ELECTRONOTES 92

NEWSLETTER OF THE MUSICAL ENGINEERING GROUP

1 PHEASANT LANE

ITHACA, NEW YORK 14850

VOLUME 10, NUMBER 92 AUGUST 1978

GROUP ANNOUNCEMENTS:

CONTENTS OF EN#92

This issue contains a nice variety of shorter features which always seems to offer a refreshing change of pace. We want to call your attention in particular to the new method of envelope extraction (page 7) which has implications for pitch extraction as well, the use of a switched capacitor filter (page 14) for electronic music, and the start of a series reviewing the perceptual abilities of the ear (page 18). Also, there is a "Reader's Comments" section that comes as close to a "letters" section as anything we have done in the past.

In lieu of listing the recent application notes, this issue contains a complete list of the first 100 notes. The notes for May - August, Numbers 85 - 100, are new listings not reported in this newsletter before.

NEW MEMBERS AND CHANGES (c)

Kirby E. Allen	23 Langfield Cr., Rexdale, ONT, Canada M9V 3L6
Analog Systems	Box 1559, Boulder, CO 80306
Robert Chidlaw	EIKONIX Corp., 103 Terrace Hall Ave., Burlington, MA 01803
Ken Conroy	1051 Soldier Hill Rd., Oradell, NJ 07649
Henry S. Dakin	3101 Washington St., San Francisco, CA 94115
Graham Hinton	64A Kingston Rd, Oxford, OX2 6RJ, England (c)

Leonard Humbird	2925 NE 61st, Portland, OR 97213
George E. Lewis	484 W. 43rd St #10-L, New York, NY 10036
Knowles G. Little	9109 Scott Dr, Rockville, MD 20850
William Pottebaum	333 St. Andrews Dr., Rochester, NY 14626
Darryl V. Powell	5425 16th Ave #T3, Hyattsville, MD 20782
Ken Reeves	Box 498, Aguilar, CO 81025
Matthew Spransy	Kinetic Systems Corp., 11 Maryknoll Dr., Lockport, IL 60441
Carlos A. Vazquez	68 W. 10th St. Apt. 37, New York, NY 10011
Frank Waters	129 Canal St., Apt. 210, San Rafael, CA 94901 (c)
Chris Wheeler	10 Cherry St., Stratford, ONT, Canada W5A 2E8 (c)
Bruce White	127 Laurel Lane, Lawrence, NY 11559
Richard Wilson	10680-D N. Blaney Ave., Cupertino, CA 95014 (c)
Charles Zych	PO Box 221, The Sea Ranch, CA 95497 (c)

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News And Notes:

New England Video, Box 34, Newton, MA 02159 (617)-965-5287, has announced a series of instructional videotapes entitled "All About Electronic Music". These tapes offer instruction in electronic music and are based around certain popular synthesizers or certain techniques such as tape composition. The instructor is Robert Ceely. Write them for a list of tapes and prices (most are $39.95).

Music Lab, 1836 Hyperion Ave., Hollywood, CA 90027 (213)-666-9000 is offering a Micro-Computer class beginning Oct. 10, 1978, and a beginning and advanced synthesizer workshop class beginning Oct. 14, 1978. Peter Hillen will conduct the microcomputer class ($475 including kit) and Alex Cima will conduct the synthesizer workshops ($95 beginning workshop, $195 advanced). Classes continue through Nov. 18.

Reader's Questions:

Q: Do you have a reasonably useful circuit for the 8038 chip VCO?

A: There are two circuits for simple "utility" or "Audio" type oscillators in the Musical Engineer's Handbook. Basically, we do not suggest the chip for synthesizer work. A lot of people have them around though. I had eight of them at one time and managed to use them up, mostly as variable clocking circuits for digital IC's.

Q: What is the standard way of specifying op-amp noise?

A: This is a very complicated question because there are all kinds of noise and many different ways of measuring it. Here is something that always seems to work - when a good audio engineer tells you that a system is noisy, then it is noisy, no matter what the specs say. The ear is the best and the fairest judge here. To answer your question more directly, you will often find listed an equivalent input noise in terms of $nV/\sqrt{Hz}$ (nanovolts per root-Hertz). A really quiet op-amp will have this figure below 10 while values of 10-20 are not bad, and value of 40 or more are rather ordinary. A good reference that will probably tell you as much or more than you really want to know is National Semiconductor's application note AN-104 "Noise Specs Confusing?" which is also reprinted in their Linear Applications Vol. 2.

Q: How did the term "state-variable filter" come about?

A: In network analysis, there is a "state-variable" analysis method which assigns certain variable symbols (for node voltages and for currents), and these are the "state" (state of the circuit) variables. There is then a flow-graph which relates these state variables (essentially, Ohm's Law and its friends). For a second-order passive network, this flow graph is the same as the state-variable filter structure. Both currents and voltages are represented by voltages in this case.

The First 100 Electronotes Application Notes - Numerical Listing

The Application Notes listed below were issued by Electronotes at a rate of four per month from August 1976 to August 1978.

AN-1	Five-Volt One Amp Power Supplies for TTL
AN-2	Bipolar 15 Volt Supplies for Op-Amps
AN-3	Use of IC Voltage Regulators for Powering Equipment from Car Batteries
AN-4	Testing and Using LED's and LED Readouts
AN-5	Notch Filters for Hum Rejection
AN-6	Using the 555 Timer as a Monostable
AN-7	Design of Second-Order Low-Pass Filters
AN-8	Design of Fourth-Order Low-Pass Filters
AN-9	Circuits for Detecting Slowly Changing Signals
AN-10	IC Amplifier Selection Guide
AN-11	Determining the "Q" of a State-Variable Filter
AN-12	The BIQUAD Active Filter Circuit
AN-13	Equalizers and Tone Controls
AN-14	How to Actually Build Something - Part 1: Parts and Supplies
AN-15	How to Actually Build Something - Part 2: Circuit Boards
AN-16	How to Actually Build Something - Part 3: Soldering in Parts
AN-17	How to Actually Build Something - Part 4: Packaging
AN-18	How to Actually Build Something - Part 5: Miscellaneous Hints
AN-19	Simple Linear Voltage-Controlled Oscillator
AN-20	Improving the Linear VCO
AN-21	Applying the CA3080 OTA
AN-22	Gain Control Applications of the CA3080
AN-23	The CA3080 as a Voltage-Controlled Resistor
AN-24	Overdriven, Saturating, and Switching Applications of the CA3080
AN-25	Low-Q Bandpass Filter
AN-26	Bandpass Filter Examples
AN-27	Simple Clocking Oscillators for Digital IC's
AN-28	Simple Op-Amp Multivibrator Oscillator
AN-29	Simple Sine-Wave Oscillator
AN-30	More Clocking Oscillators for Digital IC's
AN-31	Schmitt Trigger Circuits with IC's
AN-32	The 555 as a Collection of Devices
AN-33	Simple Interface Circuits with CMOS
AN-34	Delay Line Setup with the MN3001
AN-35	Delay Line Setup with the SAD-1024
AN-36	Applying Delay Line Evaluation Setups
AN-37	High-Q Bandpass Filter
AN-38	High-Q Bandpass Filter Example
AN-39	Design of Second-Order High-Pass Filters
AN-40	Design of Fourth-Order High-Pass Filters
AN-41	Summing Amplifiers
AN-42	RMS to DC Converters
AN-43	Selecting Resistors for Given Applications
AN-44	Selecting Capacitors for Given Applications
AN-45	Graphical Determination of Frequency Response - s-Plane Case
AN-46	Graphical Determination of Phase Response
AN-47	Graphical Relationships in the z-Plane
AN-48	Z-Plane Graphical Relationships for First-Order Systems
AN-49	Z-Plane Graphical Relationships - A Second-Order Example
AN-50	Discrete Time All-Pass Networks
AN-51	Discrete Time Notch Filters
AN-52	Relationships for Higher Order Discrete Time Filters

<u>AUTHORS</u>: All application notes were produced at Electronotes except the following:

AN-69 Chuck Rogers
AN-70 Seamour Solutions
AN-73 Lester Ludwig (Coauthor)
AN-84 Seamour Solutions
AN-96 Mark Swartwout

We are pleased to receive contributed application notes from readers. A small fee can be paid to compensate authors for actually preparing notes for photocopying.

The First 100 Electronotes Application Notes - Subject Listing

The Application Notes listed below were issued by Electronotes at a rate of four per month from August 1976 to August 1978. Full titles and an indication of authorship is given in the numerical listing. Here the listing is by subject matter.

FILTERS:

Low-Pass 2nd Order and 4th Order	AN-7, AN-8
High-Pass 2nd Order and 4th Order	AN-39, AN-40
Bandpass, Low-Q, Design and Examples	AN-25, AN-26
Bandpass, High-Q, Design, Examples, Program	AN-37, AN-38, AN-99
Notch Filters	AN-5
Equalizers, Tone Controls	AN-13
State-Variable Filters, Theory	AN-11, AN-91
Biquad	AN-12
R-Filters	AN-55
Phase Shifter (All-Pass)	AN-65
Multi-Mode Filter Based on Low-Pass	AN-71
Twin-T	AN-72

NETWORK THEORY:

Graphical Frequency and Phase Response	AN-45, AN-46, AN-47
Z-Plane, First and Second Orders	AN-48, AN-49
Discrete Time All-Pass	AN-50
Discrete Time Notch	AN-51
Higher Order Discrete Time Networks	AN-52
Obtaining Design Data, Butterworth, Chebyshev	AN-75, AN-76, AN-82
Calculator Programs, Frequency Response	AN-77, AN-78
Calculator Programs, Network Design	AN-81, AN-96, AN-99
Resonant Peaking, Second Order	AN-90

MATHEMATICS:

Decibel Relationships	AN-53, AN-84
Algebra Tricks	AN-57
Trigonometry Tricks	AN-58
Complex Number Tricks	AN-59
Factoring	AN-74
Partial Fractions	AN-92
Hyperbolic Functions	AN-60
Fourier Series, Square Wave, Reference Table	AN-61, AN-64
Low-Passed Square Wave	AN-62, AN-63

OSCILLATORS AND CLOCKS:

Linear VCO	AN-19, AN-20
Digital Clocking, Square Wave	AN-27, AN-28, AN-30, AN-67
Sine Wave	AN-29
Triangle Wave	AN-67
Sawtooth, Multiple Phase	AN-79, AN-73

TIMING, TRIGGER, LOGIC, AND INTERFACE:

555 Timer Applications	AN-6, AN-32, AN-31
Schmitt Triggers	AN-31
Slowly Varying Signals, ALarms	AN-9, AN-83
Op-Amp Logic	AN-93, AN-94
Interfaces with CMOS	AN-33

POWER SUPPLIES:

OTA APPLICATIONS:

IC DELAY LINE APPLICATIONS:

DESIGN, CONSTRUCTION, TESTING:

MISCELLANEOUS CIRCUITS AND TECHNIQUES

SCOPE OF THE NOTES:

CK = Circuit   ET = Engineering Theory   RD = Reference Data   TK = Techniques

01-CK	16-TK	31-CK	46-ET	61-ET	76-ET	91-ET
02-CK	17-TK	32-ET	47-ET	62-ET	77-TK	92-ET
03-CK	18-TK	33-TK	48-ET	63-ET	78-TK	93-CK
04-TK	19-CK	34-CK	49-ET	64-RD	79-CK	94-CK
05-CK	20-CK	35-CK	50-ET	65-CK	80-TK	95-CK
06-CK	21-ET	36-TK	51-ET	66-TK	81-TK	96-TK
07-CK	22-CK	37-CK	52-ET	67-CK	82-RD	97-TK
08-CK	23-CK	38-CK	53-RD	68-CK	83-CK	98-CK
09-CK	24-CK	39-CK	54-TK	69-CK	84-RD	99-TK
10-RD	25-CK	40-CK	55-ET	70-TK	85-TK	100-RD
11-ET	26-CK	41-CK	56-TK	71-ET	86-TK	
12-ET	27-CK	42-CK	57-ET	72-CK	87-TK	
13-CK	28-CK	43-TK	58-ET	73-CK	88-TK	
14-TK	29-CK	44-TK	59-ET	74-ET	89-TK	
15-TK	30-CK	45-ET	60-ET	75-ET	90-ET	

Fast Envelope Extraction By Maximum Station Sampling:

-by Bernie Hutchins

In EN#89 we discussed the general problem of envelope extraction, and discussed an improved method of envelope extraction. Since that time we have done some more thinking about the problem and here will look at the "Maximum Station" method of envelope extraction. This method has the potential of being very fast. By the term "station" we have in mind a point in a signal chain where the signal can be observed. If we look at enough of these points, or look at the points in a systematic manner, and then select the maximum voltage, we are likely to find the maximum.

THE TAPPED DELAY LINE AS AN ENVELOPE EXTRACTOR:

To give what is probably the best example of a "maximum station" method, we will show that a tapped analog delay line, of the type manufactured by Reticon under type number TAD-32, can function as an envelope extractor. We will see that what results is a kind of instantaneous peak detector which functions without need of damping, and hence has no inherent time limitations. Figure 1 shows the general idea.

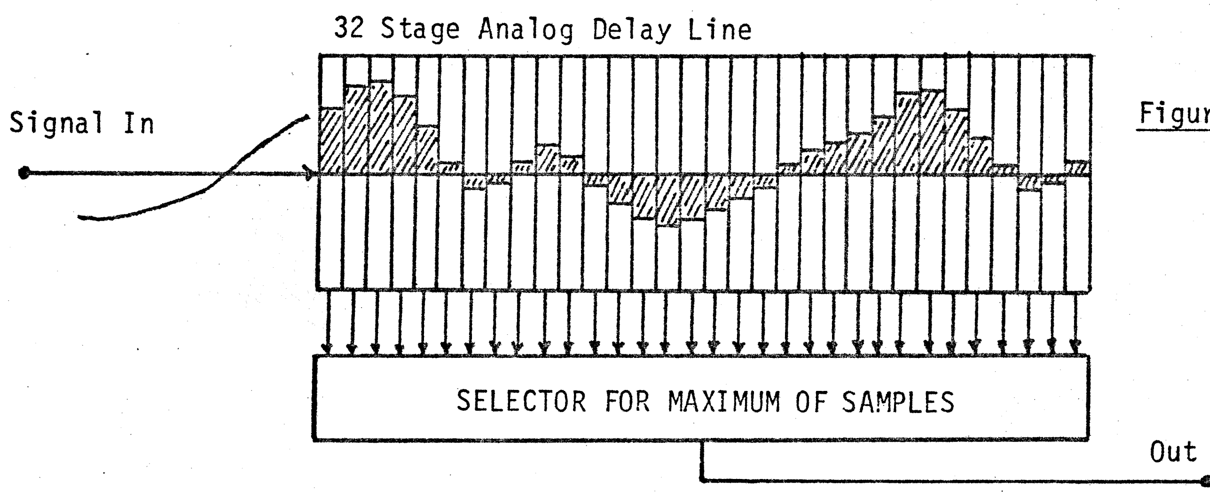

Figure 1

In Fig. 1 we see a 32 stage analog delay line which is currently holding 32 samples of an input waveform, which amounts to about one and a half cycles of the waveform as shown. The selector below the delay line examines all the samples simultaneously and outputs a voltage corresponding to the maximum. Before we go any further (assuming you have the general idea), it is desirable to say exactly what we might find in an actual realization of this idea. First there is the delay line, and the TAD-32 runs in the range of $35 to $50, although this price can be expected to drop in the future. Also, we might need more than one delay line since we must obey the sampling theorem on the high end, and have enough stages for at least one full cycle on the low frequency end. The selector for the maximum number of samples is not so bad - it could be as simple as a diode run from each stage to a common junction loaded by a resistor, and then buffered out. It should also be mentioned that the "unloading" of the delay line stages may involve individual buffers and level shifting, so this could further complicate the setup. Yet it is not by any means impossible or out of the question from an economic standpoint if this sort of envelope detection is truly needed.

Note that the output of Fig. 1 is the peak value of the waveform, just as with a peak detector circuit. A maximum value, once on the line, will walk all the way to the end. Once on the line, it will be the output voltage of the extractor unless a larger value enters the line. For a waveform such as one we find in music, we expect that successive cycles will be similar but not identical. Thus, as long as we keep at least one cycle on the delay line at all times, we do not expect extremely large changes in the maximum value output, but rather a relatively smooth variation as successive cycles slide into the delay line, along it, and are lost off the end. Note that the only delaying actions due to filters result from any possible "smoothening" filter that might be put on the output, and from the natural low-pass "transversal" filtering action that results when many cycles are on the delay line, as would be the case when the line, operating at fixed clocking rate, has a high input frequency. All and all, we would expect these effects to be relatively mild. There would be more to criticize about this particular device in that it only provides a peak voltage, which at times, may not be well correlated with subjective musical "loudness."

There are a number of additional ideas that are perhaps useful here. We could for example average (absolute values of) all samples on the line, or even use RMS if necessary. To take the peak of both the positive and the negative excursions, two sets of diodes could be used on all stages, and the maximum of the two could be then selected. Perhaps better still, we could full-wave rectify the signal before going into the delay line. The zero voltage of this full-wave rectified signal could then be shifted to a negative level corresponding to one extreme of the available dynamic range on the delay line. This would make better use of the delay line's full dynamic range, and only one set of diodes would be needed to examine both positive and negative peaks.

PHASE SHIFTERS AS STATIONS:

At first sight, the delay line, providing signal "stations" equally spaced in time would seem to provide a very promising soultion to the amplitude extraction problem. Of course, it also seems involved and expensive. What other devices could provide signal stations? The simple one stage phase shifter (all-pass network) would seem to be a possible candidate for this purpose. We could provide a number of such stations, either by driving the phase shifters in series or in parallel, and selecting the maximum instantaneous value of output voltage. An example for a parallel driven system is shown in Fig. 2.

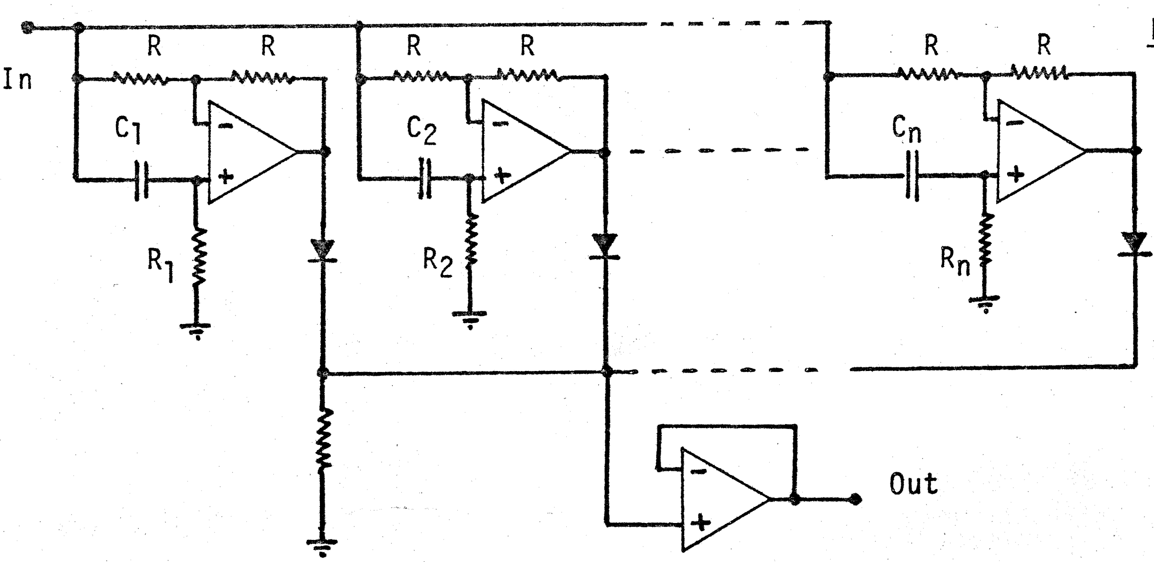

Fig. 2

At the moment, we are not prepared to suggest either values for R_1C_1, R_2C_2, up to R_nC_n, to suggest the number "n" stages that are required, or to say if a series or a parallel connection is the most desirable. This would seem to be a problem well suited to experimentation. A few points can be made however, that may be of some help. First, there is a difference between serial and parallel connection. While we might expect a longer time response as we cascade more and more sections,(a factor in favor of parallel connection instead),it is also possible that a smoother response with fewer stages could be achieved with series connection. Secondly, in the parallel case it is obvious that we do not want all the R and C values the same, but in the series case, it might be desirable to have all stages the same to achieve a uniform spread. Of course, it is difficult to suppose what an optimum strategy might be in either case and a few calculations backed up by experiment would seem to be the best approach. The third point, and a very important one, is that these phase shifters operate on a complex waveform on a component-by-component basis. A sharp pulse, for example, will not in general pass through the phase shifter and emerge as anything like a sharp pulse, but rather will be dispersed by the process of phase distortion. This means that the amplitude value we are likely to achieve with this method will not be the peak value we get in the circuit of Fig. 1, but rather some other value, which may be a better estimate, a worse estimate, or just as good an estimate as you get from the peak measurement. Here, much depends on the individual waveform that we are trying to process for amplitude, and our goal would seem to be one of finding a system that works sufficiently well for a wide range of waveforms. A fourth point is that the single stage phase shifters of Fig. 2, in parallel form will provide no more that 180° total phase shift for any frequency. Clearly, to avoid a "dip" in the output, for a sine wave for example, we need shifts in excess of 360°. It helps some here if we add in the input as a "station", but it is probably more to the point to use a second set of diodes, find the maximum negative voltage, invert it, and take the maximum of these. This effectively doubles the number of phase shifters, for the small price of a diode per stage. A fifth point to be made is that an input waveform with sharp corners has this corner by virtue of high frequency Fourier components. These high components will likely be above the corner frequency of the phase shifter, and all receive the same phase shift. This generally results in the retention of a spike, but it becomes isolated, and perhaps can be easily removed by very modest low-passing. In this way, we may be able to avoid using phase shifters in the upper region.

THE STATION APPROACH APPLIED TO THE AMPLITUDE EXTRACTOR OF EN#89:

In EN#89, we described a circuit that followed up on an idea given in a circuit by Denny Genovese in EN#88. The original circuit used cascaded full-wave rectifiers with AC coupling, and in EN#89 a pair of peak detectors were added to equalize the peak voltages of the input waveform. Clearly these units, which we cascade, can be used as stations as we have described above. What's more, it turns out that we not only can do this, but perhaps we should or even must do this. This fact was called to our attention by Ian Fritz. In EN#89 we say (page 5 at bottom, for those of you making corrections), that "a square wave is handled without difficulty although one stage works as well as many for this case." It is however the actual case that only the last part of this statement is correct. Ian points out that if a square wave is input, then the output of the first stage is a constant DC voltage, which is then blocked by subsequent stages by the AC coupling. Of course, this is correct and we apologize for overlooking this fact. The method can be salvaged by using the station approach given here. We simply take as our maximum, the output of whichever stage is the largest. With the square wave input, the first stage gives the DC constant, which is clearly larger than the zero voltage at the output of the other stages, so this, the correct amplitude, is passed to the output.

APPLYING THE IDEAS TO PITCH EXTRACTION:

Once we have developed fast methods of amplitude extraction, we can once again look at certain methods of pitch extraction to see if we can take advantage of these new methods. Certainly any method of pitch extraction, such as center clipping, which relies on amplitude information, can be expected to be improved by improved methods of amplitude extraction.

One particularly nice method that comes to mind as we look at Figure 1 is that once we know the peak amplitude, it is easy for us to detect the positions of the peak or peaks in the waveform. If we take the output of Fig. 1 and attenuate it a few percent, and then use a comparator to compare this attenuated peak with one of the taps in the middle region, we can expect to detect amplitude peaks. In the case where only one main amplitude peak occurs each cycle, the comparator triggers at the pitch rate. While we can not guarantee that any waveform will have only one peak per cycle, in many cases, this is the case, and in others, low-pass and other filters will add to the general success of this method. A pitch extractor based on this idea will be constructed soon for an evaluation of the idea.

SUMMARY:

Above we have discussed a method of arriving at a voltage related to the amplitude of an input waveform and of doing so as rapidly as possible. The method is similar to the process we would do by eye if we were measuring the amplitude on the face of a scope - we would look along the entire waveform for the highest point or points. Electronically, this is done by having the signal appear at various "stations" and then using diodes (or a similar equivalent method) to find out which station has the highest voltage. If we have enough stations, or if we are careful to place stations so that we pretty much cover the entire waveform, this should be a reliable enough method. Interestingly enough, we found that while this method results in excellent speed of acquisition, we still are not free to have as much bandwidth as we want, and still maintain our speed. Something fundamental there? Still, an improvement is an improvement.

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FILTER BANKS WITH MULTIPLE OUTPUTS: -by Bernie Hutchins

Filter banks are not difficult to design and are not really very hard to build, although they may involve a large number of individual filter units, meaning that there is a relatively high IC count to the project. In any case, you probably don't want to go out and build a large number of them. For this reason, it is desirable that the best possible choice be made for the design, and that if possible, the bank be made to have variable features, or at least to have several outputs. Basically, we know how to make a variable filter bank since we know how to make tunable filters. However, since we have many sections to tune at once, we would probably have to use voltage-control to tune them all. This too we can do, but the prospect of building a filter bank with say 40 channels of voltage-controlled filters is rather imposing. We might consider making only the Q of the filters variable, using something like a CA3080, but even this is rather involved, and if any channel becomes unstable, it may be a problem to track the culprit down and correct it. For these reasons, it seems simplest to us to just set up the filters, and get a variety by different combinations of outputs. Since we get an output for a filter bank by summation anyway, this is a logical extension of the standard procedure.

Figure 1 at the right will show the general idea. We have a bank of parallel driven sharp bandpass filters. In a standard filter bank, there would be only one summer, probably summing all filters equally. Here we will use the idea that several summers can be used. In terms of hardware and panel space, this means one more op-amp, one more set of summing resistors, and an additional output panel jack - not really a very big price to pay if we can justify the additional feature.

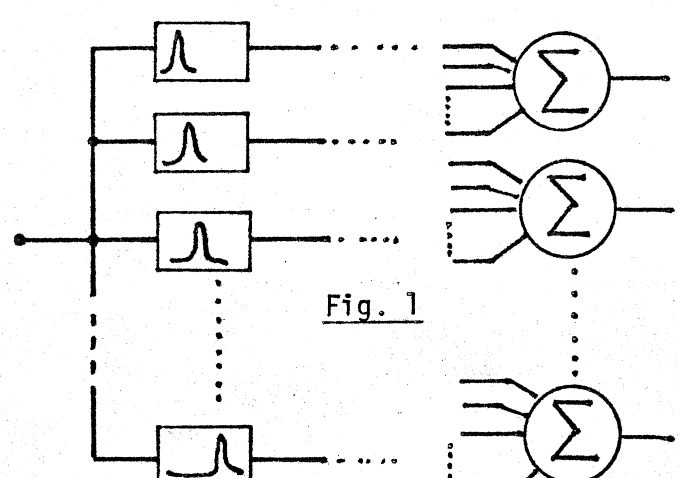
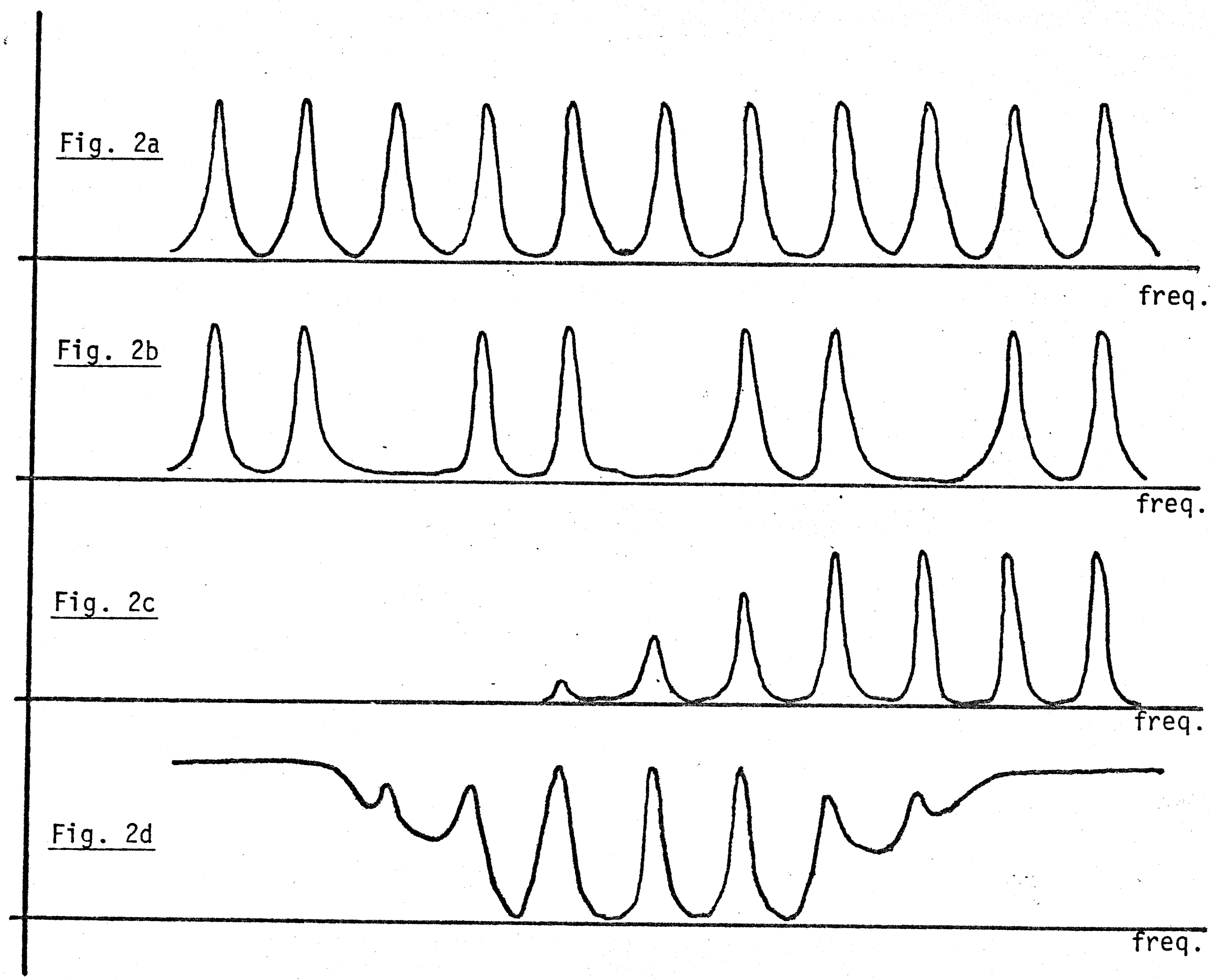


Fig. 1

We can make a list of the ways in which these additional summers may be set up:

1. We can sum all filters equally, as in the standard filter bank.
2. We could increase the spread between filters by leaving some out of the sum.
3. We could divide the bank between summers covering different frequency ranges such as low, midrange, and high. Probably the sum would be tapered at the edges - instead of summing all equally and then just stopping, the filters at the edge would be summed in decreasing amounts before stopping
4. We might wish to have the filter bank effect only over a certain frequency range, but still not completely block other ranges. In this case, the summer would include the appropriate range of sharp bandpass filters and in addition, appropriate flat response low-pass or high-pass filters.

Figure 2 shows the total response for some of the cases suggested above. Fig. 2a shows the total response for all filters equally summed, while 2b shows a case where every third filter is omitted. Figure 2c shows a tapered bank of high frequency filters while 2d shows a tapered bank of mid-range filters filled in with low-pass and high-pass on the ends. At the moment, we have nothing musically to suggest about any of these arrangements in addition to what is already known about the standard bank.



MODIFIED AD+AR ENVELOPE-GENERATOR DESIGN:

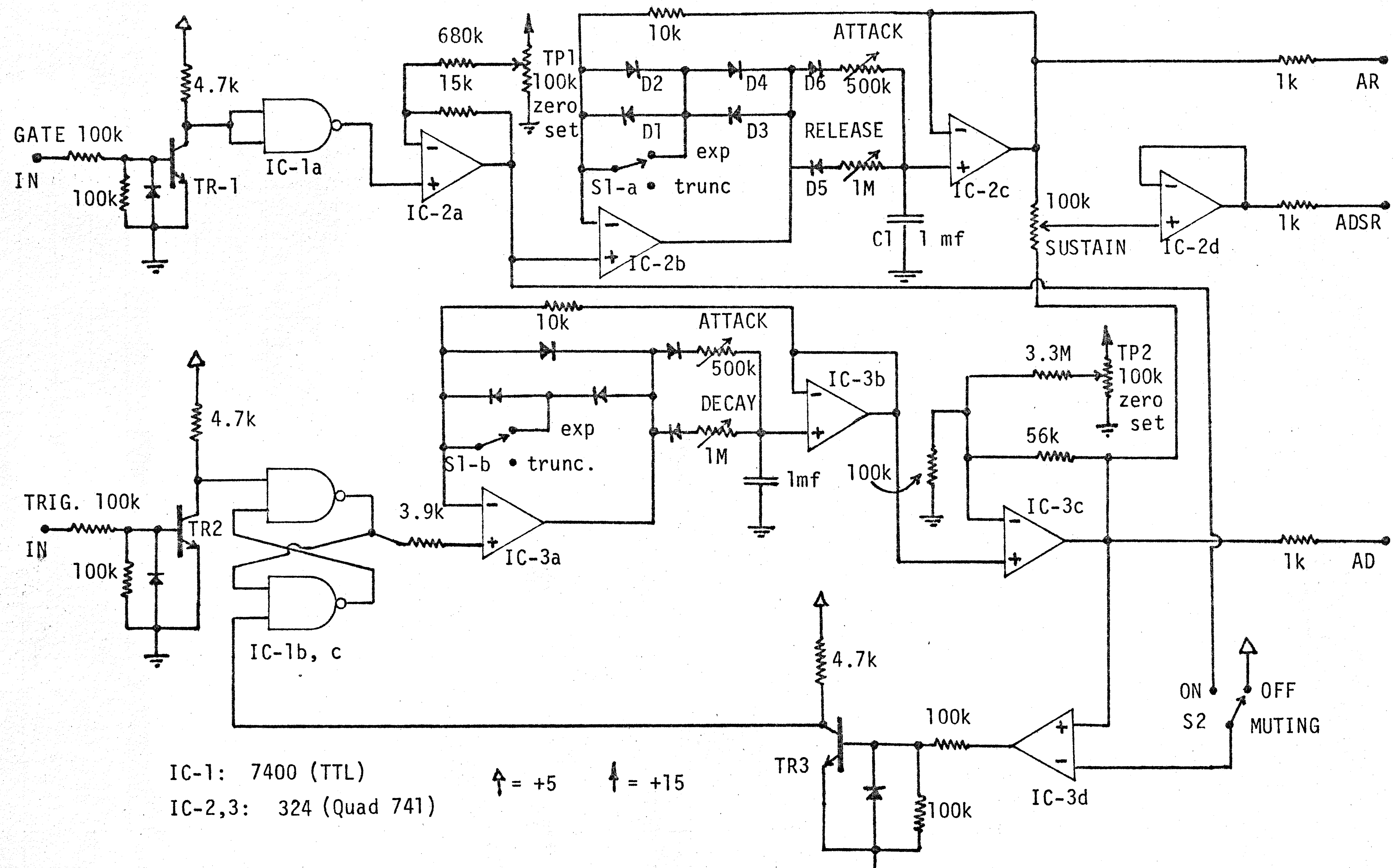
-by Ian Fritz

In the EN Mid-Month Letter #19, Bernie discussed possible advantages of employing the "older" AD+AR design for envelope generators (cf. EN#45), as opposed to the newer "standard" logic designs (cf. EN#66, EN#87). The AD+AR design has an advantage of flexibility (especially if separate pots are used for the attack controls), because the envelopes can be used either separately or in combination, and because interesting "non-standard" attacks can be produced by setting the two attack times to different values. A feature of the usual AD+AR design is that the attack phase of the AD generator is always completed after a trigger is received. The "standard" design, on the other hand, has the property (which will be referred to as "muting") that if the gate voltage goes low before the attack is complete, the generator goes immediately into a decaying mode, without completing the attack. Whether muting is advantageous or not depends of the playing situation, but it would seem desirable to at least have it as an option, as it allows variations in volume to be produced by changing the length of time the keyboard keys are held down.

The AD+AR design described in the following derives from earlier EN designs (EN#29, EN#31, EN#45). It has two new features: 1.) the AD section can be muted, with a switch being included to select either muted or non-muted operation, and 2.) an option is included, also switch-selectable, for either exponential or truncated-exponential envelopes. Truncated exponentials are useful if the envelope is to be used to modify oscillator pitch, as the long tail of the complete exponential gives pitches that never settle, and the ear is sensitive to this.

The complete circuit is given in Fig. 1, and much of it may be familiar. TTL circuitry is included, but a more modern design, eliminating the 5v supply, should not be difficult to implement. Operation, beginning with the AR section, is as follows. The input interface consists of TR-1 and IC-1a and provides consistent levels at the input of IC-2a for different gate levels. The purpose of IC-2a is to shift and amplify the TTL levels of IC-1a so that the output of IC-2a is zero or five volts (I don't know why IC-2a produces as much gain as it does). TP1 is set so the output of the AR generator is zero under quiescent conditions. The AR generator proper is built around IC-2b and IC-2c. This circuit is the same as the one given by Carl Hovey (EN#31), except for the extra diodes D1 and D2 which can be shorted by S1-a. Diodes D3 and D4 compensate for the voltage drops across D5 and D6. These drops are objectionable not only because of the 0.6 volt offset they produce, but also because the diode resistances become large as equilibrium is approached, giving extra-long (non-exponential) tails on the envelope. With S1 in the open position, the extra diode-drop voltages cause C1 to charge <u>towards</u> either 5 + 0.6 or 0 - 0.6 volts. The AR output, however, stops at 5 and zero volts, because of the feedback from the output of IC-2c to the input of IC-2b. This results in a truncated exponential envelope. The amount of truncation is perhaps a bit severe, and this can be reduced somewhat by using germanium diodes for D1 and D2.

In the AD section, IC-3a and IC-3b are analogous to IC-2b and IC-2c. The switch section S1-b allows truncation of the decay curve. (Truncation of the attack section is inherent in the design. so no extra diode is needed.) The AD envelope is zeroed under quiescent conditions by adjustment of TP2. Operation of the AD generator is, except for the minor changes just discussed, exactly the same as in the EN designs, provided that the muting switch S2 is in the "off" position. With the muting "on", the AD envelope depends on the gate via the line back to the output of IC-2a. If the gate goes down during the attack phase of the AD envelope, IC-3d switches and resets the flip-flop, thereby initiating the decay phase. It may not be obvious that the logic is otherwise correct, but careful analysis shows that there are no problems. (The analysis is a bit tricky, because of the indeterminate states of the RS flip-flop. For example, it doesn't matter whether



AD + AR ENVELOPE GENERATOR (Ian Fritz)

IC-3d is high or low under quiescent conditions.) The only problem encountered in getting the unit to work was caused by the input trigger pulse being too short. This pulse has to be longer than the time it takes IC-3d to respond to the gate, which seems to be about 0.1 to 0.2 msec.

The final ADSR output is obtained in the usual manner, as a weighted sum of the AD and AR envelopes, buffered by IC-2d.

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A QUASI-DIGITAL BI-N-TIC FILTER

-by Jan Hall
Elect. Music Dept. Hunter College
695 Park Ave
New York, NY 10021

The figure on page 15 shows my version of an article* in a 1974 issue of Electronic Design on a Biquart Filter (not Biquad, but Biquart, twice the order).

By sequentially switching eight 0.047 mfd capacitors across the 356 op-amp, multiple bandpass integrators are formed giving a filtered quantized analog output with multiple passbands (comb filter).

The damping control sets decay times to a pulsed input, and the filter can be made to ring for 2-3 seconds. It is very hard to describe the sound quality of this filter. Due to the very high Q's and f_o, $2f_o$, $3f_o$, etc. passbands, a very complex dynamically changing output can be obtained. By simply sweeping the clock frequency, at times sounds similar to voices having human features are heard, only to decay into ringing bells.

Anyone wishing to hear this filter or obtain help in building it can reach me at the address above.

- -

*Allan Lloyd, "Transform the Biquad into a Biquartic", Electronic Design, Jan. 4, 1974, pg. 120. EDITOR'S NOTES: This paper is mainly concerned with filters of the more standard RC type rather than the switched capacitor type Jan uses. The starting point is the Biquad circuit we call the "state variable". A low-pass to bandpass transformation is achived by adding inductors in parallel with the capacitors, forming resonators, and transforming zero frequency to this resonant frequency, and reflecting the response on both sides. The remaining task of the paper is to realize the inductors by capacitors and op-amps. Some interesting practical circuits result. The switched capacitor version, which Jan follows, is referred to as a "bi-n-tic" filter, and does not seem to draw much on the main part of the paper, but is rather closely related to other filter techniques of the switched capacitor type or the "commutating" type. It is good to see these filters being used for electronic music. --*Bernie*

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DIGITAL KEYBOARD REVISITED:

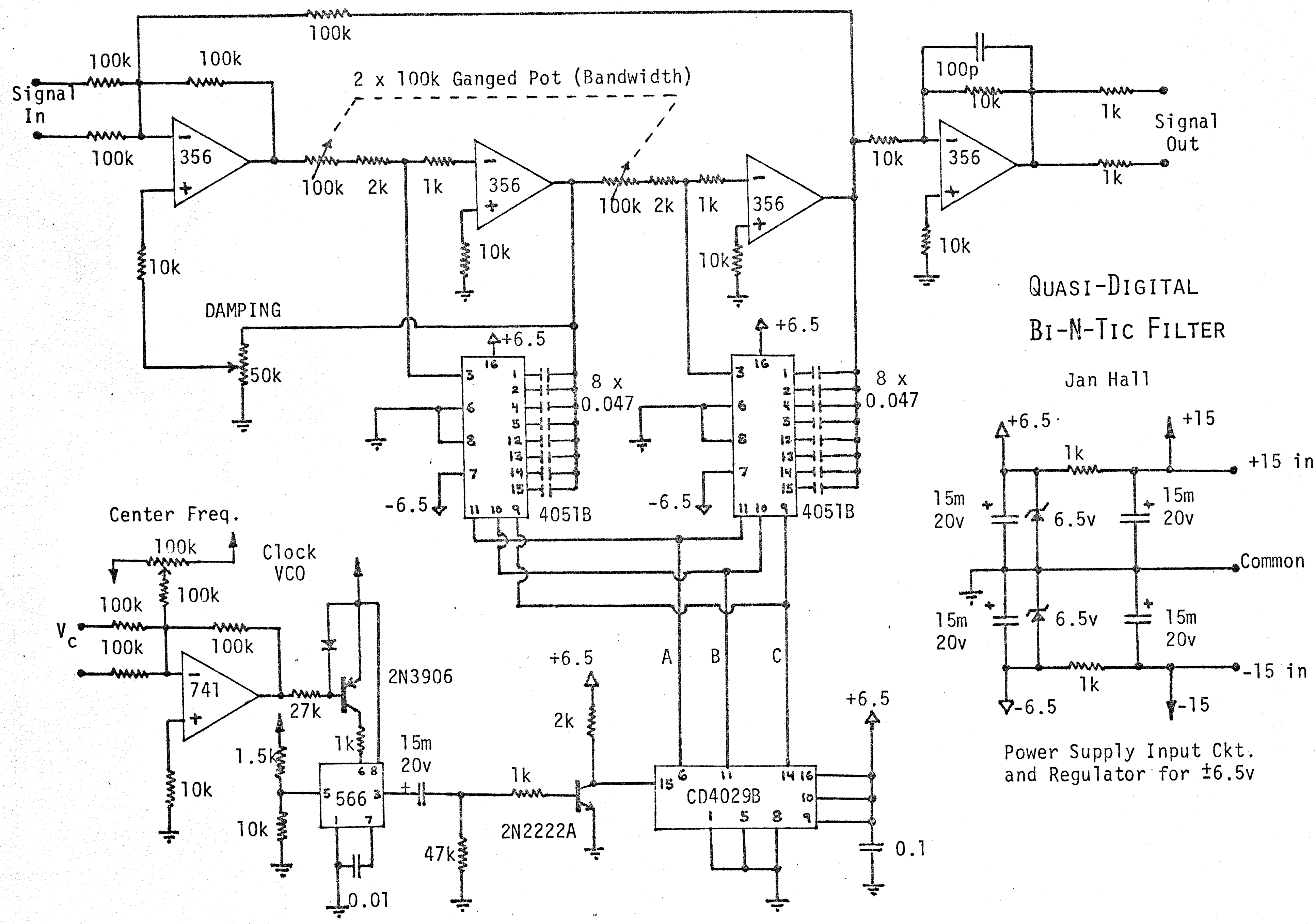
-by Richard Curcio

I have constructed a digital keyboard similar to the ENS-76 design in EN#68, and have made a few changes in the circuitry which is shown in the schematic diagrams on pages 16 and 17.

Figure 1 (page 16) shows the gate, trigger, and D/A circuitry. The upper part of Fig. 1 is the gate and trigger circuitry. Instead of going to ground, the keyboard

Quasi-Digital Bi-N-Tic Filter

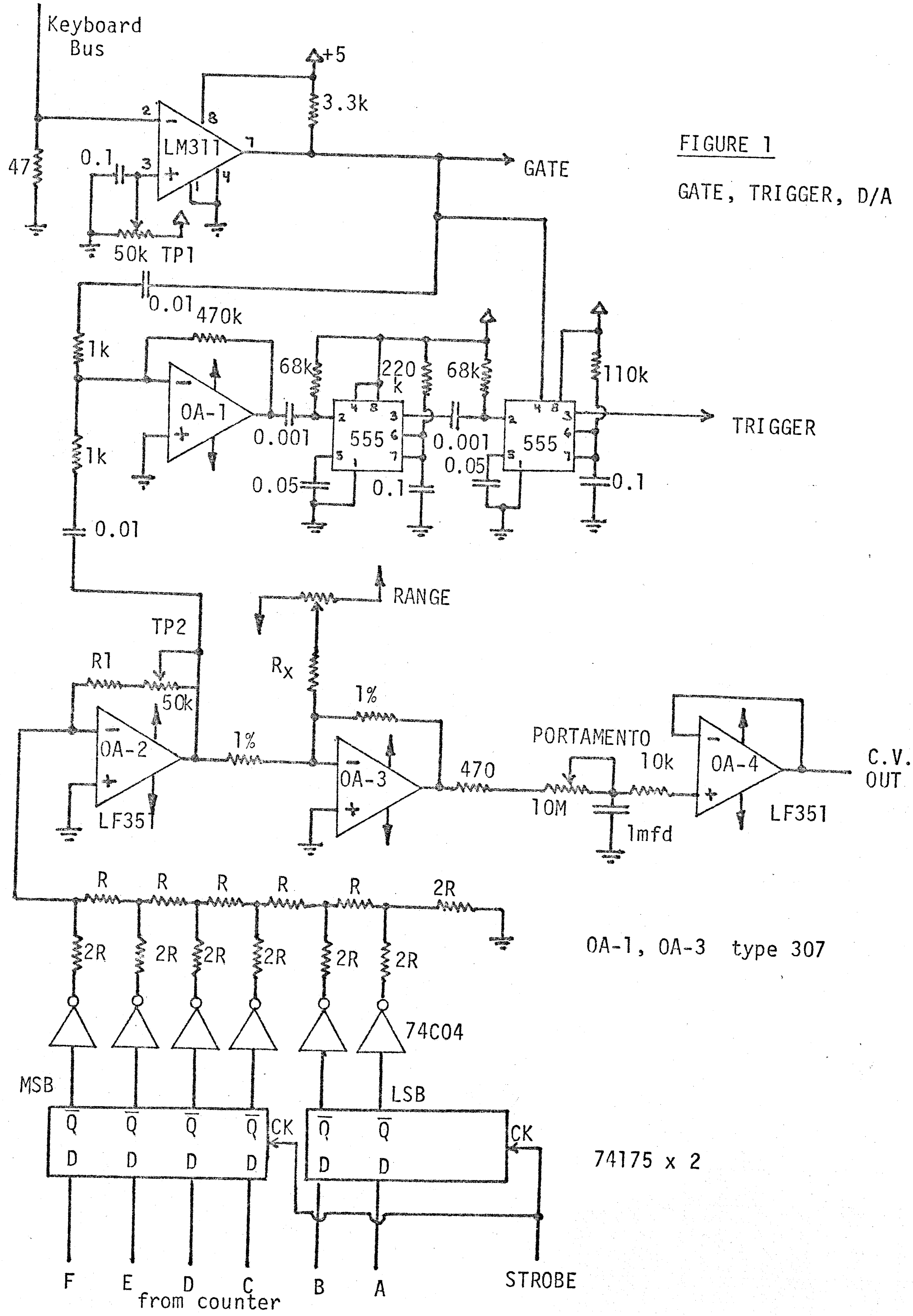
Jan Hall



Power Supply Input Ckt. and Regulator for ±6.5v

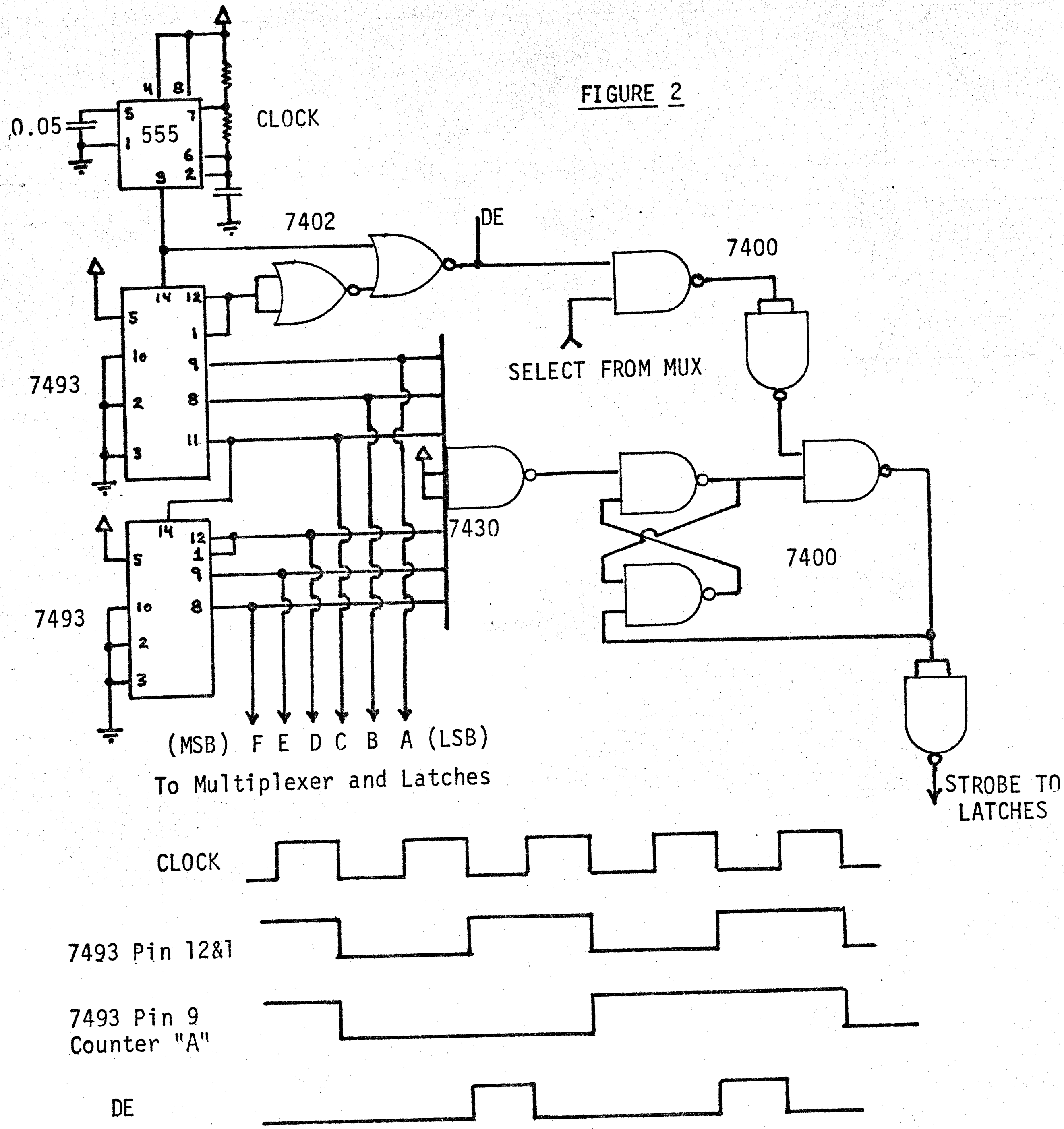
FIGURE 1

GATE, TRIGGER, D/A



bus goes to a 47 ohm resistor. This provides a logic "zero" to the 74150's but the voltage drop across the 47 ohm resistor is detected by the 311 comparator. TP1 is adjusted so that pin 7 of the 311 goes high when a key is pressed and low when released. The trigger is derived in a manner similar to the analog sample-and-hold interface in EN#45. OA-1 is a differentiator that detects a change in the D/A voltage or the rising

FIGURE 2



and falling edge of the gate. This triggers a 555 monostable and when that goes low, it triggers another 555 if pin 4 is high (key pressed). The net result is a trigger is issued when the D/A voltage changes or the gate goes high.

The lower portion of Fig. 1 is the D/A converter. I used 74175's for my latches. Taking data from $\overline{Q}$ outputs, only one inversion is needed (74C04). OA-2 is just an inverting summer. R1 should be approximately equal to R in the R-2R ladder and TP2 is adjusted for the proper gain. OA-3 is another summer for ranging. R_X is selected for the desired range, or you could use "Coarse" and "Fine" adjustments. OA-4 is just a voltage follower for portamento.

Figure 2 (above) is the clock, counter and scan control circuitry. It's essentiall the same as in the original circuit. I used 2 7493's for the counter. Instead of using the first divider output for the "A" or least significant bit, I've inverted it and NORe it with the clock. This provides a DE (Data Enter) clock as in the timing diagram. The data to the multiplexer and latches then change coincident with the next divider stage (pin 9). DE is then ANDed with the select pulse from the MUX. Since DE comes along some time after the data changes, this ANDing serves as a "Garbage Eliminator" and may also aid in debouncing without resorting to 555 one-shots. The rest of the operation of the circuit is the same as in the original.

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THE EAR - PART 1:

BASIC IDEAS ON PITCH PERCEPTION:

- by Bernie Hutchins

INTRODUCTION TO THE SERIES: Here we are starting a series on "The Ear" in which we will examine several areas relating to the perceptual abilities of the ear. We see this as becoming a fairly long series, but one which we will take in relatively small installments. We will begin with several installments on the perception of pitch. The importance of knowing what the ear can do, and how it does it (as much as is possible), is a point about which we really need make no further comment. In following this series, we expect that many readers will want to repeat many of the experiments suggested and make their own observations. Fortunately, electronic music synthesis equipment is well suited to this purpose. Making your own observations is not only educational and (to many) entertaining, but also will serve the observer by sharpening his hearing skills, engineers and musicians alike.

BASIC IDEAS ON PITCH PERCEPTION:

It often comes as a great surprise to engineers that pitch, while related to frequency, is not the same thing. Frequency is an objective propertly that is strickly assigned only to pure sine waves of infinite duration, but by common usage is used for finite duration signals which are both sinusoidal and complex, and is related to the repetition rate of these signals. This is a reasonable approach and one which the engineer is familiar with. Pitch on the other hand is subjective. It is a generalization or an abstraction - an attribute of a tone which obtains as a result of a natural process. We can become aware of a basic endowment of pitch simply by observing that a great many tones, different in numerous other properties, all may have the same pitch.

Following a mental device due to Terhardt, we can construct visual analogs for the pitch perception process. Consider an abstract subjective property of a tone which we will call pitch and to make it specific, we will call it "Pitch A." In the visual analog, we note that we can form an abstraction of "Pitch A" by forgetting about any possible meaning in the English expression "Pitch A," but rather just by considering it as a visual target. Thus, we are able to recognize the following visual devices as the same abstract thing visually:

(1) Pitch A

(2) *Pitch A*

(3) P I T C H   A

(4) PI CH A

(5) PITCH A

Cases (1), (2), and (3) are rather similar, and we are used to making a single abstraction where such similarity exists, and we do this every day (all chairs, etc.) Cases (4) and (5) are a little different. In (4), something is left out, while in (5), we have only an outline. The point is, the ear can get the right pitch, even in cases where something is left out, or where only an outline is provided. This is an indication of the strength of the generalization the ear can make.

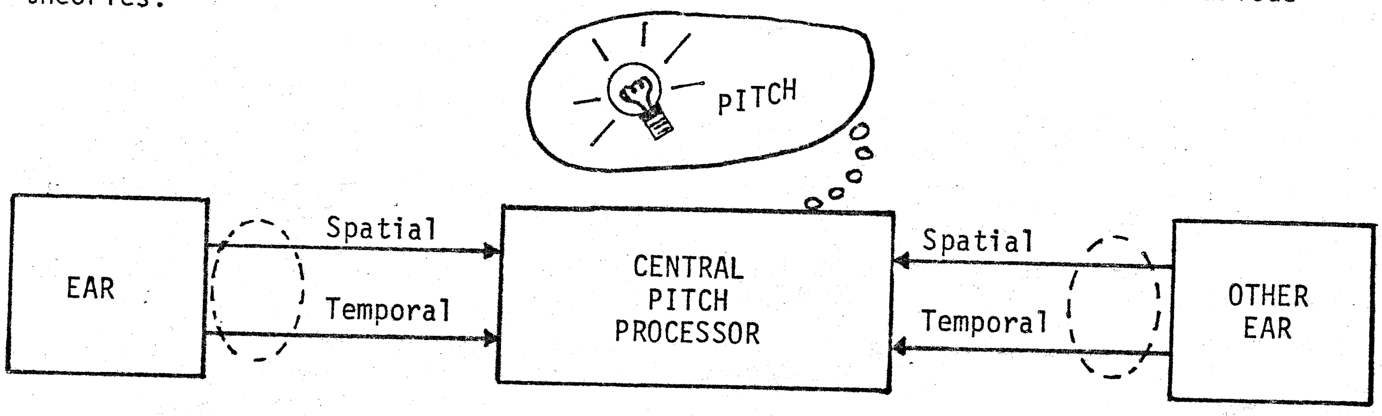
At this point, the interesting question comes up as to whether the abstracting ability of the ear is inherent or learned. In the visual analog, we know that at least some of our ability to identify the visual targets as "Pitch A" is due to our past experience. This is particularly true of target (4) where a letter is left out of the word. In the case of the ear, there are theories on both sides, some saying that the identifying mechanism is built in, others saying that it is a result of learning at an early age. Probably the truth is in the middle. To see this, consider that in a hearing experiment, the experimenter can point out a feature of a sound stimulus to the listener, proving (or at least - weakly indicating) that the listener can learn to hear something, while the experimenter's assurance that he will hear it indicates a built in feature of the ear.

Because we do not know exactly what the ear does, how it does it, or even if an objective definition of pitch exists or can be formulated, we must rely on experimental observations and try to explain the results with a model. Typically, a model comes first, and then an experimental test is made to find the limits of the model, which in turn must be refined again, and so on. This process has been going on for over fifty years (much longer, depending on your concept of the problem) and still the answer is not at hand. Furthermore, all the simple experiments probably have been done by now. In looking for tests of minor points of refined models, researchers must resort to more difficult experiments, requiring more complicated equipment, and more concentration on the part of their experimental subjects (or better "trained" subjects, possibly confounding the experiment in some way). Fortunately, we are in an age where electronics and computers at least take some of the weight off our shoulders. None the less, study of pitch perception is not a field one enters literally overnight, but perhaps overyear would be more like it. There is much literature to review and many ideas to understand before any useful new work can even be considered. It is the goal in this series to give a brief introduction and offer some suggested experiments.

It would be nice if we could say that we hear the pitches we do because the ear acts as though it were an XYZ instrument, where for XYZ you can consider things like frequency counters, spectrum analyzers, and autocorrelators. Things are not this simple however. Evolution has resulted in a human ear that is marvelous in efficiency as well as redundency, accuracy as well as crudeness, and refinement as well as versatility. While the goals of this evolutionary "developmental engineering" need not be considered here (they were probably pure survival), our view should be not that the ear is a poor instrument, because it does not seem to do anything we know about from our engineering experience, but rather that it is a unique and remarkable device which (at least) makes it possible for us to hear music. We must realize that as engineers we can say nothing about devices that receive and process information of esthetic importance (musicians can't say anything about them either for that matter). Rather, we must say that here is an example of one such device, and we wish to understand it for academic reasons, for theories of music, or for practical purposes such as building electronic pitch extractors as musical controllers for electronic musical instruments. The ear is something to be appreciated and understood as much as possible, and not to be criticized for what it is not.

We can make long list of what the ear is not, as a matter of fact. What can we say about what it is? It is for certain, at least a crude spectrum analyzer. The cochlea (inner ear) acts as a spectrum analyzer in that for a given pure tone, a certain spatial portion of the cochlea serves as a homing place for the frequency of the tone. Yet this is not enough because, for one thing, the frequency resolution of the ear, as determined by its ability to detect changes of frequency, is much greater than the frequency resolution of the cochlear spectrum analyzer. Thus, we get from the cochlea a spatial pattern of nerve firings that is something like a crude spectrum of the input sound. Obviously, we have to find a sharpening mechanism somewhere, but this we will not consider now. One other thing that we know the ear does is to provide a time ordered sequence of neural firings that is or at least may be a coding of the input waveform. Finally, we know that there is at least some processing that takes place outside the ears, somewhere in the brain. (To play it safe, we often refer to the "ear-brain.") We know this processing takes place because we can observe effects which take place as an interaction of two different sounds where these sounds are input to the two different ears.

Combining the ideas in the paragraph above, we have a simple-minded model of the pitch perception system as shown in the figure below. Start with the ear. The ear is transmitting information to a "central pitch processor" in the brain. There are two types of information. First there is the spatial pattern - a report of which places on the "basilar membrane" in the cochlea are being stimulated. Secondly, there is a sort of waveform code, an instantaneous report of the position and the velocity of the moving basilar membrane, based on the firing rate of the nerves on the membrane. In the figure, we show a dotted circle around both these transmission links from the ear to the central pitch processor, an indication that this is really just one "pipeline" and that the spatial and temporal reports are different aspects of the same thing. We must emphasize that we do not know for sure how the central pitch processor works, or how it uses spatial and temporal information. We do know that it "sharpens" the information, and is able to generalize and fill in missing details. Pitch perception theories that work with spatial patterns are called "Place Theories" while those that work with the temporal patterns are called "Fine Structure Theories" (among other names for both). In later installments of this series we will look at some observational results, and then will get into more detail on these various theories.



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READER'S COMMENTS:

▷ I know you have made a point about it at various times before, but I think it should be pointed out again that timbre modulators of the type you have been publishing have many advantages over voltage-controlled filters. I have built all of your designs and some of my own, and my feeling now is that we should stop making filters altogether. I get every bit the dynamic effects with the modulators that I do with the filters, and as you point out, they <u>don't have to be made to track</u>. They have no exponential converters, not even any voltage controlled pitch tracking at all. They are simpler, cheaper, and do the same jobs. What more can I say? Please point out these advantages again to your readers.

-Norman Rose

> One of my favorite pastimes is to examine different electronically closed loops of the type of which a phase-locked loop is one example. These can be examined by analysis or by experiment. This is a good thing to suggest to your readers who are looking for new things to do with their equipment. For example, try letting one VCO modulate a second, and this a third, and so on, and then finally, close the loop so that the one on the far end modulates the first one. The result is surprising.

Another good trick is to rig up an analog multiplier as a divider, and use this in loops. By the way, what do you get when you divide one sinusoidal by another? You get a sine times a cosecant of course, and that blows up at certain points. Using the divider, it just saturates.

-Willis Norcross

▷ I put an SR-52 calculator to work figuring out rational approximations to irrational quantities. Here are some approximations to the twelfth root of two, the ratio of semitones in the equal tempered scale.

3118/2943 1461/1379 1265/1194 1069/1009 873/824

Also, e ≈ 2721/1001.

-Nick Bodley

▷ In reference to your Hints on Setting Up an Electronic Music Working Area in EN#82, I thought you might be interested in the following concerning patch cord storage: Isao Tomita drapes his cords over the pegs on an ordinary hat rack, the pole type which can stand alone in the corner of a room. Its main advantage is that it is movable.

-Robin Graham, Roland Corp.

▷ I was interested in the reader's question about the musical effect of different low-pass filters, and in your reply. Are you familiar with John Bournes' MS thesis at M.I.T. (which claims that instruments are recognized by the "onset times" of the various harmonic envelopes), or with the Stanford AI Lab research on the extent to which simplified envelopes for the harmonics can yield good synthetic imitations? As the frequency roll-off curve of a low-pass filter slides upward to include various harmonics, there is a corresponding time-domain amplitude envelope for each harmonic. For example, if the filter's frequency response were ideal (zero above cutoff frequency), each harmonic would pop in abruptly at some delay time. It seems to me that the typical LPF (12 or 24 db/octave) is not sharp enough to give well-distinguished onset times for the successive harmonics. Fortunately, it is likely that harmonics within the ear's "critical bandwidth" of about 1/3 octave are not seriously distinguished by the ear anyhow, but regular filters aren't even this sharp. Attempts to sharpen the filter by applying feedback (or otherwise using higher Q) do not really give the desired effect. For this purpose, 36db/octave filters, preferably with the damping of each section separately adjusted, are of interest. How about Cauer (elliptical) filters? I'm about to build some goodies to aurally explore some of these areas. Example: a wiggle or dip in the rolloff curve could give a "horn blip" in the time domain envelope of each harmonic. Furthermore, there's no law that says the rolloff curve must be a constant shape as it is swept, or that the filter sections must be swept identically.

Anyhow, the fact it's the low-amplitude initial activity in each harmonic that is aurally significant is consistent with your intuition that the filter's ultimate rolloff, not its corner performance, is the critical factor.

A related area I'd like to explore is whether the hi-Q resonance of a typical swept LPF causes inharmonic components to be generated as the excitation energy bounces around in the sweeping filter. Perhaps this accounts for some of the icky overworked "wah" sound, and perhaps that would go away (while retaining a pleasant timberal attack) if a low-Q but sharp filter (such as the elliptical filter) were used.

-Chuck Cooper

EDITOR'S NOTES: The reader's question referred to above appeared in EN#86, pg. 22. It is also of interest to consider the initial results of using the 6th order, 36db/octave "high-ripple Chebyshev" filter in EN#90, which tends to bear out what Chuck suggests above.

▷ While I do feel you have the right to publish anything you want, I am mainly interested in the technical material, and not the philosophical discussions and the music reviews, etc. Actually, a little bit of the latter goes a long way with me, and I guess you don't carry that much, but I would hate to know that one of your superb technical articles is pushed out by non-technical material.

-name withheld

EDITOR'S NOTES: Technical material always gets priority. We have never held up a real interesting piece of technical material for less urgent material. Remember that the real excitement in this material <u>for us</u> is telling <u>you</u> about it.

▷ I wonder if the musicians among your readers are aware of the high cost of engineering books. It is not at all unusual for an engineering text book, even of relatively modest size, to cost from $20 to $35. Engineering students are aware of this, and accept it, like it or not. On the other side of the coin, many music books run well under $10. I became aware of this when someone expressed the feeling that your <u>Musical Engineer's Handbook</u> was expensive at $18. After telling him that $18 is cheap for an engineering book, he told me that one music professor was able to keep the size of his class down by selecting a text that cost $12.95! I don't think that we can put a price on the value of information, either to ourselves or to anyone else, and I don't know why some books cost more than others. However, since you have both engineers and musicians in your market, it might be well to point out the relative costs of different texts.

-L. W. King

▷ One of the best things I have seen in Electronotes was the interview with Bob Moog a few years back. In itself, it was very interesting, but also it was rather unique to have anyone interviewed about engineering as such. I would like to see more interviews. Perhaps some of your readers can suggest interviews and even conduct them for you if they are closer to the person to be interviewed than you are.

-L. W. King

CLASSIFIEDS:

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