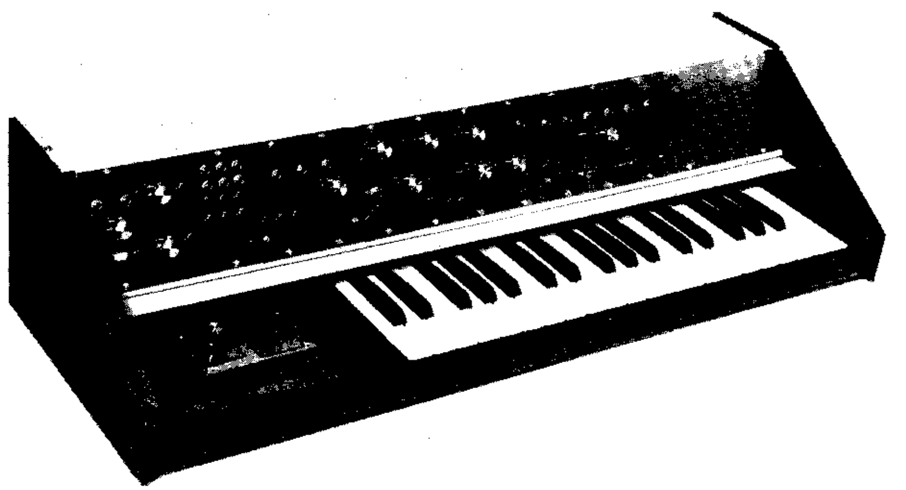
$1.00

USING THE PAIA 2720 SYNTHESIZER



There has never been a musical instrument that was conceived overnight and released to the world in an immutable state the next day, each instrument has undergone change and refinement to bring it to its present condition. The same is true of electronic music but the newness of this field coupled with the technology explosion has caused its development to be compressed in time. Whereas the piano has taken centuries to evolve, electronic musical instruments first appeared only a little over four decades ago.

Most of these first instruments were little more than exercises in technology but some were designed to overcome shortcomings in existing instruments. For instance, the piano keyboard is one of the most powerful musical operating systems available but it has one outstanding drawback in that while it provides the musician easy access to all twelve notes of the equally tempered musical scale it prohibits him from using the infinity of musical pitches between those twelve notes. By its very nature it eliminates the possibility of an easy glide from one musical pitch to another (glissando).

One electronic instrument designed to overcome this weakness was the Martinot. The Martinot is similar to modern organs in that a standard keyboard is used to control an electronic oscillator built around a frequency determining capacitance/inductance tank circuit. The inductor is tapped at points that produce frequencies corresponding to the chromatic scale; a rather straightforward, if somewhat simplistic, approach to electronicly generating a musical scale. In addition to the keyboard there is a finger ring attached to a slider that controls another oscillator. When properly adjusted this second oscillator produces the pitch corresponding to the keyboard key adjacent to the position of the ring. The combination of keyboard and slider allows the musician to glissando from one note to another or add vibrato with a simple move of the hand without sacrificing the operating ease of the keyboard.

The Ondioline was a contemporary of the Martinot but is significant because it was the first electronic musical instrument to use something other than a sine wave as its basic tone. In the Ondioline a relaxation oscillator controlled by the keyboard produced a sawtooth wave which in turn activated several frequency dividers. The output of the oscillator and frequency dividers were combined using much the same techniques employed in some modern organs so that the instrument was capable of generating a great variety of sounds. Observers report that a skilled operator could come close to making an Ondioline talk.

While the Martinot and Ondioline were both designed in France, America's contribution to freeing the musician from the restrictions of the keyboard was probably the most outstanding - not to mention bizarre. A Theremin has no visible means of control at all and is played simply by moving the hand in relation to two metal plates or rods. Inside the instrument are two high frequency oscillators, one shielded from any external influences and the second arranged so that the plate or rod forms part of the frequency determining inductance/capacitance tank circuit. The outputs of these two oscillators are combined in such a way that an audible tone that is the difference between the two frequencies is produced. As the performer's hand is brought closer to the sensing antenna the difference in the two frequencies increases and so does the pitch of the tone. A second circuit allows the performer's other hand to determine the volume of the sound produced. Since there are no frets or keys to provide visual or tactile clues to the pitch a Theremin will produce, it is a very difficult instrument to play - but loads of fun.

SYNTHESIZERS

The first equipment that would come close to meeting our current definition of a synthesizer was built by Dr. Harry Olson during the early 1940's* Produced under the auspices of the RCA Labs, the RCA Mark I and Mark II Synthesizers were something to behold. The Mark I has been disassembled for some time now but the Mark II still exists and is currently being leased to Columbia-Princeton Electronic Music Center, it measures 17 feet long by 7 feet high and is valued at anywhere between $250,000 and one and a half million dollars depending on who you're talking to.

* At the 1939 New York world's fair an electrical instrument called a "Voder" was demonstrated. By current definitions this would be one of the first synthesizers, unfortunately the Voder was intended to produce human speech and no one thought of it as a musical instrument.

The average performer might be a little disappointed in the Mark II today because even if there were some way to transport it to a gig, he would find when he got there that he couldn't actually perform a number. The Mark II was simply not capable of real-time operation, each characeristic of the sound the instrument was to produce was laboriously calculated and plotted ahead of time and the result punched into a roll of paper tape. When it came time for the Mark II to do its thing the tape was fed in - like a very large, very expensive player piano - and the results recorded on a multi-track disc, (early 40's, remember, recording tape wasn't so hot in those days.) When all the parts of a number had been recorded on the seperate tracks of the disc they were re-recorded on another disc from which a master was made.

You might think that about the only thing that the Mark I and Mark II did that was of any consequence was add the word Synthesizer to our vocabulary but that's not the case at all. They were significant first of all because they were the first to put it all together as far as electronic music production was concerned. All the oscillators, amplifiers and filters needed in one place at one time and best of all some means - no matter how cumbersome - of controlling them all. Secondly, they were the first instruments to utilize white noise sources as part of an electro-musical instrument. White noise will be covered in detail later, for now it should suffice to say that without it sounds like snare drums and cymbals, to mention only two, are impossible.

Don't get the impression that electronic music cannot be produced without a synthesizer, that's not true. Imagine that you are in a laboratory with all sorts of electronic equipment such as oscillators, filters, amplifiers, modulators, tape recorders, etc. You turn on one of the tape recorders and set the oscillator for the pitch you want, twiddle the knobs of the amplifier to shape the loudness countour and play with the filter knobs to adjust timbre. It only takes about six hands and a couple of minutes but when you're through you've got a whole note recorded on the tape. Repeat the process often enough and you've got a whole string of notes. Of course, the tempo is not right and the notes may not be in the right sequence but you can fix that by snipping the tape apart and editing out all the junk before splicing it back together again to produce the desired melody. Now you go back and do the same thing for bass, rhythm and all the other parts. About the only thing you can say for this technique is that it should certainly give you a feeling of accomplishment. Considering the complexity of the process even such monstrosities as dogs barking out the tune of "Away in a Manger" can be forgiven - all that knob twiddling has to do something to a person's mind.

In the early 1960's Dr. R. A. Moog (recognize the name?) began developing and producing a line of electronic music synthesis equipment that revolutionized the field. The feature that made the Moog equipment such a quantum jump in ease of operation sounds almost ridiculously simple, but its implications are so far reaching that it must be stressed: THE KEY PARAMETERS OF THE PROCESSING ELEMENTS ARE A FUNCTION OF THE SUM OF SEVERAL CONTROL VOLTAGES RATHER THAN THE POSITION OF A KNOB.

As an example of the operating ease of voltage control let's see what it does for a relatively simple processing element, an amplifier. As we shall see a little later, one of the things that contributes most to the way an instrument sounds is the manner in which its sound builds up and dies away. When using the classical tape splicing technique these characteristics have to be duplicated manually for every note by turning the volume control of an amplifier. Even though the Mark II allowed for automatic control of the amplifiers, information still had to be punched into its programming tape for each individual note.

With voltage control the job of setting the correct time varying amplifier gain can be turned over to an automatic electronic function generator circuit that produces a repeatable, pre-set voltage waveform each time a key is pressed. This voltage is then used to control the amplifier. The musician sets the function generator to reproduce the characteristics of some real or imagined instrument and the electronics will produce that characteristic for each note he plays. If he desires a totally different sound it's simply a matter of re-setting a couple of knobs. Summing the control voltages allows the performer to produce more than one effect from a single processing module. If, in the above example the operator decides to add a low frequency amplitude modulation (tremolo) to the sound, he needs only to sum a second voltage that is changing at the rate of the desired tremolo into one of the remaining amplifier control inputs. As the control voltage varies up and down so does the gain of the amplifier and the volume of the sound.

THE SOUND OF MUSIC

Anyone can make weird noises on a Synthesizer simply by randomly making connections and pushing buttons. It's even fun for the first hour or so, until you begin to think of specific sounds you want to make and can't. If we're going to learn to use a Synthesizer rather than just play with it it's important that we understand what sound is and what makes one sound different from another.

If your knowledge makes the following discussion seem trite, read on anyway. We have to start somewhere and if nothing else you can probably find something to disagree with.

Sound travels as waves, waves of pressure in the air. A vibrating string displaces the air around it and the air molecules that the string moves in turn bump into and move other molecules. All the things that these sound waves can bump into and be reflected off of and the effect that this has on the original wave are beyond the scope of our discussion. The only thing relevant to the subject at hand is that if a man is present the pressure of the waves will finally cause a deflection of his eardrum which in turn will vibrate three small bones inside his ear which will in turn cause a disturbance in a fluid medium which in turn excites the auditor nerves which in turn causes the man to say " Hey, listen to that ______. " Whether he fills in the blank with "noise" or "music" is personal preference.

The thing that vibrates to produce sound doesn't have to be a string. It can be a synthetic or organic membrane as in a drum, a vibrating reed as in the wind instruments or the lips of the musician in the brass instruments. Most important to us, it can also be the cone of a loudspeaker.

When a recording of a musical instrument is made a microphone converts the air pressure waves into exactly analogous electrical voltage waves. If you were to graph the vibrations of the air and the "vibrations" of the voltage side by side they would be identical except that one would be measured in volts and the other in dynes per square centimeter - or something. When these voltage variations are re-played through an amplifier and loudspeaker they are converted from electrical back into sound energy. If all the links in the chain have been faithful in their recording and reproducing functions the pressure waves generated by the loudspeaker will be exactly the same as those originally generated by the musical instrument and the two will be indistinguishable.

Since the thing that an amplifier and loudspeaker works with is not really sound but an electrical analog of sound; and since it is possible to electronically generate any imaginable voltage waveform (difficult in some cases, but possible), it seems only logical that at some point sounds should be generated not by physical musical instruments but by <u>synthesizing</u> their electronic analog and then converting that to sound.

PITCH, DYNAMICS, TIMBRE

There are really only three characteristics that determine what a musical instrument will sound like: pitch, dynamics and timbre. Of the three, pitch probably requires the least explanation.

<u>Pitch</u> and frequency are two words from two different technologies that describe the same thing. When an engineer or technician speaks of 261 Hz. they mean that the thing they are referring to is vibrating 261 times per second. When a musician mentions

middle C he is also talking about something that is vibrating 261 times per second. If the musician is dealing with conventional instruments he is probably talking about a string or reed but if he is working with an organ or Synthesizer he is likely referring to the same thing that the technologists were talking about, the frequency of the changes of an electrical waveform.

The human ear is more sensitive to changes in pitch than any other musical parameter. The intensity of a sound has to be cut significantly before a listener experiences any decrease in loudness but a skilled musician can tell when a musical semi-tone deviates by as little as 3% of the interval between that note and the next higher tone.

Dynamics is a broad term that refers to the time varying intensity characteristics of the sound; how fast it builds up and how fast it dies away.

The length of time required for a sound to build up to its greatest intensity is called attack time and this one parameter conveys more information about the way an instrument is played than any other. If the attack time is very short the instrument will be in the percussion family where the vibrating member is immediately excited to its maximum amplitude by the deforming action of being plucked or struck with a hammer or mallet. If the attack is relatively slow then the instrument is probably in the reed or bowed string groups where the action of the exciting force - the wind or bow of the performer - takes a short time to fully excite the vibrating element.

If you forget about the talent factor for a moment the primary purpose of the musician in playing most instruments is to serve as an energy source. The performer pumps energy into the system (instrument) and the system dissipates it in some way, usually either as sound or heat. I know what you're probably thinking. Heat? Yes, heat; if you were able to accurately measure the temperature of a drum head you'd find that it gets hotter as you pound on it. The energy that is converted to heat can be thought of as being lost since it does not contribute to the primary object of producing sound.

Very interesting, right? But what has this to do with the sound of a bassoon. Just this, another important characteristic of an instrument is its decay time. That is, how fast the sound dies away. Decay time is directly related to how much of the energy goes into heat and how much into sound. A vibrating string, for instance, is as close to lossless as you can get and it's decay time is very long. The stretched membrane of a drumhead on the other hand is very lossy and as a result the decay time of drums is very short.

Reed instruments have a short decay time because the reeds are relatively lossy and don't continue to vibrate for very long after the musician stops adding energy. Brass instruments have the shortest decay time because the performer can force his lips to stop vibrating and the column of air in the instrument is very lossy.

Sustain time is the interval in between attack and decay, the steady state response of the instrument. As is obvious, percussion instruments have zero sustain time - as soon as the attack is finished there is no more energy input so it's downhill the rest of the way. Instruments that have some continuous energy input from the performer, in the form of bowing, blowing or even pedaling in the case of some organs, can sustain as long as the energy holds out.

Though attack, sustain and decay are the primary phenomena of dynamics there is one other condition that is common enough to merit a seperate section. When a percussion instrument is struck very hard the vibrating member will deform beyond the point at which a smooth decay is possible, in effect more energy is put into the system than it can handle, with a resulting overload. Under these conditions the system (string, membrane or whatever) will rapidly get rid of the excess energy. With the "overload" dissipated the vibrating element will continue to dissipate the remaining energy in a normal fashion. The result is an initial rapid attack immediately followed by a release time which is then followed by a normal decay. In a natural instrument it would be all but impossible for the release time to be followed by a sustain interval but with a synthesizer this is simple.

We can graphicly illustrate the conditions discussed by plotting the overall intensity of the sound versus time as shown below. Since these graphs are drawn to show the peak amplitude of the sound at any given time and therefore "contain" the sound they are often referred to as envelopes.

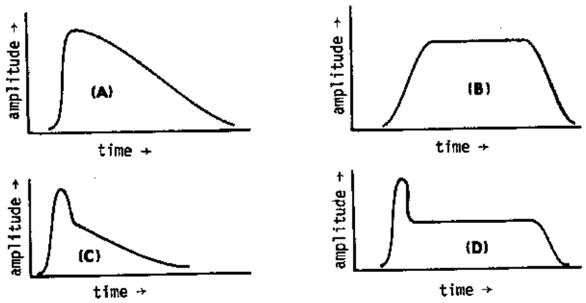


FIGURE 1. Amplitude envelopes for (A) percussion (B) reeds (C) attack-release-decay (D) attack-release-sustain-decay.

It is pretty obvious that as important as dynamics is, it doesn't account for all the differences between the sounds of instruments. For instance, the trumpet and french horn are both brass instruments with approximately the same attack, sustain and decay characteristics. They even overlap as far as pitch range is concerned but there would be little danger of mistaking the blaring, brassy sound of the trumpet for the mellow, muted tones of the french horn. These differences come about because no musical instrument produces a tone that is composed exclusively of a single frequency. Each note is composed of a number of different frequencies, and the number and amplitude of the various components are what gives each instrument its distinctive timbre.

The concept that a single musical pitch can be made up of more than one frequency can be confusing and needs further attention. The sine wave shown in figure 2 is the basic building block of any imaginable accoustic or electrical wave. It is the only waveform that is composed entirely of a single frequency and, more importantly, any waveform can be built up using nothing but sine waves.

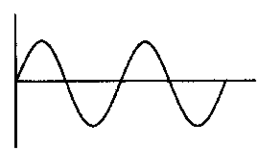


FIGURE 2. Sine wave

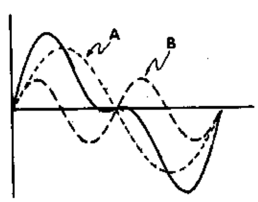


FIGURE 3. Fundamental and 2nd. harmonic

To illustrate this look at figure 3. Here we have two sine waves drawn in dotted lines which are labeled "A" and "B". As you can see from the drawing, waveform "B" goes through two cycles in the time that it takes waveform "A" to complete a single cycle. Waveform "B" is therefore twice the frequency of "A" and is said to be the second harmonic of the fundamental frequency "A". If we draw another wave that was three times the frequency of "A" it would be the third harmonic, four times would be the fourth harmonic, five times the fifth and so on.

If at every point in time we sum together the amplitudes of waveforms A and B the result is the waveform shown by the solid line. Note that while the new wave is shaped differently than either A or B it has the same frequency (and consequently pitch) as the fundamental frequency A. If third, fourth, fifth and higher order harmonics were added into this wave the result would continue to change shape but the frequency would remain the same.

It is not necessary that every harmonic of a fundamental frequency be included in a wave and indeed the most musically interesting sounds have certain harmonics deleted. The square wave shown in figure 4 is a good example. It is difficult to imagine that the sharp-edged wave illustrated could be built up from smoothly changing sine waves but it can as shown in the progression of diagrams figure 5 (a) through (c). In (a) a fundamental frequency is added to its third harmonic producing the waveform shown by the solid line. In (b) the fifth harmonic has been added to the result of (a) to produce the new solid waveform and in (c) the seventh harmonic has been added to all the rest. You can see that the trend as higher order harmonics are added is to steepen the sides of the square and flatten and reduce the ripple in the top. When enough harmonics have been added the result will be a square wave. Notice in particular that not all harmonics are added together for a square wave, only the odd harmonics (3rd, 5th, 7th, etc.) are included.

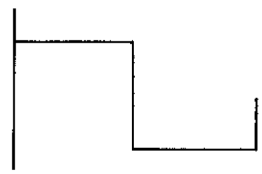


FIGURE 4. Square wave

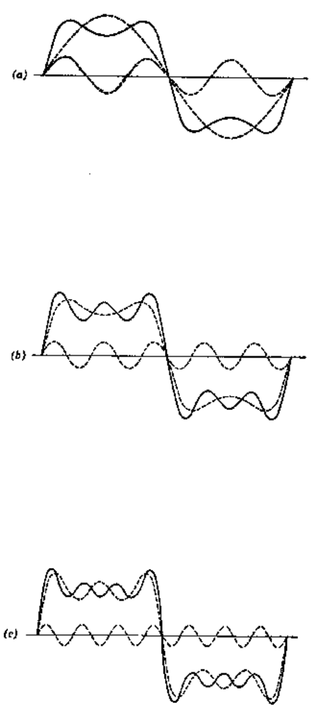


figure 5.

MAKING WAVES

Now that we have a pretty good idea of why instruments sound the way they do we can begin looking at ways of duplicating these sounds using electronic circuits.

The first method of electronicly producing a desired waveform is called frequency synthesis and the technique should be obvious from the discussion of harmonic structure. Several oscillators provide a source of sine waves of various harmonically related frequencies and combinations of the outputs are summed together to build up the desired waveform. By changing the amplitudes of the sine waves practicaly any waveform can be easily produced. One of the problems with this system is keeping all of the oscillators tuned so that they are multiples of one another. Most electronic organs that use frequency synthesis systems get around this problem by using one oscillator for the highest frequency component desired and then producing the other frequencies using a chain of frequency dividers.

The technique used in synthesizers is called formant synthesis and can be thought of as just the opposite of frequency synthesis. Rather than summing together the frequencies you do want, you start off with a source that is already rich in harmonics and then remove the ones you don't want. This may seem a rather strange way to get from here to there but there is an excellent biological precedent for formant synthesis, the most versatile musical instrument of all - the human voice.

There are other reasons for using formant synthesis than just pleasing mother nature. If we are going to be consistent in our design of a line of voltage controlled equipment then everything should be voltage controlled, including the oscillators. Designing a voltage controlled sine wave oscillator is not impossible but it is difficult and hard as it is to design one VCO (voltage controlled oscillator), designing the five or six that would be required using frequency synthesis and having them all track the control voltage would be a horrendous task.

Since synthesizers operate with harmonic rich waveforms as their primary signal source there is no need to start out with a sine wave at all. The VCOs supplied with most synthesizers provide a variety of waveforms each of which provides different harmonic structures. Common practice is to use a relaxation oscillator to generate a voltage ramp which is then converted to triangle and pulse waves using simple shaping circuits. In some cases the triangle will also be shaped into a sine wave. These waveforms and their harmonic contents are listed in table 1.

wave / harmonic	triangle	ramp	square
fundamental	$8/\pi^2$	$2/\pi$	$4/\pi$
2nd	---	$1/\pi$	---
3rd	$8/9\pi^2$	$2/3\pi$	$4/3\pi$
4th	---	$1/2\pi$	---
5th	$8/25\pi^2$	$2/5\pi$	$4/5\pi$
6th	---	$1/3\pi$	---
7th	$8/49\pi^2$	$2/7\pi$	$4/7\pi$
8th	---	$1/4\pi$	---
9th	$8/81\pi^2$	$2/9\pi$	$4/9\pi$

Table 1. Harmonic content of triangle, ramp and square wave
$\pi=3.142$ $\pi^2=9.872$

In order to use formant synthesis we need some means of getting rid of the harmonics we don't want and to do this we use filters. A filter is quite simply an electronic gadget that eliminates a single frequency or group of frequencies.

Figure 6 shows diagramatically a representation of the frequency response of a low pass filter. This drawing shows that as the frequency of the signal being fed to the input of the filter increases the amplitude of the filter's output falls off. Note that the filter does not change the frequency of the input signal, only the amplitude. If the input is a complex waveform the filter will of course change the signal's shape as it attenuates the higher frequency components but that is, after all, what we're after.

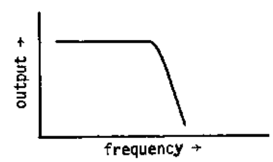


FIGURE 6. Low-pass filter

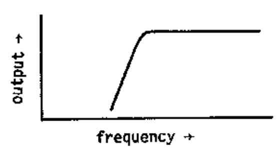


FIGURE 7. High-pass filter

Figure 7 shows the frequency response of a high pass filter. In this case the amplitude of the output falls off as the input frequency decreases.

Notice that in both of these filters the response curve is flat either up to or beyond some definite frequency. This is the frequency at which the filter begins to "take hold" and is designated the cutoff frequency or f_c. One other important parameter associated with low pass and high pass filters is the roll off rate, ordinarily measured in units of db/octave. This sounds complicated but it's really not. A decibel (db) is a measure of electrical level and when you're talking about voltage, a change of 6 db. corresponds to a halfing (if -6db.) or doubling (if +6db.) the original reference level. Octaves are of course frequencies that are double some reference frequency; therefore a low pass filter that "rolls off" at 6db/ octave simply means that every time the frequency is doubled the output of the filters falls by 1/2.

Figure 8 shows the frequency response of a band pass filter. As the diagram implies, a band pass filter attenuates all frequencies above and below a certain frequency while allowing the frequency of interest (or frequencies close to it) to pass without being effected. The frequency that is allowed to pass without attenuation is quite logically called the center frequency and is also designated f_c. There are parameters that can be used to specify how well the filter does its job of rejecting frequencies outside of its pass band but none of them are very easy to understand and for our purposes we will confine ourselves to speaking of the "Q" (quality) of the filter. The higher the "Q" the greater the frequencies outside the pass band will be attenuated.

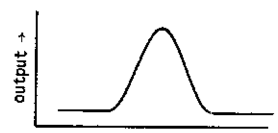


FIGURE 8. Band-pass filter

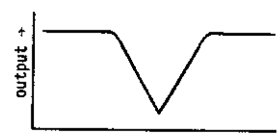


FIGURE 9. Notch filter

As the frequency response curve of figure 9 shows, you can think of a notch filter as being the opposite of a band pass filter. Instead of allowing frequencies around the center frequency through, the notch filter blocks these and allows all others to pass.

CONTROLLERS

It is about time that we looked at a problem that has plagued instrument makers since the first caveman beat on a hollow log - how to control the instrument in such a way that you realize its full potential. With most conventional instruments the control system is obvious. You control some of the elements of the dynamics by how hard you blow, pick, or strike the instrument and you control the pitch by the positions of your hands and/or lips. Timbre is in most cases a quality of the instrument and is therefore beyond the control of the performer.

This is not the case with a synthesizer; you have at least the theoretical capability of controlling and varying every characteristic of the sound. Some characteristics you can pre-set by the position of a knob and some you can turn over to automatic function generating equipment. Some parameters are varied with a manual controller such as a keyboard and some, unfortunately, you wind up forgetting about because there are no more controllers available.

Before examining some of the types of controllers that are available, make sure that you have firmly implanted in your mind that a controller for a synthesizer does only one thing; it provides a voltage proportional to some parameter that is physically changed by the performer. While in most cases the voltage produced by the controller will subsequently be used to set the pitch of a VCO, this will not always be the case.

Depending on the sound being produced, the controller may also be used to set the center frequency of a band-pass filter, roll off rate of a low pass filter or any number of other things.

KEYBOARDS

When used to control a piano, a keyboard is one of the great inventions of all time. When used with a synthesizer it is at best a compromise.

Musicians are used to keyboards being connected to polytonic instruments; that is, instruments that are capable of playing as many notes at one time as the number of keys that the performer is able to press down. This simply isn't the case with a synthesizer. A synthesizer is a monotonic instrument capable of playing only one note at a time. The reason for this is obvious, the voltage control oscillator may accept three control voltages but the pitch that is produced is proportional to the sum of those voltages. Since it is impossible that there be more than one sum at one time the oscillator can produce only one pitch. By using a rather clever switching system one brand of synthesis equipment is able to produce two different notes simultaneously (using two VCO's) but this is really not such a great improvement.

Since the electronic organ has become commonplace, performers have gotten used to the idea that they can't control the dynamics of their instrument by varying the striking force on the keys, this is also true of most synthesizers. Other than triggering signals that are generated when any key is pressed, the only control voltage that most keyboards produce is proportional to the location of the key being activated. One manufacturer has a keyboard that is an exception to this rule; in addition to the standard control voltage it also generates two voltages proportional to the velocity of the key as it is pressed down and the final pressure on the key as it is held down. This is a significant improvement since it allows the performer to directly influence three musical parameters by pressing a single key.

Strangely enough, the original objection to a keyboard that was mentioned in the first part of this booklet (unavailability of pitches between semi-tones) is not a great problem on a synthesizer. Most keyboards provide a "pitch" knob that allows some variation in tuning of the instrument and many provide for an automatic, variable rate glissando.

In spite of its drawbacks, the standard keyboard has one big thing in its favor - familiarity. It is similar to a thing that the musician already knows hot to use and re-training time is therefore reduced.

LINEAR CONTROLLERS

These are electrically and mechanically the simplest of all controllers. Most consist of a long strip of electrically resistive material with a voltage applied to each end. The potential difference between the two ends distributes evenly along the length of the strip so that the voltage between any point and electrical ground is proportional to the position of that point on the strip. When the performer presses on the controller, a conducting metal band makes contact with the resistance element and picks off the voltage present at the point of contact.

Linear controllers are generally not intended as substitutes for keyboards for a number of reasons. First, it is technically difficult to automatically produce a trigger pulse whenever the controller is pressed. This function has to be performed manually with a seperate switch that must be closed for each note or run that is to be played. Secondly, using a linear controller for pitch is like playing a fretless instrument such as a violin, it requires considerable experience to know what pitch is going to be produced at a given location.

These devices come into their own when used in conjunction with a keyboard. In this application they can provide an auxilliary control for some parameter other than pitch, like manually sweeping a filter or controlling the amount of noise mixed into a sound. The control voltages produced by this unit can also be summed into one of the VCO control inputs to produce a manually controlled glissando or vibrato or can be used with a VCA to give manually controlled tremolo.

FOOT PEDALS

Foot pedals allow you to control additional musical parameters with your feet. They are similar to the expression pedals on electronic organs except that instead of controlling only the volume they can be used to control filters, oscillators or amplifiers.

Anything you can say about linear controllers applies to foot pedals, they're intended to be used in conjunction with a keyboard.

JOY STICKS

These are the wackiest controllers imaginable and as you would expect are similar to the joy sticks used in airplanes.

The biggest thing going for this type of controller is that is offers the possibility of directly controlling four musical parameters simultaneously. One parameter could be controlled by moving the stick forward and backward, another by moving the stick from side to side, a third control voltage could be generated proportional to vertical motions (along the long axis of the stick) and a fourth proportional to the rotation of the handle. If you like you could even put a switch on top to control such vital functions as self-destruct.

A joy stick seems like a valid concept but anyone that could use one properly probably wouldn't be able to communicate with earth people.

FUNCTION GENERATORS

Function generators are automatic controllers that electronically generate a time varying voltage as pre-set by the positions of knobs or sliders. A function generator ordinarily responds to a trigger pulse by generating an electrical wavefrom that rises to some pre-set value in a pre-set time, sustains that level as long as the trigger pulse is present (or for a pre-set time in some cases) and then falling back to zero in a pre-set time. Some function generators are capable of producing the attack, release, sustain, decay type functions discussed earlier.

The output of the function generator can be used in the same ways any other control voltage source should, but these items find their most common application in controlling dynamics and time varying timbral qualities of a sound.

A low frequency oscillator can also serve as a control voltage source to provide cyclicly varying voltages for vibrato, tremolo or filter sweeping.

SEQUENCERS

Sequencers can be thought of as extremely versatile function generators. Instead of the control voltage going up to a pre-set level and back down again these devices allow a complete sequence of output voltage to be pre-programmed and reproduced on command.

DIGITAL COMPUTERS

If you are prone to wild flights of fancy here is a subject worth your consideration. It seems that whenever any of the literature mentions the use of a digital computer in conjunction with a synthesizer it always comes out as a super-sophisticated sequencer, a high technology replacement for a roll of paper tape. What a ridiculous waste of the immense computational capabilities of modern mini-computers. Imagine instead, a machine that could "hear" the sound of an instrument, analyze it and reproduce that sound for any melody the performer chose to play. If you have a touch of larceny in your soul how about a machine that could hear a person's voice and then duplicate his exact tones for any spoken phrases, or a machine programmed with compositional algorithms that could compose tunes to fit your mood. The list is endless.

THE EQUALLY TEMPERED SCALE

This is as good a place as any to bring up the subject of the equally tempered or chromatic scale. As anyone who is reading this knows there are 12 semi-tones in each octave of the chromatic scale, 7 naturals labeled A through G and 5 accidentals that are designated as either sharps or flats of the naturals. With two exceptions the sharp of one note is identical to the flat of the next highest note, the exception being that there are no accidentals between B and C, or E and F so that $B^{\#}$ is the same as C and F^{b} is the same as E.

For each octave increase in the musical scale the frequency of the note doubles so that since middle C corresponds to 261.6 cycles per second the next C above middle C is 523.2 cycles per second.

Somewhere back in antiquity (around the time of J.S. Bach) some genius decided that since there are 12 semi-tones to the octave and each octave doubles the frequency, each note should be related to the note directly below it in the scale by a factor of the twelfth root of two. Just in case you're not used to working out the twelfth root of numbers in your head, this translates to 1.059 times the frequency of the note directly below it. The significance of this is that as pitch increases, the <u>difference</u> between adjacent notes in the scale also increases.

All this may seem like academic trivia until you realize one point. All voltage controlled oscillator designs produce a device whose output frequency is directly proportional to the control voltage, identical control voltage changes produce identical frequency changes.

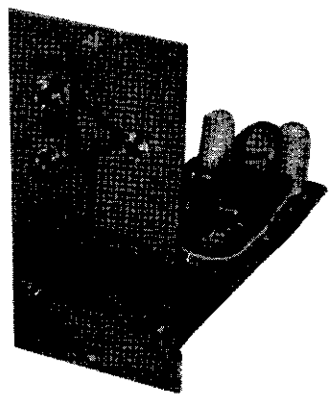
An example will most readily demonstrate the significance of these facts. Suppose that we have a keyboard that produces a control voltage of .625 volts when its lowest C key is pressed. The voltage corresponding to the next C up is quite logically 1.25 volts but don't fall into the trap of thinking that the voltage corresponding to the third C is 1.25 plus .625 or 1.875 volts because it's not, it should be twice the voltage required for the second C or 2 X 1.25 = 2.5 volts.

Many synthesizer designers use an electronic conversion device to get around this difficulty. This device converts a linear controller output voltage (1v. for the first C, 2v. for the second, 3v. for the third C, etc.) to the octavely related voltage required by the VCO. This is an excellent approach if you are willing to spend the money to do it because it allows two oscillators to be a fixed number of semi-tones apart and still track a control voltage in such a way that they maintain an equally tempered relationship.

Unfortunately, the exponential converter circuits (as these devices are known) are not only expensive but also quite often they tend to drift so that even for a fixed input voltage the output voltage (and of course pitch of the VCO) wanders from one value to another.

A simple means of getting around this is to have the keyboard generate octavely related voltages in the first place but you of course sacrifice the capability of having two oscillators track each other.

2720-1 VOLTAGE CONTROLLED AMPLIFIER



SPECIFICATIONS

Power Requirements:	18v. @ 2.5 ma. 9v. @ less than 500 microa. sink
Output Impedance:	1K short circuit protected
Input Impedance:	nominal 47K
Max. Audio In:	2v. peak to peak
Frequency Response:	1Hz. to 40kHz.
Gain Characteristics:	@ 5v. control; 0db. input; 0db. 3db. input; 3db. @ 0v. control 0db. input; -80db. 3db. input; -80db.
Control Input Imp.:	150K

The biggest concern in the design of a Voltage Controlled Amplifier is that none of the control voltage transfer into the audio channel. If there is a leak between the two, very rapid changes in control voltage (such as the fast attack of percussion waveforms) will become audible as clicks or thumps. The design of the 2720-1 is such that the control voltage appears identically in two seperate amplifiers and is then balanced out by an operational amplifier output stage. Nothing special here, this is the way practically all synthesizers do it.

The 2720-1 differs from many synthesizer modules in the way the control voltages are summed together. In expensive equipment this summation is performed by an active network built around an operational amplifier. In the 2720-1 the summing is performed by three resistors.

The advantage of this approach is obviously cost - the disadvantage (in a technical sense) is that the summation is never exact and the voltage applied to one input can have a small influence on the effect of a voltage applied to another input. If the resistors sum into a point that has a low impedance in relation to the summing resistors then this effect is for all practical purposes negligable. Our justification for using this slightly less exact summing network is that, as was mentioned in the text, the human ear is less sensitive to changes in intensity of a sound than any other parameter and even a trained listener would be able to tell the difference between active and passive summing networks.

An explanation of the control jacks follows:

<u>CONTROL</u> The gain of the amplifier is set by the algebraic summation of the control voltages that are present at the three pin jacks along the bottom of the panel. The

design of the amplifier is such that if the voltages at the control input add up to +5v. there is zero insertion loss between the 0db. audio input and the audio output. Voltages less than +5v. cause the amplifier to present a greater and greater attenuation to the audio signal until at a total control input of 0v. the amplifier can be considered off. Negative control voltages and voltages greater than +5v. will not damage the amplifier but if the control voltage input goes higher than 6v. distortion will develop.

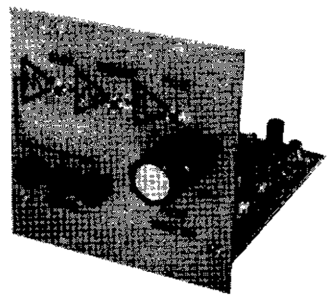
0db. INPUT At the maximum control voltage of +5v. there is no gain or attenuation between this input and the audio output.

3db. INPUT At the maximum control voltage of +5v. there is a 3db. power gain between this input and the output. Both of the inputs may be used simultaneously for mixing putposes. These two input terminals have a slight D.C. potential and must be capacitively coupled to audio sources. All audio sources associated with the 2720 Synthesizer already have capacitive output coupling but if electrified musical instruments are being processed this capacitor must be supplied externally. Two coupling capacitors have been provided on the patch panel of the 2720-7 power supply module for use in cases such as this.

INTERNAL CONTROLS

Balance - allows for exactly balancing out any control voltage feed-through into the audio channel. Screwdriver adjust.

2720-2 VOLTAGE CONTROLLED OSCILLATOR



SPECIFICATIONS

Power Requirements:	+9v. @ 25ma. -9v. @ 25ma.
Outputs:	ramp, triangle, pulse
Output Impedance:	1K for each output
Output Level:	.5v. peak to peak
Pulse Duration:	variable up to 60% duty factor
Frequency Range:	40Hz. to 2.5kHz.
Control/Freq.:	linear
Control Imput Imp.:	150K

The VCO is the one place in a synthesizer that you can't scrimp because as mentioned earlier, pitch is the parameter of a sound that the human ear is most sensitive to. Control voltage summation has to be exact and oscillator drift must be kept to a minimum.

In order to provide exact control voltage summation the 2720-2 uses an active summing network built around an integrated circuit operational amplifier. Frequency drift caused by voltage variations are minimized by using two independent voltage regulators directly on the 2720-2 circuit board.

There is one place that cost was cut in the 2720-2 and that is by the deletion of a sine wave output. Several factors entered into the decision to eliminate this common and useful waveform but in the final analysis they all boiled down to the added cost and space requirements necessary to produce the sine versus the ease of using the voltage controlled low pass filter (2720-3L) to derive the sine wave from the triangular wave.

CONTROL These three input jacks accept voltages from keyboard, linear controller, control oscillator, function generator, etc. and set the frequency of the oscillator such that it is directly proportional to the algebraic sum of the three inputs. Normal range of the sum of the three inputs is 0 to 5 volts but the oscillator will track up to 100% overange. Negative voltages and voltages greater than 5 volts will not harm the module.

RAMP The ramp output of the VCO produces a waveform such as that shown on the front panel. This waveform has a very "reedy" sound.

TRIANGLE The triangle output produces a waveform such as shown on the front panel. This waveform has the softest sound and is comparable to the voice of a flute.

PULSE The pulse output produces a waveform variable between a short pulse and a square wave. This is the "raspiest" of the three waveforms and is an excellent harmonic-rich source for use with voltage controlled filters.

PULSE DURATION This control knob varies the duty factor of the pulse output. At the "min." setting the output is just a spike and at "max." the duty factor is better than 50%.

INTERNAL CONTROLS

Zero - a control that regulates the frequency produced by the oscillator with zero input voltage. Screwdriver adjust.

Range - to regulate the amount of frequency variation caused by a given control voltage change. Screwdriver adjust.

Pulse - to regulate the duration of the pulse generated when the DURATION front panel control is set to "min.".

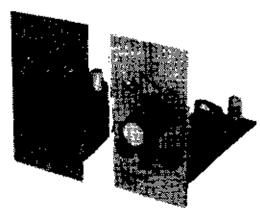
Triangle - to adjust the spectral purity of the triangle wave.

2720-3L & 2720-3B FILTERS

SPECIFICATIONS

band-pass

Power Requirements:	18v. @ 2.5 ma.
Input Impedance:	nominal 500K
Output Impedance:	1.5K
Center Freq. Range:	300 Hz. to 1.2kHz.
Control Voltage Range:	0 to 5v.
Control Input Impedance:	470K
"Q" Range:	1/2 to 300



low-pass

Power Requirements:	18v. @ 2.5ma.
Input Impedance:	100K
Output Impedance:	2K
Cut-Off Freq.:	50 Hz.
Roll Off Rate:	0 to 12db/octave
Control Voltage Range:	0 to 5v.
Control Input Imp.:	150K

It may seem strange for a manufacturer to state in a sales booklet that his product doesn't have the best technical specifications available, but that's what we're doing. We looked at a lot of filters while we were designing the 2720 series modules and some of the specs were enough to make your head swim; center and cut-off frequency ranges running up to 20 octaves, filters that tracked a control voltage with errors of only .02%, phenomenal "Q's" and lots more.

But when we started playing with these filters we discovered an interesting thing, you pay a lot for specifications and in most cases the added usefulness is not proportional to the added cost. The average user is after the sound that he can produce with a synthesizer and is not so interested in being able to exactly calculate a mathematical analysis of the sound.

Based on this, compromises have been made in the filters. For instance, the range of the band-pass filter is limited and a professional user able to compare the 2720-3B to the $300 filters of a large synthesizer could easily tell the difference. Similarly the filters will not exactly track the keyboard voltage.

What the filters will do is provide you with an economical means of controlling the harmonic content of a synthesized sound and allow you to change the harmonic content using control voltages generated by control oscillators, function generators, or manual controllers.

Operation of the band pass filter controls is as follows:

INPUT The miniature phone jack in the upper left hand corner of the front panel provides a high impedance input to the filter.

OUTPUT The miniature phone jack in the upper right hand corner of the front panel provides a capacitively coupled output from the filter.

CONTROL The three pin jacks along the lower edge of the front panel allow the summation of up to three control voltage sources. As the algebraic sum of the inputs to these jacks increases, the center frequency of the filter is shifted up. The highest possible center frequency is achieved when the algebraic sum of the inputs is 5 volts but negative voltages and voltages greater than 5 v. will not harm the unit.

"Q" CONTROL The "Q" control in the center of the front panel adjusts the width of the pass band and how much the center frequency will be accentuated. "max. Q" provides the narrowest possible pass band and the highest rejection of frequencies outside that band.

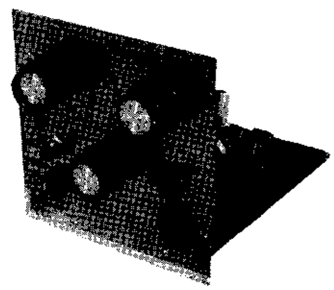
Controls of the low-pass filter are as follows:

INPUT The miniature phone jack in the upper left hand corner of the front panel provides a high impedance input to the filter.

OUTPUT The miniature phone jack in the upper right hand corner of the front panel provides a capacitively coupled output from the filter.

CONTROL The three pin jacks along the lower edge of the front panel allow the summation of up to three control voltage sources. As the algebraic sum of the inputs to these jacks increase the roll off rate of the filter increases. The highest roll off rate of 12db/octave is reached when the control voltages sum to 5v. but negative control voltages and voltages greater than 5v. will not harm the unit.

2720-4 FUNCTION GENERATOR



SPECIFICATIONS

Power Requirements:	18v. @ 4 ma.
Outputs:	5v. p-p fixed 0 to 5v. p-p var.
Output Impedance:	less than 5K
Output Form:	linear attack exponential decay
Time Range:	2ms. to 1 sec. attack 5ms. to 1 sec. decay
Trig. Required:	3v. step or pulse into 47K load

A common feature of many function generators is the attack-release- sustain-decay output used synthesizing certain percussion sounds. A single 2720-4 module does not offer this function - not because it is not desirable but because the cost of adding it would almost double the price of the module while only adding one feature.

If there were no way to produce the attack-release-sustain-decay waveform other than using a special module we would have designed the module regardless of the cost, but as it happens this function can be generated using two 2720-4's. The cost of two modules is only slightly greater than what the cost of the more elaborate single module would have been and the versatility added is great.

Figure 10 shows how this is accomplished. The 2720-6 Controller has two trigger outputs, a short duration pulse that is generated whenever any key is pressed and a voltage step that is turned on when a key is pressed and off when the key is released. The pulse can be used to trigger one of the function generators that is responsible for the attack and release portion of the wave. The second 2720-4 is

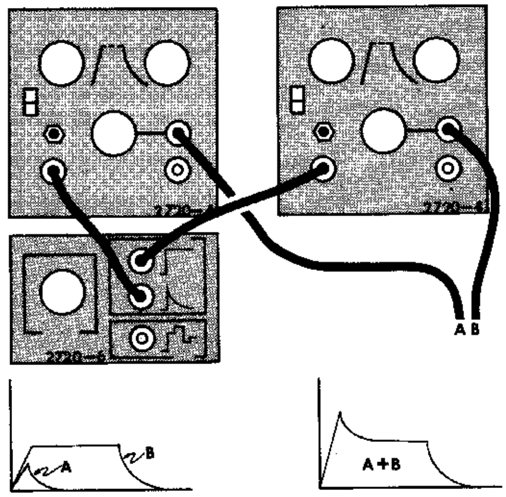


figure 10.

triggered from the step output and produces the sustain and decay part of the wave as well as contributing to the attack. The outputs of the two function generators are run to the control inputs of a single module (VCA typically) where they are summed together as shown in the drawings. Note particularly that the setting of the variable output attenuator of the second module determines the sustain level and that the peak level of the attack is the sum of the output of the two function generators, for optimum performance this sum should not exceed 5 volts.

Not only do you have a greater range of control over the function when two modules are used but you have the capability of using them seperately. For instance, one to generate dynamics while the other makes time varying timbral changes.

Operation of the controls is as follows:

ATTACK The attack control sets the rate at which the output of the function generator rises to its maximum voltage. Rotating this control in a clockwise direction increases the amount of time required for the rise.

DECAY The decay control sets the rate at which the output falls back to zero. Rotating this control in a clockwise direction increases the amount of time required for the fall.

EXPAND The attack time control is set up with two overlapping ranges. In the "expand" position of this slide switch the range of the attack control is from 30 milli-seconds (.03 sec.) to 1 second. In the "off" position the attack is variable from 2 ms. (.002 sec.) to 40 ms.

VARIABLE OUTPUT The pin jack marked "variable" is controlled by the knob immediately to its left. When this control is rotated fully counter clockwise there will be no output at the "variable" jack. Rotating the control in a clockwise direction increases the maximum value of the control voltage available at the "variable" output up to a maximum of 5 volts.

FIXED OUTPUT The pin jack marked "5v. p-p" has no variable control. At this output the control voltage will always rise to 5v.

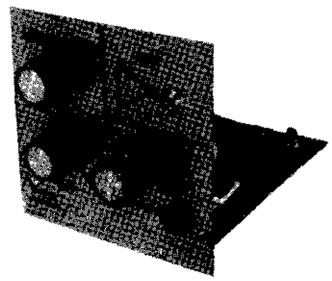
MANUAL TRIGGER The manual trigger push button is provided for testing of the function generator and to provide a means of triggering the module without using a controller. When the "manual" button is pressed the output of the generator will rise at a rate set by the "attack" control and remain at that level until the button is released. The manual trigger button should not be used when another trigger source is plugged into the "trigger" pin jack, no damage will result but the output of the 2720-4 will not meet full specifications.

ELECTRICAL TRIGGER The pin jack marked "trigger" provides an electrical means of triggering the function generator from keyboards or linear controllers. A 3v. input is required at the trigger jack to initiate and sustain the function generating process.

2720-5 CONTROL OSCILLATOR/NOISE SOURCE

SPECIFICATIONS

Power Requirements:	18v. @ 1.75 ma.
Oscillator	
Output:	Sinusoidal
Output Impedance:	5K or less
Output Level:	fixed 5v. p-p var. 5v. p-p
Output D.C. Bias:	50% of peak
Freq. Range:	1 to 25 Hz.
Noise Source	
Output:	broadband noise
Output Impedance:	1K
Output Level:	.5v. p-p



Control Oscillator/Noise Source may seem like a strange combination but there is sound reasoning behind it. Most importantly, neither or these two circuits is likely to be repeated within any one system; one of each is necessary but usually sufficient. Secondly, the required circuitry and front panel controls of the two lend themselves to the combination. Both require only single module circuit boards but the oscillator needs a double module front panel to properly arrange the controls. The single output jack of the noise source can be put just about anywhere.

The noise source has been mentioned as being necessary for the synthesis of snare drums, cymbals, wind and surf but it is instructive to explain exactly what noise is.

If you turn on an FM radio and set it between two stations you will hear noise of the type produced by the 2720-5. This familiar hissing sound is the result of summing together all audio frequencies each of which has its own random amplitude variations.

Color references are ordinarily used as a qualitative measure of the frequency content of the noise. If all frequencies are distributed within the signal (as in the FM example) then the noise is called white - drawing a parallel between it and white light, which consists of all colors. If only the lower frequencies are included in the signal the noise is referred to as pink. Many synthesizers provide "color" control directly on the noise source front panel but this is to a certain extent redundant since the voltage controlled filters can serve the same purpose.

The control oscillator can be used as a trigger source for the function generator to produce a combined element capable of producing a wide range of repeating waveforms.

Operation of the controls is as follows:

COURSE FREQUENCY The course frequency adjust is the control in the upper left hand corner of the front panel. Three over-lapping frequency ranges allow the oscillator to generate any frequency in the 1 to 25 Hz. range.

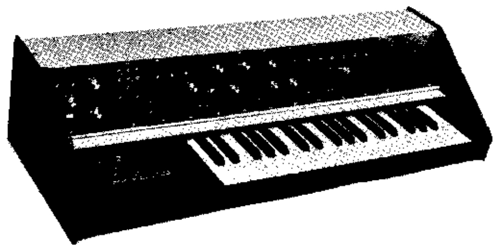
FREQ. The freq. control allows the continuously variable selection of any particular frequency within the range selected by the course frequency control

VARIABLE OUTPUT The output jack marked "variable" provides an output voltage that is continuously variable between 0 and 5v. peak to peak, adjustable by means of the control immediately to the left of the jack.

5v. p-p This output jack provides a non-adjustable source of control voltage that is always 5v. peak to peak.

NOISE The noise output jack provides a capacitively coupled source of . 5v. peak to peak to peak broadband noise.

2720-8 KEYBOARD/CASE



SPECIFICATIONS

Case Dimensions	31" X 11-1/2" X 7-1/2"
Power Requirements:	-9v. @ 1. 5 ma. 9v. @ 3 ma. 18v. @ 1 ma.
Configuration:	Keyboard arranged for three octaves of key controllable voltage C through C. Pitch knob provides additional octave.
Control Out Range:	to 5v.
Control Buffering:	Sample and hold. 20 sec. min. hold time after key released.
Tuning:	Each key tunable
Trigger Outputs:	5v. pulse when any key pressed 5v. step while key held down.

As was mentioned before, any keyboard controller for a synthesizer is going to be a misleading compromise to those that are unable to adjust their thinking and talents to a new medium.

Many professional musicians are initially confused when they begin to work with synthesizers, because the keyboard is only capable of playing one note at a time. The synthesizer keyboard must be thought of as a controller device, much like a knob or switch. The only difference is that the output voltage is variable depending on where along the keyboard you press the key. The keys farther to the right produce a higher output voltage, but there is only one output jack, and only one voltage can be produced at a time. The synthesizer as a whole must be considered as a single note instrument much like a trumpet or saxaphone. Although this may initially seem like a restriction, you will become familiar with many ways to achieve multiple voicing, and more complex multi-note sounds through the use of advanced patching. These types of effects depend on your experimentation and familiarization with the equipment.

The pulse (_/_) and step (_/‾) outputs may be used simultaneously. Note that these functions are generated every time a key is pressed with the step remaining high until the key is released. All keys must be released before another trigger pulse will be generated but pressing one key immediately before one already down is released will not keep the control voltage output from changing to the new value.

In many cases the end of the step function from the keyboard is the signal for the electronics to begin processing the decay portion of a note. Since the key must be released to end the step there must be some way to hold the last controller output voltage while final processing takes place. This task is performed by a sample and hold circuit. In the 2720-8 the sample and hold circuit not only holds the last voltage but through a unique "clocked" sampling circuit actually compensates for any corrosion on the key contacts that might cause controller malfunction.

The pitch control can be set to any intermediate value between the extremes of its octave range and the keyboard will still be in tune with itself; that is, the keys will still produce voltages that represent an equally tempered scale.

While pressing two controller keys simultaneously will not generate both notes, it will produce a new note that is pitched somewhere between the two. This peculiarity of the keyboard may be used to produce many interesting non-chromatic melody lines.

Operation of the controls is as follows:

KEYS Pressing any key causes a pre-set voltage to appear at the control voltage output jack. In most cases these voltages will be set to generate a chromatic scale but other tunings are possible.

PITCH The pitch knob on the control panel allows the entire keyboard to be lowered an octave from the standard tuning. Counter-clockwise rotation of the knob decreases the output voltage for any given key.

STEP The uppermost pin jack on the control panel provides access to a voltage that changes from 0 volts to 5 volts when any key is pressed. This voltage remains at the high level as long as the key is held down.

PULSE The middle pin jack provides access to a short duration trigger pulse that is generated when any key is pressed. All keys must be released for the pulse generator to re-set before it can produce another pulse. Re-set time is short, on the order of one-ten-thousandth of a second (.0001 sec.).

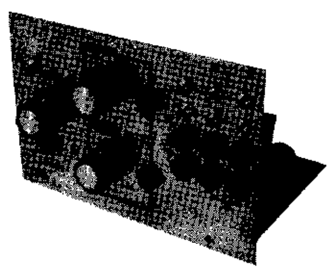
OUT The lower pin jack provides access to the control voltage output.

INTERNAL CONTROLS

Tune - 35 individual trimmers are provided for tuning each of the keys over a range of several semi-tones.

Low End Trim - This control, accessible from the rear of the case, is used to calibrate the overall keyboard span to the required 3 octaves.

2720-7 POWER SUPPLY



SPECIFICATIONS

Power Requirements:	117v. 60Hz. 3w.
Output Voltages:	18v. @ 100ma. 9v. @ 100ma. -9v. @ 100ma.
Protection:	power line fuse
Regulation:	none
Bias Outputs:	0 to 5v. and -5 to +5v.

The 2720-7 does more than simply supply power to the rest of the modules.

Occasions arise where the output of standard control voltage sources do not lend themselves to the effect desired. For instance, there will be many times that you want to shift a filter over only part of its range. The high control voltage end of the range can be decreased simply by using an attenuator but if you want to bring up the low end you need to "bias" the filter by summing in a constant control voltage. For cases such as this the power supply module has two variable voltage bias supplies. These bias supplies also provide a means of manually controlling or very slowly varying voltage controlled parameters.

There are times when the output of a control voltage source will need to be routed to more than one processing module - as when the control voltage is not only setting the pitch of the VCO but also the parameters of one of the filters. The 2720-7 provides for this with a multiple access patch panel consisting of six pin jacks and six miniature phone jacks. One of the features of the patch panel is a switching system that conditionally connects together all of the phone jacks and the first row of pin jacks. Inserting a phone plug in the first jack of either row of phone jacks isolates that row from the row directly below it. The phone jacks and pin jacks can be used as isolated rows or in a number of combinations. The placement of two capacitors in this panel allows their use as isolating elements when patching external musical instruments into the console.

An isolated front panel attenuator can be used as a master volume control, mixing level adjustment or as an attenuator for external instruments.

Controls are as follows:

BIAS The 0 to 5 volt and -5 to +5 volt bias supplies are controlled by the two knobs in the middle of the panel. Clockwise rotation of the control knob increases the voltage available at the pin jack immediately to the right of the control.

ATTENUATOR The knob on the left hand end of the panel is the control of a 5,000 ohm linear potentiometer. The high side of the pot goes to the left hand miniature phone jack and the wiper is capacitovely coupled to the right hand jack.

POWER SWITCH The slide switch in the lower left hand corner of the module supplies 117v. a.c. to the power supply.

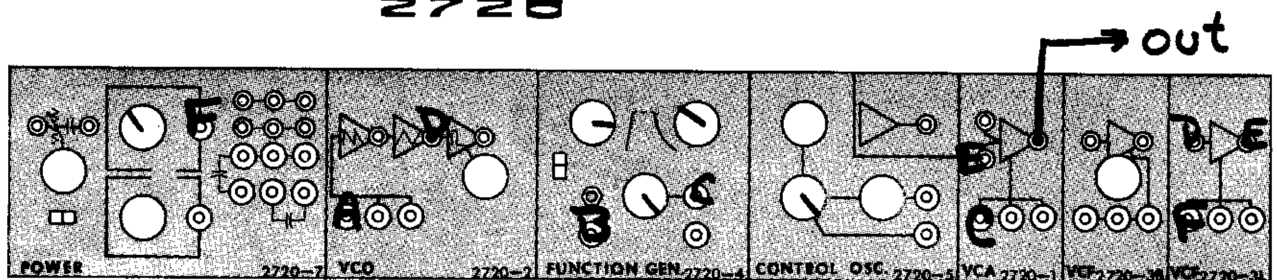
PATCH PANEL The patch panel consists of six pin jacks and six miniature phone jacks on the right end of the panel. All three of the miniature jacks in the upper row are permanently wired together. The first row of jacks is wired to the second row of jacks through a set of "conditional" contacts on the left hand jack in the top row. Inserting a plug into the left hand jack of the top row isolates that row from the second row. In a similar manner, inserting a plug into the left hand jack of the second row isolates that row from the third row (but not from the first row.) Capacitive coupling of certain of the pin jacks is as shown by the standard capacitor symbols on the front panel.

The following pages show six examples of sound production using the modules' supplied with the standard 2720 package. Needless to say, the number of possible combinations is for all practical purposes unlimited but these sheets have been prepared to demonstrate specific uses of the equipment

When patching together these demonstration set-ups play with the controls. Vary the bias settings, filter Q, rise and fall time, etc. and notice the effect that these variations have on the sound. After a short time you will find yourself listening more carefully to the world around you and analyzing what you hear in terms of the characteristics that make things sound the way they do.

Above all, don't be afraid to experiment. The 2720 series modules are as "goof-proof" as possible and there is no possible combination of inputs and outputs that will damage the circuits. We can't guard agains the input of an amplifier being plugged into a wall outlet of course, but all inputs and outputs are current limited and short circuit protected.

PAIA 2720 PROGRAMMING SHEET



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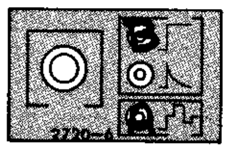
PROGRAMMER: J.S. STUDIO PAIA



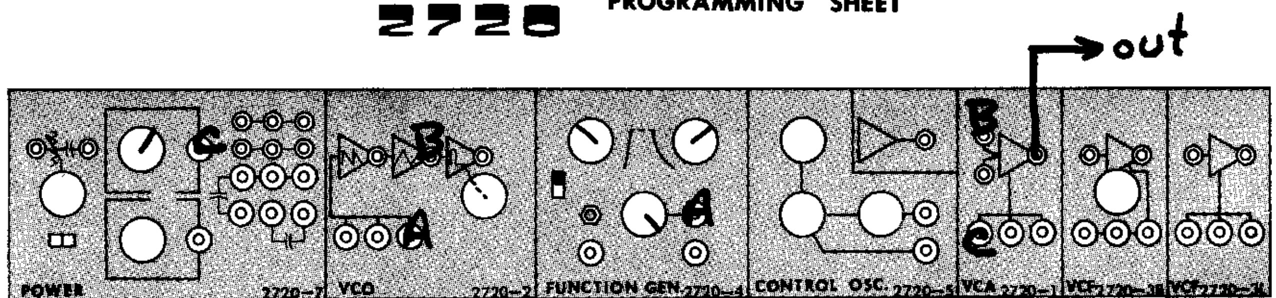
accessory modules

COMMENTS: This set-up demonstrates the use of the 2720-3b Low-Pass filter in converting the triangle output of the 2720-2 VCO into a sine wave. Referring back to table 1 on page 8 of the text you see that a triangle is mostly fundamental and that the 3rd harmonic (the first to appear in the spectrum of a triangle) is only 1/9 the amplitude of the fundamental and that all higher harmonics are correspondingly low in amplitude. The Low-Pass Filter eliminates the higher order harmonics and leaves only the sine wave fundamental.

Play with the attack and decay controls of the 2720-4 Function Generator and observe the effect that this has on the sound. Try using the pulse instead of the step output of the controller to trigger the Function Generator and observe the effect of eliminating any sustain.

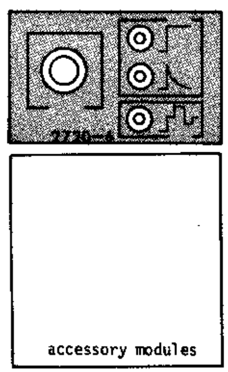
TO USE PROGRAMMING SHEET: Indicate patch cord begining and end points with identical alphabetically ordered letters. Do not use the letters O or Q. Indicate slide switch positions by darkening position of slide bat. Indicate knob positions with a straight line from center to perimeter.

PAIA 2720 **PROGRAMMING SHEET**



EFFECT: Siren DATE 9/5/72

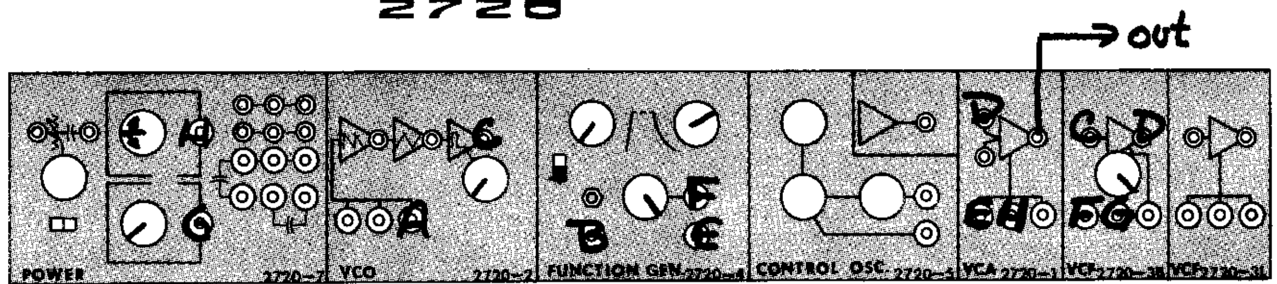
PROGRAMMER: J.S. STUDIO PAIA



COMMENTS In this effect you are using the ouput of the Function Generator to sweep the output frequency of the VCO and the 0 to 5 volt bias supply is used only as a volume control. Pressing the Function Generator "manual" button causes the VCO output to sweep up in frequency at a rate determined by the setting of the 2720-4 "attack" knob and down at a rate set by the "decay" knob. Vary the attack and decay rates of the Function Generator and observe the effect. Also vary the output attenuator of the Function Generator. Use the pulse output of the VCO in place of the triangle output to convert from a modern electronic siren sound to an older mechanical type.

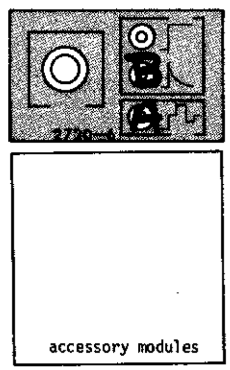
TO USE PROGRAMMING SHEET: Indicate patch cord begining and end points with identical alphabetically ordered letters. Do not use the letters O or Q. Indicate slide switch positions by darkening position of slide bat. Indicate knob positions with a straight line from center to perimeter.

PAIA 2720 PROGRAMMING SHEET



EFFECT: Automute DATE 9/20/72

PROGRAMMER: J.S. STUDIO PAIA



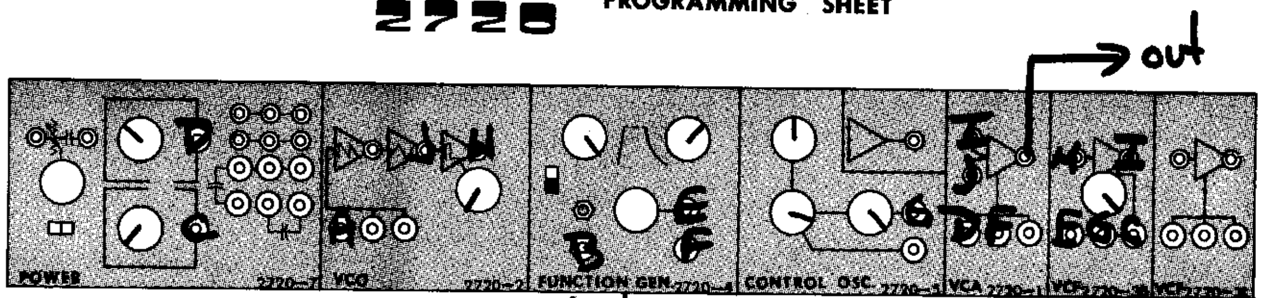
COMMENTS: The basis for this sound is the short duration pulse which is an excellent source of high order harmonics. The 2720-4 Function Generator controls both the 2720-1 VCA and the 2720-3b Band-Pass Filter. When any key is pressed the control voltage output of the 2720-6 Controller sets the pitch of the 2720-2 VCO while the pulse output of the Controller triggers the Function Generator.

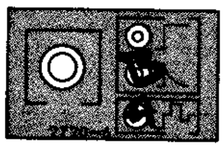
The most pleasing results are obtained when the 0 to 5 volt bias supply ("H" output) is used to set the gain of the VCA such that it is just barely off when no key is pressed. Vary the sweep range and quiescent center frequency of the VCF using the Function Generator and -5 to 5 volt bias supplies respectively and observe the effect on the sound.

TO USE PROGRAMMING SHEET: Indicate patch cord begining and end points with identical alphabetically ordered letters. Do not use the letters O or Q. Indicate slide switch positions by darkening position of slide bat. Indicate knob positions with a straight line from center to perimeter.

PAIA 2720

PROGRAMMING SHEET





EFFECT: Electronic #96 DATE 9-6-72

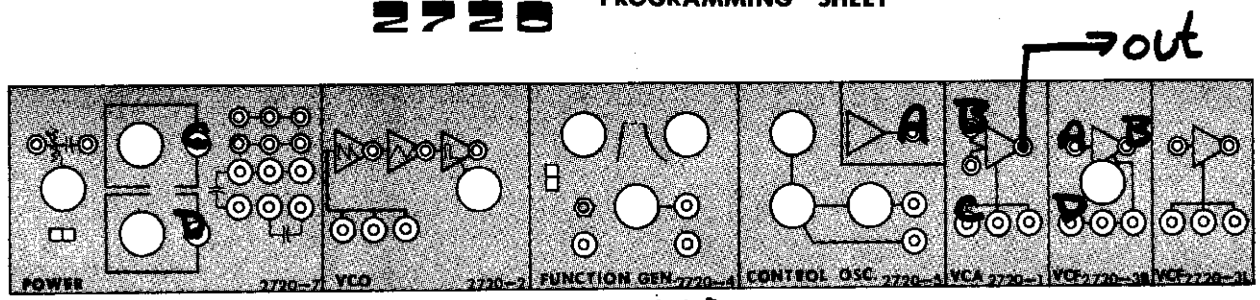
PROGRAMMER: __________ STUDIO PAIA

accessory modules

COMMENTS: With two exceptions this sound is quite similar to the Automute. The first exception is that the center frequency of the 2720-3b Band-Pass Filter is not only being swept by the Function Generator but is also cyclicly varied by a control voltage from the 2720-5 Control Oscillator. By properly adjusting the -5 to 5 volt bias supply, Control Oscillator output attenuator and Function Generator output attenuator the cyclic sweeping can be more pronounced at the output of the VCO into the "3 db." input of the VCA produces a sound with greater "body" than if the filter-swept pulse were used alone.

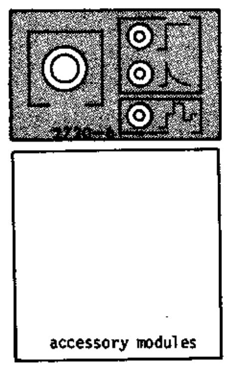
TO USE PROGRAMMING SHEET: Indicate patch cord begining and end points with identical alphabetically ordered letters. Do not use the letters O or Q. Indicate slide switch positions by darkening position of slide bat. Indicate knob positions with a straight line from center to perimeter.

PAiA 2720 PROGRAMMING SHEET



EFFECT: WIND ______________ DATE______________

PROGRAMMER:______________ STUDIO______________

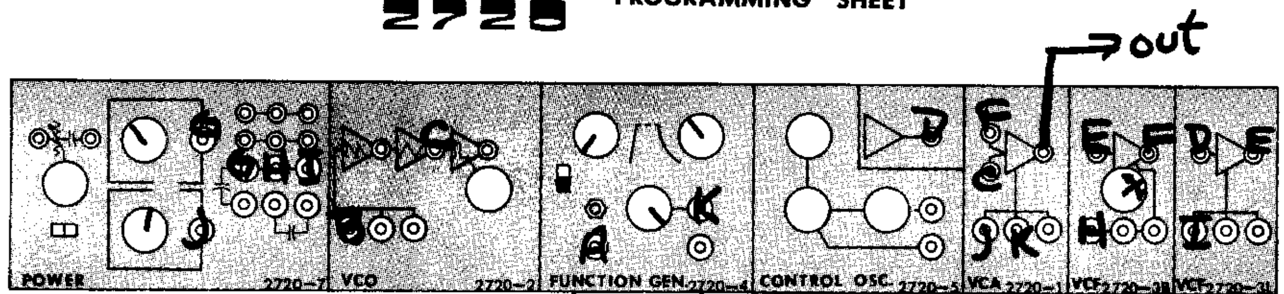


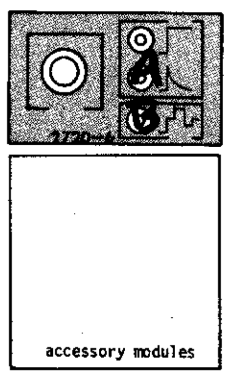
COMMENTS: This is a great demonstration of the use of white noise. The output of the noise section of the 2720-5 is routed to the input of the Band-Pass VCF which is being manuall swept by the -5 to 5 volt bias supply. The output of the filter is connected to the input of the VCA which is under the manual control of the 0 to 5 volt bias supply. Randomly varying these two bias supplies creates effects ranging from a gentle breeze to a howling blizzard.

TO USE PROGRAMMING SHEET: Indicate patch cord begining and end points with identical alphabetically ordered letters. Do not use the letters O or Q. Indicate slide switch positions by darkening position of slide bat. Indicate knob positions with a straight line from center to perimeter.

PAiA 2720

PROGRAMMING SHEET





EFFECT: Chromatic Snare Drum DATE______________

PROGRAMMER:______________ STUDIO______________

COMMENTS: This is a complicated effect but points up the power of voltage controlled techniques. The triangle output of the VCO is given a fast attack and medium decay using the VCA under the control of the Function Generator. Using the pulse Controller output to trigger the Function Generator guarantees that there will be no sustain at all. This combination produces the "strike tone" of the drum. Since the VCO pitch is controlled by the Controller output the "drum" can be played like a melodic instrument. The noise source is voiced first by the Low-Pass Filter and then by the Band-Pass to produce the sound of the snares and is mixed into the output through the "0 db." input of the VCA. The 0 to 5 volt bias supply is connected through the patch panel so that it controls both the Low-Pass and Band-Pass Filters. The -5 to 5 volt bias supply is used to bias the VCA.

Vary the -5 to 5 volt bias supply and observe the effect that working on different portions of the exponential decay curve produces.

TO USE PROGRAMMING SHEET: Indicate patch cord begining and end points with identical alphabetically ordered letters. Do not use the letters O or Q. Indicate slide switch positions by darkening position of slide bat. Indicate knob positions with a straight line from center to perimeter.

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