propellerhead

PARSEC 2 SPECTRAL SYNTHESIZER

OPERATION MANUAL

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Parsec 2
Spectral Synthesizer

Introduction

The Parsec 2 Spectral Synthesizer Rack Extension is a very advanced additive synthesizer which offers a sonic palette far beyond the ordinary. Despite its vast sonic capabilities, Parsec 2 has a simple and straight-forward user interface, designed for experimentation. You are guaranteed to come up with interesting timbres with only a few knob twists!

Each of the two sound engines in Parsec 2 can use up to 512 sine wave oscillators. This allows for extremely rich and pristine sound. Thanks to clever programming Parsec 2 is still very DSP friendly, without sacrificing audio quality.

A large number of filter and modifier algorithms make it possible to modulate and control the sine wave oscillators in very interesting ways. The extensive Modulation Bus section allows for detailed and flexible modulation and control. Parsec 2 also features global stereo Delay and Reverb effects to spice up the sound even more.

A few words about additive synthesis

Parsec 2 uses additive synthesis to generate sounds. Additive synthesis is based on a large number of sine wave oscillators that can be introduced in the sound - at various times, levels and durations. The big difference, compared to subtractive synthesis, is that overtones are added to a basic sine wave signal to form complex signals - instead of subtracted by filters from complex signals. In practice this means that you can tailor-make the frequency content in your sounds a lot more precisely with additive synthesis than with subtractive synthesis.

The sonic results of additive synthesis can vary dramatically; from standard “analog” type of synth sounds, via emulations of existing instruments, to extremely complex and animated timbres.

- **Check out these Parsec video tutorials for more info: Parsec Additive Synthesis and Additive sound design with Parsec.**

What's new in Parsec 2?

Here's a list of what's new in Parsec 2 compared to the original Parsec:

- **New graphical panel layout (see "Panel overview").**
- **Link Engines mode.**
  This allows you to route the signal from Generator A through all four Modifiers in series (see "Signal flow with the "Link Engines" function").
- **36 new Generator signals/waveforms.**
  These include a number of carefully resynthesized and modeled instruments, plus 31 wave tables (see "Waveform/ signal selector").
- **On/Off buttons for the Modifiers have been added (see "Modifier 1 and 2").**
- **Interactive Modifier displays.**
  Now you can edit (and draw) directly in the interactive Modifier displays (see "Modifier 1 and 2").
- **Five new Modifier Types (algorithms).**
  Harmonic Stretch, Pitch Warp, Filter Curve, Pitch Curve and Partial Envelopes have been added (see "Modifier 1 and 2").
- **"Drag & Drop" functionality for assigning Destinations in the Modulation Bus (see "The Modulation Bus section").**
- **Six new Sources in the Modulation Bus (see "The Modulation Bus section").**
- **Three new Destinations in the Modulation Bus (see "The Modulation Bus section").**
- **The Pan and Spread functions now work "per voice" instead of globally (see "Pan" and "Spread").**
- **Three different Spread modes (see "Spread").**
- **One new LFO waveform has been added (see "LFO 1" and "LFO 2 Global").**
- **A lot of great new and inspiring patches have been added!**
  The Parsec 2 patches were created by:

Kristoffer Wallman (effots)	Gordon Reid
Sami Rabia (Aiyn Zahev)	Ian Duncan
Daniel Thiel (eXode)	Chris Griffin
Jan Klimaschewski (Speo)	Kurt Kurasaki
Adrian Dybowski (Navi Retlav)	Leo Nathorst-Böös
Adam Fielding (AdFielding)	Loui Westin
Terence Skerritt (Arsenic)	Simon Sherbourne
Sonny Orhan (Black Reign)	Tom Pritchard
Brian Findlay (EpiGenetic)	Bryan Shields (008)
Richard Hider (TonalAxis)	Adam McClure (8piscean8)
Nicolas Delmotte (Odarmonix)	Bernardo Cubelos (BAC)
Clint Grierson (Point Zero Production)	Gregory Coulet (Deeplift)
Michel Desangles (Wong the Sane)	Marco Correia (Koshdukai)
Michael Gorman (Lizard)	Terry Grame (tgrame)
Paul Rostill (Luapy)	Peter Jubel
Nils-Erik Johansson (Nisse)	Ludvig Carlson
Jon Bouy (Ostermilk)	Mattias Häggström Gerdt
Giles Reeves (Selig)	Mats Karlöf

Panel overview

The Parsec 2 front panel contains the following sections:

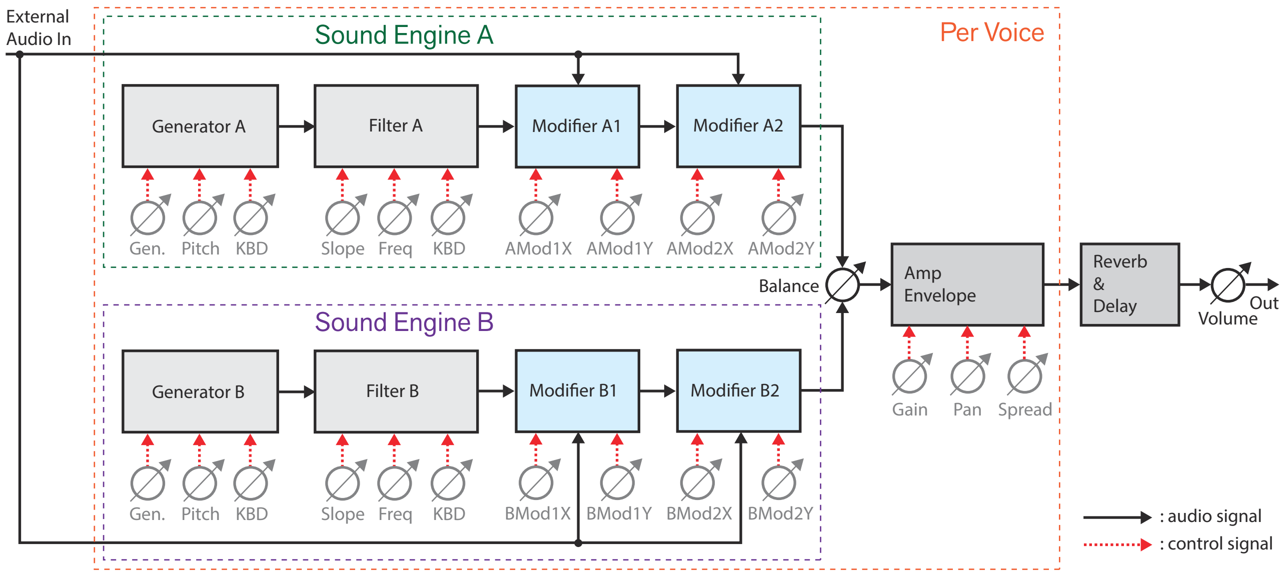
The Parsec 2 front panel sections.

- **1. Patch Selector (for browsing, loading and saving patches).**
- **2. MIDI Note On LED.**
- **3. Sound Engine A (Generator A, Filter A and Modifier A sections).**
- **4. Sound Engine A/B Balance, Link Engines, Gain and Pan controls.**
- **5. Sound Engine B (Generator B, Filter B and Modifier B sections).**
- **6. Modulation controls (common for both Sound Engines).**
- **7. Stereo Spread controls.**
- **8. Global performance and “play” controls.**
- **9. Modulation Bus section.**
- **10. Reverb and Delay sections.**

Signal flow

Default signal flow (non-linked Sound Engines)

The picture below shows the default signal flow in Parsec 2:



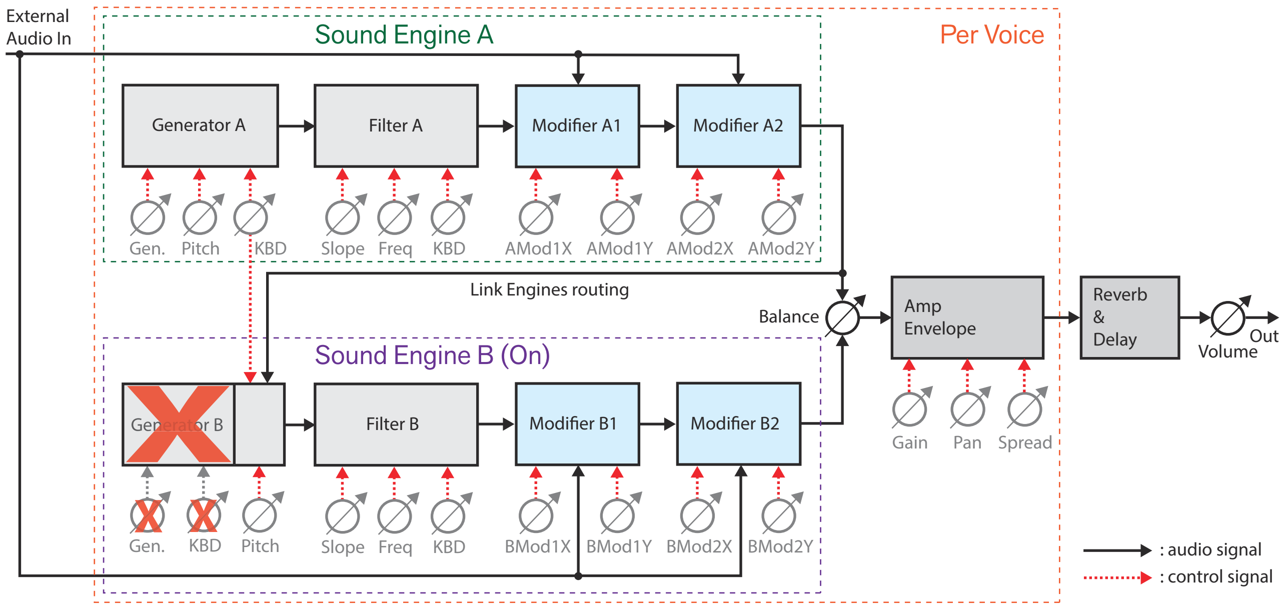
Parsec 2 default signal flowchart.

- **The "hearts" of Parsec 2 are the two Sound Engines A and B.**
- **First in each Sound Engine is a Generator which generates the audio signal.**
  The Generator could be compared to an oscillator in an "ordinary" synth, but is much more powerful and flexible in Parsec 2. Each Generator can generate up to 512 sine wave signals.
- **The signal from the Generator is routed to the Filter.**
  The Filter is a combined lowpass/high shelving filter with adjustable slope and cutoff frequency. The Filter can be used for attenuating or boosting the higher frequencies of the Generator signal.
- **The signal from the Filter is routed to Modifier 1.**
  The Modifier affects the signal according to the algorithm you have selected. The algorithms could be various types of filters - or special purpose modifiers that affect the partials in the signal in various ways.
- **The signal from Modifier 1 is routed to Modifier 2.**
  Modifier 2 features the same array of selectable algorithms as Modifier 1.
- **The signals from the two Sound Engines are routed to the Balance "mixer" where you can set the mix (crossfade) between the two Sound Engine output signals.**
- **The mixed signal is routed to the Amp Envelope and then, via the Reverb and Delay sections, to the stereo outputs.**
- **The remaining sections in Parsec 2 (Envelope 1, Envelope 2, LFO1 and LFO2) can be freely assigned to modulate destination parameters via the Modulation Bus section, see "The Modulation Bus section".**

Signal flow with the "Link Engines" function

In Parsec 2 it's also possible to route the signal of Generator A to the Modifiers of Sound Engines A and B and thus be able to modify the signal with up to four Modifiers in series. The picture below shows the signal flow when Sound Engine A is linked to Sound Engine B using the "Link Engines" function:

The "Link Engines" button.



Parsec 2 signal flowchart in "Link Engines" mode.

There are a couple of important differences when running in Link Engines mode compared to the default (non-linked) mode:

- **In "Link Engines" mode you choose the waveform/signal in Generator A only - this is the only active signal generator in this configuration. Signal generation from Generator B is automatically disabled.**
  However, you can change the pitch of a "copy" of Generator A's signal with the Pitch controls of Generator B, to generate a detuned/pitched "clone" of Generator A's signal. This is great for "fattening up" the sound - or for detuning in desired musical intervals.
- **The KBD (Keyboard Tracking) of the Generator B pitch is automatically disabled in "Link Engines" mode.**
  Instead, the Generator B pitch automatically follows the KBD setting of Generator A. This is to maintain any relative detuning between the Generator A and B pitches, regardless of the notes you play.

Using Parsec 2

Loading and saving patches

Loading and saving patches is done in the same way as with any other internal Reason/Reason Essentials device. See the "Sounds and Patches" chapter in the Reason/Reason Essentials Operation Manual pdf for details.

Global performance and "play" controls

P. Range

→ **Set the desired Pitch Bend range for the "Pitch" wheel with the up/down buttons, or by click-holding on the display and dragging up/down.**
Range: +/-24 semitones (+/-2 octaves) in steps of +/-1 semitone.

Pitch

The Pitch bend wheel can be used for bending note pitches up and down. Parsec 2 also responds to Pitch Bend MIDI data from a connected MIDI master keyboard. You set the desired Pitch bend Range with the "P. Range" control above the Pitch bend wheel.

Mod

The Mod wheel can be used for controlling almost any parameter in Parsec 2. Use the Mod wheel as a Source parameter in the Modulation Bus section and then route to the desired Destination parameter(s), see "The Modulation Bus section".

Portamento

Portamento makes note pitches glide from previous notes to new ones, at the time set with the Time knob.

Portamento can be used in all Keyboard modes (see below).

- **When On in Polyphonic Keyboard Mode (see below), the pitches will glide from any of the available voices.**
The results will be unpredictable since there is no way of controlling from which note(s) the glide(s) will commence. The effect is very nice, though.
- **When On in Mono Retrig or Mono Legato Keyboard Mode (see below), the pitch will glide between consecutive notes.**
- **In Auto mode, the pitch will glide between consecutive monophonic notes only when you play legato. If you have selected Polyphonic Keyboard Mode (see below), Auto will have no effect at all.**
If you release the previous key before hitting the new key, there will be no portamento effect.

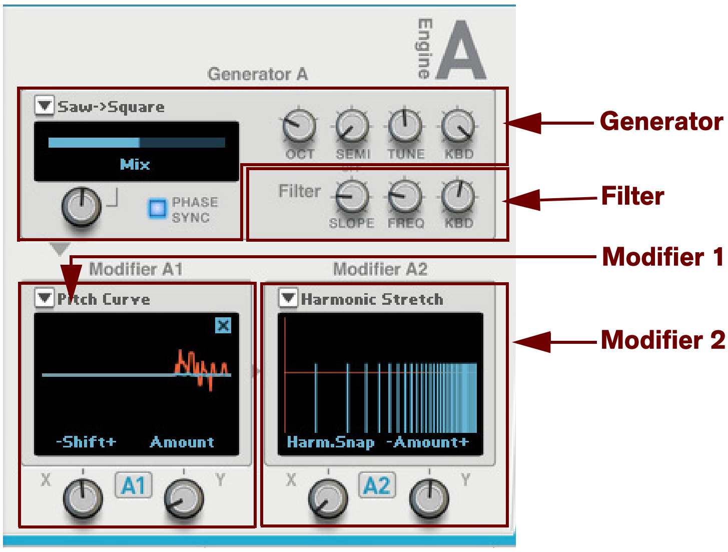
Keyboard Mode

Here you choose how Parsec 2 should respond to MIDI Note data:

- **Poly**
  Select this if you want to play Parsec 2 polyphonically. The maximum number of voices is 32 but depends on the selected Generator signal(s) and Modifier algorithm(s).
- **Retrig**
  Select this if you want to play Parsec 2 in monophonic mode and always retrigger the envelopes as soon as you play a new note.
- **Legato**
  The Mono Legato mode is also monophonic. However, if you play a new note without having released the previous one, the envelopes won't start over.

Panel controls

Sound Engines A and B



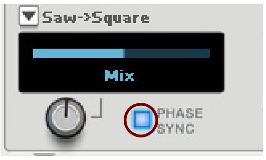
Parsec 2 features two Sound Engines. These could be considered very advanced and flexible sine wave generators. Each Sound Engine can generate up to 512 sine wave signals. The sine wave signals of each Generator can be modified via two Modifier sections, Mod 1 and Mod 2, and their dynamic X and Y controls.

Except for the On/Off button in Sound Engine B, Sound Engines A and B feature identical parameters and controls:

Sound Engine B On/Off

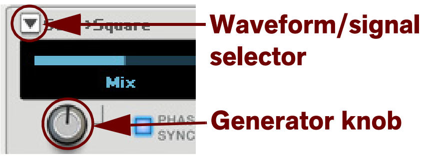
→ **Click to switch Sound Engine B on/off.**

Phase Sync



→ **Click to sync the start phases of all sine waves in the Generator.**
When active, the sound character will be the same each time you play the same note. When inactive, the sound character will vary more or less each time you play the same note.

Waveform/signal selector



→ **Select the desired waveform/signal by clicking the selection button to the left above the display, and selecting from the list that appears.**
Here you select the desired overtone content for the Generator in the Sound Engine. For example, if you want the overtone content of a square wave, select “Saw -> Square” and turn the Generator knob all the way up. If you want the overtone content of a sawtooth wave, select “Saw -> Square” or “Dual Saw” and turn the Generator knob to zero. There are also a number of special waveforms, as well as waveforms that are only possible to generate using additive synthesis.

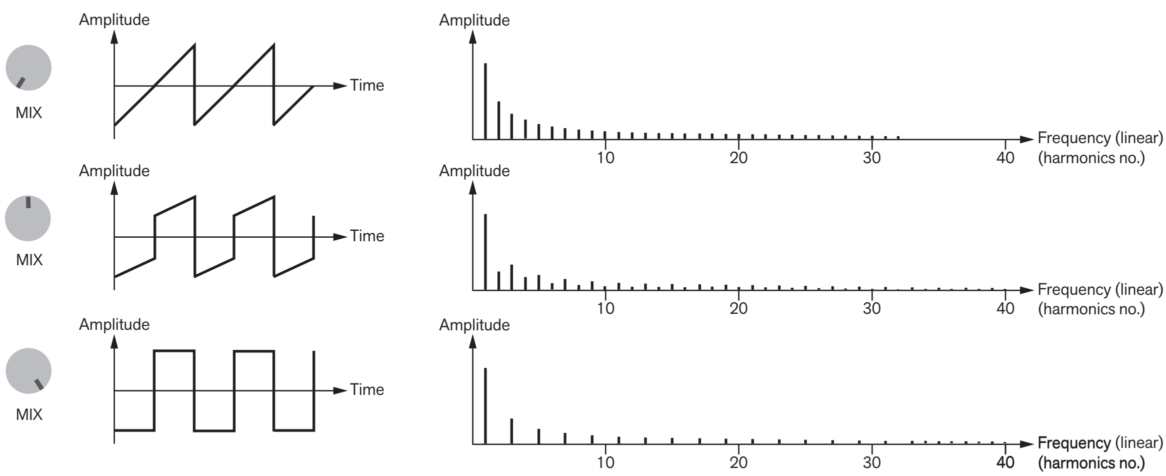
! **Waveform/signal selection is disabled for Generator B, if Link Engine is active, see “Link Engines”.**

The Generator knob below the waveform selector has different functionality depending on which waveform/signal you have selected. The actual knob functionality is indicated by the name in the display above the knob.

→ **Turn the Generator knob - or click/drag in the display - to change the waveform/signal character.**

The following waveforms/signals are available:

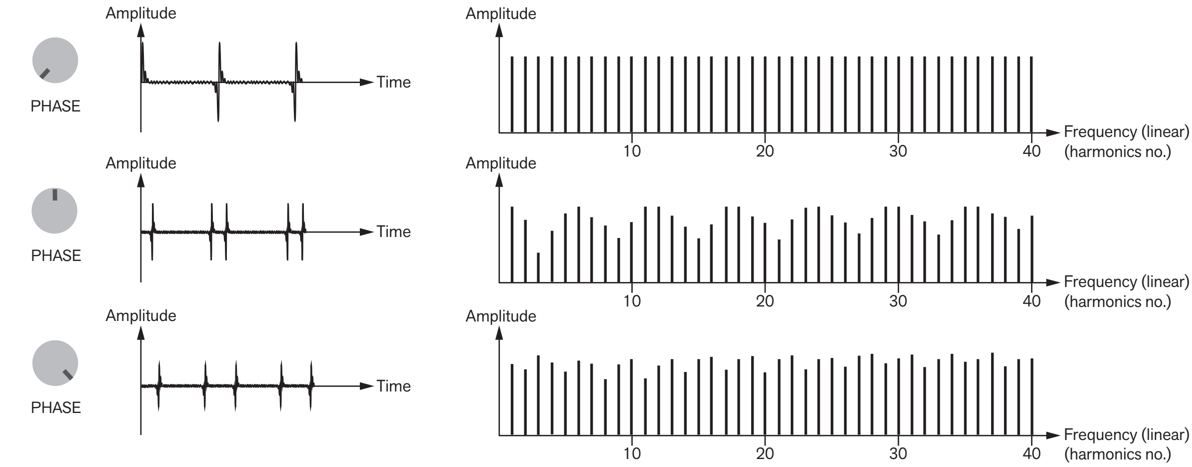
- **Saw -> Square**
  This is a sawtooth wave which can be continuously transformed into a square wave.



Examples of wave shapes and spectra of the Saw to Square signal when Mix=0% (top), 50% (middle) and 100% (bottom).

→ **Control the waveform shape with the Generator (Mix) knob.**
Range: Sawtooth wave to Square wave (50% pulse width).

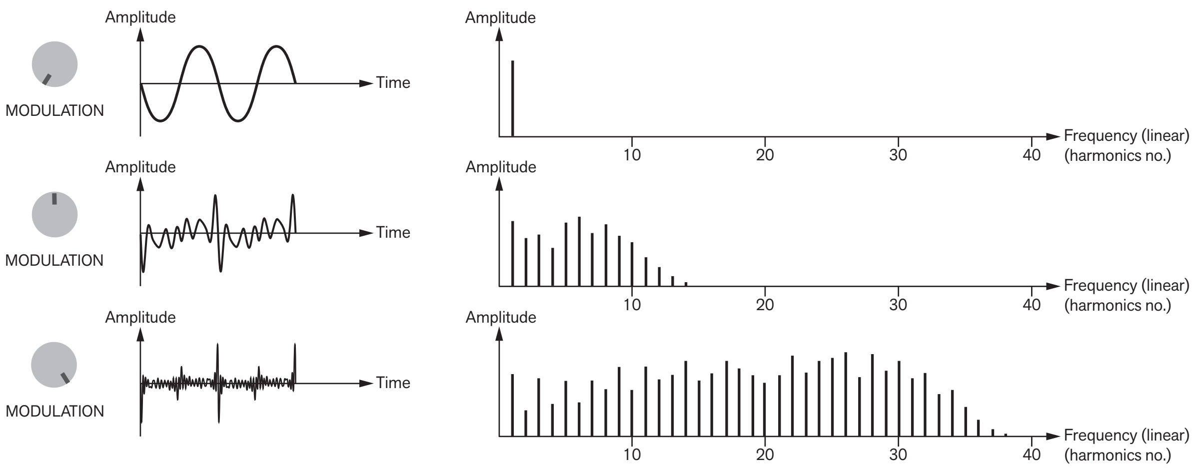
- **Pulse**
  This signal consists of repetitions of two short pulses and contains all harmonic overtones. The phase between the two pulses can be adjusted.



Examples of wave shapes and spectra of the Pulse signal when Phase=0% (top), 50% (middle) and 100% (bottom).

→ **Control the phase between the two pulses, and thus the pulse spectrum shape, with the Generator (Phase) knob.**
When the Phase is 0% the two pulses are combined into a single pulse.

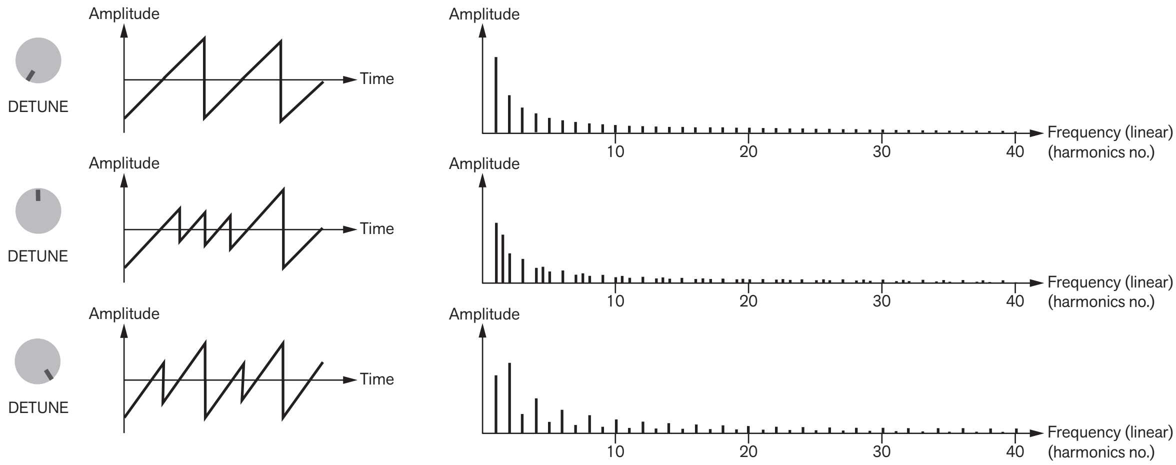
- **FM**
  This signal simulates a sine wave signal, frequency-modulated by another sine wave signal at a fixed 1:1 frequency ratio. The frequency modulation amount can be adjusted.



Examples of wave shapes and spectra of the FM signal when Modulation=0% (top), 50% (middle) and 100% (bottom).

→ **Set the frequency modulation amount with the Generator (Modulation) knob.**
Range: from a pure sine wave signal to a heavily frequency modulated signal.

- **Dual Saw**
  This signal simulates two combined sawtooth waves where one of the sawtooth waves can be detuned.

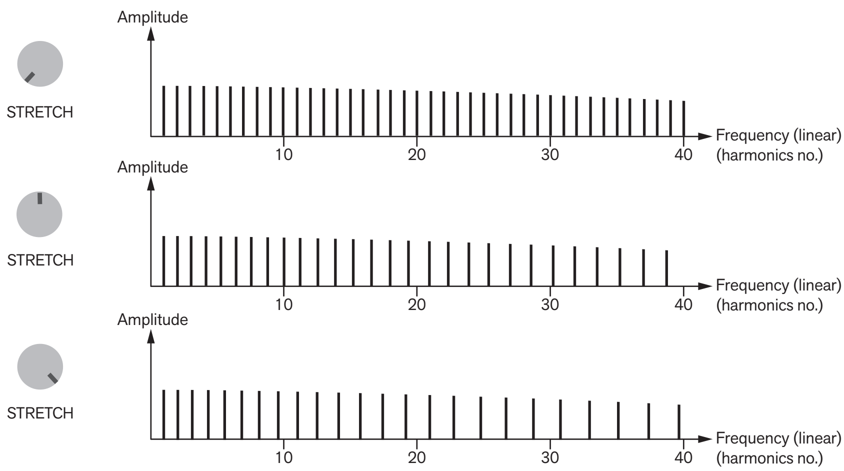


Examples of wave shapes and spectra of the Dual Saw signal when Detune=0% (top), 50% (middle) and 100% (bottom).

→ **Control the pitch detuning of the second sawtooth wave with the Generator (Detune) knob.**
Range: single sawtooth wave to dual sawtooth waves one octave apart.

► **Use the Spread control to separate the two sawtooth waves left and right in the stereo panorama, see “Spread”.**

- **String**
  This signal simulates the stretched frequency spectrum of a stiff vibrating string.

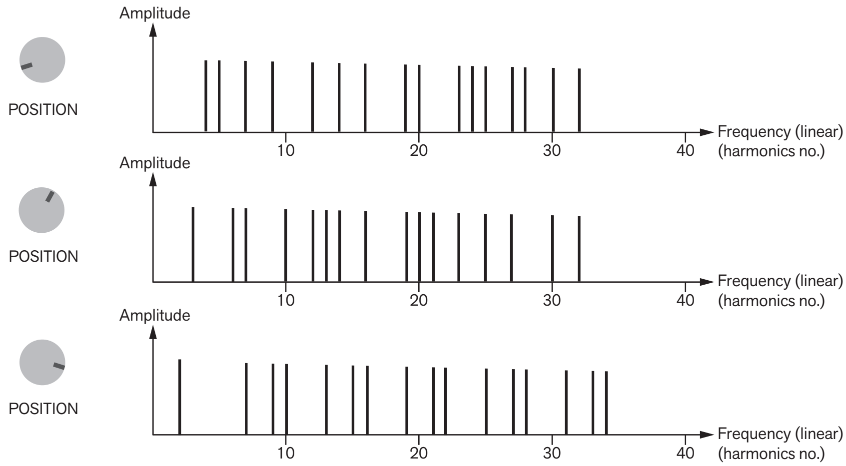


The spectra of the String signal when Stretch=0% (top), 50% (middle) and 100% (bottom).

→ **Set the sound character with the Generator (Stretch) knob.**
Range: harmonic overtone spectrum to an inharmonic spectrum.

As you turn the Generator knob clockwise from 0%, the partials above the fundamental are moved upwards in the frequency spectrum. With the knob up to around 50% this is perceived as a detuning. At around 50% and above, the signal gets more and more inharmonic.

- **Sparse**
  This signal consists of eight different harmonic overtone series, each containing 16 partials. The different harmonic overtone series can be accessed by turning the Generator knob. The transitions between the overtone series are smooth. The figure below shows three examples of how the overtone series could look in the frequency spectrum.



The spectra of the Sparse signal at three different Generator knob (Position) settings.

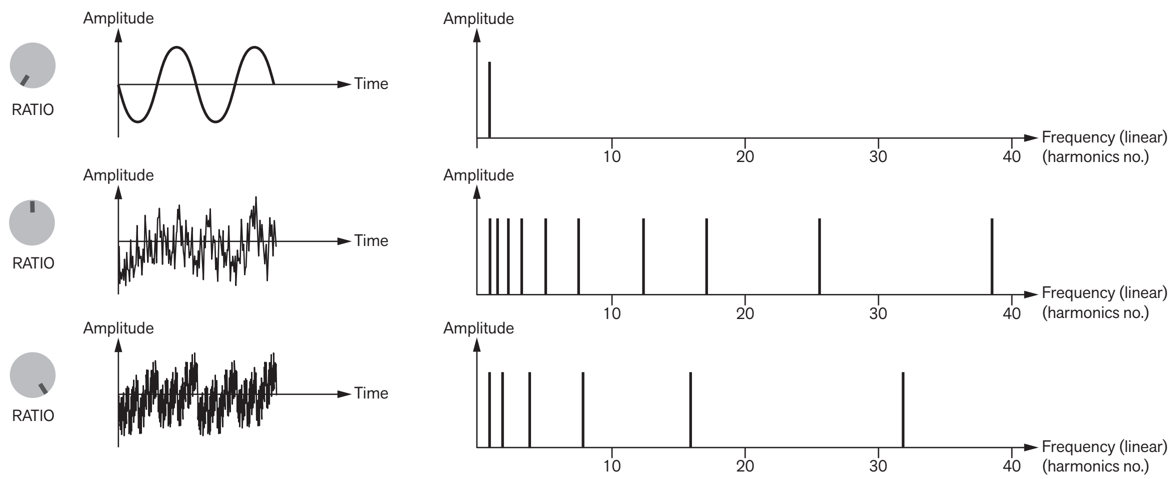
→ **Select the desired harmonic overtone series with the Generator (Position) knob.**

- **Sparse Inharmonic**
  This signal consists of eight different inharmonic overtone series, each containing 16 partials. The different inharmonic overtone series can be accessed by turning the Generator (Position) knob. The transitions between the overtone series are continuous, i.e. the partials are individually transposed from one overtone series to another.

→ **Select the desired inharmonic overtone series with the Generator (Position) knob.**

- **Ratio**
  This signal consists of several hundreds of sine wave signals (partials) that can be placed with a "spacing" between each other in the frequency spectrum, at a set ratio. The range spans from a single sine wave (at Ratio 1:1), via dissonant sounds, up to a harmonic sound (at Ratio 2:1).

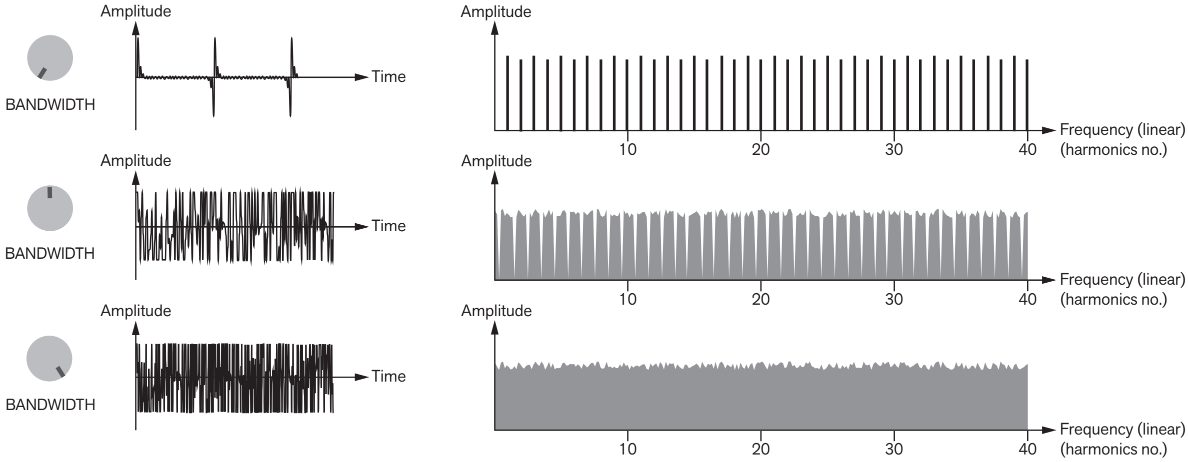
! **Deactivate Phase Sync (if it is on) to avoid loud signals when Ratio=0%, see "Phase Sync".**



Examples of wave shapes and spectra of the Ratio signal when Ratio=0% (top), 50% (middle) and 100% (bottom).

→ **Set the frequency ratio with the Generator (Ratio) knob.**
Range: from a pure sine wave (Ratio=0% (1:1)), via dissonant sounds, to a wave that only contains partials at octave intervals (Ratio=100% (2:1)).

- **Noise**
  This signal produces a range of noises, from tonal noise up to white noise, by amplitude modulating the partials with noise.



Examples of wave shapes and spectra of the Noise signal when Bandwidth=0% (top), 50% (middle) and 100% (bottom).

→ **Set the noise color with the Generator (Bandwidth) knob.**
With the Generator knob at 0%, the wave is a pulse wave with slightly noise-modulated partials. As you turn the Generator knob upwards the noise amplitude modulation is increased and at 100% the signal is white noise.

- **FM Noise**
  This signal produces a range of noises, from tonal noise up to white noise, by frequency modulating the partials with noise. This creates a little "sharper" character than the Noise described above since the lower partials are less frequency modulated than the higher partials.

→ **Set the noise color with the Generator (Modulation) knob.**
With the Generator knob at 0%, the wave is a pulse wave with the partials only slightly frequency modulated by noise. As you turn the Generator knob upwards the frequency modulation of the partials is increased and at 100% the signal is close to white noise.

! **The results when treating the FM Noise signal by Modifiers that affect the partials in the sound (e.g. the brick wall filters) might be a little unexpected at high Modulation amounts. This is due to the nature of the FM Noise signal.**

- **Perc 1 and 2**
  The two types of Perc(ussion) signals produce variable dissonant frequency spectra that are suitable for creating electronic and metallic drums and percussive sounds. The Perc 1 signal contains a little less overtones than the Perc 2 signal.

→ **Control the overtone spectrum with the Generator (Tune) knob.**
The Tune parameter changes the individual tunings of the partials in the signal.

- **Resynthesis: Grand Piano**
  This is a completely resynthesized grand piano, built on analyzed multi-samples. A special thing with the resynthesized Grand Piano samples is that they contain partials below the fundamental partial. This means that if you use any of the Modifiers that can remove partials down to the fundamental partial, the partials below the fundamental will still be audible.

→ **Control the character with the Generator (Character) knob.**
Range: "subdued" to "agitated", with natural sound at the 12 o'clock position (50%).

Subdued (0-50%) produces a warm and mellow tone whereas agitated (50-100%) generates a brighter and significantly more pronounced tone.

! **Changing the Character only has effect at the next Note On(s) - not on any held/sustaining notes.**

► **For more natural-sounding attacks, make sure Phase Sync is Off, see "Phase Sync".**

- **Resynthesis: Clav**
  This is a completely resynthesized Clavinet, with multiple velocity layers and key zones. As with the Grand Piano, the analyzed samples contain partials below the fundamental partial. This means that if you use any of the Modifiers that can remove partials down to the fundamental partial, the partials below the fundamental will still be audible.

→ **Control the playback speed with the Generator (Speed) knob.**
Range: Slow to fast, with natural sound at the 12 o'clock position (50%).

► **For authentic attacks, make sure Phase Sync is On, see "Phase Sync".**

- **Resynthesis: Guitar**
  This is a completely resynthesized steel-string guitar, with multiple velocity layers and zones. As with the Grand Piano, the analyzed samples contain partials below the fundamental partial. This means that if you use any of the Modifiers that can remove partials down to the fundamental partial, the partials below the fundamental will still be audible.

→ **Control the playback speed with the Generator (Speed) knob.**
Range: Slow to fast, with natural sound at the 12 o'clock position (50%).

► **Change the attack character with the Phase Sync button, see "Phase Sync".**

- **Resynthesis: Horn**
  This is a completely resynthesized horn, with multiple velocity layers and zones.

→ **Control the playback speed with the Generator (Speed) knob.**
Range: Slow to fast, with natural sound at the 12 o'clock position (50%).

► **For more natural-sounding attacks, make sure Phase Sync is Off, see "Phase Sync".**

- **Electromechanical**
  This is a simulation of an electric piano.

→ **Control the overtone amount with the Generator (Drive) knob.**
Range: soft/mellow to agitated, with natural sound at the 12 o'clock position (50%).

- **Wavetables**
  The 31 different wave table signals are taken from existing Reason instruments. The crossfades between the waveforms in each wave table are smooth.

→ **Control the position in the selected wave table with the Generator (Position) knob.**

► **Experiment with modulating the Generator (Position) knob from sources like Envelopes and/or LFOs in Parsec 2, for really interesting effects, see "The Modulation Bus section".**

► **Note that the Phase Sync function can have a big impact on the sound character, so feel free to experiment with different settings, see "Phase Sync".**

Pitch controls

- **OCT**

→ **Turn the OCT knob to change the Generator pitch in octave steps.**
Range: 5 octaves.

- **SEMI**

→ **Turn the SEMI knob to change the Generator pitch in semitone steps.**
Range: 12 semitones (one octave).

- **TUNE**

→ **Turn the TUNE knob to change the Generator pitch in steps of 1 cent.**
Range: +/- 50 cents (down or up half a semitone).

- **KBD**

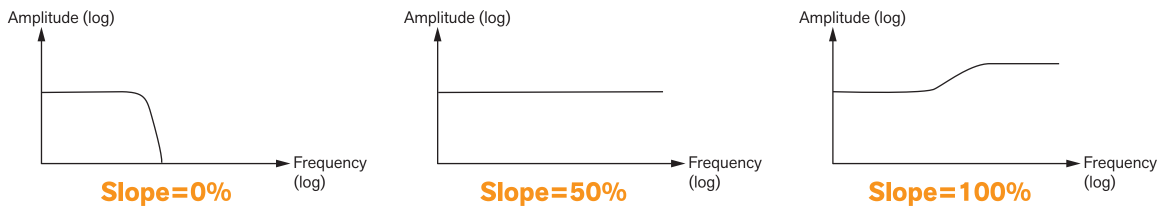
→ **Turn the KBD (Keyboard Track) knob to set how much the Generator pitch should track incoming MIDI Notes.**
Range: 0% (no tracking (constant pitch)) to 100% (1 semitone per key).

! **Keyboard Tracking is automatically disabled for Generator B, if Link Engine is active, see "Link Engines".**

Filter

Each Sound Engine has one built-in combined "lowpass/high shelving" filter per voice, which can be used for shaping the Generator signals before being modified further by the Modifiers (see "Modifier 1 and 2"). The filter features the following parameters:

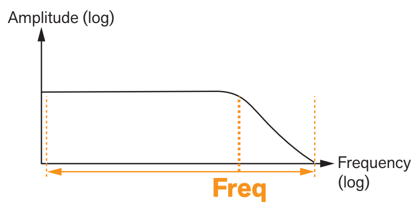
- **SLOPE**



The filter attenuation/gain curves at three different Slope settings (with the same Freq setting)

→ **Set the attenuation/gain slope with the SLOPE knob.**
Range: >100 dB/octave attenuation to +12 dB Hi Shelving.
At 50% the attenuation/gain curve is straight and does not affect the signal at all.

- **FREQ**



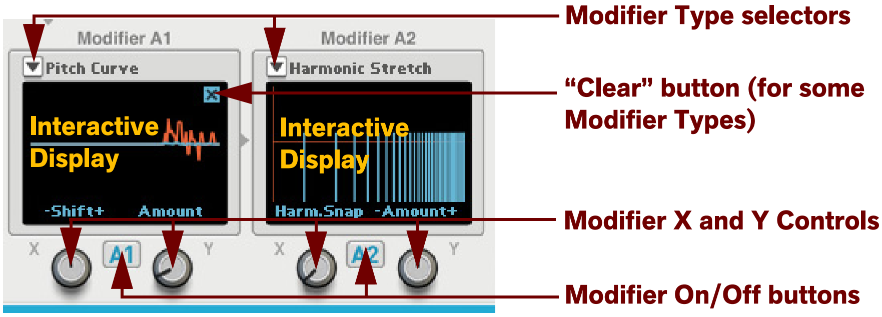
→ **Set the filter cutoff frequency with the FREQ knob.**

- **KBD**

→ **Turn the KBD (Keyboard Track) knob to set how much the cutoff frequency should track incoming MIDI Notes.**
Range: 0% (no tracking (constant frequency)) to 100% (1 semitone per key).

► **The Slope and Freq parameters can be modulated, e.g. from an Envelope source, in the Modulation Bus section, see "The Modulation Bus section".**

Modifier 1 and 2



The two Modifiers in each Sound Engine control how the frequency content of the selected waveform/signal (see "Waveform/signal selector") should be modified/affected. The modifiers are routed in series: Modifier 1 to Modifier 2.

→ **Turn On/Off a Modifier with the Modifier On/Off buttons.**
The last selected Modifier is automatically activated when you switch on the Modifier again.

→ **Select the desired modifier type (algorithm) by clicking on the selection button at the top left of the display, and selecting from the list that appears.**
The interactive display shows a graphical representation of the selected modifier type and also indicates (in text) what functionality the X and Y knobs have in the selected modifier. Texts with - and + signs indicate that the parameter is bipolar, i.e. has the value 0 in the middle (50%), negative values to the left (0-49%) and positive values (51-100) to the right.

▪ **For modifier types that display frequency on the x-axis, the readout in the display is always logarithmic.**
Some of the explaining graphics in this section of the manual show the frequency in a linear fashion, but that's just to make it easier to see what's going on in the higher frequencies/partials.

→ **Click/draw in the interactive display to edit the values or to manually draw in your custom modifications.**
See the descriptions of each of the modifier types for details about what can be drawn/edited.

→ **Turn the X and Y controls to edit the parameters described in text above each of the knobs.**
The X and Y parameter functions are described in each of the Modifier Type descriptions below.

! **Note that there is a big difference between modifier types that affect the frequency content in the sound (e.g. lowpass/highpass and bandpass filters) and modifier types that affect the partials in the voice (e.g. brick wall bandpass and brick wall notch filters).**

Modifier types that affect the frequency content in the sound do not track the keyboard by default; this has to be manually set up in the Modulation Bus section.

Modifier types that affect the partials in the voice automatically "track" the keyboard and do not require any Modulation Bus setup.

The available Modifier types are:

- **Off**

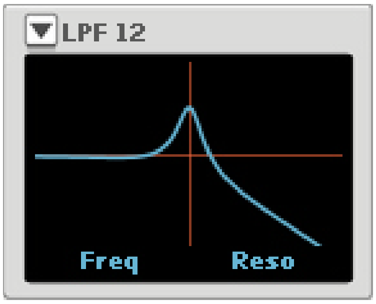
This bypasses the signal from the previous section in the Sound Engine to the next destination. Selecting the “OFF” Modifier type is exactly the same as deactivating the Modifier altogether with the On/Off button. The reason the “OFF” Modifier type is available is for backwards compatibility to the original Parsec.

If Modifier 1 is set to OFF, the signal goes from the Generator, via the Filter, straight to Modifier 2.

If Modifier 2 is set to OFF, the signal goes from Modifier 1 straight to the Balance mixer.

If both Modifiers are set to OFF, the signal goes from the Generator, via the Filter, straight to the Balance mixer.

- **LPF 12**

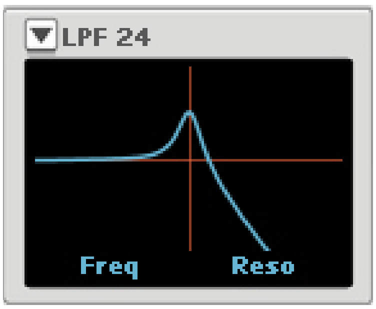


This simulates a standard 12dB/octave lowpass filter. The filter works individually on each voice and affects the frequency content in the sound.

→ **Click or click and drag in the display to change the cutoff frequency and the resonance.**
The X parameter controls the cutoff frequency and the Y parameter controls the resonance.

► **If you want the cutoff frequency to track the keyboard, assign “Note Number” as Source and “Mod X” as Destination in the Modulation Bus section. Set the Destination Amount value to 100 for full (one semitone per key) keyboard tracking. See “The Modulation Bus section”.**

- **LPF 24**

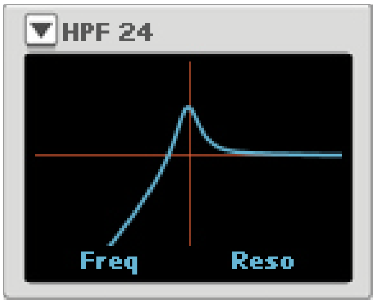


This simulates a standard 24dB/octave lowpass filter. The filter works individually on each voice and affects the frequency content in the sound.

→ **Click or click and drag in the display to change the cutoff frequency and the resonance.**
The X parameter controls the cutoff frequency and the Y parameter controls the resonance.

- If you want the cutoff frequency to track the keyboard, assign “Note Number” as Source and “Mod X” as Destination in the Modulation Bus section. Set the Destination Amount value to 100 for full (one semitone per key) keyboard tracking. See “The Modulation Bus section”.

- **HPF 24**

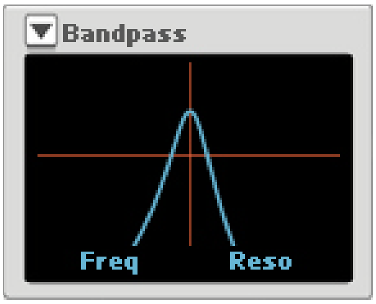


This simulates a standard 24dB/octave highpass filter. The filter works individually on each voice and affects the frequency content in the sound.

→ **Click or click and drag in the display to change the cutoff frequency and the resonance.**
The X parameter controls the cutoff frequency and the Y parameter controls the resonance.

- If you want the cutoff frequency to track the keyboard, assign “Note Number” as Source and “Mod X” as Destination in the Modulation Bus section. Set the Destination Amount value to 100 for full (one semitone per key) keyboard tracking. See “The Modulation Bus section”.

- **Bandpass**

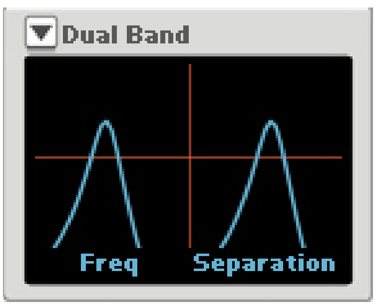


This simulates a standard 12dB/octave bandpass filter. The filter works individually on each voice and affects the frequency content in the sound.

→ **Click or click and drag in the display to change the center frequency and the resonance.**
The X parameter controls the center frequency and the Y parameter controls the resonance. When Y=100% the filter peak is so narrow that only a small frequency band (or one or a few partials) are let through.

- If you want the center frequency to track the keyboard, assign “Note Number” as Source and “Mod X” as Destination in the Modulation Bus section. Set the Destination Amount value to 100 for full (one semitone per key) keyboard tracking. See “The Modulation Bus section”.

- **Dual Band**

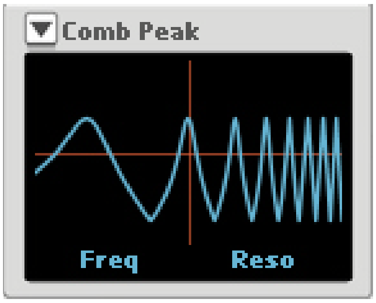


This simulates two 12dB/octave bandpass filter routed in parallel. The filter works individually on each voice and affects the frequency content in the sound.

→ **Click or click and drag in the display to change the center frequency and the separation.**
The X parameter controls the center frequency of the first bandpass filter (when Y=0%) and the Y parameter controls the separation (spacing) between the two bandpass filter peaks.

► **If you want the center frequency to track the keyboard, assign "Note Number" as Source and "Mod X" as Destination in the Modulation Bus section. Set the Destination Amount value to 100 for full (one semitone per key) keyboard tracking. See** **"The Modulation Bus section".**

- **Comb Peak**

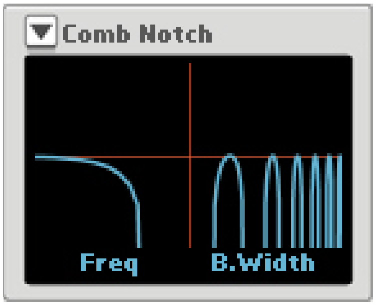


This simulates a comb filter with a positive feedback loop - but without feed forward - ideal for flanger and phaser types of effects. The filter works individually on each voice and affects the frequency content in the sound.

→ **Click or click and drag in the display to change the frequency of the second peak and the resonance.**
The X parameter sets the frequency of the second peak, and thus the spacing between the filter peaks. The Y parameter controls the resonance amount.

► **If you want the cutoff frequency to track the keyboard, assign "Note Number" as Source and "Mod X" as Destination in the Modulation Bus section. Set the Destination Amount value to 100 for full (one semitone per key) keyboard tracking. See** **"The Modulation Bus section".**

- **Comb Notch**



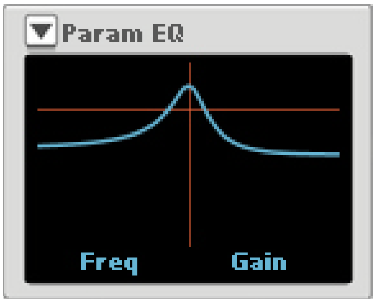
This simulates a multi notch filter, great for phaser types of effects. The filter works individually on each voice and affects the frequency content in the sound.

→ **Click or click and drag in the display to change the frequency of the first notch and the bandwidth.**
The X parameter controls the frequency of the first notch, and consequently "spacing" between the rest of the notches.

The Y parameter controls the attenuation amount, and consequently the bandwidth of the notches.

High Y values makes the attenuation notches wider, which means they remove bigger parts of the frequency spectrum.

- **Param EQ**



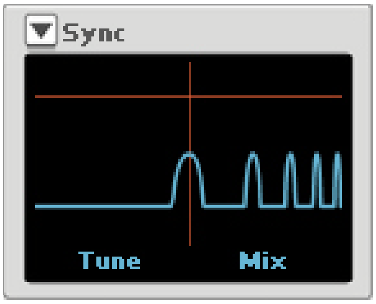
This simulates a single band parametric EQ with a fixed bandwidth. The EQ works individually on each voice and affects the frequency content in the sound.

→ **Click or click and drag in the display to change the center frequency and the gain/attenuation.**
The X parameter controls the center frequency for the EQ and the Y parameter controls the gain/attenuation amount. The attenuation/gain range is: -inf dB to +36dB.

! **Note that raising the gain towards maximum also narrows the peak bandwidth.**

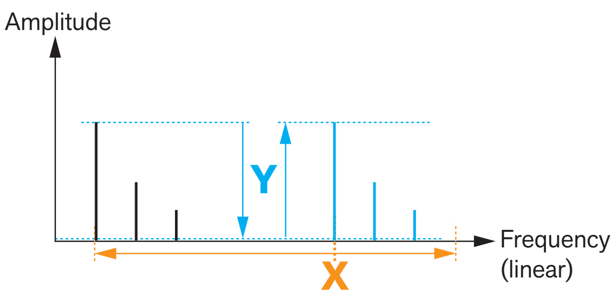
► **If you want the center frequency to track the keyboard, assign "Note Number" as Source and "Mod X" as Destination in the Modulation Bus section. Set the Destination Amount value to 100 for full (one semitone per key) keyboard tracking. See "The Modulation Bus section".**

- **Sync (tracks the keyboard)**



This simulates the frequency spectra of a "synced" oscillator, i.e. an oscillator (slave) that is constantly restarted at the rate of another oscillator (master). The Sync modifier creates this effect by multiplying the partials of the original signal with a factor and then interpolating and generating new harmonic partials above the original partials.

The sound of a "synced" signal is very characteristic, especially when the pitch of the slave oscillator is constantly changed, and thus changes the overtone character in the sound.



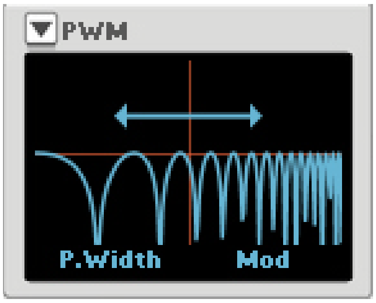
The partials of an original (3-partial) signal (black lines) and of the generated additional harmonic partials (blue lines) at a certain X value.

→ **Click or click and drag in the display to change the tuning and the mix.**
The X parameter controls the simulated sync ratio, i.e. the tuning of the simulated slave oscillator - and thus the distance between the resonance peaks in the frequency spectrum. Turn the X knob to achieve that typical sweeping “sync” sound.

The Y parameter controls the mix of the generated harmonics and the original signal in the “synced” sound. When Y=100%, only the generated “sync” harmonics are heard and the original signal is completely attenuated.

! **Since the Sync modifier actually adds completely new partials to a signal, it could be of importance in which Modifier “slot” (1 or 2) you use it. For example, placing a lowpass Filter modifier before or after the Sync modifier will render different results.**

- **PWM (tracks the keyboard)**

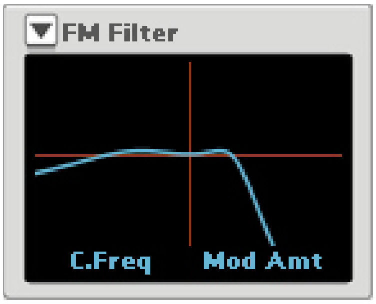


This simulates the frequency spectra you get from traditional pulse width modulation, i.e. when the pulse width of a square wave is constantly changed. Depending on what type of signal you have selected with the “Waveform/signal selector”, the character of the sound will be different. If you use a sawtooth signal the effect is almost the same as with traditional (square wave) pulse width modulation.

→ **Click or click and drag in the display to change the pulse width and the pulse width modulation amount.**
The X parameter controls the simulated pulse width frequency spectrum and the Y parameter controls a combination of pulse width modulation amount and rate.

► **If you want to control the modulation rate and amount from an LFO instead of from the Y parameter, set Y to zero and assign the X parameter as Destination to an LFO Source in the Modulation Bus section, see “The Modulation Bus section”.**

- **FM Filter (tracks the keyboard)**

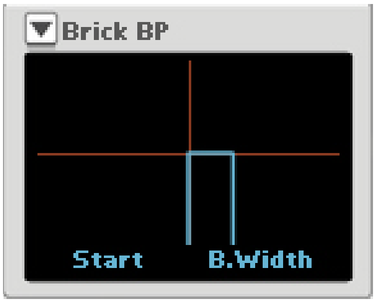


The FM Filter modifier creates a “virtual filter” which has the spectral characteristics of a frequency modulated signal.

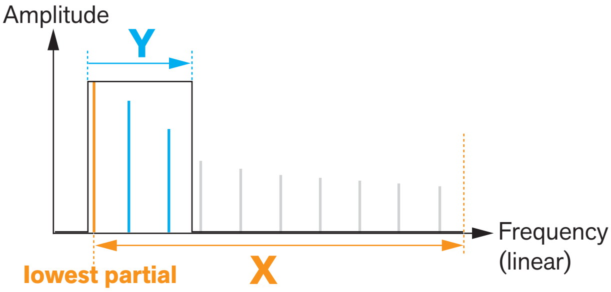
→ **Click or click and drag in the display to change the carrier frequency and the frequency modulation amount.**
The X parameter simulates the carrier frequency, and thus changes the position of the frequency spectrum. Since frequency modulated signals generate sidebands on either side of the fundamental frequency, increasing the X position will also mirror the spectrum downwards in the frequency range.

The Y parameter controls the frequency modulation amount, and thus the width of the spectrum. When Y=0% only a narrow band around the fundamental frequency is audible in each voice. If you use a harmonic Generator signal, sweeping the position with the X knob when Y=0% will create a dual sine wave “smooth glissando” effect.

- **Brick BP (tracks the keyboard)**



This simulates a brick wall bandpass filter, i.e. a bandpass filter with completely vertical slopes. Only the partials in the passband "window" pass through. The filter works individually on each voice and affects the partials in each voice.



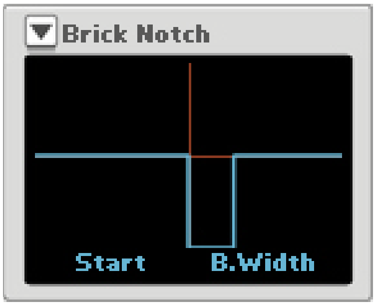
The orange and blue partials are let through. The grey lines represent attenuated partials.

→ **Click or click and drag in the display to change the start partial and the width of the passband.**
The X parameter defines at which partial of each voice the passband should begin and the Y parameter controls the width of the pass band.

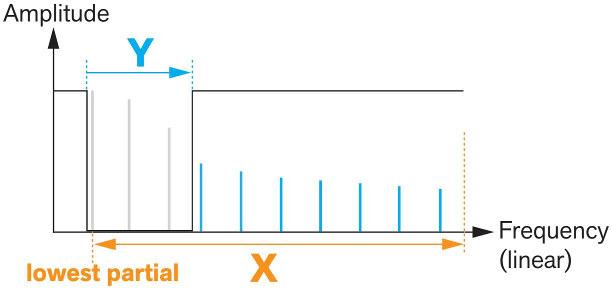
The lower limit of the X parameter is always the voice's fundamental (lowest partial). If both the X and Y parameters are set to zero, only the fundamental of each voice is heard.

► **Sweeping this type of filter generates an arpeggio-like effect, since the partials are being switched on and off instantaneously.**

- **Brick Notch (tracks the keyboard)**



This simulates a brick wall notch filter, i.e. a notch filter with completely vertical slopes. Only the partials outside the notch pass through. The filter works individually on each voice and affects the partials in each voice.



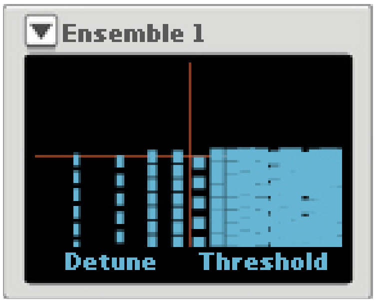
The blue partials are let through. The grey lines represent attenuated partials.

→ **Click or click and drag in the display to change the start partial and the width of the notch band.**
The X parameter defines at which partial of each voice the notch should begin and the Y parameter controls the width of the notch band.

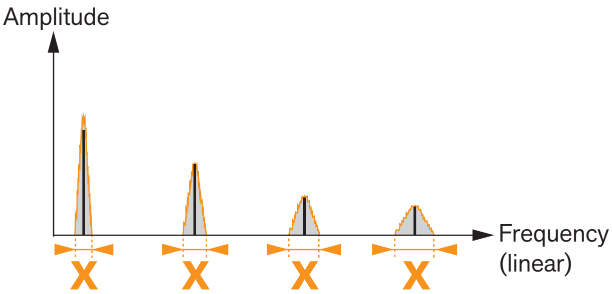
The lower limit of the X parameter is just a little below the voice's fundamental (lowest partial). If the X parameter is set to zero and the Y parameter is set to maximum, only the very highest partials of each voice is heard; in some situations no sound is heard at all (if the notch should attenuate all partials in the voice).

► **Sweeping this type of filter generates an arpeggio-like effect, since the partials are being switched on and off instantaneously.**

- **Ensemble 1 (tracks the keyboard)**



This is the perfect modifier for really dense pad sounds! The Ensemble 1 modifier simulates a type of chorus effect by utilizing noise modulation of the partials. The Ensemble 1 modifier is best suited for sounds with slow attacks, since the noise modulation always starts from zero and the transients are therefore "smeared out". The modifier works individually on each voice and affects the partials in each voice.

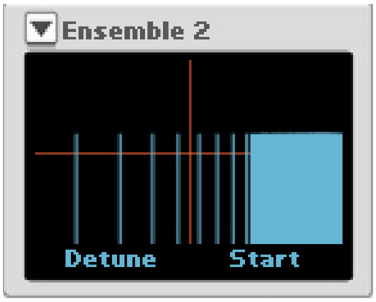


→ **Click or click and drag in the display to change the detune amount and the noise modulation threshold.**
The X parameter controls the detuning amount, in practice the "depth" of the Ensemble effect.

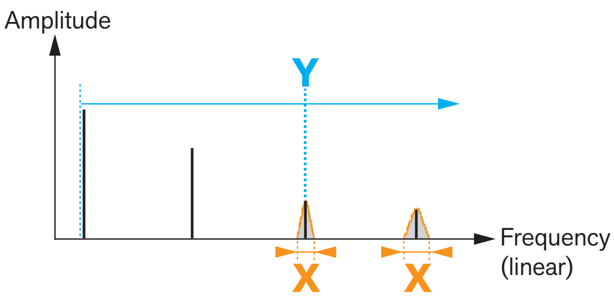
The Y parameter controls the noise modulation threshold. At low Y values the noise is dense and at high Y values the noise modulation becomes more of an intermittent "crackling" type.

► **Turn up the Spread control for really nice and wide stereo chorus effects, see** **"Spread".**

- **Ensemble 2 (tracks the keyboard)**



The Ensemble 2 modifier simulates a type of chorus effect by utilizing noise modulation of the higher partials, leaving the lower partials unaffected - or less affected. The Ensemble 2 modifier is suitable for sounds with fast attacks, since the noise modulation always starts at full level, and thus preserves any initial transients in the sound. At higher Y values the modifier is also very useful for generating "breath" and "wind" types of effects to the sound. The modifier works individually on each voice and affects the partials in each voice.



Here, the noise modulation begins at the 3rd partial, leaving the two partials below unaffected.

→ **Click or click and drag in the display to change the detune amount and at which partial the modulation should start.**

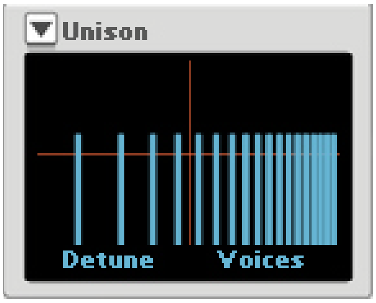
The X parameter controls the detuning amount, in practice the "depth" of the Ensemble effect.

The Y parameter controls at which partial the noise modulation should begin.

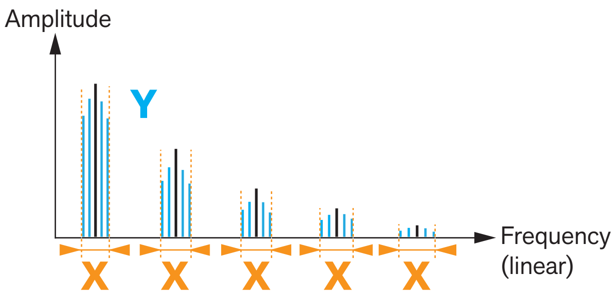
If you want all partials of the voice to be modulated, set the Y parameter to zero. If you want only the higher partials of the voice to be noise modulated, set the Y parameter fairly high.

► **Turn up the Spread control for really nice and wide stereo chorus effects, see "Spread".**

- **Unison (tracks the keyboard)**



This modifier simulates several detuned voices by generating copies of the original partials in pairs on either side of the original partials.



The black lines represent the original partials in the voice.
The blue lines represent the detuned virtual voices.

→ **Click or click and drag in the display to change the detune amount and the number of additional generated voices.**
The X parameter controls the detuning amount and the Y parameter controls the number of virtual voices. A maximum of 16 detuned voices plus the original voice are available. The Y parameter smoothly adds (0), 2, 4, 8 or 16 additional "virtual" voices.

► **Turn up the Spread control for really nice and wide stereo unison effects! See "Spread".**

- **Body Wide/Body Narrow/Body Sweep Wide/Body Sweep Narrow**

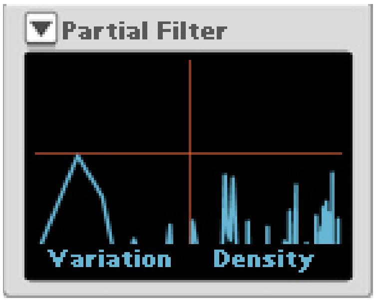
The four Body algorithms are formant filters that simulate body resonances (multi-peak+notch filters). The filters work individually on each voice and affects the frequency content in the sound.

→ **Click or click and drag in the display to shift the position (or sweep) in the formant filter table - and the amount/resonance of the filter.**
In the first two algorithms (Body Wide and Body Narrow) you can shift the position in a table of several formant filters. The X parameter controls the position of the filter in the table, which makes it possible to alter the body character of the sound. The Y parameter controls the amount (resonance).

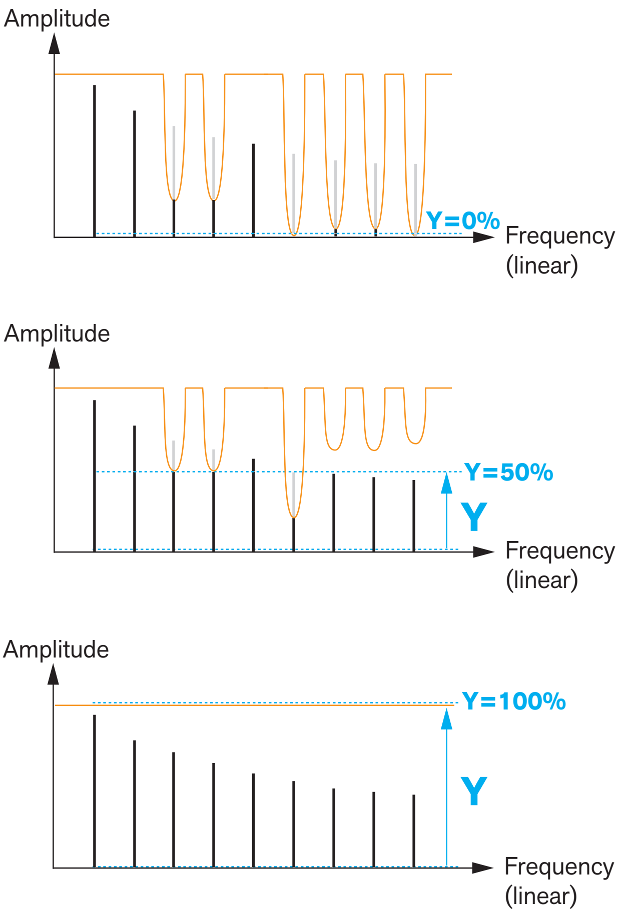
In the last two algorithms (Body Sweep Wide and Body Sweep Narrow) the formant filter peaks and notches are at a fixed relative distance from each other. Here, the X parameter controls the sweep in the frequency range, in practice the "size" of the body, from large to small. The Y parameter controls the amount (resonance).

► **If you want the body "size" to track the keyboard in the last two algorithms, assign "Note Number" as Source and "Mod X" as Destination in the Modulation Bus section. Set the Destination Amount value to 100 for full (one semitone per key) keyboard tracking. See "The Modulation Bus section".**

- **Partial Filter (tracks the keyboard)**



The Partial Filter modifier attenuates individual partials in the voice according to a number of preset variations. The filter works individually on each voice.

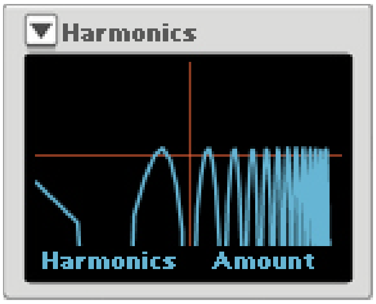


One of the Partial Filter variations (orange line) with the Y parameter at 0%, 50% and 100% (top to bottom).
The grey lines represent attenuated partials or attenuated parts of partials.

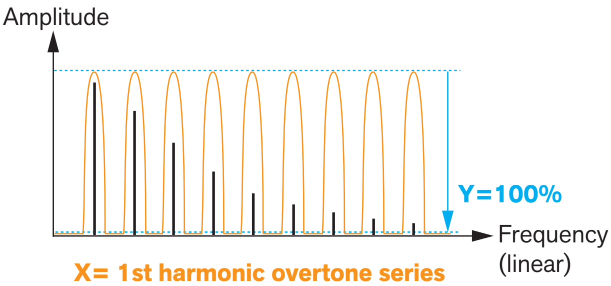
→ **Click or click and drag in the display to define the partial variation preset and the partial attenuation amount.**
The X parameter defines the desired partial variation (which preset combination of partials that should be attenuated) and the Y parameter controls the density of the sound. A low Y value means high attenuation of the partials in the selected Variation and a high Y value is no attenuation (bypass).

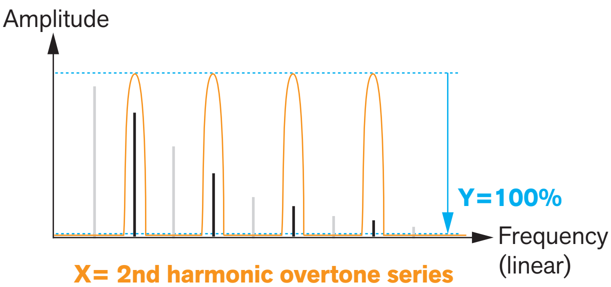
! **Note that the attenuation "rate" of the individual partials might not be linear across the Y parameter range. This is intentional and means that some partials might be attenuated faster (or slower) when the Y parameter is changed in certain intervals of its range.**

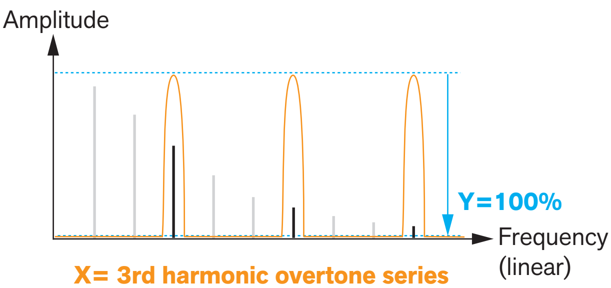
- **Harmonics (tracks the keyboard)**



The Harmonics modifier lets you alter between harmonic overtone series in the sound. The modifier works individually on each voice and affects the partials.





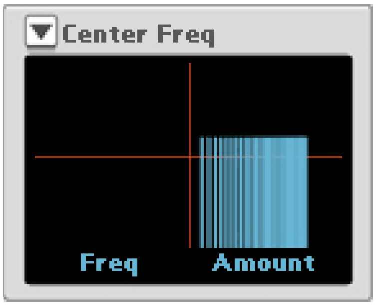


The first, second and third harmonic overtone series when Y=100%.
The black lines represent sounding partials and the grey lines represent attenuated partials.

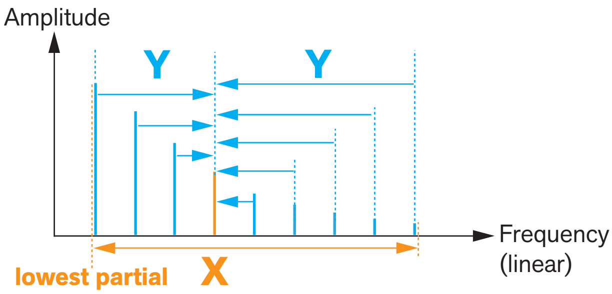
→ **Click or click and drag in the display to select the overtone series and the attenuation amount of the remaining overtones.**
The X parameter sets the desired harmonic overtone series and the Y parameter controls the attenuation amount of the remaining overtones (the overtones outside the currently selected harmonic series).

► **To create flageolet effects, raise the Y parameter fairly high and then assign the Y parameter as Destination to an Envelope Source in the Modulation Bus (see "The Modulation Bus section"). Set the modulation Amount to a low negative (-) value and adjust the Source Envelope Decay time to get the typical effect. Select the desired harmonic overtone series with the X knob.**

- **Center Freq (tracks the keyboard)**



This modifier detunes all partials in the voice upwards and/or downwards towards one single frequency in the voice.



Here, the X parameter is set to the fourth partial and the Y parameter is set to 100%.

→ **Click or click and drag in the display to set the center frequency and the detune amount.**
The X parameter defines the frequency towards which to detune the other partials in each voice.

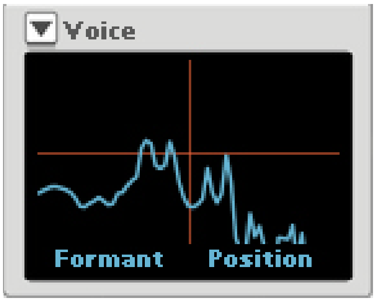
The Y parameter controls the detuning amount, i.e. how close to the set (X) partial the other partials should be detuned.

If X=0% and the Y=100%, all partials in the voice are detuned down to the first partial in the voice.

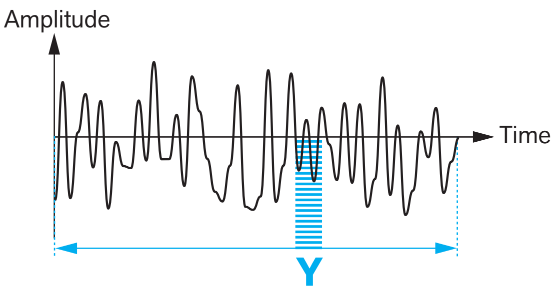
If Y=0%, no detuning takes place regardless of the X value.

► **Assign the Y parameter to an Envelope Source in the Modulation Bus section to create e.g. "partial pitch bend" effects or inharmonic attacks, see "The Modulation Bus section".**

- **Voice**



This modifier detects formants in a pre-recorded vocal sample in Parsec 2 and applies the formants to the Generator signal.

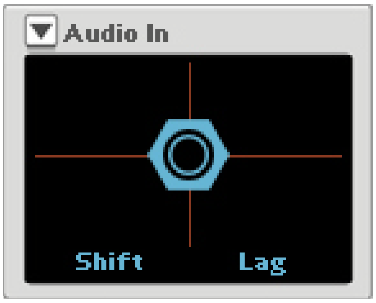


→ **Click or click and drag in the display to set the formant pitch shift amount and the position in the pre-recorded sample.**
The X parameter defines the formant pitch shift amount, i.e. the "bassy vs. chipmunk"-effect.
Range: +/-3.5 octaves. When X=50% there is no formant shift.

The Y parameter controls the position in the pre-recorded sample. When you change the position, the formant also changes.

► **Modulate the Y parameter, e.g. from an LFO, to achieve animated vocal effects, see "The Modulation Bus section".**

- **Audio In**



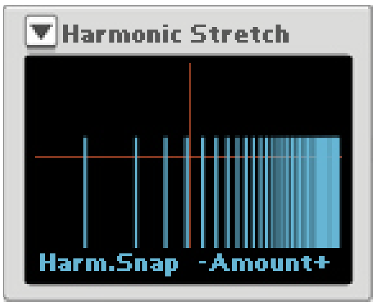
This lets you modify the Generator signal by modulating it from the Audio Input on the rear panel of Parsec 2; in practice a vocoder effect. The Audio Input signal is analyzed in real-time and the formants are applied to the Generator signal. The formants can also be shifted up and down in frequency before being applied to the Generator signal.

→ **Click or click and drag in the display to set the formant pitch shift amount and the lagging of the audio input signal.**
With the X parameter you set the formant shift amount, i.e. the "bassy vs. chipmunk"-effect.
Range: +/-3.5 octaves. When X=50% there is no formant shift.

The Y parameter controls the "lagging" of the audio input signal. At low Y values the vocoder effect is instant, and at higher Y values the audio input signal becomes delayed somewhat and also sustains for a longer time.

If the Y parameter is rapidly switched to 100%, you can "freeze" a snapshot the current audio input signal and have it modulate the Generator signal infinitely. The modulation will continue for as long as there is a signal present from the Generator.

- **Harmonic Stretch (tracks the keyboard)**



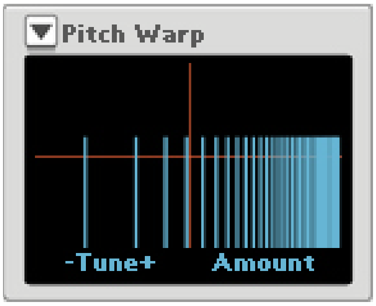
This modifier stretches or squeezes all partials (overtones) - except for the fundamental - in the signal, up or down in the frequency range.

→ **Click or click and drag in the display to set the harmonic snap value and the stretch amount.**
The X parameter defines how the frequencies should snap to harmonic frequencies of the signal - or "float freely" across the frequency range. An X value of 0% means "free float" (smooth) and 100% means full snap to the harmonic overtone series defined by the Generator pitch.

The Y parameter controls the stretch amount and direction, i.e. the total frequency range for the stretch - down or up. an Y value of 50% means no stretch at all, 0% means full squeeze down in pitch and 100% means full stretch up in pitch.

► **The Harmonic Stretch Modifier works great routed in series after the Pitch Warp Modifier (see below).**

- **Pitch Warp (tracks the keyboard)**



This modifier creates a copy of the signal - which can be detuned smoothly in eight steps - and tunes the pitch of the original and the copied signal, up or down, by smoothly pitch shifting all partials in the original and copied signal.

→ **Click or click and drag in the display to set the pitch tuning and the detuning amount of the copied signal.**
The X parameter defines the tuning pitch and direction (down/up) and the Y parameter defines the detuning amount of the copied signal.

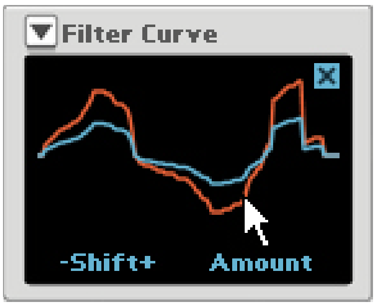
The easiest way of controlling this Modifier is probably by using the X and Y knobs instead of drawing in the display. This is especially true when you want to set the detune amount between the original signal and the copied signal. Just to describe how the modifier works, start by setting both the X and Y knobs to 0%, By turning the Y knob upwards you can see the mix between the original and copied signals in the display. As you turn the Y knob upwards, the copied signal gets detuned down in octave intervals.

► **A practical use of the modifier could, for example, be to create "tape stop" effects. Turn up the Y knob fairly high (around the 2 o'clock position) and so that you can only hear the original signal (no "chorus" effect"). Then turn the X knob from 50% to 0% to get a "tape stop" effect.**

► **The Pitch Warp Modifier is perfect to modulate from fast envelopes, to create interesting attacks.**

► **The Pitch Warp Modifier works great routed before the Harmonic Stretch Modifier (see above).**

- **Filter Curve**



With this modifier you can design your own custom filter, by drawing in your desired filter curve in the display. You can then use the X and Y knobs to change the "cutoff" frequency and the resonance/attenuation amount.

→ **Draw/edit your curve by clicking or "drawing" in the display.**
The curve you draw is displayed in red in the display. The blue curve, however, shows the actual resulting filter characteristics.

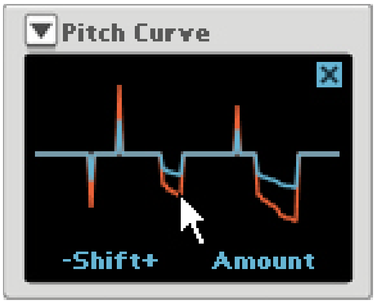
If you turn the Y knob to max, the drawn (red) curve and the actual (blue) curve will coincide. With the Y knob set to minimum, the filter curve will be completely flat and won't affect the signal.

You can also move the filter curve in the frequency range, by turning the X knob. The curve will maintain its shape across the entire frequency range.

→ **To start from scratch with a flat curve, click the X button at the top right in the display.**

! **Note that the last drawn filter curve is automatically stored in the corresponding Parsec 2 device, so if you change Modifier type in the Modifier slot and then revert, the latest filter curve is still there.**

- **Pitch Curve**



With this modifier you can pitch shift individual overtones or ranges of overtones in the signal, by drawing in your desired pitch-shift amount curve in the display. You can then use the X knob to sweep the pitch-shift curve throughout the frequency range and the Y knob to set the pitch-shift amount (deviation) of the frequencies/ frequency ranges.

→ **Draw/edit your curve by clicking or "drawing" in the display.**
The curve you draw is displayed in red in the display. The blue curve, however, shows the actual resulting pitch-shift/deviation characteristics. The "no pitch-shift" line is in the middle of the display. Values above the line mean pitched up and values below the line indicate pitched down partials.

If you turn the Y knob to max, the drawn (red) curve and the actual (blue) curve will coincide. With the Y knob set to minimum, the pitch-shift amount curve will be completely flat and won't affect the signal.

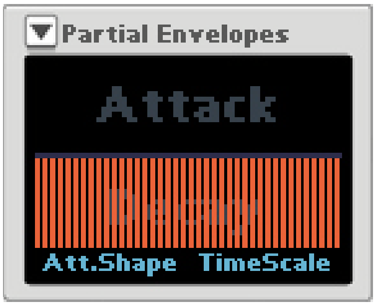
You can also move the pitch-shift curve throughout the partials in the frequency range, by turning the X knob. The curve will maintain its shape across the entire frequency range.

► **To get a pitch-shift character that tracks the notes you play, assign the X knob to Keyboard Tracking in the Modulation Bus, see** **"The Modulation Bus section".**

→ **To start from scratch with a flat curve, click the X button at the top right in the display.**

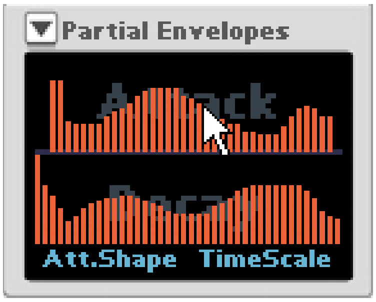
! **Note that the last drawn pitch-shift curve is automatically stored in the corresponding Parsec 2 device, so if you change Modifier type in the Modifier slot and then revert, the latest filter curve is still there.**

- **Partial Envelopes (tracks the keyboard)**



The Partial Envelopes modifier is an Attack-Decay envelope that can control the amplitude envelope of each partial in the signal individually. The Partial Envelopes display is divided into two areas: the upper area of the display where you draw the Attack times for the partials and the lower area where you draw the Decay times.

→ **Draw/edit your curve by clicking or "drawing" in the display:**



Edited (drawn) Attack and Decay times in the display.

The two orange lines to the far left in the display represent the Attack (top) and Decay (bottom) times of the fundamental partial. The lines to the right throughout the rest of the display represent the attack and decay times of the remaining partials in the signal. The two orange lines to the far right in the display represent all remaining partials above the 40th partial (which the display doesn't show).

Once you have drawn/edited your Attack and Decay times for the partials, you can control the Attack Shape character with the X knob and the Time Scale with the Y knob.

A low X value starts the Attack envelopes of all partials at exactly the same time (at their individual Attack time settings), whereas a high X value triggers the playback of the partials differently. This is great for "strumming" effects, for example.

A low Y value forces the envelopes to go really quick (for "staccato effects"), whereas a high Y value makes both the Attack and Decay times longer.

► **The Partial Envelopes Modifier routed in series after the Pitch Curve Modifier (see previous Modifier type) is great for creating interesting percussive sounds.**

! **Note that the Amp Envelope section in Parsec 2 also affects the envelope of the signal. Depending on the settings of the Amp Envelope, the result produced by the Partial Envelopes might not be what you expect.**

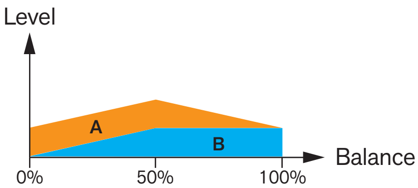
! **Note that the last drawn settings are automatically stored in the corresponding Modifier slot in the Parsec 2 device, so if you change Modifier type in the Modifier slot and then revert, the latest curve will still be there.**

Common modulation controls

The common modulation controls affect both Sound Engines.

Balance A-B

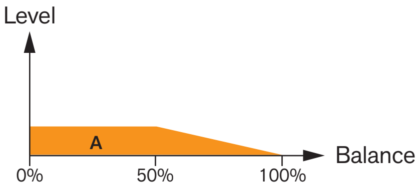
→ **Set the mix between Sound Engines A and B with the Balance knob.**
If there are signals present from both Sound Engines (A and B), the mix and level distribution works according to the picture below:



Between 0% and 50% Sound Engine A's level is constant and between 50% and 100% Sound Engine B's level is constant.

If you only want to use the signal from Sound Engine B, turn the Balance knob fully clockwise.

If only Sound Engine A is active (B switched off), the Balance control works according to the figure below:

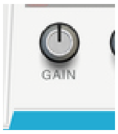


Balance levels when Sound Engine B is switched off.

Link Engines

→ **Click the Link Engines button to route the output from Sound Engine A into Sound Engine B.**
This allows you to use up to four Modifiers in series, to process the signal. See "Signal flow with the "Link Engines" function" for a signal flowchart and a description.

Gain

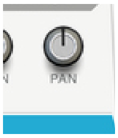


→ **Set the desired maximum level for the built-in Amp Envelope amplifier with the Gain knob.**
This is the maximum level the envelope will reach after the Attack stage is completed (see "Amp Envelope" below).

► **If you want to make the Amp Envelope velocity sensitive, assign "Gain" as Destination and "Velocity" as Source in the Modulation Bus section, see "The Modulation Bus section".**

► **If you want to create a tremolo effect, assign "Gain" as Destination and an LFO as Source in the Modulation Bus section, see "The Modulation Bus section".**

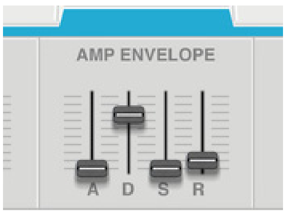
Pan



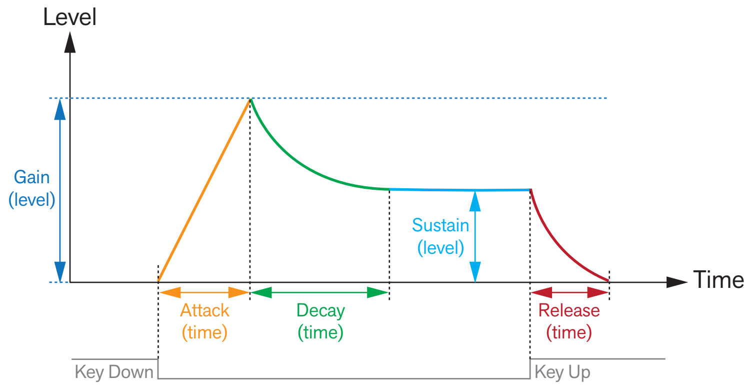
→ **Set the panning of the output of the Sound Engines in the stereo panorama.**

► **Since Pan works individually per voice, you can assign e.g. Keyboard Velocity or an Envelope in the Modulation Bus to control the Pan effect, see "The Modulation Bus section".**

Amp Envelope



This is a standard ADSR envelope which controls the amplitude of both Sound Engines equally. The picture below shows the various stages of the ADSR envelope:



The ADSR envelope stages.

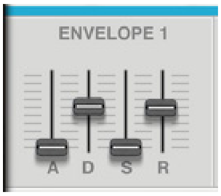
- **A(ttack)**
  When you play a note on your keyboard, the envelope is triggered. This means it starts rising from zero to the value set with the Gain slider (see above). How long this should take, depends on the Attack setting. If the Attack is set to "0", the Gain value is reached instantly. If the Attack value is raised, it will take longer time before the Gain value is reached.
- **D(ecay)**
  After the Gain value has been reached, the level starts to drop. How long this should take is governed by the Decay parameter.

  If you want to emulate the volume envelope of a note played on a piano for example, the Attack should be set to "0", the Decay parameter should be set to a medium value and the Sustain level should be set to "0", so that the volume gradually decreases down to silence, even if you keep holding the key down. Should you want the decay to drop to some other value than zero, you raise the Sustain parameter.
- **S(ustain)**
  The Sustain parameter determines the level the envelope should rest at, after the Decay stage. If you set Sustain to full level, the Decay setting is of no importance since the volume of the sound is never lowered.

  If you want to emulate the volume envelope of an organ, you theoretically only really need to use the Sustain parameter set to full level, as a basic organ volume envelope instantly goes to the maximum level (Attack "0") and stays there (Decay "0"), until the key is released and the sound instantly stops (Release "0").

  But often a combination of Decay and Sustain is used to generate envelopes that rise up to the Amount value, then gradually decreases to finally land to rest on a level somewhere in-between zero and the Amount value. Note that Sustain represents a level, whereas the other envelope parameters represent times.
- **R(elease)**
  The Release parameter works just like the Decay parameter, except it determines the time it takes for the volume to drop back to zero after you release the key.

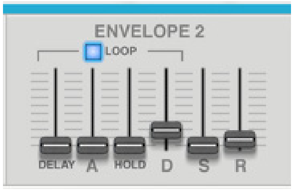
Envelope 1



This is a polyphonic (one per voice) general purpose ADSR envelope generator (see the description above) which can be assigned to control selectable parameter(s) in the Modulation Bus section. For example, you can assign it to control the X and Y parameters in the desired Sound Engine Modifier algorithms.

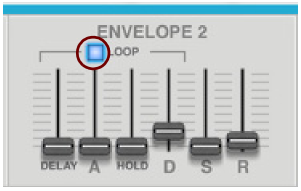
See "The Modulation Bus section" for details on how to assign Envelope 1 to the desired destination(s).

Envelope 2



This is a polyphonic (one per voice) general purpose AHDSR (Attack, Hold, Decay, Sustain, Release) envelope with a pre-delay stage before the Attack phase. The Delay to Decay phase can also be looped. Besides the standard ADSR envelope parameters, Envelope 2 has the following additional parameters:

- **Delay**
  This can be used to set a delay before the onset of the envelope.
- **Hold**
  This allows you to set a "hold" phase before the Decay. The envelope level is always at max during the hold stage.
- **Loop**

  If Loop is activated, the envelope stages will loop continuously, from the Delay stage to the Decay stage, for as long as the key is held down.

The modulation destination(s) for Envelope 2 can be assigned in the Modulation Bus, see "The Modulation Bus section".

LFO 1

An LFO (Low Frequency Oscillator) is used for generating cyclic modulation. A typical example is to have an LFO modulate the pitch of a signal to produce vibrato, but there are countless other applications for LFOs. LFO 1 applies modulation polyphonically (one LFO per voice), i.e. if LFO 1 is assigned to modulate a parameter, an individual LFO will be started for each note you play. Also, LFO 1 always restarts on Note On (key down). Use the Modulation Bus to assign modulation destination(s) for LFO 1, see "The Modulation Bus section".

→ **Select an LFO waveform by clicking the spin controls to the right of the waveform display, or by click-holding in the display and moving the mouse up or down.**
Besides the standard waveforms (sine, triangle, pulse, etc.) there are random, slope and stepped waveforms. The shape of the waveforms are shown in the display.

→ **Set the LFO frequency with the Rate knob.**

→ **Turn the Delay knob to introduce a delay before the LFO modulation kicks in after a note is played.**
Turn clockwise for longer delay times.

LFO 2 Global

LFO 2 Global is monophonic and affects all voices together. Use the Modulation Bus to set modulation destination(s) for LFO 2, see "The Modulation Bus section".

→ **Select an LFO waveform by using the spin controls to the right of the waveform display, or by clicking in the display and moving the mouse up or down.**
Besides the standard waveforms (sine, triangle, pulse, etc.) there are random, slope and stepped waveforms. The shape of the waveforms are shown in the display.

→ **Set the LFO rate with the Rate knob.**

→ **Click the Sync button to sync LFO 2 to the main sequencer Tempo in Reason/Reason Essentials.**
The Rate parameter now controls the time divisions.

Spread

The Spread control works a little differently depending on selected waveform(s) in the Sound Engines' Generators. However, the effect is that the partials in each voice of the sound are spread in the stereo panorama in a couple of different ways.

→ **Select Spread Mode with the three-way switch.**
"A-B" spreads the partials in each voice of Sound Engine A to the left and the partials in each voice of Sound Engine B to the right in the stereo panorama.

"High" spreads the high frequencies of each voice of Sound Engine A and B alternating between the left and right outputs. The lower and fundamental partials of each voice are kept fairly centered, but gets more spread for each additional note you play in a chord.

"All" spreads the odd and even partials in each voices of Sound Engine A and B alternating between the left and right outputs.

→ **Control the Spread amount with the Amount knob.**

► **Since Spread works individually per voice, you can assign e.g. Keyboard Velocity in the Modulation Bus to control the Spread effect, see "The Modulation Bus section".**

! **Note that the Spread control affects the partials in each voice in each Generator individually. This means you will get a stereo spread effect even if you are using only one Sound Engine.**

! **Note that the result of the Spread function is also affected by the Pan setting, see "Pan".**

Global effects

Reverb

This is a stereo reverb which affects all voices globally.

- **On/Off**
  Click the red On/Off button to switch the Reverb effect on/off.
- **Size**
  Sets the emulated room size, from small room to large hall. Middle position is the default room size.
  Lowering this parameter results in a closer and gradually more "canned" sound. Raising the parameter results in a more spacey sound, with longer pre-delay.
- **Decay**
  This governs the length of the reverb effect. Middle position is the default decay time.
- **Damp**
  Raising the Damp value cuts off the high frequencies of the reverb, thereby creating a smoother, warmer effect.
- **Level**
  Use this parameter to adjust the balance between the unprocessed audio signal and the reverb effect.

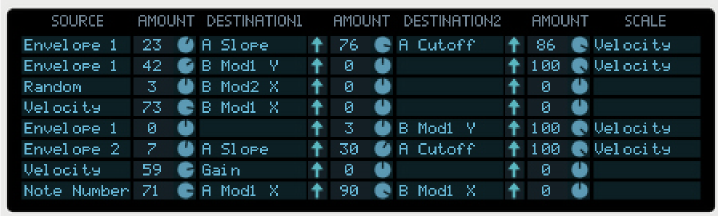
Delay

This is a stereo delay which affects all voices globally.

- **On/Off**
  Click the red On/Off button to switch the Delay effect on/off.
- **Sync**
  Click the Sync button to sync the delay time to the main sequencer Tempo.
- **Time**
  This sets the time between the delay repeats. If Tempo Sync is active (see above), the Time parameter now controls the time divisions.
- **F.back**
  The Feedback parameter determines the number of delay repeats.
- **Dry/W**
  Use this parameter to adjust the balance between the unprocessed audio signal (dry) and the effect (wet).

► **To get a wider stereo delay effect, turn up the Spread Amount knob, see "Spread". This spreads the stereo input signals to the Delay and consequently gives a wider stereo effect from the Delay.**

The Modulation Bus section



The Modulation Bus section is used for routing a modulation Source to one or two modulation Destinations each. This creates a very flexible routing system that complements the pre-wired routing in Parsec 2.

The Modulation Bus section in Parsec 2 is derived from the one in the Reason Thor Polysonic Synthesizer device, so if you are familiar with Thor, you will quickly find your way around in Parsec 2's modulation bus.

There are eight "Source –> Destination 1 –> Destination 2 –> Scale" busses, of which the first four have pre-assigned sources. However, these four pre-assigned sources can be easily changed if you like.

A Source parameter can modulate two different Destination parameters per bus (with variable Amount settings). Each bus also has a Scale parameter that affects the relative modulation Amount for both Destinations.

► **Note that it is possible to assign the same source parameter as Source in several busses. This allows you to control more than two Destination parameters from the same Source.**

1. Select the desired Source parameter by clicking in the corresponding Source box and selecting from the list.

The following parameters can be used as modulation Sources:

Parameter	Description
Velocity	This applies modulation according to the Keyboard Velocity values (how hard or soft you strike the MIDI keyboard keys).
Envelope 1 (pre-assigned in bus 1)	This allows you to modulate parameters from Envelope 1. This can also be selected as Source in any of the assignable modulation busses.
Envelope 2 (pre-assigned in bus 2)	This allows you to modulate parameters from Envelope 2. This can also be selected as Source in any of the assignable modulation busses.
Amp Envelope	This allows you to modulate parameters from the Amp Envelope.
LFO 1 (pre-assigned in bus 3)	This allows you to modulate parameters from LFO 1.
LFO 2 (pre-assigned in bus 4)	This allows you to modulate parameters from LFO 2.
Mod Wheel	This allows you to modulate parameters from the Mod Wheel.
Pitch Bend	This allows you to modulate parameters from the Pitch Bend control.
Breath	This allows you to modulate parameters from the Breath performance controller
Expression	This allows you to modulate parameters from the Expression performance controller
Aftertouch	This allows you to modulate parameters from Keyboard Aftertouch (channel aftertouch)
Sustain Pedal	This allows you to modulate parameters from a connected sustain pedal. Note that continuous sustain data (0-127) is supported - not just on/off.
Note Number	This is keyboard tracking. If a positive Amount value is used and the destination is filter frequency, the filter frequency will track the keyboard, i.e. increase with higher notes.
Random	This sends out a random value each time a new note is played.
CV Input 1 and 2	This takes the current value on the CV 1 and/or CV 2 inputs on the rear panel and sends to the desired destination.
Key In Oct.	This allows you to modulate parameters based on 12 separate note values (within each octave).
Noise	This allows you to modulate parameters from white noise.
Polyphony	This allows you to modulate parameters based on the number of Note Ons at a given time.
CV1 and CV2 Latched	This allows you to modulate parameters based on the current CV1 and/or CV2 value at a given Note On.
MW Latched	This allows you to modulate parameters based on the current Mod Wheel value at a given Note On.

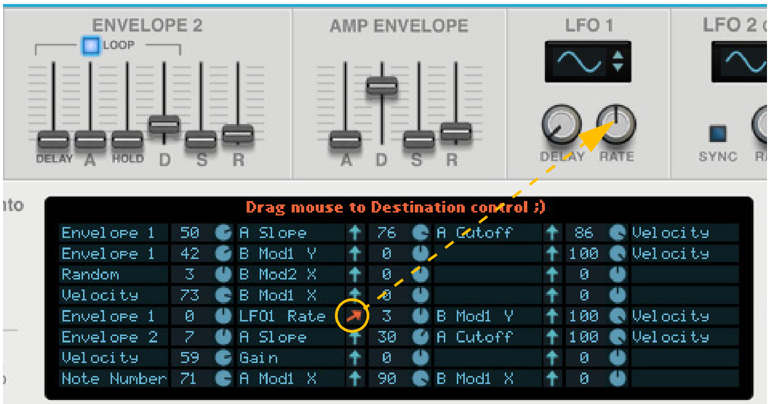
Modulation Bus Source parameters.

2. **Set the Amount for the first Destination by turning the corresponding Amount knob, or by clicking and dragging vertically in the corresponding Amount box.**

! **Note that the Amount range in the Modulation Bus section is +/-100. This means that the Amount value can exceed the modulated parameter's range. When this happens, the modulated parameter simply stays at its extreme value until the Amount value gets within the parameter's range again.**

3. **Select the first Destination parameter by click-holding the blue arrow symbol to the right of the corresponding Destination box.**
   The arrow symbol now turns orange.

4. **While click-holding, drag to the desired destination parameter on the panel:**



Assigning LFO 1 Rate as a Destination for Envelope 1

As you hover over a valid destination control on the panel, the parameter name is automatically displayed in the Destination box in the Modulation Bus.

5. **To assign the currently selected Destination control, release the mouse button.**

→ **Alternatively, click the desired Destination box and select the Destination parameter from the list.**

The following parameters can be used as modulation Destinations:

Parameter	Description
A Cutoff	This affects the Filter Freq parameter in Sound Engine A. A high modulation value opens the filter, and vice versa.
A Slope	This affects the Filter Slope parameter in Sound Engine A. A low modulation value results in high attenuation, and vice versa.
A Pitch	This affects the pitch of the Generator in Sound Engine A.
A Generator	This affects the Generator parameter in Sound Engine A.
A Mod1 X	This affects the X parameter in Modifier 1 in Sound Engine A.
A Mod1 Y	This affects the Y parameter in Modifier 1 in Sound Engine A.
A Mod2 X	This affects the X parameter in Modifier 2 in Sound Engine A.
A Mod2 Y	This affects the Y parameter in Modifier 2 in Sound Engine A.
B Cutoff	This affects the Filter Freq parameter in Sound Engine B. A high modulation value opens the filter, and vice versa.
B Slope	This affects the Filter Slope parameter in Sound Engine B. A low modulation value results in high attenuation, and vice versa.
B Pitch	This affects the pitch of the Generator in Sound Engine B.
B Generator	This affects the Generator parameter in Sound Engine B.
B Mod1 X	This affects the X parameter in Modifier 1 in Sound Engine B.
B Mod1 Y	This affects the Y parameter in Modifier 1 in Sound Engine B.

Parameter	Description
B Mod2 X	This affects the X parameter in Modifier 2 in Sound Engine B.
B Mod2 Y	This affects the Y parameter in Modifier 2 in Sound Engine B.
Gain	This affects the Gain parameter in the Amp Envelope.
Balance	This affects the Balance parameter, i.e. the mix between Sound Engines A and B.
Env1Attack	This affects the Attack time of Envelope 1.
Env1Decay	This affects the Decay time of Envelope 1.
Env1Release	This affects the Release time of Envelope 1.
Env2 Delay	This affects the Delay time of Envelope 2.
Env2 Decay	This affects the Decay time of Envelope 2.
Amp Env Attack	This affects the Attack time of the Amp Envelope.
Amp Env Decay	This affects the Decay time of the Amp Envelope.
Amp Env Release	This affects the Release time of the Amp Envelope.
LFO1 Rate	This affects the rate of LFO1.
Spread	This affects the Spread amount.
Pan	This affects the Pan control.

Modulation Bus Destination parameters.

6. **Set the Amount for the second Destination (if desired) by turning the corresponding Amount knob, or by clicking and dragging vertically in the Amount box for the second destination.**
7. **If desired, select a second Destination parameter by click-holding the blue arrow symbol to the right of the corresponding Destination box, and dragging to the desired control on the panel.**
8. **If desired, click the Scale box and select a Scale parameter.**
   The available Scale parameters are the same as the Source parameters, see "Modulation Bus Source parameters.".
9. **Turn the Scale Amount knob, or click the Amount box to the left of the Scale box and move the mouse pointer up or down to set a Scale Amount value.**
   Both positive and negative Scale Amount values can be set (+/- 100%). If you, for example, are using the Mod Wheel as Scale parameter and don't want any modulation when the Mod Wheel is set to zero, set the Scale Amount parameter to 100%. Then, there will be no effect when the Mod wheel is set to zero, and full modulation when the Mod Wheel is all the way up.

- **How much modulation will be applied when the Scale parameter is set to maximum is governed by the to Destination Amount parameter(s).**
- **How much the Scale parameter controls the modulation is set with the Scale Amount parameter.**

→ **To clear an assigned Source or Destination, hold down [Ctrl](Win) or [Cmd](Mac) and click the Source or Destination box. Alternatively, click the Source or Destination box and select "Off" from the list.**

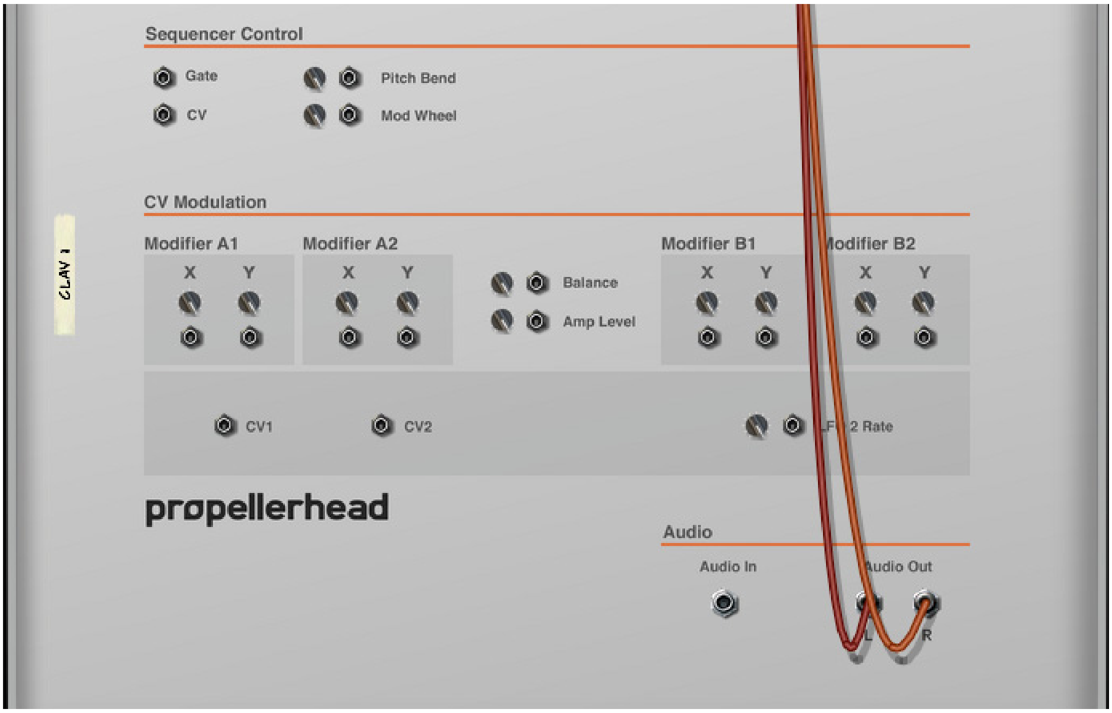
→ **To reset an Amount value to 0, hold down [Ctrl](Win) or [Cmd](Mac) and click the Amount box or knob.**

Global output controls

Volume

This is the main stereo output volume control.

Connections



! **Remember that CV connections are NOT stored in the Parsec 2 patches! If you want to store CV connections between devices, put them in a Combinator device and save the Combi patch.**

Sequencer Control inputs

The Sequencer Control CV and Gate inputs allow you to play Parsec 2 from another CV/Gate device (typically a Matrix or an RPG-8). The signal to the CV input controls the note pitch, while the signal to the Gate input delivers note on/off along with velocity. There are also inputs for modulating the Pitch Bend and Mod Wheel parameters.

CV Modulation and Engine Modulation inputs

These control voltage (CV) inputs (with associated trim pots in some cases) can modulate the X and Y parameters of the Modifiers of the two Sound Engines. There are also inputs for modulating the global parameters Amp Level, LFO 2 Rate and Balance - plus two assignable CV inputs which can be used for modulating assigned Destination parameters in the Modulation Bus section, see “The Modulation Bus section”.

Audio Input

Here you can route an external audio signal to be used for vocoder effects with the Audio In algorithm, see “Audio In”.

Audio Output

These are the main audio outputs. When you create a new Parsec 2 device, these outputs are auto-routed to the first available channel in the Reason/Reason Essentials main mixer.