SoundMagic Spectral

Real-time spectral plug-ins for creative audio processing

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1. Introduction

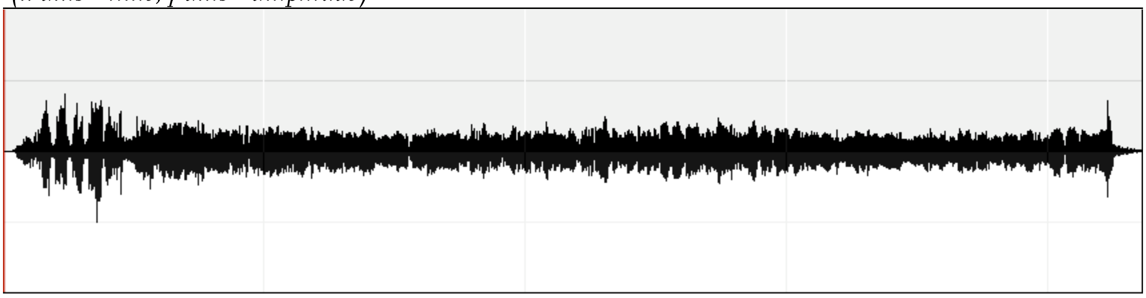
SoundMagic Spectral is a suite of freeware Audio Unit plug-ins that offer exotic real-time spectral processing effects, along with a few time-domain effects. This document explains the theory behind spectral processing and details each plug-in in the suite.

2. Understanding spectral processing

Time domain vs. frequency domain processing

A **time domain representation** of sound is one in which only the instantaneous changes in sound pressure are displayed, recorded or manipulated. This can be seen in the image below:

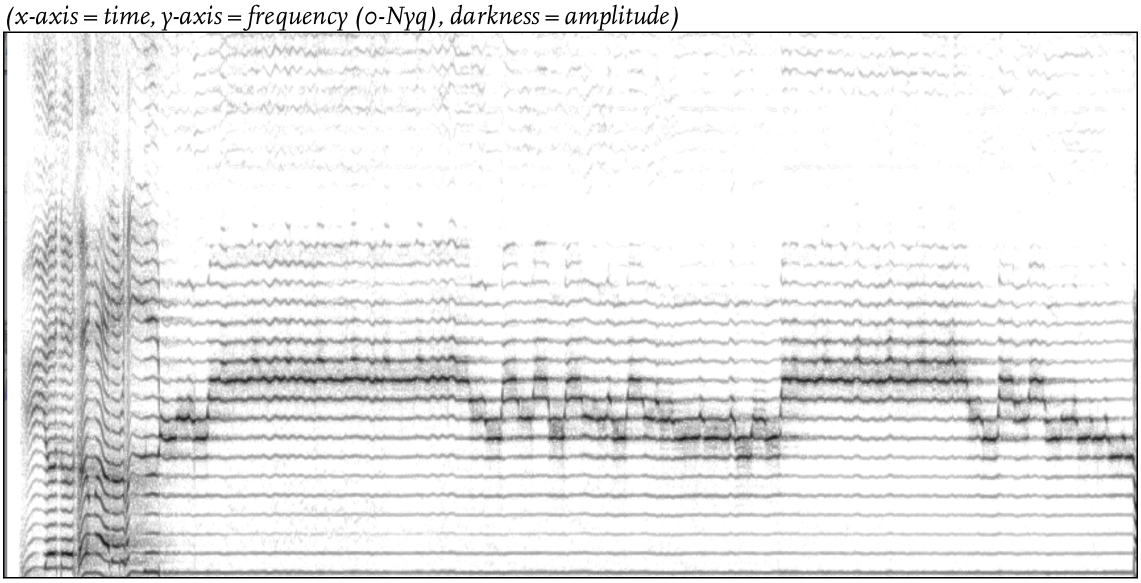
(x-axis = time, y-axis = amplitude)



The graphic here represents only the overall amplitude changes in the sound. However, this representation tells us little about the frequency content of the sound (other than some vague ideas about pitch or noise content by looking at the randomness of the waveform). It is hard to tell from this picture just *what* the sound actually is.

In the second image, however, we see the same sound converted to a **frequency domain representation** (also called a "spectral representation"):

(x-axis = time, y-axis = frequency (0-Nyq), darkness = amplitude)



Here the raw sound data has been converted into a series of frequencies and amplitudes. The frequencies are higher as you go from the bottom to the top of the graphic; the amplitudes are represented by the *darkness* of these component frequencies. In the soundfile represented above, we can see that while the overtones of the spectrum remain constant in frequency, different harmonics are being highlighted. Indeed, this soundfile is a recording of Tuvan throatsinging, and the spectrogram above graphically shows how the throatsinger can create a melodic line even without changing the fundamental frequency of his vocal cords. This sort of information could not be seen at all in the time domain representation.

Now, the main difference between a **time domain process** and a **frequency domain (spectral) process** is that spectral processes have two extra steps involved in the processing of audio: spectral **conversion** and **reconversion**. Firstly, the effect converts the raw sound data into a "spectral representation" — a series of numbers representing the strength of the frequency components within the sound at various points — before processing. Then, after operating on the spectral data in some manner, it reconverts the spectral representation back to a time domain representation (sound file format). In order for a spectral effect to operate in real-time, this conversion and reconversion must happen on the order of 150 to 200 times a second for a typical configuration.

The Fast Fourier Transform

The algorithm that drives these conversions is called the **Fast Fourier Transform** (FFT for short). FFTs are commonly thought of as computationally expensive: SoundHack, for instance, seems to use rather slow, unoptimized code (though with excellent sound quality), and as a result cannot operate in real-time. In recent years, however, the availability of hardware-optimized versions of the FFT and other maths libraries, along with general increases in processor capability, have made real-time spectral processing possible. These optimizations have been implemented as far as possible in SoundMagic Spectral.

Spectral bins and FFT sizes

In order to understand how these effects work, it's useful to have a little bit of "behind the scenes" understanding. The frequency domain representation of audio consists of an array of frequency bands (called "bins"), equally spaced across the spectrum from 0Hz up to the Nyquist limit (sampling rate/2). You can choose the number of bins that you want in the FFT, but it must be a power of 2 (e.g. 1024, 2048, etc). In choosing the right FFT size, there is a tradeoff between the amount of frequency data that can be captured (the higher, the better) and the time accuracy upon resynthesis (the smaller, the better). So if the "detail" in the sound is important, then smaller FFT sizes are better. But if you want more accuracy in frequency, then choose larger FFT sizes.

Once the FFT has converted the time domain data into the frequency domain, the particular DSP effect is then given a series of frequencies and amplitudes that represent the strength of various frequency components (for instance, harmonics) present within the sound at a given time. The job of a spectral process is to now do something with those frequencies and amplitudes to radically alter the spectral makeup of a sound. There are many, many clever things we can do with them, and the SoundMagic Spectral effects are just a start, but I'm sure there are many more than have been thought of so far. In many ways, spectral processing is still in its infancy.

3. A GUIDE TO THE SOUNDMAGIC SPECTRAL PROJECT

The SoundMagic Spectral project builds on the ideas implemented in the SoundMagic plug-in suite that I started in 1995 for the now discontinued OS 9 application SoundMaker. SoundMagic Spectral brings not only effects familiar from the SoundMagic suite to Mac OS X through the standard plug-in format called "Audio Units", but furthermore adds many new features unthinkable in SoundMaker: real-time processing, G4/G5-optimized FFT and phase vocoder routines and even more exotic and creative effects than in the original suite.

The concepts behind the SoundMagic algorithms owe much to the work of previous composers and programmers — most notably Trevor Wishart in his book *Audible Design* (a must-read for those using DSP) based upon his work with the Composers Desktop Project.

DOWNLOADING

These effects may be downloaded with a "beta" status (i.e. limited support) from the following URL: *http://www.michaelnorris.info/soundmagicspectral*. I hope to release of "final" version of SoundMagic Spectral in late January 2007. In the meantime, updates will be posted to my website.

LICENCE & SUPPORT INFORMATION

The current set of plug-ins is considered "beta" software. It's entirely possible that the effects contain some lingering bugs/quirks, though hopefully I've ironed most of them out. They are offered as freeware, so please download them, use them and spread the word. While I am open to, and interested in, suggestions for new effects and new parameters for existing effects, I can't guarantee immediate satisfaction of your request. I do have a working to-do list, however, so email through any bugs/suggestions to me at <michael.norris@nzsm.ac.nz>.

MINIMUM TECHNICAL REQUIREMENTS

SoundMagic Spectral requires a Mac with a G4 or G5 processor, running Mac OS X 10.4 or later. An AudioUnit-compatible host, such as Logic, Peak, Performer, Live, Amadeus, Sound Studio, or Max/MSP using my newly developed *au~* object, is needed.

LATENCY

Because these effects use large chunks of sound, there is often significant latency, usually of about half a second, sometimes longer, between the input and output. This means that it's a good idea to ensure there is at least a second's worth of silence at the end of your sound file, to allow for the processing "tail".

PHASE RANDOMIZATION

The optimized phase vocoder routines in these effects do not perform the usual "phase unwrapping" stage that is typically seen in most phase vocoder implementations. The reason is that it is computationally expensive, and for most processes is actually redundant. By removing it, I have made about a 10–15% saving in CPU utilization of these effects. The downside is that when you apply certain procedures such as "interpolation" to the amplitudes, you tend to get a "ringing" effect, which sounds like a comb filter with a fundamental frequency derived from the size of the FFT. To combat this, a simple trick is to randomize the phases of each component, which removes this ringing effect, as the phases no longer coincide at each FFT period. However, this also removes the temporal accuracy of phase components, tending to "smear" the sound, especially for larger FFT sizes., and also widens the stereo field. On the plus side, this can be actually quite sonically desirable for many applications, and so you will often see this option selected by default.

SPECTRAL CLICHÉS

It's very easy with these effects to end up with "spectral soup" (sine tones running up and down the spectrum). While this might be fun, it's also a spectral cliché, and best avoided. The expressive power of these effects is in their more subtle manipulations of timbre. Often the most rewarding effects are found when you move a parameter just slightly away from its "normal" position. Also, on a reasonably grunty machine, you should be able to get a couple of these effects in series for even more possibilities!

4. THE EFFECTS

I have categorized the effects in SoundMagic Spectral below by their typical function or application. I have divided them into the following categories:

- Spectral Smoothing Effects
- Spectral Sustaining Effects
- Spectral Excitement Effects
- Spectral Pitch Effects
- Spectral Filtering Effects
- Spectral Texturizing Effects
- Spectral Partial Effects
- Time Domain Effects

Common Parameters

- **Brightness**
  Applies a filter to either increase or decrease the strength of higher-frequency components. At its most aggressive, the upper frequency components can "sizzle" quite audibly.
- **Lo bin cutoff**
  Frequency components lower than a certain frequency are removed (set to zero amplitude)
- **Hi bin cutoff**
  Frequency components higher than a certain frequency are removed (set to zero)
- **Randomize phases**
  The phase information is randomized (see "Phase Randomization" section above)
- **FFT size**
  The larger the FFT size, the "smoother" the result, but also the more CPU is used in performing the FFT. On G5s, this probably isn't an issue, but on G4s, going too high can cause the audio to break up as the CPU maxes out.
- **Gain**
  The output level of the effect. Some "accumulation" effects can create clipping in the audio — reducing the gain will avoid this.
- **Feedback**
  The amount with which the output of the effect is "fed back" into the input
- **Parameter Variance**
  Some effects allow you to alter a parameter in a programmatic way, through the notion of "variance". This lets you "connect" one parameter to either the amplitude of the sound, the inverse of the amplitude, a Low Frequency Oscillator (LFO) or to a random change.

5. SPECTRAL SMOOTHING EFFECTS

The effects in this category tend to *blur*, *smooth* or otherwise slow down the rate of spectral change. The net result is that one can create strange hybrid sounds, or *drones* that are still based on the sonic characteristics of the original.

SPECTRAL AVERAGING

Averages the spectral information over n windows. Very similar effect to Spectral Blurring.

- **Number of Frames**
  The number of FFT frames over which to average. The more frames the smoother the result.

SPECTRAL BLURRING

Applies a low-pass filter to the spectral changes from one window to the next, progressively "smearing" the sound. Has similar spurious problems as Spectral Averaging, but may still be useful.

- **Blur amount**

  The amount of "smearing" that takes place. This can be varied using the variance controls detailed above

SPECTRAL DRONEMAKER

Spectral DroneMaker is the "king" of spectral drone effects. It can turn any sound into a beautiful, slowly-changing drone based on the sonic characteristics of the underlying sound. The idea came from the software called "Mammut" developed by Øyvind Hammer of Notam, which used very large FFTs to"smear" out the detail, thus creating slowly changing drones. Unlike Mammut, Spectral DroneMaker still uses normal-sized FFTs, instead creating the drone by gradually "interpolating" between frequency components sampled at a distance. Other parameters for making the drone even more sexy include a bank of comb filters added prior to the interpolation for a subtle, or not-so-subtle pitched effect, the ability to gate frequency components as in Spectral Tracing, and the ability to add a flanger to the output.

- **Interpolation length**

  Each bin is interpolated between samples taken at a distance. This parameter sets the amount of time that the bins are sampled at.

- **Interp length variance**

  The above parameter can be randomized within a range of interp length ± variance

- **Use peak amplitudes**

  With this switched on, the process will scan for the highest amplitude that occurs in each bin the input file while they are being interpolated. The process will then use that stored amplitude instead of the actual value that occurs at the next interpolation point in the input file.

- **Gate level**

  Only lets partials above a certain threshold into the output sound.

- **Comb filterbank level**

  Spectral DroneMaker allows you to "harmonize" the input sound by putting it through a comb filterbank. This parameter sets the wet/dry level of the comb filtering.

- **Comb filter fundamental**

  Sets the fundamental frequency of the filterbank. The filters are built up from this frequency.

- **Scale type**

  A list of preset scale types for building the filterbank

- **Number of octaves**

  How many octaves you want the filterbank to cover. The filterbank can be quite computationally expensive with a high number of octaves.

- **Filter resonance**

  The strength of the comb filters. Watch out for distortion from high resonance filters.

- **Flanger amount**

  Lets you apply a time-varying flanger to the output

- **Flanger depth**

  The maximum delay length of the flanger. The higher the number the more "intense" the pitch-varying aspect of the flanger

- **Flanger rate**

  The speed of the flanger.

6. SPECTRAL SUSTAINING EFFECTS

These effects in some way sustain partials for a certain duration, using slightly different algorithms to determine when and for how long partials should be sustained for.

SPECTRAL FREEZING

Sometimes also known as "Spectral Accumulation". Each bin is watched until its amplitude reaches a peak. It is then held at that peak until it is exceeded by another peak in that bin. The Freeze Factor parameter puts a decay onto the peak holding, so that you don't just end up with a whole series of very high-amplitude peaks.

- **Freeze Factor**
  Once a bin is frozen, it can then be "decayed". If this is set to 100%, no decaying takes place, allowing for a complete bin freeze, which will only be superceded by a peak.
- **Use threshold param**
  If this is checked, a freeze will only take place if the new peak is above a threshold level.
- **Threshold**
  The threshold above which a freeze can take place
- **Zero bins under threshold**
  If this is checked, unfrozen bins under the threshold level will be zeroed

SPECTRAL GATE AND HOLD

A similar effect to Spectral Freezing, in which each bin is watched until its amplitude exceeds a threshold level. The main difference between this effect and Spectral Freezing, is that a bin is frozen for a specific length of time, rather than until it is exceeded by a higher amplitude. In Spectral Gate and Hold, once a bin is frozen, it is held for a specific duration (indicated by the Hold duration and Hold variance parameters). During that time there is the possibility of an optional decay (indicated by the Decay factor parameter).

- **Threshold**
  The level above which a bin will be frozen
- **Hold duration**
  The length of time to hold a frozen bin for
- **Hold variance**
  The "hold duration" parameter can be randomized on a per-bin basis within a range of hold duration ± var
- **Decay factor**
  The amount that the frozen bin is decayed by over its lifetime. 100% = no decay, 0% = instantaneous decay
- **Zero partials under threshold**
  Partial falling under the threshold will be zeroed. If this is checked, then nothing of the original sound will come through, except the frozen bins.

7. SPECTRAL EXCITEMENT EFFECTS

SPECTRAL SHIMMER

A variety of ways to add "shimmer", "jitter", "sparkle", "pulsation" or "sizzle" to your sound.

- **Jitter amount**
  The percentage of bins that are "jittered" (a random fluctuation in amplitude)
- **Jitter only above**
  Allows jitter only above a certain frequency
- **Low-frequency jitter**
  The maximum amount of jitter (in dB) at 0Hz
- **High-frequency jitter**

The maximum amount of jitter (in dB) at the Nyquist limit. The amount of jitter is interpolated between this value and the low-frequency jitter over the range.

- **Jitter gate level**

  Only apply jitter if the bin is above a certain amplitude (a low level here can create a real "sizzle" in the higher frequencies)
- **Jitter hold (windows)**

  The number of windows for which a jitter is "held" on a bin — a larger value makes less of a "jitter" and more of a kind of "surging" sound
- **Amplitude pulse type**

  Applies an amplitude pulsation wave across the spectrum. "Square" uses blocks of bins (of size set by the "Pulse Width" parameter), either of max or min pulse level. "Sin" uses a sin wave across the spectrum ranging between max & min levels.
- **Pulse level max**

  The maximum amplification of the pulse wave
- **Pulse level min**

  The minimum amplification of the pulse wave
- **Pulse width**

  The number of bins that a pulsation wave is spread across
- **Pulse rate**

  The rate at which the pulsation wave is cycled up or down across the spectrum
- **Delay variance**

  Delays each bin in a programmatic way, creating sweeps up or down or random delay fluctuations. The Delay Variance parameter sets the maximum delay, while the Delay Type parameter sets the particular method by which bins are delayed.

SPECTRAL SHUFFLE

Randomly shuffles blocks of bins around within the spectrum. You have control over the Shuffle factor (the amount of blocks that are shuffled), the Shuffle range (how far away from the original position each block is moved) and the number of bins per block.

- **Shuffle factor**

  The number of groups that are shuffled, as a percentage of the FFT size.
- **Shuffle range**

  How far away, in bins, each group is moved from its original position.
- **Bins per group**

  The number of bins that are kept together in a block.

SPECTRAL PULSING

A rather experimental effect, that switches bins on and off across the spectrum in a pulsating fashion.

- **On duration**

  The length of time to keep bin zero switched on
- **Off duration**

  The length of time to keep bin zero switched off (after the on duration)
- **On multiplier freq**

  As you go up the frequency spectrum, multiply the on duration by this number
- **Off multiplier freq**

  As you go up the frequency spectrum, multiply the off duration by this number
- **On multiplier time**

  As you move forward in time, multiply the on duration by this number
- **Off multiplier time**

As you move forward in time, multiply the off duration by this number

- **Min on duration**

  Set the minimum length for the on duration
- **Max on duration**

  Set the maximum length for the on duration
- **Min off duration**

  Set the minimum length for the off duration
- **Max off duration**

  Set the maximum length for the off duration
- **Modulo type**

  1= ping-pong between min and max; 2 = wrap round from max to min/min to max

8. SPECTRAL PITCH EFFECTS

SPECTRAL HARMONIZER

The spectrum is transposed up or down by a set series of intervals (you have three to play with) and mixed back into the original. Each interval can then be transposed in a similar manner to the Spectral Filterbank transpositions.

- **Intervals**

  There are three "intervals" that you can define to harmonize the original sound file. If, for instance, you define interval 1 to be 400 cents, this will harmonize the sound file up a major fourth.
- **Transposition Type**

  If the "no. partials per interval" setting is higher than 1, you can define how the interval will be used to determine the other harmonizations. See "Spectral Filterbank" for more information on these transpositions
- **Transpositions per interval**

  For each interval, how many transpositions there are. For instance, 4 transpositions of a minor third, with the transposition type set to "stacked" would give you a diminished seventh chord.

SPECTRAL BIN SHIFT

Shifts the entire spectrum up or down a set number of bins. Because the bins of the FFT are equally-spaced, not logarithmically spaced, the spectrum becomes rather compressed or expanded the further away you get, and as a result, strange, inharmonic spectra result which sound quite distant from the original. Can be useful for then feeding into another effect (e.g. spectral gran, or time stretching)

- **Bins to shift**

  The number of bins to offset the entire spectrum
- **Rotate**

  Should bins that "fall off the end" be rotated around to the other side?
- **Instantaneous feedback**

  The bin-shifting is fed back within the same frame (as opposed to normal feedback, in which the output is fed back into the following input frame)
- **Num instant feedback**

  How many times the instantaneous feedback algorithm should be calculated

SPECTRAL PITCH SHIFT

A simple spectral pitch shift up or down by a set amount, using the method suggested by Mark Dolson and Jean Laroche in their paper "New Phase-Vocoder Techniques for Real-Time Pitch-Shifting, Chorusing, Harmonizing and Other Exotic Audio Modifications", in which spectral peaks are tracked and shifted independently of one another. You can optionally use a "Bins to Shift" parameter to shift all the spectral peaks up in an inharmonic manner.

- **Pitch Shift**

  The amount to shift
- **Use 'Bins to Shift' parameter**

  Instead of using an interval, shift by a set number of bins (this is different from the spectral bin shift, in that it shifts individual peaks by a set number of bins, rather than the entire spectrum)
- **Bins to shift**

  The number of bins to shift, if the above parameter is checked
- **Rotate**

  Should bins that "fall off the end" be rotated around to the other end?

SPECTRAL STRETCH

An effect to create a variety of inharmonic stretches. Spectral peaks are tracked and shifted by the quadratic equation: $\beta\omega+\alpha\omega^2$, where ω is the frequency of each bin.

- **Alpha stretch**

  The value for α in the equation above
- **Beta stretch**

  The value for β in the equation above, which can be varied using the "beta stretch variance" parameter

9. SPECTRAL FILTERING EFFECTS

SPECTRAL TRACING

Typically used to retain only the n loudest bins in a spectrum and remove the others. Alternatively you can retain the n softest bins, or set a threshold level (a "gate") below which all partials are zeroed, or above which all partials are zeroed.

- **Tracing type**

  Choose from four different algorithms: retain the n loudest, the n softest, bins above the threshold or bins below the threshold
- **Number of bins**

  The number of bins to retain (either the loudest or the softest depending on the Tracing Type parameter)
- **Threshold**

  The threshold in dB; retains bins above or below this threshold depending on the Tracing Type parameter

SPECTRAL FILTERBANK

A bank of extremely narrow, very pure bandpass filters (similar to the GRM Reson plug-in). You can choose the interval between centre frequencies of your filterbank, and how many filters should be created. Because of the inaccuracies in pitch in the FFT, the filterbank cannot guarantee the actual frequencies you ask for will be exactly rendered; however, it will be as accurate as possible within the limitations of the FFT.

- **Number of filters**

  The number of filters in the filterbank
- **Base frequency**

  The lowest frequency in the filterbank, from which the other frequencies are built-up
- **Interval**

  The interval that will be applied to the base frequency and successive "harmonics" to build up the filterbank, using the techniques described in the "transposition type" parameter

- **Transposition type**

  —Stacked: each interval is simply applied to the previous interval, "stacking" the interval on top of each other. For an interval of 300 cents (a minor third), this would create a diminished seventh chord, for instance. An interval of 200 cents would create a whole-tone scale.

  —Chord: the "fundamental" and the upper note defined by the interval are simply repeated at the octave. If the fundamental was C, and the interval was a perfect fifth (700 cents), then the notes are just C and G repeated up through the octaves

  —Harmonics: a harmonic series is based on integer multiples of a fundamental (n, 2n, 3n, 4n, etc). The harmonics setting extends this by using multiples of other intervals. It assumes that the standard harmonic series would have an interval setting of 1200 cents (an octave) — therefore, halving this to 600 cents (a tritone) would give you filters at 0.5n, n, 1.5n, 2n, etc..., while 2400 cents (two octaves) would give you 2n, 4n, 6n, 8n, etc.
- **Harmonic gain**

  "Harmonic gain" indicates how much each progressive "harmonic" as you get higher should be attenuated by.
- **Side bins**

  Allows you to create a "wider" filter by adding side bins on either side of the central filter bin. While this lets more of the original sound through, and therefore less pure filtering, it also creates some interesting "pulsating" dynamic effects
- **Side bin gain**

  How much each side bin is attenuated by as you get further away from the central filter

SPECTRAL GLIDING FILTERS

Creates a series of spectral filters that glide in pitch

- **Number of filters**

  How many filters are to be created
- **Filter width**

  The size of the filter in bins
- **Start frequency**

  The frequency at which the filter should start its life
- **Start frequency variation**

  Randomization of the filter Start Frequency parameter
- **Pitch glide per second**

  How much drift in pitch each filter should have (per second)
- **Pitch glide var**

  Randomization of the Pitch Glide parameter
- **Filter duration**

  The lifetime of each filter
- **Filter duration var**

  Randomization of the Filter Duration parameter
- **Silence duration**

  The duration of silence before a new filter is created
- **Silence duration var**

  Randomization of the Silence Duration parameter

10. SPECTRAL TEXTURIZING EFFECTS

SPECTRAL GRANULATION

“Grains” or “chunks” of the spectrum, of a certain frequency range and duration are taken and delayed by a certain length. Optionally, you can also apply a bin shift and a fade in/fade out to the grains.

- **Grain length**
  The length of each grain. NB: the actual length is rounded down to the nearest FFT length
- **Grain length variance**
  Randomization of the grain length (l ± var)
- **Grain delay**
  The length of time to delay each group
- **Grain delay variance**
  Randomization of the grain delay
- **Bin shift**
  An optional amount to shift each grain up or down the spectrum by a number of bins
- **Bin shift variance**
  Randomization of the bin shift parameter
- **Bins per grain**
  How many bins constitute a grain. The larger this parameter the more of the original sound can be detected in the output.
- **Fade amount**
  Fades in and out each grain for extra smoothness. 100% is maximum fading.
- **Density**
  The percentage of grains that are actually delayed. 100% means the entire sound is granulation. 0% means none of the sound makes it out.

11. SPECTRAL PARTIAL EFFECTS

These effects track peaks in the spectral envelope that appear to have a certain lifetime, and creates “regions of influence” between the tracked peaks. These are regions are estimates of the stable “partials” of the sound, and so these effects work best on sounds that have a stable pitched morphology. The **fuzziness** parameter allows a peak to drift in pitch within a certain range of bins and still be tracked as a stable partial. If this parameter is set too small, then a pitch drift outside that boundary will end the lifetime of that partial, and it will be tracked as a new partial. On the other hand, a setting too large might cause new partials to be tracked as a drift in an existing partials. The **max bins per partial** allows you to reduce a “region of influence” down to a smaller number of bins centred around the peak bin, creating “purer” partials, a little bit like Spectral Tracing.

SPECTRAL EMERGENCE

This effect tracks the partials, but applies an amplitude envelope to them to create a strange pulsating sound. Because this happens separately for left and right channels, you often get a “rotating” panning effect as well.

- **Fade in (windows)**
  The number of windows over which to fade in a tracked partial
- **Hold (windows)**
  Once the tracked partial has been faded in, how long should it be held at 0dB for?
- **Fade out (windows)**
  Once the tracked partial has been faded in and held, how long it should be faded out for?
- **Recycle faded peaks**

If this is checked, then if a partial is still being tracked after it has finished all three phases of its envelope, it will be restarted from the fade in section. If unchecked, it will remain silence until the end of its life.

SPECTRAL PARTIAL GLIDE

This effect tracks partials, but shifts their pitches at a set amount. However, this amount can vary for each partial, to the point where you can get partials gliding in opposite directions!

- **Pitch shift per second**
  The central pitch shift, in cents, per second
- **Pitch shift variance**
  Randomization of the Pitch Shift per Second parameter, on a per partial basis
- **Mirror bin**
  A somewhat experimental parameter: bins below this number will glide in the opposite direction from bins above this number
- **Fade in (windows)**
  Apply a gradual fade in to tracked partials
- **Zero transients**
  Partials that only exist for a single window should be removed.

12. TIME DOMAIN EFFECTS

Chorus

A time-varying chorus/flanger effect

Comb Filterbank

A bank of comb filters

Grain Streamer

"Captures" and repeats grains

Mr. Filterbank

A standard time-domain implementation of a bank of bandpass filters