SYNTHESIS

Processing & Controlling

with Danny Sofer

Most synthesizers are used as sound generating devices: you play the keyboard and the synthesizer produces sound. However, a synthesizer does not have to operate this way. You can play your synthesizer with anything (well, *almost* anything) if you've got the proper interface. But you ask, "why would I want to do that?"

Well, if you can play the hell out of a violin but only "Chopsticks" on the piano it becomes apparent that by interfacing the violin to the synthesizer you can dramatically increase your timbral options on the violin and your virtuosity on the synthesizer all at the same time.

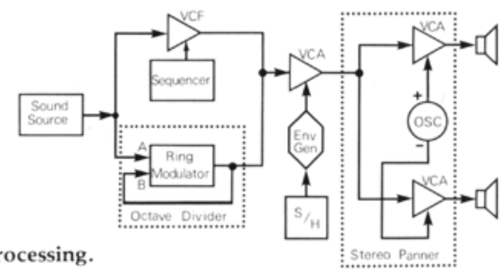

Figure One: Processing.

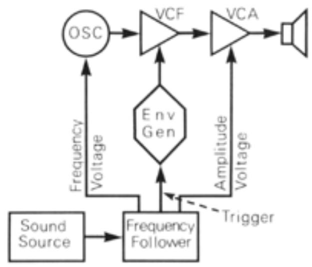

Figure Two: Controlling.

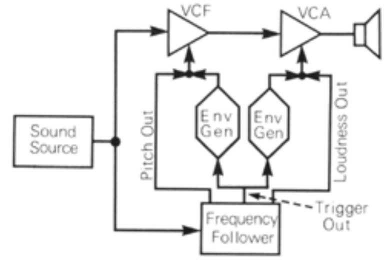

Figure Three: Processing & controlling combined.

In my mind, I divide the alternatives into two groups: Processing, when an outside sound source, such as a piano or guitar is patched into the synthesizer which modifies the timbre of the source without adding any new sound of its own. (fig. 1) The second group, external controllers uses an external device, be it a guitar, drums, microphones, light sensors, or whatever to control the sounds that the synthesizer is putting out, instead of using the keyboard for this (figure 2). If you've got a synthesizer and an electric instrument then you're ready to begin. Of course, an acoustic instrument must be electrified, whether with a microphone or better yet, a pickup such as a Barcus-Berry or Frap, and this goes for instruments that are not normally amplified such as jack hammers and electric toothbrushes. The output from any of these can be plugged directly in to the audio input or mic input of most synthesizer. (If it's not loud enough, use a small mic preamp or phono preamp as the case may be, available at any electronics store.) Usually, the audio input goes into the filter and VCA of the synthesizer which can be controlled by oscillators (LFO's), envelope generators, sequencers or even the keyboard (but the keyboard doesn't control pitch in this case). If you've got a ring modulator, by taking its output back into the second input you can get a square wave an octave lower than what you put in. Also, if you've got two VCA's and stereo outputs you can get a stereo tremolo effect (ala Rhodes) by using a low frequency oscillator to control both VCA's, one positive and negative (see figure 1). All of these effects have one limitation: The synthesizer functions independently of what's processed through it. Now, if we could just get all of these devices to work together . . .

The answer to this problem is to plug your instrument into another box called a frequency follower, pitch follower or pitch to voltage converter. These are actually three devices at once: one part figures out the fundamental frequency (pitch) that you are playing into it and puts out a proportional voltage (usually 1 volt) that can feed the oscillators in the synthesizer so that they play melodically what ever is being played into the pitch follower. The second part is the envelope follower which puts out a voltage proportional to how loud the sound is at any moment, the third section, the trigger sensor, watches for the input sound to go above a certain volume level at which point it puts out a trigger which can be used to fire an envelope generator or advance a sequencer, etc.

Two of these devices are the 360 Systems Pitch Follower ($595) and the EMS PVC (around $700). For electric guitars and basses 360 Systems offers the Slavedriver ($795) that overcomes a lot of potential problems that can result when two or more notes are played at the same time (we are talking monophonic synthesizers here). If you want six synthesized notes out of your guitar simultaneously then get out $13,000 and get 360's Polyphonic Guitar Synthesizer).*

Whatever your instrument, the synthesizer can now be controlled in all sorts of different ways that simply can't be done with a keyboard. But there is no reason to use only the synthesized sound; by combining control and processing techniques your controlling instrument can also control it's own processed sound (figure 3). In this patch the source's frequency controls the filter point, the loudness controls the VCA's output level, and the trigger fires two envelope generators which control filter & VCA envelopes (the frequency & loudness outputs "transpose" these envelopes).

Another way to process sound is by distortion. As any electric guitarist knows, distorting a sound gives it more "balls" and enables a longer sustain than would otherwise be possible. The filter and mic preamp work the best for this. Figure 4 shows a sawtooth wave in its undistorted form (a) and after it has been distorted by overdriving a VCF (b). Notice that the distortion has turned the sawtooth into a squarewave.

I've run through everything rather briefly but by experimentation you should be able to find dozens of new ways to use your synthesizer as something other than a keyboard machine.

*Moog makes an envelope follower which has the frequency control. Also, Serge and EMS synthesizers have a trigger sensor built in.

COMPUTERS

Using an A/D Converter

with Peter Hillen

When using an analog to digital (A/D) converter it is important to realize its limitations in measuring an unknown analog voltage. These limitations result from inherent errors in the measuring process. The object, of course, is to make the errors as small as possible. To better understand how to use an A/D converter and what some of its sources of error are, let's look at an analogous device: a movie camera. At first they may seem to have nothing in common but they both work on the concept of recording a continuous event by taking a series of samples of it.

A movie camera works by taking a rapid succession of still pictures. Each of these still pictures is a sample of the scene being filmed. It records the position of everything in the scene at that instant of time. Each sample is a valid representation of the scene only for the instant in time it is taken. When the movie film is played back the still pictures are shown in the same sequence in which they were taken and depict the scene as it was. The eye does not detect a series of still pictures, but rather continuous motion, because the playback rate is faster than the eye can respond. Our eye averages the pictures and that gives the effect of continuous motion. An A/D converter works in a similar way. The computer calls for the converter to measure an analog input at some specific sampling time. At each sampling time the converter takes the unknown voltage and converts to a digital word which acts like the "still picture" (see fig. 1). The computer can store each of these measurements for playback later, as in a digital delay line echo, or process them in some way as they are entered.

An important parameter to consider in both of these cases is the sampling rate. It is how often the samples are taken. The events between samples are not accounted for. If the time is short between sample times compared to the action being sampled, then it is possible to guess what happened between samples through an averaging process. If the sampling times are far apart,

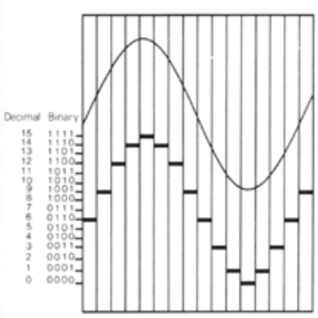

Figure One: Analog sinewave and digital equivalent.

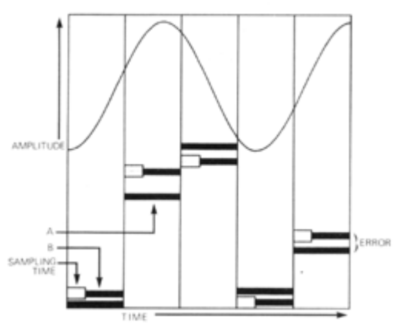

Figure Two: Effect of aperature time on accuracy of sample.

then the samples may not be an accurate representation of the event they sampled. There is a commercial on television where a crown appears on the head of some lucky person who is using a particular product. What has happened is that the camera was stopped, the crown placed on the actor's head and then the camera was started again. It can be thought of as slowing down the sampling rate of the scene and therefore all that transpired on the set was not recorded because some of it happened between samples. Therefore for an accurate conversion the sampling rate should be faster than the fastest event being sampled. Fig. 1 shows a sine wave which was sampled 16 times. The digital reproduction is not very good. Doubling the sample rate to 32 would improve it considerably. However, if only the frequency of the waveform is to be preserved and not the waveshape, then it is necessary to only sample at twice the highest frequency being converted.

In an electronic music system there are two classes of voltages to be sampled; first are the DC control voltages such as from a keyboard or ribbon controller. These change at relatively slow rates and sampling at a 30 Hz rate is sufficient. The second are audio frequencies: they can go to 20,000 Hz. The sampling rate to preserve the frequency must be 40,000 Hz. Why not just sample as fast as possible? The answer is not electronic but rather economic. Faster speed requires more film in the case of a movie camera or more memory in the case of a computer both of which are expensive.

To help complete the analogy of an A/D converter with a movie camera let us briefly discuss resolution. There is a discussion of resolution in the first part of this article on A/D converters which appeared last issue. Resolution is how close the digital word can come to the analog voltage being measured. It has to do with the number of bits (places) in the digital word. The resolution can be equated to focus in a movie camera. For example, an out-of-focus movie camera is trained on this page and from the pictures that are taken of it we want to see exactly where some of the lines are. The best we could do is determine that the lines were somewhere in a band of fuzziness. We would be tempted to say that the real line is exactly in the middle but because of angle and optical error this is not possible. This fuzzy area corresponds to the space between two consecutive numbers on the A/D converter output that the analog voltage may fall between. For example, if the converter had two place resolution the output would round off an analog voltage of 4.963 volts to 4.96. It could not resolve the .003 volts.

The final error to consider is aperture error. This error is related to how fast the sample of the event can be taken. In a movie camera if the film is slow, meaning that the shutter must be open for a long time to get enough reflected light to expose the film, there is a chance that some images will be blurred because they moved during the exposure time. The remedy is to use faster film which requires shorter exposure time. Electronically an A/D converter takes a finite amount of time to convert the analog voltage to a digital word. In the last issue the time required was shown to be greatly reduced by the use of a successive approximation register. During the conversion time the wave being converted may change causing an error to result. Figure 2 shows a wave being sampled at the same sample rate but with a different sampling or aperture time. "A" is the ideal sampling which takes zero time. "B" takes a finite sampling time. Note that the error of "B" compared to "A" is not the same from sample to sample. The solution to the aperture error problem is to use a smaller aperture time. This could be done by making a faster A/D converter. However, A/D converters get quite expensive as they get faster and in between conversion times they would just sit idle. The solution is to put something in front of the A/D converter which can *sample* the unknown waveform with a very fast aperture time and *hold* at that value until the A/D converter can convert it. The speed and cost of the A/D converter is greatly reduced because now the limit on conversion time is the sample rate, not the aperture time. There can be as much as a 1000 to 1 difference in the two requirements. You may have noticed from italicized words above that this function is performed by a Sample and Hold already familiar in electronic music. It does as its name implies: samples an unknown voltage and holds it for further processing. It is considerably less expensive than a faster A/D converter.

Next time we will explore how a Sample and Hold works and some of its uses.

Syndrum/ *from page 45*

envelope generator (instant on, variable decay) and VCA, as well as a snare drum circuit within a two position switch that apparently controls the decay of the white noise (short being approximately 50 ms, and long being approximately 250 ms). The envelope generator can be used to control the frequency of the main oscillator, (a sweep up or down) a nice effect used to simulate tablas, loose drum heads or generate further-out sounds.

There is a footpedal to control the frequencies of the main oscillator in each module simultaneously. Also a "kill" switch to cut short a long decaying sound. There are individual high impedence outputs for the modules as well as mixed high & low impedence outputs.

The Syndrum is pressure sensitive, however the dynamic range is somewhat limited and applied only ot the LFO envelope generator and VCA, so that while its effect can be detected, it is rather subtle.

The main thing that they've left out is patching; one can't bring drum voltages outside or take other voltages into the Syndrum. There is no way to control, say the main oscillator with the dynamic voltage from the drum alone. Or no way to cause several events to occur simultaneously upon one hit on the drum. Oh well. If you're into getting drum sounds and/or replacing the tom toms on your drum set, then go for the Syndrum. But if you're interested in manipulating your synthesizer with drum sticks the best bet is still the old Moog/Ludwig percussion controller. Perhaps four of them, each with its own Oberheim Expander Module. The Syndrum is made by Pollard Industries, Inc., 9014 Lindeblade Street, Culver City, California 90230.

—Danny Sofer