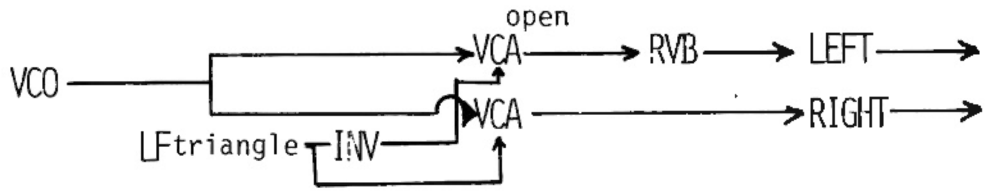
half of the signals from inputs #1 and #2 will be reverberated while half will pass through unchanged. The source control, then, determines how much of each input signal will be routed to the reverb springs and the level control determines how much it will be reverberated.

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.

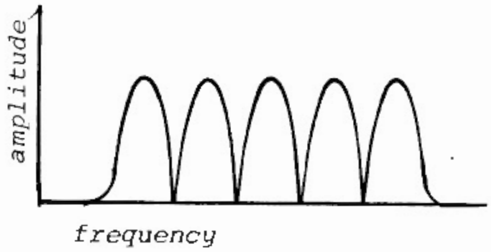
There are four audio inputs. A signal patched into any of these inputs becomes split. Part of the signal is routed to the phase shifting circuitry and part of it is not. Internally, there is a mixer near the output stage of the module that adds the phase shifted signal with the non-shifted signal. There 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.

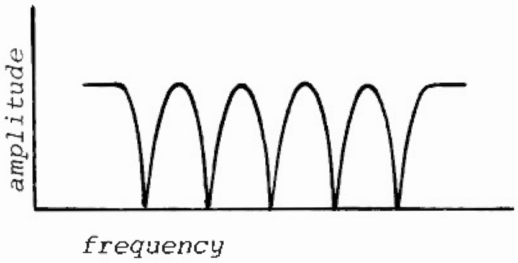
The exponential inputs are the usual control inputs with exponential converter which are found on the AR-314 and AR-327 filters. Their sensitivity is one volt per octave. The flange input allows the module to be used as an electronic flanger. The main and aux outputs are audio outputs that work in conjunction 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.



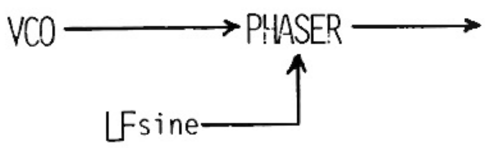
In almost all situations, the phaser produces a more "full" sound when the switch is in the even position rather than when it's in the odd position. Without getting into the math involved, here is what the switch does. At the input of the circuit, there is a phase splitting network. The outputs of the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

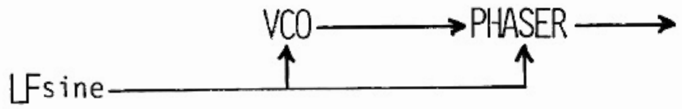
All these modules produce signals which are analogous to acoustic events. One way of producing acoustic phase shifting is to play an organ simultaneously through a leslie speaker and a stationary speaker. As each speaker in the leslie cabinet moves toward the listener, it causes a slight Doppler shift 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.

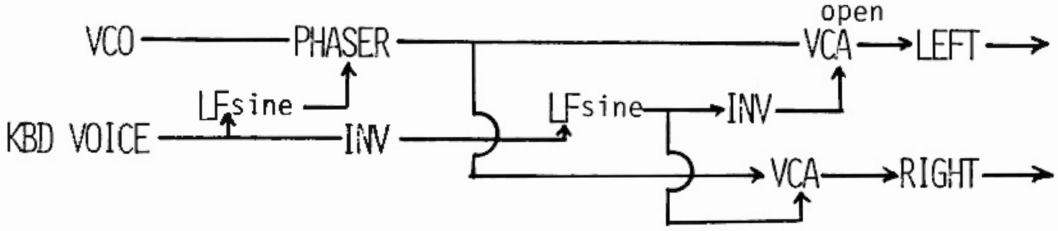


If you use the main output into one speaker and the aux output into another speaker and control the Fc with a slow sine wave or triangle, you can perceive apparent sonic motion between the two speakers.

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

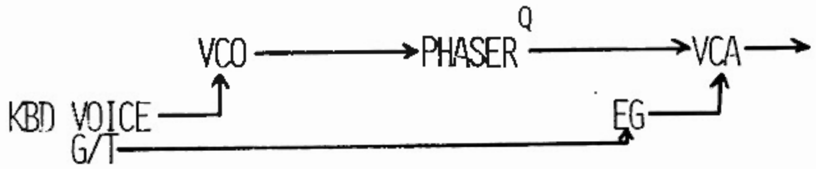


In this patch, the phasing rate is inversely proportional to the panning rate.



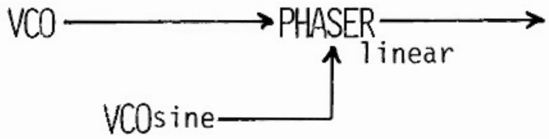
Or, you can use the phase shifter without sweeping the Fc. This will achieve some of those subtle timbral modifications we've previously mentioned. If you leave the Fc stationary and control the frequency of the input signal, some interesting formant structures will result. Try increasing the resonance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

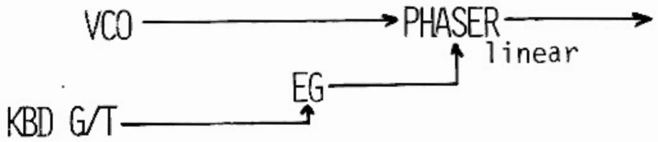


In a patch like this, you're completely on your own. Timbre is such a subjective factor that there's no way we can tell you what settings are effective or what will work in a particular situation.

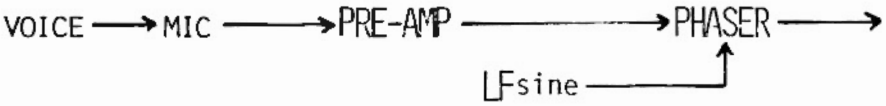
With a control voltage patched into the linear input, the phaser operates in exactly the same way. The difference is that the control voltage is not exponentially converted before actually going to control the phaser's Fc. Instead of the control sensitivity being measured in volts per octave, it is measured 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.