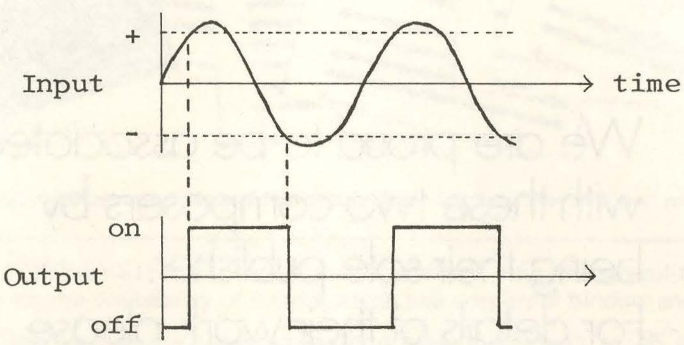
New Instruments through Frequency Division

SINCE 1945 MORE AND MORE interest has been paid to timbral transformation as a vital process of composition. Orchestration, as a transformation technique, is concerned with which instruments should share which pitches, and often which pitches should be doubled in the octave relationship. Timbral transformation, octave division and multiple octave division can also be accomplished electronically from a single source, rather than orchestrationally from multiple sources. (In these days, this is frequently an economic as well as an aesthetic consideration!)

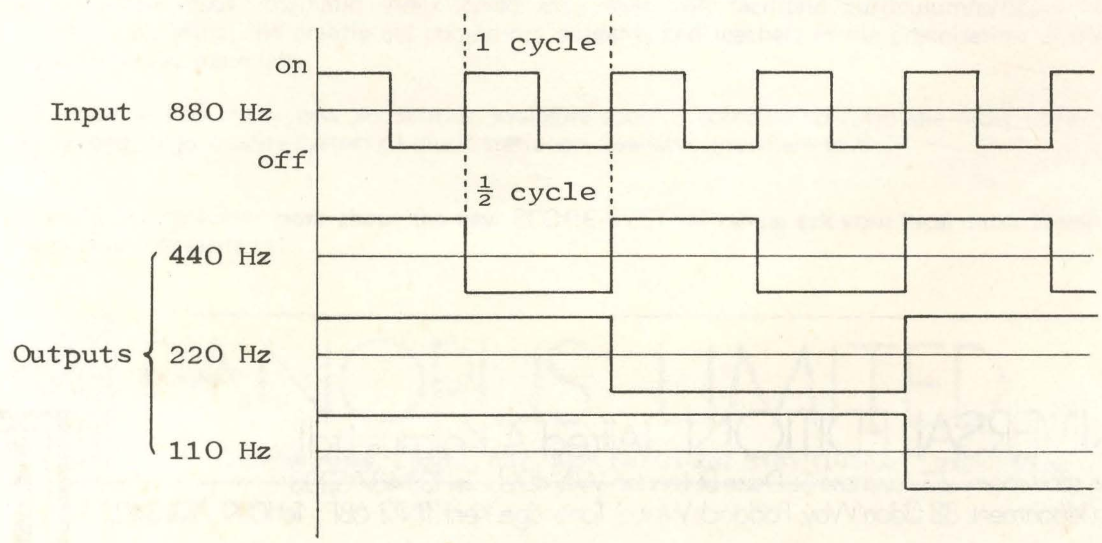
The octave relationship is logarithmic: in other words, if the note a″ (880 Hz or cycles per second) is to be lowered by one octave, its frequency must be divided by two; if it is to be lowered by two octaves, its frequency is divided by four; by three octaves, divided by eight, and so on. So what is needed is a device that, when fed a certain frequency, will output exactly a half, quarter or eighth of that number. This is what happens in an electronic device called an 'octave divider'. A signal to be transformed is dealt with in two stages. First it is passed through a digital device[1] called a Schmitt trigger, the output state of which changes from on to off, or vice versa, whenever the input voltage crosses certain preset positive and negative thresholds; the incoming signal is thus transformed into a square wave (Fig.1). Secondly, the square wave

Figure 1



output of the Schmitt trigger is fed into the divider, also a digital circuit, the output state of which changes when a positive voltage is received at the input. Since a whole cycle of the input generates a half-cycle of the output, the output frequency will equal the input frequency divided by two, or one octave down. To divide by another octave, the divided signal is itself divided, and so forth (Fig.2).

Figure 2



By means of these two very simple devices, commonly found on the same IC (integrated circuit or chip), one can transpose an instrument down any number of octaves, and transform its timbre to that of a square wave. It should be pointed out that a drawback of this system is that the input must be fairly pure, i.e. the fundamental must be the strongest component of the signal. The guitar, for example, has a very strong second harmonic, and often the divider cannot decide between whether to follow the fundamental or this the second harmonic. This results in 'hiccups' between the octaves. Manufacturers of commercially marketed dividers suggest that when using a guitar, it is best to stick to the upper three strings, which are less rich in harmonics. The unit works very well with practically every other instrument, especially the winds.

The spectrum of a square wave consists of the odd-number harmonics in the amplitude ratios of the reciprocal of the harmonic number. (For harmonic number 'n' of a fundamental 'F', frequency =n(F), amplitude = 1/n(amplitude of F); e.g. the 7th harmonic of the fundamental 110 Hz has a frequency of 770 Hz and 1/7 the amplitude of the fundamental.) Because of its harmonic construction, the sound of a square wave is quite harsh — much like a clarinet in the chalumeau register. Any of the characteristic harmonics can be amplified or attenuated by filtering and the timbre modified. In this way a rich sound source may be produced that is related in pitch to the original input.

Once two or more octaves have been generated, they can be added to one another in various amplitude relationships to create a multitude of wave-forms. Let's say, for example, that an input of 1660 Hz has been divided four times: there will be four outputs — 880 Hz, 440 Hz, 220 Hz and 110 Hz. If these are added together in equal proportions, the harmonics will relate to the fundamental, or lowest tone, 110 Hz (hereafter F_1). The spectrum of F_1 is shown in Fig.3, which displays the decay in the amplitudes of the harmonics.

Figure 3

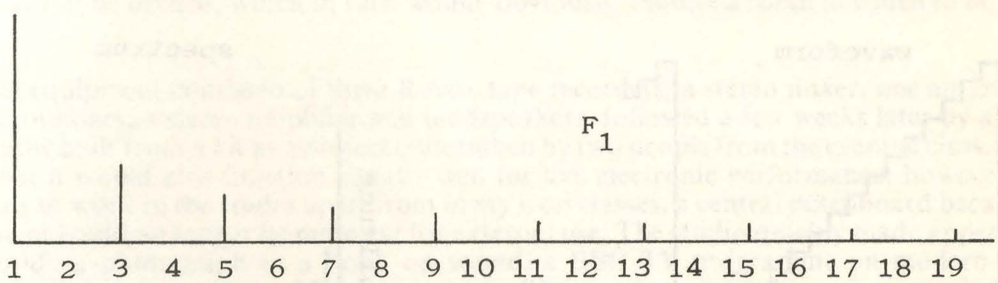


Fig.4 shows the result of adding to this the octave above (F_2).

Figure 4

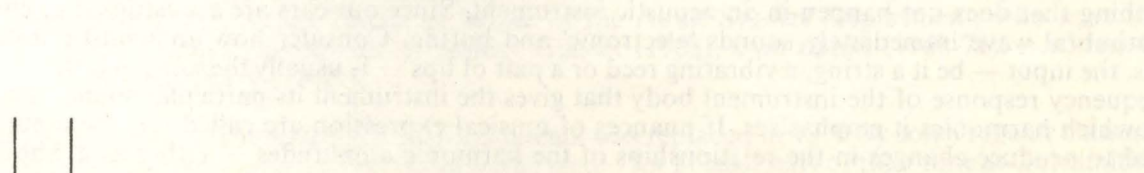
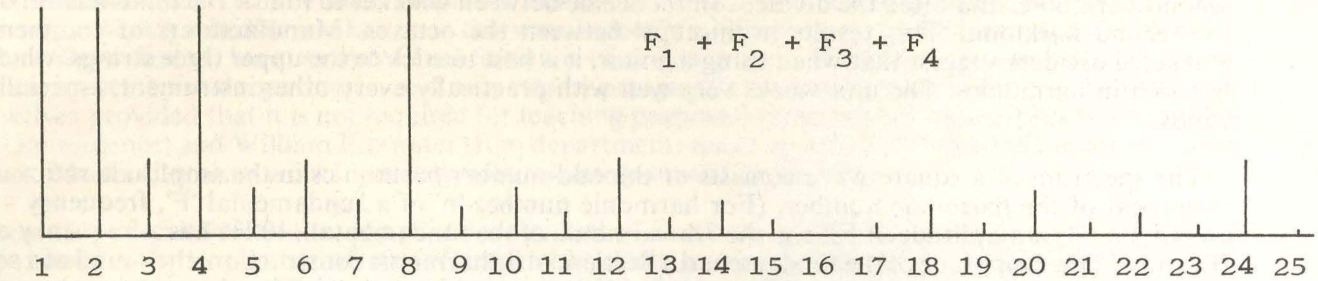


Fig.5 gives the spectrum that is produced when all four octaves are added equally.

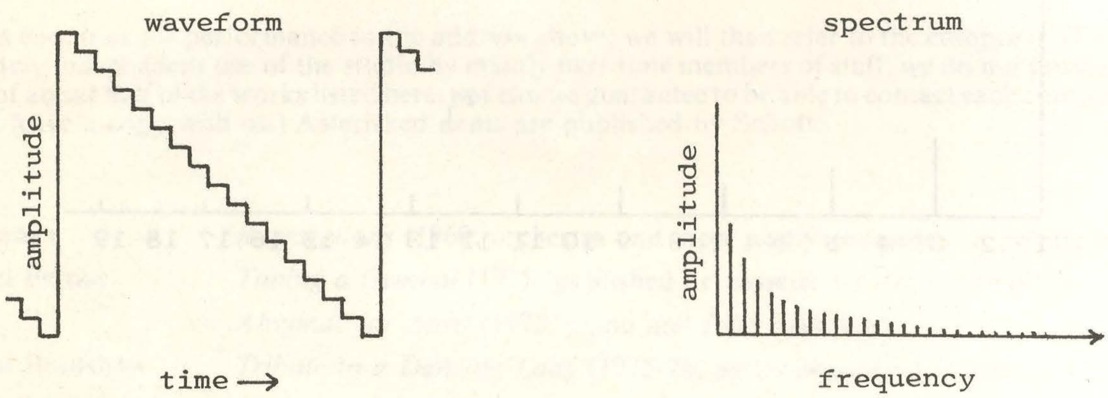
Figure 5



	1	2	3	4	5	6	7	8	9	10	11	12	13	14	15	16	17	18	19	20	21	22	23	24	25
F_1 110 Hz	*		*		*		*		*		*		*		*		*		*		*		*		*
F_2 220 Hz		*				*				*				*				*				*			
F_3 440 Hz				*																*					
F_4 880 Hz								*																*	

This is quite an unusual distribution, and as one can imagine, a very 'live' sound with such high amplitudes at the 8th, 12th, 20th and 24th harmonics. If the four octaves are added with amplitudes descending in the proportions $F_1+\frac{1}{2}(F_2)+\frac{1}{4}(F_3)+\frac{1}{8}(F_4)$ the result is a staircase wave, equivalent to a ramp wave with every 16th harmonic missing,[2] and in which there is an even decay of both even and odd harmonics (Fig.6).

Figure 6

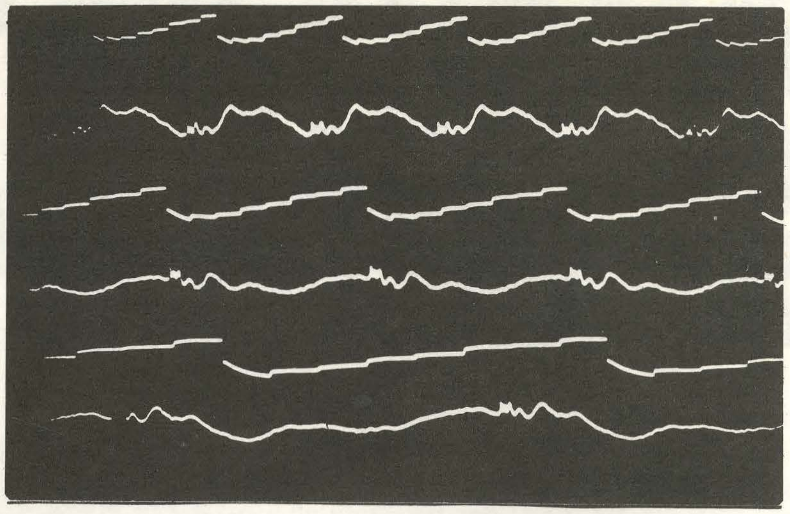


After these further harmonics have been generated through addition, the composite wave can be filtered to synthesise many more wave-forms, which will respond 'acoustically' when passed through a third-octave filter, the use of which is described below.

One of the shortcomings of synthetic tones, as many electronic musicians are now finding, is that they are too efficient. In other words, the wave-form stays exactly the same throughout the entire range, something that does not happen in an acoustic instrument. Since our ears are accustomed to change, a monotimbral wave immediately sounds 'electronic' and boring. Consider how an acoustic instrument works: the input — be it a string, a vibrating reed or a pair of lips — is usually the same; it is the resonance or frequency response of the instrument body that gives the instrument its particular sound, depending upon which harmonics it emphasises. If nuances of musical expression are called for, the input is then altered to produce changes in the relationships of the harmonic amplitudes — either as a whole (i.e. a change in volume) or individually (i.e. a change in spectrum).

This is how the third-octave filter modifies a composite-wave input. The device operates by dividing the audio spectrum into bands of one-third of an octave, each of which can be controlled independently. By virtue of this tunable response the filter can be made to act as an 'artificial body' with almost limitless variability. Fig. 7 shows an oscilloscope reading demonstrating the action of the filter. Three frequencies (1,2,3), each with the same wave-form, are passed through the filter, the setting of which remains constant. The differences in response between the three outputs can be seen easily.

Figure 7



So the end result is a very versatile instrument in which the spectrum of the input is variable according to the mixture of octaves, and the response of the output is tunable within a third of an octave.

So far we have just been dealing with octave division. With the addition of one more circuit in the process, octave multiplication is also possible, giving any input an infinite range, up or down, limited only by the human ear. But why stop at *octave* division and multiplication? Octave-related division is necessary for subtle timbral transformation, but for intervallic transformation, whole-number division and multiplication of fundamentals can also be used, i.e. the frequency of an input can be shifted by 1,2,3. . . and not just 2,4,8. . . . So, if one wanted to shift the transformed wave a major third above the input, since the ratio of a major third is 5:4, one would multiply the input by 5 and divide by 4. (This sort of logic is basic to the manufacture of electronic organs that do not use separate oscillators for every note.)

With a handful of inexpensive electronic components, an instrument can in this way be transformed extensively in both timbre and pitch. In addition, the steady-state tone that such a process produces can be modified by enveloping it (i.e. providing it with attack, sustain and decay characteristics). The envelope might either be the same as that of the input or else an entirely different one that would further disguise the identity of the instrumental source.

The real challenge arises when it comes to mixing the newly synthesised tone with the input, or the rest of an ensemble. It must be remembered that this is a new instrument, which one must learn to play, and that like any musical instrument its strengths and weaknesses must be digested in order to facilitate musical expression. With so large a dynamic, timbral and pitch range, a great deal of subtlety is needed to achieve the correct musical balance. (One can imagine the psychological problems of playing a piccolo and the sound of a tuba emerging, or vice versa!) Division has been used successfully throughout the past decade in many different kinds of music: Herbert Lawes plays divided flute on many solo jazz albums, Captain Beefheart featured the 'Air Bass' (divided trombone) in his last British concert tour, and Paul Chihara, the young Oriental-American composer, uses divided flute as the solo instrument in one of the movements of his *Forest Music* suite, commissioned by the Los Angeles Philharmonic in 1970.

The possibilities are endless and exciting. I hope that composers will take advantage of this new sound world, and that players will discover this powerful extension of their own instruments in order to become more diversified musicians, and help create a more versatile music of the future.

NOTES:

[1] A digital device is one that can be in only one of a fixed number of possible states. In the case of the Schmitt trigger there are two — on or off. An analogue device, however, can vary continuously.

[2] It is interesting to note that with octave square wave addition, the harmonic that will be missing from the spectrum is determined by the formula 2^n where 'n' is the number of octaves added. In Fig. 4 there are two octaves, $2^2 = 4$, therefore every 4th harmonic is missing, which is evident in the figure. For Fig.5, $2^4 = 16$, and so on.