PHASER FILTER SPECULATIONS

Max Mathews, Jan 2009

A phaser filter is a digital filter which introduces a single resonance into a sound file which is put through the filter. Adding a single resonance almost never makes an interesting change in the timbre of a sound, but adding many resonances--tens or hundreds of resonances--can create surprisingly interesting and beautiful timbres. Almost all successful musical instruments have many resonances. Computers are now so powerful that they can now synthesize enormous banks of resonances in real-time.

Difference equations for a resonant filter are well known but can be hard to use, particularly if one wants to dynamically retune the resonant frequency or decay time of the filter while it is processing a sound wave. Changing the filter parameters can introduce discontinuities in the sound wave which produce clicks into the sound. Phaser filters use a different difference equation in which changing resonance frequency or decay time does not generate discontinuities.

The main perceived effect of filtering a sound file with a bank of phaser filters is to change the timbres of the sound file. The effect is much stronger than I expected. Filters cannot add new frequencies to the output sound. All the output frequencies must be present in the input file. But the strength of an input component can be greatly changed and the duration of a component is greatly increased by filters with long decay times.

Perceptually, the filtering seems to throw away the input wave timbres and create highly resonant sounds, the resonant frequencies and decay times coming from the phaser filters. Often the perceived timbres are interesting and sometimes very beautiful.

What got me interested in resonant filters? I believe it is my dissatisfaction with the timbres of most computer music. It seems clear that almost all successful musical instruments have lots of resonances. I wondered why? My theory, as yet entirely speculation, is that the human auditory brain contains neurons which are "resonance" detectors, that is these neurons are activated when an sound wave is heard from an instrument with resonances. Moreover the listener likes to hear resonances. He finds these timbres interesting and beautiful.

The resonances must produce spectra that change in time at limited rates of change. Risset, in the 1960's, discovered that to synthesize interesting instrumental timbres, one cannot use constant spectra but rather the spectra must change over the duration of a note. If the change rate is slower than about one per second, the brain gets bored. If the change rate is faster than about 10 per second, the timber sounds harsh. If the change rate is very fast, such as white noise, the timbre is no longer perceived as changing but is a constant smooth noise.

There is clear evidence that some visual neurons only respond to changes in retinal images. If one projects a stationary image on the retina (stabilizing a retinal image is very difficult to do) the perceived image rapidly fades away. What is true for visual neurons may also be true for auditory neurons. So rates of change of whatever stimulus is being received by our sensors may be important to our perceptions.

Computers solve difference equations. Difference equations for resonant filters have been well known for centuries. These filters can be characterized by a resonant frequency and a decay time. If you excite the filter with an impulse or another input having energy at the resonant frequency, the filter will "ring" at the resonant frequency and the ringing sound will gradually decay according to the decay time of the filter.

The sound of one resonant filter is simply the sound of a decaying sine wave and is not at all interesting. But a bank of 10's to 100's of resonant filters, each tuned to a different resonant frequency and decay time can be interesting and beautiful. Modern computers, including laptop computers, can easily synthesize 1000 filters in real-time.

In traditional difference equations for infinite impulse response filters, changing either the resonant frequency or the decay time while the filter is actively filtering an input sound will usually change the energy in the filter and thus produce a unpleasant sounding discontinuity in the output of the filter. Phaser filters are special difference equations that eliminate the discontinuity in the waveform of their output wave resulting from changing the parameters (frequency and decay time) of the resonance. Changing the parameters does introduce a discontinuity into the derivative of the output wave, but this is less audible than a discontinuity in the wave itself.