TIMBRE & VOICING CIRCUITS FOR ELECTRONIC MUSIC

BY DON LANCASTER

Techniques for converting basic frequency references to elaborate musical notes.

THE majority of electronic musical instruments generate tonal frequency references in whatever waveform happens to be handy. To convert a sine, sawtooth, or square wave into varied and interesting musical sounds, the signal's time and frequency structure must be altered so that it simulates the sound of a traditional musical note or provides a new type of note or special effect.

To alter the time structure of the envelope of a tone (see "Keyers and VCA's," January and February 1975), a suitable waveform with the attack, fallback, sustain, decay, snubbing, and echo desired is combined with the required frequency reference. The envelope waveform is then multiplied by the frequency or tonal information to give the overall tone time shape desired.

But this is only half of the problem. A steady-sounding trumpet is very different from a steady-sounding clarinet, even when both instruments are sounding the same note and attack and time differences are ignored. The differences in sound between different types of musical instruments are called color or timbre. Circuits that alter timbre are called voicing circuits, filters, tracking filters, and voltage-controlled filters (vcf's). Only when suitable voicing is combined with a selected envelope does the final musical note result.

Elements of Tonal Color. Color differences between notes are obtained by the presence of harmonics, nonharmonic but related multiples of the basic pitch, multiple tones in chorus and warmth situations, and "extra" sources of noise (chiff, buzzes, random noise and variations, and other unrelated audio energy sources). By selecting the correct combination of these things, almost any desired timbre waveshape can be generated. Multiply the waveshape by an envelope, and the note can have any color you want.

Harmonics are the easiest with which to deal, but they are not by any means the whole story. If the zero-crossing points of a musical waveshape always occur at the same fixed fundamental frequency, that waveshape can be broken down into a fundamental-frequency sine wave and a series of harmonic sine waves of various phases through a mathematical process known as Fourier Analysis.

In Fig. 1 is shown how sine waves and their harmonics can be combined to build three basic musical waveforms, the square, sine, and sawtooth waves. Creating a new complex waveform is known as synthesis. While direct analysis and synthesis are often the best ways to find out what color you want and how to go about obtaining it, you rarely use direct synthesis to do the actual coloring process. This is because you can generate a square wave (for example) much easier with a digital flip-flop than by summing dozens of sine waves.

The undistorted sine wave has no

EMS "Synthi KS" Digital Sequencer Keyboard

harmonics present and is close to a flute-like tone. The square wave consists of diminishing odd harmonics only and produces a nominally hollow or "woody" sound like that associated with such woodwinds as the clarinet and stopped organ pipes. The sawtooth wave has all harmonics present in a uniformly decreasing series and produces a "bright" tone like that of a stringed instrument or, with bandpass modifications, a trumpet or other horn. Since everything needed is available in the sawtooth wave, it often represents a good "universal" waveform with which to work in electronic music because it is easy to use it to build a lot of realistic voices. Other basic waveforms can lead to voicing difficulties. The triangle waveform, for instance, does not have enough harmonic strength to be useful and the impulse has far too much strength.

The methods of getting back and forth between the basic waveforms are shown in Fig. 2. A sine or triangle becomes a square wave by heavily amplifying and then limiting it, but this can be done only to one sine wave at a time if intermodulation distortion is to be avoided. A sawtooth linearly becomes a square wave if minus one-half the second harmonic is summed with itself. This is called "outphasing," a technique that can handle many notes at once.

Getting from a square wave to a sawtooth wave can be accomplished by using a binary divider and a staircase generator. This particular approximation sounds much better than it looks. In a 16-step staircase, the first missing harmonic is the 16th, the next is the 32nd, and so on. Otherwise, the waveform is identical to a sawtooth.

Finally, to get from the more complex waveforms back to a sine wave, the fundamental can be sharply band-pass or low-pass filtered. A high Q needed for good second- and third-harmonic rejection directly conflicts with the need to handle many notes of different frequencies simultaneously. So either a tracking filter or fixed filters of only one-third—or at most one-half—an octave can be used.

Harmonics permit us to build many instrument imitations. Nonharmonic multiples can also be very important and are usually much more difficult to handle electronically. Nonharmonic multiples in the piano are "almost harmonics" caused by lateral stiffness of the strings, particularly on the low-frequency end. These almost-harmonics are called "partials" and get progressively sharper (higher in pitch) with harmonic number. For this reason, piano keyboards are not tuned to absolute pitch because, if they were, the nonharmonic partials of the low notes would sound out of tune with respect to the higher notes. Instead, piano keyboards are stretched while tuning to make the low notes flatter and the high notes sharper than they really are. The result is a warmer, fuller sound character.

Another feature of piano notes is that the hammer usually strikes three strings simultaneously. The strings are almost but not quite tuned in unison. This gives the characteristic warmth to a piano tone. It also explains why a chorus or an orchestra full of violins sounds richer than a single violin. Chorus and warmth effects can be introduced electronically by using several tone sources, differentially delaying ("Doppler modulating") one tone source, or using special-purpose tape delay units.

Yet another example of nonharmonic multiples are carillons and bells, which produce strong sequences of fifths (5:3 frequency) and others present in the characteristic cast-bell tone. Any other situation where you are intentionally generating chord sequences takes tonal groups related by something other than simple harmonics. It is often possible simply to use multiple combinations of notes already in hand.

The final noise effects—chiff, buzzes, etc.—are usually tacked on an as afterthought. In traditional musical instruments, they are inherent and unwanted byproducts or defects of the instrument's physical characteristics. Obvious examples are the wind noise in a pipe organ, steam in a calliope, resonance buzzes in a poorly designed or low-cost violin or guitar, sympathetic vibrations, etc. One time you do *not* tack on noise effects is when the note is predominantly filtered noise, such as that used for some percussion effects. In this case, the usual practice is to start with white or pink noise and filter it, the sharpness of the filtering determining how close you want to get to a well-defined fundamental tone. Small amounts of

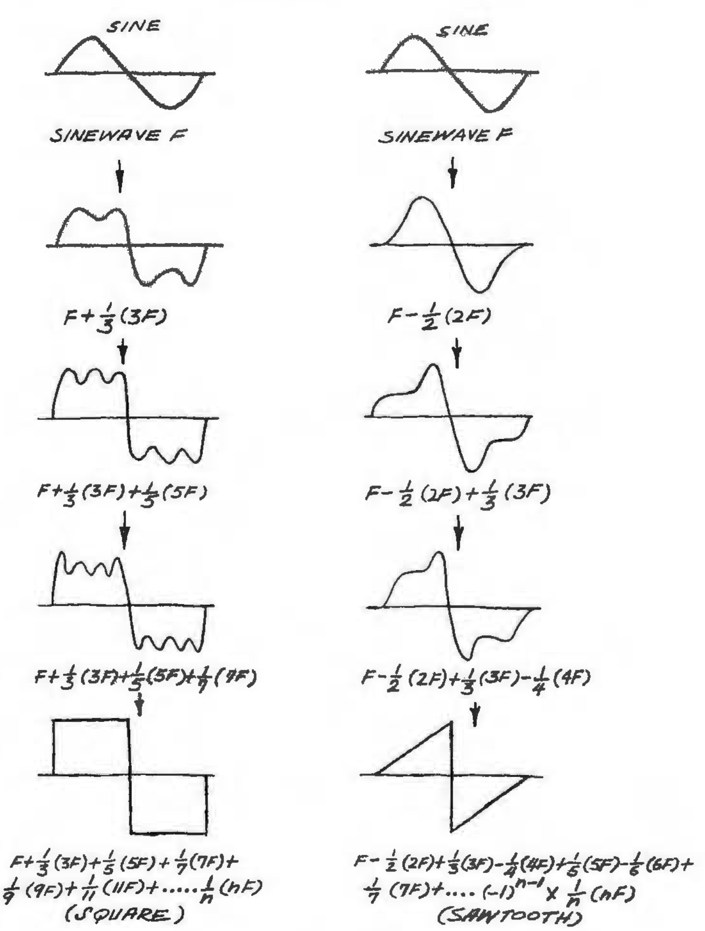


Fig. 1. Complex waveforms can be broken down into a series of sine waves and vice versa.

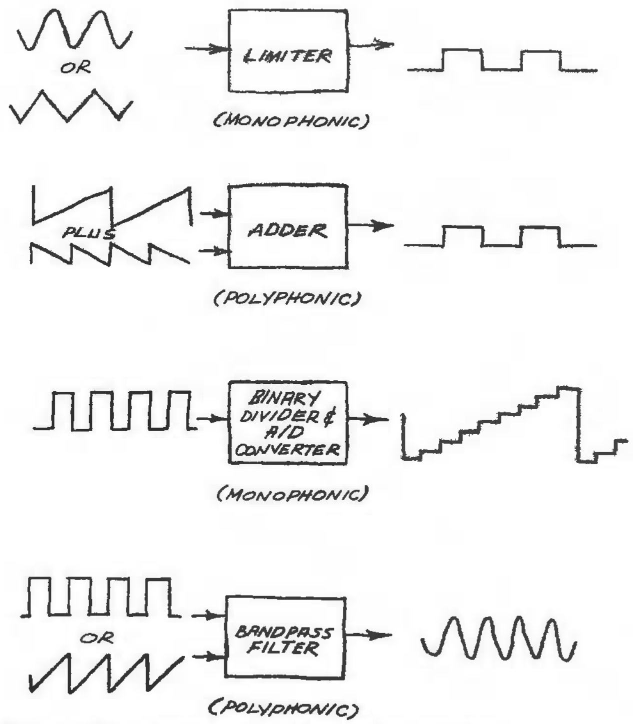


Fig. 2. Ways of transforming basic waveforms.

noise and frequency shifting can be introduced to slightly randomize or break up the exactitude of electronic pitch generation systems.

Harmonic Structure. Once you have decided what you want in harmonics, you have a second decision to make: whether you want identical harmonics on every note, changing harmonics as the notes change in frequency, or some combination of the two.

Musical instruments are basically some sort of acoustical bandpass filter. The horn on a trumpet is a pipe resonator. The guitar or violin is more or less a cavity resonator. These resonances are physical constants. Since they do not change, the harmonic structure of the notes passed through them evidently must change for notes of different frequencies. So, almost all traditional instruments introduce fixed-frequency filtering that alters the harmonic content from note to note. A fixed filter gives a variable harmonic content. If your primary goal is to imitate traditional instruments, fixed-frequency selective emphasis and de-emphasis of various frequency ranges is called for.

The sound once (and still) referred to as the "Hammond sound" is now called "synthesizer sound." Its characteristic is that all the notes have identical harmonic structures, evolving from the Hammond organ's electromechanical system in which synchronously rotated steel cams were used. As the cams rotated, each generated a series of harmonics that could be selectively added to a fundamental. All notes were treated identically and had identical harmonic structure. You can obtain the same effect in a synthesizer by using a tracking or voltage-controlled filter. Hence you need a moving or variable filter for a fixed harmonic structure.

Neither of the above techniques is "better" than the other. The one you use depends on what you are after. Traditional instrument synthesis will often sound phony if you use fixed harmonic structure. On the other hand, the fixed-harmonic-structure notes offer much in the way of flexibility and new sounds. (These two basic filter techniques are compared in Fig. 3.) One big advantage of a synthesizer is that you can handle both types quite easily.

Another reason for using variable or tracking filters is to let the harmonic content of the note change during the envelope time. In many musical instruments, higher harmonics decay faster than do the lower ones, while in the piano, sympathetic resonances may actually build up harmonics with time. These effects can often be obtained by adding an electrically variable filter to the output of the fixed formant filter, sliding the filter down as the note decays. A changing harmonic structure with envelope time, particularly with the decay cycle, is the key to realistic traditional instrument syn-

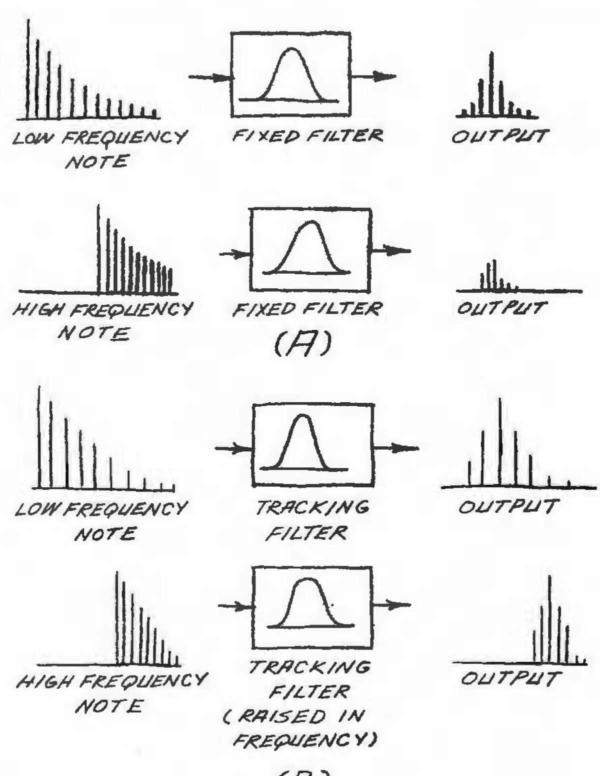


Fig. 3. Fixed filter (A) alters harmonic content from note to note. With tracking filter (B), output tone spectrum is independent of input.

thesis. It also provides many new possibilities for far-out sounds. (We will take a close look at what the traditional instruments need in harmonics and envelope in a future article in this series.)

Voicing Techniques. As a general rule, there are four basic ways to electronically generate a suitable musical voice. Starting with a waveform or waveforms containing more harmonics in a different structure than are needed, filtering, or a "subtractive" method, can be used. Also called formant filtering, it is by far the most popular fixed-filter method used today in both electronic organ and synthesizer circuits.

An "additive" method can be used. Here, harmonically related sine waves are combined in the correct proportions to yield the desired note. While this approach was often used in older electromechanical organs, it is generally complex and difficult to program, and it is somewhat limited, requiring too many harmonics to achieve good synthesis. Too, if you are attempting to synthesize the sound of a traditional musical instrument, you must also provide for changing harmonic content with changing pitch.

A third approach is called the "nonlinear" method of voicing, where a fundamental frequency is run through a diode or other highly nonlinear component to generate harmonics. The harmonics are then filtered by the subtractive method. Some monophonic pedal circuits in home organs half-wave rectify a sine wave to get the strong second harmonic needed for diapason pipe synthesis. There are serious limitations to the nonlinear method. The input amplitudes may be critical if the nonlinear system is to generate the proper harmonic structure. Also, it is strictly a one-note-at-a-time monophonic technique; two notes fed into a nonlinear circuit at the same time will produce intolerable intermodulation distortion.

The fourth method of voicing is "replication." Here, the note wanted is generated directly in exactly the shape desired by starting with some sort of model or "replica" of the tone to be produced. For example, a binary-addressed read-only memory (ROM) could be followed by a digital-to-analog (D/A) converter. Stored in the ROM would be the exact wave shape desired. The faster the ROM is clocked or addressed, the higher the output pitch, but the shape would remain the same, as would the harmonic structure. The same thing could be done with a you-program-it computer lashup in which the timbre and envelope are simultaneously programmable. Important advantages of the replication technique are its tremendous flexibility and extreme simplicity, especially when new voices are to be added to the system. You can easily try a starting system with a binary counter, 7489 ROM, and MC1406 or MC1408 (both by Motorola) D/A converter. Cost is less than $10 for the system.

We might call "noise" technique a fifth basic approach to timbre generation, but it is really a form of subtractive filtering. Instead of removing or emphasizing harmonics of a fixed tone, various portions of a noise spec-

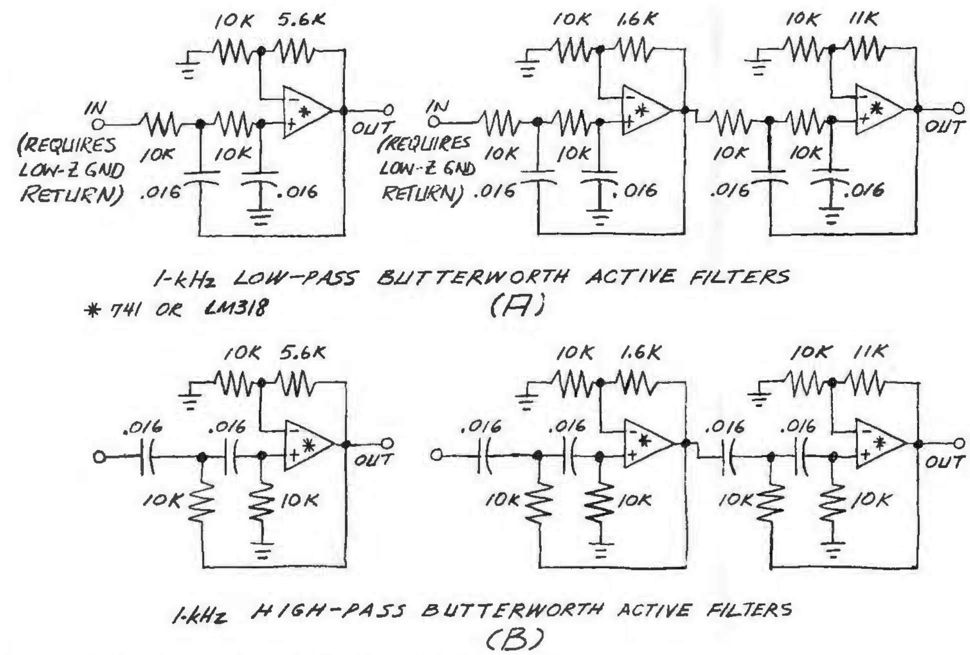


Fig. 4. Active filters on left have 12 dB/octave slopes. On right, are 24 dB/octave filters.

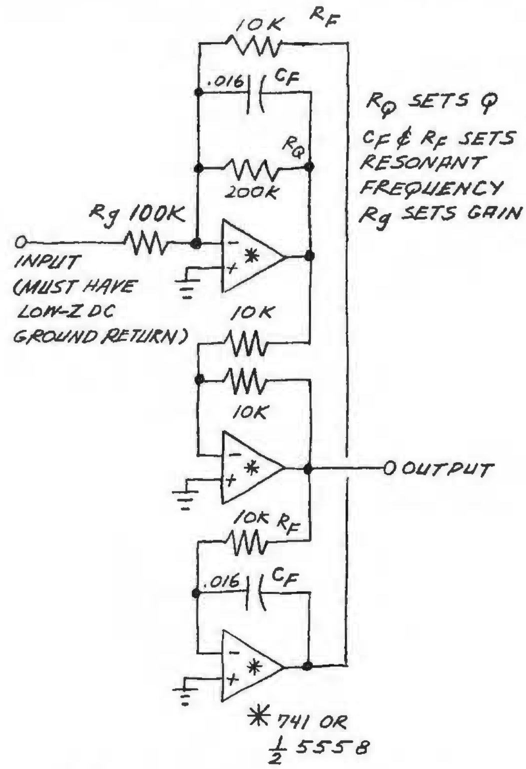


Fig. 5. Biquadratic bandpass filter set to 1 kHz resonance and Q = 20.

trum are emphasized or de-emphasized. Both are essentially subtractive filter techniques.

Filter Circuits. The subtractive filter technique is by far the most popular used in today's electronic music systems. Originally, fixed RC filters were used, with an expensive inductor or two thrown in only where absolutely necessary. This accounts for some of the rather poor instrument imitations produced by some older economy organs.

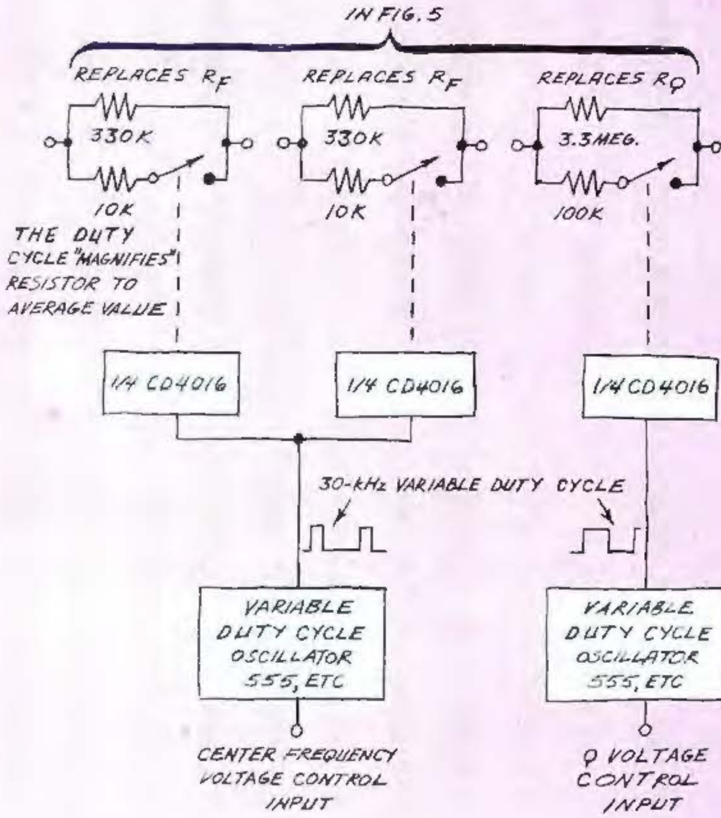
Today, we have a new approach: active filters. The active filter combines resistors, capacitors, and operational amplifiers (usually the inexpensive 741 op amp) to simulate exactly the type of filter you could normally build only with expensive, large, adjustable inductors.

Important advantages of active filters are their low cost and easy tuning and the total absence of loading problems. The last lets you combine and cascade sections in many arrangements without interaction. Better yet, is the ability of certain active-filter circuits to be electronically controllable to permit you to electronically shift the cutoff frequencies or bandwidths under voltage or digital-word control. This lets you move the filter from note to note or change the harmonic structure during the envelope time.

Pairs of high-pass and low-pass active Butterworth filters with 12- and 24-dB/octave slopes are shown in Fig. 4, while a single-pole active bandpass filter circuit is shown in Fig. 5. For higher frequencies and higher Q's, premium op amps like the National LM318 should be used to replace the 741 device, for which it is a direct pin-for-pin replacement.

To change the cutoff or resonance of the above filters, simply scale down the values of all the capacitors. For instance, doubling the capacitive values moves the 1000-Hz response of the figures down to 500 Hz. You can also scale down the resistor values, but you must keep all frequency-determining resistors and all capacitors identical in value.

The fixed active filter can be used for formant synthesis of traditional instruments, usually by starting with a sawtooth for most voices, except for some woodwinds that are better off with a square wave. Heavy sine-wave filtering can be used for flute and piccolo and certain bland organ voices. ◈



CMOS switch is added to circuit in Fig. 5 to make a high-performance voltage-controlled bandpass filter.

ELECTRONIC TRACKING

How is it possible to vary *electronically* the cutoff frequency of an active filter? The answer is that anything that can change the value of a resistor under voltage control—such as a FET, incandescent lamp, or LED and photocell—will do the job. However, linearity, power, "insertedness," and cost might be objectionable. The CA3080 circuit on page 37 of the February 1975 issue provides good filter control.

Today, there is a new and easy way to build a tracking filter, working with the inexpensive CD4016 (RCA) or MC14016 (Motorola) CMOS integrated analog switch circuit. It works in an active-filter circuit that is basically an integrator, such as the low-pass versions in Fig. 4 and the bandpass circuit in Fig. 5. The details are shown in the circuit above.

The key to the process is to make a fixed resistor appear to be variable. If the value of a pair of fixed resistors can be electronically controlled, the same thing could be done as varying a dual potentiometer and automatically tracking any note you want. Better yet, if you want to, you can change parameters *during* the note's envelope, providing all sorts of synthesis possibilities.

The CD4106 is a quad analog switch that rapidly lets you turn a given resistor on and off. If you vary the off-to-on duty cycle, you can make the resistor look—on the average—like any resistance you want, from its actual value to infinity. The only requisite is that the switching be much faster than the highest frequency of interest so that the capacitors in the circuit average out the on/off switching into an essentially proportional resistance. In a 400-Hz bandpass filter, for example, an 8000-Hz *minimum* switching rate is recommended, with proportionately higer ultrasonic frequencies needed for higher cutoff frequencies.

The switches can be driven from an ultrasonic source of rectangular waves with a variable duty cycle. This is easily accomplished with CMOS logic. Alternatively, you can use the 555 or 8038 IC devices for initial tests, switching over to a voltage-controlled duty cycle modulator later on. (Incidentally, really good wide-range active bandpass filters that are stable and track have not been available at reasonable cost before. Both these techniques give you a wide open field for serious electronic music experiments.)