COMPUTER SOUND GENERATION

Music 198

Section 1

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I. Introduction

Starting in 1958, M. V. Mathews and others at the Bell Telephone Laboratories developed computer programs[1] and related equipment[2] which could be used in generating sound by means of a digital computer. The original work has been continuously expanded[3] and now offers the composer an extremely flexible sound source. The sounds possible are essentially limited only by the imagination of the composer and his ability to represent the desired sounds mathematically, the technical limitations of the method being slight. In fact, all techniques commonly used in the production of non-concrete electronic music in an "analogue" electronic music laboratory can easily be handled by the computer method, but in addition a multitude of sounds either too difficult or simply impossible by the former method can be produced by the latter.

Two main disadvantages to the Bell Laboratories method exist. First, the cost of computer time is high, amounting to about $100 per minute of music. The other disadvantage is the problem involved in training composers in the techniques necessary to effectively use the method.

However, it is hoped that these two disadvantages will both be overcome in the situation at U.C.L.A.

Another computer sound generating technique has been developed, at the University of Illinois[4], for the purpose of reducing the cost problem. But since if a computer is available to interested parties at U.C.L.A. it will be capable of handling the Bell Laboratories program at no cost to those concerned, and since the University of Illinois method appears to be quite limited in comparison, it will not be considered here.

II. A Brief Introduction to the Bell Laboratories Method of Computer Sound Generation

The technical details of the Bell Laboratories method are well described in the references. However, it is felt that an introduction for musicians might be a help toward the understanding of the technique. Toward this end we shall briefly describe an idealized process of sound generation and perception. Refer to figure 1.

Figure 1

Sound is produced in air when a sound producer, such as a violin string or a loudspeaker cone, causes the air to undergo a series of compressions and rarefactions. These air pressure variations, when incident upon the ear drum, cause it to move and the perception of sound occurs. If the sound source is a loudspeaker cone, the air pressure variations occur in proportion to the voltage into the loudspeaker. Consider the events above at some instant of time, as if a "snapshot" could be taken of all participants. The cone of the loudspeaker will, in general, be displaced from its equilibrium or "rest" position by a certain amount in response to the particular voltage entering the loudspeaker. In turn, the resulting air pressure variation from the loudspeaker will cause a displacement of the ear drum. The process of sound generation and perception can be considered then as a series of these events occurring successively. In actuality we would have to say that the number of events occurring during any period of time is infinite, but it need not be. If we have less than an infinite number but still a very large number we can generate sound inperceptably different, to any degree desired, from a "natural" sound generation method. This idea, called "sampling", is the basis of the Bell Laboratories method.

DIGITAL SAMPLING OF A SINE WAVE

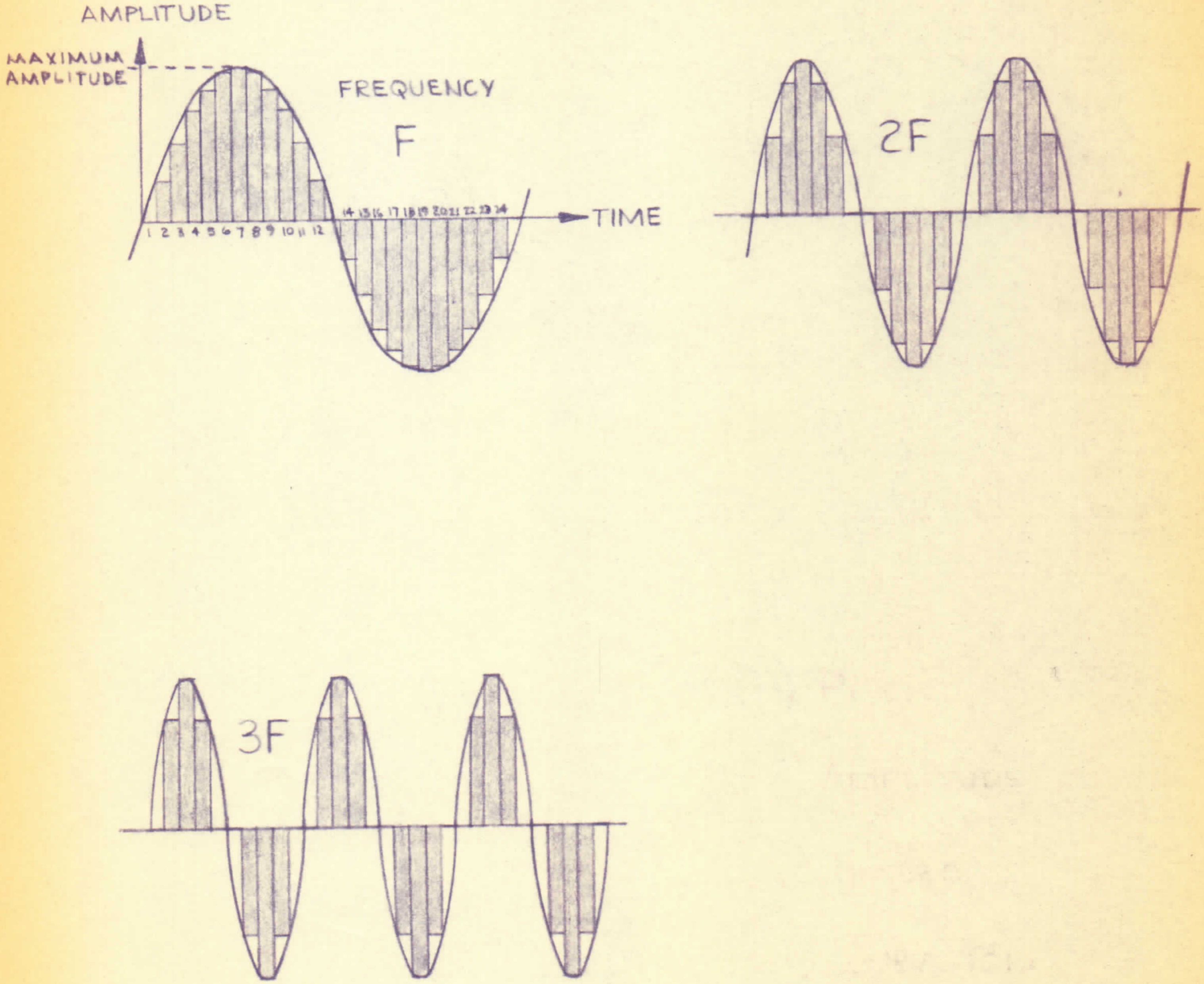


Figure 2

A sine wave is an example of a "periodic" wave, that is, a wave whose amplitude excursions are repeated each interval of time called the "period" of the wave. See figure 2. Let us choose as our starting point that point where the amplitude is zero and is going in the positive direction. The amplitude increases as time progresses at a continually decreasing rate until the positive peak of the wave is reached. At this point the amplitude starts decreasing at a continually increasing rate until the amplitude is zero, at which time the amplitude, while still decreasing, decreases at a continuously decreasing rate until the negative peak is reached. Then the amplitude starts increasing until the zero point is reached. The wave, of course, continues, but at this point it has completed one complete 'cycle'. As mentioned above, the time necessary to complete one cycle is called the period of the wave, and the reciprocal of the period is the frequency or pitch. The loudness with which the sound is perceived is governed by the amplitude of the wave. The timbre of the wave is determined by the waveform, in this case a sine wave. The duration of the sound is determined by how many successive cycles occur.

We can consider this sine wave as a succession of an infinite number of separate amplitudes. But, as mentioned above, an infinite number are not necessarily needed. It is a well known fact of information theory that if a waveform with a frequency of F cycles per second is to be represented by a discreet number of successive amplitudes, or samples, the minimum number of samples required is 2F samples per second. For example, if frequencies up to 10,000 cycles per second are required, then at least 20,000 amplitude samples per second are required.

The mechanism by which the computer performs this sampling operation is as follows. First a sine wave with a maximum amplitude of one is divided into a large number of parts, amplitude samples, and each sample is assigned a relative amplitude (between zero and one) according to its position in time. The table below lists the relative amplitudes of a sine wave if the number of samples is 24: (Refer again to figure 2)

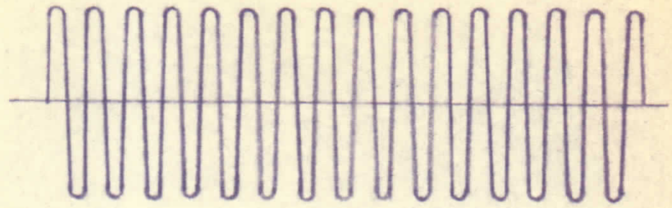
Time Position:	1	2	3	4	5	6	7	8	9	10	11	12
Relative Ampl:	0.0	.26	.50	.71	.87	.97	1.0	.97	.87	.71	.50	.26
Time Position:	13	14	15	16	17	18	19	20	21	22	23	24
Relative Ampl:	0.0	-.26	-.50	-.71	-.87	-.97	-1.0	-.97	-.87	-.71	-.50	-.26

The number of samples per cycle in the Bell method is 512. These samples are then stored in the computer memory. If we now cause the computer to "output" the amplitude samples in successive order at a constant rate, through a device called a Digital to Analogue converter, we obtain a continuously varying voltage, which if applied to a loudspeaker produces a sine wave sound.

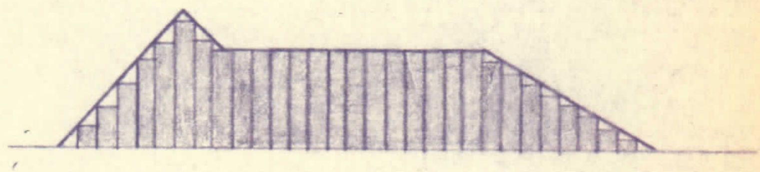
The amplitude of the sound can be varied by causing the computer to multiply the relative amplitudes of the samples by a certain factor before they are output. This factor can be changed gradually to affect a steady crescendo or decrescendo, or it can be changed abruptly. The frequency of the sound can be changed by causing the computer to skip over certain samples as it outputs them. For instance, if the computer chooses every sample the lowest of the frequencies possible with the method will be produced. If the program is set to sample at a 20,000 sample per second rate, and the computer selects every sample, the frequency of the waveform will be 39 cycles with the Bell method. If the computer selects every other sample (see figure 2) the frequency will be 78 cycles per second, 117 cycles per second if every third sample is selected, etc. If the computer selects only two samples per cycle

ATTACK & DECAY

SINE WAVE:



MODULATION:



MODULATED SINE WAVE:

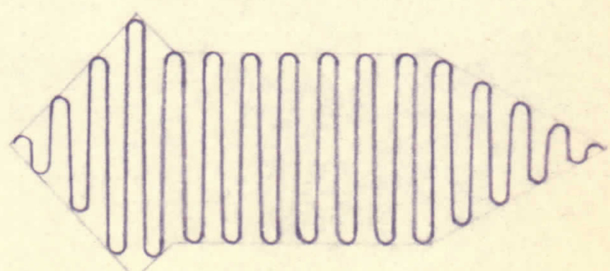


Figure 3

the frequency will be 10,000 cycles per second. The duration of a single note is determined by how long the computer continues to output samples without a break.

The timbre of the sound is determined by several factors. The primary factor is the basic waveform. The examples given have been for a sine wave but any waveform which can be represented by a succession of discreet samples can be produced. Also, synthesis of complex waveforms from harmonics of simple waves is easily accomplished by causing the computer to add together the proper samples before it produces an output. The timbre of a sound is also affected by the attack and decay characteristics of the sound. (see figure 3) The relative amplitudes of the samples can be multiplied by another slowly changing function, also stored in the computer as samples, before they are output. This causes the basic waveform to be amplitude modulated, giving it attack and decay qualities. Or a different amplitude modulation function can be used to obtain tremolo. If the number of samples the computer chooses during each cycle is caused to vary slightly from cycle to cycle, frequency modulation or vibrato is produced.

All but the simplest music involves several sounds occurring simultaneously. The computer method accomplishes this by adding together the individual samples of each sound to produce one final output sample. For example, let us assume that the computer is in the process of producing sound for a three voice composition, with sine waves, square waves, and sawtooth waves being these three voices. For each final sample, or interval of time of 1/20,000 of a second, the computer must choose the proper sample for the sine wave, multiply it by an amplitude factor, perform similar operations for the square and sawtooth waves, and then add the three samples together to produce the output sample.

In addition to being able to produce periodic waveforms by the method just described, the Bell method is also capable of subjecting any waveform to bandpass filtering, and of producing random noise and random notes within any specified range. The explanation of these features is covered in the references.

In order for a composition to be realized using the computer as a sound source, the composer must select his orchestra. For composers who are just beginning to use the method instruments such as those described in the references may be used as a start. Or, computer equivalents of electronic music laboratory sounds may be used. It is felt, however, that as the method is studied and used its potentialities will become evident and the technical barriers will disappear.

III. Outlook

The computer sound generation method can be considered, in general, as a broadening of the technical facilities available to the composer of electronic music. Assuming a continual expansion of the method, there seems no reason why it should not become the source of a large percentage of the electronic music of the future. This assumption is based on the following facts. At present the method is capable of producing most, if not all, of the sounds producable in an electronic music laboratory, but with much less work for the composer. For instance, no splicing of tape or continual re-recording of sounds is required. Techniques such as random note generators, exact control of parameters, ease of using scales with other than 12 tones, seem to indicate the limitations will be a function of the composer's imagination (and, possibly, his pocketbook) rather than the equipment itself. Also, as composers develop new "instruments" it is an easy task to document these for use by other composers, when one considers the fact that the program is general enough to be used on any IBM 7090 computer and probably also on any large-scale digital computer. The only technical problem involves the digital to analogue converter, and it is hoped that a general solution is forthcoming.

One interesting area which is worth study is that of using the process in reverse. That is, from live or recorded sounds, develop samplings in the computer which can be analysed. The possibility that immediately comes to mind here is of further developing the method in order to also include the field of concrete music. Some work in this area by the original developers of the Bell method has been done[2]. The possibility of subjecting natural sounds to computer modification seems, to this writer, a very exciting possibility.

In the next few months it is hoped that much can be learned of the details of this method, and that it can be put in operation at U.C.L.A. We are fortunate to have Dr. Strang, who worked with the method at the Bell Laboratories, and Dr. Knopoff, who has been doing computer music work for several years at U.C.L.A, both close at hand. Let us hope that once the technical problems are solved much useful music can result by local people.

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