OWNER'S MANUAL
for
THE ARIES 300 SYSTEM SYNTHESIZER

by
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This is a manual about the Aries 300 System Synthesizer. In reading it you will learn what the synthesizer does, how it works, and how to make it do what you want it to do. In reading it you will also learn something about electronics, about music, about acoustics and psycho-acoustics, about math and physics, and about your aural perception.

Learning all the facts in this manual will not make you a good synthesist, but it's the first step. Becoming a good synthesist requires a curious blend of knowledge, understanding, patience, and creativity. It requires that you throw away some pre-conceptions that you may have about what a synthesizer is and what it does. It requires that you honestly accept your Aries 300 as a synthesizer, which it is and not as a fancy organ, which it is not. It requires an openness and a willingness that you accept the fact that learning about your synthesizer will take some time. It is an on-going process that not only allows you to use your ability for insight, it requires it. It is a kind of circular learning in which each new fact you learn allows you to see, in a new light, something you thought you already understood. Because of this, it is always difficult to know where to begin teaching about synthesis.

We have decided to start like this.

With a patch cord, connect the sine wave output from the first VCO to the fourth audio input of the VCA. With a second patch cord, connect the output of the VCA to the first, or left, input of the output and power module and then take a close look at the VCO. At the top of the module are two knobs labeled "frequency." The left one is coarse and the right one is fine. Set the right knob to x1.6 and the left knob to 1K. Below the knobs is a two position switch; flip the switch to the right or x1 position, then look at the VCA. The lower left knob is a control knob labeled "initial gain." Set it to 10. Finally, turn knob #1 on the output and power module slowly to the right until you hear a sound. If there is no sound, turn to Appendix #1 and follow the troubleshooting procedure outlined there.

The sound you hear is vibration. A vibrating speaker cone causes pressure waves in the air which are alternate compressions and rarefactions. If these density changes occur between twenty times and twenty thousand times a second and if there is sufficient difference between the maximum and minimum pressure, then these fluctuations are audible and are called sound waves.

Sound waves have three perceptible characteristics: pitch, volume, and timbre. The pitch of a sound is determined by how fast the pressure changes occur. A speaker cone vibrating very quickly produces a high pitch; one vibrating slowly produces a low pitch. One that vibrates slower than twenty times a second or faster than twenty thousand times a second produces pressure changes which humans do not perceive as sound.

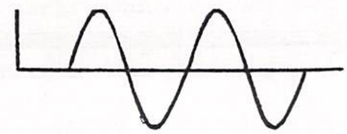
The volume of a sound is determined by the difference between the highest and lowest amounts of pressure. A speaker cone which moves a further distance out and back produces a louder sound than one which moves a shorter distance. The overall distance which the speaker cone moves determines, in part, the volume of the sound.

The sound's timbre is determined by the manner in which the air pressure changes. The compressions may happen gradually and the rarefactions suddenly. This produces a particular timbre. Another timbre is produced when the pressure changes back and forth between its most dense and least dense states suddenly. A third possibility is that it changes constantly and gradually from its least dense to its most dense state. These three motions describe three of the waveforms commonly used in audio synthesis.

Synthesizers produce no sound. Analog synthesizers are so called because they produce, process, and modify electronic signals which are analogous to sounds. Signals, like sounds, have three characteristics: frequency, amplitude, and wave shape. When these signals are converted into sounds by your loudspeaker, each signal characteristic determines a sonic characteristic. The frequency determines the pitch;

the amplitude, the volume; the wave shape, the timbre. Here's how it works.

The sound you're hearing is the result of a sine wave. It looks like this.



When the wave goes high, it forces the speaker cone out; when it goes low, it forces the cone back. Obviously, that sine wave causes the speaker cone to travel a greater distance than this sine.

The amplitude of a signal is measured in volts. The sine you are hearing is the result of a 10 volt signal. It is balanced across 0 volts so that it goes equally above and below the 0 volt line. This particular sine, then, goes from +5 volts to -5 volts and has a peak to peak amplitude of 10 volts. Try changing the amplitude by turning the initial gain on the VCA down and you can hear the resulting decrease in volume.

If you want to change the pitch of the sound, you have to change the frequency of the signal. Make this change, first, by turning the right control knob at the top of the VCO, the fine tune control. The amount of frequency change controllable by this knob should be about one octave. The other manual control which changes the frequency is the coarse tune control. It is possible with this control to change the frequency so much that it becomes either too high or too low to hear. Finally, you can change the timbre of the sound by changing the wave shape of the signal. For now, change the wave shape by patching another waveform from the VCO into the VCA. First try the triangle, then the sawtooth, and finally the pulse. With the pulse patched into the VCA, try altering the pulse width by turning the pulse width control on the VCO. You'll notice that with the pulse width control at either extreme, the sound stops. This is not a malfunction but, on the contrary, an extremely useful function. A detailed explanation of this will be forthcoming in a few pages. Once you have done all this, unpatch your synthesizer and read on.

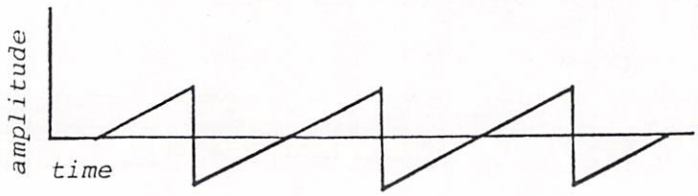
It is impossible to learn to use your synthesizer intelligently without learning how it works. We have seen books with patch diagrams and dial settings specifying how to get certain sounds from a synthesizer. This seems little more than the sonic equivalent of a "paint-by-the-numbers." By their nature, these patch diagrams are not instructional but are simply the dial settings resulting from someone else's experiments. We believe that to understand your synthesizer you must learn in detail how it works. This way you can create your own instruments and sounds and not be limited by someone else's musical taste. Although this is more work than memorizing dial settings, it offers greater flexibility and allows your synthesizer to become a musical extension of YOU, and not of a diagram maker.

The method we've chosen to graphically represent patches throughout this manual is a standard notation system called a "block diagram." A block diagram is a chart which maps the path of the signals and displays which modules the signals go into and in what order they occur in the patch. Your own understanding of the synthesizer tells you what happens to the signal as it passes through each module. Set up the following patch.

	Fc at max.	Gain at 10	
VCO sawtooth →	VCF →	VCA →	LEFT →
100 Hz.	input 4	input 4	

This block diagram tells you to set the VCO frequency control to 100 Hz. (you can approximate this) and patch the sawtooth output into audio input #4 of the VCF. If, instead of the AR-314 low pass filter, you have the AR-327 multi-function filter, take the low pass output. The filter frequency control, also called "cutoff frequency" or "Fc," should be turned all the way to the right (Fc at maximum). The filter output should be patched into audio input #4 of the VCA and the initial gain set to 10. Finally, the VCA output should be patched into the output and power module's left input. In almost all patches, the final module used is the output and power module and this is assumed in the block diagram. The only time you actually need to include the output and power module in the block diagram is when you want to specify the left or right channel or when you want to specify reverberation from the AR-328 module in the patch. The control on the output and power module, which can be thought of as a master amplitude control for the audio path, should be set to a comfortable level.

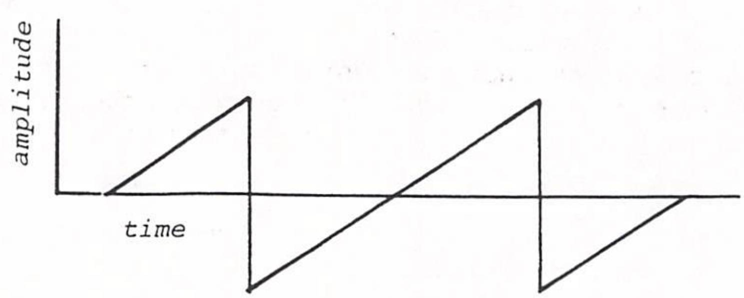
Try to put into words the difference between the sound created by the sine and that created by the sawtooth. Do words like "richer," "fuller," "brighter," "buzzier," seem appropriate? If you were to look at the signal on a scope, it would look like this.



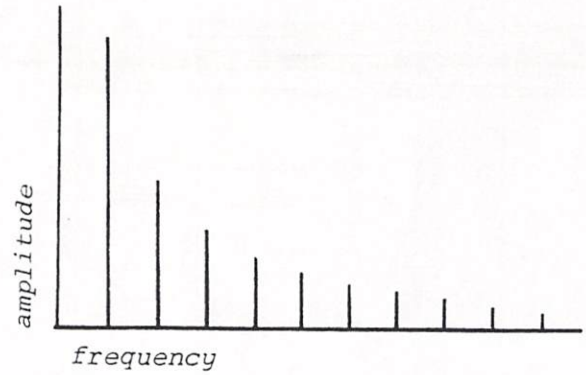
The amplitude is 10 volts peak to peak and it is a balanced waveform; the frequency is 100 Hz. (assuming you set it to 100Hz). Obviously the only difference between this waveform and a sine wave is its wave shape. Since the wave shape determines the motion of the speaker cone and the motion of the cone determines the timbre of the sound, the difference between the sound produced by a sine wave and a sawtooth wave is one of timbre.

In the case of the sawtooth wave, as the voltage increases, it pushes the speaker cone out until it reaches its highest point then snaps it back to its lowest point and begins the cycle again. In other words, it causes gradual compressions and sudden rarefactions of the air molecules. This sawtooth motion creates an audible harmonic spectrum that contains energy at the fundamental frequency, in this case 100 Hz., and at all whole number multiples of the fundamental frequency, (200 Hz., 300 Hz., 400 Hz., 500 Hz.....). This corresponds to the harmonic series.

Auguste Fourier, a French mathematician and contemporary of Robespierre, determined that all periodic, complex wave motion can be analyzed interms of a series of sine waves at specific frequencies and of specific amplitudes. "Periodic" simply means repeating so that all cycles are the same, and "complex wave motion" is any wave motion other than a sine wave. There are two kinds of graphs which are used to represent wave motion. The first of these is a time domain graph. A time domain graph plots amplitude on the vertical axis against time on the horizontal axis. This graph is a picture of one cycle of a waveform. Here's the time domain graph of a sawtooth.



The frequency domain graph plots amplitude on the vertical axis against frequency on the horizontal axis. This graph is a representation of the harmonics of a waveform. Here's the same sawtooth represented in the frequency domain.



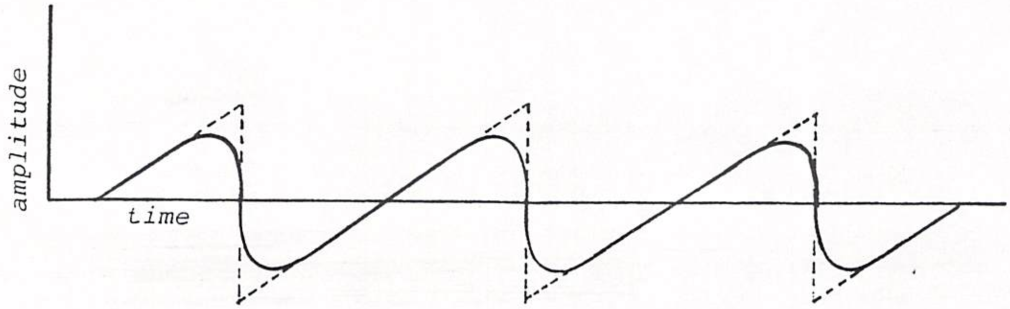
As you can see, a sawtooth wave contains harmonics that occur at all whole number (integer) multiples of the fundamental frequency and the relative amplitude of each harmonic is:

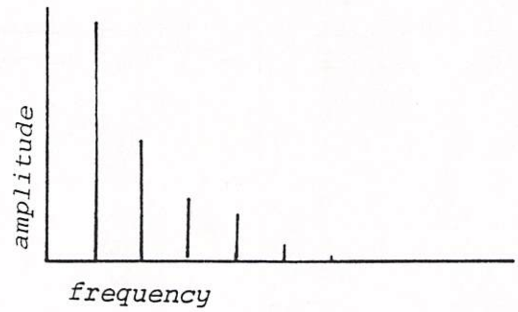
1/harmonic number x amplitude of the fundamental.

The amplitude of the fundamental in this case is 10 volts, so the amplitude of the second harmonic is 5 volts (1/2 x 10), the amplitude of the third harmonic is 3.3 volts (1/3 x 10), the amplitude of the fourth harmonic is 2.5 volts (1/4 x 10) and so forth.

With the patch that you have set up, turn the filter cutoff frequency control slowly to the left and listen to the sonic changes. Describe what's happening to the sound. Turn the control back and forth a few times until you're sure that you can identify which characteristics (volume, pitch, or timbre) of the sound are changing. Consider the VCF, now, in terms of the signal rather than the sound. If you've correctly identified the sonic change as mostly a timbral one, then you've probably figured out that the filter has changed the shape of the sawtooth wave.

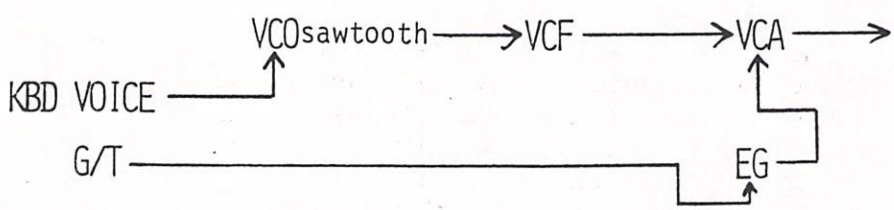
The AR-314 is a low pass filter, as is the AR-327 multi-function filter in the low pass mode. A low pass filter, just as the name implies, passes the low frequencies and filters the higher frequencies. The approximate point at which this filtering takes place is the cutoff frequency, and it is controllable manually by the filter frequency control. All the harmonics of the sawtooth with frequencies above the cutoff point have been filtered and all the harmonics with frequencies below the cutoff point have been passed relatively unchanged. Because the harmonic structure of the waveform has been changed, the signal coming from the filter is no longer a sawtooth wave. Here is a filtered sawtooth shown both in the time domain and in the frequency domain.





With low pass filters "slope" is a term used to denote the rate at which filtering occurs above the cutoff point. Both the Aries AR-314 and the AR-327 filters have slopes of -12 dB/octave. This means that all signals with frequencies one octave higher than the cutoff point have 12 dB less amplitude than they would have had they not been filtered. Signals with frequencies two octaves above the cutoff point have 24 dB less amplitude; signals three octaves above the cutoff point have 36 dB less amplitude, and so on. There is a discussion of dB (decibels) in Appendix #2 which should make this all more clear. It is not necessary to understand this on a technical level now. However, it is important that you have a good aural grasp of the sonic changes resulting from filtering a waveform.

Here is a patch, using a filtered sawtooth, that approximates a tuba.

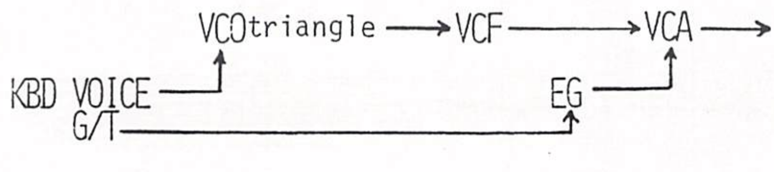


Find the 4 keyboard voice output jacks on the keyboard interface module and connect any one of them to VCO control input #4. Play a few notes on the keyboard just to make sure the keyboard is changing the frequency of the VCO. Play a note somewhere in the middle of the keyboard and hold it down; set the VCO frequency to any low note in the tuba range. (Whenever you're tuning a patch that requires the use of the keyboard, always hold down a key while tuning. This helps to insure that the correct voltage is coming from the keyboard.) Patch the keyboard gate from the interface module into the gate input of an envelope generator and the keyboard trigger into the trigger input of the same envelope generator. The keyboard gate and trigger is also available at the keyboard interface. Notice that the envelope generator has four control knobs. Set the first one, "attack," to 3; the second one, "decay," to 10; third one, "sustain," to 10; the fourth one, "release," to 3. Play a note on the keyboard. The patch should begin to sound like a tuba. Now hold a key down and lower the filter cutoff frequency slowly until you find a fairly good tuba sound. Try a couple of different keyboard notes and then readjust the filter Fc. Play the patch awhile making some small changes in the filter's cutoff frequency. Describe the different kinds of "imaginary tubas" that result. Some should sound brassier than others; some "blat" more, and still others might be more mellow. After you're convinced that you understand what the filter does in this patch, try this experiment.

Hold the lowest key on the keyboard down and adjust the VCO and VCF frequencies until you have a really mellow sounding tuba. Play up the keyboard note by note and listen to the sonic changes. As you reach the high end of the keyboard, the sound gets softer and softer and finally fades out completely. Take a few moments, think about it, and try to figure out what happened. By the way, if the sound didn't fade out by the time you reached the top of the keyboard, start over again with a <u>really mellow</u> sounding tuba.

Here's what happened. With the lowest key depressed, the VCO's frequency was say 100 Hz., and the filter cutoff frequency was set to about 400 Hz. or so. You increased the frequency of the VCO by playing up the keyboard but the filter's cutoff frequency remained the same. As the fundamental and its harmonics became higher in frequency, more of them were filtered until, finally, the fundamental itself was above the cutoff frequency and it, too, was filtered. Remember, a low pass filter filters signals with frequencies above the cutoff point. By the time you reached the top of the keyboard, all the signals had frequencies above the filter's cutoff frequency and so everything was filtered. If this is clear, go on to the next patch.

There are three changes you have to make to turn your tuba into a recorder. First, use the triangle wave output of the VCO instead of the sawtooth. Second, re-tune the VCO so that its frequency is a couple of octaves higher and within the range of a recorder. Third, re-adjust the filter's cutoff frequency to approximate a recorder's timbre.



A triangle wave, like all other periodic waveforms, has a given harmonic structure. The harmonics of a triangle wave occur at all odd-numbered multiples of the fundamental frequency. If the fundamental (F) is equal to 400 Hz., then the harmonics occur at 1200 Hz., (3 x 400), at 2000 Hz., (5 x 400), at 2800 Hz., (7 x 400), at 3600 Hz., (9 x 400) and so forth. The amplitude of each harmonic is equal to:

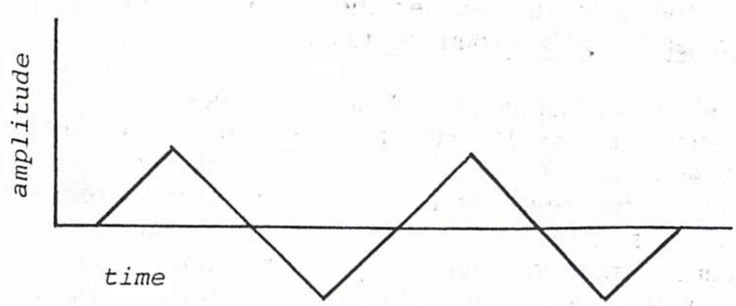
1/harmonic number squared x amplitude of the fundamental.

If the amplitude of the fundamental is 10 volts, as it is on the Aries, then the amplitude of 3F is 1.1 volts.

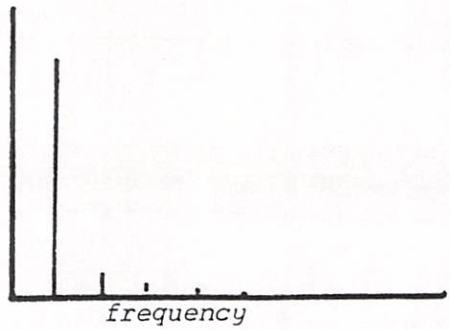
$$1/3 \times 1/3 = 1/9, \quad 1/9 \times 10 \text{ volts} = 1.1 \text{ volts}$$

The amplitude of 5F is 1/25 x 10 or 0.4 volts. The amplitude of 7F is 1/49 x 10 or 0.2 volts and so on.

As you can hear, the triangle wave sounds similar to a sine wave and not as "buzzy" as a sawtooth wave. A triangle wave has much less energy in the harmonics than a sawtooth wave. From the time domain graph, you can see how a triangle wave resembles a sine.

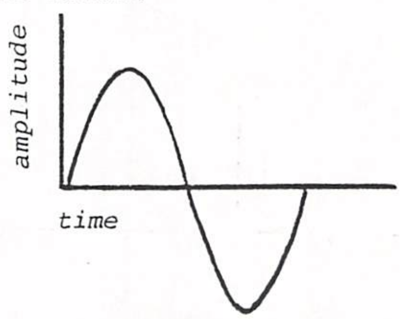


Here is the frequency domain graph of the same waveform. Compare it with the frequency domain graph of a sawtooth wave and note the relative amplitude of the harmonics.

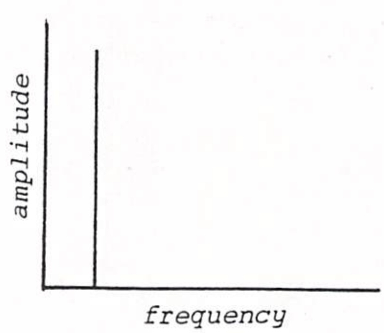


Since any waveform can be broken down and analyzed in terms of its harmonics each of which is a sine wave, it can also be built up from a series of sine waves. In the earlier days of electronic music, complex waveforms were often built from sine waves, each at a designated frequency and with a designated amplitude. This is called "additive synthesis" and computer music is often made this way. Because of the expense of having banks and banks of analog VCO's and VCA's, and the problem of how to control them musicians and composers usually prefer synthesizing timbres by subtractive synthesis--by filtering certain harmonics.

The sine wave from which all these other waveforms are made looks like this in the time domain.



And in the frequency domain, it looks like this. Note that it has no other harmonics other than the fundamental, which is the first harmonic in any harmonic spectrum.

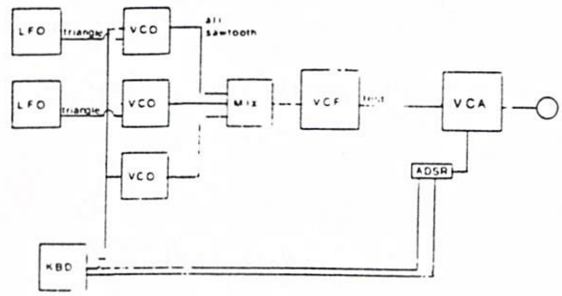


If you vary the preamp level or threshold level as you record subsequent layers of percussion, you can get the envelopes to double fire. This adds more variation and interest to your work. The sounds can be as simple or as complex as your imagination, and equipment allow. Percussion voices can consist of a VCO, VCA and ADSR; or you can use the whole synthesizer for just one sound. (Walter Carlos uses up to ten ADSR's for one patch.)

The key word here is experiment. There is no single correct way when working with synthesizers. If it works, use it!

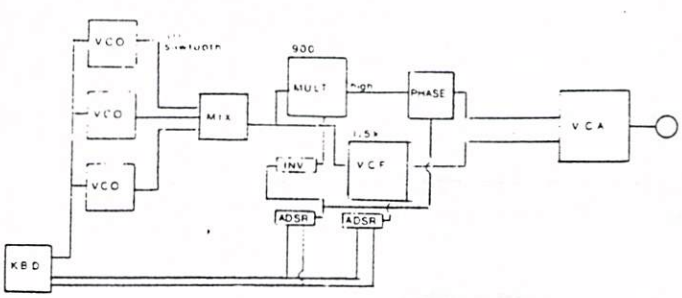
Here is a patch for string sounds. I like to use three VCO's. They should be ever so slightly out of tune for that chorus effect. The balance of the pitches with their minute variations is extremely critical. I prefer to use the 'test' output on the AR-314 VCF. This output has 6db cutoff instead of 12db. The envelope is variable depending on the phrasing of the line you are playing. If the line is very legato, long attacks and decays work nicely. If it is faster, you obviously have to shorten both the A and R. The phrasing and context of what you play is very important. To use strings effectively you have to sonically balance them with other instruments in the mix.

STRINGS:



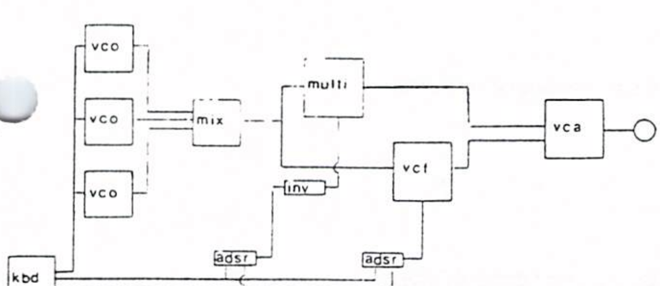
With this patch it is possible to get the sound of 3 - 6 strings. I find it much more effective to overdub a second harmony rather than patch the keyboard up for two notes. The 'blanket sound' of the strings falls apart as soon as the second note is played. This is because the chorus interaction has been interrupted.

HORN PATCH



This versatile patch was used to make horn type sounds. The use of two signal paths, each with their own filter and ADSR creates a certain kind of phasing when they are mixed in the VCA. With a longer attack on the first ADSR you get a sort of 'sheen' to the sound which comes in only when you hold the notes down long enough.

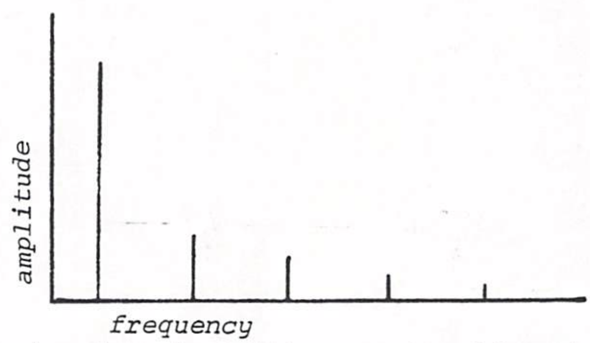
"OOYE" CHORUS



This is the patch used on the "ooye" track on side one. Set the Q to about 20 and use the bandpass output. Note the inverted envelope to the multi-mode filter. Experiment with the length of attacks on the ADSR's.

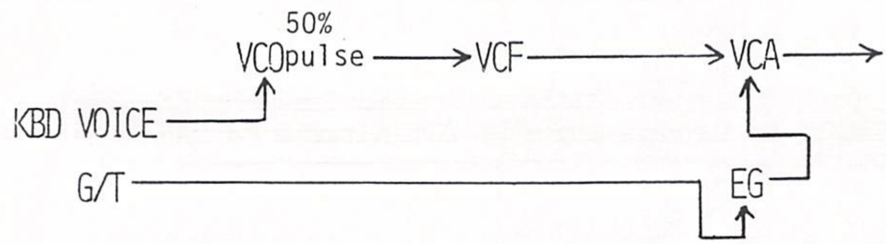
As the pulse wave changes in the time domain, there is a corresponding change in the frequency domain. Changing the pulse width re-distributes the energy of the pulse throughout the harmonics. At 50%, the fundamental has the greatest amplitude relative to the other harmonics. At 1%, the amplitude of the fundamental is audibly negligible.

A 100% pulse is a "pulse wave" that is high 100% of every cycle. Because there is no fluctuation in this signal, there is no vibration and no resultant sound. Likewise, a 0% "pulse wave" is "high" 0% of the time. This, also, is a non-fluctuating signal and can produce no sound. The denominator of the duty cycle fraction indicates which harmonics are missing from the spectrum. In the case of a 33 1/3% pulse (duty cycle of 1/3) each third harmonic is missing from the harmonic spectrum. For a 33 1/3% pulse, then, its harmonics occur at these multiples of the fundamental frequency: 1,2,4,5,7,8,10,11,13,14,16,17,-19,20, and so forth. A 10% pulse has each tenth harmonic missing and a 50% pulse wave has each second harmonic missing from the spectrum. The frequency domain graph for a 50% pulse wave is this:

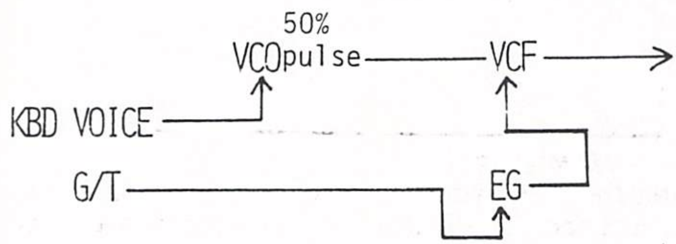


By slowly turning the pulse width control and listening to the resulting timbre, try to adjust for a 50% pulse. As you approach the square wave, you can hear the harmonics dropping out until, at one point, the timbre becomes "hollow" sounding, sort of like a clarinet. It will probably take you several tries in order to learn what sound to listen for. When you think you have it, switch the 2 position switch to the x.002 position and, if you've adjusted correctly, you will hear evenly spaced clicks. If the clicks are un-evenly spaced, adjust the pulse width control until they become even and then switch the switch back to the x1 position. This is the resultant sound of the square wave. The clicks you were hearing were the result of the instantaneous voltage changes of a low frequency pulse wave snapping your speaker cone out then back. Try adjusting for a square wave by ear until you can do it each time. Always check your attempt by switching the switch and listening for the evenly spaced clicks.

If you can adjust for a square wave, you can patch up a clarinet sound. Here's the block diagram.

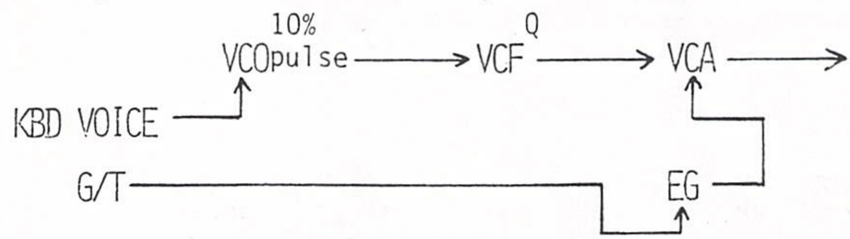


Tune the VCO initially so that its frequency is in the clarinet range and adjust the filter cutoff frequency for the desired timbre. Close the initial gain on the VCA and play the keyboard; your patch should sound something like a clarinet. You can adjust the filter cutoff frequency to simulate different kinds of clarinets, or make up a hypothetical clarinet of your own. Think about what a clarinet made out of brass would sound like, then change the filter's Fc to simulate it. Lower the frequency of the VCO and you have a brass clarinet 10 feet long. O.K., when you've finished experimenting with imaginary clarinets, set up a normal sounding one again. Try this variation of the clarinet patch.



Set the filter Fc to minimum and make sure the resonance is set to minimum. Patch the envelope generator output into the filter control input #4 and patch the filter output into the output and power module. This patch does not use the VCA. There is a sonic difference between this patch and the previous patch. Listen to the sound as you play the keys and try to describe the difference.

In the previous patch, the characteristic of the signal that was being changed was the amplitude. This patch changes the amplitude of the signal by filtering it to below the audible level, but it also changes the timbre of the sound by selectively filtering the harmonics. As you press a key, the filter Fc is opened up by the envelope generator. As it opens, it allows the fundamental to pass, then the second harmonic, then the third and so on. Although this happens quickly, it is not quickly enough to avoid detection by your ears. There are many such sounds that we hear accurately but subliminally. One of the first things that everyone learns in becoming a good synthesist is to hear more accurately. Fortunately, working with a synthesizer improves your aural acuity which leads to better synthesis technique. It's a happy chain of events that makes it almost impossible not to become a better synthesist. Go back and set up the original clarinet patch and listen to it carefully. Then set up the variation. Which one sounds more like a real clarinet to you? Why do you suppose this is true? After you can answer these questions to your own satisfaction, set up the following bassoon patch.



The "Q" above the filter is a standard symbol for resonance and it will be used throughout this manual. First, set up the patch with the resonance off, then slowly increase the resonance by turning the control to the right. Listen to the changes. Fiddle with the filter's Fc and resonance controls until you find a reasonable bassoon sound. By changing the VCO's frequency, the filter's Fc and the amount of resonance, the entire double reed family of woodwinds can be synthesized using this patch. It's just a matter of really listening the next time you hear these instruments.

As an experiment, turn the resonance control all the way up and the Fc all the way down. Patch two pulse waves and two sawtooths from the two VCO's into the audio inputs of the filter and turn "signal 1" control to 10. Slowly turn up the Fc. The sound that you hear is distorted. You have exceeded the electrical limits of the filter. Although it is impossible to make any connection within the Aries synthesizer that can damage it, you have distorted the waveforms. Of course, if you like this sound, by all means use it. It's your choice. Just the same, be aware that it is possible to distort the filter, or any other module, by exceeding the limits of the system. Unpatch the signals one by one and you will hear when the remaining waveforms are no longer distorted.

Here's another experiment. Patch the sawtooth into the audio input of the filter. Turn the resonance to maximum and tune the Fc to minimum. Patch the filter output into the output and power module. Slowly raise the Fc of the filter. As you increase the filter frequency, you can hear individual volume increases called "resonant peaks" at each of the sawtooth's harmonics. Here is why.

All filters have a tendency to resonate. The resonance control determines the degree of this tendency and the cutoff frequency control determines at what frequency this resonant tendency exists. The concept of resonance is not new for you. Perhaps you have called it "sympathetic vibration." Have you noticed that when a band is tuning up, some notes that the bass player plays will cause the snares on the drummer's snare drum to rattle? This is one example of resonance. Everyone has blown across the neck of a bottle and produced a tone. A similar situation exists electrically within the filter. Remember, all these electronic modules deal with signals. Acoustic "modules" like bottles, basses, and snare drums deal directly with sounds. The fact that some of the same sonic modifications occur in both cases is the idea behind synthesizers. It's a lot easier to patch up a 10 foot brass clarinet than it is to construct one.

You've probably noticed that some bottles produce tones more easily than others. This is due, in part, to the shape of the bottle. The resonance control determines the degree of tendency to resonate. If the bottle had a resonance control, turning it would change the shape of the bottle so that it would become more or less resonant as you blew across the neck. The overall volume, or enclosed airspace within the bottle, would remain the same. The filter's Fc control determines at what frequency this resonant tendency exists. If our hypothetical bottle had a frequency control, turning it would increase or decrease the size of the bottle, but not change its shape. As you've probably surmised, a bottle is, in fact, an acoustic filter.

In addition to bottles, basses, and snares, here's a list of some acoustic resonant filters which you may not have thought of as filters.

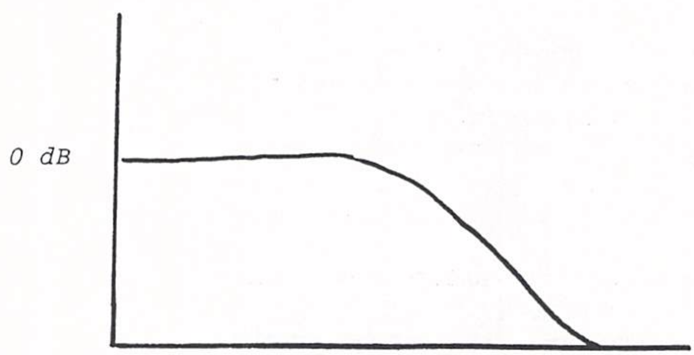
1. your mouth
2. a violin body
3. a piano string
4. your living room
5. a drum
6. the glass that Ella Fitzgerald breaks with her voice *
7. an auditorium
8. a fish
9. your ears

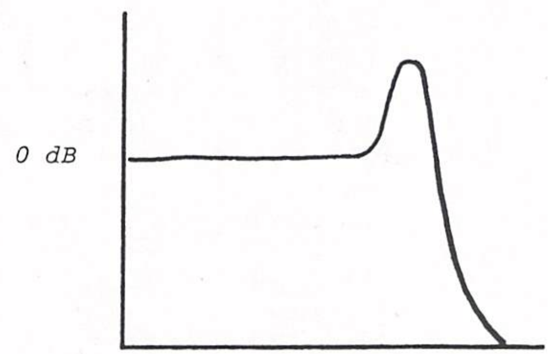
There are many more which you can add to this list.

As you swept the VCF's Fc from the highest to the lowest point, from time to time the Fc was equal to the frequency of one of the harmonics of the sawtooth. The filter resonates whenever the Fc is approximately equal to an audio input. When the Fc was equal to the frequency of any of the sawtooth's harmonics, a resonant peak occurred.

* or was it Memorex?

Another change in the filter caused by the resonance control is that it subjectively makes the filter's cutoff slope steeper. With no resonance the slope of the AR-314 and the AR-327 in low pass mode is -12 dB/octave. With maximum resonance the filter slope can be increased to more than -50 dB/octave. Here are two diagrams that represent that information graphically. The peak at the Fc in diagram #2 is the resonant peak that you heard in the patch.





Up to this point we've been dealing with basically three modules: the VCO, the VCA, and the VCF. In most patches these three modules constitute a typical audio path. In a general way, you have explored all the basic ways a synthesizer creates, processes, and modifies signals. You have heard all the basic waveforms and know now how to manually change the three variables of a signal: frequency, amplitude, and wave shape. We've presented a lot of information so far and a couple of important concepts. When you feel that you have a good understanding of this material, then go on to the next section.

Composers and musicians were making electronic music before synthesizers were available. Their instruments were laboratory oscillators, filters, amplifiers and other electronic test equipment. They were able to set up most of the patches you have done so far with your Aries, but not as easily as you have done. For them, just to make this simple patch that we've been working with,

OSCILLATOR ——> FILTER ——> AMPLIFIER ——>

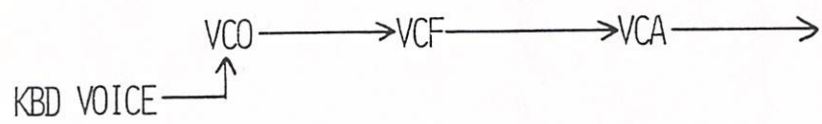
required three separate pieces of equipment; often each had its own power supply. Because of electrical differences between the separate pieces of equipment, not only did they need patch cords to patch them together, but also they sometimes needed impedance matching transformers and pre-amps. To record something as simple as a major scale using this equipment might require hours of tedious work.

The Aries 300 System offers many advantages over the classical electronic music studio. Each piece of equipment in a classical electronic music studio is a single module in an Aries synthesizer. All the modules are compatible; none require a special interface. The Aries synthesizer, unlike some others, makes no distinction between audio signals, control signals, and timing signals. A voltage output from any module can be patched into any other module without harming the synthesizer. In fact, there is no patch you can make using just your Aries synthesizer which can damage it. All Aries synthesizers are completely modular. You can get a more or less standard Aries 300 system, or any of the other standard systems if you like. Or you can make up a system of your own to fit your own musical needs. The Aries 300 System is portable and it can be played from stage or in the studio. It is possible to "normal" certain often used patches internally within the synthesizer so you can set up patches without using patch cords. This choice is yours. There is no one at the Aries factory who decides what is musically useful for you and wires the synthesizer that way. Finally, and most important, the Aries, and all other synthesizers, offer voltage control over a signal's parameters, and this is what the next section of this manual is about.

THE KEYBOARD

The keyboard (KBD) provides one of the easiest and most versatile ways of voltage controlling a synthesizer, but a synthesizer's KBD is deceptively familiar. Set up this patch and we'll start with the familiar part.

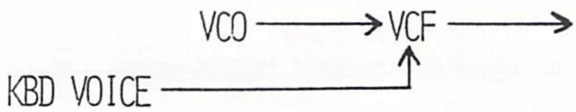
In a block diagram, voltages which control a module are drawn as arrows pointing in at the bottom of the module being controlled. Control voltages coming from a module are drawn as arrows coming either from the right side or the top of the module.



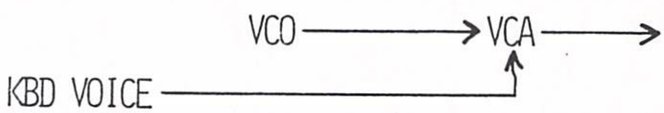
The KBD voice output is one of the 4 jacks so labeled on the KBD interface and it should be patched into one of the unattenuated jacks (jack 2, 3, or 4) of the VCO. Any time the VCF or VCA is used in a patch and it has no voltage controlling it, assume that the filter's Fc and the VCA's initial gain are set to such a level to permit the audio signal to pass. Play up and down the keyboard and you will hear the pitch of the tone change. This is the result of the voltage from the KBD changing the frequency of the VCO. Of course you hear a constant tone, so you know that the synthesizer KBD doesn't work exactly like other keyboard instruments.

On the KBD interface, find the portamento mode switch and set it to "normal." Turn the portamento on, set the control to maximum and play a few notes. The portamento control makes all the sudden KBD voltage changes from one note to another note, gradual changes. It is, in fact, a low pass filter filtering the KBD control voltage output and operating like a shock absorber in the time domain, rounding off the instantaneous voltage changes. Notice that the glide time is the same regardless of how far apart the notes on the KBD are. Switch the mode switch to linear and you will hear a marked difference. The glide is more uniform and it takes longer to glide between large intervals than it does small intervals. Try some different portamento rates by changing the portamento control knob. (It is sometimes musically useful to change the portamento rate in the middle of a glide.)

Notice that you need not keep the key depressed for the portamento to complete its glide. If you raise the key during the glide, the glide will continue until the correct frequency is reached. Switch the portamento to "off" and try another patch.



For this patch, hold the lowest key down and set the VCF Fc to minimum. Tune the VCO to some low frequency and play up the KBD, listening to the sonic change. It's apparent that the KBD is controlling the VCF, not the VCO, because the timbre of the sound is changing, not the pitch. For another example, try this patch.



Again hold the lowest key down and set the initial gain to 0. Before playing up the KBD, describe what sonic changes will happen, then try it and see if you were right.

You can see how a synthesizer's KBD is deceptively familiar. It is familiar because it looks like any other keyboard. It is divided into octaves and each octave is divided into 12 keys: 7 white keys and 5 black keys. Unlike keyboards found on most other instruments, a synthesizer's keyboard is a voltage source and its output can be used to control any module that can be voltage controlled.

The KBD voltage output is linear. This means that every time you play one KBD octave higher, you increase the voltage output by one volt. With the lowest key depressed, the voltage output is 0 volts and with the highest key depressed, the output is 5 volts. Each KBD semitone results in a 1/12 volt change. As you've noticed, the KBD has a memory that remembers the voltage even after you have lifted the key.

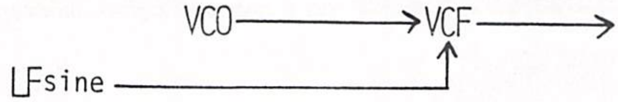
There are two more controls on the KBD interface that are associated with the KBD voice output. The first of these is a knob labeled "tuning." This control adds or subtracts a constant voltage to the KBD voice output. You can adjust this control to tune your Aries to other instruments or to transpose to other keys, much like a guitarist uses a capo. It's also possible to use this control as a pitch bend device or to provide other kinds of musical inflections. The other control is the "voice trim" control. This is used to adjust the voltage output of the KBD voice voltage. Generally, it is best to set this control so that playing the first octave on the KBD results in a 1 volt output and, if the KBD is controlling a VCO, a doubling of the VCO's frequency.

Here's a patch using a voltage other than the KBD to control the frequency of a VCO.

$$\begin{array}{ccccccc} & & \text{VCO} & \longrightarrow & \text{VCF} & \longrightarrow & \text{VCA} \longrightarrow \\ & & \uparrow & & & & \\ \text{LFsine} & \longrightarrow & \lrcorner & & & & \end{array}$$

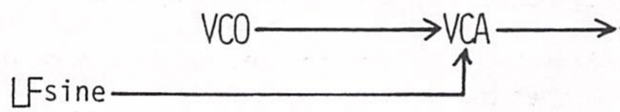
First, patch the audio path. Then, on your second VCO, switch the two position switch to the x.002 position. Set the coarse frequency control to 256 and the fine frequency control to 1.6. The switch is a range switch which, in the .002 position, makes the VCO oscillate in the low frequency range. To determine the approximate frequency of the VCO, multiply the frequency indicated by the setting of the coarse frequency control (256) times the setting indicated by the fine tuning control (1.6) times the setting of the range switch (.002). The frequency of the VCO with these settings is 256 x 1.6 x .002 = .8192 or roughly 0.8 Hz. This 0.8 Hz. sine wave controls the frequency of the other VCO and you can hear the pitch swooping up and down a little slower than once a second. Unless you attenuate the control signal at the VCO input, the audio VCO will be driven out of the audio range by the 10 volt controlling sine wave.

The sensitivity of all Aries VCO's is one volt per octave. In other words, each positive control volt patched into the frequency control of a VCO will double the frequency of the VCO; each negative control volt will halve the frequency. The 10 volt sine drives the VCO up 5 octaves and down 5 octaves for a total of 10 octaves. For a comparison, your voice has a range of about 2 octaves. A trumpet has about 3 octaves. A guitar has a range of about 3 1/2 octaves and an entire symphony orchestra has a total range of about 5 1/2 octaves. The VCO in this patch has a range almost double that of a symphony orchestra and it can go even further than that but who cares except maybe bats and dogs? Change the frequency of the low frequency sine from its highest to its lowest frequency and listen to the change. When the controlling VCO oscillates in the audio range, the frequency of the audio path VCO is being changed at audio rates. You can no longer hear the individual rises and drops in pitch but instead hear a dense sound which is called a "frequency modulation timbre." We will discuss this in detail later in this manual. Adjust the frequency of the controlling oscillator so that it's again in the low frequency range and substitute other waveforms for the low frequency sine. You can probably imagine what will happen before you hear the patch. After you've tried all the waveforms, set up this patch.



Here, the LF oscillator controls the filter's frequency. Of course you could do the same thing manually by turning the frequency control back and forth, but you can't obtain the same accuracy that this patch provides. Again, when the controlling oscillator is in the audio range, a modulation timbre results. This is a form of amplitude modulation and it, too, will be discussed later in detail.

Here's another patch. By now you can probably tell what sound will result just by looking at the block diagram.

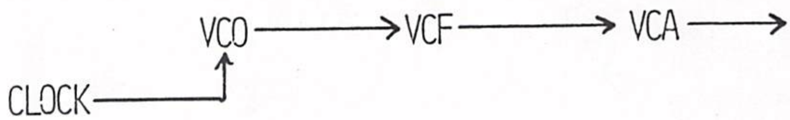


Amplitude modulation is also possible with this patch. Try putting the controlling oscillator up into the audio range and experimenting with some amplitude modulation timbres. Try some different waveforms from the controlling oscillator and from the audio path oscillator; try some different frequency settings. See if you can hear a difference when you substitute a LF triangle for a LF sine. Change the filter's Fc and the amplifier's initial gain. Expand the patch by using some of the modules we've already discussed, and some we haven't. Be adventurous! Let your patches grow! If you find a sound you like and think you can use, don't tear down the patch until you've written a block diagram of it and understand what's happening in the patch. If any of the dial settings are crucial, be sure to note them on the block diagram.

There are other modules which can generate low frequency waveforms. One of these is the DUAL LFO.

Substitute the output from either of the LFO's in the AR-324 module for the controlling oscillator. Do either of these waveforms differ from the waveforms output from the AR-317 in low frequency mode? By now, you shouldn't have expected them to. These waveforms are all 10 volts peak to peak amplitude and can be varied manually from 0.3 Hz. to 30 Hz. The LFO's in the AR-324 module cannot be voltage controlled. There is just one more variation to try and then we'll get on to something else.

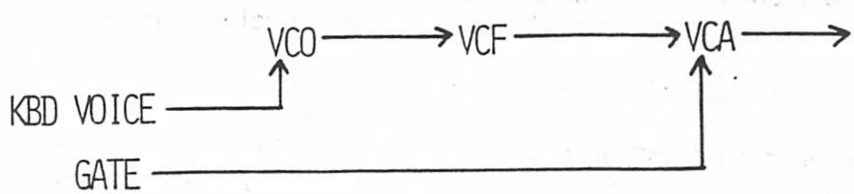
Try using the clock output from the AR-318 module to control an audio range VCO.



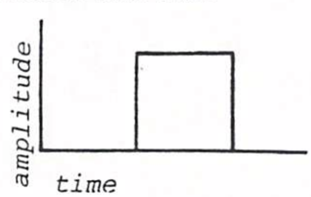
The term "clock" is just another name for a low frequency oscillator. It is so called because it is often used to generate timing signals. It is, in fact, just another LFO. The clock on the AR-318 can be voltage controlled via its FM input and its sensitivity is approximately one volt per octave.

KBD GATE

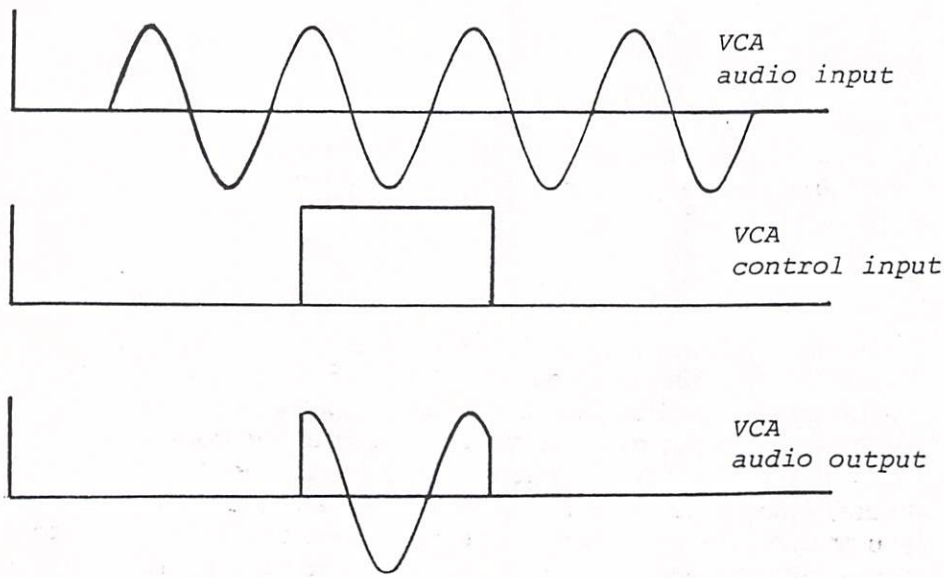
Using the same audio path we've been using all along, set the VCO frequency to whatever you like, the VCF Fc to 16 KHz. and the VCA initial gain to 0. For part of the control path, patch the KBD voice into VCO control input #4. For the other part of the control path, patch the KBD gate output into control input #4 of the VCA. Put the VCA linear/exponential switch to linear mode and play one of the keys. As you pressed the key down you heard a sound and when you lifted the key the sound stopped. The KBD gate is a voltage of 10 volts amplitude and, in this patch, it is used to control the gain of the VCA. Keep in mind that the KBD gate is different from the KBD voice and that each of these voltages is controlling a different module simultaneously. Here is a block diagram of the patch you have just set up.



The KBD gate voltage looks like this.



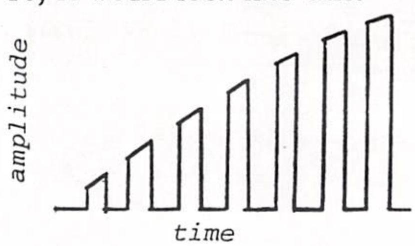
Although it looks like one cycle of a pulse wave, there is a theoretical distinction. Waveforms are periodic; they repeat and have exactly the same wave shape every cycle. The way that mathematicians deal with waveforms is to assume that they never began and they will never end--that they have and always will exist without change. We needn't be that cosmic about it. We can say that a waveform is periodic if it starts when the synthesizer is turned on, ends when it's turned off, and is the same every cycle. The gate begins when a key is depressed and ends when the key is released. A gate is not periodic. When you press the key down, the gate voltage of 10 volts opens the VCA to maximum and allows the sine wave to pass through at its maximum level. These diagrams should help you to visualize what's happening in the patch.



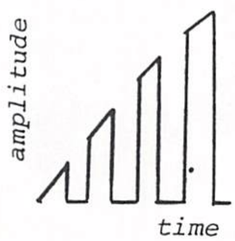
The pops you hear both at the beginning and the end of the gate are called "transients." They are caused by the gate's instantaneous voltage changes which snaps the speaker cone out, then back. Although transients help give acoustic instruments their characteristic sound, they are annoying in this patch. But then using the KBD gate is not the usual way of controlling a VCA. A more typical way is to use an

ENVELOPE GENERATOR.

Connect the KBD gate to the AR-312 envelope generator (EG) gate input and the KBD trigger output into the envelope generator trigger input. Then patch any one of the envelope generator outputs into the VCA control input #4; set the 4 EG controls to 0 and press a key. You should hear a single transient. Set the attack time on the EG to 2 and press the key again. Now you should hear the sound fading in fairly quickly and dropping out suddenly with a transient click. Increase the attack time to 5 and try it again. Finally, try the maximum attack time, being sure to hold the key down until the sound drops out. Since you know what this sounds like, you can probably imagine what it would look like on a scope. With a pulse as the audio signal, and an attack time of 10, it would look like this.



And an attack time of 5 would look like this.



The second control is the decay time control. For now, set this to 0 and we'll come back to it later. The third control is the sustain level. Set the attack time to whatever you want, and the sustain level to 10. Play and hold down a key. With the key held down, slowly turn the sustain level down and listen to the change. As you can hear, the sustain control is a level control (the other three controls are time controls). It determines the voltage level at which the envelope sustains. Lifting the key drops the KBD gate to 0 and ends the sustain level. Releasing the gate also starts the release time.

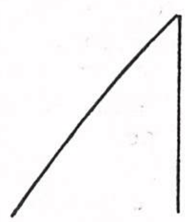
Choose another attack time, still 0 decay time, a sustain level of 10 and a release time of 0. Press a key, hold it, then release it. There is still that transient pop at the release. Now set the release time to 10 and try it again. Try a few other release times just to acquaint yourself with the control. When you feel that you understand what the release time controls, put it back to 10, play a key, and subjectively "measure" the release time. From the time you lift the key until the time the sound fades out is probably about 4 or 5 seconds, right? Without changing the release time, set the sustain level to 5 and try it again. Take a few minutes and consider how changing the sustain level can alter the release time.

Finally, go back to the decay time and try these settings: 0, 10, 10, 0. Play a key and listen to the result. Now try these settings: 0, 0, 10, 0, and compare it to the other decay time setting. Try some other decay settings and see if there's any change. Don't worry, your envelope generator is not malfunctioning. To prove it, try this setting: 0, 10, 5, 0. Again play some notes and slowly change the decay time. The decay time control determines the amount of time the envelope generator needs to go from the attack level (which is always 10 volts) to the sustain level (which is variable). With the sustain level at 10, the decay time control has no effect. By now you have a fairly good idea of what the envelope generator does; here is how it does it.

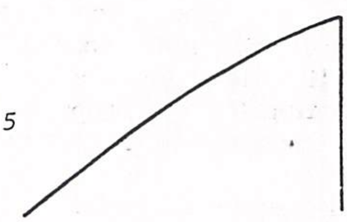
The ADSR envelope generator requires both a gate and a trigger. A trigger is a very quick pulse of about 10 volts amplitude and 10 mS. (milli-seconds) duration. The KBD produces a trigger whenever a key is depressed. When both a gate and a trigger are present at their respective inputs of the envelope generator, the attack time starts. The attack time is controlled by the control knob, but the attack level is always 10 volts. The attack time control, then, determines the length of time it takes the envelope generator to go from 0 volts to 10 volts. The minimum attack time of the AR-312 is 2 mS.; the maximum attack time is 4 seconds. Here are some examples of various attack times.

attack 0

attack 2



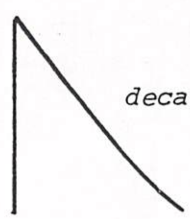
attack 5



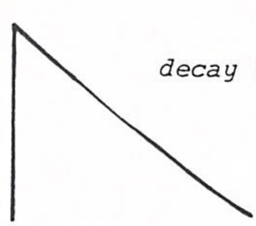
attack 8

The decay time control determines how long the envelope generator takes to go from the 10 volt attack level to the sustain level. In the following diagrams, the attack time is 0 and the sustain level is 0. Only the decay time is being changed.

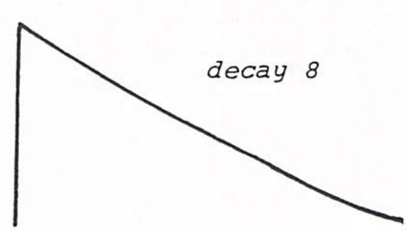
decay 2



decay 4

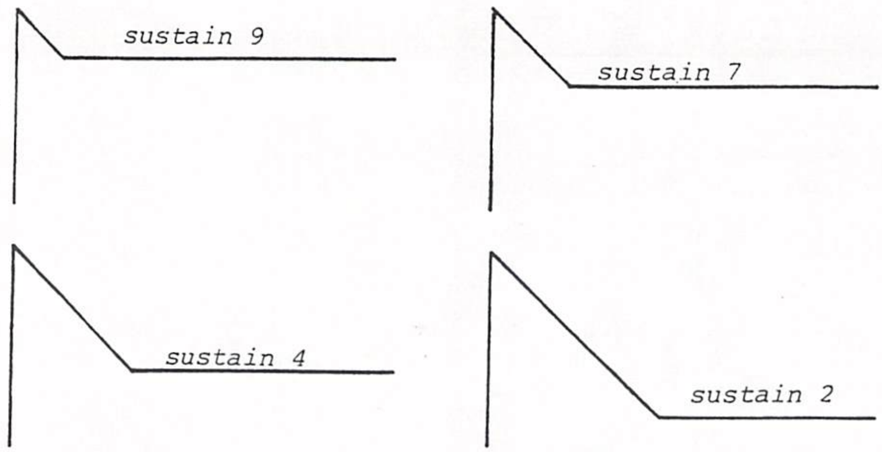


decay 6



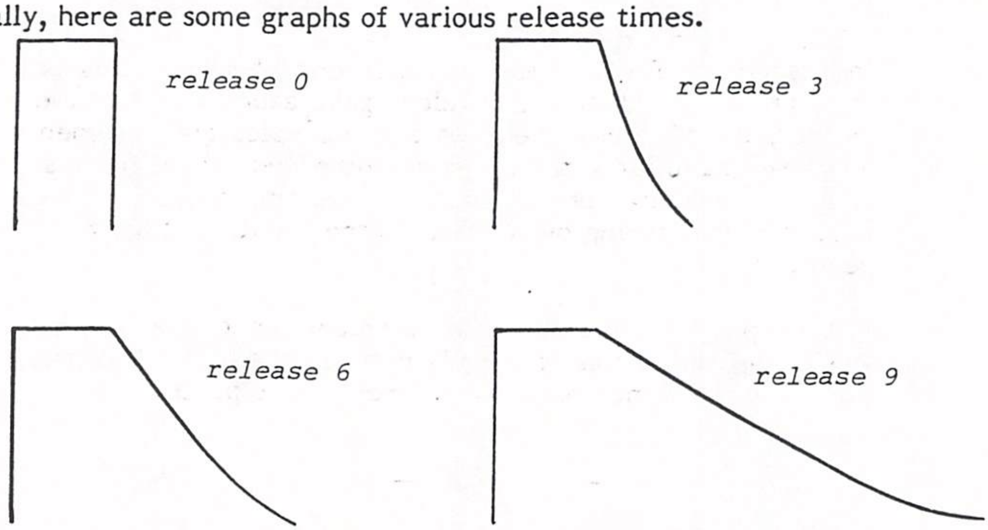
decay 8

This next set of diagrams show a constant attack time and a constant decay time. Look what happens to the decay time, though, when the sustain level is changed.



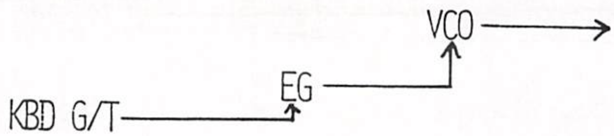
When the voltage has less distance to travel between the attack level and the sustain level, it gets there faster. In the previous patch where the sustain level was set to 10, the voltage had no distance to travel and so the decay time had no effect.

Finally, here are some graphs of various release times.

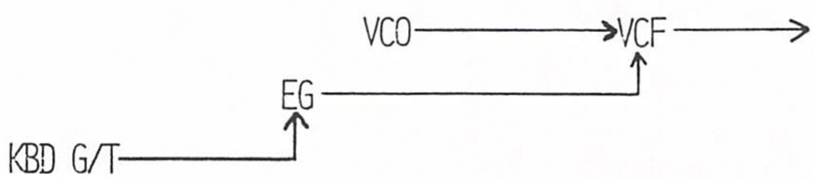


In the same way that changing the sustain level affects the decay time, changing it also affects the final release time. The release time determines how long the voltage takes to go from the sustain level to 0 volts. If the sustain level is low to begin with, it takes a shorter time to get to 0 volts.

By now, you can probably set up various envelope controls and picture the envelope generator output. Try this a couple of times and see if you are right. A good patch to check out your predictions is this one in which the envelope generator controls the frequency of a VCO.

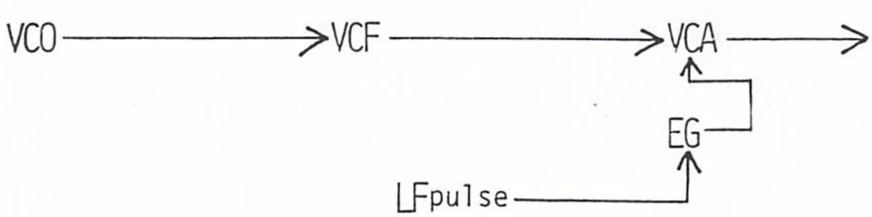


After you've tried a few different envelope settings, then set up this patch in which the the EG controls the cutoff frequency of the filter.



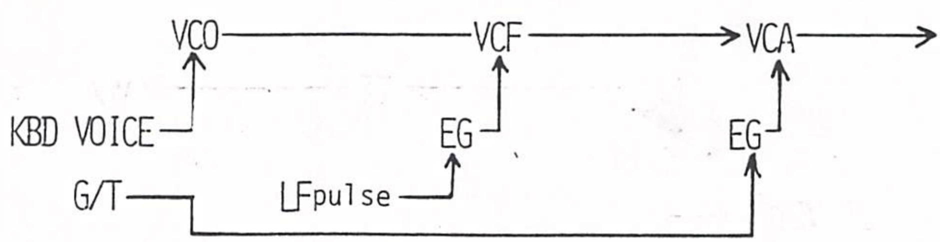
In this patch, you should be listening especially for the timbral change that occurs. Try to hear the difference between this patch and the previous one in which the envelope generator controlled the VCA's gain. In fact, all of these exercises and examples are designed not only to make you familiar with how your synthesizer works, but also to help you to learn to hear more accurately. Hearing a sound accurately is the first step in figuring out how to synthesize it. But back to the envelope generator.

It is possible to start the envelope generator on its cycle with something other than the KBD gate and trigger. One possibility is to use a low frequency pulse wave. Here is the patch.



Whenever you use a pulse to fire the envelope generator, you must remember to patch the pulse into both the gate and trigger inputs of the envelope generator. Each EG contains a set of 4 jacks labeled "patch." When you patch a signal into any one of these jacks, you can get the signal out of any or all of the other jacks. This configuration is sometimes called a "multiple" and it works like a Y cord. In this case, you can patch the pulse into one of the "patch" jacks and take it out from one of the other "patch" jacks and into the gate input. Then take it from another "patch" jack and into the trigger input. There are 4 possible LF pulses you could use in this patch: the pulse from the other VCO, either of the dual LFO pulse outputs, or the pulse from the clock. They all work equally well in this application and you might prove it to yourself by trying each of them. You might also try changing the frequency of the LF pulse and make some envelope changes and see what you can come up with. With some of the higher frequency LF pulses, it is possible that the pulse drops to 0 volts before the 10 volt attack level is reached. Whenever this happens, the envelope goes directly to a release stage and the slope of the release is determined by a combination of the decay and release controls. Also, it is possible that the leading edge of the next pulse initiates an envelope before the previous envelope has had a chance to release down to 0 volts. When this happens, the second envelope's attack starts from the amplitude at which it interrupts the first's release.

There's a host of patches you can try and experiment with. With the right settings on the envelope generators, you can patch up a pseudo "echo-plex" with this patch.

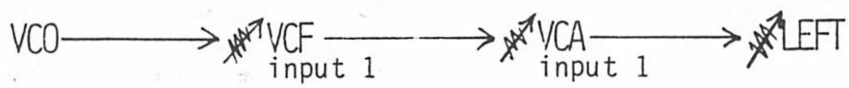


ATTENUATORS

One of the most useful functions on any synthesizer is also one of the simplest, the attenuator. To attenuate a signal means to decrease its amplitude, so an attenuator, then, is simply a manual amplitude control which can only decrease a signal's amplitude. Attenuators can be used both in an audio and in a control path. They are represented in block diagrams by this symbol,

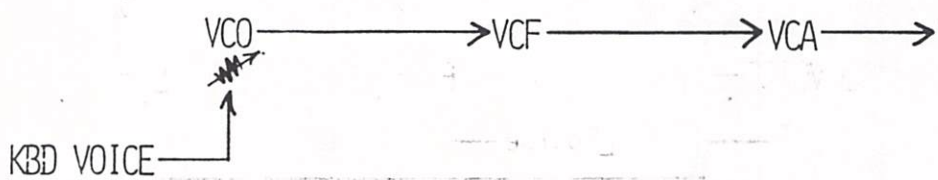
the electronic symbol for a variable resistor, which is what an attenuator actually is.

You've been using an attenuator in all these patches but probably haven't thought about it much. The controls marked "1" and "2" on the output and power module are attenuators for the inputs to that module. Don't be misled by the fact that opening an attenuator produces a greater amplitude. With the attenuator at maximum, the voltage output equals or is less than the voltage input. Other attenuators are the controls for the first audio input on both the VCA and the VCF. The control knob associated with the first control input of the VCO, VCF, and VCA are also attenuators. For convenience, almost all of the voltage controllable modules have input attenuators associated with them. In this patch,



if you use the first audio inputs to these modules, you have three possible audio attenuators: one at the input to the VCF, one at the input to the VCA, and one at the input to the output and power module. In the above patch, change the amount of attenuation at each of these three possible places. You can see that these attenuators all do the same thing--they decrease the amplitude of the signal. Two of these attenuators are redundant in this patch.

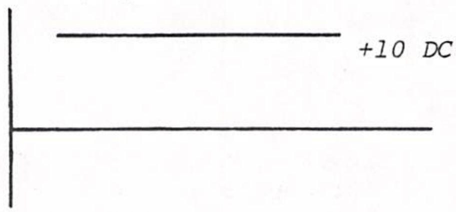
Attenuators can also be useful in a control path. Here is a useful example.



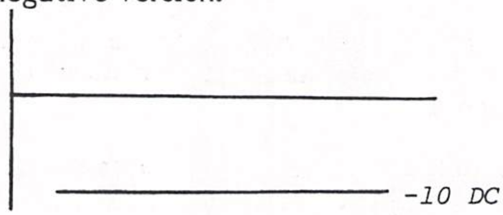
Patch the KBD voice into the first control input of the VCO. Set the attenuator about half way open and play the KBD. Each KBD octave that you play results in less than an octave pitch change. The KBD voltage is attenuated at the VCO input so that when you play a KBD octave, one volt goes into the attenuator and maybe only half a volt comes out. If you carefully set the attenuator so that as you play 2 octaves on the KBD the pitch changes by only one octave, then you have attenuated the KBD voltage to give you exact quarter tones. In this way it is possible to get all sorts of micro-tonal scales and you can invent your own intonation systems. A synthesizer's KBD is deceptively familiar.

DC BIAS

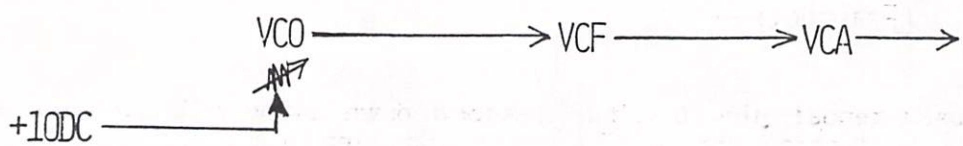
Another patch you can try is this one. On the output and power module are 4 jacks, 2 labeled "+DC" and 2 labeled "-DC." The voltage from these jacks is a non-fluctuating DC voltage called a "bias" or an "offset." The scope picture of a 10 volt positive DC bias is this.



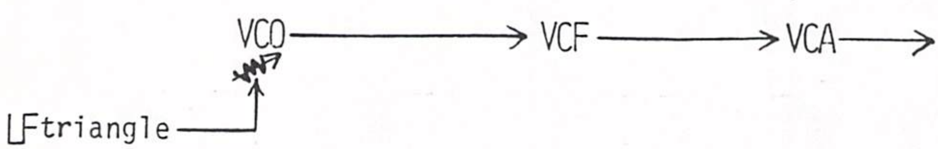
Here's the negative version.



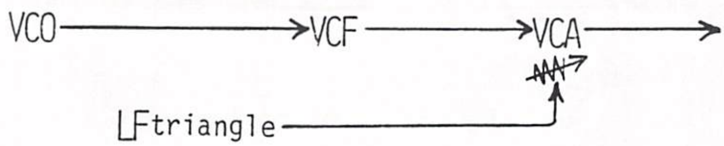
These, like any other voltages, can be used to control a signal's frequency, amplitude, or wave shape. Patch the +10 DC into the attenuated input of a VCO and change the amount of attenuation. The block diagram is this.



When you've attenuated to a point where the VCO's frequency is only an octave above its original frequency, then you've attenuated the +10 DC down to one volt. If you attenuate further so that the pitch is a perfect fifth above the original pitch, then you've attenuated the voltage to 7/12 volts. Each 1/12 volt is equal to a semi-tone change and there are 7 semi-tones in a perfect fifth. Using just a VCO, an attenuator, and your ears, you can patch up a passable volt meter, and in fact, you have just done so. Another patch to try is this one.

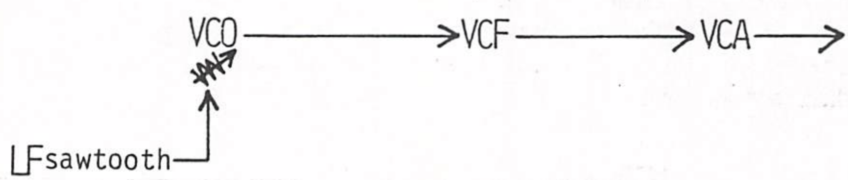


Before, we looked at a patch similar to this and, if you'll remember, the 10 volts LF sine drove the audio VCO up and down a total of 10 octaves. By attenuating the 10 volt sine down to maybe 1/20 or 1/30 volts, you can use it as a vibrato. Just so we can keep the terms straight, this pitch change is a vibrato while a similar amplitude change is a tremolo. A tremolo patch is this.



In an acoustic instrument, it is very difficult to produce either just a vibrato or just a tremolo; they almost always occur simultaneously. Naturally any other LF wavefrom can be substituted for the triangle. A pulse or square wave is particularly useful in synthesizing fast trills. If the controlling VCO is in the audio range, the result will again be a frequency modulation timbre. We'll again defer discussing this in detail until later. For now, here's another problem.

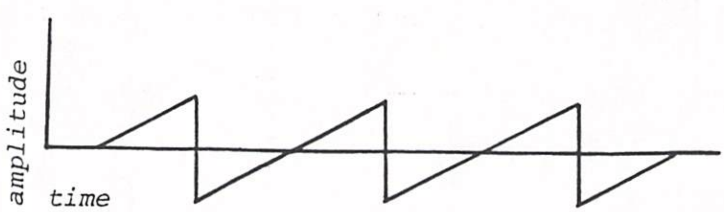
Assume you want to patch a series of one octave ascending pitch glides. That seems fairly straightforward. You'd do it like this.



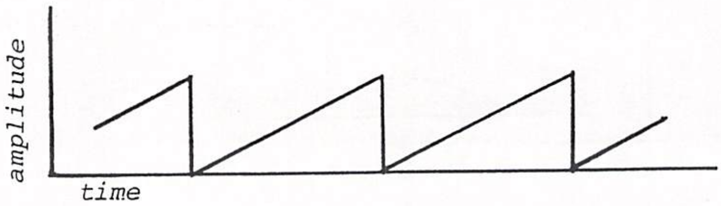
Just attenuate the 10 volt LF sawtooth down to one volt. Now un-plug the patch cord from the LF sawtooth and listen to the pitch of the VCO. When you plug the patch cord back in, you'll notice that the pitch glide starts lower than the original pitch. The problem is: how can you make the pitch start where you originally set it and then ascend upwards one octave? To accomplish this, you use the mixer.

MIXER

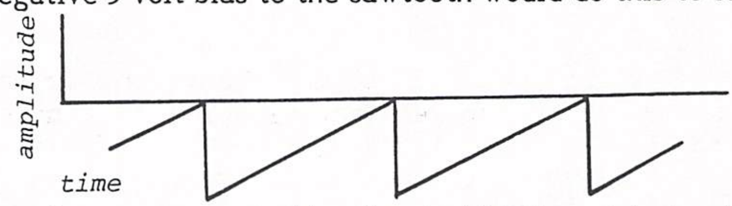
Back in the beginning, we drew a scope picture of a sawtooth as it was generated by the AR-317 VCO. It looked like this.



This waveform is balanced across 0 volts. It goes 5 volts positive and 5 volts negative. If you add a +5 DC bias to the signal, it would look like this.

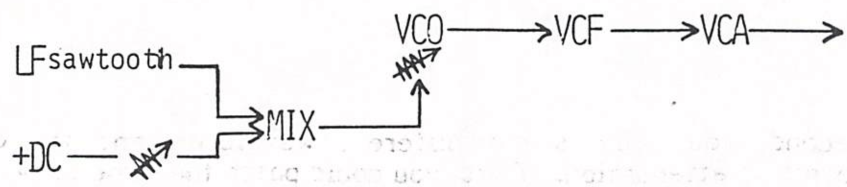


Adding a negative 5 volt bias to the sawtooth would do this to it.

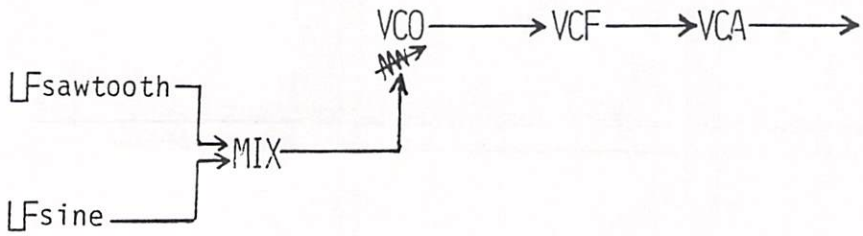


The sawtooth wave generated by the dual LFO module is already an unbalanced, positive sawtooth

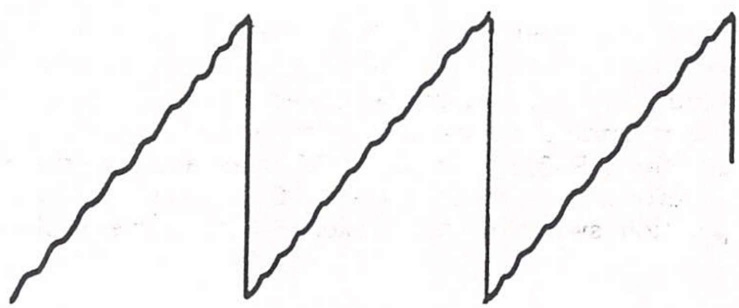
In a synthesizer, the module that allows you to add voltages together is the mixer. The AR-323 dual mixer is actually two independent mixers whose outputs can, in turn, be mixed together. So, in order to get this ascending pitch glide starting on the original pitch, mix the LF sawtooth from the AR-317 with a +5 DC bias and use the resulting unbalanced waveform to control the other VCO. Here's the patch; make sure the 2 position switch on the mixer is set to the positive (plus) position.



The order in which you attenuate will make a difference. First patch the audio path. Then patch the LF sawtooth and the + DC bias into the mixer, putting the DC into an attenuated input (input 1 or 2). Then patch the mixer output into the attenuated control input of the VCO. The first attenuator you should use is the one on the mixer. Adjust it so that the frequency of the VCO never goes below the original frequency. You are again using your ears as a volt meter to attenuate the 10 volt DC signal down to exactly 5 volts. Now use the attenuator on the VCO to adjust for exactly a one octave change. When you first start tuning these patches, being accurate may be difficult. It gets easier as you become more practiced at it. Try another patch using the mixer to mix control signals.



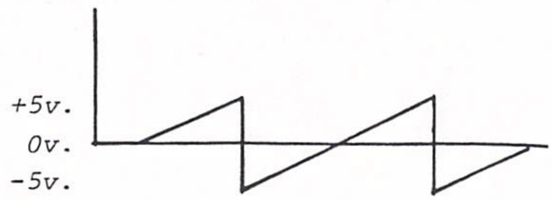
This patch offers a lot of possibilities. First you could try having the LF sawtooth at a very low frequency, say 0.2 Hz., and have the sine at about 7 Hz. Then you could attenuate the sine at the mixer so that it becomes a wide vibrato control. You should hear something that looks like this.



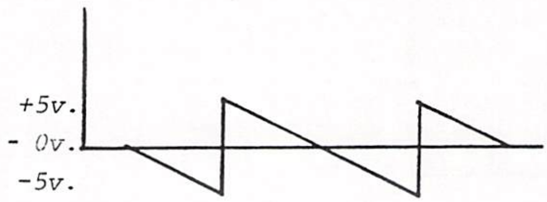
Second, you could select different waveforms and try different amounts of attenuation. Third, you could patch the same LF waveform, but from different oscillators, and have one much faster than the other. Fourth, you could mix two envelope voltages together, or mix an envelope voltage with a LF waveform, or mix 2 envelope voltages with 2 LF waveforms, or.......... the possibilities are legion.

INVERTERS

The first two inputs of each mixer have both attenuators and inverters. The inverters change the polarity of the signal. Set up this patch then flip the switch to the negative position. This waveform

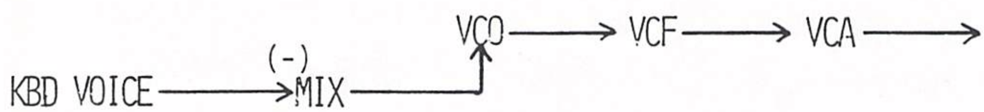


becomes this.

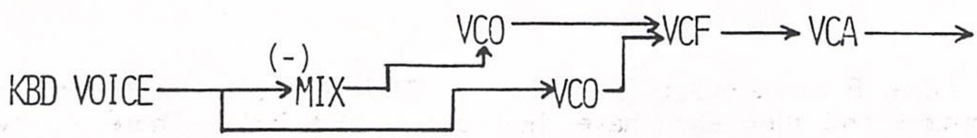


It remains a sawtooth and its harmonic spectrum remains the same. It is just put 180 degrees out of phase.

Although all voltages can be inverted, an interesting inversion is this one.

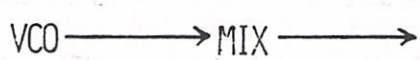


A synthesizer's KBD is deceptively familiar and as you play up the KBD, the pitches descend. Another way of using this idea is in this patch.

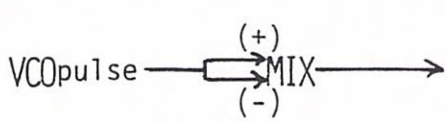


Start with the VCO's tuned to unison and, as you play up and down the KBD, the VCO's change in opposite directions but by equal amounts.

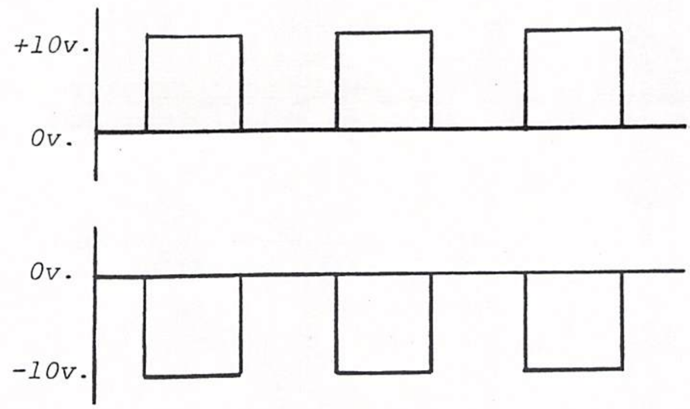
Now try this patch



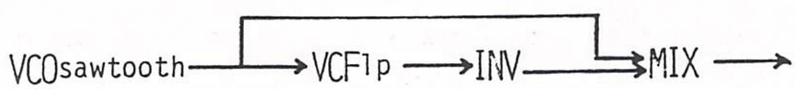
and alternately flip the switch back and forth to prove to yourself that inverting an audio waveform makes no sonic difference. Try this experiment.



An inverted waveform mixed with its uninverted form will cancel each other leaving 0 volts as the result. You can see that just by visualizing the waveforms and adding the voltages.



A useful patch using an inverter in the audio path is this one, making a high pass filter.

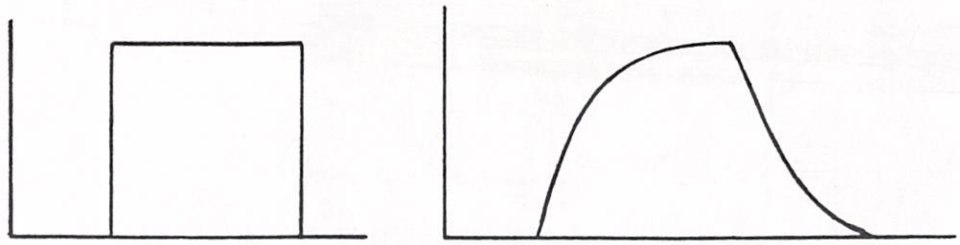


Mixer B works exactly like mixer A. They are each independent 4X1 mixers and they each have their own output jack. There are two additional output jacks. Output A + B is the combined outputs of mixer A and mixer B. It is possible to mix the outputs of the two mixers together simply by taking the output from this jack. In this case, the two mixers become a single 8x1 mixer. Output A - B is the output of mixer A added to the inverted output of mixer B. Again, this is an 8x1 configuration, but with 4 of the signals inverted.

The Aries has 6 inverters. Four of them are in the mixers. The fifth is a function of the AR-324 module and the sixth is in the AR-331 module. These last two inverters each have one input, one output and an input attenuator and they work the same way as the other inverters.

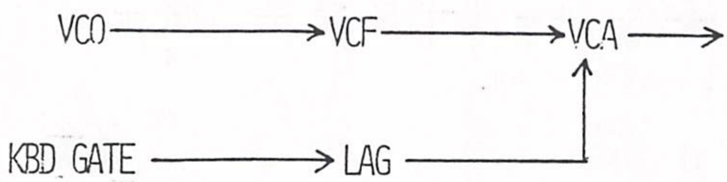
LAG

In addition to the dual LFO's and the inverter, the AR-324 also contains a lag function. The lag is a low pass filter with a -6 dB/octave slope. Although it may be used to filter audio signals, it is more commonly used to filter sharply rising control signals. A LF pulse filtered through the lag comes out looking like this.

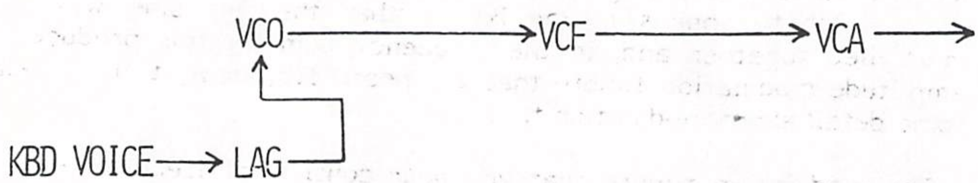


original pulse *lagged pulse*

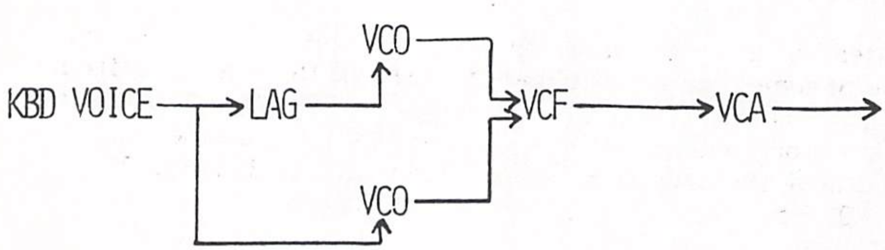
It rounds off all the sharply changing voltages. The lag time is variable from 1 mS. to 1 Sec. and is controllable by the lag (sec.) control. One of the ways in which it is commonly used is as an "envelope shaper." By lagging the KBD gate, you eliminate the transient pops from this patch.



Another way the lag is useful is in creating a KBD portamento.



But this can be accomplished at the KBD interface. Using 2 VCO's and having portamento on only one of them must be patched up.

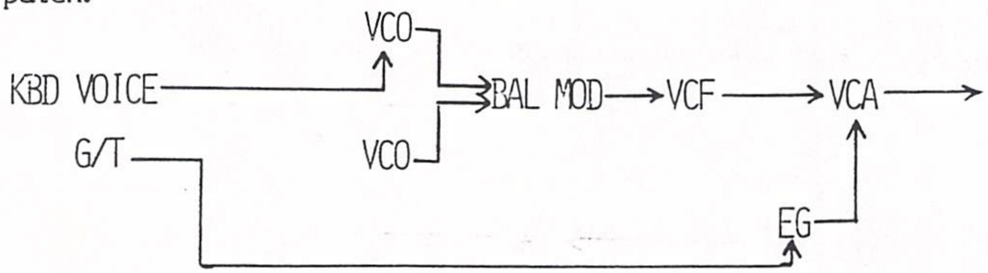


This patch is particularly useful in simulating the chorus effect of instruments playing in unison. In this application, use just the smallest amount of lag time.

There is another -6 dB/octave low pass filter available on the Aries 300 and that is the "test" output from the AR-314 low pass filter. In addition to providing a different filter slope from the normal AR-314 output, this output provides a signal shifted by 90 degrees with respect to the input signal. The signal from the normal AR-314 output is shifted by 180 degrees with respect to the input signal.

BALANCED MODULATOR

There are a few modules on the System 300 that we've said nothing about yet; the balanced modulator is one of these. You've undoubtedly heard the effects of a balanced modulator (BM). It's used to create sound effects in almost every science fiction movie. Another of its popular uses is in imitating bells, gong sounds, and klangs. Try this patch.



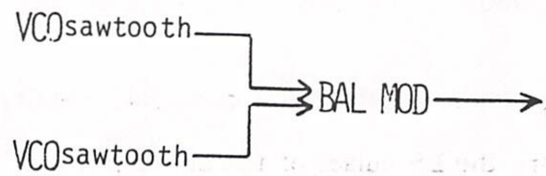
Set the envelope to simulate a bell's envelope (sharp attack, long decay, medium sustain, and long release) and try playing some different KBD notes. What happens in the BM is that the two sine waves are multiplied together and, in the frequency domain, this produces the amplitude modulation timbre that you hear. It's finally time to talk in some detail about modulation timbres.

The modulation timbre that you hear consists of frequencies which are equal to the sum and the difference of the two input frequencies. If, for example, one sine has a frequency of 600 Hz. and the other sine's frequency is 720 Hz., the resulting timbre would have harmonics at 120 Hz. (the difference) and at 1320 Hz. (the sum).

If, instead of using 2 sine waves you use a sine and a sawtooth, then the timbre becomes more complicated. Assume the sine has a frequency of 200 Hz. and the sawtooth's frequency is 350 Hz. To determine the timbre's components, first you have to determine the harmonic spectrum of the sawtooth. It has harmonics at 350 Hz., 700 Hz., 1050 Hz., 1400 Hz., 1750 Hz., 2100 Hz., 2450 Hz., etc. Then you have to add and subtract the sine's frequency from each of the sawtooth's harmonics. We'll do this one for you, but then you should do one on your own.

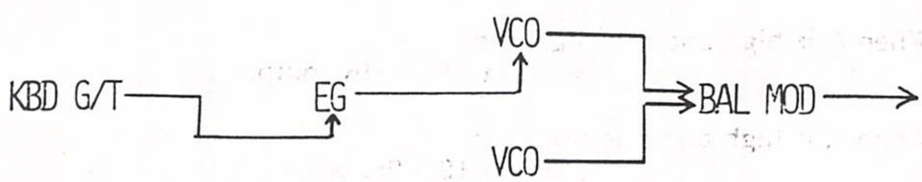
350	−	200	=	150
350	+	200	=	550
700	−	200	=	500
700	+	200	=	900
1050	−	200	=	850
1050	+	200	=	1250
1400	−	200	=	1200
1400	+	200	=	1600
1750	−	200	=	1550
1750	+	200	=	1950
2100	−	200	=	1900
2100	+	200	=	2300
2450	−	200	=	2250
2450	+	200	=	2650

If you really feel ambitious, figure the harmonic spectrum for this timbre.



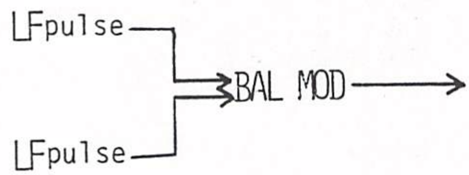
Each harmonic of each sawtooth must be added and subtracted from each harmonic of the other sawtooth. Even without doing all the arithmetic, you can see that the more harmonics that the waveforms have, the more complex the modulation timbre will be.

Photon torpedos!!



Try some variations of your own with these patches. By now you have an idea of what will work and what won't--which modules will cause what changes and how to patch up what you want your synthesizer to do. While there's not an infinite number of possibilities, there is an uncountably finite number, and it's up to you to decide which are useful for you.

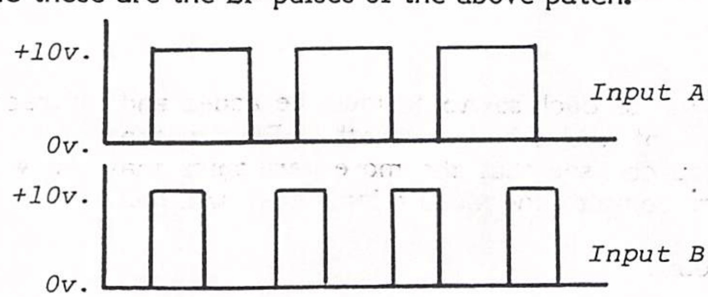
So far, the patches involving the BM have treated it in the frequency domain in terms of the timbre it produces. It's also possible to consider its output in the time domain in terms of the wave shape it produces. For the first example, consider this patch.



The BM is a four quadrant multiplier. This means that the BM will produce an output regardless of the polarity of either of the input signals. The formula for determining the output of the BM in the time domain is this.

(Voltage Input A x Voltage Input B)/10 = Voltage Output

Assume these are the LF pulses of the above patch.



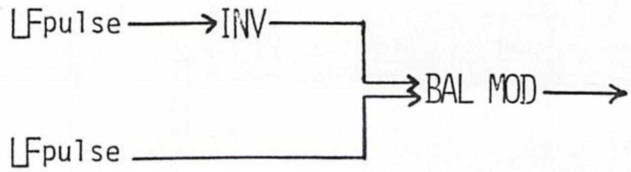
When A is high and B is high, then
10v. x 10v./10 = 10v. output

When A is high and B is low, then
10v. x 0/10 = 0v. output.

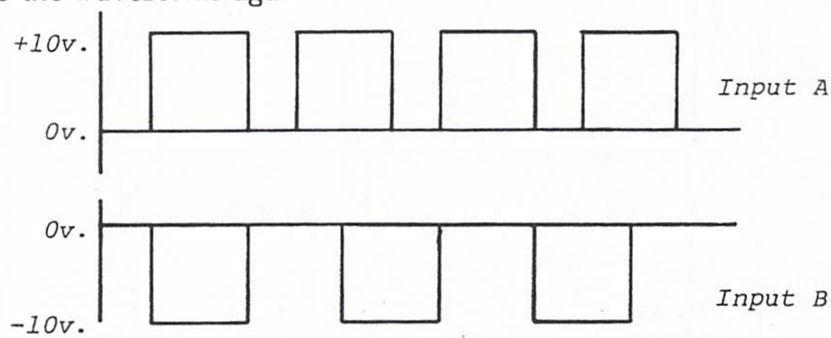
When A is low and B is high, then
10v. x 10v./10 = 10 v. output.

Of course when A and B are both low there is no output.

The BM is a four quadrant multiplier. This means that the BM will produce an output regardless of the polarity of either of the input signals. Consider this patch.



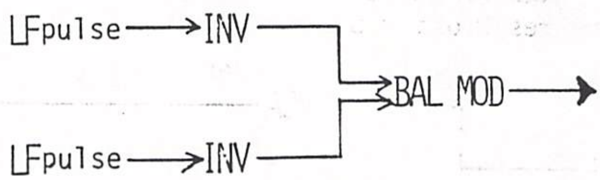
Here are the waveforms again.



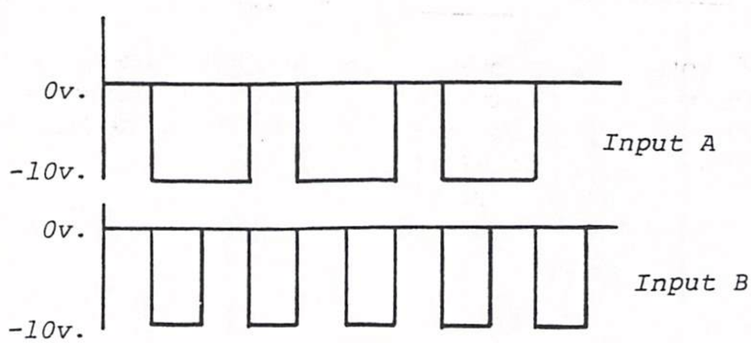
And the arithmetic

$$(+10 \times -10)/10 = -10; \quad (+10 \times 0)/10 = 0; \quad (0 \times -10)/10 = 0$$

Here's just one more to consider.



And the waveforms, again.

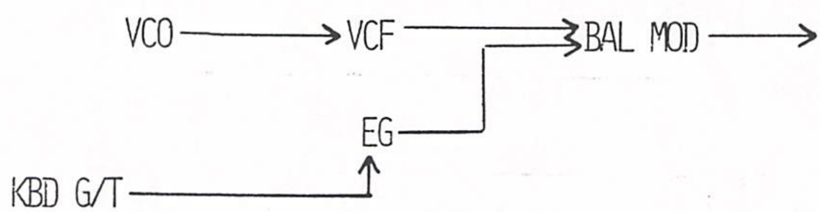


And just one formula.

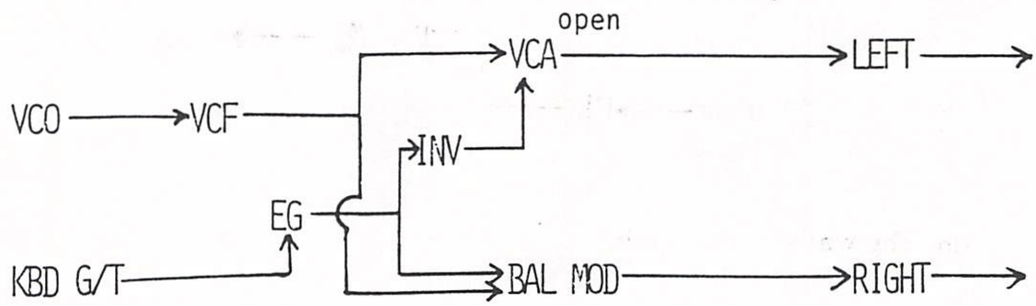
$$(-10 \times -10)/10 = +10$$

Remember that two negative numbers multiplied together produce a positive number, and so it is with the balanced modulator.

The BM can operate as an extra VCA. It multiplies one input signal times the other and it makes no difference if one is an audio signal and the other is a control signal. If your VCA is already being used in the patch, the BM used in this manner can be very useful.

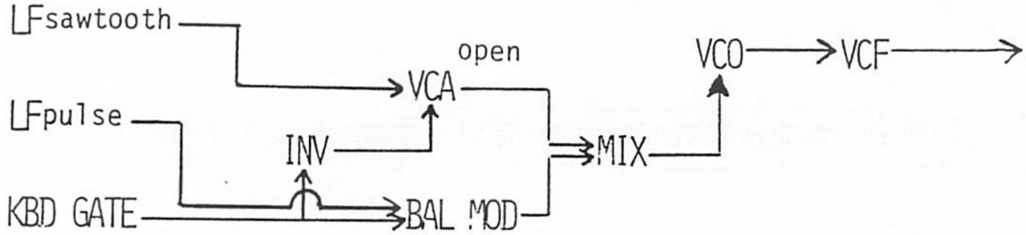


By now you have probably figured out that the VCA is also a voltage multiplier. There is one basic difference. Unlike the BM, the VCA has designated audio and control inputs. The audio input will accept either positive or negative voltage while the control input will only open the VCA with a positive voltage. If the VCA is "open" (initial gain turned up), the VCA will close when a negative voltage is applied to its control input. The following patch is a spacial location patch which uses one voltage source, the EG, to simultaneoulsy open the BM and close the VCA. Obviously, your synthesizer must be monitored through a stereo system to hear the result of this patch.



The EG voltage which opens the BM also gets inverted and closes the VCA. The result of the patch is that when a key is depressed, the sound comes from the right speaker, and it comes from the left speaker when there's no key depressed. Automatic panning is possible by using a LF pulse to fire the envelope generator.

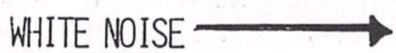
Other kinds of "either-or" switching patches are also possible. This patch switches between two control voltages.



When the LF pulse is high, the VCO is controlled by the LF sawtooth. When the LF pulse is low, the VCO is controlled by the sine. In both cases the KBD voice always controls the VCO. As you can see, there are innumerable possibilities using the BM and the VCA in combination. Again, it's up to you to determine what is musically useful for your purposes.

NOISE

Another module which we've yet to mention is the noise generator. First, just listen to white noise by patching it into the output and power module. This is the sound you hear when your FM tuner is between stations, or when you turn your amplifier all the way up with no signal going into it, or when you bounce tracks too many times on your tape recorder. In fact, these were methods used to obtain white noise sounds in the classical electronic music studio.



White noise can be specifically defined as "a signal which contains equal energy per cycle throughout the audio range." It is a completely random signal and it is aperiodic. Its spectrum consists of energy which is constantly changing in frequency and amplitude so that it is just as likely to have energy at one particular frequency as it is to have energy at any other particular frequency. White noise is electrically flat through 10 octaves, but it doesn't <u>sound</u> flat.

White noise seems to have more energy in the high frequencies than in the low frequencies; it seems to "hiss" more than it "rumbles." Here is the reason. The audio range, as you know, is divided into 10 octaves and each higher octave is achieved by doubling the frequency of the last octave. If you figure out the frequencies of the octaves starting with 20 Hz., they look like this:

20 Hz.	-	*40 Hz.*
40 Hz.	-	*80 Hz.*
80 Hz.	-	*160 Hz.*
160 Hz.	-	*320 Hz.*
320 Hz.	-	*640 Hz.*
640 Hz.	-	*1280 Hz.*
1280 Hz.	-	*2560 Hz.*
2560 Hz.	-	*5120 Hz.*
5120 Hz.	-	*10240 Hz.*
10240 Hz.	-	*20480 Hz.*

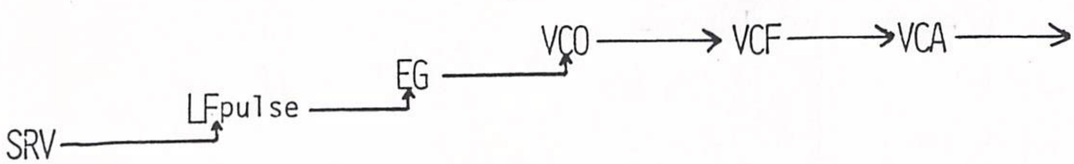
The distance from 20 Hz. to 40 Hz. is obviously 20 Hz., but it's also an octave. Likewise the distance from 10,240 Hz. to 20,480 Hz. is also an octave. Since white noise is flat from 20 Hz. to 20,480 Hz., half of its energy is found in the highest octave. Although it is electrically flat, it is not acoustically flat. Pink noise is an acoustically flat signal.

PINK NOISE ⟶

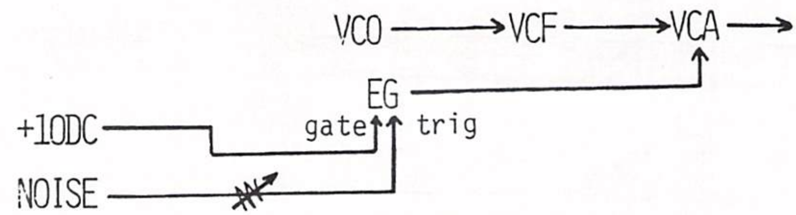
You can hear the difference. Pink noise is noise which contains equal energy per octave throughout the audio range. Pink noise sounds flat. Through a resonant filter it simulates unpitched, or sometimes quasi-pitched sounds like surf and thunder. By increasing the resonance and shortening the envelope, some percussion sounds are possible. Both white and pink noise can be used in either an audio or a control path.

Slow random voltage is LF noise which is used only as a control signal. In this patch the VCO's frequency is varying randomly.

Here, the envelope generator is being fired randomly.



Try patching a +10 DC bias into the gate input of the EG, turning the sustain level down, and patching attenuated white noise into the trigger input. By carefully attenuating the white noise, you can get



More random envelopes.

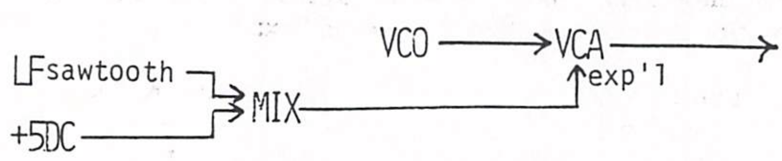
Rather than going on to some other modules, let's go back and clear up a few points about some of the modules we've been using all along. First, the mode switch on the VCA.

In the linear mode the VCA works like this. The amplitude of the signal that the VCA outputs is equal to the amplitude of the input signal times the gain factor. The gain factor is the sum of the control voltages divided by ten. For example, if you are controlling the VCA with an EG putting out a 10 volt envelope and have attenuated the envelope to 5 volts at the VCA control input, then the gain factor is 5 volts divided by 10, which equals 0.5. If the audio signal is 10 volts, then the output of the VCA is

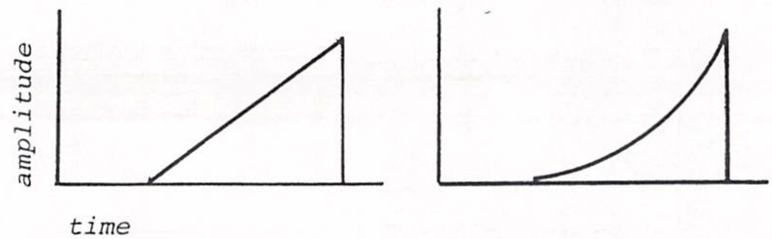
10v. x 0.5 = 5v. output

<u>The VCA does not amplify!</u> The VCA's voltage output is always less than or equal to the voltage at its input. When the voltage input equals the voltage output, then the gain factor is 1 and this is called "unity gain." Just to keep things straight, the initial gain control is actually a +DC voltage which is permanently patched into the VCA's control. All other control voltages into the VCA are summed with the initial gain.

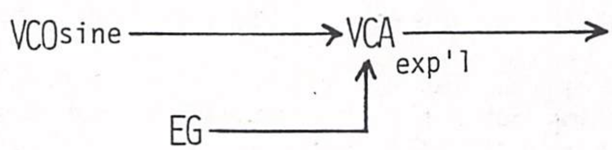
In the exponential mode, all control voltages (except the initial gain) are routed through an exponential converter before going to control the VCA. The effect of exponentializing the control voltages can be heard in this patch.



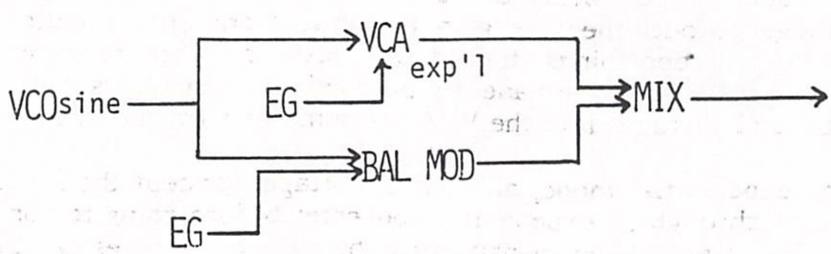
Try it first with the switch in the linear position, then switch it to the exponential mode. If you were to look at the sawtooth and the exponentialized sawtooth on a scope, you would see this happen to it.



With the switch in the exponential mode, the VCA's output increases by 10 dB for each positive control volt. The result of exponentializing an envelope's voltage is to get a quicker and more percussive event. To synthesize most wooden percussion instruments, use an exponentialized envelope. Here's a wood block patch.

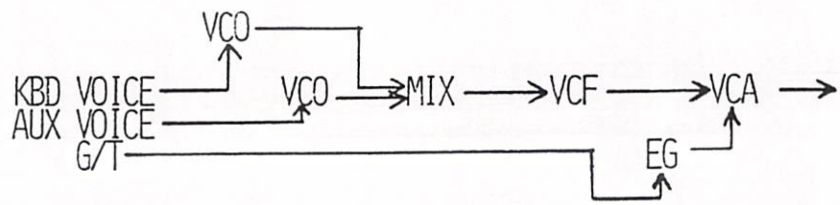


When you want a percussive attack but with a longer duration than is available with the exponentialized envelope, you can use this patch using the BM as an additional VCA.

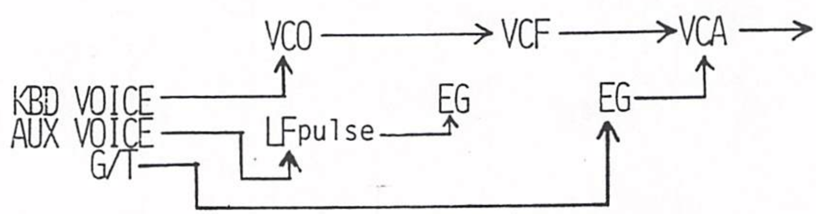


In block diagrams, the VCA mode switch is presumed to be in the linear position unless specifically marked "exp'l."

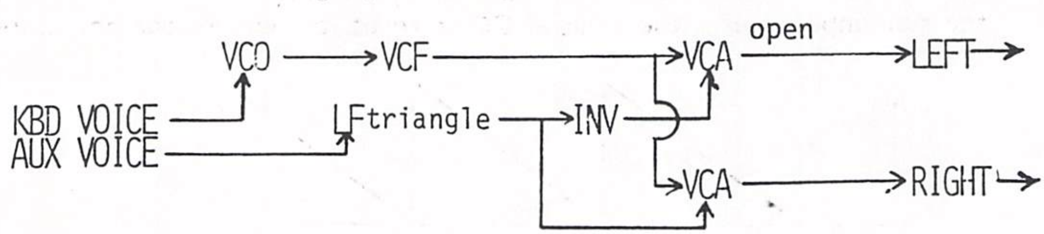
Another point to clear up is the aux voice of the KBD. The KBD can produce two control voltages simultaneously--one equivalent to the lowest key depressed and the other equivalent to the distance between the lowest key and the highest key. This patch uses this feature to control two audio frequency VCO's to simulate two-voice polyphony.



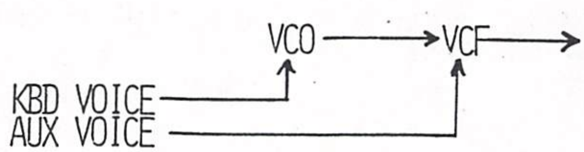
You can use the aux voice output to control other than an audio frequency VCO. In this patch, it controls the frequency of a LF pulse which fires an EG.



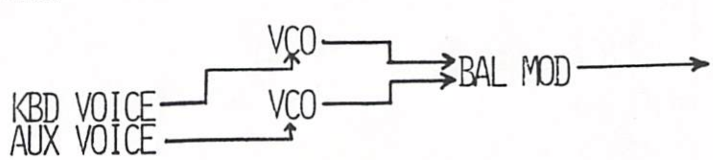
It can also control the panning rate.



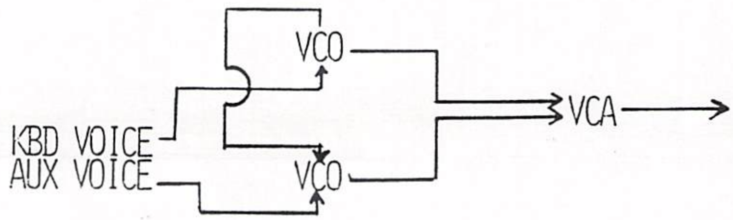
It can control the timbre when used like this.



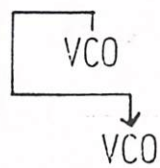
Or like this.



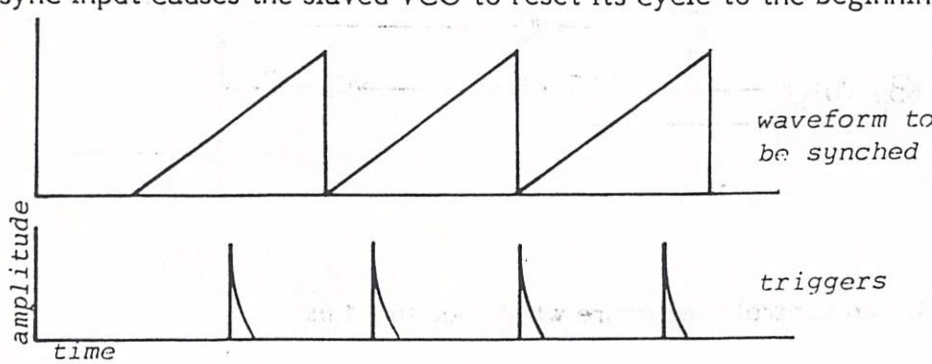
With the VCO's synched together, it is also a timbral control.



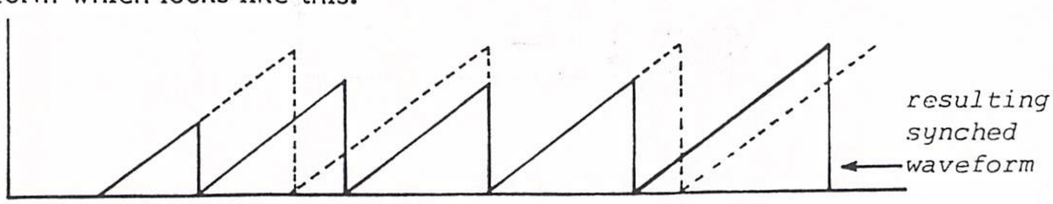
The sync input on the VCO's and LFO's permit two or more oscillators to be tuned to the exact same frequency. In order to patch it, patch the pulse output from one of the VCO's into the sync input of the other VCO. They now have exactly the same frequency and any change in the frequency of the first VCO results in a similar change in the second VCO. In a block diagram, sync is represented by an arrow coming from the top of one VCO (called the "master") into the top of the other VCO (called the "slave.")



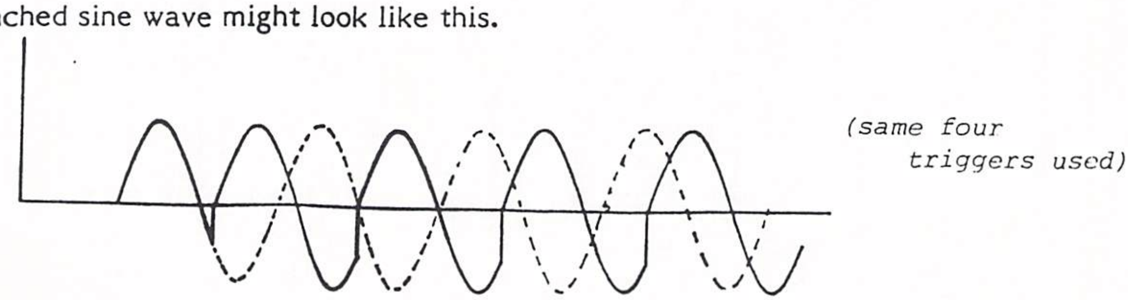
What happens internally is that any steeply rising voltage patched into the sync input causes the slaved VCO to reset its cycle to the beginning.



Instead of putting out a sawtooth, the slaved oscillator is outputting a waveform which looks like this.

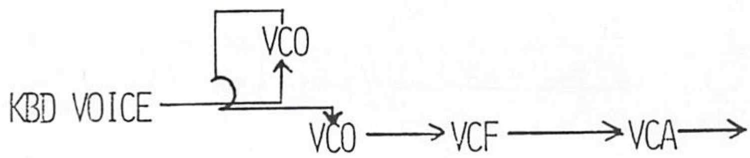


A synched sine wave might look like this.



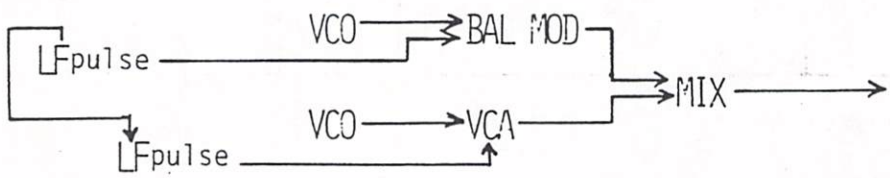
In the frequency domain these altered waveforms will produce all kinds of strange and useful timbres. Vocal sounds and timbres with interesting formants are possible using the sync.

One way of using the sync is to patch a voltage into the frequency control of a slaved VCO. This voltage cannot change the frequency of the VCO (it's synched) but it will change the waveform.

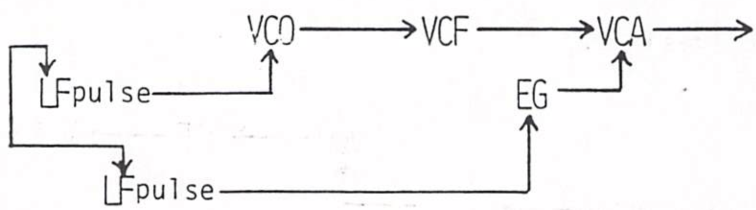


Notice that you're not listening to the master VCO in this patch.

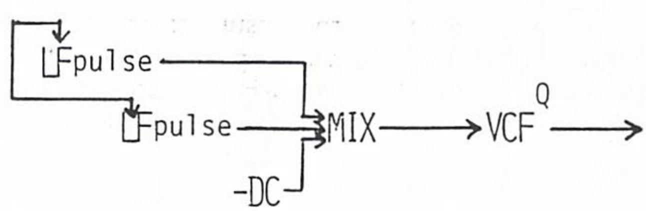
The LFO's can also be synched together and can produce repeating patterns. One example is this.



Another is this.



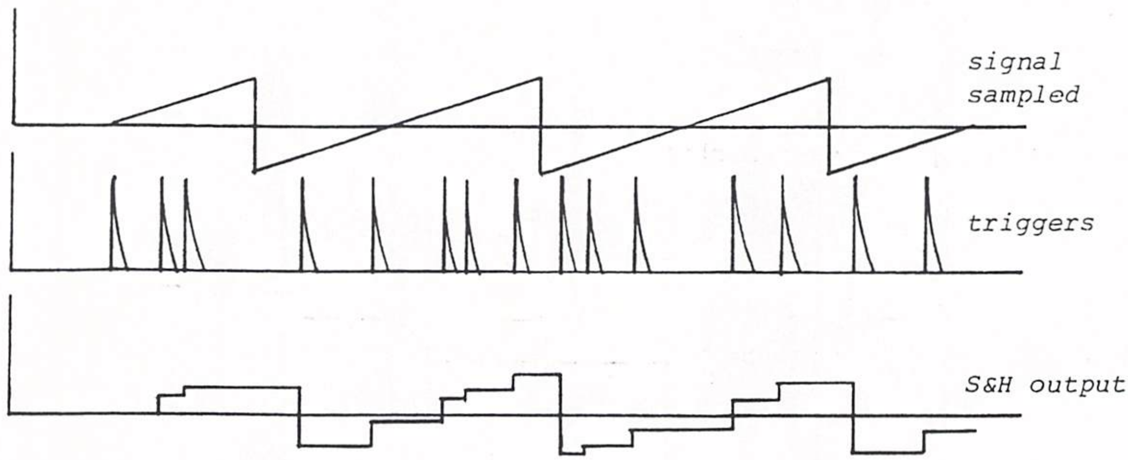
When the LFO's ring the resonant filter, the patch sounds like water drops or temple blocks.



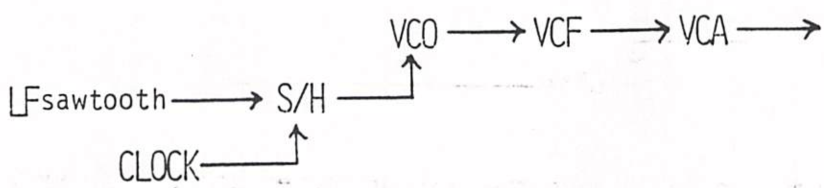
Another way of using the sync is with the sample and hold.

SAMPLE AND HOLD

The sample and hold is a module that requires two inputs: an input for the signal to be sampled and an input for a sampling trigger or gate. With the switch on the sample and hold (S/H) in the "clock" position, the clock is normalled into the S/H trigger input and the clock frequency determines the sampling rate. The S/H samples the instantaneous voltage of an input signal whenever it receives a trigger at the trigger input. It outputs the sampled voltage and holds it until it receives another trigger causing it to sample again. This diagram should help to make its operation clear.

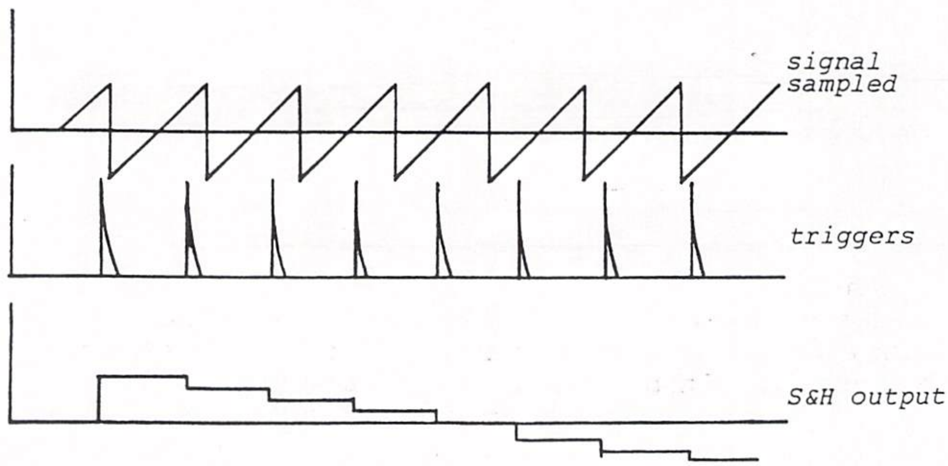


If you set up this patch,

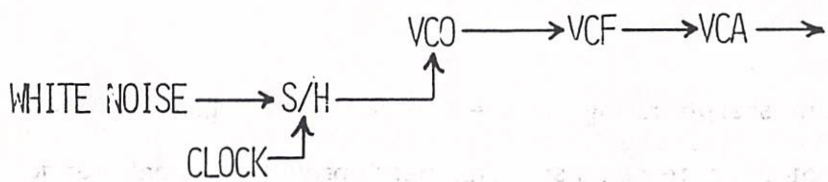


you can hear the result of the S/H. Make sure that the clock frequency is about 10 times greater than the frequency of the sawtooth. Also be sure that the S/H output level is turned up. This, by the way, is the only output attenuator on the entire synthesizer. All the others are input attenuators.

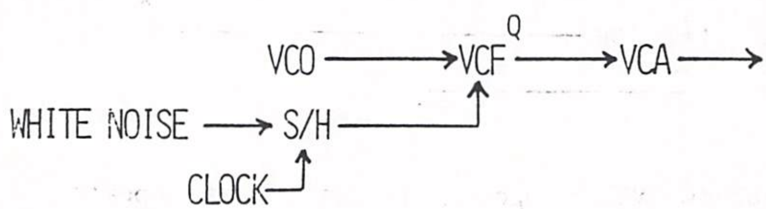
By varying the clock rate, it is possible to get a descending pitch pattern by sampling an ascending sawtooth. That would happen like this.



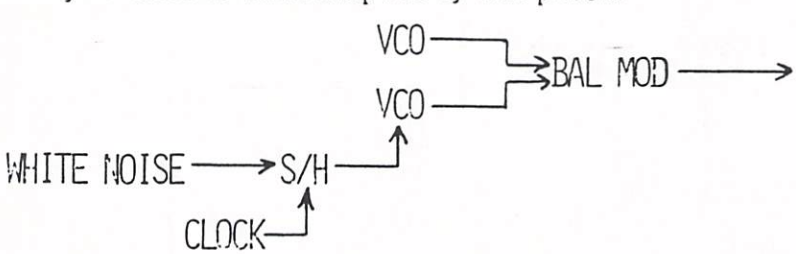
Of course, it's possible to sample any voltage. Sampling white noise in the following patch results in a series of random pitches.



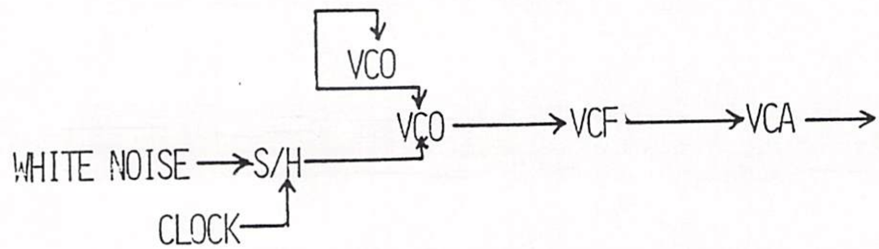
Try changing the output attenuator and limiting the overall pitch variance. The S/H output, like other voltages, can control frequency, amplitude, or wave shape. One way of controlling a signal's wave shape is shown in this patch which seems to have a lot of popular applications.



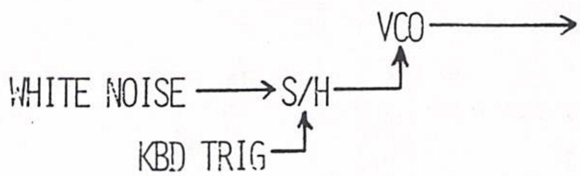
Another way to control wave shape is by this patch.



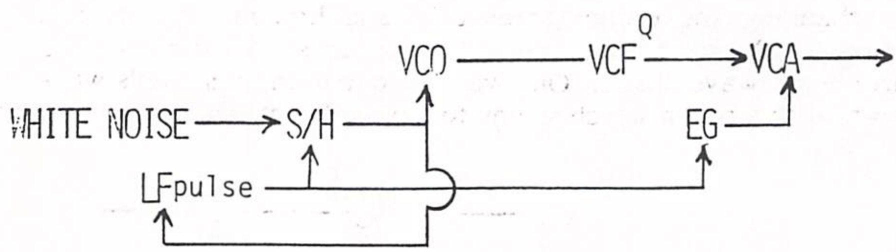
Finally, try this one.



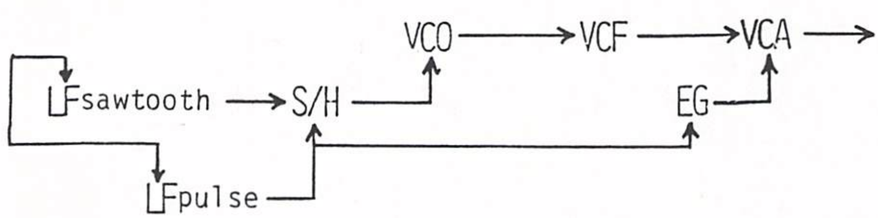
There are two additional ways of triggering the S/H. First, flip the switch to the external position, push the manual trigger button, and release it quickly. Each time you push the button, the S/H is provided with a trigger. It samples the signal and holds it until you push the button again. Another way is to patch any sharply rising voltage into the trigger input. Again the switch must be in the external position. One possible source of triggers is the keyboard trigger output.



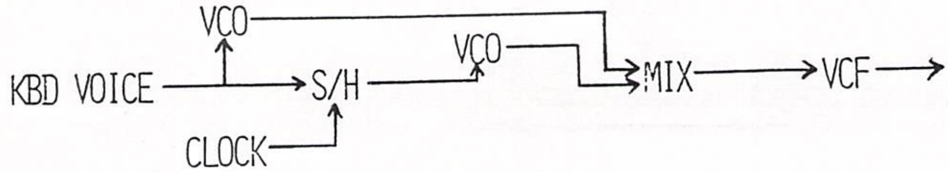
But any sharply rising voltage will do. When a gate or pulse is patched into the trigger input, the AR-318 will derive a trigger from the leading edge of the gate or pulse. This patch provides not only random pitches, but also random time events. Instant Subotnick!!



By synching two VCO's together you can generate repeating patterns.

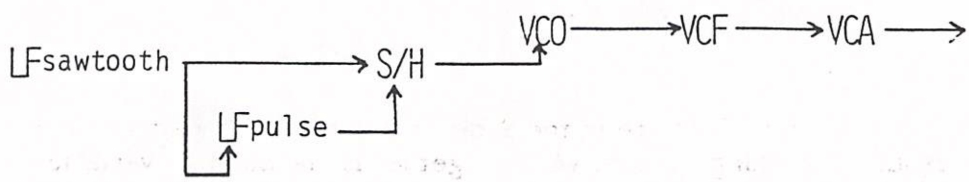


A potentially useful patch for KBD players is this one.



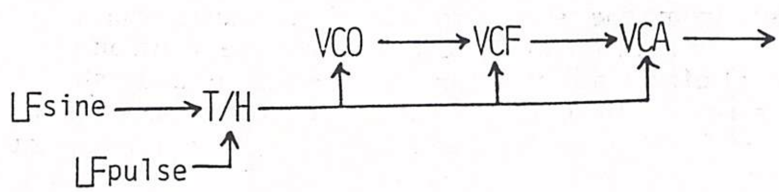
First, hold the lowest note on the KBD down and tune the 2 VCO's to unison. Second, play an octave higher on the KBD and, by using the output attenuator on the sample and hold, again tune the VCO's to unison. Third, set the clock frequency to about 1 Hz. Now, by playing a different note on the KBD each time the clock is halfway through its cycle, you can get one oscillator to follow the other one always ending up at unison.

This patch may also be useful. It produces a series of repeating pitches by patching up a kind of sync.

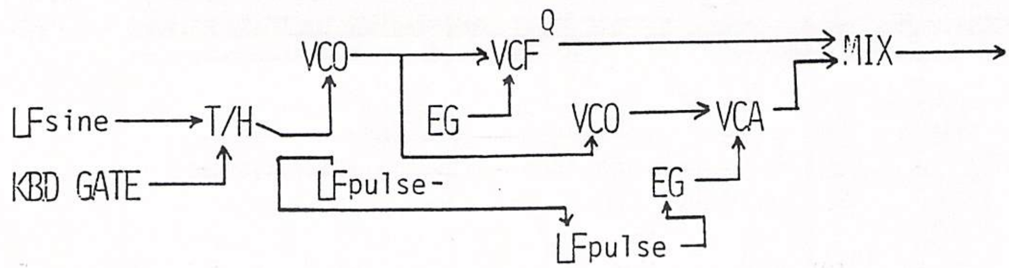


As the sawtooth voltage climbs, it gets sampled at a higher and higher voltage. The S/H output is used, then, not only to control the frequency of the audio path, but also to control the frequency of the LFO which is providing the sampling triggers. The result of the patch is that as the frequency of the audio path VCO increases, the sampling triggers occur at shorter intervals. This results in the LF sawtooth being sampled repeatedly at the same points each cycle. This patch is an example of a phase lock loop in which the relative frequencies of two signals are synched together by synching them into a constant phase relationship.

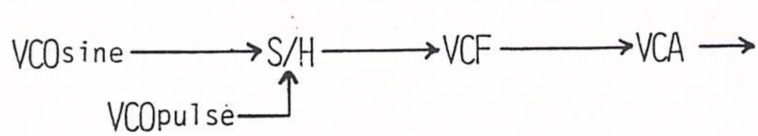
The track and hold function, like the sample and hold, can be used to control frequency, amplitude, and wave shape. In this patch it controls all three.



In order to patch the track and hold function, set the switch to "external" and patch any gate into the gate input. In block diagrams, it is represented as T/H and it is assumed that it requires a gate input. This patch has proven to have some useful applications as an "automatic percussion" patch.



Another way of using either the S/H or the T/H is as a part of the audio path. Here's the patch.



If both the VCO's are in the audio range, some interesting timbres can result. Synching the two VCO's together is one possible variation on the patch. Another variation is to sync them together and patch them into a resonant filter. Adjust the filter's Fc so that the resonant peak coincides with a strong harmonic and control both VCO's equally with the same voltage. This patch is particularly effective using the AR-327 multi-function filter in the band pass or the peak mode. Using the multi-function filter and the sample and hold (and a host of other modules) makes possible an Artoo-Detoo patch. Before diagramming the patch, we should discuss the multi-function filter.

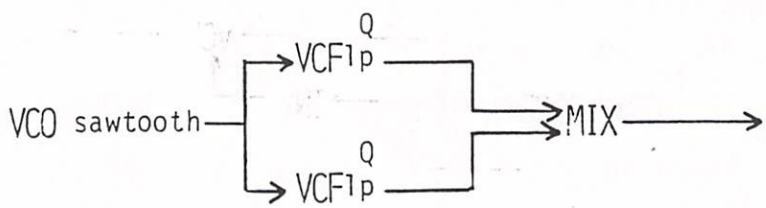
MULTI-FUNCTION FILTER

The AR-327 multi-function filter is a group of five filter functions packaged within one module. Each function is different in that each changes the input signal differently. All the functions are related in that they share the same inputs and all five filter functions are produced from the same circuit. The manual controls (frequency, resonance and attenuation) and the voltage controls (frequency and resonance) affect all the filter functions simultaneously. As the input signal is routed through the module, different portions of the circuit cause the signal to be filtered differently. The filter function outputs are taken from different places within the AR-327 circuitry and each is the result of different kinds of filtering. The five outputs are low pass, high pass, band pass, notch, and peak. With the exception of notch and peak, they all are available simultaneously. The best way to approach this module is to look at each filter function separately and explore some patches integrating that function with the rest of your modules. First, the low pass function.

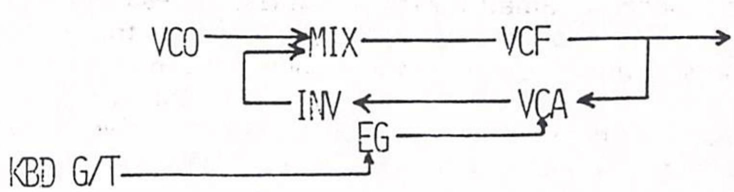
The low pass filter function is the fourth jack in the series of outputs. The low pass function is similar to the output of the AR-314 low pass filter in that it passes the frequencies below the Fc and attenuates the frequencies above the Fc at the rate of -12 dB/octave. There is, however, a significant difference. The AR-327 filter is capable of becoming more resonant than the AR-314. Not only does this produce a subtly different sound, but also it is capable of having a much steeper slope than that of the AR-314. Patching two filters together in series adds their slopes together. In this patch assume the Fc's of each of the filters are equal.

VCOsawtooth ——> VCFlp ——> VCFlp ——>

The result of the patch, then, is that you have attenuated the signals with frequencies above the Fc at the rate of -24 dB/octave. Another way to patch two filters together is in parallel. In this patch, set the Fc's of the filters an octave apart and make each filter resonant.

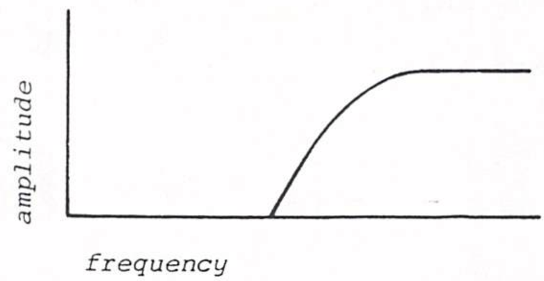


Another feature of the AR-327 is that it provides for voltage controllable resonance. Resonance, in any filter, is the result of a feed-back loop in which the output signal is routed back into the input of the filter. It's possible, using your AR-314, to set up a voltage controlled resonance patch.

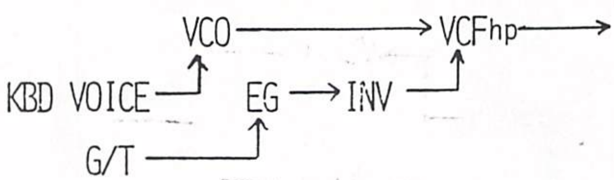


The inverter is in the patch to compensate for the phase shift caused by the filter at the cutoff frequency. The amplitude of the envelope determines how much of the output signal is fed back into the input and thus determines the amount of resonance. In th AR-327, this patch is made internally and any positive voltage patched into the resonance control inputs increases the amount of resonance.

The high pass filter function, as the name implies, passes the frequencies above the Fc and attenuates the frequencies below the Fc. The slope of this filter function is also -12 dB/octave. The important distinction to remember is that signals one octave below the Fc are attenuated by 12 dB. In the low pass function, signals an octave above the Fc are 12 dB down. Here is the graphic representation of the high pass filter function slope.

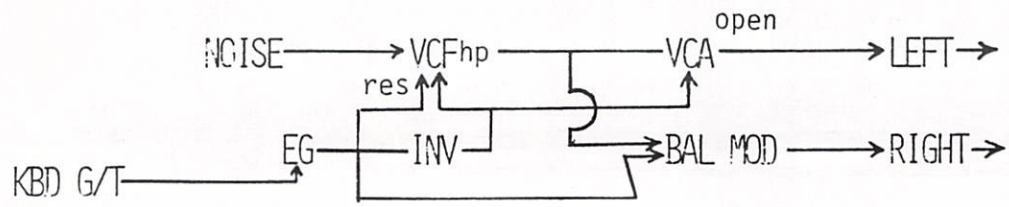


Any positive voltage applied to the filter's frequency control input will raise the cutoff frequency. When using the high pass filter function, it is usually necessary to invert the control voltage. Some interesting timbral changes are possible with this simple patch.



What makes this patch sonically interesting is that almost no acoustic event begins with the higher harmonics and then adds the lower ones. It's an unusual sound not often found in nature.

Like any other filter, the high pass filter function of the AR-327 can become resonant. Small bursts or "chiffs" of resonant high pass filtered noise were a popular sound with many of the earlier electronic composers. Among contemporary composers, both Subotnick and Berio have used this sound effectively. As with the low pass function, the resonance can be voltage controlled. Any positive voltage applied to the resonance control input increases the resonant tendency of the filter. Here's a patch using changes in Fc, resonance, and amplitude to help suggest movement between two speakers.



There are three more outputs available from the AR-327 but before discussing them, take a look at some acoustic events to which these outputs can be analogous. An acoustic sound generator, such as a violin, has inherent in its construction a series of frequencies at which it will tend to resonate. These resonant tendencies are called "formants" and they help to determine the timbral characteristics of all instruments. Each formant is a resonant peak and is, in fact, an acoustic, resonant band-pass filter.

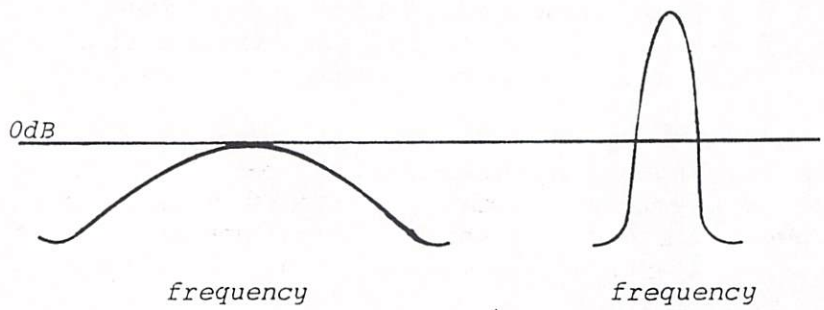
The formants are, for the most part, fixed and do not change with respect to frequency. Each different pitch produced by an acoustic instrument will have a slightly different timbre. This, among other things, is what gives an acoustic instrument its sonic interest. Take this hypothetical example.

Assume you patch a 440 Hz. sawtooth wave into the output and power module and connect the output of your Aries to a driver which is, in turn, connected to the body of a violin. Further assume that the violin body has resonant peaks at 1320 Hz., 2200 Hz., and 3520 Hz. This is, of course, an over-simplification, but since it's a hypothetical example, why not make the arithmetic simple? The violin body would become a "speaker" whose formants would accentuate the 3rd, 5th, and 8th harmonics of the sawtooth. Change the sawtooth's frequency to 660 Hz. (roughly E above middle A) and the violin would resonate with the second harmonic.

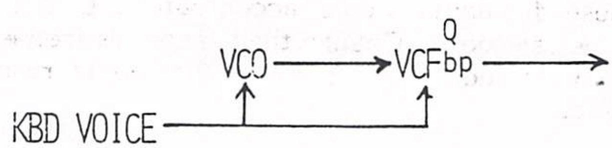
The strings on a violin, when bowed, produce an approximate sawtooth wave. The body and the air space within the body resonate with the various harmonics of the string. As the player changes the length of the vibrating string, different harmonics excite the natural resonant peaks within the instrument producing the different timbres for each different pitch. If the violin were to maintain a constant timbre throughout its range, it would have to be a one-stringed instrument which got bigger to produce a low note and smaller to produce a high note. Look at the other three outputs of the multi-function filter and you can see how they can be analogous to these acoustic phenomena.

There is some special terminology associated with band pass, notch (band reject) and peak filters. Before going any further, let's take a paragraph to talk about that.

First, the frequency control determines not the cutoff frequency, but the "center frequency." The center frequency is that point at which the least amount of attenuation takes place. A band pass filter function has two cutoff frequencies; each is a point on either side of the center frequency at which the input signal has been attenuated by 3 dB. The bandwidth is the distance from the lower to the higher cutoff frequency measured in Hz. The abbreviation, "Fc," is useful because it can refer to "cutoff frequency" with regard to high pass and low pass filters, and to "center frequency" with regard to band pass, peak, and notch filters. Like any other filter, the band pass filter function can become resonant. The resonance increases the filter's output at the center frequency, increases the filter slope and decreases the bandwidth. These two diagrams should make this apparent.



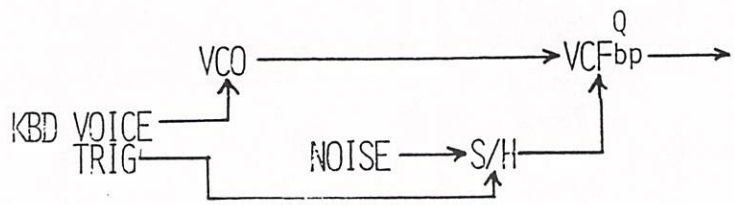
One band of a graphic equalizer is an example of a band pass filter. One way of using the band pass output is to make it resonant and tune it to one particular harmonic of the input signal. Controlling the frequency of the input signal and the Fc of the band pass filter with the same voltage assures that the same harmonic of the input signal will be resonated. Here's the patch.



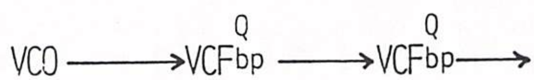
This patch is the electrical analog of the violin that changes size to produce different pitches.

Another way that the band pass filter function has been proven useful is for filtering a complex modulation timbre. Remember that a modulation timbre is the result of the sum and the difference between the input frequencies and their harmonics. Often it is difficult to obtain a modulation timbre of sufficient complexity to be sonically interesting without getting too much energy in the extreme upper and lower sidebands. Filtering a modulation timbre through a band pass filter attenuates the extreme upper and lower sidebands and can produce a more interesting timbre.

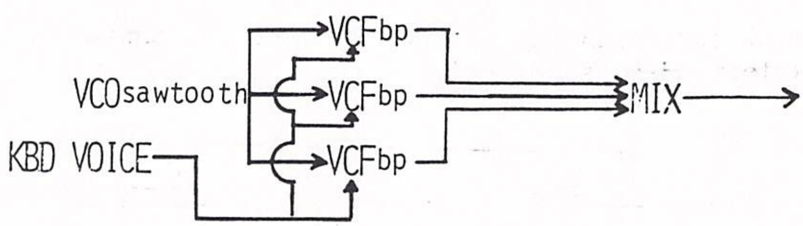
Still another way of using this filter function is to control it with one voltage while controlling the frequency of the input signal with another voltage. This will produce a different harmonic spectrum for each different input frequency. Here the resonant peak changes randomly with each different pitch.



The band pass function of the AR-327 can be used in conjunction with other AR-327's patched either in series or in parallel. In series, the slopes of the filter functions are added together producing an extremely narrow bandwidth. Making them resonant, of course, further decreases the band width. The Fc's should be tuned together.

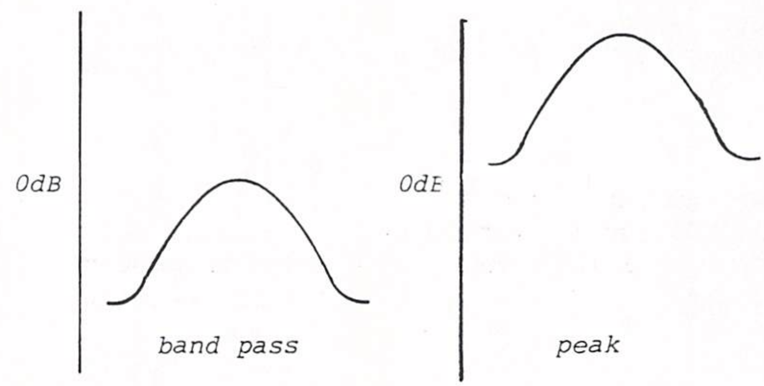


In parallel, a few AR-327's can produce some spectacular formant filtering. Tune each of the Fc's to a harmonic relationship to each other and patch the same input signal into all of them. Mixing the outputs together produces a waveform with marked resonant peaks. The filters can all be controlled by the same voltage which will change the frequency of the resonant peaks while maintaining their relationship to each other.



What you have patched is essentially a resonant, voltage controlled equalizer. It can produce timbres extremely close to those produced by acoustic instruments.

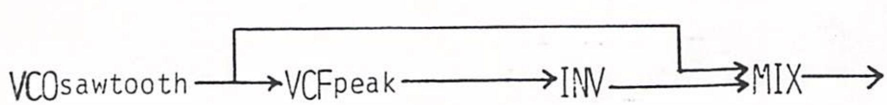
The peak filter function is similar to the band pass function. Instead on filtering the frequencies on either side of the band, the peak filter passes these frequencies relatively unchanged and boosts the frequencies near the center frequency. By increasing the resonance, you can increase the bandwidth to no wider than a fraction of a semi-tone. This diagram shows how the peak filter function differs from the band pass function.



As an experiment, patch a sawtooth wave of some low audio frequency into the AR-327 and listen to the peak output. Tune the Fc so that it is equal to the fundamental of the sawtooth and use the maximum amount of resonance. Slowly increase the Fc and count the number of the individual harmonics that you can identify. You should be able to hear at least 32 individual harmonics.

Like the band pass function, the peak filter function is useful in synthesizing instrument sounds. We don't necessarily mean duplicating the sound of a bassoon or a french horn, but rather producing a sound with enough sonic interest so that it sounds as if it might have been produced by some real or imaginary acoustic instrument. Try patching up a plucked flute, or a hammered soprano, or a bowed duck.

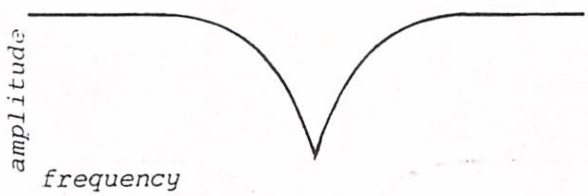
The peak filter function can also be useful in simulating a phase shifter effect. Here is the patch.



Because there is some degree of phase shift inherent in all filters, the filtered and non-filtered signals will cancel and re-inforce each other at different frequencies producing the characteristic comb filter of a phase shifter. Another way of obtaining a similar, but subtly different, effect is to use the notch output.

The notch filter function is the opposite of the band pass filter function. It passes all frequencies except those at or near the center frequency. It is a peak filter which is inverted, not with respect to 0 volts, but with respect to 0 dB.

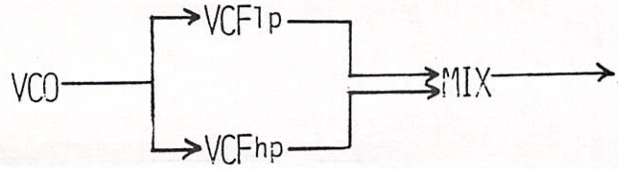
The ubiquitous diagram....



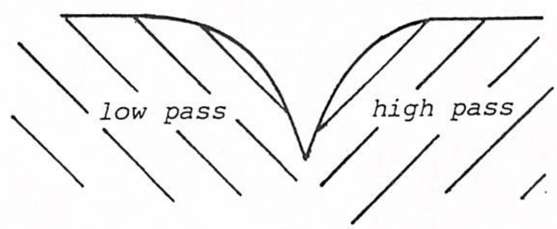
The resonance control, as you have probably surmised, will decrease the bandwidth of the notch and steepen the slope. The result is not as sonically obvious as with the peak and band pass filters because it is easier to hear one particular harmonic that has been accentuated than it is to hear one particular harmonic that has been attenuated. Still, the notch filter function is an important timbral modification device which offers some very subtle effects.

Often, the way to approach filtering a signal is not to make blatant changes in the timbre, but to make small, subtle changes. These changes may not even be consciously heard changes but rather nuances which are subliminally perceived. This is the way we are accustomed to experiencing acoustic events. Sounds are not steady state phenomena; they change with time. To imitate nature, then, make small and continual changes in the signals. These nuances provide the sonic interest that our ears are accustomed to hearing. We become quickly bored with unchanging, electrically perfect sounds.

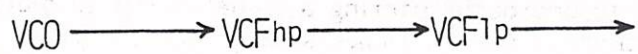
Some synthesizer manufacturers provide only high and low pass filters. With these two filters and some additional modules, it is possible to patch a band pass and a notch function. First, the notch filter patch.



Set the Fc of your low pass filter to about 300 Hz. This will, of course, pass frequencies below 300 Hz. and attenuate frequencies above 300 Hz. Next, set the Fc on the high pass filter function to about 800 Hz., passing the frequencies above that point. When the same signal is patched into these two filters in parallel and the outputs of the two filters mixed together, it is as if you have superimposed the two filter slopes. Here is the result.

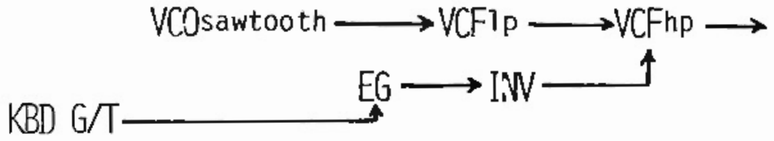


The patch for a band pass filter is this.

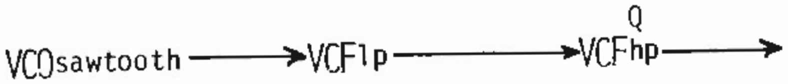


Set the Fc of the low pass filter to about 800 Hz. and the high pass Fc to about 300 Hz. Frequencies from 0 Hz. to 800 Hz. are passed by the low pass while frequencies of from 300 Hz. to 0 Hz. are filtered by the high pass. The result is a band pass filter with a bandwidth of from 300 Hz. to 800 Hz.

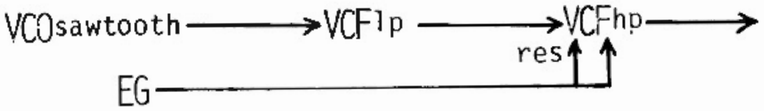
Of course, making these functions through patching is more complex than simply using the associated output of the AR-327, but it offers some advantages. First, it is possible to move just one of the cutoff frequencies. This produces a band or notch whose bandwidth increases in only one direction. An example of that patch is this.



Second, it is possible to make a band pass or notch with different filter slopes on either side of the center frequency. That patch can be accomplished by having one of the filters resonant.



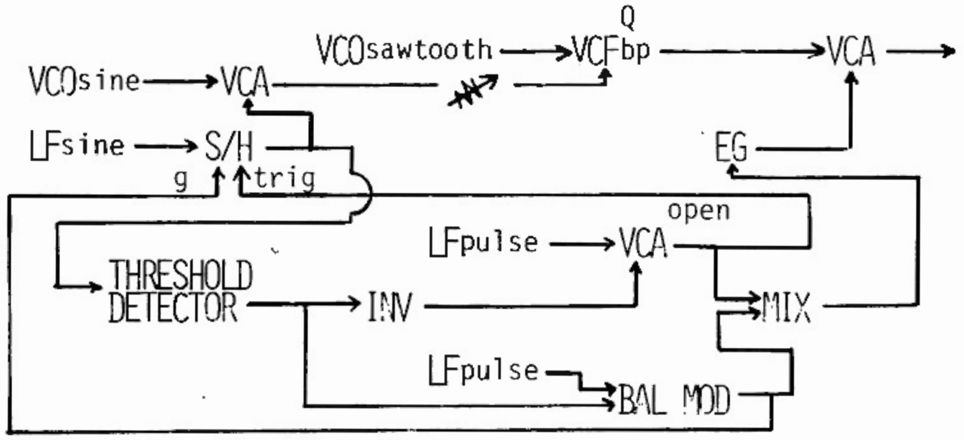
Third, you can voltage control the slope of the filter on the AR-327 by voltage controlling the resonance. In this patch, as the bandwidth increases, the upper filter slope becomes steeper.



The possibilities are legion. Having two or more filters allows you to control the Fc's simultaneously, oppositely, with different voltages, with the same voltage, with various resonances, with different audio inputs, with different filter slopes.....
and the list goes on. After having read all the information about the multi-function filter, try this one experiment to help all the information to fall into place.

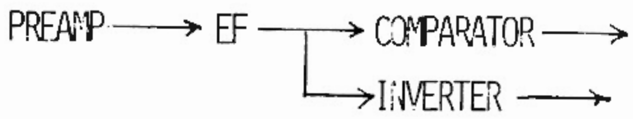
Patch white noise into the AR-327 and listen successively to all the outputs. Try various Fc's and various amounts of resonance. Listen carefully and learn to identify the sonic character resulting from the different kinds of filtering. This one particular module is capable of such a variety of timbral modification that it is only by understanding how the module works and by accurately hearing and remembering the sonic result, that you can patch up, a week from now, the sound you have just discovered.

Finally, the Artoo-Detoo patch.



PREAMPLIFIER

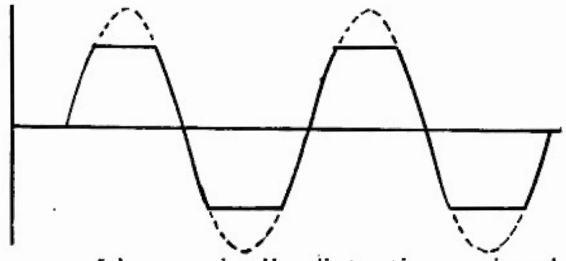
The AR-331 module is a four function module containing a pre-amp, an envelope follower, a comparator and an inverter. Internally, the output of the pre-amp is normalled to the input of the envelope follower and the output of the envelope follower is normalled both to the inputs of the comparator and the inverter. Inserting a patch cord or a mini-plug into the inputs of the envelope follower, the comparator, or the inverter, breaks the normalled connection. Patching a patch cord from the output of any module into these inputs substitutes that signal for the normalled signal. A block diagram of the normalled patch within the AR-331 looks like this.



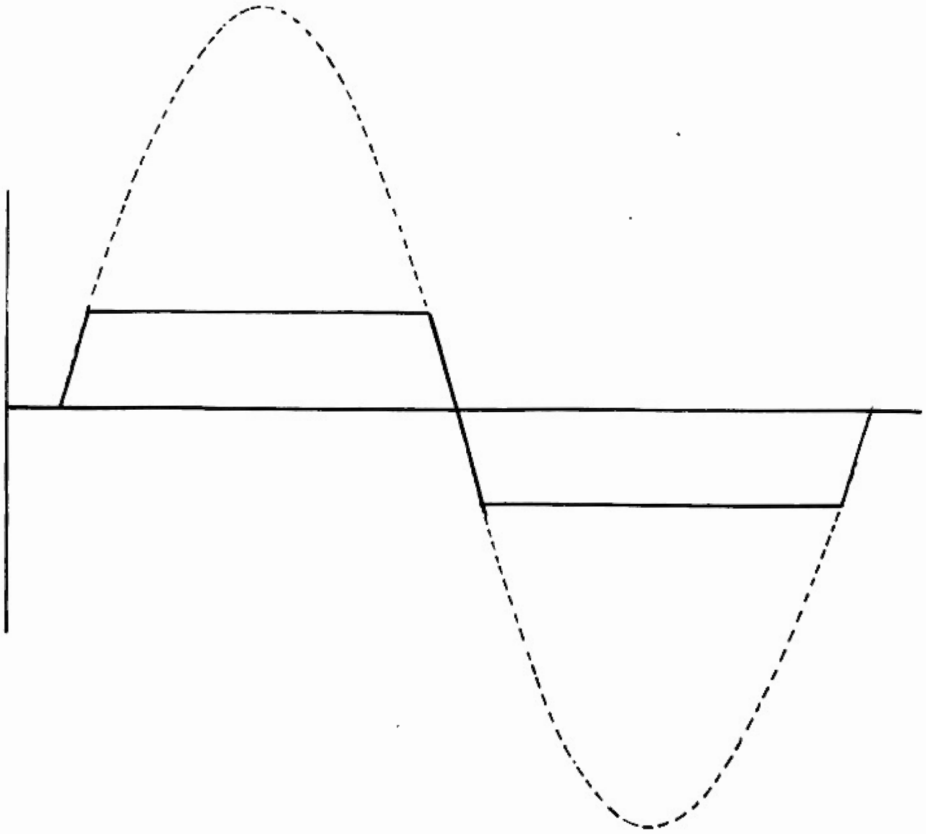
This module was designed to allow you to interface any signal generated by an electric instrument with your Aries synthesizer. Of course, there are other uses.

The pre-amp is a high gain amplifier with a gain factor of 100. Signals generated by electric guitars, electric pianos, tape recorders, microphones, etc. are measured in terms of milli-volts and must first be amplified to be used with a synthesizer. The first step in patching an externally generated signal into your Aries is to patch the signal into the pre-amp. The pre-amp's input attenuator determines the peak to peak amplitude of the output signal.

Another use for the pre-amp is as a wave shaping device. No signal within the Aries system can exceed the plus and minus 15 volts of the power supply. A hypothetical sine wave of plus and minus 20 volts, for example, becomes clipped at the limits of the system. It would look like this.

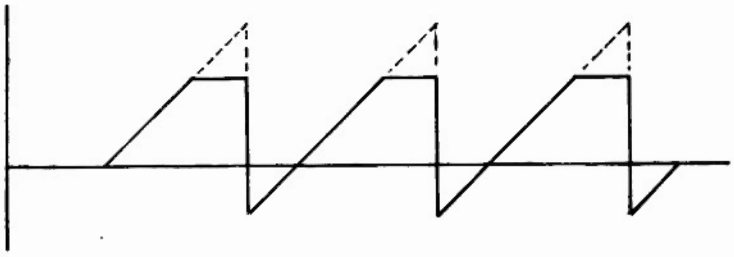


Clipping is a way of harmonically distorting a signal. It increases the energy in the harmonics. If you amplify the sine by the gain factor of 100, the pre-amp's output would begin to look like a square wave.

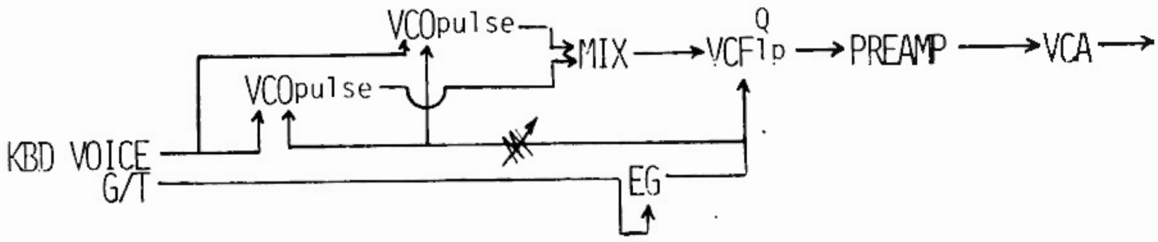


Another way to harmonically distort a signal is to clip only the top or the bottom portion of the waveform. This can be accomplished by adding either a positive or a negative bias to the signal and then pre-amping it. In this patch, only the top portion of the sawtooth is being clipped.

Resulting in this waveform.

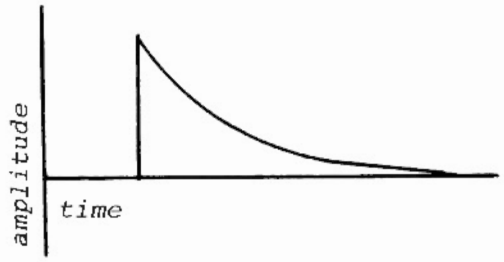


One particularly effective use of the pre-amp as a waveshaping device is in synthesizing an electric guitar sound.

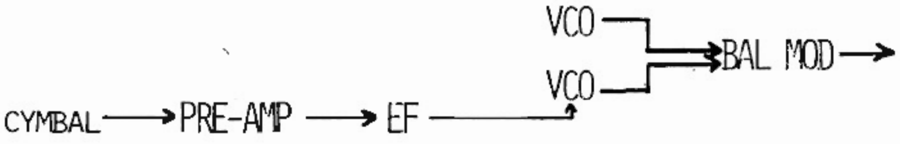


Although the output of the pre-amp is normalled into the envelope follower's input, inserting a patch cord into the pre-amp's output does not break this connection. This connection is only broken by inserting a patch cord into the envelope follower's input.

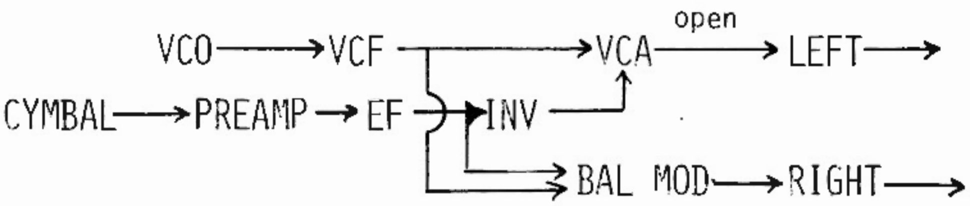
An envelope follower is a module which converts the average amplitude of an AC audio signal into a DC control voltage. In other words, it detects any significant change in the amplitude envelope of the audio input signal and reproduces that change as a DC voltage at its output. It outputs a control voltage that is proportional to the average input signal amplitude. If you were to mic a cymbal, patch the mic into the pre-amp, the pre-amp into the envelope follower and strike the cymbal once, the envelope follower's output would be a voltage representing the cymbal's amplitude envelope. It would look like this.



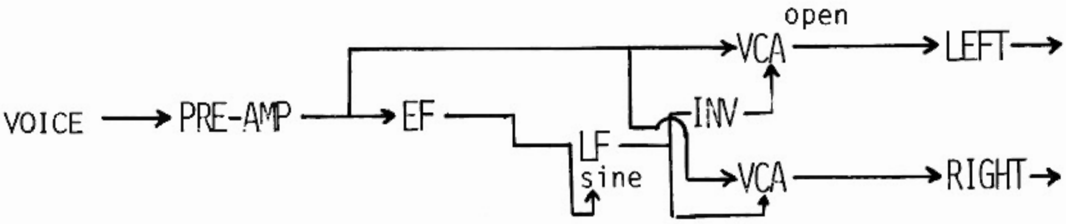
This voltage, then, could be used to control the frequency, amplitude, or wave shape of another signal. For example, here's a patch in which the harmonic spectrum of a modulation timbre changes relative to the amplitude of the cymbal.



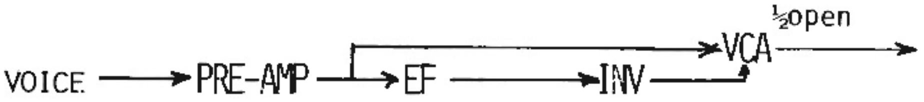
In this patch, the envelope follower's output is used to determine spacial location. The audio path appears first in the left channel then, when the cymbal is struck, jumps to the right channel and slowly pans back to the left channel as the cymbal decays.



It is also possible to control the amplitude or the wave shape of the external signal itself. In this patch, a voice is the external signal and it is normally panning between two channels. The amplitude of the external signal determines the panning rate.

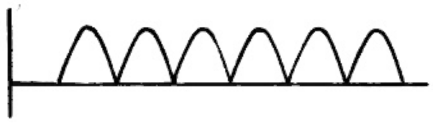


Using the pre-amp, the envelope follower, a VCA and an inverter, you can patch up a compressor.



This patch offers a good chance to experiment with the linear and exponential switch on the VCA. Switch it back and forth and see which gives the best compression effect. You will find it necessary to adjust the input attenuator to produce the desired compression ratio. As the amplitude of the input signal increases, it results in a greater signal amplitude being fed into the audio input of the VCA. Simultaneously, the amplitude of the voice is detected by the envelope follower. The envelope follower puts out an increasingly greater amplitude which is inverted and used to close down the VCA. Each increase in the VCA audio input signal is met with an equal, but inversely proportional change in the VCA's control voltage. This maintains the VCA output at equal to or less than the output with the original initial gain setting.

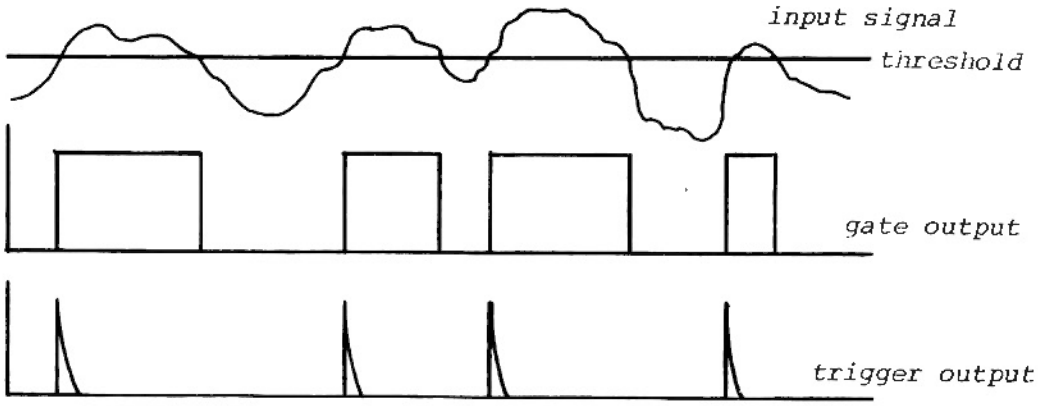
Like the pre-amp, the envelope follower can be used as a wave shaping device. It produces a "full wave rectified" output which means that all the negative portions of the input signal have been inverted. A balanced sine wave patched into the envelope follower comes out looking like this.



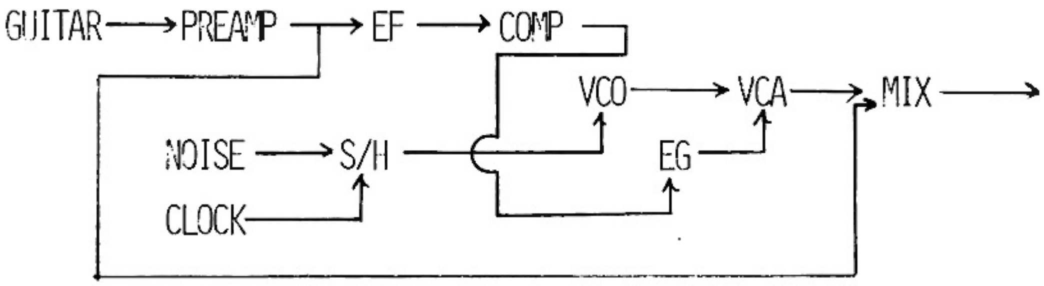
As you can see, it is no longer a sine wave, no longer balanced across 0 volts and is an octave higher than the sine wave it was made from. Full wave rectifying a balanced waveform always adds harmonic distortion and increases the frequency. It can also provide some interesting timbres.

Another function within this module is the comparator or threshold detector. This device has two inputs and one output. The first input is a DC bias that serves as a threshold level; this bias is hard-wired internally. The threshold level is adjustable by the threshold level control on the front panel. The second input is labeled "signal" and the output is a gate. The way it works is this.

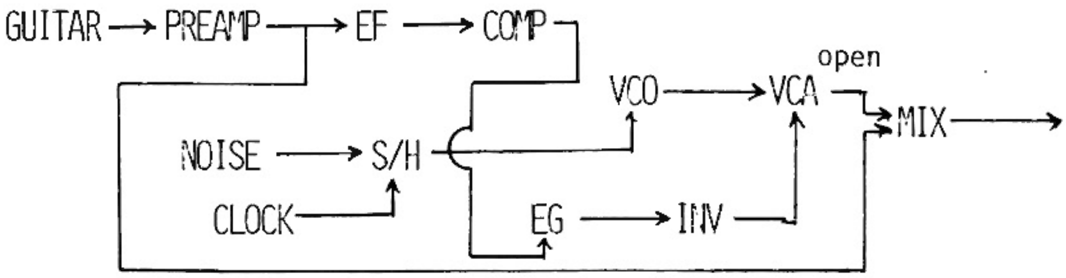
Whenever the instantaneous amplitude of the signal at the signal input exceeds the threshold level, a gate is produced. At the rising edge of each gate, a trigger is also produced and is available at the trigger output. When the signal's amplitude falls below the threshold level, the gate falls to 0 volts. When you see it graphically it becomes obvious.



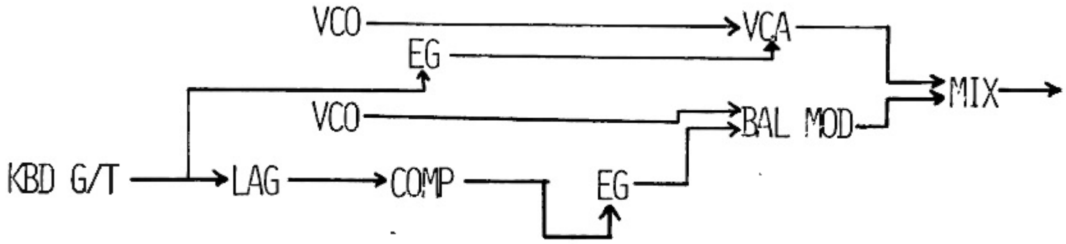
The comparator is useful in generating timing signals which are dependent upon another signal's amplitude. For example, in this patch whenever a loud note is played on the guitar, a gate is produced firing the envelope generator which opens the VCA allowing the series of random frequencies to pass.



Here the opposite happens. The series of random pitches is heard except when the guitar plays a loud note.



Another way the comparator may be used is to generate a delayed gate. In this patch, assume that EG #1 has a very slow attack time and that the comparator's threshold is set to 8 volts. When the envelope generator's output exceeds 8 volts the comparator outputs a gate which fires the second envelope generator. The attack time of EG #1, then, determines the length of the gate delay. If you want a percussive envelope with a quick attack but still followed by a delayed envelope, you can patch this up by using the lag. The patch works like this.



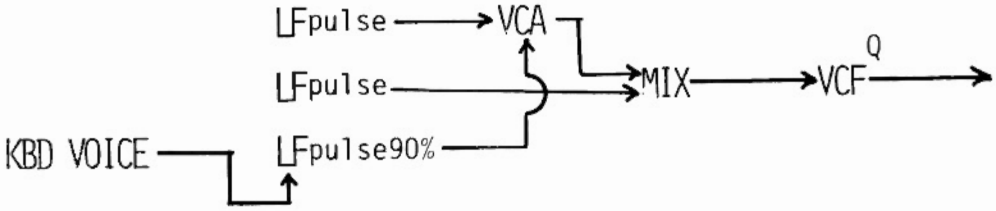
By using two percussive envelopes, you can create a single, stereo, "slap-back" echo. Be sure to hold the key down until both envelopes have fired.

You know, of course, what an inverter does. It is sufficient to mention that there's one included within this module.

DUAL VCO

The AR-332 dual VCO module contains two identical sawtooth and pulse oscillators. Except for the additional waveforms available from the AR-317 VCO, the AR-332 and the AR-317 are nearly identical. There are two differences. First, the controls are arranged concentrically so that the outside knob controls VCO A and the inside knob controls VCO B. Like the AR-317, the AR-332 has both coarse and fine tuning controls, a pulse width control, and an input attenuator on the first frequency control input. The other difference is that there is a normalled sync connection from an internal trigger output of VCO A into the sync input of VCO B. The sync connection can be broken by inserting a mini-plug into the sync input of VCO B. Of course, either VCO can be synched to any other oscillator by patching the proper connection.

The pulse width sensitivity of the AR-332 is 10% pulse change per control volt. Patching a positive control voltage into the PWM control can drive the pulse to 100%. This provides a 10 volt DC bias signal and provides a way of "turning the oscillator off and on" with a voltage. By patching sufficient voltage into the PWM control input so that the 100% pulse results, you have "turned the oscillator off" in its high state. Using a negative voltage to drive it down to a 0% pulse has "turned it off" in its low state. These patches can be particularly useful in providing some manual control over "automated rhythm" patches like this one.



Tthere are two additional modules, excluding cases and power supplies, in the current Aries 300 series. Both of these, the phase shifter/flanger and the reverberation unit, are fairly dedicated modules. They are used mainly for one purpose--to change the wave shape of an audio signal. Almost everyone has heard the effects of these electronic devices. However a detailed explanation of what they do is more appropriately found in a textbook on acoustics than a manual about a synthesizer. We'll not discuss here, in detail, what these modules do. Rather we'll suggest some of the ways to use them. First, the....

REVERB

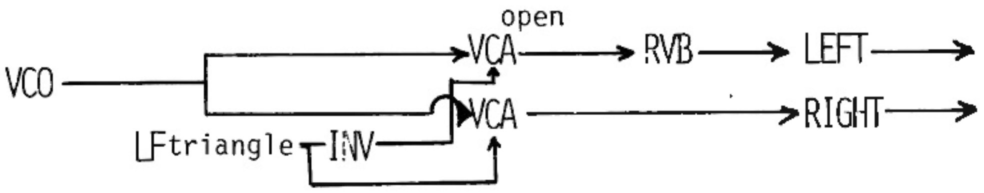
The AR-328 is a spring-type, two channel reverberation unit. It is an electro-acoustical device that can be used to simulate the natural reverberation of an acoustic space. It contains two independent sets of springs so that it can produce a discreet, two channel output. In addition, this module contains an output and power function so that it may be substituted for the output and power module. It is designated in a block diagram as "RVB."

The controls on the AR-328 are fairly straightforward. Each input has its own input attenuator, just as in the AR-326 output and power module. Each output has a rotary pan pot which can be used to manually pan each of the output signals independently or to place the output anywhere within the stereo field. These controls affect only the output and power function of the module.

The reverb function also has two sets of controls. There is a concentric knob which controls the amount of reverberation for each channel. Because there are two sets of springs, it is possible to have a different amount of reverberation for each output channel. The other control, the source control, works like this.

With the knob turned all the way to the left, all the signal from input #1 is sent to the reverb springs; none of the signal from input #2 is reverberated. With the control turned all the way to the right, all of the signal from input #2 is reverberated while the signal from input #1 will pass through the module unchanged. With the control in the center, half of the signals from inputs #1 and #2 will be reverberated while half will pass through unchanged. The source control, then, determines how much of each input signal will be routed to the reverb springs and the level control determines how much it will be reverberated.

Some interesting effects are possible by placing the signal in one speaker and the reverberation in the other. By using the reverb in your panning or spacial location patches, you can increase the possibilities of apparent sonic motion. This patch creates an apparent change in depth as well as in stereo placement.



The best way to musically use an effect like reverberation is very judiciously. The same holds true for the.....

PHASE-SHIFTER/FLANGER

Phase shifting and flanging are two different but similar effects. Both are commonly used in recording studios and you can probably recognize the characteristic "churning" or "whooshing" sound. First, we'll discuss the phase shifter.

A phase shifter is an allpass filter. Like other filters you can consider its output in both the time domain and the frequency domain. We'll consider, first, the operation of the module and then its output in both domains.

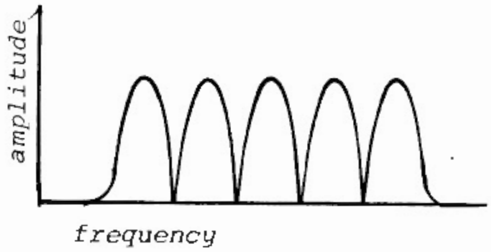
There are four audio inputs. A signal patched into any of these inputs becomes split. Part of the signal is routed to the phase shifting circuitry and part of it is not. Internally, there is a mixer near the output stage of the module that adds the phase shifted signal with the non-shifted signal. There is a back plane connector at which only the phase shifted signal is available. To utilize this, however, would require you to make a very simple internal modification and add an output jack to your synthesizer.

There are four control knobs. If you keep in mind that the phase shifter is a filter, the function of the control knobs becomes clear. The "frequency" control controls the cutoff point of the filter; the "resonance" controls the "Q." The "audio and control" knobs are input attenuators on the first audio and control inputs respectively. The bypass switch controls the inputs to the final mixer. In the "bypass mode" the signal at the audio input is routed directly to the output and it is as if you have unpatched the phase shifter from the audio path. In the "mix" position, the output is a mixture of the phase shifted signal and the non-shifted signal and the characteristic phase shifting sound results. For the moment, we'll defer the explanation of the "odd/even" switch.

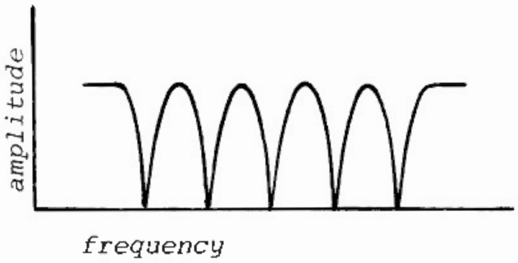
The exponential inputs are the usual control inputs with exponential converter which are found on the AR-314 and AR-327 filters. Their sensitivity is one volt per octave. The flange input allows the module to be used as an electronic flanger. The main and aux outputs are audio outputs that work in conjunction with the "odd/even" switch and they will become clear in a few paragraphs.

By definition, one stage of a phase shift circuit changes the phase of an input signal by 180 degrees. The AR-329 is a ten stage phase shifter; it changes the phase of the input signal by 1800 degrees. Since there are 360 degrees in one complete cycle, the AR-329 shifts the phase of the input signal by 5 complete cycles.

When the shifted signal is mixed with the original signal, a series of cancellations and re-inforcements occur. In the frequency domain, these re-inforcements and cancellations result in a filter slope which is characteristically called a "comb filter." With a 10 stage phase shifter, there are simultaneously 5 peaks in the slope and each peak is approximately 2 1/2 octaves apart. With the odd/even switch in the even position, here is what the filter slope taken from the main output of the phase shifter looks like.



As you can see, there are 5 distinct peaks and 4 notches. Flipping the switch to the odd position, produces this comb filter slope which contains 4 peaks and 5 notches.



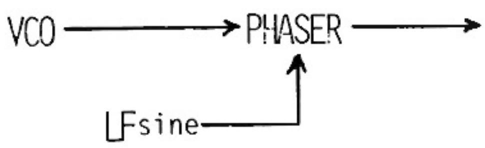
In almost all situations, the phaser produces a more "full" sound when the switch is in the even position rather than when it's in the odd position. Without getting into the math involved, here is what the switch does. At the input of the circuit, there is a phase splitting network. The outputs of the phase splitting network are coupled with the rest of the phase shifting circuit. The odd/even switch determines how they are coupled.

With the switch in the even position, the "even slope" is available from the main output; the "odd slope" is simultaneously available from the aux output. In the odd position, the opposite is true. We suggest that you treat this switch empirically. Switch it back and forth and compare the sonic results.

Controlling the Fc moves these peaks back and forth along the horizontal axis and as it sweeps, it alternately passes and attenuates the various harmonics of the input signal. Increasing the resonance makes the resonant peaks taller and decreases the bandwidth of each peak.

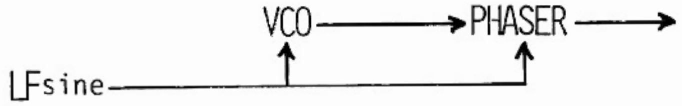
All these modules produce signals which are analogous to acoustic events. One way of producing acoustic phase shifting is to play an organ simultaneously through a leslie speaker and a stationary speaker. As each speaker in the leslie cabinet moves toward the listener, it causes a slight Doppler shift resulting in an apparent rise, then fall in the pitch. This, when mixed with the sound waves from the stationary speaker, will cause alternate cancellations and re-inforcements. The phasing "rate" is the speed at which the leslie is revolving times the number of speakers in the cabinet. The phasing "depth" is dependent upon how well the two sounds "mix" in the air.

There are so many patches involving the phase shifter that it is difficult to choose which ones to diagram. Here is a representative sampling of some of the areas in which we have found the AR-329 to be useful. To begin with, here is the typical phase shifter patch.

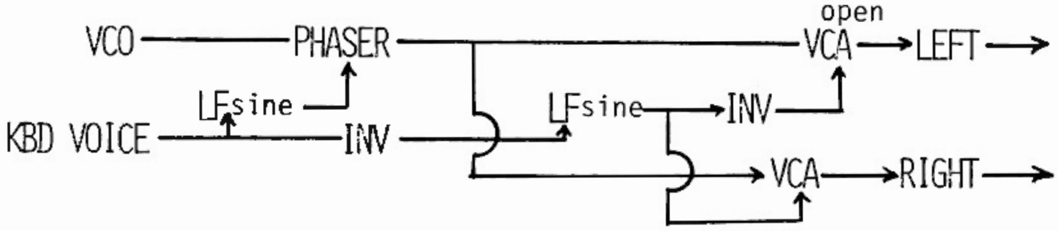


If you use the main output into one speaker and the aux output into another speaker and control the Fc with a slow sine wave or triangle, you can perceive apparent sonic motion between the two speakers.

A particularly good effect is achieved by controlling both the phasing rate and the frequency of the input signal from the same voltage source.

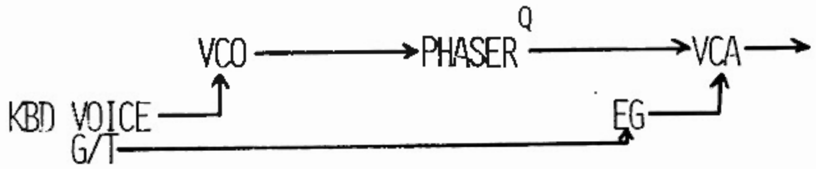


In this patch, the phasing rate is inversely proportional to the panning rate.



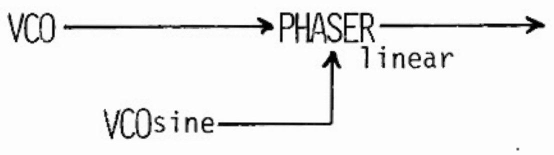
Or, you can use the phase shifter without sweeping the Fc. This will achieve some of those subtle timbral modifications we've previously mentioned. If you leave the Fc stationary and control the frequency of the input signal, some interesting formant structures will result. Try increasing the resonance to decrease the bandwidth of the resonant peaks.

You can also try controlling the input signal and the phaser's Fc from the same voltage source. Attenuating only the voltage that controls the Fc results in a slightly different formant structure for each different frequency.

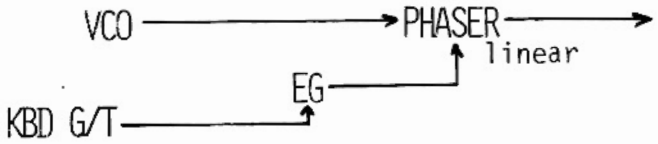


In a patch like this, you're completely on your own. Timbre is such a subjective factor that there's no way we can tell you what settings are effective or what will work in a particular situation.

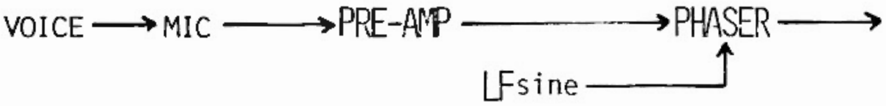
With a control voltage patched into the linear input, the phaser operates in exactly the same way. The difference is that the control voltage is not exponentially converted before actually going to control the phaser's Fc. Instead of the control sensitivity being measured in volts per octave, it is measured in volts per Hertz. A balanced waveform patched into the linear control input will drive the Fc down twice as far as it drives it up. Again, this produces a subtle effect and one which you should sonically explore on your own. We have found, however, that modulating the Fc with an audio frequency sine wave patched into the linear control input produces some interesting timbres. Experiment with the relative frequencies of the audio and control signals.



Another interesting effect is achieved by using an envelope voltage into the linear input.



Of course, you can shift the phase of an external signal.



Flanging and phasing are very closely related. However, there is a subtle difference. A true flanger is a time delay device which delays one waveform in time and then mixes it with the original waveform. Like phasing, flanging produces a series of peaks and notches in the signal in the frequency domain. Unlike phasing, flanging is frequency dependent.

The sonic difference between phasing and flanging is that the flanger appears to sweep through the higher harmonics faster than the phaser. Set up a patch in which a low frequency sine or triangle is patched into the unattenuated input to the phaser. Switch the control voltage to the flanger input and you can hear the difference. As the Fc becomes increasingly higher than the fundamental of the input signal, the reinforcements and cancellations occur more rapidly with the flanger than with the phaser. Again, you may use the flanger to create subtle and interesting timbral effects by not controlling the Fc. Review the patches in which you used the phaser and replace it with the flanger. You should be able to hear a sonic difference is each case.

This concludes the main section of the Aries 300 System Manual. In the near future we will be making new modules available and as we do, we will provide supplements to this manual explaining these new modules. We will also include patches integrating these modules with your other Aries 300 modules.

After you finish reading this manual and the following appendices and glossary, we suggest you read it again. Learning about your synthesizer is a circular process in which each new fact you learn allows you to see, in a new light, something you thought you already understood. Practicing a synthesizer is different from practicing any other instrument. It not only requires that you achieve some minimal technique with your KBD, but also requires that you learn to play with two hands: at least one hand should always be on the control panel making small changes in the patch as you play. Practicing the synthesizer is also a matter of simply thinking about it. Conceive of patches while you're away from your synthesizer and then ask yourself what interesting changes can be made in them. Consider what changing the setting of this or that knob would do to the resulting sound. Become more sonically aware of your environment. Learning to hear a sound is the first step in learning to synthesize it.

We strongly urge that you continue learning about your instrument. Just reading (and re-reading) this manual has not made you an expert. Most public libraries have a few books about electronic music. Read them! At the end of this section is a list of recommended additional reading. Also, you can find in the library some of the music your instrument and its precursors have spawned. Listen to it and see how your instrument has changed contemporary music.

You may have to throw out some pre-conceptions about what music is and what it should sound like. Synthesizers are too new to have a "music theory" or an established sonic code built up around them. The only criterion specifying what sound to use is your own good taste; the only way to patch the sound you imagine is to acquire impeccable synthesizer technique. You own an instrument capable of producing incredible sonic diversity. Learn to make it produce any sound you can imagine, then increase the scope of your sonic imagination.

APPENDICES

The following appendices and glossary are neither complete nor comprehensive. They are included here as an introduction to some concepts and as a more detailed explanation of some others. Their purpose is to clarify and expand the information in the text, but to include them within the body of the text would have been inappropriate.

APPENDIX I

What to do if there's no sound

The problem could be in your monitor system.

Are the speakers connected to the amplifier?
Is the amplifier turned on? plugged in?
Is the pre-amp plugged in? turned on?
Is the pre-amp plugged into the amplifier?
Is the Aries plugged into the proper input?
Is the pre-amp switch selector set to the proper position?
Is the volume control turned up?
Are all the connecting cables working?
Are you sure the monitor system is working?

Or the problem could be in your Aries.

Is the synthesizer plugged in? turned on?
Are the connecting cables from the synthesizer to the monitor system working?
Are all your patch cords working?
Have you set up the patch according to instructions?
Have you re-read the instructions to make sure?

If you answer "yes" to all the above questions, then call this number (617) 744-2400 and ask for Jim. You can make arrangements with him to get your synthesizer fixed.

APPENDIX 2

dB de-mystified

Before explaining decibels (dB), there's a bit of math to deal with, namely, logarithms. A logarithm is an exponent of 10 and the log of any number is the exponent, or the power to which 10 must be raised, that will equal that number. For example, the log of 100 is 2 because 10 to the second power is equal to 100. Likewise, the log of 1000 is three because 10 to the third power is 1000. Of course not all logs are this easy to figure out and it is best to consult a table of logarithms or a pocket calculator to determine the logs of less obvious numbers.

A decibel is a unit of measurement and, like all such units, it is a comparison. This page, for example, is so many inches wide and one inch is the distance between two etched marks on a platinum bar kept by the Bureau of Weights and Measures. To measure distance, then, we compare a given length against this standard length. Part of the confusion surrounding decibels is that there is no one standard reference level.

Often in tape recorder signal to noise ratios (S/N) you will see a figure like 65 dB. This means that whatever the level of the program signal, the level of the noise is 65 dB less. The noise is -65 dB relative to the signal. Another way of stating this is to say that the program signal is +65 dB relative to the noise. These statements say the same thing.

Another reason that dB seems confusing is that it is used to measure different things. Decibels are used to measure power levels, voltage levels, sound pressure levels, and sound levels. Take a look at a power level application.

Suppose you have a 100 watt amplifier and you want to know how much more powerful is a 200 watt amplifier. The formula for dB as a power level measurement is:

$$dB = 10 \log A/B$$

where A and B are the power levels.

So, 200/100 = 2. The log of 2 is 0.3 and 10 x 0.3 = 3. Therefore, doubling the power results in a 3 dB increase. Halving the power results in a 3 dB decrease.

If you want to measure voltage in terms of dB, there is a slightly different formula:

$$dB = 20 \log A/B$$

where A and B are the voltage levels.

The fundamental of a sawtooth wave is 10 volts; the second harmonic is 5 volts. Plug these values into the formula to determine the relative amplitudes of the first and second harmonics of a sawtooth wave measured in dB.

So, 10/5 = 2. The log of 2 is 0.3 and 20 x 0.3 = 6. Whenever you double the volts, you increase the dB by 6.

These two tables show the relative amplitudes of the first 16 harmonics of a sawtooth. Table 1 shows their amplitude in volts and table 2 their amplitude in dB relative to the fundamental amplitude.

	TABLE 1	TABLE 2
F	10.00 v.	0.0 dB
2F	5.00 v.	-6.0 dB
3F	3.33 v.	-9.6 dB
4F	2.25 v.	-13.0 dB
5F	2.00 v.	-14.0 dB
6F	1.67 v.	-15.6 dB
7F	1.43 v.	-17.9 dB
8F	1.25 v.	-18.0 dB
9F	1.11 v.	-19.1 dB
10F	1.00 v.	-20.0 dB
11F	0.91 v.	-20.8 dB
12F	0.83 v.	-21.6 dB
13F	0.77 v.	-22.3 dB
14F	0.71 v.	-23.0 dB
15F	0.67 v.	-23.5 dB
16F	0.63 v.	-24.0 dB

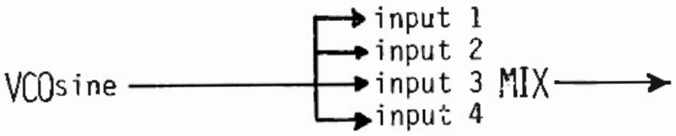
The important thing to remember is that dB is always a comparison; it compares one level with reference to another level.

APPENDIX 3

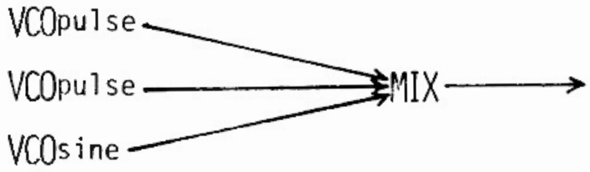
DC Overload

You've no doubt noticed that all the waveforms on the Aries oscillator are balanced across 0 volts with the exception of the pulse wave. We have purposely provided an unbalanced pulse so that, at low frequencies, it can serve as a typical 10 volt gate. Although having an unbalanced pulse makes the system easier to use, it results in one problem which you should be aware of.

The power supply supplies all the Aries modules with a positive and negative 15 volts. Any signal which exceeds either or both of these limits will become distorted. In this patch,

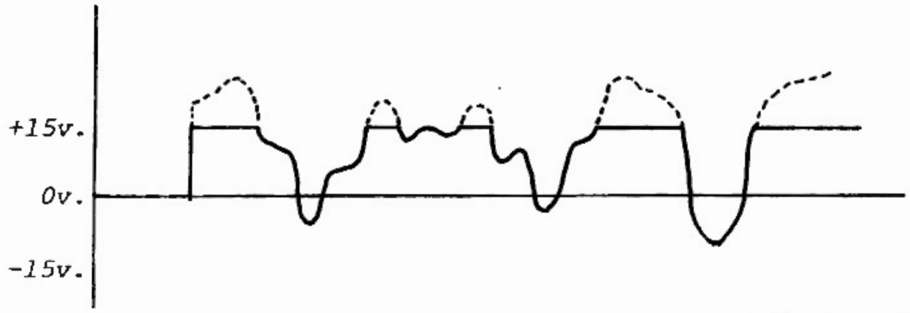


the output from the mixer is no longer a sine wave, but a clipped sine wave. Overload cannot harm your system and, in fact, you can intentionally clip signals to achieve harmonic distortion. Other than this, though the above patch has little use. A patch which you might often use, however, is this one.



And this is where the problem occurs.

Because the pulse is unbalanced the resulting waveform output from the mixer is also unbalanced. Not only does the mixer add the instantaneous voltages of the three waveforms together, it also adds the DC bias of each waveform together. If all three waveforms were at their highest amplitude level simultaneously, the resultant waveform would have a maximun isntantaneous amplitude of +25 volts. The mixer, or any other module, clips the signal at plus or minus 15 volts. The waveforms would look like this.



The way to avoid clipping, in this case, is to add a -10 DC bias. This again balances the resultant waveform across 0 volts. Keep in mind that when you are using a pulse wave as an audio signal that it is an unbalanced waveform and it might be necessary to use a negative DC with it to avoid clipping.

There is another way to solve this problem and we chose this way for two of the modules. It is a design consideration called "input coupling" and it's what the next appendix is about.

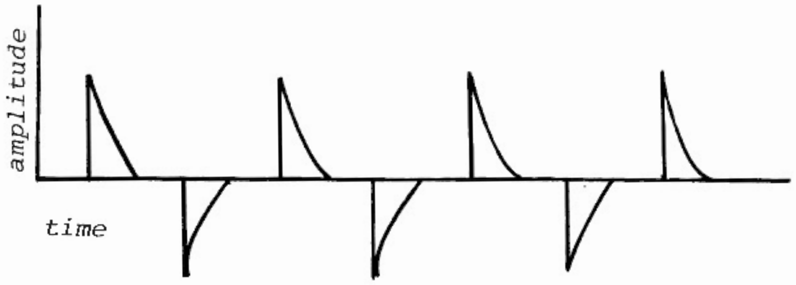
APPENDIX 4

Input Coupling

If the Aries were designed differently you would not have this problem of DC overload. At the audio input of each module we could have added a capacitor, an electronic component which allows no DC voltage to pass. An unbalanced waveform patched into a module with a coupling capacitor at its input becomes a balanced waveform. The DC bias component is eliminated.

If an audio frequency pulse from a VCO were patched into this capacitively coupled circuit, the only significant change would be that the pulse would become a balanced waveform. If a low frequency pulse were patched into this same circuit, not only would it become balanced, but also its wave shape would become changed.

A capacitor reacts to the rate of voltage change. When the low frequency pulse is patched into the capacitively coupled input, the circuit reacts to the instantaneous rates of change of the pulse. The time it takes to react is dependent upon three factors. First, whether or not the circuit output voltage reaches the high level of the pulse. Second, if it does, how long it takes it to do so. Third, how long it takes the circuit output voltage to decay. These three factors determine the circuit's "time constant." If the time constant is small, the circuit responds rapidly to the pulse and cancels the DC level to 0 volts quickly. A pulse input into a capacitively coupled circuit with a small time constant becomes "differentiated" and looks like this.



As you can see, capacitively coupling the audio inputs to the modules severely limits the flexibility of the system. It would be impossible to voltage control the amplitude of a LF pulse in the VCA without also changing the pulse's wave shape. Also, you would not be able to modulate an envelope's voltage with a LF waveform. You would need separate mixers for audio and control signals. Any patches in which you voltage control the amplitude of a low frequency DC signal, filter the KBD voice output, or use the BM as a VCA would similarly become impossible.

For these reasons we have decided to "direct couple" all the inputs except those on the phase shifter, the reverb, and the preamp. Dealing with DC overload is much easier than dealing with an inflexible synthesizer.

APPENDIX 5

Relative output of the VCA in linear and exponential modes

LINEAR					EXPONENTIAL				
V.in	V.c	G.F.	V.out	dB	V.in	V.c	G.F.	V.out	dB
10	10	1.0	10	0	10	10	1	10	0
10	9	0.9	9	-.9	10	9			-10
10	8	0.8	8	-1.9	10	8	0.1	1	-20
10	7	0.7	7	-3.1	10	7			-30
10	6	0.6	6	-4.4	10	6	0.01	0.1	-40
10	5	0.5	5	-6.0	10	5			-50
10	4	0.4	4	-8.0	10	4	0.001	0.01	-60
10	3	0.3	3	-10.5	10	3			-70
10	2.5	0.25	2.5	-12.0	10	2.5			-75
10	2	0.2	2	-14.0	10	2	0.0001	0.001	-80
10	1.25	0.125	1.25	-18.0	10	1.25			-85
10	1	0.1	1	-20.0	10	1			-90
10	0	0.0	0	-100.0	10	0	0.0	0.0	-100

GLOSSARY

ADDITIVE SYNTHESIS	the process of mixing together a series of waveforms to obtain one complex waveform
ALTERNATING CURRENT	AC, the flow of electrons which changes direction
AMPERE	Amp, a unit of measurement of electrical current
AMPLITUDE MODULATION	periodic changes in the amplitude of a signal
ATTACK	the onset of an event and the first parameter of an envelope generator
ATTENUATOR	an electrical device which controls the peak to peak amplitude of a signal An attenuator can only decrease the amplitude so that the voltage input is equal to or greater to the voltage output.
AUDIO PATH	that part of the patch which includes the audio signal source and all the modules that modify that source
BALANCED MODULATOR	a 4 quadrant multiplier which multiplies the instantaneous voltage at one of its inputs times the instantaneous voltage at the other input
BALANCED WAVEFORM	a waveform whose amplitude is alternately positive then negative by equal amounts
BAND REJECT FILTER	a filter which passes all frequencies except those at or near its center frequency; a notch filter

BAND PASS FILTER — a filter which passes all frequencies at or near its center frequency and attenuates those frequencies on either side of the band

BANDWIDTH — the distance, in Hertz, from the high and low cutoff frequencies

BIAS — a non-fluctuating DC voltage

CARRIER SIGNAL — the signal in a modulation patch which is being modulated

CLIPPING — harmonic distortion which occurs whenever the amplitude of a signal exceeds the limits of the system

CLOCK — any repeating signal which provides timing information

COMPARATOR — a threshold detector, a device that outputs a gate whenever the instantaneous voltage of the input signal exceeds the amplitude of the threshold signal

CONTROL PATH — the part of a patch which is used to voltage control a signal's amplitude, wave shape or frequency

CURRENT — a flow of electrons

CUTOFF FREQUENCY — the point at which a filter has attenuated the input to 1/2 the value of the filter slope. The cutoff frequency of a -12 dB/octave filter is the point at which the output is 6 dB down from the input

DECAY — the amount of time it takes the envelope generator's voltage to get from its attack level to its sustain level

DECIBEL	one tenth of a bel, a unit of measurement of voltage, power, sound pressure or sound pressure level
DIFFERENCE TONE	a component of a modulation timbre whose frequency is equal to the difference between the frequencies of the carrier and the program signal
DIRECT CURRENT	the flow of electrons in one direction
ECHO	discrete repetitions of a sound or a signal caused by an acoustic or electronic device
ENVELOPE	changes in volume and/or timbre during one sonic event; the output of an envelope generator or envelope follower
ENVELOPE GENERATOR	a DC voltage generator offering control over its instantaneous amplitude with respect to time
ENVELOPE FOLLOWER	a module which outputs a DC voltage proportional to the average amplitude of an audio input signal
EXPONENTIAL CONVERTER	an electronic device that changes linear current into exponential current
FEEDBACK LOOP	a generalized term referring to any situation in which the input to a module or device consists of a portion of that module's or device's output
FILTER	a frequency selective attenuator

FILTER MODULATION	periodic changes in the cutoff frequency or center frequency of a filter
FLANGING	delaying a signal in time and mixing the delayed signal with the original signal
FREQUENCY	the number of periodic fluctuations per second
FREQUENCY DOMAIN GRAPH	a graph which plots frequency on the horizontal axis against amplitude on the vertical axis
FREQUENCY MODULATION	periodic changes in the frequency of a signal
FUNDAMENTAL FREQUENCY	the first harmonic in a signal's harmonic spectrum
GATE	a non-periodic signal which has only two voltage levels, high and low; it is generally used as a timing signal
GLITCH	an electronic hiccup
HARMONIC	any sine component of a complex waveform which is an integral multiple of the fundamental's frequency
HARMONIC SERIES	all frequencies which are integral multiples of the fundamental frequency
HERTZ	a unit of measurement which expresses the frequency of a signal
HIGH PASS FILTER	a filter which passes those frequencies above its cutoff frequency and attenuates those frequencies below its cutoff frequency
INVERTER	a device which changes the polarity of a signal; it multiplies by (-1)

KEYBOARD	a module which outputs discrete voltages relative to the key depressed; it also outputs a gate and a trigger
KLUDGE	an in-elegant solution to an electrical or mechanical problem
LFO	an oscillator whose frequency is set to below 20 Hz.
LOW PASS FILTER	a filter which passes those frequencies below its cutoff frequency and attenuates those frequencies above the cutoff frequency
MACRO-TONES	adjacent pitches which are more than a semi-tone apart
MICRO-TONES	adjacent pitches which are less than a semi-tone apart
MIXER	a module which adds together the instantaneous amplitude of two or more signals
MODULATION	periodic changes in any of a signal's parameters
NOISE GENERATOR	an aperiodic signal generator
OHM	a measure of electrical resistance
OSCILLATOR	a module which outputs a periodic signal
OVERTONE	a component in a timbre which may or may not be an integral multiple of the fundamental frequency
PEAK FILTER	an allpass filter which produces a peak at its center frequency
PATCH	a particular configuration of modules

PERIOD	the amount of time in which a periodic waveform completes one cycle; period is equal to 1/frequency
PHASE	the fraction of a waveform's period which has elapsed relative to some given point. Phase is measured in degrees and one complete cycle is equal to 360 degrees
PHASE SHIFTER	a module which changes the phase of a signal
PINK NOISE	an aperiodic signal which contains equal energy per octave throughout the audio range
PORTAMENTO	a gliding between pitches
PRE-AMPLIFIER	a high gain amplifier used to interface an externally generated signal with a synthesizer
PROGRAM SIGNAL	in a modulation patch, the signal that is controlling another signal
PULSE WAVE	a periodic waveform with two voltage levels, high and low
PULSE WIDTH MODULATION	periodic changes in the width of a pulse wave
REGENERATION	a signal taken from the output of a module and patched back into the module's input, (also called resonance, emphasis, or feedback)
RELEASE TIME	the time it takes for the final stage of an envelope generator to go from the sustain level to 0 volts
RING MODULATOR	a balanced modulator, the

	term "ring" originated from the design which employed a ring (bridge) of diodes
RESONANCE	the tendency of a system to oscillate in response to another signal
REVERBERATION UNIT	an electronic, acoustic, or electro-acoustic device used to simulate the natural acoustic decay of a sound
SAWTOOTH WAVE	a periodic waveform having harmonics at all integral multiples of the fundamental frequency and whose amplitude are equal to the reciprocal of the harmonic number times the amplitude of the fundamental
SEQUENCER	a module which outputs a series of controllable voltage levels
SINE WAVE	a periodic waveform having no harmonics other than the fundamental
SLOW RANDOM VOLTAGE	a sub-audio aperiodic signal
SQUARE WAVE	a 50% pulse wave
SUMMATION TONE	a component of a modulation timbre whose frequency is equal to the sum of the carrier and the program's frequencies
SUSTAIN LEVEL	the envelope generator control which determines the voltage level of the steady state portion of the envelope generator's output
SYNC	a function which permits a sharply rising voltage to reset a waveform to a given

	position in its cycle; when 2 VCO's are synched together the "master" determines the frequency of the "slave" oscillator
THRESHOLD	a reference level
TIME DOMAIN GRAPH	a graph that plots time on the horizontal axis against amplitude on the vertical axis
TREMOLO	periodic changes in a signal's amplitude at less than audio rates
TRANSDUCER	a device used to convert intelligence in one form of energy to another form of energy--microphones and speakers are examples of transducers
TRIANGLE WAVE	a periodic waveform whose harmonics occur at all odd-numbered multiples of the fundamental frequency and the amplitude of each is equal to the inverse square of the harmonic number times the amplitude of the fundamental
TRIGGER	a non-periodic pulse of about 10 mS. duration which is generally used as a timing signal
VIBRATO	periodic changes in a signal's frequency at less than audio rates
VOLT	a unit of measurement used in expressing electromotive force
VOLTAGE	electromotive force
WHITE NOISE	an aperiodic signal containing equal energy per cycle throughout the audio range

ADDITIONAL SUGGESTED READING

Appleton and Perrera, The Development and Practice of Electronic Music

Backus, The Acoustic Foundations of Music

Benade, Horns, Strings, and Harmony

Boulez, Boulez on Music Today; Notes of an Apprenticeship

Cage, A Year from Monday

Cott, Stockhausen: Conversations with the Composer

Lang, ed. Problems of Modern Music

Mileaf, Electronics 1-7

Reich, Writings About Music

Runstein, Modern Recording Techniques

Strange, Electronic Music; Systems, Techniques, Controls

Van Bercejk, Waves and the Ear

Wells and Vogel, The Techniques of Electronic Music

Winckel, Music, Sound, and Sensation

PERIODICALS

Audio Engineering Society Journal

Electronotes

Synapse