Logic Pro X Instruments

For OS X

Apple
1 Infinite Loop
Cupertino, CA 95014
408-996-1010
www.apple.com

019-2552

Contents

Drum Kit Designer

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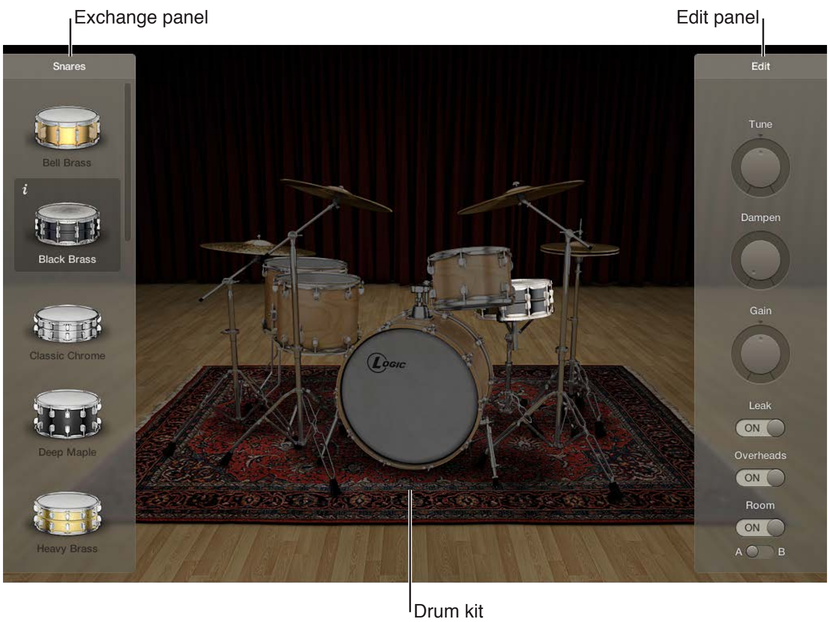
Drum Kit Designer overview

Drum Kit Designer lets you build custom drum kits from a wide selection of drum and percussion sounds. It also provides controls that change sound characteristics and the level of each piece in your kit.

Further settings allow you to use different microphones and rooms to enhance Producer kits. Producer kits are identified in the Library by a "+" at the end of the patch name. See "Add drummers to a project" in Logic Pro Help for information on Producer kits.

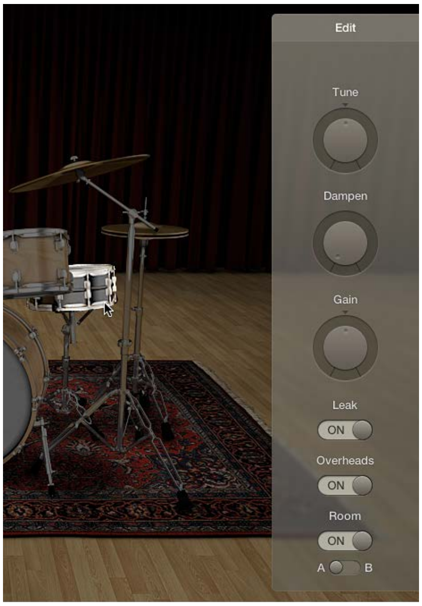


Drum Kit Designer's interface is divided into the following main areas.

- *Drum kit:* Click a drum kit piece to preview its sound and to open the Edit panel and the Exchange panel if exchange pieces are available for that drum type.
- *Exchange panel:* Shows all drums that are available for exchange (you may need to scroll).
- *Edit panel:* Shows settings that change sound characteristics.

Drum Kit Designer Edit panel

The Edit panel is used to change sound characteristics and the level of each piece in your kit.



Edit panel parameters

- *Tune knob and field:* Rotate to adjust the pitch.
- *Dampen knob and field:* Rotate to adjust the sustain.
- *Gain knob and field:* Rotate to adjust the volume.
- *Leak switch (Producer kits only):* Drag to On to include the sound in the mic of the other kit pieces.
- *Overheads switch (Producer kits only):* Drag to On to include the drum kit's overhead mic in the sound.
- *Room switch (Producer kits only):* Drag to choose between rooms A and B or to turn off the room emulation.

Use Drum Kit Designer

Drum Kit Designer shows a 3D representation of the drum kit for the currently loaded patch.

For all kits, you can preview the drums, edit the pitch, sustain, and volume of each drum kit piece, and exchange the kick and snare drums. When working with Producer kits, you can additionally exchange toms, cymbals, and hi-hat. Producer kits let you turn different microphones, such as overheads or room mics, on or off.

Note: Producer kits and some drums are only available after you download additional content.

Drum Kit Designer also has additional parameters for adjusting the gain of other instrument pieces, such as shaker, cowbell, and so on. See Drum Kit Designer extended parameters.

Preview a drum or percussion piece

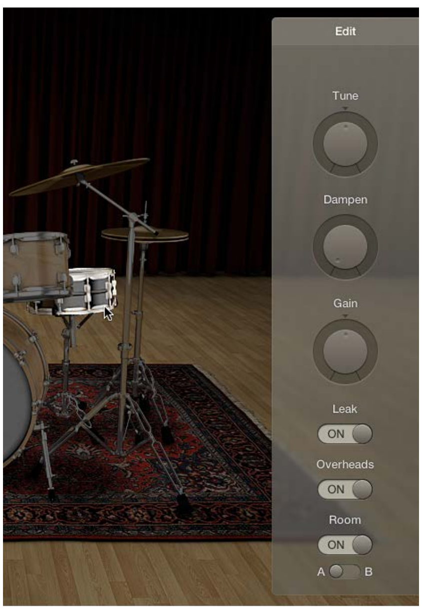
- Click a drum or percussion piece.

The first time you click any drum or percussion piece after opening the plug-in, one or two panes open. You can exchange individual sounds in the Exchange panel to the left and can edit individual drum or percussion piece settings in the Edit panel to the right.

Adjust kit piece settings (all kits)

1. Click a drum or percussion piece.

   The Edit panel opens to the right.



- *Toms:* Click the tab for the tom you want to edit, or click the All tab to adjust the tone of all toms.
- *Cymbals:* Click the tab for the crash cymbal you want to edit, or click the All tab to adjust the tone of both crash cymbals. The ride cymbal can be edited directly.
- *Kicks and snares:* There are no tabs, so make your adjustments with the controls.

2 To adjust settings, do any of the following:

- *To adjust the pitch:* Drag the Tune control vertically, or double-click the field and enter a new value.
- *To adjust the sustain:* Drag the Dampen control vertically, or double-click the field and enter a new value.
- *To adjust the volume:* Drag the Gain control vertically, or double-click the field and enter a new value.

3 To close open panels, click anywhere in the plug-in window background.

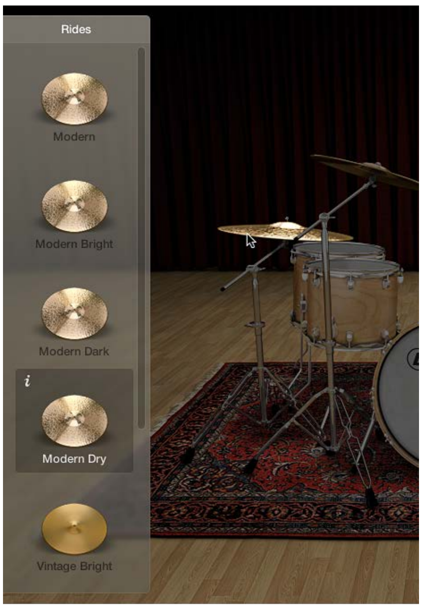
Exchange a kit piece

For all kits, you can exchange kick and snare. When working with Producer kits, you can additionally exchange toms, cymbals, and hi-hat.

Note: Producer kits and some drums are only available after you download additional content.

1 Click a drum or percussion piece.

The Exchange panel opens to the left if exchange pieces are available for that kit piece.



2 Click the Info button of a selected piece to view its description.

3 Click the kit piece that you want to exchange in the Exchange panel. You may need to scroll in order to find the piece you want to use.

The piece is exchanged and the respective drum sound is loaded.

Note: The toms and the crash cymbals can only be exchanged as a group.

4 To close the panels, click anywhere in the plug-in window background.

Adjust mic settings (Producer kits only)

In the Edit panel, do any of the following:

- *To include the sound in the mic of the other kit pieces:* Turn on the Leak switch.

This turns microphone bleed on or off, where the sound of a kit piece is picked up by the different mics from other kit pieces.

- *To include the kit piece's overhead mic in the sound:* Turn on the Overheads switch.

This turns the overhead mic for the selected kit piece on or off.

- *To choose a room emulation to use with the sound:* Choose between rooms A and B. You can also turn off the room microphones.

Rooms A and B determine which room mic setup is used with the kit piece.

Drum Kit Designer extended parameters

Drum Kit Designer provides additional parameters that are accessed by clicking the disclosure triangle at the lower left.

The Input Mapping pop-up menu lets you choose different mappings that provide enhanced control of HiHats. The maps also change the way Drum Kit Designer sounds are assigned across the MIDI note range. See Drum Kit Designer mappings.

Extended parameters

- *Input Mapping pop-up menu:* Choose a keyboard mapping mode.
  - *GM:* Drums are mapped to the GM standard.
  - *GM + ModWheel controls HiHat opening level:* The keyboard Mod Wheel is mapped for hi-hat control. Additional sounds are also mapped to keyboard zones above and below the standard GM note mapping range.
  - *V-Drum:* Drums are mapped to work with V-Drum hi-hat, cymbal, and drum triggers.
- *Gain sliders:* Drag the slider (or drag vertically in the field) to adjust the level of the corresponding sound (if available in the kit).
  - Shaker Gain
  - Tambourine Gain
  - Claps Gain
  - Cowbell Gain
  - Sticks Gain

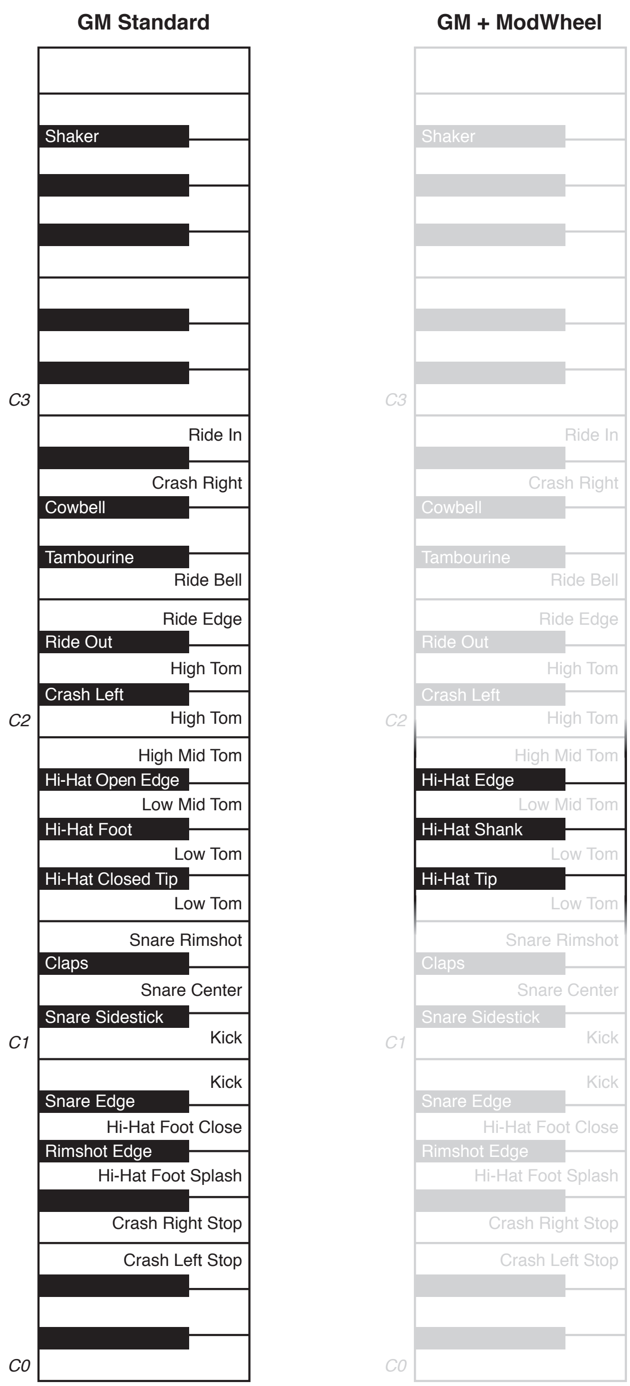
Drum Kit Designer mappings

Drum Kit Designer is compatible with the GM standard. You can also choose GM+, which maps the keyboard ModWheel for hi-hat control. This means that you can use the keyboard ModWheel to adjust the degree to which the hi-hat opens and closes during the drum performance.

Drum Kit Designer is also compatible with the V-Drum standard.

The image shows how drum sounds are remapped when different modes are chosen with the Input Mapping pop-up menu in the extended parameters.

Note: A number of alias drums sounds are included for GM compatibility purposes.



ES1

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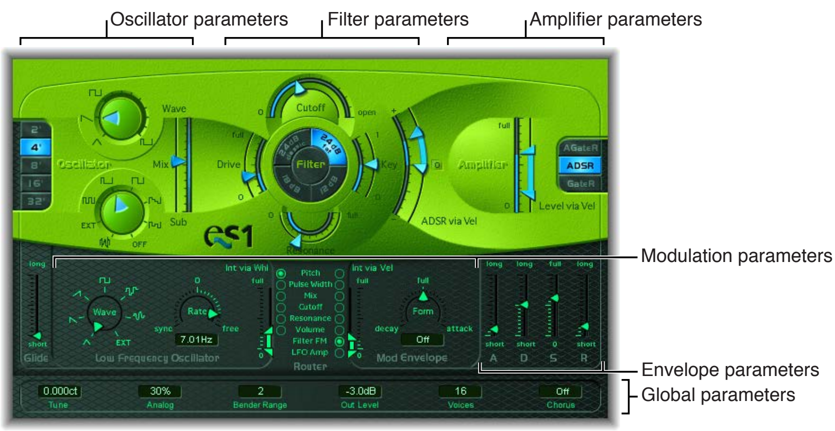
ES1 overview

ES1 emulates the circuits of analog synthesizers in a simple, streamlined interface.

ES1 produces sounds using subtractive synthesis. It provides an oscillator and sub-oscillator that generate harmonically rich waveforms. You *subtract* (cut, or filter out) portions of these waveforms and reshape them to create new sounds. The ES1's tone-generation system also provides flexible modulation options that make it easy to create punchy basses, atmospheric pads, biting leads, and sharp percussion.

If you're new to synthesizers, see Synthesizer basics overview on page 471, which will introduce you to the terminology and give you an overview of different synthesis systems and how they work.



ES1 is divided into six main areas.

- *Oscillator parameters:* Located in the upper left, the oscillators generate the basic waveforms that form the basis of your sound. See ES1 oscillator parameters overview on page 21.
- *Global parameters:* Located in the bottom green/gray strip, global sound control parameters are used to assign and adjust global tuning, activate the in-built chorus, and so on. You can use the chorus to color or thicken the sound. See ES1 global parameters on page 23.
- *Filter parameters:* Located in the upper-middle section with the circular Filter area as well as the Drive and Key scaling parameters, the filter is used to contour the waveforms sent from the oscillators. See ES1 filter parameters overview on page 24.
- *Amplifier parameters:* Located in the upper right, the amplifier parameters allow you to fine-tune the behavior of your sound's level. See ES1 amplifier parameters on page 26.
- *Envelope parameters:* Located to the right in the dark green/gray area, the ADSR sliders are used to control both filter cutoff and the amplifier level over time. See ES1 envelope parameters overview on page 27.

- *Modulation parameters:* Located to the left and middle in the dark green/gray area, the modulation sources, modulation router, modulation envelope, and amplitude envelope are used to modulate the sound in a number of ways. See ES1 modulation parameters overview on page 29.

ES1 oscillator parameters

ES1 oscillator parameters overview

ES1 includes a primary oscillator and a sub-oscillator. The primary oscillator generates a waveform that is sent to other parts of the synthesizer for processing or manipulation. The sub-oscillator generates a secondary waveform one or two octaves below that of the primary oscillator.

Oscillator parameters

- *Wave knob:* Rotate to select the waveform of the primary oscillator, which is responsible for the basic color of the tone. See ES1 oscillator waveforms on page 22.
- *Mix slider:* Drag to set the level relationship between the primary and sub-oscillator signals. (When the sub-oscillator is switched off, its output is completely removed from the signal path.)
- *Sub knob:* Rotate to generate square, pulse, and white-noise waveforms with the sub-oscillator. The sub-oscillator also allows you to route a side-chain signal through the ES1 synthesizer engine. See Use the ES1 sub-oscillator on page 22.
- *2′, 4′, 8′, 16′, and 32′ buttons:* Click to transpose the pitch of the oscillators up or down by octaves. The lowest setting is 32 feet and the highest is 2 feet. The use of the term *feet* to determine octaves comes from the measurements of organ pipe lengths. The longer and wider the pipe, the deeper the tone.

Modulate ES1 pulse width

- Rotate the Wave knob to a position between the square wave and pulse wave symbols.

The pulse width can also be automatically modulated in the modulation section (see Use the ES1 router on page 29). Modulating the pulse width with a slowly cycling LFO, for example, allows periodically mutating, fat bass sounds.

ES1 oscillator waveforms

The table outlines the basic tones of the oscillator waveforms—how they affect your synthesizer sound.

Waveform	Basic tone	Comments
Sawtooth	Warm and even	Useful for strings, pads, bass, and brass sounds
Triangle	Sweet sounding, softer than sawtooth	Useful for flutes, pads
Square	Hollow and "woody" sounding	Useful for basses, clarinets, and oboes
Pulse	"Nasal" sounding	Great for reed instruments, synth blips, basses

Use the ES1 sub-oscillator

The ES1 sub-oscillator is used to make your sound richer. Its pitch matches the frequency of the main oscillator. You can choose from the following sub-oscillator waveform options:

- A square wave that plays one or two octaves below the frequency of the primary oscillator
- A pulse wave that plays two octaves below the frequency of the primary oscillator
- Variations of these waveforms, with different mixes and phase relationships, resulting in various sounds
- White noise, which is useful for creating percussion sounds as well as wind, surf, and rain sounds
- OFF, which allows you to disable the sub-oscillator
- EXT, which allows you to run an external channel strip signal through the ES1 synthesizer engine, by using a side chain

Process a channel strip signal through the ES1 synthesizer engine

1. Set the Sub knob to EXT.
2. Choose the side-chain source channel strip from the Side Chain pop-up menu in the upper-right corner of ES1.

ES1 global parameters

The global parameters affect the overall sound, or behavior, of the ES1 and are found primarily in the strip that spans the lower edge of the ES1 interface. The Glide slider is displayed above the left end of the strip.

Global parameters

- *Glide slider:* Drag to set the amount of time it takes to slide between the pitches of each triggered note. The Glide trigger behavior depends on the value set in the Voices field (see below).
- *Tune field:* Drag to tune the instrument in cents. One cent is 1/100th of a semitone.
- *Analog field:* Drag to slightly, and randomly, change the pitch of each note and the cutoff frequency. This emulates the oscillator detuning and filter fluctuations of polyphonic analog synthesizers, due to heat and age.

  If you set the Analog parameter to 0%, the oscillator cycle start points of all triggered voices are synchronized. This can be useful for percussive sounds, when you want to achieve a sharper attack characteristic.

  If you set the Analog parameter higher than 0%, the oscillators of all triggered voices can cycle freely. Use higher values if you want a warm, analog type of sound—where subtle sonic variations occur for each triggered voice.
- *Bender Range field:* Drag to set the sensitivity of the pitch bender, in semitone steps.
- *Neg Bender Range slider (Extended Parameters area):* Drag to set the negative (downward) pitch bend range in semitone steps. The default value is Pos PB (positive pitch bend), which essentially means that there is no downward pitch bend available. (Click the disclosure triangle at the lower left of the ES1 interface to access the Extended Parameters area.)
- *Out Level field:* Drag to set the ES1 master volume.
- *Voices field:* Drag to set the maximum number of notes that can be played simultaneously—up to 16 voices.

  When Voices is set to Legato, the ES1 behaves like a monophonic synthesizer—with single trigger and fingered portamento engaged. This means that if you play legato, a portamento—glide from one note to the next—will happen. If you release each key before pressing a new one, the envelope is not triggered by the new note, and there is no portamento. Use this feature to create pitch bend effects, without touching your keyboard's pitch bender, by choosing a high Glide parameter value when using the Legato setting.
- *Chorus field:* Click to choose a classic stereo chorus effect, an ensemble effect, or to disable the effects processor.
  - Off disables the built-in chorus circuit.
  - C1 is a typical chorus effect.
  - C2 is a variation of C1 and is characterized by a stronger modulation.
  - Ens(emble) uses a more complex modulation routing, creating a fuller and richer sound.

ES1 filter parameters

ES1 filter parameters overview

This section outlines the filter parameters of the ES1.

Filter parameters

- *Cutoff slider:* Drag to set the cutoff frequency of the ES1's lowpass filter.
- *Resonance slider:* Drag to cut or boost the portions of the signal that surround the frequency defined by the Cutoff parameter. Boost can be set so intensively that the filter begins to oscillate by itself (see Drive the ES1 filter to self-oscillate on page 25).

  Tip: You can simultaneously adjust the cutoff frequency and resonance parameters by dragging vertically (cutoff) or horizontally (resonance) on the word *Filter*, found in the center of the black circle.

- *Slope buttons:* The lowpass filter offers four different slopes of band rejection above the cutoff frequency. Click one of the buttons to choose a slope (amount of rejection, expressed in decibels (dB) per octave):
  - *24 dB classic:* Mimics the behavior of a Moog filter. Turning up the resonance results in a reduction of the low end of the signal.
  - *24 dB fat:* Compensates for the reduction of low frequency content caused by high Resonance values. This resembles the behavior of an Oberheim filter.
  - *12 dB:* Provides a soft, smooth sound that is reminiscent of the early Oberheim SEM synthesizer.
  - *18 dB:* Resembles the filter sound of Roland's TB-303.
- *Drive slider:* Drag to change the behavior of the Resonance parameter, which eventually distorts the sound of the waveform. Drive is actually an input level control, which allows you to overdrive the filter.

- *Key slider:* Drag to set the effect that keyboard pitch (the note number) has on filter cutoff frequency modulation.
  - If Key is set to zero, the cutoff frequency does not change, no matter which key you strike. This makes the lower notes sound comparatively brighter than higher notes.
  - If Key is set to maximum, the filter follows the pitch, resulting in a constant relationship between cutoff frequency and pitch. This mirrors the properties of many acoustic instruments, where higher notes sound both brighter in tone and higher in pitch.
- *ADSR via Vel sliders:* Drag to determine how note velocity affects modulation of the filter cutoff frequency with the envelope generator. See ES1 envelope parameters overview on page 27.
- *Filter Boost button (Extended Parameters area):* Turn on to increase the output of the filter by approximately 10 decibels. The filter input has a corresponding decrease of approximately 10 decibels, maintaining the overall level. This parameter is particularly useful when applying high Resonance values. See Drive the ES1 filter to self-oscillate. (Click the disclosure triangle at the lower left of the ES1 interface to access the Extended Parameters area.)

Drive the ES1 filter to self-oscillate

If you increase the filter Resonance parameter to higher values, the filter begins to internally feed back and, as a consequence, begins to self-oscillate. This results in a sine oscillation—a sine wave—that is actually audible.

You can make the ES1 filter output a sine wave by following the steps below. This lets you play the filter-generated sine wave with the keyboard.

Output a sine wave from the filter

1. Switch the Sub knob to Off.
2. Drag the Mix slider to the very bottom (Sub).
3. Drag the Resonance slider to the maximum position (full).
4. If you want, click the disclosure triangle at the lower left to open the extended parameters, then click the Filter Boost button.

   Filter Boost increases the output of the filter by approximately 10 decibels, making the self-oscillation signal much louder.

ES1 amplifier parameters

The parameters in the ES1 Amplifier section allow you to fine-tune the behavior of your sound's level. These are separate from the global Out Level parameter, which acts as the ES1's master volume control. See ES1 global parameters on page 23.

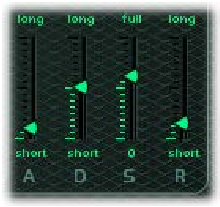
Amplifier parameters

- *Level via Vel slider:* Drag to determine how note velocity affects the synthesizer level. The greater the distance between the arrows (indicated by the blue bar), the more the volume is affected by incoming velocity messages.
  - Drag the upper arrow to set the level when you play hard (velocity=127).
  - Drag the lower arrow to set the level when you play softly (velocity=1).
  - To simultaneously adjust the modulation range and intensity, drag the blue bar—between the arrows—and move both arrows at once.
- *Amplifier envelope selector buttons:* Click one of the buttons—AGateR, ADSR, or GateR—to determine the ADSR envelope generator used for control of the amplifier envelope. See ES1 envelope parameters overview on page 27.

ES1 envelope parameters

ES1 envelope parameters overview

ES1 provides an attack, decay, sustain, and release (ADSR) envelope that can shape filter cutoff and the level of the sound over time.



Envelope Parameters

- *A(ttack) slider:* Drag to set the time it takes for the envelope to reach the initial desired level.
- *D(ecay) slider:* Drag to set the time it takes for the envelope to fall to the sustain level, following the initial attack time.
- *S(ustain) slider:* Drag to set the sustain level, which is held until the key is released.
- *R(elease) slider:* Drag to set the time it takes the envelope to fall from the sustain level to a level of 0.

ES1 filter cutoff envelope modulation

The envelope generator modulates the filter cutoff frequency over the course of a note's duration. The modulation intensity—and response to velocity information—is set by the arrows on the ADSR via Vel slider in the Filter section.

ADSR via Vel slider

The modulation range is determined by the two arrows.

- The lower arrow indicates the minimum amount of modulation.
- The upper arrow indicates the maximum amount of modulation.
- The blue bar between the arrows shows the dynamic range of this modulation. You can simultaneously adjust the modulation range and intensity by dragging the blue bar.

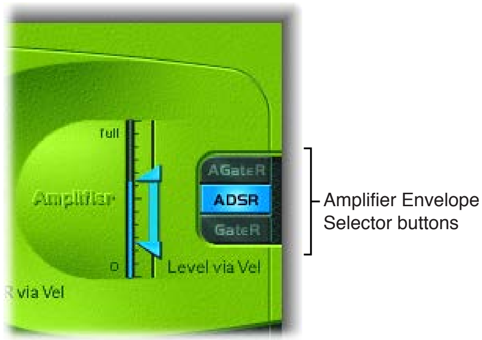
Tip: If you're unfamiliar with these parameters, set the Cutoff parameter to a low value, Resonance to a high value, and move both ADSR via Vel arrows upward. Constantly strike a note on the keyboard while changing the arrows to learn how these parameters work.

ES1 amplifier envelope modulation

The AGateR, ADSR, and GateR buttons in the Amplifier section determine which of the ADSR envelope controls affect the amplifier envelope. All ADSR parameters remain active for the filter.

The letters *A, D, S,* and *R* refer to the attack, decay, sustain, and release phases of the envelope (see ES1 envelope parameters overview on page 27).

Gate refers to a control signal used in analog synthesizers that is sent to an envelope generator when a key is pressed. As long as an analog synthesizer key is pressed, the gate signal maintains a constant voltage. When Gate is used as a modulation source in the voltage-controlled amplifier (instead of the envelope), it creates an organ-type envelope without any attack, decay, or release phase—in other words, an even, sustained sound.



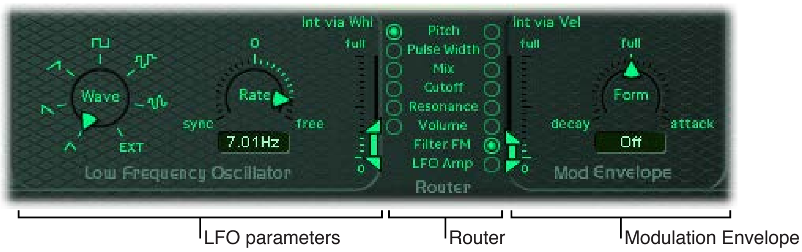
The ES1 amplifier envelope selector buttons have the following effect on played notes:

- *AGateR:* The Attack and Release sliders of the ADSR envelope control the attack and release phases of the sound. In-between these phases, the Gate control signal is used to maintain a constant level while a note is held. As soon as you release the key, the release phase begins. The Decay and Sustain sliders of the ADSR Envelope have no impact on the sound's level.
- *ADSR:* The standard operating mode of most synthesizers, where the level of the sound over time is controlled by the ADSR Envelope.
- *GateR:* The Gate control signal is used to maintain a constant level while a note is held. As soon as you release the key, the release phase begins. The Attack, Decay, and Sustain sliders of the ADSR Envelope have no impact on the sound's level.

ES1 modulation

ES1 modulation parameters overview

ES1 offers a number of simple yet flexible modulation routing options. You use modulation to add animation to your sound over time, making it more interesting, lively, or realistic. A good example of this type of sonic animation is the vibrato used by orchestral string players.

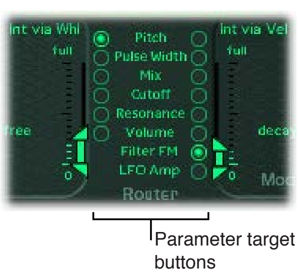


Modulation parameters

- *LFO parameters:* Used to modulate other ES1 parameters. See Use the ES1 LFO on page 30.
- *Router:* Enables you to choose the ES1 parameters that are modulated. See Use the ES1 router on page 29.
- *Modulation Envelope:* A dedicated modulation control source that can be used to control various ES1 parameters, or it can be used to control the LFO level. See Use the ES1 modulation envelope on page 31.

Use the ES1 router

The router determines the ES1 parameters (*targets*) that are modulated by the LFO and by the modulation envelope. The buttons in the left column set the target for LFO modulation. The buttons in the right column set the target for the modulation envelope.

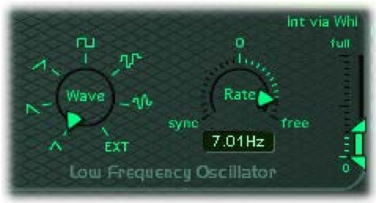


Router parameters

- *Pitch buttons:* Click to modulate the pitch—the frequency—of the oscillators.
- *Pulse Width buttons:* Click to modulate the pulse width of the pulse wave.
- *Mix buttons:* Click to modulate the mix between the primary oscillator and the sub-oscillator.
- *Cutoff buttons:* Click to modulate the cutoff frequency of the filter.
- *Resonance buttons:* Click to modulate the resonance of the filter.
- *Volume buttons:* Click to modulate the main volume.
- *Filter FM button (modulation envelope only):* Click to use the triangle wave of the oscillator to modulate filter cutoff frequency. This modulation can result in a pseudo-distortion of the sound, or it can create metallic, FM-style sounds. The latter occurs when the only signal you can hear is the self-oscillation of the resonating filter (see Drive the ES1 filter to self-oscillate on page 25).
- *LFO Amp (modulation envelope only):* Click to modulate the overall amount of LFO modulation.

Use the ES1 LFO

The LFO (low frequency oscillator) generates an adjustable, cyclic waveform that you can use to modulate other ES1 parameters.



LFO parameters

- *Wave knob:* Rotate to set the LFO waveform. Each waveform has its own shape, providing different types of modulation.
  - You can choose the following waveforms: triangle; ascending and descending sawtooth; square wave; sample & hold (random); and a lagged, smoothly changing random wave.
  - You can also choose EXT to assign a side-chain signal as a modulation source. Choose the side-chain source channel strip from the Side Chain pop-up menu in the upper-right corner of ES1.
- *Rate slider and field:* Drag to set the speed—the frequency—of the LFO waveform cycles.
  - If you set values to the right of 0, the LFO phase runs freely.
  - If you set values to the left of 0, the LFO phase is synchronized with the tempo of the host application—with phase lengths adjustable between 1/96 bar and 32 bars.
  - When set to 0, the LFO outputs at a constant, full level, which allows you to manually control the LFO speed with your keyboard's modulation wheel. This can be useful, for example, if you want to change the pulse width by moving your keyboard's modulation wheel. You would choose the pulse width as the LFO modulation target, using a button in the left router column, and set the modulation intensity range using the Int via Whl slider.
- *Int via Whl slider:* The upper arrow defines the intensity of LFO modulation if the modulation wheel is set to maximum. The lower arrow defines the amount of LFO modulation if the modulation wheel is set to 0. The distance between the arrows—shown as a green bar—indicates the range of your keyboard's modulation wheel.

  You can simultaneously adjust the modulation range and intensity by dragging the green bar, thus moving both arrows at once. Note that as you do so, the arrows retain their relative distance from each other.

Use the ES1 modulation envelope

The modulation envelope can directly modulate the parameter chosen in the router. It determines the time it takes for the modulation to fade in or fade out. At its center position (click Full), modulation intensity is static—no fade-in or fade-out occurs. When set to its full value, modulation intensity is at a constant level.

The modulation envelope allows you to set either a percussive type of decay envelope by choosing low values or an attack type of envelope by choosing high values.

Modulation envelope parameters

- *Form slider and field:* Drag to set a fade-in (attack) or fade-out (decay) time for the modulation. When set to the full position, the modulation envelope is turned off.
- *Int via Vel sliders:* The top arrow sets the upper limit for the modulation envelope—for the hardest keystrike (velocity = 127). The bottom arrow sets the lower limit—for the softest keystrike (velocity = 1). The green bar between the arrows displays the impact of velocity sensitivity on the intensity of the modulation envelope.

  You can simultaneously adjust the modulation range and intensity by dragging the green bar, thus moving both arrows at once. Note that as you do so, the arrows retain their relative distance from each other.

Modulate a parameter with velocity

1. Select a modulation target, such as Pulse Width, from the right column of the router.
2. Set the Form slider to full, and adjust the Int via Vel parameter as needed.

This results in a velocity-sensitive modulation of the oscillator pulse width.

More interestingly, you can directly control the LFO level if you click the LFO Amp(litude) button in the right column of the router.

Fade the LFO modulation in or out

- *To fade in the LFO modulation:* drag the Form slider to a positive value—toward attack. The higher the value, the longer it takes for you to hear the modulation.
- *To fade out the LFO modulation:* drag the Form slider to a negative value—toward decay. The lower the value—closer to decay—the shorter the fade-out time is.

LFO control with envelopes is most often used for delayed vibrato, a technique many instrumentalists and singers employ to intonate longer notes.

Set up a delayed vibrato

1. Drag the Form slider to the right—toward attack.
2. Select Pitch as the LFO target in the left column of the router.
3. Use the Wave knob to select the triangular wave as the LFO waveform.
4. Drag the Rate field to an LFO rate of about 5 Hz.
5. Drag the upper Int via Wheel arrow to a low value, and the lower arrow to 0.

ES1 MIDI controllers

ES1 responds to the following MIDI continuous controller numbers (CC).

Controller number	Parameter name
12	Oscillator pitch buttons
13	Oscillator waveform
14	Mix slider
15	Waveform of sub-oscillator
16	Drive slider
17	Cutoff slider
18	Resonance slider
19	Slope buttons
20	ADSR via Vel (lower slider)
21	ADSR via Vel (upper slider)
22	Attack slider
23	Decay slider
24	Sustain slider
25	Release slider
26	Key slider
27	Amplifier Envelope Selector buttons
28	Level via Velocity (lower slider)
29	Level via Velocity (upper slider)
30	Chorus parameter
31	Modulation envelope target
102	Modulation envelope form slider
103	Modulation envelope: Int via Vel parameter (lower slider)
104	Modulation envelope: Int via Vel parameter (upper slider)
105	LFO rate
106	LFO waveform
107	LFO modulation target
108	LFO: Int via Whl (lower slider)
109	LFO: Int via Whl (upper slider)
110	Glide slider
111	Tune parameter
112	Analog parameter
113	Bender Range parameter
114	Out Level parameter
115	Voices parameter

ES2

3

ES2 overview

ES2 combines subtractive synthesis with elements of FM and wavetable synthesis to help you generate an extraordinary variety of sounds. This makes it the perfect choice for creating powerful pads, evolving textures, rich basses, or synthetic brass.

If you're new to synthesizers, see Synthesizer basics overview on page 471, which introduces you to the fundamentals and terminology of different synthesis systems.

The three oscillators of the ES2 provide classic analog synthesizer waveforms (including noise) and 100 single-cycle waveforms, known as Digiwaves. This raw material forms the basis for sounds that range from fat analog to harsh digital sounds, or hybrids of the two. You can also cross-modulate oscillators, making it easy to create FM-style sounds. Further options include the ability to synchronize and ring-modulate the oscillators or to mix a sine wave directly into the output stage, to thicken the sound.

ES2 features a flexible modulation router that offers up to ten simultaneous (user-defined) modulation routings. Further modulation options include the unique Planar Pad—which provides control of two parameters on a two-dimensional grid. The Planar Pad itself can be controlled by the sophisticated Vector Envelope. This is a multipoint, loop-capable envelope that makes it easy to create complex, evolving sounds.

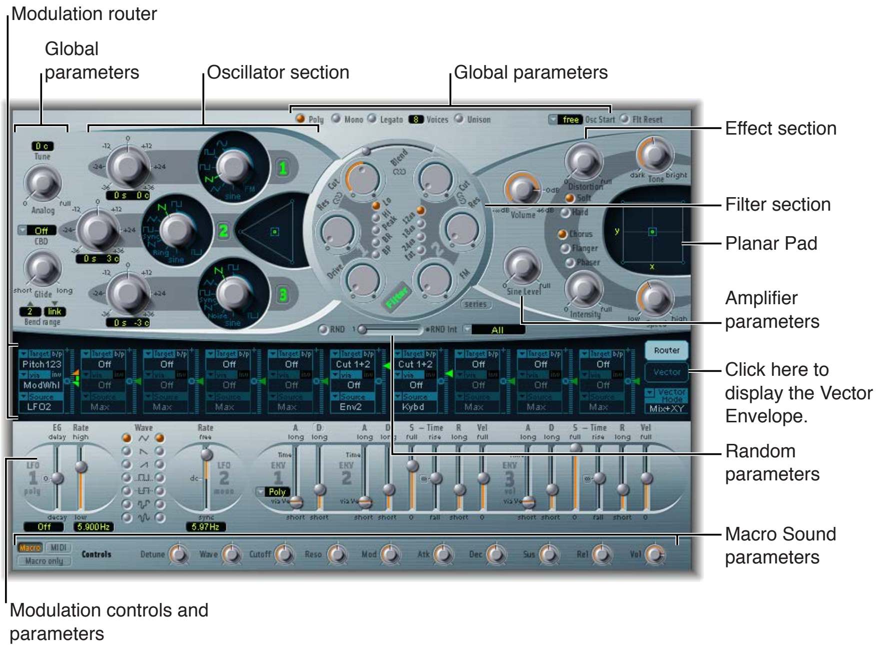
Lastly, Distortion, Chorus, Phaser, and Flanger effects are built into the ES2.

If you want to begin experimenting right away, there are a number of settings to try. There are also two tutorials that provide tips and information, and invite you to explore the ES2. See ES2 sound design from scratch overview on page 94 and ES2 sound design with templates on page 103.

Note: You will find tasks that cover the use of parameters as modulation targets or sources throughout these pages. This underlines one of ES2's greatest strengths—namely, the vast modulation possibilities it offers. Follow the steps in these tasks to create expressive, evolving sounds. See ES2 modulation overview on page 58.

ES2 interface

ES2's graphical interface is divided into the following main areas.



- *Oscillator section:* The oscillator parameters are shown in the upper-left area of the ES2 interface. The Triangle is used to set the mix relationships between the three oscillators. See ES2 oscillator parameters overview on page 36.
- *Global parameters:* A number of related global parameters that directly influence the overall output of the ES2, such as Tune, are found to the left of the oscillators, and above the amplifier and filter parameters. See Global parameters overview on page 46.
- *Filter section:* The circular area houses the filter section, including the Drive and Filter FM parameters. See ES2 filter overview on page 49.
- *Amplifier parameters:* The area at the top right contains the output parameters, where you can set the overall volume of the ES2, and add a sine signal at the output stage. See Use ES2's dynamic stage on page 57.
- *Modulation router or Vector Envelope:* The dark strip across the center of the ES2 interface is shared by the modulation router and the Vector Envelope. Use the buttons at the right end of this section to switch between the two.
  - The router links modulation sources, such as the envelopes and other parameters shown in the lower portion of the interface, to modulation targets, such as the oscillators and filters. See Use the modulation router on page 59.
  - The Vector Envelope is a flexible, powerful envelope generator that provides extensive control over your sound. See Use the Vector Envelope on page 69.

- *Modulation controls and parameters:* The area immediately below the router is where you can assign and adjust the modulation generator parameters (such as LFO and envelope controls). See ES2 modulation overview on page 58.
- *Planar Pad:* The square area at the top right is a two-dimensional controller known as the Planar Pad. The Planar Pad facilitates the simultaneous manipulation of two assignable parameters, and can be controlled with the mouse, another controller, or the Vector Envelope. See Use the Planar Pad on page 76.
- *Effect section:* The built-in effect-processing options are found to the right of the output parameters. See ES2 integrated effects processor on page 87.
- *Macro and MIDI controller parameters:* The area shown on the thin, gray strip at the bottom can display either Macro parameters or MIDI controller assignments. The preassigned macro sound parameters are perfect for quick tweaks to the ES2's sound (and that of ES2-based GarageBand instruments). You can reassign MIDI control numbers for these parameters. See ES2 macro and controller assignment overview on page 89.

ES2 sound sources

ES2 oscillator parameters overview

ES2 oscillators are used to generate one or more waveforms. This signal is then sent to other portions of the synthesizer engine for shaping, processing, or manipulation.

- Oscillators 2 and 3 are almost identical to each other, but they differ from oscillator 1.
- Oscillator 1 can be frequency modulated by oscillator 2, for FM synthesis sounds.
- Oscillators 2 and 3 can be synchronized to, or ring modulated with, oscillator 1. They also have rectangular waves with either user-defined fixed pulse widths or pulse width modulation (PWM) features.
- You can use the modulation router to simultaneously change the pulse widths of rectangular waves generated by oscillator 1 and the synchronized and ring-modulated rectangular waves of oscillators 2 and 3.

Oscillator parameters

- *Oscillator on/off buttons:* The number to the right of each oscillator activates or deactivates each oscillator independently. A green numeric button indicates an active oscillator. A gray numeric button denotes an inactive oscillator. Deactivating an oscillator saves computer processing power.
- *Wave knobs:* Rotate to choose the waveform that an oscillator generates. The waveform is responsible for the basic tonal color. See ES2 basic oscillator waveforms on page 37.
- *(Coarse) Frequency knobs:* Rotate to set the oscillator's pitch, in semitone steps, over a range of ±3 octaves. Because an octave consists of 12 semitones, the ±12, 24, and 36 settings represent octaves.
- *(Fine) Frequency value fields:* Fine-tune the oscillator frequency (pitch). The left numbers show the semitone *s* setting, and the right numbers show the cent *c* setting (1 cent = 1/100th semitone). For example, an oscillator with the value 12 s 30 c sounds an octave (12 semitones) and 30 cents higher than an oscillator with the value 0 s 0 c. Drag vertically to adjust each value.
- *Oscillator Mix (Triangle):* Move the pointer in the Triangle to cross-fade (set the level relationships) between the three oscillators. See Balance ES2 oscillator levels on page 44.

ES2 basic oscillator waveforms

All ES2 oscillators output a number of standard waveforms—sine, pulse, rectangular, sawtooth, and triangular waves—or, alternately, any of 100 Digiwaves (see Use ES2 Digiwaves on page 41). The following table covers the basic waveforms:

Waveform	Basic tone	Comments
Pulse/Rectangular	Nasal sounding	Great for reed instruments, synth blips, and basses
Square	Hollow and woody sounding	Useful for basses, clarinets, and oboes. The pulse width of (oscillator 2 and 3) square waveforms can be smoothly scaled between 50% and the thinnest of pulses.
Sawtooth	Warm and even	Useful for strings, pads, bass, and brass sounds
Triangle	Sweet sounding, softer than sawtooth	Useful for flutes and pad sounds
Sine	A pure tone	The sine wave of oscillator 1 can be frequency modulated by oscillator 2. This kind of modulation forms the basis of FM synthesis (see Use frequency modulation in ES2 on page 38).

Oscillators 2 and 3 also offer the selection of:

- A rectangular wave, synchronized to oscillator 1
- A sawtooth wave, synchronized to oscillator 1
- A ring modulator, which is fed by the output of oscillator 1 and a square wave from oscillator 2
- Colored noise for oscillator 3. See Use the ES2 noise generator on page 41.

Oscillator synchronization and ring modulation allow for the creation of very complex and flexible harmonic spectra. The principles behind oscillator synchronization are described in Synchronize ES2 oscillators on page 45. Ring modulation principles are described in Use ring modulation in ES2 on page 40.

Use pulse width modulation in ES2

You can alter the tonal color of rectangular waveforms by scaling the width of waveform pulses to any value. This is known as pulse width modulation.

ES2 pulse width modulation features are extensive. For example, if rectangular waves are chosen for all oscillators, you can simultaneously modulate the pulse width of oscillator 1 and the synchronized pulse waves of oscillator 2 (or the square wave of oscillator 2's ring modulator) and oscillator 3.

Set a basic pulse width in oscillator 2 or 3

- Drag the waveform rotary control that surrounds the Wave knob (see the highlighted area in the image above).

  Only oscillators 2 and 3 allow you to define a base (default) pulse width, prior to any pulse width modulation.

Set up a pulse width modulation (of oscillator 1) in the router

1. Choose a rectangle wave for oscillator 1.
2. In the router, choose Osc1Wave as the target, and LFO1 as the source.
3. Adjust the modulation amount slider (try a value of 0.12).
4. Choose a sine wave for LFO 1.
5. Adjust the LFO 1 Rate (around 0.160 Hz for a slow sweep).

Use frequency modulation in ES2

The principle of frequency modulation (FM) synthesis was developed in the late 1960s and early 1970s by John Chowning. It was popularized by Yamaha's range of DX synthesizers in the 1980s. Although the ES2 can't be compared with the DX series in the discipline of pure FM synthesis, it can achieve some of the signature sounds of these instruments.

In pure FM synthesis, the frequency of one signal generator, or oscillator, is altered (modulated) by another signal generator. Positive values from the second generator increase the frequency of the first generator. Negative values decrease the frequency. In a synthesizer, this type of modulation takes place in the audible range. Depending on the design of the instrument, you can hear the signals of either the first oscillator alone (being modulated by the other oscillator), or both oscillators. The interaction between the two generators alters the waveform signal of the first oscillator and introduces a number of new harmonics. This harmonic spectrum can then be used as the source signal for further sound processing, such as filtering, envelope control, and so on. See Frequency modulation (FM) synthesis on page 492 for further information.

In ES2, the frequency of oscillator 1 (with a sine wave chosen—11 o'clock position for the Wave knob) can be modulated by the output signal of oscillator 2.

- When oscillator 2 outputs a positive signal, the frequency of oscillator 1 increases.
- When oscillator 2 outputs a negative signal, the frequency of oscillator 1 decreases.

The net effect of speeding up or slowing down the frequency of oscillator 1 in each waveform cycle is a distortion of the basic wave shape. This waveform distortion also has the side benefit of introducing a number of new, audible harmonics.

Important: The impact of any frequency modulations you perform depends on *both* the frequency ratio *and* the modulation intensity of the two oscillators.

The "pure" FM synthesis method uses a sine wave for both the first and second signal generator (both oscillator 1 and 2 would be limited to generating a sine wave in ES2 if you stuck with this approach). ES2, however, provides 100 Digiwaves and countless combinations of modulation intensities and frequency ratios that can be used for either oscillator. This provides a vast pool of harmonic spectra and tonal colors for you to experiment with.

Tip: The type of modulation that occurs can vary significantly when different waveforms are chosen for oscillator 2—the modulating oscillator—in particular.

Set the frequency ratio and adjust the modulation intensity

1. Adjust the Frequency (coarse and fine tune) parameter values of one, or both, oscillators.
2. Click (or drag) in the control range between the Sine and FM icons around the oscillator 1 Wave knob.

   This determines the amount, or intensity, of frequency modulation.

Use ring modulation in ES2

Ring modulation is a powerful tool for the creation of inharmonic, metallic, bell-like sounds. The spectra resulting from its use are inharmonic at almost every frequency ratio. The ring modulator is a device that dates back to the early days of the synthesizer.

A ring modulator has two inputs. At the output you hear both the sum and difference frequencies of the input signals. If you ring modulate a sine oscillation of 200 Hz with a sine oscillation of 500 Hz, the output signal of the ring modulator consists of a 700 Hz (sum) and a 300 Hz (difference) signal. Negative frequencies result in a change to the phase polarity of output signals.

Tip: Use sawtooth and rectangular (pulse width modulated) input signals from oscillators 1 and 2, respectively, to create a much more complex output signal. The use of these harmonically rich waveforms results in a number of extra sidebands becoming audible.

Create a ring-modulated sound

1. Set the oscillator 2 Wave knob to the Ring setting.
2. Experiment with different Frequency (main and fine tune) values for one, or both, oscillators.

The oscillator 2 ring modulator is fed with the output signal of oscillator 1 and a square wave, generated by oscillator 2 itself. The pulse width of this square wave can be modulated (see Use pulse width modulation in ES2 on page 38).

Use ES2 Digiwaves

In addition to the basic synthesizer waveforms, all ES2 oscillators provide 100 additional waveforms, called *Digiwaves*. These are very short samples of the attack transients of various sounds and instruments.

Choose a Digiwave

- Set the Wave knob to Sine (6 o'clock position), then do one of the following:
  - Control-click or right-click the Sine label, then choose a waveform from the pop-up menu.
  - Drag the Sine label vertically.
  - To select the Digiwave numerically, Shift-click the Sine label, then type a value.

Use the ES2 noise generator

The sonic palette of oscillator 3 is bolstered by the inclusion of a noise generator, which can be activated by choosing the noise waveform. By default, oscillator 3's noise generator generates *white noise*.

White noise is defined as a signal that consists of all frequencies (an infinite number) sounding simultaneously, at the same intensity, in a given frequency band. The width of the frequency band is measured in Hertz. Sonically, white noise falls between the sound of the consonant "*F*" and breaking waves (surf). White noise is useful for synthesizing wind and seashore noises, or electronic snare drum sounds.

You can also modulate the tonal color of the noise signal in real time—without using the main filters of the ES2—by modulating the waveform of oscillator 3.

Change the noise color

1. Set up a modulation routing as follows: modulation target Osc3Wave, source ModWhl. The modulation amount slider behaves somewhat differently with this routing, essentially acting like a filter.
2. Use negative modulation amount values (not −1.000) to set a descending filter slope that roughly equates to 6 dB/octave. The sound becomes darker (red noise) as you adjust the mod wheel downwards.
3. To tune this pseudo filter down to 18 Hz, set the modulation amount to −1.000. When Osc3Wave is modulated positively, the noise becomes brighter (blue noise).
4. If you choose a modulation amount value of +1.000 for the Osc3Wave modulation target, the filter cutoff frequency is set to 18 kHz.

ES2 emulation of detuned analog oscillators

The Analog parameter randomly alters the pitch of each note and the filter cutoff frequency.

- Low Analog values can add a subtle richness to the sound.
- Medium Analog values simulate the tuning instabilities of analog synthesizer circuitry, which can be useful in achieving that much sought-after "warmth" of analog hardware synthesizers.
- High Analog values result in significant pitch instability, which can sound truly out of tune—but this may be perfect for your needs.

Rotate the Analog knob to randomly alter the pitch of each note, and the filter cutoff frequency.

Much like polyphonic analog synthesizers, all three oscillators maintain their specific frequency deviation from each other, but the pitches of all three oscillators are randomly detuned by the same Analog amount. For example, if the Analog detuning is set to around 20%, all three oscillators (if used) randomly drifts by 20%.

Note: If ES2 is set to Mono or Legato keyboard mode, the Analog parameter is effective only when Unison is turned on. In this situation, Analog sets the amount of detuning between the stacked (unison) voices. If the Voices parameter is set to 1 and/or Unison is not active, the Analog parameter has no effect. For more information about these parameters, see Set the ES2 keyboard mode on page 47.

Stretch tuning in ES2

The (coarse) Frequency knob of each oscillator enables you to tune oscillators 1, 2, and 3 in semitones or octaves. The (fine tune) Frequency parameter enables you to fine-tune each oscillator in cents (1/100th of a semitone). Precise detuning between oscillators can result in beats, or phasing, between the oscillator frequencies. The higher the played frequency/pitch, the faster the phasing beats. High notes, therefore, may seem to be somewhat out of tune in comparison with lower notes.

CBD (Constant Beat Detuning) can be used as a corrective tool to even out the beating between oscillators, or it can be used as a creative tool to emulate stretch tuning. The latter can be particularly important when you use an ES2 sound alongside an acoustic piano recording. This is because acoustic pianos are intentionally tuned "out-of-tune" (from equal temperament). This is known as *stretch tuning*, and results in the upper and lower keyboard ranges being slightly out of tune with the center octaves but harmonically "in-tune" with each other.

Choose a CBD value to detune the harmonics of low note frequencies in a ratio proportionate with the fundamental tone of the upper note frequencies.

CBD offers five values: off, 25%, 50%, 75%, and 100%. If you choose 100%, the phasing beats are almost constant across the entire keyboard range. This value may, however, be too high, because the lower notes might be overly detuned at the point where the phasing of the higher notes feels right. Try lower CBD values in cases where the bass notes are a little too far out of tune with the upper keyboard range.

The reference pitch for CBD is C3 (middle C): its (de)tuning is constant, regardless of the chosen CBD value.

Balance ES2 oscillator levels

The position of the pointer in the Triangle is described by two parameters—x and y coordinates—which are used when automating the oscillator mix. These parameters, called OscLevelX and OscLevelY, are available as targets in the router.

Drag the pointer in the Triangle to cross-fade—set the level relationships—between the three oscillators. This is self-evident in use. If you move the pointer along one of the Triangle's sides, it cross-fades between the two closest oscillators, and the third oscillator is muted.

The position of the pointer (x and y coordinates) in the Triangle can also be controlled with the Vector Envelope. Because the Vector Envelope features a loop function, it can be used as a pseudo-LFO with a programmable waveform. For more information about this feature, see Use the Vector Envelope on page 69.

Modulate triangle coordinates with the modulation wheel

1 Set up a modulation routing as follows: modulation target OscLevelX, source ModWhl. Adjust the intensity.

2 Set up a second modulation routing as follows: modulation target OscLevelY, source ModWhl. Adjust the intensity. You can choose other sources for these targets.

ES2 oscillator start points

The oscillators can run freely or can begin at the same phase position of their respective waveform cycles each time a note is struck.

Choose free, soft, or hard from the Osc(illator) Start pop-up menu.

- *Free:* The initial oscillator phase start point is random for each played note. This adds life to the sound. The downside is that the output level may differ each time a note is played, making the attack phase sound less punchy—even if the performance is identical each time—such as when the note is triggered by a MIDI region. This setting is useful when you are emulating sounds typical of hardware analog synthesizers.
- *Soft:* The initial oscillator phase starts at a zero crossing for each played note. This mimics the sonic character (and precision) typical of digital synthesizers.
- *Hard:* The initial oscillator phase starts at the highest level in the waveform cycle for each played note. The extra *punch* that this setting can provide is audible only if the ENV3 Attack Time parameter is set to a low value—a very fast attack, in other words. This setting is highly recommended for electronic percussion and hard basses.

Note: Osc Start soft and hard result in a constant output level of the initial oscillator phase every time the sound is played back. This may be of importance when you use the Bounce function of Logic Pro at close to maximum recording levels.

Synchronize ES2 oscillators

Typical *oscillator sync* sounds tend toward the aggressive, screaming leads that synthesizer manufacturers like to talk about. The rectangular and sawtooth waveforms of oscillators 2 and 3 feature a Sync option. When this parameter is turned on, the phase of oscillator 2 or 3 is synchronized with oscillator 1.

Every time oscillator 1 starts a new oscillation phase, the synchronized oscillator (oscillator 2 or 3) is also forced to restart its phase from the beginning. Between the waveform cycles of oscillator 1, the waveform cycles of the synchronized oscillators run freely.

You can achieve interesting synchronized oscillator sounds by modulating the frequency of the synchronized oscillator with an envelope generator. This constantly changes the number of phases within a section of the synchronization cycle, resulting in corresponding changes to the frequency spectrum.

Modulate the synchronized oscillator frequency with an envelope

1. Set the oscillator 2 Wave knob to Sync.
2. Set up a modulation routing as follows: modulation target Pitch2, source Env2.
3. Adjust the settings of Envelope 2.

ES2 global parameters

Global parameters overview

The ES2 global parameters affect the overall instrument sound produced by ES2. You can find global parameters to the left of the oscillators and above the filter and output sections.

Global parameters

- *Keyboard Mode buttons:* Switch ES2 between polyphonic, monophonic, and legato behaviors. See Set the ES2 keyboard mode on page 47.
- *Unison button:* Click to turn unison mode on or off. See Use unison and voices in ES2 on page 47.
- *Voices field:* Drag to set the maximum number of notes that can be played simultaneously.
- *Glide knob:* Rotate to set the time it takes for the pitch of a played note to slide to the pitch of the following played note. See Set the ES2 glide time on page 48.
- *Bend Range fields:* Drag to define the upward and downward pitch bend range. See Set the ES2 pitch bend range on page 48.
- *Tune field:* Drag to set the overall instrument pitch in cents. 100 cents equals a semitone step. At a value of 0 c (zero cents), the central A key is tuned to 440 Hz, or concert pitch.
- *Analog knob:* Rotate to randomly alter the pitch of each note and the filter cutoff frequency. See ES2 emulation of detuned analog oscillators on page 42).
- *Constant Beat Detuning (CBD) pop-up menu:* Choose a CBD value to detune the harmonics of low note frequencies in a ratio proportionate with the fundamental tone of the upper note frequencies. See Stretch tuning in ES2 on page 43.
- *Osc(illator) Start pop-up menu:* Choose free, soft, or hard from the Osc(illator) Start pop-up menu. See ES2 oscillator start points on page 44.

Set the ES2 keyboard mode

A *polyphonic* instrument, such as an organ or a piano, allows several notes to be played simultaneously. Many older analog synthesizers are *monophonic*, which means that only one note can be played at a time, much like a brass or reed instrument. This shouldn't be viewed as a disadvantage; instead, it allows playing styles that are not possible with polyphonic instruments.

Change the keyboard mode

- Click the Poly, Mono, or Legato button.

- In Mono mode, staccato playing retriggers the envelope generators every time a new note is played. If you play in a legato style (play a new key while holding another), the envelope generators are triggered only for the first note you play legato, then they continue their curve until you release the last legato played key.
- Legato mode is also monophonic, but with one difference: the envelope generators are retriggered only if you play staccato—releasing each key before playing a new key. If you play in a legato style, envelopes are not retriggered.

Note: On several monophonic synthesizers, the behavior in Legato mode is referred to as *single trigger*, while Mono mode is referred to as *multi trigger*.

Use unison and voices in ES2

One of the great strengths of polyphonic analog synthesizers is unison—or stacked voices—mode. Unison mode in polyphonic analog synthesizers is typically monophonic, with all voices playing simultaneously when a single note is struck. Because the voices of an analog synthesizer are never perfectly in tune, the result is an extremely fat chorus effect with great sonic depth.

Use monophonic unison mode

1. Click the Mono or Legato button, depending on the keyboard mode you want to use. See Set the ES2 keyboard mode.
2. Click the Unison button.
    - The intensity of the unison effect depends on the number chosen in the Voices parameter field. Increase the Voices value for a fatter sound. See Global parameters overview.
    - The intensity of detuning (voice deviation) is set with the Analog parameter. See ES2 emulation of detuned analog oscillators.

Use polyphonic unison mode

- Click the Poly *and* Unison buttons.

In poly/unison mode, each played note is effectively doubled—or, more correctly, the polyphony value chosen with the Voices parameter is halved. These two voices are heard when you trigger the note. Poly/unison has the same effect as setting the ES2 to mono/unison (Voices = 2), but you can play polyphonically.

Set the ES2 glide time

The Glide parameter (also known as *portamento*) sets the time it takes for the pitch of one played note to travel to the pitch of another played note.

Make portamento active

- Rotate the Glide knob.

Glide behavior is dependent on the chosen keyboard mode. See Set the ES2 keyboard mode.

- If the keyboard mode is set to Poly or Mono, and Glide is set to a value other than 0, portamento is active.
- If Legato is chosen, and Glide is set to a value other than 0, you need to play legato (press a new key while holding the old one) to activate portamento. If you don't play in a legato style, portamento won't work. This behavior is also known as *fingered portamento*.

Set the ES2 pitch bend range

The Bend range fields determine the range for pitch bend modulation, typically performed with your keyboard's pitch bend wheel.

Set independent upward and downward bend ranges

- Drag in either field to set a bend range.

Set an identical upward and downward bend range

1. Set the upward Bend range field to Link mode.

   This locks the upward and downward bend ranges, making them identical.

2. Set a downward Bend range value.

   This is mirrored in the upward Bend range field.

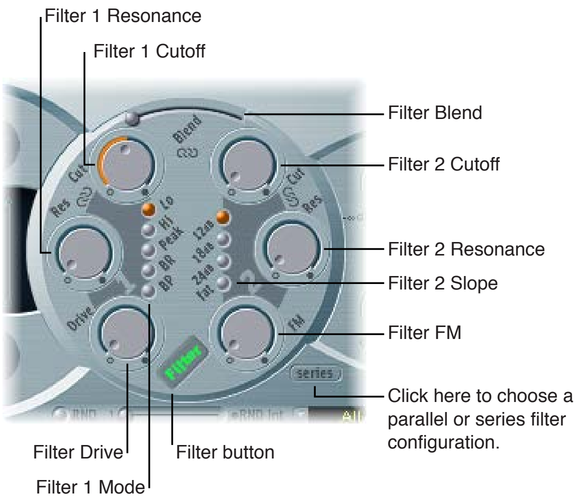
   Note: A downward bend of 4 semitones results in a combined bend range of 8 semitones—9 if you include the standard pitch, or "no bend" position.

ES2 filter parameters

ES2 filter overview

ES2 features two discrete, and different, filters.

- Filter 1 can operate as a lowpass, highpass, bandpass, band reject, or peak filter.
- Filter 2 is a lowpass filter that offers variable slopes (measured in dB/octave).

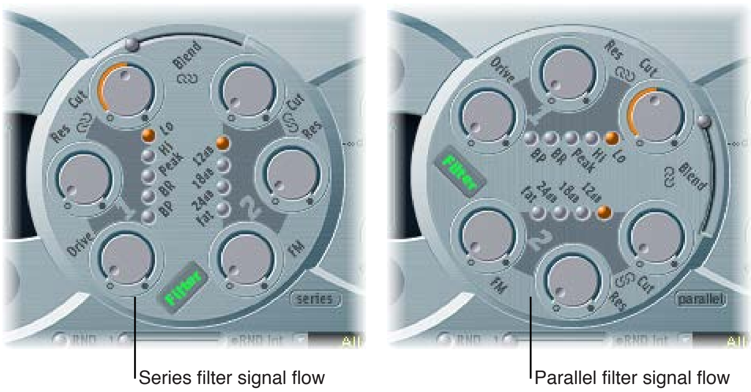


Filter parameters

- *Filter button:* Turns the entire filter section on or off. Deactivating the filter section makes it easier to hear adjustments to other sound parameters, because the filters always heavily affect the sound. Disabling the filters also reduces processor load.
- *Filter Configuration button:* Switches between a parallel or series filter configuration. See ES2 filter configuration on page 50.
- *Filter Blend slider:* Sets the balance between Filter 1 and Filter 2. See Cross-fade between ES2 filters on page 50.
- *Filter 1 Mode buttons:* Switch Filter 1 between lowpass, highpass, bandpass, band reject, or peak filter types. See ES2 Filter 1 modes on page 52.
- *Filter 2 Slope buttons:* Switch Filter 2 between different slopes. See ES2 Filter 2 slopes on page 52.
- *Cutoff and Resonance:* Rotate the Cutoff and Resonance knobs to determine the cutoff frequency and resonance behavior of each filter. See Filter cutoff and resonance overview on page 53.
- *Filter Drive knob:* Rotate to overdrive the filter, which affects each voice independently. See Overdrive ES2 filters on page 55.
- *Filter FM knob:* Rotate to set the amount of Filter 2 cutoff frequency modulation with the oscillator 1 frequency. See Modulate ES2’s Filter 2 Frequency on page 56.

ES2 filter configuration

The Filter Configuration button lets you switch between a parallel and series filter routing. When either is chosen, the entire circular filter element rotates, and the positions and direction of the filter controls clearly indicate the signal flow. The button name also changes in each mode.



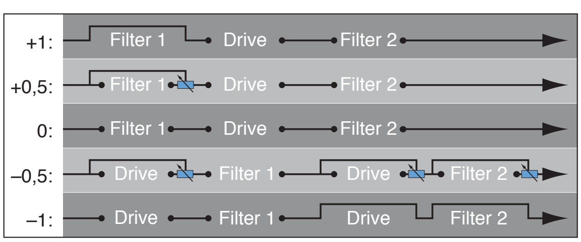
In the figure to the left, the filters are cabled in series. This means that the signal of all oscillators (combined at the Oscillator Mix Triangle) passes through the first filter, then this filtered signal passes through Filter 2, if Filter Blend is set to 0, the middle position. The output signal of Filter 2 is then sent to the input of the dynamic stage (Amplifier section).

In the figure to the right, the filters are cabled in parallel. If Filter Blend is set to 0, you'll hear a 50/50 mix of the source signal, routed via Filter 1 and Filter 2. The output signals of the two filters are then sent to the input of the dynamic stage. See Cross-fade between ES2 filters.

Cross-fade between ES2 filters

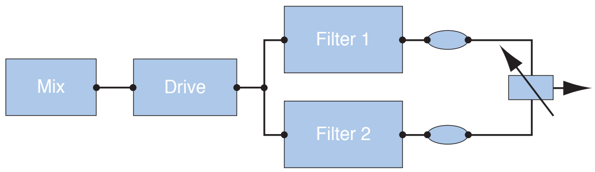
Filter Blend can have a significant effect on the ES2 signal flow. Regardless of whether parallel or series filter configurations are chosen, a Filter Blend setting of –1 results in only Filter 1 being audible. A Filter Blend setting of +1 limits audibility to Filter 2. The figures illustrate the signal flow between the Oscillator Mix stage (the Triangle) and the dynamic stage (the Amplifier). The signal flow through the filters and the filter overdrive circuit (the Drive parameter) are dependent on the Filter Blend setting.

- *Filter Blend:* Series filter configuration information
  - Use *positive* values for Filter Blend to partially bypass Filter 1.
  - Use *negative* values for Filter Blend to partially bypass Filter 2.
  - When *zero* or *positive* Filter Blend values are used, there is only one overdrive circuit for *both* filters.
  - Use of *negative* Filter Blend values introduces another overdrive circuit, which distorts the output signal of the oscillator mix stage before it is fed into the first filter.
  - If Drive is set to 0, no distortion occurs.



- *Filter Blend:* Parallel filter configuration information

In a parallel configuration, the overdrive/distortion circuit—the Drive parameter—is always wired *after* the oscillator mix stage—the Triangle—and *before* the filters. The filters receive a mono input signal from the output of the overdrive circuit. The outputs of both filters are mixed to mono via Filter Blend.



The Filter Blend parameter is available as a modulation target in the router. You can use manual control sources, such as the modulation wheel, to change the filter blend; but the Filter Blend target can also be used creatively, to rapidly switch or smoothly fade between the two filters. You can also use velocity, or a combination of the Vector Envelope and Planar Pad as sources. The latter allows for interesting filter control possibilities that evolve independently, or alongside oscillator parameters that are also being controlled with the Vector Envelope.

Cross-fade between filters

- Drag the Filter Blend slider to cross-fade between the two filters when cabled in parallel.

- If Filter Blend is set to the top position, you only hear the effect of Filter 1.
- If Filter Blend is set to its lowest position, you only hear the effect of Filter 2.
- In between these positions, the filters are cross-faded. You hear the effect of both filters.

You can also cross-fade the filters when they are cabled in series. In this situation, the distortion—controlled by the Drive parameter—also needs to be considered, as this can be positioned either before or in between the filters, depending on the Filter Blend setting you choose.

Modulate Filter Blend with an LFO

1. Set up a modulation routing as follows: modulation target FltBlend, source LFO2.
2. Adjust the settings of LFO 2.

ES2 Filter 1 modes

Filter 1 can operate in several modes, allowing specific frequency bands to be filtered (cut away) or emphasized.

Click one of the following filter mode buttons for Filter 1:

- *Lo (lowpass):* Allows frequencies that fall below the cutoff frequency to pass. The slope of Filter 1 is fixed at 12 dB/octave.
- *Hi (highpass):* Allows frequencies above the cutoff frequency to pass. The slope of Filter 1 is fixed at 12 dB/octave.
- *Peak:* Filter 1 works as a peak filter. This allows the level in a frequency band to be increased. The center of the frequency band is determined by the Cutoff parameter. The width of the band is controlled by the Resonance parameter.
- *BR (band reject):* The frequency band directly surrounding the cutoff frequency is rejected, but frequencies outside this band can pass. The Resonance parameter controls the width of the rejected frequency band.
- *BP (bandpass):* The frequency band directly surrounding the cutoff frequency is allowed to pass. All other frequencies are cut. The Resonance parameter controls the width of the frequency band. The bandpass filter is a two-pole filter with a slope of 6 dB/octave on each side of the band's center frequency.

ES2 Filter 2 slopes

Most filters do not completely suppress the portion of the signal that falls outside the frequency range defined by the Cutoff parameter. The slope, or curve, chosen for Filter 2 expresses the amount of rejection below the cutoff frequency in decibels per octave.

Slope buttons: Click any button to choose a Filter 2 slope: 12 dB, 18 dB, and 24 dB. The steeper the slope, the more severe the effect on signal levels below the cutoff frequency.

Fat button: Click the Fat button for 24 dB per octave of rejection. Fat mode has a built-in compensation circuit that retains the sound's bottom end. By comparison, the standard 24 dB setting tends to make lower end sounds less rich.

ES2 filter cutoff and resonance

Filter cutoff and resonance overview

In every lowpass filter (ES2: Lo mode for Filter 1; Filter 2 is a lowpass filter), all frequency portions above the cutoff frequency are suppressed, or cut off, hence the name. If you're new to synthesizers and the concepts behind filters, see Synthesizer basics overview on page 471.

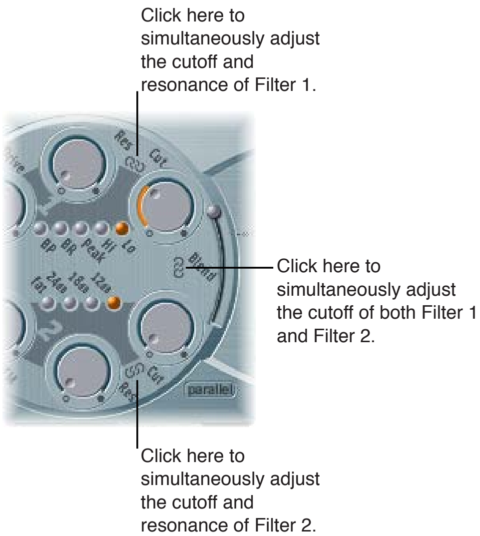
Cutoff and resonance parameters

- *Cutoff Frequency knob:* Rotate to control the brilliance of the signal.
  - In a lowpass filter, the higher the cutoff frequency is set, the higher the frequencies of signals that are allowed to pass.
  - In a highpass filter, the cutoff frequency determines the point where lower frequencies are suppressed and only upper frequencies are allowed to pass.
  - In a bandpass/band rejection filter, the cutoff frequency determines the center frequency for the bandpass or band rejection filter.
- *Resonance knob:* Rotate to emphasize or suppress portions of the signal above or below the defined cutoff frequency.
  - In a lowpass filter, Resonance emphasizes or suppresses signals below the cutoff frequency.
  - In a highpass filter, Resonance emphasizes or suppresses signals above the cutoff frequency.
  - In bandpass/band rejection filters, resonance emphasizes or suppresses the portions of the signal—the frequency band—that surround the defined frequency, set with the Cutoff Frequency parameter.

Control two filter parameters simultaneously

The ability to change the Cutoff and Resonance controls at the same time is essential for creating expressive synthesizer sounds.

- Drag one of the three chain symbols in the ES2 filter section.



  - The chain between Cut and Res of Filter 1 controls both the resonance (drag horizontally) and cutoff frequency (drag vertically) simultaneously.
  - The chain between Cut and Res of Filter 2 controls both the resonance (drag horizontally) and cutoff frequency (drag vertically) simultaneously.
  - The chain between Filter 1 Cut and Filter 2 Cut controls the cutoff frequency of Filter 1 (drag vertically) and Filter 2 (drag horizontally) simultaneously.

Force ES2 filters to self-oscillate

If you increase the filter Resonance parameter to higher values, the filter begins to internally feed back and, as a consequence, begins to self-oscillate. This results in a sine oscillation—a sine wave—that is actually audible.

To start this type of oscillation, the filter requires a trigger. In an analog synthesizer, this trigger can be the noise floor or the oscillator output. In the digital domain of the ES2, noise is all but eliminated. Therefore, when the oscillators are muted there is no input signal routed to the filter. Filter Reset provides a trigger signal that can be used to drive the filter to self-oscillate.

Use Filter Reset to drive the ES2 filters to self-oscillate

- Click the Filter Reset button to turn on.

  When this button is engaged, each note starts with a trigger that makes the filter resonate/self-oscillate immediately.

Compensate for high resonance values with the Fat(ness) parameter

- Click to turn on the Fat(ness) button—below the other filter slope buttons.

An increase of the resonance value results in a rejection of bass—low frequency energy—when using lowpass filters. Use the Fatness button to compensate for this side effect and to obtain a richer sound.

Overdrive ES2 filters

The filters are equipped with discrete overdrive modules. You can set the overdrive intensity by rotating the Drive parameter.

Drive affects each voice independently. When every voice is overdriven individually—like having six fuzz boxes for a guitar, one for each string—you can play extremely complex harmonies over the entire keyboard range. Each voice sounds clean, without unwanted intermodulation effects spoiling the overall sound.

Certain Drive settings can lead to a different tonal character for the following reason: the way analog filters behave when overdriven forms an essential part of a synthesizer's sonic character. Each synthesizer model is unique in the way its filters behave when overdriven. ES2 is very flexible in this area, allowing tonal colors that range from the most subtle fuzz to the hardest of distortions.

- If the filters are connected in parallel, the overdrive circuit is placed *before* the filters.
- If the filters are connected in series, the position of the overdrive circuits is dependent on the Filter Blend parameter. See Cross-fade between ES2 filters.

Tip: Because Filter 2 can cut away the overtones introduced by the distortion, Drive can be used as another tool for deforming oscillator waveforms.

Polyphonic distortions in the real world

ES2 provides a dedicated distortion effect in the Effects section. Given this inclusion, you may wonder what benefit the filter's Drive function offers.

The Distortion circuit in the Effects section affects the entire polyphonic output of the ES2. Every rock guitarist knows that more complex chords—other than major chords, parallel fifths, and octaves—sound "rough" when using distortion. Therefore, distorted guitar playing generally involves few voices or parallel fifths and octaves. Because the filter Drive parameter affects each voice individually, you can play complex chords without introducing the unpleasant intermodulations that the Distortion effect can add to your sound.

Modulate ES2's Filter 2 Frequency

Filter 2 cutoff frequency can be modulated by the sine wave of oscillator 1, which is always generated, even when the oscillator is switched off. The level of this sine signal can be mixed in at the output stage with the Sine Level parameter (see Sine Level enhanced ES2 sounds on page 57).

The effect of such filter modulations in the audio spectrum is unpredictable, but the results tend to remain harmonic if you avoid high modulation intensity values. The FM parameter is used to define the intensity of this filter frequency modulation.

Note: Don't confuse this filter frequency modulation with the oscillator FM feature (oscillator 1 is modulated by oscillator 2). If oscillator 1 is frequency-modulated by oscillator 2, it does not influence the sine wave signal used to modulate the cutoff frequencies. See Use frequency modulation in ES2.

Filter 2 can also be driven to self-oscillation. If you set a very high resonance value, it produces a sine wave. This self-oscillating sine wave distorts at the maximum resonance value. If you mute all oscillators, you'll only hear this sine oscillation. By modulating the cutoff frequency, you can produce effects similar to those produced by modulating the frequency of oscillator 1 with oscillator 2.

Modulate filter FM

1. Set up a modulation routing as follows: modulation target LPF FM.

   A sine wave, at the frequency of oscillator 1, is *always* used as the modulation source. Given this default assignment and the direct relationship between the filter FM intensity and oscillator 1's frequency, you can set up a second routing to modulate Oscillator 1's pitch.
2. Set up a modulation routing as follows: modulation target Pitch 1, source LFO1.
3. Adjust LFO settings.

ES2 amplifier parameters

Use ES2's dynamic stage

The dynamic stage of a synthesizer defines the level, or perceived volume, of a played note. The change in level over time is controlled by an *envelope generator*. For more information about envelope generators, see Synthesizer basics overview on page 471.

ENV 3 is hard wired to the dynamic stage of the ES2—it is always used to control the level of each note. See ES2 envelopes overview.

The dynamic stage can be modulated by any router modulation source.

Modulate the dynamic stage (Amp)

1 Set up a modulation routing as follows: modulation target AMP, source LFO1.

2 Make sure that via is set to Off.

3 Adjust LFO settings.

A tremolo effect is created, with the level changing periodically, based on the current LFO 1 Rate value.

Sine Level enhanced ES2 sounds

The Sine Level knob mixes a sine wave (at the frequency of oscillator 1) directly into the dynamic stage, independent of the filters. Even if you have filtered away the basic partial tone of oscillator 1 with a highpass filter, you can reconstitute it with this parameter.

- In cases where oscillator 1 is frequency-modulated by oscillator 2 (if you have turned up FM with the waveform selector), *only* the pure sine wave is mixed into the dynamic section, not the distorted FM waveform.
- Any modulation of oscillator 1's pitch, set in the router, affects the frequency of the sine wave mixed in at this stage.

Note: The Sine Level knob is useful for adding warmth and a fat bass quality to the sound. Extra body can be added to thin sounds with this parameter, given that oscillator 1 actually plays the basic pitch.

ES2 modulation

ES2 modulation overview

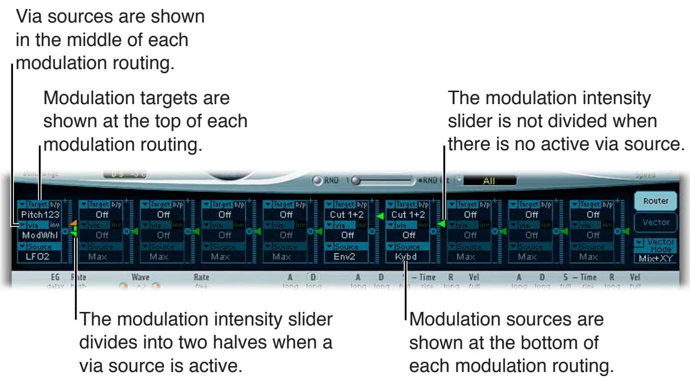
ES2 is equipped with a huge number of modulation sources and targets, making it a synthesizer that can generate extraordinary sounds that constantly evolve, sound like audio loops, or are just plain expressive to play.

- *Modulation router:* The modulation router—or router, for short—links modulation sources, such as the envelope, to modulation targets, such as the oscillators and filters. The router features ten modulation routings, arranged into columns. See Use the modulation router on page 59.
- *Modulation sources:* The modulation sources include the LFOs and envelopes. See ES2 LFO overview on page 63 and ES2 envelopes overview on page 66.
- *Vector Envelope:* The Vector Envelope is an extremely sophisticated, loop-capable, multipoint envelope that can control the Planar Pad and Triangle (oscillator mix parameter). The Vector Envelope shares the space occupied by the modulation router and can be viewed by clicking the Vector Envelope button to the right of the router. See Use the Vector Envelope on page 69.
- *Planar Pad:* The Planar Pad is a two-dimensional controller that facilitates the simultaneous manipulation of two, freely assignable, parameters. It can be controlled with the Vector Envelope. See Use the Planar Pad on page 76.

ES2 modulation router

Use the modulation router

The modulation router—or router—spans the center of the ES2 interface. Click the Router button to view it if the Vector Envelope is displayed (these components share the same section of the interface). You can hide or show the router/Vector Envelope by clicking the disclosure triangle at the lower left of the section. If you are new to synthesizer modulation routings, see Modulation overview on page 487.



Any modulation source can be connected to any modulation target, much like an old-fashioned telephone exchange or a studio patch bay.

The modulation intensity—how strongly the target is influenced by the source—is set with the vertical slider to the right of the modulation routing.

The intensity of the modulation can itself be modulated: the via parameter defines a further modulation source, which is used to control the modulation intensity. When via is active, you can specify upper and lower limits for the modulation intensity.

Ten such modulation routings of source, via, and target can take place simultaneously. It doesn't matter which of the ten modulation routings you use. You can even select the same target in several parallel modulation routings. You can also use the same sources and the same via controllers in multiple modulation routings.

Create a basic modulation routing

1 Choose the parameter you want to modulate from the Target pop-up menu.

2 Choose the parameter you want to use for modulation of the target from the Source pop-up menu.

3 Vertically drag the Intensity slider to set a fixed modulation intensity. When via is active, this slider sets the minimum modulation intensity.

Bypass a modulation routing

- Click the “b/p” button at the top right of the modulation routing next to the Target parameter.

The Bypass (b/p) parameter enables or disables individual modulation routings, without losing settings.

Control ES2 modulation intensity with via sources

In a basic modulation routing comprised of a target and source, you can set a fixed modulation intensity by vertically dragging the Intensity slider to the right of the routing. The slider value always defines a constant modulation intensity.

You can choose a further modulation source from the via pop-up menu, which controls modulation intensity.

Choosing a value other than off for via divides the Intensity slider into two halves. Each half has its own arrowhead.

- The upper half of the slider defines the maximum modulation intensity when the via controller is set to its maximum value.
- The lower half of the slider defines the minimum modulation intensity when the via controller—the modulation wheel, for example—is set to its minimum value.
- The area between the two slider halves defines the modulation range of the via controller.

Create a modulation routing that includes a via source

1 Choose a modulation target from the Target pop-up menu.

2 Choose a modulation source from the Source pop-up menu.

3 Choose the modulation source that you want to use for control of modulation intensity from the via pop-up menu.

4 Vertically drag the upper arrowhead of the Intensity slider (to the right of the modulation routing) to set the maximum modulation intensity.

5 Vertically drag the lower arrowhead of the Intensity slider to set the minimum modulation intensity.

Move the entire via range

- Vertically drag the range (the area between the two slider halves).

Both arrowheads move simultaneously.

If this area is too small to drag, drag an unused section of the Intensity slider "track" to move the area.

Set the modulation intensity to zero

- Click the zero symbol beside the via parameter.

Invert the effect of the via modulation source

- Click the via invert (inv) parameter to the right of the via parameter.

ES2 LFOs

ES2 LFO overview

ES2 features two multi-waveform LFOs. Both are available as sources in the router.

LFO 1 is polyphonic, which means that if used for any modulation of multiple voices, they will *not* be phase-locked. LFO 1 is also key-synced: each time you play a key, LFO 1 modulation of this voice is started from zero.

To understand the non phase-locked characteristic more fully, imagine a scenario where a chord is played on the keyboard. If LFO 1 is used to modulate pitch, for example, the pitch of one voice may rise, the pitch of another voice might fall, and the pitch of a third voice may reach its minimum value. As you can see, the modulation is independent for each voice, or note.

The key-sync feature ensures that the LFO waveform cycle always starts from zero, which results in consistent modulation of each voice. If the LFO waveform cycles were not synchronized in this way, individual note modulations would be uneven.

- LFO 1 is preconfigured to control the pitch of all three oscillators. It can be simultaneously used for modulation of other parameters.
- LFO 1 can also be faded in or out automatically, courtesy of a built-in envelope generator.
- LFO 2 is monophonic, which means that the modulation is identical for all voices. For example, imagine a chord is played on the keyboard. If LFO 2 is used to modulate pitch, the pitch of all voices in the played chord rises and falls synchronously. LFO 2 is ideally suited for creating rhythmic modulation effects that retain perfect synchronicity, even during project tempo changes.

LFO parameters

- *LFO 1 EG slider:* Move to set the time it takes for the LFO modulation to fade in or fade out. The value is displayed in milliseconds beneath the slider. Click the zero to turn the LFO 1 envelope generator off.
- *LFO 1 Rate slider:* Move to set the frequency (speed) of LFO 1 modulation. The value is displayed in Hertz (Hz) beneath the slider.
- *LFO 1 Wave buttons:* Choose the waveform used by LFO 1. See ES2 LFO waveforms on page 64.
- *LFO 2 Rate slider:* Move to set the frequency of LFO 2 modulation. LFO 2 can be synchronized with the host application tempo.

ES2 LFO waveforms

Choose a waveform for LFO 1 or LFO 2 with the LFO Wave buttons. The table below outlines how these waveforms can affect your sounds.

Tip: Try using the waveforms while a modulation routing of Pitch123 (the pitch of all three oscillators) is engaged and running

Waveform	Comments
Triangle	Suitable for vibrato effects
Sawtooth	Suitable for helicopter and space gun sounds. Intense modulations of oscillator frequencies with a negative (inverse) sawtooth wave lead to "bubbling" sounds. Intense sawtooth modulations of lowpass filter cutoff and resonance creates rhythmic effects. The inverted sawtooth waveform provides a different start point for the modulation cycle.
Rectangle	Rectangular waves periodically switch the LFO between two values. The upper rectangular wave switches between a positive value and zero. The lower wave switches between a positive and a negative value set to the same amount above/below zero. An interesting effect can be achieved by modulating Pitch123 with a suitable modulation intensity that leads to an interval of a fifth. Choose the upper rectangular wave to do so.
Sample & Hold	The bottom two LFO waveforms output random values. A random value is selected at regular intervals, defined by the LFO rate. The upper random wave steps between randomized values—rapid switches between values. The lower random wave is smoothed out, resulting in fluid changes to values. The term Sample & Hold (S & H) refers to the procedure of taking samples from a noise signal at regular intervals. The values of these samples are then held until the next sample is taken. *Tip:* A random modulation of Pitch123 leads to an effect commonly referred to as a random pitch pattern generator or sample and hold. Try using very high notes, at very high rates and high intensities—you'll recognize this well-known effect from hundreds of science fiction movies.

Use ES2 LFOs

The ES2 LFO's can be used to create delayed modulations, free modulations, and modulations that are synchronized with your host application.

Set the LFO 1 modulation fade time

- *To fade in the modulation:* Set a positive LFO 1 EG value.

  The higher the value, the longer the delay time.

- *To fade out the modulation:* Set a negative LFO 1 EG value.

  The lower the slider is positioned onscreen, the shorter the fade out time.

Set up a delayed vibrato

LFO envelopes are most often used for delayed vibrato—many instrumentalists and singers intonate longer notes this way.

1. Place the LFO 1 EG slider at a position in the upper half (Delay) and modulate the Pitch123 target with the LFO1 source in the router.
2. Set a slight modulation intensity.
3. Set an LFO 1 Rate of about 5 Hz.
4. Choose the triangular wave as the LFO 1 waveform.

Set a free rate for LFO 2

- Choose a value in the upper half of the LFO 2 Rate slider range to run LFO 2 freely.

  The rate is displayed in hertz.

Synchronize the LFO 2 rate with the song tempo

- Choose a value in the lower half of the LFO 2 Rate slider range to synchronize LFO 2 with the host application tempo.

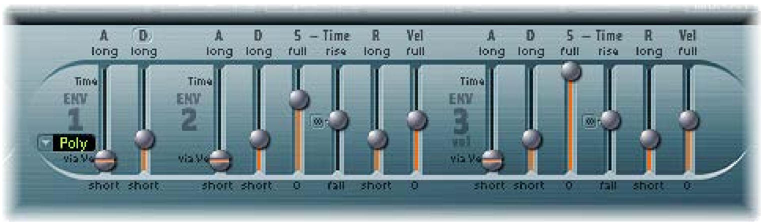
  The rate is displayed in rhythmic values (when project tempo synchronization is active).

  Synchronized rates range from speeds of 1/64-notes to a periodic duration of 32 bars. Triplet and punctuated values are also available.

ES2 envelopes

ES2 envelopes overview

ES2 features three envelope generators per voice. They are abbreviated as ENV 1, ENV 2, and ENV 3 in the interface and router. In addition, ES2 features the sophisticated Vector Envelope. See Use the Vector Envelope.



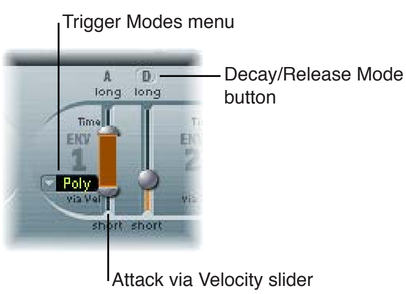
To learn more about the roots of the term "envelope generator" and its basic function, see Amplifier envelope overview on page 485.

The parameters of ENV 2 and ENV 3 are identical. ENV 3 defines the changes in level over time for each note played. You can think of ENV 3 as being hardwired to the router's AMP modulation target. ENV 2 controls the cutoff frequency of both ES2 filters.

Note: All envelopes can be used to control multiple parameters simultaneously.

ES2 Envelope 1

Although Envelope 1 (ENV 1) appears to be simplistic, it is useful for a range of synthesizer functions.



Envelope 1 parameters

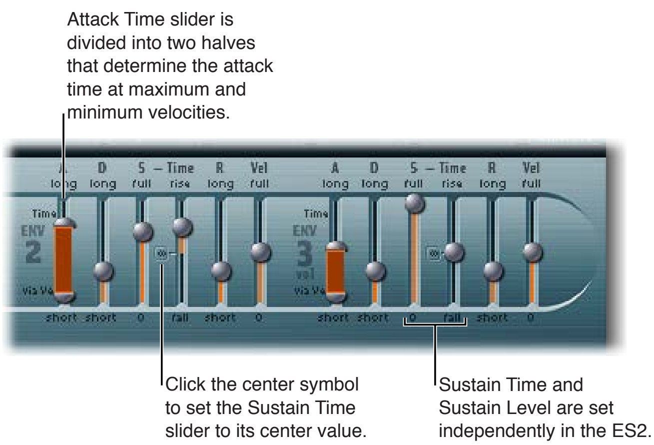
- *Trigger Modes pop-up menu:* Choose a mode to define the trigger behavior of ENV 1.
  - *Poly:* The envelope generator behaves as you would expect on any polyphonic synthesizer: every voice has its own envelope.
  - *Mono:* A single envelope generator modulates all voices in the same way. All notes must be released before the envelope can be retriggered. If you play legato, or any key remains depressed, the envelope does not restart its attack phase.
  - *Retrig:* A single envelope generator modulates all voices in the same way. The envelope is triggered by any key you strike, even when other notes are sustained. All sustained notes are identically affected by the retriggered envelope.
- *Attack via Velocity slider:* The Attack time slider is divided into two halves. The lower slider sets the attack time when keys are struck at maximum velocity. The top slider sets the attack time at minimum velocity. Drag the area between the two slider halves to move both simultaneously. If this area is too small to drag, click an unused portion of the slider, and drag vertically.

- *Decay/Release mode button:* Switches ENV 1 between an Attack/Decay or an Attack/Release envelope. The button label changes to reflect the mode that is activated (D=Decay, R=Release).
  - *In Attack/Decay mode:* The level falls to zero after the attack phase has completed, whether or not the note is sustained. It decays at the same speed, even if you release the key. The decay time is set with the D (Decay time) slider.
  - *In Attack/Release mode:* The envelope level remains at its maximum after the attack phase is over, while the key remains depressed. Following the release of the key, the level decreases over the time period defined by the R (Release time) slider.

ES2 Envelopes 2 and 3

The feature sets of ENV 2 and ENV 3 are identical, but it is *always* the task of ENV 3 to define the level of each note—to modulate the dynamic stage, in other words. ENV2 is preconfigured to control the Cutoff frequency of both filters.

Both ENV 2 and ENV 3 can also be used simultaneously as *sources* in the router. The envelope time parameters can be used as modulation *targets* in the router.



Envelope 2 and 3 Parameters

- *Attack slider:* Defines the time it takes for the level of a note to rise from a level (amplitude) of zero to the set amplitude. The Attack time sliders of ENV 2 and ENV 3 are divided into two halves.
  - The lower half defines the attack time when the keys are struck hard, at maximum velocity. The upper half defines the attack time at minimum velocity.

  Drag the area between the two slider halves to move both simultaneously. If this area is too small to drag, drag an unused portion of the slider vertically.
- *Decay slider:* Sets the time it takes for the level of a held note to fall to the sustain level, after the attack phase has completed.
  - If the Sustain level parameter is set to its maximum value, the Decay parameter has no effect.
  - When the Sustain level is set to its minimum value, the Decay parameter defines the duration or fade-out time of the note.

- *Sustain and Sustain Time sliders:* The two sustain parameters interact with each other. One controls the sustain level, and the other controls the sustain time. See Use ES2 Envelope 2 and 3 sustain controls on page 68.
- *(R) Release Time slider:* Defines the time required for the (sustain) level to decay to zero, after the key is released.
- *Vel (Velocity Sensitivity) slider:* Determines the velocity sensitivity of the entire envelope. If set to maximum, the envelope outputs its maximum level only when the keys are struck at maximum velocity. Softer velocities result in a corresponding change to the envelope levels, with a 50% velocity resulting in half-levels for each envelope-level parameter.

Use ES2 Envelope 2 and 3 sustain controls

When the Sustain Time (rise) slider is set to its center value, the Sustain (S) Level slider behaves like the sustain parameter of any synthesizer ADSR envelope.

In this position, the Sustain (Level) slider defines the level that is sustained while the key remains depressed, following completion of the Attack time and Decay time phases.

The Sustain Time slider defines the time it takes for the level to rise from the Sustain level back to its maximum level—or to fall to zero:

- Settings in the lower half of the Sustain Time slider range (fall) determine the time required for the level to decay from the sustain level to zero. The lower the slider position, the faster the sound level decays.
- Settings in the upper half of its range (rise) determine the time required for the level to rise from the sustain level to its maximum value. The higher the slider position, the faster the sound level rises.

Emulate instrument behaviors with envelope decay modulation

1. Set up a modulation routing as follows: modulation target Env3Dec, source Kybd.
2. Make sure the Intensity slider is set to a negative value.
3. Adjust Env3 settings.

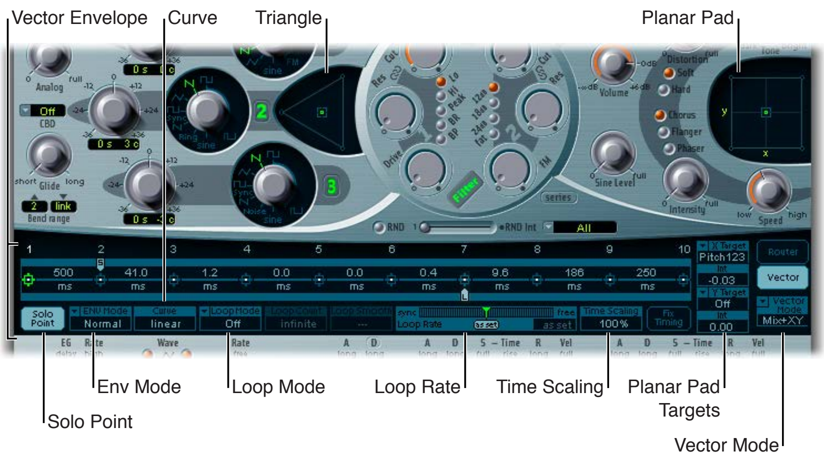
This routing simulates the behavior of pianos and plucked instruments, where high notes decay faster than low notes.

Use the Vector Envelope

The Vector Envelope is a multipoint, loop-capable control source. Its sole purpose is to provide real-time control of pointer movements in the Triangle and the Planar Pad. The Vector Envelope shares the space occupied by the modulation router.

Each played voice has an independent Vector Envelope, which is triggered from its start point with every new keystrike (MIDI note-on message).

Conceptually, the Vector Envelope—and Planar Pad and Triangle—may be difficult to grasp, but some experimentation will reveal how easy these features are to use. Combining these facilities with other synthesis options enables you to create truly unique sounds that are—quite literally—moving.



Display the Vector Envelope

- Click the Vector Envelope button to the right of the router to display the Vector Envelope.

Turn the Vector Envelope on or off

- *To make the Vector Envelope active:* Turn off the Solo Point button.
- *To deactivate the Vector Envelope:* Turn on the Solo Point button.

When Solo Point is turned on, only the currently selected Triangle and Planar Pad positions of the currently selected point are active.

Control the Planar Pad and Triangle with the Vector Envelope

- Choose the target for the Vector Envelope—the Planar Pad, Triangle, or both—from the Vector Mode pop-up menu.
  - *Off:* The Vector Envelope does not control the Triangle or the Planar Pad. It is completely turned off. You can manually set and control the pointers of the Triangle and the Planar Pad.
  - *Mix:* The Vector Envelope controls the Triangle but not the Planar Pad.
  - *XY:* The Vector Envelope controls the Planar Pad but not the Triangle.
  - *Mix+XY:* The Vector Envelope controls both the Planar Pad and the Triangle.

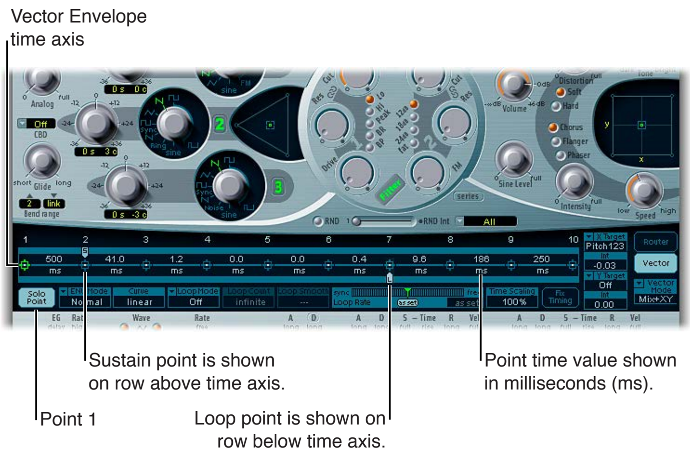
Use the Vector Envelope shortcut menu

1. Right-click or Control-click anywhere in the Vector Envelope to open a shortcut menu of commands and functions.
2. Choose any item in the menu to perform the operation.

Vector Envelope points, times, and loops

Use Vector Envelope points

The Vector Envelope time axis runs from left to right.



Up to 16 points can be displayed on the time axis (10 are shown in the figure above). Each point can control the pointer positions of the Triangle and the Planar Pad.

The points are numbered sequentially, from left to right, along the time axis.

There are always at least three points: point 1 is the start point, point 2 is defined as the Sustain point, and point 3 is the end point.

Any point can be declared the Sustain point. If a played note is held for a sufficient length of time and there's no loop engaged, any envelope movement stops when the Sustain point is reached. The Sustain point value is maintained until the key is released—until the MIDI note-off command.

Any point can be declared the Loop point. The looped area spans the time between the Sustain point and Loop point. In between these points you can create additional points that describe the movements of the pointers in the Planar Pad and Triangle.

The more points you set, the more complex the movements that can be performed.

Select a point

- Click the point to select it.

  Once selected, you can edit the point.

Create a new point

- Shift-click between two existing points.

  The segment that previously existed between the two old points is divided at the clicked position. The sum of the two new segment times is equal to the time of the original undivided segment. This ensures that any points that follow retain their absolute time positions. Existing pointer positions in the Triangle and Planar Pad are fixed, thus ensuring that newly created points don't affect any previously defined movements.

Delete a point

- Control-click the point.

Revert to the default value for a point

Do one of the following:

- Option-click the Triangle.

  The pointer is set to the center position of the Triangle, and all oscillators are set to output the same level.

- Option-click the Planar Pad.

  The pointer is set to the center position of the Planar Pad. Both axis values are set to zero.

Use Vector Envelope solo and sustain points

You use the Solo Point button to turn the Vector Envelope on or off. If the Solo Point button is on, no dynamic modulations are generated by the Vector Envelope. In this scenario, the currently visible pointer positions of the Triangle and Planar Pad are permanently in effect. These pointer positions match the currently selected Vector Envelope point.

If you select another Vector Envelope point by clicking it, the pointer positions in the Triangle and Planar Pad update to reflect your selection. If the Solo Point button is on, the newly selected point becomes the Solo point.

Note: You can independently turn off Vector Envelope modulation of the Planar Pad by setting Vector Mode to off. See Use the Vector Envelope.

Any point can be declared the Sustain point. Assuming that the played note is held long enough and there's no loop engaged, any envelope movement stops when this Sustain point is reached. The Sustain point value is maintained until the key is released—until the MIDI note-off command.

Define a point as the Sustain point

- Click in the turquoise strip above the chosen point.

  The Sustain point is indicated by an S between the point and its number shown on the turquoise strip.

Set up Vector Envelope loops

The Vector Envelope can run in one-shot mode, as long as the note is sustained; it can be set to repeat a specific number of times or it can repeat indefinitely, much like an LFO modulation. You achieve repetitions by using the loop functions.

Although the loop parameters seem similar to the loop parameters available for samples, there are significant differences between them. The Vector Envelope only supplies control signals that are used to move the pointer positions of the Triangle and Planar Pad. The audio output of the ES2 is *not* looped in any way.

Any point can be declared the Loop point. Provided that the note is held for a suitable length of time, portions of the envelope can be repeated, or looped.

The looped area spans the time between the Sustain point and the Loop point. In between these points you can define several points that describe pointer movements in the Triangle and the Planar Pad.

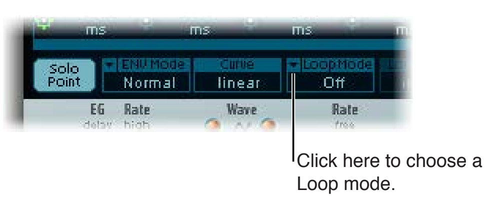
Define a point as the Loop point

- Click in the turquoise strip below the chosen point.

  A Loop point is indicated by an *L* in the strip below.

Set a Vector Envelope Loop mode

- Choose one of the following Vector Envelope Loop modes: Off, Forward, Backward, and Alternate.
  - *Off:* When Loop mode is set to Off, the Vector Envelope runs in one-shot mode from beginning to end, if the note is held long enough to complete all envelope phases. The other loop parameters are disabled.
  - *Forward:* When Loop mode is set to Forward, the Vector Envelope runs from the beginning to the Sustain point, and then begins to periodically repeat the section between the Sustain point and the Loop point in a forward direction.
  - *Backward:* When Loop mode is set to Backward, the Vector Envelope runs from the beginning to the Sustain point, and then begins to periodically repeat the section between the Sustain point and the Loop point in a backward direction.
  - *Alternate:* When Loop mode is set to Alternate, the Vector Envelope runs from the beginning to the Sustain point and then periodically switches to the Loop point, then back to the Sustain point, alternating between backward and forward directions.

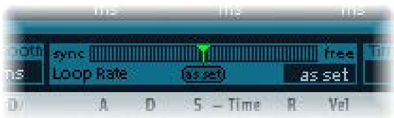


Set the Vector Envelope Loop Rate

Do one of the following:

- Drag the green indicator in the center of the Loop Rate bar to the left or right.
- Drag vertically in the value field "as set" (shown in the figure below).

The Vector Envelope loop can cycle at a defined speed. It can also be synchronized with the host application tempo.



- *As set:* If you switch the Loop Rate to "as set," the loop cycle length equals the sum of the times between the sustain and Loop points. Click the field labeled "as set" below the Rate slider to select it.
- *Rhythmic:* If you set the Loop Rate to one of the rhythmic values (sync) by dragging the Loop Rate indicator toward the left half of the slider, the loop rate follows the project tempo. You can choose from 32 bars up to a 64th triplet note value.
- *Free:* You can also set a free Loop Rate by dragging the Loop Rate indicator toward the right half of the slider (free). The value indicates the number of cycles per second.

  Note: If Loop Rate is not switched to "as set," and Loop mode (Forward, Backward, or Alternate) is active, the times of points between the Loop and Sustain points and the Loop Smooth value are shown as a percentage of the loop duration, rather than in milliseconds.

Make smooth Vector Envelope loop transitions

- When Loop mode is set to Forward or Backward, there is a transition from the Sustain point to the Loop point. Turn on Loop Smooth to even out this transition, avoiding abrupt position changes.
  - If the Loop Rate parameter is set to Sync or Free, the loop-smoothing time is displayed as a percentage of the loop cycle duration.
  - If the Loop Rate parameter is "as set," the loop-smoothing time is displayed in milliseconds (ms).

Specify a Vector Envelope loop count

- The Vector Envelope loop cycle can be repeated a specified number of times. Following the defined number of repetitions, the Vector Envelope runs from the Sustain point onward. Possible values are 1 to 10 and "infinite."

Vector Envelope release phase behavior

There are two release phase options in the Env Mode menu: Normal and Finish.

In Normal mode, the release phase—the phase after the Sustain point—begins as soon as you release the key (note off). In other words, the release phase starts from the Vector Envelope point where you released the key. The following behaviors apply:

- If looping is turned off and the Vector Envelope reaches the Sustain point, the Sustain point value is retained for as long as you hold a key.
- If looping is turned on and the Loop point is positioned before the Sustain point, the loop cycles for as long as you hold a key.
- If looping is turned on and the Loop point is positioned after the Sustain point, the Vector Envelope loop continues to cycle until the overall release phase of the sound, as determined by the ENV 3 Release parameter, has completed.

If the Env Mode menu is set to Finish, the Vector Envelope does not immediately begin the release phase when you release the key. Rather, it plays all points for their full duration until the end point is reached, regardless of whether you hold the key or release it. The following behaviors apply:

- If looping is turned off, the Sustain point is ignored. The Vector Envelope completes all points up to the end point, regardless of whether you hold the key or release it.
- If looping is turned on, the Vector Envelope plays all points until it reaches the Loop point, and then plays loop until the end point is reached. It does not matter if the Loop point is positioned before or after the Sustain point.
- If looping is turned on, and Loop Count is set to a value other than "infinite," the Vector Envelope continues on to the subsequent points—following completion of the specified number of loop repetitions. If Loop Count is set to "infinite," the points after the loop are irrelevant.

Vector Envelope point transition shapes

Curve defines the shape of the transition from point to point. Choose from nine convex and nine concave shapes, plus "hold+step" and "step+hold," which allow stepped modulations.

- *step+hold:* This curve jumps at the beginning of the transition.
- *hold+step:* This curve jumps at the end of the transition.

Note: You can use "hold+step" to create stepped vector grooves—with up to 15 steps.

Set Vector Envelope times

With the exception of the first point, which is tied to the beginning of each played note, every point has a Time parameter. This parameter defines the period of time required for the position indicator to travel from the point that preceded it. The times are normally displayed in milliseconds (ms).

Set a time value

- Drag the numerical value vertically.

Note: Changing a time value alters the absolute time positions of all subsequent points.

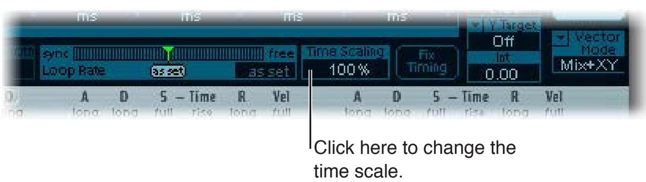
Set a time value without affecting the absolute time positions of later points

- Control-drag the Time parameter to increase or decrease the time required to reach the following point.

The time setting of the ensuing point is simultaneously adjusted by a corresponding amount. This ensures that the adjacent and all following points retain their absolute time positions.

Use Vector Envelope time scaling

You can stretch and compress the entire Vector Envelope. To double the Vector Envelope's speed, for example, set Time Scaling to 50%, rather than halving the time values of every point.



- The Time Scaling parameter ranges from 10% to 1000%. It is scaled logarithmically.
- If the Loop Rate is "as set," scaling also affects the loop.
- If the Loop Rate is set to a free or synced value, the setting is not affected by the Time Scaling parameter.

Normalize time scaling and the loop rate with Fix Timing

- Click Fix Timing to multiply the Time Scaling value by all time parameters. Time Scaling is reset to 100%.

There is no audible difference. This is simply a normalizing procedure.

In cases where Loop Rate is set to a synced value, clicking Fix Timing switches the Loop Rate to "as set," thus preserving the absolute rate.

Use the Planar Pad

The Planar Pad has two axes—X (horizontal) and Y (vertical). Two user-defined parameters can be modulated with the X and Y values, allowing you to use the mouse like a joystick.

The X and Y axes have positive and negative value ranges. When you drag the pointer (the square icon), the values of both axes are continuously transmitted.

The Vector X and Vector Y Target menus determine which parameter is modulated by pointer movements in the Planar Pad. These modulation targets are identical to those in the router. See ES2 oscillator modulation targets on page 77, ES2 filter modulation targets, and Other ES2 modulation targets.

The position (coordinates) of the Planar Pad pointer is also available in the router, as the Pad-X and Pad-Y source and via options. See ES2 modulation source reference on page 83 and Control ES2 modulation intensity with via sources on page 61.

The maximum intensity, sensitivity, and polarity of the modulation is set with the Vector X Int and Vector Y Int parameters.

Set the modulation intensity

- Drag vertically in the Vector X and Y Int fields.

Use a negative value to invert the modulation polarity.

ES2 modulation target reference

ES2 oscillator modulation targets

The table below shows all oscillator-related modulation targets.

Target	Comments
Pitch123	Modulates the frequencies (pitch) of all three oscillators. If you select an LFO as the source, this target leads to siren or vibrato sounds. Select one of the envelope generators with zero attack, short decay, zero sustain, and short release as the source for tom and kick drum sounds.
Pitch 1	Modulates the frequency (pitch) of oscillator 1. Slight envelope modulations can make the amount of detuning change over time, when oscillator 1 is sounding in unison with another (unmodulated) oscillator. This also applies to the other Pitch targets and is particularly useful for synthesizer brass sounds.
Pitch 2	Modulates the frequency (pitch) of oscillator 2.
Pitch 3	Modulates the frequency (pitch) of oscillator 3.
Detune	Controls the amount of detuning between all three oscillators. The sensitivity of all pitch modulation targets is determined by the modulation intensity. This is scaled as per the lists below, allowing you to create very delicate vibrati in the cent range (1/100 semitone), and huge pitch jumps by octaves. • Modulation intensity from 0 to 8: steps are 1.25 cents. • Modulation intensity from 8 to 20: steps are 3.33 cents. • Modulation intensity from 20 to 28: steps are 6.25 cents. • Modulation intensity from 28 to 36: steps are 12.5 cents. • Modulation intensity from 36 to 76: steps are 25 cents. • Modulation intensity from 76 to 100: steps are 100 cents. This leads to the following rules of thumb for modulation intensity values. • Intensity of 8 equals a pitch shift of 10 cents. • Intensity of 20 equals a pitch shift of 50 cents (one quarter tone). • Intensity of 28 equals a pitch shift of 100 cents (one semitone). • Intensity of 36 equals a pitch shift of 200 cents (two semitones). • Intensity of 76 equals a pitch shift of 1,200 cents (one octave). • Intensity of 100 equals a pitch shift of 3,600 cents (three octaves).

Target	Comments
OscWaves	Depending on the waveforms set in the three oscillators, this target can be used to modulate: • The pulse width of rectangular and pulse waves • The amount of frequency modulation (oscillator 1 only) • Noise color (oscillator 3 only) • The position of the Digiwaves OscWaves affects all oscillators simultaneously. For further information about the effects of these modulations, see Use pulse width modulation in ES2 on page 38, Use frequency modulation in ES2 on page 38, Use the ES2 noise generator on page 41, and Use ES2 Digiwaves on page 41.
Osc1Wave	Depending on the waveform selected for oscillator 1, you can control the pulse width of rectangular and pulse waves, the amount of frequency modulation, or the position of the Digiwave. In classic FM synthesizers the amount of FM is controlled in real time by velocity-sensitive envelope generators. Select one of the ENVs as the source for such sounds.
Osc2Wave	The same as Osc1Wave, except that oscillator 2 does not feature FM. Note that pulse width modulation also works with both the synchronized rectangular and ring-modulated rectangular waves.
Osc3Wave	Oscillator 3 is the same as Osc1Wave and Osc2Wave except that it does not feature FM or ring modulation. Oscillator 3 features noise, the color of which can be modulated with this parameter.
OscWaveB	The transitions between Digiwaves during a wavetable modulation (where you switch between different Digiwaves) are always smooth. You can use the OscWaveB target to continuously modulate the shape of the transitions from smooth to hard. This target applies to all oscillators.
Osc1WaveB	If wavetable modulation is active for a Digiwave (using the Osc1Wav target), you can use this target to modulate the shape of the transition. When you are frequency-modulating oscillator 1, the Osc1WaveB target offers much higher FM intensities than either the Osc1 FM or the Osc1Wave targets.
Osc2WaveB	The same as above for a Digiwave using the Osc2Wav target.
Osc3WaveB	The same as above for a Digiwave using the Osc3Wav target.
SineLev1	SineLevl (Sine Level) allows the *sine wave* level of oscillator 1 to be modulated. The parameter defines the level of the first partial tone of oscillator 1. See Sine Level enhanced ES2 sounds on page 57.

Target	Comments
OscLScle	OscLScle (Osc Level Scale) modulates the levels of *all three* oscillators simultaneously. A modulation value of *0* mutes all oscillators, whereas a value of *1* raises the gain of the entire mix by 12 dB. The modulation is applied *before* the overdrive stage, allowing for dynamic distortions.
Osc1Levl	(Osc 1 Level) allows modulation of oscillator 1's level.
Osc2Levl	(Osc 2 Level) allows modulation of oscillator 2's level.
Osc3Levl	(Osc 3 Level) allows modulation of oscillator 3's level.

ES2 filter modulation targets

The table below includes all filter-related modulation targets.

Target	Comments
Cutoff 1	Modulates the Cutoff Frequency parameter of Filter 1. See Filter cutoff and resonance overview on page 53.
Resonance 1	(Reso 1) Modulates the Resonance parameter of Filter 1.
Cutoff 2	Modulates the Cutoff Frequency parameter of Filter 2.
Resonance 2	(Reso 2) Modulates the Resonance parameter of Filter 2.
LPF FM	Determines the intensity of the lowpass filter frequency modulation (LPF FM) of Filter 2—with a sine wave (at the same frequency as oscillator 1). This parameter is described in Modulate ES2's Filter 2 Frequency on page 56.
Cut 1+2	Modulates the cutoff frequency of both filters in parallel. This is like applying the same modulation to Cutoff 1 and Cutoff 2 in two *modulation routings*.
Cut1inv2	Cut1inv2 (Cutoff 1 normal and Cutoff 2 inverse) simultaneously modulates the cutoff frequencies of the first and second filters inversely (in opposite directions). Put another way, when the first filter's cutoff frequency is rising, the cutoff of the second filter falls, and vice versa. In cases where you have combined Filter 1, defined as a highpass filter, and Filter 2 in *serial* mode, both act as a bandpass filter. Modulating the Cut1 inv 2 target results in a modulation of the bandpass filter's bandwidth in this scenario.
Filter Blend	(FltBlend) modulates the Filter Blend parameter. See Cross-fade between ES2 filters on page 50.

Other ES2 modulation targets

The table below includes all other modulation targets.

Target	Comments
Amp	This target modulates the dynamic stage, or level of voices. If you select Amp as the target and modulate it with an LFO as the source, the level changes periodically, and you hear a tremolo.
Pan	This target modulates the panorama position of the sound in the stereo or surround spectrum. Modulating Pan with an LFO results in a stereo or surround tremolo (auto panning). In unison mode, the panorama positions of all voices are spread across the entire stereo or surround spectrum. Nevertheless, pan can still be modulated, with positions being moved in parallel. The extended Surround Range parameter defines the angle range resulting from modulation values. For example, if pan is modulated by the maximum amount of an LFO (using a sawtooth waveform), a Surround Range value of 360 results in circular movements of the voice output.
Diversity	This parameter (available only in surround instances of the ES2) enables you to dynamically control how much the voice output is spread across the surround channels. Negative values reduce this effect.
LFO1Asym	(LFO1 Asymmetry) can modulate the selected waveform of LFO 1. If a square wave, it changes pulse width. If a triangle wave, it sweeps between triangle and sawtooth. If a sawtooth wave, it shifts its zero crossing.
LFO1Curve	This target modulates the waveform smoothing of the square and random wave. If the LFO is using a triangle or sawtooth wave, it changes between convex, linear, and concave curves.

Scaled ES2 modulation targets

All of the following modulation targets result in a scaled modulation, which means that the target parameter value is multiplied by the modulation value. This works as follows: a modulation value of 0.0 results in no change, a modulation value of +1.0 equals a 10x multiplication, and a modulation value of −1.0 equals a multiplication by 0.04.

Target	Comments
LFO1Rate	This target modulates the frequency (rate) of LFO 1. You can automatically accelerate or slow down LFO 1's rate by modulating the LFO1Rate target with one of the envelope generators (ENV) or with LFO2.
Env2Atck	(Envelope 2 Attack) modulates the Attack time of the second envelope generator.
Env2Dec	(Envelope 2 Decay) modulates the Decay time of the second envelope generator. In cases where you've selected Env2Dec as the target and Velocity as the source, the duration of the decaying note is dependent on how hard you strike the key. Selecting Keyboard as the source results in higher notes decaying more quickly (or slowly).
Env2Rel	Env2Rel (Envelope 2 Release) modulates the Release time of the second envelope generator.
Env2Time	Env2Time (Envelope 2 All Times) modulates *all* of ENV2's time parameters: Attack, Decay, Sustain, and Release times.
Env3Atck	Env3Atck (Envelope 3 Attack) modulates the Attack time of ENV3.
Env3Dec	Env3Dec (Envelope 3 Decay) modulates the Decay time of ENV3.
Env3Rel	Env3Rel (Envelope 3 Release) modulates the Release time of ENV3.
Env3Time	Env3Time (Envelope 3 All Times) modulates all ENV3 time parameters: Attack, Decay, Sustain, and Release times.
Glide	This target modulates the duration of the Glide (portamento) effect. If you modulate Glide, with Velocity selected as the source, the speed of the keystrike determines the time it takes for the played notes to reach the target pitch.

ES2 modulation source reference

The following modulation sources are available:

Source	Comment
LFO1	LFO 1 is used as a source.
LFO2	LFO 2 is used as a source.
ENV1	Envelope Generator 1 is used as a source.
ENV2	Envelope Generator 2 is used as a source.
ENV3	Envelope Generator 3 is used as a source. Envelope Generator 3 always controls the level of the overall sound.
Pad-X, Pad-Y	Define the axes of the Planar Pad as modulation sources for the selected modulation target. See Use the Planar Pad on page 76 and Use the Vector Envelope on page 69.
Max	Max sets the value of this source to +1. This offers interesting options for controlling the modulation intensity with all possible via values.
Kybd	Kybd (Keyboard) outputs the keyboard position (the MIDI note number). The center point is C3 (an output value of 0). Five octaves below and above, an output value of −1 or +1, respectively, is sent. Modulate the Cut 1+2 target with the Kybd source to control the cutoff frequencies of the filters with the keyboard position—as you play up and down the keyboard, the cutoff frequencies change. A modulation intensity of 0.5 proportionately scales cutoff frequencies with keyboard note pitches.
Velo	Velocity sensitivity serves as a modulation source.
Bender	The pitch bend wheel serves as a bipolar modulation source. This is also true when the Bend Range parameter of the oscillators is set to 0.
ModWhl	The modulation wheel serves as a modulation source. For most standard applications, you'll probably use the wheel as the via controller. Traditionally, it is used to control the intensity of periodic LFO modulations. Used here, it can be employed for direct, static modulations, such as controlling both filter cutoff frequencies (Target = Cut 1+2).
Touch	Aftertouch serves as a modulation source. ES2 reacts to poly pressure (polyphonic aftertouch). If you set the Target to Cut 1+2, the cutoff frequencies rise and fall, depending on how firmly you press a key on your touch-sensitive MIDI keyboard—after the initial keystrike.
Whl+To	Both the modulation wheel *and* aftertouch serve as modulation sources.
MIDI Controllers A-F	MIDI controllers available in the router are named Ctrl A–F and can be assigned to arbitrary controller numbers. See ES2 macro and controller assignment overview on page 89.

Source	Comment
RndN01	RndNO1 (Note On Random1) outputs a random modulation value between –1.0 and 1.0, that changes when a note is triggered or retriggered. The (random) note-on modulation remains constant throughout the note duration, until the next note-on trigger. There is no value change when playing legato while in legato mode.
RndN02	RndNO2 (Note On Random 2) behaves like Note On Random1, but it glides, rather than steps, to the new random value, using the Glide time (inclusive of modulation). It also differs from Note On Random 1 in that the random modulation value changes when playing legato while in legato mode.
SideCh	SideCh (Side Chain modulation) uses a side chain signal as a modulation (trigger) signal. The side chain source can be chosen from the Side Chain pop-up menu in the upper gray area of the plug-in window. It is fed to the internal envelope follower, which creates a modulation value based on the current side chain input signal level.

ES2 via modulation source reference

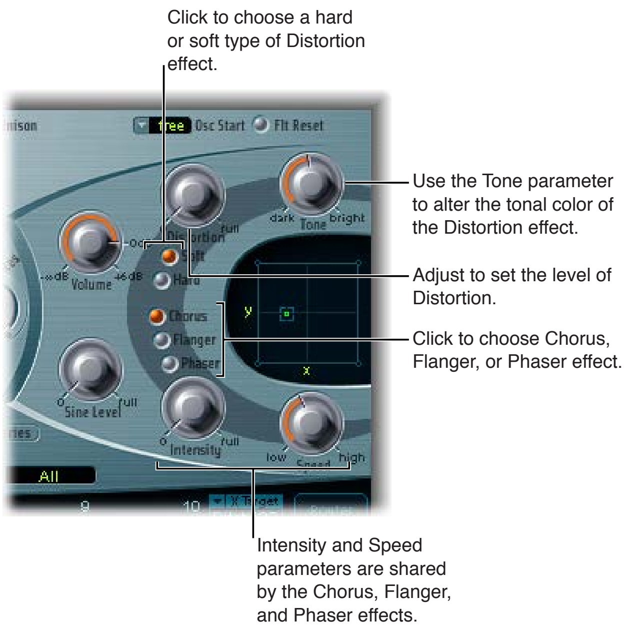
The following sources may be used to control the modulation intensity.

Via source	Comment
LFO1	The modulation undulates at the speed and waveform of LFO 1, which controls the modulation intensity.
LFO2	The modulation undulates at the speed and waveform of LFO 2, which controls the modulation intensity.
ENV1	ENV1 controls the modulation intensity.
ENV2	ENV2 controls the modulation intensity.
ENV3	ENV3 controls the modulation intensity.
Pad-X, Pad-Y	Both axes of the Planar Pad are also available as via sources, allowing you to control modulation intensities with them.
Kybd	Kybd (Keyboard) outputs the keyboard position (the MIDI note number). The center point is C3 (an output value of 0). Five octaves below and above, an output value of −1 or +1, respectively, is sent. If you select Pitch123 as the target, modulate it with the LFO1 source, and select Keyboard as the via value, the vibrato depth changes, depending on the key position. Put another way, the vibrato depth is different for notes higher or lower than the defined Keyboard position.
Velo	If you select Velo (Velocity) as the via value, the modulation intensity is velocity sensitive—modulation is more or less intense depending on how quickly (how hard) you strike the key.
Bender	The pitch bend wheel controls the modulation intensity.
ModWhl	If you select ModWhl (Modulation Wheel) as the via value, the modulation intensity is controlled by your MIDI keyboard's modulation wheel.
Touch	If you select Touch (Aftertouch) as the via value, the modulation intensity is touch sensitive—modulation is more or less intense depending on how firmly you press the key of your touch-sensitive MIDI keyboard after the initial keystrike (aftertouch is also known as *pressure sensitivity*).
Whl+To	Both the modulation wheel and aftertouch control the modulation intensity.
MIDI Controllers A-F	MIDI controllers available in the router are named Ctrl A–F, rather than Expression, Breath, and General Purpose 1–4 (MIDI Control Change Messages 16 to 19 are also known as General Purpose Slider 1/2/3/4). These can be assigned to arbitrary controller numbers with the Controller Assignments pop-up menus.

Via source	Comment
RndN01	RndNO1 (Note On Random1) outputs a random modulation intensity value between –1.0 and 1.0, which changes when a note is triggered or retriggered. The random note-on modulation remains constant throughout the note duration, until the next note-on trigger.*Note:* There is no value change when playing legato while in legato mode.
RndN02	RndNO2 (Note On Random 2) behaves like Note On Random1, but it glides, rather than steps, to the new random intensity value, using the Glide time (inclusive of modulation). It also differs from Note On Random 1 in that the random modulation value changes when playing legato while in legato mode.
SideCh	SideCh (Side Chain modulation) uses a side chain signal as a modulation intensity (trigger) signal. The side chain source can be chosen from the Side Chain pop-up menu in the upper gray area of the plug-in window. It is fed to the internal envelope follower, which creates a modulation value based on the current side chain input signal level.

ES2 integrated effects processor

ES2 is equipped with an integrated effects processor. Any changes to the parameters of these effects are saved with each sound setting.



You can activate only two effects at the same time.

- Distortion
- Choose the Chorus, the Flanger, or the Phaser effect. These effects share the same control knobs—Intensity and Speed.

  A chorus effect is based on a delay line, the output of which is mixed with the original, dry signal. The short delay time is modulated periodically, resulting in pitch deviations. The modulated deviations, in conjunction with the original signal's pitch, produce the chorus effect.

  A flanger works in a similar fashion to a chorus, but with even shorter delay times. The output signal is fed back into the input of the delay line. This feedback results in the creation of harmonic resonances that wander cyclically through the spectrum, giving the signal a "metallic" sound.

  A phaser mixes a delayed and an original signal. The delayed element is derived from an allpass filter, which applies a frequency-dependent delay to the signal. This is expressed as a phase angle. The effect is based on a comb filter, which is basically an array of inharmonic notches—rather than resonances, as with the flanger—that also wanders through the frequency spectrum.

Distortion parameters

- *Soft button:* Activates the Distortion effect Soft mode. The distortion circuit sounds somewhat like a tube overdrive.
- *Hard button:* Activates the Distortion effect Hard mode. The distortion effect sounds like a fully transistorized fuzz box.
- *Distortion knob:* Rotate to set the amount of distortion. Turn this knob to zero to disable the effect.
- *Tone knob:* Rotate to control the treble portion of the distortion signal.

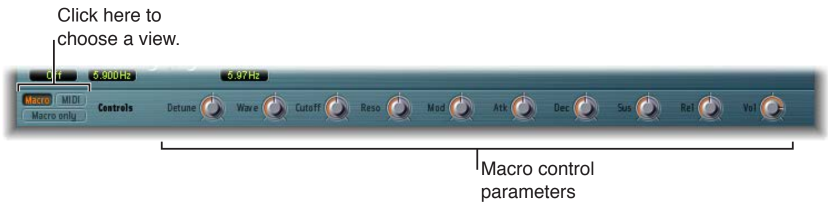
Chorus, Flanger, or Phaser parameters

- When Chorus is on:
  - *Intensity knob:* Rotate to set the depth of the effect—how "rich" the modulation is. Turn this knob to zero to turn off the effect.
  - *Speed knob:* Rotate to set the modulation rate.
- When Flanger is on:
  - *Intensity knob:* Rotate to set the depth of the effect—how "cutting" the modulation is. Turn this knob to zero to turn off the effect.
  - *Speed knob:* Rotate to set the modulation rate.
- When Phaser is on:
  - *Intensity knob:* Rotate to set the depth of the "sweeping" effect—the width of the modulation. Turn this knob to zero to turn off the effect.
  - *Speed knob:* Rotate to set the modulation rate.

ES2 macro controls and controller assignments

ES2 macro and controller assignment overview

The section at the bottom of the ES2 interface provides three views:



- *Macro:* Click to show a number of macro controls that affect groups of other parameters.
- *MIDI:* Click to assign MIDI controllers to particular modulation routings. See ES2 via modulation source reference on page 85.
- *Macro only:* Click to replace the ES2 interface with a smaller view that is limited to the macro controls.

ES2 macro controls

The macro knobs give you quick access to several linked, related parameters.

As you rotate any of the macro controls, one or more parameters in the ES2 interface update. For example, adjusting the Detune macro control simultaneously affects the Analog parameter and the coarse and fine oscillator Frequency parameters.

The macro parameters are settings-compatible with ES2-based GarageBand instruments. In other words, you can use the ES2 and some GarageBand synthesizer settings interchangeably.

Important: The impact of each macro control is completely dependent on the parameter values of the current setting. In some patches, a number of macro controls may have no effect.

Make ES2 controller assignments

The Controller Assignments area enables you to assign your MIDi keyboard's knobs, sliders, and other controls to act as control sources for ES2 parameters. There are six menus, for Ctrl A to Ctrl F. You can use any MIDI controller shown in the menus for these control sources.

These parameters are saved with each setting. They are updated only if the default setting that is loaded on instantiating the plug-in is used or if the setting was saved with a project. This approach helps you to adapt all MIDI controllers to the keyboard, without having to edit and save each setting separately.

Controllers 0 and 32 are reserved for Bank Select messages, controller 1 is used as modulation source in the router, controllers 33 to 63 work as LSB for controllers 1 to 31, controllers 64 to 69 are reserved for pedal messages, controllers 120 to 127 are reserved for channel mode messages.

In the MIDI specification, all controllers from 0 to 31 are known as Most Significant Byte (MSB) controller definitions. Each of these controllers (0 to 31) also contains a Least Significant Byte (LSB) controller definition (32 to 63). Use of this secondary LSB controller in conjunction with the MSB controller allows for a resolution of 14 bits instead of 7 bits. The ES2 recognizes these control change messages—the breath or expression controllers, for example.

To explain:

- 14-bit controllers are pairs of normal Control Change (CC) messages, where the number of the second CC message (the LSB) is 32 higher than the first CC message (the MSB). Examples of valid 14-bit pairs are: CC1/33, CC7/39, and CC10/42.
- 14-bit controllers have a resolution of 16,384 steps, allowing very precise control of plug-in parameters. The first CC message of a 14-bit pair (the MSB) has a coarse resolution of 128 steps. Each of these steps can be divided into a further 128 substeps using the second CC message (the LSB). This results in 128 x 128 = 16,384 steps.
- You don't need to create new, or special, data types to use 14-bit controllers. The finer resolution is achieved by complementing the assigned CC message (the MSB) with its LSB. The CC message assigned in the ES2 can always be used alone if your MIDI controller isn't capable of sending 14-bit messages, thus limiting the resolution to 7-bit = 128 steps.

The 14-bit capability is the reason why CC numbers 33–63 can't be assigned in the Ctrl A–F menus. Using these (LSB) CC numbers would result in changing 1/128th of the parameter range—or put another way, 128 continuous steps out of 16,384.

Assign a MIDI controller

1. Click the MIDI button in the lower-left corner to display the Controller Assignments.
2. Click any Ctrl A to Ctrl F menu, then choose the controller name/number that you want to use from the list.

Learn a MIDI controller assignment

1. Click the MIDI button in the lower-left corner to display the Controller Assignments.
2. Choose the Learn item from a control menu (Ctrl A to Ctrl F).
3. Move the selected controller on your MIDI keyboard or controller.

   Note: If no suitable MIDI message is received within 20 seconds, the selected control reverts to the previous value/assignment.

ES2 Surround mode

In surround ES2 instances, two additional parameters are shown in the slide out Extended Parameters section at the bottom of the plug-in window: Surround Range and Surround Diversity.

Surround parameters

- *Surround Range:* Determines the range of the surround angle (0 to 360 degrees). Put another way, this sets the breadth of the surround field. You can modulate the movement of sounds within the surround range by using the Pan target in the router.
- *Surround Diversity:* Determines how the output signal is distributed across your surround speakers. If you choose a value of 0, only the speakers that are closest to the original signal's position carry the signal. A diversity value of 1 distributes an identical amount of signal to all speakers. You can modulate the distribution of signals between speakers with the Diversity target in the router.

ES2 extended parameters

ES2 provides additional parameters that can be accessed by clicking the disclosure triangle at the lower left of the interface.

Extended parameters

- *MIDI Mono Mode pop-up menu:* Choose Off, On (with common base channel 1), or On (with common base channel 16).

  In either mode, each voice receives on a different MIDI channel. Controllers and MIDI messages sent on the base channel affect all voices.
- *Mono Mode Pitch Range pop-up menu:* Choose 0, 24, or 48.

  The chosen pitch bend range affects individual note pitch bend messages received on all but the common base channel. The default is 48 semitones, which is compatible with Mobile GarageBand's keyboard in pitch mode. When using a MIDI guitar, 24 semitones is the preferable setting because most guitar to MIDI converters use this range by default.

Create random ES2 sound variations

Use ES2's randomization parameters

ES2 offers a powerful feature that enables you to randomly vary sound parameters. You can define the amount of random variation and restrict variations to specific sonic elements. The random sound variation feature will inspire and aid you when creating new sounds.

You can set the amount of random parameter alteration with the Random Intensity slider.

The random sound variation feature always alters parameters as they are currently set, not based on the original setting file. Therefore, clicking RND repeatedly results in a sound that increasingly differs from the original setting.

Randomly alter a sound

- Click the Randomize button.

The randomize process is triggered by a single click and can be repeated as often as you like.

Increase the amount of random variation

- Move the Random Intensity slider farther to the right.

Create several slight variations of the current setting

- Reload the original setting after each random alteration, saving each with a new name.

Restriction of ES2 randomization

You can restrict randomization to particular groups of parameters using the Randomize Section pop-up menu.

Some aspects of your sound may already be ideal for the sound you had in mind. For example, your sound setting has a nice percussiveness, and you'd like to try a few sonic color variations while retaining this percussive feel. To avoid the random variation of any attack times, you can restrict the variation to oscillator or filter parameters. You do this by setting the RND Destination to Waves or Filters, thus excluding the envelope parameters from the variation process.

Note: The Master Level, Filter Bypass, and oscillator on/off parameters are never randomized. Also, randomizations of the Vector Envelope turn the Solo Point parameter off.

You can restrict random sound variations to the parameter groups outlined below:

Randomize section	Comments
All	All parameters, with the exception of those mentioned above, are randomized.
All except router and Pitch	All parameters, with the exception of router parameters and the basic pitch (semitone settings of the oscillators), are altered. Oscillator fine-tuning is, however, randomized.
All except Vector Env	All parameters, with the exception of Vector Envelope parameters, are altered. This maintains the rhythmic feel of a given setting.
Waves	Only the oscillator Wave and Digiwave parameters are altered. Other oscillator parameters (tuning, mix, and modulation routings in the router) are excluded.
Digiwaves	New Digiwaves are selected for all oscillators. Other oscillator parameters (tuning, mix, and modulation routings in the router) are excluded.
Filters	The following filter parameters are varied: Filter Structure (series or parallel), Filter Blend, Filter Mode, Cutoff Frequency, and Resonance for Filters 1 and 2. The Fatness and Filter FM parameters of Filter 2 are also randomized.
Envs	All parameters of all three envelopes (ENV 1, ENV 2, and ENV 3) are randomized. The Vector Envelope is *excluded*.
LFOs	All parameters of both LFOs are varied.
Router	All router parameters—in all modulation routings—are varied (all intensities, target, via, and source parameters are changed).
FX	All effects parameters are randomized.
Vector Envelope	All Vector Envelope parameters are varied, including the X/Y routing of the Planar Pad.
Vector Env Mix Pad	The oscillator mix levels of the Vector Envelope points are altered. The rhythm and tempo of the modulation (the time parameters of the points) are not changed.
Vector Env XY Pad Options	The Planar Pad pointer positions (the Vector Envelope points) are randomized. The X/Y routing, however, is not changed. The rhythm and tempo of the modulation (the time parameters of the points) are also left unaltered. You can specify a single direction for randomization by choosing either: • Vector Env XY Pad X only • Vector Env XY Pad Y only
Vec Env Times	Only the time parameters of the Vector Envelope points are altered.
Vec Env Structure	The Vector Envelope structure is altered. This includes: All times, the Sustain point, the number of points, and all loop parameters.
Vec Env Shuffle Times	The Vector Envelope shuffle times (within loops) are altered. This includes the Loop Smooth value, if Loop Mode is set to Forward or Backward.

ES2 tutorials

Create ES2 sounds from scratch

ES2 sound design from scratch overview

The “Create ES2 sounds from scratch” tutorials guide you—from the ground up (from scratch)—through the creation of commonly used sounds. The ES2 sound design with templates on page 103 tutorials also guide you through the sound creation process, but you use a number of templates as your starting point.

To see the settings for these tutorials in the ES2 window, choose Tutorial Settings from the Settings pop-up menu.

The Analog Saw Init tutorial setting is designed to be used as a starting point when you are programming new sounds from scratch. When programming entirely new sounds, professional sound designers like to use this type of setting, which has an unfiltered sawtooth wave sound without envelopes, modulations, or any gimmicks. This type of setting is also useful when you are getting to know a new synthesizer. It allows you to access all parameters without having to consider any preset values.

- Start with the filters, the heart of any subtractive synthesizer. Check out the four lowpass filter types—12 dB, 18 dB, 24 dB, and fat (Filter 2)—with different values for Cut (Cutoff Frequency) and Res (Resonance). Define Env 2 as the filter envelope. This modulation wiring is preset in the Router.
- Set Filter Blend to its leftmost position, which allows you to listen to Filter 1 in isolation. In many circumstances, you’ll probably prefer Filter 1, but Filter 2 has its advantages. In addition to the lowpass filter with 12 dB/octave slope (Lo), Filter 1 also offers a highpass, peak, bandpass (BP), and band rejection (BR) mode. Filter 2’s lowpass sounds “softer” when compared with Filter 1. It is best-suited to sounds where the filter effect is or should be less audible, such as with Strings and FM sounds. Distorted TB-303-style sounds are more easily achieved with Filter 2.
- This setting is also ideal for experimenting with different oscillator waveforms.

Create fat ES2 sounds

“Fat” synthesizer sounds have always been popular and are likely to remain so, given their use in modern trance, techno, R & B, and other styles.

Create fat ES2 sounds with oscillator detuning and unison mode

The Analog Saw 3 Osc setting features three detuned oscillators, and sounds fat as it is. The following introduces you to some additional tools to fatten the sound even more.

In many factory settings, the Unison mode is active. This demands a lot of processing power. If your computer isn’t fast enough, you can switch off the Unison mode and insert an Ensemble effect in a bus, for use with several plug-ins. This saves processing power. You can also save CPU resources by freezing or bouncing several software instrument tracks.

Do the following:

- Check out the three-oscillator basic sound with different filter and envelope settings.
- Check out the chorus effect at different Intensities and speeds.
- Engage Unison mode and select a higher setting for Analog. Because the sound is polyphonic, each note is doubled. The number of notes that can be played simultaneously is reduced from 10 to 5. This makes the sound rich and broad. Combining Unison and higher values for Analog spreads the sound across the stereo or surround spectrum.

In many factory settings, the Unison mode is active. This demands a lot of processing power. If your computer isn't fast enough, you can switch off the Unison mode and insert an Ensemble effect in a bus, for use with several plug-ins. This saves processing power. You can also save CPU resources by freezing or bouncing several software instrument tracks.

Create detuned ES2 monophonic sounds and effects
The Analog Saw Unison setting is a fat, heavily detuned, unfiltered basic sound. Three sawtooth oscillators are used, but they are further detuned. The combination of Unison and Analog (set to a high value) is essential—but this time monophonic mode is used to stack ten voices. Without further effects, the result is a huge lead sound, much like those used in countless dance and trance productions. With appropriate filter and envelope settings, electro sounds that are ideal for arpeggiation and sequencing can easily be set up.

Do the following:

- Set the Cutoff Frequency of Filter 2 to 0. This activates the preset filter envelope. Feel free to check out different envelope settings.
- Switch Osc 1 to sound one or two octaves lower.
- Increase Drive or Distortion.
- Set Env 2 to be velocity sensitive. This allows for velocity-sensitive filter modulations.
- Insert a delay effect in the instrument channel strip of the ES2 (or a bus target).

Create ES2 bass sounds

Not every sound needs to consist of several oscillators. There are numerous simple, effective, sounds that make use of a single oscillator. This is especially true of synthesizer bass sounds, which can be created quickly and easily with the Analog Bass Clean setting.

Create clean single-oscillator bass sounds
In the Analog Bass Clean setting, the basic sound is a rectangular wave that is transposed down by one octave. The sound is filtered by Filter 2. What's special about this sound is its combination of Legato and Glide (portamento). When you play staccato, no glide effect occurs. When you play legato, the pitch smoothly glides from one note to another. All keys must be released before you strike a new key, in order to retrigger the envelopes.

Do the following:

- Check out different filter and envelope settings.
- Replace the rectangular wave with a sawtooth.
- Vary the Glide settings.

Tip: It's best to make your edits while a bass line is playing. Create or play a monophonic bass line, with most notes played staccato, but some legato. This can provide some interesting results with very long Glide values.

Create distorted analog basses
In the *Analog Bass Distorted* setting, Filter 1 is engaged, with high settings for Drive and Distortion. This filter is better suited to the creation of distorted analog sounds than Filter 2.

Do the following:

- Check out Filter 2 by setting Filter Blend to its rightmost position. Notice that Filter 1 works better with distorted sounds.
- To control the filter modulation, move the green sliders of the first modulation routing in the router. This controls the modulation intensity.

Create ES2 FM sounds

In the ES2, oscillator 1 is always the carrier, and oscillator 2 the modulator. In other words, oscillator 2 modulates oscillator 1.

The FM Start setting is great for familiarizing yourself with linear frequency modulation (FM) synthesis.

Use FM Intensity and Frequency to create new sounds

Load the FM Start setting to hear an unmodulated sine sound, generated by oscillator 1. Oscillator 2 is switched on and set to produce a sine oscillation as well, but its level is set to 0: Drag the small square in the uppermost corner of the Triangle to change the settings.

Do the following:

- Adjust the intensity of the frequency modulation by slowly moving the wave selector from Sine to FM. You will hear a typical FM spectrum, with the carrier and modulator set to the same frequency.
- Alter the modulator frequency (oscillator 2) by adjusting Fine Tune from 0 c to 50 c. You'll hear a very slow frequency modulation, which can be compared to the effect of an LFO. The frequency modulation, however, takes place in the audio spectrum. It is adjusted in semitone steps by the frequency selector. Check out the entire range from –36 s to +36 s for oscillator 2. You'll hear a broad spectrum of FM sounds. Some settings will remind you of classic FM synthesizer sounds.
- Select other waveforms for oscillator 2. Sine is the classic, standard FM waveform, but other waveforms lead to interesting results as well, especially the Digiwaves.
- You will achieve further interesting results by altering the carrier (oscillator 1) frequency. Check out the entire range, from –36 s to +36 s here, as well. The odd intervals are especially fascinating. Note that the basic pitch changes when you do this.

Control ES2 FM intensity with an envelope and FM scaling

In the FM Envelope setting, you can control the FM intensity with an envelope, generated by Envelope 2. The modulation target is the range that falls between Sine and FM in the oscillator wave selector. The first Router channel is used for this modulation routing. You can control a wider range by using additional modulation routings, which have been pre-prepared for you. All you need to do is set their values. Because these modulations work without velocity sensitivity, you can set them in the Editor view by moving both the lower and upper fader halves to their topmost positions.

Do the following:

- Set the second modulation routing to 1.0. You'll hear how the modulation now "wanders" through a broader sound range.
- Set modulation routings 3 and 4 to a value of 1.0 as well, and listen to the increase in the sound range.
- After these drastic augmentations to the modulation range, the sound becomes uneven across the keyboard. In the lower and middle ranges it sounds nice, but in the upper key range the FM intensity appears to be too severe. You can compensate for this effect by modulating the Osc 1 Wave target by keyboard position (kybd) in modulation routings 5 and 6. This results in a keyboard scaling of the FM intensity.
- Because the sound range is so vast (due to the four modulations), two modulation routings are required to compensate for this. Set the lower slider halves to their lowest positions. Good keyboard scaling is essential for any FM sound.

Use FM Drive and Filter FM to change the tonal color

The FM Drive setting illustrates how dramatically the character of FM sounds can be altered when you apply Drive and Filter FM. The results are reminiscent of the feedback circuits of classic FM synthesizers.

Do the following:

- Check out different Drive and Filter FM settings.
- Lower the Cutoff Frequency of Filter 2 to 0. Envelope 2 modulates Filter 2. This modulation routing is already present in the setting.

Create FM sounds with Digiwaves

In the FM Digiwave setting, a Digiwave is used as an FM modulator. This results in bell-like spectra from only two operators. With traditional FM synthesis, this type of timbre could normally be produced only with a larger number of sine oscillators.

To create a fatter, undulating, and atmospheric quality to the sound, the polyphonic Unison mode has been engaged. Filter and amplitude envelopes have been preset to shape the sound.

Do the following:

- Check out the variety of Digiwaves as FM modulation sources.
- Check out different Analog parameter values.

Create FM sounds with wavetables

You can program the most vivid FM sounds when the modulation source morphs between different Digiwaves. The morphing in the FM Digiwave setting is controlled by LFO 2. The tempo of LFO 2, and therefore the morph, depends on the host application tempo—here, two bars.

Do the following:

- Set LFO 2 to different waveforms. Lag S/H (smooth random), in particular, should be fun.
- Check out different FM intensities and oscillator frequencies.
- Alter the modulation intensity of the first modulation routing (LFO2 modulates Osc2 Wave) and the LFO 2 rate.

Create distorted FM sounds with monophonic unison

The FM Megafat setting is well-suited for distorted basses and guitar-like sounds. This sound gets rather "rude" in the upper key range. This cannot be compensated for with key scaling, but not every sound has to be "nice" across the entire keyboard range!

Do the following:

- Check out extreme detunings by adjusting the Analog parameter.
- Check out the Flanger with this sound.
- Engage the filter envelope by lowering the Cutoff Frequency of Filter 2 down to 0.
- Add some Glide to lead sounds.
- As always when it comes to FM, you can dramatically alter the sound by varying the frequencies of the oscillators. Make sure you check out the odd intervals, as well.

Create FM sounds with unusual spectra

If you're unconcerned with the pitch of your sound, you can get the weirdest spectra out of odd frequency ratios—oscillator intervals.

The FM Out of Tune setting offers a bell-like sound, reminiscent of a ring modulator. It was achieved through a setting of 30 s 0 c, with the modulator set to a value of 0 s 0 c. Sounds like this were commonly used in the electronic music of the eighties and have had a resurgence in popularity in ambient and trance music styles.

You can further develop the sound by applying filtering, envelope modulations, and effects. There is, however, one small problem—the sound is out of tune.

Do the following:

- Use oscillator 3 as a reference for the tuning of the FM sound by dragging the pointer in the Triangle.
- You'll notice that the sound is five semitones too high (or seven semitones too low, conversely).
- Transpose both oscillators 1 and 2 five semitones (500 ct) lower. Transposing them upward is not practical, as you'd need to select 37 s 0 c for oscillator 1, which has a maximum value of 36 s 0 c.
- It's important to maintain the frequency ratio (interval) between oscillators 1 and 2. This means that oscillator 1 sounds at 25 s O c and oscillator 2 at –5 s 0 c.

Create ES2 PWM sounds

Pulse width modulation (PWM) is one of the most essential features of any analog synthesizer.

Set up a basic PWM sound

- Choose the PWM Start setting, and move the Wave control slowly back and forth between the rectangular and the pulse wave symbols. Both are green. What you will hear is a manual pulse width modulation.
- Choose the PWM Slow setting. Here, LFO 1 controls the pulse width modulation source, not your manual movements. The result should be quite similar.
- Raise the LFO 1 rate from its preset value of 0.230 to 4.400. The result is a classic, fast PWM.
- In this and the next step, the PWM is set so that it sounds slower in the lower keyboard range and faster in the upper range. This is desirable for many sounds, such as synthetic strings. First, reduce the LFO 1 Rate to 3,800.
- Change the modulation intensity of the second router channel (target = LFO1 Rate, Source = Kybd) to 0.46. This alters the scaling of the PWM, making it sound faster in the treble range. You can also hear this type of effect in the PWM Scaled setting.

  Tip: Avoid Drive and Distortion with PWM sounds.

To make the sound fatter

- Add oscillator 3, which can also be pulse width modulated. In fact, even the first oscillator can deliver PWM. In the PWM 2 Osc setting, both oscillators are detuned quite significantly. Develop your own personalized PWM string sound, using this setting as your base.
- Adjust the Chorus intensity. You'll probably choose higher values, which make the sound rather broad.
- Program Envelope 3 according to your taste. You should, at the very least, raise the attack and release times. Define it to react to velocity, if you prefer. If you want to use the sound for something other than a simple pad, a shorter Decay Time and a lower Sustain Level of about 80 to 90% may be more appropriate.
- Reduce the Cutoff Frequency and Resonance of Filter 1 to make the sound softer.
- Save the new setting.
- Compare the result with the original PWM 2 Osc setting. You'll hear that the sound has undergone a remarkable evolution.
- Also compare it to PWM Soft Strings, which was created as described above. You'll probably notice a few similarities.

Create ES2 ring modulated sounds

A ring modulator takes its two input signals and outputs their sum and difference frequencies.

In the ES2, oscillator 2 outputs a ring modulation, which is fed with a square wave of oscillator 2 and the wave of oscillator 1, when Ring is set as oscillator 2's waveform.

Odd intervals (frequency ratios) between the oscillators result in bell-like spectra, much like those heard in the Ringmod Start setting.

The third oscillator can be used as a tuning reference, to maintain a kind of basic tuning. On occasion, you may find that it's nice to leave the sound out of tune—for use as a source of overtones and harmonics for another basic wave, supplied by oscillator 3.

Create an atmospheric bell sound

Try the following with the Ringmod Start setting:

- Experiment with the various frequency ratios of oscillators 1 and 2. You may want to use the 29 s 0 c/21 s 0 c ratio, which doesn't sound out of tune at all. Ring modulation is not only useful for bell-like sounds, it's also good for a great variety of spectra that tend to sound weird at lower frequency settings. Also try alterations to the fine-tuning of the oscillators.
- Check out an Intensity of 50% and a Rate set to around 2/3 of the maximum value for the Chorus effect.
- Set the Attack and Release Times of Envelope 3 to taste.
- Check out Drive and Filter FM if you like your sounds a little "out of control."
- The rest is up to you.

Create ES2 oscillator sync sounds

If you select the synced square and sawtooth waveforms for oscillators 2 and 3, they are synchronized with oscillator 1. In the Sync Start setting, only oscillator 2 is audible, and oscillator 3 is switched off.

Typical sync sounds feature dynamic frequency sweeps over wide frequency ranges. These frequency modulations (the sweeps) can be applied in various ways.

Enhance the Sync Start setting

Do the following:

- Try the pre-programmed pitch modulation, assigned to the modulation wheel first.
- In the second router channel, an envelope pitch modulation has been preprogrammed (target = Pitch 2, Source = Env 1). Setting the minimum value to 1.0 results in a typical sync envelope. Also check out shorter Decay Times for Envelope 1.
- To avoid a sterile, lifeless sound (after the decay phase of the envelope), you may also want to modulate the oscillator frequency with an LFO. Use the third router channel, and set the minimum modulation applied by LFO 1 to about 0.50.
- Substitute the synchronized square wave with the synced sawtooth wave, and see if you like the results.

Note: Pulse width modulation is also available via the synchronized square wave of oscillators 2 and 3. A modulation of the wave parameters of these two oscillators results in a PWM when the synced square wave is selected.

Vector synthesis in the ES2

This tutorial provides some hints for Vector Envelope programming.

Familiarize yourself with the Vector Envelope

In the Vector Start setting, the "mix" of the oscillators is controlled by the Vector Envelope. Each oscillator has been set to a different waveform.

- Switch from the Router view to Vector view.
- In its basic (default) setting, the Vector Envelope has three envelope points. Point 1 is the start point, point 2 the Sustain point, and point 3 is the target in the release phase. By clicking the points, you can see that the mix is always set to 100% for oscillator 1, in the Triangle.
- Click point 2, and drag the pointer in the Triangle to oscillator 2. You'll hear a square wave, instead of oscillator 1's sawtooth.
- Engage the Vector Envelope by switching the Solo Point parameter off. When it is switched on, you hear only the selected point, with no dynamic modulation. When Solo Point is switched off, you'll hear the sound moving from saw to square, with every triggered note.
- Alter the preset time of 498 ms between points 1 and 2.
- While holding down Shift, click between points 1 and 2. A new point 2 is created, and the point formerly known as point 2 becomes point 3. The total time span between point 1 and point 3 is divided into the times between points 1 and 2, and 2 and 3. The division takes place at the click location. If you click at the exact midpoint, the new time spans are equal.
- Click the newly created point 2, and then drag its corresponding pointer in the Triangle to oscillator 2.
- Click point 3, and drag its corresponding pointer in the Triangle to oscillator 3. Listen to the three oscillators morphing from sawtooth to square to a triangular wave at the final Sustain point.

- Click point 4 (the end point) and drag its corresponding pointer in the Triangle to oscillator 1, if it's not already there. Listen to how the sound returns to oscillator 1's sawtooth wave, following the release of the key.

Vector synthesis with the Planar Pad

The Vector Envelope setting starts where the Vector Start setting left off. You have a simple Vector Envelope consisting of four points, which is set to modulate the oscillator mix (the Triangle).

In this example, the Vector Envelope is used to control two additional parameters—the Cutoff Frequency of Filter 2 and Panorama. These are preset as the X and Y targets in the Planar Pad. Both have a value of 0.50.

Do the following:

- Switch on Solo Point, to more easily listen to the settings for the single points.
- Click point 1. You will hear only oscillator 1's sawtooth.
- Drag the pointer in the Planar Pad to the far left, which results in a low cutoff frequency for oscillator 2.
- Click Point 2. You will hear only oscillator 2's rectangular wave.
- Drag the pointer in the Planar Pad all the way down, which results in the rightmost panorama position.
- Click Point 3. You will hear only oscillator 3's triangular wave.
- Drag the pointer in the Planar Pad all the way up, which results in the leftmost panorama position.
- Switch on Solo Point. The sound begins with a strongly filtered sawtooth wave and turns into an unfiltered square wave. It initially sounds from the right, and then it moves to the left while morphing into a triangular wave. After you release the key, the saw sound is heard.

Use Vector synthesis loops

The basic sound of the Vector Loop setting—without the Vector Envelope—consists of three elements:

- Oscillator 1 delivers a metallic FM spectrum, modulated by oscillator 2's wavetable.
- Oscillator 2 outputs cross-faded Digiwaves (a wavetable), modulated by LFO 2.
- Oscillator 3 plays a PWM sound at the well-balanced, and keyboard-scaled, speed of LFO 1.

These heterogeneous sound colors are used as sound sources for the vector loop. Unison and Analog make the sound fat and wide.

A slow, forward loop is preset. It moves from oscillator 3 (PWM sound, point 1) to oscillator 1 (FM sound, point 2), then to oscillator 3 again (PWM, point 3), then to oscillator 2 (wavetable, point 4), and finally it returns to oscillator 3 (PWM, point 5). Points 1 and 5 are identical, which prevents any transition from point 5 to point 1 in the forward loop. This transition could be smoothed out with Loop Smooth, but this would make the rhythmic design more difficult to program.

The distances between the points of the Vector Envelope have been set to be rhythmically exact. Given that Loop Rate has been engaged, the time values are not displayed in ms, but as percentages. There are four time values (each at 25%), which is a good basis for the transformation into note values.

Do the following:

- Switch off the Vector Envelope by setting Solo Point to on. This allows you to audition the individual points in isolation.
- Take the opportunity to alter the pointer positions in the Planar Pad according to your taste. The X/Y axes of the Planar Pad control the cutoff frequency of Filter 2, and the panorama position. Adjustments to these make the sound more vivid.
- Activate the Vector Envelope by setting Solo Point to off. Check the result, and fine-tune the pointer positions in the Planar Pad.
- Alter the Loop Rate from the preset value of 0.09 up to 2.00. You will hear a periodic modulation, much like that of an LFO. At this point, the modulation is not synchronized with the project tempo. To synchronize the loop speed with the project tempo, move the Rate to the far left, and set a note or bar value.
- You can create faster rhythmic note values by clicking between two points and setting the new time values—which result from the division that occurs—to a value of 12.5%, for example.

Create kick drums with a self-oscillating filter and the Vector Envelope
Electronic kick drum sounds are often created with modulated, self-oscillating filters. This approach can also be taken with the ES2, particularly when the Vector Envelope is used for filter modulation. An advantage of the Vector Envelope, in comparison with conventional ADSR envelopes, is its ability to define and provide two independent decay phases. The distortion effect applies the right amount of drive without sacrificing the original sonic character of the drum sound.

Note: To make the setting really punchy, you must activate Flt Reset, because all oscillators are switched off in this setting, and the filter needs a little time to start oscillating. At the start of each note, Flt Reset sends a very short impulse to the filter—making it oscillate from the outset.

By tweaking the Vector Kick setting you can create any dance-floor kick drum sound you can think of.

Change the following parameters to create sound variations:

- Filter 2 slopes: 12 dB, 18 dB, 24 dB
- Distortion: Intensity and Soft or Hard
- Envelope 3's Decay Time: (D)
- Vector Envelope Time 1 > 2: preset to 9.0 ms
- Vector Envelope Time 2 > 3: preset to 303 ms
- Vector Time Scaling

Create percussive synthesizer and bass sounds with two filter decay phases
As with the Vector Kick setting, the Vector Perc Synth setting uses the Vector Envelope to control the filter cutoff frequency, with two independently adjustable decay phases. This would not be possible with a conventional ADSR envelope generator.

Try creating further percussive synthesizers and basses by varying these parameters:

- Vector Envelope Time 1 > 2 (= Decay 1)
- Vector Envelope Time 2 > 3 (= Decay 2)
- Vector Time Scaling
- Points 1, 2, and 3 (= Cutoff Frequency) in the Planar Pad
- Waveforms (choosing other waveforms)

Create ES2 sounds with templates

ES2 sound design with templates

There are a number of tutorial templates that you can open from the Settings pop-up menu (choose the Tutorial Settings folder).

This programming tour of the ES2 is included as a part of the toolbox to help you learn the ES2's architecture through experimentation with these template sounds.

As you become more familiar with ES2 functions and parameters, you can create your own templates to use as starting points when designing new sounds.

ES2 Slapped StratENV setting

The target of this setting is the sound of a Stratocaster, with the switch between bridge and middle pickup in the middle position—in phase. It attempts to emulate the noisy twang typical of this sound's characteristics. This might be a useful template for emulations of fretted instruments, harpsichords, clavinets, and so on.

Take a look at the sound's architecture:

Osc 1 and Osc 3 provide the basic wave combination within the Digiwave field. Changing the Digiwaves of both in combination delivers a huge number of basic variations—some also work pretty well for electric piano-type keyboard sounds.

Osc 2 adds harmonics with its synced waveform, so you should only vary its pitch or sync waveform. There are a couple of values that can be changed here, which will give you a much stronger, more balanced signal.

An old trick, which delivers a punchy attack, was used—to create an effect that the use of a naked wave wouldn't deliver, even with the best and fastest filters available: You use an envelope (in this case, Env 1) for a quick "push" of a wavetable's window—or all wavetables together, where it makes sense.

Set up Envelope 1's decay time for this short push by moving the wave selectors for all oscillators. (Although it makes no sense to do this on the synced sawtooth oscillator, Osc 2, use the envelope trick regardless.) This allows you to vary the punchiness of the content between:

- Envelope 1's contribution to the overall attack noise and changing decay length—a short decay results in a peak, a long decay results in a growl, as Envelope 1 reads a couple of waves from the wavetable.
- Modulation destination—you can always assign this to each of the oscillators separately.
- Start point—you vary the wave window start with minimum and maximum control of EG1/Osc. waves modulation: negative values for a start wave before the selected wave, positive values for a start wave from a position behind the selected wave that rolls the table back.

Feel free to experiment with this wavetable-driving trick. The growl effect works well for brass sounds, and some organs absolutely shine with a little click, courtesy of a wavetable push.

Envelope 2, which controls the filter, provides a slight attack when used for "slapped" characteristics. Setting it to the fastest value eliminates the wah-like attack, while retaining the punch.

For playing purposes, you'll find that LFO 2 is used as a real-time source for vibrato. It is assigned to the mod wheel and pressure. Feel free to change the wheel and pressure settings

Velocity is set up to be very responsive, because many synthesizer players don't strike keys like a piano player would with a weighted-action "punch." Therefore, you should play this patch softly, or you may find that the slap tends to sweep a little. Alternatively, you can adjust the sensitivity of the filter modulation's velocity value to match your personal touch.

If required, increase the Voices value to maximum—six strings should be enough for a guitar, but for held or sustained notes, a few extra voices may come in handy.

ES2 Wheelrocker setting

This ordinary organ patch doesn't hold any deep, high-end sound design secrets—it is just a combination of three oscillators with mixed wave levels. You'll probably find a different combination that more closely matches your vision of what an organ sounds like. Check out the Digiwaves.

Focus your attention on the mod wheel's response—hold a chord, and bring the wheel in by moving it slowly upward until you reach the top (maximum). The intention behind the programming of this mod wheel modulation is to simulate an accelerating Leslie rotor speaker.

The modulation routings do the following:

- Modulation routing 1 assigns envelope 2 to Filter 1—the only one used for this patch—and produces a little organ key click with the envelope. The filter is opened slightly (with Keyboard as via) when you play in the higher keyboard range, with the maximum value.
- Modulation routings 2 and 3 bring in LFO 1 vibrato, and both oscillators are modulated out of phase.
- Modulation routing 4 does not need to be adjusted, but you are free to do so. It has been set up to use ENV1 to "push" the wavetable. Adjust ENV1 Decay to make the sound more pipe organ-like. Adjust ENV1 Attack to sweep through the wavetable.
- Modulation routing 5 reduces the overall volume according to personal taste, but the organ's level shouldn't increase too drastically when all modulations are moved to their respective maximums.
- Modulation routings 6 and 7 detune oscillators 2 and 3 against each other, within symmetrical values—to avoid the sound getting out of tune overall. Again, both work out of phase with modulation routings 2 and 3; oscillator 1 remains at a stable pitch.
- Modulation routing 8 brings in LFO 1 as a modulator for panorama movement—this patch changes from mono to stereo. If you prefer a full stereo sound with a slowly rotating Leslie in its idle position, just set an amount equal to the chosen minimum value, thereby achieving a permanent, slow rotation. Another modification you may wish to try is a higher value, resulting in more extreme channel separation.
- Modulation routing 9 speeds up LFO 2's modulation frequency.
- Modulation routing 10: A little cutoff was added to Filter 1, increasing the intensity of the big twirl.

Feel free to find your own values. While doing so, keep in mind the fact that there are two modulation pairs that should only be changed symmetrically—modulation routings 2 and 3 work as a pair, as do modulation routings 6 and 7. So, if you change Pitch 2's maximum to a lower minus value, remember to set Pitch 3's maximum value to the same positive amount—the same rule applies for modulation routing pair 6 and 7.

You can also use LFO 2 to increase the pitch diffusion against LFO 1's pitch and pan movements. Just exchange it for LFO 1 on modulation routings 2 and 3. Note that there will be no modulation source for the Leslie acceleration, so you'll need to use it in a static way, just fading it in. Alternatively, you'll need to sacrifice one of the other modulations in favor of a second twirl.

For another stereo modification of the static sound, you can use the patch in Unison mode with a slight detune—make sure to adjust the analog parameter for this.

ES2 Crescendo Brass setting

The oscillators are used for the following tasks:

- Oscillator 1 provides the basic brass wave—sawtooth.
- Oscillator 2 provides a not particularly "brassy" pulse wave, which brings in the ensemble. It is pulse-width modulated by LFO 1 (modulation routing 4).

Note: The following critical point should be taken into account for any modulations. There are four parameters that behave in an entirely different fashion when any one of them is changed. Therefore, all four must be changed when making adjustments:

- You may adjust the initial pulse width of oscillator 2's wave parameter. A "fat" position, close to the ideal square wave, has been chosen for this patch—in order to program a full, voluminous synth-brass sound.
- Modulation routing 4 adjusts the modulation intensity—how far the range differs from fat to narrow when being pulse-width modulated. Set with the Minimum parameter.
- The rate of LFO1 directly controls the speed of the movement of the pulse width modulation. For this patch, both LFOs are used, to achieve a stronger diffusion effect at different modulation speeds.

  Tip: You should use LFO1 for all permanent, automatic modulations because you are able to delay its impact with its EG parameter. You can use LFO 2 for all real-time modulations that you intend to access via ModWheel, pressure, or other controls while playing.
- A keyboard assignment was set up as the source for modulation routing 4. This is because all pitch, or pulse-width, modulations tend to cause a stronger detuning in the lower ranges, while the middle and upper key zones feature the diffusion effect. When using this parameter, you should initially adjust the lower ranges until an acceptable amount of detuning (resulting from the modulation) is reached. When set, check whether or not the modulations in the upper zones work to your satisfaction. Adjust the relationship between intensity (Max) and scaling (Min) values.

Oscillator 3 generates a Digiwave, which is "brassy" enough, within the overall wave mix. As an alternative to the Digiwave, another modulated pulse wave could be used to support the ensemble, or another sawtooth wave—to achieve a "fatter" sound, when detuning it with oscillator 1's sawtooth wave.

The primary aim, however, is to have a little bit of "growl," achieved through a short wavetable push, as described in ES2 Slapped StratENV setting on page 103. This configuration is set up in modulation routing 3 (oscillator 3 Wave moved by Envelope 1's Decay).

Other controls have a variety of functions:

- Envelope 1 also affects the pitch of oscillator 2 against oscillator 3. This results in both pitches clashing with each other, and also with the stable pitch of oscillator 1 in the attack phase of the sound.
- The filter envelope's design closes with a short stab in the attack phase, then reopens for a slower crescendo phase.
- A further real-time crescendo has been assigned to the mod wheel, which also brings in an overall pitch modulation, controlled by LFO 2.
- In addition to all of this, a "contrary" real-time modulation by pressure—which closes the filters—has been programmed. This allows you to play with an additional decrescendo, remotely controlled by touch. Try to get a feel for the patch's response. You'll find that it offers quite a few controls for expression—velocity, pressure after note-on, and pressure in advance. Listen to what happens when you press with the left hand before hitting a new chord with the right hand and allowing the swell to come in.

ES2 MW-Pad-Creator setting

This is an attempt to create a patch that is able to automatically generate new patches.

Oscillator 2 is used for a pulse width modulation—which creates a strong ensemble component (for more information, see ES2 Crescendo Brass setting on page 105).

Oscillators 1 and 3 are set to an initial start wave combination within their respective Digiwave tables. You can modify these, if you wish, and start with a different combination of Digiwaves from the outset.

Modulation 3 "drives" the wavetables of all three oscillators, via the mod wheel. Stated simply, you can simultaneously scroll through the oscillator 1 and oscillator 3 wavetables, and change oscillator 2's pulse width—by moving the mod wheel.

Try a careful, very slow movement of the mod wheel, and you'll hear drastic changes within the wave configuration. Each incremental position of the wheel offers a different digital pad sound. Avoid rapid movements, or this will sound like an AM radio.

Another potential modification procedure is hidden in the modulation intensity of the oscillator 1, 2, and 3 wave parameters. The value of this intensity parameter assigns both the step width and direction through the wavetables. You can try modifications to the amount using positive or negative values.

An interesting side-effect of FM assignment to Filter 2 (modulation routing 4—Lowpass Filter FM) occurs when the mod wheel is moved to higher positions: the frequency modulation of the filter is increased, causing all cyclical beats (vibrating pitches, detunes, pulse width) to be emphasized. This also adds a rough, "hissing" quality to the overall sound character. FM offers vast scope for experimentation, and you can decide between:

- An initial FM, using Filter 2's FM parameter, which you can redraw (set a negative modulation amount for modulation routing 4's maximum) by moving the mod wheel to its top position.
- Permanent FM (and another modulation setup, saved for a different assignment). You can also switch off FM, if you consider its effect too dirty sounding.

Real-time control is via pressure for a vibrato (modulation routing 10) and also for a slight opening of the Cutoff to emphasize the modulation (modulation routing 9).

ES2 Wheelsyncer setting

Never obsolete—and undergoing a renaissance in electronic music—are sync sounds.

Wheelsyncer is a single-oscillator lead sound; all other oscillators are switched off.

Although oscillator 2 is the only one actively making any sound, it is directly dependent on oscillator 1.

If you change oscillator 1's pitch or tuning, the overall pitch of the sound will go out of tune or will be transposed.

The pitch of oscillator 2 provides the tone-color (or the harmonics) for the sync sound. Pitch changes are controlled by modulation routing 7—oscillator 2 pitch is assigned to the mod wheel.

If you move the wheel, you can scroll through the spectrum of harmonics that have been programmed—for real-time changes. Any modification here starts with the pitch of oscillator 2 itself, which is set to 3 semitones below the overall pitch. Feel free to start with a different pitch for oscillator 2; it won't affect the tuning of the sound.

The next modification may be modulation routing 7's intensity (or the interval). The maximum value has been chosen—if this is too extreme for your needs, feel free to reduce it.

Another modification lies in the tone color of the lead sound itself. oscillator 1 is switched off, because the patch is OK as it is. If you switch it on, all oscillator 1 waveforms—including Digiwaves, standard waveforms, or a sine wave (which can be further modulated by FM)—are available for use.

All real-time controls are via the mod wheel, which is used for opening the filter on modulation routing 6, a panning movement on modulation routing 8, and acceleration of panning movement on modulation routing 9. If you have deeper modulation ambitions, a similar setup is used for a Leslie speaker simulation in the Wheelrocker setting (see ES2 Wheelrocker setting on page 104).

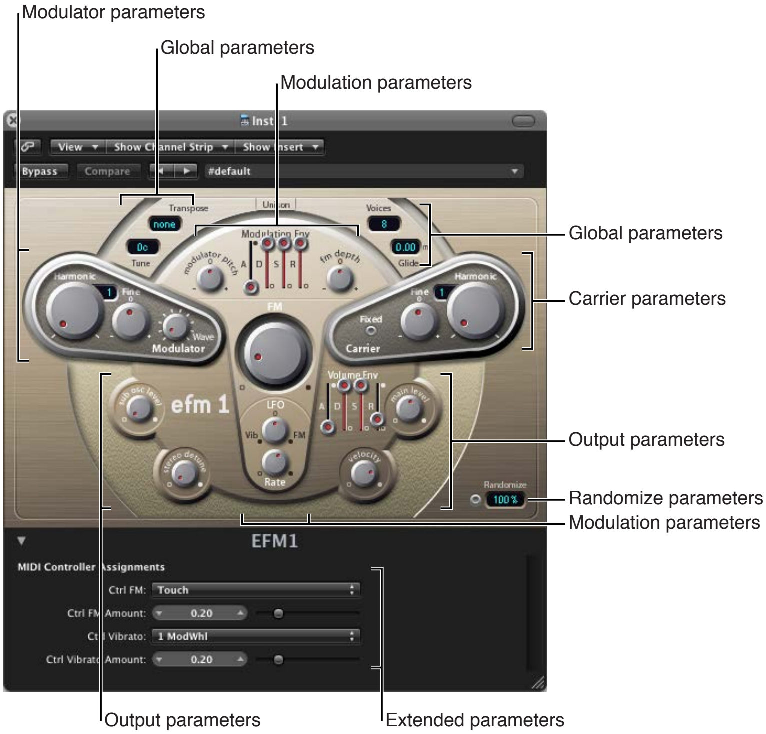
EFM1

4

EFM1 overview

The 16-voice EFM1 is a simple but powerful frequency modulation synthesizer. It can produce the rich bell and digital sounds that frequency modulation (FM) synthesis has become synonymous with.

If you're new to synthesizers, see Synthesizer basics overview on page 471, which will introduce you to the terminology and give you an overview of various synthesis systems and how they work.



EFM1 is divided into several areas.

- *Global parameters:* The top section contains parameters that set the overall tuning of EFM1. Further controls enable you to set the Glide (portamento) time, limit the number of voices, and thicken the sound with Unison. See EFM1 global parameters on page 114.
- *Modulator and Carrier parameters:* The FM engine consists of the modulator and carrier parameters (raised, darker sections), and the FM Intensity knob (in the center). These are the key controls for setting the basic tone of EFM1. See Modulator and carrier overview on page 110.
- *Modulation parameters:* The modulation envelope and LFO at the top and bottom of the mushroom-shaped area in the center respectively are used to animate the sound. See EFM1 modulation parameters on page 113.
- *Output parameters:* The bottom section houses the Output section, which includes the Sub Osc Level and Stereo Detune knobs that can be used to thicken the sound. The volume envelope, Main Level, and Velocity controls are used to set the EFM1 level. See EFM1 output parameters on page 115.
- *Randomize parameters:* The Randomize field and button in the lower-right corner are used to create random variations of the current settings, resulting in new sounds. See Create random EFM1 sounds on page 116.
- *Extended parameters:* Accessed by clicking the disclosure triangle at the lower left of the interface, these parameters allow you to assign MIDI controllers to the FM Depth and Vibrato parameters. See EFM1 extended parameters and EFM1 MIDI controller assignments on page 117.

EFM1 modulator and carrier parameters

Modulator and carrier overview

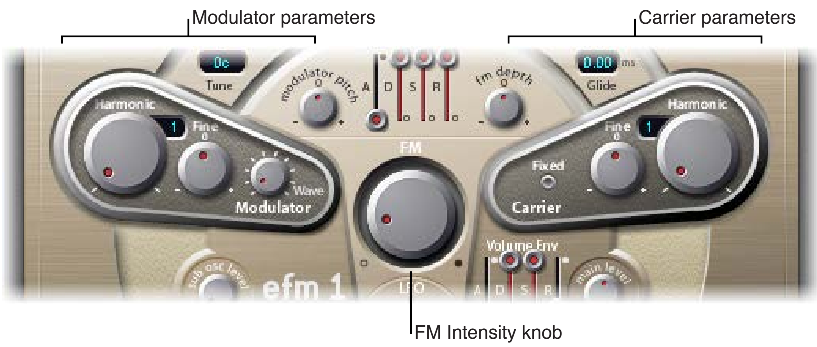
In FM synthesis, the basic sound is generated by setting different tuning ratios between the modulator and carrier oscillators and by altering the FM intensity. The tuning ratio determines the basic overtone structure, and FM intensity controls the level of these overtones.

At the core of the EFM1 synthesis system is a multiwave modulator oscillator and a sine wave carrier oscillator. The basic sine wave of the carrier oscillator is a pure, characterless tone.

To make things more sonically interesting, you use the modulator oscillator to modulate the frequency of the carrier oscillator. This modulation occurs in the audio range—you can hear it—and results in a number of new harmonics becoming audible.

The pure sine wave of the carrier oscillator is combined with the newly generated harmonics, making the sound more interesting.

Changes to the ratio of the two oscillators is achieved by adjusting the Harmonic parameters, found in both the Modulator and Carrier sections. Additional tuning control is provided by the Fine (tune) parameters.



Modulator parameters

- *Harmonic knob:* Rotate to set the tuning ratio between the modulator (left) and carrier (right) oscillators. See Set the EFM1 tuning ratio on page 112.
- *Fine (tune) knob:* Rotate to adjust the tuning between two adjacent harmonics, as determined by the Harmonic knobs of both oscillators. The range of this control is ±0.5 harmonic. In the center (0) position, Fine tune does not have an effect. Click the "0" to center the Fine tune knob. Depending on the amount of detuning, you will hear one of the following:
  - A subtle "beating" of the timbre if lower detuning amounts are used
  - New harmonic and inharmonic overtones if high detuning amounts are used
- *Wave knob:* Rotate to choose a different waveform for the modulator oscillator. See Choose a different EFM1 modulator waveform on page 112.
- *FM (Intensity) knob:* Rotate to set the amount of carrier oscillator frequency modulation by the modulator oscillator. As you adjust the FM knob, the intensity—and number—of overtones increases, making the sound brighter.

  Note: Although the technology behind it is very different, you could compare the FM (Intensity) parameter with the Filter Cutoff parameter of an analog synthesizer.

Carrier parameters

- *Harmonic knob:* Rotate to set the tuning ratio between the modulator (left) and carrier (right) oscillators. See Set the EFM1 tuning ratio on page 112.
- *Fine (tune) knob:* Rotate to adjust the tuning between two adjacent harmonics, as determined by the Harmonic knobs of both oscillators. The range of this control is ±0.5 harmonic. In the center (0) position, Fine tune does not have an effect. Click the "0" to center the Fine tune knob. Depending on the amount of detuning, you will hear one of the following:
  - A subtle "beating" of the timbre if lower detuning amounts are used
  - New harmonic and inharmonic overtones if high detuning amounts are used
- *Fixed Carrier button:* Turn on to disconnect the carrier frequency from keyboard, pitch bend, and LFO modulations, thus allowing you to produce a carrier tone that is free of these modulation sources.

Set the EFM1 tuning ratio

The carrier frequency is determined by the played key, and the modulator frequency is typically a multiple of the carrier frequency.

You can tune the modulator and carrier to any of the first 32 harmonics. The tuning relationship, or ratio, between the two significantly changes the base sound of the EFM1, and is best set by ear.

You use the Harmonic knobs to set the tuning ratio between the modulator (left) and carrier (right) oscillators.

In general, even tuning ratios between the carrier and modulator tend to sound more harmonic or musical, whereas odd ratios produce more inharmonic overtones—which are great for bell and metallic sounds.

In this respect, you can view the tuning ratio as being somewhat like the waveform selector of an analog synthesizer.

Note: The Harmonic and Fine tune knobs only affect the tuning relationship between the carrier and modulator oscillators. These should not be confused with the global Tune and Fine Tune parameters, which determine the overall tuning of the EFM1 (see EFM1 global parameters on page 114).

Experiment with basic tuning ratios

Do one of the following:

- Set the modulator and carrier to the first harmonic—a 1:1 ratio.

  A sawtooth-like sound is produced.

- Set the modulator to the second harmonic and the carrier to the first harmonic—a 2:1 ratio.

  A tone that sounds similar to a square wave is produced.

Choose a different EFM1 modulator waveform

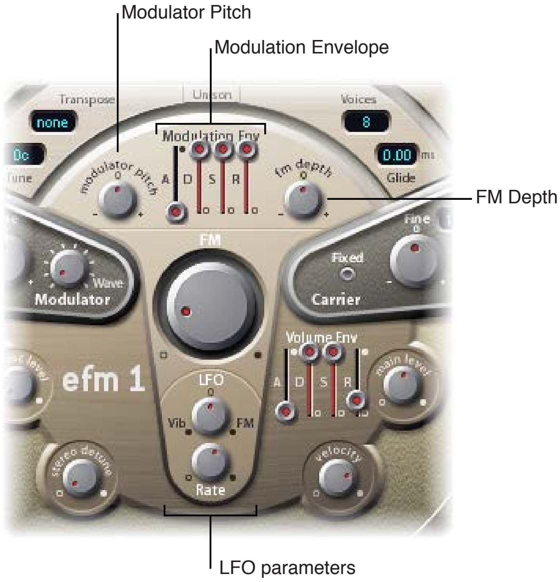
In classic FM synthesis, sine waves are used as modulator and carrier waveforms. The EFM1 modulator oscillator provides a number of additional digital waveforms, which extend its sonic capabilities significantly. These waveforms contain additional harmonics that add a new level of richness to the resulting FM sounds.

Choose a different waveform

- Turn the Wave parameter knob.
  - If you turn the knob to the full-left position, the modulator produces a sine wave.
  - If you turn the knob clockwise, you step—or fade—through a series of complex digital waveforms.

EFM1 modulation parameters

FM synthesis is, at its core, caused by the intensity and type of modulations that take place in the signal path. Therefore, the modulators outlined in this section have a different impact and role to play than equivalent envelopes and LFOs found in analog synthesizer designs.



Modulation parameters

- *Modulation Env(elope) sliders:* Control both the FM (Intensity) and Modulator pitch parameters over time. The envelope is triggered every time a MIDI note is received.
  - *A(ttack) slider:* Sets the time it takes to reach the maximum envelope level.
  - *D(ecay) slider:* Sets the time it takes to reach the Sustain level.
  - *S(ustain) slider:* Sets a level that is held until the MIDI note is released.
  - *R(elease) slider:* Sets the time it takes to reach a level of 0, after the MIDI note has been released.
- *Modulator Pitch knob:* Rotate to determine the impact of the modulation envelope on the pitch of the modulator oscillator.
  - If you turn the knob clockwise, you increase the effect of the modulation envelope. If you turn the knob counterclockwise, you invert the effect of the modulation envelope, as follows: the envelope slopes down during the attack phase and slopes up during the decay and release time phases.
  - If you click the "0" to center the Modulator Pitch knob, the envelope has no effect on the pitch of the modulator oscillator.
- *FM Depth knob:* Rotate to determine the impact of the modulation envelope on FM intensity.
  - If you turn the knob clockwise, you increase the effect of the modulation envelope. If you turn the knob counterclockwise, you invert the effect of the modulation envelope, as follows: the envelope slopes down during the attack phase and slopes up during the decay and release time phases.
  - If you click the "0" to center the FM Depth knob, the envelope has no effect on FM intensity.

- *LFO (low frequency oscillator) knob:* Rotate to set the amount of modulation applied to FM intensity or pitch.
  - If you turn the LFO knob clockwise, you increase the effect of the LFO on FM Intensity. If you turn the knob counterclockwise, you introduce a vibrato.
  - If you click the "0" to center the LFO knob, the LFO has no effect.
- *Rate knob:* Rotate to set the speed of the LFO.

EFM1 global parameters

The global parameters are used to set the tuning, number of voices, and other aspects of the EFM1's overall sound.

Global parameters

- *Transpose pop-up menu:* Choose the base pitch. You can transpose EFM1 by semitones or octaves with this control.
- *Tune field:* Drag to fine-tune the pitch of EFM1 by cents. One cent is 1/100th of a semitone.
- *Voices pop-up menu:* Choose the number of simultaneously playable voices—polyphony. You can choose mono (one voice), legato (one voice), or any number from 2 to 16 voices.
  - In Mono mode, staccato playing retriggers the envelope generators every time a new note is played. If you play in a legato style (play a new key while holding another), the envelope generators are triggered only for the first note you play legato, then they continue their curve until you release the last legato played key.
  - Legato mode is also monophonic, but with one difference: the envelope generators are retriggered only if you play staccato—releasing each key before playing a new key. If you play in a legato style, envelopes are not retriggered.

  Note: On several monophonic synthesizers, the behavior in Legato mode is referred to as *single trigger*, while Mono mode is referred to as *multi trigger*.
- *Unison button:* Turn on to layer two complete EFM1 voices, thus making the sound richer. EFM1 can be played with up to eight-voice polyphony when in unison mode.
- *Glide field:* Drag to introduce a continuous pitch bend between two consecutively played notes. Adjust the Glide value, which is in ms, to determine the time it takes for the pitch to travel from the last played note to the next.

  Note: Glide can be used in both of the monophonic modes—Mono and Legato—or in any of the polyphonic settings—where Voices is set from 2 to 16.

EFM1 output parameters

EFM1 provides the following level controls.

Output parameters

- *Sub Osc Level knob:* Rotate to introduce a sub-oscillator signal that enhances bass response. EFM1 features a sine wave sub-oscillator. This operates one octave below the FM engine, as determined by the Transpose parameter. Turning up the Sub Osc Level control mixes the sub-oscillator sine wave with EFM1's FM engine output.
- *Stereo Detune knob:* Rotate to add a chorus-like effect to the sound. This is achieved by doubling the EFM1 voice with a secondary, detuned FM engine. High values result in a wide stereo effect being added to the detuning, thus increasing the perceived space and width of your sound.

  Note: It is possible that mono compatibility could be lost with use of this parameter.
- *Volume Env(elope):* Shapes the level of the sound over time. The volume envelope is triggered every time a MIDI note is received.
  - *A(ttack) slider:* Move to set the time it takes to reach the maximum volume level.
  - *D(ecay) slider:* Move to set the time it takes to reach the Sustain level.
  - *S(ustain) slider:* Move to set a level that is held until the MIDI note is released.
  - *R(elease) slider:* Move to set the time it takes to reach a level of 0, after the MIDI note has been released.
- *Main Level knob:* Rotate to set the overall output level of EFM1.
- *Velocity knob:* Rotate to determine the sensitivity of EFM1 to incoming MIDI velocity messages. EFM1 dynamically reacts to MIDI velocity messages—harder playing will result in a brighter and louder sound. Set the Velocity control all the way to the left—counterclockwise—if you don't want EFM1 to respond to velocity.

Create random EFM1 sounds

The Randomize feature in the lower-right corner of the interface generates new sounds. It does this by randomly altering a number of key EFM1 parameter values.

This feature is ideal for creating subtle variations of a particular sound or for creating totally new sounds. It is useful when getting started with FM synthesis.

Randomize parameters

- *Randomize button:* Creates a new sound by randomizing multiple parameters.
- *Randomize field:* Determines the amount of randomization—variance from the original sound.

Create a random sound

- Click the Randomize button.

You can click multiple times. Save your settings as you go if you generate a sound you want to keep.

Limit the amount of randomization

- Drag in the numeric field to set the amount of randomization—variance from the original sound.

If you want to only randomly "tweak" the current sound, use values below 10%. Use higher values to radically change the sound with each click.

EFM1 extended parameters

Click the disclosure triangle at the lower left to open the EFM1 extended parameters. See EFM1 MIDI controller assignments for information about the controller assignment extended parameters.

Extended parameters

- *MIDI Mono Mode pop-up menu:* Choose Off, On (with common base channel 1), or On (with common base channel 16).

  In either mode, each voice receives on a different MIDI channel. Controllers and MIDI messages sent on the base channel affect all voices.
- *Mono Mode Pitch Range pop-up menu:* Choose 0, 24, or 48.

  The chosen pitch bend range affects individual note pitch bend messages received on all but the common base channel. The default is 48 semitones, which is compatible with Mobile GarageBand's keyboard in pitch mode. When using a MIDI guitar, 24 semitones is the preferable setting because most guitar to MIDI converters use this range by default.

EFM1 MIDI controller assignments

The EFM1 Extended Parameters area allows you remotely control EFM1 with your MIDI controller keyboard or other MIDI device. You can assign any unused MIDI controller to the following parameters:

- FM Amount
- Vibrato

Note: EFM1 also responds to MIDI pitch bend data. Pitch bend is hard-wired to the overall pitch of EFM1.

Assign a MIDI controller

1. Choose the controller name or number from the Ctrl FM or Ctrl Vibrato pop-up menu.
2. Set the FM or vibrato amount using the slider below the pop-up menu.

5 ES E

ES E overview

The eight-voice ES E (ES Ensemble) synthesizer is ideal for quickly creating warm, rich pad and ensemble sounds.

ES E produces sounds using subtractive synthesis. It features an oscillator that generates harmonically rich waveforms. You *subtract*—cut, or filter out—portions of these waveforms and reshape them to create new sounds.

If you're new to synthesizers, see Synthesizer basics overview on page 471, which will introduce you to the terminology and give you an overview of different synthesis methods and how they work.

ES E is divided into several areas.

- *Oscillator parameters:* The oscillator Wave and Octave parameters are shown in the area to the left. The oscillator generates the waveforms that form the basis of your sound. See ES E oscillator parameters on page 119.
- *LFO parameters:* The LFO parameters (below the Wave knob) are used to modulate the sound. See ES E LFO parameters on page 120.
- *Filter parameters:* The section to the right of the oscillator parameters includes the Cutoff (frequency) and Resonance knobs. You use the filter to contour the waveforms sent from the oscillator. See ES E filter parameters on page 121.
- *Envelope parameters:* The area to the right of the filter parameters contains the envelope parameters, which control the level of the sound over time. See ES E envelope parameters on page 122.

- *Output parameters:* The area at the extreme right houses the switches for the integrated modulation effects and the Volume knob, which is responsible for the main output level. The effects can be used to color or thicken the sound. See ES E output parameters on page 122.
- *Extended parameters:* Not shown in the image, the extended parameters are accessed by clicking the triangle at the lower left of the interface. These parameters include bend and tuning functions. See Extended ES E parameters on page 123.

ES E oscillator parameters

The synthesizer oscillator generates a waveform, which is then sent to other portions of the synthesizer engine for processing or manipulation.

Oscillator parameters

- *Wave knob:* Rotate to select the waveform of the oscillator, which is responsible for the basic color of the tone. The leftmost setting of the Wave parameter causes the oscillators to output sawtooth signals. Across the remaining range, the oscillators output pulse waves, with the average pulse width determined by the Wave parameter position.
- *4, 8, and 16 buttons:* Switch the pitch in octaves—transpose it up or down. The lowest setting is 16 feet, and the highest is 4 feet. The use of the term *feet* to determine octaves comes from the measurements of organ pipe lengths. The longer and wider the pipe, the deeper the tone.

ES E LFO parameters

The LFO (low frequency oscillator) generates a cyclic waveform that is used to modulate the ES E waveform. The behavior and effect of the LFO depend on whether a sawtooth or pulse wave is selected.

- If Wave is set to sawtooth, the LFO modulates the frequency of the waveform, resulting in a vibrato or siren effect—depending on the LFO speed and intensity.
- If Wave is set to a pulse wave, the LFO modulates the waveform's pulse width—pulse width modulation (PWM).

LFO parameters

- *Vib(rato)/PWM knob:* Rotate to define the intensity of LFO modulation.
- *Speed knob:* Rotate to set the frequency of LFO modulation.

  Note: When the pulse width becomes very narrow, the signal sounds as if it is being interrupted—"breaking up." Given this potential artifact, set the PWM intensity with care, and select the Wave parameter's 12 o'clock position (50% rectangular) for the pulse width, if you want to achieve the maximum modulation range.

ES E filter parameters

ES E includes a lowpass filter that lets you contour the output from the oscillator.

Filter parameters

- *Cutoff knob:* Rotate to control the cutoff frequency of the filter.
- *Resonance knob:* Rotate to boost or cut portions of the signal that surround the frequency defined by the Cutoff parameter.

  Note: Increasing the Resonance value results in a rejection of bass—low frequency energy—when using lowpass filters.
- *AR Int knob:* Rotate to set the amount (depth) of cutoff frequency modulation applied by the envelope generator.

  Note: ES E provides one envelope generator per voice, offering Attack and Release (AR) parameters (see ES E envelope parameters on page 122).
- *Velo Filter knob:* Rotate to set the velocity sensitivity of the cutoff frequency modulation applied by the envelope generator.

  Note: This parameter has no effect if AR Int is set to 0.

ES E envelope parameters

The AR (Attack and Release) envelope affects both the filter cutoff (AR Int) and the level of the sound over time.

Envelope parameters

- *Attack slider:* Move to set the time it takes for the signal to reach the initial signal level, known as the sustain level.
- *Release slider:* Move to set the time it takes for the signal to fall from the sustain level to a level of 0.

ES E output parameters

The ES E output stage consists of the Volume section and the Chorus/Ensemble buttons.

Output Parameters

- *Volume knob:* Rotate to set the overall output level.
- *Velo Volume knob:* Rotate to set the amount (depth) of velocity sensitivity to incoming MIDI note events. When set to higher values, each note is louder, if struck more firmly. At lower values, the dynamic response is reduced, so that there is little difference when you play a note pianissimo (soft) or forte (loud/hard).
- *Chorus I, Chorus II, and Ensemble buttons:* Turn these effect variations on or off. If no button is active, the effects processor is turned off.
  - Chorus I is a typical chorus effect.
  - Chorus II is characterized by a stronger modulation.
  - Ensemble has a fuller and richer sound, due to a more complex modulation routing.

Extended ES E parameters

ES E offers three additional parameters that are accessed by clicking the disclosure triangle at the lower left of the interface.

Extended parameters

- *Pos. Bender Range:* Move to set the positive—upward—pitch bend range in semitone steps. This allows you to use the pitch bend controller of your keyboard to bend the ES E pitch.
- *Neg. Bender Range:* Move to set the negative—downward—pitch bend range in semitone steps, by up to 2 octaves—a value of 24. The default Neg. Bender Range value is Pos PB (positive pitch bend). In essence, this means that only positive pitch bend is available.
- *Tune:* Move to tune the instrument sound in cents. A cent is 1/100th of a semitone.

ES M

6

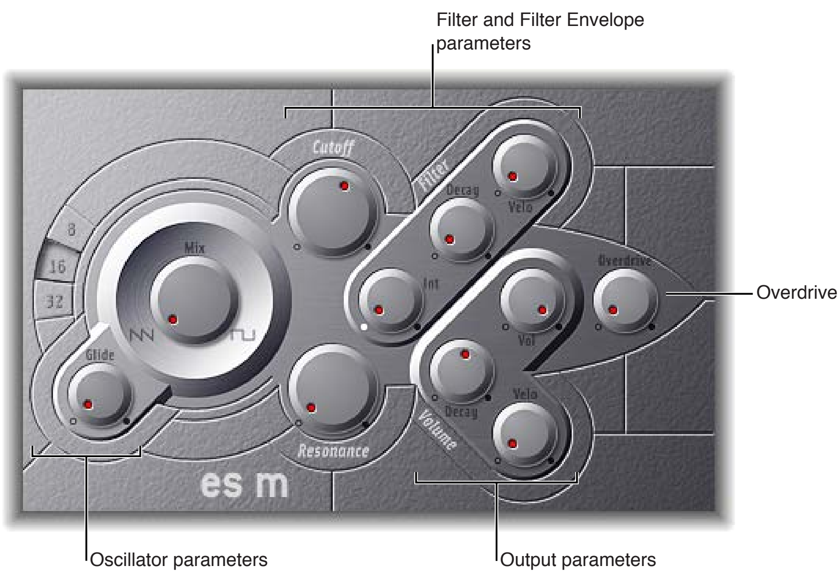
ES M overview

The monophonic ES M (ES Mono) synthesizer is a good starting point if you're looking for bass sounds that punch through your mix.

ES M features an automatic fingered portamento mode, making bass slides easy. It also provides an automatic filter compensation circuit that delivers rich, creamy basses, even when you use higher resonance values.

ES M produces sounds using subtractive synthesis. It has an oscillator that generates harmonically rich waveforms. You *subtract*—cut, or filter out—portions of these waveforms and reshape them to create new sounds.

If you're new to synthesizers, see Synthesizer basics overview on page 471, which will introduce you to the terminology and give you an overview of different synthesis methods and how they work.



ES M is divided into several areas.

- *Oscillator parameters:* The oscillator Mix and Octave parameters are shown in the area to the left. The oscillator generates the basic waveforms that form the basis of your sound. See ES M oscillator parameters on page 125.
- *Filter and filter envelope parameters:* The section to the right of the Oscillator parameters includes the Cutoff (frequency) and Resonance knobs. The filter is used to contour the waveforms sent from the oscillators. The filter envelope parameters are found toward the upper right. These control the filter cutoff over time. See ES M filter and filter envelope on page 126.

- *Output parameters:* The angle-shaped area to the lower right contains the level envelope and output parameters, which control the level of the sound over time. The Overdrive knob is located near the right edge of the interface, halfway up. The Overdrive can be used to color or add bite to the sound. See ES M level envelope and output controls on page 127.
- *Extended parameters:* Not shown in the image, the extended parameters are accessed by clicking the triangle at the lower left of the interface. These parameters include bend and tuning functions. See Extended ES M parameters on page 127.

ES M oscillator parameters

The synthesizer oscillator is used to generate a waveform, which is then sent to other portions of the synthesizer engine for processing or manipulation.

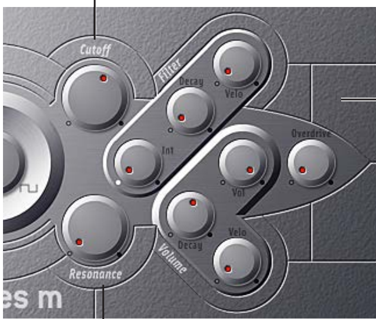
Oscillator parameters

- *Mix knob:* Rotate to set the waveform of the oscillator, which is responsible for the basic color of the tone.
  - Setting the Wave parameter all the way to the left causes the oscillator to output sawtooth signals.
  - Setting the Wave parameter all the way to the right outputs a 50% rectangular wave, which is heard one octave below the sawtooth.
  - For any Wave setting between these extreme positions, the oscillator outputs a crossfaded mix of the two waveforms.
- *8, 16, and 32 buttons:* Switch the pitch in octaves—transpose it up or down. The lowest setting is 32 feet, and the highest is 8 feet. The use of the term *feet* to determine octaves comes from the measurements of organ pipe lengths. The longer and wider the pipe, the deeper the tone.
- *Glide knob:* Rotate to introduce a continuous pitch bend between two consecutively played notes. Adjust the Glide value, in ms, to determine the time it takes for the pitch to travel from the last played note to the next. At a value of 0, no glide effect occurs.

  Note: The ES M always works in a fingered portamento mode, with notes played in a legato style resulting in a glide—portamento—from pitch to pitch.

ES M filter and filter envelope

The ES M includes a lowpass filter that lets you contour the output from the oscillator. The filter has a dedicated envelope.



Filter and filter envelope parameters

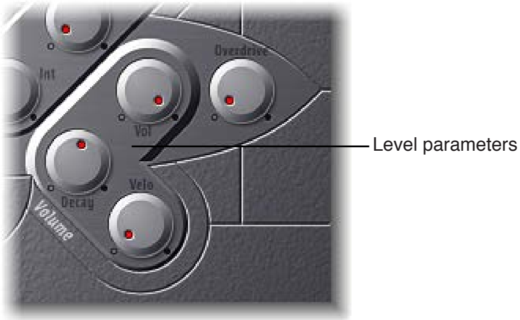
- *Cutoff knob:* Rotate to set the cutoff frequency of the ES M filter. Its slope is 24 dB/octave.
- *Resonance knob:* Rotate to boost or cut portions of the signal that surround the frequency defined by the Cutoff parameter.

  Note: Increasing the Resonance value results in a rejection of bass—low frequency energy—when using lowpass filters. The ES M compensates for this side-effect internally, resulting in a more bassy sound.
- *Int knob:* Rotate to define the amount—the intensity or depth—of cutoff frequency modulation applied by the envelope generator.
- *Decay knob:* Rotate to set the decay time of the filter envelope.
- *Velo knob:* Rotate to set the velocity sensitivity of the cutoff frequency modulation applied by the envelope generator.

  Note: The Decay and Velo parameters have no effect if Int is set to 0.

ES M level envelope and output controls

The output stage of the ES M offers the following parameters.



Envelope and output parameters

- *Decay knob:* Rotate to set the decay time of the dynamic stage. The attack, release, and sustain times of the synthesizer are internally set to 0.
- *Velo knob:* Rotate to determine the velocity sensitivity of the dynamic stage.
- *Vol knob:* Rotate to set the ES M master output level.
- *Overdrive knob:* Rotate to set the level of the integrated overdrive effect.

  Important: To avoid hurting your ears or damaging your speakers, consider turning down the Volume level before setting Overdrive to a high value; then turn it up gradually.

Extended ES M parameters

ES M offers three additional parameters that you access by clicking the disclosure triangle at the lower left of the interface.

Extended parameters

- *Pos. Bender Range:* Move to set the positive—upward—pitch bend range in semitone steps. This allows you to use the pitch bend controller of your keyboard to bend the ES M pitch.
- *Neg. Bender Range:* The default Neg. Bender Range value is Pos PB (positive pitch bend). In essence, this means that only positive pitch bend is available. Move to set the negative—downward—pitch bend range in semitone steps, by up to 2 octaves (a value of 24).
- *Tune:* Move to tune the instrument in cents. One cent is 1/100th of a semitone.

ES P

7

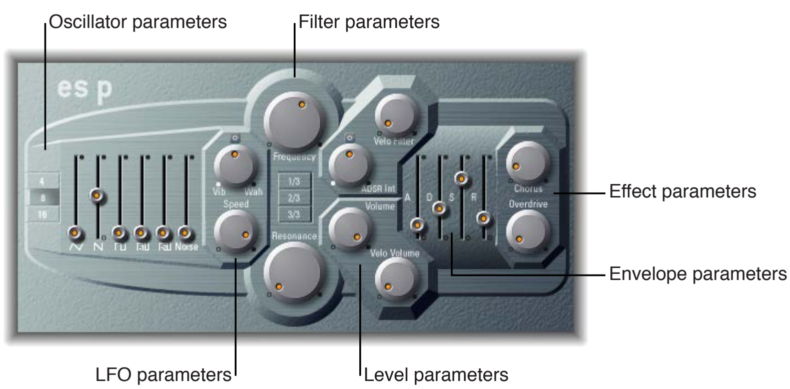
ES P overview

The eight-voice ES P (ES Poly) emulates classic polyphonic synthesizers of the 1980s.

It is a versatile instrument that is capable of producing a huge variety of useful musical sounds. The creation of classic analog synthesizer brass sounds is just one of its many strengths.

ES P produces sounds using subtractive synthesis. It features an oscillator that generates harmonically rich waveforms. You *subtract*—cut, or filter out—portions of these waveforms and reshape them to create new sounds.

If you're new to synthesizers, see Synthesizer basics overview on page 471, which will introduce you to the terminology and give you an overview of different synthesis methods and how they work.



ES P is divided into several areas.

- *Oscillator parameters:* The oscillator sliders are shown in the area to the left. The octave parameters are also found in this section. The oscillators generate the basic waveforms that form the basis of your sound. See ES P oscillator parameters on page 129.
- *LFO parameters:* The LFO parameters (to the right of the oscillator parameters) are used to modulate the sound. See ES P LFO parameters on page 130.
- *Filter parameters:* The vertical column in the center includes the (cutoff) Frequency and Resonance knobs and the key follow buttons. The filter is used to contour the waveforms sent from the oscillators. See ES P LFO parameters on page 130.
- *Envelope and level parameters:* The area to the right of the filter parameters contains the envelope and level parameters, which control the level of the sound over time. See ES P envelope and level controls on page 132.
- *Effect parameters:* The area at the extreme right contains the Chorus and Overdrive parameters. These can be used to color or thicken the sound. See Integrated ES P effects processor on page 133.

- *Extended parameters:* Not shown in the image, the extended parameters are accessed by clicking the disclosure triangle at the lower left of the interface. These parameters include bend and tuning functions. See Extended ES P parameters on page 133.

ES P oscillator parameters

ES P features several oscillators that output different waveforms. These signals can be mixed together—at different levels—providing countless variations of the raw material used for your sounds.

In addition to triangular, sawtooth, and rectangular waves, the rectangular waves of two sub-oscillators are also available. The left sub-oscillator fader is one octave lower than the main oscillators, and the right sub-oscillator fader is two octaves lower. Use these to fatten up the sound.

Oscillator parameters

- *Triangle oscillator slider:* Drag to set the level of the triangle waveform output by the oscillators.
- *Sawtooth oscillator slider:* Drag to set the level of the sawtooth waveform output by the oscillators.
- *Rectangle oscillator slider:* Drag to set the level of the rectangle waveform output by the oscillators. The pulse width is fixed at 50%.
- *Sub-oscillator -1 octave slider:* Drag to set the level of the (rectangular) sub-oscillator waveform, which is one octave lower than the main oscillators. Use this to fatten up the sound. The pulse width is fixed at 50%.
- *Sub-oscillator - 2 octaves slider:* Drag to set the level of the (rectangular) sub-oscillator waveform, which is two octaves lower than the main oscillators. Use this to fatten up the sound. The pulse width is fixed at 50%.
- *Noise generator slider:* Drag to set the level of white noise added to the mix. This is the raw material for classic synthesizer sound effects, such as ocean waves, wind, and helicopters.
- *4, 8, and 16 buttons:* Switch the pitch in octaves—transpose it up or down. The lowest setting is 16 feet, and the highest is 4 feet. The use of the term *feet* to determine octaves comes from the measurements of organ pipe lengths. The longer and wider the pipe, the deeper the tone.

ES P LFO parameters

ES P features an LFO (low frequency oscillator), which can do either of the following:

- Modulate the frequency of the oscillators, resulting in a vibrato
- Modulate the cutoff frequency of the dynamic lowpass filter, resulting in a wah wah effect

LFO parameters

- *Vib/Wah knob:* Turn to the left to set a vibrato; turn to the right to cyclically modulate the filter.
- *Speed knob:* Rotate to set the rate of the vibrato or cutoff frequency modulation.

ES P filter parameters

ES P includes a lowpass filter that lets you contour the output signals from the oscillator.

Filter parameters

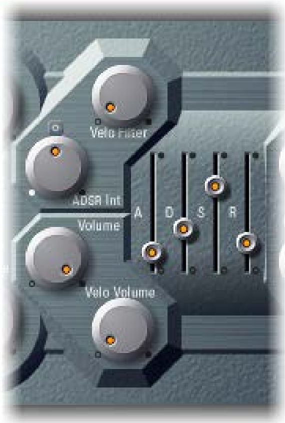
- *Frequency knob:* Rotate to set the cutoff frequency of the ES P's lowpass filter.
- *Resonance knob:* Rotate to boost or cut portions of the signal that surround the frequency defined by the Frequency knob.

  Note: Increasing the Resonance value results in a rejection of bass—low frequency energy—when using lowpass filters. The ES P compensates for this side effect internally, resulting in a more bassy sound.

- *1/3, 2/3, and 3/3 (key follow) buttons:* The cutoff frequency can be modulated by MIDI note number (keyboard position); you may know this parameter as *keyboard follow* on other synthesizers. Click a button to choose 1/3, 2/3, or full-keyboard follow (3/3). If no button is active, the key you strike won't affect the cutoff frequency. This makes the lower notes sound relatively brighter than the higher ones. If you choose 3/3, the filter follows the pitch, resulting in a constant relationship between cutoff frequency and pitch. This is typical of many acoustic instruments where higher notes sound both brighter in tone and higher in pitch.
- *ADSR Int knob:* Rotate to define the amount (depth) of cutoff frequency modulation applied by the envelope generator (see ES P envelope and level controls on page 132).
- *Velo Filter knob:* Rotate to set the velocity sensitivity of the cutoff frequency modulation applied by the envelope generator. The main envelope generator (ADSR) modulates the cutoff frequency over the duration of a note. The intensity of this modulation can respond to velocity information. If you play pianissimo (velocity = 1), the modulation is minimal. If you strike with the hardest fortissimo (velocity = 127), the modulation is more intense.

ES P envelope and level controls

ES P features an ADSR envelope that affects both the filter cutoff (ADSR Int) and the level of the sound over time. This section also covers the master level control parameters.



Envelope and level parameters

- *Attack slider:* Move to set the time it takes for the signal to reach the initial, desired signal level (the sustain level).
- *Decay slider:* Move to set the time it takes for the signal to fall from the attack level to the sustain level.
- *Sustain slider:* Move to set the signal level (the sustain level).
- *Release slider:* Move to set the time it takes for the signal to fall from the sustain level to a level of zero.
- *Volume knob:* Rotate to set the overall ES P output level.
- *Velo Volume knob:* Rotate to set the amount (depth) of velocity sensitivity to incoming MIDI note events. When set to higher values, each note is louder if struck harder. At lower values, the dynamic response is reduced, so that there is little difference when you play a note pianissimo (soft) or forte (loud/hard).
- *VCA Mode buttons (Controls view):* Click ADSR to control the amplifier with the ADSR envelope generator. Click Gate to output a constant organ-like tone when a key is played.

Integrated ES P effects processor

ES P offers integrated stereo chorus and overdrive effects. These are based on similar effects processors found in the affordable Japanese synthesizers of the 1980s—which the ES P emulates.

ES P effect parameters

- *Chorus knob:* Rotate to set the intensity (depth) of the integrated chorus effect.
- *Overdrive knob:* Rotate to set the overdrive/distortion level of the ES P output.

  Important: To avoid hurting your ears or damaging your speakers, consider turning down the Volume level before setting Overdrive to a high value; then turn it up gradually.

Extended ES P parameters

ES P offers three additional parameters that are accessed by clicking the disclosure triangle at the lower left of the interface.

- *Pos. Bender Range:* Move to set the positive—upward—pitch bend range in semitone steps. This allows you to use the pitch bend controller of your keyboard to bend the ES P pitch.
- *Neg. Bender Range:* The default Neg. Bender Range value is Pos PB (positive pitch bend). In essence, this means that only positive pitch bend is available. Move to set the negative—downward—pitch bend range in semitone steps, by up to 2 octaves (a value of 24).
- *Tune field:* Move to tune the instrument in cents. One cent is 1/100th of a semitone.

EVOC 20 PolySynth

8

EVOC 20 PolySynth and vocoding

EVOC 20 PolySynth overview

EVOC 20 PolySynth combines a vocoder with a polyphonic synthesizer and can be played in real time.

It can create classic vocoder sounds, made famous by groups such as Kraftwerk during the 1970s and 1980s. Vocoding remains popular in current electronic, hip-hop, R & B, and other music styles.

EVOC 20 PolySynth "listens" to an incoming audio signal—typically of a spoken or sung performance—and imposes the sonic characteristics and level changes of this signal onto the integrated synthesizer.

When you play notes and chords with your MIDI keyboard, the internal synthesizer "sings" at the pitches of incoming MIDI notes, but with the articulations—level changes, vowel and consonant sounds—of the incoming audio signal. This results in the classic "singing robot" or "synthetic voice" sounds that vocoders are mainly known for.

EVOC 20 PolySynth can also be used as a synthesizer, or it can be used for more subtle effects processing—such as the creation of relatively natural-sounding vocal harmonies from a solo voice performance. Not limited to vocal processing, you can also achieve interesting results by processing other audio material, such as drum or instrument loops.

To use EVOC 20 PolySynth, you need to insert it into the Instrument slot of an instrument channel strip. You also need to provide an audio signal as the analysis audio source, via a side chain.

Set up EVOC 20 PolySynth in your host application

1. Insert EVOC 20 PolySynth into the Instrument slot of an instrument channel strip.
2. Choose an input source from the Side Chain pop-up menu in the plug-in header. This can be an audio track, live input, or bus, depending on the host application.

   The EVOC 20 PolySynth is now ready to accept incoming MIDI data and has been assigned to an input, audio track, or bus—via a side chain.
3. If applicable to your host application and needs, mute the audio track serving as the side-chain input, start playback, and play your MIDI keyboard.
4. Adjust the volume levels of EVOC 20 PolySynth and the side-chain source—if not muted—to meet your needs.
5. To further enhance the sound, adjust the knobs, sliders, and other controls, and insert other effect plug-ins.

Vocoder basics

The word *vocoder* is an abbreviation for *voice encoder*. A vocoder analyzes and transfers the sonic character of the audio signal arriving at its analysis input to the synthesizer's sound generators. The result of this process is heard at the output of the vocoder.

The classic vocoder sound uses speech as the analysis signal and a synthesizer sound as the synthesis signal. This sound was popularized in the late 1970s and early 1980s. You may be familiar with tracks such as "O Superman" by Laurie Anderson, "Funkytown" by Lipps Inc., and numerous Kraftwerk pieces—such as "Autobahn," "Europe Endless," "The Robots," and "Computer World."

In addition to these "singing robot" sounds, vocoding has also been used in many films—such as with the Cylons in *Battlestar Galactica*, and most famously, with the voice of Darth Vader from the *Star Wars* saga. See Vocoder history on page 152.

Vocoding, as a process, is not strictly limited to vocal performances. You could use a drum loop as the analysis signal to shape a string ensemble sound arriving at the synthesis input.

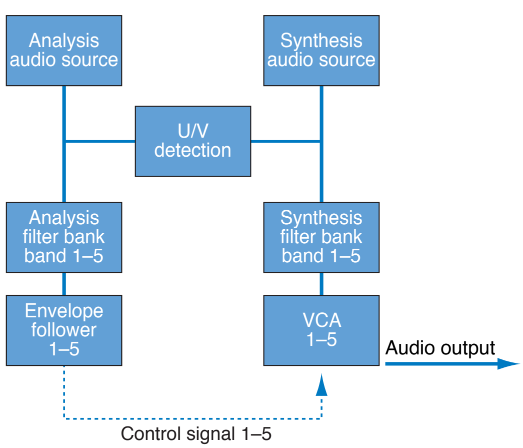
The speech *analyzer* and *synthesizer* features of a vocoder are two bandpass *filter banks*. Bandpass filters allow a frequency band—a slice in the overall frequency spectrum—to pass through unchanged. Frequencies that fall outside the band are cut.

In the EVOC 20 plug-ins, these filter banks are named the *analysis* and *synthesis* banks. Each filter bank has a matching number of corresponding bands—if the analysis filter bank has five bands (1, 2, 3, 4, and 5), there will be a corresponding set of five bands in the synthesis filter bank. Band 1 in the analysis bank is matched to band 1 in the synthesis bank, band 2 to band 2, and so on.

The audio signal arriving at the analysis input passes through the analysis filter bank, where it is divided into bands.

An envelope follower is coupled to each filter band. The envelope follower of each band tracks, or *follows*, volume changes in the audio source—or, more specifically, the portion of the audio that has been allowed to pass by the associated bandpass filter. In this way, the envelope follower of each band generates dynamic control signals.

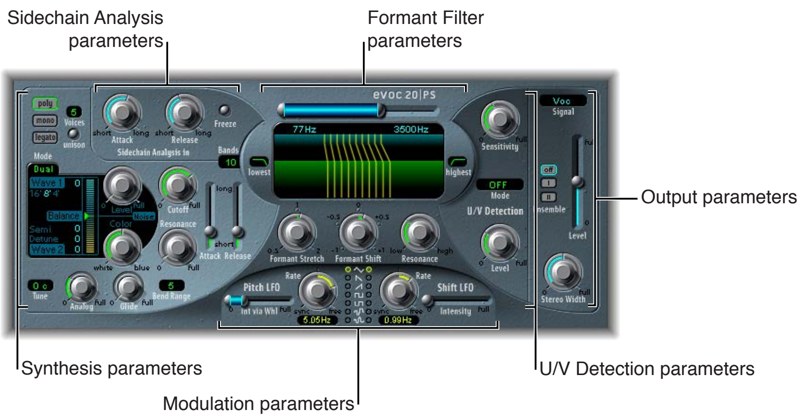
These control signals are then sent to the synthesis filter bank—where they control the levels of the corresponding synthesis filter bands. This is done with voltage-controlled amplifiers (VCAs) in analog vocoders. Volume changes to the bands in the analysis filter bank are imposed on the matching bands in the synthesis filter bank. These filter level changes are heard as a synthetic reproduction of the original input signal—or a mix of the two filter bank signals.



The more bands a vocoder offers, the more precisely the original sound's character will be reproduced by the synthesis filter bank. EVOC 20 PolySynth provides up to 20 bands per bank. See EVOC 20 block diagram on page 153 for a detailed image of the EVOC 20 PolySynth signal path.

EVOC 20 PolySynth interface

The EVOC 20 PolySynth interface is divided into six main parameter sections.



- *Sidechain Analysis parameters:* Determine how the input signal is analyzed and used by the EVOC 20 PolySynth. See EVOC 20 PolySynth analysis parameters on page 137.
- *U/V Detection parameters:* Detect the unvoiced portions of the sound in the analysis signal, improving speech intelligibility. See EVOC 20 PolySynth (U/V) detection parameters on page 138.
- *Synthesis parameters:* Control the polyphonic synthesizer of the EVOC 20 PolySynth. See EVOC 20 PolySynth synthesis parameters overview on page 140.
- *Formant Filter parameters:* Configure the analysis and synthesis filter banks. See EVOC 20 PolySynth formant filter on page 146.
- *Modulation parameters:* Modulate the synthesizer and filter banks—through two LFOs. See EVOC 20 PolySynth modulation parameters on page 148.
- *Output parameters:* Configure the output signal of the EVOC 20 PolySynth. See EVOC 20 PolySynth output parameters on page 149.

EVOC 20 PolySynth analysis parameters

The parameters in the Sidechain Analysis section control how EVOC 20 PolySynth analyzes and uses the input signal. Be precise with these parameters to attain the best possible speech intelligibility and the most accurate tracking.

Sidechain analysis parameters

- *Attack knob:* Rotate to determine how quickly each envelope follower—coupled to each analysis filter band—reacts to rising signal levels. Longer attack times result in a slower tracking response to transients—level spikes—of the analysis input signal. A long attack time on percussive input signals—a spoken word or hi-hat part, for example—will translate into a less articulated vocoder effect. Set the Attack parameter to the lowest possible value to enhance articulation.
- *Release knob:* Rotate to determine how quickly each envelope follower—coupled to each analysis filter band—reacts to falling signal levels. Longer release times cause the analysis input signal transients to sustain for a longer period at the vocoder's output. A long release time on percussive input signals—a spoken word or hi-hat part, for example—will translate into a less articulated vocoder effect. Use of extremely short release times results in rough, grainy vocoder sounds. Release values of around 8 to 10 milliseconds are useful starting points.
- *Freeze button:* Turn on to hold—or freeze—the current analysis sound spectrum indefinitely. When Freeze is enabled, the analysis filter bank ignores the input source, and the Attack and Release knobs have no effect.
- *Bands field:* Drag to set the number—up to 20—of frequency bands used by the filter banks.

Freeze the input signal

Freezing the input signal lets you capture a particular characteristic of the signal, which is then imposed as a complex sustained filter shape on the Synthesis section. Here are some examples of when this could be useful:

- If you are using a spoken word pattern as a source, the Freeze button could capture the attack or tail phase of an individual word within the pattern—the vowel *a*, for example.
- People cannot sustain sung notes indefinitely. To compensate for this human limitation, use the Freeze button. If the synthesis signal needs to be sustained but the analysis source signal—a vocal part—is not sustained, use the Freeze button to lock the current formant levels of a sung note, even during gaps in the vocal part, between words in a vocal phrase. The Freeze parameter can be automated, which may be useful in this situation.

■ Click the Freeze button to hold, or sustain, the sound spectrum of the analysis input signal.

Set the number of filter bank bands

- To set the number of frequency bands the EVOC 20 PolySynth's filter bank uses, drag the Bands field vertically.

The greater the number of frequency bands, the more precisely the sound can be reshaped. As the number of bands is reduced, the source signal's frequency range is divided up into fewer bands, and the resulting sound is formed with less precision by the synthesis engine. You may find that a good compromise between sonic precision—allowing incoming signals such as speech and vocals to remain intelligible—and resource usage is around 10 to 15 bands.

EVOC 20 PolySynth (U/V) detection parameters

Human speech consists of a series of voiced sounds—tonal sounds or formants—and unvoiced sounds. The main distinction between voiced and unvoiced sounds is that voiced sounds are produced by an oscillation of the vocal cords, whereas unvoiced sounds are produced by blocking and restricting the air flow with lips, tongue, palate, throat, and larynx.

If speech containing voiced and unvoiced sounds is used as a vocoder's analysis signal but the synthesis engine doesn't differentiate between voiced and unvoiced sounds, the result sounds rather weak. To avoid this problem, the synthesis section of the vocoder must produce different sounds for the *voiced* and *unvoiced* parts of the signal.

The EVOC 20 PolySynth includes an Unvoiced/Voiced detector for this specific purpose. This unit detects the unvoiced portions of the sound in the analysis signal and then substitutes the corresponding portions in the synthesis signal with noise, with a mixture of noise and synthesizer signal, or with the original signal. If the U/V detector detects voiced parts, it passes this information to the Synthesis section, which uses the normal synthesis signal for these portions.

A formant is a peak in the frequency spectrum of a sound. In the context of human voices, formants are the key component that enables humans to distinguish between different vowel sounds—based purely on the frequency of the sounds. Formants in human speech and singing are produced by the vocal tract, with most vowel sounds containing four or more formants.

U/V detection parameters

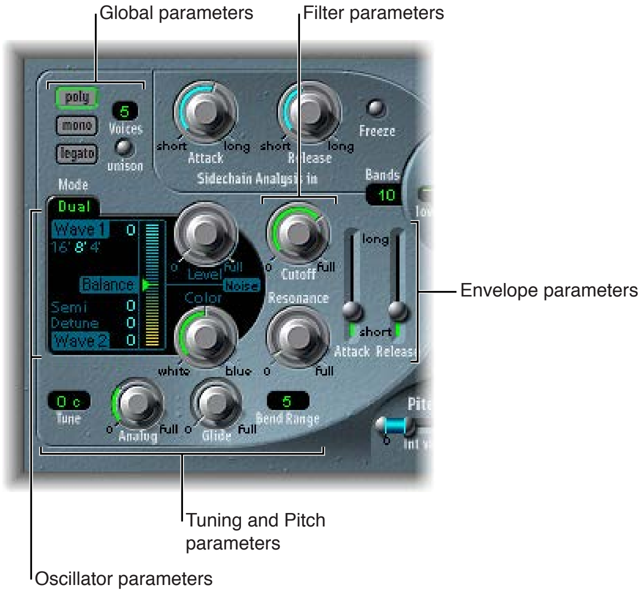
- *Sensitivity knob:* Rotate to determine how responsive U/V detection is. Turn to the right for higher settings, where more of the individual unvoiced portions of the input signal are recognized. When high settings are used, the increased sensitivity to unvoiced signals can lead to the U/V sound source being used on the majority of the input signal, including voiced signals. Sonically, this results in a sound that resembles a radio signal that is breaking up and contains a lot of static, or noise. (The U/V sound source is determined by the U/V Source menu, as described below.)
- *U/V Source pop-up menu:* Choose the sound sources that are used to replace the unvoiced content of the input signal.
  - *Noise:* Uses noise alone for the unvoiced portions of the sound.
  - *Noise + Synth:* Uses noise and the synthesizer for the unvoiced portions of the sound.
  - *Blend:* Uses the analysis signal after it has passed through a highpass filter for the unvoiced portions of the sound. The Sensitivity parameter has no effect when this setting is used.
- *U/V Level knob:* Rotate to set the volume of the signal used to replace the unvoiced content in the input signal.

  Important: Take care with the Level knob, particularly when a high Sensitivity value is used, to avoid internally overloading the EVOC 20 PolySynth.

EVOC 20 PolySynth synthesis parameters

EVOC 20 PolySynth synthesis parameters overview

EVOC 20 PolySynth is equipped with a polyphonic synthesizer that is capable of accepting MIDI note input. The parameters of the Synthesis section are described below.



Synthesis parameters

- *Oscillator parameters:* Determine the basic waveforms for the synthesis engine of the EVOC 20 PolySynth. See EVOC 20 PolySynth oscillators overview on page 141.
- *Tuning and Pitch parameters:* Control the overall tuning of the synthesizer, and aspects such as pitch bend and portamento. See EVOC 20 PolySynth tuning and pitch parameters on page 143.
- *Filter parameters:* Shape the basic waveforms of the oscillators. See EVOC 20 PolySynth filter parameters on page 144.
- *Envelope parameters:* Control the level of the attack and release phases of the synthesizer sound. See EVOC 20 PolySynth envelope parameters on page 144.
- *Global parameters:* Determine the keyboard mode and number of voices used by the EVOC 20 PolySynth. (The Global parameters are located at the top left of the interface.) See EVOC 20 PolySynth global parameters on page 145.

EVOC 20 PolySynth oscillator parameters

EVOC 20 PolySynth oscillators overview

EVOC 20 PolySynth has two oscillators, which you can switch between Dual mode and FM mode. The Synthesis section also incorporates a noise generator that can add a further color to your sound.

Click here to switch between Dual and FM mode.

- *Dual mode:* Each oscillator allows you to choose a digital waveform.
- *FM mode:* Oscillator 1 generates a sine wave. The frequency, or *pitch*, of oscillator 1 is modulated by oscillator 2. This leads to a number of different tones and harmonics becoming audible. Oscillator 2 can use any available digital waveform. See Frequency modulation (FM) synthesis on page 492).

  Each mode subtly changes the parameters shown in the oscillator section.

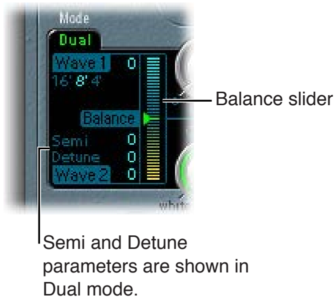
Both Dual and FM mode provide the following common parameters.

Common oscillator parameters

- *16', 8', 4' value buttons:* Click to set the octave range for oscillator 1. 16' (16 feet) is the lowest, and 4' the highest setting. The use of the term *feet* to determine octaves comes from the measurements of organ pipe lengths. The longer and wider the pipe, the deeper the tone.
- *Wave 1 and Wave 2 fields:* Choose the waveform type for oscillators 1 and 2. There are 50 single-cycle digital waveforms with different sonic characteristics.

EVOC 20 PolySynth oscillator dual mode

In dual mode, each oscillator can use any of 50 digital waveforms.

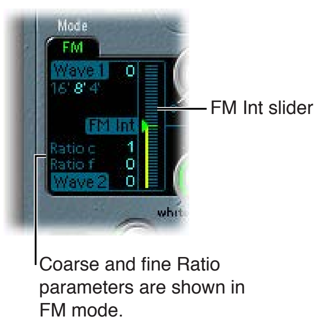


Dual mode oscillator parameters

- *Semi field:* Drag to adjust the tuning of oscillator 2 in semitone steps.
- *Detune field:* Drag to fine-tune both oscillators in cents. One hundred cents equals one semitone step.
- *Balance slider:* Drag to set the level balance between the two oscillator signals.

EVOC 20 PolySynth oscillator FM mode

In FM mode, oscillator 1 generates a sine wave. The Wave 1 parameter has no effect in this mode.



FM mode oscillator parameters

- *Ratio c(oarse) field:* Drag to adjust the frequency ratio between oscillator 2 and oscillator 1 in semitone steps.
- *Ratio f(ine) field:* Drag to adjust the frequency ratio between oscillator 2 and oscillator 1 in cents. One hundred cents equals one semitone step.
- *FM Int slider:* Drag to determine the intensity of modulation. Higher values result in a more complex waveform with more overtones.

EVOC 20 PolySynth noise generator

The noise generator provides a further sound source that can be used alongside the two oscillators.

Important: The noise generator in the oscillator section is independent of the noise generator in the U/V detection area. For further information about voiced and unvoiced signals, see EVOC 20 PolySynth (U/V) detection parameters on page 138.

Noise generator parameters

- *Level knob:* Rotate to control the amount of noise added to the signals of the two oscillators.
- *Color knob:* Rotate to set the timbre of the noise signal. Turn full-left to hear white noise. Turn full-right to hear blue noise (high-passed noise). White noise has traditionally been used to create wind and rain sound effects. It has the same energy in each frequency interval. Blue noise sounds brighter, because its bass portion is suppressed by a highpass filter.

  Tip: Set Color to the full-right position and Level to a very low value to achieve a lively and fresh synthesis signal.

EVOC 20 PolySynth tuning and pitch parameters

The parameters at the lower left of the interface are used to control the overall tuning and other pitch-related aspects of the EVOC 20 PolySynth sound.

Tuning and pitch parameters

- *Tune field:* Drag to set the overall tuning of the EVOC 20 PolySynth, in cents. One hundred cents equals one semitone step.
- *Analog knob:* Rotate to set the amount of random pitch detuning. Analog simulates the instability of analog circuitry found in vintage vocoders by randomly altering the pitch of each note. This behavior is much like that of polyphonic analog synthesizers.
- *Glide knob:* Determines the time it takes for the pitch to slide from one note to another—portamento. Also see EVOC 20 PolySynth global parameters on page 145 for information about mono and legato mode.
- *Bend Range field:* Drag to determine the pitch bend modulation range, in semitone steps.

EVOC 20 PolySynth filter parameters

The Synthesis section includes a lowpass filter that is used for coarse signal shaping, before the signal is more precisely shaped by the individual bands of the formant filter banks.

Filter parameters

- *Cutoff knob:* Rotate to set the cutoff frequency of the lowpass filter. Turn to the left to remove high-frequency content from the synthesizer signal.
- *Resonance knob:* Rotate to boost or cut the signal portion that surrounds the frequency defined by the Cutoff knob.

  Tip: Set cutoff as high as possible, then adjust resonance to achieve a brighter high-end signal. This is useful for achieving better speech intelligibility.

EVOC 20 PolySynth envelope parameters

EVOC 20 PolySynth includes an Attack/Release envelope generator that controls the levels of the oscillators over time.

Envelope parameters

- *Attack slider:* Drag to set the time it takes for the oscillators to reach their maximum level.
- *Release slider:* Drag to set the time it takes for the oscillators to reach their minimum level, after the keys have been released.

EVOC 20 PolySynth global parameters

The parameters at the top left of the interface determine the keyboard mode and number of voices used by EVOC 20 PolySynth.

Global parameters

- *Poly/Mono/Legato buttons:* Click a button to determine the keyboard mode.
  - When Poly is on, you can set the maximum number of voices in the Voices field. (When Mono or Legato is on, a single voice is heard.)
  - When Mono is on, Glide is always active and the envelopes are retriggered by every note played (multi trigger behavior).
  - When Legato is on, Glide is active only on tied notes. Envelopes are not retriggered when tied notes are played (single trigger behavior). (See EVOC 20 PolySynth tuning and pitch parameters on page 143.)
- *Voices field:* Drag to set the maximum number of voices in the numeric field (only when Poly is turned on).
- *Unison button:* Click to turn Unison mode on or off.
  - In Unison/Poly mode—where both the Unison and Poly buttons are active—each voice is doubled. This cuts polyphony in half (to a maximum of eight voices, shown in the Voices field). The doubled voices are detuned by the amount defined with the Analog knob.
  - In Unison/Mono mode—where both the Unison and Mono or Legato buttons are active—up to 16 voices can be stacked and played monophonically. The Voices field displays the number of stacked voices that are heard.

  Important: Stacking voices in Unison/Mono mode increases the output volume. To avoid overloading the instrument channel strip output, set a low Level slider value and gradually increase it. See EVOC 20 PolySynth output parameters on page 149.

EVOC 20 PolySynth formant filter

EVOC 20 PolySynth features two formant filter banks—one for the Analysis section and one for the Synthesis section. Each bank provides up to 20 individual filters. The entire frequency spectrum of an incoming signal is analyzed by the Analysis section and is divided equally into a number of frequency bands. These analysis filter bands are mirrored by a corresponding number of bands in the synthesis filter bank. Each filter bank controls the peak levels—the formants—within these frequency bands.

The Formant Filter display is divided in two by a horizontal line. The upper half applies to the Analysis section and the lower half to the Synthesis section. Parameter changes are instantly reflected in the Formant Filter display, thus providing invaluable feedback about what is happening to the signal as it is routed through the two formant filter banks.

Formant filter parameters

- *Low and High Frequency parameters:* Drag to set the lowest and highest frequencies allowed to pass by the formant filter. Frequencies outside these boundaries are cut.
  - The length of the horizontal blue bar at the top represents the frequency range for both analysis and synthesis (unless Formant Stretch or Formant Shift is used). You can move the entire frequency range by dragging the blue bar. The silver handles on either end of the blue bar set the Low Frequency and High Frequency values, respectively.
  - You can also drag vertically in the numeric fields to adjust the Low and High frequency values.
- *Lowest and Highest buttons:* Click to determine whether the lowest and highest filter bands act as bandpass filters or whether they act as lowpass or highpass filters.
  - *Lowest button:* Determines whether the lowest filter band acts as a bandpass or highpass filter. In bandpass mode, the frequencies above and below the lowest band are ignored. In highpass mode, all frequencies below the lowest band are filtered.
  - *Highest button:* Determines whether the lowest filter band acts as a bandpass or lowpass filter. In bandpass mode, the frequencies above and below the highest band are ignored. In lowpass mode, all frequencies above the highest band are filtered.
- *Resonance knob:* Rotate to determine the basic sonic character of the vocoder. Low settings result in a softer character; high settings result in a sharper character. Technically, increasing the Resonance value emphasizes the middle frequency of each frequency band.

- *Formant Stretch knob:* Rotate to change the width and distribution of all bands in the synthesis filter bank. This can be a broader or narrower frequency range than that defined by the Low and High Frequency parameters.
  - When Formant Stretch is set to 0, the width and distribution of the bands in the synthesis filter bank match the width of the bands in the analysis filter bank. Low values narrow the width of each band in the synthesis filter bank, whereas high values widen the bands. The control range is expressed as a ratio of the overall bandwidth.
- *Formant Shift knob:* Rotate to move all bands in the synthesis filter bank up or down the frequency spectrum.
  - When Formant Shift is set to 0, the positions of the bands in the synthesis filter bank match the positions of the bands in the analysis filter bank. Positive values move the synthesis filter bank bands up in frequency, whereas negative values move them down—in respect to the analysis filter bank band positions.

When combined, Formant Stretch and Formant Shift alter the formant structure of the resulting vocoder sound, which can lead to interesting timbral changes. For example, using speech signals and tuning Formant Shift up results in "Mickey Mouse" effects.

Formant Stretch and Formant Shift are also useful if the frequency spectrum of the synthesis signal does not complement the frequency spectrum of the analysis signal. You could create a synthesis signal in the high-frequency range from an analysis signal that mainly modulates the sound in a lower-frequency range, for example.

Note: The use of the Formant Stretch and the Formant Shift parameters can result in the generation of unusual resonant frequencies when high Resonance settings are used.

EVOC 20 PolySynth modulation parameters

The Modulation section contains two LFOs that can either run freely or be synchronized with the host application tempo.

- The Pitch LFO controls pitch modulation of the oscillators, enabling you to produce vibrato effects.
- The Shift LFO controls the Formant Shift parameter of the synthesis filter bank, enabling you to produce dynamic phasing-like effects.

Modulation parameters

- *Int via Whl slider:* Drag to set the intensity of LFO pitch modulation. The right half of the slider determines the intensity when the modulation wheel is set to its maximum value; the left half determines the intensity when the wheel is set to its minimum value. By dragging the area between the two slider segments, you can simultaneously move both. This parameter is permanently assigned to the modulation wheel of your MIDI keyboard, or corresponding MIDI data.
- *Rate knobs:* Rotate to set the speed of modulation. Values to the left (of the centered position) are synchronized with the host application tempo. These include bar values, triplet values, and so on. Values to the right (of the centered position) are nonsynchronized, and are displayed in Hertz—cycles per second.

  Note: The ability to use synchronous bar values could be used to perform a formant shift every four bars on a cycled one-bar percussion part, for example. Alternatively, you could perform the same formant shift on every eighth-note triplet within the same part. Either method can generate interesting results and lead to new ideas, or add life to existing audio material.
- *Waveform buttons:* Click to set the waveform type used by the Pitch LFO (left column) or the Shift LFO (right column). You can choose from the following waveforms for each LFO:
  - Triangle
  - Falling and rising sawtooth
  - Square up and down around zero (bipolar, good for trills)
  - Square up from zero (unipolar, good for changing between two definable pitches)
  - Random stepped waveform (S & H)
  - Smoothed random waveform
- *Intensity slider:* Drag to define the amount of formant shift modulation by the Shift LFO.

EVOC 20 PolySynth output parameters

The Output section provides control over the type of signal, stereo width, and level of signal that is sent from EVOC 20 PolySynth. The Output section also has an ensemble effect processor.

Output parameters

- *Signal pop-up menu:* Choose the signal that is sent to the EVOC 20 PolySynth main outputs.
  - *Voc(oder):* Choose to hear the vocoder effect.
  - *Syn(thesis):* Choose to hear only the synthesizer signal.
  - *Ana(lysis):* Choose to hear only the analysis signal.

  Note: The last two settings are mainly useful for monitoring purposes.
- *Ensemble buttons:* Turn the ensemble effect on or off and determine the type of sound.
  - *Off:* Click to turn the ensemble effect off.
  - *I:* Click to obtain a special chorus effect.
  - *II:* Click to obtain a fuller and richer sound.
- *Level slider:* Drag to set the overall volume of the EVOC 20 PolySynth output signal.
- *Stereo Width knob:* Rotate to distribute the output signals of the Synthesis section filter bands in the stereo field.
  - At the 0 position to the left, the outputs of all bands are centered.
  - At the centered position, the outputs of all bands ascend from left to right.
  - At the Full position to the right, the bands are output—alternately—to the left and right channels.

EVOC 20 PolySynth performance tips

Level and frequency tips

A vocoder always generates the intersection point of the analysis and synthesis signals. If there's no treble portion in the analysis signal, the resulting vocoder output also lacks treble. This is also the case when the synthesis signal has a lot of high-frequency content. Because this is true of each frequency band, the vocoder demands a stable level in *all* frequency bands from *both* input signals to obtain the best results.

Achieving a great "classic" vocoder effect requires both the analysis and synthesis signals to be of excellent quality, and it also requires care to be taken with the vocoder parameters. These tips will help you achieve the best possible results.

- The less the level changes, the better the intelligibility of the vocoder. You should therefore compress the analysis signal in most cases.
- Due to the way human hearing works, the intelligibility of speech is highly dependent on the presence of high-frequency content. To aid in keeping speech clear, consider using equalization to boost or cut particular frequencies in analysis signals before you process them.
  - If the analysis signal consists of vocals or speech, a simple shelving filter should be sufficient to boost the high-mid and treble range, which is important for speech intelligibility.
  - If the *synthesis* signal lacks treble energy, it can be generated with a distortion effect. The Logic Pro Overdrive effect is perfect for this purpose.

Tips to avoid sonic artifacts

A common problem with vocoder sounds involves sudden signal interruptions—ripping, breaking sounds—and rapidly triggered noises during speech pauses.

The Release parameter defines the time it takes for a given synthesis frequency band to decrease in level if the signal level of the respective analysis band decreases abruptly. The sound is smoother when band levels decrease slowly. To achieve this smoother character, use higher Release values in the Analysis section. Take care to avoid setting an over-long release time, because this can result in a less distinct, washy sound. Use short Attack values when a fast reaction to incoming signals is required.

If the analysis signal is compressed as recommended, the level of breath, rumble, and background noise will rise. These unwanted signals can cause the vocoder bands to open unintentionally. To eliminate these artifacts, use a noise gate before using compression and boosting the treble frequencies. If the analysis signal is gated appropriately, you may be able to reduce the (Analysis) Release value.

When you gate speech and vocals with the Logic Pro Noise Gate plug-in, use Threshold to define the level above which the gate will open, and use Hysteresis to define a lower Threshold level below which the gate will close. The Hysteresis value is relative to the Threshold level.

The figure above shows a Threshold setting that is well-suited for speech compression. Unwanted triggering by low or high frequency noise is avoided by the dedicated sidechain filters of the Noise Gate plug-in. The Hold, Release, and Hysteresis values are suitable for most vocal and speech signals.

Tips to enhance speech intelligibility

Keep these points in mind to achieve the best possible speech intelligibility:

- The spectra of the analysis and synthesis signals should almost completely overlap. Coupling low male voices with synthesis signals in the treble range doesn't work well.
- The synthesis signal must be constantly sustained, without breaks. The incoming side-chain signal should be played or sung legato, because breaks in the synthesis signal stop the vocoder's output. Alternatively, the Release parameter of the synthesis signal—not the Release time of the Analysis section—can be set to a longer time. You can also achieve nice effects by using a reverberation signal as a synthesis signal. Note that the two latter methods can lead to harmonic overlaps.
- Do not overdrive the vocoder. This can happen easily, and distortion will occur.
- Enunciate your speech clearly if the recording is to be used as an analysis signal. Spoken words with a relatively low pitch work better than sung vocals—even if the creation of vocoder choirs is your goal. Pronounce consonants well, as exemplified in the rolled *"R"* of "We are the Robots," by Kraftwerk, a classic vocoder track. This exaggerated pronunciation was specifically made to cater to the vocoder.
- You can freely set Formant parameters. Shifting, stretching, or compressing the formants has a minimal effect on the intelligibility of speech, as does the number of frequency bands. The reason for this is due to the human ability to differentiate the voices of children, women, and men, whose skulls and throats vary. Such physical differences cause variations in the formants that make up their voices. Human perception, or recognition, of speech is based on an analysis of the *relationships* between these formants. In the EVOC 20 plug-ins, these relationships are maintained even when extreme formant settings are used.

Vocoder history

The development of the vocoder dates back to the 1930s in the telecommunications industry.

Homer Dudley, a research physicist at Bell Laboratories in New Jersey, developed the vocoder (short for *voice encoder*) as a research machine. It was originally designed to test compression schemes for the secure transmission of voice signals over copper phone lines.

It was a composite device consisting of an analyzer and an artificial voice synthesizer, as follows:

- *Parallel bandpass vocoder:* A speech analyzer and resynthesizer.
- *Vocoder speech synthesizer:* A voice modeler, this valve-driven machine was played by a human operator. It had two keyboards, buttons to recreate consonants, a pedal for oscillator frequency control, and a wrist-bar to switch vowel sounds on and off.

The analyzer detected the energy levels of successive sound samples, measured over the entire audio frequency spectrum via a series of narrow band filters. The results of this analysis could be viewed graphically as functions of frequency against time.

The synthesizer reversed the process by scanning the data from the analyzer and supplying the results to a number of analytical filters, hooked up to a noise generator. This combination produced sounds.

In World War II, the vocoder (known then as the *voice encoder*) proved to be of crucial importance, scrambling the transoceanic conversations between Winston Churchill and Franklin Delano Roosevelt.

Werner Meyer-Eppler, the director of Phonetics at Bonn University, recognized the relevance of the machines to electronic music—following a visit by Dudley in 1948. Meyer-Eppler used the vocoder as a basis for his future writings which, in turn, became the inspiration for the German "Elektronische Musik" movement.

In the 1950s, a handful of recordings ensued.

In 1960, the Siemens Synthesizer was developed in Munich. Among its many oscillators and filters, it included a valve-based vocoding circuit.

In 1967, a company called Sylvania created a number of digital machines that used time-based analysis of input signals, rather than bandpass filter analysis.

In 1971, after studying Dudley's unit, Bob Moog and Wendy Carlos modified a number of synthesizer modules to create their own vocoder for the *Clockwork Orange* soundtrack.

Peter Zinovieff's London-based company EMS developed a standalone—and altogether more portable—vocoder. EMS is probably best known for the Synthi AKS and VCS3 synthesizers. The EMS Studio Vocoder was the world's first commercially available machine, released in 1976. It was later renamed the EMS 5000. Among its users were Stevie Wonder and Kraftwerk. Stockhausen, the German "Elektronische Musik" pioneer, also used an EMS vocoder.

Sennheiser released the VMS 201 in 1977, and EMS released the EMS 2000, which was a cut-down version of its older sibling.

1978 saw the beginning of mainstream vocoder use, riding on the back of popularity created through the music of Herbie Hancock, Kraftwerk, and a handful of other artists. Among the manufacturers who jumped into vocoder production at this time are Synton/Bode, Electro-Harmonix, and Korg, with the VC-10.

In 1979, Roland released the VP 330 ensemble/vocoder keyboard.

The late 1970s and early 1980s were the heyday of the vocoder. Artists who used them included ELO, Pink Floyd, Eurythmics, Tangerine Dream, Telex, David Bowie, Kate Bush, and many more.

On the production side, vocoders could—and can still—be picked up cheaply in the form of kits from electronics stores.

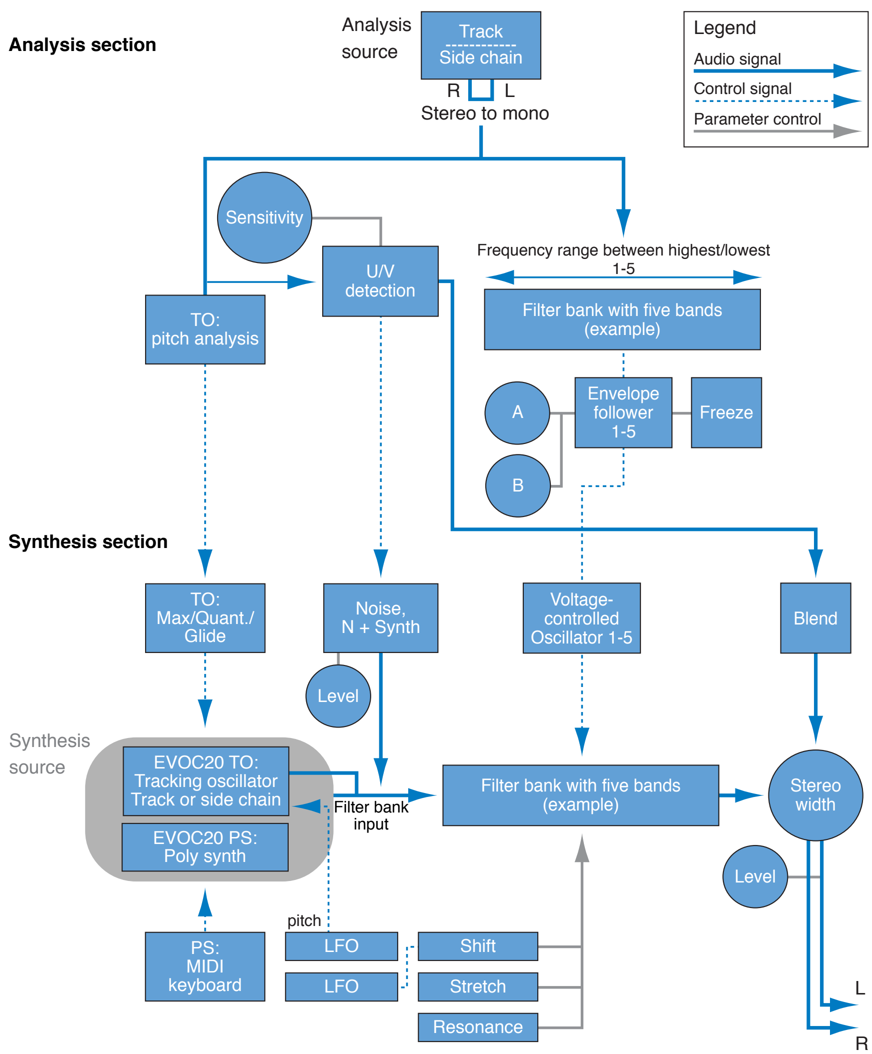
From 1980 to the present, EMS in the UK, Synton in Holland, and PAiA in the USA have been—and remain—the main flyers of the vocoding flag.

In 1996, Doepfer in Germany and Music and More joined the vocoder-producing fraternity.

From the late 1990s to the present, a number of standalone and integrated software-based vocoders—like the EVOC 20—have appeared.

EVOC 20 block diagram

This block diagram illustrates the signal path in EVOC 20 TrackOscillator and EVOC 20 PolySynth.



EXS24 mkII

9

EXS24 mkII overview

EXS24 mkII is a software sampler. It plays back audio files, called *samples*, that you load into it. These samples are combined into tuned, organized collections called *sampler instruments*. Because sampler instruments are based on audio recordings, they are ideally suited to emulating real instruments such as guitars, pianos, and drums.

EXS24 mkII lets you play, edit, and create sampler instruments. You can assign the samples in sampler instruments to specific key and velocity ranges and process them with EXS24 mkII filters and modulators.

EXS24 mkII offers powerful modulation and editing features and is a flexible synthesizer in its own right. This enables you to create expressive sounds by using any sample as a basic synthesizer waveform.

EXS24 mkII can be used as a mono or stereo instrument, or you can route loaded samples to multiple audio outputs. This enables you to independently process individual drum sounds in a drum kit, for example.

You can use samples of almost unlimited length in EXS24 mkII by streaming them directly from a hard disk. This lets you use many of the multigigabyte sample libraries available.

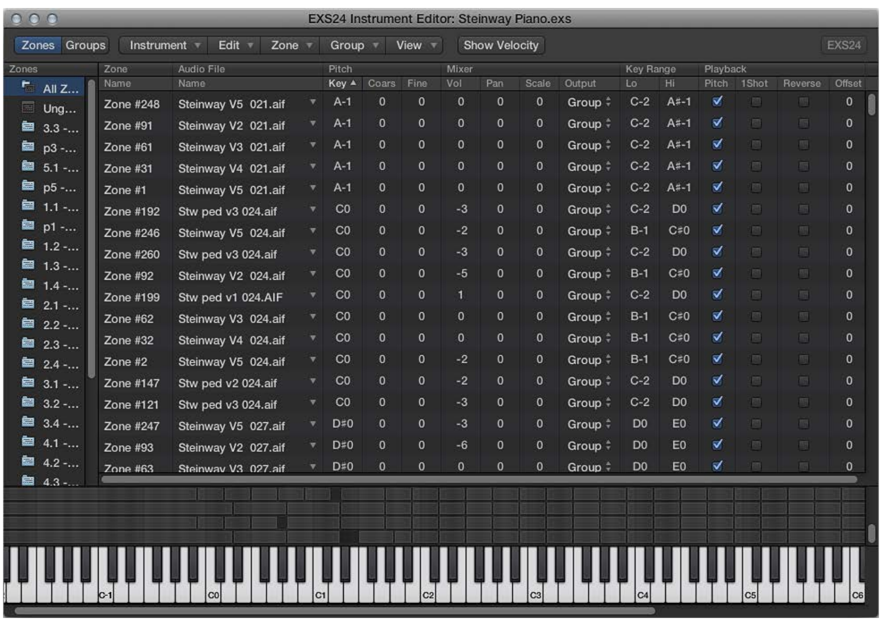
EXS24 mkII provides an extensive library of sampler instruments that includes piano, string, drum, acoustic and electric guitar, and many other sounds.

The EXS24 mkII native file format—the EXS format—is supported by most sample library providers. You can also import sampler instruments in the Gigasampler, DLS, and SoundFont2 sample file formats.

There are two EXS24 mkII windows.

- *Parameter window:* This window is used to load sampler instruments and contains synthesis and modulation options that enable you to customize your sounds. See EXS24 mkII Parameter window overview.

- *Instrument Editor window:* This window is used to create and edit sampler instruments. Click the Edit button in the Parameter window to open the Instrument Editor window. See EXS24 mkII Instrument Editor overview.



Sampler instruments

Sampler instruments overview

A sampler instrument is the file type that is loaded into EXS24 mkII. You load sampler instruments using the Sampler Instruments pop-up menu directly above the Cutoff knob. When you choose a sampler instrument, the associated audio files are automatically located on the hard disk (or disks), and are loaded into your computer's RAM. You play and record the loaded sampler instrument in the same way as any software instrument.

A sampler instrument tells EXS24 mkII which samples—audio files—to use, and how to organize them into zones and groups.

- A zone is a location into which a single sample—an audio file—can be loaded from a hard disk.
- Zones can be assigned to groups, which provide parameters that allow you to simultaneously edit all zones in the group. You can define as many groups as required.

EXS24 mkII is compatible with the following audio file formats: AIFF, WAV, SDII, and CAF. Each audio file is loaded into EXS24 mkII as a separate sample. Each audio file is then automatically assigned to a zone in the EXS24 mkII Instrument Editor window. These zones can be edited and organized into sampler instruments. See Zone and group edit overview on page 196.

Important: Audio files are *not* contained within a sampler instrument. The sampler instrument only stores information about audio filenames, parameter settings, and locations on the hard disk. If you delete or rename an audio file, any sampler instrument that uses this file will not be able to find it. You can move audio files to another location on your system, however, because EXS24 mkII automatically searches for files when you load a sampler instrument.

EXS24 mkII is compatible with the SoundFont2, DLS, Gigasampler, and ReCycle sample formats, as well as the Vienna Library. See Import SoundFont2, DLS, and Gigasampler files on page 159 and Convert ReCycle files to sampler instruments on page 162.

Sample storage locations

To be visible in the EXS24 mkII Sampler Instruments pop-up menu, instruments must be stored in the Sampler Instruments subfolder of any of the following folders:

- *~/Library/Application Support/Logic:* User-defined or edited instruments are stored here.
- */Library/Application Support/Logic:* Factory-supplied sampler instruments are installed here.
- *…/ProjectName:* Logic Pro also searches for sampler instruments in the project folder.

  Search by project is included for backward compatibility with existing projects.

  - If you use the Save As command, old project files and folders are converted to project bundles. The original files are not changed.
  - If you save a new project, sampler instruments and samples are saved with the project bundle.

Note: You can store your sampler instruments in any folder on any connected hard disk. Create an alias pointing to this folder within a Sampler Instruments subfolder (using any of the paths listed above) and they are shown in the Sampler Instruments pop-up menu.

Manage sampler instruments

As your sample library grows, the list of sampler instruments also expands. To help you keep the list of sampler instruments manageable, EXS24 mkII provides a simple, flexible file management method.

It is recommended that you copy any sampler instruments and all associated audio files to your hard drive. This provides access to your sampler instruments with no need to find and insert CD-ROM or DVD discs, and it also enables you to organize your sampler instruments to meet your needs. Load times are faster, and you can play back samples that exceed the size of your computer's RAM by streaming them from your hard disk. This feature is not practical for optical drives.

If you are using Logic Pro X, you can save sampler instruments and associated audio files with the project bundle. This makes it easy to keep all sampler instruments and audio samples in one place.

Organize your sampler instruments into a preferred hierarchy

1. Create a folder on the desktop—named *Basses*, for example—and drag it into the target folder.
2. Drag the sampler instruments you want to move into this newly created folder.

   The modified menu structure is reflected when you open the Sampler Instruments pop-up menu.

Note: Use the Refresh Menu command in the Sampler Instruments pop-up menu after changes are made to the folder hierarchy in the Sampler Instruments folder.

The Sampler Instruments pop-up menu displays submenus only for folders that actually contain sampler instrument files. Other folders are not added to the menu. Aliases pointing to folders that contain sampler instrument files outside the Sampler Instruments folders can also be added to the menu. The Sampler Instruments folder itself can be an alias to a folder on a different drive or in another location.

Copy sampler instruments to your hard drive

1. To open the Library folder in a Finder window, Option-click the Go menu from the desktop, then click Library.
2. Copy the sampler instrument file into the ~/Library/Application Support/Logic/Sampler Instruments folder.
3. Copy the associated samples into a folder named *Samples* in the same folder as the Sampler Instruments folder.

Create a backup of sampler instruments in Logic Pro

- To copy the audio and sampler instrument files of all currently active sampler instruments in the project to a specified file location, use the Backup Audio Files of All Used and Active Instruments of Current Project key command.

  Folders for the audio files associated with these sampler instruments are created in the target location.

  Tip: You can also do this by saving a project. By default, sampler instruments and samples are saved in the project bundle. For more information, see Logic Pro Help.

Use sampler instruments and settings

Sampler instruments are distinct from plug-in settings, which are loaded and saved in the plug-in window header. Each has advantages and disadvantages for handling parameter values in the Parameter window.

Typically, you store the current Parameter window settings into the loaded sampler instrument. This overrides the settings currently saved in the loaded sampler instrument. Alternatively, you can save a new sampler instrument.

A plug-in setting, by comparison, stores all parameter adjustments made in the Parameter window, but these settings are distinct from the sampler instrument being loaded. A plug-in setting contains only a pointer to an associated instrument, which means that loading a setting also loads the assigned sampler instrument.

This separation between plug-in settings and sampler instruments enables you to use sampler instruments as you would use waveforms in a synthesizer. For example, you could create a plug-in setting with guitar-like envelope, modulation, and filter parameter values. You would then use the Sampler Instruments pop-up menu to load an instrument without any existing settings, such as a flute, to create a plucked or strummed flute sound.

Important: Using sampler instruments as described requires that they contain no settings.

Remove settings from an existing sampler instrument

1. Choose Options > "Save Instrument as" to create a copy of the sampler instrument (see EXS24 mkII Options pop-up menu commands on page 167).
2. Choose Options > Delete Settings from Instrument to remove the settings from the copied instrument.

Note: All of the factory-supplied sampler instruments contain settings, so you need to follow the steps above to use these instruments in this way.

Import SoundFont2, DLS, and Gigasampler files

EXS24 mkII recognizes SoundFont2, DLS, and Gigasampler files placed inside the Sampler Instruments folder and converts them into sampler instruments.

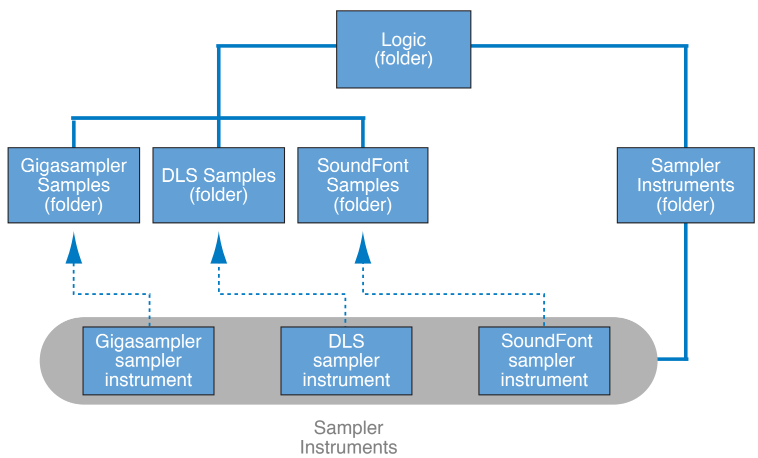
You can store your imported sampler instruments in any folder on any of your computer's hard drives. To access these instruments from the Sampler Instruments pop-up menu, you need to create an alias pointing to the target folder within the ~/Library/Application Support/Logic/Sampler Instruments folder.

Import SoundFont2, DLS, or Gigasampler files into EXS24 mkII

1. To open the Library folder in a Finder window, Option-click the Go menu from the desktop, then click Library.
2. Copy or move your SoundFont2, DLS, or Gigasampler files into the ~/Library/Application Support/Logic/Sampler Instruments folder.
3. Choose the SoundFont2, DLS, or Gigasampler file from the Sampler Instruments pop-up menu.

   EXS24 mkII automatically converts the chosen file into a sampler instrument, as follows:

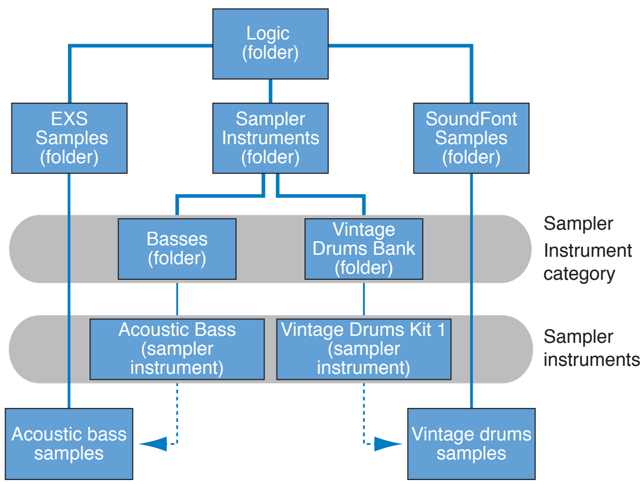
   - A sampler instrument file is created in the Sampler Instruments folder. This contains the file in its original format.
   - The raw samples associated with the sampler instrument are placed in one of the following folders, depending on the format being converted:
     - ~/Library/Application Support/Logic/SoundFont Samples
     - ~/Library/Application Support/Logic/Gigasampler Samples
     - ~/Library/Application Support/Logic/DLS Samples



The procedure outlined above also applies when you are importing SoundFont2 files. These file types contain multiple sounds in addition to single-instrument files.

When you load a SoundFont2 Bank into EXS24 mkII, it creates a Bank folder and a Samples folder, named after the SoundFont2 Bank file. The word *Bank* or *Samples* is appended to each folder name. A sampler instrument file is automatically created for all sounds in the bank, and placed in the new Bank folder. The Sampler Instruments pop-up menu automatically updates to reflect the new folder hierarchy. All samples associated with the bank are automatically added to a Samples folder inside the SoundFont folder. For example, if you load a SoundFont2 bank file named Vintage Drums, which contains over 50 individual drum kits from several different vintage drum machines, into EXS24 mkII, the following happens:

- A new folder named *Vintage Drums Bank* is created in the ~/Library/Application Support/Logic/Sampler Instruments folder.
- A second folder named *Vintage Drums Samples* is created in the ~/Library/Application Support/Logic/SoundFont Samples folder.
- The Sampler Instruments pop-up menu hierarchy is updated and the original Vintage Drums entry is replaced with a Vintage Drums.Bank entry. This new entry is a folder that contains the individual sampler instruments, which can be selected and loaded as usual.



After conversion is complete, the original SoundFont2 or Gigasampler source files can be deleted from the hard disks.

Convert audio regions to sampler instruments

You can convert audio regions to sampler instruments with the Convert Regions to New Sampler Track function (default key command: Control-E). All selected regions are sequentially mapped—in accordance with their timeline positions—to the specified key range, starting with the lowest note.

A new track is also created, with automatically created trigger notes for the converted audio regions. These trigger notes match the time positions of the source audio regions. You can reposition or edit the trigger notes in the Piano Roll editor. You can also change zone parameters in the EXS24 mkII Instrument Editor. See EXS24 mkII Instrument Editor overview.

Convert audio regions to sampler instruments and tracks

1. In the Tracks window, select the audio regions you want to convert.
2. Right-click or Control-click any selected region, then choose Convert > Convert Regions to New Sampler Track (or use the default key command: Control-E).
3. In the dialog, choose to Create Zones From either Regions or Transient Markers.
4. Enter the EXS Instrument Name in the field.
5. Set the Trigger Note Range in one of the following ways:
   - Play a low key on your MIDI keyboard to set the low note of the trigger range. Play a second key to set the top note of the trigger range.
   - Choose a low and high note from the Trigger Note Range pop-up menus.
6. Click OK to create a new sampler instrument and track or click Cancel if you change your mind.

Convert ReCycle files to sampler instruments

ReCycle, a sample editing program from Propellerhead Software, can generate a number of file types that can be read by Logic Pro and EXS24 mkII.

ReCycle separates sample material into small segments called *slices*, based on waveform peaks, or transients, in the audio file. In this way, ReCycle is able to split an audio file into musically relevant slices. The timing of these slices—in a drum loop, for example—is also mapped to an automatically generated region in Logic Pro.

EXS24 mkII supports the following ReCycle file types:

- *Old ReCycle file:* These files have the *.rcy* suffix, and the abbreviation for this file type is *RCSO*. These files are no longer commonly used.
- *Old ReCycle export file:* These files have the *.rex* suffix, and the abbreviation for this file type is *REX*. A number of older sample libraries include REX format files.
- *ReCycle 2.0 file:* These files have the *.rx2* suffix, and the abbreviation for this file type is *REX2*. These files are used extensively by Propellerhead Reason, and many popular sample libraries include REX2 format files.

Create a new sampler instrument and assign each ReCycle slice to a zone

1. Choose Instrument > ReCycle Convert > Extract MIDI Region and Make New Instrument in the Instrument Editor.
2. Select the ReCycle file, and click Open.
3. Enter a velocity factor in the Create MIDI Region dialog.

The velocity factor analyzes the loudness—transient peak—of each slice in the imported ReCycle file. It then maps this value to a corresponding velocity value for the MIDI note event that is used to trigger the slice.

- If you enter a positive value (up to 100), louder slices generate MIDI note events with higher velocity values.
- If you enter a negative value, louder slices generate MIDI note events with lower velocity values.

4. Click OK.

EXS24 mkII generates a zone for each slice of the imported ReCycle file and assigns these zones to a single group. The new sampler instrument is named after the ReCycle loop. If a sampler instrument of that name already exists, a number sign (#) and a number are appended to the name. For example, if you import a ReCycle file named *Tricky Backbeat* but Tricky Backbeat already exists as a sampler instrument, the imported instrument would be named *Tricky Backbeat#1*, thus ensuring that the filename is unique within the Sampler Instruments folder.

In addition, a MIDI region is generated on the currently selected track, at the current project position, rounded to whole bars. You use this MIDI region to trigger the imported slices at the timing defined by the ReCycle file. You can generate new MIDI regions at any time from the imported sampler instrument (see "Generate a new MIDI region from a ReCycle instrument"), so you can feel free to modify or delete the region.

Tip: The Extract MIDI Region and Add Samples to Current Instrument command also allows you to add the slices of a ReCycle loop to any sampler instrument currently opened in the Instrument Editor. This allows you to use several different ReCycle loops in a single sampler instrument.

Assign a complete ReCycle loop to a zone

- Choose Instrument > ReCycle Convert > Slice Loop and Make New Instrument to create a sampler instrument from a ReCycle loop.

Each slice is chromatically mapped across the keyboard—from low to high notes.

Each zone plays back the ReCycle loop to the end point, at the current project tempo. This means that the lowest zone plays back the entire loop, but the highest zone plays only the last slice of the loop. Notes between the lowest and highest zones play several slices. This allows for drum'n'bass-style note triggering, where the sample loop start point is determined by playing the respective notes on the keyboard.

If you want to add zones from the sliced loop to the currently active sampler instrument, you can choose Instrument > ReCycle Convert > Slice Loop and Add Samples to Current Instrument.

Paste ReCycle loops from the Clipboard

You can generate a MIDI region from imported ReCycle files. These regions trigger the imported slices at the timing defined by the ReCycle files.

- Choose Edit > Paste ReCycle Loop as New Instrument to create a sampler instrument from a ReCycle loop that was copied to the Clipboard using ReCycle's Copy Loop feature.

Instrument creation is identical to the Extract MIDI Region and Make New Instrument command.

If you want to add zones to the currently active sampler instrument, you can choose Edit > Paste ReCycle Loop to Current Instrument.

Generate a new MIDI region from a ReCycle instrument

You can generate a MIDI region from imported ReCycle files. These regions trigger the imported slices at the timing defined by the ReCycle files.

- Choose Instrument > ReCycle Convert > Extract Region(s) from ReCycle Instrument.

MIDI regions are created on the currently selected track, at the current project position, rounded to bars. A single MIDI region is generated for each imported ReCycle loop in the currently open instrument. You will also need to enter a velocity factor (see "Create a new sampler instrument and assign each ReCycle slice to a zone").

EXS24 mkII Parameter window

EXS24 mkII Parameter window overview

You use the EXS24 mkII Parameter window to change and control the entire loaded sampler instrument. You control individual samples (zones), or grouped samples, in the Instrument Editor window. See EXS24 mkII Instrument Editor overview on page 189.

The Parameter window contains the following parameter groups:

- *Sampler Instruments pop-up menu and field:* Click to access, find, and load your sampler instruments. The name of the loaded sampler instrument is displayed in the field. The related Edit and Options buttons are to the right. See Use the Sampler Instruments pop-up menu on page 165.
- *Global parameters:* Use to select and configure sampler instruments, define the polyphony, set crossfades, and so on. See EXS24 mkII global parameters overview.
- *Pitch parameters:* Use to adjust tuning, transposition, and pitch bend behavior. See EXS24 mkII pitch parameters on page 171.
- *Filter parameters:* Use to shape the tonal color of the loaded sampler instrument. See EXS24 mkII filter overview on page 173.
- *Output parameters:* Use to control the level and keyboard scaling of the loaded sampler instrument. See EXS24 mkII output parameters on page 175.
- *Extended parameters:* Click the disclosure triangle at the lower left of the interface to access additional MIDI and randomization parameters. See EXS24 mkII extended parameters.
- *Modulation router:* Use the modulation router in the strip across the center of the interface to link modulation sources, such as envelopes and LFOs, to modulation targets, such as oscillators and filters. See Use the EXS24 mkII modulation router on page 177.
- *Modulation and control parameters:* Use the modulation and control parameters in the area immediately below the router to assign and adjust the LFOs and envelopes. See EXS24 mkII modulation overview on page 176.

Sampler Instruments pop-up menu

Use the Sampler Instruments pop-up menu

This section outlines the use of the Sampler Instruments pop-up menu. The Edit button opens the Instrument Editor window. The Options button opens a pop-up menu. See EXS24 mkII Options pop-up menu commands on page 167.

Load an instrument

1 Click the Sampler Instruments field to open the Sampler Instruments pop-up menu.

2 Choose the sampler instrument you want to edit or play.

Load the next or previous instrument in your sampler instrument library

Do one of the following:

- Click the plus or minus button to choose the next or previous instrument in your sampler instrument library.

- Choose Next Instrument or Previous Instrument from the Sampler Instruments pop-up menu (or use the Next EXS Instrument or Previous EXS Instrument key command).

  If EXS24 mkII is the key focus window, you can also use the following key commands:

  - Next Plug-In Setting or EXS Instrument
  - Next Channel Strip or Plug-In Setting or EXS Instrument
  - Previous Plug-In Setting or EXS Instrument
  - Previous Channel Strip or Plug-In Setting or EXS Instrument

  Tip: You can also browse through your sampler instruments by using your MIDI keyboard. The Sampler Preferences window offers Previous EXS Instrument and Next EXS Instrument options. These allow you to assign a MIDI event, such as a MIDI note, control change, or program change to select the previous or next sampler instrument. See EXS24 mkII preferences on page 211.

Update the Sampler Instruments pop-up menu

- Choose Refresh Menu from the Sampler Instruments pop-up menu.

  Choosing this menu item scans all default file locations and updates the Sampler Instruments pop-up menu. Use this command when you have finished importing or creating new sampler instruments.

Load sampler instruments from other locations

You can manually load sampler instruments that are not shown in the Sampler Instruments pop-up menu. This is done using the Instrument pop-up menu in the Instrument Editor window.

1 To open the Instrument Editor window, click the Edit button in the Parameter window.

2 Choose Instrument > Open, then browse to the instrument in the dialog.

Open the Instrument Editor window

- Click the Edit button to open the Instrument Editor window.

Note: Clicking the Edit button when no sampler instrument is loaded also opens the Instrument Editor window and automatically creates a new, empty, sampler instrument. See EXS24 mkII Instrument Editor overview on page 189.

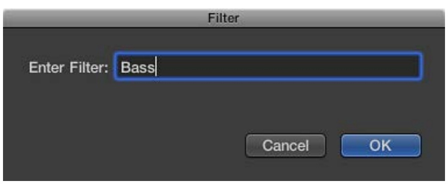
Search for sampler instruments

To minimize the number of sampler instruments displayed in the Sampler Instruments pop-up menu, you can use the Find function. This limits the Sampler Instruments pop-up menu to displaying only sampler instrument names that contain the search term.

Search for sampler instruments

1 Choose Find from the Sampler Instruments pop-up menu.

2 Enter the search term in the Filter dialog.



Disable the search filter

- Choose Clear Find from the Sampler Instruments pop-up menu.

The full Sampler Instruments pop-up menu is displayed, but the search term you typed in the Filter dialog is not cleared. You can return to the limited menu by choosing Enable Find from the Sampler Instruments pop-up menu. This allows you to switch between the two views without re-typing the search term.

Perform a different search

- Choose Find from the Sampler Instruments pop-up menu again, then enter the search term.

EXS24 mkII Options pop-up menu commands

Click Options to open a pop-up menu of settings and sampler instruments management commands, to import non-native samples and instruments, and to access EXS24 mkII preferences and memory management.

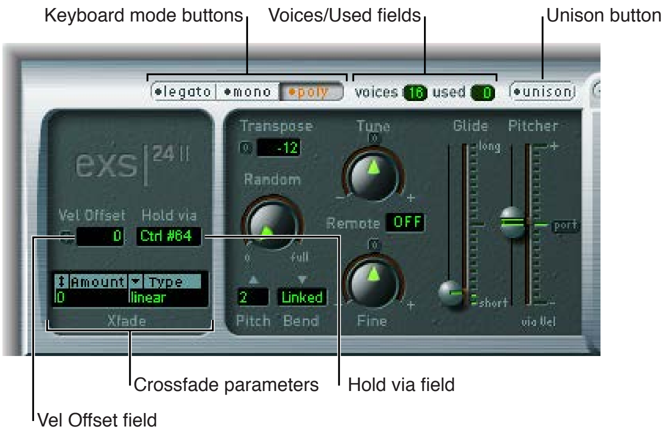
- *Recall Default EXS24 Settings:* Recalls a neutral setting for all parameters in the Parameter window. This provides a "clean slate" when you are adjusting the parameters of your sampler instrument.
- *Recall Settings from Instrument:* Recalls the original parameter settings of the loaded sampler instrument. This option is useful if you've been overzealous with your tweaking and want to return to the original sampler instrument parameter settings.
- *Save Settings to Instrument:* Stores the current parameter values of the Parameter window in the sampler instrument file. When the instrument is reloaded, these values are recalled.
- *Delete Settings from Instrument:* Removes the stored settings (Parameter window values) from the sampler instrument file.
- *Rename Instrument:* Allows you to rename the loaded instrument, which overwrites the existing instrument name.
- *Save Instrument as:* Allows you to enter an alternative name for the loaded instrument, which preserves the original name and sampler instrument file, but creates a new sampler instrument—a copy.

  Note: This is a good, safe option to use because any projects or templates that use the originally named sampler instrument will work as expected.
- *Delete Instrument:* Deletes the loaded sampler instrument.
- *(Recall Default EXS24 mkI Settings):* Recalls the parameter settings of sampler instruments created in the older version of the EXS24—notably, the modulation paths (see EXS24 mkI modulation paths on page 180). This option is not relevant for sampler instruments created in EXS24 mkII.
- *Extract MIDI Region(s) from ReCycle Instrument:* Extracts regions contained in a ReCycle instrument. If no ReCycle instrument is selected, this option is dimmed. See Convert ReCycle files to sampler instruments on page 162.
- *SoundFont Convert, DLS Convert, Giga Convert:* Each of these commands opens a dialog with instructions about performing the respective conversions. See Import SoundFont2, DLS, and Gigasampler files on page 159 for details.
- *Preferences:* Opens EXS24 mkII preferences (see EXS24 mkII preferences on page 211).
- *Virtual Memory:* Opens a configuration dialog for EXS24 mkII's virtual memory functions. Virtual memory allows samples of almost unlimited length to be played back, by streaming audio directly from the hard disk in real time. The Virtual Memory dialog also enables direct access to system memory for EXS24 mkII—in systems with 5 GB of RAM or more. See EXS24 mkII memory management on page 214.

EXS24 mkII global parameters

EXS24 mkII global parameters overview

The global parameters affect the overall behavior of EXS24 mkII.



Global parameters

- *Keyboard mode buttons:* Switch EXS24 mkII between legato, monophonic, and polyphonic behaviors. See Set the EXS24 mkII keyboard mode on page 169.
- *Unison button:* Turns Unison mode on or off. See Use Unison and Voices in EXS24 mkII on page 169.
- *Voices and Used fields:* Set the maximum number of notes that can be played simultaneously. The Used field is a real-time monitor that indicates the number of voices that are actually used when you play the keyboard.
- *Vel Offset field:* Increases or decreases incoming MIDI note velocity values by ±127, expanding or limiting the dynamic response to MIDI note events.
- *Hold via field:* Sets the modulation source used to trigger the sustain pedal function (hold all currently played notes, and ignore note-off messages until the modulation source value falls below 64). The default is MIDI controller number CC 64 (the standard MIDI "Hold/Sustain" controller number).
- *Crossfade parameters:* Use to crossfade between layered samples—zones—with adjacent velocity ranges. See EXS24 mkII crossfade parameters on page 170.

Set the EXS24 mkII keyboard mode

A polyphonic instrument, such as an organ or piano, allows several notes to be played simultaneously. Brass or reed instruments are monophonic, which means that only one note can be played at a time. EXS24 mkII lets you choose an appropriate keyboard mode for the type of instrument that is loaded. You are free to use a monophonic mode for polyphonic instruments, which allows playing styles that are not possible with polyphonic instruments.

Change the keyboard mode

- Click the Legato, Mono, or Poly button.

- In Legato mode, only one note can be played at a time. The envelope generators are retriggered only if you play staccato—releasing each key before playing a new key.
- In Mono mode, staccato playing retriggers the envelope generators every time a new note is played. If you play in a legato style (play a new key while holding another), the envelope generators are triggered only for the first note you play legato, then they continue their curve until you release the last legato played key.
- In Poly mode, several notes can be played simultaneously.

Note: In many classic monophonic analog synthesizers, the behavior in Legato mode is referred to as *single trigger*, while Mono mode is referred to as *multi trigger*.

Use Unison and Voices in EXS24 mkII

In Unison mode, up to 64 voices are played when a key is struck. This enables a richer sound, achieved by slightly detuning each voice. This is ideal when you are emulating classic analog synthesizers.

Voices are equally distributed in the panorama field and are evenly detuned. You can use the Random knob to determine the amount of tuning deviation between voices.

Note: The number of voices actually used per note increases with the number of layered sample zones.

Use monophonic unison mode

1. Click the Mono or Legato button, depending on the keyboard mode you want to use. See Set the EXS24 mkII keyboard mode.
2. Click the Unison button.
   - The intensity of the unison effect depends on the number you set in the Voices field. Increase the Voices value for a fatter sound.
   - The intensity of detuning—voice deviation—depends on the value you set with the Random knob. See EXS24 mkII pitch parameters on page 171.

Use polyphonic unison mode

- Click the Poly *and* Unison buttons.

In Poly/Unison mode, each played note is effectively doubled, or—more correctly—the polyphony value of the Voices parameter is halved. These two voices are heard when you trigger the note. Selecting Poly and Unison has the same effect as setting the EXS24 mkII to Mono and Unison with Voices set to 2, but you can play polyphonically.

Set the number of voices

- Drag the Voices field to determine the maximum number of voices (polyphony) that EXS24 mkII can play.

The Used field is a real-time monitor that indicates the number of voices in use when you play the keyboard. If the Voices and Used fields show the same value most of the time (causing voices to drop out), set a higher Voices value.

EXS24 mkII crossfade parameters

The crossfade (Xfade) parameters enable you to crossfade between layered samples—known as *zones* in EXS24 mkII—with adjacent velocity ranges.

When you assign a sample to a zone, you can set the lowest and highest MIDI note velocity that will trigger that zone. The area between these values is known as the zone's velocity range. You can layer zones—different samples—on the same keyboard note, and trigger them individually by playing at different velocities. For example, imagine the following two layered samples, zone 1 and zone 2, on MIDI note A#2:

- Zone 1 is a sample of a snare drum hit lightly and a little off-center. It has a MIDI note velocity range of 24 to 90.
- Zone 2 is a sample of a snare drum hit hard in the center of the drum head. It has a velocity range of 91 to 127.

In this example, the maximum velocity range value of zone 1 and the minimum velocity range value of zone 2 are adjacent. If you play note A#2 at velocities above or below a value of 90, you will clearly hear each sample being triggered. To make this transition less abrupt, use the crossfade parameters to smoothly fade between each zone. When you have distinctly different audio samples in adjacent zones, you will find crossfading very helpful in creating realistic-sounding sampler instruments.

Crossfade (Xfade) parameters

- *Amount field:* Expands the velocity range of all zones by applying an identical value to each layered zone. The crossfade takes place in the extended velocity range area. When the Amount parameter is set to 0, EXS24 mkII switches from one zone to another.

  Note: You can also set other modulation sources, such as the modulation wheel of your MIDI keyboard, to modulate the Amount parameter. If you do this, the Amount parameter still functions in the same way, but the crossfade is triggered by the modulation wheel rather than by velocity.

- *Type pop-up menu:* Choose the curve type for your velocity crossfade:
  - *dB lin (dB linear):* A logarithmic curve that evenly crossfades between zones
  - *linear (gain linear):* A convex crossfade curve with a rapid volume fade toward the end
  - *Eq. Pow (equal power):* A nonlinear curve with a rapid level increase at the beginning of the fade. This is useful if your crossfade seems to drop in volume part way through.

EXS24 mkII pitch parameters

The EXS24 mkII pitch parameters enable you to adjust the tuning and transposition of the loaded sampler instrument.

Pitch parameters

- *Tune knob:* Rotate to raise or lower the pitch of the sampler instrument in semitone increments. At the centered position—click the small 0 at the top—no pitch change occurs.
- *Transpose field:* Transpose EXS24 mkII in semitone increments. This not only affects the pitch but also moves the zones by the specified value.
- *Random knob:* Rotate to set the amount of random detuning applied to each voice. Use this parameter to simulate the tuning drift of analog synthesizers or to thicken the sound. Random is also effective when you are emulating various stringed instruments.
- *Fine knob:* Rotate to tune the sampler instrument in cent increments—1/100th of a semitone. Use this parameter to correct samples that are slightly out of tune or to create a thick chorus-like effect.
- *Pitch Bend Up and Down pop-up menus:* Set the upper and lower limit of pitch bends—in semitones—that can be introduced by moving your keyboard's pitch bend wheel. A value of 0 disables pitch bends.

  Note: When you choose Linked in the Pitch Bend Down pop-up menu, the bend range is identical in both directions—for example, if you assign an upward bend of 4 semitones, the downward bend is also set to 4 semitones, resulting in a combined bend range of 8 semitones (9, if you include the standard pitch, or "no bend" position).

- *Remote field:* Use to remotely change the pitch of complete EXS24 mkII instruments in real time. You can define a key on your MIDI keyboard that is used as the original, or "reference" pitch. After the reference pitch is set, playing any of the keys in a range of ±1 octave above or below this key changes the pitch of the entire instrument, rather than triggering a sample. This is similar to the Pitch Bend function but is quantized to semitones.
- *Glide and Pitcher sliders:* Glide sets the time it takes to slide from one note pitch to another. Its behavior depends on the Pitcher parameter setting.
  - When Pitcher is centered, Glide determines the time it takes for the pitch to slide from one note to another—the portamento time.
  - When Pitcher is set to a position above the center value, Glide determines the time it takes for the pitch to glide down from this higher value to the normal pitch value.
  - When Pitcher is set to a position below the center value, the pitch glides from this lower setting back up to the normal pitch value.

  Pitcher can be modulated by velocity—the upper slider half determines the setting for maximum velocity, the lower half for minimum velocity. By dragging the area between the two slider segments, you can move both simultaneously. When the upper half of the Pitcher slider is set above the center position and the lower half below the center position, lower velocity values cause the pitch to rise from the lower setting up to the original note pitch, whereas higher values cause it to fall from the higher setting down to the original note pitch. When both halves of the Pitcher slider are set either below or above the centered position, a low or high velocity will slide up or down to the original pitch. Depending on the position of the upper and lower halves of the slider—relative to the center position—the time required for the slide up or down to the original note pitch can be adjusted independently for both soft and hard velocities.

  Note: In Legato mode, Glide is active only on tied notes and envelopes are *not* retriggered when tied notes are played. In other words, playing a series of tied notes results in only a single envelope trigger. In Mono mode, Glide is always active and the envelopes are retriggered by every note played.

EXS24 mkII filter parameters

EXS24 mkII filter overview

These parameters control the EXS24 mkII filter. You can configure the type of filter, filter resonance, cutoff frequency, drive, and amount of key follow. For details about the filter envelope, see EXS24 mkII envelope overview on page 184.

Filter parameters

- *Filter On/Off button:* Turns the entire filter section and the filter envelope on or off. Disabling the filter section makes it easier to hear adjustments to other sound parameters because the filter always heavily affects the sound.
- *Filter Mode/Slope buttons:* Labeled HP, LP, and BP at the bottom of the filter section, these buttons determine the type and slope of the filter. See Set the EXS24 mkII filter mode and slope on page 174.
- *Cutoff knob:* Rotate to set the cutoff frequency of the filter. The Cutoff value also serves as the starting point for any modulation involving the filter.
  - In a lowpass filter, the higher the cutoff frequency is set, the higher the frequencies of signals that are allowed to pass.
  - In a highpass filter, the cutoff frequency determines the point where lower frequencies are suppressed, with only upper frequencies allowed to pass.
  - In a bandpass/band-rejection filter, the cutoff frequency determines the center frequency for the bandpass or band-rejection filter.
- *Resonance knob:* Rotate to boost or cut the frequency area surrounding the cutoff frequency. Very high Resonance values introduce self-oscillation, causing the filter to produce an audible sine wave.
  - In a lowpass filter, resonance emphasizes or suppresses signals below the cutoff frequency.
  - In a highpass filter, resonance emphasizes or suppresses signals above the cutoff frequency.
  - In bandpass filters, resonance emphasizes or suppresses the portions of the signal—the frequency band—that surround the defined frequency, set with the Cutoff knob. In EXS24 mkII, Resonance defines the width of the frequency band.

- *Drive knob:* Rotate to overdrive the filter input, leading to a denser, more saturated signal, which introduces additional harmonics. Drive affects each voice independently. When every voice is overdriven individually—like having six fuzz boxes for a guitar, one for each string—you can play extremely complex harmonies over the entire keyboard range. They'll sound clean, without unwanted intermodulation effects spoiling the sound. Some Drive settings lead to a different tonal character. The way analog filters behave when overdriven forms an essential part of the sonic character of a synthesizer. Each synthesizer is unique in the way its filters behave when overdriven. EXS24 mkII is flexible in this regard, allowing tonal colors that range from a subtle fuzz to a hard distortion.
- *Key knob:* Rotate to define the amount of filter cutoff frequency modulation by note number. When Key is set to the full-left position, the cutoff frequency is unaffected by the note number and is identical for all played notes. When Key is set to the full-right position, the cutoff frequency follows the note number one-to-one; if you play one octave higher, the cutoff frequency is also shifted upward by one octave. This parameter is useful for avoiding overly filtered high notes.
- *Fat button:* Click to turn on the fatness feature. Fatness preserves the bass frequency response, even when high Resonance settings are used.

  Note: The Fatness parameter applies only to lowpass filters. Fatness is nonfunctional when the highpass or bandpass filter types are used.

Control two filter parameters at once

- Drag the chain symbol between the Cutoff and the Resonance knobs to control both parameters simultaneously—vertical movements alter Cutoff values and horizontal movements affect Resonance values.

Set the EXS24 mkII filter mode and slope

The EXS24 mkII filter can operate in several modes, allowing specific frequency bands to be filtered (cut away) or emphasized.

Most filters do not completely suppress the portion of the signal that falls outside the frequency range defined by the Cutoff parameter. The slope, or curve, chosen for the lowpass filter expresses the amount of rejection below the cutoff frequency in decibels per octave. The steeper the slope, the more severe the effect on signal levels below the cutoff frequency.

Select a filter mode and slope

Click one of the following buttons:

- *HP (highpass):* Allows frequencies above the cutoff frequency to pass.

  The slope of the highpass filter is fixed at 12 dB/octave in HP mode.

- *LP (lowpass):* Allows frequencies that fall below the cutoff frequency to pass. Click any of the four filter buttons to engage the lowpass filter and to select a filter slope:
  - *24 dB (4 pole):* Use this setting for drastic effects, such as cutting off all but a few notes.
  - *18 dB (3 pole)*
  - *12 dB (2 pole)*
  - *6 dB (1 pole):* Use this setting for creating a slightly warmer sound without drastic filter effects—to smooth overly bright samples, for example.

- *BP (bandpass):* Allows a frequency band directly surrounding the cutoff frequency to pass—all other frequencies are cut.

  The Resonance parameter controls the width of the frequency band. The bandpass filter is a two-pole filter with a slope of 6 dB/octave on each side of the center frequency of the band.

EXS24 mkII output parameters

The EXS24 mkII output parameters define the level—the perceived volume—of a played note. The change in level over time is controlled by an envelope generator, ENV 2.

ENV 2 is hard-wired to the dynamic stage of EXS24 mkII—it is always used to control the level of each note. See EXS24 mkII envelope overview on page 184.

Output parameters

- *Level via Vel slider:* Determines how velocity affects the volume of the sound. The upper slider half determines the volume when the keyboard is struck at maximum velocity, and the lower half determines the volume when the keyboard is struck at minimum velocity. Drag the area between the two slider segments to move both simultaneously.
- *Volume knob:* Sets the output level of EXS24 mkII. Adjust to find the balance between no distortion and getting the best—the highest—resolution in the channel strip and the Level via Vel slider.
- *Key Scale field:* Modulates the sampler instrument level by note number—position on the keyboard. Negative values increase the level of lower notes. Positive values increase the level of higher notes. This is useful when you are emulating a number of acoustic instruments, where higher-pitched notes are often louder than low notes.

EXS24 mkII extended parameters

EXS24 mkII provides additional parameters that can be accessed by clicking the disclosure triangle at the lower left of the interface.

Extended parameters

- *MIDI Mono Mode pop-up menu:* Choose Off, On (with common base channel 1), or On (with common base channel 16).

  In either mode, each voice receives on a different MIDI channel. Controllers and MIDI messages sent on the base channel affect all voices.

- *Mono Mode Pitch Range pop-up menu:* Choose 0, 24, or 48.

  The chosen pitch bend range affects individual note pitch bend messages received on all but the common base channel. The default is 48 semitones, which is compatible with Mobile GarageBand's keyboard in pitch mode. When using a MIDI guitar, 24 semitones is the preferable setting because most guitar to MIDI converters use this range by default.

EXS24 mkII modulation overview

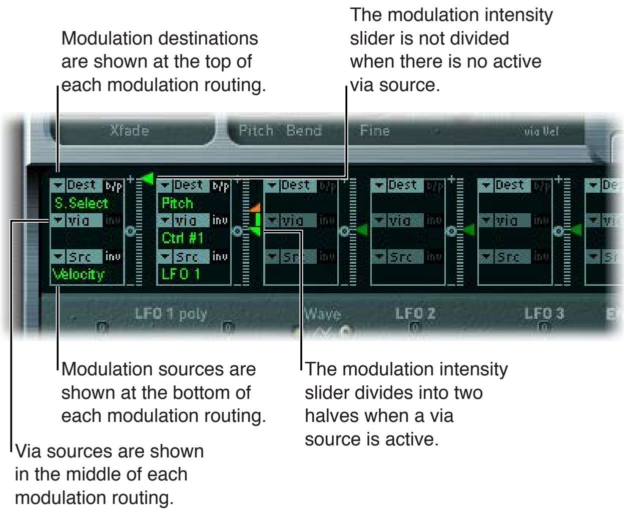
EXS24 mkII is equipped with an extensive range of modulation sources and targets, making it an instrument that can generate extraordinary sounds that constantly evolve, or are just plain expressive to play.

- *Modulation router:* Links modulation sources, such as the envelope, to modulation destinations, such as the filter. The router features ten modulation routings—arranged into columns. See Use the EXS24 mkII modulation router on page 177.
- *Modulation and control parameters:* Include the LFOs and envelopes. See EXS24 mkII LFO overview on page 181 and EXS24 mkII envelope overview on page 184.

EXS24 mkII modulation router

Use the EXS24 mkII modulation router

The modulation router spans the center of the EXS24 mkII interface. If you are new to synthesizer modulation routings, see Modulation overview on page 487.



Any modulation source can be connected to any modulation destination, or target—much like an old-fashioned telephone exchange or a studio patch bay.

The modulation intensity—how strongly the destination is influenced by the source—is set with the vertical slider to the right of the modulation routing.

The intensity of the modulation can itself be modulated: The via parameter defines a further modulation source, which is used to control the modulation intensity. When via is active, you can specify upper and lower limits for the modulation intensity. See Use EXS24 mkII via sources.

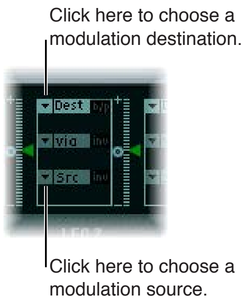
Ten such modulation routings of source, via, and target can take place simultaneously, in addition to routings that are hard-wired outside of the router. It doesn't matter which of the ten modulation routings you use.

You can even select the same destination in several parallel modulation routings. You can also use the same sources and the same via controllers in multiple modulation routings.

The following instructions apply to all ten modulation routings.

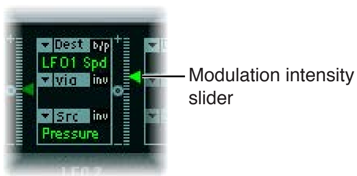
Create a basic modulation routing

1 Choose the parameter you want to modulate from the Dest pop-up menu.



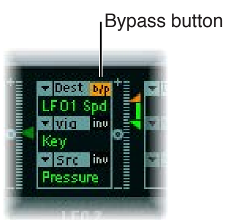
2 Choose the parameter you want to use to modulate the destination from the Src pop-up menu.

3 Drag the intensity slider to set a fixed modulation intensity.



Bypass a modulation routing

■ Click the Bypass (b/p) button at the top right of the modulation routing.



The Bypass button disables individual modulation routings without losing settings.

Use EXS24 mkII via sources

In a basic modulation routing consisting of a target and source, you can set a fixed modulation intensity by vertically dragging the arrowhead of the intensity slider to the right of the routing. The slider value always defines a constant modulation intensity.

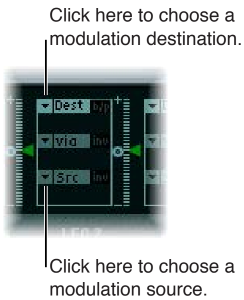
The intensity of the modulation can itself be modulated: via defines a further modulation source, which is used to control the modulation intensity.

If you choose a value other than off (shown as "---") from the via pop-up menu, the intensity slider divides into two halves, each half with its own arrowhead.

- The lower half of the slider defines the minimum modulation intensity, when the via controller—the modulation wheel, for example—is set to its minimum value.
- The upper half of the slider defines the maximum modulation intensity when the via controller is set to its maximum value.
- The area between the two slider halves defines the modulation range of the via controller.

Create a modulation routing that includes a via source

1 Choose the target parameter you want to modulate from the Dest pop-up menu.



2 Choose a modulation source from the Src pop-up menu.

3 Choose the modulation source that you want to use for control of modulation intensity from the via pop-up menu.

4 Vertically drag the upper intensity slider to set the maximum modulation intensity.

5 Vertically drag the lower intensity slider to set the minimum modulation intensity.

Move the entire via range

- Vertically drag the area between the two intensity slider halves (the range).

Both arrowheads move simultaneously.

Tip: If this area is too small to drag, just drag an unused section of the intensity slider to move the area.

Set the modulation intensity to zero

- Click the small 0 symbol that is halfway up the intensity slider control.

Invert the effect of the via modulation source

- Click the Invert (inv) button to invert the effect of the via modulation source.

EXS24 mkI modulation paths

Many of the hard-wired modulation paths that were available as sliders on the original EXS24 (mkI) are now integrated into the modulation router, which was not available in the original EXS24. To reconstitute the modulation slider configuration of the mkI version, choose Recall Default EXS24 mkI Settings from the Options pop-up menu. This loads the mkI modulation paths into the modulation router, as follows:

- Velocity to Sample Select
- LFO 1 to Pitch via ModWheel (= Ctrl#1)
- Velocity to Sample Start (inv)
- LFO 2 to Filter Cutoff via ModWheel
- Velocity to Filter Cutoff
- Envelope 1 to Filter Cutoff via Velocity
- LFO 2 to Pan via ModWheel

You can alter the settings of these modulation paths—allowing you to use modulation sources that were unavailable in EXS24 mkI.

Note: For technical reasons, the settings of the modulation router cannot be used in EXS24 mkI.

EXS24 mkII LFOs

EXS24 mkII LFO overview

EXS24 mkII includes three LFOs (low frequency oscillators). They are all available as sources and targets in the router. If you are new to synthesizers and the concept behind LFOs, see Common modulation sources on page 489.

LFO 1 is polyphonic, which means that if it is used for any modulation of multiple voices, they will *not* be phase-locked. LFO 1 is also key-synced—each time you play a key, LFO 1 modulation of this voice is started from 0.

To understand the non-phase-locked characteristic more fully, imagine a scenario where a chord is played on the keyboard. If LFO 1 is used to modulate pitch, for example, the pitch of one voice might rise, the pitch of another voice might fall, and the pitch of a third voice might reach its minimum value. As you can see, the modulation is independent for each voice, or note.

The key sync feature ensures that the LFO waveform cycle always starts from 0, which results in consistent modulation of each voice. If the LFO waveform cycles were not synchronized in this way, individual note modulations would be uneven.

LFO 1 can also be faded in or out automatically, courtesy of a built-in envelope generator.

LFO 2 is monophonic, which means that the modulation is identical for all voices. For example, imagine that a chord is played on the keyboard. If LFO 2 is used to modulate pitch, the pitch of all voices in the played chord rises and falls synchronously.

LFO 3 is also monophonic, always using a triangular waveform.

All three LFOs can either oscillate freely or be synchronized to the host application tempo in values ranging between 32 bars and 1/128 triplets.

LFO parameters

- *LFO 1 EG knob:* Rotate to set the time it takes for LFO 1 modulation to fade in or fade out (see Use EXS24 mkII's LFO 1 envelope generator on page 183).
- *LFO 1 Rate knob:* Rotate to set the frequency—the speed—of LFO 1 modulation. The value is displayed in Hertz (Hz) or note values beneath the slider. See Set EXS24 mkII's LFO rate.
- *Wave buttons:* Choose the waveform for LFO 1 (left column) and LFO 2 (right column). For details about how to use them, see EXS24 mkII LFO waveforms on page 182.
- *LFO 2 Rate knob:* Rotate to set the frequency—the speed—of LFO 2 modulation.
- *LFO 3 Rate knob:* Rotate to set the frequency—the speed—of LFO 3 modulation.

EXS24 mkII LFO waveforms

You can use the Wave buttons to choose different waveforms for LFO 1 and LFO 2. The table outlines how these waveforms can affect your sounds.

Tip: Try different waveforms while a modulation routing of Pitch is engaged and running.

Waveform	Comments
Triangle	Well-suited for vibrato effects
Sawtooth	Well-suited for helicopter and space gun sounds. Intense modulations of pitch with a negative (inverse) sawtooth wave lead to "bubbling" sounds. Intense sawtooth modulations of lowpass filter cutoff and resonance create rhythmic effects. The waveform can also be inverted, resulting in a different start point for the modulation cycle.
Rectangle	Use of the rectangular waves periodically switches the LFO between two values. The upper rectangular wave switches between a positive value and 0. The lower wave switches between a positive and a negative value set to the same amount above or below 0. You can achieve an interesting effect by modulating the Pitch destination with a suitable modulation intensity that leads to an interval of a fifth. Choose the upper rectangular wave to do so.
Sample & Hold	The two lower waveform settings of the LFOs output random values. A random value is selected at regular intervals, as defined by the LFO rate. The upper waveform steps between randomized values (rapid switches between values). At its lower setting, the random wave is smoothed out, resulting in fluid changes to values. The term *Sample & Hold* (S & H) refers to the procedure of taking samples from a noise signal at regular intervals. The values of these samples are then *held* until the next *sample* is taken. *Tip:* A random modulation of Pitch leads to an effect commonly referred to as a *random pitch pattern generator* or *sample and hold*. Try using very high notes, at very high rates and high intensities—you'll recognize this well-known effect from hundreds of science fiction movies.

Use EXS24 mkII's LFO 1 envelope generator

LFO 1 provides a simple envelope generator (EG) that sets the time it takes for the LFO modulation to fade in or fade out. At its center position (click the middle mark) the modulation intensity is static—no fade-in or fade-out occurs.

Set the LFO 1 modulation fade time

- *To fade in the modulation:* Drag the LFO 1 EG knob to a positive value.

  The higher the value, the longer the delay time.

- *To fade out the modulation:* Drag the LFO 1 EG knob to a negative value.

  The lower the value, the shorter the fade-out time.

Set up a delayed vibrato

LFO envelopes are most often used for delayed vibrato; many instrumentalists and singers intonate longer notes this way.

1. Drag the LFO 1 EG knob to a position toward the right (Delay), and modulate the Pitch destination using the LFO1 source in the router.
2. Set a slight modulation intensity.
3. Select an LFO 1 Rate of about 5 Hz.
4. Click the triangular Wave button as the LFO 1 waveform.

   Tip: Chaotic and fast modulations of frequencies (choose Pitch from the Destination pop-up menu) by the LFO 1 source—with a delayed Sample & Hold waveform, a high Rate, and short fade-out—are ideal for emulating the attack phase of brass instruments.

Set EXS24 mkII's LFO rate

LFO 2 is ideally suited for creating rhythmic modulation effects—effects that retain perfect synchronicity even during project tempo changes. LFO 3 is much the same but uses a fixed triangle waveform, making it most suitable for adding vibrato to a sound, or for use as a modulation source for the other LFOs.

The Rate parameter of all three LFOs allows the respective LFO to run freely (in the right side of the Rate knob range), or to be synchronized with the project tempo (in the left side of the Rate knob range).

Set a free LFO rate

- To run the LFO freely, choose a value in the right side of the LFO Rate knob range.

  The rate is displayed in hertz.

Synchronize the LFO rate to the host application tempo

- To synchronize the LFO with the host application tempo, choose a value in the left side of the LFO Rate knob range.

  The rate is displayed in rhythmic values (when project tempo synchronization is active).

  Synchronized rates range from speeds of 1/64-notes to a periodic duration of 32 bars. Triplet and punctuated values are also available.

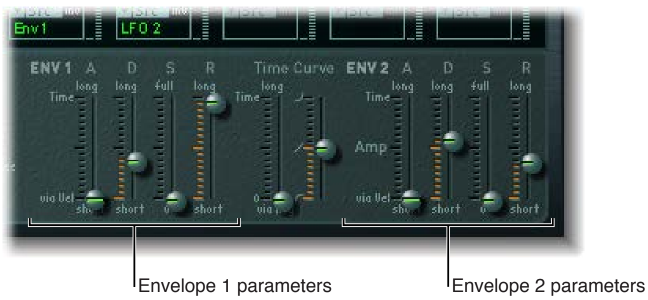
EXS24 mkII envelope overview

EXS24 mkII features two envelope generators per voice, abbreviated as *ENV 1* and *ENV 2* in the interface and router. For details on the roots of the term *envelope generator* and its basic function, see Amplifier envelope overview on page 485.

The parameters of ENV 1 and ENV 2 are identical.

- ENV 1 controls the filter over time.
- ENV 2 defines the changes in level over time—for each note played.

Both envelopes, however, are also available for simultaneous use as sources in the router. The envelope time parameters (Attack, Decay, and Release) are also available as modulation destinations in the router.



Envelope parameters

- *A(ttack) slider:* Defines the time it takes for the level of a note to rise from an amplitude of 0 to the set amplitude. The Attack time sliders of both envelopes are divided into two halves.
  - The lower slider half defines the attack time when the keys are struck hard, at maximum velocity. The upper slider half defines the attack time at minimum velocity. Drag the area between the two slider halves to move them both simultaneously. If this area is too small to drag, drag above or below the slider in the control area.
- *D(ecay) slider:* Defines the time it takes for the level of a held note to fall to the sustain level after the attack phase has completed.
  - If the Sustain level parameter is set to its maximum value, the Decay parameter has no effect.
  - If the Sustain level is set to its minimum value, the Decay parameter defines the duration or fade-out time of the note.
- *S(ustain) slider:* Determines the sustain level.
- *R(elease) slider:* Defines the time required for the (sustain) level to decay to 0—after the key is released.
- *Time via Key slider:* Applies to both envelopes. Scales (lengthens or shortens) the envelope time intervals. Note that position C3 is the center point.
  - Time intervals for zones assigned to keys above C3 can be made shorter with the left slider. Time intervals for zones assigned to keys below C3 can be made longer.
- *Curve slider:* The attack curve slider applies to both envelopes, determining the shape of the envelope attack curve.

EXS24 mkII modulation reference

EXS24 mkII modulation target reference

The following targets, or destinations, are available in the Dest pop-up menu for real-time modulation.

Destination	Comments
Sample Select	Modulates the sample (zone) that is played. By default, Sample Select is controlled by velocity—through the default Velocity to Sample Select modulation routing. This means that the received note velocity value determines which of the layered zones (in different velocity ranges) is heard as you play the keyboard softer or harder. You aren't limited to using velocity, however, to determine which sample is played. You could assign the modulation wheel source to the Sample Select destination, or use both velocity and the modulation wheel. If you choose a continuous controller such as the modulation wheel, you can step through the velocity layers during playback. If you do this, use the crossfade (Xfade) parameters to create smooth transitions between velocity split points. When you use multiple modulation sources, be aware that these can cause all velocity layers to run simultaneously—using up as many voices as there are layered zones. The CPU usage increases accordingly.
Sample Start	Modulates the sample start time. This allows you to trigger a drum loop partway through, for example.
Glide Time	Modulates the duration of the Glide (portamento) effect. If you modulate Glide, with Velocity selected as the source, the speed of the keystrike determines the time it takes for the played notes to reach the target pitch.
Pitch	Modulates the frequency (pitch) of the loaded sampler instrument. If you select an LFO as the source, this destination leads to siren or vibrato sounds. Select one of the envelope generators with 0 attack, short decay, 0 sustain, and short release as the source for tom and kick drum sounds. Slight envelope modulations can make the amount of detuning change over time, which can be particularly useful for brass sounds.
Filter Drive	Modulates the filter Drive parameter.
Filter Cutoff	Modulates the Cutoff Frequency parameter. See EXS24 mkII filter overview on page 173.
Filter Resonance	Modulates the Resonance parameter of the filter.
Volume	Controls the main output level of EXS24 mkII.

Destination	Comments
Pan	Modulates the panorama position of the sound in the stereo spectrum. Modulating Pan with an LFO results in a stereo tremolo (auto panning). In unison mode, the panorama positions of all voices are spread across the entire stereo spectrum. Nevertheless, you can still modulate pan, moving the positions in parallel.
Relative Volume	Adds or subtracts the specified amount to or from the Volume parameter.
LFO 1 Dcy./Dly (Decay/Delay)	Controls the LFO 1 EG parameter (see Use EXS24 mkII's LFO 1 envelope generator on page 183).
LFO 1 Speed	Modulates the frequency (rate) of LFO 1. You can automatically accelerate or slow down LFO 1's rate by modulating the LFO 1 Speed destination with one of the envelope generators (ENV) or with LFO 2 or LFO 3.
LFO 2 Speed	As above, for LFO 2
LFO 3 Speed	As above, for LFO 3
Env 1 Attack	Modulates the Attack time of the filter envelope.
Env 1 Decay	Modulates the Decay time of the filter envelope.
Env 1 Release	Modulates the Release time of the filter envelope.
Time	Modulates the Time via Key slider position—see Time Curve sliders description in EXS24 mkII envelope overview on page 184.
Env 2 Attack (Amp)	Modulates the Attack time of the second envelope generator.
Env 2 Decay (Amp)	Modulates the Decay time of the second envelope generator. If you select Env 2 Decay as the destination and Velocity as the source, the duration of the decaying note depends on how hard you strike the key. Selecting Key(board) as the source results in higher notes decaying more quickly (or slowly).
Env 2 Release (Amp)	Modulates the Release time of the second envelope generator.
Hold	Modulates the (alternate) controller assigned to the sustain pedal function. See the Hold parameter information in EXS24 mkII global parameters overview on page 168.

EXS24 mkII modulation source reference

The following modulation sources are available in the Src pop-up menu:

Source	Comments
Side Chain	Side Chain modulation uses a side-chain signal as a modulation signal. The side-chain source can be chosen from the Side Chain pop-up menu in the header of the plug-in window. It is fed to the internal envelope follower, which creates a modulation value based on the current side-chain input signal level.
Maximum	Max sets the value of this source to +1 (an internal value that indicates the maximum possible amount for this source). This offers interesting options for controlling the modulation intensity with all possible via values.
ENV 1	Envelope Generator 1 is used as a source.
ENV 2 (Amp)	Envelope Generator 2 is used as a source. Env 2 always controls the level of the overall sound.
LFO 1	LFO 1 is used as a source.
LFO 2	LFO 2 is used as a source.
LFO 3	LFO 3 is used as a source.
Release Velocity	The modulation occurs when you release a key (this requires a keyboard that sends release velocity information).
Pressure	Pressure (also known as *aftertouch*) serves as a modulation source. EXS24 mkII reacts to poly pressure (polyphonic aftertouch). ***Note:*** If you set the destination to Cutoff, the cutoff frequencies rise and fall, depending on how firmly you press a key on your touch-sensitive MIDI keyboard—after the initial keystrike.
Pitch Bend	The pitch bend wheel is used as a modulation source.
Key	Kybd (Keyboard) outputs the keyboard position (the MIDI note number). The center point is C3 (an output value of 0, used internally by EXS24 mkII). An output value of −1 indicates five octaves below the center point. An output value of +1 indicates five octaves above. Modulate the Cutoff destination with the Key source to control the cutoff frequencies of the filter with the keyboard position—as you play up and down the keyboard, the cutoff frequencies change. A modulation intensity of 0.5 proportionately scales cutoff frequencies with keyboard note pitches.
Velocity	Velocity serves as a modulation source.
---	Disables the source.
MIDI Controllers 1–120	The MIDI controller you choose serves as a modulation source. Controllers 7 and 10 are marked *Not available*. Logic Pro uses these controllers for volume and pan automation of channel strips. Controller 11 is marked *Expression*. It has a fixed connection to this function, but it can also be used to control other modulation sources.

EXS24 mkII modulation via source reference

The following sources may be chosen from the Via pop-up menu to control the modulation intensity.

Via source	Comments
Side Chain	Side Chain modulation uses a side-chain signal as a modulation intensity (trigger) signal. You can choose the side-chain source from the Side Chain pop-up menu in the upper-right corner of the plug-in window. It is fed to the internal envelope follower, which creates a modulation value based on the current side-chain input signal level.
Maximum	Sets the value of this source to +1.
ENV 1	Envelope Generator 1 controls the modulation intensity.
ENV 2 (Amp)	Envelope Generator 2 controls the modulation intensity.
LFO 1	The modulation uses the speed and waveform of LFO 1, which controls the modulation intensity.
LFO 2	The modulation uses the speed and waveform of LFO 2, which controls the modulation intensity.
LFO 3	The modulation uses the speed and waveform of LFO 3, which controls the modulation intensity.
Release Velocity	The modulation will be more or less intense depending on how quickly you release the key. ***Note:*** This requires a keyboard that sends release velocity information.
Pressure	If you choose Pressure (also known as *aftertouch*) from the Via pop-up menu, the modulation intensity will be touch sensitive—modulation will be more or less intense depending on how firmly you press the key of your touch-sensitive MIDI keyboard after the initial keystrike.
Pitch Bend	The pitch bend wheel controls the modulation intensity.
Key	Key(board) outputs the keyboard position—the MIDI note number. The center point is C3, an output value of 0. Five octaves below and above, an output value of −1 or +1, respectively, is sent. If you select Pitch as the destination, modulate it with the LFO1 source, and select Key as the via value, the vibrato depth changes, depending on key position. In other words, the vibrato depth will be different for notes higher or lower than the defined Key(board) position.
Velocity	The modulation intensity will be velocity sensitive; modulation will be more or less intense depending on how quickly—that is, how hard—you strike the key.
---	Disables the via source.
MIDI Controllers 1–120	Modulation intensity is determined by the MIDI controller value you choose. Controllers 7 and 10 are marked *Not available*. Logic Pro uses these controllers for volume and pan automation of channel strips. Controller 11 is marked *Expression*. It has a fixed connection to this function, but it can also be used to control other modulation sources.

EXS24 mkII Instrument Editor window

EXS24 mkII Instrument Editor overview

The Instrument Editor is used to create and edit sampler instruments. A sampler instrument consists of zones and groups:

- A zone is a location into which a single sample (an audio file) is loaded from a hard disk. You can edit zone parameters in Zones view.
- Zones can be assigned to groups, which provide parameters that allow you to simultaneously edit all zones in the group. You can define as many groups as required. You can edit group parameters in Groups view.

Note: The Instrument Editor is available only when Advanced Editing is enabled in your host application preferences.

See Create EXS24 mkII instruments on page 192, Create EXS24 mkII zones, Create EXS24 mkII groups, and Zone and group edit overview on page 196.

The Instrument Editor window has two views: Zones view and Groups view. Zones view displays zones and associated parameters in the parameter area. Groups view shows groups and associated parameters. See EXS24 mkII Zones and Groups view.

Important: Due to the level of integration between EXS24 mkII and Logic Pro, several sections cover features that are not available in MainStage.

Open the EXS24 mkII Instrument Editor

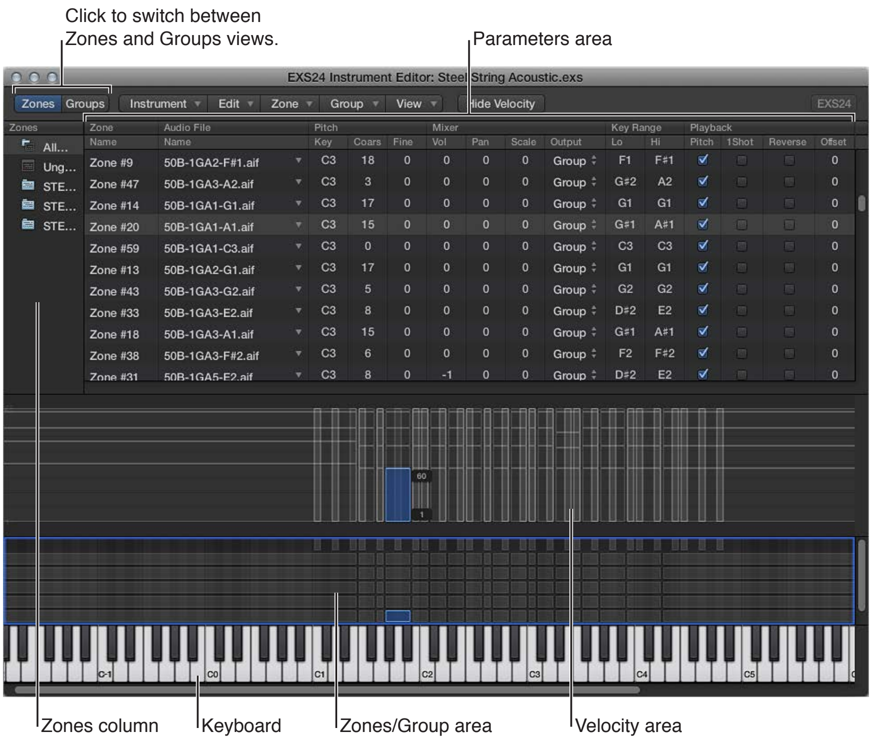
- Click the Edit button in the Parameter window.

Note: If no sampler instrument is loaded when you click the Edit button, a new instrument is automatically created.

EXS24 mkII Zones and Groups view

There are two views in the Instrument Editor window: Zones and Groups.

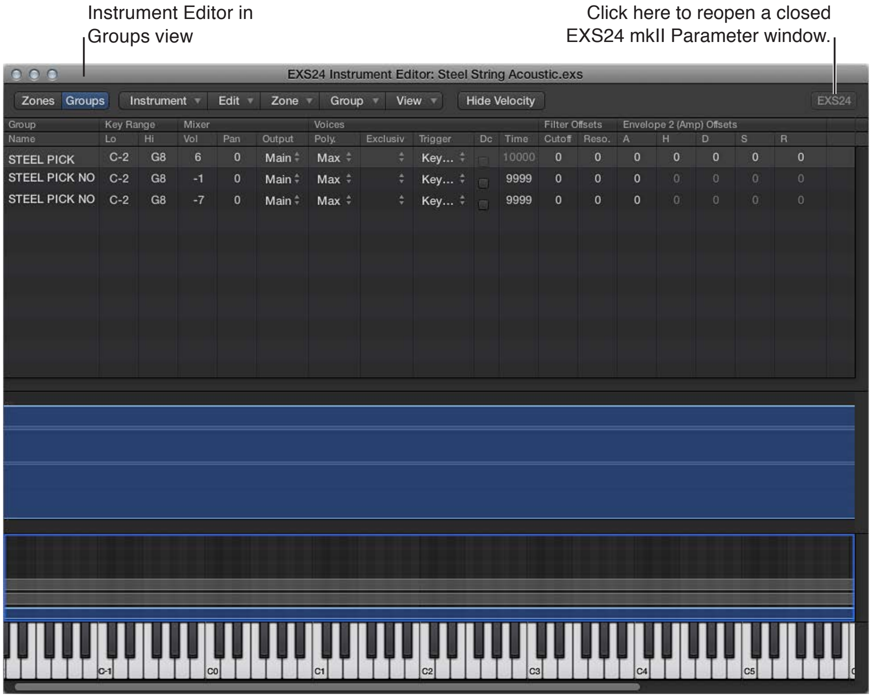
In Zones view, the area above the keyboard displays the Zones area. The general menus, buttons, and so on, are displayed in both Zones and Groups views.



- *Zones column:* Displays all zones of the instrument. By default, every instrument contains All Zones (which includes "grouped" zones) and Ungrouped. Click either All Zones or Ungrouped to display the associated zones in the Parameters area.
- *Parameters area:* Displays the parameters of the zone (individual, all, or ungrouped) chosen in the Zones column.
- *Velocity area:* Shows the velocity range of the selected zone.

  Note: The Velocity area is turned off, by default.
- *Zones/Groups area:* Visually indicates the zones or groups above the keyboard.
- *Keyboard:* Click notes to trigger the associated zone. The keyboard also serves as a visual reference for the placement of zones or groups (in the Zones/Groups area).

In Groups view, the area above the keyboard displays Groups. The general menus, buttons, and so on, are displayed in both Zones and Groups views.



Note: Clicking the EXS24 button does not display the Parameter window in the foreground if it is covered by other floating windows.

Switch between EXS24 mkII Instrument Editor views

- Click the Groups button in the upper-left corner to switch to Groups view, or click the Zones button in the upper-left corner to switch to Zones view. You can also use the Toggle Zones/Groups View key command to switch between views.

Create instruments, zones, and groups

Create EXS24 mkII instruments

You can add zones and groups to loaded instruments, or you can create a new, empty instrument and fill it with zones and groups.

Important: EXS24 mkII cannot directly record samples as you would with a hardware sampler. You need to record the samples in a suitable application, such as Logic Pro.

Create a new instrument

- *If you are in the Parameter window and no sampler instrument is loaded:* Click the Edit button.
- *If you are in the Instrument Editor window:* Choose Instrument > New.

For information about loading sampler instruments, see Use the Sampler Instruments pop-up menu on page 165. For information about saving, renaming and exporting sampler instruments, see Save, rename, and export EXS24 mkII instruments on page 208.

For information about creating zones and groups, see Create EXS24 mkII zones on page 192 and Create EXS24 mkII groups on page 195.

Create EXS24 mkII zones

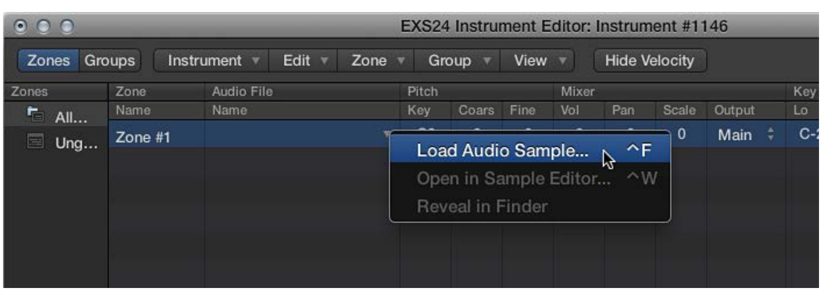
A zone is a location into which a single sample—or audio file, if you prefer this term—can be loaded. The sample loaded into the zone is memory resident—it uses the RAM of your computer. A zone offers parameters that control sample playback. Each zone enables you to determine the Key Range—the range of notes that the sample spans—and the Root Key—the note at which the sample sounds at its original pitch. In addition, Sample Start, End, and Loop points, Volume, and several other parameters can be adjusted for the zone. You can define as many zones as needed.

Create a zone, and assign a sample to it

1. Choose Zone > New Zone (or use the New Zone key command).

   A new zone entry appears in the Instrument Editor.

2. Do one of the following:
   - Double-click the empty area in the Audio File column.
   - Click the arrow in the Audio File column, then choose Load Audio Sample from the pop-up menu.



3 Locate the audio file you want and select it.

- If you select the "Hide used audio files" checkbox, files used in the currently loaded sampler instrument are dimmed.
- If you select the "Preview audio file in EXS Instrument" checkbox, the sample files in the currently selected zone are temporarily replaced. The zone is not directly triggered by selecting this option, but it can be triggered by playing MIDI notes while the file selector is open—and different files are chosen. The selected sample can be heard as part of the zone, inclusive of all synthesizer processing (filters, modulation, and so on).

4 To loop playback of the currently selected sample file, click the Play button.

- Click the Play button a second time to stop playback.
- Click the Play button, then step through the files using the Down Arrow key, or by clicking them, to audition each file in turn.

5 When you find a sample you want to use, click the Open button to add it to the zone.

When the sample is loaded, the sample's name is displayed in the Audio File Name field.

Create a zone by dragging an audio file to a key

- Drag an audio file onto one of the keys of the onscreen keyboard.

The start key, end key, and root key are all set to the note that the file was dragged to. Drag-and-drop works for audio files from the following sources: Browser, Project Audio Browser, and the Finder.

Create a zone by dragging an audio file to a range of keys

- Drag an audio file directly into the zone area to create a new zone.

The root key for the zone is the key at which the sample is played at its recorded pitch. This information is written in the sample header. If no root key is defined in the sample header, the C3 key is used by default.

Note: If you drag an audio file onto an existing zone, the file referenced by that zone is replaced with the new, dropped file. The pointer changes to reflect the replace mode.

Dragging a single sample into the empty area below Ungrouped Zones creates a default zone and a default group, with the new default zone placed in the default group.

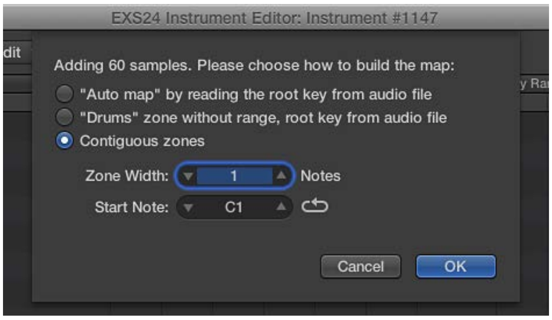
Create multiple zones in one operation

1 Choose Zone > Load Multiple Samples in the Instrument Editor (or use the Load Multiple Samples key command).

2 Browse to the source files location, then use the Add or Add All buttons to select the samples you want to use.

3 Click Done.

4 Choose one of the three automatic mapping modes in the dialog that appears:



- *"Auto map" by reading the root key from audio file:* Uses the root keys stored in the audio file headers, and places the samples, as zones, across the keyboard range. The number of keys that constitute a zone is intelligently determined by the placement of neighboring zones.
- *"Drums" zone without range, root key from audio file:* Uses the root keys stored in the audio file headers. Each zone is mapped to a single key on the keyboard—as determined by the root key information.
- *Contiguous zones:* Ignores all root key information and maps the samples across the keyboard in chromatic order. The Zone Width field allows you to specify the width—the key range—of the newly generated zones. The Start Note field defines the start note of newly generated zones.

You can also load multiple samples by dragging them into the Instrument Editor. Dragging multiple samples into a group folder assigns the samples to the respective group. Dragging multiple samples below the Ungrouped Zones area assigns the audio files to a new default group.

Note: If you drag multiple files onto one of the keyboard keys, the dialog does not include the Start Note field, because the start key, end key, and root key are all set to the note that the file was dropped on.

Create EXS24 mkII groups

Imagine a drum kit with a number of different samples used in several zones, mapped across the keyboard. You might decide that you want to change the parameters of each sample independently—to alter the decay of the snare, or to use a different cutoff setting for the hi-hat samples, for example.

In this context, EXS24 mkII's groups feature is useful. Groups allow for flexible sample organization. You can define as many groups as you need and assign each zone to one of these groups. In a drum kit, for example, you could assign all kick drums to Group 1, all snares to Group 2, all hi-hats to Group 3, and so on.

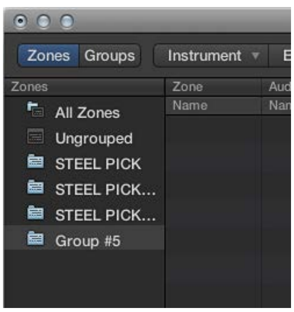
A group makes it possible to define a velocity range for all assigned zones, allowing you to specify a velocity window in which the grouped zones will be triggered, for example. Each group also features offset parameters for the amplitude envelope and filter settings made in the Parameter window.

You can also play all zones without defining and assigning any groups—in this case, changes to parameter settings affect all samples (in all zones) equally.

Create a new group

- Choose Group > New Group in the Instrument Editor.

A new group appears in the Zones column at the left side of the Instrument Editor.



Assign a zone to a group

Do one of the following:

- Select the group in the zone's Group pop-up menu.
- Select a zone in the Instrument Editor, Finder, or Project Browser—then drag it into a group displayed in the Zones column.
- Drag an ungrouped zone (or multiple selected zones) into the empty area below the Ungrouped Zones icon. This creates a new group, containing the dragged zone, or zones.
- Drag a zone (or multiple selected zones) from one group to any of the following:
  - Another group, if you want to change the previous group assignment to the new group.
  - The Ungrouped Zones icon, if you want to change the previous group assignment to unassigned—ungrouped.
  - The empty area below the Ungrouped Zones icon, if you want to create a new group containing the dragged zone, or zones.

Tip: If you press Option while dragging zones to another group, you will copy—rather than move—the selected zones.

Delete all groups that do not have a zone assignment

- Choose Group > Delete Unused Groups in the Instrument Editor.

Remap pitch bend and modulation wheel events

To create realistic-sounding performances in an easy and intuitive way, the Jam Pack 4 (Symphony Orchestra) instruments use the modulation wheel to switch between articulations—legato, staccato, and so on. The pitch bend wheel is used to change expression—crescendo, diminuendo, and so on. For more information, see the Jam Pack 4 documentation.

This is achieved by internally remapping pitch bend events to MIDI controller 11 and modulation wheel events to MIDI controller 4. To ensure compatibility with the Jam Pack 4 instruments, EXS24 mkII automatically uses this remapping behavior for Jam Pack 4 instruments.

Remap pitch bend and modulation wheel events in EXS24 mkII

- To use this remapping model for other instruments, choose Map Mod & Pitch Wheel to Ctrl 4 & 11 from the Instrument menu.

EXS24 mkII remaps incoming pitch bend and mod wheel events to controller 11 or controller 4, respectively.

Note: The default pitch bend and modulation wheel functions cannot be used in this mode.

Edit EXS24 mkII zones and groups

Zone and group edit overview

Zones and groups offer unique parameters that enable you to customize your sampler instrument.

You can use the zone parameters to edit the pitch, velocity range, panorama, looping parameters, and other aspects of zones.

You can use the group parameters to adjust the velocity and output and to offset envelopes and filters for a group of zones, for example.

Editing techniques, menu selection commands, and other parameter interactions that are shared by zones and groups are discussed in these sections:

- Zone and group Edit menu commands on page 202
- Select a zone or group for editing on page 197
- Show or hide zone and group parameters on page 200
- Graphically edit EXS24 mkII zones and groups on page 202

For information about parameters that differ between zones and groups, see EXS24 mkII zone parameters on page 204 and EXS24 mkII group parameters on page 207.

Note: Click the EXS24 button in the top-right corner of the Instrument Editor window to reopen a closed Parameter window and bring it to the foreground. This button is dimmed when the Parameter window is open.

Select a zone or group for editing

There are a number of ways you can select zones and groups for editing.

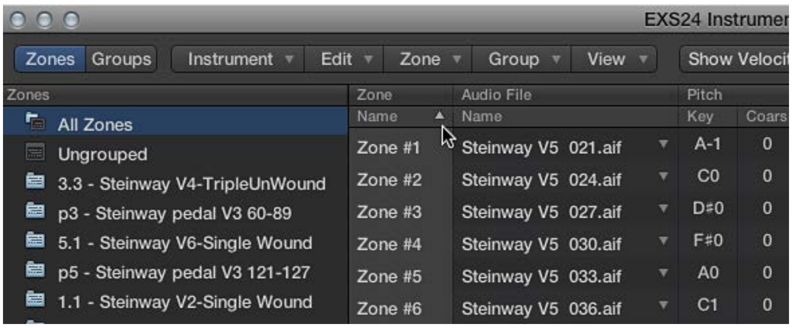
To make selection easier, you can first use the sort options.

Sort zones or groups

You can easily sort zones and groups in the Instrument Editor. For example, if you want to sort your zones by name, click the Name subcolumn heading in the Zone column and your zones are sorted alphabetically. If you want to sort groups by lowest to highest starting velocity, click the Low subcolumn heading in the Velocity Range column. The group with the lowest starting velocity range is displayed at the top of the list.

Do one of the following:

- Click the subcolumn heading that you want to sort by.
- Click the triangle to invert, or reverse, the sort order.



Select zones or groups using Edit menu commands

- Choose one of the following zone and group selection commands from the Edit menu:
  - *Select All:* Selects all zones and groups of the loaded sampler instrument.
  - *Toggle Selection:* Switches the selection between the currently selected zones or groups and all currently unselected zones or groups.

Select zones or groups by clicking them or pressing arrow keys

- Click zones and groups in the Parameters area to select them.
  - *To select a single zone or group:* Click the parameters of that zone or group.
  - *To select two nonadjacent zones and the zones between them:* Shift-click the two nonadjacent zones.
  - *To select multiple nonadjacent zones:* Command-click each zone.
- Press the Up Arrow key or the Down Arrow key to select the previous or next zone or group.

Switch EXS24 mkII zones using a MIDI keyboard

- Choose Select Zone of Last Played Key from the Zone menu to switch between zones by pressing a key on a connected MIDI keyboard. You can continue to select zones by clicking them in the Instrument Editor when this feature is enabled.

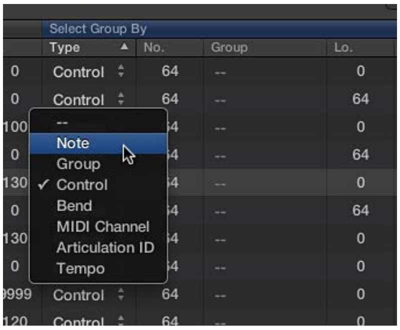
Switch EXS24 mkII groups using a MIDI keyboard

- Choose Select Group of Last Played Key from the Group menu to switch between groups by pressing a key on a connected MIDI keyboard. This is useful when you want to adjust the velocity of an instrument's groups, for example.

Make advanced EXS24 mkII group selections

You can define a specific event to use as a group selection switch. Whenever the defined selection event is triggered, zones pointing to this group can be played, while other groups selected with a different event are not played. The defined event does not play or alter a sound; it acts only as a group selection switch.

The Select Group By command is available in the View menu when you are in Groups view.



Group selection filters

The following group selection filters are available in the Type column pop-up menu:

- *Note:* Used to select a group by MIDI note. You need to assign multiple groups to a different MIDI note; if the note is triggered, playback of the other group is stopped.
- *Group:* Used to select a group based on membership of a master group. You need to assign multiple groups to one master group; if any group member within the master group is triggered, playback of the other group or groups is stopped.
- *Control:* Used to select a group based on a specified controller number.
- *Bend:* Used to select a group by bend range. Groups are played only if they fall within the specified bend range.
- *MIDI Channel:* Used to select a group by MIDI channel. Groups are played only if they match the channel.
- *Articulation ID:* Used to select a group based on the articulation ID. This parameter is available as a modulation destination, so you can switch between sample groups with a controller. For example, you could use your keyboard modulation wheel to switch between several hi-hat groups with different opening degrees.
- *Tempo:* Used to select a group by tempo. Groups are played only if they fall within the specified tempo range.

Define a "base" group and switch between groups with MIDI notes

If you want EXS24 mkII to automatically switch between two string sample groups, for example—one for staccato samples and one for legato samples—you could set the Select Group By menu to MIDI notes, and assign a different MIDI note to trigger each group. You can then use a note that is not triggering a sound as a remote group switch.

The following assumes that several groups already exist. See Create EXS24 mkII groups.

1. Click the Groups view button, then choose the Select Group By command from the View menu.

   The Select Group By fields are displayed in the Group view.

2. Choose Note in the pop-up menu shown in the Type column for the first group you want to switch.

   This is the base group. The default note number is C-2, shown in the No. column field.

3. Drag vertically in the No. column field to change the note number of the base group.

   This should be a note that has no assigned zone. When you play this note, this group is selected—all other groups are unselected.

4. Choose Note from the pop-up menu shown in the Type column for the second group you want to switch.

5. Drag vertically in the No. column field to change the note number of the second group.

   This should be a note that has no assigned zone. When you play this note, the second group is selected—all other groups are unselected.

Refine group selection conditions

You can use multiple group selection criteria to refine your group selections. For example, you could specify that only a particular range of specified controller message values switches between different articulations, for example. This could be further refined with a second Select Group By selection with MIDI channel specified as the group selection criteria.

Do one of the following:

- Click the plus sign in the upper-right corner of the Select Group By column.

  Select Group By (2) fields are displayed in the Group view.

- Click the minus sign to remove a Select Group By condition and to broaden the group selection criteria.

Set up a round robin in EXS24 mkII

The term *round robin* is used to describe sample switching when a single key is struck repeatedly. This feature can be particularly useful in live performance or for avoiding abrupt, machine-gun-like effects when switching between real instrument samples.

In EXS24 mkII, you can use a group as the selection criteria for the Select Group By condition. When one group is played, other groups are silent.

1. Click the Sampler Instruments pop-up menu and load an instrument.
2. Click the Edit button to open the Instrument Editor window. Click the Zones button if Groups view is shown.
3. Assign two zones to the same key (note C2, for example) and play the onscreen keyboard or play C2 on a connected MIDI keyboard.

   You can see and hear two samples (zones) being played. For clarity, these are zone 1 and zone 2.
4. Now assign another zone to another note (C3, for example) on the onscreen keyboard.

   Play C3 on a connected MIDI keyboard and you will hear a different zone being triggered. This is zone 3.
5. Drag vertically in the Lo column for zone 3 until C2 is shown.
6. If you repeatedly strike C2, you will hear zones 1, 2, and 3 being played.

   Repeat steps 4 and 5 for other notes, if required.
7. Click the Groups button to switch to Groups view.
8. Choose Group: Group Select By from the View menu.
9. Click the Type subcolumn on the row of zone 2, then choose Group from the pop-up menu.
10. Click the Group subcolumn on the zone 2 row and choose zone 1.
11. Repeat steps 9 and 10 on the zone 3 row, but choose zone 2 in the Group column.
12. Repeatedly click note C2 on the onscreen keyboard or play C2 on a connected MIDI keyboard.

    You can hear zones 1, 2, and 3 switched one after the other.

This example illustrates the use of groups as the selection or switching criteria. The selection of zone 1, zone 2, and zone 3 in the Group menu enables you specify the trigger group for another group. Because these trigger groups are unique in the Group column, you will sequentially step through groups.

Show or hide zone and group parameters

Use the View menu to determine which zone and group parameters are shown in the Instrument Editor:

- *View All:* Shows all available columns and subcolumns.
- *Individual Zone and Group display settings:* Choose the individual columns and subcolumns you want to display. The zone entries are available in Zones view. The group entries are available in Groups view.

  Tip: Press Option, then choose a disabled zone or group column to limit the display to the chosen column.
- *Restore to Default:* Reverts to the default view.
- *Save as Default:* Saves the current view of zone and group parameters as the default view whenever you open the Instrument Editor.

Use zone and group output labels

You can name the outputs used by instrument zones or groups and can save these names as a label set.

Set output labels for zones and groups

1. To open the Output Labels window, choose Instrument > Output Labels.
2. Click the field in the Label column to the right of the Output that you want to rename and type the new name.

   Press Enter or click outside the field to complete text entry.
3. Select the new name from the Output subcolumn for the zone or group.

Save a label set

1. Choose Save Label Set As from the pop-up menu in the Output Labels window.
2. Type a name in the Label Set Name field, then click Save.

   The label set name is shown in the pop-up menu in the Output Labels window.

Load a label set

- Choose the label set name from the pop-up menu in the Output Labels window.

Reset a label set

- Choose Reset Label Set As from the pop-up menu in the Output Labels window.

  All changes that you have made to label names are reset.

Delete a label set

1. Choose the label set name from the pop-up menu in the Output Labels window.
2. Choose Delete "label set name" from the pop-up menu in the Output Labels window.

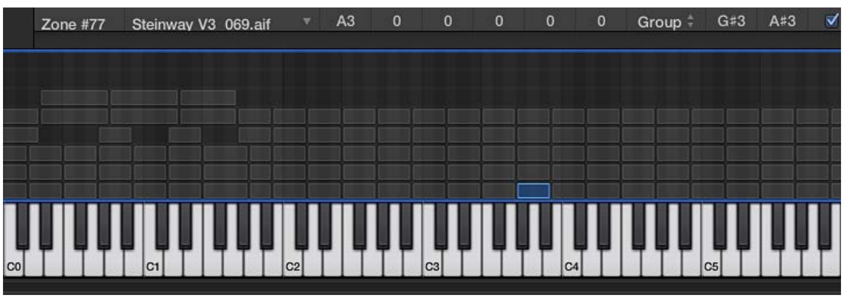
Zone and group Edit menu commands

Use the Edit menu for all basic sampler instrument editing operations, such as copying zones, undoing edit operations, and so on.

- *Undo:* Allows the most recent change to the sampler instrument to be undone.
- *Redo:* Undoes the last Undo command.
- *Cut, Copy, Paste:* The standard commands for cutting, copying, and pasting values. You can also cut, copy, and paste selected zones and groups.
  - When you copy groups in Zones view, the selected groups and their associated zones are copied. The group assignments of the zones are retained.
  - When you copy groups in Groups view, only the groups themselves are copied, not the associated zones.
- *Delete:* Deletes the currently selected zone or group.

Graphically edit EXS24 mkII zones and groups

You are not limited to editing zones and groups in the Parameters area. You can also graphically edit a number of zone and group parameters in the Zones/Groups Display area above the keyboard. If you want to edit the audio file of a zone, see Edit samples in the Logic Pro Audio File Editor on page 209.

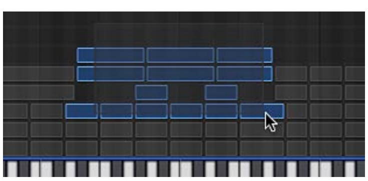


Move a zone or group

- Drag the zone or group to the target position.

Move multiple zones or groups

- Shift-click or drag to select the zones or groups, and drag them to the target position.



Change the root key when moving a zone

- Hold down Command-Option while dragging the zone.

Change the start or end note of a zone or group

1. Move the pointer to the beginning or end of a zone or group (the pointer changes to the resize icon).

2. Drag the start or end point of the zone or group to the target position.

Move a zone to the left or right

- Use one of the following key commands:
  - *Shift Selected Zone(s)/Group(s) Left:* Option–Left Arrow key
  - *Shift Selected Zone(s)/Group(s) Right:* Option–Right Arrow key

Shift both the root note and the zone position

- Use one of the following key commands:
  - *Shift Selected Zone(s)/Group(s) Left (Zones incl. Root Key):* Shift–Option–Left Arrow key
  - *Shift Selected Zone(s)/Group(s) Right (Zones incl. Root Key:* Shift–Option–Right Arrow key

Edit the velocity range of a zone or group

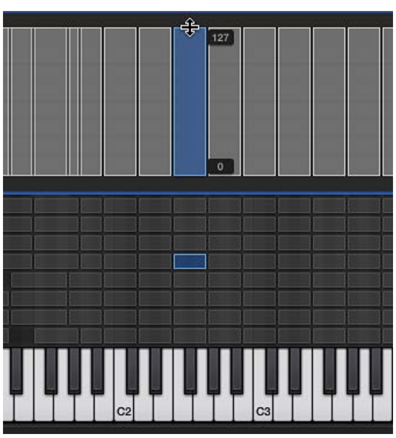
1. Click the Show Velocity button at the top of the Instrument Editor (or use the Show/Hide Velocity key command).

The Velocity Display area opens above the Zones/Views Display area.

2. Click one or more zones or groups in the Velocity Display area.

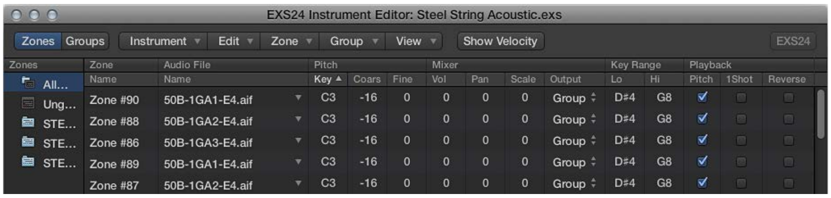
The velocity bars of the selected zones/groups are highlighted in the Velocity Display area.

3. Move the pointer to either the High or Low value of the velocity bar that you want to change (the pointer changes to the resize icon).
4. Drag upward to raise the value, or downward to lower the value.



EXS24 mkII zone parameters

The zone parameters provide extensive control over each zone, or sample, in your sampler instrument.



Zone parameters

- *Zone Name field:* Displays the zone name. New zones are automatically assigned a consecutive number. Click a zone number to enter a name.
- *Audio File menu:* Displays the audio filename. Move the pointer over a name to reveal a help tag with additional information, such as format, bit depth, sample rate, and so on. To display the full file path in the help tag, press Command before the help tag appears. Click the arrow to open a shortcut menu that contains the following commands:
  - *Load Audio Sample:* Opens a dialog where you can select an audio file. Default key command: Control-F.
  - *Open in Audio File Editor:* Opens the selected sample in the Logic Pro Audio File Editor (or the sample editor chosen in the Open External Sample Editor preference). Default key command: Control-W.

  Note: This command is available only when Destructive audio editing is enabled in your host application preferences.

  - *Reveal in Finder:* Shows the full path of the loaded audio file in the Finder.

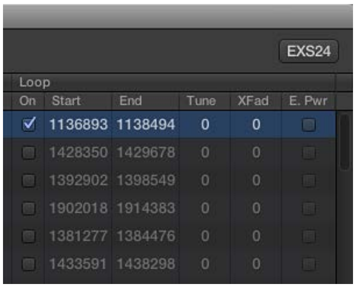
  Tip: Double-click the name of a sample in the Audio File > Name column to open the audio file in the Logic Pro Audio File Editor. When no audio file is loaded, the audio file selector opens.
- *Pitch fields:* Key determines the root note of the sample—in other words, the note at which the sample will sound at its original pitch. Use the Coarse and Fine fields to tune the sample in semitone/cent increments.
- *Volume field:* Adjusts the overall output level of the zone.
- *Pan field:* Adjusts the pan position of the zone. This parameter works only when EXS24 mkII is used in stereo.
- *Scale field:* Balances the level of a sample across the selected key range. A negative value makes lower notes softer than higher notes; positive values have the opposite effect.
- *Output menu:* Determines the outputs used by the zone. Choices include the main outputs, and paired channels 3 and 4, 5 and 6, 7 and 8, 9 and 10, or individual outputs 11 through 16. This allows individual zones to be routed independently to aux channel strips (in a multi-output EXS24 mkII instance).
- *Key Range fields:* The two key range parameters allow you to define a key range for the zone. Playing notes outside this range does not trigger the sample assigned to this zone.
  - *Lo:* Sets the lowest note for the zone.
  - *Hi:* Sets the highest note for the zone.
- *Pitch checkbox:* Click to change the sample pitch when triggered by different keys. When disabled, the sample is always played at its original pitch, regardless of the note played.

- *1Shot checkbox:* Click to make the zone ignore the length of incoming MIDI note events—resulting in the sample always being played from beginning to end whenever a note-on event is received. This is useful for drum samples, where you often don't want the MIDI note length to affect sample playback. Also see the Fade field parameter below.
- *Reverse checkbox:* Click to play the sample from the end to the beginning.
- *Group menu:* Shows the group assignment of a zone. For more information, see Create EXS24 mkII groups on page 195 and EXS24 mkII group parameters on page 207.
- *Velocity Range checkbox and fields:* Click the checkbox, then define a velocity range for the zone. Playing notes outside this velocity range does not trigger the sample assigned to this zone.
  - *Lo:* Sets the lowest velocity that will trigger the zone.
  - *Hi:* Sets the highest velocity that will trigger the zone.
- *Sample Start and End fields:* Set the sample start and end points, respectively. Control-click either field to open a shortcut menu that allows you to open the sample in the Logic Pro Sample Editor (or an external editor), where you can set the start and end points graphically. See Edit samples in the Logic Pro Audio File Editor on page 209, or Use an external instrument editor with EXS24 mkII.
- *Fade field:* Determines the fade-out time for a one-shot sample. The value is shown in samples. The difference between the value specified in this field and the value shown in the End field determines the fade-out duration. The lower the value, the longer it takes for the sample to reach a 0 level (at the sample end point). This option is dimmed when the Loop On checkbox is active. See EXS24 mkII zone loop parameters on page 206.

  Note: This parameter defaults to a value of 0, except when the Sampler Instrument is created with Logic Pro's Audio > Convert Regions to New Sampler Track command. This feature uses transient markers and results in a default Fade field value that matches the slicing offset of the following transient marker.

EXS24 mkII zone loop parameters

EXS24 mkII can loop playback of either an entire sample or a portion of it, when sustained MIDI notes are received.



Zone loop parameters

- *Loop On checkbox:* Click to enable looping and to allow access to the other Loop parameters.
- *Loop Start, Loop End fields:* Define discrete loop start and end points, allowing you to loop a portion of the audio file.

  Control-click either field to open a shortcut menu that allows access to the Logic Pro Audio File Editor (or an external editor). This enables you to set the loop start and end points graphically: Loop Start is represented by the LS marker and Loop End by the LE marker. See Edit samples in the Logic Pro Audio File Editor on page 209.
- *Tune field:* Changes the tuning of the looped portion of the audio file in cent increments.
- *Xfade (Crossfade) field:* Determines the crossfade time between the end/start of a looped sample. In a crossfaded loop, there is no "step" between the loop end and loop start points. The higher the value, the longer the crossfade and the smoother the transition between the loop end and start points. This is especially convenient with samples that are hard to loop and that would normally produce clicks at the transition point—the join in the loop.
- *E. Pwr (Equal Power) checkbox:* Click to enable an exponential crossfade curve that causes a volume boost of 3 dB in the middle of the crossfade range. This fades out/fades in the join between the loop end and start points at an equal volume level.

  Note: The ideal settings for the Xfade and E. Pwr parameters depend on the sample material. A loop that cycles reasonably smoothly is the best starting point for a perfectly crossfaded loop, but a crossfaded loop does not always sound better. Experiment with both parameters to learn how, when, and where they work best.

EXS24 mkII group parameters

Group parameters provide simultaneous control of all assigned zones.

Group parameters

- *Group Name field:* Displays the group name. Click to enter a name.
- *Key Range fields:* Define a key range for the group.
  - *Lo:* Sets the lowest note for the group.
  - *Hi:* Sets the highest note for the group. Playing notes outside this range does not trigger the zones assigned to this group.

  Note: Take your time with these parameters because as they override zone range settings, possibly making some zones inaudible.
- *Vol(ume):* Adjusts the overall level of the group—and, therefore, the volume of all zones in the group. This works much like a subgroup on a mixing console.
- *Pan:* Adjusts the pan position of the group—stereo balance for stereo samples—and the pan position of all assigned zones simultaneously.

  Note: This will affect any individual zone panning adjustments.
- *Output:* Determines the outputs used by the group. Choices include the main outputs and paired channels 3 and 4, 5 and 6, 7 and 8, 9 and 10, or individual outputs 11 through 16. This allows individual groups to be routed independently to aux channels in a multi-output EXS24 mkII instance.

  Note: This will have an impact on any individual zone output assignments.
- *Poly. (polyphony):* Determines the number of voices that the group can play. The Max option ensures that the group uses all voices allowed by the Voices parameter in the Parameter window.

  A practical use of the Poly parameter is to set up a classic "hi-hat mode" within a full drum kit that is mapped across the keyboard. For example, you could assign an open and closed hi-hat sample to a group, and set the Voices parameter of the group to 1. The most recently triggered of the two hi-hat samples mutes the other because only one voice is allowed for the group. This mirrors the real-world behavior of hi-hats. The other sounds of the drum kit can still be played polyphonically, if samples in zones are assigned to another group.
- *Trigger menu:* Determines if zones pointing to this group are triggered on key down (Key Down setting) or on key release (Key Release setting). This is useful for emulating organ key clicks, for example, where you may want the organ note triggered on key down, but the organ click triggered on key release.
- *Dc (decay) checkbox:* Select the checkbox to access the Decay Time parameter.
  - *(Decay) Time field:* Determines the time it takes for the level of a sample (triggered by key release) to decay.

  Note: The Decay parameters function only when the Trigger parameter is set to Key Release.

- *Cutoff and Reso(nance) fields:* Independently offset the Cutoff and Resonance settings for each group. This can be useful if you want the initial impact of a note to be unfiltered for one group but not other groups.
- *Envelope 1/Envelope 2 Offsets fields:* Independently offset the envelope settings in the Parameter window for each group. This is useful if you want the filter (Envelope 1) or volume (Envelope 2) envelopes to affect the samples in a group—after the initial impact of the triggered sounds.
- *H(old) field:* Determines the time that the envelope is held at the maximum attack level, before the decay phase begins.

  Note: When the Trigger parameter is set to Key Release, the Decay Time parameter controls the decay level, rather than Envelope 2 (the volume envelope). This means that when Trigger is set to Key Release, the Envelope 2 Offsets have no effect.
- *Vel(ocity) Range:* Sets a velocity range for the group. Velocity Range is useful for sounds where you want to dynamically mix—or switch between—samples by playing your MIDI keyboard harder or softer. This feature is ideal for layered sounds, such as a piano/string layer, or when switching between different percussion samples.
  - *Lo:* Sets the lowest velocity that triggers the group.
  - *Hi:* Sets the highest velocity that triggers the group. Playing notes outside this velocity range will not trigger the zones assigned to this group.

  Note: The settings made here override zone settings. When a zone's velocity range is larger than the group setting, the zone's velocity range is limited by the group setting.

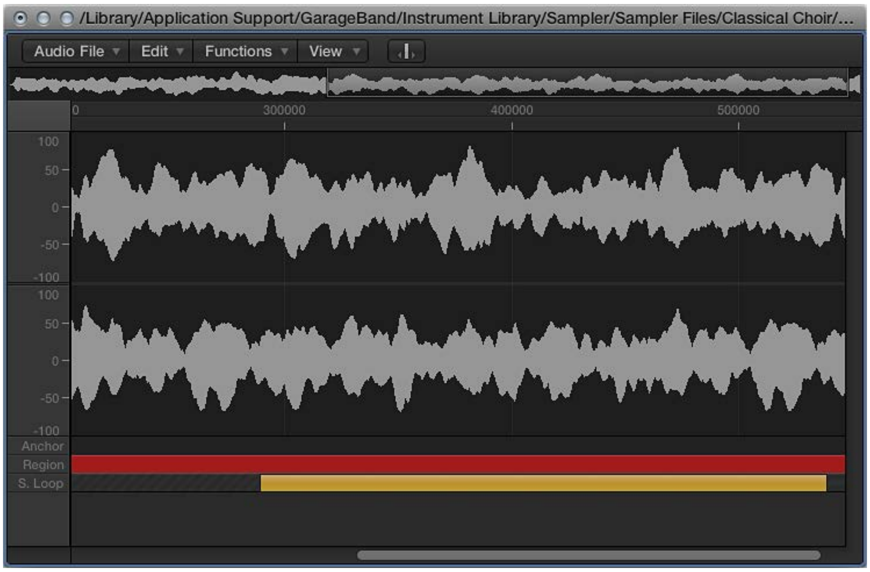
Save, rename, and export EXS24 mkII instruments

You can access all basic sampler instrument file operations using the commands in the Instrument Editor's Instrument menu.

- *Save:* Saves the currently loaded sampler instrument. When you create a new instrument and save it for the first time, you are asked to provide a name. If you have edited an existing sampler instrument and use this command, the existing filename is used and the original instrument is overwritten. You can also use the Save Instrument key command.
- *Save As:* Saves the currently loaded sampler instrument, but you are prompted to provide a different filename. Use this command when you want to save a copy or multiple versions of an edited sampler instrument, rather than overwriting the original version.
- *Rename:* Renames the loaded sampler instrument. The renamed version replaces the original version on the hard disk.
- *Export Sampler Instrument and Sample Files:* Copies the selected sampler instrument, including all associated audio files, to another folder location. Choosing this command opens a dialog where you can browse to an existing folder or create a new folder. You can also use the Export Sampler Instrument and Sample Files key command (default: Control-C).

Edit samples in the Logic Pro Audio File Editor

EXS24 mkII and the Logic Pro Audio File Editor are built to work together. The Logic Pro Audio File Editor provides an intuitive way to adjust sample and loop start and end points by working directly on a visual representation of the waveform.



Open the Logic Pro Audio File Editor

- Control-click either the Loop Start or Loop End parameter fields of the zone you want to edit in the Instrument Editor window.

This opens a shortcut menu, which you can use to open the selected sample in the Logic Pro Audio File Editor (or the external sample editor set in the preferences—see Use an external instrument editor with EXS24 mkII on page 210.

Edit sample borders and loop points

1. After the sample is opened in the Audio File Editor, drag the sample borders and Loop points graphically. Loop Start is represented by the LS marker and Loop End by the LE marker.

   To accelerate your workflow, you can also use the Audio File Editor loop commands:

   - *Sample Loop → Selection:* The loop area—defined by the loop start and end points—is used to select a portion of the overall audio file.
   - *Selection → Sample Loop:* The selected area is used to set the loop start and end points.

2. Choose Edit > Write Sample Loop to Audio File.

   The new loop values are written to the audio file header.

3. Save the sample after you complete your edits.

   The new values written to the audio file header are used by EXS24 mkII.

   Note: Edited samples may have values that are not accurately shown in the Instrument Editor.

Update zone information

After you save and reopen a sample that was edited in either the Logic Pro Audio File Editor or a sample editor not made by Apple, it is likely that either the Start and End, or Loop point values—shown in the Parameters area—will no longer be accurate.

- Choose the Update Selected Zone(s) Info from Audio File command from the Zone menu.

This command reads loop settings and start and end points directly from the audio file and updates the settings of the zone—shown in the Parameters area—accordingly.

Use an external instrument editor with EXS24 mkII

EXS24 mkII allows you to use external applications to edit your sampler instruments.

Open sampler instruments in an external instrument editor

1. Choose Options > Open in external Instrument Editor in the EXS24 mkII Parameter window.
2. Locate and select the instrument editor software.
3. Repeat step 1, but choose the Open in [name of external instrument editor] command. This is the same command as above, but it is automatically renamed after an external instrument editor is assigned.
4. Edit the instrument in the external instrument editor, then use the external instrument editor to send the instrument back to Logic Pro.

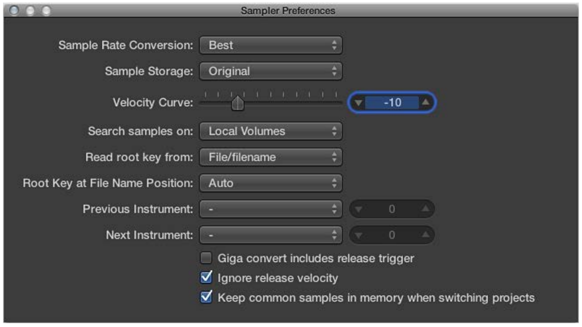
Important: You need to save the edited instrument in Logic Pro—not the external editor—to make any instrument changes permanent.

Reassign the external instrument editor

- Hold down Option while choosing Options > Open in [name of external instrument editor].

EXS24 mkII preferences

The EXS24 mkII Sampler Preferences window provides access to sample-related preferences, such as sample rate conversion quality, velocity response, sample storage, search-related parameters, and so on.



Sampler preferences

- *Sample Rate Conversion pop-up menu:* Determines the interpolation quality used by EXS24 mkII. Choose Best to maintain the highest possible sound quality when transposing.
- *Sample Storage pop-up menu:* Determines the sample format handling method used by EXS24 mkII.
  - *Original:* Loads samples into RAM at their original bit depth. These are converted to the internal 32-bit floating point format of the host application on playback.
  - *32 Bit Float:* Samples are stored and loaded in this format. This eliminates the need for real-time format conversions, meaning that EXS24 mkII handles samples more efficiently and can play back more voices simultaneously.

  Note: This requires twice as much RAM for 16-bit samples and a third more RAM for 24-bit samples.
- *Velocity Curve slider:* Determines how EXS24 mkII responds to incoming velocity values. Negative values increase the responsiveness to soft key strikes, and positive values decrease responsiveness.
- *Search samples on pop-up menu:* Determines the locations that EXS24 mkII will search for instrument samples. You can choose from:
  - *Local Volumes:* Storage media (hard disks and optical mechanisms) attached to or installed in the computer.
  - *External Volumes:* Storage media accessible over a network.
  - *All Volumes:* Both internal and network media are scanned for appropriate data.

  Note: Choosing External Volumes or All Volumes may result in a dramatic increase in the time required by EXS24 mkII to find and load sampler instruments and files.
- *Read root key from pop-up menu:* Sets the method used by EXS24 mkII to determine the root key or velocity (or both) of loaded audio files. You can choose from the following:
  - *File/filename:* Initially reads information about the root key/velocity from the audio file itself (in the header of the AIFF or WAV file) when loading it into a zone. If no information of this type exists in the file header, a smart analysis of the filename may detect a root key/velocity. If this second method doesn't provide useful results, C3 is used as the default root key in the zone.
  - *Filename/file:* As above, but the filename is read before the header.

- *Filename only:* Reads from the filename only. If no root key/velocity information exists, C3 is automatically assigned to the zone as the root key.
- *File only:* Reads from the file header only. If no root key/velocity information exists, C3 is automatically assigned to the zone as the root key.
- *File/Filename/Analysis:* Initially reads information about root key/velocity from the audio file itself (in the header of the AIFF or WAV file). If no information of this type exists in the file header, a smart analysis of the filename may detect a root key/velocity. If this method doesn't provide useful results, the initial transient in the file itself is analyzed for velocity and is applied as the velocity. The root key is derived from the initial pitch. C3 is automatically assigned to the zone as the root key if no pitch is detected.
- *Filename/File/Analysis:* As above, but the filename is read before the header.
- *Analysis only:* Analyzes the file only and applies a velocity based on the initial transient level. The root key is derived from the initial pitch. C3 is automatically assigned to the zone as the root key if no pitch is detected.

- *Root Key at File Name Position pop-up menu:* Normally, EXS24 mkII intelligently determines the root key from the file header of the loaded audio file. On occasion you may want manual control over this parameter, if you feel that the root key is not being properly determined.
  - *Auto:* Provides a smart analysis of numbers and keys from the filename. A number in the filename can be recognized, regardless of its format—60 or 060 are both valid. Other valid numbers can range between 21 and 127. Numerical values outside of these are generally just version numbers. A key number is also a valid possibility for this use—C3, C 3, C_3, A-1, A-1 or #C3, C#3, for example. The possible range is C-2 up to G8.
  - *Numeric value:* There may be cases where a sound designer has used multiple numbers in a filename, which is common with loops, with one value being used to indicate tempo—"loop60-100.wav," for example. In this situation, it isn't clear which, if either, of the numbers indicates a root key or something else: 60 or 100 could indicate the file number in a collection, tempo, root key, and so on. You can set a value of 8 to read the root key at position (letter/character) eight of the filename—namely, the 100 (E6). Alternatively, setting a value of 5 selects the 60 (C3) as the root key position.
- *Velocity at File Name Position pop-up menu:* EXS24 mkII can determine the velocity from the file header of the loaded audio file. On occasion you may want manual control over this parameter, if you feel that the velocity is not being properly determined.
  - *Auto:* Provides a smart analysis of velocity from the filename. An abbreviation in the filename can be recognized—pp or ff, for example.
  - *Numeric value:* There may be cases where a sound designer has used multiple letters and numbers in a filename, which is common with loops, with one value being used to indicate tempo—"loopff-pp.wav," for example. In this situation, it isn't clear which, if either, of the ff/pp values indicates a velocity or something else: ff or pp could be an abbreviation of "fast funk" or "power pop," for example. You can set a value of 8 to read the velocity at position (letter/character) eight of the filename—namely, the pp. Alternatively, setting a value of 5 selects the ff as the velocity indicator.

- *Previous Instrument and Next Instrument:* Determine which MIDI event type and data value are used for selection of the previous or next instrument.
  - Choose the MIDI event type from the Previous Instrument and Next Instrument pop-up menus. Choices include Note, Poly Pressure, Control Change, Program Change, Channel Pressure, and Pitch Bend. In the field next to each pop-up menu, you can enter either the note number or the value of the first data byte. If you choose Control Change, the number field determines the controller number.

  Important: These commands are unique to EXS24 mkII and are separate from the global Previous/Next Plug-In Setting or EXS Instrument commands. Therefore, you should make sure that you do not assign the same MIDI event for both. If you do this, both commands will execute, which may result in unexpected behavior.
- *Giga convert includes release trigger checkbox:* Determines whether or not the release trigger function of the Gigasampler format is performed by EXS24 mkII.
- *Ignore release velocity checkbox:* Also refers to the release trigger function of the Gigasampler format, and it should always be selected for this purpose. Regardless of whether your keyboard is able to send release velocity, you want your samples played by the release trigger function to be louder or softer than the original sample, or at the same volume, regardless of the initial velocity. When playing with release trigger, you want the release velocity value to have the same value as the initial velocity value. To accomplish this, you can switch off release velocity.
- *Keep common samples in memory when switching projects checkbox:* Determines whether the samples commonly used, or shared, by two open project files are reloaded when you switch between projects.

Open the Sampler Preferences window

Do one of the following:

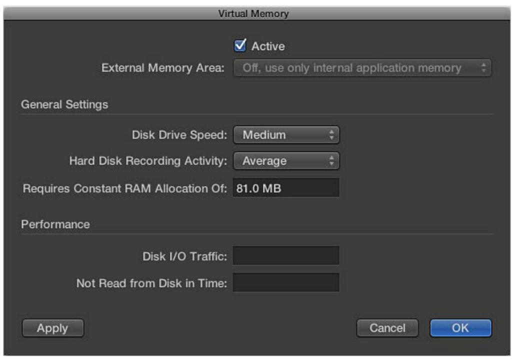
- In the Parameter window, click the Options button, then choose Preferences from the pop-up menu.
- In the Instrument Editor window, choose Edit > Preferences.

EXS24 mkII memory management

Multigigabyte sample libraries are commonplace today, delivering incredibly detailed and accurate instrument sounds. In many cases, these sample libraries are too large to fit into your computer's random-access memory (RAM). To use these huge sampler instruments, EXS24 mkII can use a portion of your hard drive as virtual memory. When you turn on EXS24 mkII's virtual memory, only the initial attacks of audio samples are loaded into the computer's RAM; the rest of the sample is streamed in real time from the hard drive.

Logic Pro automatically addresses all available system memory. The amount of RAM available for use by EXS24 mkII is determined by several factors, including:

- The amount of physical RAM installed in your computer.
- How much RAM other open applications and the operating system are using.
- How much RAM Logic Pro is using. This will vary in accordance with the number and size of audio files in the project, and other plug-ins used. Sampler plug-ins not made by Apple, in particular, can significantly affect the amount of RAM that Logic Pro uses.



Virtual memory parameters

- *Active checkbox:* Turns on the EXS24 mkII virtual memory feature.

  Note: If you have enough physical RAM to hold all the samples for a project, you will see improved performance by deselecting the Active checkbox. In projects with lots of audio tracks playing and relatively few EXS instances, this may affect performance. If the Active checkbox is deselected and there is insufficient RAM to hold all samples, Logic Pro swaps data to and from the disk, which degrades performance. Deselecting the Active checkbox also increases project load times, so you should leave it selected in most cases.

- *Disk Drive Speed pop-up menu:* Specifies the speed of your hard drive; if you have a solid-state drive, or a 7200-rpm or faster hard drive for your audio samples, choose Fast. If you are using a 5400-rpm laptop drive for your audio samples, choose Medium. You should not need to use the Slow setting with any modern Mac.
- *Hard Disk Recording Activity pop-up menu:* Specifies overall hard disk usage—how much recording and streaming of non-sampler-related audio is occurring. For example, if you are recording an entire drum kit using over a dozen microphones, streaming live guitars and bass, recording choirs, and so on, set your hard disk recording activity to Extensive. If your projects consist mostly of software instrument playback, with perhaps a recorded instrument or vocal or two, set your hard disk recording activity to Less. If you are unsure, choose Average.
- *Requires Constant RAM Allocation Of field:* Shows the memory requirements of the above parameters. The slower your hard drive and the higher your hard disk recording activity, the more RAM you will need to allocate to virtual memory.

- *Performance section:* Shows the current disk I/O traffic and the data not read from disk in time. If these numbers start rising, EXS24 mkII may glitch when trying to stream samples from disk. If you notice these values rising to high levels, change the general settings to free up additional RAM for virtual memory use. If you continue to see high Performance display values and hear audio glitches, consider installing more RAM in your Mac.

Open the Virtual Memory dialog

- In the Parameter window, click the Options button, then choose Virtual Memory from the pop-up menu.

External Instrument

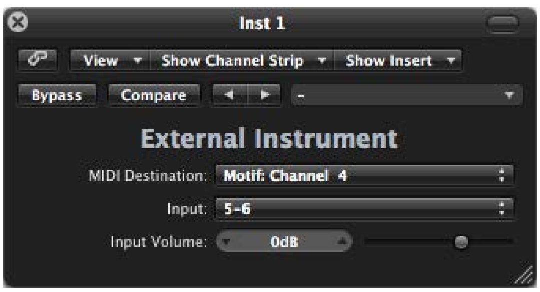
10

External Instrument overview

You can use the External Instrument to route external MIDI sound generators through the Logic Pro Mixer, which you can then process with effects.

You can also use the External Instrument to transmit and receive MIDI information through the instrument channel strip that it is inserted into. This enables you to control an external sound module—both MIDI and audio—from within one element.

To avoid constant repatching of devices, it is best to use an audio interface that supports multiple inputs and outputs. The External Instrument plug-in is inserted into instrument channel strips in place of a software instrument.



External Instrument parameters

- *MIDI Destination pop-up menu:* Choose the target MIDI instrument and channel.
- *Input pop-up menu:* Choose the inputs of your audio hardware that the MIDI sound generator is connected to.
- *Input Volume slider and field:* Move to set the incoming signal level.

Use the External Instrument

The track routed to an instrument channel strip that is being used for an external MIDI sound module behaves just like a standard software instrument track. This enables you to record and play back MIDI regions on it, with the following benefits:

- You can use the sounds and synthesis engine of your MIDI module with no overhead on your computer CPU apart from effects used in the channel strip.
- You can use insert and send effects. To use send effects, route the instrument channel strip to aux channel strips.
- You can bounce external MIDI instrument parts, with or without effects, to an audio file in real time. This makes the creation of a mix, including all internal and external devices and tracks, a one-step process.

Note: Bouncing an External Instrument track cannot happen faster than real time, as is the case with any bounce operation where MIDI hardware is involved.

When you use multitimbral MIDI sound sources, you can gain maximum flexibility by using multiple External Instrument instances. In this situation, connect a separate audio output of the tone generator (if equipped with multiple outputs) to different inputs on your audio interface—each addressed by individual External Instruments.

Process external MIDI instruments with effects

1. Connect the output (or output pair) of your MIDI module with an input (or input pair) on your audio interface.

   Note: These can be either analog or digital connections if your audio interface and MIDI sound generator are equipped with either, or both.

2. Create an instrument channel strip.
3. Click the Instrument slot, then choose External Instrument from the pop-up menu.
4. Choose the MIDI Destination from the pop-up menu in the External Instrument window.
5. Choose the input (of your audio interface) that the MIDI sound generator is connected to from the Input pop-up menu.
6. Adjust the Input Volume, if necessary.
7. Insert any required effects into the Insert slots of the channel strip (or channel strips, if you are using multiple External Instrument instances with a multitimbral sound source).

   You can also route the instrument channel strip to aux channel strips, if you want to use send effects.

Klopfgeist 11

Klopfgeist parameters

Klopfgeist is an instrument that provides a metronome click in Logic Pro. It is inserted into instrument channel strip 256 by default and is used to generate the MIDI metronome click.

Klopfgeist can also be inserted into any other instrument channel strip in Logic Pro for use as an instrument.

Klopfgeist has a number of synthesizer parameters that you can use to quickly create metronome click sounds.

Klopfgeist parameters

- *Trigger Mode buttons:* Click Poly to use Klopfgeist as a four-voice polyphonic instrument. Click Mono to use it as a monophonic instrument.
- *Tune knob and field:* Rotate to tune Klopfgeist in semitone steps.
- *Detune knob and field:* Rotate to fine-tune Klopfgeist in cents (one cent equals 1/100 of a semitone).
- *Tonality slider and field:* Drag to change the sound of Klopfgeist from a short click to a pitched percussion sound—similar to a wood block or claves.

- *Damp slider and field:* Drag to set the release time. The shortest release time is attained when Damp is at its maximum value, 1.00.
- *Level via Vel slider and fields:* Drag to set velocity sensitivity. The top slider sets the volume at maximum velocity. The lower slider sets the volume at minimum velocity. Drag the area between the two slider segments to move both simultaneously.

Retro Synth

12

Retro Synth overview

Retro Synth is a flexible 16-voice synthesizer that can produce a wide variety of sounds.

Retro Synth provides four types of synthesizer engines—Analog, FM, Sync, and Wavetable. Each engine can generate unique sounds that are difficult, or impossible, to achieve with other types of synthesizers.

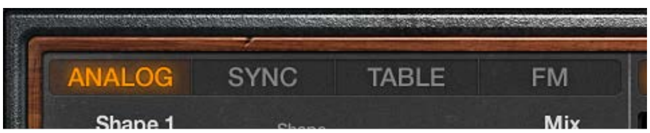
Retro Synth is very easy to use, with many identical controls found in each synthesizer engine.

If you're new to synthesizers, it might be best to start off with Synthesizer basics overview, which will introduce you to the fundamentals and terminology of different synthesis systems.

The first step in creating a new sound is to choose a Retro Synth synthesizer engine. Your choice should be guided by the type of sound you want to generate.

Choose a synthesizer engine

- Click the tab of the synthesizer engine that you want to use:



- *Analog:* Use for classic synthesizer sounds, such as leads, pads, and basses. See Retro Synth Analog oscillator controls.
- *Sync:* Use for aggressive synthesizer sounds, particularly leads and basses. See Retro Synth Sync oscillator controls.
- *Table:* Use for synthesizer and real instrument sounds, or hybrids of these sound types. See Retro Synth Table oscillator controls.
- *FM:* Use for classic digital synthesizer sounds. Of note are bells, electric piano, clavinet, and spiky bass sounds. See Retro Synth FM oscillator controls.

Your choice of synthesizer engine changes the controls available. Most of these changes are seen in the Oscillator section.

Retro Synth Analog oscillator controls

The synthesizer oscillators are used to generate one or more waveforms. You set the basic tonal color with the chosen waveform or waveforms, adjust the pitch of the basic sound, and set the level relationships between oscillators. The signal of one or both oscillators is then sent to other parts of the synthesizer engine for shaping, processing, or manipulation. See Retro Synth filter controls, Retro Synth amp and effect controls, Use Retro Synth modulation, and Retro Synth global and controller settings.

Analog synthesizer sounds are typically attributed with having a warm and rich tone. You can create a wide variety of timbres using this synthesis method, notably string and pad sounds, synthetic brass, bass, and percussion.

Analog oscillator parameters

- *Shape 1 and Shape 2 knobs:* Rotate to choose the type of waveform that each oscillator generates. The waveform is responsible for the basic tonal color.
  - *Analog engine:* The oscillators output a number of standard waveforms—noise, rectangular, sawtooth, and triangular.
  - *Table engine:* You can choose from 100 waveforms, known as Digiwaves. Digiwaves are very short samples of the attack transients of various sounds and instruments.
- *Shape Modulation knob:* Rotate to choose a waveform shape modulation source, and to set the modulation intensity. The centered (off) position disables all waveform shape modulation with the LFO or filter envelope.
- *Vibrato knob:* Rotate to set the amount of vibrato (pitch modulation).
- *Semitones knob:* Rotate to set the pitch of oscillator 2—in semitone steps, over a range of ±2 octaves.
- *Cents knob:* Rotate to precisely adjust the frequency of oscillator 2 in cents (1 cent = 1/100th semitone).
- *Mix slider:* Move to crossfade (set the level relationships) between the oscillators (Shape 1 and 2).

Retro Synth Sync oscillator controls

The synthesizer oscillators are used to generate one or more waveforms. You set the basic tonal color with the chosen waveform or waveforms, adjust the pitch of the basic sound, and set the level relationships between oscillators. The signal of one or both oscillators is then sent to other parts of the synthesizer engine for shaping, processing, or manipulation. See Retro Synth filter controls, Retro Synth amp and effect controls, Use Retro Synth modulation, and Retro Synth global and controller settings.

Oscillator sync sounds are typically more aggressive than sounds generated with other synthesis methods, making this engine most suitable for lead and hard bass instruments. The second oscillator is resynchronized with the first oscillator each time a note is triggered or each time the waveform cycle of the first oscillator restarts.

Sync oscillator parameters

- *Shape 1 and Shape 2 knobs:* Rotate to choose the type of waveform that each oscillator generates. The waveform is responsible for the basic tonal color. The oscillators output a number of standard waveforms—noise, rectangular, sawtooth, and triangular.
- *Sync Modulation knob:* Rotate to choose a sync modulation source, and to set the modulation intensity. The centered (off) position disables all waveform modulation with the LFO or filter envelope.
- *Vibrato knob:* Rotate to set the amount of vibrato (pitch modulation).
- *Sync knob:* Rotate to set the maximum amount of sync modulation. This makes the sound more or less aggressive. Technically, this control changes the waveform start point of both oscillators.
- *Mix slider:* Move to crossfade (set the level relationships) between the oscillators (Shape 1 and 2).

Retro Synth Table oscillator controls

The synthesizer oscillators are used to generate one or more waveforms. You set the basic tonal color with the chosen waveform or waveforms, adjust the pitch of the basic sound, and set the level relationships between oscillators. The signal of one or both oscillators is then sent to other parts of the synthesizer engine for shaping, processing, or manipulation. See Retro Synth filter controls, Retro Synth amp and effect controls, Use Retro Synth modulation, and Retro Synth global and controller settings.

Wavetable synthesis is useful for creating evolving textures and more clinical sounds. It is well-suited for pad creation, basses, and sound effects.

Table oscillator parameters

- *Shape 1 and Shape 2 knobs:* Rotate to choose the type of waveform that each oscillator generates. The waveform is responsible for the basic tonal color.
  - *Analog engine:* The oscillators output a number of standard waveforms—noise, rectangular, sawtooth, and triangular.
  - *Table engine:* You can choose from 100 waveforms, known as Digiwaves. Digiwaves are very short samples of the attack transients of various sounds and instruments.
- *Shape Modulation knob:* Rotate to choose a waveform shape modulation source, and to set the modulation intensity. The centered (off) position disables all waveform shape modulation with the LFO or filter envelope.
- *Vibrato knob:* Rotate to set the amount of vibrato (pitch modulation).
- *Semitones knob:* Rotate to set the pitch of oscillator 2—in semitone steps, over a range of ±2 octaves.
- *Cents knob:* Rotate to precisely adjust the frequency of oscillator 2 in cents (1 cent = 1/100th semitone).
- *Mix slider:* Move to crossfade (set the level relationships) between the oscillators (Shape 1 and 2).

Retro Synth FM oscillator controls

The synthesizer oscillators are used to generate the basic tonal color. This signal is then sent to other parts of the synthesizer engine for shaping, processing, or manipulation. See Retro Synth filter controls, Retro Synth amp and effect controls, Use Retro Synth modulation, and Retro Synth global and controller settings.

In FM synthesis, the basic sound is generated by setting different tuning ratios between the modulator and carrier oscillators and by altering the FM intensity. The tuning ratio determines the basic overtone structure, and the FM control sets the level of these overtones.

At the core of Retro Synth's FM synthesis engine, you'll find a multiwave modulator oscillator—the (Wave) Shape slider, and a sine wave carrier oscillator—the FM (Amount) slider. The basic sine wave of the carrier oscillator is a pure, characterless tone.

To make things more sonically interesting, the modulator oscillator is used to modulate the frequency of the carrier oscillator. This modulation occurs in the audio range (you can actually hear it), and results in a number of new harmonics becoming audible, thus changing the tonal color.

The pure sine wave (of the carrier oscillator) is combined with the newly generated harmonics, making the sound much more interesting.

You can make fine changes to the tuning ratio of the two oscillators (and therefore the levels of the harmonics) by adjusting the Harmonic and Inharmonic controls.

FM oscillator parameters

- *Vibrato knob:* Rotate to set the amount of vibrato (pitch modulation).
- *Modulation knob:* Rotate to choose an FM amount modulation source, and to set the modulation intensity.
- *FM (Amount) slider:* The carrier waveform is a simple sine wave. Drag to adjust the level of this basic tone.
- *FM/Harmonic switch:* Switch to control the harmonic/inharmonic content of your sound with the LFO or Filter Envelope.
  - The left switch position lets you use the LFO or Filter envelope to modulate the FM (Amount).
  - The middle switch position lets you use the LFO or Filter envelope to modulate the FM (Amount) and Harmonic content at the same time.
  - The middle switch position lets you use the LFO or Filter envelope to modulate the Harmonic content.
- *Harmonic/Inharmonic sliders:* Drag the Harmonic and Inharmonic sliders to precisely change the levels of these sonic elements, and therefore, the tonal color of your sound. Technically, you are changing the tuning ratio between the carrier and modulator oscillators, resulting in harmonic or inharmonic content becoming more or less audible.

  Note: The tuning ratio can change significantly when you adjust the (Wave) Shape slider, so avoid using it if making a subtle alteration to the harmonic or inharmonic content of your sound.
- *(Wave) Shape slider:* Drag to modulate the carrier waveform. This control and the FM slider interact as you adjust either, resulting in a range of tones with more or less harmonic/inharmonic content.
- *Mix slider:* Move to crossfade (set the level relationships) between the modulator and carrier oscillators.

Retro Synth filter controls

Retro Synth features a flexible filter that can operate as a lowpass, highpass, bandpass, band reject, or peak filter. The filter can subtly, or dramatically, affect the basic tone sent from the oscillators.

Filter use is straightforward. Choose a filter type and set a filter slope (if applicable). Adjust the filter cutoff and resonance controls to sculpt the sound. You can also control the filter cutoff and resonance controls while playing—either manually or by using keyboard position, an envelope, or the LFO to modulate these filter controls automatically. Real-time changes to filter cutoff and resonance can make your performance much more dynamic and interesting. You can do this with MIDI keyboard controllers and with the other filter section controls. See Use Retro Synth modulation and Retro Synth global and controller settings.

Filter parameters

- *On/off button:* Turns the filter section on or off. You will normally have the filter enabled (indicated by the lit button at the top left of the filter section). Disable the filter when adjusting other sound controls because this makes it easier to hear changes.
- *Filter Type pop-up menu:* Choose a filter type from the menu. There are four lowpass filters with different slopes, a highpass, bandpass, band reject, and peak filter available.
  - *LP (lowpass):* Allows frequencies that fall below the cutoff frequency to pass. You can choose one of four slopes that change the tonal characteristics of the filter, making it sound brighter, mellow, thinner, or fuller—particularly in the bass end of the sound.
  - *HP (highpass):* Allows frequencies above the cutoff frequency to pass. The slope of the highpass filter is fixed at 12 dB/octave.
  - *BP (bandpass):* The frequency band directly surrounding the cutoff frequency is allowed to pass. All other frequencies are cut. The Resonance control sets the width of the frequency band. Bandpass is a two-pole filter with a slope of 6 dB/octave on each side of the band's center frequency.
  - *BR (band reject):* The frequency band directly surrounding the cutoff frequency is rejected, but frequencies outside the band can pass. The Resonance control sets the width of the rejected frequency band.
  - *Peak:* A peak filter allows the level in a frequency band to be increased. The center of the frequency band is set with the Cutoff control. The width of the band is set with the Resonance control.

- *Cutoff control:* Drag the handle horizontally to set the brilliance of the signal.
  - *In a lowpass filter:* the higher the Cutoff frequency is set, the higher the frequencies of signals that are allowed to pass.
  - *In a highpass filter:* Cutoff sets the point where low frequencies are suppressed.
  - *In a bandpass, band reject, or peak filter:* Cutoff sets the center frequency of the band that is allowed to pass, is suppressed, or is emphasized.
- *Resonance control:* Drag the handle vertically to emphasize or suppress portions of the signal above, below, or surrounding, the defined cutoff frequency.
- *Key (Follow) slider:* Move to determine the effect that keyboard pitch (the note number) has on filter cutoff frequency modulation.

  At the top position, the filter follows keyboard pitch, resulting in a constant relationship between cutoff frequency and pitch. This mirrors the properties of many acoustic instruments where higher notes sound both brighter in tone and higher in pitch. At the bottom position, the cutoff frequency does not change, regardless of which key (pitch) you strike. This makes the lower notes sound relatively brighter than the higher ones.
- *Filter FM knob:* Rotate to set the intensity of filter cutoff frequency modulation with Oscillator 1's sine wave generator. Positions to the left set the strength of static sine wave modulations. Positions to the right set the strength of envelope-controlled sine wave modulations. The centered (off) position disables filter frequency modulation by Oscillator 1's sine wave generator.

  Note: Oscillator 1's sine wave generator always generates a sine signal—at the frequency of Oscillator 1.
- *LFO knob:* Rotate to set the strength of filter cutoff frequency modulation with the LFO. Positions further away from the centered (off) position make modulation more or less intense. See Retro Synth LFO and Vibrato.
- *Filter Env(elope) knob:* Rotate to set the strength of filter cutoff frequency modulation with the Filter Envelope. Positions further away from the centered (off) position make modulation more or less intense. See Retro Synth envelopes.

Retro Synth amp and effect controls

Retro Synth's Amp controls set the overall volume. You can also mix a sine wave directly into the output stage, which thickens the sound. You can make the sound richer with Retro Synth's integrated Chorus effect or add a sweeping, metallic character with the Flanger effect. If you're new to synthesizers and the concepts behind amp controls, see Synthesizer basics overview.

Amp parameters

- *Volume knob:* Rotate to set Retro Synth's overall output level.
- *Sine Level knob:* Rotate to mix a sine wave at the frequency of oscillator 1 (Shape 1) directly into Retro Synth's output stage. This sine signal is not processed by the filter.

Effect parameters

- *On/off button:* Turns the effect section on or off.
- *Effect Type pop-up menu:* Choose either the Chorus or Flanger effect.
  - The Chorus effect is based on a delay line, the output of which is mixed with the original, dry signal. The short delay time is modulated periodically, resulting in pitch deviations. The modulated deviations, in conjunction with the original signal's pitch, produce the chorus effect.
  - The Flanger effect works in a similar fashion to the chorus, but with even shorter delay times. The output signal is fed back into the input of the delay line. This feedback results in the creation of harmonic resonances that cyclically move through the frequency spectrum, resulting in a sweeping, metallic sound.
- *Mix knob:* Rotate to set the balance between the original and effect signals. High values result in stronger effect processing.
- *Rate knob:* Rotate to set the modulation speed.

Tip: Use your host application effects if you need more precise control of chorus and flanging or want to use both effects simultaneously.

Retro Synth modulation controls

Use Retro Synth modulation

Retro Synth's Glide/Autobend, LFOs, and envelopes are known as modulation generators. These modulation sources are used to control modulation targets, such as oscillator pitch or filter cutoff.

The Volume Envelope is dedicated to control of your sound's level over time. The Filter Envelope controls the filter over time. See Retro Synth envelopes.

Retro Synth's LFO is used as a source for multiple modulation targets. The Vibrato modulation source is dedicated to control of oscillator pitch. See Retro Synth LFO and Vibrato.

Glide—also known as *portamento*—and Autobend are hardwired to control oscillator pitch. See Retro Synth Glide and Autobend.

You should make use of all modulation options because they can help you to create expressive performances.

If you're new to synthesizers and the concepts behind modulation generators, such as LFOs and envelopes, see Synthesizer basics overview.

Modulate the oscillator waveform

The Analog, Sync, and Wavetable oscillator waveform shapes can be modulated by following these steps. In the FM synth engine, you can modulate the FM or Harmonic amount (or both).

1. Move the control toward LFO or Filter Env. Positions further away from the centered (off) position make waveform modulation more intense.
2. Adjust the controls of the LFO (click the LFO tab if the Vibrato is visible) and Filter Env sections. See Retro Synth LFO and Vibrato and Retro Synth envelopes.
3. If you are using the LFO, select a waveform.
4. If the (LFO) Sync switch is turned off:
   - Adjust the Via Amount slider to the right of the waveform graphic to set the maximum modulation intensity (the highest LFO speed).
   - Move your MIDI keyboard's modulation wheel to change the LFO speed. If you don't want to use your keyboard's modulation wheel, click the Via pop-up menu to assign a different MIDI controller.
5. If you are using the Filter Envelope, drag the handles to set the attack, decay, sustain, and release values. Drag the Velocity slider to set the maximum amount of envelope modulation by velocity.

Use vibrato to modulate oscillator pitch

1. Rotate the Vibrato knob to set the amount of vibrato (pitch modulation).
2. Click the Vibrato tab if the LFO is visible.
3. Select a vibrato waveform.
4. If the Sync switch is turned off, drag the Rate note to set the vibrato speed. If the Sync switch is turned on, vibrato speed is controlled by the host application tempo. See Retro Synth LFO and Vibrato.
5. If the (Vibrato) Sync switch is turned off:
   - Adjust the Via Amount slider to the right of the waveform graphic to set the maximum modulation intensity (the highest vibrato speed).
   - Move your MIDI keyboard's modulation wheel to change the vibrato speed. If you don't want to use your keyboard's modulation wheel, click the Via pop-up menu to assign a different MIDI controller.

Retro Synth Glide and Autobend

In glide mode, the pitch of a played note slides to the pitch of the following played note. Autobend automatically bends note pitches when you strike a key.

Glide mode behavior changes when legato is selected with the Voices control. See Retro Synth global and controller settings.

Glide/Autobend parameters

- *On/off button:* Turns the Glide/Autobend section on or off.
- *Bend Type pop-up menu:* Choose Glide or Autobend mode.
- *Mode pop-up menu:* Choose what you want to bend: Oscillator 1 + Sine, Oscillator 2, All Oscillators, Opposed—with one oscillator bending up, while the other bends down by an equal amount, or Oscillators + Filter.
- *Time knob:* Rotate to set the time it takes for the pitch of one played note to travel to the pitch of another played note.
- *Depth knob:* (Autobend mode only) Rotate to set the pitch bend range (over a range of ±3 octaves).

Retro Synth LFO and Vibrato

Retro Synth's LFO (Low Frequency Oscillator) is a multiwaveform, polyphonic modulation generator that modulates each voice, or note you play, individually. It can be used as a source for multiple modulation targets.

Retro Synth also provides a dedicated Vibrato LFO for pitch modulation.

Although they are oscillators, LFOs are not audible—but their effects can certainly be heard. The sole purpose of an LFO is to modulate other sound generating elements of the synthesizer.

LFO/Vibrato parameters

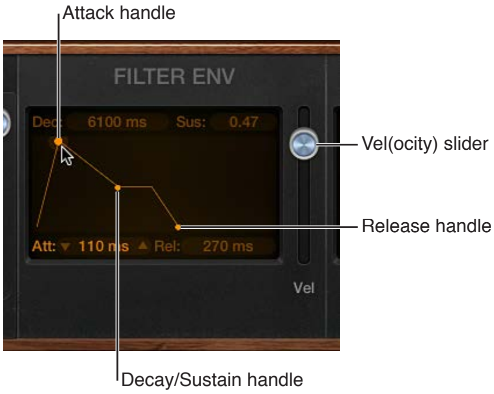
- *LFO/Vibrato tabs:* Click to change LFO or Vibrato parameters.
- *Waveform display:* Click the buttons at the top of the display to independently choose an LFO or Vibrato waveform.
  - The sawtooth waves are suitable for bubbling, rhythmic effects.
  - The triangle wave is suitable for vibrato and other evenly-modulated effects.
  - The rectangular waves switch between two values, which is useful for stepping the oscillator pitch by a fifth, for example.
- *LFO Sync button:* Turn on to independently synchronize the LFO or Vibrato speed with the host application tempo. Turn off to control the LFO or Vibrato speed manually.
- *Rate slider:* Drag to independently set the maximum LFO or Vibrato speed.
- *Source pop-up menu:* By default, your MIDI keyboard modulation wheel changes the LFO or Vibrato speed (Rate slider). You can choose aftertouch or both aftertouch and your keyboard's modulation wheel to control the Rate parameter.

Retro Synth envelopes

Retro Synth features two identical attack, decay, sustain, and release (ADSR) envelopes that shape the filter cutoff and the level of the sound over time.

When you think of different sounds, such as a snare drum, piano, or strings, they're not only tonally different, but the characteristics of the sound change over time. Both the snare drum and piano are heard immediately when struck. This is because they both have a short attack phase. Bowed strings, on the other hand, slowly ramp up in level—they have a long attack time, in other words.

If you break down any sound over time, you can emulate snare drum-like, piano-like, or string-like characteristics easily with Retro Synth's envelopes.



Envelope parameters

- *Attack handle:* Drag horizontally to set the time it takes for the envelope to reach the initial level.
- *Decay/Sustain handle:* Drag horizontally to set the time it takes for the envelope to fall to the sustain level, following the initial attack time.

  Drag vertically to set the sustain level, which is held until the key is released.
- *Release handle:* Drag horizontally to set the time it takes the envelope to fall from the sustain level to a level of zero.
- *Vel(ocity) slider:* Drag to determine how sensitive the envelope is to incoming velocity.
  - If set to maximum, the envelope outputs its maximum level only when the keys are struck at maximum velocity.
  - Softer velocities result in a corresponding change to the levels of each envelope—with a 50% velocity resulting in half-levels for the attack and sustain level parameters. Envelope attack, decay, and release times are not affected by velocity modulation.

Retro Synth global and controller settings

Retro Synth global controls are used to set the overall tuning, polyphony, and other aspects of your instrument.

The controller settings let you assign MIDI keyboard features to Retro Synth controls. You can use three MIDI controllers—velocity, modulation wheel, and aftertouch—to change Retro Synth's Filter Cutoff, Wave Shape (Pulse Width), or LFO/Vibrato Rate controls. Multiple MIDI controllers can be assigned to the same Retro Synth control, so you could change filter cutoff with both velocity and aftertouch, for example. Alternatively, a single MIDI controller can be assigned to multiple Retro Synth parameters—with aftertouch affecting both filter cutoff and LFO speed, for example.

If you're new to synthesizers and the concepts behind modulation controls, see Synthesizer basics overview.

Click the Settings label to switch between the modulation and global/controller controls.

Global parameters

- *Transpose pop-up menu:* Choose a value to transpose Retro Synth ±2 octaves.
- *Tune field:* Click the arrows or drag vertically to tune Retro Synth in semitone steps.
- *Bend pop-up menu:* Choose a value to set the maximum upward/downward pitch bend. Pitch bend modulation is typically performed with your MIDI keyboard pitch bend wheel or joystick.
- *Voices pop-up menu:* Choose a value to set the maximum number of notes that can be played. Retro Synth can play up to 16 simultaneous voices, or can be used as a monophonic (single voice) synthesizer.
  - If you choose legato and play in a legato style (strike a new key while holding another), the envelope generators are triggered only for the first note you play legato, and then they continue their curve until you release the last legato played key. This means that if you play legato, a portamento occurs (the portamento time is set with the Autobend / Glide Time control). If you release each key before pressing a new one, the envelope is not triggered by the new note, and there is no portamento.
  - If you choose mono, staccato playing retriggers the envelope generators every time a new note is played.
- *Voice Detune field:* Click the arrows or drag vertically to tune Retro Synth in cents (1 cent = 1/100th semitone).
- *Stereo Spread field:* Click the arrows or drag vertically to set the amount of voice panning, respective to the center position. Spread: 0=mono, 1=full left/right panning. Voices are panned left or right in an alternating, symmetrical pattern.

  Note: Detuning and panning works in Single and Double voice mode. In Double voice mode, detuning and panning affects the respective voice pairs.

- *Double switch:* Turns unison mode on or off. The behavior of unison mode depends on the number of voices set with the Voices parameter. One of the strengths of polyphonic analog synthesizers is unison—or stacked voices—mode. Traditionally, in unison mode classic analog polysynths run monophonically, with all voices playing simultaneously when a single note is struck. Because the voices of an analog synthesizer are never perfectly in tune, this results in a rich, chorus-like effect with great sonic depth.
  - Polyphonic unison mode: When 2–16 voices are selected, voices are stacked, but you can play polyphonically.
  - Monophonic unison mode: When Mono or Legato is selected with the Voices parameter, all voices are stacked, but you can only play monophonically or in a legato style.

Controller parameters

- *Mod Wheel to pop-up menu and slider:* Choose a target for modulation with your keyboard's modulation wheel. Drag the slider to set the maximum modulation amount.
- *Velocity to pop-up menu and slider:* Choose a target for modulation with keyboard velocity. Drag the slider to set the maximum modulation amount.
- *Aftertouch to pop-up menu and slider:* Choose a target for modulation with keyboard aftertouch. Drag the slider to set the maximum modulation amount.

Retro Synth extended parameters

Retro Synth provides additional parameters that can be accessed by clicking the disclosure triangle at the lower left of the interface.

Extended parameters

- *MIDI Mono Mode pop-up menu:* Choose Off, On (with common base channel 1), or On (with common base channel 16).

  In either mode, each voice receives on a different MIDI channel. Controllers and MIDI messages sent on the base channel affect all voices.
- *Mono Mode Pitch Range pop-up menu:* Choose 0, 24, or 48.

  The chosen pitch bend range affects individual note pitch bend messages received on all but the common base channel. The default is 48 semitones, which is compatible with Mobile GarageBand's keyboard in pitch mode. When using a MIDI guitar, 24 semitones is the preferable setting because most guitar to MIDI converters use this range by default.

Sculpture

13

Sculpture overview

This section contains key information and concepts that you need to understand before taking a look at Sculpture features and parameters. If you're new to synthesizers, it might be best to start off with Synthesizer basics overview on page 471, which will introduce you to the terminology and give you an overview of different synthesis methods and how they work.

Sculpture is a synthesizer that generates sounds by simulating the physical properties of a vibrating string. This approach to tone generation is called *component modeling*. It enables you to create a virtual model of an acoustic instrument, such as a violin or cello. Components such as the length of the neck, the material the instrument is made of—wood or metal, for example—the diameter, tension, and material of the strings—nylon or steel, for example—and the size of the instrument body can be modeled.

In addition to the physical properties of the instrument, you can determine how and where it is played—softly bowed, or plucked, on top of a mountain, or under the sea. Other aspects such as finger noise and vibrato can also be emulated. You can even hit your virtual instrument's strings with a stick, or emulate dropping a coin onto the bridge.

Sculpture is not limited to recreating real-world instruments. You are free to combine components in any way, leading to bizarre hybrids such as a six-foot-long guitar with a bronze bell for a body—played with a felt hammer.

You can also create more traditional synthesizer tones in Sculpture. These benefit from the modeling process itself, which tends to add a level of richness and an organic quality to sounds. The end results are lush, warm pads, deep and round synthesizer basses, and powerful lead sounds. If you need to create an endlessly evolving texture for a film soundtrack, or a spaceship takeoff sound, Sculpture is the perfect instrument for the job.

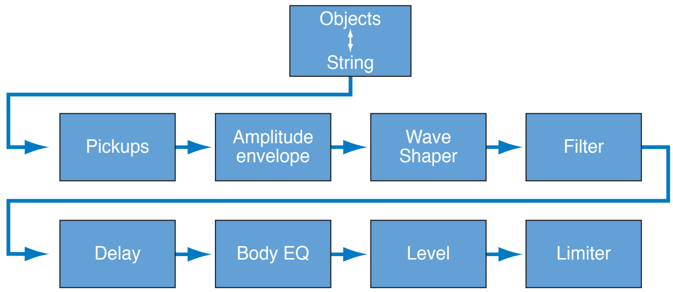
Like a real instrument, Sculpture generates sounds by using an object, such as a fingertip, wind, drumstick, or violin bow, to stimulate another object, such as a guitar string or reed.

Note: For clarity, the stimulated object is always referred to as the *string*.

As with a real instrument, the sound consists of multiple elements. It's not only the string that is responsible for the tonal color, but also the objects that stimulate or otherwise affect the string, and therefore the sound.

For example, imagine a steel-stringed guitar that is alternately strummed with your thumb and then picked strongly with your fingers. Changing to nylon strings, or 12 strings, would significantly change the tone. Now imagine the impact of pressing the strings down onto the fretboard, which not only changes the chord but also momentarily bends the strings, and therefore their pitch. Other aspects to consider are the size and material of the guitar body and how they influence the resonant characteristics of your sound. Further elements, such as the size or type of sound hole—round or F-shaped—the finger noise on the strings, and the medium that the guitar is played in, also have roles to play in the overall sound that you produce.

Sculpture enables you to virtually model the physical consistency and behavior of all components involved—hence *component modeling* synthesis.



This figure shows the signal flow of Sculpture's core synthesis engine.

Following the stimulation of the string by various objects, the vibration of the string is captured by two movable pickups—you can view these as being similar, in concept and operation, to the electromagnetic pickups found on guitars, electric pianos, or clavinets.

The pickups send the signal to the ADSR-equipped amplitude stage, a Waveshaper module, and a multimode filter. These all serve to *sculpt* your sound.

Note: All elements described above exist on a per voice basis.

The sum of all voice signals can then be processed by an integrated delay effect. From there, the signal is sent to an EQ-like module (the Body EQ), which simulates the spectral shape/body response of your instrument. The resulting signal is then fed to a level limiter section.

A vast number of modulation sources are also available, from tempo-synced LFOs to jitter generators and recordable envelopes. These can control the string and object properties, the filter, and other parameters. You can even modulate other modulation sources.

A recordable morph function also allows for smooth or abrupt transitions between up to five morph points. A morph point is essentially a collection of parameter settings at a given moment in time.

Important: The interaction between various sections of the component modeling synthesis engine is more dynamic and more tightly intertwined than that of other synthesis methods. This can lead to some truly unique sounds, but sometimes even a small parameter change can deliver dramatically different, and unexpected, results. Sculpture requires a more measured approach to sound creation than a traditional synthesizer design. Refer to the flowchart while learning the interface and programming.

Sculpture is a performance-oriented synthesizer that benefits from the use of controllers, modulations, and different playing techniques. Take time to experiment with all available controls and parameters when you initially audition some of the supplied sounds, and when you create new ones of your own.

Several tutorial sections will help you learn about creating sounds with Sculpture.

- Explore Sculpture overview on page 288
- Basic sound programming overview on page 293
- Electric bass programming overview on page 303
- Synthetic sound programming overview on page 321
- Explore Sculpture modulation options on page 292

Sculpture is an instrument that requires some investment of your time, but it will reward you with beautifully warm and organic sounds, evolving soundscapes—or a harsh and metallic "Hell's Bells" patch, if required. Don't be afraid to experiment; that's what Sculpture was created to do.

Sculpture interface

The user interface of Sculpture is divided into three main areas.

- *Sound engine:* The top two-thirds of Sculpture contains the sound engine. It is divided into five subsections:
  - *String parameters:* The circular Material Pad in the center is used to create and control the string, thus determining the basic timbre of your sound.
  - *Object parameters:* The area at the top left contains the objects that are used to stimulate or otherwise affect the string in some way.
  - *Processing parameters:* The processing parameters capture the string signal and provide further tonal control. These include the filter, Waveshaper, pickup, and amplitude envelope parameters.
  - *Global parameters:* Affect the overall behavior of Sculpture. Global parameters include bend range, keyboard mode, and tuning controls.
  - *Post-Processing parameters:* Affect the overall tone and behavior of the entire instrument. Post-processing parameters include the Delay, Body EQ, and Level Limiter parameters.
- *Modulation section:* The blue/gray area below the sound engine contains the modulation sources—LFOs, jitter generators, and recordable envelopes.
- *Global control sources:* The area at the bottom of the interface enables you to assign MIDI controllers to Sculpture parameters. This section also incorporates the Morph Pad, a dedicated controller for *morphable* parameters.

Sculpture string parameters

Sculpture string overview

The *string* is responsible for the basic tone of your sound. You can define its material—what it's made of—and determine its behavior when bowed, plucked, struck, and so on.

The string itself doesn't make a sound unless it is stimulated—excited or disturbed—by at least one object. Up to three different types of objects are used to excite, disturb, or damp the string (make it vibrate or affect its movement). See Sculpture objects overview on page 244.

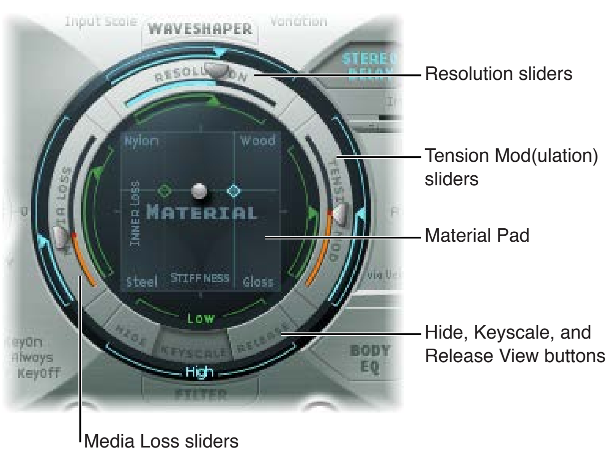
Sculpture's string and the excite/disturb objects are similar to the oscillators in traditional synthesizers. The string is considerably more sophisticated in concept than simple oscillators, however.

In essence, you are creating the waveform, or base timbre, by mathematically describing the string's properties, and the properties of its environment. These include, among others, the material the string is made of; the thickness, length, and tension of the string; its characteristics over time; the atmosphere it is being played in (such as water or air); and the way it is being played—struck, bowed, and so on.

Sculpture goes far beyond the mere creation of an infinite number of base timbres, however. One of the key differences between Sculpture's string and a traditional synthesizer waveform is that the base timbre provided by the string is in a constant state of flux. For example, if Sculpture's string is still vibrating for a specific note, retriggering that same note will interact with the ongoing vibration. This is not dissimilar to the effect of repeatedly plucking a guitar string, where the string is still vibrating when the next note is played. This will alter the harmonic spectrum each time—which is why acoustic guitars sound organic when a note is played repeatedly, and sampled guitars don't.

This is quite different from other synthesis methods where the base timbre waveform, even if modulated, does not harmonically interact with currently audible notes when retriggered. What usually happens in traditional synthesizers is that the waveform is restarted—from mid-cycle, or from the beginning—with the result being an increase in volume, or a slight cyclical wave shift.

The string parameters apply on a per-voice basis. You will note a number of parameter names followed by *(morphable)*. This indicates that the parameters can be morphed between up to five morph points. See Sculpture morph overview on page 278.



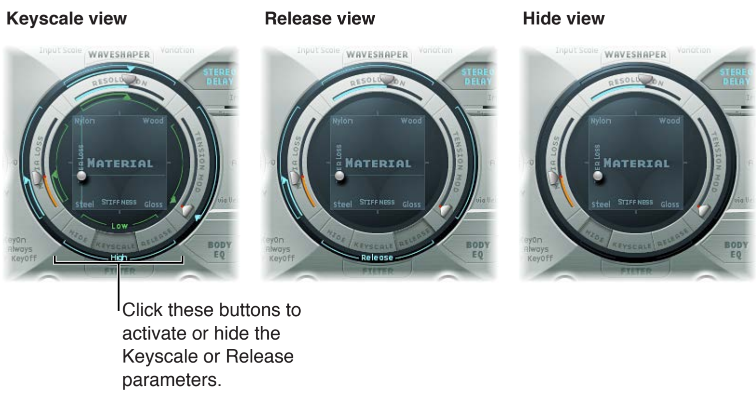
String parameters

- *Hide, Keyscale, and Release view buttons:* Show or hide different groups of parameters.
- *Material Pad:* Determines the basic tone of the string by setting the stiffness and damping properties.
- *String parameter sliders:* Shown on the outer ring of the Material Pad, the String parameter sliders further define the properties and behavior of the string.
  - *Resolution sliders:* Determine the maximum number of harmonics contained in the sound at C3 (middle C). Spatial resolution is also changed.
  - *Media Loss sliders:* Emulate the amount of string dampening caused by the surrounding environment (air, water, and so on) at C3 (middle C).
  - *Tension Mod(ulation) sliders:* Determine the momentary detuning of the sound at C3 (middle C).

  All sliders are set relative to middle C. As you play above or below this note, tuning and other elements of the string can, and will, change.

Sculpture Hide, Keyscale, and Release view

Click the Keyscale, Release, or Hide button to show or hide the corresponding parameters in the ring surrounding the Material Pad.

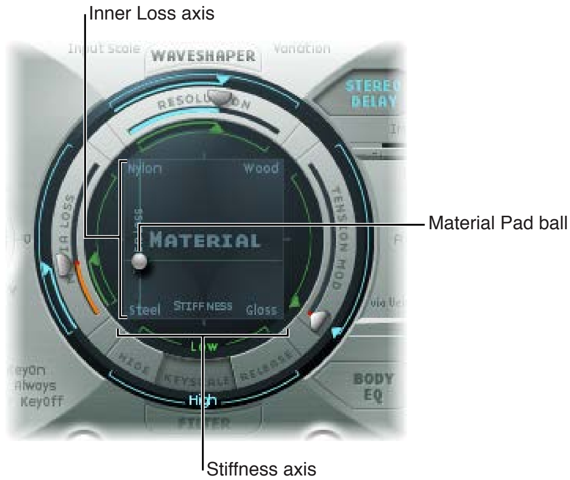


Hide, Keyscale, and Release parameters

- *Hide button:* Hides several elements, simplifying the interface and making it impossible to accidentally change Key Scale or Release parameters.
- *Keyscale button:* Sets parameters for notes that fall below C3, or notes that are positioned above it. In simple terms, the impact of these parameters can be controlled across the keyboard range. For example, a parameter such as string Stiffness could be more intense for high notes and less intense for low notes. In practical terms, this would result in more harmonic (sweeter) sounding bass notes and inharmonic overtones in treble notes (notes above C3).
- *Release button:* Turn on to set string Release parameters, which affect the vibrations of the string after the key is released.

Sculpture's basic Material Pad parameters

The Material Pad works as a matrix of Stiffness (x-axis) and Inner Loss (y-axis) values.



The four corners of the Material Pad display different material names. These each represent a combination of maximum/minimum Stiffness and Inner Loss values. The combination of the Inner Loss and Stiffness parameter positions determine the string material and, therefore, the general timbre of your sound. Here are examples of how Inner Loss and Stiffness settings can change the tonal color:

- Low Stiffness values, combined with low Inner Loss values, lead to metallic sounds.
- Higher Stiffness values, combined with low Inner Loss values, make the sound become more bell-like or glass-like.
- Higher Inner Loss values, combined with a low Stiffness level, correspond to nylon or catgut strings.
- High Stiffness values, combined with high Inner Loss values, simulate wood-like materials.

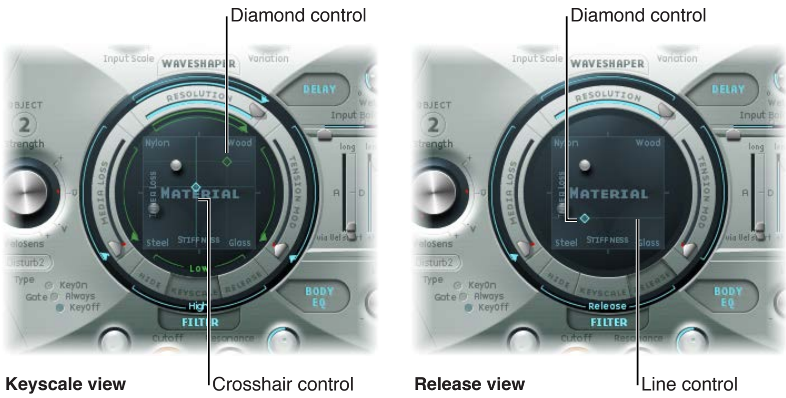
Material Pad parameters

- *Inner Loss:* Emulates damping of the string caused by the string material—steel, glass, nylon, or wood. These are frequency-dependent losses that cause the sound to become more mellow during the decay phase.
- *Stiffness:* Sets the rigidity of the string. In reality, this is determined by the string material and diameter—or, to be more precise, by its geometrical moment of inertia. Increasing the Stiffness parameter to the maximum value turns the string into a solid metal bar. Stiffer strings also exhibit an inharmonic vibration, where overtones are not integer multiples of the base frequency. Rather, they have higher frequencies, which can make upper/lower notes sound somewhat out of tune with each other.
- *Material Pad ball:* Controls both the Inner Loss and Stiffness parameters simultaneously when you drag it within the Material Pad. The ball marks a specific point on the X and Y planes.

  Note: The thickness of the string—the green horizontal line in the Pickup display—changes as you move the ball. See Use Sculpture's string parameter sliders on page 242

Use Sculpture's Material Pad in Keyscale or Release view

In Keyscale or Release view, the Material Pad shows additional controls for the Keyscale and Release parameters.



Material Pad Keyscale and Release parameters

- *Diamond controls:* Drag to adjust the Stiffness and Inner Loss Keyscale and Release parameters.
  - In Keyscale view, the diamonds indicate the intersection between the Inner Loss and Stiffness Low/High Scaling positions. You can drag these diamonds to adjust both parameters simultaneously.
  - In Release view, you only drag the diamond vertically, because you cannot adjust the release behavior of the Stiffness parameter.
- *Crosshair and line controls:* Use to control the Keyscale and Release parameters when the diamonds are hidden by the ball. The crosshair also enables you to independently change the keyscaling for one of the two axes (X/Y positions, which control the current Inner Loss and Stiffness values).

Note: Option-click any of the controls to reset parameters to their default values.

Adjust Inner Loss key scaling in the Material Pad

The Inner Loss Keyscale parameters control the amount of damping—independently for notes above and below middle C. This changes the way damping is scaled as you play lower or higher on the keyboard.

1. Click the Keyscale button.
2. Drag the green horizontal line for low notes or the blue horizontal line for high notes.

Adjust Stiffness key scaling in the Material Pad

The Stiffness Keyscale parameters adjust the string stiffness—independently for notes above and below middle C. This changes the scaling of inharmonic content that is heard as you play lower or higher on the keyboard.

1. Click the Keyscale button.
2. Horizontally drag the green vertical line for low notes or the blue vertical line for high notes.

Tip: You can simultaneously adjust both Stiffness and Inner Loss key scaling by dragging the diamond that intersects the green lines.

Adjust Inner Loss release scaling in the Material Pad

In Release view, you define changes to the damping amount when the key is released. Conservative use of this parameter—in conjunction with Media Loss Scale Release—allows a natural simulation of strings that are dampened when a note-off message is received. See Use Sculpture's string parameter sliders on page 242.

1. Click the Release button.
2. Vertically drag the blue Release line.

Use Sculpture's string parameter sliders

The sliders on the outer ring of the Material Pad further define the properties and behavior of the string.

Material Pad in Keyscale view

String parameter sliders

- *Resolution slider:* Drag to set the maximum number of harmonics contained in (and spatial resolution of) the sound at C3. Changes to the Resolution value alter the interaction of the string with the objects, which has a corresponding effect on overtone frequencies—very low Resolution values result in inharmonic spectra, even with Stiffness set to 0. Higher resolution values enhance the precision of calculations, increasing computer processing requirements. In Keyscale view, the Resolution High and Low Scaling sliders are shown:
  - *Resolution High Scaling slider (blue):* Drag to set the key tracking resolution—the accuracy of key tracking—for notes above middle C (C3).
  - *Resolution Low Scaling slider (green):* Drag to set the key tracking resolution for notes below middle C.
- *Media Loss slider:* Drag to set the amount of string damping caused by the surrounding media (the atmosphere)—for example, air, water, olive oil, and so on. These losses are independent of frequency. This provides control over the duration of the exponential amplitude decay, after the excitation of the string has stopped.
  - *Media Loss High Scaling slider (blue):* In Keyscale view, sets the key tracking resolution for notes above middle C (C3). In Release view, it sets media loss behavior when the key is released.
  - *Media Loss Low Scaling slider (green):* In Keyscale view, sets the key tracking resolution for notes below middle C. In Release view, it sets media loss behavior when the key is released.

- *Tension Mod slider:* Drag to set the momentary detuning of the string.
  - *Tension Mod High Scaling slider (blue):* Drag to set the tension modulation behavior for notes above middle C.
  - *Tension Mod Resolution Low Scaling slider (green):* Drag to set the tension modulation behavior for notes below middle C.

  Note: This nonlinear effect can deliver surprising results and can also make the entire model unstable, especially when combined with low Media Loss and Inner Loss values. If your sound spikes or drops out during the decay phase, try reducing Tension Mod, and perhaps Resolution.

Adjust Resolution key scaling

1. Click the Keyscale button.
2. Drag the green low slider inside the top of the Material Pad ring for low notes—or the blue high slider around the top of the outer ring for high notes.

Adjust Media Loss key scaling

1. Click the Keyscale button.
2. Drag the green slider inside the left side of the Material Pad ring.

Adjust the Media Loss release time

1. Click the Release button.
2. Drag the blue slider in the outer ring at the left side of the Material Pad.

Values above 1.0 cause media losses to increase when the key is released. This parameter can be used to simulate a string that is dropped into a bucket of water after initially vibrating in air, for example. Obviously, this is not what the average violinist or pianist would do, but it can be useful for some interesting sound variations.

Adjust Tension Mod key scaling

Strings, such as those of a guitar, exhibit a particularly prominent nonlinear behavior—if the string excursion is large, the string is detuned upward. Because this detuning is caused by the momentary, rather than the average, excursion of the string, the detuning occurs very quickly. This phenomenon is known technically as *tension modulation nonlinearity*. Nontechnically, setting or modulating the Tension Mod slider to values above 0.0 emulates this momentary detuning effect in Sculpture.

1. Click the Keyscale button.
2. Drag the green low slider inside the right side of the Material Pad ring for low notes—or the blue high slider around the right side of the outer ring for high notes.

Tip: If your instrument seems slightly sharp or flat as you play up or down the keyboard, consider adjustments to the Tension Mod, and perhaps Media Loss, Keyscale parameters.

Sculpture objects parameters

Sculpture objects overview

The objects are used to stimulate or otherwise affect the string in some way. The object parameters discussed in this section apply on a per-voice basis. You will note a number of parameter names followed by *(morphable)*. This indicates that the parameters can be morphed between up to five morph points. For more information, see Sculpture morph overview on page 278.

Important: At least one object must be used to excite or disturb the string, because the string itself does not make any sound.

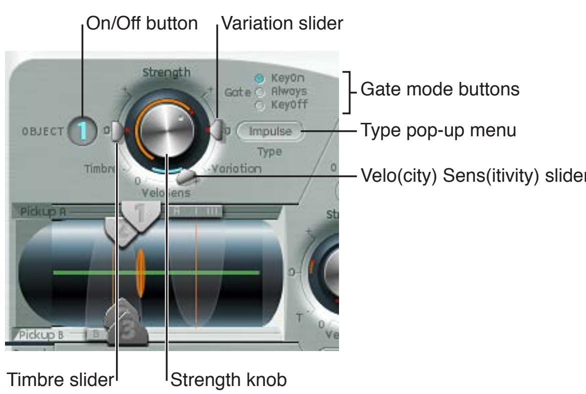
There are a number of different string *excite/disturb/damp* models available, such as blow, pluck, bow, and so on. These can radically alter the general timbre of the string's attack phase, making it possible to create bowed or plucked flute and bell sounds, or guitars with a flute-like attack, for example.

Judicious use of the object parameters can deliver very accurate emulations of real-world instruments, or sounds that are altogether more other-worldly.

It is important to note that each additional disturb/damp object that is activated will affect the string. This will, in turn, alter the interaction of *any* other active object with the string, often resulting in a completely different sonic character.

The goal of changing your sound's character is the reason you would use a new object, but the pluck and blow object combination you chose may sound like fingernails on a blackboard, rather than the plucked pan flute you were trying to create—depending on other string settings. Therefore, you need to pay special attention to the model type and strength of objects. You may find that the flavor of the excite object (Object 1), for example, has changed significantly—and you may need to adjust or change the parameters of *all* objects (and perhaps several string parameter values) after you introduce a new disturb/damp object (2 or 3). Similarly, the selection of a different *type* of excite object will affect the disturb/damp objects—and the string itself—thereby changing the character of your sound.

Repositioning objects also changes the timbre of the string. If you are emulating a guitar, for example, changing an object position would be similar to picking or strumming a string at various spots along the fretboard.



Object parameters

- *On/Off buttons (1, 2, and 3):* Turn the object on or off.
- *Type pop-up menus:* Choose each object type. See Sculpture excite table (objects 1 and 2) on page 246 and Sculpture disturb and damp table (objects 2 and 3) on page 247.
- *Gate mode buttons:* Click to determine when the object is active—that is, when it disturbs or excites the string. You can choose from:
  - *KeyOn:* Between note-on and note-off
  - *Always:* Between note-on and the end of the release phase
  - *KeyOff:* Triggered at note-off, and remaining active until the voice is released

  Note: Some object types, such as Gravity Strike, may retrigger the note when you release a key—when in Key On gate mode. If you encounter this artifact, try setting gate mode to Always, or reduce the Strength of the object.
- *Strength knob (morphable):* Rotate to set the intensity of the excitation/disturbance (depending on type). A value of 0.0 effectively disables excitation/disturbance. In contrast to the On/Off button of each object, you can fade in the Strength parameter with modulation or morphing options.
- *Timbre slider (morphable):* Determines the timbre (tonal color) of the chosen excitation/disturbance type. Zero (0.0) is the default value for the object. Positive values make the sound brighter. Negative values lead to a more mellow sound.
- *Variation slider (morphable):* An additional timbre parameter, which is also type dependent.
- *VeloSens slider (Objects 1 and 2 only):* Enables you to reduce velocity sensitivity to 0. Excite objects are velocity sensitive, but this may not be appropriate for all sounds, which is where this parameter is useful.

  Note: An object is velocity sensitive only when a type that actively excites the string is selected. The Velocity Slider is available only for objects that are velocity sensitive. Object 1 is velocity sensitive. Object 2 can be both, depending on the object type you choose. Object 3 is not velocity sensitive.

Sculpture excite table (objects 1 and 2)

Before you look at the tables of object types and properties, you should note the following:

- Object 1 can only use the excite types found in the first table.
- Object 2 can use the excite and disturb/damp types available in both tables.
- Object 3 can only use the disturb/damp types found in the second table. See Sculpture disturb and damp table (objects 2 and 3) on page 247.

The following table lists all excite types available for Objects 1 and 2, and information about the controls available for each object type.

Name	Description	Strength controls	Timbre controls	Variation controls
Impulse	A short impulse excitation	Impulse amplitude	Width	Velocity dependence of width
Strike	Short excitation, like a piano hammer or mallet	Hammer start speed (velocity dependent)	Hammer mass	Felt stiffness
GravStrike	Like hammer but with gravitation toward the string, leading to multiple hammer-string interactions and disturbed string vibrations	Hammer start speed	Felt stiffness	Gravitation
Pick	Finger or plectrum picking	Pickup force and speed	Force/speed ratio	Plectrum stiffness
Bow	Bowing of the string	Bow speed	Bow pressure	Slip stick characteristics
Bow wide	Same as bow, but wider, resulting in a more mellow tone, especially suited for smooth bow position changes	Bow speed	Bow pressure	Slip stick characteristics
Noise	Noise injected into the string	Noise level	Noise bandwidth/ cutoff frequency	Noise resonance
Blow	Blow into one end of the string (an air column, or tube). At various positions, starting from 0.0 (far left), move the blowing direction and position from along the string toward one end. The string is blown sideways at the chosen position.	Lip clearance	Blow pressure	Noisiness
External (available only for Object 2)	Feeds side-chain signal into string.	Level	Cutoff frequency of lowpass filter being used to process side-chain signal	Width (size) of the string area being affected by the side-chained signal

Sculpture disturb and damp table (objects 2 and 3)

The following table lists all disturb and damp types available for Objects 2 and 3. For information on excite types available for Objects 1 and 2, see Sculpture excite table (objects 1 and 2) on page 246.

Name	Description	Strength controls	Timbre controls	Variation controls
Disturb	A disturb object that is placed at a fixed distance from the string's resting position	The hardness of the object	The distance from the resting position • *Negative values:* Push the string away from the resting position. • *Positive values:* Do not affect the string in the resting position.	Controls width. • *Negative values:* Affect only a small section of the string. • *Positive values:* Affect a broader section of the string.
Disturb 2-sided	Somewhat like a ring placed around the string, which limits the string's vibration in all directions	The hardness of the ring	The clearance of the ring (the distance between the ring and string) • *Negative values:* The sides of the damping ring overlap, influencing the string if any movement occurs. • *Positive values:* There is an amount of clearance inside the ring. The string will be influenced only if moved sufficiently to actually touch the ring.	No effect
Bouncing	Emulates a loose object lying or bouncing on, and interacting with, the vibrating string. This is very random by nature and can't be synchronized.	Controls the gravity constant for the object lying/bouncing on the string.	The stiffness of the object	The damping of the object

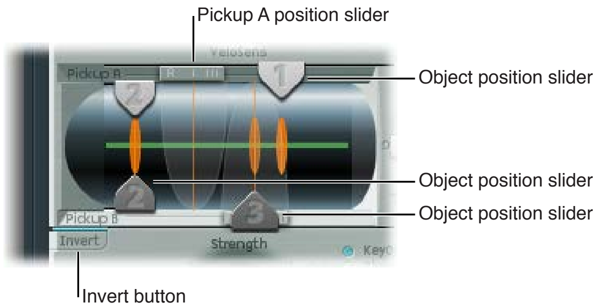
Name	Description	Strength controls	Timbre controls	Variation controls
Bound	A boundary that limits and reflects string movement. This is much like a fingerboard that limits string movement when the string is plucked very firmly.	The distance from the boundary center position to the string's resting position	The slope (steepness) of the boundary. A value of 0.0 places the boundary parallel to the string. Other values will position the boundary closer to the string on one end and farther away on the other.	The amount of reflection at the boundary limits
Mass	Used to model an additional mass attached to the string. This can lead to inharmonic sounds, and very interesting results, if the position of this mass is modulated along the string.	The mass size/weight	No effect	No effect
Damp	Localized damper, which is useful for soft damping	The intensity of the damping	The damping characteristics	The width of the damped string section

Sculpture pickups parameters

Use Sculpture pickup parameters

The pickup parameters discussed in this section apply on a per-voice basis. You will note a number of parameter names followed by *(morphable)*. This indicates that the parameters can be morphed between up to five morph points. See Sculpture morph overview on page 278.

The pickups are the first element beyond the sound-generating portion of Sculpture—consisting of the string and objects—and act as the input to the virtual signal processing chain. You can view the pickups as being like those of an electric guitar or clavinet. Obviously, changing their positions alters the tone of your instrument, just like the pickups in Sculpture.



The green horizontal line within the Pickup display represents the string. As the Stiffness parameter value of the string is increased, the line becomes thicker. The line can be animated and will show the range of the string's motion.

The Pickup A and B ranges are shown as transparent bell curves, which represent the position and widths of pickups A and B.

Pickup parameters

- *Object position sliders (morphable):* Drag sliders 1, 2, and 3 to determine the respective position of each (excite/disturb/damp) object along the string.
- *Pickup A position slider (morphable):* Drag to set the position of Pickup A along the string. Values of 0.0 and 1.0 determine the left and right ends of the string, respectively.
- *Pickup B position slider (morphable):* Drag to set the position of Pickup B along the string, shown underneath Object position slider 3 at the bottom of the figure.
- *Invert button (Pickup B phase):* Click to invert the phase of Pickup B, found at the lower left of the Pickup display. Options are: normal or invert(ed).

  Note: If the phase of Pickup B is inverted, the sound can become thinner due to portions of the Pickup A and Pickup B signals canceling each other out. Depending on the position of the pickups, however, the reverse may happen, with the sound actually becoming richer.

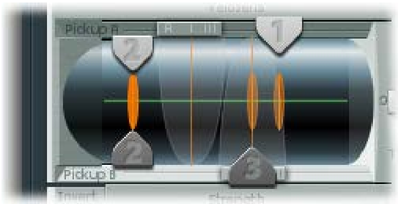
Adjust the Pickup A or Pickup B position

- Drag the slider handle at the top or bottom of the Pickup display.

Adjust an object pickup position

- Drag the corresponding numerical slider handle (the 1, 2, or 3 arrows) for each object.

  Adjustments to object positions will disturb/excite a given portion of the string. The vertical orange lines represent the positions of Objects 1, 2, and 3. The thickness and brightness of these lines indicate the strength of the objects. Object 1 can be an exciter. Object 3 can be a damper. Object 2 has two arrows, indicating that it can be used as either an exciter or damper.



Turn string animation on or off

- Control-click the green horizontal line—the string—to enable or disable string animation.

  When animation is active, the string vibrates, making it easier to visualize the impact of the objects and pickups. Note that string animation increases CPU overhead, so disable it if your computer is struggling to process all data in real time.

Sculpture's spread controls

Although they are not found in the actual Pickup display, two additional pickup parameters are available to the right of the Material Pad.

Key/Pickup Spread parameters

- *Key Spread button:* Drag vertically to set the amount of panning (pan position) by MIDI note number. Depending on settings, the farther up or down the keyboard you play, the more the voice is panned left or right. Two lines in the ring that surrounds the Spread parameters indicate values.
- *Pickup Spread button:* Drag vertically to spread the two pickups across the stereo or surround base. Two dots in the ring that surrounds the Spread parameters indicate values.

In surround instances, these two parameters can be affected by the Surround Range parameter. For more information, see Sculpture surround range and diversity on page 261.

Tip: You can create animated width and chorus effects by modulating the Pickup Position parameters with an LFO or other modulator.

Sculpture global parameters

These are found across the top of the Sculpture interface, unless otherwise specified.

Global parameters

- *Glide Time field:* Drag to set the time required to slide from the pitch of one played note to another. The Glide parameter behavior depends on the keyboard mode you choose.
  - If you set the keyboard mode to Poly or Mono and set Glide to a value other than 0, portamento is active.
  - If you choose Legato and set Glide to a value other than 0, you need to play legato (press a new key while holding the old one) to activate portamento. If you don't play in a legato style, portamento won't work. This behavior is also known as *fingered portamento*.
- *Tune field:* Drag to fine-tune the entire instrument, in cents. A cent is 1/100th of a semitone.
- *Warmth field:* Drag to slightly detune each voice, much like the random fluctuations caused by the components and circuitry of analog synthesizers. As the parameter name suggests, this warms up or thickens the sound.
- *Transpose field:* Drag to tune the entire instrument by octaves. Given the ability of component modeling to radically alter pitch with certain settings, coarse tuning is limited to octave increments.
- *Voices field:* Drag to specify the number of voices that can be played at any one time. Sixteen voices is the maximum polyphony of Sculpture.
- *Keyboard Mode buttons:* Click to choose polyphonic, monophonic, and legato behaviors. A *polyphonic* instrument, such as an organ or piano, allows several notes to be played simultaneously. Many older analog synthesizers are *monophonic*, which means that only one note can be played at a time, much like a brass or reed instrument. This shouldn't be viewed as a disadvantage in any way, because it allows playing styles that are not possible with polyphonic instruments.
  - In Mono mode, staccato playing retriggers the envelope generators every time a new note is played. If you play in a legato style (play a new key while holding another), the envelope generators are triggered only for the first note you play legato. They then continue their curve until you release the last legato played key. Mono mode is also known as *multi trigger* mode.
  - Legato mode is also monophonic, but with one difference: the envelope generators are retriggered only if you play staccato—releasing each key before playing a new key. If you play in a legato style, envelopes are not retriggered. Legato mode is also known as *single trigger* mode.

  Note: All modes retrigger a potentially sounding voice with the same pitch, instead of allocating a new one. Therefore, multiple triggering of a given note results in slight timbral variations, depending on the current state of the model at note-on time. If Sculpture's string is still vibrating for a specific note, retriggering that same note interacts with the ongoing vibration, or current state of the string. A true retrigger of the vibrating string will happen only if both Attack sliders of the amplitude envelope are set to 0. If either slider is set to any other value, a new voice is allocated with each retriggered note. See Sculpture amplitude envelope parameters on page 252.

- *Bender Range Up/Down fields:* Drag to set the upward/downward pitch bend range.
  - Separate settings are available for upward and downward pitch bends—using your MIDI keyboard's pitch bend controller.
  - When Bender Range Down is set to Linked, the Bender Range Up value is used for both (up and down) directions.

  Note: Bending the string, just like the string on a real guitar, will alter the shape of the modeled string, rather than merely act as a simple pitch bend.

Sculpture amplitude envelope parameters

The parameters discussed in this section apply on a per-voice basis. This is a straightforward ADSR envelope that scales the pickup signals before passing them on to the Waveshaper and filter.

The positioning of the amplitude envelope at this point in the signal path produces more natural-sounding results because you can control signal levels before sending them to the Waveshaper (if used). The Waveshaper can have a significant impact on the spectral content of the sound, which can lead to synthetic-sounding results.

Amplitude envelope parameters

- *A(ttack)—Soft and Hard slider(s):* The lower slider (Soft) sets the attack time for a note played at maximum velocity. The top slider (Hard) sets the attack time for a note played at minimum velocity. You can adjust both slider halves simultaneously by dragging in the space between them.

  Important: The attack time parameters of the amplitude envelope have a major impact on the way a single note is retriggered. When both Attack Soft and Hard are set to a value of 0, the vibrating string is retriggered. If either of these parameters is set to a value *above 0*, a new note is triggered. Sonically, the retriggering of a vibrating string results in different harmonics being heard during the attack phase.

- *D(ecay) slider:* Drag to set the time it takes for the signal to fall to the sustain level, following the initial strike/attack time.
- *S(ustain) slider:* Drag to set the sustain level. The sustain level is held until the key is released.
- *R(elease) slider:* Drag to set the time it takes for the signal to fall from the sustain level to a level of 0. Short Release values help to reduce CPU load, because the voice is no longer processed after the release phase has completed.

Note: Even with long decay and release times, the sound may decay quickly. This can be caused by high Inner or Media Loss values in the string material section or by objects (2 or 3) that are used to damp the string.

Use Sculpture's Waveshaper

The Waveshaper imposes a nonlinear shaping curve on each voice of the signal coming from the pickups and amplitude envelope. This reshaped signal is then passed on to the filter. This process is quite similar to the waveshaping of oscillators in synthesizers such as Korg's O1/W.

Waveshaper parameters

- *Waveshaper On/Off button:* Turns the Waveshaper on or off.
- *Type pop-up menu:* Select one of four waveshaping curves. See the table.
- *Input Scale knob (morphable):* Rotate to cut or boost the input signal, prior to processing by the Waveshaper. Positive values result in a richer harmonic spectrum. Any level increase introduced by this parameter is automatically compensated for by the Waveshaper.

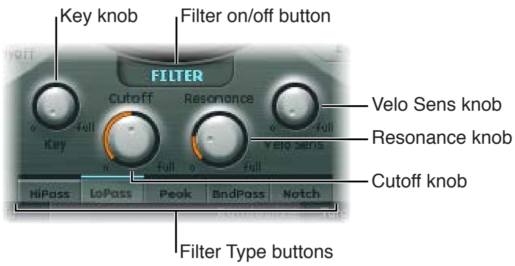
  Note: Given its impact on the harmonic spectrum, Input Scale should be viewed and used as a timbral control, rather than a level control. Also note that extreme Input Scale values can introduce processing noise at the Waveshaper output.
- *Variation knob (morphable):* Rotate to change. Impact is dependent on the selected Waveshaper curve. See the table.

Type	Variation controls	Value of 0.0	Negative values	Positive values
VariDrive	Wet/dry ratio	Provides shaped signal only.	Reduces shaped signal and adds dry signal.	Raises shaped signal and adds phase-inverted dry signal, making sound sharper.
• SoftSat • Tube Dist. • Scream	Bias—alters the symmetry of the shaping curve.	Results in symmetrical shaping.	Alters symmetry.	Alters symmetry.

Sculpture filter parameters

The parameters discussed in this section apply on a per-voice basis. A number of parameter names are followed by *(morphable)*, which indicates that the parameters can be morphed between as many as five morph points. For more information, see Sculpture morph overview on page 278.

The filter parameters provide further timbral/spectral control over your sound. They will be familiar to you if you have any experience with synthesizers. If you're new to the concepts behind synthesizer filters, see Filters overview on page 482.



Filter parameters

- *Filter On/Off button:* Turns the filter section on or off.
- *Filter type buttons:* Click to choose the filter mode.
  - *HiPass:* Allows frequencies above the cutoff frequency to pass. Because frequencies below the cutoff frequency are suppressed, it's also known as a *low cut filter*. The slope of the filter is 12 dB/octave.
  - *LoPass:* Allows frequencies that fall below the cutoff frequency to pass. Because frequencies above the cutoff frequency are suppressed, it's also known as a *high cut filter*. The slope of the filter is 12 dB/octave.
  - *Peak:* Allows the center of a frequency band to be specified with the Cutoff knob. Bandwidth and gain are controlled with the Resonance knob. Frequencies outside the band are left at their current level. Peak filters are generally used to enhance a frequency range.
  - *BandPass:* The frequency band surrounding the center frequency is allowed to pass. Resonance controls the width of this band. All other frequencies are cut. The bandpass filter is a two-pole filter with a slope of 6 dB/octave on each side of the band.
  - *Notch:* The frequency band surrounding the center frequency is cut. Resonance controls the width of this band. All other frequencies are allowed to pass. Notch filters are generally used to suppress noise or a particular frequency.
- *Cutoff knob (morphable):* Rotate to set the cutoff or center frequency, depending on the chosen filter type. In a *lowpass* filter, all frequency portions *above* the cutoff frequency are suppressed, or cut off, hence the name. The cutoff frequency controls the brilliance of the signal. The higher the cutoff frequency is set, the higher the frequencies of signals that are allowed to pass through the lowpass filter.

- *Resonance knob (morphable):* Rotate to set the filter resonance value.
  - In highpass and lowpass modes, Resonance emphasizes the portions of the signal that surround the center frequency.
  - In Peak, Bandpass, and Notch modes, Resonance controls the width of the band that surrounds the center frequency.
- *Key (tracking) knob:* Rotate to determine how cutoff frequency responds to key position. The farther up or down the keyboard you play, the more bright or mellow the sound becomes. Technically speaking, the cutoff frequency is modulated by the keyboard position. A value of 0.0 disables key tracking. A value of 1.0 makes the cutoff frequency follow the fundamental of the note across the entire keyboard range. Play an octave higher and the cutoff frequency also changes by an octave.
- *Velo Sens knob:* Rotate to determine how cutoff frequency responds to incoming note velocities. The harder you strike the keyboard, the higher the cutoff frequency—and, generally, the brightness of the sound—becomes. A value of 0.0 disables velocity sensitivity. A value of 1.0 results in maximum velocity sensitivity.

Sculpture delay effect parameters

Sculpture delay effect overview

This is a (project) tempo-syncable stereo or true surround delay. It can also run freely (unsynchronized). The Delay section features all the general delay parameters you'd expect from a delay plus the Groove (delay timing) Pad.

Delay effect parameters

- *Delay On/Off button:* Turns the Delay effect on or off.
- *Wet Level knob:* Rotate to set the Delay output level.
- *Feedback knob:* Rotate to set the amount of delay signal that is routed back from the delay unit output channels to the delay unit input channels. Negative values result in phase-inverted feedback.
- *Xfeed (Crossfeed) knob:* Rotate to set the amount of delay signal that is fed from the delay unit's left output channel to the right input channel and vice versa. Negative values result in phase-inverted feedback of the crossfed signal.

  In surround instances, the Xfeed knob controls crossfeedback between the delay lines, but offers additional crossfeed modes. You can access these in Sculpture's Extended Parameters area.

- *LoCut slider:* Determines the cutoff frequency of the highpass filter at the delay line output/feedback loop.
- *HiCut slider:* Determines the cutoff frequency of the lowpass filter at the delay line output/feedback loop.
- *Groove Pad:* Use to graphically adjust delay times in stereo or surround instances. See Sculpture's Groove Pad (stereo) on page 256 and Sculpture's Groove Pad (surround) on page 257.
- *Input Balance slider:* Drag to move the stereo center of the Delay input to the left or right, without the loss of any signal components. This makes it ideal for ping-pong delays.

  In surround instances, this parameter moves all channels toward the front left or front right channel.
- *Delay Time slider and field:* Drag to set the delay time. This can be in either musical note values—1/4, 1/4t (1/4 triplet), and so on (see "Sync button" below)—or in milliseconds.
- *Sync button:* Set either tempo-synced or free-running delay modes.
- *Output Width slider:* Drag to change the stereo or surround base of the wet signal. A value of 0.0 results in mono output. A value of 1.0 results in full stereo or surround output—the left delay line output channels are panned hard left, and the right delay line output channels are panned hard right, but the stereo center is unaffected.

  Note: This parameter is aimed primarily at achieving pure delay grooves in multiple channels, without hard left/right ping-pong panning.

Sculpture's Groove Pad (stereo)

When used in a stereo instance of Sculpture, the Spread and Groove parameters are displayed in the two-dimensional Groove Pad.

Drag the diamond in the center of the crosshair to adjust the values. You can also independently adjust the Spread and Groove parameter values by dragging the lines that intersect the diamond.

You can also Control-click the Groove Pad to open a shortcut menu that contains Copy, Paste, and Clear commands. These can be used to copy and paste delay settings between multiple Sculpture instances or between consecutively loaded settings. The Clear command resets the current delay settings.

Groove Pad parameters

- *Spread:* Adjust for wide stereo delay effects. Values on the y-axis (above the default, centered position) increase the delay time of the right delay line or decrease the delay time of the left delay line—in effect, smearing the delay times of the left and right channels. Negative values invert this effect.
- *Groove:* Distributes delay taps to the left/right channels, rather than smearing them, like the Spread parameter. Values on the x-axis reduce the delay time of one delay line by a given percentage, while keeping the other delay line constant. Keep an eye on the small help tag while adjusting.

  For example, a value of +50% reduces the right delay time by half. If a value of 1/4 was used as the Delay Time, the right delay would equal 1/8 of a note and the left delay would remain at 1/4 of a note. Needless to say, this parameter is perfect for the creation of interesting rhythmic delays—in stereo.

Sculpture's Groove Pad (surround)

When used in a surround instance of Sculpture, the Delay Time Pad converts into a pure groove pad that controls the delay time relationship between:

- Left and right channels (speakers) in the horizontal direction
- Front and rear channels (speakers) in the vertical direction

The Spread parameter is shown as a numerical field that can be edited at the top left of the Groove Pad. Drag, or double-click and type, to change the value.

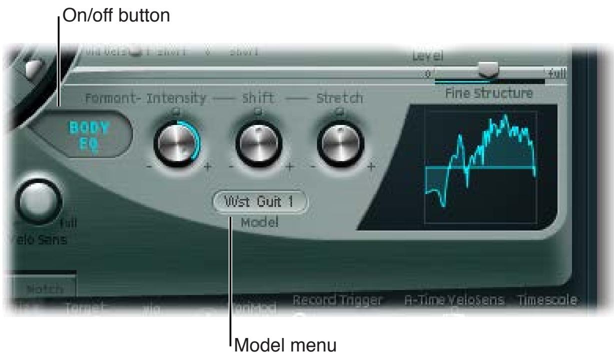
Sculpture Body EQ parameters

Sculpture Body EQ overview

The Body EQ can work as a simple EQ, as a complex spectral shaper, or as a body response simulator. In effect, the Body EQ can emulate the resonant characteristics of a wooden or metallic body—such as that of a guitar, violin, or flute.

The various models are derived from impulse response recordings of actual instrument bodies. These recordings have been separated into their general formant structure and fine structure, enabling you to alter these properties separately.

The Body EQ affects the summed signal of *all* voices, rather than each voice independently.



Body EQ global parameters

- *Body EQ On/Off button:* Turns the spectral shaping (Body EQ) section on or off.
- *Model pop-up menu:* Choose from various emulations of acoustic instrument bodies or the Basic EQ model. Your selection is reflected in the graphical display to the right.

  Note: When Basic EQ or another Body EQ model is chosen, the three knobs and slider parameter names and behaviors change. See Use Sculpture’s Basic EQ model on page 259, and Use Sculpture’s Body EQ models on page 260.

Use Sculpture's Basic EQ model

The Basic EQ parameters differ from other EQ models.

Basic EQ parameters

- *Low knob:* Rotate to set the gain of a low shelving filter.
- *Mid knob:* Rotate to set the gain of a peak filter (sweepable—see "Mid Frequency slider" below).
- *High knob:* Rotate to set the gain of a high shelving filter.
- *Mid Frequency slider:* Drag to sweep the center frequency of the mid band between 100 Hz and 10 kHz.

Adjust the Basic EQ (Lo Mid Hi model) graphically

- *To control the Low parameter:* Drag the left third of the graphic vertically.
- *To control the Mid parameter:* Drag the center third of the graphic vertically.
- *To control the Mid Frequency parameter:* Drag the center third of the graphic horizontally.
- *To control the Hi parameter:* Drag the right third of the graphic vertically.

Use Sculpture's Body EQ models

All other Body EQ models provide the following parameters:

Body EQ parameters

- *Formant–Intensity knob:* Rotate to scale the intensity of the model's formants. Any formants (harmonics) in the model become louder or are inverted, depending on how this parameter is used. A value of 0.0 results in a flat response. A value of 1.0 results in strong formants. Negative values invert the formants.
- *Formant–Shift knob:* Rotate to shift the formants logarithmically. A value of –0.3, for example, shifts all formants one octave downward, and a value of +0.3 shifts the formants up one octave. A value of +1.0 shifts up by a factor of 10—from 500 Hz to 5000 Hz, for example.
- *Formant–Stretch knob:* Rotate to stretch the formant frequencies, relative to each other. This parameter alters the width of all bands being processed by the Body EQ, extending or narrowing the frequency range. Low Formant Stretch values move the formants closer together (centered around 1 kHz), whereas high values move the formants farther apart from each other. The control range is expressed as a ratio of the overall bandwidth.

  Note: When combined, Formant Stretch and Formant Shift alter the formant structure of the sound and can result in some interesting timbral changes.

- *Fine Structure slider:* Drag to enhance the spectral (harmonic) structure, making the overall harmonic makeup of the sound more precise. This results in a more detailed sound that is harmonically richer and—depending on the model selected—more guitar-like or violin-like, for example. In other words, the resonant cavities of the instrument become more resonant—somewhat like the increased depth of tone provided by a larger-bodied guitar. A value of 0.0 denotes no fine structure. A value of 1.0 results in enhanced/full fine structure of the selected model.

  Note: Heavy use of Fine Structure may be quite CPU intensive. Also note that Fine Structure may not actually result in much difference in your sound. It is highly dependent on several string, Waveshaper, and Body EQ model parameter settings. As always, use your ears!

Adjust Body EQ models graphically

- *To control the Formant Intensity parameter:* Drag the graphic vertically.
- *To control the Formant Shift parameter:* Drag the graphic horizontally.

Sculpture output parameters

The Level Limiter is useful for softening some of the more aggressive aspects—such as snarling or roaring sonic artifacts—that you may encounter when using Sculpture.

Output parameters

- *Level knob:* Rotate to set the overall output level of Sculpture.
- *Level Limiter mode buttons:*
  - *Off:* Turns the Level Limiter off.
  - *Mono:* Turns on a monophonic limiter that processes the summed signal of all voices.
  - *Poly:* Turns on a polyphonic limiter that processes each voice independently.
  - *Both:* Turns on a combination of both limiter types.

Sculpture surround range and diversity

In surround instances, Sculpture's Extended Parameters area displays the Surround Range and Surround Diversity parameters:

Surround parameters

- *Surround Range slider:* Drag to set the range of the surround angle—the breadth of the surround field. Imagine an LFO routed to a pickup's pan position with an amount of 1.0. Setting the LFO waveform to sawtooth and the Surround Range to 360 results in circular movement—around the entire surround circle—of the voice output. The Surround Range parameter also influences the Key and Pickup spread parameters in the same way.
- *Surround Diversity slider:* Drag to determine how the output signal is spread across your surround speakers. If you choose a value of 0, only the speakers closest to the original signal's position will carry the signal. A diversity of 1 means that all speakers will carry an identical amount of the signal.

Sculpture modulation controls

Sculpture modulation overview

Sculpture is equipped with a huge number of modulation sources and targets, making it an instrument that can generate extraordinary sounds that constantly evolve, sound like audio loops, or are just plain expressive to play.

Some of the modulation sources provided are like those found on traditional synthesizer designs. These include:

- Two freely assignable LFOs, with (project) tempo-syncable rates.
- A third LFO that is dedicated to vibrato—pitch modulation.
- Two envelopes that can be used as standard envelopes, but which can also be used quite differently.

Sculpture also includes a number of specially designed modulation sources that are less conventional. These include:

- Two jitter generators with adjustable bandwidth—used to create random variations.
- Two Randomizers that change values only at note start/on—perfect for emulating the lip, breath, and tongue effects of brass instrument players, for example.
- Two recordable envelopes that can be used as MIDI controlled modulators—with the ability to polyphonically play back on a per-voice basis, and modify incoming MIDI controller movements.

Sculpture does not provide a centralized modulation router. All modulation routings—choosing a modulation target and/or via source—are made within each modulation source pane.

Open a modulation source pane

- Click the button of the modulation source you want to use. When a modulation source is active, the corresponding button label is highlighted.

Sculpture LFOs

Sculpture LFO overview

Sculpture offers two multiwaveform LFOs. Both can be used either polyphonically, monophonically, or somewhere in-between.

If used monophonically, the modulation is identical for all voices. Imagine a scenario where a chord is played on the keyboard. If LFO 2 is used to modulate pitch, for example, the pitch of all voices in the played chord will rise and fall synchronously. This is known as a phase-locked modulation.

In the same scenario, if LFO 2 is used polyphonically—to modulate multiple voices—they will *not* be phase-locked.

If a random (in-between) value is used, some notes will be modulated synchronously, and others won't.

Furthermore, both LFOs are key-synced: Each time you play a key, the LFO modulation of this voice is started from 0.

To understand the non-phase-locked characteristic more easily, imagine a situation where a chord is played on the keyboard. If LFO 2 is used to modulate pitch, for example, the pitch of one voice might rise, the pitch of another voice might fall, and the pitch of a third voice might reach its minimum value. As you can see from this example, the modulation is independent for each voice, or note.

The key sync feature ensures that the LFO waveform cycle always starts from 0, which results in consistent modulation of each voice. If the LFO waveform cycles were not synchronized in this way, individual note modulations would be uneven.

Both LFOs can also be faded in or out automatically, courtesy of built-in envelope generators.

LFO parameters

- *Waveform pop-up menu:* Choose the waveform used for LFO modulation. See Sculpture LFO waveforms on page 265.
- *Waveform display:* Shows the results of changes to the Waveform pop-up menu and Curve knob parameter settings.
- *Curve knob:* Rotate to change the shape of modulation waveforms. A pure waveform of the chosen type is active at a value of 0.0. The +1 and −1 positions deform the wave. For example, with a sine wave chosen as the LFO waveform type:
  - *Curve value of 0.0:* Wave is sine-shaped.
  - *Curve values above 0.0:* Wave is smoothly changed into a nearly rectangular wave.
  - *Curve values below 0.0:* The slope at the zero crossing is reduced, resulting in shorter soft pulses to +1 and −1.
- *Rate knob:* Rotate to set the rate of LFO modulation. This is either a freely definable Hz value (when the Free button is active), or a rhythmic value (when the Sync button is active). When synchronized with the project tempo, available rates range from 1/64 notes to a periodic duration of 32 bars. Triplet and punctuated values are also accessible.
- *Sync/Free buttons:* Click to choose either synchronized or free-running LFO rates. These buttons interact with the Rate knob. The synchronized value is derived from the host application tempo and meter.
- *Envelope knob:* Rotate to set the time it takes for the LFO modulation to fade in or fade out. See Modulate Sculpture LFOs on page 266.
- *Phase knob:* Rotate to choose between monophonic or polyphonic LFO modulations. These can have similar phases, completely random phase relationships, key-synced phases, or anything in-between.

  Tip: If you move the Phase knob slightly away from the mono position, you get nonlocked modulations for all voices running at similar, but not identical, phases. This is ideal for string-section vibratos.
- *RateMod Source pop-up menu:* Choose a modulation source for the LFO Rate parameter.
- *RateMod Amount slider:* Move to set the intensity—the amount—of LFO rate modulation.

Sculpture LFO waveforms

The LFO Waveform pop-up menus set different waveforms for the LFOs. The table below outlines how these can affect your modulations.

Waveform	Comments
Sine	Ideal for constant, even, modulations.
Triangle	Well-suited for vibrato effects.
Sawtooth	Well-suited for helicopter and space gun sounds. Intense modulations of the oscillator frequencies with a negative (inverse) sawtooth wave lead to "bubbling" sounds. Intense sawtooth modulations of lowpass filter cutoff and resonance create rhythmic effects. The waveform can also be inverted, resulting in a different start point for the modulation cycle.
Rectangle	Use of the rectangular waves periodically switches the LFO between two values. The Rectangle Unipolar wave switches between a positive value and 0. The Rectangle Bipolar wave switches between a positive and a negative value set to the same amount above and below 0. An interesting effect can be achieved by modulating pitch with a suitable modulation intensity that leads to an interval of a fifth. Choose the upper rectangular wave to do this.
Sample & Hold	The two Sample & Hold (S & H) waveform settings of the LFOs output *random* values. A random value is selected at regular intervals, as defined by the LFO rate. The S & H waveform steps between randomized values (rapid switches between values). The S & H Lag setting smooths the random waveform, resulting in fluid changes to values. The term *Sample & Hold* refers to the procedure of taking samples from a noise signal at regular intervals. The values of these samples are then *held* until the next *sample* is taken. *Tip:* A random modulation of pitch leads to an effect commonly referred to as a *random pitch pattern generator* or *sample and hold*. Try using very high notes, at very high rates and high intensities—you'll recognize this well-known effect from hundreds of science fiction movies.
Filtered Noise	Can be used for chaotic modulations, but it is principally of use in conjunction with the LFO envelope function, where you would introduce a brief modulation at some point in the note phase—for example, to introduce breath in a brass emulation, or to control an organ key click or piano hammer noise. The random nature of the noise waveform means that such modulations would vary slightly each time.

Modulate Sculpture LFOs

Two modulation targets can be assigned per LFO. An optional via modulation can also be assigned.

The LFOs also feature a simple envelope generator, which is used to control the time it takes for the LFO modulation to fade in or fade out. At its center position, which is accessed by clicking the middle mark, the modulation intensity is static—in other words, no fade-in or fade-out will occur.

Click the 1 or 2 buttons to activate each source.

LFO target and source parameters

- *LFO Modulation On/Off buttons (1 and 2):* Turn each LFO on or off, independently.
- *Modulation Target pop-up menus:* Choose the modulation targets.
- *Via Source pop-up menus:* Choose (or disable) the via sources that control the modulation scaling for each LFO.
- *Amt sliders:* Move to set the modulation amount (when the incoming via signal is 0)— for example, when the modulation wheel is at its minimum position.

  In cases where the via source is set to off, only one amount slider is visible (the via amount slider is hidden). In cases where any via source other than off is selected, there are two sliders.
- *Via (Amount) sliders:* Move to set the via modulation amount (when the incoming via signal is at its maximum)—for example, when the Modulation wheel is at the maximum position.

Set the LFO modulation fade time

- *To fade in the modulation:* Rotate the Envelope knob to a positive value.

  The higher the value, the longer is the delay time.
- *To fade out the modulation,* rotate the Envelope knob to a negative value.

  The farther to the left the knob is positioned, the shorter is the fade-out time.

Set up a delayed vibrato

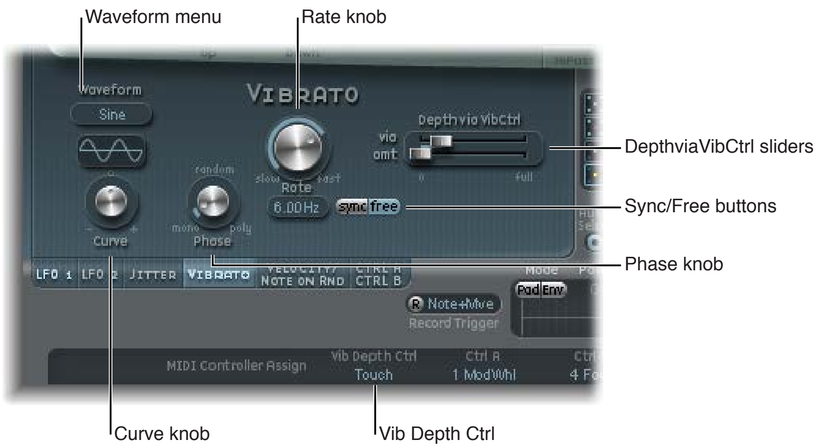
LFO envelopes are most often used for delayed vibrato—many instrumentalists and singers intonate longer notes this way.

1. Set the LFO Envelope knob toward the right (Delay) and choose pitch as the target.
2. Set a slight modulation intensity.
3. Select an LFO Rate of 5 Hz.
4. Choose the triangular wave as the LFO waveform.

Tip: Chaotic and fast modulations of pitch by an LFO source—with a delayed Sample&Hold waveform, a high Rate, and short fade-out—are ideal for emulating the attack phase of brass instruments.

Sculpture Vibrato parameters

One LFO is hard-wired to pitch, for vibrato effects, periodic pitch modulations. You can adjust the strength of the vibrato effect by the MIDI controller you choose from the Vib Depth Ctrl pop-up menu, which is in the MIDI Controller Assignment section. See Define Sculpture MIDI controllers on page 287.



Vibrato parameters

- *Waveform pop-up menu:* Choose the waveform used for vibrato—sine, triangle, sawtooth, and so on. See Sculpture LFO waveforms on page 265.

  Note: There are two special rectangular waves, Rect01 and Rect1—the former switching between values of 0.0 and 1.0 (unipolar), and the latter switching between values of –1.0 and +1.0 (bipolar, like the other waveforms).

- *Curve knob:* Rotate to change the shape of modulation waveforms. Such variations can result in subtle or drastic changes to your modulation waveforms.

  Note: The waveform displayed between the Curve knob and the Waveform menu shows the results of these two parameter settings.

- *Phase knob:* Rotate to choose between strictly monophonic or polyphonic vibrato with variable phase relationships. These can be similar phases, completely random phase relationships, key-synced phases—or any value in between. For more details, see Sculpture LFO overview on page 263.
- *Rate knob:* Rotate to set the rate of vibrato, which can be either synced to the current project tempo or set independently in Hz (Hertz) values.
- *Sync/Free buttons:* Select either a synchronized or free-running vibrato rate. These buttons interact with the Rate knob. The synchronized value is derived from the host application tempo and meter.
- *DepthviaVibCtrl sliders:* Drag to define the impact of the controller assigned to Vib Depth Ctrl (see Define Sculpture MIDI controllers on page 287).
  - *Via slider:* Drag to determine the modulation intensity of the controller assigned to Vib Depth Ctrl.
  - *Amt slider:* Drag to determine the maximum modulation amount.

Sculpture Jitter generators

Many sounds can benefit from small, random modulations to parameters. These can emulate the subtle variations that occur when particular instruments are played.

The two jitter generators are special LFO sources that are designed to produce continuous, random variations—such as those of smooth bow position changes. The jitter generators are equivalent to general purpose LFOs set to a noise waveform.

Note: Jitter modulation of pickup positions as the target produces great chorus-like effects.

Jitter generator parameters

- *Rate knobs:* Rotate to set the speed of the modulation (jitter) signal for each jitter generator.
- *Jitter On/Off buttons (1 and 2):* Turn each jitter generator on or off, independently.
- *Target 1 and 2 pop-up menus:* Choose modulation targets 1 and 2, for each jitter generator.
- *Amount 1 and 2 sliders:* Drag to determine the amount of modulation for each jitter source.

Sculpture note-on random modulators

The two note-on random sources are intended for random variations between different notes or voices. Values are randomly generated for each note and remain constant until the voice is released. Such randomizations are useful for adding interest or thickening the sound when you play polyphonically. Note-on random is also useful for emulating the periodic fluctuations that a musician introduces when playing an instrument—even when repeating the same note.

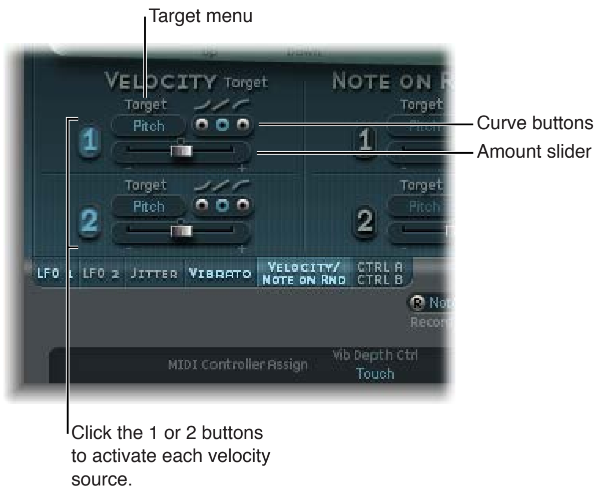
Note-on Random parameters

- *On/Off buttons (1 and 2):* Turn note-on random modulation on or off.
- *Target pop-up menus:* Choose the modulation target—what parameter is randomly modulated when a note is played.
- *Amount sliders:* Move to set the modulation amount—the strength of the modulation.

Sculpture velocity modulators

The excite objects and the filter have dedicated velocity sensitivity controls. Many other modulation routings also allow you to select velocity as a via source.

In some cases, however, it may be useful to directly control other synthesis core parameters by velocity. This can be done in this section—where two independent target/amount/velocity curve slots are available.

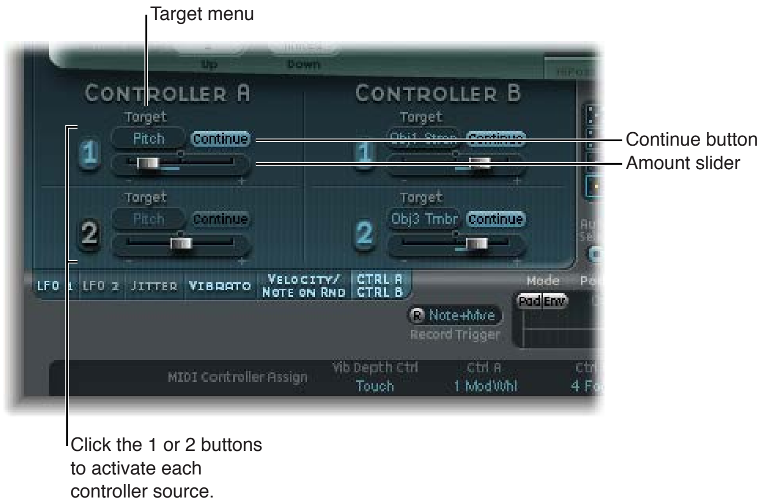


Velocity modulator parameters

- *On/Off buttons (1 and 2):* Turn velocity modulation on or off.
- *Target pop-up menus:* Choose the target parameter that you want to modulate with velocity.
- *Amount sliders:* Move to set the amount, or strength, of modulation.
- *Curve buttons:* Choose from concave, linear, and convex velocity curves.

Use Controller A and B in Sculpture

These parameters define two discrete modulation targets. The modulation intensity, or strength, is assigned to Controller A, Controller B, or both.



Controller A and B parameters

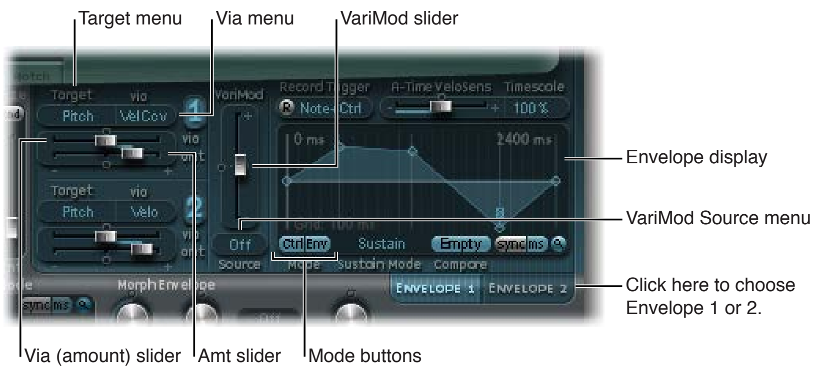
- *On/Off buttons (1 and 2):* Turn the controller A and B modulation sources on or off.
- *Target pop-up menus:* Choose the target parameter that you want to modulate with the specified controller. Each target features a two-state button (the label changes in each state).
- *Continue/Note On buttons:* Select either continuous modulation, or a modulation value that is updated only when a note-on message is received.
- *Amount sliders (1 and 2):* Move to set the amount, or strength, of modulation.

Sculpture envelope parameters

Sculpture envelope overview

Among Sculpture's control sources are two envelopes. In traditional synthesizer designs, envelopes are used to control amplitude and filter levels over time. By comparison, Sculpture's envelopes are somewhat special in that they can be used as:

- Traditional four-segment envelopes
- MIDI controller modulations
- A combination of both—as MIDI controller movement *recorders* with ADSR-like macro parameters, for polyphonic playback



Control envelope parameters

- *Envelope 1 and 2 buttons:* Choose control envelope 1 or 2, and access the parameters of each.
- *On/Off buttons (1 and 2):* Turn the control envelope 1 and 2 modulation sources on or off.
- *Target pop-up menus:* Choose modulation targets 1 and 2. Two targets can be assigned per envelope, with an optional via modulation. Targets include string, object, pickup, Waveshaper, and filter parameters.
- *Via (source) pop-up menus:* Choose the modulation source used to scale the amount of envelope modulation.
- *Amt and Via (Amount) sliders:* Set the modulation amount. In cases where any via source other than "off" is selected, both sliders are available for use.
  - *Amt slider:* Move to set the modulation amount when the incoming via signal is 0—for example, a modulation wheel at its minimum position.
  - *Via (amount) slider:* Move to set the modulation amount when the incoming via signal is at full level—for example, a modulation wheel at its maximum position.
- *VariMod Source pop-up menu:* Choose a modulation source. (VariMod is available only for recorded envelopes.)
- *VariMod slider:* Move to set the amount of envelope variation.
- *Mode buttons (Ctrl and Env):* Select either controller (run mode) or standard envelope behavior. If both are activated, the controller value is *added* to the envelope output, resulting in a modulation offset.

  Note: When the envelopes are used as polyphonic modulation recorders and playback units, each voice is handled independently, with a separate envelope being triggered as each note is played.

- *Envelope display:* Shows the envelope curve, and enables you to record and edit envelopes. See Use Sculpture's envelope display on page 274 and Sculpture active envelope parameters on page 273.

Sculpture active envelope parameters

The following parameters are active only if the envelope is engaged (Mode buttons set to either Env or Ctrl+Env).

Envelope parameters

- *A-Time Velosens slider:* Move to set velocity sensitivity for the attack phase of the envelope. Positive values reduce the attack time at lower velocities. Negative values reduce the attack time at higher velocities.

- *Timescale field:* Drag to scale the duration of the entire envelope between 10% (ten times faster) and 1000% (ten times slower). This also affects the appearance of the envelope curve displayed as it is shortened (sped up) or lengthened (slowed down).
- *Sustain Mode pop-up menu:* Choose the behavior of the envelope while a note is held. Choices are Sustain mode (default), Finish mode, and three loop modes—Loop Forward, Loop Backward, and Loop Alternate. See Loop Sculpture envelopes on page 276.
- *Sync and ms buttons:* Select either a tempo-synced envelope with note value options, such as 1/8th or 1/4, or a free-running envelope (with segment times displayed in milliseconds).

  Note: Switching between values forces a recalculation of times to the nearest note value or ms time, respectively, based on the current project tempo.

- *Compare button:* Toggles between the original recording and the edited version.

  Note: This is available as an option only if an envelope curve has actually been recorded and edited.

- *VariMod-Source and Amount:* Controls the strength of envelope variation with a user-defined modulation source (available only for recorded envelopes).
  - Source options include Off, Velocity Concave, Velocity, Velocity Convex, KeyScale, Ctrl A, and Ctrl B.
  - Variation in the envelopes means the deviation of a recorded envelope path from straight interconnecting lines between the points. After you record an envelope, you can reduce or exaggerate the amplitude-jitter (variation) of the recording by Command-dragging the curves between points down (to reduce) or up (to exaggerate).

Use Sculpture’s envelope display

The envelope curve is shown in the display to the lower right of the Sculpture interface. The envelope display is active only if the envelope is engaged (Mode buttons are set to either Env or Ctrl+Env).

- The overall time/length of the envelope is indicated by the numerical entry at the top right of the window (2400 ms in the figure).
- The maximum time/length of the envelope is 48 bars/40 seconds.
- The lines on the background grid are placed 100 milliseconds apart.
- The background lines are placed 1000 ms apart when very long envelope times are displayed. In sync mode, this is displayed as 1 quarter.
- The envelope is *zoomed automatically* after you release the mouse button. This displays the entire envelope at the highest possible resolution. You can disable or enable this feature by clicking the Autozoom button—the small magnifying glass.
- Autozoom is automatically disabled when you perform a manual zoom—by clicking the envelope display background, and dragging horizontally. As a reference, the current display width is displayed in the numerical entry at the top right of the display. You can re-engage automatic zooming by clicking the Autozoom button.
- If you click the handles (nodes) or lines between the nodes in the envelope display, the current envelope segment becomes highlighted. A small help tag also indicates the millisecond value of the current segment.

Copy, paste, or clear Sculpture envelopes with the shortcut menu

You can copy and paste envelopes between Envelopes 1 and 2, between settings, or between multiple open Sculpture instances. The Clear command deletes the selected envelope.

- Control-click the envelope buttons or the envelope display background, then choose Copy, Paste, or Clear from the shortcut menu.

Use Sculpture envelope nodes

When an envelope is first opened, a default envelope curve is automatically created for each envelope. Click the (Mode) Env button to view it.

A few handles (nodes) are placed—from left to right—along a straight line within the envelope. These are indicators of the following parameters.

- *Node 1:* Start level—cannot be repositioned
- *Node 2:* Attack time position/level
- *Node 3:* Loop time position/level
- *Node 4:* Sustain time position/level
- *Node 5:* End time position/level

As you move the pointer along the line, or place it over the nodes, the current envelope segment is highlighted.

You can create your own envelopes manually by manipulating the nodes and lines, or you can record an envelope, as discussed in Record Sculpture envelopes on page 276.

Adjust the time between nodes

- Drag a handle left or right.

As you drag, the overall length of the envelope changes—with all following nodes being moved. When you release the mouse button, the envelope display automatically zooms to show the entire envelope.

Note: You cannot move a node beyond the position of the preceding node. You can, however, move nodes beyond the position of the following node—even beyond the right side of the envelope display—effectively lengthening both the envelope segment and the overall envelope.

Set the level of each node

- Drag a handle up or down.

Change the shape of the curve between nodes

- For simple (nonrecorded) envelopes, drag the line that connects nodes up or down.
- For recorded envelopes, which may have a more complex curve between nodes, Control-drag the curve.

Loop Sculpture envelopes

The envelope can run in one-shot mode, like any envelope—the envelope phases run for as long as the note is held. It can also run through each phase several times or in an infinite cycle, much like an LFO. You can achieve this through the use of loops.

You can synchronize loops to the project tempo automatically by using the sync and ms buttons.

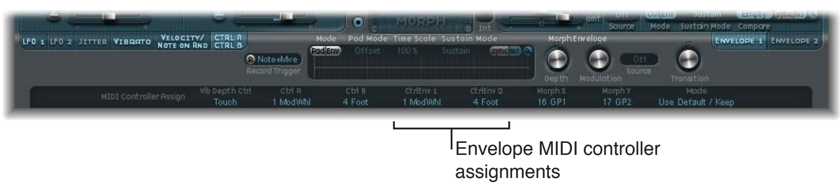
When you are in any of the loop modes, the loop always cycles between user-defined envelope handles that indicate the Loop start point (L icon), and the Sustain point (S icon). You can drag these handles to any position.

- When set to Finish, the envelope runs in one-shot mode from beginning to end—even if the note is released before all envelope phases have completed. The other loop parameters are disabled.
- When set to Loop Forward, the envelope runs to the Sustain point, and begins to periodically repeat the section between the Loop and Sustain points—always in a forward direction.
- When set to Loop Backward, the envelope runs to the Sustain point, and begins to periodically repeat the section between the Sustain and Loop points—always in a backward direction.
- When set to Loop Alternate, the envelope runs to the Sustain point, then periodically returns to the Loop point and back to the Sustain point, alternating in both a backward and forward direction.

Note: If the Loop point lies behind the Sustain point, the loop will start after the key is released.

Record Sculpture envelopes

It is important to note that you can only record the movements of the assigned MIDI controller. MIDI controller assignments for the envelopes must be set in the MIDI Controller Assignment section at the bottom of the Sculpture interface (see Define Sculpture MIDI controllers on page 287).



Envelope recording parameters

- *R(ecord) button:* Starts or stops envelope recording. Recording can also be stopped using the trigger mode function described below.

- *Record Trigger Mode pop-up menu:* Choose different record trigger modes to start—and stop—recording when Record is active.

  - *NoteOn:* Recording starts when a note is played.
  - *Note + Ctrl Movement:* Recording starts when MIDI control change messages arrive while a note is held (for information about assigned MIDI controllers, see Define Sculpture MIDI controllers.
  - *Note + Sustain Pedal:* Recording starts when the sustain pedal is depressed while a note is held.

Record an envelope

1. Choose a Record Trigger Mode, such as Note + Ctrl.
2. Click the Record button (the "R") to start recording.
3. Play, and hold, a key—and start moving the controllers assigned to envelope controls 1 or 2 or both, such as the modulation wheel.

Stop an envelope recording

Do one of the following:

- Click the Record button (the "R") to disengage it.
- Release all *voices*.
- Play a new note after releasing all *keys*.

Note: When a controller movement has been recorded, R(ecord) is automatically set to off and Mode is set to Env. This ensures that only the recorded movement is active, regardless of the stop position of the recorded controller.

Play a recorded envelope

- Play a key to begin polyphonic playback of the recorded envelope.

Note: The Mode parameter must be set to Env and the R(ecord) parameter must be set to off.

You can also turn on both the Env and Ctrl buttons of the Mode parameter, which enables you to use controllers assigned to Ctrl Env1 or Ctrl Env2 to manipulate the envelope in real time, alongside playback of the recorded envelope.

Note: When both Env and Ctrl are turned on, the controller value is added to the envelope output, resulting in a modulation offset.

Prepare a recorded envelope for editing

- The envelope segments and handles are set automatically after recording. Drag the vertical lines that intersect the handles to enable editing.

Note: This will not change the shape of the envelope.

Sculpture morph parameters

Sculpture morph overview

Sculpture has a number of *morphable* parameters, indicated in the Sculpture interface by an orange value bar, rather than a blue or turquoise one. This makes it easy to identify, and edit, the values of these parameters.

All morphable parameters can be independently adjusted and stored in a morph point. In essence, the values of all morphable parameters are captured at a particular moment in time, much like a photograph. You can smoothly change the sound—in a subtle or radical way—by transitioning between up to five morph points.

The Morph Pad and Morph Envelope enable you to create, and precisely control, the movements and blending between morph points.

Morph Pad

Morph envelope

The Morph section consists of two parts:

- *Morph Pad:* Used to display and edit, or draw, morph point paths. There are five morph points—the four corners and the center—plus menu options for randomizing and copying and pasting morph points or Morph Pad states.
- *Morph Envelope:* Used to display and edit morph points—either by segment (with a mouse or trackpad), or recorded MIDI controller movements. For example, you could use a vector stick (Morph X/Y controllers) or drag the morph ball (on the Morph Pad).

The current morph point position is indicated by the ball in the Morph Pad. This can be moved with MIDI controllers, such as a vector stick, or with the mouse. Such movements can be recorded and played back independently—with each voice being morphed in a different way.

The dot indicates the current morph position.

The line indicates the current morph position.

During a morph, the red line in the Morph Envelope timeline shows the current time position, and the Morph Pad displays a moving dot that indicates the current morph position.

Note: The current morph position is only shown if one note is being played.

Use Sculpture's Morph Pad

Morph points in Sculpture's Morph Pad

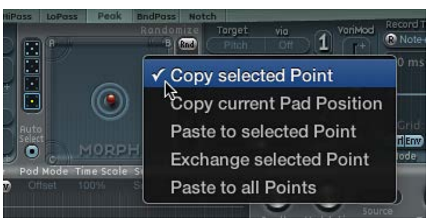
One of the five Morph Pad points (A, B, C, D, and Center) is always selected for editing. This selected point is indicated by two concentric circles that surround it.

When you turn on Auto Select mode, the nearest morph point will be automatically selected when you move the ball in the Morph Pad.

You can also click in the circles around A, B, C, D, or the center, to manually select a Morph Pad point.

Sculpture Morph Pad menu commands

You can open the shortcut menu by Control-clicking the Morph Pad. The menu contains Copy, Paste, and Exchange commands.



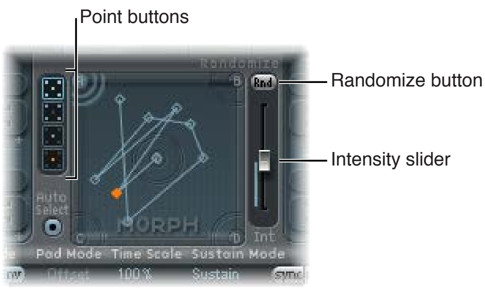
- *Copy selected Point:* Copies the current morph point into Sculpture's Clipboard.
- *Copy current Pad Position:* Copies the current morph state into Sculpture's Clipboard.
- *Paste to selected Point:* Pastes the Clipboard content to the selected point.
- *Exchange selected Point:* Swaps previously copied data with the selected point.
- *Paste to all Points:* Pastes the Clipboard content to all selected points.

Randomize morph points in Sculpture’s Morph Pad

The randomize feature creates random variations of selected morph points. When combined with the copy/paste function, randomizing lends itself to using the Morph Pad as an automatic sound generator.

Use of the Morph Pad can yield interesting composite sounds—hybrids of the original and morphed sound. You can copy this hybrid sound to a corner of the Morph Pad, or to several corners, and randomize it by a definable amount. The morphed sound then becomes a new timbral element that can, in turn, be moved to the corners, randomized, and so on.

In effect, you are “breeding” a sound, while maintaining some control by selecting parent and child sounds. This approach can result in new, complex sounds—even if your sound programming knowledge is limited.



Morph point randomize parameters

- *Point buttons:* Click to define the number of morph points that you want to use for randomization. The active button indicates which points will be randomized.

  The bottom button, when selected, limits randomization to the currently selected morph point.
- *Auto Select button:* Turn on to automatically select the closest morph point.
- *Randomize button:* Click to create randomized values for all parameters of the chosen morph points.
- *Int(ensity) slider:* Move to set the amount of randomization from 1% (slight deviation) to 100% (completely random values).

Randomize morph points

This example provides a general approach that you can follow for morph point randomization.

1. Select a Point button (the top, five-point button, for example).
2. Turn on Auto Select.
3. Drag the Int(ensity) slider to a value of 25%.
4. Click the Rnd button.

   Note the movement of a number of the parameters in the core synthesis engine.

5. Drag the morph ball to each corner in the Morph Pad. Do this along the edges and through the center of the Morph Pad.

   Note how this affects the morph.

6. Strike a few notes on your MIDI keyboard while dragging the ball.

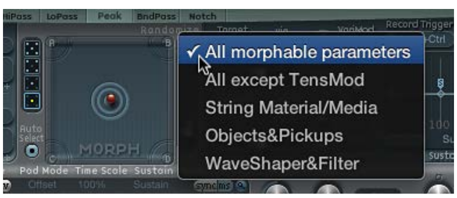
   Note: The morph ball is visible only when the Record Trigger button is active.

As you move the morph ball around, you'll see ghost controls in the Pickup display and the ball in the Material Pad move. If you look closely, you should see a number of red dots moving in the string and object parameters, which indicate the current morph position.

Note that positions on the Morph Pad that fall in between the various morph points cause the randomized parameters to interpolate between values. You can use the Copy and Paste commands to make use of these in-between values.

Sculpture randomize menu commands

You can Control-click the Rnd button to open a shortcut menu containing commands that determine which parameters are randomized with the Rnd button and Int slider.



Randomize menu commands

- *All morphable parameters:* Randomizes all parameters in the following groups, which can lead to unusual sounds. This can lead to some interesting results, but it is uncontrolled, making it unsuitable for the "sound breeding" idea discussed in Randomize morph points in Sculpture's Morph Pad.
- *All except TensMod:* Identical to All morphable parameters, but excludes the TensionMod parameter from randomization.
- *String Material/Media:* Includes the Material Pad position, Stiffness, Inner Loss, Media Loss, Resolution, and Tension Modulation parameters for randomization.
- *Objects&Pickups:* Alters the positions of objects and pickups, plus various object parameters, when randomization is used.
- *Waveshaper&Filter:* Alters the positions of all Waveshaper and filter parameters when randomization is used.

Use Sculpture's Morph Envelope

Use Sculpture's Morph Envelope display

The Morph Envelope contains nine points and eight segments, and it has recording behavior that is much like that of the controller envelopes.

The selected (orange) point in the lower panel (the Timeline) corresponds to the selected point in the Morph Pad trajectory.

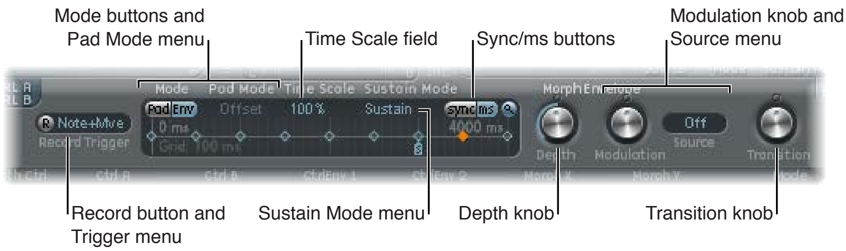
- The overall time/length of the Morph Envelope is indicated by the numerical entry at the top right of the display.
- The maximum time/length of the Morph Envelope is 48 bars/40 seconds.
- The lines on the background grid are placed 100 milliseconds apart.
- If you click the handles (nodes) or lines between the nodes, the current envelope segment becomes highlighted. A small help tag also indicates the millisecond value of the current segment.
- As you move your pointer along the line, or place it over the nodes, the current envelope segment is highlighted.
- You can create your own envelopes manually by manipulating the nodes and lines, or you can record an envelope (see Record Morph Envelopes in Sculpture on page 286).

Adjust the time between nodes

- Drag a handle to the left or right. As you drag, the overall length of the Morph Envelope changes, with all following nodes being moved.

Sculpture Morph Envelope parameters

The following section describes the Morph Envelope's parameters.



Morph Envelope parameters

- *Record button:* Click to arm the envelope for recording. See Record Morph Envelopes in Sculpture on page 286.
- *Trigger Mode pop-up menu:* Choose the event type that triggers recording.
- *Mode buttons:* Turn on the Morph Envelope. See Sculpture's Morph Envelope mode on page 284.
- *Pad Mode pop-up menu:* Choose a Morph Envelope mode.
- *Timescale field:* Drag to scale the duration of the entire envelope between 10% (ten times faster) and 1000% (ten times slower). This also affects the appearance of the envelope curve displayed as it is shortened (sped up) or lengthened (slowed down).
- *Sustain mode pop-up menu:* Choose the behavior of the Morph Envelope while a note is held. The menu items are Sustain mode, Finish mode, three loop modes—Loop Forward, Loop Backward, Loop Alternate—and Scan via CtrlB mode. See Sculpture Morph Envelope Sustain and loop mode on page 285.
- *Sync and ms buttons:* Select either a tempo-synced envelope with note value options, such as 1/8 or 1/4, or a free-running envelope, with segment times displayed in milliseconds.

  Note: Switching between values forces a recalculation of times to the nearest note value or ms time, respectively, based on the current project tempo.
- *Depth knob:* Rotate to scale the amount of morph movement caused by the Morph Envelope. The effect of the Depth parameter is visually displayed in the Morph Pad. As you increase or decrease the value, the morph trajectory is also scaled.
- *Modulation knob:* Rotate to set the scaling amount for Morph Envelope movements.
- *Modulation Source pop-up menu:* Choose a modulation source that is used to scale Morph Envelope movements.
- *Transition knob:* Rotate to control transitions between morph points. This can be the original (possibly recorded) movement to linear, or stepped, transitions. The latter remains at one morph state and then abruptly switches to another morph state at the following envelope point. This parameter (and the Morph Envelope itself) can lead to interesting, evolving sounds, or even rhythmic patches.

Sculpture's Morph Envelope mode

The Mode buttons activate the Morph Envelope and provide a choice of several modes:

- *Both buttons off:* Morph function is disabled.
- *Pad only:* Envelope is deactivated, and morphing is controlled by the morph ball or X/Y MIDI controllers only.
- *Env only:* Envelope is running, but the morph ball and X/Y MIDI controllers are deactivated.
- *Env + Pad:* Envelope is running, and the position of the morph ball or X/Y MIDI controllers is used as an offset for any envelope movements.
- *Offset button:* When in Env + Pad mode, choose from several menu items:

  - *Offset:* The default mode. Behavior is the same as Env+Pad mode: Envelope is running, and the position of the morph ball or X/Y MIDI controllers is used as an offset for any envelope movements.
  - *Point Set:* Envelope is running. The selected envelope point can be edited by moving the morph ball or with a MIDI controller (MorphX and MorphY Controller Assignments).
  - *Point Solo:* Envelope is in a kind of *snapshot* mode. The selected envelope point can be edited by moving the morph ball.

Sculpture Morph Envelope Sustain and loop mode

The Sustain mode pop-up menu lets you choose one of the following modes: Sustain, Finish, Loop Forward, Loop Backward, Loop Alternate, and Scan via CtrlB.

When you are in any of the loop modes, the loop always cycles between the loop and sustain envelope handles—the nodes indicated by the L and S icons. The Morph Envelope can, like any envelope, run in one-shot mode—it runs normally, for as long as the note is sustained. It can also run several times, or in an infinite cycle, much like an LFO. You can achieve the latter through the use of loops.

The Loop and Sustain point handles can be grabbed and repositioned. Note that doing so can potentially alter the loop (and the overall morph envelope) length. The loop modes behave as described below:

- *Finish:* The envelope runs in one-shot mode from its beginning to its end—even if the note is released before the envelope has completed. The other loop parameters are disabled.
- *Loop Forward:* The envelope runs to the Sustain point and begins to repeat the section between the Loop point and Sustain point periodically—always in a forward direction.
- *Loop Backward:* The envelope runs to the Sustain point and begins to repeat the section between the Sustain point and Loop start point periodically—always in a backward direction.
- *Loop Alternate:* The envelope runs to the Sustain point and returns to the Loop point and back to the Sustain point periodically, alternating in both a backward and forward direction.
- *Scan via CtrlB:* The timeline position within the envelope is disconnected from normal, real-time operation, enabling you to manually scan the overall time range with the MIDI controller assigned to Ctrl B (in the MIDI Controller Assign section).

Tip: It is also possible to drag the red time position marker.

Note: If one of the three loop modes is selected, and the Loop point is positioned *before* the Sustain point, the loop remains active until the key is released. Following key release, the envelope then continues beyond the Sustain point, as usual. If the Loop point is positioned *after* the Sustain point, the loop begins as soon as the key is released, and cycles continuously until the complete voice has finished the amplitude envelope release phase.

Record Morph Envelopes in Sculpture

The following section describes the steps you follow to record a Morph Envelope.

Choose a record trigger mode

- Click the Trigger mode pop-up menu to the right of the R button, then choose one of the following trigger modes, which will start recording when R(ecord) Enable is turned on:
  - *NoteOn:* Recording starts when a note is played.
  - *Note + Move Morph Point:* Recording starts when MIDI control change messages (as assigned in the Morph X and Y parameters of the MIDI Controller Assign section) arrive while a note is held.
  - *Note + Sustain Pedal:* Recording starts when the sustain pedal is depressed while a note is held.

Record a Morph Envelope

1. Choose a trigger mode if you don't want to use the Morph Pad.

   Skip this step if using the Morph Pad.
2. Click the R(ecord) Enable button to arm the morph envelope record function.
3. Play a note on your MIDI keyboard, and do either of the following:
   - Drag the silver ball in the Morph Pad.
   - Move an external controller (see Define Sculpture MIDI controllers on page 287).

   Following the recording of a controller movement, R(ecord) Enable is automatically set to off and Mode is set to Env only. This ensures that only the recorded movement is active, regardless of the controller's position or further movements after you finish recording.

   Note: The mode defaults to (Morph) Pad as soon as you click the R button. See Sculpture Morph Envelope parameters on page 283.

Stop a morph envelope recording

Do one of the following:

- Click the R(ecord) Enable button (or trigger) a second time.
- Release all keys, and allow all voices to complete their decay phase. This automatically ends the recording.

Note: You can stop recording early, before the decay phase completes, by releasing all keys and then pressing a single key.

Define Sculpture MIDI controllers

The bottom strip of the Sculpture interface is used to define MIDI controllers—for vibrato depth control or Morph Pad movements, for example. You can use any MIDI controller shown in the menus for these control sources.

These parameters are saved with each setting. They are updated only if the default setting that is loaded on instantiating the plug-in is used, or if the setting was saved with a project. This approach helps you to adapt all MIDI controllers to the keyboard without having to edit and save each setting separately.

MIDI controller parameters

- *Vib Depth Ctrl pop-up menu:* Choose the MIDI controller used for vibrato depth control.
- *Ctrl A and Ctrl B pop-up menus:* Choose two controllers that can be used for side-chain modulations or as via modulation sources—set in the CtrlA and CtrlB modulation routing panes.
- *CtrlEnv 1 and CtrlEnv 2 pop-up menus:* Choose controller assignments for the two control envelopes—used as a modulation signal or an offset, the latter in cases where the control envelope is set to Ctrl only or Ctrl+Env modes. They are also used to define the source for recording controller movements.
- *Morph X and Morph Y pop-up menus:* Choose controller assignments for the x and y coordinates of the Morph Pad. After they are assigned, the controller can be used to manually move the morph point, program single Morph Envelope points, shift the entire Morph Envelope, and serve as a source for recording morph movements.
- *Mode menu:* Choose whether you want to use the default MIDI controller assignments or the assignments loaded from the setting. If you choose Use Default, the assignments remain unchanged. If you choose Load From Setting, you use the assignments you saved with the setting. (The default assignments are taken from the #default.pst setting, if it exists, which is loaded when Sculpture is inserted into an instrument channel strip.)

Learn a MIDI controller assignment

1. Open a control pop-up menu, and choose the Learn item.
2. Move the controller on your MIDI keyboard or MIDI controller.

   Note: If no suitable MIDI message is received within 20 seconds, the selected control reverts to the previous value/assignment.

Sculpture tutorials

Explore Sculpture

Explore Sculpture overview

The following sections contain information to assist you as you start to explore sound creation in Sculpture. See Explore Sculpture's string, Explore Sculpture's objects, Explore Sculpture's pickups, and Other Sculpture processing parameters.

The creation of basic instrument sounds is discussed in Basic sound programming overview on page 293. For a more detailed look at programming particular types of sounds, see Electric bass programming overview on page 303 and Synthetic sound programming overview on page 321.

Given the flexibility of Sculpture's synthesis core, you can take a number of different approaches to sound design.

- If you prefer to build sounds from scratch—parameter by parameter—you can.
- If you prefer to use Sculpture's morphing capabilities to create new sounds, you can. See Randomize morph points in Sculpture's Morph Pad on page 280.
- If you prefer to tweak existing settings, it may be more suitable to use features that affect the entire instrument. See Sculpture Body EQ overview, Sculpture filter parameters, Use Sculpture's Waveshaper, and Sculpture modulation overview.

Whatever approach you favor, you will be able to achieve new and interesting results. Experiment and familiarize yourself with each approach. You will find that each has its strengths and weaknesses, and that a combination of methods may strike the best balance for your needs.

When programming a sound from scratch in Sculpture, the best approach is to work on each component of the sound in isolation. As you're probably new to Sculpture, you won't be familiar with the impact of each parameter on your end results. See String and object interactions in Sculpture.

To start, you will need a plain vanilla setting. When you first open Sculpture, this is exactly what you get—a default set of neutral parameters. It is sonically uninteresting, but provides a starting point for most examples. This setting is saved as the "#default" settings file. It is best to save a copy of this setting before starting.

Save a default setting

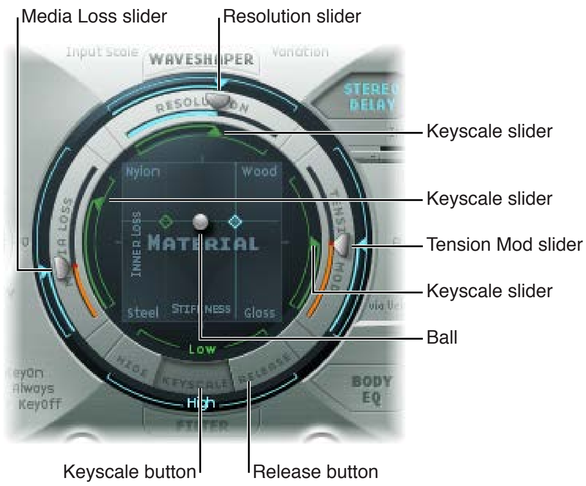
- Chose Save Settings As from the Settings pop-up menu, then enter the name you want—maybe "neutral" or "vanilla"—and click Save.

This setting can be reloaded as you work through the examples.

Explore Sculpture's string

The string is the central synthesis element of Sculpture and is responsible for the basic tone. It offers parameters that enable you to adjust its material—what it's made of, in other words—and to define the environment that it's being played in—water or air, for example.

Tip: Before starting, Control-click the string (the green horizontal line in the Pickup display), then choose "enable string animation" from the shortcut menu. When active, the string vibrates when you play a note, making it easier to visualize the impact of the objects and pickups.



Set a basic tone

1. Click the Keyscale button at the bottom of the Material Pad ring.
2. Strike and hold or repeatedly strike middle C on your keyboard. Middle C is the default pitch of the string.
3. While striking middle C, drag the ball around the Material Pad. Listen to the sonic changes as you move between the Nylon, Wood, Steel, and Glass materials. Keep an eye on the string (the green horizontal line in the Pickup display, to the left) as you're doing so.
4. When you find a basic tone that you like, release the mouse button.

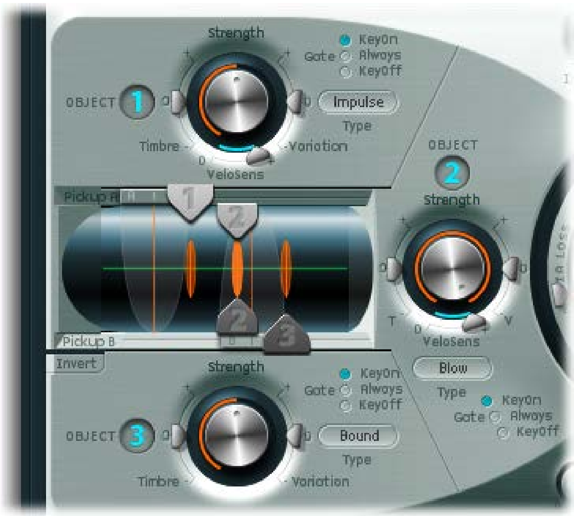
Change the string properties

1. Now experiment with the values of each of the sliders that surround the Material Pad—the Media Loss, Tension Mod, and Resolution parameters—while continuing to strike middle C. Note the changes that each makes to the sound, and also to the string animation in the Pickup display. Play a few notes above and below middle C, again keeping your eye on the string.
2. You probably noticed that moving the Media Loss, Tension Mod, and Resolution sliders also had an effect on the green and blue Keyscale sliders inside and outside the ring. Drag each of these Keyscale slider arrowheads to different positions—one by one—while you play a few notes either side of middle C. Notice the changes that happen up or down the keyboard range.
3. When you're done, click the Release button at the bottom of the Material Pad ring, and adjust the blue Media Loss Release slider while you strike notes.

Explore Sculpture's objects

Up to three objects of different types are used to excite or disturb the vibration of the string.

Tip: Before starting, Control-click the string (the green horizontal line in the Pickup display), then then choose "enable string animation" from the shortcut menu. When active, the string vibrates when you play a note, making it easier to visualize the impact of the objects and pickups.



The three string object dials/controls are shown, along with the Pickup section at the center left.

Explore the objects in Sculpture

1. Reload the #default (or your vanilla) setting file by choosing Reset Setting from the Settings pop-up menu.
2. Click the Object 1 button to deselect it while you are repeatedly striking a key. The sound stops after Object 1 is deselected. The string itself doesn't make a sound unless it is stimulated by at least one object. Click the button again to reactivate it.
3. Choose each menu item from Object 1's Type pop-up menu. Strike a note repeatedly while you choose each item to hear the impact of each object type on the string. Keep an eye on the string animation. Note that Object 1 can make use of excite types only. Object 2 can make use of either excite or damping types. Object 3 can make use of damping types only.
4. Adjust the Strength knob by dragging vertically for large changes or horizontally for fine adjustments. Strike a note repeatedly while doing so.
5. Drag the Timbre and VeloSens arrowheads to different positions while striking a key to audition the changes that they bring.
6. The impact of the Variation parameter is different for each type of object. Feel free to experiment with this as well.
7. Try out each of the Gate settings.

String and object interactions in Sculpture

Each parameter has an impact on the overall tone of the string and—more often than not—an impact on the string interaction of other parameters.

As you introduce or make changes to parameters, the modeled string is affected. This, in turn, affects the interaction of each parameter with the modeled string. Therefore, parameter settings that you already made for Object 1, for example, may need to be adjusted when Object 2 is turned on.

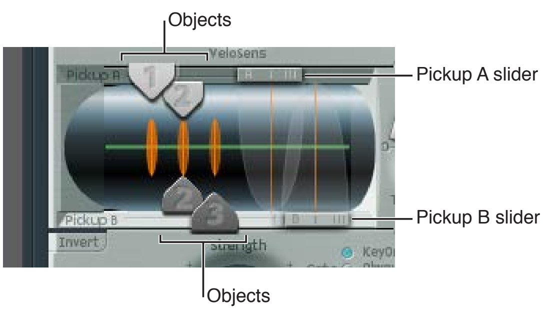
Generally, such adjustments won't need to be radical, and may involve only a small tweak to the Strength parameters, or perhaps to the pickup positions of each object, for example. These parameters have the greatest impact on the tone and level of the objects and should be the first things you look at if enabling Object 2 results in an unwanted change to the color of your sound.

You may want to fine-tune the objects further through use of the Timbre and Variation controls.

Small changes—rather than radical ones—will retain the general tonal character of the string and Object 1, while introducing the new flavor of Object 2.

Explore Sculpture's pickups

The vibration of the string is captured by two movable pickups. The pickup section also houses three object sliders—used to set the position of each object along the string.



Change the object positions along the string

1. Reload the #default (or your vanilla) setting file.
2. Drag the Object 1 pickup left or right while striking a key. Note that adjustment of the object pickup position alters the tonal characteristics of the string.
3. Adjust Object 1's Strength control to hear things better, or to adjust the tone. You can also use Object 1's Timbre and Variation parameters to alter the tone.
4. Feel free to adjust the positions and parameters of the other objects, if they are active.

Change the pickup positions along the string

- Drag the Pickup A and Pickup B sliders. Note that changes to the pickup positions result in quite different string vibrations and tonal qualities. If you want to increase the overall volume, adjust the Level knob on the right of Sculpture's interface, directly opposite the Pickup section.

Other Sculpture processing parameters

From the pickups, the signal is sent to the processing section, which consists of the ADSR-equipped amplitude stage (just to the right of the circular Material Pad in the center), a Waveshaper with selectable types of waveshaping curves (above the circular Material Pad), and a multimode filter (below the Material Pad). All elements covered thus far exist on a per-voice basis.

All voice signals coming from the pickups are summed, and then they are processed by an integrated Delay effect (to the upper right of the circular Material Pad).

From there, the signal is sent to an EQ-like module (Body EQ, to the lower right of the Material Pad), which globally simulates the spectral shape or body response of your instrument. There are several body types to choose from.

The resulting signal is then fed to a Level/Limiter section (at the far right).

Tip: Feel free to experiment with all these parameters—using the #default (or your vanilla) setting file each time. This will give you a general feel for each parameter and its impact on the sounds you hear.

All other parameters on the lower portions of the Sculpture interface (Modulation, Morph, Envelope, and Controller Assignments) are not part of the core synthesis engine, although they can affect it.

Explore Sculpture modulation options

The modulation options can be very important for the emulation of acoustic instruments, such as with the introduction of vibrato into a trumpet sound over time.

Many classic synthesizer sounds also rely as much on modulation as they do on the basic sound source components—the VCO, VCF, and VCA.

Here are some quick modulation tips:

- Imagine that you want to modulate the timbre of Object 2 with the LFO, for example. To do so, click the LFO 1 or 2 tab, click the 1 or 2 button, choose a source and target from the Source and Target pop-up menus, then drag the “amt” and “via” sliders to the values you want.
- To control any modulation with an external controller, such as your keyboard’s modulation wheel, open the “via” pop-up menu and choose Ctrl A (1 ModWhl) or Ctrl B (4 Foot) respectively. By default, the Mod Wheel is set to Ctrl A.
- The Bouncing damp type available to Object 3 affects the sound in an interesting way, but it cannot run synchronously with the project tempo. To create a similar effect to the Bouncing Object—but in sync with the project tempo—you could use a Disturb object type, and move it by modulating its vertical position (Timbre) with an LFO that is synchronized with the host application.

Breath control is available when you use Sculpture, even if you don’t own a breath controller.

Use breath control without a breath controller

1. Record breath controller modulations into the recordable envelopes by using your keyboard modulation wheel or another controller.
2. Reassign the recorded modulation routing to either, or both, the CtrlEnv 1 and CtrlEnv 2 parameters.
3. Choose NoteOn from the Record Trigger pop-up menu.

Incoming note-on messages will trigger the CtrlEnv 1 and 2 parameters.

Create basic sounds in Sculpture

Basic sound programming overview

This section covers the creation of basic types of sounds, such as organs, basses, guitars, and so on. See Acoustic instrument programming examples, Stringed instrument programming examples, and Classic synthesizer programming examples.

For a detailed look at programming particular types of sounds, see Electric bass programming overview on page 303 and Synthetic sound programming overview on page 321.

The idea here is to provide you with a starting point for your own experiments and to introduce you to different approaches for tone creation with Sculpture. As you become more familiar with Sculpture and component modeling, you'll find that there are many ways to achieve an end result. In other words, each component of the sound can be modeled using different techniques and parameters. This flexible approach allows you to create a brass sound, for example, in several ways—using the Waveshaper as a major tonal element in one sound or the filter and Body EQ to emulate the same sonic component in another sound.

It is helpful to have a good understanding of the physical properties of the instrument you are trying to emulate. Although you can do some research on the Internet to obtain this type of specialized knowledge, for most sound creation tasks you can follow the general approach set forth below.

- *How is the sound of the instrument created?*
  - Is it a string that is vibrating and resonating in a box (such as a guitar or violin)?
  - Is it a column of air that is vibrating in a tube (a flute or trumpet)?
  - Is it a solid object that is struck, causing vibration (a woodblock)?
  - Is it a hollow object that is struck, causing vibration or resonance (a drum or bell)?
- *What is the instrument made of?*

  When you answer this question, don't just consider the body of the instrument. Take into account the string material—nylon or steel on a guitar, or perhaps the thickness and material of the reed in a clarinet or oboe, or a mute in a trumpet.
- *Is the instrument polyphonic or monophonic?*

  This is a significant factor and relates to the next question about how the instrument is played. Some differences between monophonic and polyphonic instruments are obvious, such as the inability to play chords on a flute. A more subtle difference involves the way a modeled string will interact with any currently active string. This, of course, can't happen in a flute, which is strictly a one-note instrument.
- *How is the instrument played?*

  Is it bowed, blown, struck, or plucked?
- *Are there other elements that form part of the instrument sound?*
  - Changes in lip pressure and mouth position with brass and wind instruments
  - Breath or mechanical noises
  - Momentary pitch changes—for example, when fingers are pressed into a fretboard, or when a string is plucked
  - Momentary tonal or level changes—such as when brass players are running out of breath, or fluttering the valves

  After you mentally, or physically, construct a list of properties, try to emulate each *component* that contributes to the sound's character. This is what component modeling is all about.

Before you begin, it should be stressed that the examples discussed in the subtopics provide one or two approaches to the task at hand. There are many ways to model each component of the sound. With this in mind, consider the following:

- Experiment with the suggested parameters to create your own versions of sounds. Use your own parameter values if the supplied values don't match your ideal bass sound, for example.
- Subtle changes—particularly to Keyscale parameters—result in more controlled sounds. Take your time, and try everything as you follow the examples.
- Make use of other user settings, and the factory settings—either as a starting point for your own sounds or as an object of study. Looking at existing settings provides an insight into how the sound was created. Enable and disable different parameters to see what each does.

Have fun and take risks—you can't break anything.

Acoustic instrument programming examples

The tasks below provide programming guidelines, tips, tricks, and information to assist you in creating particular types of acoustic instrument sounds in Sculpture.

Create a bell sound

At a basic level, bell-like sounds are quite easy to produce with Sculpture. The creation of truly interesting bells involves a little more effort, but the harmonic richness and detuning during the decay/release phase makes all the difference.

1. Load the #default (or your vanilla) setting file.
2. Choose Strike from Object 1's Type pop-up menu.
3. Drag the Material Pad ball to the very bottom of the pad, and place it halfway between Steel and Glass. Play a few notes, and notice that the sound is already more bell-like.
4. Drag the Media Loss slider nearly all the way down. Again play a few notes, and you'll hear that the release phase of the sound is considerably longer.
5. Drag the Resolution slider all the way to the right.
6. Drag the Pickup A slider to about halfway (0.48).
7. Drag Object 1's pickup position to a value of 0.10. You should be starting to get pretty bells now … play a few notes.
8. To activate the Delay unit, click the Delay button in the upper-right section.
9. Click the Sync button at the bottom of the Delay section, and drag the Delay Time slider to a value of 20 ms.
10. Adjust the Wet Level knob to 66%.
11. Click the Body EQ button in the lower right to activate it. Make sure that Lo Mid Hi is chosen from the Model pop-up menu.
12. Adjust the Low knob to 0.55, the Mid knob to 0.32, and the Hi knob to 0.20.

    At this point, you have a working bell sound, but you'll probably find that there is a tuning issue below C3 in particular. This programming approach was taken because the harmonics of the sound are most noticeable after all other parameters have been set. The solution to the tuning issue primarily lies in the Inner Loss and Stiffness Keyscale parameters.
13. To adjust, first select the Keyscale button, then drag the green horizontal line within the Material Pad up or down for low notes, or drag the blue horizontal line up or down for high notes.
14. Choose Save Setting As from the Settings pop-up menu, save your settings with a new name, and use it as the basis for new bell sounds, or for your next Christmas album.

Create a generic brass sound

Brass instruments are notoriously difficult to recreate with electronic instruments. Samplers do a reasonable job in the right hands, and with the right sample library, but they lack the organic warmth of a real brass player. This is a simple and generic brass setting that can be played as a solo instrument or as a brass section.

1. Load the #default (or your vanilla) setting file.
2. Set Object 1's type to Blow.
3. Activate Object 2, and set its type to Noise.
4. Adjust the Strength of Object 1 to around 0.90.
5. Set Object 1 VeloSens to around 0.30.
6. Drag the Material Pad ball to a position that is diagonally between the "I" of Inner Loss, and the "I" of the word Steel, while playing middle C. The sound should be quite brassy.
7. Play the E above middle C and you'll hear a weird "mandolin meets a telephone ring" kind of sound.
8. Drag the Resolution slider to the left or right while playing middle C and a few notes down an octave or so. You'll discover that a range of sounds that cover everything from sitars to flutes is possible, just through manipulation of this parameter.
9. Click the Keyscale button and—while playing up and down the keyboard—independently adjust the Resolution slider, plus the Resolution Low and High Keyscale sliders until the range of the keyboard you wish to play (an octave or so around middle C, for example) doesn't suffer from those mandolin/phone artifacts. Make sure your sound retains the "brassy" quality.
10. Move Pickup A's position to around 77%.
11. Turn on the Waveshaper and select Scream as your preferred type. Adjust the Input Scale and Variation parameters to taste.
12. Turn on the Filter. Select HiPass mode, and adjust the Cutoff, Resonance, and other filter parameters to taste. (As a suggestion, set Cutoff at 0.30 and Resonance at 0.41).
13. Choose Save Setting As from the Settings pop-up menu and save the setting with a new name.

There are countless directions this sound could be taken in—as a muted trumpet, French horns, and even sitars or flutes.

Make further changes to your brass sound

Do any of the following:

- Use the Waveshaper to radically alter your sound.
- Use the Delay to emulate a space for your instrument.
- Use the Body EQ to cut the lows and boost the Mids and His.
- Drag the Material Pad ball toward the Nylon corner to see how this affects the nature of the sound.
- Choose Blow as Object 2's type, and then experiment with the Object 1 and 2 positions. This can also result in different brass sounds.

Create flute-like sounds

Use this approach as the basis for instruments in the wind family, including flutes, clarinets, shakuhachis, pan pipes, and so on.

1. Load the #default (or your vanilla) setting file.
2. Make sure Keyboard Mode is set to mono, as flutes and other wind instruments are monophonic. After you've created the setting, feel free to experiment with this parameter while playing, and make your choice.
3. Set Object 1's type to Blow.
4. Set Object 2's type to Noise.
5. Set the Gate of both objects to Always.
6. Adjust Object 2's Strength to a value of around 0.25.
7. Adjust Object 1's Velosens parameter to a value around 0.33.
8. Move the Material Pad ball to a position between the end of the Inner Loss text and below the Nylon text.
9. Play the keyboard and you should hear a flute-like sound, but with a long release—which obviously isn't ideal. Drag the Amplitude Envelope Release slider down to around 0.99 ms.
10. Pickup A should be set to a value of 1.00 (far right).
11. Set Object 1's pickup position to around 0.27.
12. Set Object 2's pickup position to around 0.57.
13. Activate the Waveshaper and select the "Tube-like distortion" type.
14. Play a few notes, and adjust the Waveshaper Input Scale and Variation parameters to taste (try Input Scale = 0.16 and Variation 0.55, for example).
15. As you play sustained notes, you may notice a distinct lack of interesting timbral shifts (typical of real flute sounds—due to changes in the player's breath, lip position, and so on) as the note is held.
16. You can use a number of approaches to add interest to the sustained sound. These include using the vibrato modulator (assigned to aftertouch, perhaps), recording or drawing in an envelope, and controlling the Waveshaper Input Scale via Velocity and/or String Media Loss. You could even use the Loop Alternate Sustain Mode. Feel free to experiment!
17. Choose Save Setting As from the Settings pop-up menu and save the setting with a new name.

Create an organ sound

Organ sounds are among the easiest and quickest sounds to emulate in Sculpture, because they have no release phase. This simplifies things in that you don't need to set Keyscaling parameters for the basic tone. You may, however, do so at a later stage—for modulation routing or specific sound design purposes.

1. Load the #default (or your vanilla) setting file. (Object 1's type should be set to Impulse. If it isn't, change it now.)
2. Set the Voices parameter to a value of 8, or higher if you wish.
3. Drag the Material Pad ball to the top-left corner.
4. Activate Object 2 and set the type to Bow.
5. Set the Object 2 Gate mode to Always.
6. Drag the R(elease) slider of the amplitude envelope all the way down.
7. Play a C chord, and you'll hear a flute-like sound.
8. Drag Pickup A to the far right.
9. Play a C chord, and you'll hear a cheesy organ sound. As you can see, Pickup A's position has a significant effect on the overall sonic character of the sound.
10. Drag the Object 2 pickup while holding down the C chord. When you find a position that meets your "that sounds like an organ" criteria, release the object pickup.
11. Very slightly adjust Object 2's Timbre parameter upward.
12. Carefully adjust Object 2's Variation parameter downward and upward until you find a tone you like.
13. You may at this point want to move the Object 2 pickup parameter to another position. Hold down a chord while doing so.
14. You can make further tweaks to the Variation and Timbre parameters of Object 2.
15. To introduce a little key click, change Object 1's type to Strike, and adjust the Strength and Timbre parameters.
16. To add a little of the detuned organ vibe, set the Warmth parameter between 0.150 and 0.200.
17. Choose Save Setting As from the Settings pop-up menu and save the setting with a new name.

You can use this as the basis for your next organ setting.

Tip: Play notes or chords adjusting parameters, so you can hear what each parameter is doing to the sound. You probably notice some intermodulations that are introduced when you're playing chords. Apart from the pitch differences between notes in the chord, this is a result of the interactions between each *voice* being produced by Sculpture. These slight variations between each voice—or string—and their harmonic interactions with each other are not dissimilar to the harmonic interactions of a violin section in an orchestra—even when playing identical lines.

Create a percussion sound

Percussive sounds, such as drums, tend to share a similar type of envelope. They contain a strike element, where most of the sonic character is exhibited, followed by a short decay phase. The release phase will vary depending on the instrument itself—a snare drum as opposed to a woodblock, for example—and depending on the ambient space it is placed in—a cavern, a bathroom, and so on.

1. Load the #default (or your vanilla) setting file.
2. Set Object 1's type to Strike.
3. Activate Object 2, and set its type to Disturb 2-sided.
4. Set Object 2's Gate mode to Always.
5. Object 1's Strength should be about 0.84.
6. Object 2's Strength should be about 0.34.
7. Drag the Media Loss slider up and down while playing to hear its effect. Find a suitable setting.
8. Similarly, you can change the Material Pad ball position—although its effect on the overall tone of the sound is heavily reliant on the Media Loss value.
9. Activate the Body EQ and Filter, then adjust the settings to taste.
10. Choose Save Setting As from the Settings pop-up menu and save the setting with a new name.

Stringed instrument programming examples

The tasks below provide programming guidelines, tips, tricks, and information to assist you in creating particular types of acoustic instrument sounds in Sculpture.

Create a bass sound

1. Load the #default (or your vanilla) setting file.
2. Choose the +1 Oct. parameter from the Transpose pop-up menu at the top of the interface, and play a few notes around C2.

   The general color of an acoustic bass is already there
3. You can certainly drag the ball on the Material Pad toward the Nylon corner, but first choose Pick from Object 1's Type pop-up menu.
4. Play your keyboard, and adjust the ball position while doing so.
5. Take a look at the Strength, Variation, Timbre, and VeloSens parameters of Object 1, and adjust each in turn, to taste.
6. You may also wish to adjust the amplitude envelope's Release parameter (the vertical R slider in the section to the right of the circular Material Pad).
7. To make your bass more woody, adjust Object 1's pickup position toward the right (drag the #1 slider in the Pickup section, which is at the left side of the interface). At extreme positions (the left or right end), you'll find that the bottom end of your bass is lost. Try it out.
8. Adjust the position of Pickup A and Pickup B by dragging the horizontal sliders. As you'll hear, you can quickly recreate a picked acoustic or electric bass sound.
9. To instantly make the sound a hybrid (or full-on) synthesizer bass, click the Waveshaper button (directly above the circular Material Pad and choose one of the types from the Type pop-up menu above the button.
10. Choose Save Setting As from the Settings pop-up menu and save your settings with new names as you go.

You'll probably come up with several new sounds in just a few minutes. Each of these can be used as is, or as templates for future bass sounds you will create.

Create a guitar sound

Guitar, lute, mandolin, and other plucked-type instruments, including harps, can be created from this basic setting.

1. Load the #default (or your vanilla) setting file.
2. Set the Voices parameter to a value of 6—there are only six strings on a guitar. Obviously, pick 7 for a banjo, or as many as possible for a harp.
3. Set Object 1's type to Impulse, if not already chosen.
4. Activate Object 2 and set its type to Pick.
5. Now move Pickup A's position to the extreme right.
6. Move Object 2's Pickup position to a value of 0.14.
7. Activate the Body EQ, and select one of the Guitar models.
8. Adjust the various Body EQ parameters. These have a major impact on the overall brightness and tone of your guitar sound. (Try Model Guitar 2, Intensity 0.46, Shift 0.38, and Stretch 0.20, for example.)
9. Set Fine Structure to a value of around 0.30 to 0.35—let your ears be the judge.
10. Drag the Spread Pickup semicircle vertically to increase the perception of stereo width (a value around the 10 o'clock/2 o'clock mark is nice).
11. Activate the Filter, and select Lo Pass mode.
12. Adjust the Cutoff and Resonance parameters to taste (try both at 0.81).
13. Adjust the Tension Mod slider upward, and play the keyboard to see how the momentary detuning effect caused by this parameter affects the sound. Set it to an appropriate amount.
14. Set the Level Limiter mode to "both."
15. Choose Save Setting As from the Settings pop-up menu and save the setting with a new name.

You may notice that a different approach was taken in the creation of this setting. The reason for this is the major impact that the Body EQ model has on the sound. In some cases, like this one, it may be better to work slightly out of sequence, rather than to strictly follow the signal flow.

Create other guitar-like sounds

Do any of the following:

- Adjust the Object Strength, Variation, and Timbre parameters.
- Reposition the Material Pad ball to create a completely different tone to your guitar.
- Use Delay or Vibrato to emulate the double-strike picking of mandolins.

Create a solo string sound

Solo stringed instruments that are played with a bow, such as violins and cellos, can be created in much the same way. This sound can also be played polyphonically.

1. Load the #default (or your vanilla) setting file.
2. Set Transpose to −1 Oct.
3. Set Object 1's type to Bow.
4. Play the lower half of your MIDI keyboard, and you'll hear a viola/cello-like sound, which could obviously be improved.
5. Set the Object 1 Velosens slider to match your playing style and that of the music, as you're playing the keyboard. Adjust later, if necessary.
6. Drag the Tension Mod slider slightly upward, so that the arrowhead covers the "D." This emulates the momentary detuning effect of the bow stretching the string.
7. Move Pickup A to a position around 0.90.
8. Move Object 1's pickup position to a value around 0.48.
9. Activate Body EQ, and select the Violin 1 model.
10. Set the Body EQ parameters as follows: Intensity 0.73, Shift +1.00, and Stretch+1.00.
11. Adjust the Fine Structure slider to taste.
12. Drag the Spread Pickup semicircle downward until the light blue dots reach the 10:30 and 1:30 positions.
13. Set the Level Limiter mode to "both."
14. Choose Save Setting As from the Settings pop-up menu and save the setting with a new name.

Customize your solo string sound

Do any of the following:

- Set up a modulation, such as a vibrato, that is introduced into the sound after a short period.
- Follow the example above to create higher-pitched solo string instruments, but pay special attention to *all* Keyscale parameters. Careless settings can lead to an out-of-tune violin or viola.
- Use the Body EQ to alter the sound. Take care with settings because they can have a large impact on the upper octaves in particular.
- For a truly radical change (using the example settings above), change Object 1's type to Pick, and you'll have a round and rubbery synth bass sound in the lower octaves and a passable harp across the rest of the keyboard.

Classic synthesizer programming examples

One of Sculpture's great strengths is the ability to create endlessly evolving pad and atmospheric sounds. It can also easily do fat synth basses, powerful leads, and other types of typical synthesizer sounds.

Sculpture has an advantage over traditional synthesizers in that its core synthesis engine produces a wider variety of basic tones, and these tones have an organic quality and richness to them.

The tasks below provide programming guidelines, tips, tricks, and information to assist you in creating classic synthesizer sounds in Sculpture.

Create a basic synthesizer pad sound

1. Load the #default (or your vanilla) setting file.
2. Set the Voices parameter to 16.
3. Set Object 1's type to Bow.
4. Set Object 2's type to "Bow wide."
5. Drag the Material Pad ball to a position at the extreme left of the Pad, exactly halfway between the top and bottom—on a line with the Material label.
6. Play a C chord (middle C).

   You'll hear a pad sound.
7. Move Pickup A to a position around 0.75.

   The pad will become a little sweeter.
8. Move Object 1's position to a value of 0.84.
9. Move Object 2's position to a value of 0.34.
10. Click the Points icon that has five dots in the Morph Pad section.
11. Drag the Int slider in the Morph Pad Randomize section to a value of 25%, for example.
12. Click the Morph Rnd button one time.
13. Choose File > Save Setting As, and enter a new name, such as "vanilla pad," for example.

You'll be using this basic pad sound for several other examples. Don't be shy about doctoring the "vanilla pad"—anything goes, so make use of any of the Filter, Delay, EQ, and Waveshaper parameters to quickly create new sounds.

Create an evolving synthesizer pad sound

1. Load the #default (or your "vanilla pad") setting file.
2. Click the LFO 1 tab at the bottom left of the interface.
3. Click the 1 button, and play the keyboard.

   The difference you will hear is subtle.
4. While holding down a chord, drag the amt slider left and right. Finally settle on a value of 0.15.
5. Choose Object 1 Strength from the Target pop-up menu near the 1 button.

   You'll hear a fluttering sound.
6. Click the sync button, and adjust the Rate knob to a value of 1/8t.
7. Activate the second LFO 1 object by clicking the 2 button, and then choose Object 1 Position from the Target pop-up menu by the 2 button.
8. If you play the keyboard, there's not much that's different.
9. Choose Velocity from the via pop-up menu near the 2 button.
10. Play the keyboard at different velocities, and you'll hear some shifting of the Object 1 pickup position.
11. Choose Sample&Hold from the Waveform pop-up menu, then play the keyboard at different velocities. If you've got a sustain pedal, use it. Listen to the endlessly evolving sound.
12. You might want to experiment with the project tempo and the LFO rate.
13. You may want to alter the Spread Pickup value, and introduce LFO 2 or the other modulators.

Create a morphed synthesizer sound

1. Load the #default (or your vanilla) setting file.
2. Click the R(ecord) button in the Morph Trigger section.
3. Play a chord on the keyboard, and drag the Morph Pad ball in a circle.
4. When you're done, click the R(ecord) button again.
5. Now change the Morph Mode to Env only, and you should see your Morph circle.
6. Play the keyboard. There's your morphed pad.
7. Feel free to adjust the morph envelope parameters.

   If you created and saved the vanilla pad setting discussed in "Create a basic synthesizer pad sound", you were asked to use the Morph Points, Intensity, and Rnd parameters as part of the setting. This was to ensure that there would be several morph points already available for your use when morphing.

   You can, if you like, retain the path of your morphed pad, and continue to click the Rnd button and adjust the Int(ensity) slider for an endless variety of sounds.

Advanced Sculpture tutorial: electric bass

Electric bass programming overview

This section concentrates on a single instrument type—the electric bass, including all of its important variations and articulations. The physical nature of electric basses is not as complex as their acoustic counterparts. This instrument is therefore an excellent choice for the sound programming tutorials, the goal of which is to acquaint you with the art of using Sculpture to accurately reproduce detailed sounds.

Note: To see the settings for these tutorials in the Sculpture window, choose Tutorial Settings from the Settings pop-up menu.

To build a bass and all its components in Sculpture, you need to understand the basic, physical process of sound production within the instrument. In general, the electric bass has four strings. The lowest string is usually tuned to E 0 or E (MIDI note number 28). The strings above the low E are tuned in fourths—thus A, D, and G. There are basses that have five, six, or more strings, but because Sculpture has no tonal limits, this is unimportant.

What is much more important for sound programming is the overtone content of the bass sound, which depends primarily on the qualities of the strings.

- *Round wound strings:* A very fine wire is wound around a steel cable core, which results in a metallic sound that's full of overtones.
- *Flat wound strings:* The fine wire wrapping is ground down or polished smooth, and the sound has far fewer overtones in comparison. (These are much less popular today.)

In contrast to guitar strings, the structure and workmanship are the same for all strings in a set. Sets combining wound and nonwound strings do not exist.

The relationship between string length and string tension has a significant impact on the overtone content. Disregarding basses that can be adjusted to different scale lengths (different vibrating string lengths), the actual playing position that is used plays an important role. When you play D at the tenth fret on the low E string, it sounds more muffled than the same pitch played on the open D string.

The number of frets differs from bass to bass and depends on the scale length. Don't worry about pitches higher than a single ledger line C; the actual functional range of this instrument is primarily in its two lower octaves—between E 0 and E 2.

Also worth mentioning is the fretless electric bass. Like all instruments of this type, it is freely tunable and possesses a distinctive, individual sound. See Program a fretless bass sound with Sculpture.

There are three types of articulations that are discussed:

- *Fingered:* The strings are played with the alternating index and middle fingers.
- *Picked:* The strings are played with a pick. See Program a picked bass sound with Sculpture.
- *Thumbed/Slapped:* The strings are either played with the side of the thumb on the fingerboard or plucked strongly with the fingers. See Program a slap bass sound with Sculpture.

The vibration of the strings is captured by an electromagnetic pickup. When the string is vibrating, its steel core affects the magnetic field. The pickups are almost always found some distance to the side, nearer to the bridge and stop tailpiece. There are different pickup concepts for electric basses, and often two or more pickups are combined to make the sound. To avoid getting into too much detail at this point, there is a rule of thumb that applies: The farther you move the pickup toward the middle of the string, the bassier the sound will be and the more hollow it will sound. The farther you move the pickup toward the end of the string, the more the sound's overtone content will increase, becoming more dense and compact. The sound will have more mid-range frequencies, or buzz, and less bass. If the pickup is positioned at the very end of the string, the sound becomes very thin. This behavior mirrors the actual playing position of a real string: If you play more toward the middle of the string, you get a smooth, even, and powerful sound that contains limited harmonic denseness (overtones). If the string is played at the bridge, the sound develops a nasal twang and features more buzz and more overtones.

Now to the body of the instrument, and its resonant properties. Almost all electric basses have a steel rod running through the neck, to strengthen it, and a body made of solid wood. This construction allows the strings to vibrate relatively freely (sustain), even though very little direct sound is generated. The pickups and the amplifier and speaker systems are responsible for the actual sound of the instrument.

The acoustic interaction between body, strings, and external sound sources is much less complex than with pure acoustic instruments.

The vibration of the strings is, of course, naturally hampered by several physical factors: the radius of motion of the string (the *antinode*) is impeded by the left bridge or by the first fret that's pressed down upon, and the frets in between. This can lead to the development of overtones that can take the form of anything from a slight humming or buzzing to a strong scraping or scratching sound.

In addition, factors such as the material properties of the strings and the instrument, as well as the softness of your fingertips, also serve to dampen the vibration of the string.

Program a basic bass sound with Sculpture

This section covers programming of a basic bass sound, which will serve as the foundation for the different bass sounds you will create. See Program a picked bass sound with Sculpture, Program a slap bass sound with Sculpture, and Program a fretless bass sound with Sculpture.

Sequentially follow the tasks in this section and Refine the basic bass sound to learn how different components can be modeled and to gain a fuller understanding of how Sculpture parameters interact.

Create the proper working environment for design of your own bass sound

1. Make sure the range from C 0 to C 3 is available on your keyboard by either transposing your master keyboard, or by using the Transpose function in the Region parameters of your host application.

   Note: You can, of course, transpose sounds within Sculpture, but this isn't the best solution in this case, for the following reason: Sounds would not be compatible with MIDI regions in which note number 60 as middle C is considered to be the measure of all things.

2. Choose the #default setting from the Settings pop-up menu in Sculpture.

Recreate the sound characteristics of a typical bass instrument

1 Set the Attack value of the amplitude envelope to its minimum value (0.00 ms). The A(ttack) slider is just to the right of the Material Pad.

2 Shorten the Release time of the amplitude envelope to a value between 4 and 5 ms.

Play a key on your keyboard. The note should stop abruptly when you release the key and should be free of artifacts (a digital crackle or snap). If you encounter any artifacts, carefully increase the Release time.

3 Play some sustained notes in the range above E 0. These will die away too quickly. Correct this quick die-out with the Media Loss parameter by dragging the slider to the left of the Material Pad almost all the way down to the bottom. Note that the low E string on a high-quality bass can sound for over a minute.

Your basic bass should simulate a fingered articulation, which means that the sound is created by striking the strings with fingers.

4 Choose Pick from the Type pop-up menu of Object 1.

Don't be confused by the name of the object type; despite the name *pick*, this model is appropriate for simulating the playing of strings with your fingers.

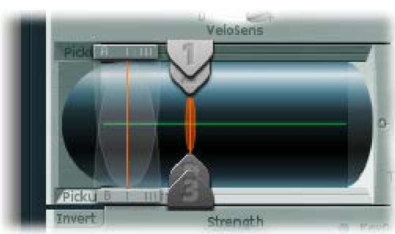
Play some notes in the lower range. You'll hear that the sound is very muffled, hollow, and distorted. Before you adjust further parameters in Object 1, you need to set the position of the pickup.

This is accomplished in Sculpture's Pickup display located to the left of the Material Pad. You'll find three arrow-shaped sliders, representing Objects 1 to 3. The two transparent bell-shaped curves help you to visualize the position and width of Pickup A and Pickup B.

On electric basses the pickups are found quite a way off to the side and near the bridge. This particular bass has only a single pickup.

The behavior of a single pickup is simulated by placing both pickups at exactly the same position.

5 Drag Pickup B to the exact position of Pickup A while keeping an eye on the help tag. The two thin orange lines should overlap perfectly. As a suitable value for the example, set both pickups to 0.10.



Note: Make sure the Invert switch to the lower left of the Pickup display isn't turned on, because this would cause the pickups to completely cancel each other out.

It's now time to determine the playing position.

6 Drag the Object 1 slider in the Pickup display in a horizontal direction. Play the keyboard while doing so, to hear the changes it makes.

7 You'll quickly realize that you can achieve a precise, crisp sound only when you drag the slider relatively far away from the middle of the string. Move Object 1 closer to the pickup (position 0.15 in the figure below).

8 The low notes are still distorted. You can remedy this by adjusting the Level knob to the right of the amplitude envelope. Set a value of −10 dB.

Recreate the material properties of a set of round wound strings

Although you can already recognize the sound of an electric bass, it doesn't sound wiry enough yet. Now it's time to focus on the bass strings themselves.

1 Drag the ball in the Material Pad up and down at the left edge. Pay attention to how the overtones react.

Drag the ball to the lower-left corner. The sound should vaguely remind you of the sound of a low piano string. Because the overtones sustain too long, the tone sounds somewhat unnatural.

2 Drag the ball upward until you hear an acceptable sound. Try the position shown in the figure:

Note: In general, a splaying of the overtones in low wound strings is typical. You can recognize it by the slightly impure, metallic sound. This occurs because the partials (overtones) are not exact whole number multiples of the fundamental frequency but rather are shifted somewhat higher. An example of this effect in the real world of electro-acoustic instruments is the low strings on a Yamaha CP70. This is overkill, but your bass model will benefit from a small amount of this effect.

Splay overtones in Sculpture

1. Drag the ball in the Material Pad gradually to the right.

   The sound takes on a more pure, bell-like character.

2. To realistically simulate the splaying of overtones, try the following example setting:

Emulate string and fret vibrations with Object 2

The vibration of a bass string does not occur in a vacuum. The antinode of the string frequently encounters the natural, physical limitations of the instrument. This is heard as the typical buzzing and rattling that occurs when the strings touch the frets.

1. Activate Object 2, and choose Bouncing from the Type pop-up menu.

   The sound should now vaguely remind you of a mandolin tremolo. This is far too strong an effect for this kind of sound.

2. Move Object 2 all the way to the right (a value of 1.00).
3. Experiment with Object 2's parameters. A discrete and realistic result can be achieved with the following parameter values: Strength 0.33, Timbre –1.00, and Variation –0.69.

   Play some low notes, and you'll find that once again the overtones sustain a little too long—somewhat like the lowest notes or strings on a piano. this can be corrected by dampening the string.

Use Object 3 to dampen overtones

1. Activate Object 3 and choose Damp from the Type pop-up menu.
2. Move Object 3 all the way to the right (value 1.00).
3. Set the Strength parameter to 0.18.

   Note: Experiment with how the Strength parameter of Object 3 interacts with the Inner Loss Material Pad parameter. The higher the Inner Loss value, the smaller the Strength value can be, and vice versa.

Set the range for the basic bass sound

To more realistically replicate the different tonal ranges of the bass, use Sculpture's scaling function.

When turned on, the key-scaling function is used to adjust the timbre of the sound, independent of pitch. Before using the blue sliders to do this, try the Resolution parameter.

- Click the Keyscale button at the bottom of the Material Pad. The key scale below C3 is displayed in green, the range above in light blue. The Material Pad with its Keyscale parameters activated is shown here:

Note: The most relevant performance range for basses is found exclusively below C3. For this reason, you should make use of the green sliders to set the actual timbre of the sound. The primary sliders found around the ring determine the timbre of the sound above C3. For the moment, ignore the blue sliders (which control high key scaling) and simply set them to the same positions as the main sliders.

Refine the basic bass sound

This section covers programming of a basic bass sound, which will serve as the foundation for the different bass sounds you will create. See Program a picked bass sound with Sculpture, Program a slap bass sound with Sculpture, and Program a fretless bass sound with Sculpture.

Sequentially follow the tasks in this section after reading Program a basic bass sound with Sculpture to learn how different components can be modeled and to gain a fuller understanding of how Sculpture parameters interact.

Use the Resolution parameter to control the timbre, independent of pitch

The Resolution parameter is normally used to set the balance between DSP load and sound quality. It can, however, also be used to shape the sound.

1. Play some notes at the higher end of the bass's range (around C2), then drag the Resolution slider all the way to the right and then gradually back toward the left.

   You can hear how the sound loses overtones yet simultaneously becomes louder. At low Resolution values, an inharmonic metallic rattling is heard in the sound.

2. Increase the Resolution value until the metallic rattling disappears. Set the slider to the following position:

3. Play some notes in the bottom range (around E 0). You'll note that the sound is quite muffled and vintage-like. Move the green Low Keyscale slider (found below the main Resolution slider) all the way to the right; the low range should now sound a little more wiry.

   With most stringed instruments, the overtone content decreases as the pitch becomes higher. Strictly speaking, this is true only of open strings, and even then in a limited sense. If the strings are fingered, the length of the string is shortened, especially in the high register, and the effect becomes more significant.

Use the Inner Loss parameter to scale the overtone content, dependent on pitch

1. Move the Material Pad ball above the words *Inner Loss*. Try to move the ball solely in a vertical direction to maintain a constant Stiffness value.
2. Drag the green line next to the ball toward the bottom until the small green diamond is located directly above the word *Steel*.

   When playing, you'll recognize the smooth transition that takes place between the wiry, overtone-rich sound at the bottom end and the extremely dampened sound in the upper register. This exaggerated setting was chosen to clearly demonstrate the scaling principle in stringed instruments. To achieve an authentic sound and timbre, try the following setting:

Set sustain levels for the basic bass sound, dependent on pitch

In basses in particular, low notes sustain far longer than high notes. Sculpture allows you to authentically and convincingly simulate this behavior with the Media Loss parameter.

1. Play a few held notes in the range around C2 and above. You'll hear that these notes die out much too slowly. Drag the Media Loss slider up until this range begins to fade out quickly enough. The downside is that the lower notes now die out too quickly.
2. Drag the green Media Loss Key Scale slider down until the fade-out phase of the lower range is sufficiently long.
3. Compare your results with these recommended values:

You've now created a basic bass that's articulated with your fingers. Save this as *E-Bass Fingered Basic*. You'll be using this basic bass as a foundation for the construction of further bass sounds.

The scope for sound design, by altering the frequency spectrum of electromagnetic instruments, is far more flexible than that offered by acoustic instruments. In addition to the number of pickups, a major role is also played by the choice of amplifier, the equalization setting within the amplifier, and—last but not least—the physical properties of the speakers and their enclosing cabinet.

The central features of your electric bass sound are complete, but the sound can be improved by paying close attention to some details. Here are a few general suggestions:

- Vary the position of the pickups. Try placing each of them in different positions. This cancels out certain frequencies, and others are summed together.
- Try turning on the Invert switch, even though this effect is not typical for electric basses.
- What is typical for bass sounds is the placement of the pickups in the outer-left third of the string model. The farther you move them to the left, the thinner and more nasal the sound becomes.
- Shifting Object 1 will have a similar effect. Try different combinations here as well.

Alter the frequency spectrum of your basic bass with the Body EQ

The Body EQ is ideal for giving the bass sound that final, finishing touch. Your electric bass sound could be a little less smooth, and a bit more precise in its attack phase. Bassists like to use the terms *drier* and *more bite* to describe this phenomenon.

1. Load the E-Bass Fingered Basic setting.
2. Choose the standard Lo Mid Hi model from the Model pop-up menu in the Body EQ section.
3. Reduce the low bass frequencies by setting the Low knob to a value of –0.30.
4. Boost the mid-range frequencies substantially by setting the Mid knob to a value of 0.50. Drag the Mid Frequency slider to a value of 0.26.
5. You'll probably find that the boosting of the low mid frequencies is a little too strong at this point, so return the Mid value to 0.30.

6. The sound could stand to be a little more wiry, so set the High knob to a value of 0.30.
7. To finish off, set the Level knob (to the right of the amplitude envelope) to a value of –3 dB.

   The sound is now as loud as possible, without the low notes distorting.
8. Save this sound setting as *E-Bass Fingered Basic EQ1*.

Program a picked bass sound with Sculpture

The basic bass is played with the fingers. In the following example, you will simulate playing the strings with a pick, using the Pick object type. The Timbre parameter will be used to adjust the relationship between the speed and intensity at which the string is struck. The Variation parameter will be used to define the virtual material density, or hardness, of the pick.

If you imagine the fingers to be very soft picks, it makes sense to alter the Pick parameters so that a hard plastic pick is the outcome.

Simulate playing with a pick

1. Load the E-Bass Fingered Basic setting.
2. Set the Timbre parameter of Object 1 to its maximum value of 1.00.

   The attack is now stronger.
3. Try several different Variation settings to get a feel for the material qualities of the pick.

   Note: Not all positions will deliver usable results for the entire range of the instrument.
4. You'll get a consistent, working setting for the two octaves above E 0 with the following parameter settings: Position 0.17 (Pickup display), Strength 1.00 (maximum), Timbre 0.90, and Variation 0.56.

   When these settings are used, you'll find that the sound has become softer and very thin. In fact, it's somewhat reminiscent of a clavinet.

Compensate for thinning side effects with the Body EQ

1. Activate the Body EQ and add a healthy portion of bottom end to the sound by setting the Low parameter to 0.60. Mid should be set to 0.33.
2. Set the High knob to −0.45 because the sound is now so bright that rolling off a few of the highs can't hurt.
3. Bring the volume into line. If you adjust the Level knob to 2.5 dB, nothing should be distorting. If this isn't the case, try reducing some more of the bottom end with the Low knob.
4. Save this setting as *Pick Open Roundwound*.

Emulate bass guitar damping

Playing with a pick is often combined with a damping technique that employs the ball of the thumb. The right hand, which also holds the pick, should physically lie on top of the strings at the bridge. This technique results in the sound having less overtone content but becoming more percussive and punchy at the same time. You can variably control the timbre of the sound through the angle and pressure of your hand while playing.

Object 3 will be used to emulate the virtual ball of the thumb in this example. The Timbre parameter determines the kind of damping that occurs, and Variation dictates the length of the string section that is being dampened.

1. Set the Object 3 type to Damp.
2. Set Object 3's Strength parameter to 0.50.
3. Move Object 3 a little bit to the right in the Pickup display (to position 0.95) to simulate the width and position of the ball of the thumb lying on the bridge.
4. Set Timbre to its minimum value (–1.00) to achieve a very soft damping effect.
5. Set the Variation parameter to its maximum value of 1.00.

   A metallic ringing occurs during the attack phase and still can be heard in the octave above E0.
6. To suppress the ringing, move the small green diamond on the Material Pad to a position directly under the ball. In doing so, you've just increased the Inner Loss value for the low key range.

   Note: To place the diamond exactly under the ball, you can click it while pressing the Option key.
7. Save this setting as *Pick Bass Half Muted*.

Simulate harmonics created by fingers lightly touching the strings

Harmonics are single partials (overtones) of the overall sound. They can be heard by damping certain points along the string. This is done by lightly laying the fingers of the left hand (assuming a right-handed bass player) on the string—not pressing down—before the note is articulated. The first overtone, the octave, is achieved by placing your finger at the exact middle of the string—in effect separating the string into two halves. The next overtone is the fifth above the octave, and the position of your finger should divide the string into a ratio of one-third to two-thirds. The next overtone separates the string into proportions of one-quarter to three-quarters, and so on.

1. Object 3 is used as a damper. Choose the Damp type.
2. Adjust Object 3's Timbre parameter to its maximum value of 1.00.
3. Adjust Variation to its initial value of 0.00 by clicking the Variation slider while holding down the Option key.
4. Move Object 3 to the exact middle (0.50) of the Pickup display. Play the keyboard, and you'll hear the first overtone as a harmonic.
5. While playing, very slowly move Object 3 toward the left of the Pickup display. In doing this, you are effectively scrolling through the overtone series, so to speak.
6. Save this setting as *Flageolet Xmple*.

Emulate a vintage flat wound pick bass

1. Load the Pick Bass Half Muted setting.
2. Drag the Material Pad ball upward and the sound becomes more muffled.
3. Increase the Object 3 Strength parameter to 0.70. The result is a muted pick bass with flat wound strings.

   Tip: If you turn off Object 3, you'll hear a sound that is reminiscent of a 1970s Fender Precision Bass.
4. Save this setting as *Flatwound Pick Damped*.

Emulate a Bert Kaempfert-style percussive bass

1. Turn Object 3 back on.
2. Move both pickups a little to the left (position 0.08).
3. Move the virtual pick (Object 1) a little farther to the outside (position 0.10).
4. Enhance the sound with the Body EQ by turning the Low knob to its maximum value (1.00).

5. To remove the smacking in the attack phase, use the graphical display to choose a value of 0.48 for the Body EQ Mid frequency, then use the knob to increase this value to 0.51. Option-click the Body EQ High parameter to set it to a value of 0.00.
6. Save this setting as Easy Listening Pick Bass.

Program a slap bass sound with Sculpture

You're actually dealing with two different articulations here. The low notes originate when the thumb literally slaps the strings on the upper part of the fingerboard. The high notes are produced when the strings are strongly plucked or popped with the fingers. This is achieved by hooking a finger under the string, pulling it away from the instrument, then allowing it to slap back onto the fingerboard. In conjunction, these articulation methods make up the typically aggressive and overtone-rich slap bass sound.

Emulate a slap bass sound

1. Load the E-Bass Fingered Basic EQ1 setting.
2. Turn off the Body EQ.
3. Also turn off Object 2 and Object 3 for now.

   Because the basic sound of a slap bass is brighter than a standard fingered bass, you need to adjust some Material Pad settings.

4. Return the Low Keyscale parameter to its initial value by Option-clicking the small green triangle (found below the main Resolution slider).
5. Drag the ball down a little, and the sound becomes more wiry. The ball should now be directly above the word "Steel" on the horizontal axis.

   From the models at your disposal, Strike is the most suitable for simulating a thumb physically striking the strings from above. This model is not, however, as appropriate for the slapped (popped) strings. It makes the most sense to choose the Pick model for this purpose.

6. To be safe, turn the Level knob to −25 dB.
7. Choose the Pick model for Object 1.
8. Drag Object 1 to position 0.90 in the Pickup display. This position corresponds to a playing position above or on the fingerboard.

   Note: Given its universal concept, Sculpture will not react exactly like a bass, where one would tend to play in the middle of the string on the upper part of the fingerboard. Try moving Object 1 to this position and see how it sounds. You'll find that the sound is a little too smooth.

Set the parameters for object 1

1. Set Timbre to a value of 0.38, which corresponds to a rapid attack.

   Timbre determines the angle of the obstacle to the string.

2. Set the Strength parameter to 0.53.
3. Set the Variation parameter to –0.69.

   This defines the softer material that constitutes the fleshy part on the side of your slapping thumb. Put more technically, Variation defines the type and degree of reflection.

4. Choose Bound in the Object 2 Type pop-up menu to emulate the typical bright rattling that is created when the string strikes the fingerboard.

   Bound limits the antinode of the string in exactly the same way as the fingerboard on a real electric bass.

Adjust object 2 parameters

1. Set Timbre to 0.39.

   This corresponds to a fingerboard that runs almost parallel to the string.

2. Set the Strength parameter to 0.33.

   Note: Try some higher values as well. You'll see that the sound becomes softer and softer until it's completely dampened by the obstacle.

3. Set Variation to 0.64. Despite the overtone-rich reflection, the string can still vibrate freely.

   Note: Try some negative values—you'll see that the reflections can no longer develop in an unhindered fashion.

4. Set the Level knob to –3 dB. The Bound obstacle has made the sound softer.
5. Notice that the sound is still too smooth for a real slap bass, so try using the Body EQ again. Turn on the Body EQ, and adjust the parameters as follows: Low 0.25, Mid 0.43, High 0.51, and drag the Mid Frequency slider to 0.59.
6. Save this sound as Slap Bass Basic#1.

Program a fretless bass sound with Sculpture

With the exception of shared playing techniques, the fretless bass differs from a normal bass through its buzzing, singing sound. Because the frets on the fingerboard of a standard bass function as a collection of mini-bridges and allow the string to vibrate in an unobstructed fashion, the direct collision of the string's antinode with the fingerboard on a fretless bass is responsible for its typical sound. The string length on a fretless bass is markedly shorter than the string length on an acoustic double bass. The upshot of this is that a controlled buzzing is produced, even when a fretless bass is played with a weak attack. This buzzing can be consistently reproduced in the high register, even on fretless basses that have very short string lengths. The use of the comparatively soft tip of your finger—instead of a hard, metallic fret—to divide or shorten the string also plays a role.

Program a fretless bass

1. Load the E-Bass Fingered Basic EQ1 setting.
2. Turn off Object 3. You'll come back to it later.
3. Choose Disturb from the Object 2 Type pop-up menu.

   Tip: In the Disturb model, the Timbre parameter determines how far the string is deflected from its resting position by the obstacle. Positive values precipitate no deflection of the vibration from its resting position. Variation defines the length of the string section that is disturbed—positive values correspond to a longer section of string, negative values to a correspondingly shorter section of string.
4. Adjust Object 2's parameters to the following values: Strength 0.14, Timbre –0.05, Variation –1.00.
5. Drag the Object 2 slider, which remains at the far right in the Pickup display, to see its value of 0.99. You'll note that the range between C2 and C3 already sounds quite acceptable, but the buzzing in the lower notes is still too strong. It is somewhat sitar-like, so keep this disturb model in mind when it comes to creating a home-spun sitar.
6. Try different settings for the Strength parameter for both the higher and lower playing ranges. You'll see that, at best, only a compromise is possible. The buzzing is either too loud in the low range or not present enough in the high range.

   Obviously, the effect needs to be scaled over the relevant tonal range. Unlike the parameters for the string, Objects 1 to 3 don't have a directly addressable key scaling function. There's a clever way around this: Both LFOs offer a key scaling function. As you probably don't want the buzzing to be modulated by a periodic oscillation, you need to reduce the LFO speed to infinitely slow or 0. In this way, you can deactivate the LFO itself, but still use its modulation matrix.
7. Activate LFO2 by clicking the LFO2 button at the bottom left, and set the Rate knob to a value of 0.00 Hz.
8. Click the 1 button (next to the RateMod slider, to the upper right) to activate the first modulation target.

9. Choose Object2 Strength from the Target pop-up menu.
10. Choose KeyScale from the via pop-up menu.

11 Drag the amt slider to the right while you are playing. You will hear that the singing buzzing fades out in the lower range, while gradually being retained as you move toward C3. Drag the slider to a value of 0.15. The buzzing is now far more moderate in the low range.

12 Switch Object 3 back on. Set Timbre to its minimum value (–1.00) and Variation to its maximum value (1.00). Object 3 should be positioned all the way to the right, at a value of 1.00.

13 Vary the Object 3 Strength parameter. You'll discover that the overtone content of the buzzing can be controlled very effectively. A Strength value of 0.25 is recommended here.

14 Save this setting as Fretless Roundwound#1.

Add effects to your Sculpture bass sound

Detuning and ensemble effects are normally achieved using a modulation effect or by combining doubling and detuning. When you are using a fretless bass for a solo part, a broad chorus effect adds a nice touch.

Because Sculpture can synthesize only one note at a time at any given pitch, simple doubling isn't an option. There are, however, alternatives for bringing movement and life into the sound. Almost all of the *type* parameters of the different objects can be modulated by LFOs, resulting in a vast number of possible combinations.

As a rule, basses are mixed without effects (dry), but a small amount of reverb can be quite appealing on a fretless bass, when used as a solo instrument. Use Sculpture's Delay section to emulate this. Heavy delays are, however, used on basses in "dub" style reggae.

Emulate a chorus effect by modulating the pickup positions

1 Make sure the Fretless Roundwound#1 setting is loaded.

2 Adjust the position of Pickup B to 0.20.

3 Drag the Spread Pickup semicircle—which is beside the Level knob—upward.

Both of the light blue dots move downward toward the letters L and R.

You can hear how the stereo breadth of the fretless sound has increased. Pickup A is sent out on the right channel, while Pickup B occupies the left channel.

Note: Although only modern basses offer such stereophonic features, it's still fun to process conventional sounds with this effect. Note that not all pickup positions are monophonic-compatible. You can check this by returning the Spread Pickup setting to monophonic—click the Spread Pickup semicircle while pressing the Option key.

Animate the pickup positions

1 Select LFO1.

2 Click the 1 button (next to the RateMod slider, at the upper right) to activate the first modulation target.

3 Choose Pickup Pos A-B as the modulation target.

4 Set the Rate knob to 1.00 Hz.

5 To hear the effect, you need to set the modulation intensity (amount). Familiarize yourself with this effect by moving the amt slider gradually to the right. Set it to a final value of 0.15, a moderate rate that doesn't wobble too much.

6 Save this setting as Fretless Chorus Dry.

Tip: At the maximum stereo breadth, effects based on detuning are not as prominent, especially when the beats heard in the sound result from signal differences between the left and right channels. This is valid only to a certain degree, because the motion of the pickup doesn't create a true chorus or harmonizer effect. Try it out and see what happens when the stereo breadth is reduced a little. Also test other modulation targets, such as Pickup Pos A+B, Pickup Pan A+B, Pickup Pan A–B, and String Stiffness.

Create an unobtrusive atmospheric space

This example shows that the Delay section can be used as a substitute reverb for small spaces. For sophisticated reverb effects, it's best to process Sculpture's output with a Logic Pro reverb plug-in.

1 Load the Fretless Chorus Dry setting.

2 Click the Delay button to turn on the Delay section.

3 Drag the Input Balance slider to 1.00.

4 Click the small Sync button—directly to the right of the Delay Time slider—to deselect it, which switches off the tempo synchronization of the delay.

5 Drag the Delay Time slider to 90 ms.

6 Set the Xfeed knob to 0.30.

The individual reflections are still too brash. To make the effect more discrete and unobtrusive, adjust the frequency spectrum and amplitude of the reflections. Start with the frequency spectrum.

7 Drag the LoCut slider to 200 Hz and the HiCut slider to 1000 Hz in the Delay section.

The LoCut parameter at 200 Hz excludes the low frequencies in the reflections, thus preventing a muddy sound. The comparatively drastic cut to the highs with the HiCut parameter blurs the individual reflections, thereby creating the impression of a small room with soft surfaces.

8 Set the Wet Level knob to 25% to reduce the total level of the effect.

9 Save this setting as Fretless Chorus+Ambience.

Create a "drowned in delay" effect

1. Reload the Fretless Chorus Dry setting.
2. Switch the Delay section on.
3. Drag the Input Balance slider all the way to the right, to 1.00.
4. Set the Delay Time value to 1/4t (quarter-note triplet).
5. Set the Feedback knob to a value of 0.20.
6. Adjust the Xfeed knob to a value of 0.30.
7. Drag the LoCut slider to 200 Hz and the HiCut slider to 1600 Hz.
8. Now adjust the overall level of the effect—try setting the Wet Level knob to a value of 45%.
9. Vary the stereo position and rhythmical structure of the delay, by moving the small light blue diamond around the Delay Pad.
10. Save this setting as Fretless Chorus+Wet Delay.

Advanced Sculpture tutorial: synthesizer sounds

Synthetic sound programming overview

The Electric bass programming overview on page 303 section covers programming of natural bass sounds by authentically reproducing the real physical interaction that occurs between a string and the exciting agent that acts upon it. While producing such lifelike models is undoubtedly a forte of Sculpture's architecture, its sonic capabilities extend to the creation of very different sounds as well.

Sculpture contains a number of functions you can use to create new and novel synthesized sounds. This includes the Morph Pad, which can be automated, as well as recordable and programmable envelopes that can be used in a rhythmic context. See Create morphed sounds in Sculpture.

Such features are usually unnecessary when reproducing natural bass sounds, because no electric bass that exists can alter the tonal characteristics of the string during the decay phase of a note—perhaps from wood to metal—and rhythmically synchronize this change to the tempo of the project. These functions are useful, however, when creating sustained, atmospheric sounds where slow and interesting modulations help them come alive. See Create a sustained synthesizer sound with Sculpture.

Create a sustained synthesizer sound with Sculpture

First, load Sculpture's default setting—the very simple sound consisting of a plucked string that vibrates and fades away. This sound needs some drastic changes to become a sustained or extended pad sound.

Examine the three objects. Notice that only Object 1 is active, and acts on the string with an Impulse object type: the string is briefly excited when the note is played, then the sound decays. A sustained pad sound requires an exciting agent that constantly acts upon the string. The appropriate object types are Bow or Bow wide (the string is played with either short bow strokes or long, extended bow strokes), Noise (excited by a random noise signal), or Blow (excited by being blown—much like a clarinet or flute).

Test the above mentioned object types one after the other. Drag the Object 1 Pickup slider, responsible for the exact position of the exciting agent, up and down the string while you're playing. You will come to two conclusions: First, the sound is now sustained for as long as you hold down a key. Second, dragging the Object 1 slider with the Bow type selected results in the most pronounced sonic changes. This setting promises the most rewarding possibilities for varying the sound, and that's why this type has been chosen.

The sonic variations created by the Bow type are very appealing when the virtual bow stroke is moved along the string. You can control this movement by using an envelope, thus creating the foundation of your pad sound.

It makes more sense and is more convenient to record the envelope rather than program it, even if it is easy to program it with the graphic display.

Record an envelope

1 Make sure you have loaded the default setting, then drag the Object 1 slider all the way to the left. Starting from this position, where it generates only an overtone-rich scratch, start animating it by using the envelope.

2 Locate the Envelope section in the lower-right corner of the Sculpture window. Select the first of the two envelopes by clicking the envelope 1 button to select it. In the left part of the Envelope section, notice the two routing possibilities that are used to assign a modulation target to the envelope.

3 Click the 1 button to activate the first routing link, and choose Object1 Position from the Target pop-up menu as the modulation target. Drag the amt slider all the way to the right to set the modulation intensity to its maximum value.

The envelope can now be recorded. It is assumed that your MIDI keyboard has a modulation wheel that outputs the corresponding MIDI controller message (CC number 1) and that option 1 ModWhl is selected for control of Envelope 1 (choose 1 ModWhl from the CtrlEnv 1 pop-up menu in the dark bottom edge of the Sculpture window).

4 Click the R button near the top of the Envelope section below Record Trigger, to prepare the envelope for recording. Choose Note + Ctrl Movement from the Record Trigger pop-up menu.

This option specifies that the recording of the controller messages from the modulation wheel begins the instant the first note is played.

5 Play a note when you want to start the recording, and move the modulation wheel slowly upward while keeping the key on the keyboard depressed. Notice the sound variations you create while moving the modulation wheel.

6 At the end of the recording, return the wheel to its initial starting position and, after you release the depressed note, click the R button to deactivate the recording mode.

Increase stereo breadth and chorus

To give the very dry-sounding "0001 raw pad" setting a little more stereo breadth and chorus effect, modulate the Pickup positions, and assign them to the left and right channels.

1. Load the "0001 raw pad" setting.
2. Drag the Spread Pickup semicircle upward until the light blue dots come to rest near the line that separates both semicircles.

   This separates the stereo pan positions of the Pickups.

3. Click the 1 and 2 buttons to activate both of the modulation links in LFO1.
4. For the first link, choose PickupA Position from the Target pop-up menu, and then drag the amt slider to a small positive value of about 0.03 Hz to modulate the position of Pickup A.

5. For the second link, choose PickupB Position from the Target pop-up menu, and then drag the amt slider to a small negative value of about –0.03 Hz to modulate the position of Pickup B.

   You will hear a pleasant beating or chorus effect in the sound, which makes it broader and more full, alleviating the unpleasant, dry character. Another unpleasant aspect is that the sound is too strong in the mid frequency range and could use some equalization. You can use the Body EQ to correct this.

6. Activate the Body EQ, and experiment with the Lo Mid Hi model—the standard setting. Try reducing Mid to –0.5 and dragging the Mid Frequency slider to 0.37.

7. To give the pad a little depth, activate the Delay. Set the Delay Time to 1/4 and adjust the Xfeed knob to 30%.

The pad now has a pleasant and unobtrusive ambience; you can leave the other Delay parameters at their original values.

Make the sound more lively using the jitter modulators

You can make the sound more animated with some subtle modulation, which makes the jitter modulators the perfect tool for the job. The jitter modulators are basically LFOs that use a random waveform.

1 Click the Jitter button below the LFO section to activate the display for both of the jitter modulators.

2 Click the 1 button to turn on the first link in Jitter 1, and choose Object1 Timbre from the Target pop-up menu.

3 Drag the slider below the Target pop-up menu to –0.40 to adjust the Intensity, and reduce the Rate parameter to 1 Hz. There should be subtle inconsistencies in the pressure applied by the bow to the string. To better recognize this effect, temporarily increase the Intensity level.

   You can use the second jitter modulator for random position deviations with the modulation target Pickup Pos A+B (pickup position A and B).

4 Activate Jitter 2 and choose the Pickup Pos A+B setting from the Target pop-up menu.

5 Drag the slider below the Target pop-up menu to an Intensity of about 0.2, and adjust the Rate knob to 1.5 Hz. As you increase the Intensity, the sound develops a distinct clinking or rattling—adjust this effect to taste.

You now have a satisfactory pad sound, which you should leave alone at this point, even though a few Sculpture features such as the Filter and the Waveshaper lie idle—not to mention the two additional Objects—but sometimes it's a good idea to quit while you're ahead.

Create morphed sounds in Sculpture

The Morph Pad is in the middle of the lower part of Sculpture's window. Each corner of the Morph Pad can contain a different setting for a diverse number of parameters. You can cross-fade between these settings and morph the sound by dragging the red ball in the center of the Morph Pad.

Vary the sound with the Morph Pad

1. To start, Control-click the Morph Pad, then choose Paste to all Points from the shortcut menu to copy the current setting into all four corners of the Morph Pad. (If Paste to all Points is unavailable, first choose Copy selected Point.)
2. Drag the ball to one corner to select the corresponding partial sound, indicated by highlighted arches that appear in the corner.
3. Adjust several parameters.
4. Repeat steps 2 and 3 in each corner.
5. Carefully drag the ball around the Material Pad to find a position where your pad sound takes on a new and interesting character. Also try the extreme corners, for example.

After you choose different settings for the Morph Pad corners, moving the morph ball creates marked sound variations—even though the intermediate stages do not all exhibit a tonal character. You can automate the morphing process by assigning two MIDI controllers to the MorphX and MorphY pop-up menus at Sculpture's bottom edge. You can also automate the Morph Pad using a recorded envelope (for more information, see Record Morph Envelopes in Sculpture on page 286).

Use the randomizing function

On each side of the Morph Pad there is a randomizing feature that randomly varies sounds to a chosen intensity level, or amount of randomization. This is especially useful for subtle changes to natural sounds, but it can also provide for rewarding variations to synthesized sounds as well.

1. Select one of the squares on the left side of the Morph Pad to determine the number of corners that are to be varied.
2. To adjust the intensity of the random deviations, drag the slider on the right side of the Morph Pad.
3. Click the Rnd button at the top of the slider to perform the randomization.

When you next move the morph ball, you'll hear the variations you just created.

Ultrabeat

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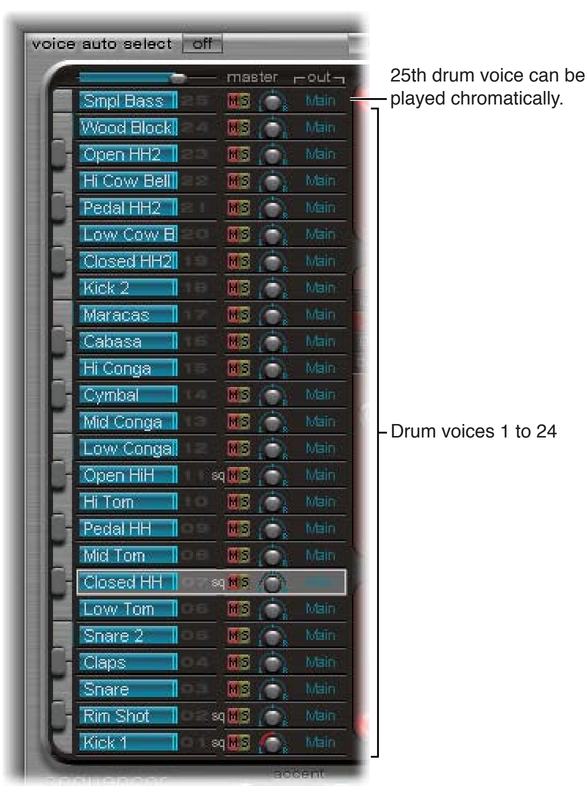
Ultrabeat overview

Ultrabeat is a synthesizer that is designed to create percussive sounds and polyphonic rhythms. It also incorporates a powerful, integrated step sequencer, which you can use to create polyphonic rhythmic sequences and patterns. See Ultrabeat step sequencer overview.

Most software synthesizers offer one synthesizer per plug-in instance. Ultrabeat, however, places 25 independent synthesizers at your disposal. These synthesizers—known as *drum voices* in Ultrabeat—are optimized for the generation of drum and percussion sounds.

You can compare Ultrabeat with a drum machine that features 24 drum pads, plus a built-in multioctave keyboard that can be used for polyphonic accompaniments, or bass or melody lines. See Play and select Ultrabeat drum sounds.

The distribution of drum voices across the MIDI keyboard is as follows: a single drum voice is assigned to each of the first 24 MIDI keys (spanning MIDI notes C1 to B2). The 25th drum voice is assigned to the keys from the 25th MIDI key (C3) and above, enabling you to play this sound chromatically.

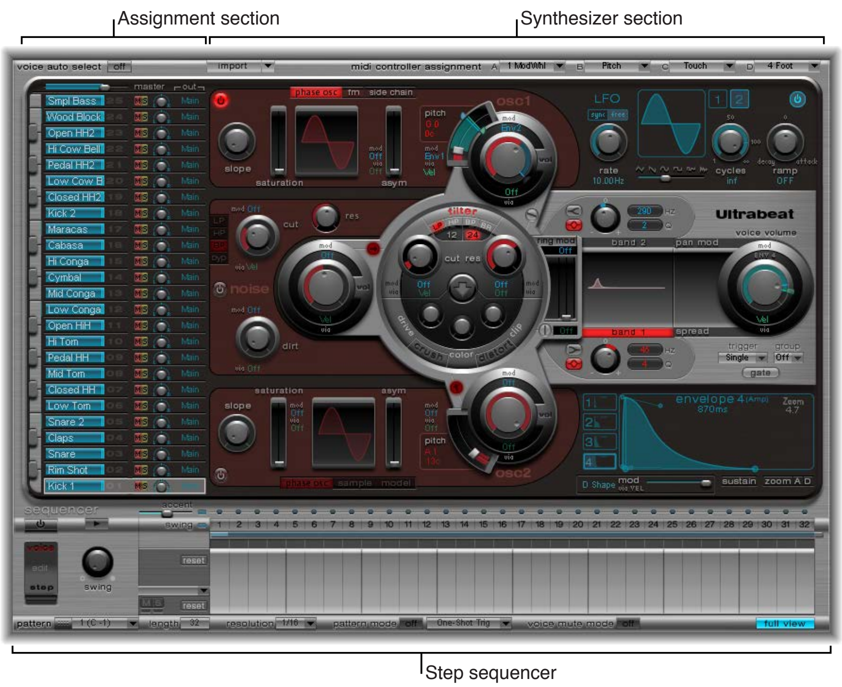


Ultrabeat's 24-drum pad assignment is compatible with the widely adopted GM (General MIDI) MIDI Drum note mapping standard. If your MIDI keyboard is limited to two octaves or does not support transposition, use the Transpose parameter of your host application to shift incoming MIDI notes up or down one or more octaves.

Note: For clarity, and to maintain the drum machine analogy, this guide refers to the independent synthesizers as *drum sounds*. A combination of drum sounds forms a *drum kit*.

Ultrabeat interface

Ultrabeat's user interface is divided into three main sections.



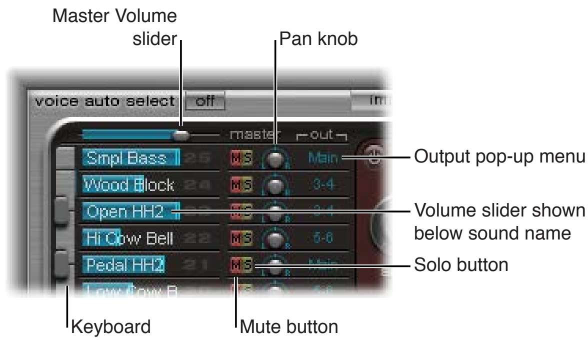
- *Assignment section:* Displays all drum sounds in a drum kit. You can select, name, and organize drum sounds here. This section also includes a small mixer, used to adjust the level and pan position of each sound. See Ultrabeat Assignment section overview on page 328.
- *Synthesizer section:* Used to create and shape individual drum sounds. The parameters of the drum sound selected in the Assignment section are displayed in the Synthesizer section. See Ultrabeat Synthesizer section overview on page 336.
- *Step sequencer:* Used to create and control sequences and patterns. A sequence triggers a single drum sound, and can consist of up to 32 steps. A pattern contains the sequences for all 25 sounds. You can trigger and control sounds with the step sequencer in place of—or in addition to—MIDI notes entering Ultrabeat from your host application or keyboard. See Ultrabeat step sequencer overview on page 368.

Ultrabeat Assignment section

Ultrabeat Assignment section overview

The Assignment section displays all sounds in a drum kit. You can:

- Select, organize, and name sounds
- Import sounds from other Ultrabeat settings or EXS instruments
- Set relative levels and pan positions for each sound
- Mute or solo sounds in the drum kit

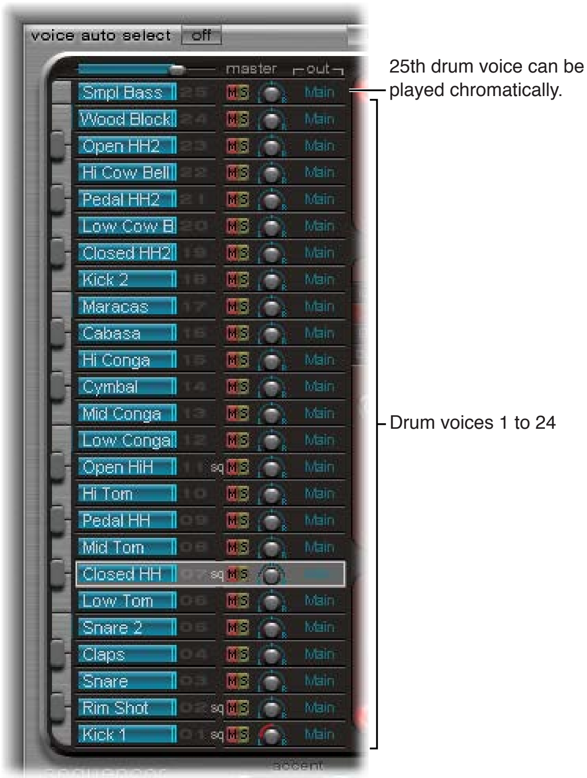


Assignment section parameters

- *Master (Volume) slider:* Controls the levels of all drum sounds in the drum kit—the overall mix level of all drum sounds.
- *Volume slider:* Drag the blue slider to set the volume for the sound. All drum sound levels are indicated by blue sliders, providing an overview of relative levels within the kit.
- *Keyboard:* Acts as a display when MIDI information is received. Click the keys to play the sound on the corresponding row.
- *M(ute) buttons:* Click to mute (silence) one, or multiple, sounds in the drum kit.
- *S(olo) buttons:* Click to hear one, or multiple, drum sounds in isolation. All other drum sounds (unsoloed) are muted.
- *Pan knobs:* Rotate to position drum sounds in the stereo field (panorama).
- *Output pop-up menus:* Use to independently route each drum sound to individual outputs, or output pairs. Ultrabeat features eight separate stereo and mono outputs when inserted as a multi-output instrument.
  - Drum sounds that are routed to an output pair other than *Main (1–2)* are automatically removed from the main output channel strips.
  - Choosing an output pair other than Main (1–2) routes the sound to an aux channel strip.

Play and select Ultrabeat drum sounds

The 25 sounds of an Ultrabeat drum kit are mapped to the onscreen keyboard at the left side of the Ultrabeat interface. Sounds start from the bottom of the onscreen keyboard and correspond to note values on a connected MIDI keyboard, starting at C1.



Play a drum sound

Do one of the following:

- Play a note on a connected MIDI keyboard.
- To trigger the sound on the adjacent row, click a key on the onscreen keyboard.

The corresponding key of the keyboard beside the drum sound name turns blue when it is clicked or when it receives MIDI information.

Select a drum sound

- Click the name of the sound in the Assignment section.

The selected sound is indicated by the gray frame around the assignment row. The parameters of the selected sound are shown in the Synthesizer section to the right. See Ultrabeat Synthesizer section overview on page 336.

The image below shows drum sound 2 being played (indicated by the blue key) while drum sound 4 is selected (the gray frame).

Select a sound by playing your MIDI keyboard

- Turn on Voice Auto Select to select sounds by playing your MIDI keyboard.

Name, swap, and copy Ultrabeat drum sounds

This section outlines the steps required to rename and rearrange the positions of drum sounds within a drum kit.

Rename a sound

1 Double-click the name of a sound to open a text entry field.

2 Enter the name and press Return, or click anywhere outside the text entry field, to complete the naming operation.

You can swap and copy drum sounds within an Ultrabeat kit by using a drag-and-drop or shortcut menu operation.

Swap or copy drum sounds using drag and drop

- Drag the sound name to the target position (do not click a button or menu).

The target is shaded as you drag across the list of sound names.

- A standard drag-and-drop operation *swaps* the two drum sounds (including Mixer settings: volume, pan, mute, solo, and output configuration). Sequences are *not* swapped.
- Command-drag to *swap* the two drum sounds *and* sequences.
- Option-drag to *copy* the sound. Sequences are *not* copied.
- Command-Option-drag to *copy* the sound *and* sequences.

Swap or copy drum sounds using a shortcut menu command

1 Control-click or right-click the sound name.

2 Choose one of the following commands from the shortcut menu:

- *Copy (Voice & Seq):* Copies the selected sound, including mixer settings and all sequences, to the Clipboard.
- *Paste Voice:* Replaces the selected sound with the sound from the Clipboard but does *not* replace existing sequences.
- *Paste Sequence > (submenu):* Enables you to replace all, or individual sequences, of the target drum sound. Sound parameters are not affected.
  - *1 to 24:* A single sequence replaces the currently active sequence (as set in the Pattern menu) of the target drum sound. This enables you to copy sequences into any of the 24 possible pattern locations.
  - *All:* Replaces all sequences. In situations where a sound only has several sequences (not all 24 are used), "Paste Sequence > all" places these sequences into the same positions; sequence 5 (in the Pattern menu) is pasted to position 5 in the target sound, for example. If a sequence exists at this location in the target sound, it is replaced. If no sequence exists at this location, the copied sequence is added to the target sound.
- *Swap with Clipboard:* Exchanges and replaces the selected sound with the sound from the Clipboard.
- *Init > (submenu):* Opens a submenu that contains a few starting-point (Init) sounds. Select one of these to replace the target drum sound. The Sample Init sound initializes the filter and pitch parameters to neutral settings, which are ideal as a starting point for programming sample-based drum sounds.

  These commands affect only the selected drum sound. The sequence and sound data of the other 24 sounds are unaffected.

Note: The shortcut menu commands Paste and Swap with Clipboard require an initial Copy command—to place data in the Clipboard—before you can use them.

Import sounds and EXS instruments into Ultrabeat

Ultrabeat can directly import EXS instruments, in addition to loading audio samples and its own settings. This provides enhanced sound design and rhythm processing options for EXS instruments, along with the ability to play and control EXS drum kits with Ultrabeat's intuitive drum mixer layout.

There are two key elements that you use to import sounds and sequences into Ultrabeat:

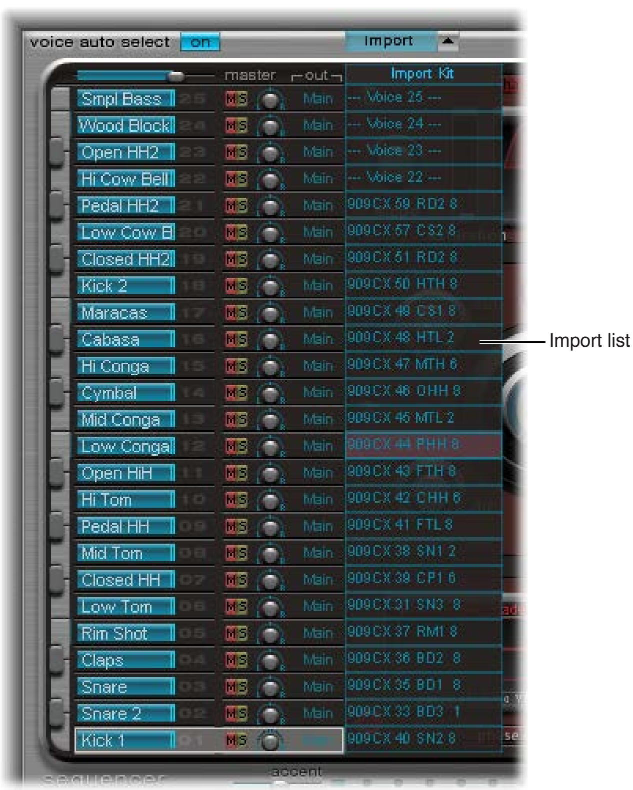
- *Import button:* Click to choose a kit that contains sounds or sequences you want to import.
- *Import list button/Import list:* Click to hide/show the import list, then add drum sounds/sequences from Ultrabeat settings/EXS instruments to your active kit.

Open an Ultrabeat setting or EXS instrument with the import list

1 Click the Import button.

2 Select the source Ultrabeat setting or EXS sampler instrument in the dialog, then click Open.

A list of all sounds found in the selected setting, or samples in the EXS instrument, is shown to the right of the Assignment section Mixer.



Note: If you import EXS instruments that include more than 25 sample zones, you can navigate through pages (of 25 zones) by using the up and down arrows to the left and right of the EXS instrument name at the top of the import list.

Drag sounds into your drum kit

- Drag the sound name from the import list into the target row in the Mixer section.

Note: Hold down the Command key to include all sequences.

Transfer sounds using shortcut menu commands

1. Control-click (or right-click) the sound name in the import list, then choose Copy (Voice & Seq) from the shortcut menu.

   The selected sound and its sequences are copied to the Clipboard.

2. Control-click (or right-click) the sound you want to replace in the current drum kit, then choose one of the following shortcut menu commands:

   - *Paste Voice:* Replaces the selected sound with the sound from the Clipboard but does not replace existing sequences.
   - *Paste Sequence > (submenu):* Opens a submenu that enables you to replace all, or individual sequences, of the target drum sound. Sound parameters are not affected.

     Pasting a single sequence replaces the currently active sequence (as set in the Pattern menu) of the target drum sound. This allows you to copy sequences into any of the 24 possible pattern locations.
   - *Swap with Clipboard:* Exchanges and replaces the selected sound (and associated sequence) with the sound from the Clipboard.

Import an EXS instrument into Ultrabeat

- Drag an EXS instrument filename from the Finder directly onto the Assignment section.

Ultrabeat reproduces the EXS layout as closely as possible. Layered EXS zones are set up as layered drum sounds, using the sample playback mode of oscillator 2. See Use Ultrabeat oscillator 2 sample mode on page 343.

Note: This method does not allow paging through the EXS instrument if it contains more than 25 sounds (samples). Ultrabeat maps only sample zones and layers that fall within Ultrabeat's drum sound range of C1 to C3. All other samples (zones) are ignored.

Ultrabeat settings

Ultrabeat settings are saved and loaded in the same way as other Logic Pro instruments.

An Ultrabeat setting contains:

- The drum kit, which consists of 25 drum sounds, including assignment and mixer settings.
- The complete parameter settings of all 25 sounds
- The sequencer settings and all patterns, including step automation, trigger, velocity, and gate information for all 25 sounds

The joint recall of all this information when loading an Ultrabeat setting is practical because the musical effect of the patterns, especially those with sequenced gate and velocity parameters, is often tightly tied to the tones of the sounds being used.

Note: When you save a drum kit with the Settings pop-up menu, only the location of the sample is saved with the setting. An Ultrabeat setting doesn't save the audio files—merely a reference to their hard disk location. If you load a setting that contains a reference to a sample that has been moved or erased, a dialog prompts you to specify or find the sample. To avoid this problem, you can use the Finder to create and manage a dedicated Ultrabeat sample folder—for all sounds and kits.

Ultrabeat Synthesizer section overview

Ultrabeat's sound engine is optimized for creating electronic and acoustic drum and percussion sounds. It combines several synthesis approaches—phase distortion, sample playback, FM (frequency modulation), and physical modeling—to create tones. You can also use an audio side-chain input as a sound source. The sound engine provides comprehensive modulation functions, enabling nearly every Ultrabeat element to be modulated.

The Synthesizer section is the heart of Ultrabeat. Each drum sound in a drum kit is an independent synthesizer and has its own set of synthesizer parameters—its own synthesizer section.

The interface and signal flow of Ultrabeat's synthesis engine are based on classic synthesizer designs. If you're new to synthesizers, it might be best to start with Synthesizer basics overview on page 471, which will introduce you to the fundamentals and terminology of different synthesis systems.

The Synthesizer section runs from left to right, following the layout and signal flow of a subtractive synthesizer. The basic tonal material is created by the oscillators, noise generator, and ring modulator. A filter then takes away certain frequencies from the raw sound, followed by volume shaping–envelopes.

Note: Although the structure and layout mirrors classic subtractive synthesizer designs, Ultrabeat incorporates a number of different tone generation (synthesis) methods, including frequency modulation, component modeling, sample playback, and phase distortion. These provide unique qualities that greatly expand the range of sounds you can create.

The details of Ultrabeat's functions and their importance become more apparent when you look at the three-dimensional nature of the interface and recognize the different levels from front to back. The following descriptions refer to the third dimension, so keep this in mind while reading and exploring Ultrabeat's interface.

The large, round, elevated Filter (and Distortion) section is in the center. Its placement and design are both symbolic and practical, as the filter plays a central role in Ultrabeat. The Filter is discussed in Use Ultrabeat's filter section on page 349.

The Filter receives its signal from the following sound sources: oscillator 1, oscillator 2, the noise generator, and the ring modulator. The outputs of these sources are represented by the three round objects, and the rectangular ring modulator section to the right, that surround the Filter.

One level down—from front to back—each sound source output object provides modulation controls. These determine how modulation sources, such as the LFO and envelopes, affect each sound source. See Ultrabeat modulation overview on page 358.

Each sound source also features a small Signal Flow button (red, when active). This is used to determine (and indicate) whether the signal of the associated sound source should proceed through the Filter or bypass it—before being routed to Ultrabeat's Output section.

The Output section is shown to the right. Signals sent from the Filter can pass through two equalizers and a stage for stereo expansion or panoramic modulation. You can also set the initial output level and trigger behavior in this section. See Ultrabeat Output section overview on page 353.

The output of the drum sound is then sent to the Assignment section mixer. See Ultrabeat Assignment section overview on page 328.

Ultrabeat sound sources

Ultrabeat oscillator overview

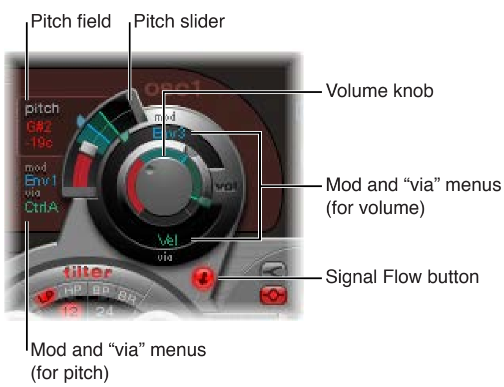
Ultrabeat oscillators are used to generate waveforms. oscillator 2 can use a sample in place of a waveform. The signal of one or both oscillators is then sent to other portions of the synthesizer engine for shaping, processing, or manipulation.

- Oscillator 1 can be frequency-modulated by oscillator 2, for FM synthesis sounds.
- Oscillator 2 can be ring-modulated with oscillator 1.
- Oscillator 2 can use an audio file (a sample), in place of a synthetic waveform. The sample is output as the oscillator 2 signal.

Other sound sources include a separate noise generator and ring modulator that can produce additional signals to those generated by the oscillators. See Ultrabeat ring modulator on page 347 and Ultrabeat noise generator on page 348.

Common oscillator parameters

These parameters are available to both oscillators.



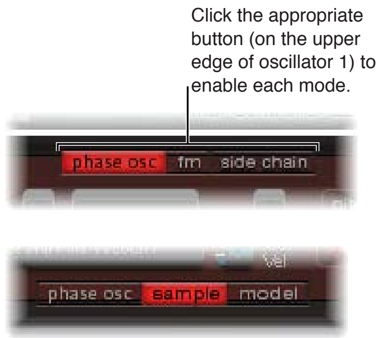
- *Oscillator on/off button:* Click the button (at the top left of oscillator 1 or bottom left of oscillator 2) to turn each oscillator on or off independently.
- *Volume knob:* Sets the level of oscillator 1 or 2. Volume can be modulated by the sources found in the mod and pop-up menus.
- *Pitch slider and field:* Drag to set the oscillator pitch in semitone steps. Press Shift to adjust the pitch in cent intervals (1 cent = 1/100th semitone). Pitch can be modulated by the sources found in the mod and via pop-up menus.
- *Signal Flow button:* Click to route the signal of the associated oscillator through the filter, or directly to the EQ (in the Output) section. When turned on, the button is lit and an arrow indicates the direction of the signal flow.
- *mod and via pop-up menus:* Determine the modulation sources for oscillator pitch and level. See Ultrabeat modulation overview on page 358.

Switch between Ultrabeat oscillator synthesis modes

Oscillator 1 can be switched between three different synthesis engines: phase oscillator, fm, and side chain (external audio input), which extends your sonic palette significantly. Each mode (engine) provides different parameters and features.

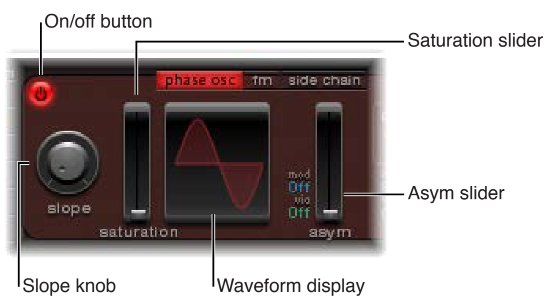
Oscillator 2 can be switched between three different types of synthesis engines: phase oscillator, sample, and model. Each mode offers different parameters and features.

- Click a button on the upper edge of oscillator 1 to select a mode (synthesizer engine).
- Click a button on the lower edge of oscillator 2 to select a mode (synthesizer engine).



Ultrabeat oscillator 1 phase oscillator mode

The waveform of the phase oscillator can be reshaped into almost any basic synthesizer waveform. The effects of parameter changes are immediately reflected in the waveform display. See Create classic waveforms with Ultrabeat's phase oscillator.



Phase oscillator parameters

- *Slope knob:* Rotate to set the slope or steepness of the waveform. The higher the value, the steeper the waveform. The resulting sound takes on an increasingly nasal character as the incline becomes more vertical.
- *Saturation slider:* Move to Increase the gain, eventually causing the waveform to clip. Higher values result in a distortion of the waveform shape, making it more rectangular. This results in a corresponding increase in odd-numbered overtones.
- *Asym(metry) slider:* Move to change the waveform angle. Higher values skew the waveform toward a sawtooth wave. Asym can be modulated by the sources shown in the mod and via pop-up menus, enabling dynamic sound changes at the oscillator level. See Ultrabeat modulation overview on page 358.

Note: Oscillator 2's phase oscillator operates in a nearly identical fashion to the phase oscillator of oscillator 1. The key difference is that Saturation can be modulated in oscillator 2, rather than Asymmetry (in oscillator 1). This means that when both oscillators are in phase oscillator mode, they can produce different sounds.

Use Ultrabeat oscillator 1 FM mode

FM (frequency modulation) synthesis is well suited for creating bell-like digital tones and metallic sounds. The principle of frequency modulation (FM) synthesis was developed in the late 1960s and early 1970s by John Chowning. It was popularized by Yamaha's range of DX synthesizers in the 1980s. Although Ultrabeat can't be compared with the DX series in the discipline of pure FM synthesis, it can achieve some of the signature sounds of these instruments.

In pure FM synthesis, the frequency of one signal generator, or oscillator, is altered (modulated) by another signal generator. Positive values from the second generator increase the frequency of the first generator. Negative values decrease the frequency.

In a synthesizer, this type of modulation takes place in the audible range. Depending on the design of the instrument, you can hear the signals of either the first oscillator alone (being modulated by the other oscillator), or both oscillators. The interaction between the two generators alters the waveform signal of the first oscillator and introduces a number of new harmonics. This harmonic spectrum can then be used as the source signal for further sound processing, such as filtering, envelope control, and so on. See Frequency modulation (FM) synthesis on page 492.

In Ultrabeat's FM synthesis mode, oscillator 1 (the carrier) generates a sine wave. The frequency of oscillator 1's sine wave is modulated by the waveform of oscillator 2 (the modulator).

- When oscillator 2 outputs a positive (or higher) frequency signal, the frequency of oscillator 1 increases.
- When oscillator 2 outputs a negative (or lower) frequency signal, the frequency of oscillator 1 decreases.

The net effect of speeding up or slowing down the frequency of oscillator 1 in each waveform cycle is a distortion of the basic wave shape. This waveform distortion also introduces a number of new, audible, harmonics. The more complex the oscillator 2 waveform, the more partials are created by increasing FM Amount. Watch the display to see how the sine wave takes on an increasingly complex shape.

Important: The impact of any frequency modulations depends on *both* the frequency ratio and the modulation intensity of the two oscillators.

Set the frequency ratio and adjust the modulation intensity

1. Turn on FM mode for oscillator 1.
2. Turn on oscillator 2.
3. Adjust the Pitch parameter values of one, or both, oscillators.
4. Adjust the amount (intensity) of frequency modulation with the FM Amount knob.

FM Amount can be modulated by the sources shown in the mod and via pop-up menus. See Ultrabeat modulation overview on page 358.

Use Ultrabeat oscillator 1 side chain mode

In side chain mode, Ultrabeat uses an external side-chain input as the source for oscillator 1. The signal of any audio channel strip, bus, or live input can be routed through Ultrabeat's filters, envelopes, LFO, and step sequencer. Using busses as side-chain sources makes it possible to route signals to the side-chain input from any channel strip type that offers busses as outputs or sends. This includes software instrument channel strips, aux channel strips, or a mix of multiple channel strips that are routed into a common aux (subgroup), that has a bus as the output destination.

This feature enables you to use an audio input from oscillator 1, along with the synthesis engine of oscillator 2, to create a part live audio, part synthesized drum sound, for example. As another creative option, you could use one drum sound in a kit to filter an external audio signal with a sequenced groove.

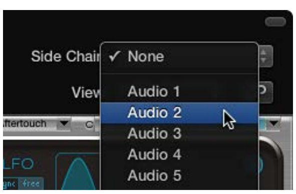
There are two points to note about side-chain use in Ultrabeat:

- The side chain affects only the selected drum sound—Ultrabeat's other drum sounds and sequences are not altered.
- A side-chain audio signal alone is not enough to trigger Ultrabeat. To hear the side-chained audio signal, you need to make sure that Ultrabeat is triggered by MIDI or the internal step sequencer.

Use Ultrabeat's side chain

1. Turn on the side chain for oscillator 1.

2. Choose the channel strip that you want to use as the side chain input source from the Side Chain pop-up menu at the top of the plug-in window.



3. Start playback of your audio source/host application.
4. Play a note on your MIDI keyboard (that corresponds to the side-chained drum sound). Alternatively, you can use the step sequencer to play a pattern for the side-chained drum sound.

Use Ultrabeat oscillator 2 phase oscillator mode

The waveform of the phase oscillator can be reshaped into almost any basic synthesizer waveform.

Oscillator 2 operates in a nearly identical fashion to oscillator 1 when in phase oscillator mode. The key difference is that Saturation can be modulated in oscillator 2, rather than Asymmetry in oscillator 1. This results in the production of different sounds when both oscillators are in phase oscillator mode.

Phase oscillator parameters

- *Slope knob:* Rotate to set the slope or steepness of the waveform. The higher the value, the steeper the waveform. The resulting sound takes on an increasingly nasal character as the incline becomes more vertical.
- *Saturation slider:* Move to increase the gain, eventually causing the waveform to clip. Higher values result in a distortion of the waveform shape, making it more rectangular. This results in a corresponding increase in odd-numbered overtones.
- *Asym(metry) slider:* Move to change the waveform angle. Higher values skew the waveform toward a sawtooth wave. Asym can be modulated by the sources shown in the mod and via pop-up menus, enabling dynamic sound changes at the oscillator level. See Ultrabeat modulation overview on page 358.

Create classic waveforms with Ultrabeat's phase oscillator

The basic waveforms of classic analog synthesizers can be reproduced with the phase oscillator: sine, rectangular, and sawtooth waves will result from different Slope, Saturation, and Asym parameter value combinations.

Do one of the following:

- To produce a classic square wave, set Slope and Saturation to their maximum values and Asym to the minimum value.
- To produce a sawtooth wave, set Slope to –0.20, Saturation to the minimum, and Asym to the maximum value.
- To produce a sine wave, set Slope, Saturation, and Asym to 0 values .

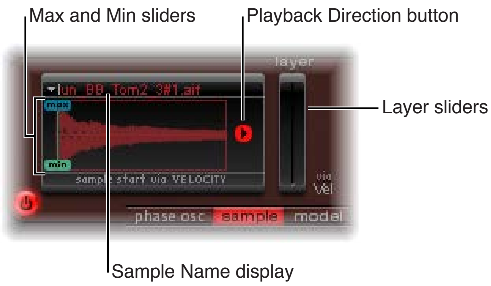
Basic waveform characteristics

The table provides an overview of the tonal qualities of each basic waveform.

Waveform	Basic tone	Comments
Rectangular	Nasal sounding	Great for reed instruments, synth blips, basses
Square	Hollow and woody sounding	Useful for basses, clarinets, and oboes. The pulse width of (oscillator 2 and 3) square waveforms can be smoothly scaled between 50% and the thinnest of pulses.
Sawtooth	Warm and even	Useful for strings, pads, bass, and brass sounds
Triangle	Sweet sounding, softer than sawtooth	Useful for flutes, pads
Sine	A pure tone	The sine wave of oscillator 1 can be frequency-modulated by oscillator 2. This kind of frequency modulation forms the basis of FM synthesis.

Use Ultrabeat oscillator 2 sample mode

In sample mode, oscillator 2 uses an audio file as a sound source.



Sample mode parameters

- *Sample Name display:* Use to load and unload samples or to display the loaded sample in the Finder. Click the arrow in the upper-left corner of the waveform display to open.
- *Max/Min sliders:* Move to set the start point of the sample—depending on the dynamics (incoming velocity level) of the performance.
  - *Min:* Determines the start point of the sample at the minimum velocity level (velocity = 1).
  - *Max:* Determines the start point of the sample at the maximum velocity level (velocity = 127).

  Note: If Min and Max are set to the same value, velocity has no effect on the sample start point.
- *Playback Direction button:* Changes the playback direction of the sample (forward or backward).

- *Layer sliders:* Both factory Ultrabeat samples and sounds imported from EXS instruments often consist of different layers that are dynamically switched by incoming MIDI note velocities. The precise sample layer that incoming velocity values switch to is determined by the green Layer slider (min), or the blue Layer slider (max).
  - The green Min slider determines which layer is triggered at a MIDI note velocity = 1.
  - The blue Max slider determines which layer is triggered at a MIDI note velocity = 127.

  Note: If you have loaded a single sample that does not have multiple layers, the Vel Layer slider has no effect.

Load a sample into Ultrabeat oscillator 2
Ultrabeat's factory settings include a selection of multilayer drum and percussion samples. You can also load your own samples in AIFF, WAV, CAF, or SDII stereo interleaved format.

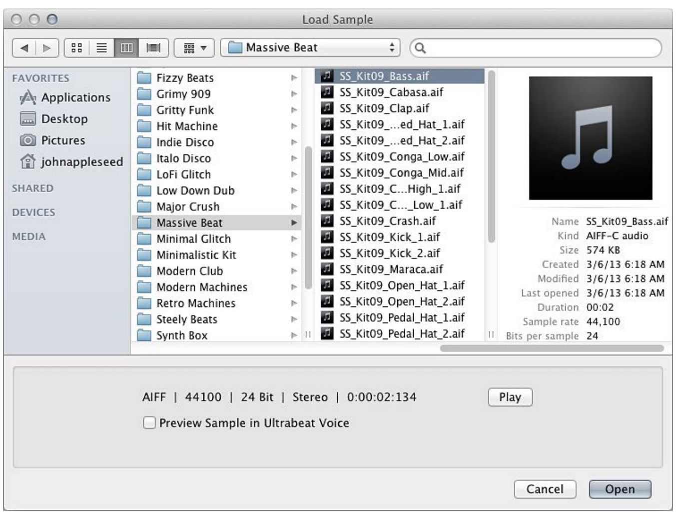
Note: The velocity layering function is not available for user-loaded samples.

1 Choose Load Sample from the pop-up menu in the upper-left corner of the waveform display.

2 In the Load Sample window, browse to the audio file you want to use, and do one of the following:

- To load the chosen file into oscillator 2, click Open.
- To retain the current state, click Cancel.

Use Ultrabeat's load sample preview
You can audition samples before importing them into Ultrabeat.



Do one of the following:

- To preview audio files (AIFF, WAV, SD2, CAF, UBS) before loading, click the Play button. Click the button again to stop playback.

  Clicking the Play button loops playback of the currently selected sample file. The sample is played with no manipulation: all filters, EQ, envelopes, and other synthesizer parameters are ignored.

- To audition multiple files, click Play once, then step through the files by pressing the Up Arrow and Down Arrow keys or by clicking each filename.

  Note: Independent auditioning of all layers is not possible. In multilayer UBS files, the audition function plays the sample at a fixed velocity of 75%. Only the layer addressed by this velocity value is played.

- To temporarily replace the sample files of the currently selected drum sound, click the "Preview Sample in Ultrabeat Voice" checkbox.

  The drum sound can be triggered as usual (played notes, MIDI region events, or Ultrabeat sequencer events) while the Load Sample window is open and different files are being selected. The selected sample can be heard as part of the current drum sound, inclusive of all synthesizer processing.

  Important: Any effects inserted into the Ultrabeat instrument channel strip are heard when you preview samples.

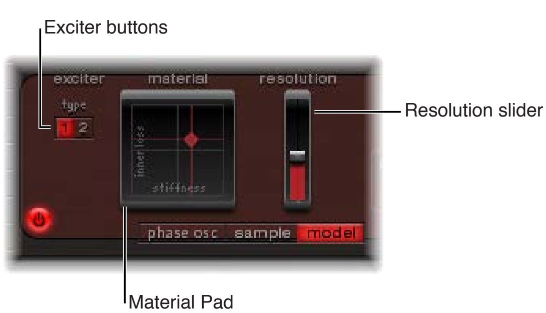
Use Ultrabeat oscillator 2 model mode

This oscillator 2 mode employs a method of synthesis known as component modeling. This tone generation technique mimics the physical properties of an object, such as a guitar string. Further objects are used to stimulate the string, emulating the way that it is played: plucked, bowed, and so on. Although the term *string* is used, model mode enables you to create sounds that don't sound like traditional stringed instruments.

The string is the element that is responsible for the basic tone. Ultrabeat offers parameters that enable you to adjust its material—what it's made of, in other words.

The exciters make the string vibrate (move) in different ways. The string itself doesn't make a sound unless it is stimulated, or excited.

The signal of the vibrating string is sent to the filter, amplifier, and so on, in the Synthesizer section.



Model mode parameters

- *Exciter buttons:* Click to select one of the two contrasting exciters. Each offers different sound characteristics (Type 1 and Type 2).

  Note: In this context, an exciter is the agent or triggering device used to initiate the vibration of the string. Don't confuse it with the effect plug-in of the same name.

- *Material Pad:* Determines the basic tone of the string with the string Stiffness and damping (Inner Loss) parameters.
  - *Inner Loss:* Emulates damping of the string, as caused by the string material—steel, glass, nylon, or wood. These are frequency-dependent losses that cause the sound to become more mellow during the decay phase.
  - *Stiffness:* Sets the rigidity of the string. In reality, this is determined by the string material and diameter—or, to be more precise, by its geometrical moment of inertia. Stiffer strings exhibit an inharmonic vibration, where overtones are not integer multiples of the base frequency. Rather, they have higher frequencies, which can make upper or lower notes sound somewhat out of tune with each other.
- *Resolution slider:* Move to determine the precision of the calculation. Higher values produce additional harmonics. Lower values produce fewer harmonics, or inharmonic spectra.

Use Ultrabeat's Material Pad

The combination of the Inner Loss and Stiffness parameter positions determines the string material and, therefore, the general timbre of your sound. In general synthesizer terms, use of these parameters could be viewed as being similar to the waveform selector/generator in the oscillator section. The default pitch of the string is C3 (middle C).

- To simultaneously adjust the Inner Loss and Stiffness parameter positions, drag the ball (which correlates to the x and y coordinates) in the Material Pad.
  - Low Stiffness values, combined with low Inner Loss values, lead to metallic sounds.
  - Increase the Stiffness to make the sound more bell-like, or glass-like. Extreme Stiffness values turn the string into a solid metal rod.
  - Increase the Inner Loss value while maintaining a low Stiffness level to emulate nylon or catgut strings.
  - High Stiffness values, combined with high Inner Loss values, simulate wood-like materials.

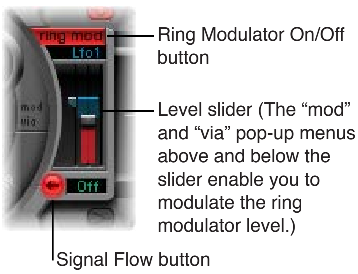
Note: Option-click the ball to reset all string parameters to their default values.

Ultrabeat ring modulator

Ring modulation is a powerful tool for the creation of inharmonic, metallic, bell-like sounds. Ultrabeat's ring modulator functions as an independent sound source—its signal can bypass or be sent into the filter, independent of the oscillator 1 and 2 signals. Its volume can also be regulated.

Important: Although the ring modulator signal is independent of the signals generated by oscillators 1 and 2, both oscillators need to be enabled if you want to use the ring modulator signal. Because the ring modulator is reliant on the signals of both oscillators to produce an output signal, it is automatically muted when one of the oscillators is switched off.

The actual sound produced by the ring modulator is largely dependent on the parameter settings of both oscillators. In particular, the tuning relationships of each oscillator have a direct effect on the sound of the ring modulator signal. The individual levels of the oscillators, however, have no effect on the process (or output) of ring modulation.



Ring modulator parameters

- *Ring Modulator On/Off button:* Turns the ring modulator on or off.

  Note: If you want to hear the ring modulator's signal in isolation (to better judge your settings), temporarily set the volume of both oscillators to a value of 0.

- *Level slider:* Move to set the output volume of the ring modulator.
- *Mod and via pop-up menus:* Choose the modulation source (and optional via source) for the Level parameter. When either source is active, small sliders (handles) appear on either side of the Level slider. See Ultrabeat modulation overview on page 358.
- *Signal Flow button:* Determines the routing of the ring modulator signal. It is sent either to the filter (the Signal Flow button turns red), or directly to the EQ section (the Signal Flow button remains gray). The direction of the arrow on the Signal Flow button illustrates the routing.

  Note: The Signal Flow button determines how the ring modulator output signal is routed. It doesn't turn the ring modulator on or off.

Ultrabeat noise generator

Ultrabeat's flexible noise generator enables you to create a wide range of percussive sounds and sound elements. The noise generator has its own filter, which functions independently of the main Ultrabeat filter, although the noise generator filter can also be used on the overall sound.

Technically, a noise signal contains all tonal frequencies, at a roughly equal volume level. As all frequencies in the spectrum are audible, it makes it difficult for human beings to hear any tonality (pitch) in a noise signal. Despite this, or as a direct result of it, noise is an indispensable ingredient when creating drum sounds.

Noise generator parameters

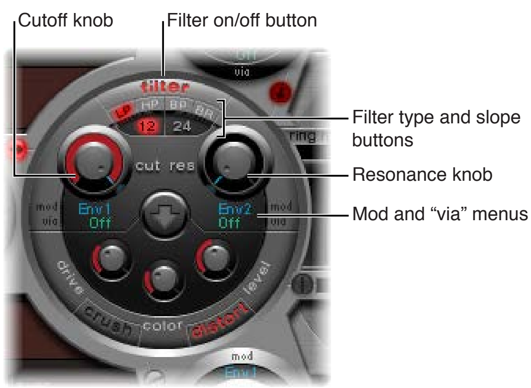
- *On/Off button:* Turns the noise generator on or off. When programming a drum sound, you can turn the individual sound sources on or off. You can also listen to, or remove, individual components of the sound separately in this way.
- *Filter Type buttons:* Switch the noise generator's integrated filter between lowpass, highpass, and bandpass filter types.
  - *LP (lowpass):* This filter type allows frequencies that fall below the cutoff frequency to pass. The filter slope is fixed at 12 dB/octave.
  - *HP (highpass):* This filter type allows frequencies above the cutoff frequency to pass. The filter slope is fixed at 12 dB/octave.
  - *BP (bandpass):* The frequency band directly surrounding the center frequency (determined with the Cutoff knob) is allowed to pass. All other frequencies are cut. The Resonance parameter controls the width of the frequency band. The bandpass filter is a two-pole filter with a slope of 6 dB/octave on each side of the center frequency of the band.
  - *byp (bypass):* Disables the integrated filter.
- *Cut(off) and Res(onance) knobs:* Rotate to set the cutoff/center frequency and resonance/bandwidth behavior of the integrated filter.
  - The Cut knob defines the point in the frequency spectrum where the signal is boosted or cut. Depending on the selected filter type, you can make a sound darker (LP), thinner (HP), or more nasal (BP) by adjusting the Cut value. Cutoff can be modulated by sources in the mod and via pop-up menus.
  - Increasing resonance boosts frequencies that surround the cutoff frequency. Values range from 0 (no increase) to self-oscillation of the filter at high resonance values. Self-oscillation is typical of analog filter circuits. It occurs when the filter feeds back into itself and begins to oscillate at its natural frequency when high resonance values are used.

- *Dirt knob:* This parameter was specifically developed for the noise generator. Higher values alter the white noise signal, making it more grainy. The Dirt parameter is particularly effective at high resonance values. Dirt can be modulated by sources in the mod and via pop-up menus.
- *Volume knob:* Rotate to set the output level of the noise generator. Volume can be modulated by sources in the mod and via pop-up menus.
- *Signal Flow button:* Determines whether the noise generator signal is routed through the main Ultrabeat filter, or directly to the EQ (Output) section. When turned on, the button is lit and an arrow indicates the direction of the signal flow.

  Note: The Signal Flow button has no effect on the independent filter within the noise generator. The independent filter is deactivated with the "byp" button. It is therefore possible to filter the noise generator signal twice. In some instances you may want the noise generator signal to bypass the main filter, thus freeing the main filter for other drum sound processing duties.

Use Ultrabeat's filter section

Ultrabeat provides a powerful multimode filter that can dramatically, or subtly, alter the timbre of your drum sounds.



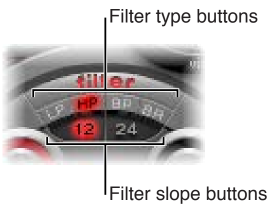
Filter parameters

- *Filter (On/Off) button:* Turns the entire Filter section on or off. Deactivating the Filter section makes it easier to hear adjustments to other sound parameters, as the filter always heavily affects the sound. If the Filter label is red, the filter is engaged. If gray, the filter is disabled.
- *Filter Type buttons:* Switch the filter between lowpass, highpass, bandpass, or band-rejection filter types.
- *Filter Slope buttons (12 and 24):* Switch the filter between different slopes.
- *Cut(off) and Res(onance) knobs:* Rotate to set the cutoff/center frequency and resonance/bandwidth of the filter.
- *mod and via pop-up menus:* Choose the modulation source (and via source) for the Cutoff and Resonance parameters. See Mod and via modulations in Ultrabeat on page 358.

Set Ultrabeat's filter type

Ultrabeat's filter can operate in several modes, allowing specific frequency bands to be filtered (cut away) or emphasized.

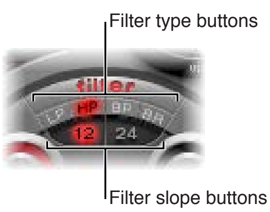
- To select a filter type, click one of the following buttons:
  - *LP (lowpass):* This filter type allows frequencies that fall below the cutoff frequency to pass. When set to LP, the filter operates as a lowpass filter. The slope of the filter can be set to 12 or 24 dB/octave in LP mode.
  - *HP (highpass):* This filter type allows frequencies above the cutoff frequency to pass. When set to HP, the filter operates as a highpass filter. The slope of the filter can be set to 12 or 24 dB/octave in HP mode.
  - *BP (bandpass):* The frequency band directly surrounding the center frequency (set with the Cutoff knob) is allowed to pass. All other frequencies are cut. The Resonance parameter controls the width of the frequency band. The bandpass filter is a two-pole filter with a slope of 6 or 12 dB/octave on each side of the center frequency of the band.
  - *BR (band rejection):* The frequency band directly surrounding the center frequency (set with the Cutoff knob) is rejected, while the frequencies outside this band can pass. The Resonance parameter controls the width of the rejected frequency band.



Set Ultrabeat's filter slope

Most filters don't completely suppress the portion of the signal that falls outside the frequency range defined by the Cutoff parameter. Frequencies that are located close to the cutoff frequency are generally reduced less than those that are farther away. The higher the slope value, the more apparent the level difference is between frequencies that are near to the Cutoff frequency and those that are farther away from it.

- Click the 12 dB or 24 dB button. The slope (curve) chosen for the Filter expresses the amount of rejection in decibels per octave. The steeper the slope, the more severely the level of signals below the cutoff frequency is affected in each octave.



Use Ultrabeat's filter Cutoff parameter

- Rotate the Cutoff Frequency (Cut) parameter to control the brilliance or determine the center frequency of the signal.
  - In a lowpass filter, the higher the cutoff frequency is set, the higher the frequencies of signals that are allowed to pass.
  - In a highpass filter, the cutoff frequency determines the point where lower frequencies are suppressed, with only upper frequencies allowed to pass.
  - In a bandpass/band-rejection filter, the cutoff frequency determines the center frequency for the bandpass or band-rejection filter.

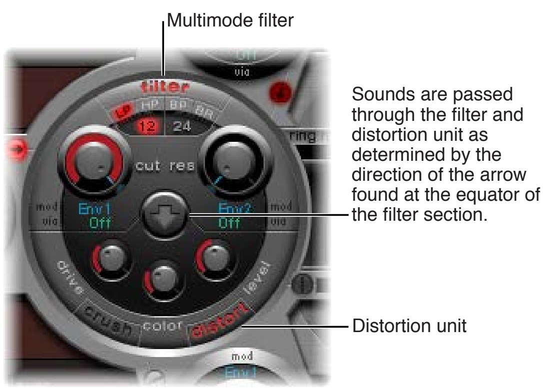
Use Ultrabeat's filter Resonance parameter

- Rotate the Resonance (Res) parameter to emphasize or suppress portions of the signal above or below the defined cutoff frequency, or to determine the width of the band that surrounds the cutoff frequency.
  - In a lowpass filter, Resonance emphasizes or suppresses signals below the cutoff frequency.
  - In a highpass filter, Resonance emphasizes or suppresses signals above the cutoff frequency.
  - In bandpass/band-rejection filters, resonance determines the width of the frequency band that surrounds the center frequency (set with the Cutoff Frequency parameter).

Set the signal flow order through Ultrabeat's filter and distortion unit

The output signals of both oscillators, the ring modulator, and the noise generator are sent to Ultrabeat's central Filter section (if not bypassed with the various Signal Flow buttons). The Filter section offers a multimode filter and a distortion unit.

- Click the arrow to change the signal flow order between the following:
  - First the Distortion unit, then the Filter circuit (arrow pointing up)
  - First the Filter circuit, then the Distortion unit (arrow pointing down).



Ultrabeat distortion circuit

The Distortion circuit provides either a bit crusher or distortion effect.

The bit crusher reduces the digital resolution of the sound, measured in bits, achieving an intentionally digital coloration of the sound. The distortion effect is modeled on an analog distortion unit, which distorts the sound by overdriving the level.

Both methods lead to distortions that are as tonally divergent as the two approaches. Distortion offers a more analog tone while the bit crusher sounds unmistakably digital.

Note: The arrow in the Filter section determines whether the Distortion circuit is inserted before or after the multimode filter (see Set the signal flow order through Ultrabeat's filter and distortion unit on page 351).

Distortion parameters

- *Crush and Distort buttons:* Activate the mode you want to use. The name of the active effect type is indicated in red. If neither button is active, the Distortion circuit is bypassed.
- *Drive knob:* Controls the amount of distortion.
- *Color knob:* Determines the basic tone of the distortion. Higher values result in a brighter sound. Lower values lead to a darker, warmer tone.
- *Level knob:* Sets the output level of the Distortion effect when in Distortion mode. In Bit Crusher mode, this knob sets a threshold level for incoming signals from the sound sources that must be reached before distortion (bit crushing) begins.

Ultrabeat Output section

Ultrabeat Output section overview

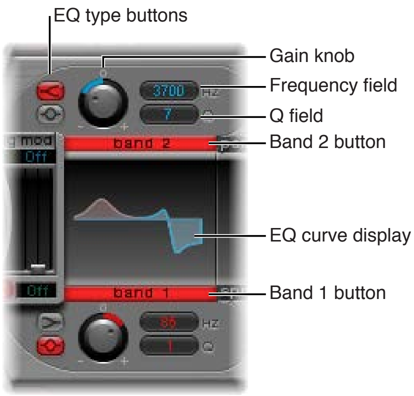
Depending on the status of each Signal Flow button, the output signals of both oscillators, the ring modulator, and the noise generator are routed to the Output section of Ultrabeat. This routing is either direct or through the Filter and Distortion section.

The Output section passes signals through both equalizers (EQ), then on to the Pan Modulation/ Stereo Spread section (in a preconfigured order) before the final level is set for the selected sound and the trigger behavior is adjusted.

- *Two-band EQ:* Provides precise tonal control of each drum sound.
- *Pan Modulation and Stereo Spread parameters:* Pan Modulation varies the panorama position of a drum sound. Stereo Spread broadens the stereo image.
- *Voice volume:* Sets the default level of each drum sound.
- *Trigger Mode controls:* Determine the way that Ultrabeat reacts to incoming MIDI notes. This is independently defined for each sound.

Adjust Ultrabeat's two-band EQ

Both equalizer bands have almost identical features. You can adjust each band separately.



Two-band EQ parameters

- *Band 1 and Band 2 buttons:* Click to turn each band on or off. When active, the label is red. If neither EQ is activated, the signal passes through unaffected. Band 1 is a low shelving EQ. Band 2 is a high shelving EQ.
- *EQ type buttons:* Switch between two different types of EQs: shelving (top button) and peak (lower button).
  - In shelving mode, all frequencies above or below the set frequency are either increased or reduced.
  - In peak mode, only frequencies located near the set frequency are affected.

  Note: Shelving EQs behave much like synthesizer lowpass and highpass filters. The key difference is that lowpass and highpass filters merely dampen certain frequencies (filter them out), whereas shelving EQs also allow these frequencies to be boosted.
- *Gain knobs:* Positive values boost a certain frequency range as determined by the EQ type and Hz settings. Negative gain values reduce the gain of the frequency range. If the Gain knob is set to a value of 0, the EQ has no effect. Option-click a Gain knob (or click the 0 above the Gain knob) to set it to a neutral position.
- *Frequency (Hz) field:* Drag vertically to set the frequency range to be boosted or reduced. Option-click the parameter to set the value to a neutral position: this is 200 Hz for band 1, and 2000 Hz for band 2.
- *Q field:* Drag vertically to set the Q (quality) factor. The effect of Q on the sound depends on the selected EQ type:
  - Shelving filters selected: as the Q value increases, the area around the threshold frequency becomes more pronounced.
  - Peak EQ selected: Q determines the width of the frequency band selection, low Q values select a broad band, and high Q values select a very narrow band to be boosted or reduced with the Gain knobs.

Each EQ band displays parameter changes on a frequency response curve. The display provides access to the Gain, Hz, and Q parameters of each band.

Edit the graphical EQ curve

Do any of the following:

- To change the EQ frequency, drag horizontally.
- To change the Gain, drag vertically.
- To change the Q factor, drag the handle shown at the peak (maximum point) of the EQ curve.

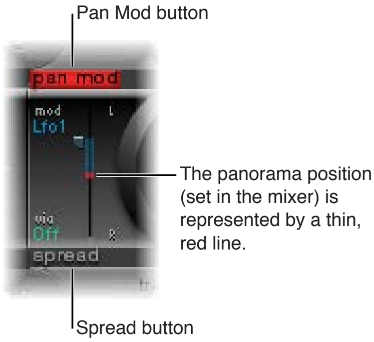
Ultrabeat pan and stereo spread

Ultrabeat pan modulation mode

The EQ's output signal is passed along to the Pan Modulation/Stereo Spread section where the placement of the sound in the stereo field can be modulated (pan modulation mode), or the stereo basis of the sound can be broadened (stereo spread mode).

Pan Modulation varies the panorama position of a drum sound with a mod (and via) source.

Note: The modulation set here is relative to the panorama position set in Ultrabeat's mixer in the Assignment section.



Pan modulation parameters

- *Pan Mod button:* Click to turn on pan modulation mode. If neither mode is activated, the signal passes through unaffected.
- *Mod and via pop-up menus:* Choose the modulation and via sources for pan modulation.
- *Mod and via sliders:* The blue and the green controls set the amount (intensity) of mod and via modulation.

Note: You cannot directly move the red line that represents pan position shown in this section. To move the line, rotate the Pan knob in the Mixer section.

Ultrabeat stereo spread mode

The EQ's output signal is passed along to the Pan Modulation/Stereo Spread section where the placement of the sound in the stereo field can be modulated (pan modulation mode), or the stereo basis of the sound can be broadened (stereo spread mode).

Stereo Spread broadens the stereo image, making it wider and more spacious.

Stereo spread parameters

- *Spread button:* Click to turn on stereo spread mode. If neither mode is activated, the signal passes through unaffected.
- *Lo Freq(uency) slider:* Move to set the width (the spreading effect) of bass frequencies: the higher the value, the more prominent the effect becomes.
- *Hi Freq(uency) slider:* Move to set the width of high frequencies.

Ultrabeat voice volume control

The Voice volume knob adjusts the output volume of individual drum sounds. To be more exact, when you adjust the Voice volume knob, you set the maximum level for the selected drum sound with Envelope 4, following the attack phase of Envelope 4.

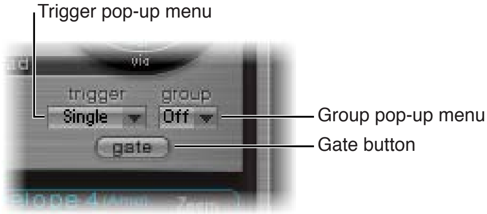
Note: Envelope 4 (Env 4) is hard-wired to level control for the selected sound. Each sound in the kit also has a further three envelopes and other modulation sources available for control of other synthesis parameters.

The intensity of Envelope 4's impact on Voice volume can also be modulated with a via source.

Note: Voice volume precedes the level sliders in the mixer. This approach allows the starting volume of individual drum sounds to be set independently of their relative levels in the drum kit mix, which you change in the Assignment section's mixer. See Ultrabeat Assignment section overview.

Change Ultrabeat's trigger mode

The way Ultrabeat reacts to a succession of incoming notes is set independently for each sound. Parameters that provide control over this aspect of Ultrabeat's behavior are found in the trigger mode section.



Trigger mode parameters

- *Trigger pop-up menu:* Choose either Single or Multi trigger mode.
  - *Single:* A new trigger note cuts off the note that is currently playing.
  - *Multi:* When a new note is played, currently playing notes continue to decay in accordance with their respective amplitude envelope settings (Env 4).
- *Group pop-up menu:* Choose between Off and settings 1 through 8. If two different sounds are assigned to the same group, they cut each other off when a new note message is received. Only one sound in the group can be played at a time.

  A typical use of this feature is when programming hi-hat sounds: when playing a real hi-hat, the closed hi-hat note cuts off or mutes the ringing of the open hi-hat. This feature is often referred to as "hi-hat mode."

  Note: While in Single Trigger mode, only the currently sounding note of the same sound is cut off. A sound that is assigned to a group cuts off all other sounds in the group, regardless of the note played.
- *Gate button:* Turns the Gate function on or off. When active, the sound is immediately cut off when the MIDI note is released (MIDI note off), regardless of envelope settings.

  Note: The Gate function ensures that a specific sound does not play—it can't be heard—after a note-off event sent from Logic Pro or Ultrabeat's internal sequencer. Note length can be an important creative element when programming rhythm tracks.

Ultrabeat modulation

Ultrabeat modulation overview

Most sound parameters can be controlled dynamically (modulated) in Ultrabeat. Ultrabeat provides two LFOs, four envelope generators, velocity, and four user-definable MIDI controllers as modulation sources.

Ultrabeat's modulation routings feature three key elements:

- *The modulation target:* The synthesizer parameter that you want to modulate.
- *The modulation source:* The parameter that modulates the target.
- *The via source:* A secondary modulation source that can influence the intensity of the first modulation source.

Note: You can use the same sources and the same via controllers in multiple modulation routings.

Mod and via modulations in Ultrabeat

You can modulate a sound parameter using an adjustable value—the modulation depth or intensity—with the mod parameter. You can choose between two LFOs, four envelope generators, and Max as sources for this modulation.

Via influences the modulation effect as follows. The depth of the first modulation (mod) can be modulated by a separate, independent source. The intensity of this secondary modulation is set with the via parameter. The sources for via modulations include velocity and four user-definable MIDI controllers.

A typical via modulation usage would be to increase a pitch sweep as you play at higher velocities. The harder a key is played, the higher in pitch it sounds—which is ideal for synthesized tom-tom sounds, for example. To create this routing, you would use an envelope (Env) as the mod source for oscillator pitch, and Velocity (Vel) as the via source.

Consider the following example:

The default Cut (Cutoff) parameter value is 0.50. No modulation source has been chosen in either the (blue) mod or (green) via pop-up menu. Both are Off in the image below.

When a modulation source is chosen from the mod pop-up menu (Env 1 in the image below), the ring around the knob is turned on. Drag in the ring to set a value (0.70 in the example) for the Cut parameter—when affected by the mod source.

Note: Exact values are shown in the help tags when adjusting parameters.

As soon as a modulation source is chosen from the via pop-up menu (Ctrl A in the image below), a movable slider appears on the mod ring. Drag this slider to set the maximum modulation value that can be reached through use of the via source (0.90 in this example).

The mod and via controls indicate the minimum and maximum values that the modulated parameter can attain, in comparison to the default value.

In the example, the Cut(off) frequency of the filter is set to a default value of 0.50. The mod source (Env 1) drives the Cut value up from 0.50 to 0.70 during the sound's attack phase and back down to 0.50 during the decay phase.

When the via source (Ctrl A) is introduced, the following occurs: when Ctrl A is at its minimum value, nothing changes; Cutoff continues to be modulated between values of 0.50 and 0.70 by the envelope (Env 1). A maximum value for Ctrl A causes the envelope generator to vary the parameter between the values of 0.50 (the default Cut value) and 0.90 (the via amount).

You can see the degree of maximum influence on basic parameters by the mod and via modulation sources—the area between the mod and via points shows the amount that the modulation depth can be further altered by the via modulation source. In the example, the Cutoff can reach values between 0.70 and 0.90, depending on the value being sent by Ctrl A.

Another example:

Cutoff is again set to 0.50, Env 1 now drives the value down to 0.25, and a maximum Ctrl A value reduces the Cutoff frequency down to 0.

The example below illustrates the simplicity and speed of Ultrabeat's modulation options:

In this example, the modulation intensity of Env 1, which affects Cutoff, is controlled with the dynamics of the performance (Vel). The secondary via modulation also controls its direction. Try this setting in Ultrabeat to create some interesting sounds.

Create a modulation routing in Ultrabeat

The following applies to all parameters that offer mod (and via) modulation options.

Create a modulation routing

1. Click the mod label of the parameter you want to modulate.
2. Choose a modulation source from the mod pop-up menu.

- *Off:* Deactivates the mod routing, and the mod control can no longer be adjusted. In this situation, no via modulation can occur either, because via no longer has a modulation target, and the via control disappears.
- *Lfo1-Lfo2:* Choose one of the LFOs (low frequency oscillators) as the modulation source.
- *Env1-Env4:* Choose one of the envelope generators as the modulation source.
- *Max:* Produces a static modulation at maximum level. When the mod value is set to Max, the via parameter is routed directly to the modulation target. Velocity can then be used as a direct modulation source, even though Vel is not available as a source in the mod pop-up menu.

  Tip: You can also set up an external MIDI fader unit with Ctrl A, B, C, or D (see Assign Ultrabeat MIDI controllers A–D on page 361). Use the Max menu item to route a via source—Ctrl A, B, C, or D—to the parameter you want to control with a fader on your MIDI fader device.

3 If you want to assign a via source, choose one of the following from the via pop-up menu.

- *Vel:* Velocity is used as the via modulation source.
- *CtrlA to CtrlD:* Choose one of these continuous controllers that can be assigned to four external MIDI controllers. These assignments apply to all sounds in the current Ultrabeat plug-in instance. See Assign Ultrabeat MIDI controllers A–D.

4 Adjust the mod and via controls.

Assign Ultrabeat MIDI controllers A–D

The MIDI Controller Assignments area enables you to assign any MIDI controller shown in the menus to each of the four controller slots—Ctrl A, B, C, or D.

These assignments enable external MIDI controller hardware—such as sliders, knobs, aftertouch, or the modulation wheel of your MIDI keyboard—to control via modulation sources in Ultrabeat.

Assign a controller

- Click the control menu (Ctrl A–D) that you want to assign, then choose the controller name or number that you want to use.

Learn a MIDI controller assignment

1 Open the control menu (Ctrl A–D) that you want to assign, then choose the Learn menu item.

2 Move the controller on your MIDI keyboard or MIDI controller unit.

Note: If no suitable MIDI message is received within 20 seconds, the selected control reverts to the previous value/assignment.

Use Ultrabeat LFOs

Two identical LFOs are available as modulation sources in the mod pop-up menus.

The LFO (low frequency oscillator) signal is used as a modulation source. In an analog synthesizer, the LFO frequency generally ranges between 0.1 and 20 Hz, which is outside the audible frequency spectrum. Therefore, this type of oscillator is used only for modulation. The speed of the LFO in Ultrabeat can reach up to 100 Hz, which affords a number of possibilities that analog synthesizers don't offer.

LFO parameters

The parameters for both Ultrabeat LFOs are described below. You can adjust LFO 1 and LFO 2 independently of each other.

- *LFO 1 and 2 buttons:* Click to select the corresponding LFO, allowing independent parameter adjustments for each.
- *On/off button:* Turn the selected LFO on or off.
- *Sync/free buttons:* The LFO speed (Rate) can be synchronized (Sync) to the host application tempo or set independently (Free). Click either button to activate the corresponding mode.
- *Rate knob:* Rotate to set the speed of the LFO. Depending on the Sync/Free setting, the rate is displayed in Hertz or rhythmic values—the latter when tempo synchronization is active. Rates range from speeds of 1/64 notes to a periodic duration of 32 bars. Triplet and punctuated values are also available.
- *Waveform shape slider (and display):* Drag to determine the shape of the LFO waveform.
- *Cycles knob and field:* Rotate to set the number of times the LFO waveform repeats.
- *Ramp knob and field:* Rotate to set the time it takes for the LFO modulation to fade in or fade out. The Ramp value is displayed in milliseconds.
  - Rotate Ramp to the right to set the LFO fade-in time.
  - Rotate Ramp to the left to set the LFO fade-out time.
  - At the middle position, Ramp has no effect on the LFO.

Set Ultrabeat LFO waveforms

- Drag the Waveform Shape slider from left to right to morph the waveform from a triangle, to a sawtooth, sine, square, and finally a rectangular wave shape—including all variations in between. At the far right position, the LFO produces random waveforms.

  The graphical display shows the current LFO waveform shape.

The table outlines how different waveform shapes can affect your sounds. Intermediate waveform shapes result in hybrid waveforms and hybrid behaviors.

Waveform	Comments
Triangle	Well suited for vibrato effects
Sawtooth	Well suited for helicopter and space gun sounds. Intense modulations of oscillator pitch with a sawtooth wave lead to "bubbling" sounds. Intense sawtooth modulations of lowpass filter cutoff and resonance create rhythmic effects.
Sine	Ideal for smooth, even modulations. Its position on the Waveform Shape slider enables you to smoothly morph between sawtooth and square/rectangular waves.
Square and Rectangle	Square/rectangular waves periodically switch the LFO between two values. The right-hand rectangular wave switches between a positive value and zero. The left-hand rectangular wave switches between a positive and a negative value set to the same amount above/below zero.
Sample & Hold	The right-hand waveform on the Waveform Shape slider outputs *random* values. A random value is selected at regular intervals, as defined by the LFO rate. The term *Sample & Hold* (S & H) refers to the procedure of taking samples from a noise signal at regular intervals. The values of these samples are then *held* until the next *sample* is taken. *Tip:* A random modulation of oscillator pitch leads to an effect commonly referred to as a *random pitch pattern generator* or *sample and hold*. Try using very high notes, at very high rates and high intensities—you'll recognize this well-known effect from hundreds of science fiction movies.

Set Ultrabeat LFO waveform cycles

An LFO normally oscillates continuously. On percussive signals it can, however, be interesting to limit the LFO cycles (repetitions of the entire waveform) to a defined number. Ultrabeat enables you to set the number of LFO cycles with the Cycles parameter. After completing the defined number of cycles, the LFO stops oscillating.

- Rotate the Cycles knob to set the number of LFO waveform cycles. The range of Cycles parameter values extends from 1 to 100. The Cycles parameter can also determine whether the LFO waveform is started from the beginning, at a zero-crossing point, with each note trigger, or continues oscillating.
  - A Cycles value of 1 allows the LFO to function as an additional, very basic, envelope generator.
  - Set Cycles to its maximum value (full right position) for an infinite number of cycles (standard LFO behavior). The LFO is not reset by incoming MIDI note-on messages.
  - When Cycles is set to values under 100, the LFO is reset by each new MIDI note-on message (Note On Reset).

    Your choice to trigger an LFO cycle from the same spot or to allow it to oscillate freely, regardless of phase, should be based on the needs of the sound. The random element of free-running LFOs can make many sounds richer. This, however, can be at the expense of a percussive attack—which is generally inappropriate for drum sounds.

Tip: Try small Cycles parameter values, with the LFO source used to control the Volume (Level) of one or both oscillators. This results in drum flams or hand claps. You can also use minor shifts of the LFO phase, with the Cycle value set to Infinity, to add an analog character to a drum sound.

Ultrabeat envelope overview

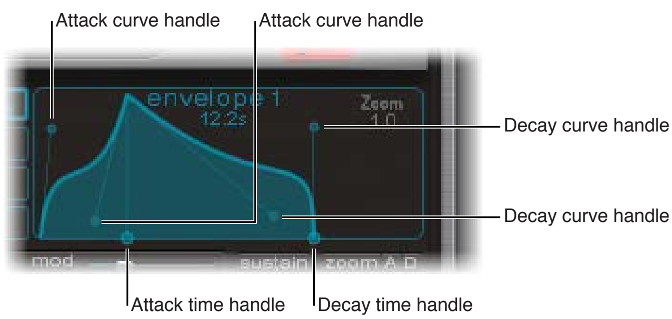
Ultrabeat features four identically specified envelope generators per voice. They are abbreviated as Env 1 to Env 4. In addition to potential use as a modulation source (in the mod pop-up menus of various sound parameters), Env 4 is permanently connected to the Voice Volume parameter. In other words, each Ultrabeat drum sound has a hard-wired volume envelope generator—Env 4.

See Attack, decay, sustain, and release on page 486 for information on the roots of the term *envelope generator* and its basic function.

The default behavior of the envelope generators is known as the *one-shot envelope mode:* after a key is pressed (note-on message), the envelopes run their course, regardless of how long the note is held. This setting is ideal for percussive signals, because it emulates the natural behavior of acoustic percussion instruments.

For special cases, such as sustained pad or cymbal sounds, you can activate a sustain mode where the envelopes take the lengths of the played notes into account.

Ultrabeat's envelope display provides a unique envelope design, consisting of Bezier curves in which two segments—attack and decay—constitute the entire envelope.



In the envelope graphic, you can see various handles (junction points) of two different sizes. Drag these handles to adjust the envelope shape.

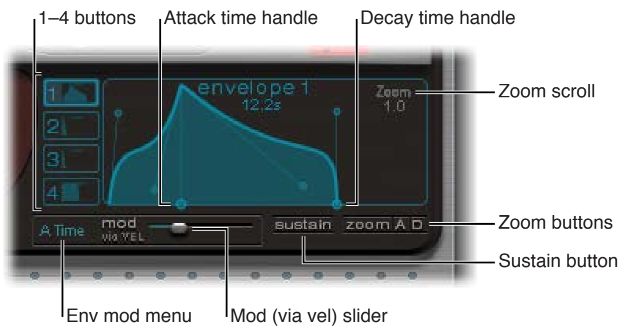
Both of the larger handles on the x-axis (the horizontal, or time axis, at the bottom) control the attack and decay times, respectively. A vertical line extends up from the first of the two handles (attack), and divides the envelope into an attack and decay phase.

Both segments have two small curve handles. You can drag these in any direction to deform the contour of the envelope and shape its amplitude.

You can also directly drag anywhere on the curve itself to reshape the envelope.

Ultrabeat envelope parameters

To edit envelope parameters, you first need to select one of the four envelopes with the 1-4 buttons. The parameters of the corresponding envelope can then be changed in the envelope display window.



Envelope parameters

- *Buttons 1–4:* Click to select one of the four envelopes. Only the selected envelope can be edited. The button frame of the chosen envelope is highlighted, and the envelope display immediately updates to reflect your selection.
- *Attack time handle:* Drag to set the time it takes for the envelope to reach its maximum value after receiving a note-on message. This period is called the *attack phase*.
- *Decay time handle:* Drag to set the time it takes for the envelope to fall to an amplitude of zero, after it has reached its maximum value (defined in the attack phase).
- *Zoom scroll field:* Drag horizontally to resize the visible contents of the envelope display.
- *Env mod pop-up menu:* Choose the modulation target (either the time or shape of the envelope attack or decay phase) by velocity. Choices are A Time, A Shape, D Time, and D Shape.
- *Mod (via vel) slider:* Drag to set the intensity of velocity modulation (of the target specified in the Env mod pop-up menu).
  - When you modulate Shape, low velocity values make the envelope concave. Higher values make the envelope convex.
  - When you modulate Time, high velocity values reduce the length of the envelope segment. Lower velocity values increase the length of the envelope segment.
- *Sustain button:* Click to turn on and display a red handle (and vertical line) on the x-axis. This handle can be dragged horizontally, but only within the envelope decay phase. The amplitude that the envelope reaches at the Sustain junction point is retained until the MIDI note is released.

  Note: If the Sustain button is not turned on, the envelope functions in one-shot mode, and the note length (MIDI note-off command) is disregarded.
- *Zoom (to fit) button:* Click to enlarge the envelope to fill the entire width of the envelope display. This makes it easier to adjust junction points and curves.

  Note: When the Zoom function is turned on, the decay handle can be dragged beyond the right edge of the envelope display area, enabling you to lengthen the decay time. After you release the mouse button, the envelope graphic is automatically resized to fit the display area.
- *Zoom A/D buttons:* Click to show only the attack (A) or decay (D) phase across the entire width of the envelope display. This enables you to perform more accurate edits to envelope shapes (down to millisecond values).

Use Ultrabeat's modulation target display

Ultrabeat has a feature that makes it easy to find the modulation targets of LFOs and envelopes.

Find modulation targets of LFOs and envelopes

- Click the numerical field of the modulation source to highlight all of its modulation targets.

Highlighted modulation
target of Envelope 1

Click here to highlight
all modulation targets of
Envelope 1.

Ultrabeat step sequencer

Ultrabeat step sequencer overview

Ultrabeat incorporates a powerful, integrated step sequencer, which you can use to create polyphonic rhythmic sequences and patterns. The sequencer displays running light-style controls like those of classic drum machines and shares many of the sequence and pattern creation methods employed in these devices.

Ultrabeat's step sequencer expands on the features of hardware drum machines by providing extensive automation and editing features. These enable you to precisely vary the timbre of the sound and the overall dynamics at any point in the pattern. The step sequencer plays an important role in shaping the rhythms and sounds that you can produce with Ultrabeat.

The step sequencer allows all Ultrabeat sounds to be combined in patterns, based on sequences for each individual sound. Its design and use—commonly referred to as step programming—are based on analog sequencers and drum machines. Unlike these analog precursors, Ultrabeat enables you to program automated changes for nearly every synthesizer parameter.

Depending on your working preference and favored musical style, you can control Ultrabeat with either the integrated step sequencer or Logic Pro when programming rhythms. Combining both sequencers is also possible; they can be active at the same time and are automatically synchronized with each other. Logic Pro acts as a master clock in this situation, determining the tempo of Ultrabeat's internal step sequencer.

If you're unfamiliar with the concept of step sequencing, see Step sequencer basics.

Step sequencer basics

The fundamental idea behind analog step sequencers is to set up a progression of control voltages and output these step by step, typically in an endlessly repeating pattern. This principle helped to spawn a number of electronic music styles based on the mesmerizing effect that repeating patterns can have.

In early analog sequencers, three control voltages were usually created per step to drive different parameters. The most common usage was control of a sound's pitch, amplitude, and timbre (cutoff) per step.

The control surface of analog sequencers often contained three rows of knobs or switches aligned on top of (or beside) each other. Each row commonly contained 8 or 16 steps. Each row provided a control voltage output that was connected to a control input (for a particular parameter) on a synthesizer. A trigger pulse determined the tempo between steps. A running light (an LED) indicated the step that was currently being triggered.

The running light programming concept also appeared in later drum computers, the most well-known examples being the Roland TR series drum machines.

The introduction of the MIDI standard and increased use of personal computers for music creation led to a rapid decline in the step sequencer and related technology. More flexible recording and arranging concepts that didn't adhere to the step and pattern principle were possible with the far more powerful personal computer.

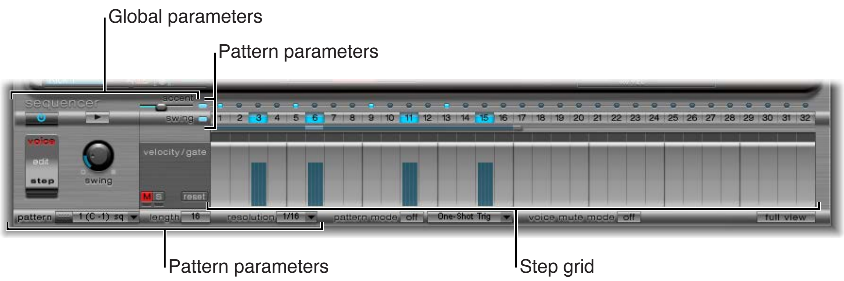
Despite these technological advances, step sequencers haven't disappeared completely. Hardware groove boxes have experienced a renaissance in recent years due to their intuitive nature, which has made them a favored tool for rhythm programming.

Ultrabeat's integrated step sequencer couples the advantages and general working principles of its analog forebears with significantly more flexible control options, making it a powerful tool for rhythm creation.

Ultrabeat step sequencer interface

Ultrabeat's step sequencer contains a *sequence* for each sound in a drum kit. Each sequence can consist of up to 32 steps.

A *pattern* is a container for all sequences in a drum kit. Up to 24 patterns can be saved and recalled with each Ultrabeat setting.



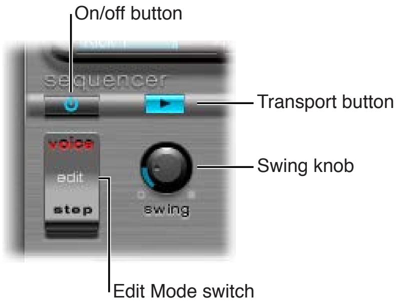
The step sequencer is divided into three sections.

- *Global parameters:* Turn the step sequencer on or off, control playback, provide access to various modes, and control the overall playback feel. See Ultrabeat global sequencer controls.
- *Pattern parameters:* Provide control over the length and resolution of the currently selected pattern. You can also accentuate individual steps in the pattern—for each drum sound. See Ultrabeat pattern controls.
- *Step grid:* This is where actual sequencing takes place. A sequence of up to 32 steps, for the sound that is currently selected in the Assignment section, is shown. You can add, remove, or alter events in the grid. See Ultrabeat Step grid overview.

Note: An alternate view allows you to simultaneously see and edit the steps of all drum sounds in the pattern. See Ultrabeat Step grid full view on page 376.

Ultrabeat global sequencer controls

The following parameters apply globally to all patterns.



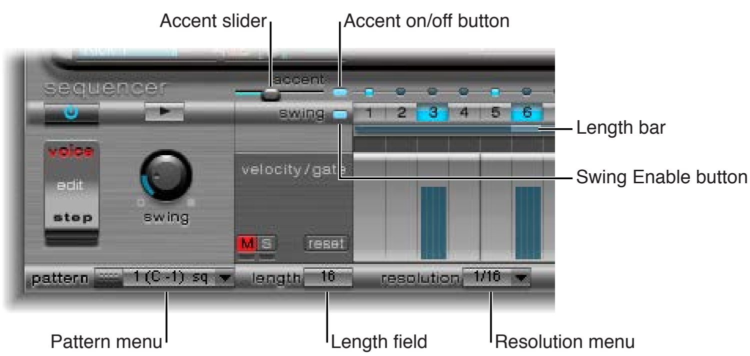
Global sequencer parameters

- *On/Off button:* Turns the step sequencer on or off.
- *Edit Mode switch:* Select either Voice or Step mode.
    - *Voice mode (default):* In Voice mode, editing a drum sound's parameters changes the drum sound itself.
    - *Step mode:* In Step mode, you can automate a sound's parameters from one step to the next. See Ultrabeat step automation overview on page 377.
- *Transport button:* Starts and stops the sequencer pattern. The step sequencer is always synchronized to the host application tempo.

  Note: If the Transport button is blue, the step sequencer interprets incoming MIDI notes between C-1 and B0 as performance information. See MIDI control of Ultrabeat's step sequencer on page 380.
- *Swing knob:* Rotate to set the swing intensity for all sounds that have the Swing function turned on. See Use Ultrabeat's swing function on page 372.

Ultrabeat pattern controls

A pattern contains all events, stored in sequences, for all 25 sounds. You can select one of 24 patterns and set parameters that globally affect all sounds in the kit.



Pattern parameters

- *Pattern pop-up menu:* Click to choose one of the 24 patterns.
- *Length field and bar:* You can adjust the length of the grid (and therefore the pattern) by dragging the value in the Length parameter field or the bar beneath the swing buttons.

- *Resolution pop-up menu:* Click to choose the resolution of the pattern. Resolution defines the metric unit of a measure that is represented by the individual steps. For example, the 1/8 setting means that each step of the grid represents an eighth note. Given a pattern length of 32 steps, the pattern would run for 4 measures (the 32 setting applies to the entire grid and, therefore, all sounds).

  Note: The interplay between Length and Resolution values enables you to create different time signatures. For example, a Length value of 14 and a Resolution of 1/16 results in 7/8 time; a Length of 12 and a Resolution of 1/16 results in 3/4 time; and a Length of 20 and a Resolution of 1/16 results in 5/4 time.
- *Accent button and slider:* Allows individual steps to be strongly emphasized, or accentuated. See Set Ultrabeat step sequencer accents on page 376.
- *Swing Enable button:* When turned on, the grid of the currently selected sound is played in accordance with the Swing knob setting. See Use Ultrabeat's swing function on page 372.

You can reorganize the 24 patterns in the Pattern pop-up menu, using Copy and Paste commands.

Copy a pattern using a shortcut menu

1. Choose the pattern that you want to copy from the Pattern pop-up menu.
2. Control-click (or right-click) the Pattern pop-up menu, and choose Copy from the shortcut menu.
3. Choose the target pattern from the Pattern pop-up menu.
4. Control-click the Pattern pop-up menu, and choose Paste from the shortcut menu.

Copy a pattern using the key command

You can also use a key command to copy patterns.

1. Choose the pattern that you want to copy in the Pattern pop-up menu.
2. Press Option, open the Pattern pop-up menu, and choose another Ultrabeat pattern.

   The pattern in the target position is replaced with this one.

   Note: All existing sequencer data in the target pattern is replaced. If you change your mind during the process, choose the source pattern number.

Clear a pattern

1. Choose the pattern that you want to clear in the Pattern menu.
2. Control-click (or right-click) the Pattern pop-up menu, and choose Clear from the shortcut menu.

Use Ultrabeat's swing function

Swing changes the distance between notes. Only even-numbered steps are affected by the Swing parameter, with notes on odd-numbered steps remaining where they are.

Which beats are affected depends on the selected Resolution parameter setting, as illustrated in the following example: At a Resolution of 1/8 and a Length of 8, the notes on steps 1, 3, 5, and 7 represent quarter notes in the measure. These remain unchanged. Only the eighth notes found between them (steps 2, 4, and so on) are shifted by the Swing function. The amount of shift is equal to the swing intensity (set with the Swing knob).

Note: Swing is active only for grid resolutions of 1/8 and 1/16. See Ultrabeat pattern controls on page 370.

Use the swing function

1. Click the Swing Enable button.

   This forces the grid of the currently selected sound to be played in accordance with the Swing knob setting.

2. Adjust the Swing knob.

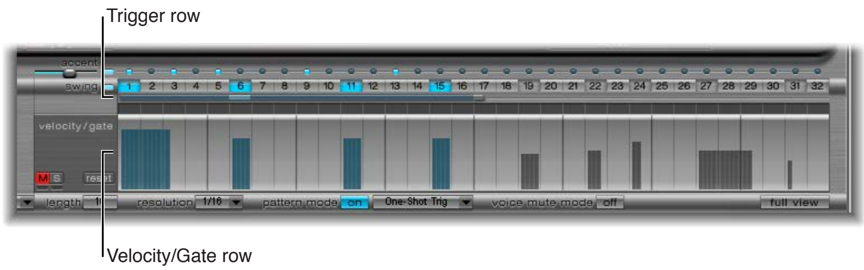
   At a zero setting (full left position), the Swing function is disabled. Rotate the knob to the right to move affected notes toward the following note.

Ultrabeat Step grid

Ultrabeat Step grid overview

The Step grid displays sequence steps on two rows. The steps shown in these rows correspond to the sound that is currently selected in the Assignment area. Choosing a different sound switches the sequencer display to show the rows that correspond to the newly selected sound.

The Step grid area contains two rows, each with 32 fields (steps).

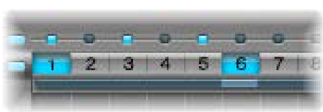


- *Trigger row:* Click a button to activate or deactivate the sound on the corresponding beat.
- *Velocity/Gate row:* Sets the length (gate time) and velocity of steps entered in the Trigger row. Both parameters are displayed as a single graphical bar.
  - The bar's height represents the velocity.
  - The bar's length, from left to right, indicates the note length.

Create and remove steps

The Trigger row consists of buttons 1 through 32, which represent steps or beats in the selected sequence. Note trigger events are placed on corresponding steps. This is where you designate when (on which beat) the selected sound plays.

Note: You can create and remove steps while the step sequencer is running.



Create steps

1. Select the drum sound that you want to create steps for in the Assignment section.
2. Click the On button to start the step sequencer.
3. Choose a pattern and set the length and resolution that you want to use. See Ultrabeat pattern controls on page 370.
4. Click the buttons you want—1 through 32—to activate or deactivate the selected sound on the corresponding beat. In the example shown above, these are steps 1 and 6.

Note: An alternate view allows you to simultaneously see and edit the steps of all drum sounds in the pattern. See Ultrabeat Step grid full view on page 376.

Remove a step

1. Select the drum sound that you want to remove steps from in the Assignment section.
2. Click the buttons—1 through 32—that correspond to the step or steps that you want to remove.

Note: Drag horizontally across the buttons to quickly turn trigger events on or off.

Ultrabeat trigger shortcut menu

Control-click (or right-click) any of the trigger buttons to open the Trigger shortcut menu, which contains the following commands:

Copy, Paste, and Clear commands

- *Copy:* Copies all activated triggers (steps) to the Clipboard.
- *Paste:* Pastes all triggers from the Clipboard.
- *Clear:* Turns off all activated triggers.

Create Beat commands

- *Add Every Downbeat:* Adds triggers on every downbeat in the sequence. The determination of steps as downbeats depends on the grid resolution. For example, if the resolution is set to 1/16, Add Every Downbeat would create triggers on every fourth step. Starting with the initial downbeat at step 1, this would create trigger events on step 5, step 9, step 13, and so on. This command doesn't erase existing trigger events; it only adds trigger events.
- *Add Every Upbeat:* Adds triggers on every upbeat in the sequence. The determination of steps as upbeats depends on the grid resolution. For example, if the resolution is set to 1/16, Add Every Upbeat would create triggers on every 4th step. Starting with the initial upbeat at step 3, this would create trigger events on step 7, step 11, step 15, and so on. This command doesn't erase existing trigger events; it only adds trigger events.

Alter, Reverse, and Shift commands

- *Alter Existing Randomly:* Randomly reorders steps while retaining the number of active triggers.
- *Reverse Existing:* Reverses the order of existing steps.
- *Shift Left 1 Step:* Shifts all steps in the sequence one step to the left.
- *Shift Left 1/2 Beat:* Shifts all steps in the sequence one-half beat to the left. The number of steps that equals one-half of a beat depends on the current grid resolution. For example, at a resolution of 1/16, a beat equals four steps, so half of one beat is two steps; at a resolution of 1/8, a beat equals two steps, so half of one beat equals one step, and so on.
- *Shift Left 1 Beat:* Shifts all steps in the sequence one beat to the left. The number of steps that equals a beat depends on the current grid resolution. For example, at a resolution of 1/16, a beat equals four steps; at a resolution of 1/8, a beat equals two steps, and so on.
- *Shift Right 1 Step:* Shifts all steps in the sequence one step to the right.
- *Shift Right 1/2 Beat:* Shifts all steps in the sequence one-half beat to the right. The number of steps that equals one-half of a beat depends on the current grid resolution. For example, at a resolution of 1/16, a beat equals four steps, so half of one beat is two steps; at a resolution of 1/8, a beat equals two steps, so one half of one beat is one step, and so on.
- *Shift Right 1 Beat:* Shifts all steps in the sequence one beat to the right. The number of steps that equals a beat depends on the current grid resolution. For example, at a resolution of 1/16, a beat equals four steps; at a resolution of 1/8, a beat equals two steps, and so on.

Create and Replace commands

- *Create & Replace Randomly:* Erases, then randomly creates new, steps in the sequence; in other words, a brand new sequence is created. The number of events that are created depends on the grid resolution.
- *Create & Replace Few:* Similar to Create & Replace Randomly, but a limited number of new steps are created. The number of steps that are created depends on the grid resolution.
- *Create & Replace Some:* Similar to Create & Replace Few, but more new steps are created. The number of steps created depends on the grid resolution.

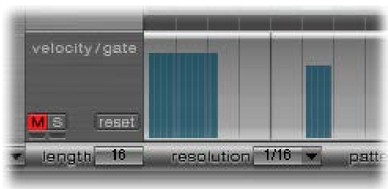
- *Create & Replace Many:* Similar to Create & Replace Some, but a large number of new steps are created, effectively filling the pattern.

  For example, start with an empty sequence of 32 steps at 1/16 resolution. Using Create & Replace Few creates 4 new steps; using Create & Replace Some creates 8 new steps; and using Create & Replace Many creates 16 new steps.

Set step lengths and velocities

The Velocity/Gate row enables you to set the length (gate time) and velocity of the notes entered in the Trigger row. Both parameters are displayed as a single graphical bar.

- The bar's height represents note velocity.
- The bar's width indicates the note length (gate time).



Change the length and velocity values for each step

Do either of the following:

- To change the velocity, drag the blue bar vertically.
- To change the note length (gate time), drag the blue bar horizontally.

The gate time is divided into four equal sections, making it easy to set rhythmically accurate note lengths. For the one-shot envelope to react to gate time, it is necessary to either turn on the Gate function in the sound itself (see Change Ultrabeat's trigger mode on page 357) or use envelopes in sustain mode (see Ultrabeat envelope parameters on page 366), in conjunction with rhythmically useful (short) decay times.

Reset all velocity and gate values to default settings

- Click the Reset button at the left of the velocity/gate row.

The default velocity setting is 75 percent. The default gate value is all four sections active.

Use the Ultrabeat velocity/gate shortcut menu

- Control-click (or right-click) any step in the velocity/gate row to open a shortcut menu that has the following commands:
  - *Alter Vel(ocities):* Changes the velocity values of all steps by a random amount, while retaining the selected beats (the Trigger row remains unchanged).
  - *Alter Gate (Time):* Changes the note lengths of all steps by a random amount, while retaining the selected beats (the Trigger row remains unchanged).
  - *Randomize Vel(ocities):* Creates a new, random velocity value.
  - *Randomize Gate (Time):* Creates a new, random gate value.

Set Ultrabeat step sequencer accents

The Accent setting can be switched on or off individually for each drum sound. This enables you to turn accents on for cymbals but turn accents off for the kick drum, for example.

Turn on accents and set the accent level

1. Click the blue LED to the right of the Accent slider to turn on the accent function.
2. Drag the Accent slider to globally set the volume of programmed accents.

Program an accent for a particular step

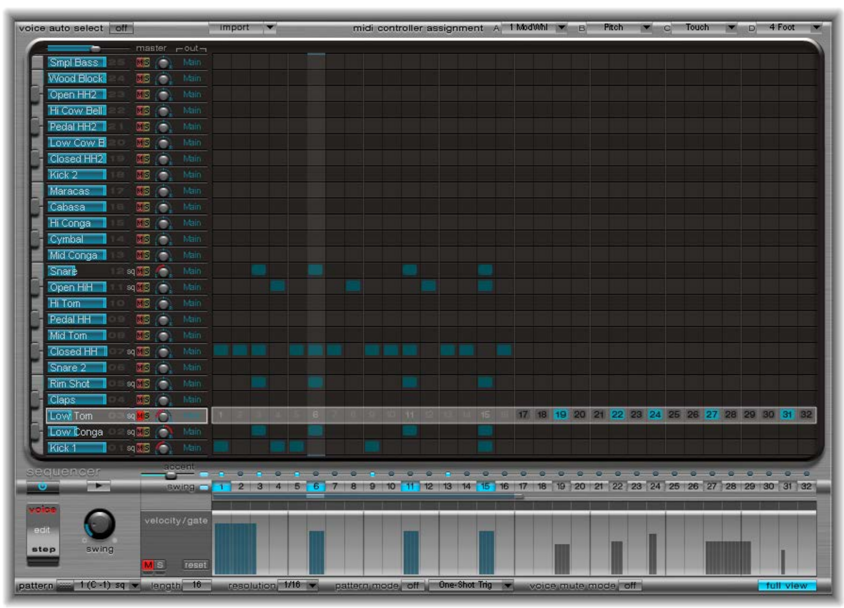
- Click the blue LED above the step (steps 1 and 3 in the figure).

The sound at this step position is accentuated (played louder).

Ultrabeat Step grid full view

Click the Full View button in the lower-right corner to see a large sequencer grid filled with trigger buttons. The large grid simultaneously displays the 32 trigger buttons for all 25 drum sounds.

Full view provides an overview of the whole pattern, rather than a single drum sound sequence. Because the trigger events for all sounds are shown, you can quickly identify accidental notes. Full view also accelerates the pattern creation process.



The selected sound is highlighted with a gray box in the step sequencer area, making it easy to set the velocity and gate time for each step or offsets in Step mode (see Ultrabeat step automation overview on page 377), within the context of all sequences/sounds in the pattern.

Both the grid and the Trigger and Gate/Length rows are displayed for the selected drum sound. This makes it simple to create trigger events in the full view grid, then set accents in the Gate/Length rows, for example.

Automate parameters in Ultrabeat's step sequencer

Ultrabeat step automation overview

Click the Edit Mode switch to turn on Ultrabeat's step automation feature. Step automation enables you to program parameter changes on a per-step basis for each drum sound. Sound parameters that can be automated include all functions in the Synthesizer section except:

- Menus (modulation routings and so on)
- Buttons (oscillator-type buttons, the buttons in the trigger/group section)
- Pan/Spread parameters

When you turn on Step mode, Ultrabeat's interface changes in the following ways:

- Yellow frames appear around all parameters that can be automated in the Synthesizer section. Parameters that cannot be automated are still visible but are disabled.
- The velocity/gate row in the Step grid changes to show the (parameter) offset row.

Tip: When creating offsets in step mode, you may want to make a change to the original drum sound. Rather than switching back and forth between edit modes for this adjustment, you can press Command-Option to temporarily flip Ultrabeat back into Voice mode.

Use Ultrabeat's offset row

This row enables you to view and enter offset values on a per-step basis for all Synthesizer section parameters that can be automated. Adjustments made in the offset row are relative to the current parameter value. Values shown in the offset row will be either added to or subtracted from the parameter value set in the Synthesizer section. In other words, parameter offsets increase or decrease, but do not specify, an absolute value for the parameter.

Parameter edits can be made in three ways:

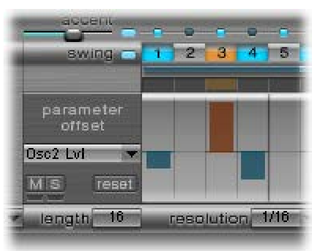
- By editing offset values in the offset row.
- With the shortcut menu.
- By directly adjusting the controls in the Synthesizer section. All parameters that you automate (create an offset for) are shown in the Parameter Offset pop-up menu.

Important: Moving a control element in the Synthesizer section will add the parameter to the Parameter Offset pop-up menu, and will create an offset, so take care.

Create a parameter offset

1. Select the drum sound that you want to offset.
2. Click a position in the (parameter) offset row that corresponds to the step you want to edit.
3. Make the parameter change in the Synthesizer section. Your change will be recorded as an offset value for this step.
4. Repeat step 3 for each parameter that you want to edit for this step.

A parameter offset that has been created for a given parameter on a given step is represented in two ways.



- A yellow bar is drawn on the parameter that indicates the deviance (the offset) between the original parameter value and the new parameter value.
- In the (parameter) offset row, the offset from the original parameter is represented as a bar starting from the zero point (horizontal center line).
  - Positive offsets are shown as a bar above the center line.
  - Negative offsets are shown as a bar below the center line.

Use the Ultrabeat parameter offset shortcut menu

- Control-click (or right-click) any step in the (parameter) offset row to open a shortcut menu, then choose one of the following commands.
  - *Alter:* Changes the (selected) parameter values, for all steps, by a random amount.
  - *Randomize:* Creates a new, random value for the selected parameter.

    Note: Consider saving your sequence/pattern before using either of the above commands.
  - *Delete:* Deletes all steps for the currently selected parameter.

Mute, solo, and reset Ultrabeat parameter offsets

- Click the M(ute), S(olo), and Reset buttons in the parameter offset row.

  - *M(ute):* Mutes the offsets of the selected parameter. This does not remove or reset existing offsets.
  - *S(olo):* Enables you to hear the effect of your offsets on the selected parameter only.
  - *Reset:* All offset values for the selected parameter are set to zero (no offset). A second click on the Reset button removes the parameter from the Parameter Offset menu.

    Note: The Reset button at the left of the velocity/gate row changes to Delete when clicked once. This Delete button mirrors the behavior of the Delete command: It deletes all steps for the currently selected parameter.

Export Ultrabeat patterns as MIDI regions

Patterns programmed in Ultrabeat's internal step sequencer can be exported as MIDI regions into Logic Pro.

Export an Ultrabeat pattern into Logic Pro

1. Choose the pattern that you want to export from Ultrabeat's Pattern pop-up menu.
2. Select the area to the left of the Pattern pop-up menu.

3. Drag the pattern to the target position on the respective Ultrabeat track.

   A region is created containing all MIDI events, including Swing and Accent parameter settings. Accents are interpreted as polyphonic pressure events. Step automation events, created in Step mode, are also exported as part of the MIDI region.

   Note: To avoid double-triggering while playing back the exported MIDI region, turn off Ultrabeat's step sequencer.

MIDI control of Ultrabeat's step sequencer

Pattern performance can be influenced by incoming MIDI notes. This enables spontaneous interaction with the step sequencer, making Ultrabeat an excellent live performance instrument. The way Ultrabeat reacts to MIDI control is determined by the chosen pattern, playback, and voice mute mode options.

MIDI control parameters

- *Pattern Mode (On/Off) button:* Allows you to choose and start patterns with MIDI note messages. The Transport button turns blue to indicate that Ultrabeat is ready to receive incoming control commands.

  MIDI notes C-1 to B0 switch between patterns: C-1 selects pattern 1, C#-1 selects pattern 2, and so on up to pattern 24 (selected when MIDI note B0 is received).
- *Playback Mode pop-up menu:* Determines pattern playback behavior when an incoming MIDI note is received. You can choose one of the following options:
  - *One-Shot Trig(ger):* The reception of a MIDI note starts the pattern, which plays once through its cycle, then stops. If the next note is received before the pattern has reached its final step, the new note stops playback of the first pattern and the next pattern begins playing immediately—this can be a different pattern or the same pattern, depending on the MIDI note received. Note-off events are ignored.
  - *Sustain:* The reception of a MIDI note starts the pattern and it continues playing in an infinite loop until the corresponding MIDI note is released (a note-off event is received).
  - *Toggle:* The reception of a MIDI note starts the pattern and it continues playing in an infinite loop until the next note is received. If it is the same note, the pattern stops immediately. If it is a different note, the sequencer immediately switches to the new pattern.

  Toggle mode allows you to switch between patterns in the middle of a bar—the sequencer stays in time and automatically jumps to the corresponding beat of the new pattern. This isn't the case in One-Shot Trig mode, which starts the new pattern from the beginning as soon as you play a MIDI note.
  - *Toggle on Step 1:* The behavior is the same as Toggle mode except that the pattern change or stop occurs the next time beat 1 is reached—at the beginning of the next pattern cycle.
- *Voice Mute Mode button:* Playing MIDI note C1 and above mutes the corresponding sound in the Ultrabeat mixer. A subsequent MIDI note of the same pitch unmutes it. This is ideal for spontaneous rearranging of patterns and/or muting single elements of a pattern without deleting them. This is especially useful in a live performance or remixing situation.

All the creative pattern switching options discussed in this section are achieved with MIDI note messages and can be recorded, edited, arranged, and automated in Logic Pro.

Ultrabeat tutorials

Ultrabeat sound programming overview

The Ultrabeat tutorials presented in these sections cover a number of specific sound creation tips. These tutorials will help you explore the possibilities available to you in Ultrabeat. You'll discover that there is hardly a category of electronic drum sound that Ultrabeat can't create easily.

As you become familiar with drum sound programming, you may begin thinking in building blocks, realizing that drum sounds usually consist of different components.

After you mentally—or physically—write down your list of components, try to emulate each component that contributes to the sound's character, making use of the different sound generators available in Ultrabeat. Assigning dedicated amplitude envelopes to the different components allows you to control their temporal behavior individually. For example, you can emulate the body of a drum with oscillator 1 and the sound of the stick hitting the skin (or first transient) with the noise generator. Additional overtones and harmonics can be provided by oscillator 2 or the ring modulator.

When you begin thinking that drum sounds consist of several building blocks or layers, the design of the Volume controls in the individual sound generators might make more sense to you—this is the place where the blocks are combined, balanced, and controlled.

Note: Choose the Tutorial Kit from Ultrabeat's Settings > 03 Tutorial Settings folder. This kit contains all drum sounds discussed in the tutorials. The Tutorial Kit also includes the Standard Tut(orial) drum sound, which is a default set of neutral parameters that provide an excellent starting point for many of the examples.

All Ultrabeat tutorial sections are listed below:

- Create Ultrabeat kick drums on page 382
- Create Ultrabeat snare drums on page 386
- Create Ultrabeat tonal percussion on page 391
- Create Ultrabeat hi-hats and cymbals on page 391
- Create metallic Ultrabeat sounds on page 392
- Tips for extreme Ultrabeat sounds on page 392

Create Ultrabeat kick drums

Electronically produced kick drum sounds are based primarily on the sound of a deeply tuned sine wave. Follow the tutorials sequentially to get the most from these examples.

Program a basic kick drum in Ultrabeat

1. Choose Settings > 03 Tutorial Settings > Tutorial Kit, and select Standard Tut from the Assignment section.

   Note that oscillator 1 is in Phase Oscillator mode.

2. Find a suitably tuned pitch in the lower octaves by soloing the bass drum along with other important tonal elements of the song (a bass or pad sound, for example). Drag the Osc 1 Pitch slider to adjust the pitch until appropriate.
3. Use Env 4 to shape the volume of the bass drum.

   For slower beats you'll want a longer decay phase, whereas at faster tempos you'll choose a shorter decay time. The attack time of Env 4 should be very short in any case (0, in most cases) or the sound will lose its percussiveness and its ability to be clearly heard in the mix.

Give your drum more kick by controlling the pitch with an envelope

The kick drum still sounds very soft and is somewhat reminiscent of the famous TR-808 bass drum. It's still missing a clearly defined attack.

1. Make sure that Env 1 is chosen from the mod pop-up menu of oscillator 1's Pitch parameter.
2. Set the degree of modulation by dragging the blue Mod slider approximately 3 to 4 octaves above the original pitch.

3. Set the attack time in Env 1 to 0 by dragging the leftmost of the two junction points on the x-axis all the way to the left.

4. Experiment with the decay time by dragging the rightmost of the two junction points on the x-axis. You'll discover that higher decay values (shifting the Bezier handle to the right) result in sounds similar to synth toms, whereas shorter decay values (shifting to the left) provide the kick character.
5. Change the Mod amount (the blue slider) of Osc 1 Pitch again (see step 2).

   The interaction of this parameter with the envelope's decay time provides numerous possibilities for shaping the kick or punch of the bass drum sound.

Note: This simple bass drum sound is listed as Kick 1 in the Tutorial Kit, at a pitch of C1.

Reduce kick drum tonality using the 2 Band EQ

One advantage of bass drums based on sine waves is that their sound can be precisely tuned to match the song. The disadvantage is that a recognizable pitch is not always desirable. Ultrabeat offers several methods to reduce the tonality of the sound. A very effective tool is the 2 Band EQ.

1 For band 1, select the Shelving mode at a frequency of about 80 Hz, a high Q value, and a negative Gain value.

2 For band 2, select the Peak mode at a frequency of around 180 Hz, a medium Q value, and also a negative Gain value.

On the EQ graph, notice how the frequencies around 80 Hz are boosted, while the surrounding frequencies are reduced.

3 Vary the frequency of band 2 (easily recognizable in the blue part of the EQ graph) to influence the extent of bass drum tonality.

Reduce kick drum tonality with a lowpass filter

A further method for reducing the tonality of a drum sound that is rich with overtones is to use a lowpass filter. In this example you will control the cutoff frequency of the filter with an envelope.

1 Reload the Standard Tutorial sound, choose A#0 as the basic pitch in oscillator 1, and modulate it with Env 1.

2 Increase the Saturation parameter value to enhance the overtones of the drum sound.

Note that the output of Osc 1 is directed to the filter, because the filter bypass button (the arrow between Osc 1 and the filter) is activated.

3 Set Filter type to LP 24.

4 Set Cutoff value to 0.10.

5 Set Mod Source for Cut to Env 3.

6 Set Mod Amount for Cut to 0.60.

7 Set Resonance to 0.30.

8 Set the attack time of Env 3 to 0. Use the Decay time of Env 3 to shape the sound of the filtered bass drum.

9 You may also choose to control the filter resonance with an envelope. Make sure you dedicate a single envelope to this function (in this case, use Env 2 as a Mod source for Res). Choose a Mod amount for Res of about 0.80. Select a longer decay time in Env 2 than in Env 3 and listen carefully to the fatter and more atonal bass drum sound achieved through this Res modulation (due to the higher filter resonance).

Note: The bass drum described in the above example is listed as Kick 2 in the Tutorial Kit, at a pitch of C#1. It also features an interesting EQ setting, as described in "Add body and bite to your kick drum."

Add body and bite to your kick drum

Do any of the following:

- *Add some bass to your sound:*

  Use the Kick 2 filtered bass drum sound as a starting point, and try out the remaining parameters in the phase oscillator. You will discover that high saturation values make the sound rounder and add more bass, for example. The character of the example is beginning to head in the direction of a TR-909.

- *Enhance the attack transients of your sound:*

  To get even closer to the TR-909, use an EQ setting as shown in the following figure. Note that the low frequency pressure point around 60 Hz (in the red area on the EQ graph) as well as the assertive punch or kick (the blue area starting at 460 Hz and up) of a 909 bass drum are strengthened. (This EQ setting is already part of the Kick 2 setting.)

- *Use envelopes to change the color of your sound:*

  In the example, all four envelopes are being used. Take some time to play with the shapes of the envelopes, while maintaining the attack and decay settings. Experiment with the junction points of the decay phase in the different envelopes to familiarize yourself with the sound-shaping options available. Start with the decay phase of Env 4, which controls both the volume of oscillator 1 and filter resonance, and observe how reshaping the belly of the envelope can change the character of the sound from crisp and short to round and full.

Create an LFO-modulated "Ultrabeat" kick drum

You can create bass drum sounds that are uniquely "Ultrabeat." Try modulating pitch with an LFO instead of an envelope, for example.

1. Start with the Standard Tutorial sound at a pitch of A#0 (Osc 1 Pitch), and choose LFO 1 as the Mod source in the Osc 1 Pitch section.
2. Set the degree of modulation by dragging the blue Mod slider to a value of A3.
3. Set LFO 1 to a low number of Cycles (25 to 35), a high Rate (starting with 70 Hz and higher) and a medium value for Decay (set the Ramp knob to about –190).
4. Experiment with the LFO waveform and you'll discover that you can attain different nuances in the character of the bass drum attack.
5. Modulate the Asym(metry) parameter with the same LFO, and also vary the Slope and Saturation values.

This method enables you to create very different bass drum sounds with a single oscillator, one LFO, and one envelope (for volume). The character of the sounds can range from soft to punchy, and the degree of tonality in the sound can be adjusted to taste.

Note: The bass drum sound described is listed as Kick 3 in the Tutorial Kit, at a pitch of D1.

Use the second oscillator (with similar settings or with a sample), or use the filter and the ring modulator—the sky's the limit as far as your imagination is concerned, so go ahead and create that next "gotta have it" drum sound.

Note: You can find an "emulation" of the legendary 808 bass drum in Kick 4 in the Tutorial Kit, at a pitch of D#1.

Create Ultrabeat snare drums

The sound of an acoustic snare drum consists primarily of two sound components: the sound of the drum itself and the buzzing of the snare springs. Try to approximate this combination in Ultrabeat with a single oscillator and the noise generator. Follow the tutorials sequentially to get the most from these examples.

Create a basic snare drum

1. Load the Standard Tutorial setting.Turn off oscillator 1, and turn on oscillator 2 (in phase oscillator mode).
2. Choose LFO 1 from the mod pop-up menu of Osc 2 Pitch.
3. Set the pitch value for Osc 2 to around G#2 and set the Mod amount (the blue Mod control) to about 3 to 4 octaves higher.

   You have modulated Osc 2 Pitch with a rapidly vibrating LFO with a medium Ramp Decay value. This eliminates the sine wave—which is not especially desirable for a snare sound, in contrast to the bass drum.
4. Set LFO 1 to a high Rate. Choose a value of 20 for Cycles and –20 for Ramp. Set the LFO Waveform parameter to a value of about 0.58, which is a square wave.
5. Use Env 1 to control the volume of oscillator 2 by setting Vol to the lowest possible value (–60 dB), choosing Env 1 from the mod pop-up menu, and adjusting the modulation intensity to a point just below its maximum value.

   The figure shows the settings of oscillator 2 and Env 1.

6. Experiment with different Slope and Asym values to impart a more or less electronic character to the sound.
7. Turn on the noise generator and control its volume with the same quick envelope used in Osc 2 Volume.
8. Use the filter parameters of the noise generator to roughen up, refine, or add bright frequencies to the noise component of the snare drum sound. Select an LP filter type, and try a filter frequency between 0.60 and 0.90. Modulate it with LFO 1, which you're already using to control the pitch of oscillator 2.

Note: The snare drum sound is listed as "snare 1" in the Tutorial Kit, at a pitch of E1.

Refine the snare drum sound using FM synthesis

1. Turn on oscillator 1 in FM mode. Use Env 1 to control the volume of Osc 1 as well.
2. Choose a pitch for oscillator 1 that's about an octave lower than oscillator 2. Consciously avoid even intervals between the oscillators and detune them slightly from each other. For example, try a pitch setting of F#2 in Osc 2 and E1 in Osc 1, then fine-tune Osc 1 a few cents higher by holding down Shift while adjusting the Osc 1 Pitch slider.
3. Experiment with FM Amount, and add more tone (low FM Amount) or noise (more FM Amount) as desired. Also try modulating the FM Amount with a fast LFO.

   Higher FM Amount values lead to considerably more overtones and a very electronic sound character. If you want to make the sound more acoustic, feed oscillator 1 (and possibly oscillator 2 as well) into the main filter. Use these settings to start: LP 24 mode and a Cutoff value of about 0.60.

   Note: An exemplary snare drum sound that uses FM can be found in the Tutorial Kit at a pitch of F1. It is listed as "snare 2."

Recreate the 808 snare sound

The famous 808 snare is based on two resonating filters and a noise generator, fed through a highpass filter. The mix ratio of the two filters and the volume of the noise generator can be adjusted. This structure cannot be 100% replicated in Ultrabeat.

1. Load the Standard Tutorial setting.

   You are now ready to replicate the resonating filters of the 808 snare using two cleverly programmed phase oscillators.

2. Assign slightly different Slope values to two phase oscillators, and detune them by almost an octave.
3. Adjust the tonal relationship between the oscillators so that it is uneven—from E3 to F2, for example.
4. Control the volume of each oscillator with a different envelope. Adjust the decay times so that the envelope for the lower-tuned oscillator has a longer decay time than the very snappy envelope setting for the higher oscillator.
5. Feed the output of both oscillators into Ultrabeat's main filter, and hollow out the sound with a highpass filter. Activate the filter bypass button in both oscillators. Choose the HP 12 setting in the filter, a Cutoff value around 0.40, and a Resonance value of about 0.70.

   You have just cleverly emulated both of the 808's resonating filters. Shifting the pitch of both oscillators simulates the behavior of the 808's Tone control.

Complete the 808 emulation by adding some noise

1. Switch the noise generator on, and activate the highpass mode in its filter (HP).
2. Set the Cutoff value to about 0.65, Resonance to 0.35, and add a little Dirt (around 0.06).

   The noise generator provides the sustained snare sound. It should be shaped by its own envelope—independent of the decay phases of both oscillators—to get 808-like results. Changing the volume of the noise generator simulates the snap parameter of the 808.

   Note: The 808 snare drum described is listed as "snare 3-808" in the Tutorial Kit, at a pitch of F#1. It also features an interesting EQ setting.

Use velocity modulation on your 808 snare
Use the 808 snare drum sounds in the Tutorial Kit to explore the possibilities Ultrabeat offers for implementing velocity.

1. Select the "snare 3-808" sound.
2. Chose Vel from the via pop-up menu below the oscillator 1 Volume knob.

   A slider appears on the ring around the knob.

3. Drag the slider clockwise. When you drag the slider, a help tag displays the value. Set it to 0 dB.

4. Repeat steps 2 and 3 in both oscillator 2 and the noise generator.

   You can now dynamically play the sound using velocity.

Increase the performance dynamics of your 808 snare

1. Reduce the values of the individual volumes by turning down the Volume knobs in both oscillators and the noise generator. Note how the mod ring and its via sliders also move back. Change the via slider positions until all three Volume knobs look like this:

If you use differing intensities for each Volume knob when completing this step, you'll have the potential of individual velocity reactions for each sound component.

2. Increase the dynamics of the sound as a whole by assigning the following setting to the Voice Volume knob:

You now have an 808 snare that is exceptionally responsive to velocity. As you may know, this wasn't possible with the original—not even an 808 sample could offer the dynamic volume control of individual sound components that's demonstrated here. A sample offers you only the whole sound, not its constituent parts.

In the next step, you use velocity to control the character of the sound—individually for each component—plus volume, of course.

3. Choose Max from the saturation mod pop-up menu of oscillator 2, and then choose Vel(ocity) from the corresponding via pop-up menu.

4. Set the additional control that appears as shown in the figure below, to control the character of the sound with velocity:

5 Repeat this with the other parameters of oscillator 2, as well as pitch:

6 Modulate the noise generator as follows:

- *Cut parameter:* Choose Max as modulation source, then set the modulation control as shown below.
- *Dirt parameter:* Choose LFO 2 as modulation source, then set the modulation control as shown below.

The sound is now nothing like an 808 snare, which was your goal. Keep experimenting with velocity and figure out when it makes sense to use it as a direct or indirect modulation source, in either its positive or negative form.

Recreate the Kraftwerk snare sound

Another classic electronic snare drum sound is the highly resonant lowpass filter of an analog synthesizer that quickly closes with a snap. This sound was used extensively by Kraftwerk.

1 Select the Snare 1 sound.

2 Direct the signals of both oscillators and the noise generator to the main filter.

3 Modulate Cutoff with Env 1 (which is already modulating the volume of the noise generator).

4 Modulate the filter resonance with Env 2.

5 Experiment with the parameters mentioned in steps 2 to 4 (especially the envelopes), introduce EQ into the sound, and discover how much "playing room" these basic settings allow you.

Note: An exemplary sound is listed as "snare 5-KW" in the Tutorial Kit, at a pitch of G#1. Analyze this sound, and compare it to your own creation.

Create Ultrabeat tonal percussion

Tonal percussion sounds such as toms or congas are relatively easy to emulate electronically with sine or triangular wave oscillators. Ultrabeat's phase oscillator offers you a broad spectrum of suitable basic sounds with which to start. Control the pitch of the oscillators with envelopes, and use the programming techniques discussed in the kick and snare drum sections to alter tonality. You should find it easy to create a broad range of toms and similar sounds. See Create Ultrabeat kick drums and Create Ultrabeat snare drums.

Use oscillator 2 model mode to create tonal percussion sounds

- Turn on oscillator 2 Model mode, then familiarize yourself with the effect of each parameter. You should find it is quite easy to create your own tonal percussion sounds, ranging from toms, to small tabla drums, to glass bowls.
  - Note pitches A1 to B0 in the Tutorial Kit contain typical 808 toms. Analyze these sounds and modify them as you see fit.
  - Note pitches C2 and C#2 in the Tutorial Kit contain tabla and glass sounds that combine both Osc 2 Model and FM. They are also good examples of the complex use of velocity as a modulation source.

Create Ultrabeat hi-hats and cymbals

Electronic hi-hat sounds are easy to create in Ultrabeat.

Create a hi-hat in Ultrabeat

1. Load the Standard Tutorial sound.
2. Turn off oscillator 1 and turn on the noise generator.
3. In the noise generator, make sure the Cutoff parameter is modulated by Env 1, the modulation is negative, and the position of the Mod slider is below that of the base parameter value.

4. Use rather short decay values for Env 1 and Env 4.
5. Set the attack time of Env 4 to a value of 0. The attack time of Env 1 should also be rather short, but not equal to 0.

   Note: You'll find a similarly constructed sound listed as "hihat 1" at a pitch of F2 in the Tutorial Kit. Also analyze the hi-hat sound "hihat 2" at pitch F#2.

Create cymbals in Ultrabeat

It's not far from the hi-hat to the crash cymbal. The main difference between a hi-hat and crash cymbal sound is the length of the decay time. Correct assignment of the envelopes is the key to producing different cymbal sounds.

- Select the Cym 1 and Cym 2 sounds in the Tutorial Kit and try different envelope assignments and settings for Cutoff and Volume in the noise generator, Cutoff and Volume in the main filter, and so on.

Create metallic Ultrabeat sounds

If you want to create metallic sounds with Ultrabeat, the ring modulator and the Model oscillator are the ideal tools.

Use the ring modulator to create metallic sounds

1. Load the Standard Tutorial sound.
2. Activate a phase oscillator and the Model oscillator. Choose a pitch for each oscillator above C3 so that a slightly detuned interval is created.
3. In the Material Pad of the Model oscillator, choose a setting with plenty of overtones, as in the figure below.

4. Set the volume of each oscillator to a value of –60 dB, and click "ring mod" to turn on the ring modulator.

You've just created a bell-like sound that you can filter with a high resonance value if required.

Note: You can find a similar sound listed as Ring Bell at a pitch of A2 in the Tutorial Kit.

Tips for extreme Ultrabeat sounds

Ultrabeat features extremely fast envelopes and uncommonly powerful LFOs. Use these modulation sources to perform extreme modulations of the oscillator and filter parameters.

Do any of the following:

- Try modulating as many targets as possible.
- Use a quick envelope to drive the filter to self oscillation for a fraction of a second.
- Use a few LFO cycles at a much higher rate than other cycles.
- Experiment with the Dirt parameter or the bit crusher.

Vintage B3

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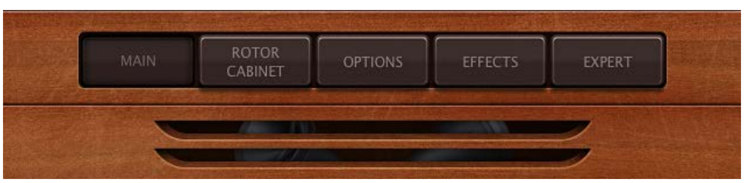
Vintage B3 overview

Vintage B3 emulates the sound and features of the Hammond B3 organ with two manuals (keyboards) and a pedalboard, each of which can have its own registration (sound setting). You can play all registers with a single-manual master keyboard, or you can use two manuals and a MIDI pedalboard. Vintage B3's Component modeling synthesis engine faithfully replicates the tonewheel generators of an electromechanical Hammond organ, down to the smallest detail. This includes charming flaws, such as the Hammond's enormous level of crosstalk and the scratchiness of the key contacts. You can adjust the intensity of these characteristics, enabling tones that range from flawlessly clean through to dirty and raucous—or anything in between.

Vintage B3 also simulates various types of Leslie sound cabinets—with rotating speakers, with and without deflectors. A flexible integrated effects section provides tube overdrive, an equalizer, a wah wah, and a reverberation effect.

The interface is divided into two main areas with a third area visible in some windows. The control bar at the top lets you access different controls in windows shown in the central display section. The lower edge gives you direct access to Leslie speed controls when the Main or Rotor Cabinet window is shown.

Click the control bar buttons to update the central display.



- *Main button:* Shows the draw bars, which are used to make changes to the basic organ sound in real time. Additional performance and setup controls can be accessed with the Control, Preset, and Split buttons in the lower-right corner. See Vintage B3 Main window overview.
- *Rotor Cabinet button:* Shows the Leslie speaker cabinet model and control parameters. See Vintage B3 Rotor Cabinet window overview.
- *Options button:* Shows several tone controls that provide quick access to various aspects of your sound. Advanced controls for Percussion, Scanner Vibrato, and Morph are also found in the Options window. See Vintage B3 Options window overview.
- *Effects button:* Shows controls for the built-in EQ, Wah, Distortion, Chorus, and Reverb effects. See Use Vintage B3 effects.
- *Expert button:* Shows Organ, Pitch, Condition, Sustain, and Miscellaneous controls. These provide precise control over the tone of the organ and over other aspects, such as tuning, draw bar leakage, key click characteristics, and crosstalk levels. You generally access these parameters only when editing or creating an organ sound. See Vintage B3 Expert window overview.

Vintage B3 Main window

Vintage B3 Main window overview

The Vintage B3 Main window is divided into three areas. The draw bars change the basic organ sound in real time. See Vintage B3 draw bar controls. Click the Control, Preset, and Split buttons at the lower right to show different parameters below the draw bars. The lower edge gives you direct access to Leslie speed controls.

Click to show different parameters below the draw bars.

- The Scanner Vibrato, Distortion Drive, and Percussion controls are shown below the draw bars when the Control button at the lower right is turned on. These add a vibrato or chorus-like effect, an overdrive, or percussive element to your organ sound. See Vintage B3 Scanner Vibrato and Chorus, Vintage B3 Distortion effect, and Vintage B3 Percussion effect.
- The Preset (registration) and Morph controls are shown below the draw bars when the Preset button at the lower right is turned on. See Use Vintage B3 preset keys and Vintage B3 Morph parameters.
- MIDI keyboard controls are shown below the draw bars when the Split button at the lower right is turned on. See Vintage B3 MIDI setup overview.

Vintage B3 draw bar controls

Vintage B3 provides 20 draw bars, nine each for the upper and lower manuals, and two for the pedalboard. The upper manual draw bars are on the left, the pedal draw bars are in the center, and the lower manual draw bars are to the right.

The draw bars behave like reversed mixer faders—the farther down you drag the draw bars, the louder the selected sine choirs will be. MIDI control of the draw bars is also reversed when using a standard MIDI fader unit.

Each sine choir is a sine wave that is mixed in at a particular level, determined by the draw bar position. You add sine choirs in this way to build up the overall organ sound for the upper or lower manual. This is a basic form of additive synthesis; for more information, see Additive synthesis with draw bars on page 427. You can intuitively pick up the fundamental principles of additive synthesis by playing a little with the draw bars.

Two draw bars are available for the bass pedals. The waveform used for the bass pedal sound is not a pure sine wave, like the waveforms used for the upper and lower manuals. The pedalboard sound uses a mixed waveform, which accurately emulates the B3's bass tones. The two registers differ in pitch, with the left, 16-foot register containing more octave harmonics. The right, 8-foot register has a more prominent fifth portion (fifth harmonics are enhanced). The term *foot* is derived from pipe organ lengths.

You can simulate the behavior of the Model A, the first Hammond organ ever made. This model had no foldback for the 16' draw bar in the lowest octave, with the bottom 12 tone generator outputs available on the first draw bar of the manuals' bottom octave. Without foldback, the sound is more strident and similar to the pedal sound. Click the disclosure triangle at the lower left, then choose "all the way down" from the Bass pop-up menu to simulate the Model A.

Vintage B3 Scanner Vibrato and Chorus

Vintage B3 emulates the Scanner Vibrato of the original B3. Few organ players use the Scanner Vibrato, preferring to work with a Leslie in isolation. Others, like B3 virtuoso Brian Auger, prefer the integrated organ vibrato over the Leslie. Compare the chorus and vibrato effects with the sound of the rotor cabinet simulation to see which you prefer.

The Scanner Vibrato is based on an analog delay line, consisting of several lowpass filters. The delay line is scanned by a multipole capacitor that has a rotating pickup. It is a unique effect that cannot be simulated with low frequency oscillators (LFOs). The vibrato of the organ itself should not be confused with the Leslie effect, which is based on rotating speaker horns. Vintage B3 simulates both.

The (Scanner Vibrato) Chorus effect is derived from mixing the vibrato signal with the original, statically pitched signal. The organ's chorus sounds different from modern chorus effects.

Important: Scanner Vibrato and Chorus controls are spread across two windows. Click Main in the control bar, then click the Control button at the lower right to view the On and Off switches and to choose the vibrato or chorus type. Click the Options button in the control bar to use the Rate and Depth controls in the Options window.

Scanner Vibrato and Chorus parameters

- *Upper and Lower switches (Main window):* Switch the scanner vibrato on or off, independently, for the upper and lower manuals. The treble portion of the organ is boosted slightly when any vibrato setting is used. Because the B3 mixes the bass register (pedal) signal with the lower manual signal, the pedal register is also affected by the lower manual's scanner vibrato settings.
- *Type knob (Main window):* Choose from three Vibrato positions (V1, V2, and V3) or three Chorus positions (C1, C2, and C3). In the Vibrato positions, only the delay line signal is heard, and like the Hammond B3, Vintage B3 vibrato types have different intensities. The three Chorus positions (C1, C2, and C3) mix the signal of the delay line with the original signal.
- *Rate knob (Options window):* Rotate to set the vibrato or chorus speed.
- *Depth knob (Options window):* Rotate to mix the dry signal with the vibrato signal. This parameter is active only when one of the chorus settings is engaged (C1, C2, or C3).

Vintage B3 Percussion effect

Vintage B3 emulates the (Key) Percussion features of the original B3. The Percussion function is available only for the upper manual. The effect adds the second or third harmonics to the attack envelope of a note. These harmonics quickly fade out, leaving the chosen draw bar tones.

The Percussion effect is polyphonic, but is only (re)triggered after all keys have been released. If you release all keys, new notes or chords sound with percussion. If you play legato, or sustain other notes on the upper manual, no percussion is heard.

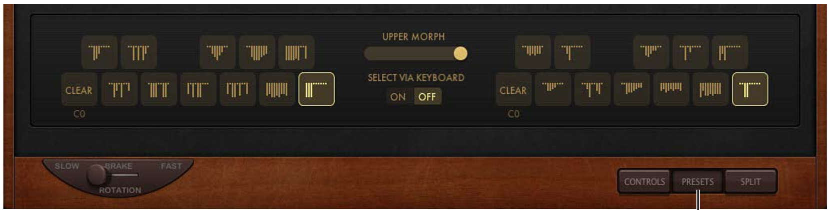
Important: Percussion controls are spread across two windows. Click Main in the control bar, then click the Control button at the lower right to view the Percussion, Harmonic, Time, and Volume switches in the Main window. Click the Options button in the control bar to use the advanced percussion controls in the Options window.

Percussion effect parameters

- *Percussion switch (Main window):* Turn on to activate percussion for the upper manual.
- *Harmonic switch (Main window):* Determines which harmonic is heard (the button toggles between the 2nd and 3rd harmonic).
- *Time switch (Main window):* Switch between a slow or fast decay. The time is set with the Time knobs in the Options window.
- *Volume switch (Main window):* Switch between a low or high decay level. The level is set with the Volume knobs in the Options window.
- *Mode switch (Options window):* Choose Mono to trigger the percussion effect with each key press. Choose Poly for a polyphonic percussion effect which is only (re)triggered after all keys have been released. If you release all keys, new notes or chords will sound with percussion. If you play legato, or sustain other notes on the upper manual, no percussion is heard.
- *Perc on Preset switch (Options window):* Set to B0 to simulate the B preset key restriction. Choose All if you want percussion to always be available.
- *Time knobs (Options window):* Rotate to set an independent percussion decay time for the slow and fast Time switch settings. When the Slow knob is set to maximum, the percussion doesn't decay at all.
- *Volume knobs (Options window):* Rotate to set an independent low and high decay level for the low and high Volume switch settings. This is an improvement from the B3, where Time and Vol could only be turned on or off.
- *Upper Level knob (Options window):* Rotate to set the balance between the upper (percussive) manual and the lower manual/pedals. On the B3, percussion is available only if the B preset key is selected (see Use Vintage B3 preset keys).
- *Velocity knob (Options window):* Rotate to set the percussion velocity sensitivity (unlike the original B3, which is not velocity sensitive). Engaging percussion on a B3 slightly reduces the volume of the normal, nonpercussive registers.

Use Vintage B3 preset keys

The Hammond B3 is equipped with 12 buttons, located below the draw bars. These preset keys are laid out like a keyboard octave, but with black keys and white sharps. They are used to recall draw bar registrations (draw bar positions).



Click to view preset keys.

Upper manual preset keys are to the left of the Upper Morph slider, and lower manual preset keys are to the right. Draw bar positions are indicated by small vertical lines on each key. These miniature draw bar representations update in real time. You can edit the draw bars of recalled presets immediately, with changes to draw bar positions being automatically memorized as you make them. You can, however, use the Save To function to save a registration to a specific preset key (see Save a registration while morphing).

Important: The presets relate only to the registration (draw bar) settings of a single manual. They do not store vibrato or other parameter settings. If you want to save and recall the overall instrument settings (including effects), use the Settings pop-up menu in the plug-in window header.

On keys C# to A#, the percussion works only if the Perc parameter is set to Always (see Vintage B3 Percussion effect).

Vintage B3's default range for preset (registration) keys spans MIDI note numbers 24 to 35 (C0 to B0). This means that the lowest playable MIDI note number is 36 (C1). You can transpose the keyboard range in your host application or Vintage B3 itself. A 61-note keyboard—which spans notes C to C—can be played across the entire range when the Transpose values of your host application are set to 0. The preset (registration) keys are positioned one octave below this transposed or non-transposed range. See Transpose keyboard zones by octaves on page 402.

Choose a registration

1. Click Main in the control bar, then click the Preset button at the lower right.
2. Click a preset key shown to the left (upper manual) or right (lower manual) of the Upper Morph slider.
3. Play *one* of the preset key MIDI notes (MIDI note numbers 24 to 35).

Initialize a registration

1. Click Main in the control bar, then click the Preset button at the lower right.
2. Click the C key for the upper or lower manual. The lowest preset key (shown as "C") is the Clear key. The other 11 keys, from C# to B, recall registrations.
3. Play MIDI note number 24.

Switch Vintage B3 registrations while playing (organ gate effect)

1. Click Main in the control bar, then click the Preset button at the lower right.
2. Hold the Clear key (C) on your master keyboard with the small finger of your left hand, while sustaining a chord with your right hand.
3. Press the preset keys with the other fingers of your left hand.

   The chord being played with your right hand is retriggered (with the new registration) each time you play one of the preset keys. This two-handed technique results in an organ-specific gate-type effect. Each time you switch to a new registration, the chord is retriggered.

Disable MIDI preset key switching

Disabling the switching of presets with MIDI notes 24 to 35 eliminates problems that may arise from transpositions.

1. Click Main in the control bar, then click the Preset button at the lower right.
2. Turn off Select via Keyboard.

Switch Vintage B3 registrations with a two-draw bar controller

When you use a two-draw bar hardware controller, there is an additional mode that allows Hammond-like switching between two registrations. By default, moving draw bars always changes the registration of the currently active preset registration key. This works differently in a real Hammond organ, where the draw bars affect only the A# (upper manual) and B (lower manual) preset registrations. This feature allows you to prepare a new registration with the draw bars while playing, then switch to the new registration when needed.

1. Open the Options window, then choose the (Edit/Preset Key) A# and B only switch position.

   The upper manual draw bars can now change the registration of the A# preset key, and the draw bars of the lower manual affect the B preset key.

2. Change the draw bars of the A# preset key. You can play the keyboard while doing so, without changing the currently chosen registration.
3. Switch to the prepared registration with the A# preset key.

Set up Vintage B3 for your MIDI equipment

Vintage B3 MIDI setup overview

Vintage B3 is unique among the instruments in that it can be played with three simultaneous controllers—namely, a MIDI bass pedal unit and two 73-key MIDI keyboards. This mirrors the two 73-key manuals (organ terminology for keyboards) and the 2-octave pedalboard configuration of the original B3. See Use multiple or multichannel controllers for more information.

Vintage B3 can also be played with a standard 61-key (5 octaves C to C) MIDI keyboard. See Use a single-channel controller for more information.

Vintage B3 also emulates the B3's preset keys—the lowest octave of attached MIDI keyboards can switch between Vintage B3 registrations. This matches the behavior of the original B3, which features a number of inverted (black) keys in the lowest octave of each manual. These inverted keys are used as buttons that recall preset registrations (a preset of your draw bar settings). See Use Vintage B3 preset keys.

For information about setup and use of dedicated MIDI draw bar controllers, see Choose a Vintage B3 MIDI control mode.

Use multiple or multichannel controllers

By default, Vintage B3 receives the notes for the upper and lower manuals, and for the pedalboard, on three consecutive MIDI channels, mapped as follows:

- MIDI channel 1: You play the upper manual sound.
- MIDI channel 2: You play the lower manual sound.
- MIDI channel 3: You play the pedalboard sound.

This allows you to simultaneously play Vintage B3 with up to three MIDI controllers. You can also use a single-manual master keyboard—with different keyboard zones, or a keyboard split feature—that sends data on different MIDI channels to address all three Vintage B3 sounds simultaneously. Each keyboard zone can be transposed independently. See also Choose a Vintage B3 MIDI control mode. You can use any of your MIDI interface inputs for your master keyboard or pedalboard. Regardless of the input devices used, the only relevant factor is the MIDI send channel.

Note: See the user manual for your master keyboard to learn how to set up splits and zones or how to set its MIDI transmission channel (often called *TX Channel*).

Set the keyboard mode

There are three keyboard modes: single, split, and multi.

1. Click Main in the control bar, then click the Split button at the lower right.
2. Set the switch to the left of the keyboard in the central display to Single, Split, or Multi.
   - *Single:* Uses the entire keyboard. You can only play the upper sound.
   - *Split:* Divides the keyboard into two. You can play the upper and lower sounds in different keyboard zones.
   - *Multi:* Divides the keyboard into three. You can play the upper, lower, and pedalboard sounds in different keyboard zones.

Change the default MIDI channels

Changing MIDI channels can be useful when you perform live and require quick access to another sound module.

1. Click Main in the control bar, then click the Split button at the lower right.
2. Set the switch to the left of the keyboard in the central display to Multi.
3. Change the channel numbers for the upper, lower, and pedal manuals.

Set keyboard zones

1. Click Main in the control bar, then click the Split button at the lower right.
2. Set the switch to the left of the keyboard in the central display to Split.
3. Horizontally drag the split icons to create the pedal/lower zone and the lower/upper zone.

   If you select the same value for both split points, the lower manual is turned off. If the lower/pedal split is moved above the upper/lower split, the other split point is moved (and vice versa).

Transpose keyboard zones by octaves

You can make transpositions that are independent of the global Tune parameter or transposition features of the host application. These have no impact on the preset keys which is particularly important when you want to use preset switching (see Use Vintage B3 preset keys) when split keyboard mode is active.

1. Click Main in the control bar, then click the Split button at the lower right.
2. Set the switch to the left of the keyboard in the central display to Split.
3. Choose an octave value (+/– 2 octaves) from the Pedal Transpose, Lower Transpose, or Upper Transpose pop-up menu.

Use a single-channel controller

If you don't have a master keyboard that allows multichannel transmission, you can use a MIDI keyboard that is capable of transmitting on only one MIDI channel. You can use Vintage B3 Split parameters to split the keyboard in order to play upper, lower, and pedal sounds in different keyboard zones. Each keyboard zone can be transposed independently.

There are three keyboard modes: single, split, and multi.

- *Single:* Uses the entire keyboard. You can only play the upper sound.
- *Split:* Divides the keyboard into two. You can play the upper and lower sounds in different keyboard zones.
- *Multi:* Divides the keyboard into three. You can play the upper, lower, and pedalboard sounds in different keyboard zones.

Technically, Vintage B3 remaps the incoming single-channel MIDI data into two or three MIDI channels when either split or multi keyboard mode is active.

Set the keyboard mode

1. Click Main in the control bar, then click the Split button at the lower right.
2. Set the switch to the left of the keyboard in the central display to Single, Split, or Multi.

Change the default MIDI channels

Changing MIDI channels can be useful when you perform live and require quick access to another sound module.

1. Click Main in the control bar, then click the Split button at the lower right.
2. Set the switch to the left of the keyboard in the central display to Multi.
3. Change the channel numbers for the upper, lower, and pedal manuals.

Set keyboard zones

1. Click Main in the control bar, then click the Split button at the lower right.
2. Set the switch to the left of the keyboard in the central display to Split.
3. Horizontally drag the split icons to create the pedal/lower zone and the lower/upper zone.

   If you select the same value for both split points, the lower manual is turned off. If the lower/pedal split is moved above the upper/lower split, the other split point is moved (and vice versa).

Transpose keyboard zones by octaves

1. Click Main in the control bar, then click the Split button at the lower right.
2. Set the switch to the left of the keyboard in the central display to Split.
3. Choose an octave value (+/– 2 octaves) from the Pedals Transpose, Lower Transpose, or Upper Transpose pop-up menu.

   These transpositions are independent of the global Tune parameter or transposition features of the host application. They also have no impact on the preset keys which is particularly important when you want to use preset switching (see Use Vintage B3 preset keys) when split keyboard mode is active.

Vintage B3 Rotor Cabinet window

Vintage B3 Rotor Cabinet window overview

The Hammond story can't be fully told without discussing the rotor cabinets manufactured by Leslie. In fact, playing the B3 organ without a rotor cabinet is viewed as something of a special effect these days. Vintage B3 not only simulates the speaker cabinet itself, but also allows you to change the listening position by placing virtual microphones in different locations.

Some of the speaker cabinet models are mathematically simulated, and others use a recording of the spatial characteristics of the speaker. The latter is known as an impulse response. Detailed information on impulse responses can be found in the Space Designer section of Logic Effects Help. If you're unfamiliar with the concepts of the Leslie rotating speaker cabinets, see The Leslie cabinet on page 430.

The Leslie rotation speed control is shown at the lower-left corner of the Vintage B3 Main and Rotor Cabinet windows. Advanced speed controls are shown in the central display when you click Rotor Cabinet in the control bar.

The advanced Leslie rotating speaker cabinet controls are useful for specialized sounds, or when you are creating realistic emulations. See Advanced Cabinet parameters, Advanced Motor parameters, and Advanced Brake parameters.

For information about microphone parameters, see Vintage B3 Microphone parameters on page 408.

Basic rotor speaker controls

Basic rotor speaker parameter

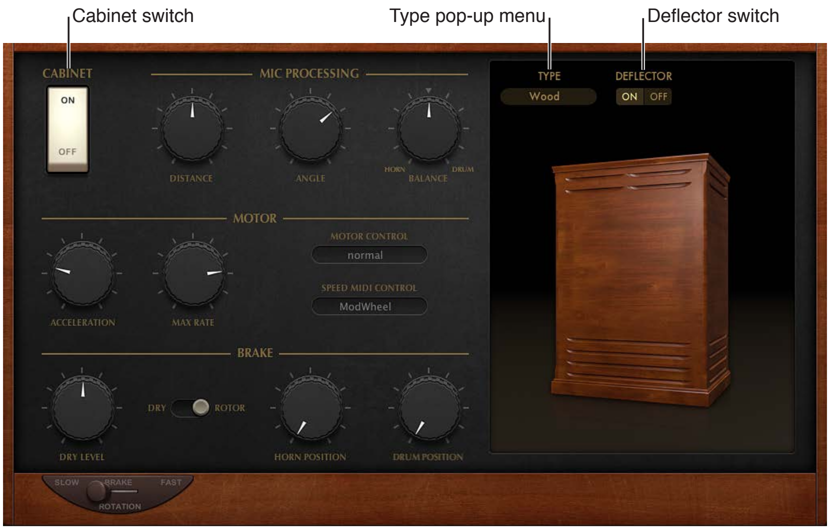
This parameter is shown at the lower left in the Main and Rotor Cabinet windows.

- *Rotation switch:* Switch the rotor speed in the following ways:
  - *Slow:* Slow rotor movement.
  - *Brake:* Stops the rotor.
  - *Fast:* Fast rotor movement.

Advanced Cabinet parameters

The advanced cabinet parameters are divided into three groups: Cabinet, Motor, and Brake. Also see Advanced Motor parameters and Advanced Brake parameters.

The microphone parameters are described in Vintage B3 Microphone parameters on page 408.



Cabinet parameters

- *Cabinet switch:* Turns the Leslie cabinet emulation on or off.
- *Type pop-up menu:* Choose a cabinet model:
  - *Real Cabinet:* Uses an impulse response recording of a Leslie cabinet. Click the microphones to change the type of microphone.
  - *Wood:* Mimics a Leslie with a wooden enclosure, and sounds like the Leslie 122 or 147 model.
  - *Proline:* Mimics a Leslie with a more open enclosure, similar to a Leslie 760 model.
  - *Single:* Simulates the sound of a Leslie with a single, full-range rotor. The sound resembles the Leslie 825 model.
  - *Split:* The bass rotor's signal is routed slightly to the left, and the treble rotor's signal is routed more toward the right.
  - *Wood & Horn IR:* Uses an impulse response of a Leslie with a wooden enclosure.
  - *Proline & Horn IR:* Uses an impulse response of a Leslie with a more open enclosure.
  - *Split & Horn IR:* Uses an impulse response of a Leslie with the bass rotor signal routed slightly to the left, and the treble rotor signal routed more to the right.
- *Deflector switch:* Emulates a Leslie cabinet with the horn deflectors removed or attached. A Leslie cabinet contains a double horn, with a deflector at the horn mouth. This deflector makes the Leslie sound. You can remove the deflector to increase amplitude modulation and decrease frequency modulation.

Advanced Motor parameters

The advanced cabinet parameters are divided into three groups: Cabinet, Motor, and Brake. See also Advanced Cabinet parameters and Advanced Brake parameters.

The microphone parameters are described in Vintage B3 Microphone parameters on page 408.

Motor parameters

- *Acceleration knob:* Rotate to set the time it takes to get the rotors up to the speed set with the Max Rate knob, and the length of time it takes for them to slow down. The Leslie motors need to physically accelerate and decelerate the speaker horns in the cabinets, and their power to do so is limited. Turn Acceleration to the far left position to switch to the preset speed immediately. As you rotate the knob to the right, it takes more time to hear the speed changes. At the default, centered, position the behavior is Leslie-like.
- *Max Rate knob:* Rotate to set the maximum possible rotor speed.
- *Motor Control pop-up menu:* Set different speeds for the bass and treble rotors in the pop-up menu. Use the Rotation switch to choose slow, brake, or fast mode. See Vintage B3 Rotor Cabinet window overview.
  - *Normal:* Both rotors use the speed determined by the Rotation switch position.
  - *Inv (inverse):* In fast mode, the bass compartment rotates at a fast speed, while the horn compartment rotates slowly. This is reversed in slow mode. In brake mode, both rotors stop.
  - *910:* The 910 (also known as "Memphis"), stops the bass drum rotation at slow speed, while the speed of the horn compartment can be switched. This is useful when you're after a solid bass sound but still want treble movement.
  - *Sync:* The acceleration and deceleration of the horn and bass drums are roughly the same. This sounds as if the two drums are locked, but the effect is clearly audible only during acceleration or deceleration.

  Note: If you choose Single Cabinet from the (Cabinet) Type pop-up menu, the Motor Control setting is not relevant because there are no separate bass and treble rotors in a single cabinet.
- *Speed MIDI Control pop-up menu:* Choose a MIDI controller that is used to remotely switch the rotor speed. All items (except ModWheel) in the pop-up menu switch between fast and the speed set with the Rotation switch positions—either switching between slow and fast, or switching between brake and fast. If fast is chosen, the rotor speaker switches between fast and slow.
  - *Modwheel:* Assigns the modulation wheel to switch between all three speed settings. Brake is selected around the modulation wheel's center position, slow is selected in the lower third, and fast in the upper third of the modulation wheel's travel.
  - *Modwhl Toggle:* Switches as soon as the modulation wheel moves away from the centered position. If the modulation wheel passes the center position when moved from a high to low position, no switching occurs. This caters to Roland keyboards with combined pitch bend and modulation controls.

- *Modwhl Temp:* Switches as soon as the modulation wheel passes the center position, regardless of whether you have moved the modulation wheel from high to low or from low to high positions. This caters to Roland keyboards with combined pitch bend and modulation controls.
- *Touch:* Switches with aftertouch on messages. No switching occurs on aftertouch release.
- *Touch Temp:* Switches with aftertouch on messages. A second switch occurs with aftertouch release messages.
- *SusPdl Toggle:* Switches when you press the sustain pedal. No switching occurs when the sustain pedal is released.
- *SusPdl Temp:* Switches when you press the sustain pedal. A second switch occurs when you release the sustain pedal.
- *CC #18 and CC #19 Toggle:* Switches when you press controller 18 or 19. No switching occurs when either controller is released.
- *CC #18 and CC #19 Temp:* Switches when you press controller 18 or 19. A second switch occurs when you release controller 18 or 19.

Advanced Brake parameters

The advanced cabinet parameters are divided into three groups: Cabinet, Motor, and Brake. See also Advanced Cabinet parameters and Advanced Motor parameters.

The microphone parameters are described in Vintage B3 Microphone parameters on page 408.

Brake parameters

- *Dry Level knob:* Rotate to set the level of the dry signal, which can also be useful if Dry is active for the Output switch.
- *Output switch:* Choose one of two positions that modify Vintage B3 brake mode.
  - *Dry:* The rotor cabinet is bypassed when stopped, with a delay time of one second. This is useful when you are using the modulation wheel to switch between fast and slow rotor speeds. If you then switch to brake mode, the rotors are slowed down during the transition to the dry sound.
  - *Rotor:* The movement of the rotor is gradually slowed down to a total stop.
- *Horn/Drum Position knobs:* Rotate to set an exact stop position for the Leslie horn or drum (bass) rotator, respectively. The original Leslie did not provide this type of control. This occasionally resulted in a horn aimed at the back of the cabinet when it came to a halt, leading to a muffled sound.

Vintage B3 Microphone types

Vintage B3 provides modeled microphones that pick up the sound of the Leslie cabinet. You can specify the microphone type with these parameters.

Click the microphone icons to choose a microphone type for the horn and drum speakers when Real Cabinet is chosen in the Type pop-up menu. See Advanced Cabinet parameters.

- *Dynamic:* Emulates the sound of a dynamic cardioid microphone. This microphone type sounds brighter and more cutting than the Condenser mic.
- *Condenser:* Emulates the sound of a studio condenser microphone. The sound of condenser microphones is fine, transparent, and well-balanced.
- *Mid-Side Mic:* A Middle and Side (MS) configuration where two microphones are positioned closely together. One is a cardioid (or omnidirectional) microphone that directly faces the cabinet—in a straight alignment. The other is a bidirectional microphone, with its axes pointing to the left and right of the cabinet at 90° angles. The cardioid microphone captures the middle signal to one stereo side. The bidirectional microphone captures the side signal to the other stereo side.

Vintage B3 Microphone parameters

Vintage B3 provides modeled microphones that pick up the sound of the Leslie cabinet. You can set the listening position with these parameters.

Microphone parameters

- *Mic Position switch:* Choose either the front or rear position for the virtual microphone.
- *When Real Cabinet is chosen in the Type pop-up menu:*
  - *Horn knob:* Rotate to define the stereo width of the Horn deflector microphone.
  - *Drum knob:* Rotate to define the stereo width of the Drum deflector microphone.
- *When other cabinets are chosen in the Type pop-up menu:*
  - *Distance knob:* Rotate to determine the distance of the virtual microphones (the listening position) from the emulated speaker cabinet. Turn to the right for a darker and less defined sound.
  - *Angle knob:* Rotate to define the stereo image, by changing the angle of the simulated microphones between 0 and 180 degrees.
- *Balance knob:* Rotate to set the balance between the horn and drum microphone signals.

Vintage B3 Options window

Vintage B3 Options window overview

You can adjust the Vintage B3 output level, tuning, key click volume, and other basic sound aspects with the Options controls. See Vintage B3 Master and Click controls.

Advanced controls for Percussion, Scanner Vibrato, and Morph are also found in the Options window. See Vintage B3 Percussion effect, Vintage B3 Scanner Vibrato and Chorus, and Vintage B3 Morph parameters.

The Edit Preset Key parameter is discussed in Switch Vintage B3 registrations with a two-draw bar controller.

Vintage B3 Master and Click controls

Master parameters

- *Tune knob:* Rotate to change Vintage B3 tuning in cents. A cent is 1/100th of a semitone. 0 c is equal to A=440 Hz.
- *Volume knob:* Rotate to set the overall output level. The Volume knob must be lowered whenever crackling or other digital distortion occurs. Volume levels over 0 dB can occur if you maximize the levels of all registers, play numerous notes, and make use of the Distortion effect.
- *Expression knob:* Rotate to set the sensitivity for a connected expression pedal (on a MIDI keyboard with an Expression or assignable controller input). Extensive, often rhythmic, use of the expression (volume) pedal forms part of the style of many organ players. The expression control also emulates the tonal changes of the B3 pre-amplifier, where bass and treble frequencies are not attenuated as much as the mid frequencies. Your master keyboard should transmit MIDI control change #11 when the pedal is moved. Vintage B3 defaults to the use of CC #11 for Expression.

Click parameters

Advanced click controls are available in the Expert page. See Vintage B3 Condition controls.

- *Key On/Key Off knobs:* Rotate to set the level of the key click sound heard during note on or note off messages.
- *Pedal knob:* Rotate to set the level of the key click sound heard during note on and note off messages for the pedal register.
- *Velocity knob:* Rotate to set the velocity sensitivity of the click parameters.

Vintage B3 Morph parameters

You can switch—or smoothly crossfade (morph)—between the presets of the upper manual. See Use Vintage B3 Morph controls.

Important: Morph controls are spread across two windows. Click Main in the control bar, then click the Preset button at the lower right to view the Morph slider. The Options window contains advanced Morph controls.

Morph parameters

- *Upper Morph slider (Main window):* Drag left or right to control the switching or morphing. You can also assign and use a MIDI controller such as your keyboard modulation wheel to control the Morph slider.
- *Select via Keyboard buttons (Main window):* Click the On button, then play a lower and upper key to set the range of Preset keys affected by the morph. Click Off to disable the morph range set up.
- *Mode switch (Options window):* Click Step to abruptly switch, or click Linear to smoothly morph (crossfade) between presets.
- *Range pop-up menu (Options window):* Choose a range of keys for morph presets.
- *MIDI Controller pop-up menu (Options window):* Assigns a MIDI controller to the Morph slider. You can choose any MIDI controller number shown (or channel aftertouch). You can also click Learn to teach the Morph slider to respond to any incoming message.

Use Vintage B3 Morph controls

You can switch—or smoothly crossfade (morph)—between the presets of the upper manual. See Vintage B3 Morph parameters.

Important: Morph controls are spread across two windows. Click Main in the control bar, then click the Preset button at the lower right to view the Morph slider. The Options window contains advanced Morph controls.

Learn a MIDI controller for morphing

1. Click Options in the control bar.
2. Choose Learn from the (Morph) MIDI Controller pop-up menu.

   When Learn is active, the parameter is assigned to the first appropriate incoming MIDI data message.
3. Move the MIDI controller on your MIDI device.

   Learn mode has a 20-second time-out function: if Vintage B3 does not receive a MIDI message within 20 seconds, the parameter reverts to its original MIDI controller assignment.

Set the Vintage B3 morph range

Once you have chosen a controller to use for switching or morphing between upper manual registrations, you can determine the number of preset keys that are affected. You can morph between two and eleven presets for the upper manual.

Morphing always begins with the top preset key, the B.

- If Range = A#, you morph between two presets.
- If Range = G#, you morph between four presets (B, A#, A, and G).
- If Range = F#, you morph between six presets (and so on).

1. Click Main in the control bar.
2. Click the Preset button at the lower right.
3. Click the Select via Keyboard On button.
4. Play two keys on your MIDI keyboard to set the morph range.

Save a registration while morphing

In Linear mode (morphing), the seamless crossfades result in a variety of new draw bar registrations that you might want to save. You can change draw bar positions manually, before saving.

- Click View > Controls, then choose a preset key from the "Save Morph to" pop-up menu (near the bottom of the Preset parameters).

Vintage B3 Effects window

Use Vintage B3 effects

Vintage B3 features a three-band equalizer, a reverberation effect, a pedal-controllable wah wah effect, and a distortion effect that simulates the sound of an overdriven tube amplifier. In addition, the signal can be routed through the Leslie rotor speaker emulation.

The default effect signal flow is as follows: the organ's signal runs through the Equalizer, Wah, and Distortion effects. This treated signal is then fed into the Reverb, and finally passed to the Leslie rotor effect.

Activate or bypass the effects

Do one of the following:

- Turn the Master switch on or off to enable or disable the entire Vintage B3 effects section.
- Use the On/Off switches to independently enable or disable the Reverb, EQ, Wah, and Distortion effects.

Change the routing of the EQ, Wah, and Distortion effects.

- Drag the name of each effect to create the signal flow you require.
  - *EQ-Wah-Dist:* This routing is perfect for a classic B3 patch—an equalized organ, plugged into a wah pedal, amplified by an overdriven Leslie.
  - *EQ-Dist-Wah:* The sound of the overdrive changes if the input signal is being filtered, either by the EQ or the Wah. Placing the EQ before the Distortion provides further sonic flexibility. Although the output signal of the Distortion effect always contains high frequency content, this content can be suppressed by positioning the Wah as the final effect in the chain.
  - *Wah-Dist-EQ:* If you want to create a screaming sound (achieved by distorting the Wah effect output), you can minimize any harshness by choosing this routing.
  - *Dist-EQ-Wah:* Choose this routing to suppress the harsh overtones of extreme distortions with two filters.

Bypass effects for the pedal register

- Set the Pedal switch to Byp. If you choose FX, the entire output of the organ is processed.

Bypassing the Distortion, Wah, and EQ effects separately for the pedal register avoids suppression of the bass portion of your organ sound by the Wah effect. It also avoids intermodulation artifacts when the Distortion effect is used.

Vintage B3 EQ

Vintage B3 features a simple but effective EQ section.

Vintage B3 EQ parameters

- *EQ On/Off switch:* Turn on or bypass the equalizer.
- *Output Level knob:* Rotate to adjust the overall EQ level.
- *Low knob:* Rotate to adjust the level of the low frequency range.
- *Mid knob:* Rotate to adjust the level of the mid frequency range.
- *High knob:* Rotate to adjust the level of the high frequency range.

Vintage B3 Wah effect

The name wah wah comes from the sound it produces. It has been a popular effect with electric guitarists since the days of Jimi Hendrix. The pedal controls the cutoff frequency of a bandpass, lowpass, or—less commonly—highpass filter. The wah pedal is also used extensively with the Hammond organ.

For the most dynamic and musical performance of the Wah effect, consider attaching an expression pedal to your MIDI master keyboard. Your master keyboard should transmit MIDI control change #11, which would normally be used to control Vintage B3 volume while playing.

Vintage B3 Wah parameters

- *Wah On/Off switch:* Turn on or bypass the Wah effect.
- *Sweep MIDI Ctrl pop-up menu:* Use to assign a MIDI controller to the Wah effect.
- *Type pop-up menu:* Choose one of several Wah effect types.
  - *Classic Wah:* This setting mimics the sound of a popular wah pedal with a slight peak characteristic.
  - *Retro Wah:* This setting mimics the sound of a popular vintage wah pedal.
  - *Modern Wah:* This setting mimics the sound of a distortion wah pedal with a constant Q(uality) Factor setting. The Q determines the resonant characteristics. Low Q values affect a wider frequency range, resulting in softer resonances. High Q values affect a narrower frequency range, resulting in more pronounced emphasis.
  - *Opto Wah 1:* This setting mimics the sound of a distortion wah pedal with a constant Q(uality) Factor setting.
  - *Opto Wah 2:* This setting mimics the sound of a distortion wah pedal with a constant Q(uality) Factor setting.
  - *Resonant LP:* In this mode, the Wah works as a resonance-capable lowpass filter. At the minimum pedal position, only low frequencies can pass.
  - *Resonant HP:* In this mode, the Wah works as a resonance-capable highpass filter. At the maximum pedal position, only high frequencies can pass.
  - *Peak:* In this mode, the Wah works as a peak (bell) filter. Frequencies close to the cutoff frequency are emphasized.

- *Range knob:* Rotate to determine the sensitivity of the Wah effect to incoming MIDI controller data.
- *Bite knob:* Rotate to boost the levels of signals surrounding the cutoff frequency. Bite is effectively a filter resonance parameter, where high values makes the Wah effect sound more aggressive.

Use an expression pedal to control the Wah effect

1. Click Options in the control bar, then set the Expression knob to a value of 0.
2. Click Effects in the control bar, then choose controller 11 from the Sweep MIDI Control pop-up menu.

   This enables control of the Wah's cutoff frequency with the expression pedal and requires no further setup of your master keyboard. If step 1 is overlooked, the expression pedal is used to control both Vintage B3 main volume and the Wah effect.

   Note: Consult the user manual for your keyboard to learn more about use of an expression pedal.
3. Adjust the Range knob to set the sensitivity of the Wah to incoming expression pedal controller data.

Learn a MIDI controller to control the Wah effect

1. Click Effects in the control bar.
2. Choose Learn from the Sweep MIDI Ctrl pop-up menu.

   When Learn is active, the parameter is assigned to the first appropriate incoming MIDI data message.
3. Move the MIDI controller on your MIDI device.

   Learn mode has a 20-second time-out facility: if Vintage B3 does not receive a MIDI message within 20 seconds, the parameter reverts to its original MIDI controller assignment.

Vintage B3 Distortion effect

The Distortion effect simulates an overdriven two-stage tube amplifier. Its primary role is to simulate the Leslie amplifier or another amplifier used to feed the Leslie speaker cabinet.

Vintage B3 Distortion parameters

- *Distortion On/Off switch:* Turn on or bypass the Distortion effect.
- *Type pop-up menu:* Choose a tube amplifier model.
  - *Growl:* Simulates a two-stage tube amplifier. It closely resembles the Leslie 122 model, the classic partner for the Hammond B3 organ.
  - *Bity:* Reminiscent of a bluesy guitar amp.
  - *Nasty:* Delivers hard distortions and is suitable for very aggressive sounds.
  - *6550 AB V3:* Tube distortion sound modeled on class AB amp head.
  - *6550 AB V2:* Tube distortion sound modeled on class AB amp head.
  - *6550 GL:* Tube distortion sound modeled on tube amp head.
- *Tone knob:* Rotate to change the distorted portion of the sound. This has no effect on the dry signal portion. Limiting changes to the distorted signal allows for very warm overdriven sounds that do not become scratchy if you try to get more treble out of Vintage B3.
- *Drive knob:* Rotate to set the amount of overdrive distortion. The output level is automatically compensated for, so there's no need for a distortion output volume control. A level of 0 effectively turns off the Distortion circuit.

  Note: The Drive knob shown in the Main window is linked with the Drive knob shown in the Effects window. When you rotate either Drive knob, your change is mirrored in the other window. In essence, this is a convenience feature that addresses the same parameter in both locations.

Vintage B3 Reverb effect

Vintage B3 features a simple reverberation effect that emulates several room types and a spring reverb.

Vintage B3 Reverb parameters

- *Reverb On/Off switch:* Turn on or bypass the reverb effect.
- *Pre/Post switch:* Patches the reverb effect before (Pre) or after (Post) the rotor effect. The reverb is always patched after the EQ, Wah, and Distortion effects, but before the Leslie rotor cabinet. Switch to Post if you don't want the reverb to sound like it is played back through the rotor speaker.
- *Type pop-up menu:* Choose from six reverb algorithms: Air, Box, Small, Medium, Large, Big, and Spring.
- *Level knob:* Rotate to set the balance between the reverb and original signal levels.

Vintage B3 Expert window

Vintage B3 Expert window overview

The Expert window shows model parameters that provide precise control over your organ sound. These include not only basic level and tonal balance controls, but a number of parameters that emulate the sonic characteristics—and technical "flaws" or limitations—of the original Hammond B3. You can even emulate the quirks of the B3 by aging the virtual components. See Vintage B3 Pitch controls, Vintage B3 Sustain controls, Vintage B3 Condition controls, and Vintage B3 Organ Model controls.

See Choose a Vintage B3 MIDI control mode for information on the Hardware Controller parameter shown in the Misc section of the Expert window.

For more information about the technical aspects of the Hammond B3 and the concepts behind tonewheel sound generation, see A brief Hammond history on page 429 and Tonewheel sound generation on page 428.

Vintage B3 Pitch controls

Vintage B3 provides several parameters that change its pitch behavior, adding flexibility that isn't possible with the original instrument.

Vintage B3 is tuned to an equal-tempered scale. As a deviation from this standard tuning, you can stretch the tuning in the bass and treble ranges, much like acoustic pianos (especially upright pianos). The tones of clavinets, harpsichords, and pianos have inharmonicities in their harmonic structure. The frequencies of these overtones (harmonics) are not exact, whole-number multiples of the base frequency. This means that the overtones of lower (tuned) notes are more closely related to the main frequencies of the upper notes. Due to the lack of strings, this inharmonic relationship is not true of organs. The stretch feature was principally included for situations in which you want to use Vintage B3 in an arrangement alongside a recording of an acoustic piano.

You can also randomly detune the sound using the Warmth parameter, and you can even use the pitch bend wheel of your keyboard to bend the sound. The latter isn't true to the original, but it's a nice creative option.

Pitch parameters

- *Upper Stretch slider:* Drag to set the amount of deviation from the equal-tempered scale in the treble end of the sound. The higher the value, the further up the high notes are tuned. At a setting of 0, Vintage B3 is tuned to an equal-tempered scale, with each octave up exactly doubling the frequency.
- *Lower Stretch slider:* Drag to set the amount of deviation from the equal-tempered scale in the bass frequencies. The higher the value, the further down the low notes are tuned. At a setting of 0, Vintage B3 is tuned to an equal-tempered scale, with each octave below exactly halving the frequency.
- *Warmth slider:* Drag to set the amount of random deviation from an equal-tempered scale.

  Note: Use of both Warmth and Stretch may result in a detuned sound, which is similar to a heavy chorus effect. Set Warmth to 0 if you're after a purer sound.
- *Pitchbend Up/Down sliders:* Drag to independently set the pitch bend up/down sensitivity in semitone steps. The maximum sensitivity for upward bends is one octave. The Hammond B3 has no pitch bend facilities. Therefore, use of pitch bend is not suitable for realistic organ simulations, but it does provide a number of creative options.

  Note: If you drag the Pitchbend Down slider to the far right, the tonewheels gradually slow down until they totally stop—when your keyboard's pitch bend control is at the minimum position. This setting re-creates an effect heard at the end of "Knife Edge" by Emerson, Lake, and Palmer. Keith Emerson's virtuoso Hammond work was recorded on a reel-to-reel tape recorder that was gently slowed to a total stop.

Vintage B3 Sustain controls

The time it takes for a note to fade out to silence, after the key has been released, is called the release time in synthesizers. Vintage B3 provides control of this parameter, known as *sustain* in organ terminology.

Sustain parameters

- *Upper Manual slider:* Drag to control the sustain (release) phase of the upper register.
- *Lower Manual slider:* Drag to control the sustain (release) phase of the lower register.
- *Pedals slider:* Drag to control the sustain (release) phase of the pedal register.
- *Mode buttons (Controls view only):* Click to choose one of two sustain behaviors:
  - *Smart:* Cuts the sustain phase of released notes when you play new notes.
  - *Normal:* Allows polyphonic sustain phases. All released notes will continue to sustain, even if new notes are played.

  Note: Smart mode allows for long sustain times, even in the bass register, which would cause rumbling dissonances if you used normal mode.

Vintage B3 Condition controls

Technical limitations of electromechanical draw bar organs, with tonewheels, can cause some strange tonal artifacts, such as crosstalk. These quirks form an integral part of the B3's charm. You can adjust a number of parameters to define the age and condition of your Vintage B3.

The key contacts of electromechanical tonewheel organs tend to saw a little on the busbar, thus introducing a short click sound. Corrosion of the key contacts or busbar increases the length and level of this click. This aspect of the B3's design causes irregular scratching noises (commonly referred to as *key click*) when striking and releasing keys. Hammond fans like these clicking noises because they introduce a transient, percussive quality to the note.

Vintage B3 allows you to adjust the volume and sound of the key click. The tonal color and volume of clicks are altered randomly, and independently, from the click on and click off (release) volume settings.

Condition parameters

- *Click Minimum/Maximum sliders:* Combined, these sliders determine a range for click duration, which can vary between a short "tick" and a longer "scratch." A random click duration (that falls within the defined range) is used as you play.

  Note: Even if both parameters have identical values, there is a random variation in sound that makes some clicks seem shorter than the value set with Click Min.
- *Click Color slider:* Drag to set the tonal color of the click. This acts as a global control for the treble portion of the click sound, which overrides (but works alongside) the random click color variations.
- *Filter Age slider:* Drag to set the center frequencies of the filters, which emulates aging capacitors. The high frequency output signals of the B3's tonewheel generators are passed through bandpass filters. The center frequency of these filters changes as the capacitors (used for filtering) get older.

  Note: This colors the sound of the jitter applied by Random FM and the background noise resulting from leakage. Filter Age also influences the intonation of the organ, if you use a pitch bend.

- *Leakage slider:* Drag to add a sound resulting from the crosstalk between all tonewheels—including the tonewheels of notes that you don't play. Adjust this slider to add a "breathy" quality to your organ sound.
- *Drawbar Leak slider:* Drag to set the minimum output level of the draw bars when they are at their minimum positions. The B3 tonewheel generators aren't completely quiet, even if all draw bars are at their minimum positions. This is due to leakage of the tonewheels, causing crosstalk at the output.
  - Use the minimum setting to completely eliminate draw bar leakage.
  - Use the maximum setting to make draw bar leakage clearly audible.
- *Crosstalk slider:* Drag to set the crosstalk level. There are two tonewheels that are four octaves apart for each key (pitch), on each rotating shaft. The signal of the lower wheel has a small amount of audible crosstalk, induced by the higher wheel, and vice versa. For more information, see Tonewheel sound generation on page 428. Because crosstalk is audible only on certain B3 tonewheels, any "rumble" when chords are played is avoided.
- *Random FM slider:* Drag to simulate irregular rotation of tonewheels in an old B3. If the tonewheel generator of a B3 is clean, all frequencies are even and in tune. The three-fold decoupling of the tonewheels—via springs, flexible couplings, and flywheels—is effective, but it can't compensate for irregularities that come with dirt and grease in the driving gears. A gradual build-up of grime in the mechanism makes the tonewheel assembly turn unevenly on its axis. This irregular rotation is transmitted to the tonewheels, and therefore, the higher frequency ranges of the sound.

Vintage B3 Organ Model controls

The Organ Model controls change the basic tonal quality.

Organ Model parameters

- *Maximum Wheels slider:* Drag to set the number of tonewheels that are emulated. Reduce the value to minimize the computer processing load. Reducing the value diminishes some overtones, so keep the number high if you're after an ultrarealistic simulation.
- *Tonal Balance slider:* Drag to change the mix relationship of the higher and lower tonewheels. Use positive values to achieve a lighter and brighter sound. Experiment with different tonal balance and equalizer settings. See Vintage B3 EQ on page 413, for further information.
- *Lower Manual Volume slider:* Drag to set relative levels between the upper and lower manuals (and pedalboard).
- *Pedals Volume slider:* Drag to set relative levels between the upper (and lower) manual and the pedalboard.
- *Shape slider:* Drag to alter the waveforms of the tonewheel generator, allowing you to produce sounds that resemble the tones of Farfisa, Solina, or Yamaha organs. The Hammond's tone generators produce pure sine waves (albeit with a few artifacts), whereas some other organs deliver distorted waveforms. The Shape parameter is placed after the filters that follow the sine generators.
  - Move the Shape slider to the right to make the tone brighter (and louder).
  - Move the Shape slider to the left to make the tone duller (and softer).
- *Bass Filter slider:* The tone of the pedal draw bars can sound bright, within the context of the combined upper/lower/pedal sound. Adjust to circumvent this issue, and to suppress the treble of the bass register. At the maximum position you will only hear a solid bass organ fundamental in the bass register.
- *Ultrabass switch:* Turn on to add another low octave to the playable range of both the upper and lower manuals.These additional low octaves, and the ability to independently transpose both manuals (see Transpose keyboard zones by octaves on page 401), are not available on the original B3.

Use a MIDI controller with Vintage B3

Choose a Vintage B3 MIDI control mode

MIDI controller assignments allow you to control Vintage B3 with an external MIDI controller or a host application such as Logic Pro.

The Hardware Controller parameter determines the way Vintage B3 draw bars respond to remote MIDI control change messages. Most users won't need to change anything here.

If you own a MIDI draw bar organ, you'll want to use its hardware draw bars to control Vintage B3. Most hardware draw bar organs use an independent MIDI control change number for each draw bar.

Choose a MIDI hardware controller

1. Click Expert in the control bar.
2. Choose a device (mode) from the Hardware Controller pop-up menu. Choose Off if you do not own a supported device and don't want to use a special assignment mode.
   - Choose Vintage B3 MIDI mode: Roland VK or Korg CX if you use a Roland VK series or Korg CX-3 draw bar organ as a remote controller for Vintage B3.
   - Choose Vintage B3 MIDI mode: Hammond Suzuki if you use a Hammond XB series organ as a remote controller for Vintage B3.
   - Choose Vintage B3 MIDI mode: Native Instruments B4D if you use a Native Instruments B4D remote controller for Vintage B3.
   - Choose Vintage B3 MIDI mode: Nord Electro if you use a Clavia Nord Electro 2 as a remote controller for Vintage B3.

Vintage B3 MIDI mode: Roland VK or Korg CX

This table shows the MIDI controller assignments when MIDI mode is set to Roland VK or Korg CX. Choose either setting if you use a Roland VK series or Korg CX-3 draw bar organ as a remote controller for Vintage B3.

Controller number	MIDI mode VK or CX: assigned parameter
70	draw bar 16'
71	draw bar 5 1/3'
72	draw bar 8'
73	draw bar 4'
74	draw bar 2 2/3'
75	draw bar 2'
76	draw bar 1 3/5'
77	draw bar 1 1/3'
78	draw bar 1'
Rotor Cabinet	
80, 92	Slow/Brake/Fast
81	Slow/Brake
Reverb	
82	Reverb Level

Controller number	MIDI mode VK or CX: assigned parameter
Vibrato	
85	Upper Vibrato on/off
86	Lower Vibrato on/off
87	Chorus Vibrato Type
Percussion	
94	on/off
95	2nd/3rd
102	Percussion Volume
103	Percussion Time
Equalizer	
104	EQ Low
105	EQ Mid
106	EQ Hi
107	EQ Level
Wah	
108	Wah Mode
109	Wah Bite
Distortion	
110	Distortion Type
111	Distortion Drive
112	Distortion Tone
Click Levels	
113	Click On Level
114	Click Off Level
Balance	
115	Main Volume
116	Lower Volume
117	Pedal Volume
Rotor Fast Rate	
118	Rotor Fast Rate

Vintage B3 MIDI mode: Hammond Suzuki

This tables shows the MIDI controller assignments when MIDI mode is set to Hammond Suzuki. This setting matches the controller mapping of Hammond XB-series organs.

Controller number	MIDI mode Hammond Suzuki: assigned parameter
80	All upper draw bars
81	All lower draw bars
82	Pedal draw bars, Scanner Vibrato, Bass Filter
Rotor Cabinet	
Leslie On	Rotor Cabinet on/off
Leslie Fast	Slow/Brake
Leslie Brake	Controls Brake function of Rotor Cabinet
Vibrato	
Vibrato On	Upper Vibrato on/off (on XK-3 only)
Vibrato Mode	Vibrato type (V1-C3, XK-3 only)
87	Chorus Vibrato type
Drive	Distortion Drive
Reverb Level	Reverb Level
Perc 2nd and Perc 3rd	Percussion Harmonic, 3rd harmonic has priority over 2nd. Translation from XK buttons to Vintage B3 is as follows: • 2nd off, 3rd off x Vintage B3: Percussion off • 2nd on, 3rd off x Vintage B3: 2nd Harmonic • 2nd off, 3rd on x Vintage B3: 3rd Harmonic • 2nd on, 3rd on x Vintage B3: 3rd Harmonic
Perc Fast	Selects a preset decay time for fast or slow decay
Perc Soft	Selects a preset level for either soft or normal percussion
Vibrato Mode	Selects either Vibrato Off, V1/V2/V3, or C1/C2/C3 (XK-2 only)
Vibrato VC	Switches between Vibrato or Chorus Vibrato (XK-2 only)

Vintage B3 MIDI mode: Native Instruments B4D

This table shows the MIDI controller assignments when MIDI mode is set to Native Instruments B4D. This setting matches the controller mapping of the Native Instruments B4D controller.

Controller number	MIDI mode Native Instruments B4D: assigned parameter
12	Upper draw bar 16'
13	Upper draw bar 5 1/3'
14	Upper draw bar 8'
15	Upper draw bar 4'
16	Upper draw bar 2 2/3'
17	Upper draw bar 2'
18	Upper draw bar 1 3/5'
19	Upper draw bar 1 1/3'
20	Upper draw bar 1'
21	Lower draw bar 16'
22	Lower draw bar 5 1/3'
23	Lower draw bar 8'
24	Lower draw bar 4'
25	Lower draw bar 2 2/3'
26	Lower draw bar 2'
27	Lower draw bar 1 3/5'
28	Lower draw bar 1 1/3'
29	Lower draw bar 1'
Vibrato	
31	Upper Vibrato on/off
30	Lower Vibrato on/off
Brightness	Vibrato
Attack Time	Chorus Intensity
Percussion	
Sostenuto	Percussion on/off
Release Time	Percussion Harmonic (2nd/3rd)
Sound Variation	Percussion Volume
Harmonic Content	Percussion Time
Equalizer	
90	EQ Low
70	EQ Mid
5	EQ High
Distortion/Click	
76	Distortion Drive

Controller number	MIDI mode Native Instruments B4D: assigned parameter
78	Distortion Tone
75	Click On Level
Leslie	
Pan MSB	Microphone Angle
3	Microphone Distance
GP 8	Leslie Accelerate/Decelerate
GP 7	Leslie Fast
ModWheel MSB	Leslie Speed
68	Controls Brake function: if Value = 0.0, switches Leslie to Brake. All other values switch Leslie to previous speed.

Vintage B3 MIDI mode: Nord Electro

This table shows the MIDI Control Change Message number assignment when MIDI mode is set to Nord Electro. This setting matches the controller mapping of the Clavia Nord Electro 2.

Controller number	MIDI mode Nord Electro: assigned parameter
16	Upper draw bar 16'
17	Upper draw bar 5 1/3'
18	Upper draw bar 8'
19	Upper draw bar 4'
20	Upper draw bar 2 2/3'
21	Upper draw bar 2'
22	Upper draw bar 1 3/5'
23	Upper draw bar 1 1/3'
24	Upper draw bar 1'
70	Lower draw bar 16'
71	Lower draw bar 5 1/3'
72	Lower draw bar 8'
73	Lower draw bar 4'
74	Lower draw bar 2 2/3'
75	Lower draw bar 2'
76	Lower draw bar 1 3/5'
77	Lower draw bar 1 1/3'
78	Lower draw bar 1'
Chorus/Vibrato	
85	Upper Vibrato on/off
86	Lower Vibrato on/off
84	Vibrato mode (selection goes from V1 to C3, C0 is excluded)
Percussion	
87	Percussion on/off
88	Percussion Volume (soft/normal) and Time (short/long)
95	Percussion Harmonic (2nd/3rd)
Equalizer	
113	EQ High
114	EQ Low
Distortion/Click	
111	Distortion Drive
Leslie	
GP 6	on/off
GP 7	Leslie Speed
GP 8	Controls Brake function

B3 and Leslie information

Additive synthesis with draw bars

The Hammond B3 is *the* classic draw bar organ. As with an air-driven pipe organ, the registers (draw bars, or "stops" on a pipe organ) can be pulled out to engage them. In contrast to a pipe organ, however, the B3 allows seamless mixing of any draw bar registers. The closer toward you that the draw bars are dragged, the louder the corresponding tones.

Despite characteristics such as key clicks, variable intonation, distortions, and crosstalk (all of which Vintage B3 emulates), playing a single note, with a single register, results in a pure sine tone. Mixing sine tones results in more complex harmonic spectra; this is known as *additive synthesis*. Organs—even pipe organs—can be regarded as additive synthesizers. Several limitations should be considered before viewing the instrument in this way. These limitations, on the other hand, constitute the charm and character of any real musical instrument.

The naming of the draw bars is derived from the length of organ pipes, measured in feet ('). This naming convention is still used with electronic musical instruments.

- Halving the length of a pipe doubles its frequency.
- Doubling the frequency results in an upward transposition of one octave.

The lowest register—16' (far left, brown draw bar)—and the higher octave registers—8', 4', 2', and 1' (white draw bars)—can be freely mixed, in any combination. 16' is commonly described as the *sub-octave*. With the sub-octave regarded as the fundamental tone, or first harmonic, the octave above 8' is the second harmonic, 4' the fourth harmonic, 2' the eighth harmonic, and 1' the sixteenth harmonic.

With the 5 1/3' register—the second brown draw bar—you can add the third harmonic. This is the fifth above the 8'. Basically, the draw bars are arranged by pitch, with one exception. The second draw bar (5 1/3') sounds a fifth higher than the third draw bar. See The residual effect on page 428 for an explanation.

The 2 2/3' register generates the sixth harmonic, 1 3/5' the tenth harmonic, and 1 1/3' the twelfth harmonic.

An electromechanical tonewheel organ offers the choice of the following registers/harmonics: 1 (16'), 2 (8'), 3 (5 1/3'), 4 (4'), 6 (2 2/3'), 8 (2'), 10 (1 3/5'), 12 (1 1/3'), and 16 (1'). As you can see, the harmonic spectrum is nowhere near complete. This is one of the main reasons for the common practice of using overdrive and distortion effects with electromechanical tonewheel organs—they enrich the harmonic spectra by generating more harmonics.

Note: 2 2/3' is the fifth over 4'. 1 3/5' is the major third over 2'. 1 1/3' is the fifth over 2'. In the bass range, this can lead to inharmonic tones, especially when playing bass lines in a minor key. This is because mixing 2', 1 3/5', and 1 1/3' results in a major chord.

The residual effect

The residual effect is a psychoacoustic phenomenon. Human beings can perceive the pitch of a note, even when the fundamental tone is completely missing. If you pull out all registers of a draw bar organ, except for the fundamental—16'—you'll still perceive the same pitch. The sound becomes thinner, with less bass and less warmth, but the pitch remains the same.

If human beings didn't hear this way, it would make listening to music on a small transistor radio impossible. The tiny speaker of a small radio can't accurately play back the fundamental tone of the bass line because this frequency is far below the range that the speaker can reproduce.

Setting draw bar registrations often involves this psychoacoustic phenomenon. In the lower octaves, mixing the 8' and 5 1/3' sine draw bars creates the illusion of a 16' sound, although the lower frequency is missing.

Old pipe organs also make use of the residual effect, by combining two smaller pipes, thus eliminating the need for long, heavy, and expensive giant pipes. This tradition is continued in modern organs and is the reason for arranging the 5 1/3' under 8': the 5 1/3' tends to create the illusion of a pitch that is one octave lower than 8'.

Tonewheel sound generation

Tonewheel sound generation resembles that of an air horn, or a siren. Of course, there's no air being blown through the holes of a revolving wheel. Rather, an electromagnetic pickup, much like a guitar pickup, is used to capture the sound.

A notched metal wheel, called a *tonewheel*, revolves at the end of a magnetized rod. The teeth of the wheel cause variations in the magnetic field, inducing an electrical voltage. This voltage/tone is then filtered, has vibrato and expression applied to it, and is then amplified.

An AC synchronous motor drives a long drive shaft. Twenty-four driving gears with 12 different gear sizes are attached to the shaft. These gears drive the tonewheels. The frequency depends on the gear ratios and the number of notches in the wheels. The Hammond is tuned to an (almost exact) equal-tempered scale.

As with pipe organs that feature multiplexed registers, the Hammond organ uses certain generators for more than one purpose. Some high frequency wheels serve as the fundamental for high notes and provide harmonics for lower notes. This has a positive impact on the overall organ sound, avoids detuning, and stabilizes levels between octaves.

A brief Hammond history

Three inventions inspired Laurens Hammond (1895–1973), a manufacturer of electric clocks, to construct and market a compact electromechanical organ with tonewheel sound generation. The Telharmonium by Thaddeus Cahill was the musical inspiration; Henry Ford's mass production methods and the domestic synchron clock motor were the other factors.

The Telharmonium (built around 1900) was the first musical instrument that made use of electromechanical sound generation techniques. Its immense tonewheel generators filled a two-story building in New York. For a short period around this time, subscribers could order Telharmonium music over the New York telephone network (the streaming audio system of the time). The only amplification tool was the telephone's mechanical diaphragm because a proper tube amplifier and acceptable speakers had not yet been invented. The Telharmonium was a commercial flop, but its historical status as the predecessor of modern electronic musical instruments is undeniable. The Telharmonium also introduced the principles of electronic additive synthesis (see Additive synthesis with draw bars on page 427).

Laurens Hammond began producing organs in 1935 in Chicago, Illinois, making use of the same sound generation method. However, he used much smaller tone generators and fewer registers. The patent for his model A organ dates from 1934.

Hammond also holds the patent for the electromechanical spring reverb, still found in countless guitar amplifiers today.

The Hammond B3 was manufactured between 1955 and 1974. It is the Hammond model preferred by jazz and rock organ players, such as Fats Waller, Wild Bill Davis, Brother Jack McDuff, Jimmy Smith, Keith Emerson, Jon Lord, Brian Auger, Steve Winwood, Joey DeFrancesco, and Barbara Dennerlein.

In addition to the B3, there are a number of smaller Hammond instruments, known as the spinet series (M3, M100, L100, T100). Bigger console models, many of which were designed to suit the needs of American churches or theaters (H100, X66, X77, E100, R100, G-100), were also manufactured.

The production of electromechanical organs ceased in 1974. Thereafter, Hammond built fully electronic organs.

The Hammond name lives on in the Hammond-Suzuki range of electronic draw bar organs, starting with the 2002 release of a digital B3 model that mimics the design and functions of the classic B3 (without the weight). This model, as well as newer units, can be partnered with real, mechanical, rotor speaker cabinets, also from the company.

The Leslie cabinet

Don Leslie developed his rotor cabinets in 1937 and began marketing them in 1940. Laurens Hammond wasn't keen on the concept of rotating speakers at all.

Leslie's approach was to simulate a variety of locations in the pipes (as in pipe organs), resulting in a new spatial perception for every note. The rotor speaker cabinets could simulate this effect, and the sense of space that they impart is incomparable, when placed side-by-side with any fixed speaker. The periodic undulations in sound and volume and the vibrato caused by the Doppler effect (see below) aren't all there is to the Leslie sound—it's the spatial effect, too.

The "classic" Leslie speaker design features two drivers—a treble driver with horns (only one works; the other simply acts as a counter-weight) and a bass driver. The horns of the treble driver and the sound baffle of the bass driver are physically rotated by electric motors.

Because the speakers rotate toward the front of the cabinet (the listening position), then toward the back of the cabinet, you hear a "Doppler effect"—where sounds become louder and brighter as their position changes. To give you an idea of this effect, it is much like the sound of a train going past if you were standing on the platform. On approach, the sound is muffled, but then it becomes both louder and brighter as the train passes, and finally it becomes more muffled as it moves away from you.

The rotating driver/sound baffle can be switched between two speeds—fast/Tremolo or slow/Chorale (or stopped completely with a mechanical brake). The transition between the two speeds, or the use of a fixed speed, produces the characteristic "Leslie" vibrato, tremolo, and chorus effects.

The first Leslie, the model 30, had no Chorale—just tremolo and stop. The Chorale idea (which came much later) was born of a desire to add a vibrato to the organ. Chorale, which offers far more than a simple vibrato, was first introduced to the market with the 122/147 models. At this time, Leslie also added the "Voice of the pipe organ" label to his cabinets.

It wasn't until 1980 that the two companies and brand names came together, six years after the last tonewheel organ was built. Mechanical Leslie rotor cabinets are still being built today, by the Hammond-Suzuki company.

Vintage Clav

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Vintage Clav overview

Vintage Clav emulates the classic Hohner D6 Clavinet. The sound of the D6 is synonymous with funk, but it was also popularized in the rock, pop, and electric jazz of the 1970s by artists like Stevie Wonder, Herbie Hancock, Keith Emerson, Foreigner, and the Commodores. If you've heard "Superstition" or "Higher Ground" by Stevie Wonder, then you know the D6 sound. See D6 Clavinet history.

Vintage Clav uses a component modeling synthesis engine that not only simulates the basic sounds of the D6 but also the various string buzzes, key clicks, and the tone of the pickups found in the original instrument. Vintage Clav accurately emulates the pluck and bite of the attack phase as well as the sticking of the hammer pads. See D6 Clavinet mechanical details.

The Vintage Clav synthesis engine improves on the Hohner D6 Clavinet with a stereo, rather than mono, output. The 60-key range (F to E) of the original D6 has also been extended across the full MIDI range (127 notes).

Vintage Clav provides extensive sound control options. You can radically alter the tone of the instrument, enabling you to simulate an aging clavinet or to create unique new timbres that have little in common with the sound of a clavinet.

Vintage Clav also incorporates an effects processor that provides classic wah, modulation, and distortion effects—often used with the original instrument. The effects are modeled on vintage effect pedals and adapted for optimized use with Vintage Clav.

For more information about component modeling synthesis, see Component modeling synthesis on page 493.

Vintage Clav interface

Vintage Clav is divided into four areas. The control bar at the top lets you choose a clavinet model and provides access to further controls shown in the central display. The lower area gives you direct access to tone, level, and damper controls. Extended parameters are available at the bottom of the interface.

Use the control bar buttons to update the central display.

- *Model pop-up menu:* Choose a basic type of tone, or model. Each model offers a unique tonal characteristic and different harmonic structure, designed to create different sounds. See Vintage Clav models.
- *Main button:* Shows the Pickups, Stereo Spread, Brilliance, and Decay controls in the central display. See Vintage Clav Main window overview.
- *Effects button:* Parameters for the integrated effects are shown in the central display. See Vintage Clav Effects window overview.
- *Details button:* Opens the Details window where you can modify sound parameters and set global parameters such as the tuning of Vintage Clav. See Vintage Clav Details window overview.
- *Extended parameters:* Click the disclosure triangle at the lower left to access Vintage Clav extended parameters if needed.

Vintage Clav Main window

Vintage Clav Main window overview

The Main window provides access to the most commonly used parameters.

Main window parameters

- *Pickup Position display:* The two pickups shown in the Pickup Position display indicate the positions and angles of the upper (above the strings) and lower (below the strings) pickups. See Use Vintage Clav Pickup parameters.
- *Stereo Spread parameter:* This two-part parameter alters the stereo imaging of Vintage Clav output—controlled by key position. This parameter also provides control of the pickup panning position. See Use Vintage Clav Stereo Spread parameters.
- *Brilliance knob:* Rotate to set the level of the harmonic content caused by string excitation. Positive values—to the right—result in a sharper sound. Negative values result in a more muted sound.
- *Decay knob:* Rotate to set the decay time of the strings, following the attack phase of a played note. Positive values increase the decay time. Negative values reduce the decay time.
- *Filter switches:* The four filter switches emulate the original tone control switches of the D6.
  - *Brilliant:* Click to make the sound nasal—cuts bass.
  - *Treble:* Click to make the sound sharper—gently cuts bass.
  - *Medium:* Click to make the sound thinner—slight bass reduction.
  - *Soft:* Click to make the sound softer—more muted.
- *Pickup switches:* Click the AB and CD switches to alter the "wiring" of the virtual pickups, changing the tone of Vintage Clav. See Use Vintage Clav Pickup parameters.
- *Volume knob:* Rotate to set the overall Vintage Clav output level.

  Note: MIDI controller 11 scales the output level—unless it is assigned to control Wah or Damper parameters.
- *Damper slider:* Drag to mute the strings. The Damper parameter can also be controlled with a MIDI controller. For information on assigning a controller to the Damper slider, see Vintage Clav Misc parameters.

Vintage Clav models

The Model pop-up menu in the control bar lets you choose a basic type of tone, or model. Each model offers a unique tonal characteristic and different harmonic structure, designed to create very different sounds. See Vintage Clav model characteristics.

The individual models are fully realized instruments and are immediately playable, without further modification. You can shape the tonal character of any loaded model with Vintage Clav model editing parameters. See Vintage Clav Details window overview.

In some respects, you can view the choice of model as being similar to selecting an oscillator waveform in a synthesizer. As with raw synthesizer waveforms, parameters can affect the model quite differently. For example, using identical parameter settings can make one model more nasal sounding and another model more noisy.

Note: When playing, you may notice some points on the keyboard where the sound changes significantly between adjacent keys. This is intentional and reflects the behavior of some of the clavinet models emulated by Vintage Clav. The original D6 has some strong key-to-key timbral differences, the most obvious being between the highest wound string, and the lowest, non-wound string. If you like the original's sound but not the original's mechanical timbre jumps, try the Mellotone model.

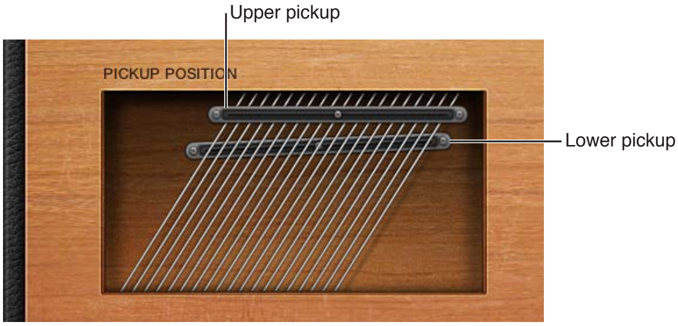
Vintage Clav model characteristics

The table outlines the characteristics of each clavinet model.

Model name	Comments
Belltone	A bell-like model with strong inharmonic overtones (inharmonicities).
Classic I and II	Classic I is a near exact emulation of the original D6. It includes string noises on long decays and accurate behavior following the release of keys. Each D6 was unique in its way, so you can adjust the sound to match the tone of D6 clavinets you have heard. Classic II is brighter and more punchy.
Dulcitone	A dulcimer-like model.
Funktone	This model invites heavy, funk-style bass playing in the lower octaves, coupled with sustained chords in the mid-to-upper octaves. It works well with phaser and delay effects. In the lower bass-octave ranges, the string oscillations become increasingly resonant over time, until they finally collapse (after 20 to 30 seconds). Higher notes have a much shorter decay, which has a corresponding impact on the resonating behavior.
Harpsitone I and II	Harpsichord-like models.
Mellotone	This model is smooth and mellow sounding across the entire keyboard range.
Plectratone I and II	These models emulate a picked string. Change the pickup positions to make the sound more guitar-like. For a harp-like sound, position the lower pickup near the mid point of the Pickup Position display and increase String Release and Excite Shape in the Details window, while decreasing Brilliance in the Main window.
Sitartone	A sitar-like sound, rich in resonance.
Vintage I and II	These models emulate a D6 with aged and worn hammers and strings. The sound of the sticky hammer heads is modeled as well as the richer bass range.
Woodtone	This model sounds wooden, thin, and contains inharmonic overtones. It can sound slightly detuned in some contexts.

Use Vintage Clav Pickup parameters

The original D6 is equipped with two electromagnetic pickups, much like those found in electric guitars—one below the strings (lower) and one above the strings (upper). In contrast to the fixed pickups of the original instrument, Vintage Clav pickups can be set to arbitrary positions *and* angles.



Try moving pickup positions while repeatedly striking a note to hear the effect that the pickup position has on the overall tone. Interesting, phaser-like effects can be achieved by automating the pickup positions.

Settings with both pickups placed near the upper end of the strings and active Brilliant and Treble filter switches result in a weak fundamental tone. Therefore, you will mostly hear the overtones of the chosen model. These can be "out of tune," particularly for models such as Wood, which has strong inharmonic content. Move the pickups toward the center of the Pickup Position display, halfway along the strings, and deactivate all filter switches to circumvent this detuned effect.

You can cross-over the pickups in the Pickup Position display. This may lead to a "hole" (silent or very quiet notes) in your keyboard range. This is due to phase cancelations between the pickups. If you encounter this phenomenon, adjust the position of one or both pickups—until the quiet or silent notes are playable.

Adjust a pickup angle

- Drag the "dot" at one end of a pickup to another position.

Reposition a pickup

- Drag the "dot" in the middle of the pickup to a new position along the strings.

Change the Vintage Clav pickup mode

- Click the AB and CD switches to change pickup modes.

The internal wiring of the two pickups changes with different switch positions, as does the sound at the combined pickup output.

Use Vintage Clav Stereo Spread parameters

Unlike the original D6, Vintage Clav has a stereo output that you configure with the Stereo Spread parameter. It is divided into two halves: Key and Pickup.

The Key parameter sets a key scale modulation of the panning position. In other words, the played keyboard note position determines the panning position.

The Pickup parameter allows you to spread the two pickup signals across the stereo spectrum—when both pickups are active (Upper+Lower and Upper-Lower pickup modes).

You can use both spread types at the same time. They are automatically mixed. The ring around the Stereo Spread button graphically displays the effect of both parameters, as follows:

- The key scale range is indicated by orange areas in the ring.
- The pickup panning position is indicated by red lines in the ring.

Adjust the keyboard position

- Vertically drag in the upper (Key) half of the circular Stereo Spread button. The center position is MIDI note number 60 (C3).

  Set this parameter to the maximum value for extreme left/right panning (semitones) at MIDI note number 60.

Adjust the pickup position in the stereo field

- Vertically drag in the lower (Pickup) half of the circular button. Higher Pickup values move the signals of both pickups away from the center position—one to the right, and the other to the left.

  Set this parameter to the maximum value for extreme left/right panning.

Vintage Clav Effects window

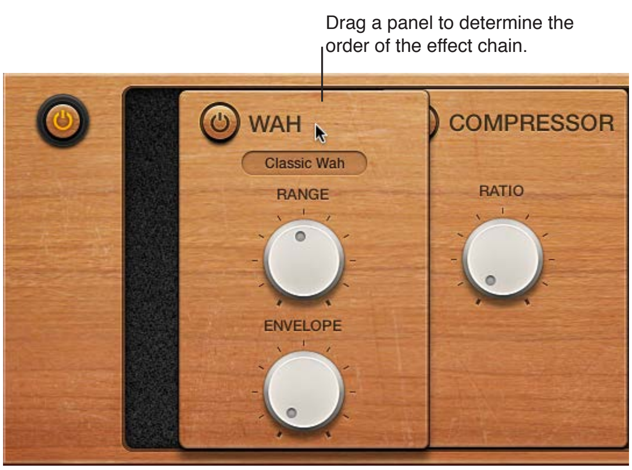
Vintage Clav Effects window overview

No clavinet simulation would be complete without a selection of effect processors. Vintage Clav incorporates three "classic" foot-pedal effect emulations: Distortion, Modulation, and Wah. Each effect is modeled on pedals that were available in the heyday of the Clavinet—the 1970s—adding an authentic sound to your performances. A simple compression circuit is also included and can be placed anywhere in the effects chain.

Click the Master button to turn the entire effects section on or off. Each effect in the effects chain can be independently turned on or off.

Vintage Clav effects work in series—where the output of one effect is fed into the next in an effects chain. The routing order lets you choose whether a distorted signal should be wah-filtered (for funkier sounds) or the wah-filtered sound should be distorted (for screaming sounds)—as an example.

Horizontally drag the name of the effect to determine the order of the effects chain.



Vintage Clav Compressor effect

The Compressor effect is normally used just before the Distortion effect. This allows you to increase or decrease the perceived gain, thus providing a suitable input level to the Distortion circuit. You can, however, place the Compressor at any position in the effects chain or can disable it completely.

Compressor effect parameters

- *On/off button:* Turns the Compressor effect on or off.
- *Ratio knob:* Rotate to adjust the compression slope. The additional gain offered by the compression circuit—when directly preceding the Distortion effect—lets you create crunchy distortions. The Compressor is also useful for enhancing the key click sound and for emphasizing harmonics in different clavinet models.

Vintage Clav Distortion effect

The Distortion effect can provide warm overdrive or aggressive distortion.

Distortion effect parameters

- *On/off button:* Turns the Distortion effect on or off.
- *Gain knob:* Rotate to set the level of the Distortion effect. If Gain is at the minimum value, no distortion is heard.
- *Tone knob:* Rotate to change the tonal color of the Distortion effect.
  - Use low Tone and Gain settings to create warm overdrive effects.
  - Use high Tone and Gain settings for bright, screaming distortion effects.

Vintage Clav Modulation effect

Vintage Clav features a choice of three modulation effect types: Phaser, Flanger, or Chorus.

Modulation effect parameters

- *On/off button:* Turns the Modulation effect on or off.
- *Mode pop-up menu:* Choose Phaser, Flanger, or Chorus as the modulation effect.
- *Intensity knob:* Rotate to set the depth of the phasing, flanging, or chorus effect. Use of high Intensity values leads to ensemble-type effects when the Chorus effect is active.

> ***WARNING:*** When the Phaser effect is active, high Rate and Intensity values lead to very deep, self-oscillating phase shifts that can damage ears and speakers.

- *Sync button:* Turn on to synchronize the Phaser or Flanger effect to the host application tempo. The Rate knob sets bar and beat values, including triplets.
- *Rate knob:* Rotate to set the speed of the phasing, flanging, or chorus effect. The rate is set in hertz values, or bar/beat values when the Sync button is turned on.

Vintage Clav Wah effect

Vintage Clav provides simulations of several classic wah effects, as well as some basic filter types. The name *wah* comes from the sound it produces. It has been a popular effect (usually a pedal effect) with electric guitarists since the days of Jimi Hendrix. The pedal controls the cutoff frequency of a bandpass, lowpass, or—less commonly—highpass filter. Wah wah pedals are also used extensively with the D6. For information about assigning a controller to the Wah effect, see Vintage Clav Misc parameters.

Wah effect parameters

- *On/off button:* Turns the Wah effect on or off.
- *Mode pop-up menu:* Choose a Wah effect type.
  - *Classic Wah:* This setting mimics the sound of a popular wah pedal with a slight peak characteristic.
  - *Retro Wah:* This setting mimics the sound of a popular vintage wah pedal.
  - *Modern Wah:* This setting mimics the sound of a distortion wah pedal with a constant Q(uality) Factor setting. The Q determines the resonant characteristics. Low Q values affect a wider frequency range, resulting in softer resonances. High Q values affect a narrower frequency range, resulting in more pronounced emphasis.
  - *Opto Wah 1:* This setting mimics the sound of a distortion wah pedal with a constant Q(uality) Factor setting.
  - *Opto Wah 2:* This setting mimics the sound of a distortion wah pedal with a constant Q(uality) Factor setting.
  - *Resonant LP:* In this mode, the Wah works as a resonance-capable lowpass filter. At the minimum pedal position, only low frequencies can pass.
  - *Resonant HP:* In this mode, the Wah works as a resonance-capable highpass filter. At the maximum pedal position, only high frequencies can pass.
  - *Peak:* In this mode, the Wah works as a peak (bell) filter. Frequencies close to the cutoff frequency are emphasized.
- *Range knob:* Rotate to set the cutoff frequency of the filter. At the extreme left position, Range limits cutoff modulation to a narrow frequency range. To provide a wider control range, turn the Range knob to the right.
- *Envelope knob:* Rotate to determine the sensitivity of the (filter) envelope to incoming note velocity messages. An *auto wah* effect is produced by using the integrated envelope follower function, which controls the depth of filter cutoff modulation. In practical terms, this means that the dynamics of your performance directly control the depth of the Wah effect.

Vintage Clav Details window

Vintage Clav Details window overview

The Details window lets you precisely control the modeling parameters of Vintage Clav and also provides global parameters that affect the overall instrument. See these sections for details:

- Vintage Clav Excite and Click parameters
- Vintage Clav String parameters
- Vintage Clav Pitch parameters
- Vintage Clav Misc parameters

Vintage Clav Excite and Click parameters

The Excite parameter describes the string excitation, emulating the characteristics and power of the hammers striking the string, and other elements that form part of the initial key strike.

The rubber hammers of the original D6 age and decay, just like piano hammer felts. Worn out D6 units produce a distinctive "click" when a key is released. This is due to the string sticking to the rubber hammer before being released. The characteristics of this release click are part of each model and can be precisely adjusted with the Click parameters.

Excite and Click parameters

- *Shape slider:* Drag to contour the attack shape, simulating the hardness of the rubber hammers in a D6. As the instrument ages, hammers wear and split, changing brightness and tone. Negative values—to the left—provide a softer attack, and positive values result in a harder attack.
- *Intensity slider:* Drag to set the level of the release click. A negative value of −1.00 eliminates the release click. To simulate an old D6, increase the value.
- *Random slider:* Drag to control the amount of click level variance across the keyboard. This slider simulates the wearing of some hammers, but not others, emulating the real-world "wear and tear" of a D6. The farther to the right the slider is moved, the greater the variation between key clicks on some keys. At the leftmost position, all keys have an identical key click level.
- *Velocity slider:* Drag to set the velocity sensitivity for the key click sound. The maximum key click level is set with the Intensity slider and the velocity mode is determined with the Velocity mode switch.
- *Velocity mode switch:* Turns attack (key on) or release (key off) velocity on or off. The Auto setting senses if the connected MIDI keyboard is sending release velocity values. If this is the case, the received release velocity is used to shape the sound; otherwise, it acts as if it is turned off.

Vintage Clav String parameters

The selected model determines the basic qualities of the strings and has a significant bearing on the behavior, and impact, of each String parameter. This is primarily due to the different harmonic content present in each model.

String parameters

- *Release slider:* Drag to set the release time of the strings, following the decay phase of a played note. Positive Release values provide a longer release time—after you have released a key.
- *Damping slider:* Drag to change the damping behavior of the strings. Damping is essentially a faster decay for the higher harmonics in a sound. Damping is directly related to the string material properties—high damping for catgut strings, medium damping for nylon strings, and low damping for steel strings. Depending on the model, damping results in a more mellow and rounded, or woody, sound. A positive Damping value makes the sound more mellow. A negative Damping value allows more high harmonics through, making the sound brighter.
- *Stiffness and Inharmonicity sliders:* Drag to intensify or reduce the inharmonic overtones in the sound. When set at different levels, you can create metallic, bell-like sounds, or DX-style electric piano sounds. Stiffness and Inharmonicity can also be useful for wood bass sounds.
  - The Inharmonicity parameter determines the lowest harmonic—the harmonic threshold. Inharmonic content above this threshold is stretched or spread across the frequency spectrum.
  - The Stiffness parameter controls the intensity of the stretching or spectral spreading set by the Inharmonicity slider.

  Note: The fundamental note pitch is not affected by the Stiffness and Inharmonicity parameters.
- *Tension Mod slider:* Drag to add a slight upward pitch bend effect, immediately after being plucked, struck, or strummed. This type of modulation is common to stringed instruments like the D6, guitars, and so on. A predefined Tension Modulation characteristic is built into each model, but this can be altered with the Tension Mod parameter. The impact of this parameter can be significant, enabling you to generate strange sound effects with Vintage Clav. You can also use it to simulate an out-of-tune clavinet, or a sitar-like sound.
- *Pitch Fall slider:* Drag to set the intensity of a D6 characteristic, where the pitch of each note falls immediately after you release the key. This sonic quirk is due to the physical construction of the D6. The intensity of this effect varies with each model, but it can be completely deactivated by setting Pitch Fall to the leftmost position.

Vintage Clav Pitch parameters

The Pitch parameters affect the tuning of the selected model.

Pitch parameters

- *Tune slider:* Drag to adjust tuning in one-cent intervals. A value of 0 equals concert pitch A 440 Hz.
- *Stretch Tuning slider:* Vintage Clav is tuned to an equal-tempered scale. You can deviate from this standard tuning by using Stretch to alter the tuning in the bass and treble ends of the sound. This simulates the way stringed keyboard instruments such as pianos are tuned (see information below).

  Note: Use of both Warmth and Stretch may result in a detuned sound that is quite similar to a heavy chorus effect. In some instances, this effect may be so extreme that Vintage Clav sounds out of tune with your project or concert.

- *Warmth slider:* Drag to set the amount of random deviation from an equal-tempered scale. High values add life to sounds. The Warmth parameter can be useful when you are emulating an instrument that has not been tuned for a while, or for slightly thickening a sound. When you are playing chords, the Warmth parameter creates a slight detuning or beating effect between notes.
- *Pitch Wheel slider:* Drag to determine the pitch bend range in semitone steps. You can use your keyboard's pitch bend wheel/stick to control pitch bends.
- *Pitch Pressure slider:* On an original D6, applying pressure (aftertouch) to a depressed key raises the pitch slightly. Pitch pressure emulates this behavior. Pitch Pressure parameter values to the left of the center position lower the pitch slightly with aftertouch messages. Values to the right raise pitch.

Stretch tuning in acoustic instruments
The tones of upright pianos, and to a lesser extent grand pianos (due to their longer strings), have inharmonicities in their harmonic structure. This also applies to other stringed instruments, but it particularly affects pianos due to the length, density, and tension of the strings. If a piano is perfectly tuned to equal temperament across the keyboard range, the overtones of the low strings and the fundamentals of the high strings will sound out of tune with each other.

To circumvent this problem, piano tuners use a technique known as *stretch tuning*, where the high and low registers of the piano are tuned higher and lower, respectively. This results in the harmonics of the low strings being in tune with the fundamental tones of the upper strings. In essence, pianos are intentionally "out of tune" (from equal temperament), so that the lower and upper registers will sound in tune.

Because the original D6 is a stringed instrument, this inharmonic relationship also applies to Vintage Clav and the instruments it emulates. The stretch feature, however, was primarily included for situations where you want to use Vintage Clav alongside an acoustic piano recording or performance.

Vintage Clav Misc parameters

The Misc(ellaneous) parameters affect the entire Vintage Clav instrument, rather than an individual model.

MIDI controller assignments allow you to control Vintage Clav with an external MIDI controller or with your host application.

Misc parameters

- *Voices pop-up menu:* Choose the maximum number of voices that can be played simultaneously. Lowering the value of this parameter limits the polyphony and processing requirements of Vintage Clav. There are two monophonic settings: mono and legato. Each setting provides a single voice when playing Vintage Clav.
  - *Mono:* Vintage Clav voice is triggered each time a key is pressed.
  - *Legato:* Vintage Clav sound-shaping processes are not triggered if the notes are played legato—only the pitch changes. If the notes are played staccato, a Vintage Clav voice with all sound-shaping processes is triggered.

- *Velocity Curve pop-up menu:* Choose one of nine preset velocity curves to suit your playing style or the selected model. The nine curves available are: fx25%, fx50%, fx75%, fx100%, convx1, convx2, linear (the default), concv1, and concv2.
  - *Fixed (fx) curves:* These are linear curves with a fixed dynamic range of 25%, 50%, 75%, and 100%.
  - *Convex (convx) curves:* These curves are more dynamically responsive in the center octaves of the keyboard range.
  - *Concave (concv) curves:* These curves are less dynamically responsive in the center octaves of the keyboard range.
- *Damper Control pop-up menu:* Choose the MIDI controller (or MIDI velocity or aftertouch) you want to use to control the Damper parameter. Off disables MIDI control.
- *Wah Control pop-up menu:* Choose the MIDI controller you want to use as a manual Wah effect control. MIDI foot controllers such as Expression pedals are commonly used for this type of task, but any controller can be assigned. You can also use MIDI velocity or aftertouch messages to control the Wah effect. Off disables MIDI control. Choose the Learn menu item to automatically assign the parameter to the first appropriate incoming MIDI data message, then move the controller on your MIDI keyboard. Learn mode has a 20 second time-out feature: If Vintage Clav does not receive a MIDI message within 20 seconds, the parameter reverts to its original MIDI controller assignment.

  Note: You can simultaneously control the Wah effect with both the integrated envelope follower function ("auto-wah"—see Vintage Clav Wah effect on page 441) and a manual controller. In this situation, the controller events of the envelope follower and manual controls are mixed.

- *Wah Pedal Position slider (Controls view):* Choose View > Controls to access the Wah Pedal Position slider. The value of this parameter represents the current pedal position, ensuring that it is saved with the setting. Choose the Learn menu item to automatically assign the parameter to the first appropriate incoming MIDI data message, then move the controller on your MIDI keyboard. Learn mode has a 20 second time-out feature: If Vintage Clav does not receive a MIDI message within 20 seconds, the parameter reverts to its original MIDI controller assignment.

Vintage Clav extended parameters

Vintage Clav provides two parameters that can be accessed by clicking the disclosure triangle at the lower left of the interface.

Extended parameters

- *MIDI Mono Mode pop-up menu:* Choose Off, On (with common base channel 1), or On (with common base channel 16).

  In either Mono mode, each voice receives on a different MIDI channel. Controllers and MIDI messages sent on the base channel affect all voices.

- *Mono Mode Pitch Range pop-up menu:* Choose 0, 24, or 48.

  The chosen pitch bend range affects individual note pitch bend messages received on all but the common base channel. The default is 48 semitones, which is compatible with Mobile GarageBand's keyboard in pitch mode. When using a MIDI guitar, 24 semitones is the preferable setting because most guitar to MIDI converters use this range by default.

D6 Clavinet information

D6 Clavinet history

The German company Hohner, manufacturer of the D6 Clavinet, was known mainly for its reed instruments (harmonicas, accordions, melodicas, and so on) but had made several classic keyboards prior to the first incarnation of the Clavinet, known as the Cembalet.

Musician and inventor Ernst Zacharias designed the Cembalet in the 1950s. It was intended to be a portable version of the cembalo, or harpsichord—which could be amplified. Its mechanism worked by plucking the end of a flat reed with the key, which was then picked up and amplified, in much the same way as an electric guitar.

A year or two after the Cembalet's release, two Pianet models appeared. Both the CH and N models used flat reeds for tone generation but employed a very different plucking/striking action. When a key was depressed, it engaged a sticky pad with a foam backing, which actually stuck to the reed. When the key was released, the weight of the key caused the pad adhesive to free itself from the reed. This made the reed vibrate, and this vibration was then amplified.

The model T Pianet was released several years later and utilized a soft rubber suction pad on the reeds, rather than the adhesive of the CH and N models. This method still had several drawbacks, however, because the dynamics available from the keyboard were limited. As a further shortcoming, all reeds were damped on release, thus negating any possibility of sustaining the sound via a foot pedal. Despite these problems, the sound of the model T Pianet was popularized by bands such as the Zombies and Small Faces in the 1960s.

In the years between the releases of the Pianet N and T models, Zacharias invented what was to become Hohner's most successful, and certainly funkiest, keyboard—the Clavinet. The Clavinet was designed to replicate the sound of a clavichord, but with an altogether fuller sound (the clavichord was notoriously thin sounding).

The early models—Clavinet I with a built-in amp, Clavinet II with tonal filters, Clavinet L with its bizarre triangular shape—all led to the Clavinet model C. This, in turn, was refined into the more portable D6. The D6 uses a hammer action, which strikes a string against a metal surface to produce a tone. It has a fully dynamic keyboard because the striker is directly beneath the key—the harder you hit, the louder and more vibrant the tone.

Mention the Clavinet today and most people will automatically think of Stevie Wonder's "Superstition"—a recording that owes as much to the D6 as it does to the artist who wrote and performed it. The D6 was later superseded by the E7 and the Clavinet/Pianet Duo. These were basically the same as the D6 but more roadworthy, quieter, and better protected against proximity hums than previous models.

D6 Clavinet mechanical details

Each D6 keyboard key forms a single arm lever. When a key is depressed, a plunger below the key strikes the string and presses it onto an anvil. The string hits the anvil with a strength determined by key velocity, thus affecting both the dynamics and harmonics of the sounding string.

The mechanical vibrations of the action are captured by magnetic pickups and converted into electrical signals, which are amplified and reproduced through speakers.

As the key is released, contact between the plunger/striker and the anvil is immediately broken, leaving the wool-wound part of the string free. This immediately suppresses the string vibration.

When experimenting with Vintage Clav, or auditioning some of the included settings, you may encounter sounds that seem to be triggered on both the note on *and* the note off.

This is actually a feature that emulates the original D6. The real D6 has the "problem" of strings sticking to worn-out hammers, producing a second trigger when the key is released. You can adjust the intensity of this key-off click using the Intensity slider. See Vintage Clav Excite and Click parameters on page 442.

Vintage Electric Piano

17

Vintage Electric Piano overview

Vintage Electric Piano simulates the sound of various Rhodes and Wurlitzer pianos as well as the sound of the Hohner Electra Piano. See Rhodes models on page 458 and Hohner and Wurlitzer models.

The unmistakable tones of Fender Rhodes pianos are some of the best-known keyboard instrument sounds used in the second half of the 20th century. Various Rhodes models have been popularized in a wide range of musical styles, encompassing pop, rock, jazz, and soul, as well as more recent genres such as house and hip-hop. Nearly as popular was the Wurlitzer piano, which enjoyed most of its success in the 1970s.

Vintage Electric Piano's sound engine uses component modeling techniques to generate ultrarealistic electric piano sounds, with smooth dynamics and scaling over the entire 88-key range. Component modeling has no abrupt changes between samples, sample looping, or filtering effects during the decay phase of notes.

Vintage Electric Piano also simulates the physical characteristics of the original instruments, including the movement of the electric piano reeds, tines, and tone bars in the (electric and magnetic) fields of the pickups. It also emulates the ringing, smacking, and bell-like transients of the attack phase as well as the hammer action and damper noises of the original instruments. See Component modeling synthesis on page 493.

The integrated effects include classic equalizer, overdrive, stereo phaser, stereo tremolo, and stereo chorus effects that are commonly used with electric piano sounds.

Vintage Electric Piano interface

Vintage Electric Piano is divided into four areas. The control bar at the top lets you choose an electric piano model and provides access to further controls shown in the main display. The central area gives you direct access to Bass Boost and Volume controls. Extended parameters are available at the bottom of the interface.

Click the control bar Effects and Details buttons to update the main display.

- *Model pop-up menu:* Choose an electric piano model. Several Rhodes models are available, plus Hohner Electra Piano and Wurlitzer models. See Rhodes models on page 458 and Hohner and Wurlitzer models.

  Note: When you choose a new model, all currently sounding voices are muted and all parameters are reset to default values.

- *Effects button:* Click to show the EQ, Drive, Phaser, Tremolo, and Chorus effect parameters in the main display area.
- *Details button:* Click to show parameters in the main display area that enable you to alter the tone and playing behavior of the selected instrument model.
- *Bass Boost knob:* Rotate to enhance the low end of the sound. This parameter emulates the behavior of the control found on the original Rhodes piano.
- *Volume knob:* Rotate to set the overall output level of Vintage Electric Piano.
- *Extended parameters:* Click the disclosure triangle at the lower left to access Vintage Electric Piano extended parameters if needed.

Vintage Electric Piano Effects window

Vintage Electric Piano EQ

The EQ allows you to boost or cut the high and low frequency ranges of your Vintage Electric Piano sound. The EQ is positioned after the Drive circuit in the Vintage Electric Piano effects chain.

EQ parameters

- *On/off button:* Turns the equalizer on or off.
- *Bass knob:* Rotate to control the low frequency range. Either shelving or peak-type filters are used—depending on the piano model selected. Optimized frequency ranges are preselected for each model.
- *Treble knob:* Rotate to control the high frequency range. Either shelving or peak-type filters are used—depending on the piano model selected. Optimized frequency ranges are preselected for each model.

  Tip: You can achieve a sound with a more dominant mid-range by suppressing the treble and bass frequency ranges. If you require more precise equalization, you can insert any of the equalizer plug-ins in the instrument channel strip. You can also use the Tone control of the Drive effect to contour the harshness of your sound.

Vintage Electric Piano Drive effect

Electric pianos sound best when played through tube amplifiers. Tube amplifiers offer a wide range of tones—from the subtle warmth or crunch of guitar amplifiers to psychedelic, screaming rock distortions. The Vintage Electric Piano Drive effect simulates the saturation characteristics of a tube amplifier stage. The Drive effect is the first signal processing circuit in the Vintage Electric Piano effects chain.

Drive effect parameters

- *On/off button:* Turns the Drive effect on or off.
- *Distortion Type switch:* Switch between two types of distortion effect.
- *Gain knob:* Rotate to set the amount of harmonic distortion.
- *Tone knob:* Rotate to equalize the sound before it is amplified or distorted by the virtual tube amplifier circuit.
  - Use low Tone values to set a mellow tonal color. If the sound becomes too soft, boost the treble portion of your sound with the EQ Treble control.
  - Use higher Tone values for harsh distortion characteristics, typical of overdriven transistor stages. If the sound is too aggressive, suppress the treble portion of your sound with the EQ Treble control.

Vintage Electric Piano Chorus effect

Chorus is the most commonly used effect on electric piano sounds. The Vintage Electric Piano Chorus effect is based on a delay circuit. The delay time is modulated by an LFO. The delayed effect signal is mixed with the original signal.

Chorus parameters

- *On/off button:* Turns the Chorus effect on or off.
- *Rate knob:* Rotate to set the speed of the Chorus effect, in Hz. High values may result in the piano sounding detuned.
- *Intensity knob:* Rotate to set the intensity of the Chorus effect (technically, the amount of delay time deviation).

Vintage Electric Piano Phaser effect

The Vintage Electric Piano Phaser effect is based on analog phaser pedals used by electric guitarists in the 1960s and 1970s, including the subtle analog-style distortion typical of these units. These phaser pedals were also popular among electric pianists—especially in the electric jazz, jazz-rock, and pop styles of the 1970s.

The Phaser effect runs the original signal through a series of four filters that enhance particular aspects of the Vintage Electric Piano frequency spectrum. This filtered signal is slightly phase delayed and mixed with the original signal, resulting in notches in the frequency spectrum. The notches in the phase-delayed signal are moved up and down through the frequency spectrum by an LFO (low frequency oscillator) modulation. This results in the amplitudes of the two signals reaching their highest and lowest points at slightly different times.

Note: Logic Pro offers a sophisticated Phaser effect (and other modulation plug-ins) that can be used alongside, or to replace, the integrated Vintage Electric Piano Phaser effect.

Phaser effect parameters

- *On/off button:* Turns the Phaser effect on or off.
- *Rate knob:* Rotate to set the speed of the phasing effect. The rate is set in Hz values, or bar/beat values when the Sync button is turned on.
- *Sync button:* Turn on to synchronize the Phaser effect to the host application tempo. The Rate knob sets bar and beat values, including triplets.
- *Color knob:* Rotate to set the amount of Phaser output signal that is fed back into the Phaser effect input. This changes the tonal color of the phasing effect.
- *Stereo knob:* Rotate to determine the relative phase shift between the left and right channels.
  - At a value of 0 the effect is most intense, but not stereophonic.
  - At a value of 180 the effect symmetrically rises in the left channel while falling in the right channel, and vice versa.

Vintage Electric Piano Tremolo effect

A periodic modulation of the amplitude (level) of the sound is known as a *tremolo*. This modulation is controlled with an LFO in Vintage Electric Piano. The Fender Rhodes suitcase piano features a stereo tremolo. Other electric pianos have a simple, often obtrusive, mono tremolo that can introduce an unusual polyrhythmic feel to performances.

Tremolo effect parameters

- *On/off button:* Turns the Tremolo effect on or off.
- *Rate knob:* Rotate to set the speed of the tremolo effect (LFO frequency). The rate is set in Hz values, or bar/beat values when the Sync button is turned on.
- *Sync button:* Turn on to synchronize the Tremolo effect to the host application tempo. The Rate knob sets bar and beat values, including triplets.
- *Intensity knob:* Rotate to set the amount of amplitude modulation.
- *Stereo knob:* Rotate to determine the relative phase shift between the left and right channels.
  - A value of 0 changes the level of both channels—in phase.
  - A value of 180 (out-of-phase modulation), results in a stereo tremolo effect that is also known as *auto panning*. This is similar to manually turning the pan pot from side to side.

  Tip: The original Wurlitzer Piano has a mono tremolo with a fixed modulation rate of 5.5 Hz. For an authentic Wurlitzer sound, select a Stereo value of 0 degrees. For Rhodes sounds, set the Stereo value to 180 degrees. The settings in between result in spacious effects—especially when low Rate knob values are used.

Vintage Electric Piano Details window

Vintage Electric Piano model parameters

Click the Details button on the control bar to use the model parameters. The model parameters affect the currently selected model.

Model parameters

- *Voices knob:* Rotate to set the maximum number of voices that can sound simultaneously. Lower the value to limit polyphony. When Voices is set to 1, Vintage Electric Piano is monophonic. The maximum value is 88, allowing for glissandi over the entire keyboard range when the sustain pedal is depressed.
- *Decay knob:* Rotate to set the decay time of the piano sound. The lower the value, the less the sound sustains and the higher the level of damping applied to the vibration of the tines. When short values are set, the main tone is more pronounced and is heard for a longer period than the transient harmonics. Sonically, the effect is reminiscent of an electric guitar string being damped with the palm of the picking hand. Electric pianos can be modified in a similar way. Higher values (longer settings) result in more sustain and a less dynamic feel.
- *Release knob:* Rotate to set the amount of damping applied after the keys are released. Extremely long settings (high Release values) let you play the piano like a vibraphone.
- *Stereo Width knob:* Rotate to set the sound's stereo field. At high values, bass notes are heard in the left channel and treble notes are heard in the right channel.

  Tip: Avoid using this parameter if you are trying to faithfully recreate a vintage electric piano because these instruments were not equipped with stereo outputs.
- *Tine Bell knob:* Rotate to set the level of the (inharmonic) treble portion of the tone. This is useful for emulating classic electric piano sounds.
- *Damper Noise knob:* Rotate to set the level of damper noise caused by the damping felt hitting the vibrating tine of the original instruments.

Vintage Electric Piano pitch parameters

Click the Details button on the control bar to use the pitch parameters. Vintage Electric Piano is tuned to an equal-tempered scale. You can deviate from this scale and can stretch the tuning in the bass and treble ranges, much as you can do with acoustic pianos (especially upright pianos). You can also modulate the tuning of each note randomly.

Pitch parameters

- *Tune knob:* Rotate to tune Vintage Electric Piano in one-cent increments. A value of 0 equals concert pitch A (440 Hz). The range is plus or minus half a semitone.
- *Down/Up (Bend Range) knobs:* Rotate to set the pitch bend range in semitone steps.
- *Warmth knob:* Rotate to set the amount of (random) deviation from the equal-tempered scale. Each note is slightly detuned from the next, adding life and richness to the sound, particularly when high Warmth values are used.

  Note: Use of both Warmth and Upper or Lower Stretch can result in a detuned sound that is similar to a heavy chorus effect. In some instances, this effect may be so extreme that Vintage Electric Piano sounds out of tune with the rest of your project or concert.
- *Lower (Stretch Tuning) knob:* Rotate to set the amount of deviation from the equal-tempered scale in the bass end of the sound. The higher the value, the farther *down* the low notes are tuned. At a setting of 0, Vintage Electric Piano is tuned to an equal-tempered scale, with each octave down halving the frequency.
- *Upper (Stretch Tuning) knob:* Rotate to set the amount of deviation from the equal-tempered scale in the treble end of the sound. The higher the value, the farther *up* the high notes are tuned. At a setting of 0, Vintage Electric Piano is tuned to an equal-tempered scale, with each octave above (up) doubling the frequency.

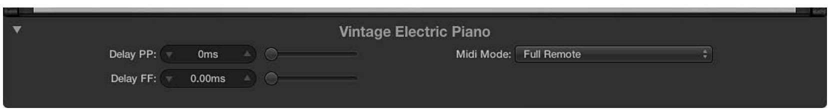
Stretch tuning in acoustic instruments
The tones of upright pianos, and to a lesser extent grand pianos (due to their longer strings), have inharmonicities in their harmonic structure. This also applies to other stringed instruments, but it particularly affects pianos due to the length, density, and tension of the strings. If a piano is perfectly tuned to equal temperament across the keyboard range, the overtones of the low strings and the fundamentals of the high strings will sound out of tune with each other.

To circumvent this problem, piano tuners use a technique known as *stretch tuning*, in which the high and low registers of the piano are tuned higher and lower, respectively. This results in the harmonics of the low strings being in tune with the fundamental tones of the upper strings. In essence, pianos are intentionally "out of tune" (from equal temperament), so that the lower and upper registers will sound in tune.

Electric pianos don't have strings, so this inharmonic relationship doesn't apply to Vintage Electric Piano nor to the original instruments it emulates. The stretch feature was primarily included for situations where you want to use Vintage Electric Piano alongside an acoustic piano recording or performance.

Vintage Electric Piano extended parameters

Click the disclosure triangle at the lower left of Vintage Electric Piano to show or hide the extended parameters.



Extended parameters

- *Delay PP slider and field:* Drag to set the delay time (in milliseconds) when the keys are struck pianissimo (PP-soft).
- *Delay FF slider and field:* Drag to set the delay time (in milliseconds) when the keys are struck forte (FF-hard).
- *Midi Mode pop-up menu:* Determine how Vintage Electric Piano responds to MIDI controllers. Choose from: Off, Modwheel to Tremolo, and Full Remote.

Vintage Electric Piano emulations

Rhodes models

Harold Rhodes (born 1910) constructed what is arguably the best known and most widely used electric piano. Designed in 1946—as a piano surrogate for practice, education, and army entertainment—the Rhodes piano was marketed by guitar manufacturer Fender from 1956. The Fender Rhodes is one of the most popular musical instruments in jazz, especially electric jazz. CBS took over production of the Rhodes in 1965, enhancing its popularity in pop and rock music. There are also a number of Rhodes synthesizers, developed by former manufacturer ARP. Japan's Roland corporation owned the Rhodes name for a while and released several digital pianos under the Rhodes moniker. From 1997 until his death in December 2000, Harold Rhodes again owned the name.

The Rhodes piano was also made available as a suitcase piano (with pre-amplifier and two-channel combo amplifier) and as a stage piano, without amplifier. Both of these 73-key "portable" versions have a vinyl-covered wooden frame and a rounded plastic top. In 1973, an 88-key model was introduced. Smaller Celeste and bass versions were less popular. The MkII (1978) had a flat top that allowed keyboardists to place extra keyboards on top. The Mark V, introduced in 1984, had a MIDI output.

The mid-1980s saw a decrease in Rhodes production, as most keyboard players invested in the lighter, more flexible digital synthesizers that became available around this time. These keyboards could easily emulate the Rhodes sound and also offered a range of new piano sounds.

The Rhodes piano bases its method of sound generation on metal reeds, which function much like a tuning fork. These reeds are struck with a velocity sensitive hammer action that works in a similar fashion to that of a grand piano. The asymmetrical tuning fork consists of a thin tine and a large tone bar that are bolted together. Due to construction considerations, some of the tone bars are rotated by 90 degrees. The piano is kept in tune by the mass of a spring, which can be moved along the tine. The tine oscillates in front of an electric pickup, similar to that of an electric guitar. This oscillation functions along inductive principles, with permanent magnets placed around the tine that have a damping effect on tine movement and therefore, the sound.

The Rhodes output signal is like that of an electric guitar and requires pre-amplification. The Rhodes sound is not harmonically rich. This is why so many performers use a treble boost or an overdrive effect when playing the Rhodes piano. The Rhodes sounds best when played through tube amplifiers.

The characteristic sound of each Rhodes piano depends more on the adjustment and maintenance of the individual instrument than on the model. Early models had hammers covered with felt, resulting in a smoother sound than later models with neoprene-covered hammers. The suitcase piano featured a pre-amplifier that could create a sound with a very dominant mid-range. Appropriate pre-amplification and equalization can, however, deliver an identical tone from almost any stage piano. The MkII does not have the treble range resonance clamps of earlier models; it has less sustain in the treble range. The most significant sonic differences are dependent on the proximity of the tine to the pickup. When the tine is moved closer to the pickup, the bell characteristic becomes more prominent. In the 1980s, many Rhodes pianos were adjusted to have more "bell."

Note: The Vintage Electric Piano Metal Piano and Attack Piano models feature "idealized" sound qualities that could only be aimed at with the original Rhodes instruments. Although these models may not sound realistic, they have at least partially achieved the ideals that the Rhodes technicians might have had in mind when preparing their keyboards.

Hohner and Wurlitzer models

Not to be confused with the all-electronic RMI Electrapiano, the extremely rare Hohner Electra Piano offers striking hammers like those of the Rhodes, but a stiffer keyboard action. It was designed to resemble the look of a conventional acoustic upright piano. Led Zeppelin's John Paul Jones played it on "Stairway to Heaven," "Misty Mountain Hop," and "No Quarter."

Wurlitzer, best-known for manufacturing music boxes and organs, also built electric pianos that helped write pop and rock music history. The 200 series, notably the 200A and 240V, Wurlitzer pianos are smaller and lighter than the Rhodes pianos, with a keyboard range of 64 keys (A to C) and an integrated amplifier and speakers.

The velocity sensitive hammer action resembles that of a conventional acoustic piano. The Wurlitzer sound generation system is based on spring steel reeds that can be tuned with a solder weight. The Wurlitzer has electrostatic pickups: The reeds are supplied with a 0-volt current and move between the teeth of a comb, connected to a 150-volt current. The tone of the Wurlitzer, which was first manufactured in the early 1960s, features a number of odd harmonics.

The Wurlitzer is best known as the signature piano sound of the band Supertramp, as heard on their "Crime of the Century" album. You might also recognize the Wurlitzer sound when listening to Pink Floyd's "The Dark Side of the Moon" or "I Am the Walrus" by The Beatles.

Note: The Vintage Electric Piano Funk Piano model offers a special synthetic piano engine sound, with an exaggerated bass. This is not based on any real-world Wurlitzer instruments, but it can be a very useful sound nonetheless.

Vintage Electric Piano MIDI controllers

Vintage Electric Piano responds to the following MIDI continuous controller numbers (CC).

Controller number	Parameter name
1	Volume knob
12	Model pop-up menu
13	Model parameters: Decay knob
14	Model parameters: Release knob
15	Model parameters: Tine Bell knob
16	Model parameters: Damper Noise knob
17	Model parameters: Stereo Width knob
18	EQ parameters: Treble knob
19	EQ parameters: Bass knob
20	Drive parameters: Gain knob
21	Drive parameters: Tone knob
22	Phaser parameters: Rate knob
23	Phaser parameters: Color knob
24	Phaser parameters: Stereo knob
25	Tremolo parameters: Rate knob
26	Tremolo parameters: Intensity knob
27	Tremolo parameters: Stereo knob
28	Chorus parameters: Intensity knob

Legacy instruments

A Appendix

Legacy instruments overview

The legacy instruments are less CPU- and memory-intensive versions of equivalent Logic Pro instruments. All legacy instruments feature a few carefully chosen parameters that provide maximum impact and flexibility, making it easy to create great sounds.

Legacy instruments are automatically loaded when a GarageBand project is imported, or a Logic project or MainStage concert that uses these instruments is opened.

Display and insert legacy instrument plug-ins in Logic Pro

1. Press Option, then click the instrument slot on a channel strip.

   The plug-in menu opens, with a Legacy submenu shown below Vintage Electric Piano.

2. Choose the legacy instrument plug-in that you want to insert from the Legacy submenu.

Emulated instruments

Bass

Bass emulates electric and acoustic basses.

- *Volume slider:* Sets the overall volume level of the instrument.
- *Filter Cutoff slider:* Allows less sound through at low values and more at high values—damping the sound or making it brighter.
- *Release slider:* Determines the time it takes for notes to fade out after you let go of the keys on your keyboard.

Church Organ

Church Organ emulates a pipe organ.

- *Volume slider:* Sets the overall volume level of the instrument.
- *Registration pop-up menu:* Provides a number of preset registrations. Registrations are combinations of different pipe organ stop (lever) settings that change the tonal character of the sound by enabling or disabling specific pipes, thus altering the harmonics that are heard when you play a key.
- *Bass buttons:* You can activate the lower (bass) pipes with these buttons, adding these lower harmonics, which makes your sound richer and fuller.

  Note: The bass buttons are not available in all registrations.
- *Cutoff slider:* Allows less sound through at low values and more at high values—damping the sound or making it brighter.
- *Attack slider:* Makes the sound start faster or slower. A fast setting makes it sound like striking a piano key, whereas a slow setting makes it sound like bowing a violin string.
- *Release slider:* Determines the time it takes for notes to fade out after you let go of the keys on your keyboard.

Drum Kits

Drum Kits includes: rock, pop, jazz, electronic, orchestral, and Latin kits, among others.

- *Volume slider:* Sets the overall volume level of the instrument.
- *Filter Cutoff slider:* Allows less sound through at low values and more at high values—damping the sound or making it brighter.
- *Release slider:* Determines the time it takes for notes to fade out after you let go of the keys on your keyboard.

Electric Clav(inet)

Electric Clavinet, which is based on Vintage Clav, emulates the Hohner D6 clavinet.

- *Voices pop-up menu:* Choose the maximum number of voices that can be played simultaneously. Lowering the value of this parameter limits the polyphony and processing requirements of Electric Clav. There are two monophonic settings: mono and legato. Each setting provides a single voice when playing Electric Clav.
  - *Mono:* Electric Clav voice is triggered each time a key is pressed.
  - *Legato:* Electric Clav sound-shaping processes are not triggered if the notes are played legato—only the pitch changes. If the notes are played staccato, a voice with all sound-shaping processes is triggered.
- *Damper slider:* Changes the tone, making it less sustained and more woody sounding as you move toward the high setting.
- *Volume slider:* Sets the overall volume level of the instrument.
- *Auto Wah slider:* Sets the sensitivity of the (filter) envelope to incoming note velocity messages.

  An *auto wah* effect is produced by using the integrated envelope follower function, which controls the depth of filter cutoff modulation. In practical terms, this means that the dynamics of your performance directly control the depth of the Wah effect.
- *Phaser slider:* Sets the overall level of the integrated Phaser effect.

  The Phaser effect adds a sweeping, whooshing quality to your clavinet sound.

Electric Piano

Electric Piano, which is based on Vintage Electric Piano, sounds like the Fender Rhodes and Wurlitzer electric pianos.

- *Model buttons:* A more bell-like tone is achieved when the Tines button is selected.
- *Decay slider:* A short value makes the sound almost plucked, whereas a long setting sustains the sound while the keys are held.
- *Bell Volume slider:* Makes the sound more bell-like, with a stronger ringing tone.
- *Voices pop-up menu:* Choose the maximum number of voices that can be played simultaneously.
- *Tremolo slider:* Sets the maximum tremolo (wobbling pitch) intensity.
- *Chorus slider:* Sets the level of the integrated Chorus effect.

  Chorus makes the sound richer and thicker.
- *Volume slider:* Sets the overall volume level of the instrument.
- *MIDI Mode pop-up menu:* Choose Off or Modwheel to Tremolo.

  When the latter is chosen, move your keyboard modulation wheel to set the tremolo (wobbling pitch) intensity.

Guitar

Guitar emulates a number of acoustic and electric guitar sounds.

- *Volume slider:* Sets the overall volume level of the instrument.
- *Filter Cutoff slider:* Allows less sound through at low values and more at high values—damping the sound or making it brighter.
- *Release slider:* Determines the time it takes for notes to fade out after you let go of the keys on your keyboard.

Horns

Horns emulates several brass sections and a number of individual brass instruments.

- *Volume slider:* Sets the overall volume level of the instrument.
- *Filter Cutoff slider:* Allows less sound through at low values and more at high values—damping the sound or making it brighter.
- *Attack slider:* Makes the sound start faster or slower. A fast setting makes it sound like striking a piano key, whereas a *slow* setting makes it sound like bowing a violin string.
- *Release slider:* Determines the time it takes for notes to fade out after you let go of the keys on your keyboard.

Piano

Piano emulates a number of classical and jazz piano sounds as well as several accordions and a harpsichord. It also provides a number of pad sounds.

- *Volume slider:* Sets the overall volume level of the instrument.
- *Filter Cutoff slider:* Allows less sound through at low values and more at high values—damping the sound or making it brighter.
- *Release slider:* Determines the time it takes for notes to fade out after you let go of the keys on your keyboard.

Sound Effects

Sound Effects provides a number of nature sounds, laughter, applause, and so on.

- *Volume slider:* Sets the overall volume level of the instrument.
- *Filter Cutoff slider:* Allows less sound through at low values and more at high values—damping the sound or making it brighter.
- *Release slider:* Determines the time it takes for notes to fade out after you let go of the keys on your keyboard.

Strings

Strings emulates string sections and a number of individual stringed instruments. These include violins, violas, cellos, harps, and several traditional instruments, such as the sitar, koto, and zither.

- *Volume slider:* Sets the overall volume level of the instrument.
- *Filter Cutoff slider:* Allows less sound through at low values and more at high values—damping the sound or making it brighter.
- *Attack slider:* Makes the sound start faster or slower. A fast setting makes it sound like striking a piano key, whereas a slow setting makes it sound like bowing a violin string.
- *Release slider:* Determines the time it takes for notes to fade out after you let go of the keys on your keyboard.

Tuned Percussion

Tuned Percussion emulates a vibraphone, xylophone, timpani, steel drums, and other tuned percussion instruments.

- *Volume slider:* Sets the overall volume level of the instrument.
- *Filter Cutoff slider:* Allows less sound through at low values and more at high values—damping the sound or making it brighter.
- *Release slider:* Determines the time it takes for notes to fade out after you let go of the keys on your keyboard.

Voice

Voice emulates a mixed choir.

- *Volume slider:* Sets the overall volume level of the instrument.
- *Filter Cutoff slider:* Allows less sound through at low values and more at high values—damping the sound or making it brighter.
- *Attack slider:* Makes the sound start faster or slower. A fast setting makes it sound like striking a piano key, whereas a slow setting makes it sound like bowing a violin string.
- *Release slider:* Determines the time it takes for notes to fade out after you let go of the keys on your keyboard.

Woodwind

Woodwind emulates the sound of wind instruments, such as flutes, clarinets, saxophones, and several other instruments from various world cultures.

- *Volume slider:* Sets the overall volume level of the instrument.
- *Filter Cutoff slider:* Allows less sound through at low values and more at high values—damping the sound or making it brighter.
- *Attack slider:* Makes the sound start faster or slower. A fast setting makes it sound like striking a piano key, whereas a slow setting makes it sound like bowing a violin string.
- *Release slider:* Determines the time it takes for notes to fade out after you let go of the keys on your keyboard.

Tonewheel Organ

Tonewheel Organ, which is based on Vintage B3, emulates the Hammond B3 organ, but it can also emulate Farfisa, Wurlitzer, and other electric organs.

- *Drawbars slider:* Drag to increase or decrease the level of sine tones and harmonics, resulting in a richer or thinner sound.
- *Percussion Time slider:* Sustains the second or third harmonic when set to a long value. If a short value is selected, the harmonics are heard only during the initial keystroke.
- *Distortion slider:* Makes the sound rough, dirty, and noisy.
- *Rotary Speaker pop-up menu:* Choose one of three speaker effects.
  - *Slow:* Makes the sound swirl.
  - *Brake:* Makes the sound swirl initially and then slow down.
  - *Fast:* Makes the sound wobble.
- *Click slider:* Introduces a click sound to the keystroke. Select a high level if you'd like this to be clearly heard.
- *Volume slider:* Sets the overall volume level of the instrument.
- *Percussion Level slider:* Sets the level of the second or third harmonic added to the sound with the Perc. Harmonic buttons.

Synthesizers

Analog Basic

Analog Basic, which is based on the ES2, is a simple analog synthesizer that is useful for a range of musical styles.

- *Volume slider:* Sets the overall volume level of the instrument.
- *Mix slider:* Determines the balance between the oscillator signals.
- *Tuning slider:* Sets the overall pitch of the instrument.
- *Cutoff slider:* Allows less sound through at low values and more at high values—damping the sound or making it brighter.
- *Resonance slider:* Emphasizes the frequency range around the point determined by the Cutoff parameter.
- *Attack slider:* Makes the sound start faster or slower. A *fast* setting makes it sound like striking a piano key, whereas a *slow* setting makes it sound like bowing a violin string.
- *Decay slider:* Makes the bright portion of the sound sustain for a longer time at slow values. Faster values move to the Sustain level more quickly.
- *Sustain slider:* Determines the level of the sound after the Attack or Decay phase has completed.

Analog Mono

Analog Mono, which is based on the ES2, is an analog lead synthesizer that is monophonic—only one note can be played at a time.

- *Volume slider:* Sets the overall volume level of the instrument.
- *Glide slider:* Determines the time it takes a note pitch to change, or slide, to another note pitch.
- *Mix slider:* Determines the balance between the oscillator signals.
- *Cutoff slider:* Allows less sound through at low values and more at high values—damping the sound or making it brighter.
- *Resonance slider:* Emphasizes the frequency range around the point determined by the Cutoff parameter.
- *Attack slider:* Makes the sound start faster or slower. A *fast* setting makes it sound like striking a piano key, whereas a *slow* setting makes it sound like bowing a violin string.
- *Decay slider:* Makes the bright portion of the sound sustain for a longer time at slow values. Faster values move to the Sustain level more quickly.
- *Richness slider:* Determines the complexity of the sound texture, making the sound fuller.

Analog Pad

Analog Pad, which is based on the ES2, can generate warm analog synthesizer pad sounds that are useful for a range of musical styles.

- *Volume slider:* Sets the overall volume level of the instrument.
- *Modulation slider:* Makes the sweeping movement of the pad faster or slower.
- *Character slider:* Determines whether the sound is soft or sharp.
- *Cutoff slider:* Allows less sound through at low values and more at high values—damping the sound or making it brighter.
- *Resonance slider:* Emphasizes the frequency range around the point determined by the Cutoff parameter.
- *Cutoff Envelope slider:* Determines the strength of the sweeping movement.
- *Duration slider:* Determines the duration of the sweeping movement.
- *Animation slider:* Determines the envelope's effect on the pad sound.

Analog Swirl

Analog Swirl, which is based on the ES2, can generate chorused, swirling analog synthesizer pad sounds.

- *Volume slider:* Sets the overall volume level of the instrument.
- *Modulation slider:* Makes the sweeping movement of the pad faster or slower.
- *Cutoff slider:* Allows less sound through at low values and more at high values—damping the sound or making it brighter.
- *Resonance slider:* Emphasizes the frequency range around the point determined by the Cutoff parameter.
- *Attack slider:* Makes the sound start faster or slower. A *fast* setting makes it sound like striking a piano key, whereas a *slow* setting makes it sound like bowing a violin string.
- *Decay slider:* Makes the harmonic, or bright, portion of the sound sustain for a longer time at slow values. Faster values move to the Sustain level more quickly.
- *Sustain slider:* Determines the level of the sound after the Attack or Decay phase has completed.
- *Release slider:* Determines the time it takes for notes to fade out after you let go of the keys on your keyboard.

Analog Sync

Analog Sync, which is based on the ES2, emulates tones from analog synthesizers that synchronize two oscillators to produce their sound. The Analog Sync instrument is most useful for hard-edged analog synthesizer lead sounds.

- *Volume slider:* Sets the overall volume level of the instrument.
- *Sync slider:* Determines the synchronization (or lack of it) between the two oscillators, and therefore the harshness of the sound.
- *Sync Modulation slider:* Determines how much the synchronization of the two oscillators is modulated, resulting in more complex (and harder) tones.
- *Sync Envelope slider:* Determines the amount that envelope parameters affect the sound.
- *Cutoff slider:* Allows less sound through at low values and more at high values—damping the sound or making it brighter.
- *Attack slider:* Makes the sound start faster or slower. A *fast* setting makes it sound like striking a piano key, whereas a *slow* setting makes it sound like bowing a violin string.
- *Decay slider:* Makes the harmonic, or bright, portion of the sound sustain for a longer time at slow values. Faster values move to the Sustain level more quickly.
- *Sustain slider:* Determines the level of the sound after the Attack or Decay phase has completed.

Digital Basic

Digital Basic, which is based on the ES2, can generate simple digital synthesizer sounds that are useful for a range of musical styles.

- *Volume slider:* Sets the overall volume level of the instrument.
- *Mix slider:* Mixes two tones together.
- *Tuning slider:* Sets the overall pitch of the instrument.
- *Harmonics slider:* Increases or decreases the number of harmonics, or overtones, in the sound. This can change the sound dramatically or subtly, so feel free to experiment.
- *Timbre slider:* Changes the color of the sound from dark to bright.
- *Attack slider:* Makes the sound start faster or slower. A *fast* setting makes it sound like striking a piano key, whereas a *slow* setting makes it sound like bowing a violin string.
- *Decay slider:* Makes the harmonic, or bright, portion of the sound sustain for a longer time at slow values. Faster values move to the Sustain level more quickly.
- *Release slider:* Determines the time it takes for notes to fade out after you let go of the keys on your keyboard.

Digital Mono

Digital Mono, which is based on the ES2, is ideal for monophonic digital synthesizer lead sounds.

- *Volume slider:* Sets the overall volume level of the instrument.
- *Tuning slider:* Sets the overall pitch of the instrument.
- *Harmonics slider:* Increases or decreases the number of harmonics, or overtones, in the sound, with *more* values making the sound a little thicker and *less* values making it thinner.
- *Timbre slider:* Changes the color of the sound from dark to bright.
- *Timbre Envelope slider:* Dynamically changes the color of the sound, depending on how hard you strike the keyboard.
  - Low values result in little or no effect on the color of the sound, no matter how hard you play the keys.
  - High values result in significant changes in the sound, in response to firmer or softer keyboard playing.
- *Decay slider:* Makes the harmonic, or bright, portion of the sound sustain for a longer time at slow values. Faster values move to the Sustain level more quickly.
- *Richness slider:* Subtly detunes each played note from the others, making the sound a little thicker, particularly when *high* parameter values are used.
- *Distortion slider:* Distorts the overall sound, making it quite nasty and aggressive.

  Important: Be careful with the Distortion parameter, which can significantly increase the overall volume of the instrument and possibly cause damage to your speakers or ears.

Digital Stepper

Digital Stepper, which is based on the ES2, is a digital synthesizer that can step through a number of tones, creating a rhythmic pattern.

- *Volume slider:* Sets the overall volume level of the instrument.
- *Balance slider:* Sets the balance between a harder and more spiky sound (digital) and a warmer, softer sound (analog).
- *Modulation slider:* Applies more or less modulation, making the sound more lively when high settings are used.
- *Harmonics slider:* Increases or decreases the number of harmonics, or overtones, in the sound, with *more* values making the sound a little thicker and *less* values making it thinner.
- *Harmonic Steps slider:* Determines how noticeable the tonal steps are, with *large* values making them more noticeable and *small* values less noticeable.
- *Cutoff slider:* Allows less sound through at low values and more at high values—damping the sound or making it brighter.
- *Cutoff Steps slider:* Sets the amount of cutoff applied to each step, with *large* values making the cutoff effect more pronounced and *small* values less pronounced.
- *Duration slider:* Sets the length of the steps.

Hybrid Basic

Hybrid Basic is a sample-based synthesizer that can create spectacular sounds.

- *Volume slider:* Sets the overall volume level of the instrument.
- *Waveform pop-up menu:* Choose the sample set used to generate the basic synthesizer sound.
- *Glide slider:* Determines the time it takes a note pitch to change, or slide, to another note pitch.
- *Wheel to Vibrato slider:* Determines the amount of pitch modulation by your keyboard's modulation wheel.
- *Wheel to Cutoff slider:* Determines the depth of Cutoff modulation by your keyboard's modulation wheel.
- *Cutoff slider:* Allows less sound through at low values and more at high values—damping the sound or making it brighter.
- *Cutoff Type pop-up menu:* Choose from a number of preset filter curves. Try them out, and experiment with the Cutoff and Resonance parameters.
- *Resonance slider:* Emphasizes the frequency range around the point determined by the Cutoff parameter.
- *Cutoff Attack slider:* Determines the time it takes before the Cutoff parameter begins to affect the sound.
- *Attack slider:* Makes the sound start faster or slower. A *fast* setting makes it sound like striking a piano key, whereas a *slow* setting makes it sound like bowing a violin string.
- *Decay slider:* Makes the harmonic, or bright, portion of the sound sustain for a longer time at slow values. Faster values move to the Sustain level more quickly.
- *Sustain slider:* Determines the level of the sound after the Attack or Decay phase has completed.
- *Release slider:* Determines the time it takes for notes to fade out after you let go of the keys on your keyboard.

Hybrid Morph

Hybrid Morph is a sample-based synthesizer that can create spectacular sounds. It differs from Hybrid Basic in that each waveform is based on two sample layers, which leads to a different sonic character.

- *Volume slider:* Sets the overall volume level of the instrument.
- *Waveform pop-up menu:* Choose the sample set used to generate the basic synthesizer sound.
- *Morph slider:* Controls crossfades between the two sample layers.
- *Morph Envelope slider:* Controls the morph over time. For example, setting the Morph parameter to B and the Morph Envelope to "From A to B" morphs the Wave from A to B, in accordance with the ADSR envelope settings.

  Note: If you set the Morph parameter to A and the Morph Envelope to "From A to B," some ADSR (envelope) settings will result in no sound. This lets you use the modulation wheel to offset the Morph parameter during live performances, resulting in interesting sounds.
- *Cutoff slider:* Allows less sound through at low values and more at high values—damping the sound or making it brighter.
- *Cutoff Type pop-up menu:* Enables you to choose from a number of preset filter curves. Try them out, and experiment with the Cutoff and Resonance parameters.
- *Resonance slider:* Emphasizes the frequency range around the point determined by the Cutoff parameter.
- *Cutoff Envelope slider:* Determines the strength of the envelope shaping applied to the Cutoff parameter.
- *Attack slider:* Makes the sound start faster or slower. A *fast* setting makes it sound like striking a piano key, whereas a *slow* setting makes it sound like bowing a violin string.
- *Decay slider:* Makes the harmonic, or bright, portion of the sound sustain for a longer time at slow values. Faster values move to the Sustain level more quickly.
- *Sustain slider:* Determines the level of the sound after the Attack or Decay phase has completed.
- *Release slider:* Determines the time it takes for notes to fade out after you let go of the keys on your keyboard.

Synthesizer Basics

B Appendix

Synthesizer basics overview

If you are new to synthesizers, this appendix will help you understand the basics of sound itself and how this applies to synthesizers.

Important facts about synthesizers are discussed and explained, including the differences between analog, digital, and virtual analog synthesizers. You will also be introduced to the major synthesizer terms as you learn about the basic workings of these hardware- or software-based devices.

This appendix is not a detailed, scientific treatise on the inner workings and mathematical theories of synthesis. It is a basic guide to what you need to know, including some extras that are useful to know.

Experiment with the ES1, ES2, and other instruments while you read. Seeing and using the parameters and other elements that are discussed will help you understand the conceptual and practical aspects of synthesizers.

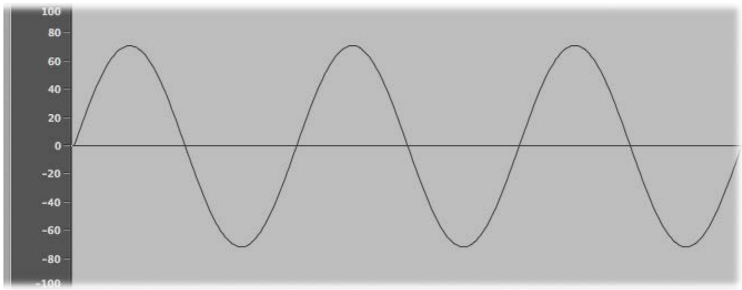
Sound basics

Sound basics overview

Prior to considering any of the sound-generating components you will find in a synthesizer, it's important that you understand sound itself.

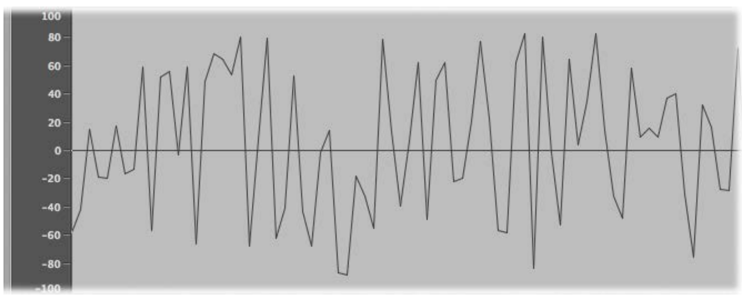
Technically, sound is the conversion of physical energy—such as a hand clap—to an air pressure disturbance. This change in air pressure is transmitted as a series of vibrations—a sound wave—through the air. Sound vibrations can also be transmitted through other matter, such as a wall or floor.

If the vibrations follow a periodic pattern, the sound is said to have a *waveform*.



The figure above shows an oscillogram—a graphical representation—of a sine wave, the simplest and purest kind of waveform.

If the vibrations do not follow a discernible pattern, the sound is called *noise*.



A repetition of a waveform—each peak and trough in the oscillogram—is known as a *cycle*. The number of cycles that occur per second determines the basic pitch of the waveform—commonly known as the *frequency*. Most instruments provide an oscillator frequency control, measured in Hertz (Hz), that determines the number of cycles per second—and therefore the basic pitch of your sound.

Tones, overtones, harmonics, and partials

The base, or core, frequency of a sound is known as its *fundamental tone*.

The waveforms of all sounds, apart from a basic sine wave, consist of the fundamental tone *and* many other tones of different frequencies.

Nonfundamental tones that are whole-number multiples of the fundamental tone are known as *overtones* or *harmonics*. (A tone with a frequency that is a fraction of the fundamental tone is referred to as a *subharmonic*.)

- The fundamental tone is referred to as the *first harmonic*. This is generally louder than the other harmonics.
- A tone played at twice the frequency of the first harmonic is called the *second harmonic*.
- A tone played at four times the frequency of the first harmonic is called the *fourth harmonic*, and so on.

Each of these harmonics has a timbral quality that is different from that of the fundamental tone. In general, harmonics that can be multiplied or divided by a whole number, such as octaves, odd-numbered or even-numbered harmonics, and so on, sound more "musical."

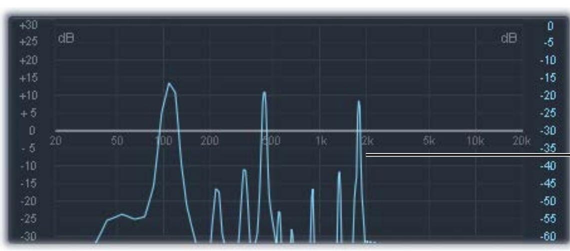
Tones that cannot be multiplied or divided by a whole number are known as *inharmonic overtones,* or *partial tones*. When you combine a number of these inharmonic overtones, it tends to sound "noisy."

Nonfundamental tones that are multiplied by fractional amounts—not whole numbers—are called *partials*.

The frequency spectrum

A fundamental tone, when combined with various harmonics of different levels, is perceived as a sound. The level relationships between these sonic elements change over time (controlled by *envelopes*, as described in Amplifier envelope overview). The combination of a number of harmonics is referred to as the *harmonic spectrum* or, more commonly, the *frequency spectrum*.

The frequency spectrum shows all individual sonic elements in a sound. It is shown low to high, and runs from left to right over time. The respective levels of all harmonics are reflected vertically, with taller spikes indicating higher levels.

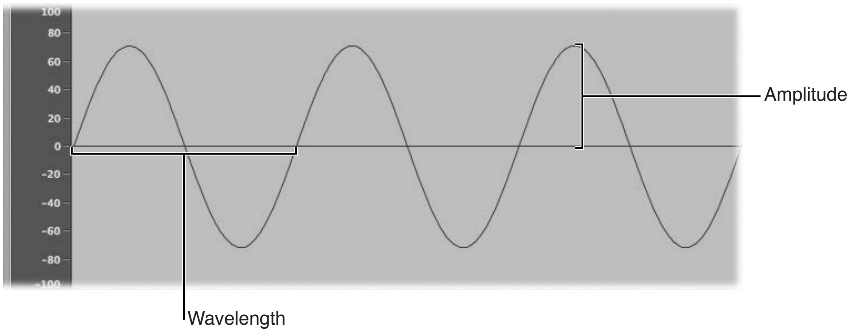


Frequency spectrum graphic of an organ sound

The illustration shows the level and frequency relationships between the fundamental tone and the harmonics at a particular moment in time. These relationships constantly change over time, which results in continuous changes to the frequency spectrum and, therefore, changes to the sound.

Other waveform properties

In addition to *frequency*, other properties of sound waves include *amplitude, wavelength, period,* and *phase*.

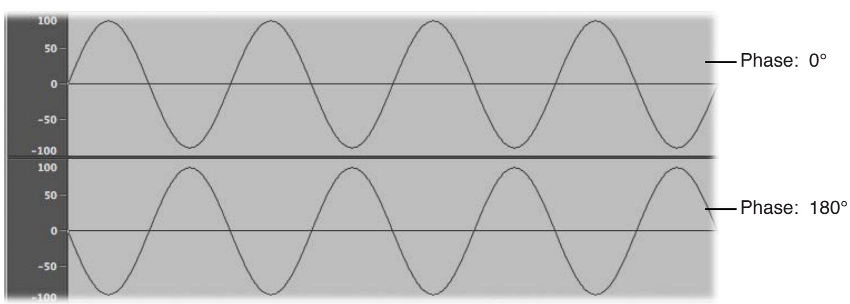


- *Amplitude:* The amplitude of a waveform indicates the amount of air pressure change. It can be measured as the maximum vertical distance from zero air pressure, or "silence" (shown as a horizontal line at 0 dB in the illustration). Put another way, amplitude is the distance between the horizontal axis and the top of the waveform peak, or the bottom of the waveform trough.
- *Wavelength:* The wavelength is the distance between repeating cycles of a waveform of a given frequency. The higher the frequency, the shorter the wavelength.
- *Period:* The wave period is the amount of time it takes to complete one full revolution of a waveform cycle. The higher and faster the frequency, the shorter the wave period.
- *Phase:* Phase compares the timing between waveforms and is measured in degrees—from 0 to 360.

  When two waveforms begin at the same time, they are said to be *in phase* or *phase aligned*. When a waveform is slightly delayed in comparison to another waveform, the waveforms are said to be *out of phase*.

  Note: It is difficult to discern a constant phase difference over the entire wave period, but if the phase of one of the waveforms changes over time, it will become audible. This is what happens in common audio effects such as *flanging* and *phase shifting*.

  When you play two otherwise identical sounds out of phase, some frequency components—harmonics—can cancel each other out, thereby producing silence in those areas. This is known as *phase cancelation*, and it occurs where the same frequencies intersect at the same level.



Fourier theorem and harmonics

According to the Fourier theorem, every periodic wave can be seen as the sum of sine waves with certain wave lengths and amplitudes, the wave lengths of which have harmonic relationships—that is, ratios of small numbers. Translated into more musical terms, this means that any tone with a certain pitch can be regarded as a mix of sine tones consisting of the fundamental tone and its harmonics, or overtones. For example, the basic oscillation—the fundamental tone or first harmonic—is an "A" at 220 Hz, the second harmonic has double the frequency (440 Hz), the third harmonic oscillates three times as fast (660 Hz), the next harmonics four and five times as fast, and so on.

Synthesizer fundamentals

Sound synthesis is the electronic production of sounds—starting from basic properties such as sine tones and other simple waves.

Synthesizers are so named because they can emulate, or *synthesize*, a wide variety of sounds—such as the sound of another instrument, a voice, a helicopter, a car, or a barking dog. Synthesizers can also produce sounds that don't occur in the natural world. The ability to generate tones that cannot be created in any other way makes the synthesizer a unique musical tool.

The simplest form of synthesizer would be a basic sine wave generator that provided little or no control over pitch. Such a synthesizer would not be able to synthesize anything except a sine wave. Combining multiple sine generators with pitch control, however, can produce interesting and useful tones.

In a synthesizer, the task of tone generation falls to a component known as an *oscillator*. Most synthesizer oscillators generate harmonically rich waveforms such as *sawtooth, triangle, square*, and *pulse* waves, in addition to sine waves. These waveform names are based on the resemblance of their respective shapes to a tooth on the blade of a saw, to a triangle, to a square, and so on. For information about the most common synthesizer waveforms, see Oscillators on page 479.

Sculpting the fundamental tone and related harmonics into another sound is achieved by routing the signal from one component, also known as a *module*, to another in the synthesizer. Each module performs a different job that affects the source signal.

In a modular synthesizer, signal routing is achieved by physically cabling modules to each other. In most modern synthesizers the signal routing between modules is internally prewired and is typically changed using switches, knobs, and other controls.

For a discussion of synthesizer components and their interaction with each other to control and shape sound, see How subtractive synthesizers work on page 477.

Synthesizers have existed far longer than you might imagine. In the days that preceded the use of digital technology, all electronic synthesizers were analog. Prior to the use of electricity, synthesizers were mechanical. There are significant differences between analog and digital synthesizers:

- *Analog:* An analog synthesizer combines voltage-controlled circuits—such as oscillators, filters, and amplifiers—to generate and shape sounds. The amount of voltage is typically related directly to the waveform pitch, with higher voltages equaling higher pitches.
- *Digital:* In a digital synthesizer, the signal flow is digital. Binary *descriptions* of the signal—a string of zeros and ones—are fed from one algorithm to another.

- *Hybrid analog and digital synthesizers:* Some synthesizer designs feature digital oscillators that generate signals—using binary descriptions of waveforms. The digital oscillator signal is then sent to analog filters and amplifiers. The main advantage of this approach is that digital oscillators don't drift in pitch, which is a common problem in analog oscillators.
- *Virtual analog:* A virtual analog synthesizer is a digital synthesizer that mimics the architecture, features, and peculiarities of an analog synthesizer. The behaviors and functions of the oscillators, filters, and other modules that you would find in an analog synthesizer are emulated by computer algorithms.

ES1 is a virtual analog synthesizer. Its virtual signal flow is that of a typical analog synthesizer, but all components and signal processing—the virtual oscillators, filters, and so on—are calculated by the central processing unit (CPU) of your computer.

ES1 emulates some of the idiosyncrasies of particular analog circuits—in cases where they tend to sound nice—such as high oscillator levels overdriving the filter. Other analog synthesizer phenomena, such as slowly drifting out of tune (as the instrument heats up), are not simulated.

Virtual analog synthesizers have other advantages over their analog counterparts as well. They're programmable, which means that you can save sound settings; they can be automated, so you can record and play back fader and knob movements; and they are often multitimbral, which allows you to play different sounds at the same time, on different instrument channels. Aspects such as polyphony—the ability to play multiple notes—and velocity sensitivity are found in most virtual analog synthesizers but in very few analog instruments.

Subtractive synthesizers

How subtractive synthesizers work

There are many approaches to sound creation with a synthesizer. (See Other synthesis methods overview on page 491.) There are also numerous differences between synthesizer models, but most follow a fundamentally similar architecture and signal flow that is based on subtractive synthesis principles.

According to legend, when Michelangelo was asked how he managed to carve David out of a block of stone, he replied, "I just cut away everything that doesn't look like David."

In essence, this is how subtractive synthesis works. You filter, or cut away, parts of the sound that you don't want to hear. In other words, you subtract parts of the frequency spectrum, consisting of the fundamental tone and associated harmonics.

Subtractive synthesis assumes that an acoustic instrument can be approximated with a simple oscillator that can produce waveforms with different frequency spectrums. The signal is sent from the oscillator to a filter that represents the frequency-dependent losses and resonances in the body of the instrument. The filtered (or unfiltered) signal is shaped over time by the amplifier section of the synthesizer.

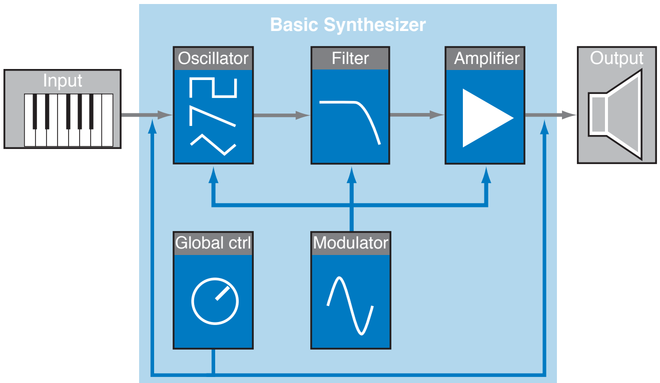
The distinctive timbre, intonation, and volume characteristics of a real instrument can theoretically be recreated by combining these components in a way that resembles the natural behavior of the instrument you are trying to emulate.

In reality, however, subtractive synthesizers aren't perfect at emulating real-world instruments. No synthesized clarinet is going to be mistaken for a real clarinet—particularly when compared to samplers like the EXS24 mkII, which are able to recreate real instruments far more convincingly by using multigigabyte sound libraries.

The true strength of subtractive synthesizers is that they offer a unique sound palette of their own.

Subtractive synthesizer components

The front panel of most subtractive synthesizers provides similar signal-generating and processing modules—coupled with a number of modulation and control modules. The signal-generating and processing modules typically run from left to right, mirroring the synthesizer's signal flow.



Signal-generating and processing components

- *Oscillators:* Generate the basic signal. This is usually a waveform that is rich in harmonics (see Oscillators on page 479). Many synthesizers provide more than one oscillator, and almost all synthesizer oscillators can generate several waveform types.
- *Filter section:* Used to alter the basic signal by filtering out (removing) portions of the frequency spectrum. Many synthesizers have a single filter that is applied universally to all oscillator signals. Multioscillator synthesizers can provide multiple filters, allowing each oscillator signal to be filtered in a different way (see Filters overview on page 482).
- *Amplifier section:* Used to control the level of the signal over time. The amplifier has a module known as an *envelope*, which is divided into several elements that provide level control for the beginning, middle, and end portions of your sound. Simple synthesizers generally have a single envelope, which is used to control the oscillator (and filter) over time. More complex synthesizers can provide multiple envelopes (see Amplifier envelope overview on page 485).

Modulation and control components

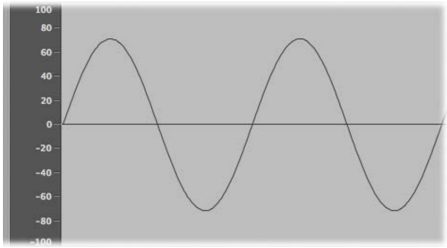
- *Modulators:* Used to modulate the signal-generating and processing components. Modulations can be machine-based—automatically generated by a synthesizer component—or can be manually activated by using the modulation wheel, for example. Most synthesizers have a component called an *LFO* (low frequency oscillator) to provide a waveform that modulates the signal. See Modulation overview on page 487.
- *Global controls:* Set the overall characteristics of your synthesizer sound, such as glides between notes, pitch bends, and monophonic or polyphonic playback (see Global controls on page 490).

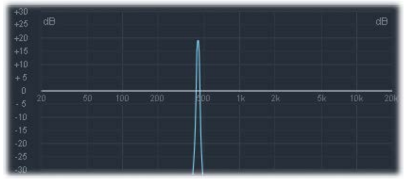
Oscillators

The audio signal of a synthesizer is generated by the oscillator. You can choose from a selection of waveforms that contain various types and amounts of harmonics. The level relationships between the fundamental tone and the harmonics of the chosen waveform are responsible for the basic sound color or timbre.

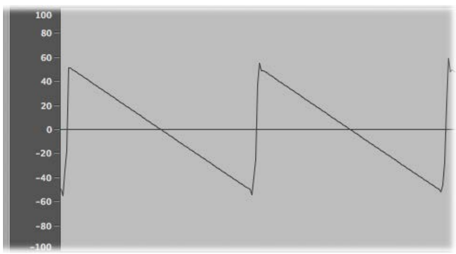
Waveform types

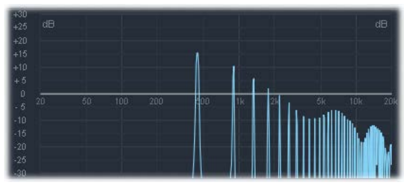
- *Sine wave:* Clean and clear-sounding, a sine wave contains only the first harmonic; in other words, it is the fundamental tone. The sine wave, used alone, can create "pure" sounds like whistles, the sound of wet fingers on the rim of a glass, tuning forks, and so on.



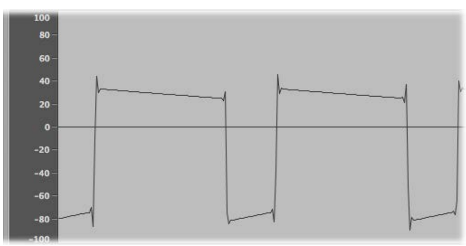


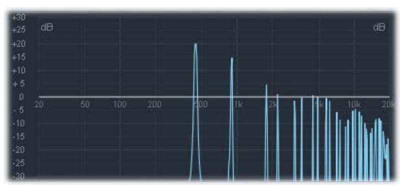
- *Sawtooth wave:* Clear and bright-sounding, a sawtooth wave contains both odd and even harmonics, as well as the fundamental tone. It is ideal for creating string, pad, bass, and brass sounds.



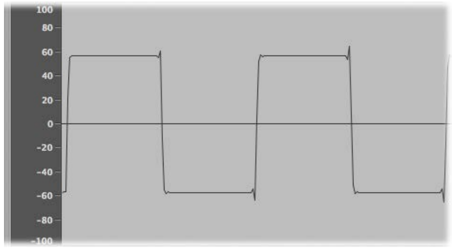


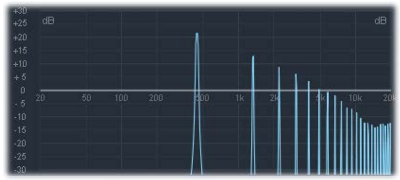
- *Square and pulse waves:* Hollow and woody-sounding, a square wave can contain a wide range of odd harmonics, as well as the fundamental tone. It is useful for creating reed instruments, pads, and basses. It can also be used to emulate kick drums, congas, tom-toms, and other percussive instruments—often when blended with another oscillator waveform, such as noise.



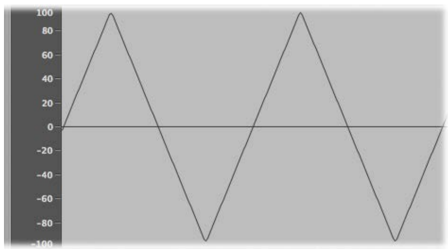


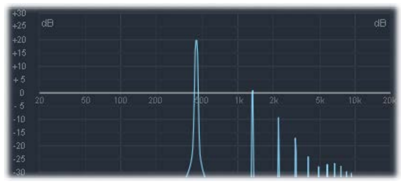
The square wave can be reshaped to make the waveform cycles, or pulses, more rectangular, by using a *pulse width modulation* (PWM) control. The more rectangular the wave becomes, the more nasal it sounds. When modulated in this way, the square wave is known as a *pulse wave*, and contains fewer harmonics. It can be used for reeds, basses, and brass sounds.



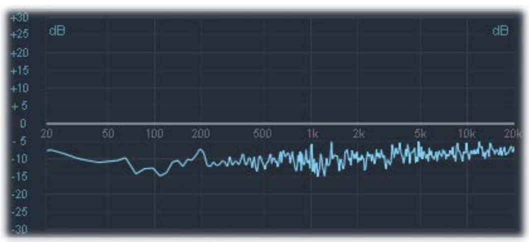


- *Triangle wave:* A triangle wave contains only odd harmonics, as well as the fundamental tone. The triangle wave's higher harmonics roll off faster than those of a square wave, making the triangle wave sound softer. It is ideal for creating flute sounds, pads, and vocal "oohs."

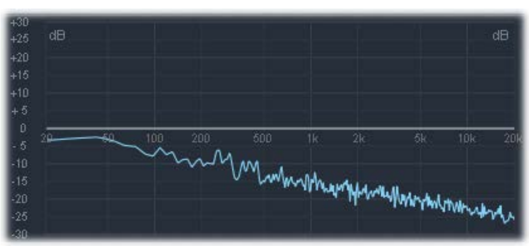




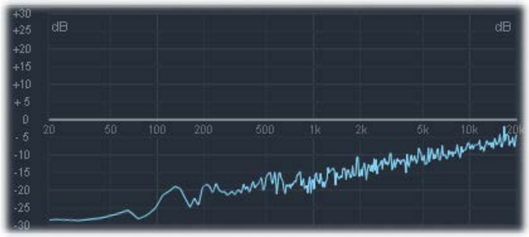
- *Noise: white, pink and red, blue:* Noise is useful for emulating percussive sounds, such as snare drums, or wind and surf sounds. There are more noise wave colors than those listed, but they are rarely found in synthesizers.
  - *White noise:* The most common noise waveform found on synthesizers. White noise contains all frequencies—at full level—around a center frequency.



  - *Pink and red noise:* These noise colors also contain all frequencies, but they are not at full level across the frequency spectrum. Pink noise decreases the level of higher frequencies by 3 dB per octave. Red noise decreases the level by 6 dB per octave.



• *Blue noise:* Blue noise is inverse pink noise, and *increases* the level of all frequencies in higher octaves by 3 dB.



You can deform the basic waveforms to create new waveforms, which results in a different timbre, or tonal color, thus expanding the palette of sounds you can create.

There are many ways to reshape a waveform, the most common of which is changing the pulse width of a square wave. Other ways include changing the phase angle, moving the start point of a waveform cycle, or combining multiple waveforms in multioscillator synthesizers.

When waveforms are reshaped in these and other ways, the relationships between the fundamental tone and other harmonics change, thus altering the frequency spectrum and the basic sound being produced.

Filters

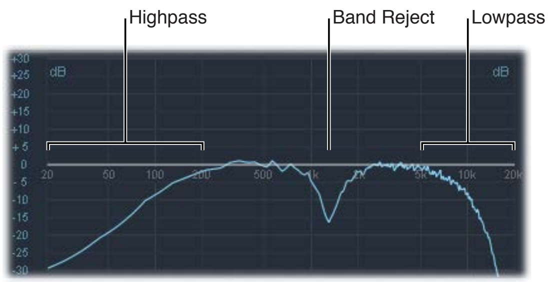
Filters overview

The purpose of the filter in a subtractive synthesizer is to remove portions of the signal—the frequency spectrum—sent from the oscillators. After being filtered, a brilliant-sounding sawtooth wave can become a smooth, warm sound without sharp treble.

The filter sections of most subtractive synthesizers contain two primary controls known as *cutoff frequency*—often abbreviated to *cutoff*—and *resonance*. Other common filter parameters are *drive* and *slope*. The filter section of most synthesizers can be modulated by envelopes, LFOs, the keyboard, or other controls such as the modulation wheel.

Filter types

There are several filter types. Each has a different effect on various portions of the frequency spectrum:

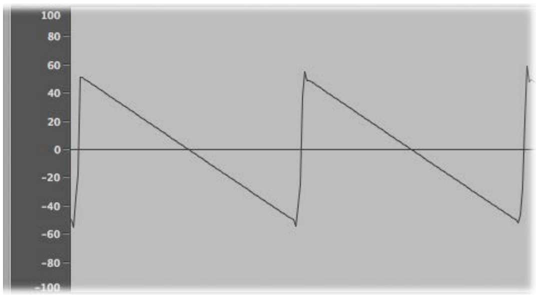


- *Highpass filter:* High frequencies are passed; low frequencies are attenuated.
- *Lowpass filter:* Low frequencies are passed; high frequencies are attenuated.
- *Bandpass filter:* Only frequencies within a frequency band are passed.
- *Band reject filter:* Only frequencies within a frequency band are attenuated.
- *Allpass filter:* All frequencies in the spectrum are passed, but the phase of the output is modified.

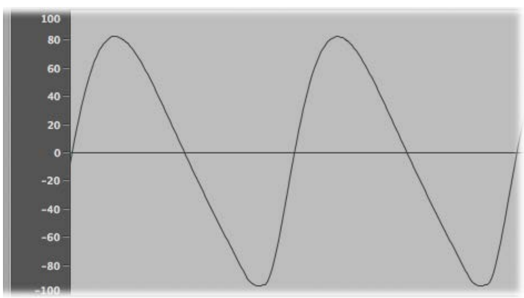
Cutoff frequency

The cutoff frequency, or *cutoff*, determines where the signal is cut off. Simpler synthesizers have only lowpass filters. If a signal contains frequencies that range from 20 to 4000 Hz and the cutoff frequency is set at 2500 Hz, frequencies above 2500 Hz are filtered. The lowpass filter allows frequencies below the cutoff point of 2500 Hz to pass through unaffected.

The figure below shows a sawtooth wave. The filter is open, with cutoff set to its maximum value. In other words, this waveform is unfiltered.



The figure below shows a sawtooth wave with the filter cutoff near a 50% value. This filter setting results in suppression of the higher frequencies and a rounding of the edges of the sawtooth waveform, making it resemble a sine wave. This setting makes the sound softer and less "brassy."

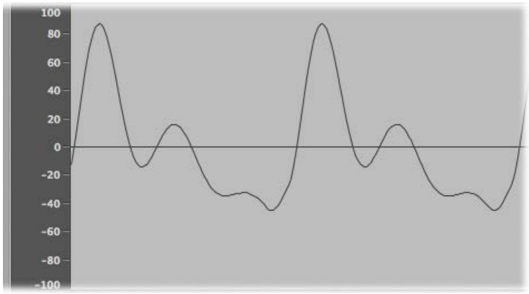


This example illustrates how using a filter to cut away portions of the frequency spectrum alters the waveform shape, thus changing the timbre of the sound.

Resonance

The resonance control emphasizes or suppresses signals around the cutoff frequency.

The figure below shows an ES1 sawtooth wave with a high resonance setting and the cutoff frequency set to 660 Hz.



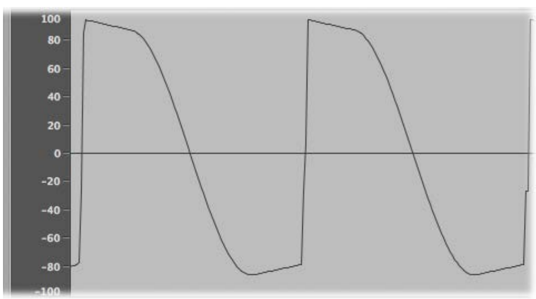
This resonant filter setting results in much brighter and harsher signals close to the *cutoff frequency*. Frequencies below the cutoff point are not affected.

The result of using filter resonance is a change in the basic waveform shape and, therefore, the timbre of the sound.

Very high filter resonance settings can cause the filter to self-oscillate, resulting in the filter generating an audible sine wave.

Filter drive

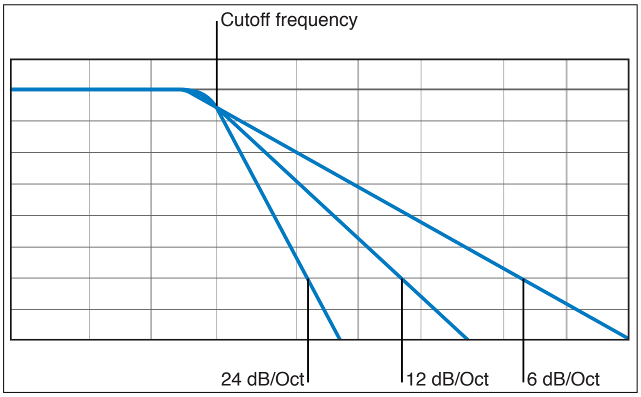
Filter drive adds an amount of gain to the waveform as it enters the filter—an input gain control—overdriving the filter and distorting the waveform. This waveform distortion changes the timbre of the sound, making it much harsher.



The figure shows an unfiltered sawtooth wave, with drive set to a value of 80%. Note the wave cycles touching the floor and ceiling of the filter's dynamic range.

Filter slope

A filter will cut off the signal at the cutoff frequency you set. This cutoff doesn't happen abruptly but rather at a given slope, which is measured in decibels (dB) of gain reduction per octave. You can define how steep the "cliff" is at the cutoff point by choosing a severe or gentle slope.



Envelopes in the amplifier

Amplifier envelope overview

The amplifier module of a synthesizer is responsible for controlling the level, or loudness, of the signal over time.

Consider the sound of a violin, for example. The sound slowly ramps up to a peak, or maximum, level as the bow is dragged across a string, then it is sustained for a period until the bow is moved away from the string, at which point it cuts off abruptly.

In contrast to the violin example, hitting a snare drum with a drumstick results in a very fast peak level with no sustain portion, then the sound immediately dies out—although there will be some *decay*, the time it takes to fall from the peak level.

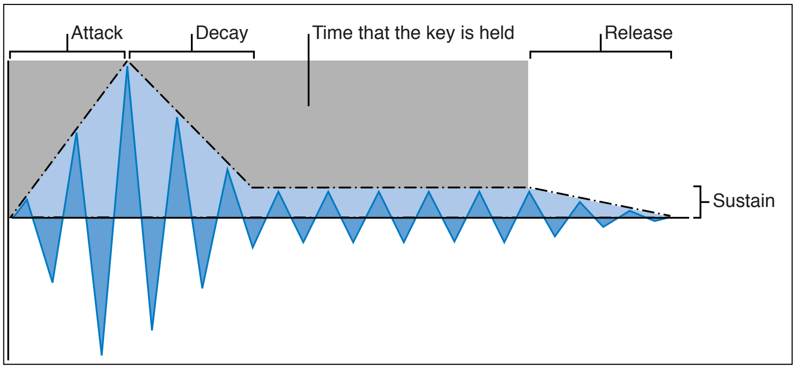
These two sounds clearly have different characteristics over time.

Synthesizers emulate these sonic characteristics by providing control over different parts—the beginning, middle, and end—of a sound's level over time. This control is achieved using a component called an *envelope generator*.

Note: Envelope generators are not limited to controlling signal amplitude. They can also control the rise and fall of the filter cutoff frequency or they can modulate other parameters. In short, envelope generators can be used as a modulation source—or as a "remote control" for a given parameter. See Modulation overview.

Attack, decay, sustain, and release

The oscillogram of a percussive tone shown below illustrates the level rising immediately to the top of its range and then decaying. If you drew a box around the upper half of the oscillogram, you could consider it the "envelope" of the sound—an image of the level as a function of time. The role of the envelope generator is to set the shape of this envelope.



The envelope generator usually features four controls—Attack, Decay, Sustain, and Release, commonly abbreviated as *ADSR*.

Envelope controls

- *Attack:* Sets the time it takes for the signal to rise from an amplitude of 0 to 100% (full amplitude).
- *Decay:* Sets the time it takes for the signal to fall from 100% amplitude to the designated sustain level.
- *Sustain:* Sets the steady amplitude level produced when a key is held down.
- *Release:* Sets the time it takes for the sound to decay from the sustain level to an amplitude of 0 when the key is released.

Note: If a key is released during the attack or decay stage, the sustain phase is usually skipped. A sustain level of 0 produces a piano-like—or percussive—envelope, with no continuous steady level, even when a key is held.

Modulation

Modulation overview

Without modulation, sounds tend to be uninteresting and fatiguing to the ear. They also sound synthetic, rather than natural, in the absence of some type of sonic modulation. Vibrato is a type of modulation commonly used by orchestral string players to add animation to their instrument's pitch.

To make sounds less static, you can use a range of synthesizer controls to modulate basic sound parameters. To this end, many synthesizers, including ES1, ES2, and EXS24 mkII sampler, provide a modulation router.

The router enables you to direct, or *route*, one or more modulation *sources* (the parameter or control that is modulating another parameter) to one or more modulation *targets* (the parameter being modulated).

You can affect modulation targets, such as oscillator pitch or filter cutoff frequency, by using modulation sources that include the following:

- *Velocity modulation:* You can modulate a target in different ways with the impact of your keyboard playing (harder or softer). The most common example of modulation controller use is a velocity-sensitive keyboard, set to control the filter and level envelopes. The harder you strike the notes, the louder *and* brighter the sound is.
- *Key scaling:* You can modulate a target in different ways by adjusting the position you play on the keyboard (low or high notes).
- *Controls:* You can use controls such as the modulation wheel, ribbon controllers, or pedals attached to your keyboard.
- *Automatic modulation:* You can use envelope generators or LFOs to modulate signals automatically.

Modulation sources can be—and often are—triggered by something you've done, such as playing a note on the keyboard, or moving the modulation wheel.

The modulation wheel, pitch bend ribbons, foot pedals, keyboard, and other input options are referred to as *modulation controllers, MIDI controllers,* or just *controllers*.

Modulation routing in ES1 and ES2

ES1 and ES2 provide an easy way to route a control—a modulation source—to part of the sound engine—a modulation target.

ES1 modulation routing

You create an ES1 modulation routing by selecting a modulation target in the left or right column of buttons in the Router section.

- You use the left column to set a modulation target that can be controlled, in amount, with the modulation wheel of your keyboard.
- The target you select in the right column will dynamically respond to keyboard velocity.
- The amount, or range, of this modulation is determined by the two arrows shown in the sliders, Int via Whl and Int via Vel. The upper arrow determines the maximum amount of modulation, and the lower arrow determines the minimum amount of modulation.

ES2 modulation routing

ES2 provides ten modulation routings, in columns. Each routing column is quite similar to the modulation controls found in ES1.

In the first routing column shown above:

- The modulation target is Pitch123. The pitch—the Frequency parameter—of Oscillators 1, 2, and 3 is affected by LFO2, the modulation source.
- LFO2 is the modulation source. The two arrows to the right of the column indicate the modulation amount. To make the modulation more intense, drag the upper and/or lower arrows up or down, thereby increasing the range of the modulation amount. The upper arrow determines the maximum amount of modulation, and the lower arrow determines the minimum amount of modulation.
- The via control is the ModWhl. Its range is determined by the sliders to the right of the channel. The amount of modulation is directly controlled with the modulation wheel of your keyboard. When the modulation wheel is at the minimum setting, at the bottom of its travel, the amount of oscillator pitch modulation is minimal, or off (no modulation). As you move the modulation wheel upward, the frequency of all three oscillators is directly controlled by the LFO (within the range determined by the sliders).

Common modulation sources

The main envelope generator of the synthesizer not only controls levels over time, but it also is often used to modulate other sound parameters when you press or release keyboard keys. Many synthesizers, such as ES2, feature multiple envelope generators.

The most common use of envelope modulation is to control the filter cutoff and resonance parameters with the keyboard velocity or keyboard scaling modulation sources (see Modulation overview on page 487).

A modulation source found on nearly all synthesizers is the LFO (low frequency oscillator). This oscillator is used only as a modulation source and does not generate any audible signals that form part of your actual synthesizer sound, because it's too low to be heard. It can, however, affect the main signal by adding vibrato, filter sweeps, and so on.

Common LFO controls

- *Waveform:* Allows you to choose the type of waveform—triangle waves and square waves are common.
  - Triangle waves are useful for filter sweeps—slow changes to the filter cutoff frequency—or when simulating an ambulance siren—slow changes to the oscillator frequency.
  - The square waveform is useful for rapid switches between two different pitches, such as vibratos or octaving.
- *Frequency/Rate:* Determines the speed of the waveform cycles produced by the LFO. When it is set to low values, very slow ramps are produced, making it easy to create sounds such as ocean waves rolling in—when white noise is chosen as the waveform in the main oscillator.
- *Sync mode:* Allows you to choose between free running—a user-defined LFO rate—or synchronization with an external tempo source, such as a host application.
- *LFO Envelopes:* The LFO can also be controlled with an envelope generator in some synthesizers. For example, imagine a sustained string section sound where vibrato is introduced a second or two into the sustained portion of the sound. If this can happen automatically, it allows you to keep both hands on the keyboard. Some synthesizers include a simple LFO envelope generator for this purpose. Often, this envelope consists only of an attack parameter—some may also include decay or release options. These parameters perform in the same way as the amplitude envelope parameters (see Attack, decay, sustain, and release on page 486), but they are limited to control of LFO modulations.

Global controls

Global controls affect the overall output signal of your synthesizer.

Common global controls

- *Level:* Sets the overall loudness of your sound. This control is the master output volume control of your synthesizer.
- *Glide (portamento):* Sets the amount of time that it takes for one note pitch to slide up or down to another note pitch. This control is useful for emulating wind instruments that slide from note to note, rather than move directly to another clear and distinct pitch.
- *Bender/bend range:* Bends the pitch—the oscillator frequency—up or down. This control is generally hard-wired to a pitch bend wheel on a keyboard. As the name suggests, moving the wheel up or down from its centered position bends the pitch up or down. The Bender/Bend Range parameter usually has an upper and lower limit of one octave but is typically set to around three semitones up or down. This setting is ideal for emulating small (or extreme) pitch fluctuations that occur in some instruments—such as when moving between notes with a trumpet, or bending the strings during a guitar solo.
- *Voices:* Sets an upper limit to the number of notes that can be played at a given time. Producing notes simultaneously is known as the *polyphony*—literally, "many voices"—of the instrument. The Voices parameter sets an upper limit to the number of notes that can be produced simultaneously.
- *Unison:* Used to "stack" voices—with the unison voice being heard one octave above the frequency of the played note. Because two voices are being used when you play a note, unison has two effects—it makes the sound richer and fuller, and it halves the polyphony.
- *Trigger mode:* Determines how the polyphony of the instrument is handled when the number of notes played exceeds the number of available voices. Trigger mode also allows you to assign legato mode. Essentially, this control changes the way the synthesizer responds to your playing technique and is invaluable when you are emulating monophonic instruments, such as flutes, clarinets, and trumpets. When you use the trigger mode control and assign a last note priority, a playing note will be cut off by playing another note.
  - *Last note priority:* When new notes are triggered while all voices are playing, the synthesizer frees up polyphony (voices) by ending the notes played earliest. This is the default trigger mode of Logic Pro synthesizers when in a monophonic mode.
  - *First note priority:* Notes played earlier are not stopped. In this mode you need to stop playing notes in order to play a new one after you have reached the limit of the polyphony (voices) of the instrument.

Note: The Trigger Mode parameter can also allow you to set priorities for lower- or higher-pitched notes when playing monophonically (one voice at a time) in some synthesizer designs.

There are many other global controls found on different synthesizer models that have an impact on your overall sound.

Other synthesis methods

Other synthesis methods overview

There are many ways to create sounds, using different technologies and approaches to synthesis. This section covers all the main methods, with reference to Logic Pro instruments where applicable.

Many of the methods described incorporate at least some elements of the subtractive synthesis design. See How subtractive synthesizers work. The most common modern approach is based on samples of real instruments and sounds.

Sample-based synthesis

Sample-based synthesis, which is sometimes known as *Pulse Code Modulation* (PCM), or *sampling and synthesis* (S&S) synthesis, is differentiated from subtractive synthesis mainly by the use of samples in place of oscillator waveforms.

The samples—digital recordings of existing sounds—are mapped across the keyboard. Typically, each sample is mapped to a note in the center of a keyboard range that spans a few notes that are unique to that sample. The reason for this limited range of notes is that samples tend to sound much less like the source sound if played more than a few notes higher or lower than the original pitch—due to the relationship between the pitch and playback speed of samples.

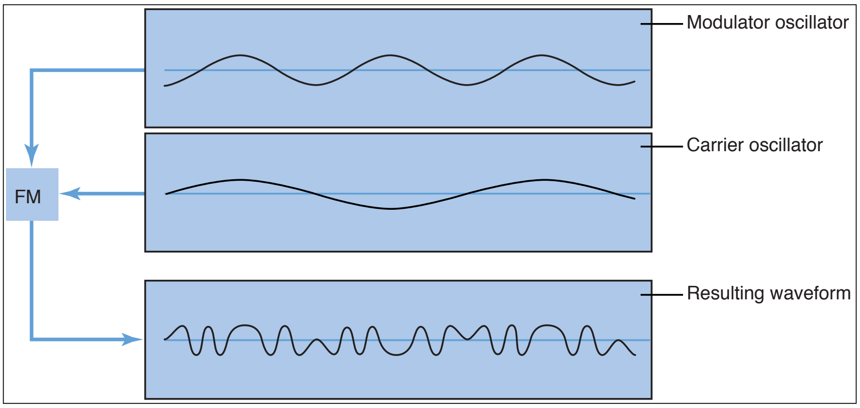
The pitch of each sample isn't changed with a frequency control, unlike the oscillator waveform of a synthesizer. Rather, a sample is played back at a faster or slower speed to alter its pitch, which has a corresponding impact on the sample playback time. For example, a sample played back at twice the speed requires half the time to play through.

EXS24 mkII is a sample player that can be used much like a sample-based synthesizer, due to the subtractive synthesis features that it offers.

Popular instruments that use this synthesis approach include Korg's M1, O1/W, and Triton; the Roland JV/XP instruments; Yamaha's Motif series; and many others.

Frequency modulation (FM) synthesis

FM synthesis uses a *modulator* oscillator and a sine wave *carrier* oscillator. The modulator oscillator modulates the frequency of the carrier oscillator within the audio range, thus producing new harmonics. These harmonics are known as *sidebands*.



Typically, FM synthesizers don't incorporate a filter. You can generate some subtractive synthesizer style sounds with FM synthesis, but it is difficult to recreate the sound of a resonant subtractive synthesizer filter using this method. FM synthesis is extremely good, however, at creating sounds that are difficult to achieve with subtractive synthesizers—sounds such as bell timbres, metallic tones, and the tine tones of electric pianos. Another strength of FM synthesis is punchy bass and synthetic brass sounds.

The EFM1 and Retro Synth FM synthesizers can produce many of the classic FM sounds made famous by Yamaha's DX series of synthesizers. The DX7, sold from 1983 to 1986, remains the most commercially successful professional-level hardware synthesizer ever made.

ES2 also features some FM techniques that allow you to modulate one oscillator with another. You can use these FM techniques to partially bridge the gap between the digital sound of FM synthesis and the fat analog sound that ES2 is noted for.

Component modeling synthesis

Also known as *physical modeling,* this synthesis method uses mathematical models to simulate instruments. Parameters are used to describe an instrument's physical characteristics, such as the materials the instrument is made of, the dimensions of the instrument, and the environment it is played in—under water, or in the air, for example. Equally important are descriptions of how the player would interact with the instrument—whether it is played by plucking, bowing, or strumming strings; by hitting it with sticks; by placing fingers on sound holes; and so on.

To model a drum sound, for example, the following aspects need to be taken into account. Of primary importance is the actual drum strike—how hard it is and whether the drumhead is struck with a wooden stick, a mallet, a beater, and so on. The properties of the drumhead (the skin or membrane) include the kind of material, its degree of stiffness, its density, its diameter, and the way it is attached to the shell of the drum. The volume of the drum cylinder itself, its material, and the resonance characteristics of all of the above need to be mathematically described.

To model a violin, you need to take into account the bow against the string, the bow width and material, the bow tension, the string material, the string density, the string tension, the resonance and damping behavior of the strings, the transfer of string vibrations through the bridge (materials, size, and shape of the bridge), and the materials, size, and resonance characteristics of the violin body. Further considerations include the environment that your modeled violin is played in and the playing style—"hammering" or tapping with the bow as opposed to drawing it across the strings.

The Sculpture component modeling synthesizer can produce convincing recreations of acoustic (and electronic) instruments. It is also exceptionally good at creating atmospheric, constantly evolving pad sounds. Other instruments that include physical modeling components and techniques are Ultrabeat, Vintage B3, Vintage Clav, and Vintage Electric Piano.

Wavetable, Vector, and Linear Arithmetic synthesis

Wavetable synthesis uses a number of different single-cycle waveforms, laid out in what is known as a *wavetable*.

Playing a note on the keyboard triggers a predetermined sequence of waves. In general, this is not a stepped transition but rather a smooth blend from one waveform into another, resulting in a constantly evolving waveform. Multiple wavetables can also be used simultaneously—either played one after the other, or blended together—resulting in more harmonically complex waveforms.

A single wavetable can emulate filter cutoff with a series of bright, less bright, then dull-sounding waveforms played in sequence—which resembles a reduction of the filter cutoff frequency in a subtractive synthesizer.

Wavetable synthesis isn't well-suited for emulating acoustic instruments. It is noted for producing constantly evolving sounds; harsh and metallic, or bell-like sounds; punchy basses; and other digital tones.

Wavetable synthesis was championed by the PPG and Waldorf instruments. The ES2 and Retro Synth also include wavetable features.

Roland LA (Linear Arithmetic) synthesizers such as the D-50 work on a similar principle. In these synthesizers complex sampled attack phases are combined with simple sustain or decay phases to create a sound. In essence, this is a simple wavetable that consists of two samples.

Where LA and wavetable synthesizers differ is that the latter were designed to create new, original, digital sounds. LA synthesizer designers, in contrast, wanted to emulate real instruments using a minimum of memory. To achieve this goal, they combined samples of the attack phase—the crucial part of a sound—with appropriate decay and sustain phases.

Vector synthesis—used in the Sequential Circuits Prophet-VS and Korg's Wavestation—allows you to move through wavetables and sequences arranged on a two-dimensional grid (two different vectors, or less technically, on the X or Y axis). The main benefit of this approach is that the balance between samples and waves is achieved in real time by moving a joystick. You can also use the ES2 to perform vector synthesis by modulating the Oscillator Mix (Triangle) parameter with the Vector Envelope.

Additive synthesis

Additive synthesis overview

Additive synthesis could be considered the reverse approach to subtractive synthesis. See Sound basics overview, Tones, overtones, harmonics, and partials, and How subtractive synthesizers work.

To obtain an insight into the additive synthesis approach, consider the fact that all sounds are a sum of various sine tones and harmonics.

In additive synthesis, you start out with nothing and build a sound by combining multiple sine waves of differing levels and frequencies. As more sine waves are combined, they begin to generate additional harmonics. In most additive synthesizers, each set of sine waves is viewed and used much like an oscillator.

Depending on the sophistication of the additive synthesizer you are using, you will either have individual envelope control over each sine wave or you will be limited to envelope control over groups of sine waves—one envelope per sound and its harmonics, or all odd or all even harmonics, for example.

Logic Pro doesn't provide a true additive synthesizer, but aspects of the additive synthesis approach are used in Vintage B3 and all other drawbar organs. In Vintage B3 you start with a basic tone and add harmonics to it, to build up a richer sound. The level relationships between the fundamental tone and each harmonic are determined by how far you pull each drawbar out. As Vintage B3 doesn't provide envelope control over each harmonic, it is limited to organ emulations.

Resynthesis

You can analyze the frequency components of a recorded sound and then resynthesize—reconstruct—a representation of the sound using additive techniques. By calculating the frequency and amplitude of each harmonic in the overall frequency spectrum of the sound, an additive resynthesis system can generate a series of sine waves, with appropriate levels over time, for each harmonic.

After the sound has been resynthesized in this fashion, you can adjust the frequency and amplitude of any harmonic. Theoretically, you could even restructure a harmonic sound to make it inharmonic, for example.

Phase distortion synthesis

Phase distortion synthesis creates different waveforms by modifying the phase angle of a sine wave.

In essence, you can bend a sine wave until it becomes a sawtooth wave, a triangle wave, a square wave, and so on. The synthesizer engine beyond the waveform generators typically follows a subtractive synthesizer design.

Phase distortion synthesis was commercially introduced in the 1984 Casio CZ series synthesizers.

Granular synthesis

The basic premise behind granular synthesis is that a sound can be broken down into tiny particles, or grains. These sampled grains—usually no more than 10 to 50 milliseconds long—can then be reorganized, or combined with grains from other sounds, to create new timbres.

In many respects, granular synthesis is similar to wavetable synthesis, but it works on a much finer scale. As you might expect, this method is ideal for creating constantly evolving sounds and truly unique tones.

The downside is that granular synthesis is extremely processor-intensive, and it wasn't possible to do in real time until relatively recently. For this reason, it has remained largely ignored by all but a few in academic institutions. Today's computers, however, have sufficient processing power to make this synthesis method a practicality, so there are a number of commercial products now available.

A brief synthesizer history

Precursors to the synthesizer

The earliest seeds of modern electronic synthesizers began in the twilight years of the 19th century. In 1897, an American inventor named Thaddeus Cahill was issued a patent to protect the principle behind an instrument known as the Telharmonium, or Dynamophone. Weighing in at 200 tons, this mammoth electronic instrument was driven by 12 steam-powered electromagnetic generators. It was played in real time using velocity-sensitive keys and, amazingly, was able to generate several different sounds simultaneously. The Telharmonium was presented to the public in a series of "concerts" held in 1906. Christened "Telharmony," this music was piped into the public telephone network, because no public address systems were available at the time.

In 1919, Russian inventor Leon Theremin took a markedly different approach. Named after the man who masterminded it, the monophonic Theremin was played without actually touching the instrument. It gauged the proximity of the player's hands as they were waved about in an electrostatic field between two antennae, and used this information to generate sound. This unorthodox technique made the Theremin enormously difficult to play. Its eerie, spine-tingling—but almost unvarying—timbre made it a favorite on countless horror movie soundtracks. R. A. Moog, whose synthesizers would later garner worldwide fame, began to build Theremins at the age of 19.

In Europe, Frenchman Maurice Martenot devised the monophonic Ondes Martenot in 1928. The sound generation method of this instrument was akin to that of the Theremin, but in its earliest incarnation it was played by pulling a wire back and forth.

In Berlin during the 1930s, Friedrich Trautwein and Oskar Sala worked on the Trautonium, an instrument that was played by pressing a steel wire onto a bar. Depending on the player's preference, it enabled either infinitely variable pitches—much like a fretless stringed instrument—or incremental pitches similar to that of a keyboard instrument. Sala continued to develop the instrument throughout his life, an effort culminating in the two-voice Mixturtrautonium in 1952. He scored numerous industrial films, as well as the entire soundtrack of Alfred Hitchcock's masterpiece *The Birds*, with this instrument. Although the movie does not feature a conventional musical soundtrack, all bird calls and the sound of beating wings heard in the movie were generated on the Mixturtrautonium.

In Canada, Hugh Le Caine began to develop his Electronic Sackbut in 1945. The design of this monophonic instrument resembled that of a synthesizer, but it featured an enormously expressive keyboard that responded not only to key velocity and pressure but also to lateral motion.

The instruments discussed thus far were all designed to be played in real time. Relatively early, however, people began to develop instruments that combined electronic sound generators and sequencers. The first instrument of this kind—named the Automatically Operating Musical Instrument of the Electric Oscillation Type—was presented by the French duo Edouard Coupleux and Joseph Givelet in 1929. This hybrid married electronic sound generation to a mechanically punched tape control. Its name was unofficially shortened to Coupleux-Givelet Synthesizer by its builders, the first time a musical instrument was called a "synthesizer."

The term was formally introduced in 1956 with the debut of the RCA Electronic Music Synthesizer Mark I, developed by American engineers Harry F. Olson and Herbert Belar. Its dual-voice sound generation system consisted of 12 tuning forks, which were stimulated electromagnetically. For its time, the instrument offered relatively sophisticated signal-processing options. The output signal of the sound generator could be monitored by loudspeakers and, amazingly, recorded directly onto two records. A single motor powered both turntables and the control unit of the Mark 1. The synthesizer was controlled by information punched onto a roll of paper tape, enabling continuous automation of pitch, volume, timbre, and envelopes. It was extremely complicated to use, it was unreliable, and spontaneous playing was impossible.

Early voltage-controlled synthesizers

With the exception of the Telharmonium, which was conceived prior to the advent of the thermionic valve, the precursors to the modern-day synthesizer were all based on tube circuitry. This made these instruments unwieldy and volatile. After the transistor became available in 1947/48, more rugged, smaller, and thus portable, instruments were soon to come.

At the end of 1963, American innovator R. A. (Bob) Moog met the composer Herbert Deutsch. Deutsch inspired Moog to combine a voltage-controlled oscillator and amplifier module with a keyboard, and in 1964 the first prototype of a voltage-controlled synthesizer was constructed. This collaboration with the German musician prompted Moog to extend his range of modules and to combine them into entire systems. It wasn't until 1967, however, that Moog actually called his diverse mix-and-match systems *synthesizers*.

Moog's achievements spread by word of mouth, and Moog, always keen to elicit the feedback of his customers, continued to add further modules to his line. Wendy Carlos's LP release *Switched-On Bach* (1968) was responsible for the breakthrough of Moog's instruments. The record featured Moog's modular synthesizers and was one of the earliest commercial multitrack recordings. The album's success introduced the synthesizer to a wider audience and made the name "Moog" synonymous with the instrument. Hoping to capitalize on the new sounds that synthesizers made available, and match Carlos's commercial success, numerous studios, producers, and musicians acquired Moog modular synthesizers. In 1969, as many as 42 employees produced two to three complete modular systems every week at Moog's production facility.

Working independently, an engineer named Donald Buchla had conceived and implemented the concept for a modular, voltage-controlled synthesizer. This coincided with Moog's version. Buchla also developed his first instruments in close cooperation with users. The inspiration for his first synthesizer originated with composers Morton Subotnik and Ramon Sender, of the San Francisco Tape Music Center. Although he began working on this instrument in 1963, it didn't make its public debut until 1966. By design, Buchla's instruments catered primarily to academia and avant-garde musicians, so they never garnered the public attention and acclaim of Moog's synthesizers.

The Minimoog

Moog and Buchla's voltage-controlled synthesizers were modular. One chassis, or several, housed the power supply and the actual modules. The inputs and outputs of the modules had to be interconnected via a confusing tangle of patch cords before the synthesizer would make a sound. Establishing these connections properly was an art unto itself, and obtaining useful settings on the modules required significant expertise.

Moog realized that these modular synthesizers were too complex and expensive for the average musician and were likely to fail if sold through traditional music retailers. In 1969, Moog collaborated with engineers Jim Scott, Bill Hemsath, and Chad Hunt to design a compact, portable, affordable, and easy-to-use synthesizer. After three prototypes were built, the Minimoog Model D was released in the summer of 1970.

In contrast to previous modular synthesizers, it was neither necessary nor possible for players to connect the modules of the Minimoog as they saw fit. All of the modules' connecting circuitry was hard-wired at the factory. The type and number of modules was also fixed. This simplified manufacturing considerably, and cut costs dramatically. A major marketing campaign saw the Minimoog become a huge success. Without alteration to its basic design, 13,000 Minimoogs were sold worldwide, right up to 1981.

Storage and polyphony

Customers weren't entirely satisfied with the Minimoog and contemporary synthesizers, however. Although musicians no longer had to contend with countless cords in order to play a synthesizer, they still had to deal with numerous knobs and switches before they could do something as simple as switch from one sound to another. Moreover, keyboardists were bored with playing monophonic melody lines on synthesizers—they wanted to play chords. Although dual-voice keyboards that connected two monophonic synthesizers were available as early as 1970, customers wanted more.

Attempting to satisfy these demands, two schools of thought emerged in synthesizer design. One approach called for an independent, monophonic synthesizer to be assigned to every key on the keyboard. To this end, designers married the design principles of electronic organs to synthesizer technology. Although this breed of instrument was fully polyphonic—all notes of the keyboard could be heard simultaneously—it wasn't as versatile in its control options as a true synthesizer. The first fully polyphonic synthesizer to feature this type of design was the Moog Polymoog, released in 1975. Developed primarily by David Luce, it featured 71 weighted, velocity-sensitive keys.

In the second approach to polyphonic sound generation, a synthesizer was assigned to a key only when the key was pressed—in effect, semi-polyphony. As early as 1973, American company E-MU Systems introduced the Modular Keyboard System Series 4050, a digital keyboard that could be connected to up to ten monophonic synthesizers, and thus had ten-voice polyphony. The problem with this approach was that very few people owned ten synthesizers, and the amount of time and effort involved in programming a new sound was an overwhelming deterrent. Digital memory was still waiting to be developed, and, once again, the evolution of semi-polyphonic synthesizers required the qualities that only digital keyboards could provide.

The same prerequisite—digital engineering—eventually led to synthesizers that allowed sounds to be stored. Without the benefit of digital technology, early attempts at storing sounds included some unusual solutions. For example, a synthesizer with analog programmability required a dedicated row featuring all of the instrument's control elements for every "memory" slot. In this case, a selector switch accessed one of the many identical control panels and connected it to the sound generator.

The first synthesizer featuring storage slots implemented in this manner was the 1975 Yamaha GX1. The control elements for the system's storage slots were so small that they could be adjusted only by using jeweler's screwdrivers and complicated tools—called *programmers* and *comparators*.

It was not until 1978 that the problem was resolved. The five-voice polyphonic Prophet-5, released by the American company Sequential Circuits, was the world's first synthesizer with a global storage feature. All settings for each of its five onboard monophonic synthesizers were stored in memory slots—40 in the debut model. Moreover, all five synthesizers shared a single user interface, which simplified matters considerably. In spite of its initially high price, this instrument proved extremely popular and approximately 8,000 were built up until 1985. In addition to its digitally implemented polyphony and memory, the success of the Prophet-5 is due to the quality of its analog sound generation system.

Digital synthesizers

Modern digital synthesizers featuring variable polyphony, memory, and completely digital sound generation systems follow a semi-polyphonic approach. The number of voices that these instruments are able to generate, however, no longer depends on the number of built-in monophonic synthesizers. Rather, polyphony depends entirely on the performance capability of the computers that power them.

The rapid developments in the digital world are best illustrated by the following example. The first program that emulated sound generation entirely by means of a computer was Music I, authored by the American programmer Max Mathew. Invented in 1957, it ran on a university mainframe, an exorbitantly expensive IBM 704. Its sole claim to fame was that it could compute a triangle wave, although doing it in real time was beyond its capabilities.

This lack of capacity for real-time performance is the reason why early digital technology was used solely for control and storage purposes in commercial synthesizers. Digital control circuitry debuted in 1971 in the form of the digital sequencer found in the Synthi 100 modular synthesizer—in all other respects an analog synthesizer—from the English company EMS. Priced out of reach of all but the wealthiest musicians, the Synthi 100 sequencer featured a total of 256 events.

Ever-increasing processor performance made it possible to integrate digital technology into parts of the sound generation engine itself. The monophonic Harmonic Synthesizer, manufactured by Rocky Mountain Instruments (RMI), was the first instrument to do so. This synthesizer had two digital oscillators, combined with analog filters and amplifier circuits.

The Synclavier, introduced in 1976 by New England Digital Corporation (NED), was the first synthesizer with completely digital sound generation. Instruments like the Synclavier were based on specialized processors that had to be developed by the manufacturers themselves. This development cost made the Synclavier an investment that few could afford.

An alternative solution was the use of general-purpose processors made by third-party computer processor manufacturers. These processors, especially designed for multiplication and accumulation operations—common in audio processing tasks—are called *digital signal processors* (DSPs). Peavey's DPM-3, released in 1990, was the first commercially available synthesizer completely based on standard DSPs. The instrument was 16-note polyphonic and based mainly on three Motorola 56001 DSPs. It featured an integrated sequencer and sample-based subtractive synthesis, with factory presets and user-definable samples.

Another solution was to design synthesizers as a computer peripheral, rather than as a standalone unit. The growing popularity of personal computers from the early 1980s made this option commercially viable. Passport Soundchaser and the Syntauri alphaSyntauri were the first examples of this concept. Both systems consisted of a processor card with a standard musical keyboard attached to it. The processor card was inserted into an Apple II computer. The synthesizers were programmed via the Apple keyboard and monitor. They were polyphonic and had programmable waveforms, envelopes, and sequencers. Today's sound cards, introduced in countless numbers since 1989, follow this concept.

Exploiting the ever-increasing processing power of today's computers, the next evolutionary step for the synthesizer was the software synthesizer, which runs as an application on a host computer.

The sound card (or built-in audio hardware) is needed these days only for audio input and output. The actual process of sound generation, effects processing, recording, and sequencing is performed by your computer's CPU—using the Logic Pro software and instrument collection.