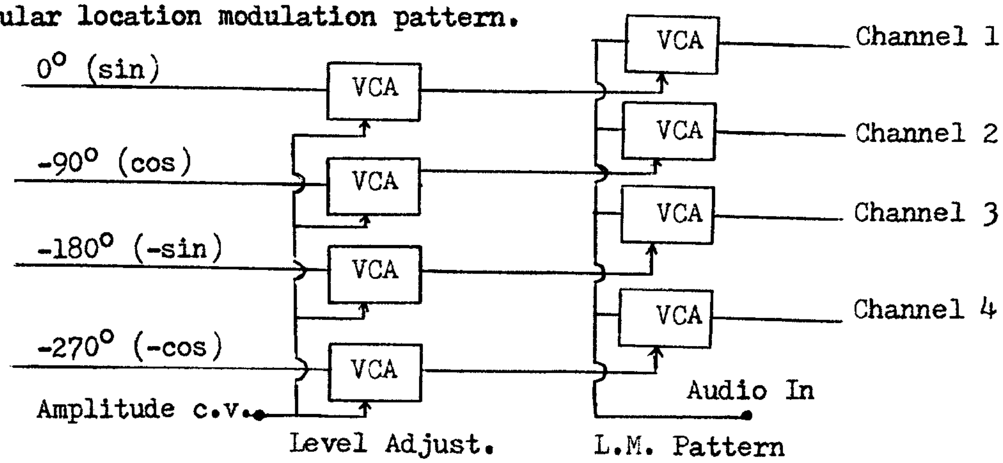
The circuit works on a single supply of +15, and is similar to the one in the National applications notes AN-72, but has been revised for greater range and for a different DC bias level. The circuit takes in a ±5 volt signal and gives out a 0 to +10 volt signal at maximum control voltage (+10). AC coupling of the output can be used to give a ±5 volt output. The low point of the waveform remains around zero volts at the output, so as increased control voltage is applied, the output stands up as indicated in the diagram.

APPLICATIONS:

1) The circuit is inexpensive and can be used in a simple "classical studio" type of synthesizer. If there is enough interest, we will publish a complete set of simple circuits with relaxed musical requirements (relaxed in the sense that no attempt to tune a keyboard would be made). The overall device could perhaps be used as a device for creating the more abstract types of electronic music.

2) These could be used in a VCA bank for additive synthesis in a manner similar to the one used with the Walsh functions (EN#20).

3) Four VCA's can be made for each LM3900 chip. Because the circuit above gives an output from a standard 5 volt source which is 0 to 10 volts, the same as the circuit's control voltage, the potential for cascading these in a location modulation scheme is attractive. In particular, the scheme below can be used to convert the quadrature oscillator outputs described earlier in this newsletter to a circular location modulation pattern.



Use of LM3900 VCA's to adjust DC level and amplitude of quadrature VCO, and to provide a circular location modulation pattern

9. <u>CLASSIFIEDS</u>:

<u>AVAILABLE</u>: Four channel computer music interface for Digital Equipment PDP-10 systems. Complete software, including timesharing system modifications. T. Joseph Zingheim, 3314 Kirk Rd., San Jose, CA 95124

<u>FOR SALE</u>: ICL-8038CC VCO IC's in 14 pin dip. Factory first quality: 4.80 each, $42 for 10 pcs. 7 Segment Readouts, factory prime: .33" character height, direct replacement for: Man-1, Man-7, SLA-1, SLA-7, etc. l.h. dec. pt., 15 ma/segment. Order #7799 $2.25 ea, $20/doz. Selected 741 op-amps, guaranteed 2 mv max offset plus other tight parameters, 65¢ each. Send Check or M.O. to: C.F.R., Box F, Newton, NH 03858

<u>FOR SALE</u>: Electrocomp 101, less than year old, barely used, $1200 or best offer: Walter Crosby, Freetown St., Lakeville, MA 02346 617-947-4368

<u>FOR SALE</u>: Precision Sequencers and Memory Units, sequencer set up three rows by 16 levels, can be used as voltage controlled waveform generator. Memory unit stores up to 256 notes in a very flexible usable format. Excellent for live performances when many different sequences are needed. Units available separate or combined in one box, compatible with most synthesizers. Write for details. Sequential Circuit Co., 7101 Rainbow Dr. #7, San Jose, CA 95129

<u>FOR SALE - KEYBOARDS</u>: For ENS-73 and others. 49 note single contact keyboards holding at $88, 61 note single contact keyboards at $110. MEG members take 10% discount. Payment cash in advance, include shipping for 11 lbs (49 note) or 13 lbs (61 note), UPS or Parcel Post. DEVTRONIX, 5872 Amapola Dr., San Jose, CA 95129

<u>WANTED</u>: Source for big matrix boards of the type EMS uses on their Synthi's. Need one approx. 100 x 100 size. Marty Maney, 75 N. 7th Ave., Des Plaines, Ill 60016

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