Sampling your own drum sounds (or anybody else's for that matter) and accessing them from your drumkit via MIDI has become all the rage. This is the first in a series designed to indoctrinate, educate, and de-intimidate you about the world of sampling. Text by Chan G. Ling.

"AMPLE IT!" IN the last few months, I've heard musicians yell those two words everytime something of even marginal aural interest happens. Audio feedback, the scrape of a chalkboard, the screech of tires, or the sound of an unlucky busboy dropping a stack of cafeteria trays calls up this new battle cry of the electronic drummer.

Samplers are the hottest items in an electronic musicians bag of hardware. Almost every company currently manufacturing electronic instruments now includes a sampler or two in their product line. You can find sampling keyboards, rack-mounted samplers, sampling drum machines and electronic drumkits, and even sampling reverb units. Samplers are everywhere.

If you walk into your friendly neighborhood toy store, you can find samplers on almost every aisle. My daughters have a little parrot that samples whatever you say and plays the sound back through a built-in speaker (Poily want a frontal lobotomy?). While this can be called a sampler, it doesn't quite qualify as an electronic instrument because it can't be controlled by a performer — at least not easily, that is.

Samplers come with different shapes, sizes, abilities, and prices (always an important consideration). At the beginning of this decade, the only sampler that was available for the normal income-producing human was the Fairlight C.M.I. (Computer Musical Instrument). The price for this baby was only $28,000 back in late 1979. Luckily, electronics go down in price (they seem to be the only things that do). Remember when the digital watch that costs six dollars today cost around $150 (and that was without the stopwatch and calendar features)? Technological advances and decreasing prices for electronic chips have made samplers affordable. Currently, samplers range in price from about $129 for the Casio SKI to over 1000 times that (that's $129,000) for a loaded Synclavier from New England Digital.

Just what are these magical mystery machines? What do samplers do that make them such a hot product? Do you want one? What can you do with them? Can you live without them? Should you sell your car to buy one?

During the course of this series, I will try to answer these questions, and discuss what samplers are and how they work, what features are available, and how you can use a sampler if you should decide to add one to your electronic percussion setup.

Analog Versus Digital

SAMPLING INSTRUMENTS ALLOW you to record any sound from the outside world (or inside world, if you can get a
microphone in there) by changing analog signals into digital signals. I'm sure that you've heard these two terms bandied about when talking about anything from musical instruments to alarm clocks, but what is the real difference between digital and analog devices? **Analog** is simply a term which is used to describe something that is infinitely variable. An analog knob can be set to any position between its upper and lower limits. Your oven's temperature knob, for example, is most likely an analog style control. Between the two extremes of 'off' and 'broll', you can dial up any temperature. If your dial is accurate enough, it would be possible to bake your famous chocolate chip cookies at 375.07864 degrees.

The term **digital** is used whenever there are a fixed number of positions between full 'off' and full 'on'. If your home FM-stereo tuner has a digital display, you can tune to channel 10.1 or 101.7, but you can't tune into a station that may transmit on the frequency of 101.65. That frequency just can't be picked up by the tuner because its digital format doesn't allow for the extra decimal place.

Another analogy can be made by comparing the fingerboard on a violin with the fingerboard on a guitar. In theory, at least, on either instrument you can place your fingers anywhere between the bridge and the nut. The violin uses your finger as the point that determines the length of the string. Any pitch along the string can be played by placing a finger down at the proper point. In addition, between any two positions (like the pitches C and C#) there are an infinite number of other pitches that are available; that's why violinists sometimes play out of tune. But the guitar works differently, as it will adjust your finger position to the nearest fret. If the guitarist wants to play a pitch between C and C# he's out of luck because the system of using frets doesn't allow this. If the strings of the guitar are perfectly in tune to begin with, all the frets will also be in tune. Of course, we all know that a guitar player can bend the pitches of his instrument, but when he does, he's cheating.

Another way to add notes in between pitches would be to add more frets. If the number of frets on a guitar were doubled, then it would be possible to play the note between C and C# but I'm getting a little ahead of myself.

A digital sampling unit performs a series of very basic tasks. Its first task is to record sounds and turn them into numbers (this is where the name digital comes from - 'digits'). The second task is to store the numbers in some way so that they might be manipulated by the processor using the sampler. The last task is the reverse of the first - taking the digital numbers that represent the sound and turning them back into signals that can be played over a speaker system. Let's take a look at each of these different tasks and see how a typical sampler might perform them.

**What Is Sound?**

THE FIRST STEP of converting analog sound into numbers is the hardest to understand. So before we get into how it's done, let's review a few basic concepts about sound itself - how it works and how it is turned into electrical energy. I'm going to keep this at a fairly basic level, so you don't need to pull out your slide rule yet. By the way, what happened to all those slide rules? Weren't they replaced with digital calculators?

After striking a cymbal with a stick, you can see the outer edge of the cymbal vibrate. In fact, whenever you hit anything, be it a drum, cowbell, or your sister, if it makes a sound, it's vibrating. Sound itself is nothing more than vibrations moving through the air. Before the cymbal was struck, the surrounding air was stationary and at rest. But when the cymbal starts vibrating up and down, the surrounding air molecules get pushed around. If you place your hand near the edge of the cymbal, you can actually feel the air moving. As the cymbal vibrates in one direction, air molecules are compressed, and as the cymbal vibrates in the other direction, the molecules are made less dense.

When our ears hear the sound of the cymbal crash, they are really responding to the subtle differences in air pressure that are created by the vibrating metal. The distinctive patterns of compression and rarification (the portion of the sound where the molecules are less dense) is how we distinguish the difference between sounds. A cymbal makes a different pattern of air pressure than a flute, a snare drum, a bongo, or a floor tom.

Take a look at examples 1 and 2. The first is a waveform of a cymbal crash and the second is the waveform of a floor tom. One of the first things to understand when looking at these examples is that the waveforms are only a visual representation (or a graph) of the varying amounts of air pressure. On the left side, there are numbers that represent the changes in air pressure (given as a percentage). The zero point can be looked upon as air pressure in its normal at-rest state. As the numbers go up on the positive end of the scale, the air is compressed, and as the numbers move down to the negative end, the air pressure is rarefied. The time continuum is moving from left to right.

There are many aspects to the sound that can be seen by comparing these two examples. First, you can see that the cymbal crash has several more 'spikes' to the wave, while the tom's waveform changes from the positive to the negative more slowly. These changes (from the positive side to the negative side of the scale) show the wave's period or cycle. A single cycle of a wave determines its pitch or frequency. Musical pitches are measured in Hertz, named after a 19th Century German physicist. Hertz (usually abbreviated 'Hz') are a measurement of the sound's cycles related to a second. In other words, a sound having a frequency of 200 Hz will go through 200 of its cycles (both positive and negative readings) each second, while a sound of 2,000 Hz will have 2000 cycles during each second of time, and seem much higher to our ears. By looking at the two waveforms in the example, you can see that the cymbal crash has a higher pitch than the floor tom.

As well as 'seeing' the frequency of these two examples, you can also determine something about how the sound's amplitude. Amplitude is a measurement of volume, and is determined by the amount of change from the positive to the negative. Sounds with a higher amplitude (read this as higher-volume) have a more dramatic swing from the positive to the negative, because louder sounds will push more air. Have you ever stood in front of an 18" woofer at a heavy metal rock concert and wondered why your clothes were almost ripped from your body? It's the high amplitude and massive amounts of air that are being moved. Don't be deceived by the example, the floor tom isn't necessarily louder than the cymbal, it's just that the floor tom was recorded at a louder level. Even though you're only looking at the relative amplitudes of each sound, you can see that the overall shapes of the sounds are quite different.

The cymbal's volume grows just a little bit after the strike, stays fairly steady for a while, and then begins to decay (in the example, the sound was stopped during the decay). The tom, on the other hand, reaches its maximum amplitude immediately, falls very quickly, and then has an extended decay which is at a very low amplitude.

OK, now we've taken a look at two things that make up any acoustic sound (frequency and amplitude), but there is still one more - **timbre**. Timbre is a word which describes a sound's color. It's very

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difficult to talk about a sound’s quality with words, but a sound’s timbre is determined by the shape of the wave during a single cycle. Early electronic instruments used a series of waves that were named for their shape, like sine, sawtooth, triangle, square, and others. Acoustic instruments produce very complex waveforms that actually change in shape over the duration of the sound. Let’s go back to the crash cymbal again. The instant you strike it, the sound of the stick is included in the timbre. Then, as the sound continues, some overtones will decay faster than others. In other words, the cymbal’s timbre changes over time. And this is the primary advantage that samplers have over any other type of electronic synthesis. Samplers have the ability to capture a sound’s frequency, its amplitude, and its timbre all at once — and for an extended period of time. Let’s look at how this is done.

**Analog/Digital Conversion**

**ANALOG TO DIGITAL** conversion is accomplished by a small chip inside the sampler called (appropriately enough) an Analog to Digital Converter. This chip reads voltages from the audio input and assigns numbers to them. How is this done? Think of a microphone plugged into a sampler. When you say something into the mike, the element inside vibrates in accordance with the changing air pressures. Keep in mind that the changing pressures are determined by the frequency, the amplitude, and the timbre of the original sound. The vibrations from the mike’s element are turned into analog voltages which are then read by the A/D converter and assigned numbers.

Now comes the tricky part. How are the analog voltages transformed into numbers? While it isn’t done with mirrors, sampling is very similar to a magic trick. Only this time, the trick is for your ears instead of your eyes. Let’s look at a magic trick for your ears and see how they compare. A movie is nothing more than a series of still photographs that are projected on the screen at a rate fast enough to fool your eyes into thinking that there is smooth, continuous movement on the screen. Even though it might look like an actor walks across the screen from left to right, it’s possible to stop the projection of the film at any particular point, and see what’s actually happening. When you look at the first frame, you’ll see a still photo with the actor in the far left of the frame. Each subsequent frame will reveal that the actor is just a little further to the right. As the film runs, your eyes (and your brain) will fill in all the necessary details to create the ‘moving picture’. However, your ears are much harder to fool than your eyes. While a film speed of only twenty-four

frames per second will give your brain the impression of fluid movement, a speed of thirty thousand or more ‘frames’ per second is required to fool your ears.

As the analog voltages enter the sampler, the A/D converter ‘looks’ at the input several thousand times per second. This is called the sampling frequency or the sample rate. To give you a feeling for how often samplers look at the input, a compact disk was sampled at a rate of 44.1 thousand samples per second. Why do our ears require such a fast rate? There is something called the Nyquist theorem which states that the sample rate must be at least twice as fast as the frequency that you want to sample. This seems to make sense, as you would need at least a positive and a negative reading for each cycle. In other words, if the sound you want to sample has a frequency of 100 cycles per second, the minimum sample rate would be 200 samples per second. Since the highest note on the piano is 4186.01 Hz, the minimum sample rate is 8372.02. While this may capture the fundamental frequency of this note, what about the overtones that are included, the slight alterations in the wave’s shape and all the subtle nuance that determines tonal quality? These features will be lost with such a slow sample rate. If you consider that the range of normal human hearing is from 20 Hz to 20,000
Hz, then the minimum sample rate would be 40,000 samples per second in order to capture every detail of the sound.

**Resolution**

SO LET'S SAY that we're using a sampler which is looking at the incoming wave 40,000 times every second. After it looks at the wave, what happens then? Well, each time the A/D converter looks at the input, the wave's analog voltage is assigned a number (remember, this is the digital part) which serves as the measurement of air pressure. If the voltage is low, then a lower number is assigned; as the voltage increases, the assigned digital numbers also increase. But remember that digital numbers are finite, while analog voltages are infinite. This requires a certain amount of 'rounding off' when the conversion takes place. The amount of numbers that can be assigned to the voltage is called the sample resolution. Generally speaking, the higher the resolution, the clearer the sound.

Because we're working with sounds in the digital domain, and therefore have to deal with computers, the sample resolution is usually given in 'bits'. As a quick review, bits are the individual numbers of 0 and 1 that make up a computer's language. A sampler's resolution is determined by the number of bits that the A/D converter works with. A two-bit computer (not to be confused with a computer that's only worth two bits) can only signify four different digital values (00, 01, 10, and 11). A four-bit computer can express sixteen values while an eight-bit computer deals with 256 values. As a rule of thumb, each additional bit adds about 6 dB to the signal to noise ratio and dynamic range. In the world of digital sound, you can interpret 'signal' as an accurate wave representation, and 'noise' as garbage, or inaccurate representation.

Example 3 shows a typical analog wave. Just for the sake of comparison, let's say that this wave's frequency is 100 Hz. Example 4 shows the resulting digital waveform sampled at a rate of 1,600 samples per second (sixteen samples of the 100 Hz wave), and a sample resolution of four-bits (sixteen possible values). You can see that the digital wave shown by the heavy line is quite different from the original waveform.

In example 5, the sample resolution is doubled, even though the sample rate remains the same. Because there are more digital numbers available for measurement, the sampled wave begins to look a little more accurate. In example 6, the sample rate is also doubled so the wave is measured twice as often. While this wave may look pretty good, imagine how close it would come if the original wave were sampled 40,000 times per second and had 65,536 possible values (sixteen-bit resolution)! It would be very difficult to tell the sampled digital wave from the original analog wave. Is it live or is it digital?

**Playing With The Bits**

ONCE THE SOUND is in the sampler in a digital format, what happens then? The numbers are stored in the sampler's RAM (random access memory), ready for playback whenever the proper instructions are received, such as pushing a key or striking a MIDI drum pad. When the sampler plays the digital waveform, it simply plays back those numbers. If the number of 'frames' or samples are played back at the same speed that they were recorded, then the sampled sound will be an accurate representation of the original. If the samples are played back at a slower rate, then the pitch or frequency of the digital sound will be lower than the original, and if played back at a faster rate, the pitch will be higher. In a way, this is also similar to film. Playing the film back at thirty frames per second will make all the movements faster, and playing it at fifteen frames per second will seem like slow motion.

Since our ears respond to differences in air pressure rather than differences in numbers, the digital representations must be converted back into analog voltages before we can hear them. Inside every sampler is another converter (this time a D/A converter) that reads the digital signal and converts it back into the analog information that can be understood by mixers, amplifiers, and speaker systems.

Let's stop for just a second and think about something. We've got an analog sound consisting of voltages that are fed into an A/D converter which turns the voltages into numbers. Then, those numbers are sent through a D/A converter and turned back into analog voltages so that we can hear it. Doesn't it seem like a big waste of time to convert the waveform into numbers if we have to convert them back again in order to hear it? Well, the big advantage of digital music is that once the sound is in digital form, you can use the power of the computer to alter those numbers in various ways. After all, if there's one thing that computers can do really well, it's crunch numbers. Here's just one example: what do you think would happen if you tell the computer inside the sampler to play all the numbers backwards? The last sample will now be the first, and so on. That's right, "Lo Kawdot tob ealp ebb druoos eht".

Next month, we'll cover several other types of digital processing that samplers can handle. We'll also take a look at features that are important to drummers who want to use samplers as a sound source for their electronic system.