ORCUS TECHNICAL
PUBLICATION-TP- 3003
SECOND EDITION
Manford L. Eaton

ELECTRONIC MUSIC-
A HANDBOOK OF SOUND SYNTHESIS & CONTROL

ORCUS

OPERATIONAL RESEARCH COMPANY UNIVERSAL SYSTEMS

Box 16022 - Kansas City, Mo. 64112 - U.S.A.

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 The circuit diagrams included in this book are for illustration of typical applications and are not intended as constructional information. Although reasonable care has been taken in their preparation to assure their technical correctness, no responsibility is assumed by ORCUS Research for any consequences of their use.

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Note: Electrical, Electronic, and Logic Diagrams have been prepared in conformance with USA Standards Institute Publications Y32.2 and Y32.14

ELECTRONIC MUSIC -

A HANDBOOK OF SOUND SYNTHESIS & CONTROL

M. L. Eaton, ORCUS Research

FIRST EDITION

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SECOND EDITION

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INTRODUCTION

This book contains information which will enable the reader to design and evaluate electronic music systems. It is the first authoritative presentation of the basic theories of sound generation and control. The few books and manuals that do exist deal either with various advanced technical problems or are instruction manuals for specific electronic music instruments.

The material contained in the following pages is drawn from many fields; music, mathematics, acoustics, biology, electronics and psychology. Some of the subject matter of each of these fields is relevant to electronic music but much of it is not. It is the purpose of this book to help the student determine what is of importance. Its purpose is to give the reader the perspective he needs in order to organize his further studies.

If each of the subject areas mentioned above was given a thorough treatment, the present book would consist of many volumes. This however, is not our aim; rather is is to present an outline of the basic areas and technologies which are integral to electronic music. Thus, few of the topics are pursued to great length. The one area which has been given a certain special treatment is electronics. There are at present few books which approach electronics from a systems point of view. For the student studying electronics for the first time or for the student who has been previously acquainted with basic electronics the material presented here will quickly place him in a position to understand the functioning of electronic systems without the necessity for long study of circuit operation and design. This is a quite significant approach to electronics for persons in fields such as electronic music since without knowledge of electronic systems the composer and theorist is at the mercy of commercial producers of electronic music equipment whose instruments are frequently incapable of the desired performance. The systems approach to electronics design is also consonant with the changes which have been occurring in engineering the past ten years. With the advent of integrated analog and digital circuits and the production of discrete component modules and logic cards it is practical to approach electronics at the systems level. The reader is, of course, encouraged to pursue his study of electronics beyond that which is presented in this book. However, the material here should give him the ability to use electronics effectively to accomplish his design aims.

The suggestions for further reading in the back of the book are recommended to the reader with the idea of leading him into basic books of the fields mentioned above and to enable him to make a smooth transition from the material in this book to the more advanced literature. The student will find that the present book is in fact a concentrated source and reference book enabling him to chart his own path through the myriad books and articles which relate to electronic music.

Those workers in this field who have had to sift through hundreds of books and articles to arrive at an understanding of the problems and potentials of electronic music will appreciate the value of the approach exemplified by this book. Armed with the information presented here and specifically with the ability to design and evaluate black box electronic generation and control systems the composer or theorist is in an excellent position to make a worthwhile contribution to the field.' All too frequently, persons interested in electronic music, but without this knowledge, invest in commercial electronic music equipment which is incapable of fulfilling their needs. They find themselves inventing ways of composing music which the instrument is capable of generating instead of specifying the characteristics of a system which will generate the music they wish to compose. This latter is the principal advantage of electronic music; it is possible to generate and control sounds with a high degree of precision and to implement compositional procedures which are quite impossible with conventional instrumentation.

It is a rather unfortunate fact that the role of the music theorist in contemporary musical life is as purveyor of harmonic and compositional rules of a bygone age. It is tragic that the theorist as a designer of possible and desirable software and hardware for musical systems is a virtually non-existent entity in musical life. Electronic music, fortunately, is changing this situation since it is so dependent upon new procedures and equipment that it is virtually impossible to make any significant contribution without an understanding of the various facets of sound synthesis and control.

Until such time as composers and theorists learn to design and evaluate both the procedures and the equipment needed to implement their ideas, electronic music will remain in its present adolescent state. It is our hope that this book may in some small way assist in easing the growing pains.

ORCUS RESEARCH

INTRODUCTION TO THE SECOND EDITION

We at ORCUS Research wish to take this opportunity to thank you, the user of this book for making it the standard English language text on electronic music. At the time of publication of the first edition of Electronic Music-A Handbook Of Sound Synthesis And Control we felt there was a need for a text presenting the more basic principles of sound generation, control and applications of electronics to the art of music. It was also a goal of the book to kindle the students interest in the possibilities for interactions between music, electronics, and the life sciences. The number of requests we have received from individuals and organizations concerning Bio-Music and sensory stimulation systems has induced us to publish much additional information on our research in these areas. The information contained in the present edition provides a basic foundation for study in the more complex interdisciplinary areas of Bio-Music, hallucinogenic, and electronic sensory stimulation systems.

I am personally indebted to my colleagues, fellow consultants and others who have contributed to the success of this publication. I wish to thank Carol Hodges for her considerable efforts in typing and editing the manuscript, Holcomb McKinley for his meticulous proofreading of this edition, Viera Jagosova for preparation of the drawings as well as for the construction of engineering models of sound generation and control circuitry for electronic and Bio-Music experimentation. Special thanks are also due to Ray Stellhorn for suggestions concerning some errors present in the first edition and to Mr. John Tsividis for pointing out some errors and oversights in the electronics section of that edition.

But most of all thank you, the students of the art of music who have responded to the need for increased quality of artistic communication that is possible through the application of electronic techniques to the arts. The electronic interactions between the life sciences such as psychology, psychiatry, neurology, physiology and the technological arts of music, television, film, radio, kinetic art, form an important part of the efforts of the ORCUS Consulting Group. Your interest in these possibilities is greeted with sincere appreciation.

Kansas City, Missouri USA
February, 1971

Manford L. Eaton,
Consultant

ORCUS Research

SOUND, ELECTRONICS & TRANSDUCERS

SOUND, as perceived by the ear, is a VIBRATION of air or, for that matter, of any other gas. Vibrations in the gas are caused by vibrations of solid matter. The solid matter vibrating in the gas may be in the form of an essentially ONE-DIMENSIONAL body such as a string, a TWO-DIMENSIONAL object such as a steel plate, or a THREE-DIMENSIONAL object such as a block of wood. Of course vibrating solid matter may also have such physical structures as vocal cords and the special devices for making gas vibrate called musical instruments. There can be no sound in a vacuum since there is no gas to vibrate.

When the air is made to vibrate back and forth at any rate between 20 times a second and 20,000 times a second, the vibrations of the air are perceived by the ear as sound. The faster the air is made to vibrate, the higher the pitch of the sound. Vibration is a rather vague term, however, and even though most people intuitively feel that they know what vibrations are, it is perhaps better to be more explicit concerning the movement of the air which causes sound. Suppose that we strike a steel plate which is hanging from a string. At the moment of impact the plate bends in a direction away from the force of the blow. If the plate is in a medium of gas, the gas is pushed in front of the plate and compressed. The air in back of the plate can now expand. After the plate has traveled as far as it can in the direction away from the striking force, it begins to return to its original position. Its momentum, however, carries it back past this position and bends the plate in the other direction. The plate continues this back and forth motion until it again comes to rest. This vibration of the plate (or more properly OSCILLATION) alternately compresses and rarefies the air or other gas on each side of the plate, and causes a listener, who is in the same gaseous medium, to perceive sound.

The rate at which an object vibrates (the number of times a second it makes a complete oscillation from one extreme position back to that extreme position) is its FREQUENCY. Each such oscillation will produce one sound wave; that is, one area of compressed air and one area of rarefied air. The number of sound waves emanating from a vibrating body each second is the frequency of the sound.

The tendency of a body to return to an equilibrium position after having been forced or deformed out of position is a function of the ELASTICITY of the body. Actually, such materials as steel and glass are far better examples of elastic substances than is rubber, but they are so hard that deformation is too slight and recovery too rapid to be easily noticeable. Nevertheless, steel or glass balls will bounce if allowed to fall on a hard surface. Substances such as wax, putty, and lead, on the other hand are relatively inelastic and when deformed, as by a fall against a hard floor, will not have much tendency to push back into their original shape.

The speed with which portions of air oscillate back and forth about some point of equilibrium depends in part upon the elasticity of air, just as the speed of oscillation of a spring depends in part upon the elasticity of the metals composing it. Upon the natural speed of oscillation of the air depends, in turn, the velocity of propagation of the sound wave. The velocity of sound in air is equal to the square root of the elasticity of air divided by the density of air. The velocity of sound in air at 0^{o} Centigrade is 331 meters per second or 740 miles per hour. The elasticity of air increases with temperature and so, consequently, does the velocity of sound. The increase in the velocity of sound is roughly half a meter per second for each Centigrade degree rise. Speeds of sound in some common materials are: Water 3240 mph; Hydrogen 2840 mph; Steel 11,200 mph; Glass up to 13,500 mph. In the example of the steel plate mentioned above, the oscillations become weaker after the initial blow until the plate finally comes again to rest. This is properly called DAMPED OSCILLATION. However, it is possible to make the plate continue to oscillate for an indefinitely long time if we apply energy to it in an appropriate fashion.

Suppose that we hired two very nimble elves to help us; we could station one on each side of the steel plate to wait until it came as close to him as it was going to before starting its travel in the opposite direction. At that very instant he could push the plate in the opposite direction very easily. The other elf would do the same thing on the other side of the plate. This, then, would constitute SUSTAINED OSCILLATION for as long as we kept the elves at work. A quite familiar example of sustained oscillation is a children's swing with someone to push it. It is much easier to push the swing when it is at the high point of its travel than it is, either to oppose it when it is traveling toward you, or to push it away faster than it will go naturally. It is not only whole solid bodies that can be made to oscillate; electrons within materials can also be made to exhibit such behavior. When electronic components are arranged so that they make electrons oscillate continuously, the resulting device is called an OSCILLATOR. Arrangements of electronic components which enlarge the numbers of electrons participating in these oscillations are called AMPLIFIERS. There are many different kinds of oscillators and amplifiers, but the above describes the basic job of each. Sometimes one finds that he has too many electrons participating in oscillations and it is desired to exclude some of them. This is done with an arrangement of electronic components, called an ATTENUATOR.

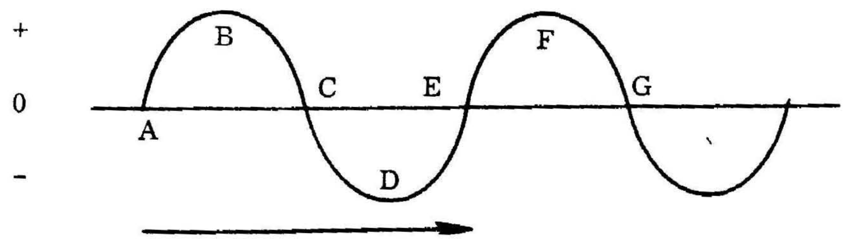
A natural question at this point is, "How does the oscillation of electrons relate to sound?" The answer is that there are devices called TRANSDUCERS. These are devices which transform energy from one form to another. For example, if we want to change sunlight to electricity, we need a transducer called a solar cell. If we wanted to change burning fuel oil into rotational motion, we would want a transducer called a diesel engine. And if we want to change sound vibrations into electronic vibrations, we need a transducer called a MICROPHONE and, conversely, if we had some electronic vibrations which we wanted to change into sound vibrations we would want a LOUDSPEAKER. There are many different types of microphones and loudspeakers, but the basic function of all of them is as described above.

It is difficult to construct transducers which convert energy efficiently and accurately. In any sort of sound system the microphones and speakers are the elements

which limit the performance of the system. Ideally, the form of the oscillations in the air and the oscillations of electrons at the microphone's output should be identical; and the electrical oscillations being fed into the loudspeaker and the sound coming out should also be identical. Any difference between what goes in and what comes out is called DISTORTION.

SOUND WAVES & MUSIC

One of the commonest types of motion in nature is the so-called SINE WAVE. It is also called SIMPLE HARMONIC MOTION. If a pendulum like that in a clock has a pen attached to the bottom of it, such that it will draw a line on a piece of paper placed underneath, and if this paper is made to move at right angles to the direction of the pendulum motion, a sine wave will be drawn on the paper. A sine wave is shown below.



The "0" line running through the middle of the sine wave form indicates the instants at which the pendulum is at its rest position. In the case of oscillating electrons, the intersection of the sine wave and the "0" line indicates the points where the electrons are at rest. Point "B" indicates the time when the maximum number of electrons are flowing through the oscillator in one direction and Point "D", the time when the maximum number of electrons are flowing through the oscillator in the opposite direction. If this electronic energy is converted into acoustical energy the compression and rarefaction of the air at each instant will be the same as the amplitudes of the sine wave shown above. A good example of a musical device which produces a sine wave is the tuning fork. Its output is an almost pure sine wave.

The time between similar points on the wave represents the time for one complete oscillation. For example, from B to F, or C to G, or A to E represents one complete oscillation. The proper name for one oscillation is the HERTZ or its abbreviation Hz. (The name comes from the German researcher, Heinrich Hertz, who discovered electromagnetic waves in the nineteenth century). In older books you will find the term CYCLE instead of Hertz, but the meaning is the same.

In the illustration above, if the time for 1Hz (for example from B to F) is 0.5 second, then the frequency of oscillation is $\frac{1}{0.5}$ or 2Hz. In engineering units, if we have a frequency of 1,000Hz we would write 1KHz. (The K stands for the prefix "Kilo" which means thousands). Thus, 1KHz means 1,000Hz. That is, to

obtain the actual value the decimal point must be moved three places to the right. Standard prefixes and their meanings are presented below.

SYMBOL	ABBREVIATION FOR	Number of places decimal point must be moved; + = move to right/ - means move to left.
p	pico	-12
u	micro	-6
m	milli	-3
K	kilo	+3
M	mega	+6
G	giga	+9

The sine wave gets its name from the fact that a certain mathematical function called the SINE OF AN ANGLE gives the proper value for each instant to describe the wave illustrated above. However, for your present purposes it is not necessary to mathematically generate a sine wave.

The AMPLITUDE of a sine wave is a measure of the height of the peaks (B and F in the above illustration) **and** the valleys (D) of the sine wave. The higher the peaks and the lower the valleys, the greater the amplitude.

Every sine wave has three variable quantities. These are called PARAMETERS. The three parameters of a sine wave are its FREQUENCY, its AMPLITUDE and its DURATION. Duration is merely the length of time in seconds that the sine wave continues at an unchanging amplitude. This limiting of our discussion to unvarying or STEADY-STATE wave forms is traditional in acoustical theory. Problems which arise from this simplification will be discussed in a moment.

HARMONICS is the name given to sine waves which are exact multiples of any given frequency. Musicians frequently call harmonics OVERTONES. However, harmonics is a much more universally understood term. In conventional musical instruments the lowest frequency sine wave component almost always has the greatest amplitude. In traditional acoustical theory this lowest frequency is called the FUNDAMENTAL. The frequency that is twice as high as the fundamental is called the SECOND HARMONIC, and the frequency which is three times higher, the THIRD HARMONIC, etc.

Non-harmonic sine waves are those which are not direct multiples of a given fundamental frequency. For example, if the given frequency is 100Hz, another sine wave of 167Hz is not harmonically related. Generally, sine waves which are not harmonically related to the fundamental frequency are called PARTIALS to distinguish them from harmonics. Any sound which contains more than one sine wave is called a COMPLEX WAVE. Complex waves, in which all of the sine waves are harmonically related to the fundamental are called HARMONIC WAVE FORMS. All sounds in which one or more of the sine waves are not related to the fundamental frequency are called NON-HARMONIC WAVE FORMS.

It might be well to point out here that the sounds utilized in electronic music often have quite different structures from those generated by traditional instruments. The differences between the sounds and the reasons are outlined below. As most music students already know, traditional musical instruments fall into one of four classes: string, woodwind, brass, and percussion instruments. All of these types of instruments are rather limited by the fact that there is relatively little control over the frequency content of the sound. As a matter of fact, to most musicians there is an ideal trumpet sound or ideal violin tone for example. Even if the player wished to change the structure of the sounds he was producing on a particular instrument, he would find it a rather discouraging job. Electronic circuitry, since it does not rely on the physical properties of strings, vibrating air columns, or other physically fixed objects, is capable of producing any sound.

Acoustical musical instruments, that is to say, all instruments which produce sound directly (except percussion instruments) have the following general characteristics:

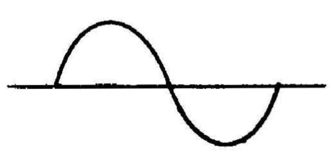
1. The sounds are composed solely of sine wave components.
2. A fundamental sine wave is produced. This fundamental is, as mentioned above, the lowest frequency in the sound and has the greatest amplitude.
3. Harmonics of this fundamental are invariably produced and are always of smaller amplitude than the fundamental.
4. Whenever an acoustical musical instrument produces a steady musical tone, there are a considerable number of frequencies present which are rapidly changing in amplitude, duration, and frequency. (Be sure to note that these components of the sound are not objects of study in traditional musical acoustics.) In our previous description, frequency and amplitude were considered to be constant for the duration of each sound. These varying components of the sound are called TRANSIENTS.

Percussion instruments do not fall into the category of sounds which have sine waves of steady amplitude for most of their duration. Percussion instruments' sounds are primarily composed of transients. As you will probably realize by now, the theory of musical sound; the theory of steady sine waves is quite inadequate to describe the variety of sounds used, even in traditional music. Percussion instruments, which are almost totally outside the realm of traditional musical acoustics, contain a large number of partials which are transient in character. The amplitudes and frequencies of these partials are constantly changing with respect to each other. It is quite interesting to note that non-percussive instruments, which generate only a relatively small amount of transients, are distinguished from one another by listeners by their transient characteristics. For example, if we generate all of the constant frequency and amplitude sine components of trumpet and violin sounds and leave out the transients, most listeners cannot tell them apart. Thus we have a rather difficult situation; traditional musical acoustics deals with collections of sine waves (generally having a simple harmonic structure), which have constant frequency and amplitude values for their entire duration. Yet the most important characteristics of musical sounds do not behave in this manner.

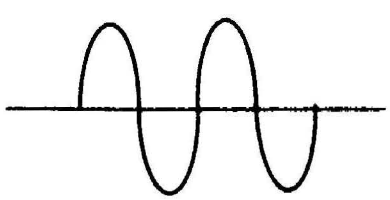
In the nineteenth century Herman Helmholtz developed the theory of musical acoustics in his book, Die Lehre von dem Tonempfindungen (1862) which was translated into English by J. A. Ellis under the title, Sensations of Tone as a Physiological Basis for the Theory of Music. The book is a masterpiece of scientific inquiry. But the measuring instruments of his day were not capable of detecting and measuring transients accurately. Therefore, he tended to neglect their importance and until quite recently, acoustical theory virtually ignored the existence of transient sounds in music. The importance of transients is outlined below:

1. All musical instruments generate some transients which are outside traditional musical and acoustical theory.
2. Percussion instruments, which form an extremely broad and important collection of musical instruments, produce sounds which are almost completely made up of transients.
3. Most listeners identify instruments, including those with only a small transient content by their transients. You can demonstrate the importance of transients in conventional musical instruments very easily. Piano tones have a high transient content when the hammer first strikes the string. If you record a piano tone on tape, and then cut off the first part of the tone, it is quite difficult to tell what instrument is producing the sound. In most conventional musical instruments the transient content is much greater at the beginning and the end of the sound than in any other part.

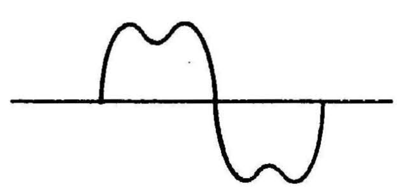
All COMPLEX WAVES (that is, all sounds containing more than one frequency component) are composed of simple harmonic motions, (sine waves). This is true of all sounds (musical or otherwise) without exception. This was shown mathematically in 1807 by the French mathematician and physicist, Jean Baptiste Joseph Fourier. He was able to prove that any wave form can be analyzed into component sine waves. Each of these sine waves differs from the others in amplitude and frequency and duration in such a way that when these sine waves are added together, they will reproduce the original complex wave form. One can see that if it is possible to analyze any complex wave into a group of simple sine waves then, conversely, one can synthesize any complex wave by adding sine waves together. Let's see what happens when we add two sine waves together.



Sine wave #1



Sine wave #2



Resultant

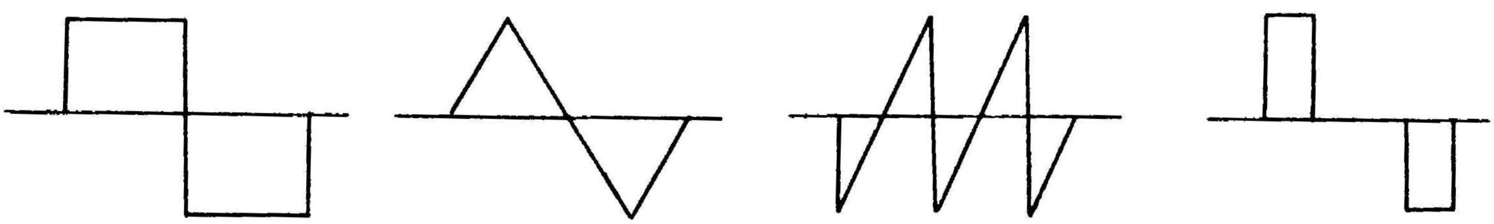
The resultant is merely the sum of the two component sine waves at each instant.

When the two waves start at the same time, as they do in the above illustration, they are said to be in PHASE. When they do not start at the same time they are said to be out of phase. In the above illustration the two sine components are in phase with each other. Suppose, however, that the second wave is out of phase with the first, but that we still have the same two frequencies and amplitudes. Now the resultant wave will look different, but it will sound the same because the ear is not sensitive to phase difference. The possibility of constructing any complex wave form from sine waves is of fundamental importance in electronic music, and many systems for generating complex waves rely upon this discovery by Fourier.

Electronic engineers have known how to construct circuits to generate sine waves for many years. These sine wave oscillators have long been used for a variety of research and testing in the electronic laboratory. One of the earliest approaches to electronic music was simply to borrow sine wave oscillators from the electronics' laboratories and use them to create complex waves.

There are several other special electronics circuits which generate certain special wave forms. It is important to realize that it is impractical to construct a special circuit for each complex wave desired. This would require the design of an astronomical number of circuits. The circuits which we will discuss generate SQUARE WAVE, RAMP, TRIANGULAR, STAIRCASE and PULSE WAVEFORMS. Each of these represents a rather special type of complex wave; they all have rather obvious geometric shapes. It is important to realize that most complex wave forms do not have obvious geometric shapes, and that the geometric shapes of the above-mentioned wave forms are the reason that it is relativeLY EASY to design circuits to generate them directly.

GEOMETRIC WAVEFORMS



SQUARE TRIANGULAR RAMP PULSE

These geometrical wave forms all have the characteristic of being composed of very simple harmonic relationships. For example, the square wave is composed of a fundamental plus all of the odd harmonics of this fundamental sine frequency. One could construct a square wave by generating a fundamental sine wave and generating sine waves at all the odd harmonic frequencies of this fundamental. It is interesting to note here that in order for the fundamental and odd harmonics to look square, the frequency components must have definite phase relationships. But since the ear is insensitive to phase relationships, the resultant wave form may not look square, but yet will sound exactly the same as a square wave. However, since a circuit can be built to generate square waves that require fewer

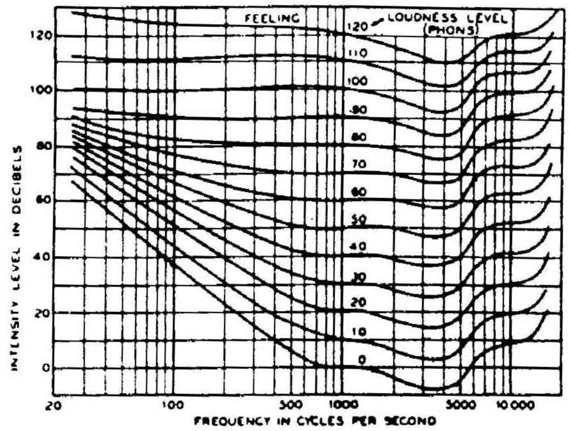
components than many sine wave oscillators, it would be rather foolish to generate square waves by using several sine wave oscillators. The same sort of thing is true of the other special wave forms above.

PARAMETERS OF SOUND (FREQUENCY)

As already indicated, the frequency of a sound (or its representation in electrons in a circuit) is the number of times it vibrates back and forth each second. The more times per second that it vibrates, the higher the pitch of the sound seems to be. When we lower the number of vibrations per second, the sound seems lower in pitch and, below about 20 cycles per second, the vibrations are no longer heard as continuous sound but as separate pulses. It is interesting to note that if we have a complex wave with a fundamental frequency of 100Hz and harmonics extending up to 10,000Hz, then if we change the fundamental frequency of this same complex wave to a frequency of 1,000Hz, the partials which were previously in the 10,000Hz range will now be 100,000Hz, and thus fall outside the range of human hearing. Thus, one cannot generate a complex wave and then produce it at various fundamental frequencies and expect it to have the same overall characteristics in all cases.

It might be well here to mention the range of conventional musical instruments. The fundamental tones on the piano, for example, have a range from about 25Hz to 4900Hz. The harmonics of these fundamental tones extend upwards to infinity. However, we can hear only those that are 20,000Hz or lower. This is the reason the highest notes on the piano have a very thin sound; the harmonic content of these high pitch notes soon goes outside the range of human hearing.

Even though the range of human hearing is considered to extend to 20,000Hz, most people cannot hear a sine wave at this frequency. Only during childhood can one hear frequencies as high as 20,000Hz. By the age 21 or so many peoples' hearing has an upper frequency limit of about 15 or 16,000Hz. By age 65 the upper limit generally decreases to 10 or 12,000Hz. Thus the subtle nuances of complex waves, having partials between 12 and 20,000Hz, are perceptible only to younger listeners. Another interesting and important property of the frequency of sound is that the human ear is not equally sensitive to all frequencies. In order for all frequencies to seem to be of equal loudness, some frequencies must be produced at a much higher amplitude than others. The ear is generally much more sensitive to frequencies in the range of 400Hz to 4KHz.



EQUAL LOUDNESS CURVES

A frequency ratio of 2:1 is a difference in pitch of one octave. The term octave is frequently used in electronics, however the name pitch is a musician's term. There are ten octaves within the range of human hearing:

20Hz to 40Hz
40Hz to 80Hz
80Hz to 160Hz
160Hz to 320Hz
320Hz to 640Hz
640Hz to 1280Hz
1280Hz to 2560Hz
2560Hz to 5120Hz
5120Hz to 10,240Hz
10,240Hz to 20,480Hz

AMPLITUDE

What the musician calls LOUDNESS, the acoustician calls INTENSITY and the electronics engineer calls AMPLITUDE. Furthermore, the units used to measure these parameters differ depending on the source of the sound. The characteristic of the ear, with regard to amplitude of sine waves, is not linear, that is, if the amplitude of the sine wave is doubled the sine wave does not seem twice as loud.

As mentioned earlier, the frequency and the amplitude of sine waves are interrelated. If we produce a sine wave of constant amplitude and vary its frequency, it will seem loud at some frequencies and barely perceptible at others.

DURATION

The duration of a sine wave or complex wave is measured simply in seconds or fractions of a second. As the duration of a sound becomes shorter, the listener begins to lose his ability to determine the frequency of the sound, and as the sounds become even shorter, all frequencies have a click-like sound.

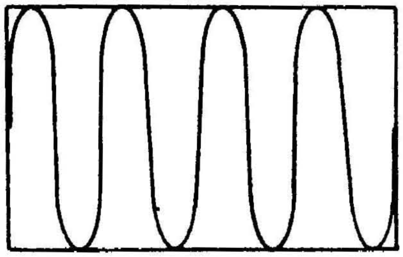
WAVE ENVELOPE

As we have pointed out already, traditional musical-acoustic theory deals only with sound having the following parameters:

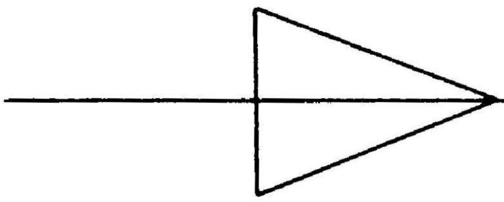
1. Unvarying frequencies for the entire duration of the sound.

2. Unvarying amplitudes of all the partials in the sound.

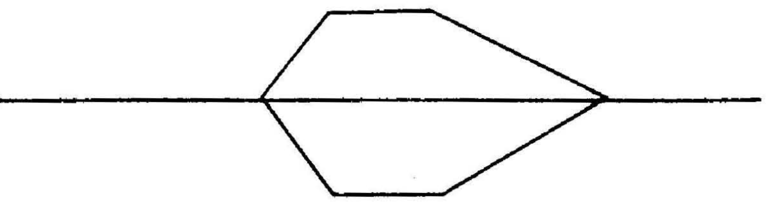
Wave characteristics as treated in traditional a-coustical theory.



These limitations are quite restrictive, and most of the sounds that are encountered, even in traditional music, exceed these limitations. A characteristic of most sounds, musical or otherwise, is that they have an overall change in amplitude during the duration of the sound. This overall shape of the sound is called the WAVE ENVELOPE. There are two particularly important parts of the wave envelope. These are the ATTACK TIME and the DECAY TIME. Both of these terms come from the field of electronic music. Typical, simple wave envelopes are shown below:

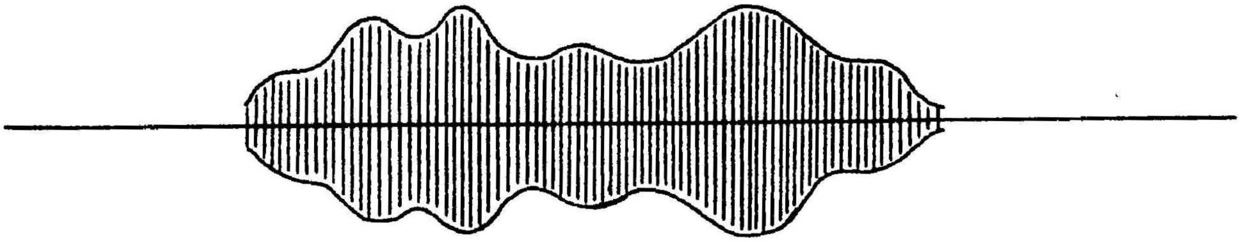


Rapid attack-Slow decay times



Slow attack-Very slow decay times

Wave envelopes are not necessarily of such simple structure as the ones pictured above. For example:



COMPLEX WAVE ENVELOPE

SOME PROPERTIES OF HEARING AND SOUND

The human ear is capable of a rather surprising range of perception. For example, even though the eardrum responds to the overall pattern the brain can, to a certain extent, analyze the sound into its component parts. It is quite easy to identify various instruments playing a traditional musical composition, and it is possible to distinguish the sine wave components of relatively simple electronically produced complex waves. In the case of light, however, the eye cannot distinguish the components of various colors. For example, orange cannot be broken down by the eye into its components, yellow and red.

Furthermore, the ear is also capable of distinguishing sounds in the presence of a great deal of interference. We can easily react to the sound of our own voice across the room in a crowd. Another important ability of the human ear is as a sort of natural sonar device. Experiments by Karl Dallenbach at Cornell University have shown that blind people and people who can normally see, but were blindfolded, were able to detect the presence of a large wood screen which was in their path, several feet before they reached it. If the ears of the subjects were plugged, this ability immediately disappeared.

Another quite interesting property of sound and man's relationship to it is the Doppler effect. Its name comes from the Austrian Physicist, Christian Johann Doppler, who first explained the phenomenon in 1842. The pitch of a sound depends upon the frequency of oscillations within the ear, and thus depends on the number of sound waves entering the ear each second. If the source of sound is not moving, with respect to the listener, the frequency of oscillations in the ear is equal to the frequency of oscillation at the sound source. Suppose, though, that the source of sound is moving toward the listener, more waves are entering the ear each second than are actually being produced. Therefore, the sound seems to be of a higher pitch. If the sound source is moving away from the listener, fewer sound waves will be entering the ear each second than are being produced. In this case, the sound will seem to be of lower pitch. The Doppler effect is used in modern radar systems to detect not only the position, but also the speed of objects.

There are many other interesting characteristics of sound and of hearing. Sound itself is fairly easy to study, but the perception of sound by the human organism is a much more difficult problem. Traditionally, music has not involved itself at all deeply in this area. The only field that takes any serious and continued interest in the perception of sound is acoustics. But, almost all of the classical experiments in acoustics are based upon simplifications and assumptions that make the research results relatively useless from a musical point of view. As an example, when we use a sine wave to construct equal loudness curves the curves assume considerably different shapes than when complex sounds are employed.

One of the fundamental problems of electronic music is that it is suddenly possible to generate and control sound structures with much greater precision than was previously possible. But the theories of musical acoustics are completely inadequate when we try to construct theories of perceived sound at this same level. Until electronic music researchers move outside of the realm of artistic whim and involve themselves in the real problems of electronic music, the compositions will continue to fall into two categories; they tend to be either completely unfounded on any perceptual base, or they tend to be trite electronic realizations of traditional musical concepts.

Some of the problems which are so pressing for electronic music research are enumerated below:

1. The perception of pitch is a partial function of duration but the relationship is quite unclear.

2. At very high frequencies a ratio of 2:1 no longer sounds like an octave, and thus all complex wave structures assume different characteristics at high frequencies.

3. The relationship between intensity and pitch has never been adequately studied. For most listeners, as the intensity becomes greater the pitch seems lower.

4. At high levels of intensity the ear adds distortion to generated sounds.

5. Probably the most difficult of all problems facing music is the differences in ability, training and physiology of various listeners.

6. There is also a psychological problem that the "meanings" of sounds change with time. Thus it is quite difficult to assess the merits of any given approach to sound generation and control.

It is absolutely imperative that scientific and technologically oriented persons, as well as from the field of music, begin thinking in terms of the perception of sound when organized into structures that could legitimately be called music. Compared with what is needed to organize the potentials of electronic sound, present theories of music are ludicrous and, as mentioned before, classical experiments with sound and the perception of sound quite inadequate.

ELECTRONICS

(1) All matter is composed of ATOMS. All matter is electrical in nature. The atom is composed, basically of two parts, the NUCLEUS, which contains NEUTRONS and PROTONS, and a number of ELECTRONS which revolve with great speed around this nucleus. The protons are positively charged. The electrons are charged negatively. The neutrons have no electrical charge. In electronics, it is the electrons which are of primary interest.

(2) Electrons are negatively charged particles. The number of electrons which orbit around the nucleus varies for different substances. For example, the hydrogen atom has one electron, while the aluminum atom has 13 electrons. Under normal conditions there are enough electrons revolving around the atom to neutralize the positive charge of the protons. When the number of electrons and the number of protons are equal, the atom is said to be electrically balanced.

(3) The electrons orbiting around the nucleus are in rings around the nucleus. The electrons orbiting closest to the nucleus are tightly bound to it. The outer rings of electrons are not tightly bound to the nucleus, and through collisions with electrons in the outer ring of other atoms may leave the influence of the nucleus and drift through the space between atoms. When this occurs the drifting electron is called a FREE ELECTRON. If the drifting motion of these free electrons is controlled in such a manner that all free electrons tend to move in the same direction, an electron flow occurs and is called an electric CURRENT.

(4) All substances fall into one of three categories. These are: CONDUCTORS, SEMICONDUCTORS, AND INSULATORS. Into which of these categories a substance falls depends upon its atomic structure; in particular, the number of free electrons which are present in the material. A material which is a good conductor of electricity has a relatively large number of free electrons. Some examples of highly conductive substances are silver, copper, aluminum, and iron. Most metals are good conductors; however, there are some metals (e.g. the ones above) which are better conductors than others.

(5) Insulators are materials whose electrons are all bound to the nucleus. They, therefore, have few free electrons. Materials such as quartz, glass polystyrene, glass and various ceramics are good insulators. Since these materials have very few free electrons, it is very difficult to make an electrical current flow through them. Therefore, these materials are used when it is necessary to prevent electricity from flowing.

(6) Current flow is expressed in terms of the number of electrons which flow past a given point. The COULOMB is the unit of quantity of electricity. One coulomb is approximately equal to 620 million electrons. Usually the rate of current flow is of more interest than the quantity. The unit for expressing this is the AMPERE and is equal to a rate of flow of one coulomb per second. In most electronics circuits with which the student will be associated, the current flow will not be this

great. More frequently he will find the unit MILLIAMPERE; that is, one-thousandth of an ampere, (abbreviated ma). He will also commonly find the unit, MICROAMP; that is, one-millionth of an ampere (abbreviated ua).

(7) The symbol for current flow is "I." Electrons, being negatively charged particles, are attracted by a positive POTENTIAL, and repulsed by a negative potential. If a point is positive, or at a positive potential, it has a deficiency of electrons and therefore has an attraction for electrons. A point having a negative potential is one that has an excess of electrons which it will release to a positive potential in an attempt to establish an equilibrium. A point is only positive with respect to a point of greater electron content. A point is never said to be positive or negative without stating or inferring that it is so in relation to a reference point. The difference in potential between two points is the electrical pressure which causes current to flow and is measured in VOLTS. Two other commonly used terms for this potential difference are VOLTAGE and ELECTROMOTIVE FORCE, abbreviated emf.

If a conductor is connected between two points which have a difference in potential, one positive with respect to the other, a current will flow from the negative point through the conductor to the positive point. The amount of electron flow will depend, in part, upon the magnitude of the difference in potential, or voltage, between the two points.

Since electrons are negatively charged, they are attracted to points which have a deficiency of electrons, that is, points which are positive. Furthermore, electrons are repulsed by points which have an excess of electrons; that is, points which are negatively charged. This is summed up in one of the basic laws of electricity; "Like charges repel; and unlike charges attract." If a point is positive, it has a deficiency of electrons with respect to some other point, and therefore has an attraction for electrons. A point which is negative is one that has an excess of electrons which it will release to the positive point in an attempt to establish equilibrium. It is extremely important that the student realize that a point is only positive with respect to a point of greater electron density. It is meaningless to speak of a single point as being positive. One must always specify that it is positive or negative with respect to some other point. It should also be noted that it is quite possible for a given point to be positive with respect to some point, and at the same time to be negative with respect to some other point.

The symbol for volts is "E." In descriptions of electronic circuits the student will frequently find the author referring, apparently, to a single point as being some voltage. This apparent contradiction is clarified by the following, and the student should constantly keep it in mind. In an electronic circuit all points which are at the same voltage are connected to one common point, often the metal chassis on which the equipment is built. This common point is called GROUND. When a writer refers to a single point as being 18 volts, for example, he means that the point is 18 volts different from ground. The ground voltage is taken to be zero volts. The electronic symbol for ground is:

All points which are connected to it are at ground potential. This ground symbol does not necessarily indicate a physical connection to ground; only that there is no difference between the ground point and physical ground.

(8) When a conductor is placed between two points which have a potential difference as described above, the amount of current which will flow depends on two factors. The first of these is the voltage between the two points. The other factor is the RESISTANCE of the conductor. A material which has many free electrons is, as pointed out in paragraph 4 above, a good conductor. A good conductor has only a small amount of resistance. That is, for a given voltage a large amount of current will flow. If a material has few free electrons, it has a high resistance. The unit of resistance is the OHM. A material is said to have a resistance of one ohm when a voltage of one volt causes a current of one ampere to flow through it. The symbol for ohms is the Greek upper case letter Omega.

(9) POWER is the rate at which electrical energy is used. The unit of power is the WATT. One watt is equal to one volt multiplied by one ampere. The student will commonly find, also, the term milliwatt (one thousandth of a watt). If a material having a very small resistance is connected between two points which have a large potential difference, a large amount of current will flow (Para. 8). Since power is equal to the voltage multiplied by the current, it is easy to see that a large amount of power would be expended. The symbol for power in WATTS is "P".

(10) The relationship between voltage, current, and resistance is expressed by OHMS LAW: "The current, in amperes, in an electrical circuit or any part of a circuit is equal to the voltage, in volts, divided by the resistance in ohms." Using the conventional symbols of:

I = current

E = voltage

R = ohms

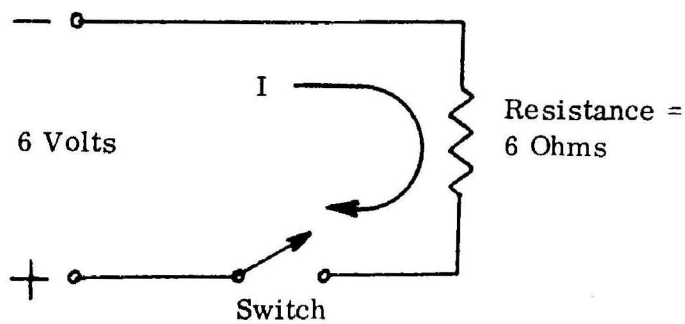
The above can be expressed mathematically as: $I = \frac{E}{R}$

This basic formula can be rearranged by algebraic manipulation to obtain an expression which gives the value of R when E and I are known: $R = \frac{E}{I}$ and also to obtain the value of E when R and I are known: $E = IR$. By further manipulation expressions for power can be obtained. Following is a table which gives the values of R, E, I, or P when two of the other values are known.

All of the formulae given below hold for circuits containing only direct current and voltages. However, only those formulae which do not contain the term P (Power) are applicable to AC circuits. (For AC formulae using P as one of the terms see chart in the back of this book).

Known Values	Formulae for determining unknown values of... I	R	E	P
I&R			IR	I^2R
I&E		$\frac{E}{I}$		EI
I&P		$\frac{P}{I^2}$	$\frac{P}{I}$	
R&E	$\frac{E}{R}$			$\frac{E^2}{R}$
R&P	$\sqrt{\frac{P}{R}}$		$\sqrt{PR}$	
E&P	$\frac{P}{E}$	$\frac{E^2}{P}$		

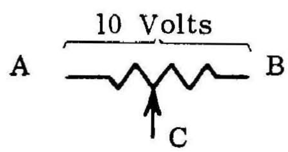
(11) As discussed above, electrons will flow from a point of negative potential to a point which is positive with respect to it, if a path is provided between the two points. The amount of current which will flow is determined by ohms law. This path is known as a circuit. All circuits contain a path or paths through which current flows. This path always has some resistance. If the path from the point of negative potential to the point which is more positive is broken, no current will flow. The circuit drawn below illustrates these points:



With the switch open, as shown, no current will flow since the electron path is broken. When the switch is closed, current will flow; the amount is given by ohms law. The student should note that even though each element in the circuit has resistance, the resistance of the wire itself is usually negligible. The element labeled Resistance in the circuit above is actually an electronic component called a RESISTOR. Resistors are manufactured in a wide variety of values and are used to limit current flow to the desired value. As pointed out (para. 9), if a material having a very small resistance (such as a length of copper wire), is placed between the negative and positive terminals of the VOLTAGE SOURCE, a very large amount of current would flow.

A very common and useful type of resistor is the POTENTIOMETER. The schematic symbol for this element is shown below:

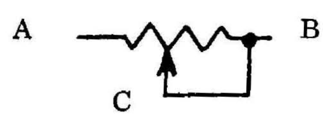
POTENTIOMETER
SYMBOL



It is like two resistors whose resistance ratio can be varied. Suppose that the total resistance from A to B is 50K. The slider (C) can be set at any position along the 50K from A to B. As the slider is moved, the ratio of the resistance from A -- C to that between C -- B changes. The potentiometer is a voltage DIVIDER.

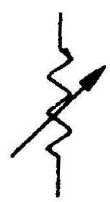
Suppose that the slider is half way between A and B, then the voltage between A and C equals that between C and B; both are 5 volts. As the slider (C) is moved closer to A, the voltage AC becomes smaller while the voltage BC becomes larger.

The most familiar example of a potentiometer is the volume control of a radio. If a potentiometer is connected as shown below it becomes a VARIABLE RESISTOR.



POTENTIOMETER
CONNECTED AS A
VARIABLE RESISTOR

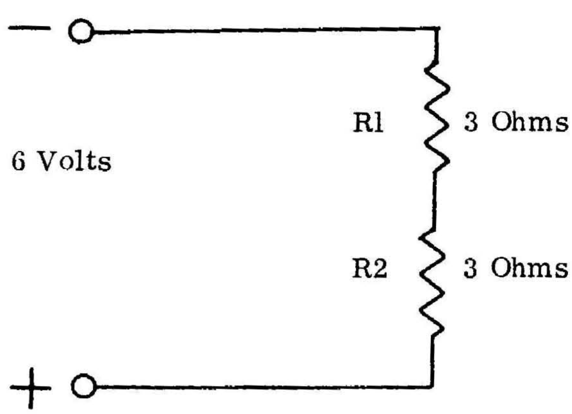
Now the resistance from C to B is bypassed and the total resistance is the resistance between A and C.



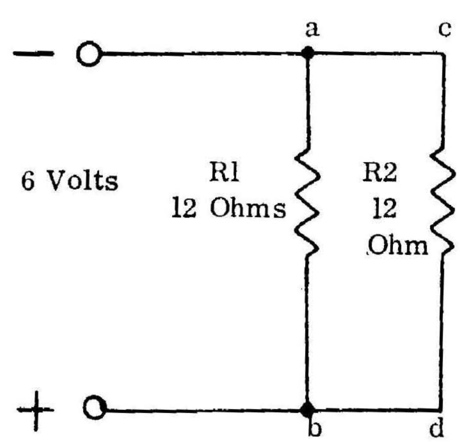
Variable
Resistor
Symbol

The variable resistor is also sometimes drawn as shown above.

(12) There are two basic ways that electronic components are connected in circuits. These are in SERIES and in PARALLEL. The figures below illustrate two Resistors connected in series and two resistors connected in parallel.



(A) Series Connection



(B) Parallel Connection

In a series circuit, the individual resistances are added to find the total resistance to the current flow. The total current flow is given by ohms law as follows:

$$I = \frac{E}{R_t} \qquad \text{where } R_t = R_1 + R_2 \ldots\ldots\ldots\ldots R_n$$

The same amount of current flows in every part of a series circuit. This fact can be used to calculate the voltage across each resistor as follows: $E_{R_1} = IR_1$ and across R_2 one has $E_{R_2} = IR_2$. The sum of the voltages across the resistors in any series circuit is equal to the applied voltage. In the figure (a) above, therefore, the voltage across the two resistors will add up to six volts. The power in the circuit is equal to the voltage multiplied by the current. In the above figure (A) we have:

Known values: Applied voltage = 6 volts
Resistor R1 = 3 ohms
Resistor R2 = 3 ohms

Unknown values: $I = \dfrac{E}{R_1 + R_2} = \dfrac{6}{6} = 1$ Ampere

$$E_{R_1} = IR_1 = 1(3) = 3 \text{ volts}$$

$$E_{R_2} = IR_2 = 1(3) = 3 \text{ volts}$$

$$P = \text{Power} = EI = 6(1) = 6 \text{ watts}$$

In a parallel circuit, the voltage across each branch is the same. Thus in figure (B) above the voltage between points a and b is the same as the voltage between points c and d. This fact can be used to find the current through each resistor as follows:

$$I_{R_1} = \frac{E}{R_1} \qquad I_{R_2} = \frac{E}{R_2}$$

The total current in any parallel circuit is equal to the sum of the currents in each branch. The total resistance in the circuit is equal to the applied voltage divided by the total current: $R_t = \dfrac{E}{I_t}$

In the parallel circuit of figure (b) above we have:

Known values: Applied voltage = 6 volts
Resistor R_1 = 12 ohms
Resistor R_2 = 12 ohms

Unknown values:

$$I_{R_1} = \frac{E}{R1} = \frac{6}{12} = 0.5 \text{ amp}$$

$$I_{R_2} = \frac{E}{R2} = \frac{6}{12} = 0.5 \text{ amp}$$

$$R_t = \frac{E}{I_t} = \frac{6}{0.5 + 0.5} = \frac{6}{1} = 6 \text{ ohms}$$

The power in the circuit is equal to $EI_t = 6\ (1) = 6$ watts.

The student should note that if it is required to solve either a series or a parallel circuit in which the known values are different from those in the examples, this can readily be done by appropriate use of the chart for ohms law (page 16).

(13) All of the material up to this point has been concerned with what is called DIRECT CURRENT. That is, current which flows continuously in one direction. The amount of current which flows in any given circuit flows from the negative terminal of the voltage source of the positive terminal the instant that a path for it is provided. The current flow remains constant until the path is again broken. A direct current is caused by a direct voltage; that is, a voltage which constantly maintains the same potential difference and POLARITY at all times.

(14) Another fundamental type of current flow is called ALTERNATING CURRENT. An alternating current is one which flows first in one direction in the circuit and then reverses polarity and flows in the opposite direction. This alternating current is caused by an ALTERNATING VOLTAGE; that is, a voltage which changes polarity periodically. With an alternating voltage, it is meaningless to indicate positive and negative terminals of the voltage source since they are constantly changing. One fundamental method of generating electricity is, in its most basic form, to rotate a loop of wire about its axis in a uniform magnetic field. In an ac generator this produces an alternating voltage and thus an alternating current. A device such as just described, is an electrical generator. Thus, alternating current is as basic as direct current. Below are listed the basic sources of electricity.

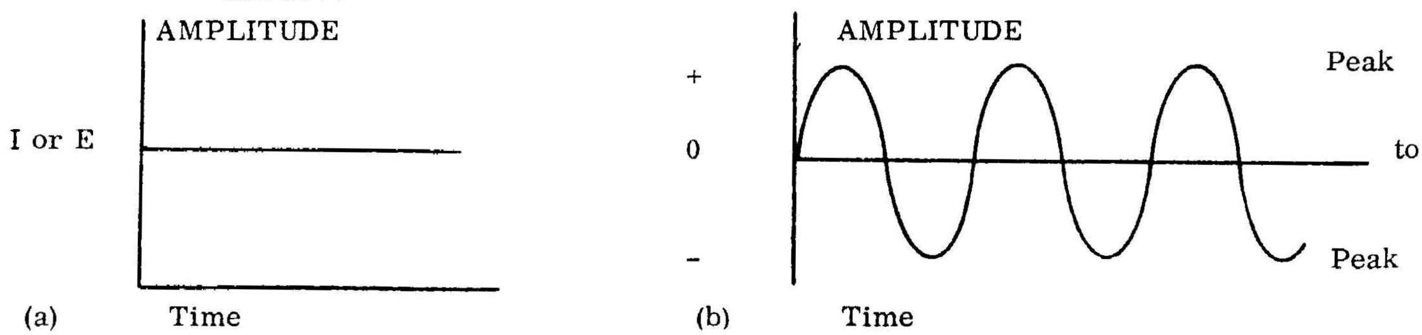
1. BATTERIES: A battery generates electricity by electrochemical action. There are two general types of batteries.

The first is the PRIMARY CELL. It consists of two dissimilar metals, often carbon and zinc which are called the electrodes. The space between

the two metals is filled with ELECTROLYTE, often sulphuric acid. Chemical action with the battery creates a potential difference between the electrodes.

The second type of battery is the SECONDARY CELL. Secondary cells are rechargeable while Primary cells must be discarded when there is no longer a useable potential difference between the electrodes. The usual secondary cell consists of plates of lead peroxide and plates of spongy lead. The spaces between these two electrodes is filled with dilute sulphuric acid, which is the electrolyte. This is the type of battery commonly used to power the ignition and electrical accessories in automobiles. All batteries produce direct voltage and direct current.

2. GENERATORS: There are two types of electrical generators, ac and dc. The generators of almost all power stations generate alternating current. In the United States the current and voltage are sine wave forms at a frequency of 60Hz. In Europe the frequency is 50Hz. In the United States the amplitude of the voltage at the wall tap is 110 volts; in Europe it is 220 volts. Most electronic circuits require direct current for operation. This direct current is sometimes obtained from batteries and sometimes by converting the 110 volt, 60Hz power line to the required DC value with an electronic circuit called a RECTIFIER. Usually a rectifier is one part of a device called a POWER SUPPLY, which supplies a fixed or variable DC voltage. Transistor circuits generally require DC voltages between 1.5V and 35V.



The symbol for current in AC circuits is a lower case i, while for DC circuits the upper case I is used. This is a convention employed to avoid confusion over whether one is discussing ac or dc. This same convention is followed for voltage.

The sine wave shown in the figure above is of the same type as that generated by an oscillator for electronic music purposes. However, its frequency is fixed at 60Hz. The hum that one hears from a loudspeaker which is being driven by an amplifier, whose power is supplied by the AC line, is the residual alternating current which the rectifier-power supply in the amplifier was unable to convert to direct current.

(15) Since alternating current and voltage are continuously changing their amplitude, they cannot be expressed in the same terms as direct current and voltage.

There are two common terms used when referring to values of alternating current or voltage; they are PEAK to PEAK value and the R. M. S. (Root mean square) value. The most commonly used is r. m. s. Most meters for measuring voltage and current are calibrated to read r. m. s. The r. m. s. value of voltage or current is defined as follows: An alternating current has a r. m. s. value of one ampere when it produces the same heating effect in a given resistance as a direct current of one ampere. The peak to peak value of an alternating current or voltage is the difference between the highest and the lowest amplitude of each complete cycle.

Upon occasion it is necessary to calculate the r. m. s. value of an alternating current or voltage when the peak to peak value is known. This can be done by multiplying the peak to peak value by $\frac{.707}{2}$.

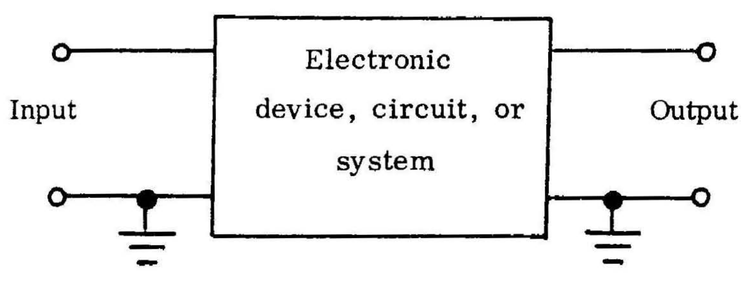
When resistors are the sole elements in an alternating current circuit, their effect on current flow is quite similar to that in a dc circuit. Resistors offer opposition to the flow of current. An ideal resistor has the same effect on AC current flow at all frequencies. In an ac circuit, just as in a dc circuit, the voltage across a resistor is equal to the resistance multiplied by the current. However, in an ac circuit the current is usually expressed in r. m. s. values. If the r. m. s. value of current is used then the voltage is also an r. m. s. value. It is conventional in electronics to assume r. m. s. values unless other values are specifically stated. If an ac voltage or current is given as 15 volts, r. m. s. values are assumed. If peak to peak values are used, this must be indicated. In an ac circuit, the opposition to current flow is called IMPEDANCE. The symbol "Z" is used for this. The reason is that some circuit components (not resistors) offer different amounts of opposition to current flow at various frequencies.

(16) As implied above, there exist several basic electronics components other than resistors, such as CAPACITORS, INDUCTORS, TRANSFORMERS, VACUUM TUBES, TRANSISTORS, DIODES, etc. The justification for not discussing these components in detail is that the majority of electronics circuits with which the student will be associated will appear as boxes whose inputs and outputs are purely-resistive in nature. Since we are making no attempt to impart a knowledge of circuit design or even more than a very elementary understanding of circuit analysis to the student, a coverage of this material in a book of this type would not serve to enhance his knowledge of electronic music. The components mentioned above will be discussed later with a view to acquainting the student with their use as coupling devices, that is as components to help transfer signals from one box to another. In later chapters circuits will be treated as BLACK BOXES. They will be presented as having INPUTS and OUTPUTS, and as having specific TRANSFER FUNCTIONS. (The transfer function of a circuit is a statement of how the output differs from the input). It is hoped that this approach will serve to introduce the student to the basics of the field in the shortest possible time. The student is encouraged to pursue his study of electronics circuits, of course, since it is only through a thorough understanding of circuit operation that he can become adept at specifying his needs and understanding the limitations and possibilities

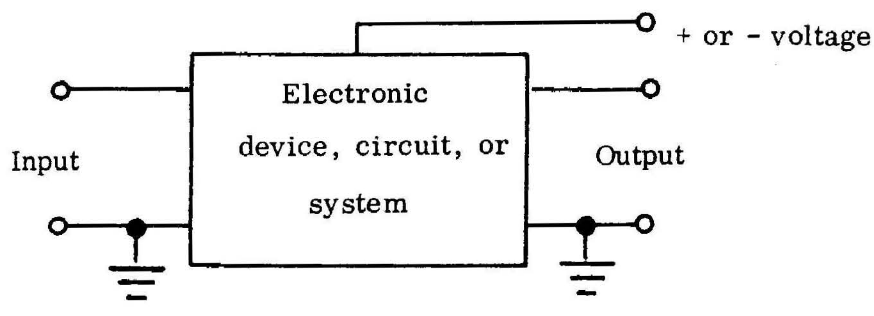
for future development of electronic music. The approach used in the remainder of this book is to describe the nature of the inputs and outputs of circuits which the student will be using; to enable him to determine what effect a given circuit will have on a signal (transfer function); how to interconnect circuits; the types of signals which can be generated; and in general, to orient his thinking in terms of system design and specification. This is the only feasible way to approach the subject, since the music student may have neither the interest nor inclination to become adept in the intricacies of circuit design. If, on the one hand, the student is provided with no reliable information on what operations are electronically feasible, he will not be able to speak coherently of his needs, and will find himself at a loss as to what is possible and what is not. If the music student is required to spend several years becoming familiar with the design and analysis of circuits and systems, he is likely to become so involved with these aspects that he will, in essence, no longer be a composer of music, but solely a designer of instrumentation. This concept of orienting the student toward system specificiation is, furthermore, quite consistent with present changes taking place in electronic engineering. With the advent of INTEGRATED CIRCUITS, circuit design is, to a much higher degree, being performed by the device designer and manufacturer, while the circuit designer of yesterday is today finding it necessary to think in terms of systems.

BLACK BOX SYNTHESIS

Some electronics circuits require the application of a specified direct voltage between one terminal and ground in order to function correctly. These circuits are termed ACTIVE. Other circuits do not require this and are called PASSIVE. Below is the symbol which we will use for passive networks:



Below is the symbol for active networks:

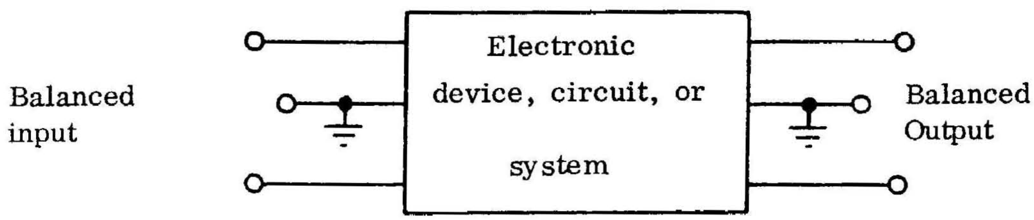


Some networks also have CONTROL TERMINALS which make it possible to control the transfer function of the circuit by applying various control signals. The boxes represent electronic circuits about which only a limited amount of information is known.

SINGLE ENDED & BALANCED CIRCUITS

The two symbols above represent circuits having SINGLE ENDED inputs and outputs. That is, the input has two wires, one of which is at ground potential. The same statement applies to the output.

There is another type of input/output arrangement that is commonly encountered. This is the BALANCED or DIFFERENTIAL input or output. The symbol for this type of circuit is shown below.



BALANCED CIRCUIT

Balanced circuits, like unbalanced ones, may be either passive or active. Combinations of single ended and balanced inputs and outputs are possible and common. For example, there are circuits having a single ended input and a balanced output. And there are circuits having a balanced input and a single ended output. It is important to note, however, that if a circuit has a balanced output it cannot be directly connected to a single ended input. A method for connecting balanced and unbalanced circuits together will be described later.

The reason for the existence of balanced inputs and outputs is as follows: single ended inputs and outputs are susceptible to stray external signals. These external signals can introduce noise and distortion into a single ended circuit. Balanced inputs are much less sensitive to this, since the circuit responds only to the difference between the two ungrounded terminals. This consideration becomes of importance when one is dealing with very low level signals or when a signal must be transmitted by cable over a considerable distance. The commonest sources of interference, which can be greatly reduced with balanced circuitry, are radiation from the 60Hz power line and from electrical devices such as power tools, and auto ignition systems. In electronic music the commonest application of balanced inputs and outputs is in studio tape recording equipment; the input and output impedances are usually 600 ohms balanced. That is, in the above symbol the impedance between positive and ground is 300 ohms, and the impedance between negative and ground is 300 ohms.

Every Black Box has an input resistance and an output resistance. If the input resistance to a circuit or system is low, it is referred to as a CURRENT DEVICE.

You will remember that Ohms Law shows that for a given voltage a greater amount of current will flow through a low resistance than through a high resistance. When the input resistance to a circuit or systems is high, it is referred to as a VOLTAGE DEVICE.

Some applications require a large amount of power. For example, loudspeakers usually require 50 to a few hundred watts. In electronic music this is usually the only part of the system which requires very much power. Most other operations on signals can be done at very low power levels; most circuits operate on fractions of a watt.

Circuits cannot be indiscriminately connected together. When a circuit has nothing connected across its output terminals, it is referred to as an OPEN CIRCUIT. Under this condition, it is obvious that no current can flow out of the circuit. If the output terminals are simply connected together, maximum current will flow. In many cases the current flow will be so large as to destroy the circuit. This condition is called a SHORT CIRCUIT. Some circuits have automatic short circuit protection so that if the output is accidentally shorted, the circuit won't be destroyed. If a resistor is connected across the output terminals of a circuit, it is called the LOAD. The amount of current which will flow depends on the resistance of this load. Now, in practice, this load resistor is actually the input resistance of the circuit to be fed. If the input resistance is very low, a large amount of current will flow. Conversely, if the input resistance of the second circuit is much higher than the output resistance of the first circuit, then a small amount of current will flow.

If the only concern is to transfer a voltage from the first circuit to the second, then the only requirement is that the input resistance to the second circuit be much greater than the output resistance of the first circuit. This is necessary to avoid OVERLOADING the first circuit. Usually the input resistance of the second circuit is made to be at least ten times larger than the output of the first. If it is necessary to transfer as much <u>power</u> as possible from the first circuit to the second, the requirements are more stringent. The expression of these requirements is called MAXIMUM POWER TRANSFER THEOREM. It can be stated as follows: THE MAXIMUM POWER WILL BE ABSORBED FROM ONE NETWORK BY A SECOND JOINED TO ITS OUTPUT TERMINALS, IF THE OUTPUT RESISTANCE OF THE FIRST NETWORK IS EQUAL TO THE INPUT RESISTANCE OF THE SECOND.

In order to intelligently interconnect various circuits, it is necessary to know certain things about their inputs and outputs. The list below represents the important characteristics of any black box. Knowing these values it is possible to determine which circuits can be connected to each other directly and which ones will require some sort of coupling circuit in order to effectively transfer energy from one circuit to another.

1. Input Impedance
2. Output Impedance

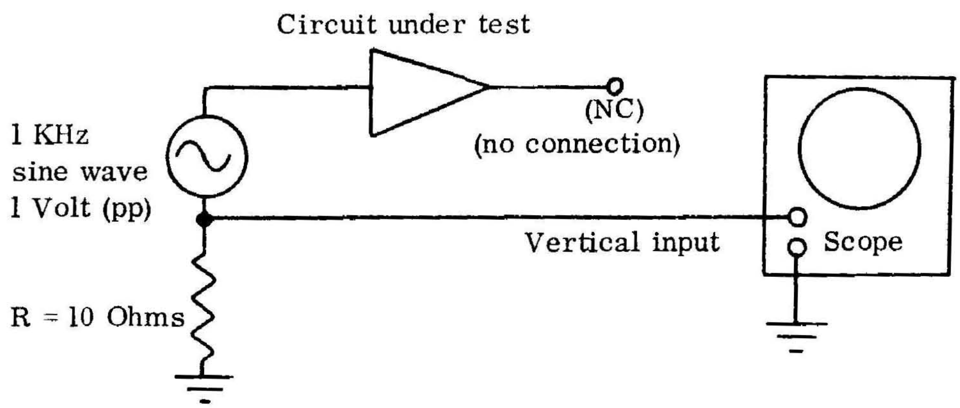
3. Maximum Acceptable Input Signal.
4. Sensitivity (Minimum Acceptable Input Signal).
5. Output Voltage, Current or Power.
6. Transfer Function (The difference between the output and the input).
7. Frequency Response.

Knowing the above information it is possible to design and construct sophisticated electronic systems without the necessity of understanding the intricacies of circuit design. This type of approach is very common in modern electronics because the field has become so large that it is not feasible for one person to be highly skilled in circuit or device design, as well as system design and applications. With the advent of integrated circuitry, complete circuits are bought pre-packaged and, knowing the information in the above list or similar information in the manufacturer's literature, it is possible to design sophisticated electronics systems with only a minimal knowledge of the operation of the devices.

Unfortunately, all of the information on the above list isn't always available. Therefore, it is very useful to know how to make measurements to determine the characteristics of the circuit which you wish to use. Therefore, we will describe simple straight-forward techniques which can be used to measure the above circuit parameters.

INPUT IMPEDANCE

To measure the input impedance of any circuit proceed as follows:

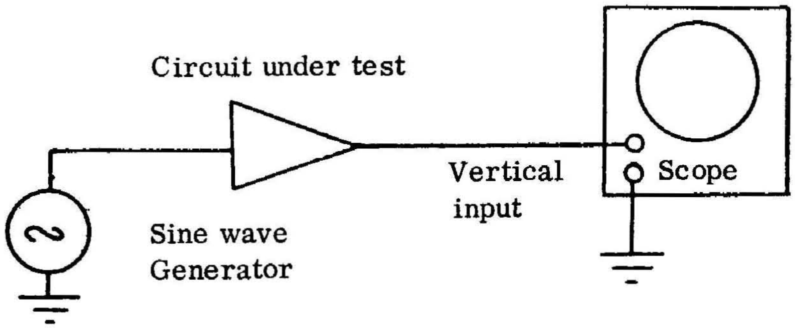


1. Measure the voltage drop across the 10 ohm resistor.
2. Calculate the input current. $i = \frac{e}{R}$
3. Input impedance $= \frac{1 \text{ volt}}{i}$

4. If the voltage drop across R is not large enough to get a convenient reading on the oscilloscope, increase R to 100 ohms, increase the oscillator output voltage or both. (There may be some details of the procedures which are being described here that are unfamiliar to the student at this point in his study of electronics. We will try to clarify these now, but in any case, any book on elementary electronics, (see bibliography), will give further information about the operation of

test equipment and test procedures. What we are attempting to do here is to present very simple algorithms for the measurement of black box parameters.)

MEASUREMENT OF OUTPUT IMPEDANCE



1. Drive the circuit with a signal generator set at 1KHz. (Be sure to keep the input signal small enough so that the output has the proper wave shape. In the case of an amplifier, for example, the output should have the same wave shape as the input.)

2. Measure the output voltage on the oscilloscope.

3. Place a variable resistor across the output of the circuit and change its value until you find the resistance which will cause the output voltage to decrease to one-half the amplitude measured in step 2.

4. Disconnect the variable resistor from the output and measure its resistance. This is the output impedance of the circuit.

MAXIMUM ACCEPTABLE INPUT SIGNAL

This is quite easy to measure, since it is always known what the circuit is supposed to do. The maximum acceptable input signal is the largest signal which can be fed into the circuit while maintaining the output wave form as defined by the transfer function. The simplest example is an amplifier where the output wave form should be the same as the input wave form, except larger. If, for example, we feed a sine wave into an amplifier and we get an output which has the top and bottom of the sine waves clipped, then the input signal is too large.

SENSITIVITY

Every circuit generates some noise internally. This internally generated noise is a limiting factor on how small a signal we may process through the circuit. Suppose, for example, that we have an amplifier which generates 50 uv noise and we try to put a 25 uv signal through the circuit. Now, when we attempt to view the circuit output on an oscilloscope, we will be unable to find the input signal

which has been amplified because the internal noise has also been amplified and our signal is lost in the noise. Suppose, however, that the circuit had 50 uv noise and we put a 100 uv signal into the circuit. We would now be able to observe the signal, but there would be so much noise present that the circuit would probably not be useable.

OUTPUT VOLTAGE, CURRENT OR POWER

This parameter is also quite easy to measure. The circuit is driven from an appropriate signal source and the output voltage measured with a voltmeter or oscilloscope. Once the output voltage and the output impedance are known, it is a simple matter to calculate the current and the power using Ohms Law.

TRANSFER FUNCTION

The transfer function is merely a statement of the operation which the circuit performs on the input signal. As mentioned above, an amplifier makes the wave form larger, and this is its transfer function. $E_o = E_{in}$ (10) for an amplifier which has a gain of 10, for example. It is necessary to know the transfer function of a circuit in order to know what it does. And it is also necessary to know the transfer function in order to determine the maximum acceptable input signal.

FREQUENCY RESPONSE

Some circuits used in audio and electronic music operate over the entire audio range of 20 to 20,000Hz, while others operate only in narrow frequency bands. To determine the frequency characteristics of any circuit, it is only necessary to feed a sine wave into the input of a circuit and observe the output wave form and amplitude, as the frequency of the input sine wave is varied from 20 to 20,000 Hz.

Suppose that we have a number of black boxes and that we know the input/output characteristics of each. And suppose that we wish to interconnect these circuits. We will find that it is sometimes possible to directly connect two circuits together, and that sometimes special interconnection networks must be devised. As we have already mentioned, all circuits fall into one of two categories. They are either voltage or power circuits. Voltage circuits are those having input impedances from about 10KΩ up to infinity and output impedences of about 5,000Ω or less. Power circuits have a Z from a few ohms to infinity and output Z of fractions of an Ω to a few hundred Ω. Usually the only point in a system which requires power circuits is the final output where it is necessary to drive some electro-mechanical device such as a loudspeaker. There are three situations which commonly occur in interconnecting black boxes. A circuit used to drive another is called the SOURCE. The circuit being driven is called the LOAD.

1. A voltage circuit feeding a voltage circuit.
2. A voltage circuit feeding a power circuit.
3. A power circuit feeding a power circuit.

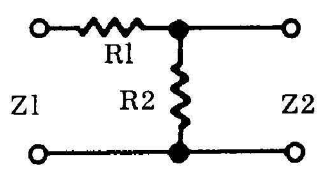
There are other possibilities, but for practical reasons they are rarely encountered. We will now give some rules for interconnecting circuits as mentioned above.

1. In order to drive one voltage circuit with another, it is only necessary that the input impedance of the circuit to be driven is at least 10 times larger than the output impedance of the source.

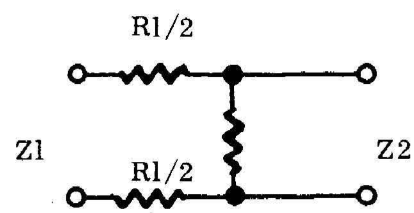
2. When we feed a power circuit from a voltage circuit, the same requirements as mentioned above must be met. The lower the output impedance of the source circuit and the higher the input impedance of the load circuit, the better the situation.

3. When we wish to transmit a maximum amount of power from one circuit to another, it is not enough for the load circuit to have a high input impedence. This is fine for transmitting voltages from one circuit to another, but it is very inefficient for transmitting power. As we have already indicated, the maximum power will be absorbed from one circuit by a second joined to its output, if the output resistance of the first circuit is equal to the input resistance of the second. There are two principal ways of making the two resistances equal; by using MINIMUM LOSS PADS and TRANSFORMERS. (It should be noted here that transformers have several applications in electronic circuit design. However, we will discuss them here only as impedance matching elements.)

<u>MINIMUM LOSS PADS</u>



UNBALANCED



BALANCED

For matching two impedances where Z1 is much greater than Z2.

$$R1 = \sqrt{Z1(Z1 - Z2)} \qquad R2 = \frac{Z1\,Z2}{R1} \qquad \text{db loss} = 20 \log_{10}\left(\sqrt{\frac{Z1}{Z2}} + \sqrt{\frac{Z1}{Z2} - 1}\right)$$

If only the larger impedance is to be matched, use a resistor R_L in series with the smaller impedance such that-- $R_L = Z1 - Z2$

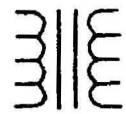
If only the smaller impedance is to be matched, use a resistor R_S in parallel with the larger impedance such that-- $R_S = \dfrac{Z1\,Z2}{Z1 - Z2}$

In both of these last two cases the db loss is equal to-- $20 \log_{10}\sqrt{\dfrac{Z1}{Z2}}$

It can be seen from the above that there is always a considerable power loss when circuits are connected together using minimum loss pads.

TRANSFORMERS AS IMPEDANCE MATCHING ELEMENTS

A transformer is indicated by the symbol shown below:

Primary  Secondary

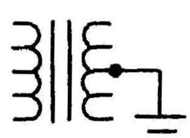
TRANSFORMER SYMBOL

Physically a transformer is two lengths of wire, each of which is wound into the form of a coil. When the two coils are placed close to one another and energy is fed into one of the coils, (called the PRIMARY), energy is transferred electromagnetically to the second coil, (called the SECONDARY). The TURNS RATIO of a transformer is the ratio of the number of turns in the primary to the number of turns in the secondary. A transformer is an impedance changing device. Its impedance ratio is equal to the square of the Turns Ratio. A transformer reflects across the primary whatever load impedance is placed across the secondary terminals, multiplied by the above-mentioned impedance ratio. For example, if a transformer has a Turn Ratio of 10:1 from primary to secondary, its impedance ratio is $\frac{Z_s}{Z_p} = \frac{N_s}{N_p}^2$ or 100:1, primary to secondary, where

Z_S = impedance of secondary
Z_P = impedance of primary
N_S = number of secondary turns
N_P = number of primary turns

Thus, if the impedance connected across the secondary is 8 ohms, the reflected primary impedance will be 100 times larger or 800 ohms.

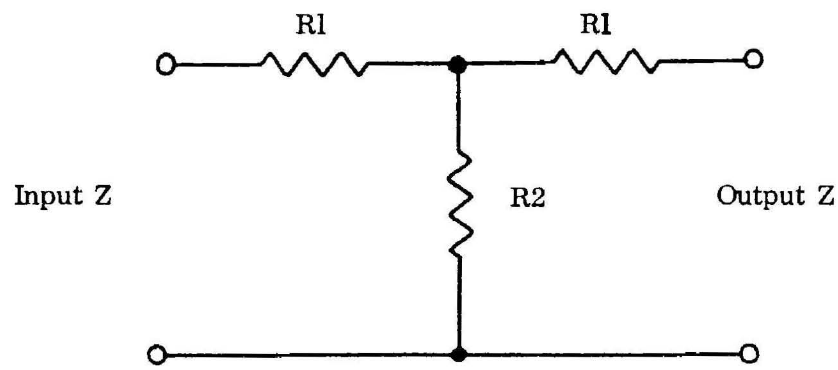
Another quite frequent use of transformers is to connect balanced and unbalanced circuits together. The symbol below represents a transformer with a center tapped secondary. When the center is grounded the signal presented to a circuit connected to the secondary will be balanced around this ground reference.

Primary  Balanced Secondary

Actually these impedances refer to the input and output impedances of the circuits which the transformers are designed to interconnect. For example, if we wish to connect a circuit having an output impedance of 150 ohms to one having an input impedance of 600 ohms, we would look for a transformer with a 150 ohm primary and 600 ohm secondary.

Transformers which operate over a wide frequency range are not inexpensive, and it is necessary to be sure that the frequency characteristics of the transformer chosen are consistent with the frequency characteristics of the circuits to be interconnected.

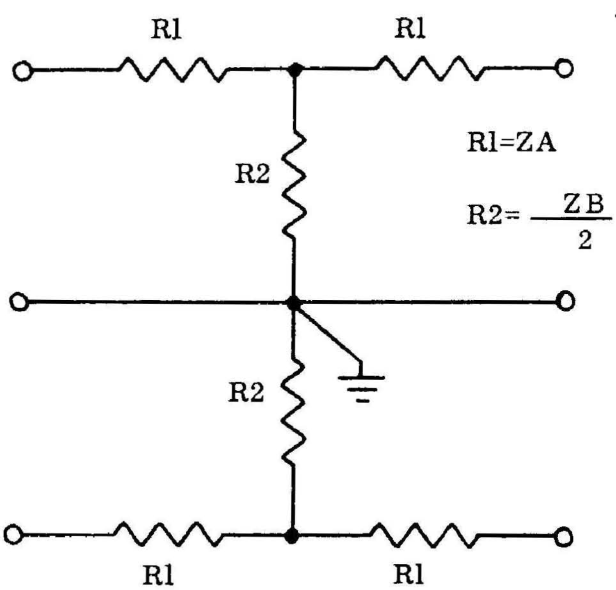
Another frequently encountered problem in interconnecting circuits is that the output signal of one circuit may be too large to be fed into another circuit. When this situation arises, it can be alleviated by the insertion of an ATTENUATOR. There are a number of different types of attenuator networks, but they all do the same thing and are quite similar in performance. Therefore, we will describe here only one of the more common types. The attenuators shown here are for use between equal impedances. If the impedances are unequal, some method such as transformer coupling as described above, must be used to match them. The network shown below is a Single Ended T-Attenuator.



R1=ZA
R2=ZB

SINGLE ENDED T-ATTENUATOR NETWORK

The values for A and B can be found in the chart below for various desired values of attenuation.



BALANCED T-ATTENUATOR NETWORK

	A	B		A	B
1.0	057501	8.6667	**27.0**	.91448	.089515
1.5	086133	5.7619	**27.5**	.91907	.084490
2.0	11462	4.3048	**28.0**	.92343	.079748
2.5	14293	3.4268	**30.0**	93869	.063309
3.0	.17100	2.8385	**32.0**	.95099	.050269
3.5	19879	2.4158	**32.5**	.95367	.047454
4.0	22627	2.0966	**33.0**	.95621	.044797
4.5	25340	1.8465	**34.0**	.96088	039921
5.0	28013	1.6448	**35.0**	.96506	.035577
6.0	33228	1.3386	**36.0**	.96880	.031706
7.0	38247	1.1160	**37.5**	.97368	026675
7.5	40677	1.0258	**38.0**	.97513	.025183
8.0	.43051	.94617	**39.0**	.97781	022443
9.0	47622	.81183	**40.0**	.98020	020002
10.0	51949	.70273	**42.0**	98424	.015888
11.0	.56026	61231	**42.5**	98511	.014999
12.0	.59848	53621	**44.0**	.98746	012620
12.5	61664	50253	**45.0**	.98882	011247
13.0	.63416	47137	**47.5**	.99160	.0084341
14.0	66732	.41560	**48.0**	.99207	.0079623
15.0	.69804	.36727	**50.0**	.99370	.0063246
16.0	72639	.32515	**51.0**	.99438	.0056368
17.0	75246	28826	**52.0**	.99499	.0050238
17.5	.76468	27153	**54.0**	99602	.0039905
18.0	77637	.25584	**55.0**	.99645	.0035566
19.0	.79823	.22726	**56.0**	.99684	.0031698
20.0	81818	.20202	**57.0**	.99718	.0028251
21.0	.83634	17968	**60.0**	.99800	.0020000
22.0	.85282	15987	**64.0**	.99874	.0012619
22.5	.86048	.15083	**65.0**	.99888	.0011247
24.0	88130	.12670	**66.0**	.99900	.0010024
25.0	89352	11283	**68.0**	99920	.0007962
26.0	90455	.10049	**70.0**	.99937	.0006325

MICROPHONES, LOUDSPEAKERS, and TAPE RECORDERS are examples of equipment used in electronic music which are partly electronic and partly mechanical. All electro-mechanical equipment such as this have unique problems, since it is generally much more difficult to achieve the same quality of operation with mechanical devices as it is with electronic ones. Microphones and loudspeakers are examples of TRANSDUCERS; the microphone changes acoustical energy to electrical energy, and the loudspeaker changes electrical energy into acoustical energy. When you look at the specifications for loudspeakers, they are generally extremely vague. This is because they are usually very poor compared to equipment such as amplifiers and oscillators. Furthermore, the performance of a loudspeaker is dependent on the enclosure in which it is housed, and the acoustical characteristics of the room in which it is used. A somewhat similar situation exists with microphones.

If a sound is suddenly increased in magnitude, the listener receives the impression of increased loudness, which is proportional to the logarithm of the ratio of the two acoustical powers. This is a very general rule and it is true for a decrease in power as well as for an increase. The ultimate effect of any change in electrical power in an audio circuit is to produce a change of acoustical power from the loudspeaker. Therefore, it is convenient to adopt a logarithmic basis for indicating changes in electrical power.

Suppose, for example, that an audio amplifier which is driving a loudspeaker is delivering 1 watt, and then suppose it is increased to 2 watts. If we say that the power has increased by 1 watt, it is misleading unless we also state that the original level was also 1 watt. A far more satisfactory way is to state that a 3db rise has occurred. Similarly a decrease from 2 watts to 1 watt is a change of approximately -3db. It is important to realize that decibels express a ratio between two values. When we say that a 3db rise has occurred, it tells us nothing about the absolute values.

A change in level of 1db is barely perceptive to the human ear, while an increase of 2db is only a slight increment. Therefore, most variable attenuators are calibrated in one decibel steps.

In addition to the application of decibels to indicate a change in level at one point, they can also be used to indicate a difference in level between two points, such as the input and output terminals of a circuit. For example, suppose we have an amplifier into which we feed an input power of 6mW. If the output power is now 6W, we have a power gain of 6 divided by .006. By referring to the decibel tables in the back of the book, it can be seen that this is a gain of 30db, this power ratio being irrespective of input or output impedances.

Although the decibel is a unit which is based on the ratio between two powers, it can also be used as an indication of absolute power provided that we know the ref-

erence level (or zero level). The most common standard reference level is 1mW. Most audio equipment considers 1mW in 600 ohms as the ZERO LEVEL. You will find a chart in the back of this book showing decibels above and below a reference level of 1mW into 600 ohms. Thus, in this context, a decibel level of +8, for example, is an absolute value of power.

When an instrument is used to measure power under steady conditions, no particular difficulties are encountered and the scale can be calibrated in decibels. However, when measuring musical material the power is constantly fluctuating, and the indications of the measuring instruments will depend on their speed of response. The VU METER is a standardized instrument which is found on most tape recorders for monitoring and controlling signal levels. The zero level is the maximum input signal level which can be recorded without excessive distortion. It is important to remember that a VU Meter measures complex waves. It is usual to assume that the peak values of the complex waves are 10db above the sine wave peak. Therefore, an amplifier for audio system is tested with a sine wave input at a level 10db higher than the maximum level at which it is intended to be used.

A high quality audio power amplifier might have the following specifications: frequency response, 20Hz to 20,000Hz $\pm \frac{1}{2}$ db; harmonic distortion, .5% at full rated output. A 30 watt power amplifier having these characteristics could easily be purchased for less than $100.00. But a speaker system and enclosure, which would give the same quality, would cost several thousand dollars. The point of this is that it is wise to spend much more money on speaker systems than on amplifiers. Unfortunately, since specifications on amplifiers are easy to obtain, most people spend a great deal of time in deciding which amplifier to buy and then buy speakers almost completely on faith.

When selecting amplifiers and speakers the following points should be followed:

1. Obtain speaker and enclosure specifications and read them carefully.
2. Plan to spend at least five times more money on speakers than on amplifiers.
3. Be sure the speaker system you use is capable of dissipating at least twice as much power as you plan to feed into them from the amplifiers.
4. Select amplifiers so that the amplitude of the sound in the room is adequate when the gain control is set slightly more than halfway between minimum and maximum.

Even after the above points are observed, there is no guarantee that the system will sound good in the auditorium or room in which it is used. It is necessary to install the system and then to take measurements at different points in the room in order to determine its acoustical characteristics. Changing these characteristics involves the selection of sound reflecting and sound absorbing materials. The shape of the room is also quite important, but we will assume here that room dimensions are already established.

Basically, hard materials tend to reflect sound and soft materials, such as carpets and drapes, tend to absorb sound. Therefore, if we find that there is a narrow band of frequencies that are being reflected about the room much more than other frequencies, we may wish to install sound absorbing material to reduce this. Unfortunately, the acoustical treatment of rooms is much more an art than a science, and one soon discovers that there are several very difficult problems involved in acoustically treating rooms.

1. Even when the problem frequency bands are found, it is very difficult to predict just how much of what kind of material in what position will correct the situation.

2. Most people are used to listening to relatively poor sound systems, and tests have shown that when listeners who have never heard good sound systems hear them for the first time, they prefer the more familiar, inferior system. However, after listening to a high quality sound system for several weeks, they almost invariably prefer it to the one of lower quality.

3. The nebulous situation of what happens to sound in a room after it is emitted from the speakers is only matched by the situation that exists when the sound comes through the human ear to the brain. Hearing varies considerably from one person to the next, and frequently an individual will prefer a sound system which has exaggerated frequency characteristics in bands where his ear is less sensitive than average. This is because he now hears sounds that he has never heard before, and concludes that he is listening to a better sound system. And perhaps for this particular listener the sound system is, in fact, better. But for a listener with more average hearing, the sound system will seem to be of erratic performance.

Microphones are also troublesome devices. However, the performance of microphones is generally better than that of speakers. Without going into a long discussion of microphone characteristics, it should suffice to say that condenser microphones are much better than the lower priced and frequently used dynamic microphones. For all professional music recording condenser microphones are a necessity. The following points should be followed in selecting microphones:

1. Purchase condenser microphones.
2. Condenser microphones use battery powered amplifiers built into the microphone housing. These batteries must be replaced at intervals specified by the manufacturer. And if the microphone is not used for several weeks at a time, the batteries should be removed.
3. Plan to spend at least $175.00 per microphone.
4. Insist on microphone specifications which are at least as good as the following:
   a. Frequency response 40Hz - 15KHz $\pm$ 3db.
   b. Distortion less than .75% at 200 dynes/cm^2, over the entire frequency range.

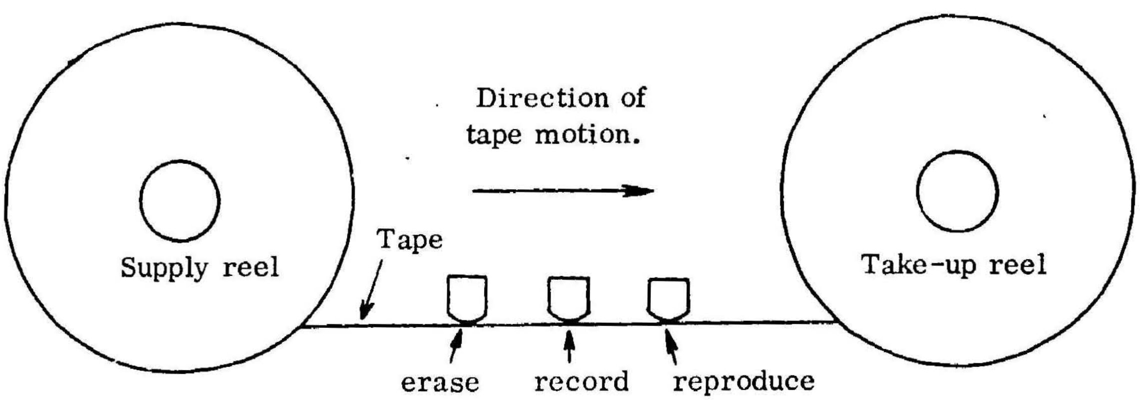
c. Equivalent noise level not more than 25db above threshold of hearing.
d. All professional microphones have low impedence outputs, (less than 200Ω). Don't use high impedance output microphones, since cables longer than 15 feet or so between microphones and amplifiers will result in a high hum level.

We will discuss tape recorders here because they are so basic to most electronic music. The important characteristics for tape recorders are the same as those for amplifiers, loudspeakers, and microphones. But there are, in addition, some other important specifications which are peculiar to the mechanical tape transport mechanism. By far the most important of these - the WOW and FLUTTER specifications. Wow is the irregularities of tape motion caused by imperfections in the transport mechanism. Flutter is basically the same problem, but the irregularities called flutter are at a higher frequency. The effect of too much wow and flutter is that when a steady musical tone, such as a single sine wave or piano tone, is reproduced it appears to waver slowly in frequency (wow) or rapidly (flutter).

Standard professional characteristics for tape recorders are approximately as follows:

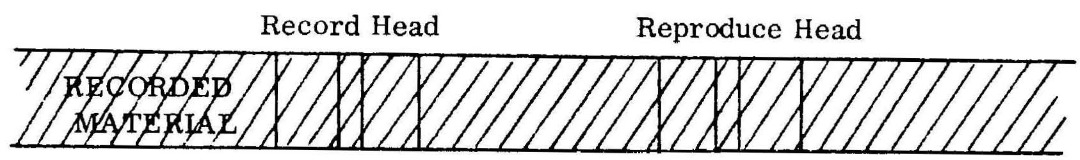
1. Tape speeds for music 7.5ips and 15ips, (generally the higher the tape speed the better the performance characteristics).
2. Frequency response 7.5ips 50 to 12KHz ± 2db
   15ips 40 to 15KHz ± 2db
3. Wow and flutter 7.5ips .15% maximum
   15ips .12% maximum
4. Signal to noise ratio--for a full track machine 50db.
5. Input and output impedance--all professional audio equipment uses standard 600 ohm balanced inputs and outputs. All tape recorders for electronic music use should be of this type.

There are a large number of head arrangements for tape recorders. Below is a basic description of these. First of all, tape recorders for audio use have three HEADS over which the tape passes. These are invariably positioned as shown below:



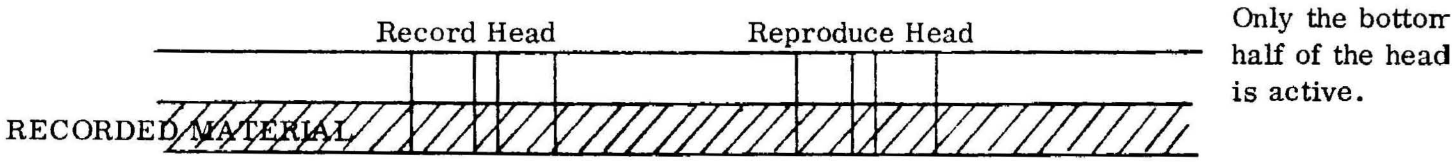
The purpose of the erase head is to erase previously recorded material on the tape. If this head is electronically or mechanically bypassed, any previously recorded material will not be erased. The purpose of the record head is to impress new audio material on the tape, and the purpose of the reproduce head is to play back this recorded material. Since the tape passes over the reproduce head after passing over the record head, a tape can be listened to (MONITORED) as it is being recorded. The significance for simple electronic music composition of these head arrangements will be discussed in more detail when we describe composition methods.

1. Full track (monaural) - In this head arrangement the entire width of the tape is used. For example, if we have four microphones whose outputs are mixed together and fed into a full track tape recorder, the sound will be deposited on the full width of the tape. If we reproduce this tape on a machine having full track heads, and after it has been reproduced flip the reel over to play the other side, we will hear the same material as we heard before except that it will be played backward.



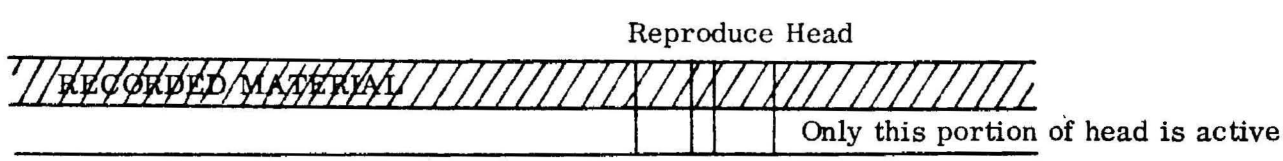
FULL TRACK MONAURAL

2. Half track (monophonic) - In this head arrangement the material is recorded on one-half the width of the tape. Again supposing that we have four microphones whose output is mixed together and fed into a half track recorder, we will have the composite sound on half the width of the tape as shown in the illustration below.



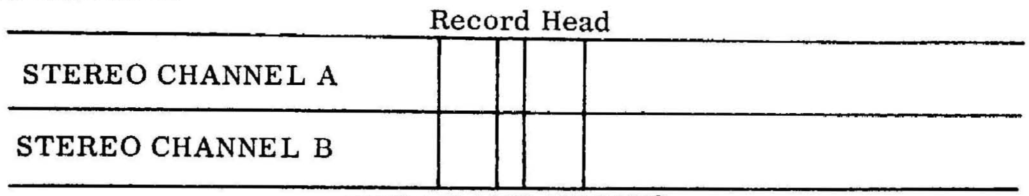
HALF-TRACK MONAURAL

Now if we turn the tape over and attempt to reproduce the tape we will not hear the music backwards as we did in #1 above, we will hear nothing. (See below)



This bottom half of the tape is the other half track which has no material on it. Material can be recorded on it just as it was recorded on the other track. It should be noted here that with half track tapes it is very difficult to edit the tape by cutting out unwanted sounds, because it is impractical to cut out only half the width of the tape. However, it is quite simple to edit full track tapes.

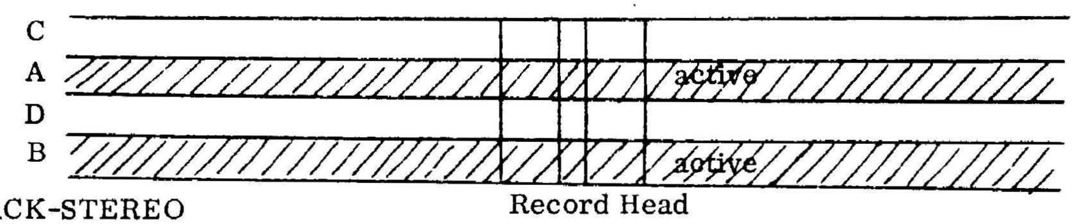
3. Two track (stereo) tapes - In this head arrangement the entire width of the tape is channel A and the other half of the tape is channel B. Suppose that we have two microphones. One of these would be connected to the channel A and the other to the channel B.



TWO TRACK-STEREO

Again we have the situation described in #1 above. If we flip the tape over after it has run through the machine once and play it in the reverse direction, we will hear the music in reverse. However, the sounds which were picked up by the microphone on the left, when the recording was made, will now come out of the speaker on the right and vice versa.

4. Four track (stereo) - Four track recorders are the stereo counterpart of half track monaural recorders. Only one-half the width of the tape is used at a time.



FOUR TRACK-STEREO

Tracks A and B are the two stereo channels that are recorded when the tape is passed through the machine in one direction. When the tape is flipped over and reproduced nothing will be heard, because channels C and D are being reproduced and nothing has yet been recorded on them. The same editing difficulties, which were mentioned above regarding half track tapes, are found when one tries to edit four track stereo tapes.

The basic arrangements discussed above explain the simplest and most common tape recording arrangements. However, frequently in electronic music and popular music recording, multi-channel machines are used. The use of these machines will become more apparent when we discuss electronic music composition techniques.

TECHNIQUES OF GENERATING COMPLEX WAVES

From this point on we will be discussing possible approaches to making music electronically. Up to now we have been speaking of the physical properties of sound and basic principles of electronics. All of these things are in the realm of necessary Physical Laws. The next topics which we will discuss are in the realm of the possible. We will be discussing possible methods of generating complex waves and of making music. It is basic and thus quite important that you

understand the difference between basic necessary principles of sound and electronics, and the possible methods of employing these principles to make music.

It is also important to have in mind the distinction between methods of generating sound and methods for controlling sequences of these sounds. The first problem is one of synthesizing the sound we want, and the second is ordering this material into the sequences that we want. A third, and very important, distinction that must be made between electronic circuitry used as an aid in arriving at compositional decisions, and its use as a work saving device for realizing in sound the compositional decisions of the composer. The point is that electronic circuitry (especially computers) are neither good or bad, human or inhuman. If a composer develops a computer program which is based on the theory of the molecular movements in gasses, and allows no other sounds into his compositions than those which the computer calculated were necessary from his theory, we would have a music much more detached from humanistic concerns than if we had the following situation: Suppose that a composer had some specific ideas on a composition he wished to make, and suppose that he wrote down on paper the necessary specifications for the sounds that he wanted and then programmed a computer to produce these sounds. It is easy to see that this use of a computer is merely that of any other piece of machinery. It saves work, and, in this last example, in no way interferes with the composer's freedom to make decisions.

The foregoing is not intended to be an endorsement or a condemnation of any method of composing electronic music, it is rather to clear up some of the rather muddled thinking that one finds with regard to people's relationship to their technological environment. There are, then, three main areas of activity within electronic music which are listed below. It is important to realize the distinction between them.

1. Methods of generating sounds.
2. Methods for controlling this sound.
3. Ways of making compositional decisions.

The first thing that we will do is discuss methods of generating complex waves. One of the oldest and simplest methods is to use several sine wave oscillators whose frequencies and amplitudes can be varied. The outputs of these oscillators are mixed together in a circuit called a MIXER and can then be recorded on tape. This is a basic COMPLEX WAVE SYNTHESIZER. The next step in making our synthesizer more versatile is to add square wave generators, sawtooth generators, and the other special function generators described earlier. The next thing we might do with our synthesizer is to add a keyboard. This would allow us to use the perhaps all too familiar piano type keyboard to set the oscillator outputs instead of turning dials. There are several other gadgets that we might add to our synthesizer which would make it seem even more useful. Complex Wave Synthesizers, such as the one described above, are quite useful in some ways and quite limiting in others.

1. The primary advantages of synthesizers such as this are that they are comparatively inexpensive.
2. They can usually be expanded at a later date with little difficulty.
3. They usually call for methods of operation that can be easily understood by someone unskilled in acoustics, electronics or computer technology.

The disadvantages of synthesizers, however, are rather formidable. They include the following:

1. The complexity of the waves which can be generated is directly related to the number of oscillators in the synthesizer. (Supposing that we have three sine wave oscillators and one of each of the special purpose oscillators mentioned earlier. In this case our music will tend to sound like it was produced on a synthesizer with just that collection of oscillators.)

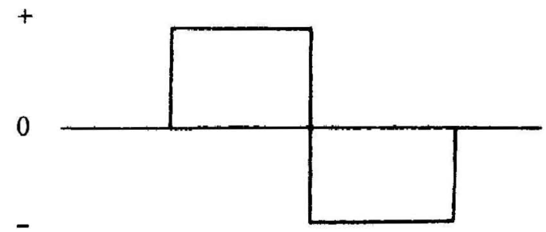
2. The methods used for controlling the sequences of sounds from this type of synthesizer are generally quite crude. They depend either on a keyboard or on splicing sections of tape together.

Despite the fact that the list of disadvantages is shorter than the list of advantages, this type of equipment can be quite frustrating if you have in mind what you want compositionally, because it will be possible to get some of the sounds desired but not others. As a matter of fact, in the final analysis the sound generation capabilities of such synthesizers are quite restrictive. The use of a keyboard to control complex sound sequences may be a delight to frustrated pianists, but it is a rather awkward method. The procedure of tape splicing, as anyone who has ever spliced thousands of sections of tape by hand will tell you, has serious drawbacks. It is feasible to control synthesizers by other methods. For example, a punched paper tape can be prepared which contains data that will program the synthesizer to generate (within its capabilities) the sounds required by the composer. Unfortunately, this type of equipment becomes very expensive as the synthesizer becomes more complicated. By the time we have enough oscillators in the synthesizer to satisfy our needs, the programming equipment becomes expensive and unwieldy.

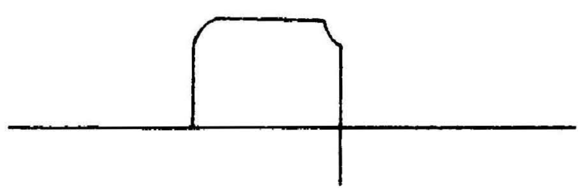
We will now discuss some of the circuitry which is used to generate electronic music in a complex wave synthesizer. The heart of any synthesizer is its oscillators; types which will generate sine, square, and sometimes some of the other special wave forms, such as triangular, ramps, pulses and staircases. There are a number of important specifications on oscillator performance with which you should be familiar, and we will now discuss these in detail. The first is distortion in sine waves. No oscillator generates an absolutely perfect sine wave. Most also generate somewhere between 10% and .01% distortion. The amount of distortion that can be tolerated without being perceptible depends on several variables; for example, the frequency of the sine wave in question, its amplitude, its duration, whether there are other sine waves being generated by other oscil-

lators at the same time, the frequency, amplitude and duration of these waves, whether the listener knows what a poor sine wave sounds like, how good the listener's hearing is, etc. Obviously, there are many factors influencing the amount of distortion which the oscillator can generate without the listener being aware of it. As a general rule though, sine wave oscillators for electronic music use should have less than 1% distortion.

The important characteristics of the other special wave forms that we have mentioned earlier are as follows: An ideal square wave should have absolutely vertical sides. In reality, though, it takes the circuit some time to get from the zero level to the maximum positive level, or from the zero level to the maximum negative level. A perfect square wave should be absolutely square as shown below:

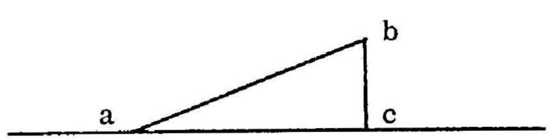


In reality, however, square waves generally look like:



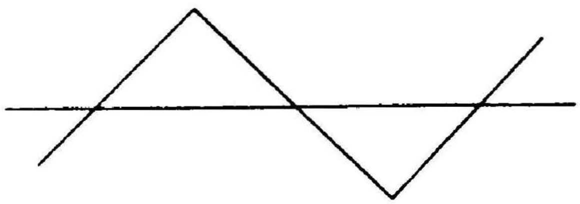
The time required for the square wave to get from zero to 90% of the final positive value is called its RISE TIME. The time required for it to get from the maximum positive level to within 10% of its zero level is called its FALL TIME. (These deviations from a perfect square wave indicate the lack of appropriate harmonics in certain regions. For example, when the front of the square wave is round at the top it indicates that the square wave does not have as much high frequency content as it should.) The measurement of rise time and fall time is applicable to any wave form which ideally should have vertical sides.

Relevant specifications of ramp wave forms are LINEARITY and fall time.

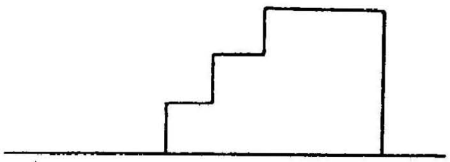


RAMP (sometimes called sawtooth wave)

Linearity merely refers to the straightness of the portion from a to b. If it bulges or sags it is non-linear. The fall time at the end of the ramp is specified in the same way as the fall time for square waves. The other special wave forms are specified quite similarly to these that we have described above.



TRIANGULAR



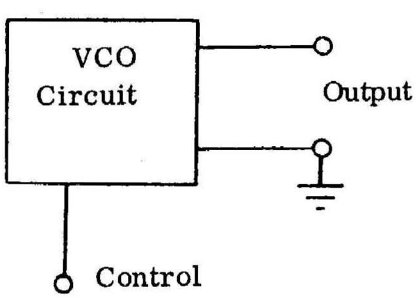
STAIRCASE

The specifications that we have described above vary depending upon the wave form generated. The specifications below are important for any kind of oscillator, no matter what type of wave form it generates.

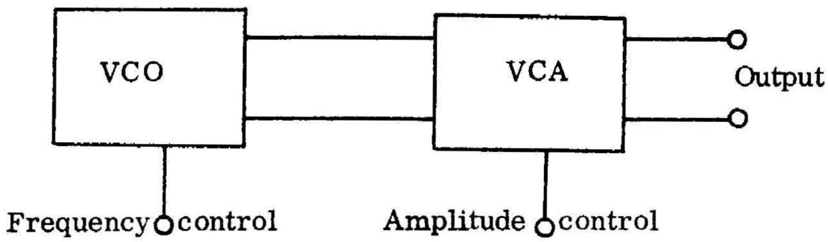
An important characteristic of any oscillator is its frequency stability versus supply voltage changes, and its frequency stability versus changes in ambient temperature. Frequently the power that is supplied to the oscillator varies, and when it does the output frequency changes and the markings on the dial become more and more erroneous. The output frequency also changes when the temperature of the air around the oscillator changes. Under laboratory conditions, where the temperature varies only a few degrees, this is usually not a problem.

The oscillators that we have been discussing are controlled manually. They have a dial which is turned to change the frequency, and another to change the amplitude. If we wish to have a synthesizer which can be programmed by electrical signals, we need some way of controlling the frequency and the amplitude of the oscillator output. Thus we need what is called a VOLTAGE CONTROLLED OSCILLATOR (VCO).

The Voltage Controlled Oscillator is a circuit quite similar to a manually controlled unit. However, the frequency and/or amplitude of the output can be controlled by externally applied direct voltages. The frequency linearity of the control voltage versus oscillator output frequency is the most important parameter of the VCO. A good voltage controlled oscillator has a linearity of .1% or better. The symbol for a voltage controlled oscillator is shown below:



In order to make the output amplitude variable, the usual procedure is to take a voltage controlled oscillator and attach a voltage controlled amplifier to its output.

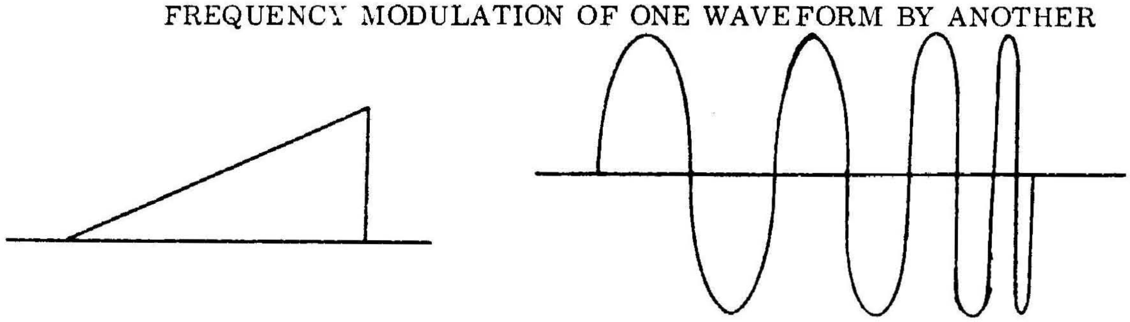


We now have a very simple system for generating electronic music by means of external electronic signals.

MODULATION

Suppose that we apply a direct voltage to the frequency control input of a VCO. As we increase the direct voltage, the frequency increases. Suppose that we have a circuit which will automatically make the control voltage input go from zero to some specified voltage and then drop back to zero. We would, thus, make the VCO generate a sine wave of rapidly ascending frequency. This is the situation we would have if, instead of applying a steady direct voltage to the VCO control input, we applied a ramp wave form to its input. This is shown in the illustration below:

FREQUENCY MODULATION OF ONE WAVEFORM BY ANOTHER

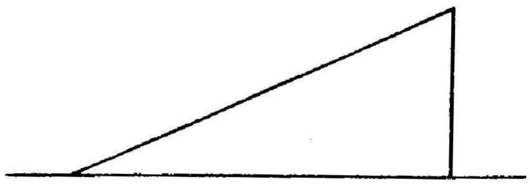


Frequency control voltage

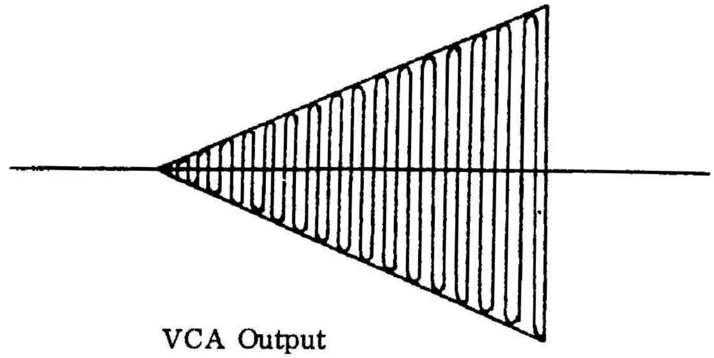
VCO Output

We could, of course, instead of applying only one ramp, apply a continuous series of ramps to the control input. We could also apply a ramp to the voltage controlled amplifier control input. In this case we would have the following wave forms:

AMPLITUDE MODULATION OF ONE WAVEFORM BY ANOTHER



VCA Input control voltage

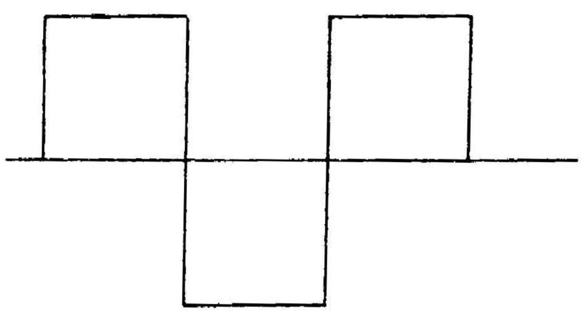


VCA Output

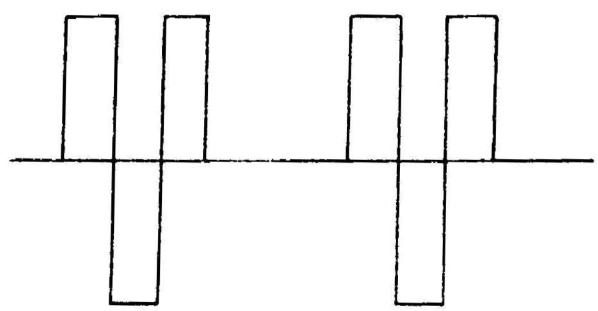
(The above assumes that the VCO frequency control is DC)

(The above assumes that the VCO frequency control input is a steady direct voltage)

You will probably be realizing by now that it is possible to apply one wave form to the control inputs and thus change the character of the output. There is no limit to how far this procedure can be extended. This process is called MODULATION of one wave form by another. For example, we can modulate the amplitude of a square wave by another square wave.

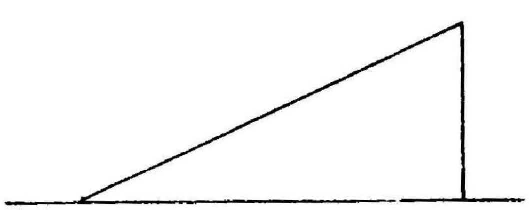


VCO Control Input

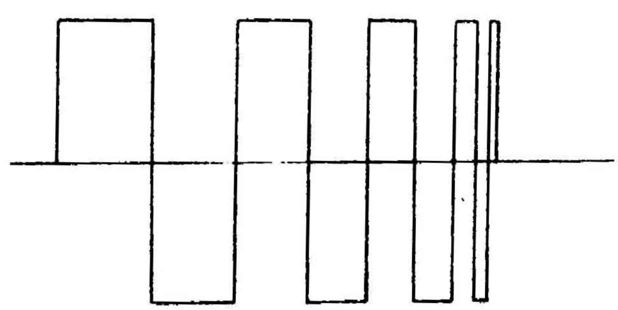


VCO (square wave) Output

Or we could modulate the frequency of a square wave oscillator with a ramp wave form, and the amplitude with a sine wave. (See below).

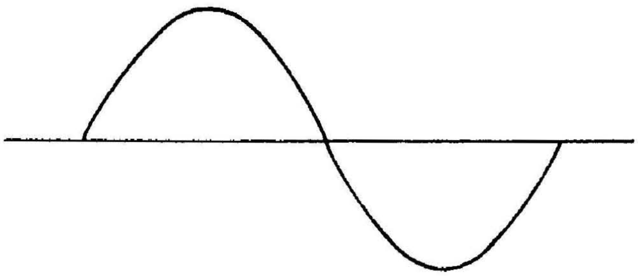


VCO frequency control input

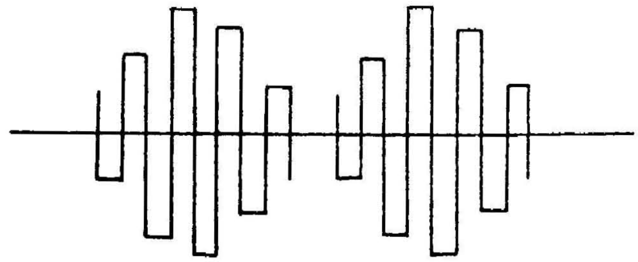


VCO (square wave) Output

And at the terminals of the voltage controlled amplifier we will have the following:



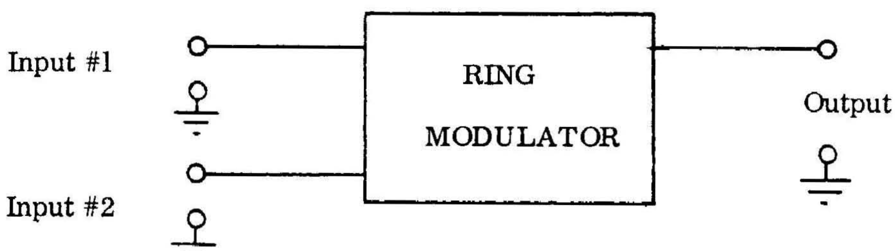
VCA Input Control



VCA Output
(square wave VCA)

This is only a small sample of the kind of thing that can be done with voltage controlled oscillators. And this is one of the reasons for their popularity with electronic music composers.

In addition to the oscillators described above, there are other miscellaneous circuits employed in some synthesizers. One of these is the RING MODULATOR. The symbol for a ring modulator is shown below:



These devices operate as follows: Whenever sine waves are put into the two inputs of the ring modulator, the output will contain ~~not~~ only ~~the original two sine waves but will also contain a sine wave having~~ a frequency equal to the difference between the two input sine waves, and another frequency which is the sum of the two input frequencies. For example, suppose input 1 equals 500Hz and input 2 equals 750Hz. In this case the difference frequency will be 250Hz, and the sum frequency will be 1250Hz. Suppose now that we have the following input frequencies: input 1 equals 500Hz and input 2 equals 1,000Hz. Now the difference frequency is equal to 500Hz. This is the same as input 1 and, therefore, supposing that the two input frequencies had an equal amplitude, the output will be reinforced at 500Hz. In this example the sum frequency will equal 1500Hz.

REVERBERATION

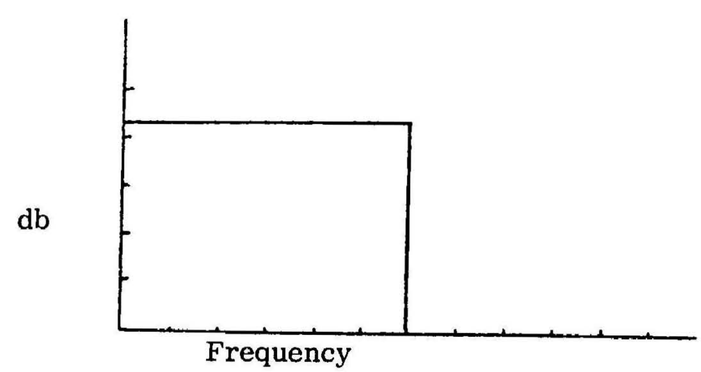
When a sound comes from a loudspeaker the wave travels through the air until it meets some obstacle. If the obstacle is of small dimensions, the sound tends to

go around it. However, suppose that the speaker is mounted in one wall of a room. When the sound wave reaches the other side of the room, it is reflected back in the opposite direction. When it gets back to the first wall it is reflected once again, and this process continues until all of the energy of the sound wave is dissipated. Of course, in listening to actual music, this effect is much more complicated by the fact that during the time that the first sound wave is being reflected back and forth the speaker will be emitting new sounds. The situation is also more complex in that the sound spreads out after it is emitted from the speaker, and does not simply travel back and forth between the two walls. It is reflected from the ceiling, floor, and other walls at various angles. This bouncing back and forth is called REVERBERATION. In the case described above, the time for the sound to cross the room and return is very small. However, there is a well-known situation where we have reverberation on a large scale, the echo. As everyone knows, when in the mountains it is frequently possible to make a sound which will be echoed back a few seconds later. The reason for the long delay is that the reflective surface (the mountain across the valley) is a sizable distance from the source of the sound.

The type of reverberation described above is acoustical. That is, it happens in a medium of air. It is also possible to achieve reverberation electronically. There are a number of devices available commercially to accomplish this; among them the variable reverberation unit for car stereo systems. A variable reverberator is merely one in which the reverberation time and intensity can be controlled over a wide range.

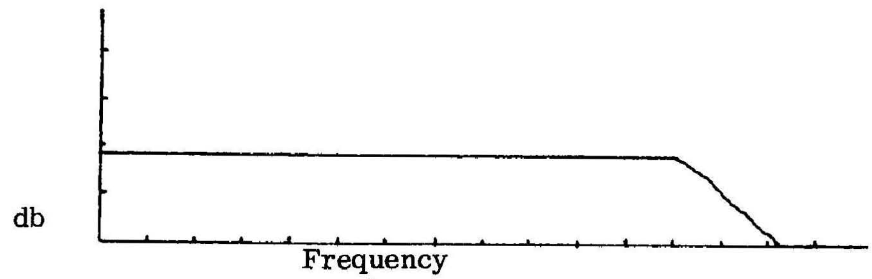
FILTERS

Filters are quite common devices in electronics and electronic music. Basically, a filter is a device which allows certain frequencies to pass through unchanged, and completely blocks other frequencies. An ideal filter has a very sharp CUT-OFF. This means that the frequency at which the filter begins completely blocking the signals, and the frequency at which the filter passed the signals unchanged, are very close together. See the drawing below.



IDEAL FILTER

In reality, simple filters don't operate this way. Their cutoff is rather gradual. The cutoff frequency of a filter is usually taken to be that frequency at which the amplitude has fallen 3db.

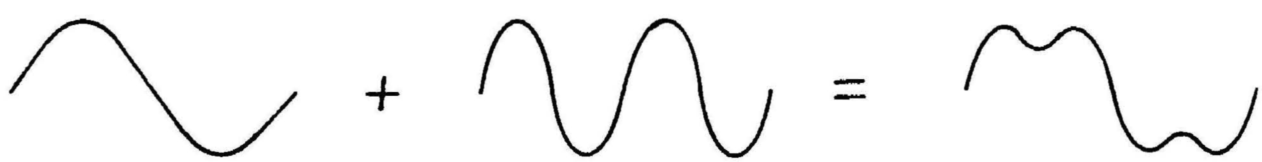


There are four basic types of filter action:

1. Low Pass
2. High Pass
3. Band Pass
4. Band Stop

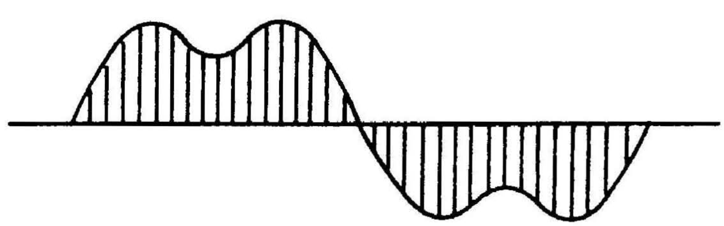
PULSE SEQUENCE GENERATION

As mentioned earlier, Fourier demonstrated that all sound can be analyzed into component sine waves of various frequencies, amplitudes and durations. It is, therefore, also possible to synthesize complex waves from simple sine waves. When we put two sine waves together, as shown below, the result looks like a modified sine wave.



The position of the notch on the wave form indicates a particular phase relationship between the component sine waves. However, since the ear is insensitive to phase differences between component sine waves in a complex wave form, the actual position of the notch is of no practical importance. For this reason it is not necessary to generate a wave form that looks like this resultant wave form. It is only necessary that the resultant wave form be analyzable into the two component sine waves.

There are a number of possible methods of simulating complex wave forms by generating pulses of various heights as shown below:



Each of the vertical segments in the drawing represents a pulse of the correct height to approximate the correct value of the wave at that particular point. The narrower the pulses, and thus the more of them for each half cycle, the greater the accuracy of the approximation. The vertical portions of these pulses represent very high frequency components; frequencies outside the audio spectrum. These components are easily filtered out. This pulse series generation can be accomplished in several ways. The first one which we will discuss was developed at Bell Telephone Laboratories in the late 1950's. A computer program was developed for this called Music IVB. The purpose of the program is to translate the composer's numerical instructions regarding frequency, amplitude, duration and wave envelope into numbers which represent the height and sequence of pulses. Basic operation of the system from the composer's standpoint is as follows: He writes down numbers corresponding to the sound parameters mentioned above. This data is then sent to a data processing department. The Music IVB program is entered into the computer. The purpose of the IVB program is to enable the computer to know what to do with the numerical data which is to be processed. After the program has been fed into the computer, the composer's numerical data, which has been put into some such form as punched cards or digital magnetic tape, is entered into the machine. The computer then proceeds to calculate mathematically the required height of each pulse. After these calculations the appropriate pulse height series' are generated with equipment which converts the numerical value of each pulse to an actual pulse of the correct height. After the high frequency components are filtered out, the signal is fed to a conventional audio tape recorder. This tape is then returned to the composer along with the original numerical data. The principle advantage of this synthesis system is its generality; that is, its ability to generate virtually any sound. Its principal disadvantages are discussed below.

1. A very high speed computer must be used. In order for a series of pulse heights to closely approximate a complex wave, we need about 1,000 pulses for each half cycle. Therefore, to generate a complex wave at 20KHz we will need to calculate pulse heights at a rate 1,000 times higher, or 20,000,000 pulses per second. Since the computer must perform several operations in order to calculate a single pulse height, it must operate at an even higher speed. It is not unusual for three minutes of music to require 9 or 10 hours of computer calculation time.

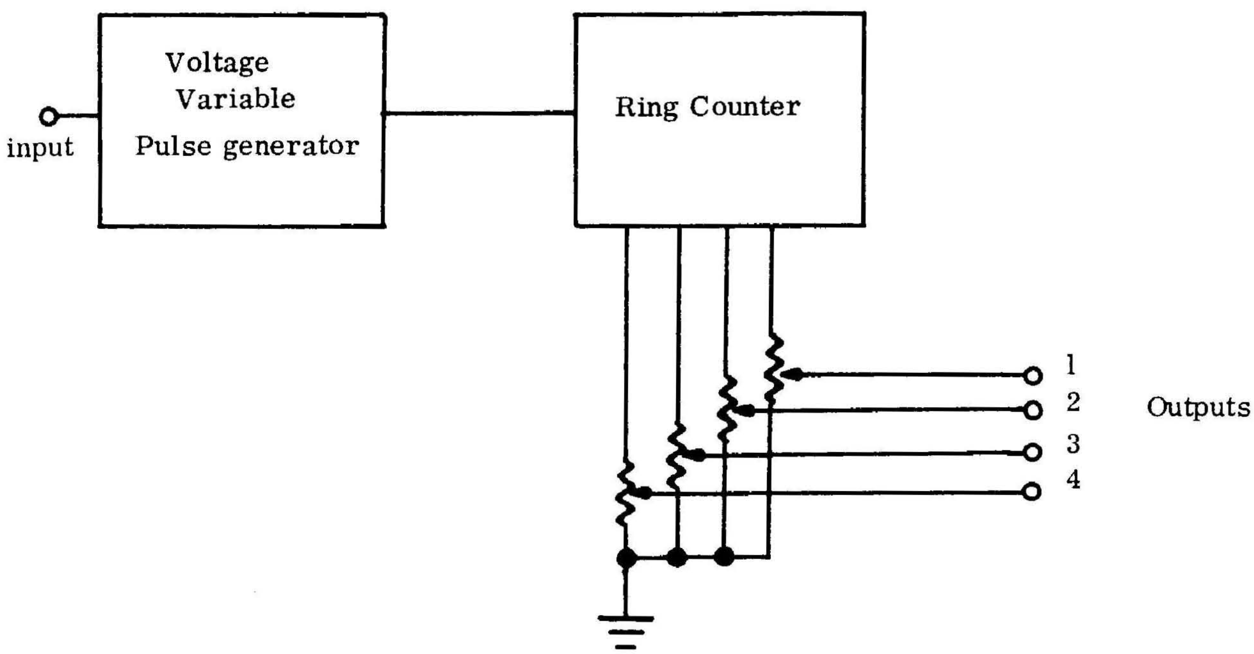
2. There is no way for the composer to actually hear the sound which will result from the numerical data without sending it to data processing and waiting for the return of the audio tape. It is, therefore, sometimes several days between the time the composer writes down the data and when he hears the result in sound.

3. Many composers do not have access to computers.

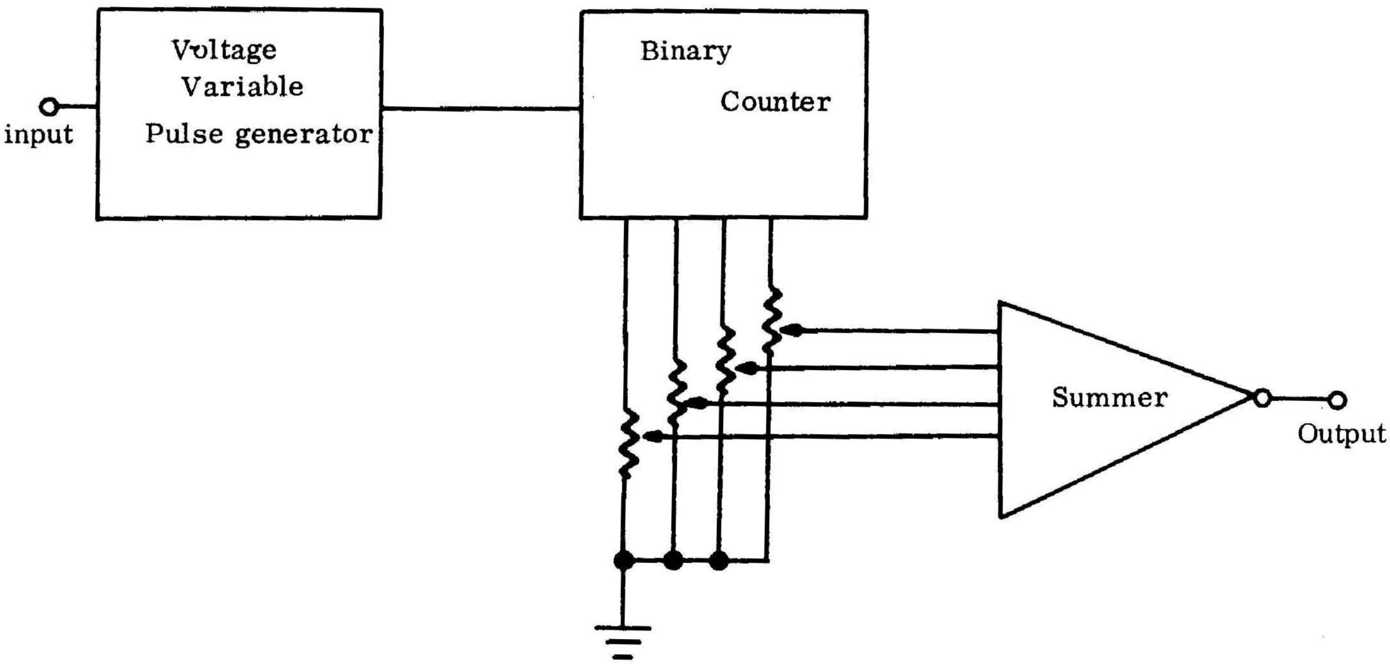
Despite these disadvantages the system represents one of the most useful methods of synthesizing sound.

Considerable research was done at Bell Telephone Laboratories on synthesizing speech and conventional instrument sounds. Some of these sounds, particularly trumpet and violin tones, are quite good; that is, they are almost indistinguishable from actual instruments. The speech sounds generally are of lower quality. The synthesis of conventional instrument sounds is not of great importance to most people involved in electronic music, but it is very important in establishing that the system is capable of accurately generating numerically specified sounds. If a system can generate good conventional instrument sounds, it gives us some confidence that the theoretically described sounds that we generate with the system are accurate.

There are methods for generating pulse height series other than the Bell method. The most straight-forward of these is to use a RING COUNTER as shown below:



By making the clock frequency voltage variable, it is possible to generate pulse heights having specific harmonic content over a wide range of frequencies. Each of the outputs of the ring counter has a voltage adjustment which can be programmed manually or electronically to produce a specified pulse height. Since the outputs of the ring counter are "on" sequentially, the desired pulse height series can be generated. The same basic circuitry can also be used for controlling wave envelopes. The principal disadvantage of this approach is that a relatively large number of ring counter states must be used in order to generate finely detailed sound structures.

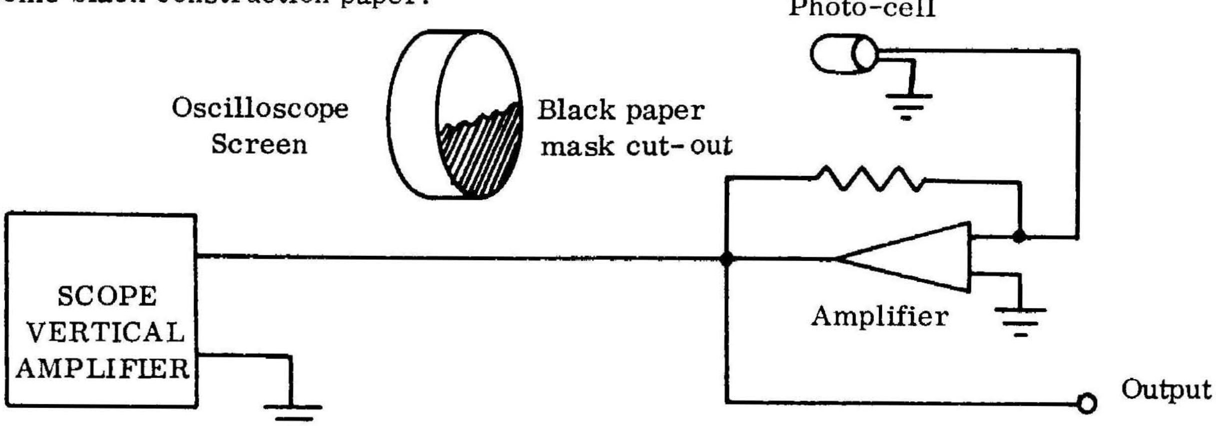


One solution to this problem is to use binary counters in place of ring counters. By doing this we are able to generate many more pulse heights with less circuitry. As can be seen from the above illustration, the outputs of the binary counter are summed to produce the pulse heights. By this and by other means, it is possible to generate series' of pulse heights with a minimum of circuitry. It is on this basic principle that the ORCUS 3001A Universal Analog Function Generator (UAFG) is based.

It can readily be seen that the pulse series approach to complex wave generation is much more elegant than the use of multiple sine wave generators. Furthermore, the pulse sequence generation equipment, briefly discussed above, has the important advantage of being capable of frequency changes over a wide range without change in waveform structure programming.

There are a number of miscellaneous methods for generating complex sounds which should be mentioned here. The systems described above are characterized by being of a general character; that is, capable of generating a wide variety of complex waves. The methods which we will mention now are more limited, but they are sometimes quite useful, particularly for studios not having extensive funds.

It is possible to generate any wave form and any wave envelope with nothing except a simple oscilloscope, a photo cell, operational amplifier, one razor blade and some black construction paper.



SIMPLE COMPLEX WAVE/COMPLEX WAVE ENVELOPE GENERATOR

The shape of the desired complex wave form is cut from construction paper. The cathode ray beam is focused slightly above the mask and the horizontal sweep is set to be equal to the desired fundamental output frequency. When the cathode ray is above the mask, the photo cell in front of the screen couples a signal into the operational amplifier. The polarity of the amplifier output is such that when it is fed to the scope vertical amplifier, it will cause the spot to move down. As the spot tries to move down below the mask, the photo cell sees less light and, therefore, the signal into the amplifier decreases to zero. Thus the spot, as it is swept across the screen, is made to follow the mask outline. Unfortunately with this system, every time it is desired to change the waveform it is necessary to cut a new mask. Another limitation of the system is that the resolution is not very good, because when the spot is made bright it becomes larger and therefore can't follow the small variations in mask contour so well. And when the beam intensity is lowered to decrease the spot size, there is less light for the photo cell.

The most interesting use of the technique is for the generation of various types of wave envelopes. Since the beam sweeps the mask at very low speeds, and since the lack of resolution is not so important in this application, the technique is very useful.

Another approach to the generation of complex sounds, which is sometimes useful, is the noise-filter system. You will recall that in the multiple waveform generator systems it is necessary to use a large number of sine generators in order to

obtain complex sounds. An alternative is to use one generator which produces WHITE NOISE. White noise contains all frequencies within a given frequency range; for musical purposes between 20 and 20,000 Hertz. This white noise is passed through various combinations of filter circuits which allow only a selected frequency or band of frequencies to appear at the output. Thus, the system starts from the other extreme relative to the sine wave method. Instead of starting with the most elemental waveform and building up complex waves, it begins with noise, which contains all frequencies, and filters out the undesired portions. Whether the system has any real advantages over the multiple sine wave or multiple waveform generation systems is debatable. While it is easy to generate bands of complex sounds which would be difficult to achieve with multiple sine waves, it is quite difficult to produce accurate multiple sine waveforms. And this means that it is difficult to apply the Fourier theory to the synthesis of complex sounds.

MUSIQUE CONCRETE

One of the principal methods of generating electronic music is to record acoustical sounds and then physically manipulate the resulting magnetic tape. This is called Musique Concrete. It is not electronic music in the more narrow sense because the origin of the sound is not electronic, but acoustical. Since this book is specifically about electronic music in its more narrow sense, we will not dwell on Musique Concrete. However, the reader should not assume that since we will not pursue a lengthy discussion here, Musique Concrete is not of importance in the area of avant garde composition. However, Musique Concrete will be considered here from a practical point of view. It is much easier and much less expensive to obtain sophisticated sounding music with Musique Concrete than it is with electronically generated music. However, it is much more difficult to achieve a sophisticated degree of control over the sound parameters in Musique Concrete. In essence, Musique Concrete is a study of very complex, real sounds, and electronic music is a study of highly controlled, simple sounds.

The reason for the considerable historical activity in Musique Concrete can be attributed to the fact that it is so simple to achieve complex sounds with little expense. There is also an esthetic reason for the popularity of Musique Concrete; that is, that the acoustical sounds are "real" sounds and are a part of common human experience. This, it is often claimed, gives Musique Concrete some greater artistic value than electronically generated music.

The basic equipment for creating Musique Concrete is a microphone, a tape recorder and tools for tape splicing. The basic process is to record any and all types of acoustical sounds such as wind in trees, city sounds, striking metallic objects together and any other sounds that agree with the composer's ideas. These sounds are then manipulated on the tape recorder. There are four basic tape manipulations.

1. <u>Speed Change</u> - Most tape recorders operate at two speeds. If a sound is recorded at the lower speed, it may be reproduced at twice the speed. And when

this is done the frequencies are doubled and its duration halved. When the sound is recorded at the higher speed and reproduced at one-half that speed, the frequencies are halved and the duration is doubled. Some tape recorders have continuously variable speed controls which enable the composer to change the duration and frequency of the sounds in small increments.

2. Retrograde - If the composer is using a full track tape recorder, the sound will be reproduced backward or retrograde when the reel of tape is turned over. This manipulation produces the quite radical changes in acoustical sounds. As pointed out earlier, if one is using a half track tape recorder it is not possible to play the sound backwards by turning the tape over.

3. Collage - It is quite possible to record one acoustical sound and then to record another acoustical sound on top of the first one without erasing the first sound. You will recall that on most tape recorders the tape passes first over the erase head, then over the record head and then over the reproduce head. In normal operation all previous sounds recorded on the tape are first erased by the erase head before new material is recorded on the tape. However, if we arrange for the tape to follow a physical path such that it does not contact the erase head, the new material will be recorded over the old material. The same result may be accomplished by electrically disconnecting the erase head. This process can be repeated several times to build up complex waves, which are mixtures of several acoustical sounds.

4. Reverberation - If the output of a tape recorder is fed back to its own input, an echo or reverberation effect can be achieved. The reason for this is as follows: Suppose that we have one short sound recorded on the tape, and that we physically or electrically bypass the erase head. Now when the reproduce head reproduces the sound on the tape, it is immediately recorded about two inches later on the same tape by the record head. As the tape continues to move this recorded sound is reproduced by the reproduce head, and is immediately recorded another two inches on down the tape. This process continues for a length of time dependent upon the amplification between the reproduce output and the record input. If this amplification is very low, the echo or the reverberation effect is hardly noticeable. If it exceeds a certain amount, the sound begins to oscillate and a continuous squealing sound is heard. It is relatively easy to see that the length of time between the short sounds on the tape depends on two factors; the speed of the tape and the distance between the record and reproduce heads.

From the above it can be seen that it is a great help to have two tape recorders. For example, if we wish to record one sound then reproduce it retrograde and mix it with another sound, which was previously recorded at one speed and which is now being reproduced at another speed, we must have two recorders. The above description of Musique Concrete represents only the most basic operations. It is possible to do many more things than we have discussed, but each of these requires some additional, rather specialized equipment. For example, there is

a device called a Springer machine which makes it possible to change the duration of sounds without changing their frequencies. However, this equipment costs several thousand dollars; rather expensive for the results achieved.

Musique Concrete is a very popular method for creating electronic music. But it should be kept in mind that Musique Concrete, as well as most other inexpensive methods of making electronic music, is rather limited in its ability to free the composer from the sound characteristics of the equipment which he is using.

CONTROL TECHNIQUES

In order to intelligently discuss control techniques for electronic music, there are several terms with which the student must be familiar. The first of these is a definition of the words DIGITAL and ANALOG. A DIGITAL device is anything which can be in only one of two or more states at any given time. A simple and familiar example of a digital device is a conventional light switch. Not all digital devices, however, have only two positions on and off, but all of them do have a finite number of well-defined states. For example, some switches have click-settings for a large number of positions. But the point is that it is not possible to set such devices to intermediate positions.

An ANALOG device is continuously adjustable. A simple example of an analog device is the volume control on a radio. It can be set in any position. The devices used in electronic music are sometimes analog and sometimes digital.

The student can find detailed explanations of computer logic and number systems in many books, some of which are listed in the bibliography. And even though it is hoped the student will pursue more information in this area, it is probably worthwhile here to outline some basic practical information about logic circuits and number systems.

The number system with which most of us are familiar is the decimal system. In it there are ten numbers, 0 through 9. However, it is not very easy to make electronic circuits which have so many different states. But it is easy to make circuits which have 2 states, "on" and "off." Thus, if we can find some way of representing digital numbers with only 2 states, it will be much easier to build digital systems. The BINARY NUMBER SYSTEM accomplishes this. It uses only 2 numbers, 0 and 1, and works as follows: If we wish to represent the number 0 we write 0, if we wish to write the number 1 we write 1. If we wish to write the number 2 we must use more than 0 or 1 alone. Let us represent 2 as 01. We can represent the number 3 as 10 and the number 4 as 11. If we only use 2 numbers or digits, this is as high as we can count because this is all the possible combinations of 0 and 1 when used one or two at a time.

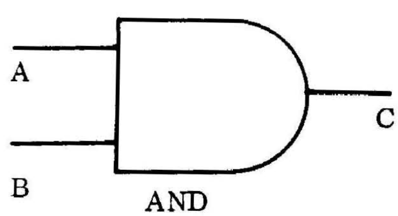
The number of possible combinations of 2 digits is 2^2. If we use 3 digits at a time the number of combinations is $2^3 = 8$. And if we used digits 4 at a time there are

$2^4 = 16$ combinations. In the above formulae the 2 means that there are 2 possible values for each digit (0 and 1), and the superscript means that there are that number of digit positions. You will find a table of the number of combinations that can be made with various numbers of digit positions in the back of this book.

			Digital equivalent
0	0	0	0
0	0	1	1
0	1	0	2
0	1	1	3
1	0	0	4
1	0	1	5
1	1	0	6
1	1	1	7

The above diagram shows all of the possible combinations for 3 digits. The chart for 4 bit positions is exactly the same except that there would be another column to the left, which would have 8 0's and 8 1's and would have 16 possible combinations. The above information is presented certainly not as a complete exposition of binary arithmetic, but it should give the student the basic ideas involved.

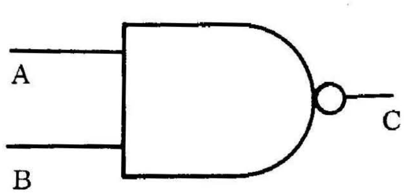
The next important area with which the student must be acquainted is simple logic circuitry. One of the most basic logic operations is performed by the AND gate. AND circuits have two or more inputs and one output. When all of the inputs are presented a logical 1 (or "on") signal, (that is a signal having a voltage greater than 0), the output also has the value 1. This is shown in the TRUTH TABLE below.



AND

INPUTS		OUTPUT
A	B	C
0	0	0
0	1	0
1	0	0
1	1	1

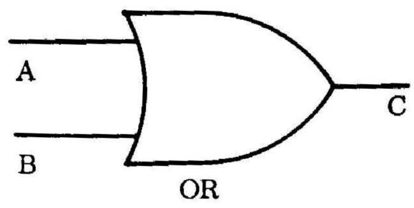
It can be seen that under all other input conditions the output is zero. A circuit whose truth table is as the one above, except that all the 0's have been changed to 1's and all the 1's to 0's, is called a NAND circuit.



NAND

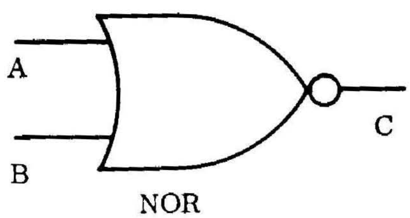
INPUTS		OUTPUT
A	B	C
0	0	1
0	1	1
1	0	1
1	1	0

Another fundamental logical operation is the OR gate.



INPUTS		OUTPUT
A	B	C
0	0	0
0	1	1
1	0	1
1	1	1

In this circuit the output is 1 when either or both of the inputs is 1. The complement of this circuit is the NOR circuit and is shown below.



INPUTS		OUTPUT
A	B	C
0	0	1
0	1	0
1	0	0
1	1	0

These above circuits are by far the most common logic elements. It is possible to use any single type logic gate described above and, through appropriate interconnection and through the use of INVERTERS, to achieve any complex logic function. An INVERTER is a circuit which has one input and one output. The output is the logical complement of the input, as shown below.

The circles at the outputs of the NOR and NAND gates are inverters.

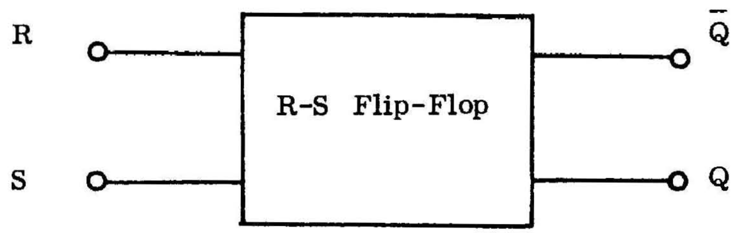
OUTPUT

input	output
1	0
0	1

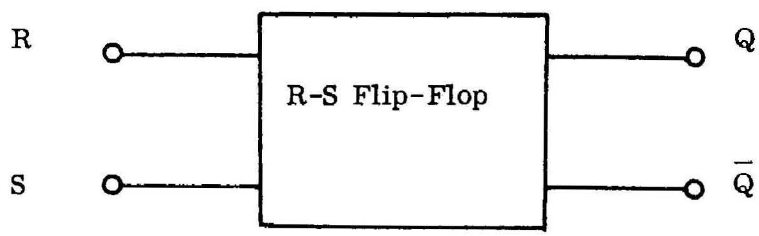
Be sure to remember that it is possible to start with AND, NAND, OR, or NOR gates and INVERTERS, and to make any complex logic function that is desired. For example, a complete computer can be built using only NOR gates and INVERTERS. The most striking example of this approach, perhaps, is the guidance computer of the Apollo spacecraft. The logic functions of this computer are all constructed from basic NOR gates and inverters.

The four logic gate types mentioned above are called DECISION ELEMENTS. That is, the circuit decides what the output shall be on the basis of what signals are present at its input. The other principal type of logic element is the MEMORY

ELEMENT. A memory element is a device that has one or more inputs to which a signal may be presented. The element will store this signal for an indefinite time, even when the signal is removed from the input. The most common sort of memory element is the FLIP-FLOP. There are a number of different types of FLIP-FLOP's, but the most basic and simplest type is the R-S FLIP-FLOP.



Let us assume that the circuit is initially as shown in the above picture. There is no input signal present, the output line Q is in the logical 1 state and the output line $\bar{Q}$ is in the logical 0 state. If we present a logical 1 to the S (SET INPUT), the output will change state as shown by comparison of the drawings above and below.

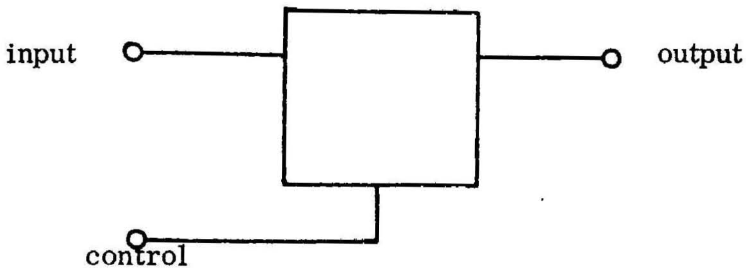


If we reapply the logical 1 signal to the set input, it will have no effect on the output. If we wish to change the circuit from its present state, it is necessary to apply a logical 1 signal to the R or RESET INPUT. The R-S FLIP-FLOP does not allow for the condition where both inputs are presented a logical 1. The circuit does not know how to handle this situation, and whether the output will change is indeterminate. Therefore, it is necessary to insure that this situation cannot arise.

Flip-Flop's are quite useful devices in control circuitry, because they make it possible to store one bit of information for any length of time desired. The most common application to electronic music of both Memory and Decision Elements is in the area of control logic for electronic music systems. There are other types of Memory Elements which are used when a large number of bits must be stored, such as in a computer memory. But all perform the same basic function as the Flip-Flop.

The next type of element is partly digital and partly analog, and it is the interface between the control circuitry and the generation circuitry in electronic music

equipment. This is the TRANSMISSION GATE or ANALOG SWITCH. (See below).



TRANSMISSION GATE (Analog Switch)

Mechanical Switch Equivalents

The function of the Transmission Gate of Analog Switch is to either pass or block analog signals which are present at the input. This is controlled by logic signals presented to the control line. Sometimes Transmission Gates are constructed so that they are normally closed. That is, when there is a logical 0 on the control line the analog signal is able to pass through the circuit. And when a logical 1 is on the control line, the signal is blocked.

Sometimes these circuits are built so that the switch is normally open. That is, when there is a logical 0 on the control line the signal is blocked, and when there is a 1 on the control line the signal is passed.

1	pass
0	block

Normally Open Switch

1	block
0	pass

Normally Closed Switch

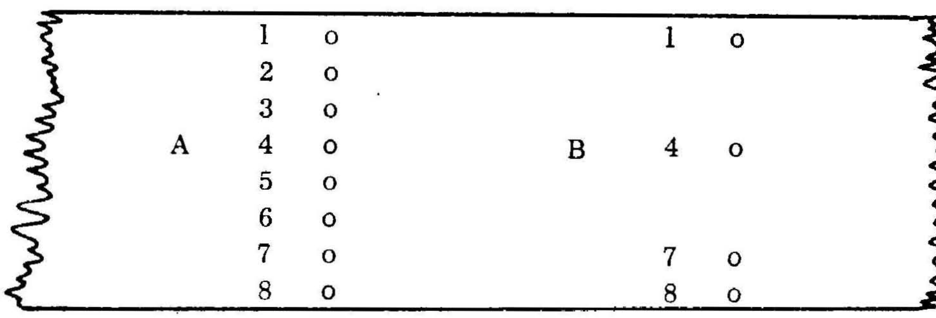
In previous chapters our discussion of electronic music has been in terms of methods for generating sounds. We have yet to discuss possible methods of controlling sound sequences. It is perhaps interesting to note here that electronic music has occupied itself primarily with the generation of sound. Only quite recently has electronic music used anything but quite simple control techniques. In many studios one finds a large number of oscillators with no means of control except the manual setting of dials and switches. A much more useable set-up would be a few oscillators with some appropriate means of programming or controlling them.

The first control technique that we will present is what is sometimes called CLASSICAL STUDIO TECHNIQUE. This is an incredibly ostentatious name for a quite primitive method. Classical Studio Technique means the splicing together of segments of tape, each containing one sound event, into the final composition. This method has a number of serious limitations, the greatest of which is that a piece might contain hundreds or thousands of individual sections of tape which must be manually spliced together in the appropriate sequences. The earliest electronic

music compositions were done in this way, but the method is definitely not recommended for any but the simplest and shortest compositions.

Moving up the scale of complexity and cost we next come to SEQUENCERS. Suppose that we have a number of oscillators whose frequency and amplitude have been set manually. With a sequencer we may program these predetermined sounds to occur in any sequence that we wish. The sequencer allows each oscillator output to pass through it for a duration selected by the composer. By the manual setting of switches on the front of the sequencer, a number of oscillators can be made to produce the desired sequence of frequencies, amplitudes and durations. The sequencer, although having a great deal of utility, is only suitable for controlling rather short sound sequences. Since each of the oscillators is pre-set, it is not possible to generate more sound events than one has oscillators. The usefulness of sequencers rapidly diminishes as they are made larger.

Another and perhaps more useful programming approach is PUNCHED PAPER TAPE. The principle advantage of paper tape over the sequencer is that a large number of sound events can be generated sequentially. Conventional punched paper tape looks generally as below:



It is usually 8 or 12 holes wide. For the moment let's suppose that we are discussing tape that is 8 holes wide, as shown at A. In each of the 8 positions we may either punch a hole or we may not punch a hole as shown at B in the drawing. Let us say that when we punch a hole we represent this by the number 1, and let us say that when we do not punch a hole in a given position we represent this by the number 0. Each 1 or 0 is called a BIT of information.

The application of punched paper tape to electronic music programming is as follows: A paper tape is prepared which has holes punched in it in the appropriate positions to create the desired sound events. After being punched the tape is put on a paper tape reader. This is a device that looks very similar to a tape recorder, but instead of playing magnetic tape it plays punched paper tape. And instead of having heads like a conventional tape recorder, the tape is passed over a device called a PAPER TAPE READER. The reader works basically as follows: A small lamp is directed toward each of the 8 bit positions. When a hole is punched in the paper, the lamp is able to shine through the paper tape and onto a light sensitive device. This produces an electrical pulse at the output of the paper tape reader. When there is no hole punched in the paper, there is no electrical output pulse.

Let us suppose that the first column of 8 bit positions is used to program the frequency of an oscillator and that the second is used to program the amplitude. Once the amplitude and frequency are programmed, the sound will continue, unchanging, until such time as a new frequency and amplitude are programmed into the equipment.

If we have several oscillators that we wish to program, we may do this by having the first pair of columns control the first oscillator, the second pair of columns on the tape control the second oscillator, etc. Since there is a limit to how rapidly we may read the bit positions on the tape, it may be seen that the number of oscillators which can be programmed is limited by the paper tape reader's ability to read the data on the tape.

The above description is a rather simplified outline of the use of punched paper tape. In practice it is usually necessary to provide some method of programming the second sound event during the time when the first sound event is being performed. This requires additional circuitry, but makes it possible to control a much larger number of oscillators.

There are other methods of controlling the output of generation equipment. One of these is simply to use the memory capacity of a digital computer, and to program the computer to read this information out serially through appropriate circuitry such that the desired sound sequences are produced.

There are a great number of different approaches which can be taken to the generation and control of sound by electronic means. It is well here to indicate some general rules for evaluating various types of generation and control systems. The first requirement of an electronic synthesis system is that it be capable of generating a broad range of sound. Ideally, it should be capable of generating any sound within the limits of human hearing. There are some systems presently which do this. However, they generally do not meet one or more of the other requirements discussed below. The next requirement is that the system be accessible from the user's standpoint. The simplest electronic music synthesis systems use familiar but crude programming devices such as keyboards. This makes the use of the equipment simple for the composer, but severely restricts possibilities for sound synthesis. More sophisticated synthesis systems require some sort of symbolic input programming data. The simpler and faster the user can acquaint himself with the program techniques the better the system. The third requirement of an ideal system is that it be operable in real time. Generally, simple synthesis systems are capable of such operation while the more general systems are not. The last requirement is that the equipment be within reasonable cost limits. Present synthesis systems range in cost all the way from very crude simple equipment costing a few hundred dollars, all the way to ones employing sophisticated computers costing many hundreds of thousands of dollars.

The control techniques that we have been discussing above all assume a traditional approach to music composition; traditional in the sense that they all imply that the

composer knows objectively, verbally the frequencies, amplitudes and durations of the sounds he wishes to produce. This is the generally accepted method of composition on the part of music school faculty. In the next section we will discuss topics which might be under the heading of control techniques, but are more properly included under the heading of compositional procedures because they involve the methods of selection of the sounds to be generated. The previous discussion of control techniques assumed that somehow the composer already knew what sounds he wanted, whereas our next discussion will be of methods of making compositional decisions.

COMPOSITIONAL PROCEDURES

Traditionally music has claimed to be based upon acoustical laws. However, history has demonstrated that the musical theories developed during the Renaissance are not so closely related to natural laws as their inventors might have wished to think. Nevertheless, the formulation of compositional procedures on the basis of acoustical principles is a much to be commended endeavor. So long as music uses sound as a medium music must be heard in order to have any effect, and in the final analysis it is what is heard that is of importance in music. No matter how wonderful a computer printout may appear to the eye, it is what the listener hears that is of consequence, and therein lies the difficulty. Different composers have different hearing capabilities and abilities as do listeners. The basing of music on acoustical laws must be kept within the realm of hearability, and the functioning of the human ear and the brain is the greatest obstacle to this goal.

The next method, which is sometimes used, is mathematical in nature. Under this heading comes the use of logic techniques for making compositional decisions as well as random or ALEATORIC procedures. It is important for the student to realize that all of the techniques for making compositional decisions, that we have mentioned here, are possible only with electronic synthesis techniques. If anyone is in doubt about this he should develop a logical system for the generation of frequencies, amplitudes and durations without reference to the limiting characteristics of conventional instruments. He will find it extremely difficult to get a performance of his work without employing electronic techniques. Mathematical procedures for making compositional decisions may range from simple arithmetical procedures to very sophisticated computerized, randomizing techniques. For some this approach to composition is an extremely sterile one, while to others it represents the most significant trend in music today. It should be noted, however, that it is quite easy, given this compositional approach and the use of electronic techniques, to exceed the capabilities of human perception.

The last compositional technique which we will discuss is the use of biological potentials as control information for electronic music synthesis equipment. The two techniques that we have been discussing above are basically SOFTWARE, that is, we were speaking solely of procedures which could be written on paper. The

use of biological potentials represents a compositional decision system which relies upon HARDWARE, that is, it depends upon actual electronic circuitry. The SOFTWARE, in the case of the biological system, is whatever causes biological potentials to change from one state to another.

There are many bio-potentials which can be used for musical purposes. Some of them generate enough information to be used with a relatively small amount of processing while others are of a more auxiliary character. The electrical signals generated by the heart are, for example, quite easily detected over the entire body and can be used as a source of information. Heart rate and waveform are affected enough by auditory experiences to make this source useable. Generally, however, these changes are not great enough to allow the EKG to serve as the sole source of information.

The psycho-galvanic skin reflex, due primarily to changes in skin resistance during emotional stress, can also serve as an auxiliary source. But, since these signals vary quite slowly and predictably, they must be used in conjunction with other bio-potential sources. The signals generated by involuntary muscular contraction can also be used; facial areas are particularly subject to this type of activity. These signals are quite large compared to most other bio-potentials, and are for this reason quite easy to employ. The placement of the sensors is, of course, a determining factor in the nature of the signals which can be detected.

One of the easiest sources to employ, and one which is more indicative of mental activity concerning shape and form, is the movement of the eyes within their orbits. When the eyes are stationary a steady potential is generated between a pair of electrodes attached equidistant from them. However, eye movement produces a change in the ionic concentration of the eye fluids, thus generating a changing electrical potential. The use of eye movement potentials is quite simple since normal eye movement generates signals in excess of 100 mv.

The source of bio-potentials which holds the greatest interest and the greatest challenge, however, is the cerebral cortex. It is primarily to this source that the present author has directed his attention. The discovery in 1929 by Hans Berger of the alpha rhythm, a basic rhythmic brain signal, produced a renewed interest in the dream of "think-work"; of mental remote-control. However, the problem of correlating electroencephalographic (EEG) data with observable behavior has proved to be extremely slow and frustrating over the past forty years. Many postulates and thousands of experiments have failed to produce any widely accepted central theory of correlative interpretation.

There are, though, two important advances in cerebral research, which have occurred during this century, that have given researchers cause for some optimism. The first of these is the determination of functional cortical areas within the brain. It is now quite clear that certain areas of the brain control specific types of behavior and physical action. For example, the parietal area controls fine hand and finger movements; the temporal lobes are associated with memory functions.

The second important advance is the discovery of spontaneous electrical signals of cortical origin. The first of these (the alpha rhythm mentioned above) has proved to be extremely important in electroencephalographic research. A large portion of the EEG literature is devoted to it since it is the most dominant cortical signal. It can be detected over a large portion of the skull, is relatively easy to identify and maintains certain stable characteristics during the entire adult life of an individual. It is generally agreed that the alpha rhythm becomes the dominant brain rhythm by the age of five; has a frequency of 10-12Hz dependent on mental activity; that its amplitude and harmonic content often vary greatly with mental activity, and that it has the greatest amplitude in the vicinity of the occipital lobe.

The basic elements of any EEG music system are: (1) The signals generated by the cerebral cortex and the rest of the brain. (2) Circuitry to process this raw data into appropriate digital information. (3) Memory and decision circuitry to store and direct this data. (4) Sound generation equipment. (5) The human ear and brain which complete the feedback loop. It is evident that a plethora of difficult philosophical problems arise in choosing the nature of the feedback loop elements. Many of the same philosophical problems found in conventional electronic music confront the theorist in ELECTROENCEPHALOMUSIC; but there are, in addition to these, several new difficulties. Questions also arise concerning the nature of "spontaneous music". As an approach to these problems, then, let us consider the following. The developments of any given age are a function of the relationship between the creative thought of the artist and the technical means which are available to him. These two facets of artistic production are intimately linked together. Only rarely does the artist find the technical means available to him adequate for the expression of his artistic ideas. This problem is generally ignored by musicologists and by theorists; it is generally accepted that the limits imposed by given instrumentation correspond to the limitations that the composer would place upon himself from aesthetic or philosophical considerations.

Probably the most significant decision in the history of music was the choice of "pre-conceived" over "spontaneous music". The necessity for the decision arose due to a severe disparity between creative thought and technical means. Technical capabilities of that age provided the composer with various instruments, but with no method for combining their sounds except by "pre-conceiving" the music through some type of musical notation. The choice of pre-meditated instead of spontaneous music came about very simply and naturally; spontaneous music evolved into "improvised" music. Busoni in his prophetic book, The Essence of Music and Other Writings, stated that "Improvisation would stand nearest the true essence of art if it lay within human capacities to master its promptings," (page 100). The use of bio-potentials and electronic circuitry provides the technical means to create a sophisticated spontaneous music.

Many problems in electroencephalomusic, which appear at first to be serious, are in fact quite illusory. The most important of these is the role of the "will" of the composer in producing sounds. Jazz musicians are often asked how they

can think of the right notes so fast. Of course, anyone who has ever improvised realizes the absurdity of the question; the player "guides" the course of the playing, he isn't thinking of each individual note. Furthermore, most will agree that even in the composition of written music the role of the "will" is usually small. Nevertheless, the fact that it is often impossible for the composer of "electroencephalomusic" to "will" a specific sound is disconcerting to some. (It is possible to construct EEG music systems in which the composer can "will" specific sounds). The success of any sophisticated EEG music system, though, depends on the ability of the composer to guide the course of the composition by initiating, subconsciously, changes in the cortical signals which are producing the sounds; the degree of attention is frequently used to regulate the complexity of the flow of events. Of course, the type and complexity of feedback loop circuitry are also quite important in determining the final audio output.

The nature of the feedback loop circuitry can be such that each bit of cortical information is immediately processed into sound material, or the cortical information can assume the role of selecting predetermined and stored sound sequences. In the first case extreme, the cortical signals containing sufficient information to initiate sound would do so immediately. In the second case extreme, cortical signal data would be stored for a duration dependent upon some cortical signal parameters; and would then produce a predetermined sound sequence. This sound sequence would constitute the composition. In practice the decision and memory circuitry never closely approach either of these extremes.

The next problem which confronts the theorist is the significance and nature of various cortical potentials. Within the confines of the present introduction a complete survey of various theories of cortical function is, of course, impossible. However, the function of the various areas of the cerebral cortex, as mentioned previously, is well established. It must be realized that detailed information on the exact site from which a signal emanates is difficult to determine even when the surface of the cortex is exposed, and quite impossible under normal conditions. Furthermore, the correlation between behavior and recorded data remains in a relatively primitive state. However, in using cortical potentials for musical purposes, behavior correlates are of little concern; the same is true in conventional music. The EEG signals can be treated as raw data which can be processed to produce any desired system characteristics. The measure of the control by the feedback loop is the effect of the audio output on the EEG signal compared statistically with the change in EEG parameters with no audio feedback.

The last problem which should be mentioned is that of the role of the composer in employing electroencephalomusic systems. It will be quickly realized that in EEG music systems the mental activity of the composer is "automatically" becoming sound; and that the perception of this sound is "automatically" changing his mental activity. This represents an ideal situation from the point of view of the creator of "spontaneous" music, but it creates a problem rarely considered in "pre-conceived" music; the effect of cogitation upon the final musical product. In "spon-

taneous music" the mental effort is the composition. The process of exclusion becomes a portion of the composition. Thus the EEG composition unfolds in "real time". The mental methods themselves become important in determining the aural output; and the composer soon discovers that the discipline involved in EEG music is a discipline of mental states.

ELECTRONIC MUSIC IN THE FUTURE

Electronic music and the development of control systems, such as exemplified by BIO-MUSIC, represent the most significant change in music since man began employing artificial instruments for the production of sounds. Electronic music is not a new and startling development, it is a logical extension of man's musical endeavors. It is the next step in man's efforts to be able to control sound for music communication. Perhaps in terms of the conventional symphony orchestra, electronic music seems to present a radical departure. In reality, however, such is the case only because many people in the field of music regard present instrumentation and musical groups as being in some way permanent. The instruments presently in use in conventional music actually have a very short history, are very limited, and will be quite archaic in the future.

The art of music is, or perhaps should be, based on man's desire to extend his capabilities in communication. And with this in mind, the ideas in this book are not at all surprising or revolutionary. It is a sad fact that music all too frequently becomes a struggle with the idiosyncracies of stretched strings, hollow tubes, and taut membranes which, quaint though they may be, lead the composer and musician into a state of physical struggle against nature.

There is an increasing tendency in music to remove the necessity for acceptance on the part of the listener. In the past music has required the listener to exert some self-control and some effort in order to understand or perceive the music. In the future, however, it is entirely possible for the listener to be transformed by the music whether he wishes to be or not. It is the same situation as exists when one makes the choice of taking aspirin instead of curing his headache by simple psychological means, or when one takes hallucinogens instead of working with his mind. If the effects desired from a given piece of music can be put into some objective statement, then electronic equipment to measure the efficacy of the sound and the results can be monitored. And this can be used to change the subsequent sound output so as to induce the desired results. Thus, it is possible for music to be composed in much the same way as chemicals are combined to form medicines for various specific purposes. Actually this is no more than music therapy of a highly refined type. The only reason that this type of effort has not been pursued at this time is the lack of research funds and the lack of people who are skilled both in the art and science of music.

It is conceivable that music in the future will dispense with sound altogether and become an art of induced psychological, physiological states. Should this occur

all arts would come closer together. It is important that people involved in the arts concern themselves in a quite fundamental way in the potential of mankind. The arts, and perhaps music is the purest manifestation, are capable of transfiguring man's view of the universe and of himself.

BIO-MUSIC

APPENDIX A

"And he shewed me a pure river of
water of life, clear as crystal
proceeding out of the throne of
God and of the Lamb.

In the midst of the street of it,
and on either side of the river was
there the tree of life, which bare
twelve manner of fruits and yielded
her fruit every month: and the leaves
of the tree were for the healing of
the nations."

BIO-MUSIC is the the term used by ORCUS Research to describe a class of electronic systems that use biological potentials in real-time feedback loops to induce powerful, predictable, repeatable, physiological and psychological states which can be elegantly controlled. The programmable states are more powerful than those possible utilizing chemical drugs and the sensory and hallucinogenic powers of electronic sensory feedback systems can be controlled and guided with a precision utterly impossible with chemical methods.

ORCUS research into the utilization of bio-potentials for artistic purposes dates back to 1960. At that time, we began to investigate the possibilities for generating a spontaneous music based upon the conversion of various bio-potentials into sound. Our goal was to devise an improvisational system which would be entirely free of acoustic and mechanical limitations as they exist in all conventional instruments. As our experiments and theories progressed, we began to see that we were getting into an area of great power because of its subtle influence and freedom from inertia. This we called, "physiological parameter control through sensory and electrical feedback stimulation". Because physiological parameters could be monitored in real time, and because we had some basis for knowing the effects of stimuli which could be monitored, we found that we could control the physiological and psychological state of performer/listener. It became apparent immediately that one of the most frustrating and abiding problems of communication would be solved with these techniques: the problem of the same physical/sensory stimulation having different meanings to various individuals. With biological real-time monitoring and electronic generation of visual, aural, and electrical stimulation, it would be possible to adjust in real time, the stimuli presented to the organism. And in such a manner that we could write--not compositions of musical notes that would have some indeterminate effect on the performer/listener--but a physiological/psychological state program that would control the generation of whatever sensory and electrical stimuli were needed to realize the physiological/psychological state program.

At that point, we sat down and had a long series of heart-to-heart discussions about the philosophy and traditions involved in art, science, technology, religion, and their relationships one with another. And we realized that we had stumbled upon something so powerful, so awesome that we became afraid to speak of it. So, we let it be for some time. Then we began to detect a growing realization on the part of others in various fields: psychology, neurology, pharmacology, psycho-acoustics, etc., of the power of bio-feedback. Biological feedback stimulation is arriving in the public consciousness with a speed and force that makes the dissemination of useful, accurate, and honest information about bio-music techniques essential. It is our hope that ORCUS efforts will lead to an understanding of biological music and electronic hallucinogens that avoids the tangle of information and error that surrounds chemical hallucinogens and narcotics.

Biological feedback sensory stimulation is the process for amplifying electrical signals originating in the body; then converting them to aural, visual, or other electrical phenomena. In turn, these phenomena are the feedback stimuli that influence the bio-potentials; depending upon the degree of attention and interest applied. Thus, a closed loop is formed which facilitates the voluntary control of normally involuntary body functions. Or normally voluntary functions are made involuntary. An example of the former mode is in the use of aural feedback of heart rate. An individual can gain control over heart rate quickly because the feedback provides him with conscious influence. An example of the latter mode is the use of muscle stimulation to control facial expression or eye movements.

To increase the probability that stimulation will have the desired effect with minimum feedback adjustment, it is desirable to know the physiological/psychological state and receptivity to external stimulation. This may be done through sensory deprivation: the performer/listener is initially deprived of all aural, visual, or tactile stimulation. A more efficient method is sensory bombardment. The performer/listener is bathed in random, high-level, audio/visual/tactile stimulation. If sensory bombardment is continued for a quarter hour or so, most individuals will begin to hallucinate. Thus, sensory bombardment has application in psychiatry and psychology as well as in experiential exploration and in bio-music. In bio-music, sensory bombardment is used primarily to engage the sensory system before and between presentations of specific aural, visual, or electrical stimulation. Sensory bombardment is not a dangerous technique.

The concept of real-time biological feedback control is one of the most powerful tools ever conceived. It has applications in virtually every area of human activity: music, visual arts, psychiatry, medicine, neurology, education, religion, sociology, etc. It is not the purpose of this article to discourage the use of bio-feedback to achieve any type of physio-psychic experience. But one must be alert to potential dangers to prevent personal injury. Biological real-time feedback stimulation is more powerful than drugs and has fewer undesirable side effects. And it is more controllable. Its correct use can lead to new experiential worlds. Its irresponsible use can lead quickly to permanent physical or psychic damage, and death.

WHAT ARE BIO-POTENTIALS?

The bodies of all living organisms produce electrical signals. These are of two basic types:

> Signals arising from vital life functions such as the heart, respiration, etc.
> Signals from non-vital functions such as eye movement potentials, muscular action potentials, etc.

BIO-FEEDBACK

Biological feedback is the process of converting bio-potentials to sensory stimulation. The reaction of the body's physical senses to the stimulation causes, in turn, a change in the bio-potentials. Positive feedback implies that the reaction causes augmentation of the potential; negative feedback, suppression of the potential. "Transfer Function" is the difference between the bio-potential originally applied and the bio-potential resulting from feedback influence. "Real-Time Bio-Feedback" is the monitoring, conversion, and re-presentation of aural, visual, and electrical stimuli to the organism in a continuous manner.

CAUTION

For those who are experimenting now or who plan to do so:

1. Always provide circuitry to limit the feedback stimulus. For example, if the feedback stimulus is an electric shock, the potential must not be allowed to reach dangerous levels. Never apply electro-stimulation between electrodes which will cause current to pass through the spinal cord or the heart. All circuitry designs should contain fail-safe provisions. Then if the circuit should fail, output drops to zero.

2. Use monitors to control application of stimulus to safe levels for either positive or negative feedback to vital functions.

3. Human organisms vary widely in their responses to stimulus. If there is any uncertainty, begin by making the limits described in 1 and 2 very narrow; then increase them slowly.

Contact ORCUS Research if there is doubt about the application. The Company will provide whatever assistance is required.

ORCUS (An International Consulting Consortium) is active in a wide variety of research into Life Sciences, Technological Arts (such as film, kinetic art, music, recording, radio, and television), Civil Rural Technology, Operational Research, and Specialized Electronics Data Acquisition. For further information about these subjects, contact the address below or the ORCUS Applications Engineer.

In the bio-music field, ORCUS generates information, devices, and special systems for experimentation and general use. For further information on equipment, design, publications or applications, please don't hesitate to contact us by phone or mail.

An ORCUS Bibliography on BIO-MUSIC

BIO-MUSIC (Biological Feedback Experimental Music Systems)-Manford L. Eaton. The definitive book on Bio-Music Techniques & Philosophy.
$10.95 (Add $1.50 outside USA for insured air mail.)

ELECTRONIC MUSIC-A HANDBOOK OF SOUND SYNTHESIS & CONTROL
Most widely used text on electronic music basics.
$10.95 (Add $1.50 outside USA for insured air mail.)

BIO-POTENTIALS AS CONTROL DATA FOR SPONTANEOUS MUSIC-
Manford L. Eaton.
The now classic paper presented to the 1st International EM Congress-Florence, Italy 1968.
$1.50 (Add 45¢ outside USA.)

"Production d'ondes par passage numerique-analogique et utilisation de circuits de commande biologique en temps reel en musique electronique" by Manford L. Eaton (Presentation to Festival du Son-Paris, 1970)
$1.75 Xerox (Add 45¢ outside USA.)

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APPENDIX B

CHARTS

1. SOUND TRANSMISSION IN AIR
2. MUSICAL SCALES
3. RESISTOR COLOR CODE
4. OHM'S LAW FORMULAE FOR AC CIRCUITS
5. DECIBEL CHART
6. POWER AND VOLTAGE OR CURRENT RATIOS
7. BINARY NUMBERS

SOUND TRANSMISSION IN AIR AT 20°C AND 760 M. M.

Frequency	30	50	100	200	400	1000	4000	c/s
Wavelength	452	271	136	67.7	33.9	13.6	3.39	inches
	37.7	22.6	11.3	5.65	2.82	1.13	0.282	feet

Velocity of sound in air = 13550 inches per second

= 1129 feet per second

= 344 metres per second.

$f \times \lambda = 1129$

where f = frequency in cycles per second

and λ = wavelength in feet.

RESISTOR

COLOR CODE

Color	Significant Figure	Decimal Multiplier	Tolerance %
Black	0	1	
Brown	1	10	
Red	2	100	
Orange	3	1,000	
Yellow	4	10,000	
Green	5	100,000	
Blue	6	1,000,000	
Violet	7	10,000,000	
Grey	8	100,000,000	
White	9		
Gold	--		$\pm$ 5
Silver	--		$\pm$ 10
No Color	--		$\pm$ 20

Band A indicates the first significant figure.
Band B indicates the second significant figure.
Band C indicates the decimal multiplier.
Band D, if any, indicates the tolerance limits about the nominal resistance value.

MUSICAL SCALES

Every musical tone has a frequency which is measured in cycles per second. A scale is a series of tones ascending or descending in frequency by definite intervals. The octave is the most important interval; two tones are separated by one octave when the frequencies are in the ratio 2 : 1. Each octave is subdivided into a number of smaller intervals. In the equally tempered scale the octave is divided into twelve equal intervals (1.0595 : 1) to allow a change of key without retuning. Thus in the equally tempered scale, if any frequency is multiplied by 1.0595 it is raised by one semi-tone; if multiplied by 1.1225 it is raised by one tone. All normal musical instruments follow the equally tempered scale.

In addition, there are several systems of what is called the natural scale, or just intonation, in which for example C# has a different frequency from Db. The table below gives one version of "just intonation."

Tone Interval		Frequency ratio	
		Natural (just) scale	Tempered Scale
C	Unison	1 = 1.000	1.000
C#	Semitone	25/24 = 1.042	
			1.0595
Db	Minor second	27/25 = 1.080	
D	Major second (= tone)	9/8 = 1.125	1.1225
D#	Augmented second	75/64 = 1.172	
			1.189
Eb	Minor third	6/5 = 1.200	
E	Major third	5/4 = 1.250	
			1.260
Fb	Diminished fourth	32/25 = 1.280	
E#	Augmented third	125/96 = 1.302	
			1.335
F	Perfect fourth	4/3 = 1.333	
F#	Augmented fourth	25/18 = 1.389	
			1.414
Gb	Diminished fifth	36/25 = 1.440	
G	Perfect fifth	3/2 = 1.500	1.498
G#	Augmented fifth	25/16 = 1.563	
			1.587
Ab	Minor sixth	8/5 = 1.600	
A	Major sixth	5/3 = 1.667	1.682
A#	Augmented sixth	125/72 = 1.736	
			1.782
Bb	Minor seventh	9/5 = 1.800	
B	Major seventh	15/8 = 1.875	
			1.888
Cb	Diminished octave	48/25 = 1.920	
B#	Augmented seventh	125/64 = 1.953	2.000
C	Octave	2 = 2.000	2.000

OHM'S LAW FORMULAE FOR A-C CIRCUITS

Known Values	Formulae for determining unknown values of...			
	I	Z	E	P
I & Z			IZ	$I^2Z \cos\theta$
I & E		$\frac{E}{I}$		$IE \cos\theta$
I & P		$\frac{P}{I^2 \cos\theta}$	$\frac{P}{I \cos\theta}$	
Z & E	$\frac{E}{Z}$			$\frac{E^2 \cos\theta}{Z}$
Z & P	$\sqrt{\frac{P}{Z \cos\theta}}$		$\sqrt{\frac{PZ}{\cos\theta}}$	
E & P	$\frac{P}{E \cos\theta}$	$\frac{E^2 \cos\theta}{P}$		

DECIBELS ABOVE AND BELOW REFERENCE LEVEL
1 mW INTO 600 OHMS
(The power holds for any impedance, but the voltage holds only for 600 ohms.)

db down		Level	db up	
Volts	Milliwatts	dbm	Volts	Milliwatts
0.7746	1.000	-0+	0.7746	1.000
0.6905	.7943	1	0.8691	1.259
0.6167	.6310	2	0.9752	1.585
0.5484	.5012	3	1.094	1.995
0.4887	.3981	4	1.228	2.512
0.4356	.3162	5	1.377	3.162
0.3882	.2512	6	1.546	3.981
0.3460	.1995	7	1.734	5.012
0.3084	.1585	8	1.946	6.310
0.2748	.1259	9	2.183	7.943
0.2449	.1000	10	2.449	10.000
0.2183	.07943	11	2.748	12.59
0.1946	.06310	12	3.084	15.85
0.1734	.05012	13	3.460	19.95
0.1546	.03981	14	3.882	25.12
0.1377	.03162	15	4.356	31.62
0.1228	.02512	16	4.887	39.81
0.1094	.01995	17	5.484	50.12
0.09752	.01585	18	6.153	63.10
0.08691	.01259	19	6.905	79.43
0.07746	.01000	20	7.746	100.00
0.04356	.00316	25	13.77	316.2
0.02449	.00100	30	24.49	1.000W
0.01377	.000316	35	43.56	3.162W
0.007746	.000100	40	77.46	10.00W
0.004356	3.16×10^{-5}	45	137.7	31.62W
0.002449	1.00×10^{-5}	50	244.9	100W
0.001377	3.16×10^{-6}	55	435.6	316.2W
0.0007746	1.00×10^{-6}	60	774.6	1000W
0.0004356	3.16×10^{-7}	65	1377	3162W
0.0002449	1.00×10^{-7}	70	2449	10000W
0.0001377	3.16×10^{-8}	75	4356	31620W
0.00007746	1.00×10^{-8}	80	7746	100000W

POWER AND VOLTAGE OR CURRENT RATIOS
EXPRESSED IN DECIBELS

Ratio	db (Power Ratio)	db (Voltage or Current Ratio)	Ratio	db (Power Ratio)	db (Voltage or Current Ratio)
1.0	0	0	5.7	7.559	15.117
1.1	0.414	0.828	5.8	7.634	15.269
1.2	0.792	1.584	5.9	7.709	15.417
1.3	1.139	2.279	6.0	7.782	15.563
1.4	1.461	2.923	6.1	7.853	15.707
1.5	1.761	3.522	6.2	7.924	15.848
1.6	2.041	4.082	6.3	7.993	15.987
1.7	2.304	4.609	6.4	8.062	16.124
1.8	2.553	5.105	6.5	8.129	16.258
1.9	2.788	5.575	6.6	8.195	16.391
2.0	3.010	6.021	6.7	8.261	16.521
2.1	3.222	6.444	6.8	8.325	16.650
2.2	3.424	6.848	6.9	8.388	16.777
2.3	3.617	7.235	7.0	8.451	16.902
2.4	3.802	7.604	7.1	8.513	17.025
2.5	3.979	7.959	7.2	8.573	17.147
2.6	4.150	8.299	7.3	8.633	17.266
2.7	4.314	8.627	7.4	8.692	17.385
2.8	4.472	8.943	7.5	8.751	17.501
2.9	4.624	9.248	7.6	8.808	17.616
3.0	4.771	9.542	7.7	8.865	17.730
3.1	4.914	9.827	7.8	8.921	17.842
3.2	5.051	10.103	7.9	8.976	17.953
3.3	5.185	10.370	8.0	9.031	18.062
3.4	5.315	10.630	8.1	9.-85	18.170
3.5	5.441	10.881	8.2	9.138	18.276
3.6	5.563	11.126	8.3	9.191	18.382
3.7	5.682	11.364	8.4	9.243	18.486
3.8	5.798	11.596	8.5	9.294	18.588
3.9	5.911	11.821	8.6	9.345	18.690
4.0	6.021	12.041	8.7	9.395	18.790
4.1	6.128	12.256	8.8	9.445	18.890
4.2	6.232	12.465	8.9	9.494	18.988
4.3	6.335	12.669	9.0	9.542	19.085
4.4	6.435	12.869	9.1	9.590	19.181
4.5	6.532	13.064	9.2	9.638	19.276
4.6	6.628	13.255	9.3	9.685	19.370
4.7	6.721	13.442	9.4	9.731	19.463
4.8	6.812	13.625	9.5	9.777	19.554
4.9	6.902	13.804	9.6	9.823	19.645
5.0	6.990	13.979	9.7	9.868	19.735
5.1	7.076	14.151	9.8	9.912	19.825
5.2	7.160	14.320	9.9	9.956	19.913
5.3	7.243	14.486	10.0	10.000	20.000
5.4	7.324	14.648	100	20	40
5.5	7.404	14.807	1000	30	60
5.6	7.482	14.964	10000	40	80

BINARY NUMBERS

B I N A R Y – F O R M					DECIMAL
0	0	0	0	0	0
0	0	0	0	1	1
0	0	0	1	0	2
0	0	0	1	1	3
0	0	1	0	0	4
0	0	1	0	1	5
0	0	1	1	0	6
0	0	1	1	1	7
0	1	0	0	0	8
0	1	0	0	1	9
0	1	0	1	0	10
0	1	0	1	1	11
0	1	1	0	0	12
0	1	1	0	1	13
0	1	1	1	0	14
0	1	1	1	1	15
1	0	0	0	0	16
1	0	0	0	1	17
1	0	0	1	0	18
1	0	0	1	1	19
1	0	1	0	0	20
1	0	1	0	1	21
1	0	1	1	0	22
1	0	1	1	1	23
1	1	0	0	0	24
1	1	0	0	1	25
1	1	0	1	0	26
1	1	0	1	1	27
1	1	1	0	0	28
1	1	1	0	1	29
1	1	1	1	0	30
1	1	1	1	1	31

SUGGESTIONS FOR FURTHER READING

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GLOSSARY AND INDEX

(Numbers following the definitions refer to page numbers in the book where further information can be found).

GLOSSARY AND INDEX

ACTIVE - A device or circuit which requires the application of DC power for correct operation. This power is applied between the supply voltage terminal and ground. It is distinct from either the input or output terminals. (22)

ALEATORIC - Random or chance events. (59)

ALTERNATING CURRENT or VOLTAGE - Electric current (voltage, etc.), which changes direction rhythmically, rising from zero to a maximum in the positive direction, falling to zero again and then increasing to a maximum in the negative direction before returning to zero, after which the cycle repeats. (19)

AMPERE - Practical unit of electric current. Equivalent to the flow of $6 \cdot 25 \times 10^{18}$ electrons per second. (13)

AMPLIFIERS - Electronic apparatus capable of producing a magnified version of an input signal. (2)

AMPLITUDE - Maximum instantaneous value (positive or negative) of a periodic quantity such as an alternating voltage. (4, 9)

ANALOG - General name for computing devices in which the variables in a given problem are represented by physical quantities such as lengths, pressures, electric charges, etc., the calculations consisting in the manipulation and measurement of these quantities, the values of which may change continuously. (52)

ANALOG SWITCH - A device or circuit having an input, output and one control terminal. Application of a DC voltage to the control input changes the resistance between the input and output terminals to a very low value or vice versa. Thus, the transmission of analog signals is controlled by the application of control signals. This circuit is also called a TRANSMISSION GATE. (56)

AND - A logic circuit that has two or more inputs and one output. The output has the logical value 1 only when all of the inputs have the logical value 1. For all other input conditions the output has the logical value 0. (53, 54)

ATOMS - Smallest quantity of a chemical element which can enter into combination or take part in a chemical reaction. (13)

ATTACK TIME - The time after initiation of a control signal for the sound to reach maximum amplitude. (10)

ATTENUATOR - Electrical network used to deliberately reduce the input signal to some piece of apparatus. The simplest continuously variable attenuator is a Potentiometer or voltage divider connected across the input terminals. (2, 30)

BALANCED - Input/Output arrangement of circuits such that the input signal is simultaneously fed to two identical circuits which are out of phase with each other. Thus, only the difference between the two inputs is amplified. This arrangement is less susceptible to extraneous noise than conventional circuits. (23)

BATTERIES - A fundamental source of electricity. They operate on the principle of potential difference between chemical elements. Batteries are classified either as primary cells which cannot be recharged or secondary cells which can. Batteries produce direct voltages and direct currents. They are available in voltages ranging from 1.2 volts to several hundred volts, and they are capable of supplying various amounts of current. (19)

BINARY COUNTER - An arrangement of Decision and Memory Elements such that with each input pulse the outputs of the Memory Elements assume states representing successive Binary numbers. (48)

BIO-MUSIC - Any type of music which uses biological electronic signals for the generation and/or control of electronic sounds. Developed at ORCUS Research. (63) (65)

BIT - The word is a contraction of the words "Binary Digit." The state of a single output or input terminal of any digital device, circuit or system at any instant contains one "bit" of data; either "1" or "0". (57)

BLACK BOXES - Circuits, devices, or systems having various inputs, outputs and control terminals, and having specific operating characteristics and transfer function, contents of which are unspecified when the input/output characteristics and the transfer functions of Black Boxes are known they may be successfully interconnected without knowledge of the contents. (21)

CAPACITOR - Device consisting of two conductive bodies (the "plates") separated by insulating material (the "dielectric"), and thus possessing the property of Capacitance, that is the ability to store up electrons. The dielectric may be impregnated paper, mica, various plastics or air. (21)

CLASSICAL STUDIO TECHNIQUE - The splicing together of segments of magnetic tape, each containing one sound event, into the final composition. (56)

COMPLEX WAVE - Any waveform which contains more than one sine wave. (4, 6)

COMPLEX WAVE SYNTHESIZER - Any electronic device for the purpose of generating complex waves from one or more basic waveforms such as sine, square, triangular or sawtooth. (37)

CONDUCTORS - A substance in which free electrons are available to move under the influence of an electric field, and thus to produce the phenomenon known as an electric current. In a solid conductor the current consists in the drift of free electrons through the inter-atomic space, in the general direction of the positive pole of the electric field. (13)

CONTROL TERMINALS - Inputs to circuits which alter the transfer function of the circuit. Control Terminals may be such that they require either analog or digital signals. (23)

COULOMB - Unit of quantity of electricity, equal to the amount of electricity delivered when a current of one ampere flows for one second (one ampere-second). (13)

CURRENT - When the drifting motion of free electrons is controlled in such a manner that all free electrons tend to move in the same direction, an electronic current results. (13)

CURRENT DEVICE - Name applied to a device, the input resistance of which is low. (23)

CUTOFF - A device is said to be cut-off when the operating conditions are such that it becomes non-conductive. Cut-off can be achieved, for example, by applying a sufficiently high DC voltage to the control input of a voltage controlled device. (44)

CYCLE - Complete series of changes occurring in or performed periodically by a system, such that the condition at the end of the series is the same as that at the commencement. (3)

DAMPED OSCILLATION - Oscillations the amplitude of which progressively decreases. (2)

DECAY TIME - Period of time required for oscillations to die away to zero once the stimulus has been removed. (10)

DECIBEL - (Symbol dB) Unit used when comparing two amounts of power, current, voltage, etc., e.g. when indicating the gain of an amplifier. Equal to one-tenth of a Bel. (31, 32)

DECISION ELEMENTS - One of the two types of elements in all digital computation equipment. Decision Elements are those having multiple inputs and one output. The state of the output is a function of the states of the inputs. That is, the circuit decides what its output should be on the basis of the input conditions which are presented to it. (54)

DIFFERENTIAL - The difference between two quantities. A differential amplifier, for example, is one which amplifies the difference between two inputs. (23)

DIGITAL - A device which can be in only one of two or more well-defined states at any given time. (52)

DIODES - Devices which will allow an electron flow in one direction only. Diodes are mainly employed as rectifiers, i.e. for delivering a uni-directional output from an alternating input.

DIRECT CURRENT or VOLTAGE - Current which flows continuously in one direction. (19)

DISTORTION - The departure of the output waveform of an amplifier, etc., from the waveform of the input. (3)

DIVIDER - A circuit which performs the operation of division on the frequency, voltage, current or any other electrical parameter. (17)

DURATION - The length of time during which any thing continues or lasts. (4, 9)

ELASTICITY - The tendency of any material to revert to its original shape after physical distortion. (1)

ELECTROENCEPHALOMUSIC - A specific type of bio-music which employs electroencephalographic (EEG) potentials as control data for electronic music. (61)

ELECTROLYTE - Conductor, normally a liquid, in which the flow of current consists of the migration of ions, negative ions moving towards the positive electrode and positive ions towards the negative electrode. (20)

ELECTROMOTIVE FORCE - Frequently abbreviated e.m.f. Force which causes the movement of electric charges. The unit of electromotive force is the volt. (14)

ELECTRONIC MUSIC - That type of music whose sound source is electronic circuitry. It is thus distinct from Musique Concrete in that the sound source is different. (63)

ELECTRONS - One of the fundamental constituents of the Atom, having a mass $\frac{1}{1840}$ that of a hydrogen atom. It carries a negative electric charge. (13)

FALL TIME - The time required for a square wave to get from the maximum positive level to within 10% of its zero level. (39)

FILTERS - Any circuit whose transfer function is dependent on input frequency. An ideal filter is one which passes some input frequencies without change, and completely blocks all others. There are four basic types of filter action: low-pass, high-pass, band-pass, band-stop. (44)

FLIP-FLOP - A type of memory element frequently used in digital computing equipment. There are several different types of flip-flops, but they all have the common ability to store one bit of information for an indefinite period after the input signal has been removed. They are frequently used in electronic music control circuitry. (55)

FLUTTER - Is the same as wow, only the irregularities are at a higher frequency. (34)

FREE ELECTRON - Any electron which is not tightly bound to an atom and is thus free to move within inter-atomic space. In electronics materials such as copper have many free electrons and are called conductors. Materials which have few free electrons are called insulators.

FREQUENCY - The number of times a periodic phenomenon repeats itself in unit time. For audio and radio frequency waves, the frequency is expressed in cycles per second or Hertz; for the higher frequencies, in kiloHertz per second or mega-Hertz per second. (41)

FREQUENCY RESPONSE - The relative impedance of, and hence the relative voltage developed across a circuit when signals of various frequencies are applied to it. (27)

FUNDAMENTAL - The lowest frequency in any complex wave. (4)

GENERATORS - Any apparatus for producing some specific action. In electronics, a circuit which produces a specified waveform. (20)

GROUND - The common connections in a circuit. All points of a circuit which have no voltage between them and physical ground. (14)

HARDWARE - The electronic equipment which is used to implement operational rules. (60)

HARMONICS - Oscillations the frequencies of which are whole multiples of the frequency of the Fundamental. Harmonics are designated second, third, fourth harmonics, etc., i.e. harmonics having twice, three times, four times, etc., the fundamental frequency. (4)

HARMONIC WAVE FORMS - Any complex waves in which all of the sine components are exact multiples of the lowest sine component. (4)

HERTZ - Term for unit frequency (abbreviation Hz), i.e. one cycle per second. Frequencies in excess of 1,000 c/s are usually expressed in Kilo-Hertz or in Mega-Hertz. The name Hertz comes from the 19th century researcher Heinrich Hertz, the discoverer of electromagnetic radiation. (3)

HEAD - The electromagnetic devices employed in tape recorders to "Record", "Reproduce" and "Erase" electromagnetic signals on magnetic tape. (34)

IMPEDANCE - (Symbol Z) The total opposition offered by a circuit to the flow of alternating current, and equal to the ratio of the r.m.s. value of the applied voltage to the r.m.s. value of the current. Impedance is a complex property, having three components: (a) the resistance of the circuit, (b) a reactive component due to the inductance of the circuit, and (c) a further reactive component due to the capacitance of the circuit, the last two being frequency-dependent. (21)

INDUCTORS - Components included in a circuit in order to exploit its inductance. It consists essentially of a coil of wire, the interior of the coil being either an air space or a core of ferromagnetic material. (21)

INPUT IMPEDANCE - The total opposition to current flow offered by the input of a device, circuit or system. (25)

INPUTS - The terminals of a device, circuit or system to which are applied the signals upon which operations are to be performed. (21)

INSULATORS - (1) In general a non-conductor of electricity. (2) A structure made of porcelain, glass or some other non-conducting material and used to support a current-carrying conductor. (13)

INTEGRATED CIRCUITS - Devices which contain entire circuit functions on a small silicon semiconductor chip. These devices fall into two categories; Linear (or analog) and Digital. (22)

INTENSITY - Specifically an acoustical term. The intensity of a sound is roughly equivalent to its loudness. Intensity is a physically measurable quantity, whereas loudness is a psychological phenomenon. (9)

INVERTERS - Circuits which have one input and one output, the output always assuming the logical complement of the input. Widely used in digital control and computation equipment. (54)

LINEARITY - The degree to which the relationship between two variable quantities is linear. (39)

LOAD - That part of an electric circuit in which the output voltage and current is developed. (27, 24)

LOUDNESS - Is the psychological parameter of intensity. The "phon" is the unit of loudness level. The loudness level in phons of a sound is numerically equal to the intensity level in decibels of a 1KHz sine wave which is judged by listeners to be equally loud. (9)

LOUDSPEAKER - An electro-acoustical transducer capable of radiating a large volume of sound. Electro-magnetic speakers employa vibrating reed or a balanced armature to drive a diaphragm; condenser loudspeakers operate by electrostatic action; crystal loudspeakers by piezo-electric action, and in moving-coil loudspeakers or electro-dynamic speakers a small flexibly-mounted coil carrying the audio-frequency current moves longitudinally in a strong magnetic field and drives a cone-shaped diaphragm, which in turn vibrates the surrounding air. (2, 31)

MAXIMUM ACCEPTABLE INPUT SIGNAL - The largest input signal which can be applied to a device, circuit, or system without altering the transfer function. (26)

MAXIMUM POWER TRANSFER THEOREM - The maximum power will be asorbed from one circuit by a second joined to its output terminals, if the output resistance of the first circuit is equal to the input resistance of the second. (24)

MEMORY ELEMENT - One of the two major types of digital system elements. They are capable of storing binary information for an indefinite period of time after the input signal has been removed. Flip-flops are the most familiar example of this. (54,55)

MICROAMP - One-millionth of an ampere, 1 x 10^{-6} ampere. (14)

MICROPHONE - An electro-acoustical transducer in which the acoustical energy of sound waves is converted into electrical energy. (2, 31)

MILLIAMPERE - One-thousandth of an ampere, 1 x 10^{-3} ampere. (14)

MINIMUM LOSS PADS - A resistive network which is placed between two circuits for the purpose of minimizing the power lost in transferring energy from one circuit to the other. (28)

MIXER - Arrangement of adjustable potentiometers whereby two or more input signals (e.g. from different microphones or electronic sound sources) can be combined in any desired proportions to form a composite signal. (37)

MODULATION - Process of varying the amplitude, frequency or phase of a signal in accordance with the amplitude, frequency or phase of a second signal. (41, 42)

MONITOR - Equipment used for sampling a signal transmission to check quality, without interfering with normal operation. Also the act of sampling a signal transmission. (35)

MUSIQUE CONCRETE - A type of music developed in France after World War II. The sound elements of this music are acoustical. This is in distinction to Electronic Music, in which the source of the sounds is electronic circuitry. (50)

NAND - The "not and" logic function. It is derived by interchanging the logical 1's and 0's in the output column of the truth table for the "and" function. (53,54)

NEUTRONS - Uncharged particles, of slightly greater mass than protons, and forming a constituent part of the nucleus of all atoms except hydrogen atoms, which consist of a single proton revolving around the nucleus. A neutron may be considered as the equivalent of one proton and one electron. (13)

NOISE - In general, any unwanted electrical signal(s) within or generated by circuits, devices or systems. (34, 50)

NON-HARMONIC WAVE FORMS - All waves in which the sine waves are not harmonically related to a fundamental frequency. (4)

NOR - The "not or" function. The NOR function is derived from the OR function by interchanging all the 1's and 0's in the output column of the truth table for the OR function. (54)

NUCLEUS - The central "core" of an atom, in which the greater part of the mass resides. The nucleus carries a positive electric charge which, in the normal atom, is neutralized by the combined negative charges of the orbital electrons. (13)

NUMBER SYSTEM - The base on which a system of indicating numerical quantities is arranged. Thus the decimal scale employs 10 digits (0,1,2,... 9) and the binary scale only two digits (0 and 1). (52)

OHMS LAW - A conductor has a resistance of 1 Ohm when it is necessary to apply a potential difference of 1 volt in order to drive a current of 1 ampere through the conductor. Algebraically, $E = I R$, where E equals potential difference in volts, I equals current in amperes, and R equals resistance in ohms. (15)

ONE-DIMENSIONAL - Any physical structure in which the dimension of length is much greater than the breadth or thickness. (1)

OPEN CIRCUIT - A normally conducting circuit which has been interrupted so that a current cannot be maintained through it. (24)

OR - The logic function represented by a circuit which has multiple inputs and one output, and whose output has the logical value 1 when any one or more of its inputs has the logical value 1. The output is 0 only when all of the inputs are 0. (54)

OSCILLATION - Rhythmic variation set up in or transmitted by a system possessing both elasticity and inertia or their electrical counterparts, capacitance and inductance. (1)

OSCILLATOR - A device, usually incorporating one or more amplifier circuits capable of generating electrical oscillations. (2)

OUTPUT IMPEDANCE - The total opposition offered by the output terminals of a device, circuit or system to the flow of current. A load equal to this output impedance when connected across the output terminals will result in a maximum transfer of power to the load. (26)

OUTPUTS - (1) The useful voltage, current or power delivered by a piece of apparatus such as a generator or amplifier. (2) The terminals from which this voltage, current or power is available. (21)

OUTPUT VOLTAGE, CURRENT OR POWER - The amplitude of these quantities at the output terminals of any device, circuit or system. (27)

OVERLOADING - The application of an excessive input, resulting either in distorted output or, in some cases, damage to the equipment. A familiar case is when the amplitude of the input signal to an amplifying circuit is so great that it results in a cutoff condition during a portion of each half cycle. (24)

OVERTONES - Those components of a complex sound having frequencies which are exact multiples of the fundamental. Overtone is a musicians' term for the more acceptable term HARMONIC. (4)

PAPER TAPE READER - A device for reading the digital information contained in the hole patterns in punched paper tape. Paper tape readers are always used in conjunction with punched paper tape handling equipment. This latter equipment is similar to magnetic recording tape transports, except that the medium is punched paper tape instead of magnetic tape. (57)

PARALLEL - Two or more circuits or circuit elements are said to be connected in parallel when the total current flow is divided between them. (17)

PARAMETER - A variable quantity, the value of which determines or affects the values of two or more other quantities or the relationship between them. The term is used in a variety of senses, the following being the more usual: (1) A variable quantity which may be kept constant while the effect of other variables is being investigated. (2) When two or more variables are all functions of some other variable, that other variable is termed a parameter. (4)

PARTIALS - Those sine components of a complex wave which are not harmonically related to the fundamental frequency. (4)

PASSIVE - A device, circuit or system is said to be passive when it does not require the application of DC power in order to perform its transfer function. This is in distinction to active devices, circuits or systems. (22)

PEAK TO PEAK - The arithmetical (not algebraic) sum of the positive and negative peak values of an alternating quantity. (21)

PHASE - When two alternating quantities have the same frequency but their maximum values do not occur at the same instant of time, they are said to have a phase difference. (7)

POLARITY - The distinction between the positive and negative terminals of a source of electromotive force. (19)

POTENTIAL - According to the concept of electrical potential, if an electron tends to move from one point to another, the point towards which it moves is said to be at a higher potential than the first point, or to be at a positive potential with respect to the first point. This first point, is thus at a negative potential with respect to the second. (14)

POTENTIOMETER - A potential divider. More particularly one in which the ratio between the parts of the divided potential is continuously variable by adjustment of the position of a sliding contact. (16)

POWER - The rate at which electrical energy is used. It is equal to Voltage multiplied by Current in DC circuits and to Voltage times Current times cos θ for AC circuits. (15)

POWER SUPPLY - Any circuit or device used in conjunction with active circuits for supplying DC power. (20)

PRIMARY - The terminals of a transformer into which a signal is introduced. It is in distinction to the secondary, which is the pair of terminals of a transformer from which energy is taken. (29)

PROTONS - Charged particles, one of the fundamental constituents of the nucleus of an atom. It carries a positive charge equal to the negative charge of an electron and has a mass about 1840 times that of an electron. (13)

PULSE GENERATORS - A specific type of electronics circuit in which the output has two voltage states, one of which is very long relative to the other per unit of time. (7)

PULSE SEQUENCE GENERATION - A technique for approximating complex waves by rapidly generating pulses of various heights. These sequences of pulses are made to approximate the irregularities of the desired complex wave. This is one of the two major methods for generating complex waves; the other being to use multiple sine wave generators. (45)

PUNCHED PAPER TAPE - A control system which uses holes punched into paper tape as digital information for programming the operation of equipment. (57)

RAMP - A specific type of waveform (sometimes called a sawtooth wave) whose amplitude changes linearily from zero to some voltage and then quickly returns to zero. (7)

RECTIFIER - Device the conductivity of which is unidirectional or at least asymmetrical, so that it is able to rectify an alternating current by suppression of alternate half waves. (20)

RESET INPUT - The terminal of a flip-flop which will change the states of the output terminals to the complement of their states when a signal is presented to the set input. (55)

RESISTANCE - Opposition to the flow of electric current exhibited by matter. In a solid conductor, resistance may be envisaged as the effect of collisions between charge carriers (electrons) and the more massive positive ions. The fact that in most conductors resistance increases with temperature is attributable to increased thermal agitation with corresponding increase in the number of collisions. In conductors whose resistance decreases with rise of temperature this is due to increased ionization (release of electrons) at the higher temperature. (15)

RESISTOR - Device composed of a substance having a specified resistance, and included in a circuit for the deliberate introduction of resistance. May be employed to produce a voltage drop or to determine the amount of current which can flow in the circuit. (16)

REVERBERATION - Acoustic effect at a given point due to the simultaneous arrival of sound waves of random phase, magnitude and direction resulting from multiple reflections. (43, 44)

RING COUNTER - A digital circuit having multiple outputs, only one of which has the logical value "1" at any given time. Each input pulse to the ring counter shifts the "1" output one position. (47)

RING MODULATOR - An electronic circuit having two inputs and one output. When two frequencies are applied to the inputs, there are four frequencies present at the output. These are the two original input frequencies, a frequency that is equal to the sum of the input frequencies and a frequency that is equal to the difference of the input frequencies. (43)

RISE TIME - The time required for a square wave or pulse to go from zero to 90% of its maximum value. (39)

R-S FLIP-FLOP - A set-reset type of flip-flop. It has two input terminals, the set and reset line, and of output lines, one of which always has the logical value 1 when the other has the logical value 0. When an input signal is applied to the set line the output lines change state. Further signals to the set line have no effect. When a signal is applied to the reset line, the output lines again change state. Further signals applied to the reset line will not change the circuit's state. (55)

SECONDARY - The terminals of a transformer from which energy is taken. It is distinct from the primary which is the terminals to which energy is applied. (29)

SEMICONDUCTORS - A material having electrical properties intermediate between those of good electrical conductors and those of insulators. Of particular interest are those whose conductivity varies with change of working conditions. The two most important types are Thermistors, whose conductivity varies greatly with change of temperature, the temperature ooefficient being negative, and the Unilaterally-conductive type which conducts electricity more readily in one direction than in the reverse direction, and thus has a rectifying action. All modern electronics is based on the control of semiconductor parameters, especially silicon and germanium. (13)

SENSITIVITY - The minimum signal which can be presented to the input of a circuit and which will produce an acceptable output. (26)

SEQUENCERS - Equipment frequently used in electronic music for the purpose of serially ordering sequences of sound events. Usually the number of sound events which can be ordered by this method is limited to 10 or 12. (57)

SERIES - Method of connecting the elements of an electric circuit so that the same current passes through them. (17)

SET INPUT - One of the input lines to a set-reset flip-flop. See R-S Flip-flop. (55)

SHORT CIRCUIT - A connection between two points by means of a conducting path of negligible resistance. (24)

SIMPLE HARMONIC MOTION - (s.h.m.) Form of mechanical vibration represented graphically by projecting on a diameter the successive positions of a point which travels at uniform angular velocity on a circular path. (3)

SINE WAVE - Waveform represented graphically by a curve, the amplitude of which at any instant is proportional to the sine of the angular displacement of a point which travels at constant angular velocity on a circular path; simple harmonic motion. (3)

SINGLE ENDED - Applied to inputs/outputs having two wires, one of which is at ground potential. (23)

SOFTWARE - A description of procedures for arriving at specific results. Sets of rules for arriving at specific results are sometimes called algorithms. When one has a description of operations to be performed, it is called software. And when equipment is built to implement these operations, it is called hardware. (59, 60)

SOUND - The sensation experienced when the ear is acted upon by vibrations within a certain range of frequencies. Also the vibrations themselves. (1)

SOURCE - When two devices, circuits or systems are connected together, the one which presents signals to the second is called the source. The device, circuit or system which receives the signals is called the load. (27)

SQUARE WAVE - Alternating current or voltage the waveform of which is approximately square or rectangular. (7)

STAIRCASE - A specific type of waveform in which all portions of the wave are vertical or horizontal. Each successive horizontal portion of the wave is at a higher voltage than that of the preceding horizontal signal. (7)

STEADY-STATE - Those parameters of a device, circuit or system which are unvarying with time during normal operation. (4)

SUSTAINED OSCILLATION - Oscillations the peak amplitudes of which never change, but always remain the same. (2)

SUB-HARMONICS - Frequencies which are submultiples of a given frequency.

TAPE RECORDERS - Devices for recording sound in the form of variations of magnetization along a continuous tape of ferromagnetic material. During recording the tape is drawn at a constant speed through the airgap of an electromagnet energized by the audio-frequency current derived from a microphone. Each point on the tape is thus magnetized to a degree proportional to the strength of the audio-frequency current at the instant when it passes a recording magnet or "head," the variations of magnetization along the tape thus corresponding to the waveform of the signal current. The sound is reproduced by passing the tape between the poles of a second electromagnet the coils of which are not energized. Changes in the magnetic flux in the core due to the varying intensity of magnetization of the tape induce a correspondingly varying voltage in the windings, and this voltage is applied to an amplifier and loudspeaker combination to reproduce the signal. Usually the tape is passed between the poles of an "erasing" magnet in order to remove any previous recording. Unless deliberately erased in this way, tape recordings are to all intents and purposes permanent. (31)

THREE-DIMENSIONAL - Any physical structure in which the three dimensions of length, breadth and thickness are of the same order of magnitude. (1)

TRANSDUCERS - Devices for transforming or converting power from one form to another, e.g. from electrical to mechanical, or vice versa. Thus, a microphone is a transducer which converts variations of air pressure into variations of electric current, and a loudspeaker is a transducer which transforms variations of electric current into variations of air pressure. (2, 31)

TRANSFER FUNCTION - A quantitative statement of the operation performed on an input signal to any circuit, device or system. (21, 27)

TRANSFORMERS - Devices consisting of two or more inductors in close physical proximity to each other, often but not necessarily wound on a core of ferromagnetic material. If an alternating, or variable, or intermittent electric current is passed through one inductor (the primary winding), corresponding alternating, varying or intermittent voltages will be induced in the other (secondary) winding or windings. The ratio of the e.m.f.s appearing at the terminals of each winding to that applied to the primary is equal to the ratio of the number of turns in the secondary to the number of turns in the primary. (21, 28)

TRANSFORMERS, MATCHING - Transformers coupling a source to a load and so designed as to ensure maximum energy transfer in spite of the fact that the load impedance differs from the source impedance. For optimum matching the square of the turns ratio of the transformer should equal the ratio of the two impedances. (29)

TRANSIENTS - Said of a phenomenon originated by a sudden change in conditions, and persisting for only a very short time after the change has taken place. In particular, the term is applied both as an adjective and as a noun to sound components of short duration and to the corresponding voltages and currents in an electronic sound system. (5)

TRANSISTOR - Device incorporating an arrangement of semiconductor material (silicon or germanium) and suitable contacts which is capable of amplification of an input signal. By use of auxiliary components various types of amplifiers, oscillators, and switching circuitry can be constructed. (21)

TRANSMISSION GATE - Also called analog switch. A device having one input, one output and one control input. The transmission of analog signals through the device is controlled by the application of either analog or digital signals presented to the control input. (56)

TRIANGULAR WAVE - A specific type of waveform which changes in a linear fashion from zero to some voltage and then decreases again to zero in the same manner. (7)

TRUTH TABLE - A matrix showing all of the possible input and output combinations for a digital circuit or logic function. (53)

TURNS RATIO - The ratio of the number of turns in the primary to the number of turns in the secondary or a transformer. (29)

TWO-DIMENSIONAL - Any physical structure in which the two dimensions of length and breadth are of the same order of magnitude and are much larger than the thickness of the structure. (1)

VARIABLE RESISTOR - Circuit component the resistance of which can be adjusted. Typical constructions include a carbon ring, or a high-resistance wire wound on a ring-shaped insulating former. Variation of the resistance is obtained by means of a radial contact arm. (17)

VIBRATION - The continuous change of state of any physical structure which is introduced by external means. Generally the change of a physical structure from a position of rest to some extreme position, a return to the position of rest followed by an extreme position in the opposite direction. (1)

VOLTAGE - Difference of potential expressed in volts. (14)

VOLTAGE CONTROLLED OSCILLATOR (VCO) - An oscillator having a control input which allows the frequency of the output to be controlled by the application of voltages to the control input. (40)

VOLTAGE DEVICE - Name applied to a device whose input resistance is high. (24)

VOLTAGE SOURCE - A device or circuit which is capable of supplying a required voltage. (16)

VOLTS - Unit of potential difference or e.m.f. The final velocity of an electron which, starting from rest, traverses a space across which a difference of potential of 1V exists, is 593 km/sec. (14)

VU METER - A standardized instrument which is found on most tape recorders for monitoring and controlling signal levels. It is used to measure complex waves. (32)

WATT - Unit of electrical power, equivalent to $\frac{1}{746}$ h.p. and corresponding to the rate at which work is done when the application of an e.m.f. of one volt results in the passage of a current of one ampere. (15)

WAVE ENVELOPE - The overall change in amplitude of the waveform during the duration of a sound. (10)

WHITE NOISE - An electronic output from a device or circuit which contains all frequencies and all amplitudes varying in a random fashion. (50)

WOW - Slow variations of pitch of sound reproduced by a tape recorder due to variation of the speed of the tape transport or other irregularities of the drive mechanism. (34)

ZERO LEVEL - The maximum input signal level, as measured on a VU Meter, which can be recorded without excessive distortion. (32)