The following appeared in edited form in Muzik ETC in Graham Collins' <u>Synthesizer Basics</u> column.

More than any other single component, a synthesizer's filter defines it's sound and overall feel. In our ongoing investigation into the various sections of common synthesizers, this month we're going to be dealing with the filter section. Filters have a certain sense of mystery about them. Most people have heard about them and know that they can change a sound, er.. somehow. Most novices when they first discover an old synth will somehow naturally find their way to the filter's cutoff frequency knob as the "knob of choice" to let their fingers dwell upon for a while to twiddle.

In the natural world, most sound generating objects have certain formant characteristics. What is a formant? It is an area of accented or de-accented frequency activity within an instrument's pitch range. Huh? Ok, lets try this another way. When you play a violin or guitar, your fingers move up and down the neck to shorten or lengthen the effective string length thus changing the note. The body of the instrument however never changes in size, and *it* is also a determining factor in the overall tone of the instrument. String players know that things like shape, size and design of an instrument's body changes the tone of the instrument. The result is that with a resonating cavity such as this, certain frequencies are reinforced and others won't be regardless of what actual note you are playing. This gives an instrument a particular sonic 'fingerprint' if you will. These areas of frequency boosting are known as formants.

So what's a filter? Well, it's a device that *removes* (and/or to a lesser degree *adds)* a particular group of frequencies from/to a sound. Most people use filters all the time and simply don't know it. Ever change the tone controls on your stereo? Ever used a parametric eq on a mixer board? Ever use a guitarists 'wah-wah' pedal? ...ever made *coffee??* Ok ok...*that* groaner is out of the way, but it does reinforce the idea that you are removing things from something complex to make it simpler in nature. This incidentally is where the term *subtractive* synthesis comes from.

Most filters only have a couple of parameters--*Frequency cutoff* (sometimes just called *cutoff* or abbreviated as *fc*) and *resonance* (also called *feedback*, *peak* or *Q).* Simple. The frequency cutoff point determines the frequency above/below/around which the filter will remove the unwanted frequencies. If that statement seems indecisive it's because there are actually different *types* of filters. The most common type is a *low-pass filter* (lpf) which removes all frequencies above the cutoff point. In second-place in the popularity department is the *high-pass filter* (hpf) which removes all frequencies *below* the cutoff point. Third most common is the *band-pass filter* (bpf) which removes frequencies *around* the cutoff point. Band-pass filters also have the additional parameter of *bandwidth* to determine the range of frequency-cutting. (Parametric EQ's are a perfect example of a common type of bpf) There are other types as well, but those are the big three, and low-pass filters are most often assumed when someone refers to an average synth's filter without specifying. Resonance is the second parameter, and a little trickier to get your head around. In short, it is a variably controlled feedback loop feeding-back the filter's output into it's own input. Yes I mean feedback, as in a microphone howling out of control or a guitarist making *that* face during a solo. Turning resonance up full results in that "ZAP!" sound so often associated with synthesizers and electronic music. Resonance is the feature of the filter that actually can *add* new frequencies to a sound. In fact, the frequencies being boosted by an increase in resonance are directly centred around the cutoff frequency. Careful tweaking of only the cutoff and resonance parameters can yield drastically different sounds from a synth.

There is one other feature of a synthesizer filter and I hate to say it, but there's math folks. Usually predetermined by design (but occasionally adjustable) is a filter's *rolloff slope.* This is measured in terms of decibels per octave (dB/oct). Most filters are either 24dB/oct or 12dB/oct, also known respectively as a *4-pole filter* and a *2-pole filter*. Saying "pole" is equivalent to saying "6dB per octave". It's a kooky system to be sure, but all you need to know is that the higher the rolloff rate the more "efficient" the filter is at its job. This isn't to say that a 4-pole filter is *better* than a 2-pole jobby, it simply has a *different* sound ...and I don't mean that in the "look how fair I'm being to all involved parties" kind of way that that sounds. Different jobs require different tools.

In the heyday of the 60's and 70's, manufacturers watched each other's designs carefully, occasionally trying to mimic certain qualities and occasionally running into litigation. Each company had a particular 'sound' to their machines and it was almost entirely due to the instrument's filters. Moog filters sounded 'fat'. Korg filters were 'squawky'. Roland filters were "smooth". Sequential Circuits machines sounded 'biting' and 'raw'. An interesting factoid about the Minimoog filter is that some have ascribed it's warm 'fatness' to the physical layout of it's circuitboard, component juxtaposition and heat dissipation!! ..and they say electronic instruments can't sound "organic"? Pah!

Don't miss our next installment folks, when we'll learn about a few more components *and* start putting them all together!

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