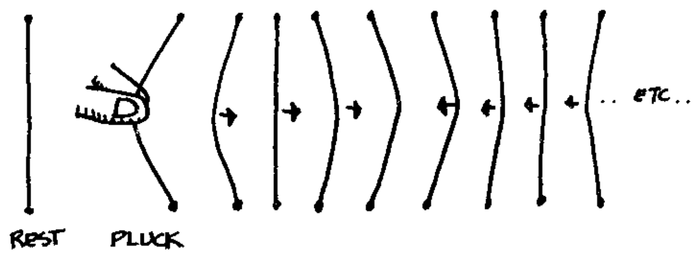
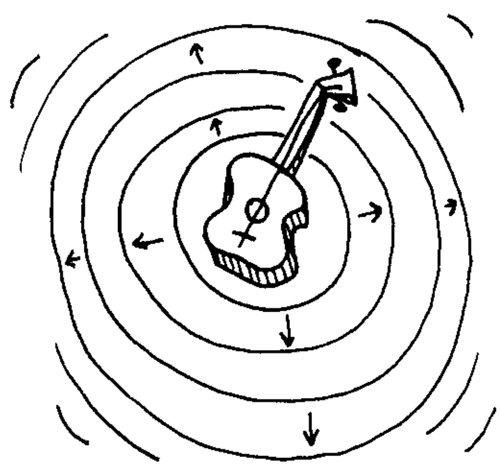
ELECTRONIC MUSIC THEORY

SOUND

Sounds are vibrations of the air caused by vibrating objects. Take a simple musical example--the string on a guitar. When it is plucked, it is pulled in one direction and released. Because it was under tension from the pulling, it snaps back to its original position and because of its momentum, it keeps going through its at-rest position to an opposite state of tension.



It proceeds to move back and forth, each time with a little less power, until it comes to rest in its original position. Almost all struck or plucked instruments vibrate in some variation of this action. When the string is released it pushes the air in front of it causing a slight extra compression of the air molecules or, put another way, a slightly higher pressure. This is called "compression". When the string flicks back, it causes a slight vacuum, or low pressure area. This is called "rarefaction". As the string vibrates back and forth more and more of these compression and rarefaction areas are created. They act like ripples in a pond, spreading out quickly and always at the same speed, the speed of sound.



If you are standing some distance away from the vibrating string when these ripples reach you, if there were some way of counting how many waves occur per second, many things could be told about the string itself! For one thing, because the speed of sound is constant, you would know how many times the string vibrates in a second. This number is called the FREQUENCY of a sound. The second thing you would want to determine is how strong the ripples are, that is, how compressed the compression wave is and how vacuous the rarefaction wave is. This strength is called the AMPLITUDE of the wave. When working with sound it can also be called the VOLUME or LOUDNESS of the sound. The Amplitude can tell you one or both of two things: how strong the source of vibrations was (i.e. how powerfully it could push air around) and/or how far away the source of the vibration is, because the amplitude of the ripples decreases with distance.

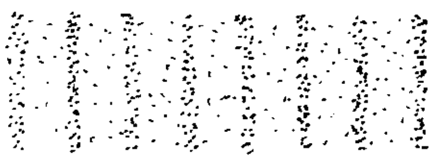
There are a number of other things we wish to detect about the sound waves that reach us. No object vibrates simply. Each has a characteristic "waveform" that, when perceived, can identify that object. This is called the TIMBRE or quality of the sound and is how we can distinquish a piano from a violin. We would want to detect these variations and have a sense of where the sound is coming from.

We perceive these complex waves with our ears. We hear different Frequencies as different PITCHES and we can hear them over the range of about 20 to 20,000 cycles (vibrations) per second.

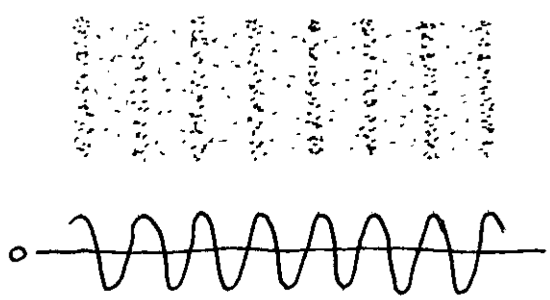
We perceive Amplitude as loudness, and remarkably, we can sense the amplitudes of rustling leaves or those of a jet plane. The jet produces compressions and rarefactions nearly one million times greater than the leaves!

Without going into much detail, this is the way the ear works: The pressure inside the human head remains constant (though adjusted to the normal pressure of the atmosphere of the air.) When there are no sound waves in the air, the eardrum is at rest between two areas of equal pressure. However, when a sound wave ripples past, with its fluctuating bands of high and low pressure, the eardrum is pulled slightly outward during a rarefaction wave and pushed slightly in by the high pressure part of the wave. This means that the eardrum is going in and out at the same rate (with the same frequency) as the original sound source. The eardrum's vibrations are transmitted by means of small bones to the cochlea, a spiral organ in the inner ear filled with a liquid and coated on the inside with millions of small hairs. Each of these hairs is connected to a nerve ending through which these signals are sent to the brain.

If we could take a picture of a small section of air through which a sound wave is moving, it might look like this:

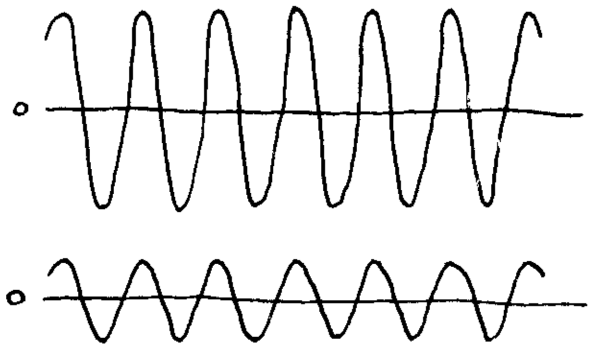


In this drawing each dot represents a few million air molecules, but even with this simplification it is a rather clumsy way of describing how a wave "looks". Here is a better way is to describe the "pressure" at each point of such a wave:

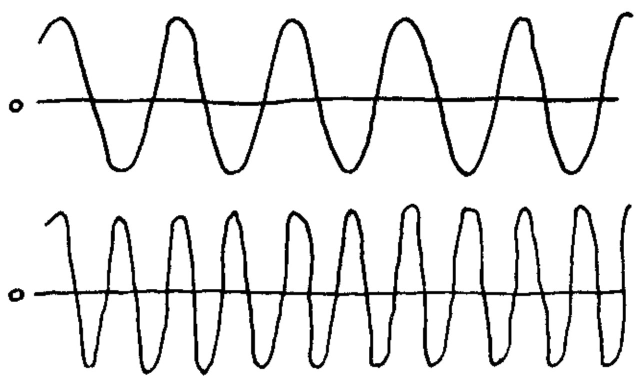


The line labelled "0" is normal pressure and the wavy line is a graph of the pressure of the wave. When the wavy line is above the 0 line, the pressure is greater than normal air pressure, when below the 0 line, it is less than air pressure.

Below are two sound waves drawn using pressure graphs:

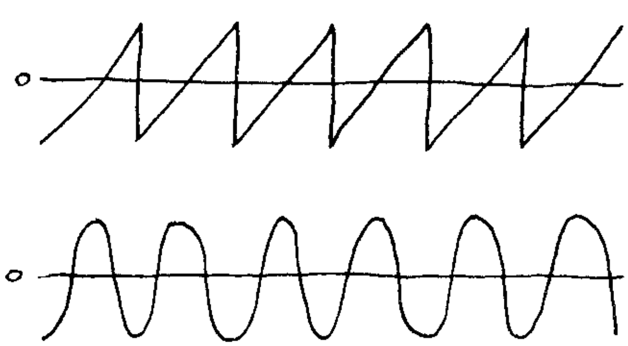


The difference between these two waves is that the top one goes further above and below the 0 line than the bottom wave. This indicates that its Amplitude or loudness is greater and is measured from "peak to peak", from the top of the highest peak to the bottom of the lowest trough. Below are two more waves.



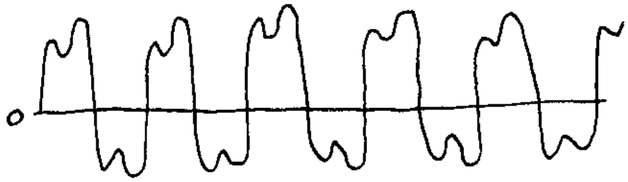
Notice that in this case the amplitude of the two waves is the same, but that in the same length of time there are twice as many excursions up and down in the bottom wave as in the top -- that is the bottom wave has twice the frequency of the top wave. The bottom wave will sound ONE OCTAVE HIGHER than the upper wave. If a wave has twice the frequency of another wave, we hear it as one octave higher. Notice that if the first octave starts out at 80 cycles per second (or 80 Hertz which means the same thing), then the next octave starts at 160 Hertz (twice the first), the third will start at 320 Hertz, the next at 640 Hertz, then 1280, 2560, and 5120 Hertz. Whereas the first octave had a range of only 160 cycles per second, the top octave had a range of 2560 cycles per second! But to our ear/brain both sound like a single octave.

Below are two waves:



These are two wave types that you will find on most synthesizers; the top one being a SAWTOOTH wave and the bottom being a SINE wave. The two waves in this drawing both have the same frequency and the same amplitude but a different SHAPE. The shape of a wave affects its TIMBRE or sound quality. Picture your eardrum being pulled in and out by the two waves shown above to see the difference in the kind of motion the liquid in the cochlea would have. In the real world, of course, nothing can vibrate in quite these shapes and if it could, the air cannot ripple in quite this fashion and if it could the eardrum cannot be moved in precisely this way. But it can all come remarkably close.

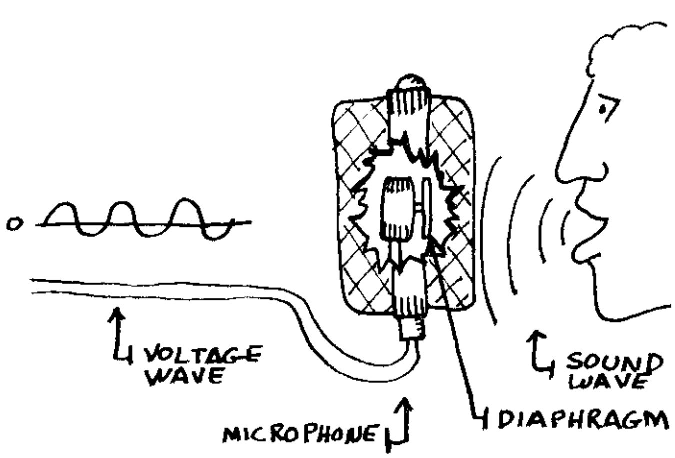
Below is what a guitar sound wave might look like:



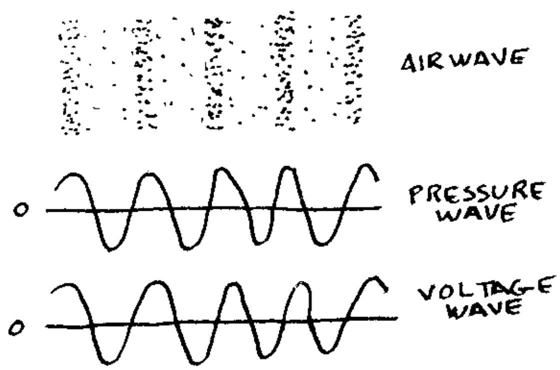
VOLTAGE

Voltage can be considered to be electric pressure. By the middle of the 19th Century, many of the advantages of converting sound waves (rapidly changing atmosperic pressure) into voltage were discerned. Primary among them was that while sound waves died out relatively rapidly, voltage waves could be sent thousands of miles over wires, around corners and through walls,. The main problem was how to convert sound waves into voltage waves and then, after a journey of perhaps a hundred miles, convert the voltage waves more or less accurately, back into sound waves. In other words, the problem was the invention of the telephone.

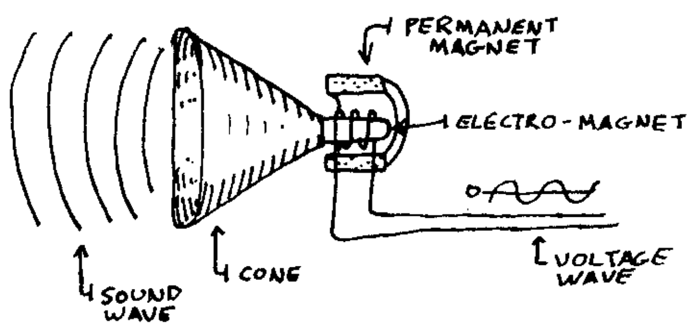
A Microphone is a device for converting sound waves into voltage waves, or atmospheric pressure into electric pressure. The simplest microphones have a diaphragm which acts much like the eardrum in its response to sound waves. It is pushed inwards by a compression wave and pulled outward by a rarefaction wave. This diaphragm is attached to a device which, when it is pushed inward creates a Positive Voltage and when it is pulled outward creates a Negative voltage. When the diaphragm is at rest, its output is Ground-- or 0 volts.



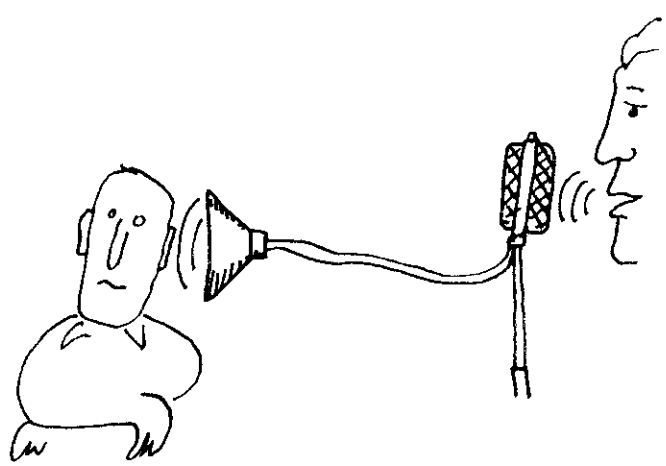
Because of this one-to-one correspondence the voltage output of a microphone is said to be isomorphic with the sound wave input.



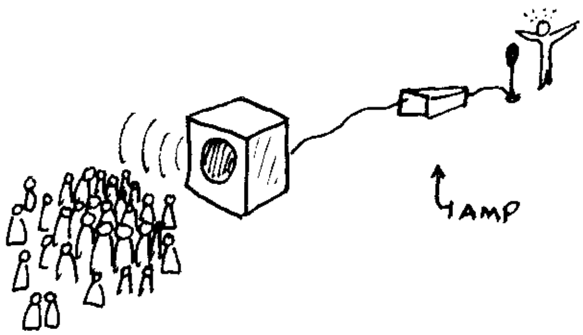
A Speaker is a device that takes a voltage wave and converts it into a sound wave. Though there are many kinds of speakers the most common ones work by moving a cardboard speaker "cone" with an electro-magnet.



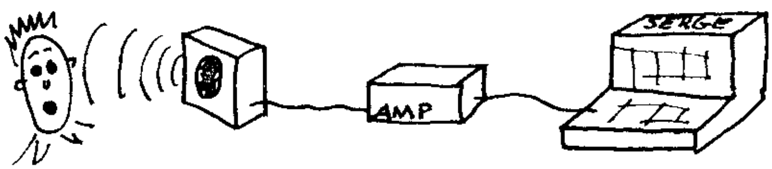
In this kind of speaker the coil of wire attached to the speaker cone sets up magnetic fields which push and pull itself in and out from the permanent magnet as the voltage changes, thereby pushing and pulling the cone in and out. This creates rarefaction and compression waves in front of the cone. The speaker cone, therefore, reproduces the movement of the diaphragm of the microphone and in so doing reproduces the original sound wave.



It was the ability of such a system to transmit sound over long distances that first attracted attention. It soon became clear that there were other advantages. Once the sound wave was converted into a voltage wave, it was far more malleable. It could be amplified, for instance, so that when the speaker re-created the sound it could be louder than the sound originally picked up by the microphone.

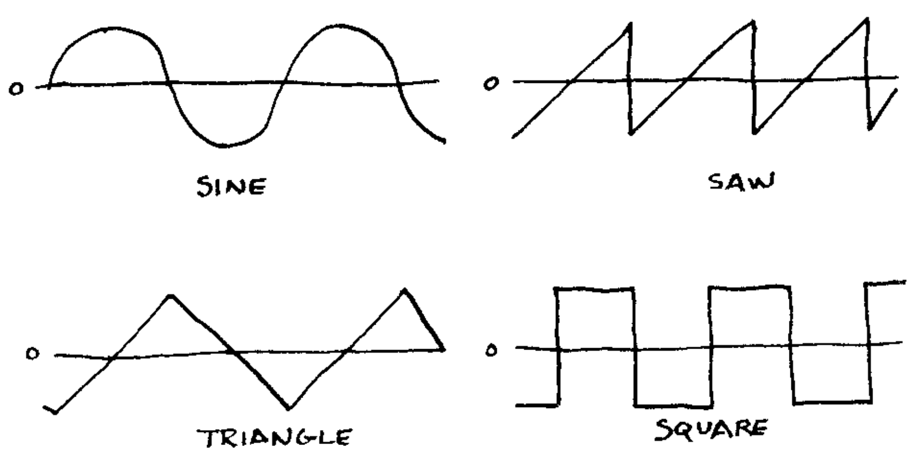


A speaker doesn't know where the voltages it is receiving are coming from. Its cone will move in response to any varying voltage. A SYNTHESIZER is a device which creates and sculpts voltages of various shapes that, when directed to a speaker, create sound that can be used in musical settings.



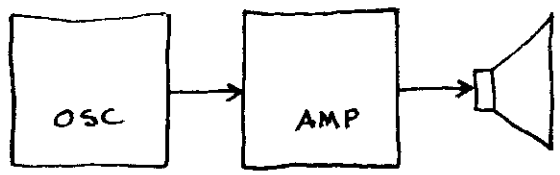
THE DEVELOPMENT OF THE SYNTHESIZER

From the earliest days of electronics, there have been various devices to create and alter voltages of audio frequency. We've already discussed the amplifier which takes an input of a varying voltage and puts out that same varying voltage magnified in amplitude. Another device is the oscillator, which simply puts out a varying voltage in a number of simple shapes:

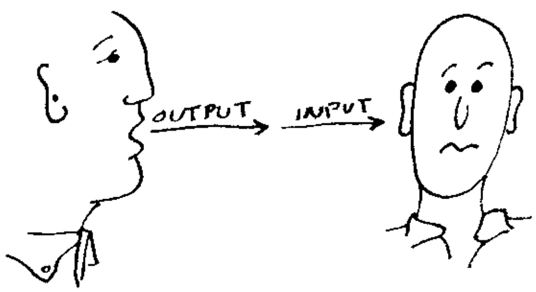


A knob, or POT (short for POTentiometer, which is the device the knob turns) on the front of the oscillator would determine the frequency of these waves, that is, how often in one second the wave would rise and fall.

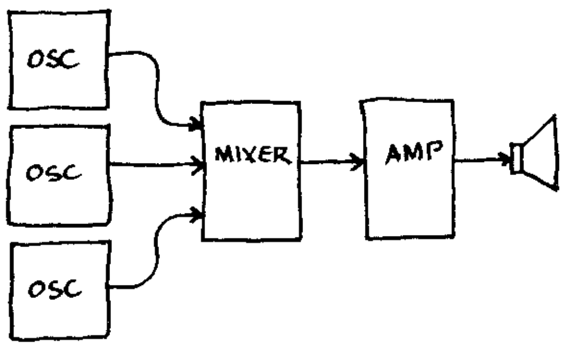
The first step towards electronic music was taken when the OUTPUT of the oscillator was connected, or PATCHED to the INPUT of the amplifier. The OUTPUT of the amplifier was sent to the speaker.



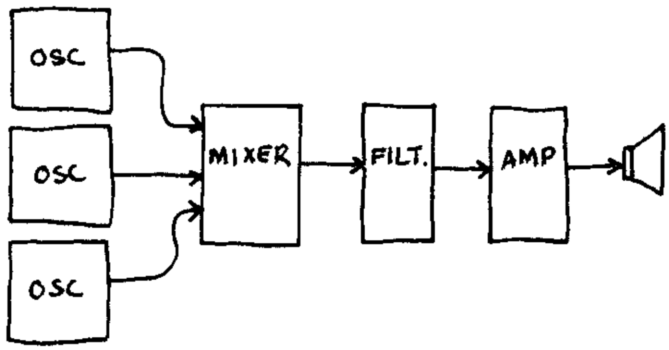
Note the "block diagram" used above. In this form of notation a block indicates an electronic device. The arrow coming out of a device is its output, while an arrow going into a device is its input. An output of one device is always the input to another device. The output of a speaker goes to the input of your ear. What outputs from your mouth inputs into someone else's ear.



Another device was the Mixer, which takes inputs and adds them together to produce a single output. Unlike the amplifier the mixer has more than one input.



Still another important device was the Filter. A filter is a device that can eliminate or accentuate various frequency components of a complex sound. For instance it can be used to eliminate all the very high components (the hiss) in a sound, by only allowing those frequencies in the range of the human voice to pass. A pot on the front of the filter controls which frequencies will be attenuated or eliminated.



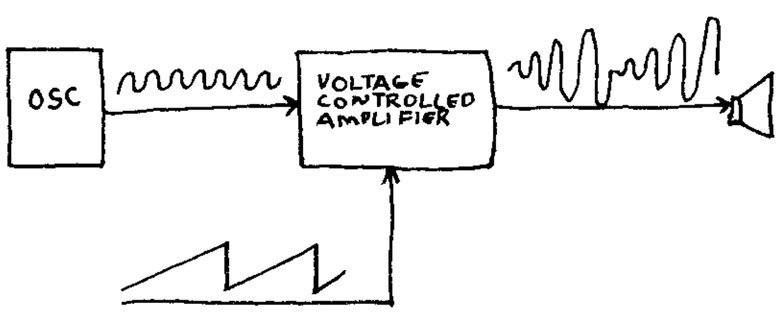
There were two problems with this procedure of adding device after device. The first was that very quickly there were just physically too many knobs to twiddle. The second problem was that the knobs couldn't be turned quickly or precisely enough. The amplifier could not be turned up and then quickly down again fast enough to make the "sound envelope" of a single whack of a drum.

The invention of the tape recorder, just after World War II, solved some of these problems. A single sound could be produced electronically, recorded on to a short piece of tape, and spliced onto another previously made sound and so on until a string of sounds had been made. Two of these tapes could be mixed together through a mixer and recorded on a third tape. The speed of the tape machines could be varied, and the segments could be reversed or even cut to form spliced "envelopes". This was (and still is) a very tedious process, but it is a very rich and flexible one. A studio built to be able to produce electronic tapes in this way is called a Classical Electronic music studio.

The first major improvement in the classical studio came from Columbia University where they devised a controller which could set all the dials instantaneously from the instructions given on a punched paper tape.

It wasn't until the Sixties that the synthesizer as we now know it was designed by Don Buchla and Robert Moog by adding Voltage Control to the classical studio.

The Voltage Controlled Electronic Music Synthesizer solved both of the two major problems of the classical studio by employing Voltage Control which works in the following manner: Each device is given a special input called a Voltage Control Input. This input accepts a voltage such that as this voltage INCREASES it is JUST LIKE TURNING UP THE KNOB ON THE FRONT OF THE DEVICE. And when the voltage goes down it is like turning down the pot on the front of the device. That is, a voltage can be used to CONTROL the device. For instance, in a voltage controlled amplifier, if the voltage at the voltage control input increases, it turns the amplifier up and makes its output louder.

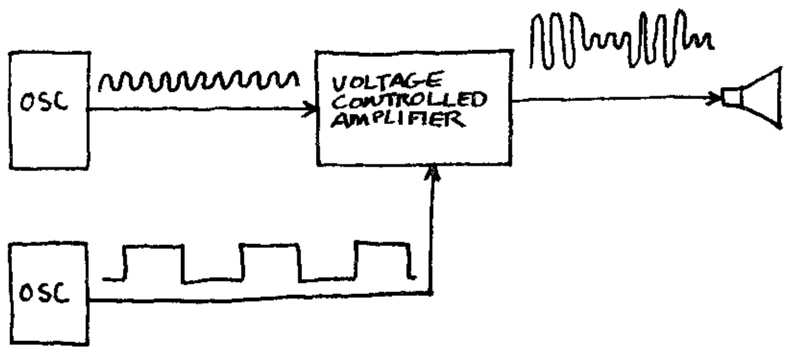


(Note that in these block diagrams, as a matter of convention, the control voltage input is on the bottom of the device, the "signal" input is on the left side and the output is on the right side.)

In a voltage controlled oscillator, a rising voltage at its voltage control input would make its frequency rise. For each device the control voltage affects only the function of that device.

Control voltages solved both problems of the classical music studio: With enough control voltages you could change all the settings of all your devices. And secondly you could change these settings so rapidly as to seem instantaneous. You could change the settings very, very slowly, or you could change them at audio frequencies, for instance 500 times per second. When a devices's settings are changed at those rates, some very strange things begin to happen, many of which can be musical.

The only problem left, of course, is where to get all these control voltages. This problem is not as great as it seems for a control voltage is identical to any other kind of voltage. For instance, we could use an oscillator to control an amplifier since the output of an oscillator is a voltage!



In the above example Oscillator #2 is controlling the amplifier, making the signal from Oscillator #1 louder and softer.

Most of the early synthesizers have two different sets of patch cords, one for the control voltages and one for the signals, even though the voltages themselves are indistinguishable. The Serge System does not make this distinction.

THE SERGE SYSTEM

The SERGE SYNTHESIZER is a Voltage Controlled Modular Music Synthesizer. By MODULAR it is meant that it is composed of separate devices or modules which must be patched together to produce a complex sound. By Voltage Controlled is meant that almost all of these devices can be controlled by a voltage as well as by their own pots. By music is meant that the Serge can be used to create complex, ordered sound, and by Synthesizer is meant that it needs no other input (though it is able to accept one) and that it can create, or synthesize, sound.

There are Four basic kinds of Modules on the Serge. Many modules can serve more than one of these functions:

SOUND SOURCES. The basic sound source is the oscillator though there are others such as white noise. Sounds from the external world, so long as they have been converted into appropriate voltages (by the use of microphones or pickups) can also be used as sound sources. Oscillators are completely voltage controllable.

SOUND PROCESSORS. Processors are devices that input one or more signals, operate on these signals, and then output a different but related signal. Mixers, filters, wave shapers, amplifiers are all processors. Almost all of these devices are voltage controllable.

CONTROL VOLTAGE SOURCES. Control voltage sources are devices that are used to create the voltages which are used to control other devices. The keyboard, for instance, puts out a voltage which can be used to control the setting of an oscillator. Other devices are envelope generators, sequencers, sample/hold devices and envelope followers. These devices are voltage controllable themselves, making possible complex levels of control.

CONTROL VOLTAGE PROCESSORS. These devices input a control voltage, operate on it, and output a related but different voltage. Processors and portamentoes are examples of these modules.

Each module on the Serge is surrounded by a border with the name of the device at the top and the Serge logo at the bottom. In some cases there is more than one device in a module and these are referred to as "dual" or "triple" modules. These dual or triple modules are two or three completely separate, though functionally identical modules.

Every module has at least one output. Outputs are usually enclosed within a border of their own.

All processor type modules have at least one input as well as an output.

Most modules have control voltage inputs which control the function of the module. These inputs are of two basic types:

PROCESSED INPUTS which have a pot associated with the input jack that can attenuate, amplify and/or invert the control voltage.

UNPROCESSED CONTROL VOLTAGE INPUTS affect the given module in a predetermined way.

Most modules have one or more pots that can control the function of the module without a control voltage. In most modules these pots control the basic setting of the module on which the control voltages operate. These pots are labelled with their function, for instance. GAIN, FREQ, etc.

The jacks on the Serge are color coded.

BLACK JACKS indicate audio or signal voltages.

BLUE JACKS indicate control voltages.

RED JACKS are pulse input/output jacks and are used to turn modules on and off, have them step through stages and control the timing of various functions.

OTHER COLOR JACKS are special jacks whose function will be described individually.