# ELECTRONOTES 93

NEWSLETTER OF THE MUSICAL ENGINEERING GROUP

1 PHEASANT LANE

ITHACA, NEW YORK 14850

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## GROUP ANNOUNCEMENTS:

CONTENTS OF EN#93



This issue is an issue and a half - next month's issue will be just a half to compensate. In the next issue, we will be giving details on our publishing plans for 1979. We have decided to keep a monthly publishing schedule, but there will be only eight mailings total, so some issues will be combined and will be smaller in total pages. This is being done to save postage costs, and we expect that much of the lost editorial material can be made up by size reduction. No, we won't be reducing the size of circuit diagrams for construction, so those of you who went out and bought microscopes during our first years of publishing will not have cause to get them off the shelf again!

In this issue, we have a couple of features that follow up on our first part of the series on the ear - again these are concerned with the perception of pitch. The third report is another of our animation devices, and is a form of "Generalized Resonator."

NEW MEMBERS AND CHANGES (c):

| | |
|---|---|
| R. Charles Blakey | 13 The Brooklands, Wren Green, Preston, Lancs, England |
| Steven C. Denney | 6348 Holbrook Dr., Dayton, OH 45424 |
| T. A. Donofrio | 150 Wedgewood Circle, Eatontown, NJ 07724 |
| Hal Freedman | 415 W. 118th St. Apt 2, New York, NY 10027 |
| Steven George | 1611 Walnut St., Berkeley, CA 94709 |
| Alan K. Hunt | 10 Grant St., Potsdam, NY 13676 |
| Robert Jacobs | 710 West End Ave., New York, NY 10025 |
| Jerry Lemay Music | 611 Charles St. Apt 1, Fall River, MA 02724 |
| Nick Martello | 706 North Chillicothe, Aurora, OH 44202 |
| Douglas Masla | 551 Wilmot Rd., New Rochelle, NY 10804 |
| Mari & Gregg D. Moore | 1946 Homewood Dr., Altadena, CA 91001 |
| Kent Nickerson | 73 Compata Way, Ottawa, ONT, Canada K1B 4X1 |
| Paul D. Nuber | 2302 University Ave Apt 346, Madison, WI 53705 |
| J. Robinson | Kimball Electronics, 1038 E. 15th St., Jasper, IN 47546 |

| | |
|---|---|
| Jim Speedy | 2509 McCallum. Austin, TX 78703 |
| Lloyd E. St. Jean | 400-26 River Ave Apt 100, Potchogue, NY 11772 (c) |
| Jeffrey Zurita | 200 E. University Blvd #208, Melbourne, FL 32901 (c) |

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## NEWS AND NOTES:

The Boston School of Electronic Music is offering courses as usual after being "evicted" by a fire at their old location. They are now offering an expanded curriculum with more equipment, and a new course is designated "Synthesizers for Humans." Their new location is 28 Highgate St., Allston, MA 02134 and the phone is 617-782-9100.

"Evenings for Electronic Muisc" are being held at the Philadelphia College of the Performing Arts. The next dates for these evenings are Dec. 7, Feb. 1, and March 1. The programs are held at 7 PM in Room 604, 250 S. Broad St., in Philadelphia.

A small two-man shop specializing in the design and construction of custom synthesis products has been formed and is called "On Key Enterprises" and is located at 12628 SE 42nd, Bellevue, WA 98006.

Polyphony has issued a collection of patch diagrams in its new book, The Source. The book is available from Polyphony Publishing Co., 1020 W. Wilshire Blvd., Oklahoma City, OK 73116 at $4.00 postpaid. Like Polyphony magazine, the book is oriented toward PAIA equipment, but as we have pointed out in the past, the diagrams they have published are one of the few sources of ideas for achieving unusual sounds, and many useful ideas are contained here. If you have not followed these in Polyphony, this book will be well worth looking at.

Sequential Circuits is looking for employees as synthesizer production technicians. Analog experience is necessary, and microprocessor experience is desirable. Contact Dave Smith, Sequential Circuits, 1172 G. Aster Ave., Sunnyvale, CA 94086.

INTERVAL / A Microtonal Newsletter is being published on a quarterly basis and is edited and published by Jonathan Glasier. The subscription rate is $8 per year and the address is PO Box 8027, San Diego, CA 92102.

Due to the large amount of interest in guitar electronics, our readers will be interested in DEVICE, the Newsletter for the Electronic Guitarist/Musician which is being edited by Craig Anderton and Roger Clay. Subscription rates are $15/year in the USA, $16/year for Canada and Mexico, and $18/year international. Free sample issues will be available after Jan. 1, 1979 from Device, PO Box C, Carmichael, CA 95608.

The Zetaphon is a polyphonic guitar synthesizer manufactured by HEAR. Write Holt Electro-Acoustic Research, 1122 Univ. Ave., Berkeley, CA 94702 or phone (415)-848-6262.

The winter session at Dondisound Studios Inc., 12 St. John St., Red Hook, NY 12571 includes an intensive short course in recording studio operations to be held Jan 2 through Jan 20, 1979. Tuition is $450 and their phone is 914-758-5167.

## READER'S QUESTIONS:

Q: Can I run your circuits on ± 9 volts instead of ±15?

A: Probably many of them could be run as is or with minor changes on ±9, but I wonder why if you are investing in new circuits why you don't make a new supply. These ±15 supplies are not difficult or too expensive. The only place you might need ±9 is where you must run a small circuit on batteries, and two 9v transistor batteries will do. By the way, if you run on two 9v batteries this way, you have to use a DPST switch to cut the power. Just switching out ground won't prevent the overall loop that will continue to drain the batteries even when off.

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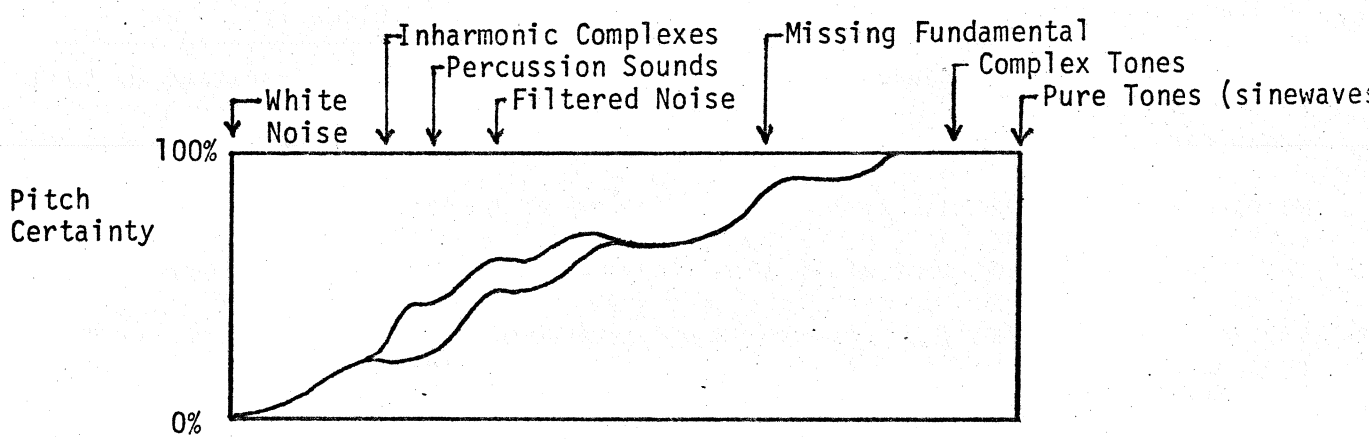
# THE EAR - PART 2:

## THE OBSERVATIONAL BASIS FOR PITCH PERCEPTION THEORIES: -by Bernie Hutchins

"EVERYDAY" OBSERVATIONS:

In Part 1, we made a point of the fact that frequency is an objective (physically measurable) property of a sound or signal, while pitch is a subjective (mental) property of the same sound. The reader should be clear in that we are not saying that pitch and frequency are subjective and objective properties, respectively, of the same thing, but rather that they are two different, but related things.

By the time the engineer, researcher, or musician first thinks about the true nature of pitch (if indeed, he ever does), he has already heard pitched sounds all his life and every day of his life. He is skilled at doing the thing he is about to investigate. Music is probably the most important case where he has heard pitched sounds, at least as far as pitch training goes. For everyday living, pitch can also be an important aspect of speech. Add to this the squeaking doors, animal sounds, and even the wind as it "whistles" around a corner, and you can understand the barrage of pitched sounds we hear all the time. There are also many unpitched sounds, which we often consider to be noise. These include the "crunches" and "thuds" as well as many scraping noises where nothing really gets going on a regular basis - there is no vibration after impacts, nor any "slip-and-stick" action such as the sound of chalk squeaking on a blackboard. We can make a highly schematic "graph" of various sounds where we plot "pitch certainly" which is a measure of whether or not a sound is pitched and the "strength" of the pitch. The plot is shown below:




The plot above is not to be considered to any scale, but just indicative of the types of sounds and their possible pitch content. On the left, we have white (flat spectrum) noise with no pitch whatsoever. On the right, in contrast, we have the pure tones (sinewaves) and the complex tones (periodic waveforms consisting of a fundamental frequency plus integer multiples of this frequency - the harmonics) which possess virtually 100% pitch certainly. Already we can see the close relationship between pitch and frequency because tones normally assigned objective frequency have high pitch certainly while white noise, with no frequency features at all, has no pitch at all. It is thus the middle region that really deserves our attention. It is here that strange and interesting things happen, and where observation will lead to the most significant discoveries about pitch perception.

In the middle region, on the right side, we have tones with missing fundamental. Especially if there are a large number of fairly low order harmonics, we can expect a strong pitch certainly, even though there is no spectral energy at a frequency that corresponds to the fundamental. Finally, there is a region of filtered noise, inharmonic complexes (tones whose frequency components or partials are not integer multiples of a fundamental) and percussive musical sounds. Here we show a double branch in the curve of certainty to indicate that not only may the pitches be obscure (of low certainty), but also that more than one pitch may be heard by different individuals or at different hearings by the same individual.

We should, at this point, inquire about the position of pitched musical sounds. Nearly all such sounds fall in the region of the complex tones and the missing fundamental. Most musical sounds are complex tones in that they have at least some energy at the fundamental frequency, and in addition, musical sounds have the added dynamical effect of having harmonics whose amplitudes change relative to each other and relative to the fundamental. Yet some instruments (notably, the violin!) have relatively weak fundamentals, and the missing fundamental phenomenon has been exploited for many years by the builders of pipe organs who produce lower fundamental frequencies, or reinforce them, with shorter pipes which produce a series of harmonics. Also in the realm of traditional acoustical musical sounds, we find the percussive sounds, a remarkable group of sounds which consist of partials (and possibly some noise) which combine to produce sensations of varying degrees of pitch certainty. [Note that when we say partials, we mean component frequencies, which are not necessarily integer multiples. If they are integer multiples, they are called harmonics. In fact, most traditional musical sounds are composed of partials, not harmonics, since the spacing at integer multiples is seldom exact. However, they are close enough that the terminology as harmonics is acceptable.] Finally, we mention in passing the fact that in electronic music, any and all types of sound can be expected to find use in musical composition.

For the moment, let's consider just how much of our everyday experience we can explain in terms of place theory, and then we will go on to discuss some laboratory type listening experiments. Place theory, which can be traced back to Ohm and von Helmholtz, assumes that any frequency has its own homing place in the ear - the idea of a spectrum analyzer. Clearly we can explain the pitch response to pure tones with this model, and also the response to white noise can be understood in terms of no preferred locations (all locations equally stimulated). The response to complex tones can also be understood if we assume that a simple inhibition mechanism blocks responses to higher frequencies when a lower frequency is stimulated, but even here we have to reconcile this with the response to white noise. Even greater problems come up when we consider the missing fundamental. We would have to have a mechanism that looked for a constant difference in frequencies. It rapidly becomes clear that we must look to neural processing at a higher level (i.e., in the brain) since simple mechanical functions of the known structures of the ear will not perform in a manner that explains the sophisticated behavior of the pitch perception mechanism. None the less, note well that place theory serves quite well for most sounds of everyday speech, music, and environment.

## OBSERVATIONS ON A SINGLE PURE TONE:

The simplest tone of all is the pure tone or sine wave. Although sine waves isolated as pure tones are seldom present in even electronic music, we can learn a good deal by listening to a single tone. When we listen to a sine wave, as we might by changing the output from a synthesizer from sawtooth to sine, two things immediately impress us: the smoothness, and the weakness of the sensation. Due to the lack of harmonics, the sine wave is very smooth compared to complex waveforms, and consequently lacks a "bright timbre." This smoothness, along with the relative insensitivity of the ear to low harmonics (to low frequencies in general actually) combine to give the sine wave an overall impression of weakness, and many synthesizer users will be moved to check to output amplitude to see if in fact something has gone wrong with the output stage.

In addition, a single sine wave can be used to examine the frequency resolution of the ear. Depending on the circumstances, (rate of change and smoothness of change) changes of frequency of from 3% at low frequencies, to as little as 0.5% to 0.02% (two parts in 1000) at high frequencies can be detected. These figures are much more impressive than one would expect from the known (mechanically observed) resolution of pure tone excitations on the basilar membrane of the inner ear. This points out the necessity of finding a sharpening mechanism as we mentioned before.

In addition, we can use a pure tone to examine the dynamic range of the ear, and will find a rather astounding range of 120db or more. Finally, we can use the pure tone to examine the "aural harmonics" effect. This is simply, the generation of harmonic distortion as a result of non-linearities in the ear, a rather subtle effect. To make this test, it is necessary to have a rather pure sine wave (if you use a synthesizer sine wave, be sure to filter it with the VCF) and an amplifier capable of producing a relatively loud signal to the ear <u>without</u> introducing non-linearities on its own. The difficulties in achieving this effect are most important in that they tell us that non-linearities are minor, and can not necessarily be used to explain observed effects. We will use non-linearities successfully to explain "combination tones" and unsuccessfully to explain the missing-fundamental effect.

## <u>OBSERVATIONS WITH TWO PURE TONES</u>; <u>BEATING</u>, <u>ROUGHNESS</u>, <u>AND DISCRIMINATION</u>:

Things get a little more interesting when two pure tones are applied to the ear simultaneously. We will assume here that both tones are mixed together and then presented to both ears at the same time. However, we will note at times, that the same results, or similar results may occur when different tone are applied to separate ears, or that certain effects will not appear if the tones are applied to separate ears. The use of two different ears in this way is one of the easiest and most significant tests that can be performed for different hearing effects.

For our observations, we will start with one fixed pure tone at frequency $f_1$, and to this we will mix, with equal amplitude, a second pure tone of frequency $f_2$ where $f_2$ varies from $f_1$ to $2f_1$ or a little higher. This is an experiment that we suggest you do before reading further. Even if you don't do this experiment right now, it would be an excellent idea to stop for a moment and list the effects you expect as $f_2$ varies. After making the list, check out your ideas experimentally if possible, and then come back and read below. Note that your electronic music synthesizer VCO's will suffice here - perhaps you may want to low-pass filter the fixed frequency with the VCF.

The thing that should have been on everyone's list of expected observations is a "beating" effect as $f_2$ rises just a few Hertz away from $f_1$. While mixing (adding) the two sine waves is just a simple superposition, and mathematically the ear is never presented with anything except frequencies $f_1$ and $f_2$, what we hear is a slow beating of amplitude at a pitch that is the average of $f_1$ and $f_2$. The beat rate is the difference frequency $f_2$-$f_1$, and this is easy to understand on the basis of the interference of the two frequencies. To understand the fact that the pitch $(f_1+f_2)/2$ is perceived, we must consider that these two frequencies are too close together to be separately resolved, and that the time constant of the ear is too short to analyze the effects of the beat envelope. Beating effects can be heard easily for difference frequencies of from 0.1 Hz to about 10 Hz, and with concentration can be heard up to about 16 Hz. It is significant that if we input separate tones to separate ears, the beating stops. This indicates that these "first-order" beats are associated with something going on in a single ear structure.

When the difference between $f_2$ and $f_1$ exceeds about 16 Hz, the general effect of the beating continues, but now you are less conscious of a beating of amplitude, and more of a general feeling of "roughness." At this point, it would be useful to run the frequency on up until the roughness seems to be gone. This will occur where $f_2$ is about 1.25 $f_1$, a musical minor third. You can now "hear out" the two tones individually, and it is said that you have "discriminated" them. This discrimination is easiest where roughness has disappeared, but if you turn $f_2$ back down some into the roughness region, you will hear that you can discriminate the two tones even in a portion of the roughness region. Depending on $f_1$, you will find that you can discriminate the two tones in the roughness region down to a half-tone in the region of 300 - 500 Hz, a musically important region, and to a full-tone for most of the rest of the musical range. [A half tone is about 1.06 $f_1$ while a whole tone is about 1.12 $f_1$ ].

There are some important points to look at here. First, note that we can discriminate tones only when they are separated by at least 6% while we can detect a change in an individual tone of only 0.5% or less. If the basilar membrane is capable of discriminating two tones 6% apart, this would be an indication of a 3% resolution, and yet we can detect a change of 0.5%, indicating a sharpening mechanism. A second point is that conventional 12-tone scales space the tones at half-steps which pretty well corresponds to the ear's ability to discriminate tones. Finally, if we associate musical dissonance with roughness, note that the transition point from roughness to smoothness (with tone discrimination) is at about a minor third, the traditional region. While we can not make too much of this, the result is rather striking, and certainly makes it easy for us to remember the general magnitudes of these numbers.

For the moment, let us summarize the results where $f_2$ is less than a minor third, and at the same time present the traditional terminology. Two tones are discriminated when their difference is on the order of a half to a whole-tone, and this is called the "limit of frequency discrimination." The sensation of roughness, which takes over at a frequency difference of about 16 Hz, does not disappear, but rather fades at a frequency difference of about a minor third, and this is called the "critial bandwidth." This pretty well covers the territory, slow beating giving way to roughness and then smoothness at about a minor third, with tone discrimination in the middle of the rough region. Other effects in the rough region may be present, but are masked by the roughness.

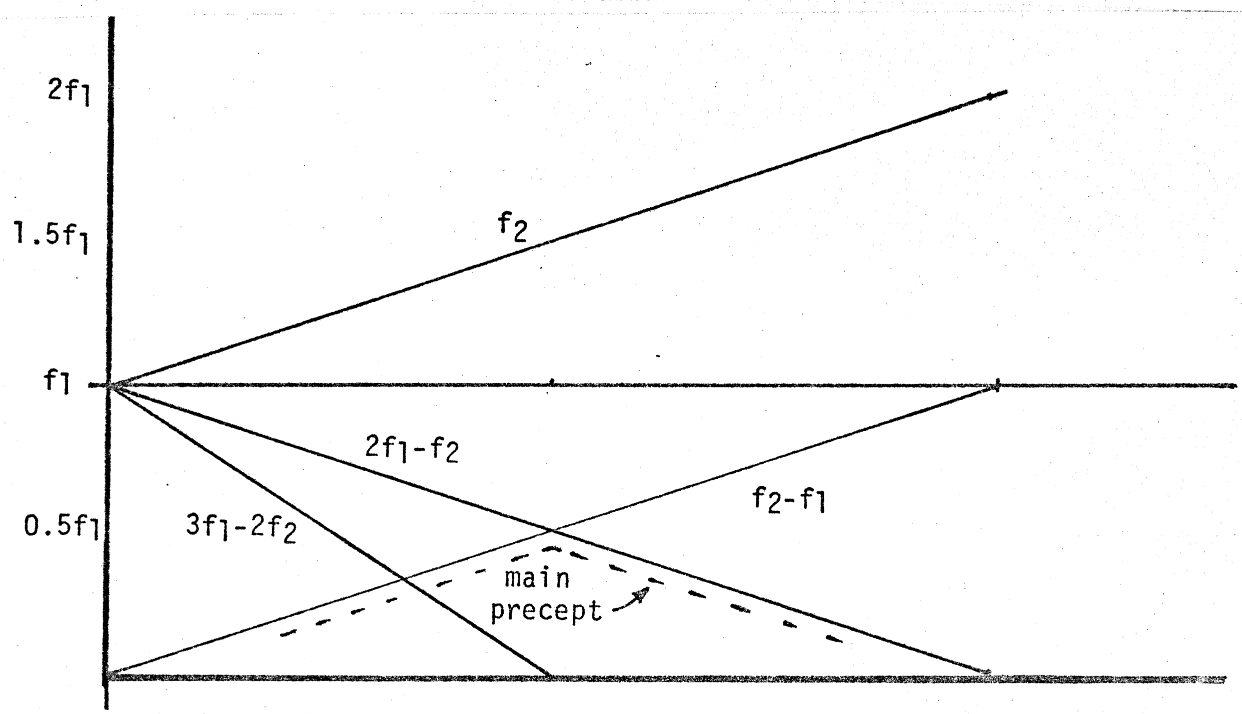
<u>BEYOND THE MINOR THIRD</u>; <u>SECOND-ORDER BEATS</u>, <u>COMBINATION TONES</u>:

It might be useful here to check your list of the effects you expected. Probably some readers expected things similar to the effects of frequency discrimination and critical bandwidths, as well as beating. Above the minor third, you probably have things listed which you are not going to get, and are going to get things you did not have listed. One of the reasons for this is that we are using pure tones, and the things you listed you probably expected based on your experience with musical tones, which are complex (and more).

A quick scan of the region from the minor third to the octave will convince the listener that nothing very dramatic is happening, but a more careful sweep will bring out some interesting effects. These will include some weak beat-like effects that you will hear around the octave, and also down around a fifth (1.5 $f_1$), a fourth (1.333.. $f_1$), and possibly around the major third (1.25 $f_1$) as well. In addition, you can hear some tones <u>lower</u> than $f_1$, and these may be moving <u>down</u> as well as up when $f_2$ goes up. The first effects are called "second-order" or "subjective" beats while the latter effects are called "combination tones." The second-order beats are evidently the result of neural processing, since they occur even when the two tones are fed to separate ears. On the other hand, the combination tones seem to be the result of nonlinear response in the cochlea of the inner ear.

The most striking fact about the second-order beats is probably that they are so mild. Possibly you had on your list of expected results some dissonant effects around the fourth and fifth, and generally, dissonance where $f_2$ is not a simple integer multiple of $f_1$. In fact, there is no roughness, and no dissonance - things are very smooth! Yet, musically, we expect something like a minor fifth to sound dissonant. The simplest explanation here is that we are using pure tones, not complex musical tones. If we were using complex tones, there would be actual first-order beats occuring between upper harmonics of the complex tones, and this would result in slow beats or roughness, and a subjective feeling of dissonance. Here we have only the weak, subjective beats of neural processing. If we examine the waveforms of close to integer ratios, we find that the "beats" thus produced are devoid of large variations in envelope amplitude, except near a 1:1 ratio. The second-order beats are rather subtle, and their exact cause is not understood. The important thing about them is that they are caused by neural processing beyond the ears, and not by actual interference on the basilar membrane, and thus there is no roughness or dissonance at these simple ratios for pure tones. It is worth noting that second-order beats do occur even near a 1:1 ratio and can be heard when the two pure tones are presented separately to the two ears.

To understand "combination tones," it is necessary to understand that when two frequencies are input to an electrical (or mechanical) system that is non-linear, a process which would otherwise be a simple summation, results in frequency components that are linear combinations of the input frequencies such as the sum, the difference, and many others such as $2f_1 - f_2$, $3f_1 - 2f_2$, etc. Because nonlinearities do exist in the inner ear to a degree, we can expect some extra frequencies, especially if we input relatively loud sounds to "drive" the ear harder. For reasons not understood at the present time, these combination tones in the ear do not appear at sum frequencies, but only at difference frequencies with $f_2-f_1$, $2f_1-f_2$, and $3f_1-2f_2$ being the easiest to hear. The easiest of these to hear is the difference tone $f_2-f_1$, which starts at zero and goes up to $f_1$ as $f_2$ goes to $2f_1$. You can hear this tone just beyond the roughness region. When $f_2$ hits $1.5 f_1$ (the fifth), the combination tone $f_2-f_1$ coincides with the combination tone $2f_1-f_2$, which has been moving down from $f_1$ to zero. At this point, you will likely hear what you suppose to be the original rising difference tone ($f_2-f_1$) reverse direction and begin to fall. Actually, you are now tracking the other tone ($2f_1-f_2$) downward. The diagram below will illustrate this main impression. Careful listening will permit you to make out some portions of the other combination tones.




## THE (NOTORIOUS) MISSING FUNDAMENTAL:

Probably there is no observation more important to the development of modern pitch perception theory than the study of the missing fundamental and related effects. Simply stated, when you present the ear with frequencies that are integer multiples of a fundamental f, even when the frequency f is completely absent from the stimulus, you still hear a pitch corresponding to f. The effect is persistent and continues even with only high-order harmonics (e.g., 9f, 10f, and 11f). The effect is important for musical instruments such as the violin and oboe, and is present in telephone speech where the limited lower bnadwidth of the telephone channel often removes the fundamental of vowel sounds, without loss of intelligibility of the speech.

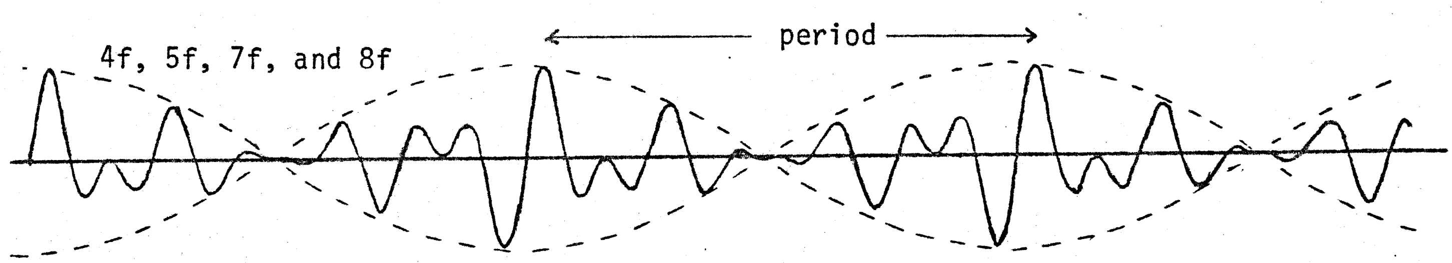
In an attempt to explain the missing fundamental effect, von Helmholtz and others (over 100 years ago) quite logically suggested non-linear distortion as a mechanism for supplying real spectral energy at the fundamental frequency. The hypothesis here was that the middle ear was non-linear, regenerated the fundamental, and the cochlea of the inner ear actually was stimulated with the fundamental frequency, exactly following the ideas of place theory. A glance at the figure above on combination tones will show that to a degree, this is true. Note that when $f_2$ is $1.5f_1$, we have the equivalent of harmonics 2f and 3f, and note that the difference tone is $0.5f_1$, corresponding to the fundamental. Yet, 40 years ago, an overwhelming degree of evidence was uncovered to indicate that the non-linear effects are too small, and in other ways inadequate, to explain the experimental observations.

The initial evidence against nonlinear distortion as the mechanism for restoring the fundamental was found by Schouten, and additional evidence has been discovered since then. This caused a void in the structure of hearing theory, and it was necessary to retrieve some earlier ideas of Seebeck which dated back before Helmholtz, and Schouten's modern version of this idea of "repetition pitch" led to "residue" theory, and later other "fine-structure" theories.

The most important observations against the nonlinear hypothesis of restoring the missing fundamental include information on the level at which combination tones can be detected, and the failure of attempts to interfere with the supposed regenerated fundamental. Once the existence region of combination tones was determined, it becomes evident that nonlinear effects exist only for relatively high amplitudes while repetition pitch, the pitch of the missing fundamental, exists even at low amplitudes. [Note that combination tones exist continuously, even in cases where a missing fundamental would be subaudio. For example, if $f_2$ is $1.222 f_1$, then the difference combination tone $f_2 - f_1$ is $0.222f_1$ while the "missing fundamental" would be $0.002f_1$. Thus there are many regions where combination tones exist where a missing fundamental, in the usual observable sense, does not exist.] In addition, if the missing fundamental is the result of a nonlinear process in the ear itself, then it should exist, physically and objectively. in the ear, and would be subject to beating with inserted frequencies close to the missing fundamental, and could be "masked" by a band of noise covering the missing fundamental. In fact, efforts to interfere with the preceived pitch in this way are unsuccessful.

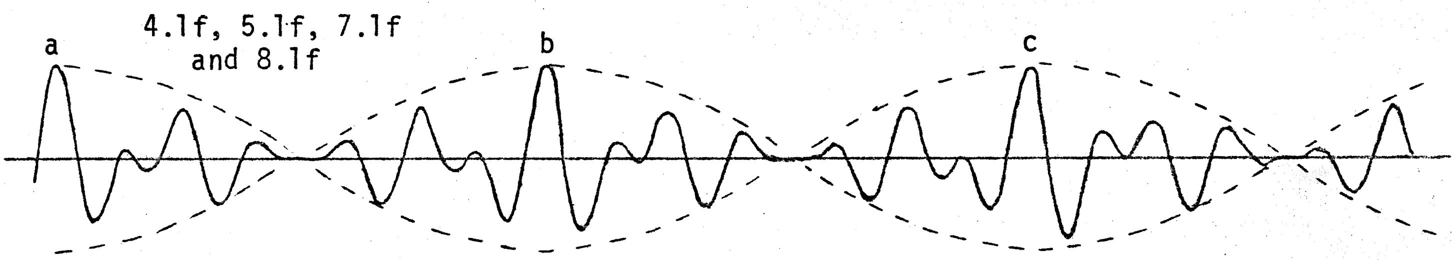
THE PITCH SHIFT OF TONAL COMPLEXES (TO MAKE MATTERS WORSE):

If we add together two frequencies $f_1$ and $f_2$, we get a beating effect which is an audible change of amplitude for small differences, roughness for larger differences, and may be a pitch for still larger differences. The beat pattern is characterized by its "beat envelope" which varies at a rate equal to the difference frequency. This beat envelope is present even for many superimposed frequencies. For example, the waveform below shows the superposition of 4f, 5f, 7f, and 8f, where f is the missing fundamental. The sum of these four frequencies repeats at the rate f, and this can be




seen in the drawing as a beat envelope or an amplitude envelope, which is drawn in as a dashed line in the figure. Any combination of frequencies that are integer multiples will have an envelope of this type. It is thus tempting to associate this envelope periodicity with a pitch perception mechanism, since it explains the missing fundamental. However, this does not work out in other cases, and we should point out that examination of the fine structure in the waveform (the time distance between the maximum peaks, for example) will also indicate the period of the fundamental.

We now want to discuss the pitch shift effect, sometimes called the "first effect of pitch shift." In our waveform, we summed 4f, 5f, 7f, and 8f. What happens if we shift all these frequencies by the same amount? Let's choose 4.1 f , 5.1 f , 7.1 f , and 8.1 f. The sum is shown in the figure below:



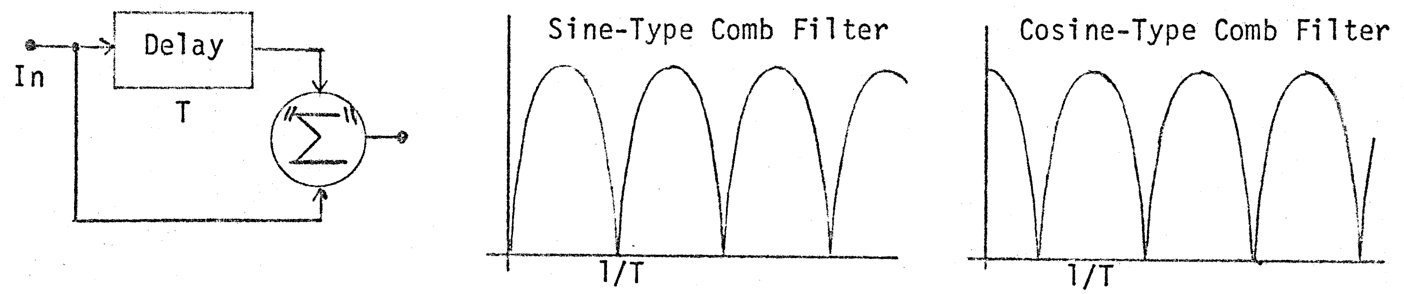

From the figure, we see that the envelope periodicity is exactly the same as it was when the frequency components were integers - we have fitted the exact same beat envelope to this new waveform. However, unlike the first waveform, here the fine structure within each beat cycle changes from cycle to cycle. This means that we cannot examine the time period between waveform features that are exactly the same between corresponding beat cycles. Rather, we must resort to the time periods between features such as the strongest maxima (such as the time between "a" and "b" and then between "b" and "c") and these change from beat cycle to beat cycle.

The pitch, as determined by experimental pitch matching, does shift upward. The shift is approximately proportional to the shift of the center of the complex, or about 10% in the example. Pitch shifts of about 20% in either direction are not difficult to observe experimentally. Thus, we must rule out envelope periodicity as a mechanism in pitch perception since the envelope beat rate is unchanged by the shift of the component frequencies. On the other hand, what was a clear application of fine structure analysis has become clouded since we have to decide which features of the fine structure must be examined - the whole structure does not repeat exactly as it did when the components were not shifted. Appropriately chosen, fine structure methods can predict the pitch shift in the direction observed. [It should be mentioned here that place theory in its original forms is completely unable to handle even the missing fundamental, let alone the pitch shift effect. Yet refinements of place theory that concentrate on sharpening mechanisms and transformations of spatial patterns are among the most generally successful theories available today. It should also be pointed out here that fine structure theories are phase sensitive. In the drawings of waveforms above, we could have chosen many different phase arrangements which would greatly strengthen or weaken the fine structure analysis, and yet no drastic change in pitch perception as a result of such phase shifts has been demonstrated.]

## EXPERIMENTS WITH NOISE AND PITCH:

Since white noise has no pitch of its own, it is a useful test signal for pitch perception because it is then possible to see what can be done with the noise signal to install a sense of pitch in it. As it turns out, just about anything that is done to the noise will bring out a pitch. If the noise is bandpass filtered, a pitch corresponding to the center frequency of the filter will be heard, and this seems logical and intuitive on the basis of the spectral content of the filtered noise and the place theory. Experiments also show that low-pass, high-pass, or even all-pass filtering (presented to separate ears) results in a weak pitch in the region where the characteristic is changing, and the explanation is less obvious here. In particular, the all-passed noise has a flat spectrum, and yet a weak pitch is heard.

In an attempt to examine repetition pitch directly, it seems reasonable to listen to a mixture of noise and a delayed version of the noise. In such experiments, we have a "summer" which may be an analog adder (with inverting as well as non-inverting inputs), or the summer may be at a "higher center" meaning that the original and delayed signals are fed to separate ears and the summation is done neurally. In such binaural experiments, results somewhat similar to those obtained with external electrical mixing before presenting the same signal to both ears, are obtained, indicating that something like an actual summation is taking place in the brain. In the figure below, we show the basic model of the experiment with a delay line and a summer where the summation sign is in quote marks to remind us that the summation may be done in the brain and in some unknown manner. In the case where an electrical summer is used, we have the well known comb filter structure, two cases of which are shown in the figure. With the summation done in antiphase, the characteristic response has a series of sinusoidal lobes while cosinusoidal lobes result from in-phase summing.




Based on what we know about the pitch of bandpassed noise, and on how the upper harmonics of a complex tone result in a pitch at the fundamental frequency, it seems reasonable that the pitch that results from filtering of white noise by the cosine-type comb filter should have a pitch corresponding to 1/T where T is the delay time. Such pitches are indeed found experimentally, and are relatively strong. The more interesting case is the one where white noise is passed through the sine-type comb filter. Here too, a relatively strong sense of pitch is obtained experimentally, but pitch matches are made strangely enough to frequencies around 0.87F and 1.14F where F = 1/T. That is, there are two apparent pitches on either side of the first notch. The explanation of this result has understandably consumed much thought on the part of the concerned researchers, and a number of possible explanations are still operable, with interesting implications for pitch perception in general.

A PARTIAL SUMMARY: In this part and the first part of our series, we have set up the understanding of pitch perception as a mini-battle between place and time theories. In fact, the momentum has switched several times. The early experiments of Seebeck in 1841 brought out the time theory in the form of repetition pitch as an outgrowth of the missing fundamental discovery. However, Ohm, and later Helmholtz in 1862 put forth the place theory, explaining the missing fundamental as a nonlinearity of the ear. Békésy's anatomical observations (about 1930) found the "spectrum analyzer" needed for the place theory, but at the same time, Schouten and others were uncovering evidence against it, and from about 1940 to 1970, while both sides were continuing in their efforts, it was time theory that was most favored. In the early 1970's, Wightman, Goldstein, and Terhardt arrived at new theories which while uniting some features of both approaches, were basically refinements of place theory. Basically, these theories united the spatial patterns of place theory with the central processing (transmitted information) of time theory. There are however still results that yield best to fine structure theories. A very recent paper by Ohgushi indicates that time as well as spatial clues are involved in pitch perception. The final answer is not at hand. In part three, we will be taking a closer look at some of these theories and seeing how the experimental results above reflect on these.

## WHO DID ALL THE RESEARCH?

In general, we prefer to have all our materials published in Electronotes documented within the text. In an overview of the type we have above and in part 1, such documentation can be done, but due to the general nature of the comments, this always comes out rather combersome in our experience. For this reason, we have prepared a separate sourcelist on pitch perception which appears elsewhere in this issue. As you can see, it is rather long, but on the other hand, it would not have been much of a challenge to make it ten times longer either. Below, we will add a few comments that will give some of the most basic credits, and the full description of the references can be found in the sourcelist.

For a start, some excellent overviews of the pitch perception problem can be found in Wightman and Green (1974) and in Roederer (1975) while basic ideas about the ear structure are found in von Békésy (1957). Historical perspective is generally very useful and such comments are found in Small (1970), Wightman and Green (1974), and as introductory material in many of the other papers. Direct reference to Helmholtz (1863) and to the papers of Fletcher will also prove useful.

Second-order beats were examined by Plomp (1967a) and combination tones, although known for many years, were critically examined by Plomp (1965), Goldstein (1970) and Smoorenberg (1972), among others.

Data on the interference with a supposed nonlinerally generated missing fundamental are available in Small and Campbell (1961a, 1961b) and in Small (1970).

Fine structure theories can be examined begining with Schouten (1940) with a somewhat more recent update in Schouten, Ritsma, and Cardozo (1962). Additional evidence refining residue theory was provided by Ritsma (1962, 1963) and by Ritsma and Engel (1964) while some of the troublesome aspects of the theory were mentioned by Wightman (1973a).

The pitch shift effect was uncovered by de Boer in his doctoral thesis (unpublished and some of this data appears in de Boer (1956) and is repeated in Small (1970), with extension of the data in Schouten, Ritsma, and Cardozo (1962).

Much of the information on noise experiments can be found in the works of Bilson with Bilson (1977) providing an up-to-date review of the observations.

Some of the more recent pitch perception theories are those of Goldstein (1973), Wightman (1973b), Terhardt (1974), with a work of Ohgushi (1978) offering the idea that time clues still need to be included.

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## A Sourcelist On Pitch Perception - 1978:

The following sourcelist contains a listing of many of the important literature references concerned with the perception of pitch by the human ear, and related topics. The vast majority of these papers were published in the Journal of the Acoustical Society of America, so to save room, we will just use JASA for this journal rather than any longer abbreviation. Two other short forms will be used: P&S refers to the compilation by Plomp and Smoorenburg, 1970; and FOMAT refers to the monograph edited by J. Tobias, 1970. The references are listed alphabetically by author's last name, and then chronologically for a given author. This form is the one used in JASA and is becoming quite popular since it allows open referencing (e.g., Wightman and Green, 1974) within the text, does not require excessive space, and at the same time provides knowledge of the exact reference to many readers knowledgeable in the field. We have provided (with some reservations!) dots (·) to the left of some references to indicate a reference to be particularly noted. The reason for this may be that it is considered useful for initial reading, that it is a fundamental block to understanding pitch perception, or simply that it is very often referenced. We have not included in this list references that are not readily available (such as conference papers) or references not in English. These additional references can be found referenced in the papers we list below.

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# A Cascaded-Monostable Type Generalized Resonator:

## The Synthesis of "Animated" Sounds - Part 3:

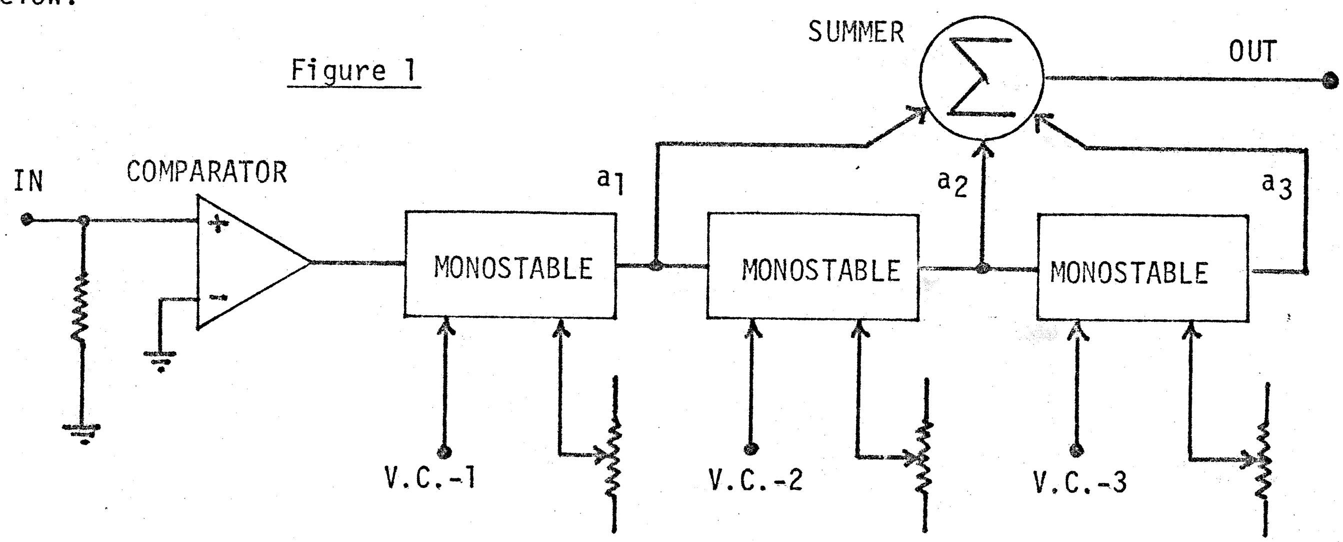
-by Bernie Hutchins

INTRODUCTION TO THIS THIRD PART: We have previously discussed the idea of a "Generalized Resonator" in Mid-Month Letters No. 1 (Jan. 1977) and No. 20 (August 1978). For those who have not seen these discussions, we can say briefly here that a Generalized Resonator is a device made in imitation of the classical "bandpass resonator". That is, it can be periodically excited, and thus has a response related to the rate of the periodic excitation, but it also has a response (a "ringing") of its own which is independent of the excitation rate. This sort of system is known to have a "Formant" type of response - it has a fixed background against which the input signal must react. The way the cascaded-monostable type generalized resonator works will become evident as we proceed. It is also worth noting here that we have made the monostable times voltage-variable, and thus we can make the formants track the input, which is the same as saying that the formants disappear. However, the main purpose of the voltage-control is not to provide a variable formant effect, but rather to allow the device to work as a "timbre modulator." With the resonator ability and the timbre modulator ability, this device fits well into our "animated sound" series.

BASIC OPERATION:

A block diagram of the Cascaded-Monostable Generalized Resonator (CMGR) is shown below:

Figure 1




In the figure, note that the comparator accepts any input signal, and for the normal type of synthesizer waveform, you will get out of the comparator only one negative-going transition per cycle (the usual monostable trigger), and hence the identity of the input waveform is lost, except for its frequency. The three monostables then fire in series, each with its own characteristic time as determined by an external control voltage, or by the manual pots which also serve as initial setting controls. The pulses out of the monostables are then fed to a summer with a weight that can be user set. The total result is that in response to a negative transition from the comparator, a three step predetermined output results. For example, with relative times of 2, 0.5, and 1, and for $a_1 = 1$, $a_2 = -0.6$, and $a_3 = 0.3$, the output is as shown in Fig. 2.

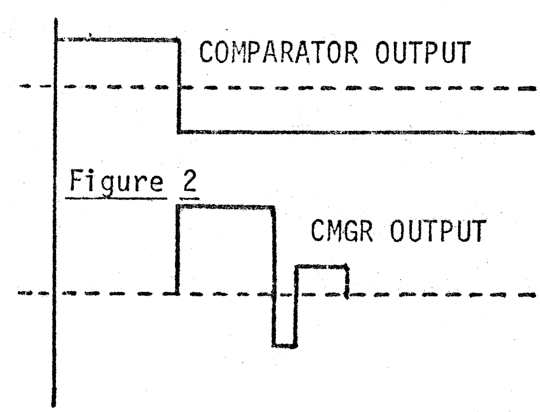



Figure 2

MODES OF OPERATION:

The CMGR, while basically a one mode device, does have three different modes that relate to the delay time settings, and also, we may want to distinguish between the voltage-controlled and the manual mode. While pictures and sketches of waveforms are very often misleading when it comes to understanding how a mode will sound, we really must resort to them here. You may think of them as "timing diagrams" for the moment. Later, when we discuss the test results, we will say more about the sounds.

Figure 3

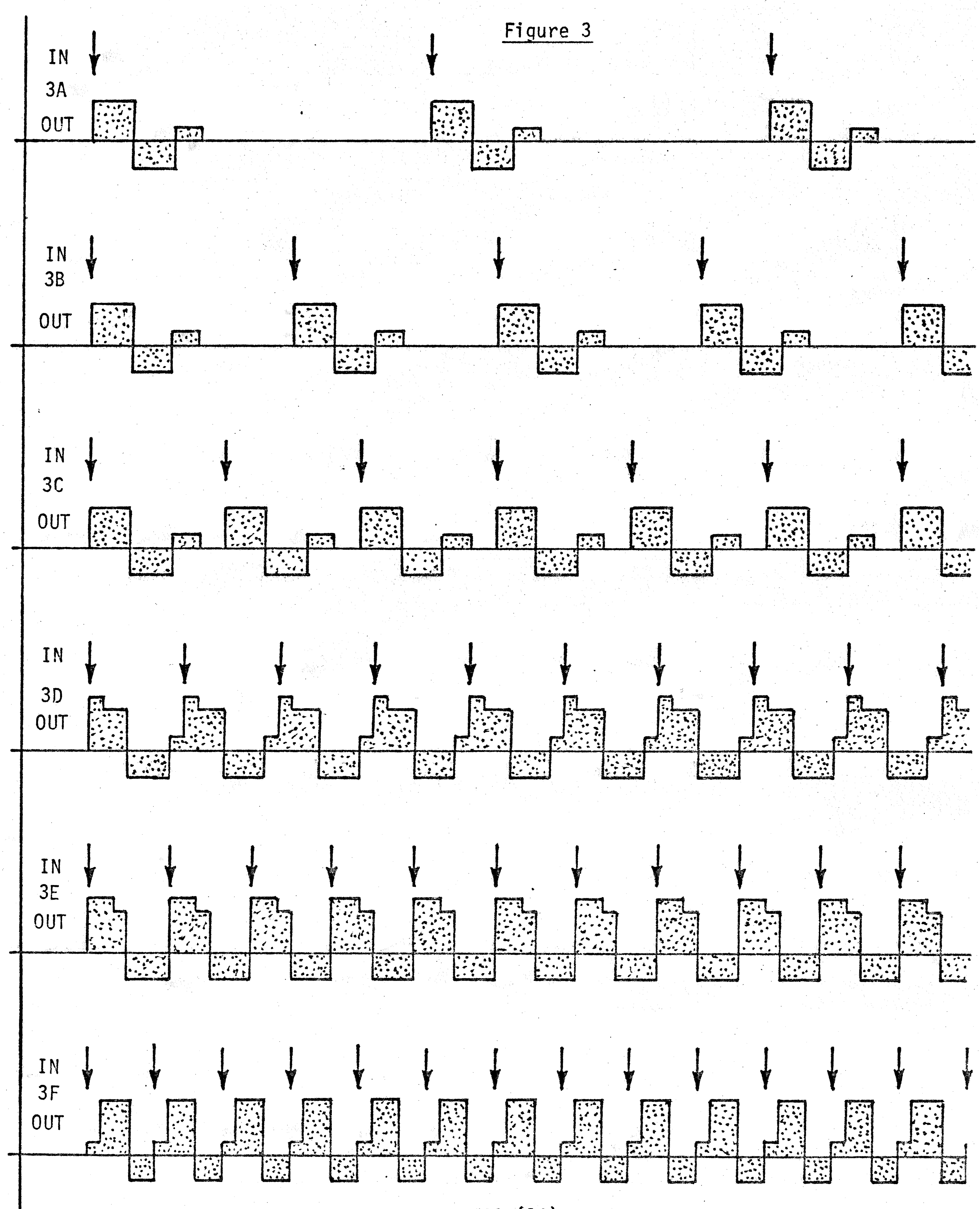
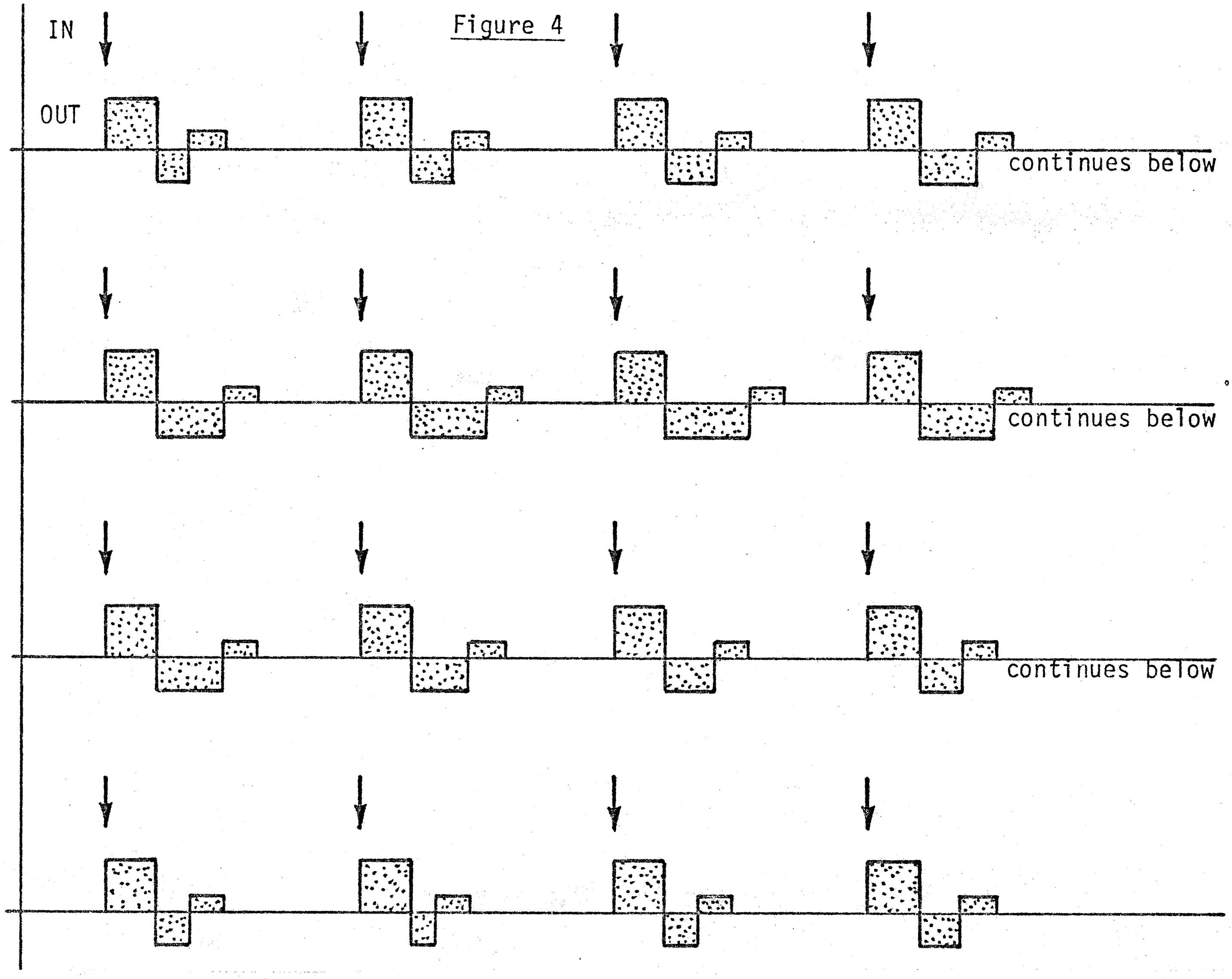


Figure 4



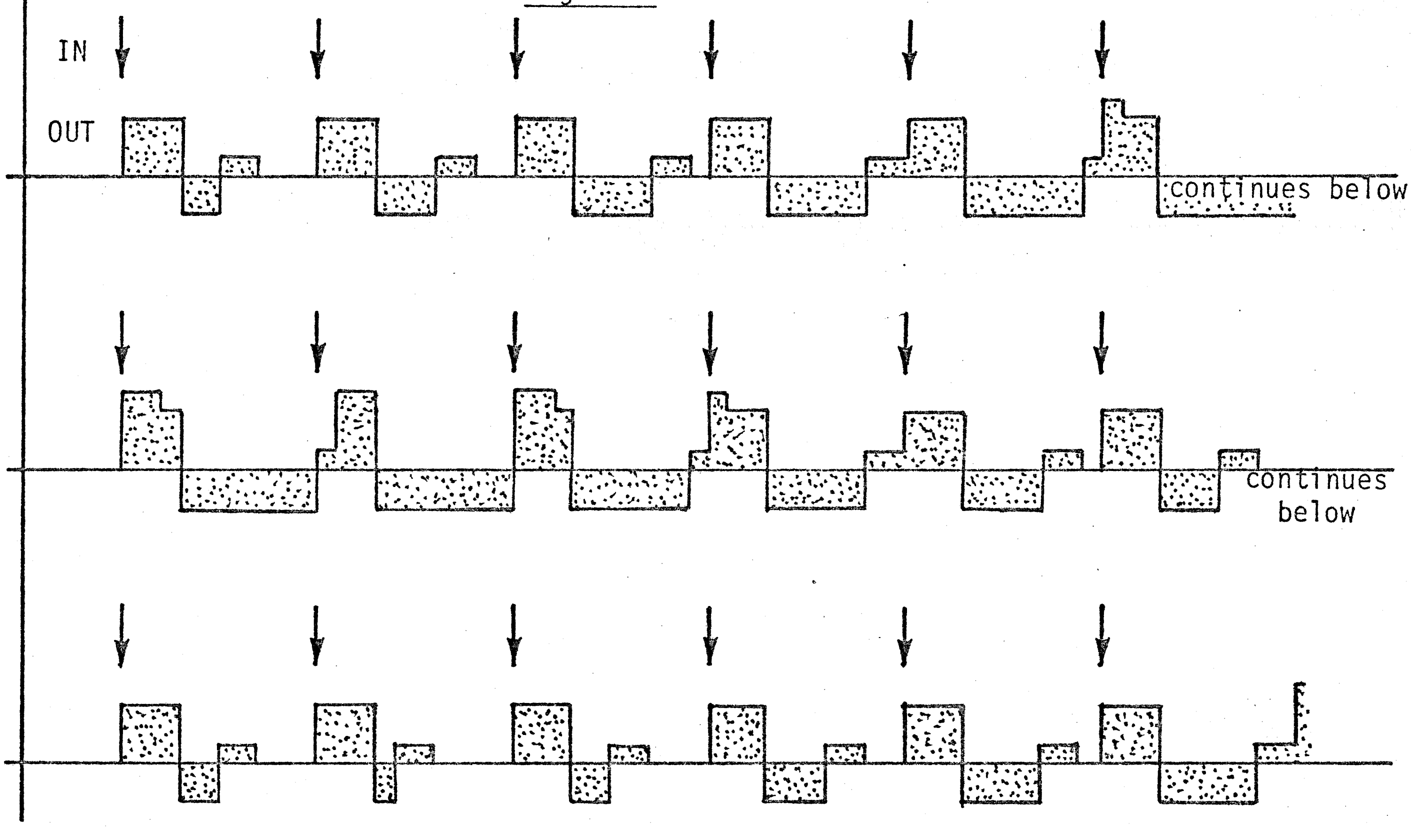

It should be noted first of all that the exact waveforms we obtain, and even their general nature, depends on our choice of time settings and relative summation amplitudes. We have chosen a set of decaying amplitudes which alternate in amplitude. This is selected due to the similarity to the decaying sinusoidal of the "prototype" bandpass resonator. Thus, while the effects we illustrate do occur, and for the reasons given, the overall impact of the effects depends on the exact parameters chosen.

First, we will consider the normal "Generalized Resonator" mode, which is illustrated in Figure 3, with lines 3A, 3B, and 3C giving the best picture of this mode of operation. Here, the delay times may be voltage controlled, but they are not changing. What is changing is the input frequency, which is higher for consecutive lines as we go down the page. The input frequency is shown as downward pointing arrows representing negative transitions of the input comparator. The response to these arrows is independent of frequency and is always as shown in the upper left corner on the first line. At higher frequencies, these responses are of course crowded closer together.

Lines 3D, 3E, and 3F show what happens when the total response time exceeds the period between the arrows. Note that the responses overlap and are added together in these cases. There is an apparent drastic change in the waveform at this point as the eye looses its ability to pick out the resonance pattern so evident in 3A, 3B, and 3C. One should not always draw conclusions about what the ear will hear from what the eye sees. Although such comparisons can be useful, they are often misleading, as for example when changes of phase of component frequencies greatly alter the waveshape of a complex waveform with very little difference in the perceived sound. Here we will call your attention to the fact that it is the case that while the changes in lines 3A to 3C seem mild visually, there is a very great change in the Fourier content mathematically, relatively as much change as we find in the waveforms of lines 3D to 3F.

We will now consider a second mode, the "Timbre Modulation" mode. This is illustrated in Figure 4 which is set up similar to that of Figure 3, but here the consecutive lines are a continuation of the line above. The variable here is the time delay of the second segment of the resonator response. While it makes no difference for this individual tone, we can assume that the delay times in their initial states have been voltage controlled and track the input frequency. Thus, the overtone structure is constant relative to the fundamental and the time into the tone. The result is similar to pulse-width modulation, only with more general abilities. Here we have effectively removed the formant features, and the CMGR is now just another waveshaper, with provisions for waveshape control. Note however that we also have the option, although not illustrated, of making one or more of the response segments track while the others do not, retaining some, but not all of the formant ability. Nothing prevents us from letting the total period of the response exceed the period of the input frequency, just as we saw happening in lines 3D through 3F of Figure 3. An example of this type is shown in Figure 5, where the second segment gets so long that it forces the third segment to overlap the first segment of the next cycle, and even enters this segment itself (under the second arrow of the second line). Again, as in the case of Figure 3, one should not draw premature conclusions about the effect of this overlap on the actual sound.

Figure 5




It is important to note that above we have shown overlaps and that these overlaps were due to the period of the total response exceeding the period of the input arrows. A very different thing happens when the period of the monostables themselves exceed the period between the arrows, and this results from the fact that we assume that the monostables are of the non-retriggerable type. The type 555 and improved versions such as the 355 are also of the non-retriggerable type. Simply stated, this means that as long as the output of a monostable is high, it can not be retriggered to start a new cycle. A monostable that is high when a trigger arrives will thus ignore that trigger (and any others that arrive while it is high) and wait until it goes low, and another trigger arrives. For example, if the monostable time is slightly longer than the space between triggers, every other trigger will be ignored. This results of course in a drop in pitch of one octave. Basically, this octave drop is the major effect of having any one stage in the CMRG exceed the time between input arrows. The total effect of this depends on which stage it is that is overtimed, and on the amplitude of this and any succeeding stages. The effect can be made gross or subtle by changing these variables.

An example is shown in Figure 6, with Figure 6-1 showing the case just before the time of the first stage exceeds the period between arrows, and Figure 6-2 showing the case just after it exceeds the period. Here we also show the outputs of the individual monostables as open boxes (with the combined output, as usual, being dotted in). Here, it can be seen that the T1 segment drops every other cycle as we go from

Figure 6-1

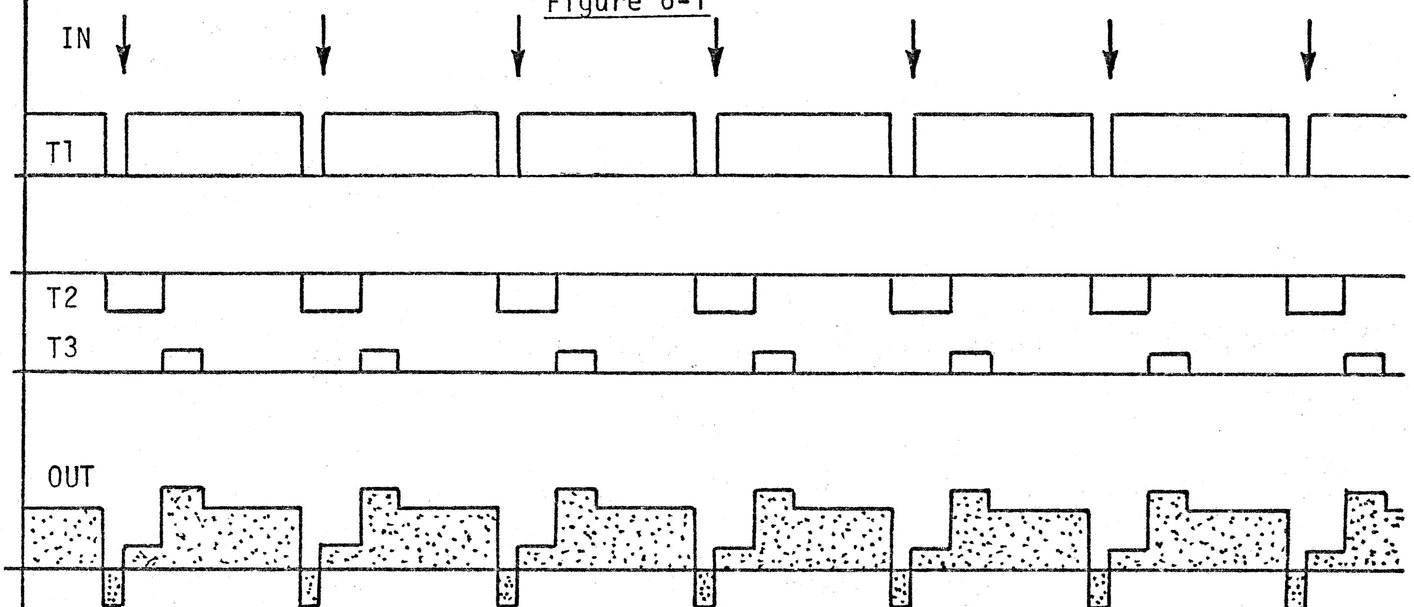



Figure 6-2

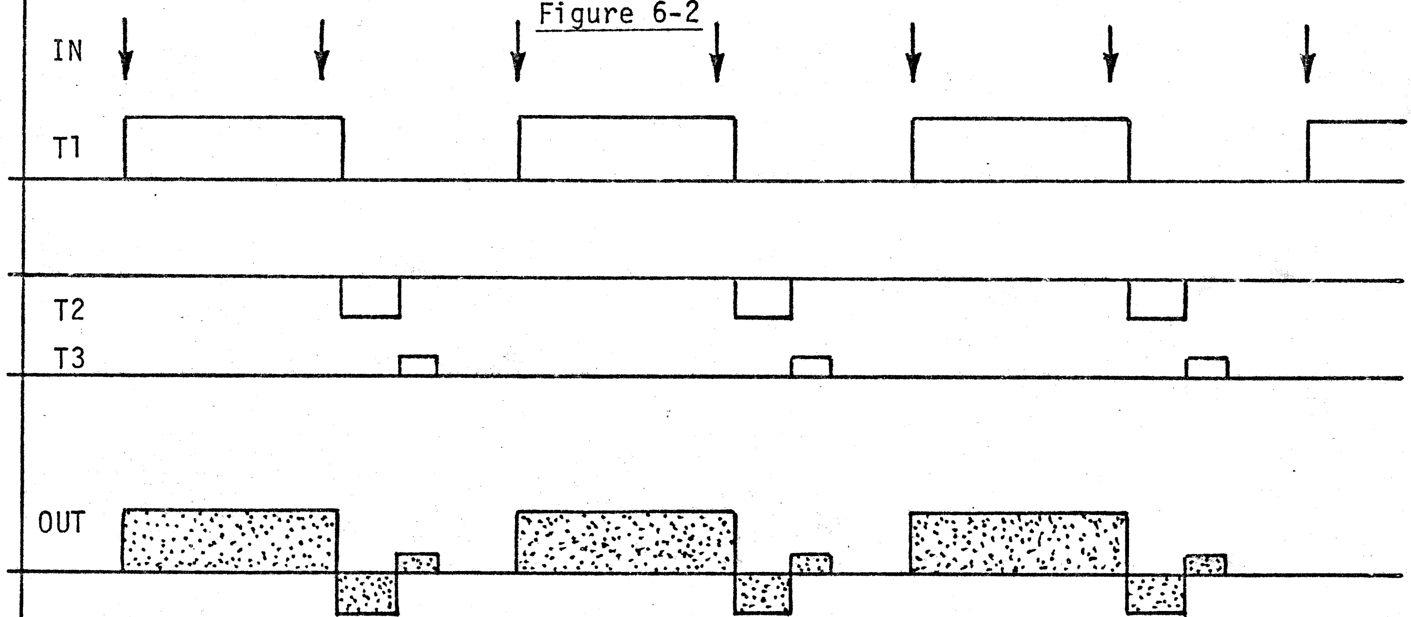


Figure 6-1 to Figure 6-2. Since the rest of the monostables are in cascade, they too must drop out for every other trigger, The result is a clear drop of one octave as the time between arrows is exceeded, and a significant change in timbre can be expecte as well. In the figure, the time between arrows is exceeded because the time T2 increases, but the same result occurs if T2 is constant and the input frequency goes up enough. In general, this sort of response - a large change as a result of a very subtle change of conditions - is unmusical in the usual sense, although it can result in interesting special effects. Note the similarity of this to the "aliasing" effect in sampling systems. Some additional results can be found by considering that the first stage amplitude can be set to zero, and it then makes its presence known only by its delay, and subsequent blockage of certain triggers. In such a case, the response of the other two stages is the same, and you end up with an octave jump to a condition that would have existed at a lower frequency.

Things get more interesting and subtle as we leave the first time T1 at a value shorter than the period between arrows, and then let a later stage exceed this limit. Figure 7 shows the situation where the second stage exceeds the limit, and Figure 8 shows the case where the third stage exceeds the limit. A quick glance at the bottom

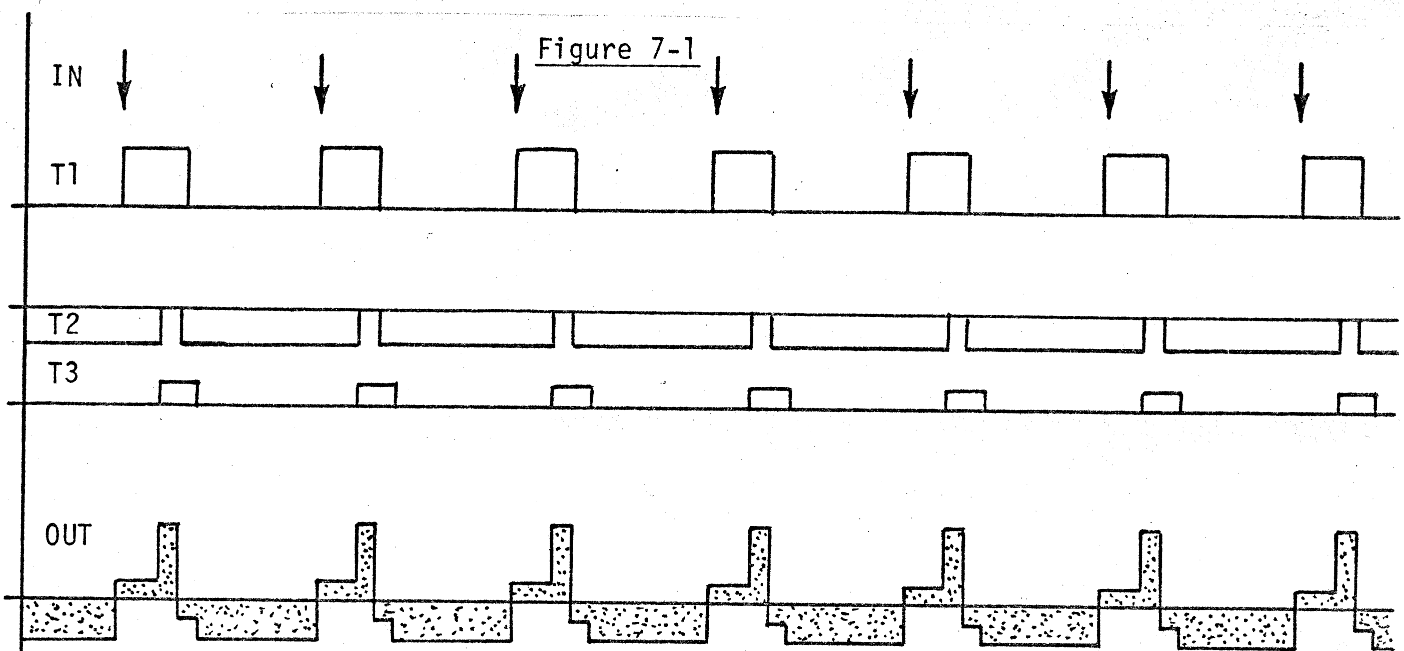


Figure 7-1

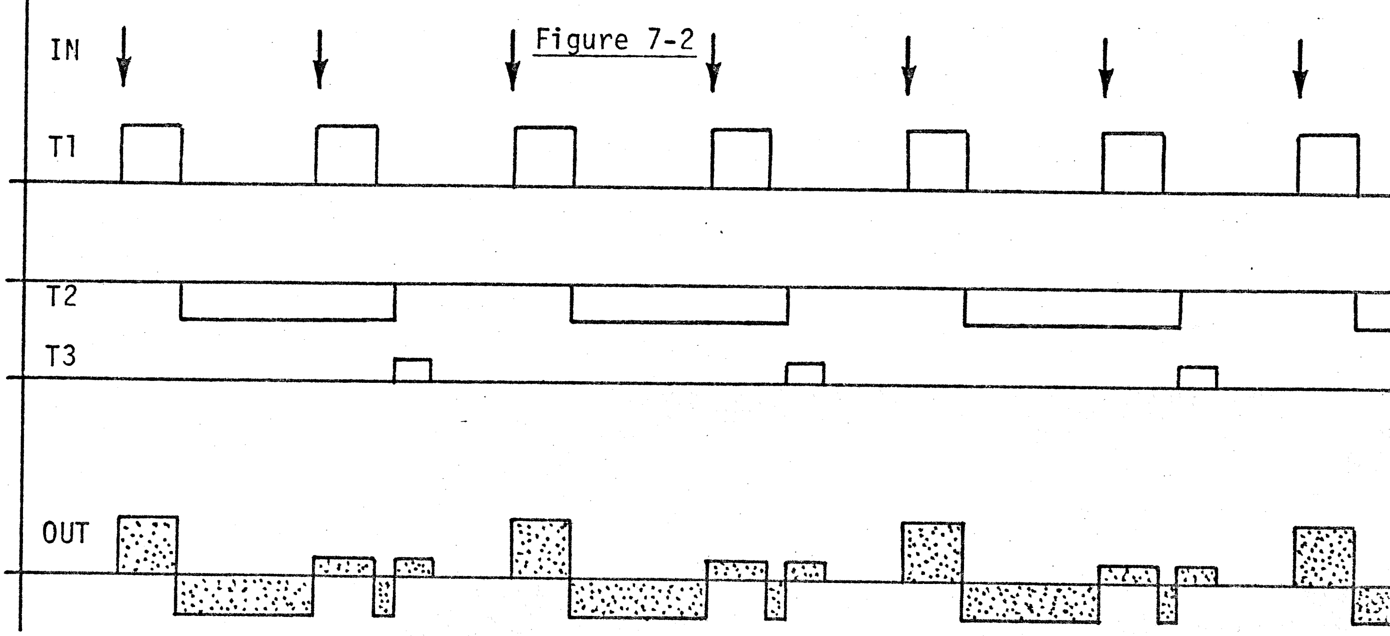


Figure 7-2

line of either figure will show two things: that there is evidence of an octave drop, and that there is less of a change with these later stages as compared to Figure 6. We should mention here that much of the feeling that the change is becoming more subtle is a result of our choice of decaying amplitudes for successive segments. In fact, the drop vanishes for Figure 8 if the amplitude of the T3 segment is made to vanish as well.

Two ways of obtaining effects that vary from non-existence to a full and obvious shift of pitch to a lower value are obtained as follows. In Figure 8, the on time of the 3rd segment can be made to exceed the the time between arrows. Then the amplitude is adjusted for a slight feeling for the pitch an octave lower. Finally, the duration of the 3rd segment is decreased, and placed under envelope control so that the octave jump down will appear as a subtle enrichment during the tone as controlled by an envelope. The second way of obtaining a subtle effect is to leave the amplitude of T1 relatively high and the duration short. The duration of T2 is then made to exceed the time between arrows, and its amplitude is set to zero. The third segment is then used to install an extra feature - a pulse which will occur every cycle, every second cycle (an octave lower), or every third cycle (making the original pitch seem to be the fifth!), etc. This sort of subtle reaching for a lower subharmonic is interesting in

Figure 8-1

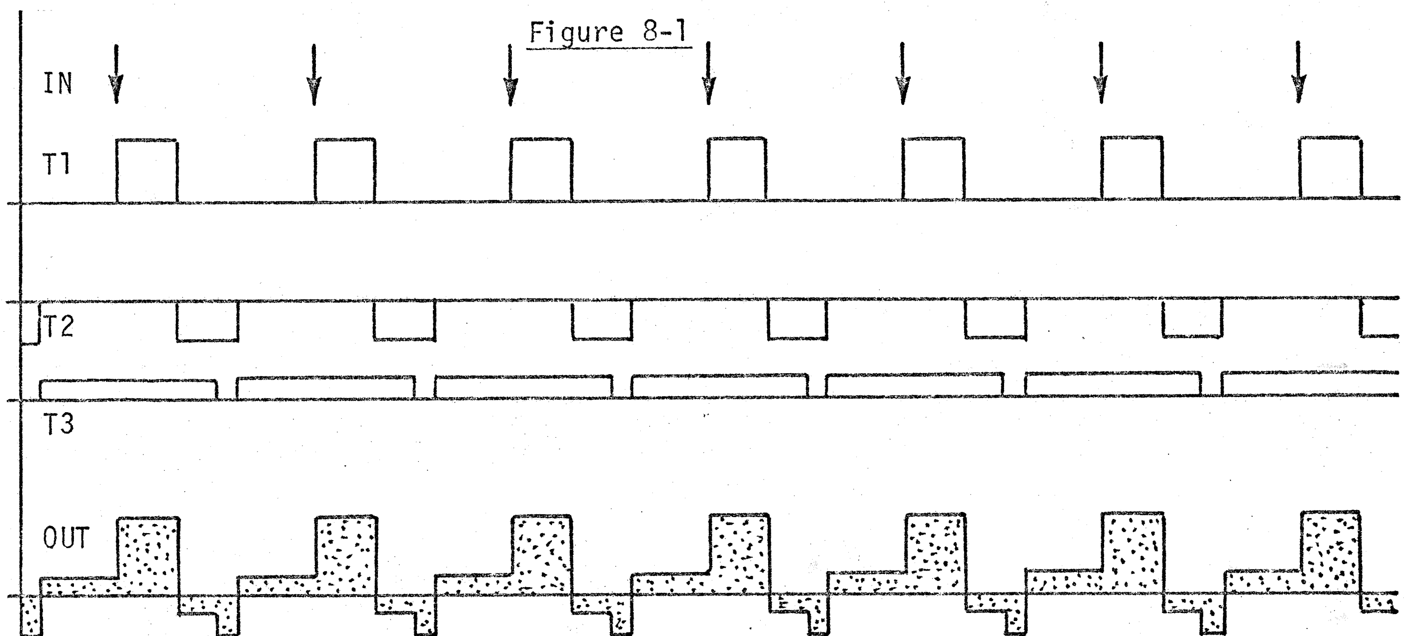
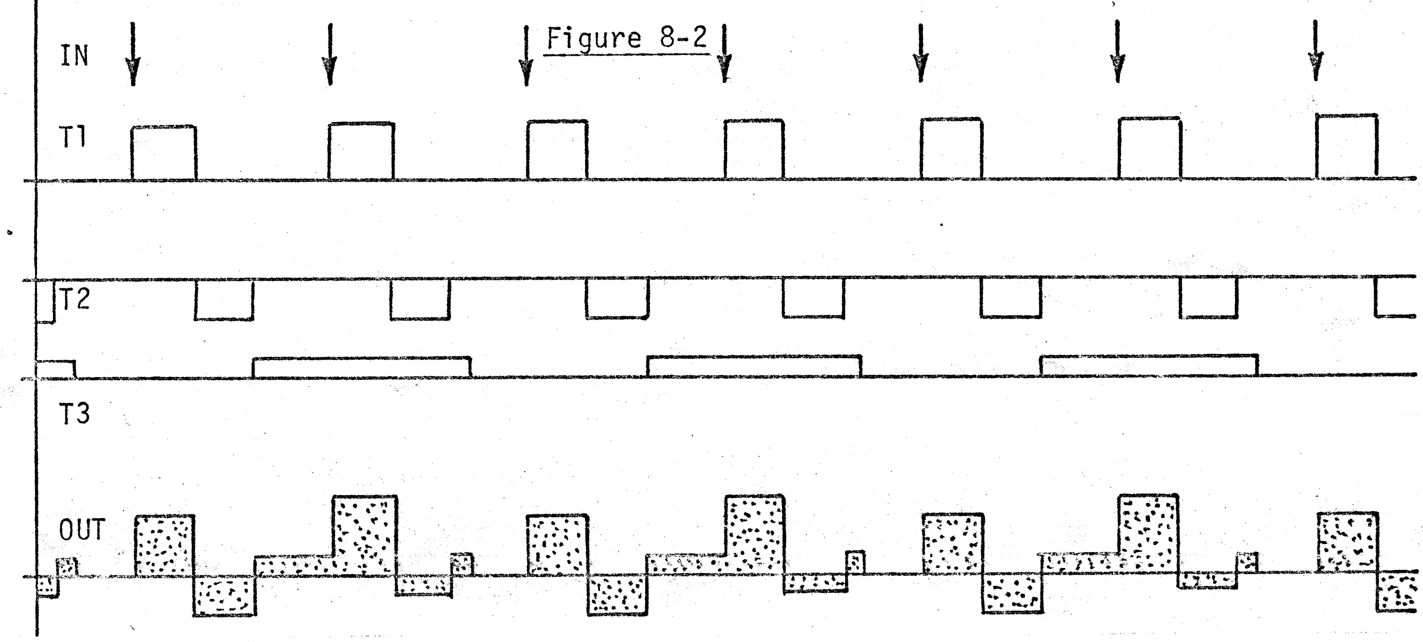



Figure 8-2



itself, and can be used to arrive at new tone colors under manual control. Note also that the appearance of this effect is under the control of the time T2, and thus can be made to appear and shift position as T2 is changed by voltage-control.

Many of the more subtle effects can be used either in a tone color control mode (as in a timbre modulator), or as a formant feature. It is of interest that they tend to strengthen the lower range of pitches as the input frequency goes up. Thus, when used as a formant device, the CMGR can behave as a standard generalized resonator, or it can act as a device which tends to add a fullness to the upper range of an instrument Many of the features possible with this device will yield to experimentation, or perhaps to the drawing of more diagrams of the type we have given here.

CIRCUIT DESCRIPTION:

The circuit diagram of the CMGR is shown in Figure 9. The top third of the diagram consists of three exponential current stages. The middle third is the cascaded monostable chain. The bottom third is the summer for weighting the output.

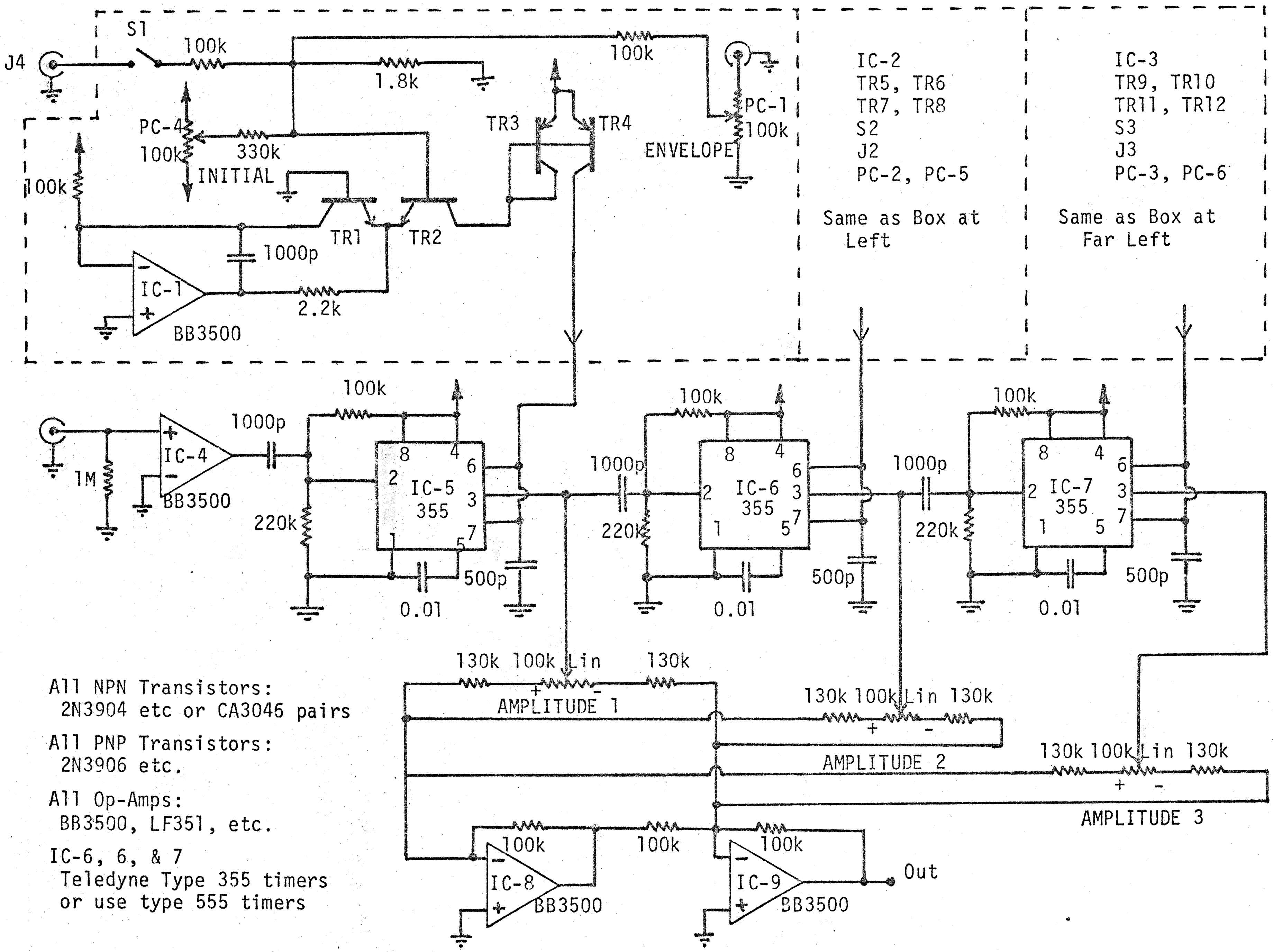
Most of the circuitry of the exponential stages should be very familiar to the reader. The reference current source, IC-1 and the exponentially converting transistors TR-1 and TR-2 are virtually the same as we have used in the past. The summing at the base of TR-2 is done by the approximate method of having large resistors drive a much smaller one (100k or greater driving 1.8k in this case) instead of using the op-amp as in VCO designs for example. Finally note TR-3, which along with TR-4 forms a current mirror, so that the exponential current is sourced from a + supply potential, as is needed for the monostables. Two sections not shown in detail are identical to the one in the dotted box in the upper left, and these supply the exponential currents for stages two and three. Note that J4, the 1v/octave input is common to all three sections

There are several design changes that are associated with the exponential current stages which may be desirable. For example, I used ordinary 2N3904 type transistors for TR-1 and TR-2, although an array such as the CA3046 would probably be better. The current mirror is formed from two type 2N3906 transistors. You could use a better PNP matched pair, or a CA3080 OTA for example (see EN#54, pg. 20, or AN-106, to be published) Another change that we looked at was to use a PNP type of exponential converter and run the monostables on -15 and ground instead of ground to +15 as in the actual schematic. This would save the current mirrors. The output of the monostables in response to a trigger would then be a pulse jumping from -15 to ground, then returning to -15 after the duration. This would be a negative going pulse that is the complement of the positive going pulse we actually used. This would be more difficult to used in the weighted summer we have selected, so you would probably need a CMOS inverter to get a pulse that is really just the inversion of the one we use. All this seemed to be as much trouble as just using the current mirrors. Perhaps a reader can suggest a better way.

The monostables we used are of the Teledyne 355 type which is very similar to the popular type 555 except it does not result in the large current glitches so familiar with the 555. We did not try 555 timers here, but they should work fine. It would be a good idea to bypass the 555's positive supply at each chip with 0.1 mfd, the usual 555 practice. Actually, the circuit was designed with 555's in mind, but it was 355's which we actually plugged in. The circuit is basically the standard monostable configuration so well known. The input trigger pins (pin 2) are pulled down slightly to assure reliable triggering in cascade. As usual, a timing capacitor is connected to pins 6 and 7, but note that the current to charge this capacitor is supplied through the current sources, and not just simply by a resistor as in the common application. This is how voltage-controlled pulse width is achieved here.

The summing network at the bottom third of the diagram is formed around IC-8 and IC-9 which are configured as inverting summers. Note that when the pot wiper is

Figure 9 CIRCUIT DIAGRAM OF CASCADED MONOSTABLE GENERALIZED RESONATOR




All NPN Transistors:
2N3904 etc or CA3046 pairs

All PNP Transistors:
2N3906 etc.

All Op-Amps:
BB3500, LF351, etc.

IC-6, 6, & 7
Teledyne Type 355 timers
or use type 555 timers

centered, identical currents are fed to the (-) summing nodes of both IC-8 and IC-9. IC-8, being inverting, supplys the opposite current to IC-9, and the two currents thus cancel at the input of IC-9. Now, with the wiper a little bit to either side of cente the two currents are out of balance and a net positive or negative excursion results at the output of IC-9. In this way we achieve a "center off" control capable of giving a positive or negative weighting to pulses from the monostable. For more information on this type of summer, or to find the "side" resistors on the pots for pots other than 100k, see the analysis in AN-102 (to be published).

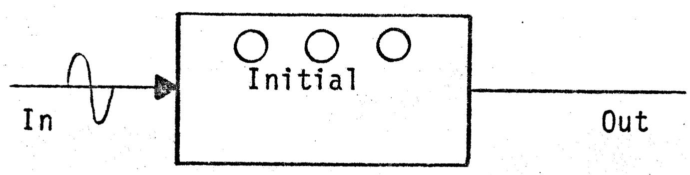
There is nothing very critical about the construction of this device. Once completed and found to work in its basic functions, there is some "tune up" work you may want to do. The main thing to check here is to be sure that all the monostables have approximately the same delay time for the same control voltage. To do this, first disconnect the 1v/octave controls by opening switches S1, S2, and S3, and also set the envelope control pots to zero (or just simply do not plug anything into the corresponding jacks). Next set the initial delay controls (PC-4, PC-5, and PC-6) to their cente positions, verifying and marking this center position by using a voltmeter indicating zero volts, if possible. It is then a matter of observing the output delay time of the three stages. If they are not fairly similar (within 20 or 30% say), you may want to try changing one of the two NPN transistors of the exponential current stage corresponding to that section. Be sure to let the transistor cool several minutes before testing again after soldering. If this in no improvement, or makes things worse, try a couple of other transistors in this same position. If there is still no improvement, it is probably the transistor in the adjacent position that is toward the extreme side of the specs, and this one should be changed. You should keep in mind that these initial controls are not precision controls, and you will not be able to set them exactly in actual use, so there is no real point in going for a match much better than 10% at best.

## APPLICATIONS:

Many of the applications of the CMGR are implied by the "Modes of Operation" section of this report. However, here it is desirable to sum these up as "patch diagrams" since the exact way of implementing these ideas may well not be clear from the timing diagrams.

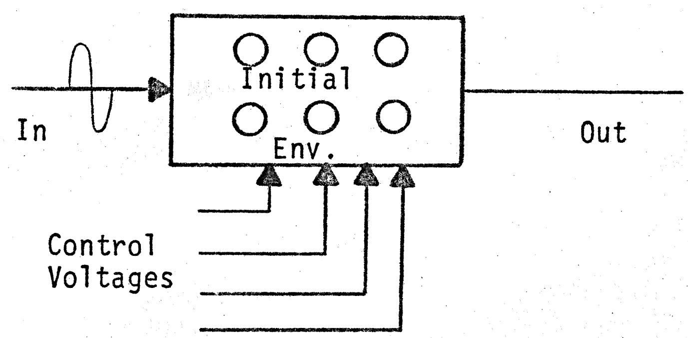
The first application, the basic "Resonator" mode is shown in Fig. 10. Here we are interested in only the input and the initial setting controls. In this application, the invarient structure of the resonator response provides a fixed background or formant to the sound. The input here, and in the rest of the examples is shown as a sine wave, but any waveform may be input with the same results.

Fig. 10 Resonator




A second application, Fig. 11, shows a "Voltage-Controlled Resonator" mode where the delay times are not fixed, but rather are (or may be) controlled by an external voltage. These control voltages are not specified here, they may be the 1v/octave tracking, envelopes, or periodic waveforms. Their effect may enhance the formant structure, or it may reduce or completely remove the structure. This patch may also produce some timbre modulation as well.

Fig. 11 Voltage-Controlled Resonator



Two patches which produce controlled timbre modulation are shown in Figures 12a and 12b. The patch of Fig. 12a is a "LFO Controlled Timbre Modulator" while the patch of Fig. 12b is an "Envelope Controlled Timbre Modulator." The former patch provides animation to long sustained sounds while the latter patch provides a time-dependent variation in tone color in response to control envelopes that follow their perscribed contour once for each tone. Thus, the LFO-Controlled patch enriches the signal prior to additional processing while the Envelope-Controlled patch is a processing similar to that achieved with a VCF. The LFO-Controlled processing sounds basically similar to that obtained with the Multi-Phase Waveform Animator (EN#87) although somewhat less rich, while the Envelope-Controlled process is basically multiple pulse-width modulation. The degree of enrichment or animation with either process might well be judged something like twice the degree achieved with other more familiar processes.

The next application to discuss is the one of providing the suboctave cue to a sound. The corresponding timing diagram most relevant here is Fig. 8 on page 24. The patch setup is shown in Fig. 13, where we show the settings of the third stage only. We assume that the initial delay setting of this third section is adjusted to about 95% of the input period. The envelope control then is adjusted relatively shallow, to about 10% of the delay period. Thus, at the peak of the envelope, the suboctave cue just begins to appear as the total delay of stage three exceeds the input period. In this application, the amplitude of the third stage will most likely be relatively small.

It is even possible to use the CMGR as a sequence generator. To do this, we set up the system much as we did for the timbre modulators. One such patch is shown in Fig. 14. The main difference here is that the input is "overclocked" - the input frequency is much greater than the output frequencies. To see what is happening here, refer to Figure 16 on page 29. This figure shows an input frequency of 10,000 Hz, and some of the subharmonics, 5000 Hz, 3333 Hz, 2500 Hz, and so on. When you get down to lower sub-harmonics, they become closer and closer together (for example, the 54th, 55th, and 56th subharmonics are about 185 Hz, 182 Hz, and 179 Hz). The period of 182 Hz is about 5.5ms. If we trigger a monostable at 182 Hz, and the period of the monostable is slightly less than 5.5 ms, the monostable on time will be less than the period and the monostable will be triggered each cycle of the 182 Hz

Fig. 12a Timbre Modulator LFO Control

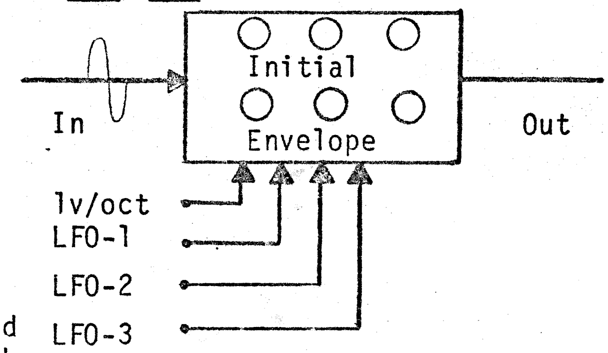
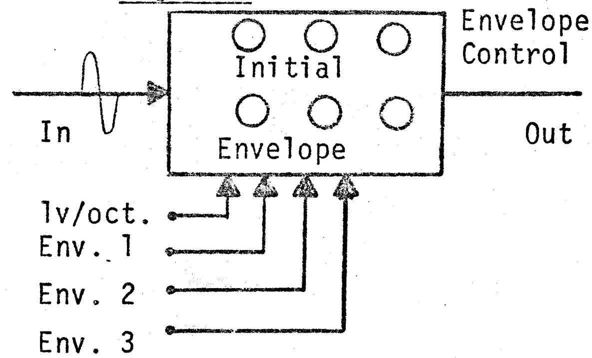



Fig. 12b Timbre Modulator Envelope Control



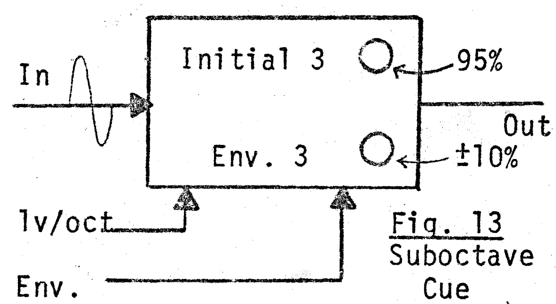




Fig. 13 Suboctave Cue

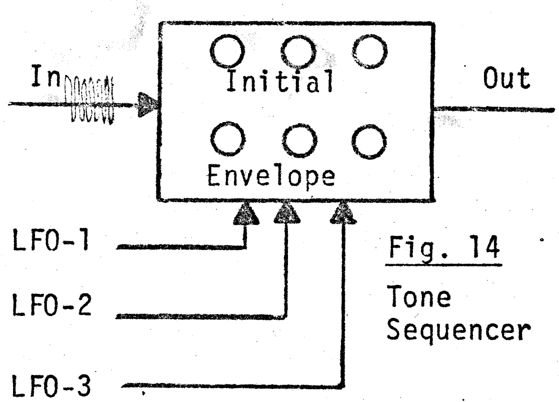



Fig. 14 Tone Sequencer

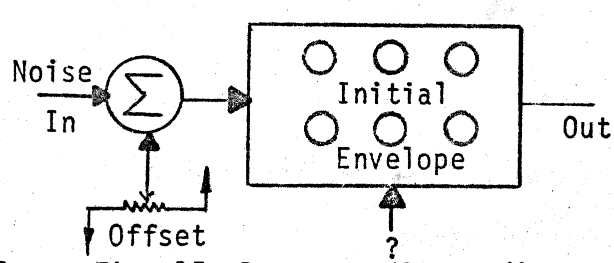



Fig. 15 Resonator Responding to Noise Input

drive rate. If, on the other hand, the monostable time increases to a value just above the period of 182 Hz, every other trigger will be missed and the frequency of the output will drop one octave to 61 Hz. Now, see what happens when we use a much higher input frequency like 10,000 Hz. At some time, the monostable triggers, and goes high for just a little over 5.5 ms. During this time, the 10,000 Hz input tries to retrigger the monostable 55 times, but these are ignored because the timers used here are non-retriggerable. The 56th attempted trigger will be successful, since the output of the monostable drops before this one arrives. The process continues again, and the monostable outputs for every 56th input pulse, resulting in a 179 Hz output. Thus, we can see that the time delay of the monostable limits the output frequency. As the delay changes, the output frequency changes, but in a quantized manner, since the output must be a subharmonic of 10,000 Hz. Thus, by changing the voltage-controlled delay, a discrete sequence of tones can result.

Things get much more complicated and musically quite interesting as one goes to a setup such as that in Fig. 14 where several different low-frequency oscillators are used. Recall from Figures 6, 7, and 8 that lower octaves may be perceived as a result of time delays other than the first exceeding the drive period. It is much the same with the overclocking. The result is always a tone sequence however. Unlike some other methods of achieving tone sequences however, this one has a constantly changing tone color as well.

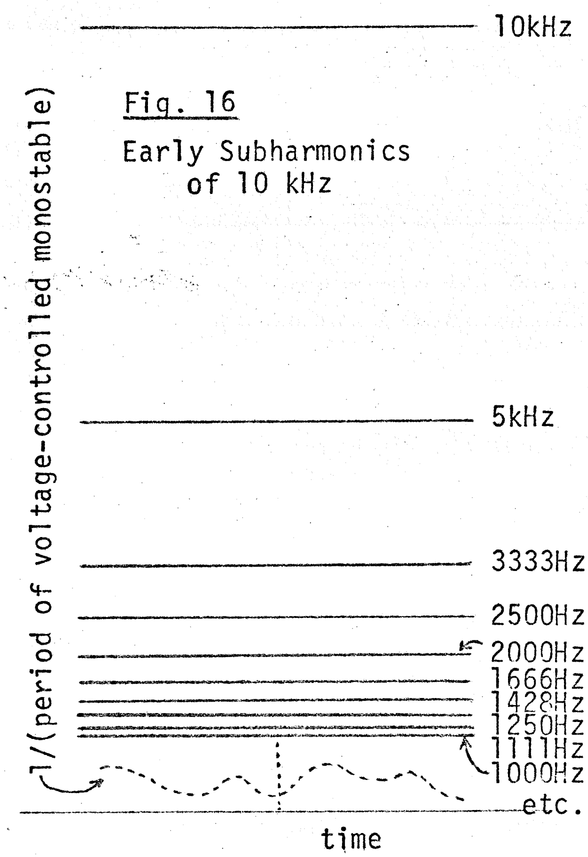



Fig. 16

Early Subharmonics of 10 kHz

In the application above, we see that after the period of a monostable expires, the monostable is retriggered by the next pulse that arrives at the input of the CMGR. For a periodic drive, this occurs at a regular and predictable rate, and only subharmonics are excited. What happens if the input is not periodic, but rather random noise? Here, when the monostable time expires, the monostable waits for the next trigger, which will occur after a random interval. If this interval is short relative to the monostable delay time, then we have regular long pulses from the monostable with relatively short, but irregular delays between these pulses. This results in a pitched sound, but one with some obvious "phase jitter." One is strongly reminded of bandpass filtered white noise, bringing home once again the idea of a "generalized resonator." Note that the average delay between input pulses can be controlled by offseting the white noise before it reaches the input comparator, as shown in Fig. 15. Since amplitude distributions for the usual type of noise source are Gaussian, as we displace the noise signal, we lessen the probability of the signal exceeding a certain level. The fastest rate occurs when there is no offset in which case every zero crossing of the white noise produces a potential trigger. This application is important both conceptually and practically, but it may be difficult to achieve with some noise sources due to insufficient bandwidth and a resulting low value for the maximum clocking rate. None the less, even low bandwidth noise sources are useful because even though they do not produce strongly pitched sounds, they do process the noise signal in some way, and along with a VCF can produce noisy sounds with a more reverberant quality than the usual types.

## SUMMARY:

The CMGR has been domonstrated in theory and in practice. The basic device consists of a series string of monostables, each one triggering the next one down the line. The delay times of the monostables are voltage-controlled, and the output of the CMGR is obtained as a weighted sum of the outputs of the various monostables.

In addition to more or less standard applications where the CMGR is used as a generalized resonator, there are a number of other applications, including some which behave much as conventional timbre modulators. An important group of applications occur when a monostable is driven at a rate which exceeds its ability to complete a pulse. In such cases, ignored triggers can lead to subtle subharmonic cues, and even to tone sequence generators and semi-random tones.

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## QUESTIONS (Continued from Page 2)

Q: Here is something that has always bothered me, and others I have talked to share my confusion. Why is it that when you find the frequency response of a filter to a real frequency, you measure the response along the imaginary axis in the s-plane?

A: This is something that is probably best answered by just saying something like "It just works out that way." Let's see if we can say briefly why it works out that way. The solution to many problems is found by solving the appropriate "differential equations" of the system involved. Such solution methods are well formulated and are a very secure branch of mathematics. However, in many, if not most engineering problems the solution is found more easily by transform methods (Fourier and Laplace transforms) and such transform methods are so universally used that many engineers don't even stop to consider the question you have asked above. The Laplace transform method involves the so-called "complex" frequency "s" where $s = \alpha + j\omega$. Here, j is the base of the imaginary number ($\sqrt{-1}$) and $\alpha$ and $\omega$ are real numbers. In part, this answers your question, since $\omega$ really is a real number, even though it is multiplied by j in the actual mathematics. No, we won't stop with this cop-out!

Even in the differential equation methods, it is common to find a "trial solution" of the form $e^{j\omega t}$ being used. This is often simpler than just trying a sinusoidal. It is generally the case, that using such a complex trial solution allows us to solve the equation more simply, and at the same time, the real and imaginary parts of the solution remain separated (because we have only real numbers in the rest of the problem). This allows us to throw away the part of the soultion we get, but did not need. Suppose for example we excite a system with $\sin\omega t$. You may recall the remarkable "Euler Relation"

$$e^{j\omega t} = \cos\omega t + j\sin\omega t$$

By using the exponential function, we are essentially solving for a Sine and a Cosine solution at the same time, and keeping them separate by using the complex base j. This is handy, and we think it useful to review this separation idea here, but it is not the reason that ordinary frequency is on the imaginary axis in the complex s-plane.

Consider a point s on the s-plane, and use the Euler relation on it:

$$e^{st} = e^{(\alpha+j\omega)t} = e^{\alpha t}[\cos\omega t + j\sin\omega t] = e^{\alpha t}\cos\omega t + j\,e^{\alpha t}\sin\omega t$$

Pay close attention to where $\alpha$ (the "real" part of the complex frequency) and $\omega$ (the imaginary part of the complex frequency) went. The $\alpha$ appears in an exponential term, and represents the growth or decay of a sinusoidal, with $\omega$ being the frequency of the sinusoidal. Thus, by using $e^{st}$ as an excitation function, we can obtain a soultion (using standard Laplace transform methods) for a whole range of growing or decaying exponentials. Now, when we want to know frequency response, we want the response to a sinusoidal of constant amplitude, and this occurs only when $\alpha = 0$, on the imaginary axis. Thus, using the Euler relation, it happens that the part of the solution we are most interested in, the oscillatory part, has as its parameter the imaginary part of the complex frequency.

Note that there is some unfortunate terminology involved here, and it begins with the inappropriate term "imaginary" for the number j. The second thing is that we would refer to $\omega$ as a real frequency (physically real - a number on a function generator dial for example) while the real part of the complex frequency s is $\alpha$, not $\omega$, and $\alpha$ is not a frequency, but a damping (or growth) time constant. Thus, the real part of the complex frequency is real, but not a frequency, while the imaginary part of complex frequency is the real frequency in the ordinary sense!

▷ Q: Bernie, for the benefit of those of us who are new to Electronotes could you tell us a little bit about how Electronotes got started and something about yourself.

A: I suspect that like most who will read this, the exact point in time and exact manner in which an interest in electronic music developed will be hard to pin down. Also, like most others, my interest grew out of the desire to work both in music and in engineering. Certainly we are lucky that electronic music came along at the right time.

As far as the origin of Electronotes, that too just seemed to appear and grow, but like many if not most electronic music enterprises, we trace our start back to Bob Moog. Let's back up a bit. In high school, I worked more in music, and had some hopes of actually becoming a musician. However, my real strengths were in math and science, so I entered the engineering program at Cornell. After a few months of engineering, and a glance into beginning "music theory" classes, I felt that I could not handle the music program, and further, the engineering program was going to take most of my time. Little was done in music at that time. I do remember however, a lecture-demonstration by a graduate student in Engineering Physics sometime during my undergraduate years. It concerned electronic music equipment and the speaker was a man who had a small company in Trumansburg, New York. Now Trumansburg is a nice enough small town just about 10 miles out of Ithaca, and I went through it every time I got a vacation from Cornell. Frankly, you never expected that a revolution in electronic music would begin there, and probably at that time, the man giving the lecture didn't either. It was something you could not quite forget however.

I was graduated from Cornell in Engineering Physics in 1967 and that was a time when a student had to either think about graduate school or the army. After four years of engineering studies, three in the army did not seem like such a bad deal. While I was in the army, the "Switched on Bach" album came out, and in finding out all I could about how it was made, the Tramansburg-Moog connection became clear. At the same time it was rapidly becoming clear that electronic music was coming into its own, and could be something that an individual could deal with, and computers or massive machines like the RCA synthesizer were not the only way. Further good news came when my father sent me a clipping saying that Bob Moog was giving a course at Cornell that "resided" somewere between the music and electrical engineering fields. This was my first real indication that something like musical engineering could exist.

After certain formalities I reactivated a previous acceptance in Applied Physics at Cornell, obtained support, and found out about Bob's course. Even though I was supposed to work in physics, and even did a little, I guess my heart was with the music and the engineering. I took Bob's course, and got an A+ I am happy to say. However, shortly after that both Bob and his company moved to Buffalo.

While still doing some physics, I also did manage some electronic music work. A friend built a "Psych-Tone" and I made one too, and shortly thereafter I found it lacking and went ahead with my own design using RTL IC's and circuits from Moog's 1965 JAES paper. While preparing the diagrams to share with a few friends, I remarked to my wife that some day I might like to start a publication on electronic music and she suggested that I do it right away. A letter to Popular Electronics started the thing going. In fact, I began receiving inquiries about "my idea" several days before the issues of PE with my letter arrived. It was then time to actually create the newsletter. Now, for those who don't know it, graduate students are on a very very limited budget. My notes and drawings on the improved "psych-tone" which had previously been Xeroxed in only a few copies were reproduced for mimeograph, and became the first three issues of Electronotes. It was necessary to have some explanation of what I wanted to do, and this became issue 4. We actually backed into the business side of the newsletter, not by choice, but because there just was no money except the few subscription checks that were coming in. Soon, it became obvious that we had to do something about printing, and the only economical system was a hand cranked mimeograph which we bought for about $70 probably using rent or food money at the time. I can remember bringing home mimeo paper from the campus store one ream or two at a time. But the thing about investing essentially no money is that there is little to loose. And the thing did fly! For getting us going, we have mostly to thank those few early subscribers who by word of mouth helped us to grow. We also were fortunate enough to have our newsletter mentioned in a number of other publications.

We have never had more than three people working part time on the newsletter. If it were not so much fun, we probably would not want to do all the work, as while we do make some money on it, we are never at the point of feeling rich, or of considering taking on an employee or two. I think it is about the right size business, or it might be twice as big and we could forget outside work, but I don't think we would want it much more than that. In fact, with such a special interest subject, there is perhaps a real limit to what can be done with growth. At the moment it is just my wife and myself working on the project, with our five year old daughter taking on some responsibility for stamps and address labels with more than a little enthusiasm.

I would rather talk about Electronotes than myself, but it is perhaps useful for readers to know a little about the editor in addition to what may be found above. My musical interests are in addition to following the developments in electronic music, mainly of the "classical" type. I am interested in Bach in particular and in what I call the "late romantic" composers: Bruckner, Mahler, Nielsen, Vaughan-Williams. I hope there are some who will understand my thinking here. My engineering interest also extend to speech research and to all types of circuitry. My "hobby" science is astronomy with astrophysics and cosmology combining this interest with a phycics background that although losing to electronic music, just won't go away completely. I am also very much interested in the search for extraterrestrial life, not in UFOs flying over to be sure, but rather in radio waves going by. I might even prefer this search to electronic music were it not for the fact that I prefer activity that provides a reasonable ratio of successes to attempts. Having spent some of my early years as a part-time farmer, I don't mind putting a few seeds in the ground to see what comes up, and I do enjoy gardening and building things around the house. Actually, come to think of it, what I enjoy most is being able to sleep till noon when the snow is three feet deep outside!

Q: What do you consider to be the best approach to using analog delay lines to achieve artificial reverberation?

A: I think that at the moment there is no final solution to this problem. We have to consider first that while there are some apparent absolutes for good artificial reverberation (reverb times of a second or more, lack of discrete detectable echos, and sufficient echo density of 1000/sec or more), there is still room for individual taste, just as there is in the concert hall. Those engineers working with purely electronic music have the further consideration that the reverb unit may often be just another link in the synthesis process, and not related to physical reverberation, except in a general sense.

Regardless of our final goals (audio processing or new synthesis) we basically have the standard bucket-brigade type of analog delay lines as the only economical and readily available delay devices. Combining the available devices with the general requirements, we arrive at several conclusions. First, to get sufficient echo density we need to use at least several delay lines, and this will probably be limited by economic considerations and available building time. Secondly, to achieve sufficient echo density with a limited and relatively small number of delay lines, we have to use feedback structures (recirculation). Thirdly, we must choose a feedback structure that does not color the sound spectrum (non-flat frequency response), or at least we should know what the coloration is and allow for it. Finally, we must consider that the already questionable signal-to-noise ratio of bucket-brigade devices becomes more of a problem as signals are recirculated and more noise accumulates.

After a little study, you will see that what we have said above tends to force us into a fairly tight corner. Currently engineers are searching for new structures (such as cross-coupling) or other tricks to get an acceptable sound with available resources. The individual builder will find experimentation the best approach. He should keep in mind that he will need at least three delay lines, should start with the simple structures such as Schroeder's all-pass (see MEH Chapter 6c for examples), and should not expect miracles from a signal-to-noise point of view. It is quite likely that you will come up with some interesting sound processors even if you don't achieve your idea of true reverberation.

▷ Q: I am interested in achieving a synthesized pipe organ sound. What can you suggest

A: I think this is quite interesting because I have always felt there is a close connection between synthesizers and pipe organs as far as the control of sound goes. If you consider it for a moment, a keyboard for a pipe organ offers about the same degree of control over the sound that a synthesizer does. This control may be just a simple on/off action, or there may be some type of additional sensing of key position or pressure. It then seems logical to me that both should fail for the same reasons and under similar situations. I think this is true. I find that both synthesizer music and organ music are most interesting to me when they are moving rapidly from note to note. I also find both synthesizer and pipe organ music to be boring (boring in the sonic sense, not a total musical sense) when asked to play extended notes. There is one saving factor that may appear in both cases. The long notes may be interesting if they provide the ear with information unavailable from other sources. In the case of the pipe organ, this may be the fullness, the volume, and the rich bass that can almost literally move you. In the case of the synthesizer, it is a new tone color or new type of animation that keeps the ear from turning off these long and otherwise relatively stationary notes. Thus, it seems to me that many of the factors that are needed for an overall synthesis (synthesis of tone color and control mode, etc.) are already common factors in pipe organs and synthesizers, and in particular, are found in polyphonic synthesizers.

Very likely the discussion above was not the main thing you had in mind when you asked the question. Probably, it was the synthesis of tone color that you were more interested in. There are a few papers on pipe organs that may provide useful clues, but I have not studied them carefully enough to make useful comments here. One thing I would point out is that many of the enrichment techniques developed for synthesizers can be applied (such as parallel VCO's). What you are basically after is a sound with many spectral components provided by tracking (which is always approximate) rather than by a complex waveform with "locked" harmonics. You may want to place the tracking voices much as you would find them in organ ranks. As a last point, never underestimate the effect of artificial reverberation (even a spring unit) in providing a subjective enhancement to organ sounds. By tradition (if not by cliché) organ sounds are associated with playing environments that provide more reverberation that would be required, or tolerated, for other music. In the synthesis or pipe organ sounds, unlike the true pipe organ, you can control this reverberation, reducing it when necessary to prevent the washing out of fast passages.

▷Q: The current sources you use for VCO's and VCF's are exponential - right? So why are the sawtooth and triangle waveforms you get out of them straight lines and not curved like an exponential?

A: The problem here is that we have to be more careful about our terminology. A current source, linearly controlled, exponentially controlled, or any other kind will form a linear ramp when it is used to charge a capacitor, as long as the current is being held constant. A glass with straight sides will fill with water at a linear rate when held under a faucet as long as the water flows out at a constant rate. The fact that this constant rate depends on line pressure, other nearby faucets on or off, whether or not you paid your bill, and so on, makes no difference. What we should be looking at is the manner in which this current, momentarily held constant we will assume, changes from time to time. The exponential current source just means that the current's magnitude is determined as an exponential function of a controlling voltage. Perhaps we should say that a current source is a current source, except in the way it is controlled?

▷Q: I store my synthesizer and parts in an unheated area that often goes below zero on a winter night. Might this cause any damage to active parts? How about when the area is heated during work periods (condensation?)?

A: I can't see that this could cause any actual damage except for some very long term effects which will probably never make any difference. There is no problem storing components at 0°F at least from what the manufacturer's specs show. You may well have however, some significant inconvenience as a result of this situation. The thing that comes to mind first is that things will probably be drifting wildly as you begin to warm things up, and it may take hours to get things under control. You might also want to take some care with any components that get quite hot, as any thermal stresses as a result of self-heating will be greater from 0° to operating temperature than from say 70°. If you have such components, you might wait till the room warms before turning on the power switch. I don't see any real problems with circuits of the type we have published. Moisture as a result of condensation can be a real problem with electronic circuits, but it seems to me that any moisture that condences as a result of heating the room will very soon evaporate again. The moisture you want to look out for is the "damp cellar" type that usually occurs during warmer weather and may be around for days, weeks, or even months. In such a situation, a standard dehumidifier of the type used to prevent tools from rusting is one answer. Another is to seal the equipment up while it is dry, and there are several ways of doing this. Another way, which will protect an area about the size of a closet is to mount a few 60 watt bulbs around the base of an area and leave them on all the time damp conditions exist.

## CLASSIFIEDS:



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