Logic Pro Instruments

for Mac



Contents

Instruments overview **9**

Instruments introduction 9

Add, remove, move, and copy plug-ins 10

Use multi-output instruments 14

Alchemy **16**

Alchemy overview 16

Alchemy user interface 17

Alchemy Name bar 20

Alchemy file locations 21

Alchemy Preset browser 22

Alchemy sound sources 27

Alchemy source editors 70

Alchemy main filters 96

Alchemy master voice section 101

Alchemy modulation section 103

Alchemy Perform section 125

Alchemy arpeggiator 133

Alchemy effects section 142

Alchemy Extended parameters 151

Alchemy tutorials 152

Drum Kit Designer **182**

Drum Kit Designer overview 182

Edit pane 183

Use Drum Kit Designer 184

Extended parameters 186

Drum Kit Designer mappings 187

Drum Machine Designer **189**

Drum Machine Designer overview 189

Drum Machine Designer interface 191
Kit controls 192
Pad controls 195

Drum Synth 202

Drum Synth overview 202
Drum Synth Kicks 203
Drum Synth Snares and Claps 204
Drum Synth Percussion 205
Drum Synth Hats and Cymbals 206

ES1 207

ES1 overview 207
ES1 oscillator parameters 208
Global parameters 210
ES1 filter parameters 211
Amplifier parameters 213
ES1 envelope parameters 213
ES1 modulation 215
MIDI controller assignments 218

ES2 220

ES2 overview 220
ES2 interface overview 221
ES2 sound sources 222
ES2 global parameters 231
ES2 filter parameters 234
ES2 amplifier parameters 241
ES2 modulation 242
ES2 integrated effects processor controls 267
ES2 macro controls and controller assignments 269
Use ES2 in Surround mode 271
Extended parameters 271
Create random ES2 sound variations 272
ES2 tutorials 274

EFM1 289

EFM1 overview 289
EFM1 modulator and carrier parameters 290
Modulation parameters 293

Global parameters 294
Output parameters 295
Create random sounds 296
Extended parameters 296
Assign MIDI controllers 297

ES E 298

ES E overview 298
Oscillator parameters 299
LFO parameters 299
Filter parameters 300
Envelope parameters 301
Output parameters 301
Extended parameters 302

ES M 303

ES M overview 303
Oscillator parameters 304
Filter and filter envelope controls 305
Level envelope and output controls 306
Extended parameters 306

ES P 307

ES P overview 307
Oscillator parameters 308
LFO parameters 309
Filter parameters 310
Envelope and level controls 311
Integrated effects processor 312
Extended parameters 312

EVOC 20 PolySynth 313

EVOC 20 PS and vocoding 313
EVOC 20 PS interface 315
Analysis controls 316
(U/V) detection parameters 318
EVOC 20 PS synthesis parameters 319
Formant filter controls 324
Modulation parameters 326
Output parameters 327
EVOC 20 PS performance tips 328

Vocoder history 330
EVOC 20 block diagram 332

Retro Synth 333

Retro Synth overview 333
Analog oscillator controls 334
Sync oscillator controls 335
Retro Synth Table oscillator controls 336
FM oscillator controls 338
Filter controls 340
Filter types 342
Amp and effect controls 343
Retro Synth modulation controls 344
Global and controller settings 348
Extended parameters 350

Sample Alchemy 351

Sample Alchemy overview 351
Interface overview 352
Add source material 353
Edit mode 354
Play modes 354
Source overview 355
Synthesis modes 358
Granular controls 359
Additive effects 360
Additive effect controls 360
Spectral effect 363
Spectral effect controls 364
Filter module 366
Lowpass, bandpass, and highpass filters 367
Comb PM filter 367
Downsampler filter 368
FM filter 368
Envelope generators 368
Mod Matrix 369
Modulation routing 370
Motion mode 371
Trim mode 372
More menu 373

Sampler 375

Sampler overview 375
Load and save instruments in Sampler 379
Sampler Synthesizer parameters 382
Sampler Modulation 389
Sampler Edit panes 403
Create instruments, zones, and groups 434
Edit Sampler zones and groups 450
Sampler articulation handling 459
Manage sampler instruments 460
Sampler memory management 463
Sampler extended parameters 463

Sculpture 464

Sculpture overview 464
Sculpture interface 466
Sculpture string parameters 467
Sculpture objects parameters 474
Sculpture pickups parameters 478
Global parameters 480
Amplitude envelope parameters 482
Use the Waveshaper 483
Filter parameters 484
Sculpture Body EQ parameters 485
Sculpture delay effect parameters 488
Output parameters 490
Use surround range and diversity 491
Sculpture modulation controls 491
Sculpture morph parameters 505
Define MIDI controllers 514
Extended parameters 515
Sculpture tutorials 516

Studio Bass 552

Studio Bass overview 552
Main view parameters 553
Details view parameters 555
Keyswitch mapping 557
String and position MIDI Out channel assignments 558

Studio Horns **560**
Studio Horns overview 560
Extended parameters 562
Keyswitch mapping 564
Section MIDI channel assignments 565

Studio Piano **566**
Studio Piano 566

Studio Strings **568**
Studio Strings overview 568
Extended parameters 570
Keyswitch mapping 572
Section MIDI channel assignments 573

Ultrabeat **574**
Ultrabeat overview 574
Ultrabeat interface 576
Ultrabeat Assignment section 577
Synthesizer section overview 585
Ultrabeat sound sources 586
Filter section controls 598
Distortion circuit controls 601
Ultrabeat Output section 602
Ultrabeat modulation 606
Ultrabeat step sequencer 614
Ultrabeat tutorials 628

Utilities **640**
Utility instruments 640
External Instrument 640
Klopfgeist 643

Vintage B3 Organ **644**
Vintage B3 overview 644
Vintage B3 Main window 645
Vintage B3 Rotor Cabinet window 653
Vintage B3 Options window 660
Vintage B3 Effects window 662
Vintage B3 Expert window 666

Use a MIDI controller with Vintage B3 671
B3 and Leslie information 678

Vintage Clav 682
Vintage Clav overview 682
Vintage Clav interface 683
Vintage Clav Main window 683
Vintage Clav Effects window 688
Vintage Clav Details window 692
Extended parameters 697
D6 Clavinet information 697

Vintage Electric Piano 699
Vintage Electric Piano overview 699
Vintage Electric Piano interface 700
Vintage Electric Piano Effects window 701
Vintage Electric Piano Details window 705
Extended parameters 707
Vintage Electric Piano emulations 708
MIDI controller assignments 710

Vintage Mellotron 711
Vintage Mellotron overview 711
Vintage Mellotron controls 712

Legacy instruments 714
Legacy instruments overview 714
Emulated instruments 714
Synthesizers 719

Synthesizer basics 726
Synthesizer basics overview 726
Sound basics 726
Synthesizer fundamentals 730
Subtractive synthesizers 731
Other synthesis methods 744
A brief synthesizer history 751

Copyright 756

Instruments overview

Instruments introduction for Logic Pro for Mac

Logic Pro provides an extensive suite of software instruments that can be used in your productions.

Sample-based instruments include Alchemy, Sampler, Sample Alchemy, and Quick Sampler. These allow you to create and play back recordings of acoustic instruments. Alchemy provides extensive sample manipulation and resynthesis options that can result in sounds that bear little resemblance to the source. Also included are the sample-based Studio Bass, Studio Piano, Studio Strings, and Studio Horns plug-ins. The latter two offer a variety of individual and grouped instruments with multiple articulations.

Specialized percussion synthesizers and instruments include Drum Synth, Ultrabeat, Drum Kit Designer, and Drum Machine Designer. Ultrabeat combines several percussion-oriented synthesis engines and a step sequencer, making it a powerful tool for the creation of new sounds and beats. Drum Machine Designer is a track instrument that directly integrates and interacts with the Logic Pro Mixer and Step Sequencer.

Four vintage instruments emulate the classic B3 organ, Rhodes, Wurlitzer, and Hohner electric pianos, the Hohner Clavinet, and the Mellotron. See Vintage B3, Vintage Electric Piano, Vintage Clav, and Vintage Mellotron.

When it comes to synthesizers, you have a huge range to choose from. These include simple plug-ins such as ES P, ES M, ES E, and EFM1. More advanced options include Retro Synth, ES1, and ES2, up to the extremely sophisticated Alchemy and Sculpture instruments.

Alchemy provides several synthesis and sample manipulation engines that you can use alone or with each other to produce a staggering variety of sounds. It features extensive modulation options, built-in arpeggiation, and sequencing.

Sculpture is a modeling synth that you can use to recreate physical instruments or to create instruments that don’t exist. It has a number of unique modulation facilities and can generate a huge array of sound types.

Rounding out the collection are the External Instrument and Klopfgeist Utility instruments, and the EVOC 20 PolySynth vocoder. This instrument tracks incoming MIDI notes and processes audio signals to create classic robotic voices and choirs that follow played note and chord pitches.

Add, remove, move, and copy plug-ins in Logic Pro for Mac

You can add plug-ins, replace a plug-in with a different one, and remove plug-ins from a channel strip. You can also move, copy, and bypass plug-ins. If you're unhappy with any of your plug-in adjustments, you can undo them.

Add an instrument plug-in

In a Logic Pro channel strip, do one of the following:

- Click the Instrument slot, then choose a plug-in from the pop-up menu.



- Option-click an Instrument slot.

  You can now choose legacy plug-ins from the pop-up menu.

Add a MIDI effect plug-in

In a Logic Pro channel strip, do one of the following:

- Click the MIDI Effect slot, then choose a plug-in from the pop-up menu.



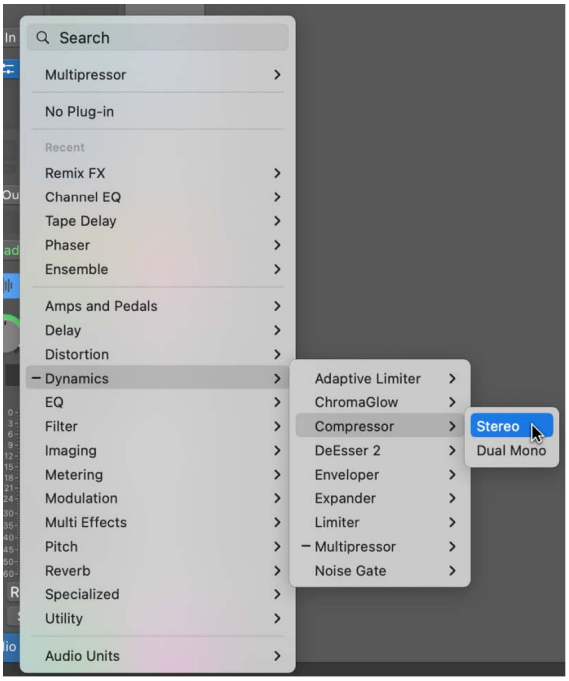
- Place the pointer above or below an occupied MIDI Effect slot, click the green line that appears, then choose a plug-in from the pop-up menu.



Add an audio effect plug-in to an instrument channel

In a Logic Pro channel strip, do one of the following:

- Click an Audio Effect slot, then choose a plug-in from the pop-up menu.



  The last visible empty Audio Effect slot in a channel strip is shown at half its height; use it in the same way.

- Option-click an Audio Effect slot.

  You can now choose legacy plug-ins from the pop-up menu.

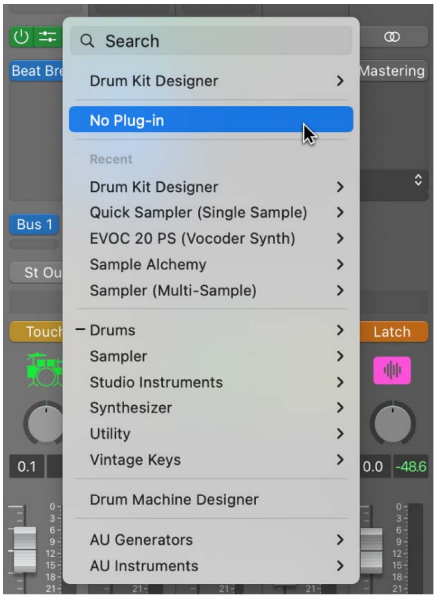
Replace a plug-in

- In a Logic Pro channel strip, place the pointer over the plug-in slot, click the arrows that appear to the right, then choose a plug-in to replace the existing one.



Remove a plug-in

- In a Logic Pro channel strip, place the pointer over the plug-in slot, click the arrows that appear to the right, then choose No Plug-in from the pop-up menu.



Move an effect or MIDI plug-in

- In a Logic Pro channel strip, drag the plug-in up or down in the channel strip, or to another channel strip.

  For guidance, use the colored line that appears when moving the plug-in.

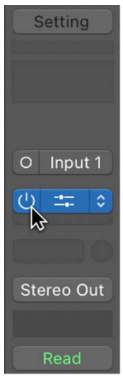
Copy an effect or MIDI plug-in

- In a Logic Pro channel strip, press and hold Option and drag the plug-in to an unused slot.

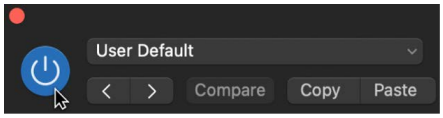
Bypass a plug-in

If you want to deactivate a plug-in, but don't want to remove it from the channel strip, you can bypass it. Bypassed plug-ins don't drain system resources.

- In Logic Pro, to bypass a single plug-in, do one of the following:
  - Place the pointer over the plug-in slot, then click the Bypass button that appears to the left.



  - Option-click the plug-in slot.
  - Click the center area of the plug-in slot to open the plug-in window, then click the Bypass button at the left side of the plug-in window header.



- To bypass multiple plug-ins, click a plug-in slot, then drag the pointer up or down.

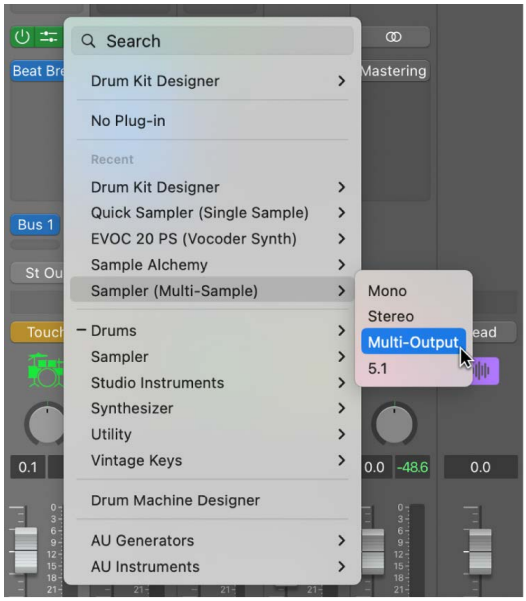
  All swiped plug-in slots switch to the same state.

Use multi-output instruments in Logic Pro for Mac

There are two reasons to use multi-output instruments in your mix:

- To process each sound individually in a drum kit, for example, with different effects
- To route the different software instrument outputs to different outputs on your audio interface

Logic Pro supports the multiple outputs of all instruments that provide a multi-output configuration (including Drum Kit Designer, Sampler, Studio Horns, Studio Strings, Ultrabeat, and all Audio Unit instruments). One or more multi-output options may be displayed in addition to the mono and stereo versions shown in the plug-in pop-up menu of the Instrument slot.



Note: Not all instruments offer multi-output options.

Insert and set up a multi-output instrument

1. In a Logic Pro channel strip, click the Instrument slot, then choose the multi-output instance from the pop-up menu.

   The first two outputs of a multi-output instrument are always played back as a stereo pair by the instrument channel strip that the plug-in is inserted into. Additional outputs (3 and 4, 5 and 6, and so on) are accessed via aux channel strips.

2. In the plug-in window, set up the output routing for individual sounds or samples. This is generally done using a pop-up menu with entries such as Main, 3-4, 5-6, and so on.

3. In the Mixer, click the Add button (+) of the instrument channel strip that you inserted the multi-output instrument into.



Note: The Add button (+) appears only on multi-output instrument channel strips.

An aux channel strip is created to the right of the multi-output instrument channel strip. Use this new channel strip to isolate and route a sound for independent mixing and processing.



4. Repeatedly click the Add button (+) to create more aux channel strips, for all stereo or mono outputs available to the instrument plug-in.

   You should create only as many aux channel strips as are required for the number of outputs used by the multi-output instrument.

   After you create the first aux channel strip for your multi-output instrument, a Delete button (–) appears beside the Add button (+).

   Click the Delete button (–) to remove aux channel strips.

Alchemy

Alchemy overview in Logic Pro for Mac

Alchemy is an easy-to-use, yet powerful sample manipulation synthesizer. It offers numerous real-time performance controls and an extensive preset library.

Alchemy features additive, spectral, and granular synthesis and resynthesis, sampling, and virtual analog engines. You can analyze imported samples and can manipulate them using one or more of these synthesis methods. Alchemy provides extensive sample mapping, looping, and grouping facilities that make it easy to create instruments containing hundreds of samples and layers. If you want to create purely synthetic sounds, the additive, spectral, and virtual analog synthesis engines are full-featured, matching or exceeding the power and facilities of many standalone instruments.

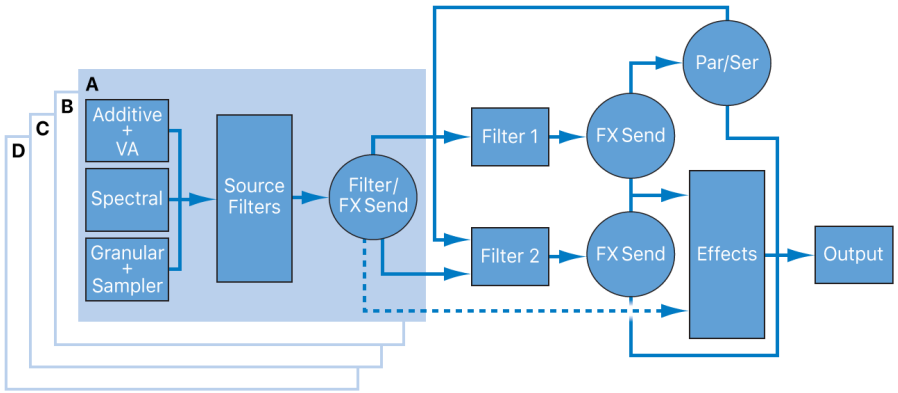
An Alchemy preset can contain up to four sources, each using one or more synthesis engines. You can morph or crossfade between these sources. Dozens of modeled analog and digital filters are available, in addition to multiple racks of integrated effect units and an extensive modulation section. Alchemy also features a powerful arpeggiator that can control each source independently and provides flexible pattern modulation options.

If you're new to synthesizers and different synthesis methods, see Synthesizer basics overview. Also see the Alchemy tutorial introduction for detailed tasks on the use of several advanced Alchemy synthesis features.

If you're new to using plug-ins in Logic Pro, see Add, remove, move, and copy plug-ins.

Before you can design new sounds with Alchemy, you need to understand how its different parts fit together, and how each of them contributes to what you hear.

At first glance Alchemy may seem complex, but its layout is relatively simple:



The diagram shows signal flow from left to right through the different sound generating and processing modules.

There are three basic stages:

- Each sound can contain up to four sources: A, B, C, and D. Each source consists of additive, spectral, granular, sampler, and virtual analog elements. Multiple elements can be active in each source. Each source has three independent filters that can operate in parallel or in series. These sources are used to create and shape the basic tone of the sound. All modulation in this section applies per voice. See Alchemy source overview, Source filter controls, and Source modulations.
- There are two main filters that can operate in parallel or in series. The main filters are used to shape or otherwise alter the combined sound from the four sources. All modulation in this section applies per voice. See Main filter controls.
- After the individual voices are mixed together and filtered, they pass through the effects stage. Any modulation of the Effects section is applied to the entire audio signal sent from the main filter section. You can also directly route sources (post source filters, if used) to the Effects section, bypassing the main filters altogether. See Alchemy effects overview.

Working with projects from Logic Pro for iPad

Logic Pro for Mac provides playback compatibility with Logic Pro for iPad projects that use Sample Alchemy. You can load presets for this plug-in in Logic Pro for Mac Alchemy, but can only edit parameters in Sample Alchemy in Logic Pro for iPad.

To continue working with Sample Alchemy settings in Logic Pro for Mac, replace the plug-in with Alchemy. See the Share a project with Logic Pro for iPad topic in the Logic Pro for Mac User Guide, and the Export and Share projects chapter in the Logic Pro for iPad User Guide.

Alchemy user interface

Alchemy interface overview in Logic Pro for Mac

View buttons: Alchemy has three main views. You access each view by clicking these buttons. Each view is designed for different tasks and situations.

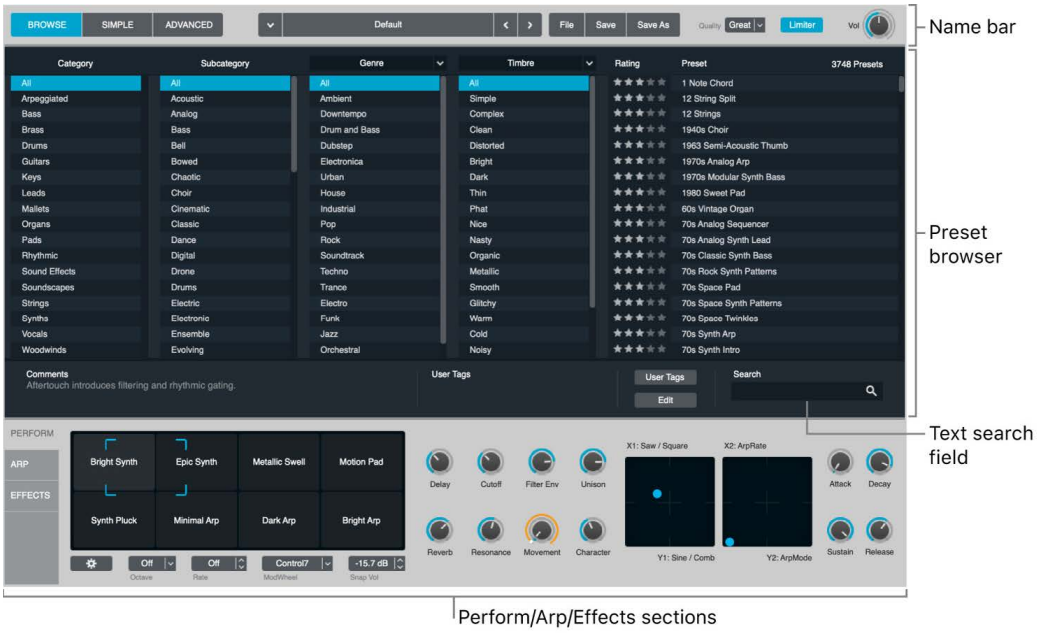


All views feature the Name bar at the top and the Perform section at the bottom. See Alchemy Name bar and Performance controls overview.

The Perform section of the interface is shared with the Arpeggiator and Effects panes. Click the Perform, Arp, or Effects buttons to view the parameters of each pane.

Tip: For users running in Controls view, all parameters are available. This includes menu commands and options such as those accessed from the File button at the top of the graphical interface, the synthesis import modes in the Import window, Modulation menus, the Source file menu, and Spectral effects menu.

- *Browse button:* Browse view is ideal for performance or exploration and for use of presets when composing. The Preset browser displays all available presets and offers a powerful search engine. The Perform section, located beneath the Preset browser, features a number of real-time controls that you can map to almost any Alchemy parameter.



  - *Preset browser:* See Preset browser overview.
  - *Text search field:* See Browser parameters.
  - *Arp(eggiator) section:* Click the Arp button to view and make changes in the arpeggiator control panel. See Alchemy arpeggiator overview.
  - *Effects section:* Click the Effects button to view and make changes in the effects control panel. See Alchemy effects overview.
- *Simple button:* When simple view is turned on, Alchemy provides a compact interface ideal for live performance.



- *Advanced button:* Advanced view is particularly useful for creating new sounds or manipulating existing presets. This view unlocks all Alchemy synthesis, modulation, and editing capabilities.



- *Source, main filters, master voice, and morph sections:* See Alchemy source overview, Main filter controls, Alchemy master voice section, and Morph controls.
- *Source editors:* When individual sources are selected, you can also access the source editors. See Alchemy source edit window overview.
- *Modulation section:* See Alchemy modulation overview.
- *Arp(eggiator) section:* Click the Arp button to view and make changes in the arpeggiator control panel. See Alchemy arpeggiator overview.
- *Effects section:* Click the Effects button to view and make changes in the effects control panel. See Alchemy effects overview.

Alchemy Name bar in Logic Pro for Mac

The Name bar provides tools for preset management and other functions. It is shown in all views. See Alchemy interface overview.



Name bar parameters

- *Preset field and pop-up menu:* Displays a list of presets available to load. By default, all presets are shown. To quickly scroll through and load a preset, use the Previous and Next buttons (the arrows) adjacent to the Preset field. The order of presets is determined by settings in the Preset browser. See Preset browser overview.

  The field shows the current preset name and category, or *Default* if no preset is loaded. The category of the selected preset is displayed to the left of the preset name when a preset is loaded.

- *Category button:* Show a pop-up menu of preset categories.
- *File button:* Open a pop-up menu containing preset management commands.
  - *Save Consolidated:* Save a copy of the active preset along with copies of audio files used.

    Note: Because supplied samples use audio files available to all Alchemy users, these files are not copied.

  - *Initialize Preset:* Initialize Alchemy, providing a basic sawtooth wave that can be used as a starting point for your own presets.

    The Master Volume is modulated by Velocity and ADHSR 1, and the Filter 1 Cutoff parameter is modulated by Ctrl C (74 Brightness), which is utilized by some Bass and Keyboard Session Player styles.

    Important: Use of Initialize Preset automatically turns on wide unison mode. See Alchemy wide unison mode.

  - *Refresh Library:* Scans all default file locations, and updates the preset library database.
- *Save button:* Save your changes to the active preset. If the preset belongs in the App Presets bank, a Save As dialog is displayed, preventing you from accidentally overwriting the preset.
- *Save As button:* Open a browser window where you can choose a filename (*.acp) and a location to save the currently active preset. Newly saved presets are automatically added to the Preset browser with a category, several predetermined attributes, and the artist name specified in Settings > My Info > Artist Name. If this preference is not set, “Unknown Artist” is shown in preset results.
- *Quality pop-up menu:* Choose the processing resolution on output. This setting remains active when loading other presets or when recalling the project.
- *Limiter button:* Turn the Limiter on or off. This prevents clipping of the signal from all voices at the end of the signal path. This setting remains active when loading other presets or when recalling the project.
- *Volume knob:* Increase or decrease the overall preset volume. This happens at the end of the signal path and controls the effected and dry signals of all voices.

Alchemy file locations in Logic Pro for Mac

You can find Alchemy files in the following locations.

Logic Pro Library

All the sound files that come with MainStage are stored inside a single file, the Logic Pro Library.bundle. It is located in the Music folder of the user directory, but you can move it to any location on your local or external drive. See Sound Library settings.

Logic Pro Library.bundle is technically a folder. To open it in the Finder, Control-click the file and choose Show Package Contents. The Alchemy files are located in the following subfolders:

- *Alchemy presets (ACP):* /Plug-In Settings/Alchemy/
- *Sampler instruments (EXS):* /Plug-In Settings/Sampler/
- *Sample data (CAF, AIF, AAZ):* /Samples/Alchemy Samples/

User files

Files that you save in Alchemy are stored in the user directory ~/Music/Audio Music Apps/, in the following subfolders:

- *Alchemy presets (ACP):* /Plug-In Settings/Alchemy/
- *Sample data (AIF, CAF, AAZ):* /Alchemy Samples/
- *Sampler instruments (EXS):* /Plug-In Settings/Sampler/
- *Preset database file:* Ratings, comments, tags, and attributes are stored in the preset database file Alchemy_Preset_Ratings_And_Tags.db.

Additional locations

When you save an ACP (Alchemy preset) file, Logic Pro also automatically saves references to AAZ files in the application samples folder. If AAZ files are in any other location, they are saved in the same folder as the ACP file, using the ACP filename.

User samples may be located anywhere on disk, with AAZ data being saved in the patch or preset location. Logic Pro for Mac uses the default locations listed above when File > Project Settings > Assets > “Copy Alchemy audio files into project” is not active.

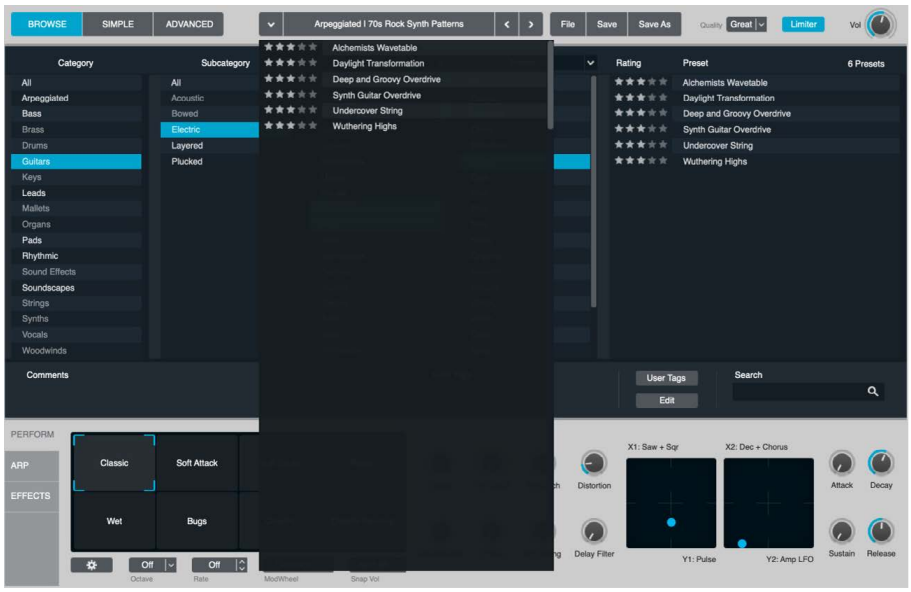
When you save a preset as part of a channel strip setting (CST file), patch, or project:

- The location of previously saved AAZ data (the Alchemy analyzed sample format) is retained.
- If AAZ data has not already been saved, it is saved in the ~/Music/Audio Music Apps/Alchemy Samples folder.

Alchemy Preset browser

Alchemy Preset browser overview in Logic Pro for Mac

The Preset browser is used for loading, searching, sorting, rating, and tagging presets.



There are two different views for the Preset browser.

- *Browser view:* Click the Browse button to open browser view.
- *Name bar list:* Click the preset field in the Name bar to display a list of presets.

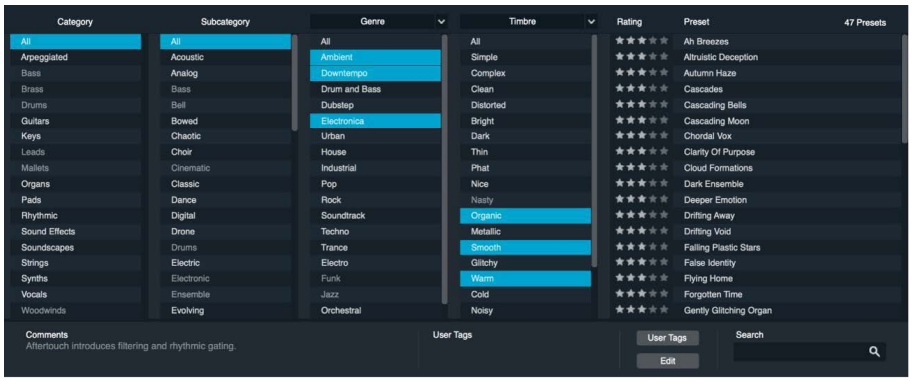
When a new instance of Alchemy is inserted, the browser shows all available presets. The Name bar list field displays the preset category and preset name.

Both lists are synchronized. If the Preset browser is used to select a particular category or to sort results, the Name bar list reflects your choices, and vice versa.

When a preset is loaded, the area below the columns displays Comments and User Tags fields. The area to the right shows additional parameters. See Browser parameters.

Use the Alchemy browser in Logic Pro for Mac

The browser displays all presets matching the current search criteria. A number of columns are shown from left to right.



Browser parameters

- *Category column:* Shows preset categories. Command-click multiple categories to refine your criteria. Click All to show all categories.
- *Subcategory column:* Shows subcategories of preset categories. Command-click multiple subcategories to refine your criteria. Click All to show all subcategories.
- *Attribute menu selection:* Choose a preset attribute type to refine the preset results list and to apply attributes to presets when in Edit mode.

  Tip: Combine Attributes such as Newer Than and Sound Designer to show presets that you created within a date range, without also showing supplied presets installed in the same date range.
- *Attributes columns:* Show the Articulation, Genre, Newer Than, Older Than, Sound Designer, Sound Library, Timbre, or User Tags attributes. Command-click multiple attributes to refine your criteria. Click All to show all matching attributes.
- *Rating column:* Shows user preset ratings. Click the column header to sort the results list. A second click reverses the current sort order. Click the stars to rate a preset.

  The highest-rated presets appear first in the list. User-rated presets always take precedence over unrated presets. Three dim stars indicate that a preset has not yet been rated.
- *Browser results list:* Shows preset names. Click the column header to sort the results list. A second click reverses the current sort order.

  The number of presets returned by the current search criteria is displayed to the right of the Preset column header.
- *Matching Presets:* Shows the number of presets that match the search criteria chosen in the Category, Subcategory, and Attributes columns, and the typed search term.
- *Comments field:* Shows comments stored with the selected preset. In preset edit mode, click to edit the comments.
- *User Tags field:* Shows user tags stored with the selected preset. In preset edit mode, click to edit the user tags.
- *User Tags button:* Open a pop-up menu with commands for creating and deleting user tags. A list of existing user tags is shown below the menu commands.

- *Edit button:* Switch to preset edit mode where you can edit user tags and can enter comments for the current preset.
- *Search field:* Use to type a search term for presets. This is restricted by the criteria chosen in the Category, Subcategory, and Attributes columns.
- *Clear Search field button:* Clear the search term you have entered in the search field.
- *Search button:* Perform a search for the term you have entered in the search field.
- *Close button:* Exit preset edit mode.

Set criteria to refine the preset results list

1. In Alchemy in Logic Pro, click an entry such as Guitars in the Category column to limit the preset results list to include only presets belonging to the Guitars category.
2. Click an entry such as Acoustic in the Subcategory column to further limit the preset results list to include only presets belonging to the Acoustic subcategory of the parent category (Guitars, in this example).
3. Click the header of one or more attribute columns to open a pop-up menu where you can choose the attributes you want to use.
4. Click a keyword in each attribute column.
5. You can choose multiple entries in any column by doing one or more of the following:
    - Click an entry, then hold down Shift while clicking another entry in the same column. All items between the clicked entries are selected and used as search criteria for the preset results list.
    - Hold down Command, then click any entries you want to use as search criteria.
    - Click the All entry at the top of a column to reset the selection criteria.

Use the search field

The search field is used for manual searching of presets by name, attributes, user comments, artist name, or tags. The most recent search term is saved and recalled with the project.

1. In Alchemy in Logic Pro, type your search term, then click the magnifying glass icon or press Enter.

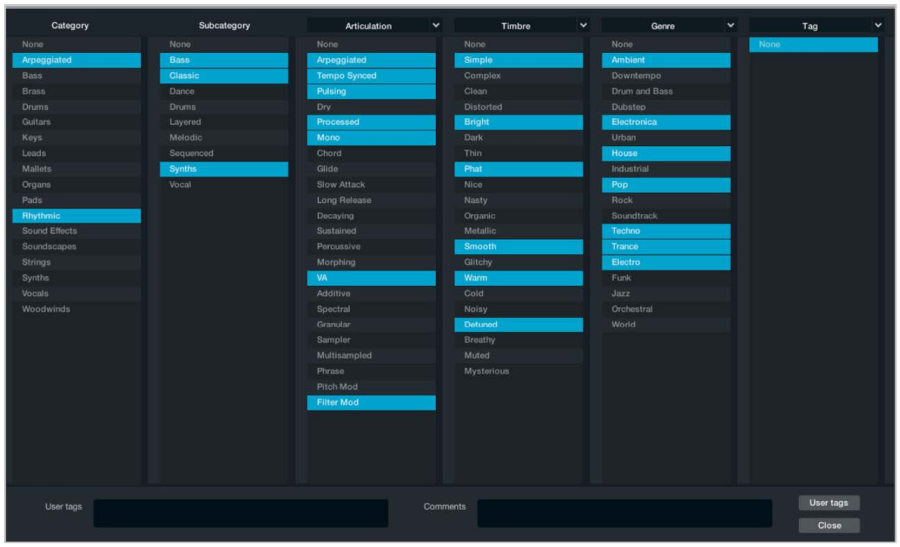
    The Preset browser results list updates to reflect your search criteria.
2. You can refine a text search with the minus symbol. For example, to search for all presets matching analog but not bass, use the search term *analog -bass*.

    This may also be used to remove all bass presets from the results list by entering only *-bass* in the search field. Presets in the Bass category and presets with bass or basses in the preset name, user tags, or comments field are removed from the preset results list.
3. To clear the current search term, click the Clear Search field button.

Use preset edit mode

Preset edit mode displays six columns in an enlarged Preset browser window. Edits made to multiple selected presets apply attributes, user tags, or comments to all presets in the group. All changes made in edit mode are immediately applied and saved in the preset database. The Sound Designer is determined by the settings in Logic Pro > Settings > My Info > Artist Name. If this is not set, “Unknown Artist” is shown.

Note: It is not possible to change the Sound Library for a preset; this column is displayed for reference only.



1. In Alchemy in Logic Pro, click the Edit button to switch to preset edit mode.
2. Choose a preset from the Preset pop-up menu in the Name bar if a preset is not already loaded. You can also use the Previous and Next buttons in the Name bar to select a preset.

   To select multiple presets, do one of the following:

   - Click a preset name from the Preset pop-up menu in the Name bar. Reopen the Preset pop-up menu in the Name bar, then hold down Shift while clicking another preset name. All items between the clicked entries are selected. Edits performed in the Preset Edit window apply to all selected presets.
3. Type in the User Tags or Comments fields, as required.

   Terms entered in these fields can be searched using the text search function.
4. Change the Category, Subcategory, or Attributes for the preset.

   Changing the preset category updates only the preset category database. The preset itself is not moved to a new category folder on disk.
5. Click the Close button to exit preset edit mode.

Create, remove, and delete user tags

1. In Alchemy in Logic Pro, select the preset that requires user tag changes.
2. Click the User Tags button, then choose New Tag from the pop-up menu.

   The Enter New Tag field is shown.
3. Type in the Enter New Tag field, then click OK. Click Cancel if you don't want to switch to a new tag or want to close the Enter New Tag field.

   The Enter New Tag field closes, and the user tag is added to the User Tags field. It is also added to the New Tags pop-up menu with a checkmark, indicating that it is assigned to the selected preset.
4. To remove an assigned tag, click the User Tags button, then choose the tag name from the pop-up menu. Assigned tags are indicated by a checkmark beside the tag name.

   The tag is removed from the preset, but the tag is not deleted. No checkmark is shown beside the tag name. This tag can be reassigned to another preset.
5. To delete a user tag, click the User Tags button, then choose the tag from the Delete Tag From All Presets submenu.

   The tag is immediately removed from all presets.

Rate a preset

1. In Alchemy in Logic Pro, click a preset name to select it.
2. Click a star in the Ratings column to set the rating between 1 and 5. Unrated presets display 3 stars.

   Orange stars indicate a user ratings. Presets without a user rating display a default rating of 3 white stars.

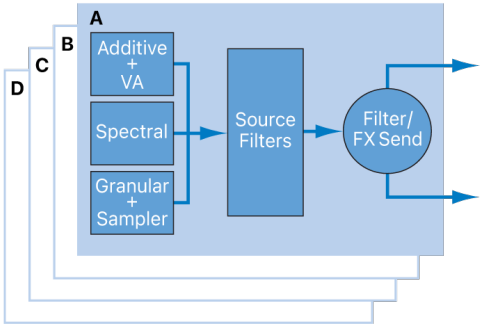
   Note: You can limit the preset results list to match any star rating by entering the number of stars in the text search field. A search for **** displays only 4-star rated presets.
3. Drag left on orange stars to remove a custom rating and set it back to 3 white stars.

Alchemy sound sources

Alchemy source overview in Logic Pro for Mac

Source components are shown only in advanced view. Click the Advanced button to switch to advanced view.

Alchemy sound generating modules are called *sources*. Four independent sources are available in each preset (A, B, C, and D). Each source has an identical set of components.



Each source provides multiple sound generating elements based on different methods of synthesis. See Source elements overview.

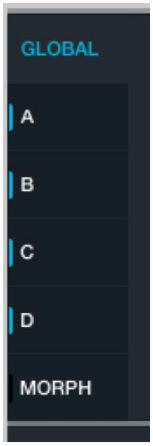
Elements can be turned on or off independently within each source, allowing you to combine synthesis methods. Some elements require a sample import or the creation or import of other content before they can be used.

There are a handful of restrictions on element combinations within a single source. You can circumvent such restrictions when creating your sound by using a different synthesis method (or combination of methods) for each of the four sources.

If you're new to synthesizers and different synthesis methods, see Synthesizer basics overview.

Alchemy source master controls in Logic Pro for Mac

Source components are shown only in advanced view. Click the Advanced button to switch to advanced view.



At the left of the source section are six buttons: Global, A, B, C, D, and Morph. Indicators to the left of each button denote the following: blue = active source, green = morphing turned on for the source.

Source mode buttons: Click Global to view all sources. Click A, B, C, or D to view a single source. Click Morph to use morph mode. Option-click A, B, C, or D to view and solo the source.

When the Global button is on, the source master controls are displayed. The main filter and master voice controls are also displayed when Global is active. See Main filter controls and Alchemy master voice section.

When the A, B, C, or D button is on, the subpage controls for the selected source are displayed. See Source subpage controls.

When the Morph button is on, source morphing parameters are displayed. See Morph controls.

The source select field in each source does double duty as a level meter that shows the current output level from the source into the main filter section. When the output level exceeds 0 dB, it clips the input of the main filter section, and the level meter turns red. This is useful for quickly determining the cause of unwanted distortion.

Note: Sources can also be routed directly to the Effects section, bypassing the main filters. The level meter is shown regardless of the source output destination.



Source master controls

- *A/B/C/D buttons:* Turn the source on or off. Option-click to solo a source.
- Click the Previous and Next buttons (the arrows) to step through available waveform data. This field also acts as a level meter during playback.
  - *Import Audio:* Import audio data into the source. You can also drag and drop audio from Logic Pro windows (Main window, Loop Browser, File Browser), you can use the Import browser, or you can drag and drop audio files from the Finder.
  - *Load VA:* Load waveform data into the VA element (Virtual Analog synthesis engine) from categorized waveform type submenus. This automatically turns on the VA engine and turns off all other synthesis engines for this source. See VA element controls.
  - *Load Source:* Open a dialog where you can load source (SRC) data from ~/Music/Audio Music Apps/Plug-Ins Settings/Alchemy/Libraries/Sources/.

    SRC file format data includes all source control settings and a reference to any loaded or imported audio data. Source control modulations are not loaded as part of the SRC file.

- *Save Source:* Open a dialog where you can write source data to disk in SRC file format in the ~/Music/Audio Music Apps/Plug-Ins Settings/Alchemy/Libraries/Sources/ folder.

  Saved data includes all source control settings and a reference to any loaded or imported audio data. Source control modulations are not saved as part of the SRC file.

- *Copy Source:* Copy source data to the Clipboard. This is useful for duplicating the content and settings of one source into another. Copied data includes settings of all source controls and a reference to loaded or imported audio data. Source control modulations are also copied.
- *Paste Source:* Paste source data from the Clipboard.
- *Clear Source:* Reset the source to the default state.
- *Randomize Source:* Apply random variations to source controls, depending on active elements in the source.
- *Swap With Source:* Choose a source from the submenu to swap with the current source. This allows you to quickly experiment with the parameter settings of the two swapped sources.

- *Vol knob:* Set the source output level (-inf dB to 0 dB).
- *Tune knob:* Tune the source in semitone increments. A further Fine Tune control is available on each source subpage. See Source subpage controls.
- *Pan knob:* Set the source output position in the stereo field. This parameter works as a pan control if the source Stereo button (next to the Edit button) is off or as a left/right balance control if the source Stereo button is on. The behavior of the Pan control changes when wide unison mode is turned on. See Alchemy wide unison mode.
- *Send knob:* Set the source output balance (post source filters, if used) sent to the destinations specified with the Send pop-up menu.

Alchemy Import browser in Logic Pro for Mac

You can import WAV, AIFF, and Sampler sound files as additive, spectral, a combination of additive and spectral, granular, or sampler elements. See Source elements overview.

You can quickly replace the sound for Alchemy on a software instrument track by dragging an audio file, audio or MIDI region, or Apple Loop to the track header. When you drag content to one of the Alchemy zones to replace the existing sound, you can choose whether the new sound uses additive, granular, or spectral synthesis elements.

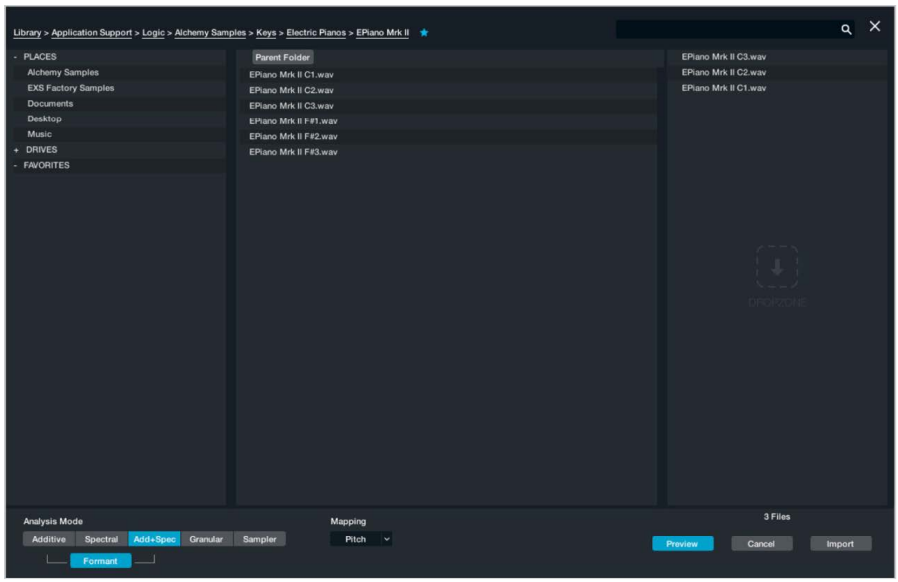
The additive element allows for the most detailed manipulation of sound and is especially good for sound files that represent single notes, rather than chords or more complex sounds and textures.

Importing to the spectral element allows effective manipulation of polyphonic sounds, such as chords, drum loops, and other complex sounds and textures.

An additive+spectral import may deliver the best results when an additive import fails to capture the noisy components of a sound, such as the hammer strike of a piano or the breath noise of a flute.

The granular element is good for drum loops, percussive sounds, and any sound that you want to use granular effects with.

Use the sampler element when you want to play a sound file in its original form, transposed across the keyboard.



Import browser parameters

- *Path navigator:* Shows path to current folder. Click underlined folder or drive names to navigate to any folder in the path hierarchy.
- *Favorite button:* Add the selected file (or files) to the Favorites category shown in the Locations column to the left.
- *Parent folder:* Click to navigate to the folder that contains the files shown in the Folders, samples, EXS instruments column.
- *Search field:* Type a search term to find matching items within the selected folder or location.
- *Search Clear button:* Clear the search term shown in the search field.
- *Search button:* Perform a search for the term entered in the search field.
- *Close button:* Exit the Import Browser window.
- *Locations column:* Choose a location to search only that part of your hard drive. Places, Drives, and Favorites categories are shown at the top level of the hierarchy. The Alchemy Samples and EXS Factory Samples folders are shown at the top of the Places category. Click to view each folder.
- *Folders, samples, EXS instruments column:* Shows the folders, samples, or EXS instruments within the selected folder, preset, or sampler instrument. See the tasks in this section for information on sample selection.

- *Dropzone:* Shows the names of samples added to this area. Samples can be from multiple locations or instruments. See the tasks in this section for information about use.

  Control-click files in the Dropzone list to open a shortcut menu that contains the following commands.

  - *Delete:* Remove the file from the Dropzone.
  - *Move to New RR Group:* Create a new round-robin group and add the selected file or files to the group. See Inspector group controls.
- *Analysis Mode buttons:* Set the type of analysis that is performed when sample data is imported.
  - *Additive button:* Use to import samples that you want to resynthesize using additive synthesis. Good results depend on accurate identification of the root note. If the filename has a pitch value appended to it, this is used to set the root note. In other cases, analysis of the waveform pitch determines the root note. The Mapping mode may also be important.
  - *Spectral button:* Use to import samples that you want to resynthesize using spectral synthesis. The root note determines the MIDI note that plays the resynthesized sound at its original pitch. If the filename has a pitch value appended to it, this is used to set the root note. In other cases, analysis of the waveform pitch is used to determine the root note.
  - *Add+Spec button:* Use to import samples that you want to resynthesize using a combination of additive and spectral synthesis. By default, the spectral component is imported and played with the spectral element set to Noise-Resynth mode. Playing the spectral data in Resynth mode produces a markedly different effect, which you may sometimes prefer. The spectral element internal highpass control is automatically set to a high value, thus excluding frequencies that would compete with those produced by the additive element. If the spectral element is too subtle, you can make it more prominent by setting the highpass control to a lower value.
  - *Formant button:* Use in conjunction with the Additive, Spectral, or Add+Spec buttons to perform additional analysis of formants in the sample material. The values of this formant analysis are mapped to the Analyzed formant filter parameters in the source subpage. See Formant filter controls.
  - *Granular button:* Use when you intend to perform granular-specific manipulations such as time stretching. The root note determines the MIDI note that plays the sound file at its original transposition. If the filename has a pitch value appended to it, Alchemy automatically sets the root note to match.
  - *Sampler button:* Use to create straight sample-playback presets. The root note determines the MIDI note that plays the sound file at its original transposition. If the filename has a pitch value appended to it, Alchemy automatically sets the root note to match.
- *Mapping pop-up menu:* Choose a mapping mode suitable for pitched or unpitched imported samples.
  - *Pitch:* Suitable for sounds that you want to play across the keyboard chromatically. Use this mode to automatically place imported samples in keyboard zones for optimum playback when re-pitched.
  - *Drum:* Maps each sample to a single key. Use this mode to map sounds such as a set of drum hits or chords that won't transpose correctly if re-pitched. Imported samples are mapped to individual keys (starting at C1).

- *Max Size pop-up menu:* Choose how samples contained in Sampler Instruments (.exs format files) are handled. This lets you limit the size of the import.
  - *Single Sample:* Maps a single sample across all keys.
  - *Octaves:* Maps one sample to each octave.
  - *Two per octave:* Maps two samples to each octave.
  - *One layer:* Maps all samples to a single layer.
  - *Four layers:* Maps all samples to four layers.
  - *Unlimited:* Maps all samples across multiple keys and layers.
- *Preview button:* Turn Preview mode on or off. When selected, clicking a filename in the Folders, samples, EXS instruments column or the Dropzone automatically previews that sample.
- *Cancel button:* Cancel the import operation. You can also click the close button at the upper-right corner to cancel the import operation, and to exit the Import Browser window.
- *Import button:* Start the import operation. Depending on the chosen mode, when you click Import, an import progress dialog is briefly displayed and the Import browser window closes. Click Cancel next to the progress bar to cancel a long import.

Drag an audio file into a source

1. In Alchemy in Logic Pro, open advanced view, then drag an audio file onto any source select field from one of the following:
   - Loop Browser
   - File Browser
   - Logic Pro Main window
   - Finder: Drag and drop.
2. Drop the file onto one of the analysis mode labels shown on the grid of the target source.

   The chosen analysis mode is highlighted.



   If Additive, Spectral, or Granular mode is chosen, analysis may take a moment or two. No analysis occurs when Sampler mode is used.

   Note: If you want to perform an Add+Spec analysis, or want to perform a further formant analysis of the audio material, use the Import browser methods discussed in following tasks.

Replace the sound for Alchemy on a software instrument track using drag and drop

1. In Logic Pro, drag an audio file, region, or Apple Loop to a software instrument track with an Alchemy instrument plug-in as the instrument.
2. When the “Replace existing sound with” dialog appears, drag the item to one of the available options to choose the synthesis type Alchemy uses to process the content: Additive, Granular, or Spectral.

Import a single audio file into a source with the Import browser

1. In Alchemy in Logic Pro, open advanced view then click a source select field and choose Import Audio from the pop-up menu to open the Import browser window.
2. Click one of the Analysis Mode buttons to choose an import analysis mode.

   If the Additive, Spectral, or Add+Spec button is active, you can also turn on the Formant button. This performs a further analysis of the audio material and sends the results to the Analyzed section of the formant filter. See Formant filter controls.

   If the audio data is imported to the additive, spectral, add+spectral, or granular elements, analysis may take a moment or two. No analysis occurs when the Sampler analysis mode button is active.
3. Click the Preview button to turn automatic preview of selected files on or off.

   Sound files can be mono or stereo, 8-, 16-, 24-, or 32-bit, at any sample rate. Note that rates above 44.1 kHz don’t provide significant improvements in quality.
4. Click a filename, then click the Import button to import the sound.

   An import progress dialog is shown. The Import browser closes when the import is complete, and the previous window is displayed.

Import multiple audio files into a source with the Import browser

Multiple WAVE and AIFF files may be imported to a single source. Other file types are not supported for multifile import.

If the note name is included in the filename, samples are mapped to corresponding keyboard zones for all analysis types. When importing using additive or spectral analysis, samples without a note name are analyzed to determine pitch information that is then used for keyboard mapping. Samples are mapped to the highest key of the zone and pitched down for the remaining notes in the zone.

When no pitch information or note names are available, samples are mapped evenly across keyboard zones based on file selection order. The root key pitch is set to the middle of each zone.

1. In Alchemy in Logic Pro, open advanced view then click a source select field and choose Import Audio from the pop-up menu to open the Import browser window.
2. Click one of the Analysis Mode buttons to choose an import analysis mode.

   If the Additive, Spectral, or Add+Spec button is active, you can also turn on the Formant button. This performs a further analysis of the audio material and sends the results to the Analyzed section of the formant filter. See Formant filter controls.

   If the audio data is imported to the additive, spectral, add+spectral, or granular elements, analysis may take a moment or two. No analysis occurs when the Sampler analysis mode button is active.

3. Click the Preview button to turn automatic preview of selected files on or off.

   Sound files can be mono or stereo, 8-, 16-, 24-, or 32-bit, at any sample rate. Note that rates above 44.1 kHz don't provide significant improvements in quality.

4. Use standard modifier keys to select multiple files: Command-click to select or deselect files, Shift-click to select a range of files.

   If you are creating a sound that uses samples from multiple instruments or folders, drag the selected file or files to the Dropzone shown at the right side of the Import browser. Once at least one file is added to the Dropzone, you can double-click a filename to add it to the list of Dropzone files. Drag filenames in the Dropzone list to change their order.

   Tip: You can also drag files directly into the Dropzone from the Main window, the File Browser, the Loop Browser, or Finder. You can use standard modifier keys to select or deselect files.

5. Choose a mode from the Mapping pop-up menu.

   - *Pitch:* Suitable for sounds that you want to play across the keyboard, chromatically.
   - *Drum:* Map each sample to a single key.

6. Click the Import button to import all selected files.

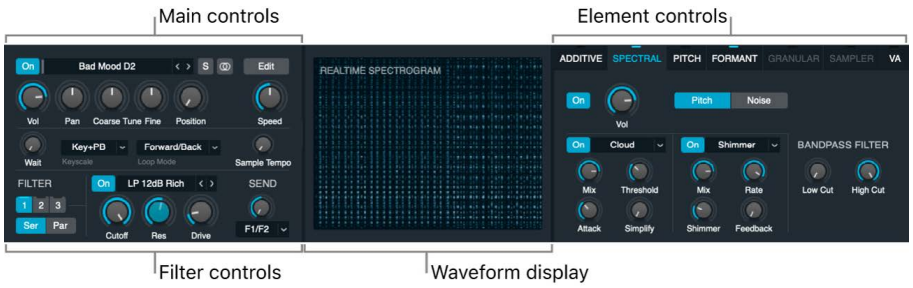
   Note: Clicking Import when at least one file is in the Dropzone imports files from the Dropzone, rather than files selected in the file list.

Alchemy source subpage controls in Logic Pro for Mac

Source components are shown only in advanced view. Click the Advanced button to switch to advanced view, then click the A/B/C/D button to select the source you want to edit.

Each of the four sources has an identical set of controls for more in-depth editing. Several source subpage controls are duplicates of those found in the source master controls. See Source master controls.

For information on source filter controls, see Source filter controls.



The waveform display and element controls section updates to show relevant data and parameters when different synthesis engines are active. The Spectral element is active for source A in the image.

Source subpage controls

- *On/Off button:* Turn the source on or off.
- *Source select pop-up menu:* Displays the name of the current source audio data. Click to open a pop-up menu with source content handling commands. See Source master controls.
- *Solo button:* Isolate the source by turning off all other sources. Click again to restore the on/off status of other sources.

  Note: If you save a preset with one source in solo mode, the resulting preset retains the On state of the source, but solo mode is turned off. All other sources are off.
- *Stereo button:* Turn stereo mode on or off. If stereo mode is off and a stereo file is loaded or imported, only the left channel is played. When stereo mode is on, loaded or imported sounds are played in stereo. This makes it possible to pan individual oscillators in the additive element or individual grains in the granular element, for example.
- *Edit button:* Open the source edit window. See Alchemy source edit window overview.
- *Volume knob:* Set the source output level (-inf dB to 0 dB).
- *Wait knob:* Set a delay between the keystrike and triggering of the source.
- *Coarse Tune knob:* Tune the source pitch in semitone increments.
- *Fine Tune knob:* Tune the source pitch in increments of one cent (one hundredth of a semitone).
- *Position knob:* Determine the playback start position in the audio data. A value of 0% indicates the absolute start point of the audio data. A value of 100% indicates the absolute end point. See Source modulations.

  When the Loop Mode pop-up menu (see entry below) is set to Start/End or Start/Length, the Position (and Speed) knob is replaced with the following two controls:

  - *(Loop) Start knob:* Determine the playback start position in the audio data. A value of 0% indicates the absolute start point of the audio data. A value of 100% indicates the absolute end point.
  - *(Loop) End/Length knob:* Loop End sets the playback end point. Loop Length determines the playback length, expressed as a percentage of the overall length of the sample.
- *Speed knob:* Set the audio data playback rate in additive, spectral, or granular mode. Speed has no effect when the element is set to sampler mode. A setting of 100% represents the original playback rate. Higher settings (up to 500%) represent faster playback, while lower settings (down to 0%) represent slower playback.

  Playback begins at the point set with the Position knob and travels through the audio data on a path determined by the Loop mode. Speed determines the rate of this travel. Playback remains at the normal pitch regardless of the rate. Set Speed to 0% to halt playback at the point set with the Position knob.

- *Sample Tempo knob:* Displays the analyzed sample tempo of imported audio material. You can set it to Off or a value between 5 and 990 to match the length of beats in the audio file with beats in the project. The file tempo is automatically analyzed for tempo (BPM) info on import.

  Note: When Sample Tempo is set to any value other than Off, the speed of sample playback is a multiple of the project tempo (BPM) and the analyzed audio file BPM. This results in a beat in the audio file matching the playback time of a beat in the project, when the Speed parameter is set to a value of 100%.

- *Keyscale pop-up menu:* Choose from three key scaling options that affect source pitch response to incoming MIDI data.

  - *Key+PBend:* The pitch of the source responds to MIDI note and pitchbend data.
  - *Key:* The pitch of the source responds to MIDI note data but does not respond to pitchbend data.
  - *Off:* The pitch of the source does not respond to either MIDI note or pitchbend data.

    Note: Global pitchbend behavior is determined with the PitchBend Up and Down controls in the master voice section. See Alchemy master voice section.

- *Loop Mode pop-up menu:* Choose from seven loop mode options that affect playback of the source audio data.

  - *None:* Ignores the loop start and end points, and plays the entire sound once without looping.
  - *Continuous:* Plays from the beginning, enters the loop region, and loops continuously in a forward direction while a note is held and during the envelope release phase.
  - *Sustain:* Plays from the beginning, enters the loop region, loops continuously while a note is held, and exits the loop region to play the normal sound release phase.
  - *Forward/Back:* Like Continuous, but plays the loop region alternately forward and backward.
  - *Start/End:* In this mode, playback starts from the defined start point, playing at 100% speed to the defined loop end position, then skips to the start point and repeats. These positions are set with the Loop Start and Loop End knobs (see entries above).

    Note off messages have no impact on loop repetitions. If the loop end point is placed before the loop start point, playback is reversed (at the loop end point) when the loop start point is reached.

  - *Start/Length:* In this mode, playback starts from the defined start point, playing at 100% speed to the defined loop end position, then skips to the start point and repeats. The length of the loop, controlled with the Loop Length knob, is set as a percentage of the overall length of the sound, such that loop end point is equal to the loop start point plus loop length.

    If the settings of Loop Start and Loop Length controls (see entries above) cause playback to go beyond the end of the underlying sample, silence is automatically inserted at the end of the sound.

- *All:* Ignores the loop start and end points, and loops the entire sound continuously.

  Note: The loop start and the loop end points, and sample start and end points, can be edited in the Main edit window. See Alchemy source edit window overview and Zone waveform editor. The VA noise component is not affected by the Loop mode setting.

- *Waveform overview display:* Shows the waveform for the source when the granular or sampler element button is active. The display also shows real-time additive or spectral data, or a representation of the waveform, if the Position or Speed controls are selected. When Position is adjusted, a position indicator is shown, allowing you to fine-tune the start position.
- *Synthesis element buttons:* Choose the synthesis type, and view and change related parameters in the area below. A variable combination of the Additive, Spectral, Pitch, Formant, Granular, Sampler, and VA buttons is available for use, depending on the audio data specified in the source select field and the import method used, if applicable. See Source elements overview and Import browser.

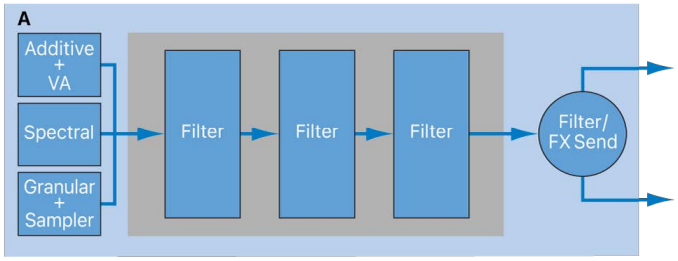
Alchemy source filters in Logic Pro for Mac

Source components are shown only in advanced view. Click the Advanced button to switch to advanced view, then click the A/B/C/D button to select the source you want to edit.

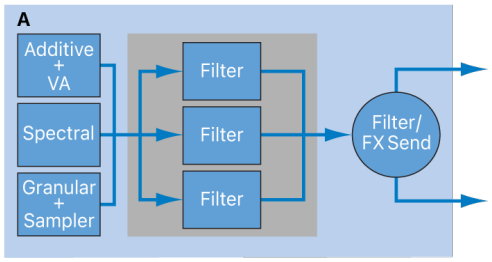
The source filter module provides three multimode filters, which can be configured either in series or in parallel. The source filters let you filter each source independently. The main filters, by comparison, process a mix of all four sources. See Main filter controls.

Tip: Though you can use filters at multiple locations in the signal path, you can attain identical or similar results by careful use of fewer filters, which helps to reduce CPU load.

- When the source filters are configured in series, the signal passes through the three filters, one after the other.



- When the source filters are configured in parallel, the signal is split and fed through the three filters simultaneously.



Source filter parameters

The most common filter controls are listed. Note that the chosen filter type can change both the name and function of available controls, notably the Cutoff, Resonance, and Drive parameters. See Filter types and Source filter use tips.

- *1/2/3 buttons:* Select each filter. The three filters are independent and can have unique settings. The LED at the top of each button shows on (lit) or off (unlit) status.



- *On/Off buttons:* Turn each filter (1, 2, or 3) on or off.
- *Filter type pop-up menu:* Choose the filter. Use the descriptive names Clean, Edgy, Gritty, Rich, Sharp, Smooth or another filter type, such as Formant or Bitcrusher. You can step through all available filter types with the Previous and Next buttons (the arrows).
- *Ser/Par buttons:* Set the filter routing configuration. Series runs from the first filter into the second filter into the third filter. Parallel runs the three filters side by side.
- *Cutoff knob:* Set the cutoff frequency for the chosen filter type.
- *Resonance knob:* Boost or cut frequencies above, below, or surrounding the value set with the Cutoff knob. Resonance behavior changes when different filter types are chosen.
- *Drive knob:* Overdrive the filter. This can lead to intense distortions and aliasing, depending on filter type.
- *Send knob:* Set the level of the knob(post source filters, if used) sent to the destinations specified with the Send pop-up menu.
- *Send pop-up menu:* Determine where the source output is source filters, if used.
  - *F1/F2:* Source output is routed to main filters (1 and 2). The balance between these targets is set with the Send knob.
  - *F1/FX A/B/C/D:* Source output is routed to main filter (1) and effects rack A, B, C, or D. The balance between these targets is set with the Send knob.
  - *F2/FX A/B/C/D:* Source output is routed to main filter (2) and effects rack A, B, C, or D. The balance between these targets is set with the Send knob.

    Note: Signal effects rack A, B, C, or D destinations bypass the main filters when the Send knob is set to the full right position. See Alchemy effects overview.

Alchemy source filter use tips in Logic Pro for Mac

Alchemy provides multiple filter types at different positions in the signal path. You can use filters at the source level, and the main filters, and you can also insert filters in the Effects section. The positioning can have a significant bearing on the sound produced and can also impact on the processing resources required. Depending on currently available resources, you may need to pay close attention to envelope settings, the number of voices, and other parameters to achieve the sonic result you are chasing.

- Source-level filtering provides the most precision and potentially the greatest sonic impact on your sound. The downside is that source-level filtering requires more processing resources. Processing is per voice.
- Use of the main filters is more CPU-efficient and can have significant sonic impact. Processing is per voice.
- Effects section filters process the entire signal, rather than each voice independently. Filtering at this stage of the signal path is often used to refine the overall sound or to provide a performance control variation.

Use an FM filter as a sound source

The FM filter produces a sine wave that is modulated by your source signal. Because the FM process adds harmonics to the sound, the more complex the sound you feed into the filter (and the louder it is) the more quickly the sound is distorted. FM in Alchemy is great for aggressive sounds, but is also useful for basses and other sounds.

Unlike dedicated FM synths, Alchemy does not have preconfigured algorithms or a modulation matrix set up for FM synthesis. It does, however, offer the option to run a huge number of “operators” (FM filters) in various series and parallel configurations. FM in Alchemy also has a sonic character that’s different from classic FM synths. This means following the approach outlined in the steps below is not always the best option in Alchemy if you want to recreate classic digital FM sounds. Such sounds are often more easily achieved by other means, such as with additive synthesis or resynthesis. FM in Alchemy is more like working with FM on analog synths where you modulate oscillator frequency rather than phase.

1. In Alchemy in Logic Pro, click the File button in the Name bar and choose Initialize Preset from the pop-up menu to reset all Alchemy parameters to default settings.

   Important: Use of Initialize Preset automatically turns on wide unison mode. See Alchemy wide unison mode.
2. Switch to advanced view, then click the A button to show source A parameters.
3. Use a sine or triangle wave as your source sound.
4. Click the source A filter On button to turn on the filter, then choose FM from the Filter type pop-up menu.
5. Control-click the Frequency knob, and choose Envelope Follower > Source A.

   The filter is assigned to track keyboard pitch. By default, the centered knob at 523 Hz provides a medium pitched sound that works well. If you want to change the octave, multiply or divide by two, and round to the nearest whole number that sounds best. For example, set the Frequency knob at 262 Hz for one octave lower.

6. Adjust the modulation Depth on the filter to increase the impact the source signal has on the sine wave generated by the FM filter, and listen to the results. For more grit, try adjusting the Feedback control, which allows the filter output to apply modulation to itself.
7. FM in Alchemy requires experimentation to develop more complex sounds. Here are a few things to try:
   - Use other types of VA oscillators or samples as your modulation source.
   - Run multiple FM filters in series or parallel to see what best meets your needs. When doing so, resist high initial modulation and feedback depths so you can get a feel for the degree of control you have in shaping the overall sound.
   - Use independent envelopes for the frequency and volume of your source (or on FM filters earlier in the signal path) to hear the results this provides.

   Important: Due to technical requirements, FM is often best done at the source filter level. As you progress through the signal path, gain increases (and therefore FM) become increasingly heavily modulated and distorted. You will find that it is easier to work with FM at the source level than as a master filter or effect.

Use a comb filter as a sound source

When making comb filters your primary sound source, you may find that longer master envelope release settings are required for a natural feel when playing. This results in more overlapping notes, higher polyphony, and therefore more CPU overhead. Because of this, you may need to carefully adjust envelope release times and reduce the maximum number of voices in the Master section.

1. In Alchemy in Logic Pro, click the File button in the Name bar and choose Initialize Preset from the pop-up menu to reset all Alchemy parameters to default settings.

   Important: Use of Initialize Preset automatically turns on wide unison mode. See Alchemy wide unison mode.
2. Switch to advanced view, then click the A button to show source A parameters.
3. Click the source A filter On button to turn on the filter, then choose Comb PM from the Filter type pop-up menu.
4. Control-click the Frequency knob, and choose Envelope Follower > Source A.

   The filter is assigned to track keyboard pitch. By default, the centered knob at 523 Hz will provide a medium pitched sound that works well. If you want to change the octave, multiply or divide by two, and round to the nearest whole number that sounds best. For example, set the Frequency knob at 262 Hz for one octave lower.
5. Control-click the Volume knob for source A (choose the lowest feasible level), and assign a New AHDSR from the menu.

6. Set up an envelope for your impulse signal to excite the comb filter. You can choose any impulse type, from VA noise to resynthesized samples. The aggressive nature of FM also makes this a good choice of impulse for comb filters.

   The impulse requires its own envelope, separate from the master or any envelope you have controlling the comb. The best settings for the envelope depend on the needs of the sound, but a good rule of thumb is to start with zero attack, zero hold, a very short decay, zero sustain, and zero release. This provides a quick spike that starts comb movement and allows the remainder of sound generation be handled by the comb itself.

   Tip: The chosen impulse can have a large impact on the tone so it is worthwhile experimenting with different sound sources. One approach is to import a sample with a strong initial attack using the Additive import method, then use the Additive Harmonic effect knobs to adjust the tone. These controls plus comb filtering can provide numerous fast and easy adjustments, letting you dramatically alter the perceived hardness, material, and tone of your modeled sound. You can also import a drum loop and set it to sustain with Continuous Loop mode. Because drum loops typically contain short bursts of sound that vary in tone, they work well with comb filters.

7. Use the Damp control to reduce ringing or other artifacts in the sound, if required.

Create vowel sounds with parallel source filters

Vowel sounds in human speech are the result of formants produced by changing the shape of the vocal cavity. By mimicking these formants, Alchemy can produce speech- or song-like effects, even in VA mode without samples of actual speech.

1. In Alchemy in Logic Pro, click the File button in the Name bar and choose Initialize Preset from the pop-up menu to reset all Alchemy parameters to default settings.

   Important: Use of Initialize Preset automatically turns on wide unison mode. See Alchemy wide unison mode.

2. Click the source A button, then make the following parameter changes:
   - Set Vol to 0 dB.
   - Set Coarse Tune to –12 semis.
3. Click the source A Fine Tune knob.
4. In the top slot of the modulation rack, choose LFO > LFO 1 from the pop-up menu, then set the Mod Depth knob to a value near 50%.
5. In the LFO 1 control panel, turn off Sync, set Attack to approximately 0.50 sec, and set Rate to approximately 5 Hz.
6. Turn on the first source filter by clicking the On button next to the filter type pop-up menu. Do the same for filters 2 and 3. Click the Par(allel) button to turn on parallel configuration of the source A filters.
7. Click the Global button to view all sources, then click the source A content field and choose Copy Source from the pop-up menu.
8. Click the source B button to turn on source B, then click the content field, and choose Paste Source from the pop-up menu. Repeat for sources C and D.

9. Click the global A button to view source A, then make the following (approximate) parameter changes to create an “ahhh” sound:
    - Cutoff 1 = 800 Hz, Res 1 = 87%
    - Cutoff 2 = 1200 Hz, Res 2 = 87%
    - Cutoff 3 = 2800 Hz, Res 3 = 87%
10. Click the global B button to view source B, then make the following (approximate) parameter changes to create an “eeeh” sound:
    - Cutoff 1 = 230 Hz, Res 1 = 87%
    - Cutoff 2 = 2600 Hz, Res 2 = 75%
    - Cutoff 3 = 3200 Hz, Res 3 = 75%
11. Click the global C button to view source C, then make the following (approximate) parameter changes to create an “oooh” sound:
    - Cutoff 1 = 200 Hz, Res 1 = 87%
    - Cutoff 2 = 880 Hz, Res 2 = 65%
    - Cutoff 3 = 2400 Hz, Res 3 = 50%
12. Click the global D button to view source D, then make the following (approximate) parameter changes to create an “ehh” sound:
    - Cutoff 1 = 530 Hz, Res 1 = 65%
    - Cutoff 2 = 1850 Hz, Res 2 = 75%
    - Cutoff 3 = 2500 Hz, Res 3 = 50%

Control parallel source filters with morphing

The previous task set up a number of vowel sounds with source filters. Follow these steps to morph between these sounds with your keyboard modulation wheel.

1. In Alchemy in Logic Pro, click Morph to view morph parameters, then click the Morph Lin button.
2. Set the X knob to 0%.

    In the modulation rack, Morph All X is shown at the top.
3. Choose Perform > Control7 in the first modulation rack slot. Set the modulation depth value to 100%.

    The modulation wheel is linked by default to performance control number 7 (Control7).
4. Play your keyboard, and move your modulation wheel to morph between the source filtered sounds.

    If the output level seems low, you can boost it by increasing the value of the Vol knob in the Master section or the Volume knob on the Name bar.

Alchemy source elements overview in Logic Pro for Mac

Source components are shown only in advanced view. Click the Advanced button to switch to advanced view, then click the A/B/C/D button to select the source you want to edit.

Each source can make use of multiple synthesis elements that operate on different functional principles. You can use a synthesis method independently, or you can combine multiple synthesis types by turning on all required elements. For example, you can combine granular synthesis with virtual analog synthesis, or additive synthesis with spectral synthesis. Each synthesis method has inherent strengths and weaknesses, making them more suitable for certain sound types than other synthesis engines. If you're new to synthesis or are unfamiliar with different synthesis approaches, see Synthesizer basics overview.



The Additive, Spectral, Pitch, Formant, Granular, Sampler, and VA buttons on the source subpage show synthesis engine parameters. Not all of these buttons are available for use at any one time. The combination of active buttons updates to reflect the audio data specified in the source select field and the import method used, if applicable. See Import browser.

See these sections for detailed information on parameters and functions.

- Additive element controls
- Additive element effects
- Spectral element controls
- Spectral element effects
- Pitch correction controls
- Formant filter controls
- Granular element controls
- Sampler element controls
- VA element controls
- Wide unison mode

Alchemy additive element controls in Logic Pro for Mac

Source components are shown only in advanced view. Click the Advanced button to switch to advanced view, then click the A/B/C/D button to select the source you want to edit.

The parameters in this section are shown when the Additive button is active in a source subpage. The additive element controls also include a number of additive effects. See Additive element effects.



Additive element parameters

- *On/Off button:* Turn the additive element in the source on or off. This can be useful when an audio file has been imported using Add+Spec mode and both the additive and spectral engines are used to recreate different aspects of the sound. Use the On/Off button to turn off either section if you need to listen to the additive or spectral component in isolation.
- *Sine/Complex buttons:* Use Sine to resynthesize each partial with a sine wave. Use Complex to choose a resynthesis waveform from the Shape pop-up menu.

  Sine mode results in the most accurate resynthesis of the original sample and makes it easy to work with the additive effects and formant controls. In Complex mode, choosing any non-sine waveform can have a dramatic and often unusual effect on the overall timbre of the sound.

  Note: The additive effects and formant controls are named on the assumption that each partial is a sine wave. If one of the complex waveforms is used, the Pulse/Saw knob and formant shape selectors behave in a more complex manner. To simplify working with these controls, it is recommended that the Sine option is chosen in most cases.
- *Shape pop-up menu:* Choose a waveform used to resynthesize each partial in your sound.
- *Volume knob:* Set the output level of the additive element within the source. When multiple elements are used in a source, use this control to set the relative level of the additive component.
- *Num Partials knob:* Set the number of additive partials that are generated (up to 600).

  The number of oscillators required depends on the sound. For example, a flute has a limited number of harmonics and requires fewer partials than a cello or a violin. The playable register can also affect the number of oscillators required: high notes will accommodate only a small number of higher harmonics before reaching the limits of audibility, while low bass notes may have hundreds of harmonics without reaching the limit. Alchemy automatically sets a suitable Num Partials value when re-synthesizing additive data from imported audio files. You can reduce this value in some cases, but removing higher partials can cause unwanted changes to certain sounds.

  Note: The additive engine processes partials in groups of four. Set the Num Partials parameter to a multiple of four to achieve the best compromise between CPU load and sound quality. Always set Num Partials to the lowest number of partials that are *required* by the sound because this helps to reduce CPU load.

- *PVar knob:* Tune all partials simultaneously. This occurs before processing by the additive effects modules which stretch/shift partial tunings. Set to 0% to tune all partials in a perfect harmonic series. Set to 100% to make each partial follow the pitch fluctuations detected in the original audio file. The sonic impact of this parameter is highly dependent on the audio material: sounds with strong inharmonic content such as bells are dramatically changed by reducing pitch variations. If all partials are tuned to the harmonic series, however, the knob has no influence on the sound.

  The pitch variation knob is most useful when dealing with resynthesized audio. For example, if you analyze a vocal sample recorded with vibrato, this knob lets you reduce the vibrato depth, or remove it entirely with a setting of zero. Removing all pitch variations from a vocal can result in a synthetic, artificial character.

- *Sym knob:* Alter the symmetry, or shape, of sine waves by lengthening the first half of the waveform while shortening the second, or the reverse. The audible effect is similar when the knob is turned in either direction. Symmetry alters waveforms until they are no longer pure sine waves in shape, resulting in each partial developing independent harmonics and making the sound brighter.

Resynthesize a sound and modify it with the main additive controls

1. In Alchemy in Logic Pro, click the File button in the Name bar and choose Initialize Preset from the pop-up menu to reset all Alchemy parameters to default settings.

   Important: Use of Initialize Preset automatically turns on wide unison mode. See Alchemy wide unison mode.

2. Click a source select field, then choose Import Audio from the pop-up menu.
3. In the Import browser, click the Additive button to change the analysis mode.
4. Choose an audio file, then click Import.

   Single notes with a strong harmonic character tend to work well. A good source of such files is the Vocals subfolder of the EXS Factory Samples folder.

5. Play the file up and down the keyboard and reduce the value of the Num Partials knob to remove upper harmonics. Note that when playing higher notes you will need to turn this knob down further before you hear it start to take effect.
6. Reduce the PVar knob value to remove all pitch variations and force all partials to a perfect harmonic series. A PVar value of zero completely removes any vibrato or pitch changes in the source sound.
7. Adjust the Sym knob value to change the symmetry of source sine waves, and note the extra brightness that is introduced by new harmonics that are generated.

Alchemy additive element effects in Logic Pro for Mac

Source components are shown only in advanced view. Click the Advanced button to switch to advanced view, then click the A/B/C/D button to select the source you want to edit.

Three effects units are available in the lower half of the additive parameters shown in the source subpage. See Additive element controls. These are not audio effects in the traditional sense; rather they provide ways to control entire groups of partials simultaneously. Each unit provides a different selection of effects. Unit 1 is devoted to amplitude effects, unit 2 to pitch effects, and unit 3 to panning effects.

The initial configuration of the effects units changes when you use the default preset and when you import a sample. A preset with initialized settings loads the Pulse/Saw module into unit 1. A sound with an imported sample loads the Harmonic module into unit 1.

Note: Some effects are included only for compatibility with older Alchemy versions. These effects are shown when a preset based on the older Alchemy architecture is loaded, but you cannot see, nor insert, these “legacy” effects in new presets or in presets designed for the new Alchemy architecture.



Additive effect 1 (amplitude) parameters

- *Additive effect On/Off button:* Turn the additive effect unit on or off.

  WARNING: Turning off additive effects modules, particularly additive effect 1, can result in a significant level increase. It is best to turn down your amplifier or mixer levels to avoid signal spikes that may damage your speakers or hearing.

- *Additive effect 1 pop-up menu:* Choose an effect type that is used to adjust the levels of different partial groups. Your choice determines the controls that are shown. Choosing None results in an equal volume level for all partials, which can sound extremely bright and harsh.

  - *Harmonic:* Allows you to set groups of partial levels.

    - *Fundamental knob:* Set the level of the fundamental tone and all partials above it. Set to zero to completely remove the fundamental tone. Set to 100% to hear the fundamental tone in isolation. Higher values tend to make the sound thicker.

    - *Octaves knob:* Set the level of the fundamental tone and all partials at whole octave intervals above it. Set to zero to completely remove harmonics 1, 2, 4, 8, 16, and so on, while boosting the levels of non-octave harmonics. Set to higher values to boost the levels of octave harmonics while reducing the levels of other harmonics.

- *Odd/Even knob:* Set the balance between odd or even harmonics. Low values increase the level of odd-numbered partials (1, 3, 5, 7, and so on), making the sound more hollow. High values increase the level of even harmonics (retaining the fundamental tone: harmonic/partial 1) to make the sound brighter and sweeter.
- *Fifths knob:* Set the level of the fundamental tone and all partials at fifth intervals (7 semitones) above it. Higher values boost harmonics 1, 3, 9, 27, and so on, with a corresponding reduction in the levels of other harmonics. Low values have the reverse effect and can make the sound more cutting and edgy.

- *Pulse/Saw:* Scales partial levels to make higher partials softer than low partials. Loaded by default when no sample has been imported into the additive element of the source.
  - *Pulse/Saw knob:* Set the mix ratio between a pulse wave and a sawtooth wave, created by summing sine wave partials tuned to a harmonic series. Turn toward Pulse to mute even harmonics. Note that if one of the complex waveforms is used, the Pulse/Saw knob behaves differently.
  - *Sync knob:* Turn on oscillator sync and set the pitch that the main oscillator is synchronized with.
  - *Odd/Even knob:* Set the balance between odd or even harmonics. Low values increase the level of odd-numbered partials 1, 3, 5, 7, and so on, making the sound more hollow. High values increase the level of even harmonics (retaining the fundamental tone: harmonic/partial 1) to make the sound brighter and sweeter.
  - *Tone knob:* Tilt the overall spectrum from flat to steeply angled, affecting the tonal color.
- *Saw+Noise:* Saw wave with noise applied to the amplitude of harmonics.
  - *Noise knob:* Set the amount of noise applied to amplitudes of harmonics.
  - *Rate knob:* Set the rate at which the noise value is changed.
  - *Smooth knob:* Set a response that varies between an instant change at each new value through to a gradual ramp over the entire time period.
  - *Tone knob:* Tilt the overall spectrum from flat to steeply angled, affecting the tonal color.

Additive effect 2 (pitch) parameters

- *Additive effect On/Off button:* Turn the additive effect unit on or off.
- *Additive effect 2 pop-up menu:* Choose an effect type that is used to adjust the tuning of different partial groups. Your choice determines the controls that are shown.
  - *Beating:* Allows a group of partials to be detuned to create beating effects as they move in and out of phase with adjacent partials or to add a clangorous, atonal character to a sound. The group of partials affected is defined by the Partial knob.
    - *Amount knob:* Set the amount of detuning for selected partials. The range of this knob is determined by the position of the Tuned knob.
    - *Partial knob:* Define the pattern of (groups of) partials affected. Set to 2 to limit the detuning affect to only partials 2, 4, 6, 8, and so on; set to 3 to apply detuning to only partials 3, 6, 9, 12, and so on. Higher values affect fewer partials, which in turn changes the impact of the Amount knob, making it more subtle.
    - *Tuned knob:* Set the range for the Amount knob. This knob behaves like a switch.

      On: At the zero position, the selected partial is tuned down to the pitch of the second partial below. When the Amount knob is at 100%, the partial is unaffected.

      Off: At the zero position, the selected partial is tuned down to the pitch of the second partial below. When the Amount knob is at 100%, the selected partial is tuned up to the pitch of the next partial above.
  - *Stretch:* Allows the tuning of partials to be stretched upward, sharpening upper partials while retaining the original fundamental tone pitch. This stretching of the harmonic series is typical of instruments that use vibrating strings or tines.
    - *Amount knob:* Stretch the tuning of all partials equally. Higher values increase the intervals between partials and shift higher partials upward. Lower values decrease the intervals between partials and shift higher partials downward. This is a quick way to tune upper partials without the need to manually edit partial pitch values in the additive editor. See Additive edit window overview.
    - *String knob:* Stretch the tuning of higher partials more intensely than lower partials. Small increases can result in a subtle sweetening of the sound without altering its basic character. Larger increases can add an inharmonic, metallic, or bell-like quality to upper partials. Modulate this parameter with an envelope to add a plucked string type articulation to the start of a note.
  - *Shift:* An unusual frequency shifter type effect that moves all partials up or down by the same amount in hertz, thereby breaking the harmonic relationships between them. In a sound with a fundamental frequency of 220 Hz and a second harmonic an octave higher at 440 Hz, an upward frequency shift of 100 Hz results in partials at 320 Hz and 540 Hz, with the second partial no longer an octave higher than the first. This effect type can radically alter the harmonic structure, leading to inharmonicities and atonalities, in addition to a perceived change of the fundamental pitch.
    - *Pitch knob:* Set the amount of shift for the first harmonic. All subsequent partials are shifted by the same amount in hertz rather than in semitones because this would result in a simple pitch change. Defining the frequency shift in this way retains a consistent timbre as you play up and down the keyboard.
    - *Frequency knob:* Define a static shift in hertz. Because the shift does not track the keyboard, the timbre of the resulting sound changes from note to note. When combined with another harmonic sound (in the VA section, for example), small shifts can create beating, chorus-type effects between the source elements.

- *Magnet:* Allows you to shift the tunings of all partials toward a specified target pitch. This can lead to unusual effects and can result in dramatic transformations of the sound.
  - *Amount knob:* Specify the amount of pitch shift as a percentage. At 100%, all partials are set to the pitch determined by the Pitch knob. At 50%, all partials are moved halfway to the target pitch. Subtle use of this parameter can turn harmonic pitched sounds into atonal bell-like timbres, for example.
  - *Pitch knob:* Set the target pitch. All partials are shifted toward the target when the Amount knob is rotated. At low positions, partial pitches shift down. Toward 100%, partial pitches shift up. At 50%, high partial pitches shift down and low partial pitches shift up. Adjust the knob to balance the shift and to control the brightness of any atonal, inharmonic elements that may be introduced to the sound.
- *Noise:* Applies noise to shift the tunings of partials. This can lead to chaotic effects.
  - *Amount knob:* Set the amount of noise applied to partials.
  - *Rate knob:* Set the rate at which the noise value is changed, with a smooth ramp of pitches between these points.
  - *Smooth knob:* Set a response that varies between an instant change at each new value through to a gradual ramp over the entire time period.
  - *Min. Partial knob:* Set the amount that noise affects low or high partials by specifying the minimum partial that is altered by the effect.

Additive effect 3 parameters

- *Additive effect On/Off button:* Turn the additive effect unit on or off.
- *Additive effect 3 pop-up menu:* Choose an effect type that is used to process the output of the additive element.
  - *Auto Pan:* Modulates the pan positions of all partials from left to right in a regular pattern.
    - *Amount knob:* Set the depth of the stereo panning effect.
    - *Rate knob:* Set the speed of the stereo panning effect.
    - *Ramp knob:* Set the panning amount for lower partials. This can create a subtle widening of the sound, with less obvious left to right movement in the lowest partials.
    - *Cycles knob:* Set phase differences between partials. At zero, all partials are in-phase. Higher values lead to a finer, more subtle and complex effect.
  - *Comb:* Applies a comb filter to the output of the additive element.
    - *Amount knob:* Set the depth of the comb filter effect.
    - *Damp (Damping) knob:* Set the duration of each dip (or notch) in the comb.
    - *Frequency knob:* Set the frequency of the filter. The range is from 16 Hz to 20 kHz.
    - *Feedback knob:* Set the amount of feedback.

- *EQ:* Applies an EQ to the output of the element. Crossover frequencies are centered at 120 Hz, 800 Hz, and 5,000 Hz, with a one octave transition between bands.
  - *Low knob:* Set the amount of low frequency cut or boost.
  - *Low Mid knob:* Set the amount of lower mid frequency cut or boost.
  - *High Mid knob:* Set the amount of upper mid frequency cut or boost.
  - *High knob:* Set the amount of high frequency cut or boost.
- *Filter:* Applies a continuously variable lowpass/highpass filter to the output of the additive element.
  - *Amount knob:* Set the depth of the filter effect.
  - *LP - HP knob:* Set the shape of the filter blending between a low pass, band pass, and high pass filter.
  - *Cutoff knob:* Set the cutoff frequency. The range is from 16 Hz to 20 kHz.
  - *Resonance knob:* Adjust the level of frequencies surrounding the cutoff frequency.
- *Spread:* Modulates the pan positions of all partials to create a wide pseudo-stereo effect. Partials are panned left to right in a regular pattern, with every second partial position inverted.
  - *Amount knob:* Set the depth of the stereo panning effect.
  - *Rate knob:* Set the speed of the stereo panning effect.
  - *Ramp knob:* Set the panning amount for lower partials. This can create a subtle widening of the sound, with less obvious left to right movement in the lowest partials.
  - *Cycles knob:* Set phase differences between partials. At zero, all partials are in-phase. Higher values lead to a finer, more subtle and complex effect.
- *Strum:* A filter that modulates the amplitudes of partials to create interesting effects, some of which can be reminiscent of guitar strumming.
  - *Amount knob:* Set the depth of the effect.
  - *Partial knob:* Select the partial (number) that you want to set to full amplitude. Modulating the partial number can create rhythmic melodic effects.
  - *Release knob:* Set the release (or decay) time for the selected partial. Shorter release times and slower modulation of the partial number can create strumming of partials.
  - *Width knob:* Set the number of partials in a range surrounding the selected partial.
- *Ripples:* Applies a sine wave to the amplitude of partials, resembling waves or ripples in water.
  - *Amount knob:* Set the depth of the effect.
  - *Group knob:* Select a group of partials to be processed.
  - *Period knob:* Set the number of partials over which one full cycle of the sine wave is applied. For example, a value of 10 applies a sine wave shape over partials 1-10, 11-20, 21-30, and so on.
  - *Phase knob:* Set the phase of the sine wave. Modulate this with a ramp up LFO for smooth sound changes.

Alchemy spectral element controls in Logic Pro for Mac

Source components are shown only in advanced view. Click the Advanced button to switch to advanced view, then click the A/B/C/D button to select the source you want to edit.

The parameters in this section are shown when the Spectral button is active in a source subpage. Two Spectral element effects units are available in the lower half of the spectral parameters shown in the source subpage.

Note: You must first perform a sample import with a spectral analysis or draw in the Spectral edit window before you can use any of the spectral engine parameters. See Import browser and Spectral edit window.

Spectral resynthesis analyzes the changing frequency spectrum of a signal and attempts to recreate these spectral characteristics. In Alchemy, the audible spectrum of a signal is split into a large number of *spectral bins*. Energy distribution across these bins is analyzed and the sound is recreated by filling each spectral bin with the required amount of signal, using either sine waves or filtered noise. The results are then summed.



Spectral element parameters

- *On/Off button:* Turn the spectral element in the source on or off. This can be useful when an audio file has been imported using Add+Spec mode and both the additive and spectral engines are used to recreate different aspects of the sound. Use the On/Off button to turn off either section if you need to listen to the additive or spectral component in isolation.
- *Pitch/Noise buttons:* Use Pitch to fill spectral bins with sine waves. Use Noise to fill spectral bins with filtered noise. The chosen audio import mode automatically changes the state of these buttons, but you are free to change this to meet the sound's requirements.
  - *Spectral import:* Turns on only the spectral element and sets the mode to pitch. The spectral bins are filled with sine waves, which is generally the best choice to recreate the entire original signal. Noise mode can be useful for transforming normal speech into whispering, for example.
  - *Add+Spec import:* Turns on both the additive and the spectral engines. The spectral engine is used only to recreate noisier aspects of the sound because this is not a strength of additive resynthesis. In this case, the mode is set to Noise.

- *Volume knob:* Set the output level of the spectral element within the source. When multiple elements are used in a source, use this control to set the relative level of the spectral component.
- *Low Cut knob:* Set a cutoff frequency. All signals above this frequency are allowed to pass. Signals below the frequency are cut.
- *High Cut knob:* Set a cutoff frequency. All signals below this frequency are allowed to pass. Signals above the frequency are cut.

  The Low Cut and High Cut parameters work in conjunction with each other to act as a bandpass filter, where signals that fall within the two cutoff ranges are allowed to pass.

Alchemy spectral element effects in Logic Pro for Mac

Source components are shown only in advanced view. Click the Advanced button to switch to advanced view, then click the A/B/C/D button to select the source you want to edit.

Alchemy spectral element effects provide a number of creative options in the spectral synthesis engine. Two effects units are available in the lower half of the spectral parameters shown in the source subpage. See Spectral element controls for information on other spectral element parameters.

Note: You must first perform a sample import with a spectral analysis or draw in the Spectral edit window before you can use any of the spectral engine parameters. See Import browser and Spectral edit window.



Spectral effect parameters

- *Spectral effect On/Off buttons:* Turn the spectral effect unit on or off.
- *Spectral effect pop-up menus:* Choose a spectral effect type. See spectral effect descriptions below.
- *Mix knob:* Set the balance between the original signal and the processed sound. This parameter is common to all spectral effect types.
- *Parameter knob 1:* Set the value for the parameter assigned to the chosen spectral effect. The parameter name and function vary with each effect type.
- *Parameter knob 2:* Set the value for the parameter assigned to the chosen spectral effect. The parameter name and function vary with each effect type.
- *Parameter knob 3:* Set the value for the parameter assigned to the chosen spectral effect. The parameter name and function vary with each effect type.

Bloom

Bloom produces a burst of frequencies based on the source sound. Note that this effect requires a small amount of calculation time to collect and release a group of frequencies. As a result, there may be a gap between playing a note and hearing the effect.

Blur

Blur produces a frequency blurring effect.

Tip: Try a melodic loop with pitch variations to best hear the impact of this effect.

- *Mix knob:* Set the balance between the original signal and the processed sound.
- *Length knob:* Set the time period that frequencies are sustained (blurred over time).
- *Variance knob:* Set the degree of variation for frequency selection (frequencies that are blurred).
- *Gate knob:* Determine the impact of the source sound envelope on the effect and the number of audible frequencies. For example, when used on a loop, higher settings produce a simplified sound with more frequent gaps in the effect output.

Cloud

Cloud produces what might best be described as a cloud of frequency grains, resulting in a textured chorus effect.

Tip: Try this effect on vocal samples.

- *Mix knob:* Set the balance between the original signal and the processed sound.
- *Threshold knob:* Set a sustained emphasis on the highest amplitude harmonics.
- *Attack knob:* Set the time it takes for frequencies emphasized by the Threshold setting to fade in.
- *Simplify knob:* Reduce lower amplitude frequencies to enhance prominent frequencies, lessening detail in favor of an increasingly pure tone.

Freq Shift

Freq(uency) Shift moves the spectral bins higher or lower in the spectrum, reducing the level of (or entirely dropping) some frequencies while emphasizing others. This is a powerful sound design tool that can dramatically alter the sound and can introduce inharmonic overtones.

Tip: Try single note samples with a strong initial attack, such as a piano, and blend the mix level so that the effect comes in as part of the tail of the sound.

- *Mix knob:* Set the balance between the original signal and the processed sound.
- *Shift knob:* Move bins up (positive) or down (negative) the frequency spectrum. Start with small adjustments because this parameter has a wide range.
- *Feedback knob:* Set the intensity of selected frequencies.
- *LP Cutoff knob:* Set the cutoff frequency. Higher frequencies are attenuated.

Freq Stretch

Freq(uency) Stretch is an unusual and powerful effect in that spectral peaks are shifted based on a combination of the Alpha and Beta knobs, resulting in a series of inharmonic stretches and randomizations.

Tip: This effect is highly dependent on the available frequency range in the imported sample. For example, the Alpha and Beta knobs are useful across the entire range with drum loops, whereas the most useful Alpha and Beta ranges are small positive or negative deviations from the center position when used with spoken vocals.

- *Mix knob:* Set the balance between the original signal and the processed sound.
- *Alpha knob:* Shift harmonics to produce an effect reminiscent of a formant filter. A setting of 1 (centered) is closest to the source sound.
- *Beta knob:* Set the frequency range of the effect. A setting of 1 (centered) is closest to the source sound.
- *Variance knob:* Introduce randomization to frequency shifts. Depending on source material and Alpha/Beta settings, these can range from “underwater” sounds to powerful noise clouds.

Gate

Gate is best described as a combination of a square wave LFO and envelope follower modulation for frequencies in the spectrum. Depending on your settings and source material, this can either produce a choppy sound or a smoother one.

Tip: Drum loops are an ideal starting point when learning uses for this effect.

- *Mix knob:* Set the balance between the original signal and the processed sound.
- *Threshold knob:* Set the amount of lower amplitude frequencies that are allowed to pass through the gate. This reduces detail and enhances prominent frequencies in the source.
- *Attack knob:* Set the time it takes for effect-controlled frequencies to fade in.
- *Decay knob:* Set the time it takes for effect-controlled frequencies to fade out.

Glide

Glide creates adjustable, repeating upward filter sweeps that are based on the source content. Note that this effect requires a small amount of calculation time before the results of your adjustments are heard.

Tip: Sources with wide frequency ranges produce a more pronounced filter sweep sound, whereas sources with limited frequencies can result in unique melodic drones as narrow filters sweep across their ranges.

- *Mix knob:* Set the balance between the original signal and the processed sound.
- *Frequency knob:* Set the starting frequency for the sweep.
- *Width knob:* Set the width of the filter (the frequency range, or band, affected by the filter).
- *Shift knob:* Set the height of the sweep.

Metallize

Metallize produces classic comb filter style effects.

Tip: Experiment with drum loops to clearly hear the impact of the controls.

- *Mix knob:* Set the balance between the original signal and the processed sound.
- *Interval knob:* Set the effect frequency.
- *Simplify knob:* Reduce the number of effect-generated frequencies, resulting in a simplified sound.
- *Feedback knob:* Set the intensity of the effect. Higher settings emphasize harmonics, creating metallic resonances.

Shimmer

Shimmer sweeps the frequencies to varying degrees and speeds, imparting either slow frequency shifts or fast shimmering sounds.

Tip: Try pure organ samples to clearly see the results of the effect in the realtime spectrogram display, particularly at moderate rate settings.

- *Mix knob:* Set the balance between the original signal and the processed sound.
- *Rate knob:* Set the sweep speed.
- *Shimmer knob:* Set the intensity of the frequency sweep.
- *Feedback knob:* Adjust to introduce resonant, metallic harmonics that are reminiscent of sounds that can be attained with comb filters.

Shuffle

Shuffle randomly rearranges blocks of bins, resulting in an increasingly abstract and scattered sound.

Tip: Import a bell sample, and start with very low settings to see and hear the impact of controls on the sound.

- *Mix knob:* Set the balance between the original signal and the processed sound.
- *Factor knob:* Set the number of blocks (of bins) that you want to shuffle.
- *Range knob:* Set the distance (number of bins) that you want to move each block.
- *Bins knob:* Set the number of bins contained in each block. Note that bins are numbered (and selected) sequentially.

Smear

Smear averages between blocks of frequencies to create a smoother, more consistent sound. It delivers different results to the Blur effect.

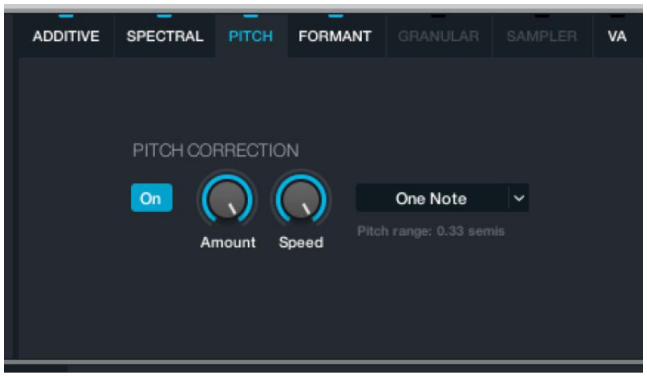
Tip: Try melodic loops that have pitch variations to showcase this effect.

- *Mix knob:* Set the balance between the original signal and the processed sound.
- *Smooth knob:* Set the number of frequency blocks to smooth between. Higher settings have less sonic variation, so make small changes.
- *Bright knob:* Emphasize higher frequencies.
- *Phase On/Off knob:* Turn on to preserve the original phases of the source sound. This natural variation in the sound provides a more organic cloud-like effect. Turn off to lock the phases of the source, resulting in a tight, metallic sound.

Alchemy pitch correction controls in Logic Pro for Mac

Source components are shown only in advanced view. Click the Advanced button to switch to advanced view, then click the A/B/C/D button to select the source you want to edit.

The parameters in this section are shown when the Pitch button is active in an additive or spectral source subpage.



Pitch correction parameters

- *On/Off button:* Turn pitch correction on or off.
- *Amount knob:* Set the intensity of pitch correction. Higher values result in stronger correction.
- *Speed knob:* Set the delay time before pitch correction occurs, following a change in the pitch of the sound. This is shown as a percentage.
- *Pitch pop-up menu:* Choose a criteria that snaps all pitches to the chosen value.
  - *One Note:* All pitches are snapped to a single pitch.
  - *Octaves:* All pitches are snapped to the nearest octave.
  - *Fifths:* All pitches are snapped to the nearest fifth.
  - *Octaves & Fifths:* All pitches are snapped to the nearest octave or fifth.
  - *Chromatic:* All pitches are corrected to a chromatic scale.

Alchemy formant filter controls in Logic Pro for Mac

Source components are shown only in advanced view. Click the Advanced button to switch to advanced view, then click the A/B/C/D button to select the source you want to edit.

The parameters in this section are shown when the Formant button is active in an additive or spectral source subpage.

When audio is imported into the additive or spectral engines with the Formant option turned on, the signal is analyzed and resonances in the original signal are extracted and converted into a formant filter shape. See Import browser.

The formant filter scales the amplitude of additive partials or spectral bins over time to recreate the characteristic resonances of the instrument, rather than processing the audio signal like a conventional filter.



Formant filter analyzed parameters

The analyzed controls work only if you activate the Formant button in the Import browser when importing your audio files. This more detailed analysis attempts to determine the resonant frequencies of the source audio data.

- *On/Off button:* Turn the analyzed formant section on or off.
- *Shift knob:* Shift the formant filter up or down in semitones. Higher values can make sounds seem brighter or thinner. Lower values can create a darker, thicker character.
- *KTrack knob:* Determine how the formant filter tracks notes on the keyboard. At 100%, filter resonances shift up or down in pitch with the note. Set to lower values to reduce key tracking which may make some sounds playable over a wider keyboard range.
- *Size knob:* Stretch the formant filter to change the perceived size of the resonant chamber. Use this parameter to alter the size of a guitar body or to make a child's vocal sample sound like that of a giant, for example. The Size knob works in conjunction with the Center parameter.
- *Center knob:* Set the center frequency for formant stretching (controlled with the Size knob). Resonances below the center frequency are shifted upward as the Size knob value is increased. A corresponding downward shift occurs to resonances above the center frequency.

  Note: The Center knob has no effect when the Size knob is set to 100%.
- *Smooth knob:* Set the rate of change for the formant filter. High values smooth and slow down formant changes. Low values exaggerate and speed up changes. Low values can also introduce an unusual "chattering" distortion that may be suitable for drum sounds.

Formant filter synthesized parameters

The synthesized controls work with any additive or spectral material and do not require the formants to be analyzed on import. Use these parameters to impose new resonant characteristics on the original signal. The default settings provide a range of vowel sounds: use the Select knob to morph through the default "a," "e," "i," "o," and "u" vowel sounds loaded into the four slots to the right.

- *On/Off button:* Turn the synthesized formants section on or off.
- *Shift knob:* Shift all synthesized formant filters up or down in semitones. Higher values can make sounds seem brighter or thinner. Lower values can create a darker, thicker character.
- *Size knob:* Stretch the formant filter to alter the perceived size of the resonant chamber. Size works in conjunction with the Center knob.

- *Center knob:* Set the center frequency for the formant stretch set with the Size knob. Resonances below the center frequency are shifted upward as the Size value is increased. A corresponding downward shift occurs to resonances above the center frequency.
- *Select knob:* Morph smoothly through the four filter units. The displayed value indicates position. Whole numbers indicate a particular filter unit, and fractional values indicate a position between filters.
- You can step through the available filter types with the Previous and Next buttons (the arrows).
    - *Off:* Passes the original data, as if the synthesized section were turned off. Assign this type to use one of the four filter units as a bypass. Adjust the Select knob to quickly turn off synthesized formant processing.
    - *Comb filters:* Comb filters are so-named because they create a distinctive comb-like pattern of boosts and cuts in the frequency spectrum, arranged in a harmonic series.
        - *Comb Neg:* The position of the first harmonic is determined by the Shift knob. The Size knob can be used to stretch the pattern of cuts and boosts up or down the frequency spectrum, or both, depending on the setting of the Center knob. The negative filter name is used because it recreates the effect of a phase-inverted delayed signal that boosts only odd harmonics, resulting in a hollow sound.
        - *Comb Pos:* This filter emulates the effect of mixing in a positive phase delayed copy of the original signal, resulting in a boost of both even and odd harmonics. This filter has a brighter sound than the negative comb filter.
        - *Metal Combs:* The Brass, Iron, Steel, and Tin comb filter variants provide a distinctive tonal coloration that emphasizes different harmonics in the frequency spectrum. To help you visualize these differences, the “teeth” (harmonics) on the comb are of varying thicknesses and are spaced at different intervals in each filter. Experiment with each comb to determine the best choice for your sound.
    - *Classic and Parallel filters:* The classic variants are two-pole designs. The parallel filters are multipole designs.
        - *Low Pass (LP):* Emulates a two-pole lowpass filter that gently reduces the levels of higher frequencies. The Shift knob sets the cutoff and the Size knob changes the filter slope.
        - *Band Pass (BP):* A bandpass filter with a gentle slope. Signals above or below the set center frequency are attenuated.
        - *High Pass (HP):* Emulates a two-pole highpass filter that gently reduces the levels of lower frequencies. The Shift knob sets the cutoff frequency. The Size knob changes the filter slope.
        - *Peak:* A filter that boosts a narrow band of frequencies around the set frequency. The frequency band can be moved up or down the frequency spectrum with the Shift knob. The Size knob sets the width of the band.
        - *Notch:* A filter that attenuates a narrow band of frequencies near the set frequency. The frequency band can be moved up or down the frequency spectrum with the Shift knob. The Size knob sets the width of the band (notch).

- *Vowel Bright, Classic, and Smooth filters:* Bright vowel sounds contain more high frequency content and are the most aggressive-sounding. Classic vowel sounds are warmer, and are similar to synthesizer vowel sound filtering. Smooth variants are more natural-sounding vowel shapes with a gentler filter slope.

  Tip: Most of the "vowel" filters are not strictly designed to create that exact vowel. Each Bright, Classic, and Smooth vowel filter is more of a unique variation on that general sound, with not only brightness differences, but also overall character differences. Additionally, any vowel filter can be independently modulated, alone or in conjunction with Select knob morphing between filters (even from mismatched sets). Use these facilities to dramatically expand your filtering options.

  - *Vowel A:* Mimics a set of vocal chords to impose an "a" vowel sound on the audio. Each variation of this complex filter shape has prominent peaks at different frequencies. It is, generally speaking, an open-sounding filter.
  - *Vowel E:* Mimics a set of vocal chords to impose an "e" vowel sound on the audio. Each variation of this complex filter shape has prominent peaks at different frequencies. It is, generally speaking, an open-sounding filter.
  - *Vowel I:* Mimics a set of vocal chords to impose an "i" vowel sound on the audio. Each variation of this filter shape has prominent upper midrange peaks and a further low-mid peak, making this filter sound thinner and less open than the "a" and "e" vowel filters.
  - *Vowel O:* Mimics a set of vocal chords to impose an "o" vowel sound on the audio. This filter shape has gentler midrange and upper midrange peaks with a dominant low-mid resonance. The result is a rounder sound with less brightness and presence than the vowel types above.
  - *Vowel U:* Mimics a set of vocal chords to impose a "u" vowel sound on the audio. This filter shape has gentler midrange and upper midrange peaks with a prominent low-mid resonance. The "u" filter variants also emphasize higher-range content around 7 and 8 kHz, for example, making them sound thinner than the "o" filter.

Modify formants in a resynthesized additive guitar sound

1. In Alchemy in Logic Pro, click the File button in the Name bar and choose Initialize Preset from the pop-up menu to reset all Alchemy parameters to default settings.

   Important: Use of Initialize Preset automatically turns on wide unison mode. See Alchemy wide unison mode.

2. Select source A, then click the source select field and choose Import Audio from the pop-up menu.
3. In the Import browser window, click the Additive and Formant Import Mode buttons.
4. Navigate to the Guitars subfolder in the EXS Factory Samples folder, and choose a single guitar sample.
5. When loading is complete, click the Formant button to the right side of the source A window. Note that the upper Analyzed section is turned on.
6. Adjust the Shift knob to move resonances up or down in frequency and to change the timbre. Small amounts of Shift variation work well for subtle changes: try a few semitones in either direction.

7. Play some very low notes, then some very high notes. Gradually turn down the KTrack knob to reduce key tracking for the formant filter, and note the difference when you replay the high and low notes.
8. Adjust the Size knob value to change the apparent size of the guitar body. Also adjust the Center knob value, and note the effect it has on the tone of the resulting larger or smaller guitar body.

Modify formants in a resynthesized spectral drum loop

1. In Alchemy in Logic Pro, click the File button in the Name bar and choose Initialize Preset from the pop-up menu to reset all Alchemy parameters to default settings.

   Important: Use of Initialize Preset automatically turns on wide unison mode. See Alchemy wide unison mode.
2. Select source A, then click the source select field and choose Import Audio from the pop-up menu.
3. In the Import browser window, click the Spectral and Formant Import Mode buttons.
4. Navigate to the Loops subfolder in the EXS Factory Samples folder, and choose a drum loop.
5. When loading is complete, click the Formant button to the right of the source A window. Note that the upper Analyzed section is turned on.
6. Adjust the Size knob value to make the drums seem bigger or smaller.
7. Adjust the Smooth knob value to alter the rate of change for the formant filter. Higher values smear the timbre of one drum into the next. Lower values exaggerate changes and create an unusual distortion near the bottom of the knob range.

Create a talking additive sound with synthesized vowel formants

1. In Alchemy in Logic Pro, click the File button in the Name bar and choose Initialize Preset from the pop-up menu to reset all Alchemy parameters to default settings.

   Important: Use of Initialize Preset automatically turns on wide unison mode. See Alchemy wide unison mode.
2. Select source A, then turn off the oscillator in the VA section to the right.
3. Click the Additive button, and turn on the additive section. You will hear an additive sawtooth sound if you play some notes.
4. As an option, increase the Num Partials value. This helps to prevent the sound becoming dull if played in lower registers.
5. Click the Formant button, and turn on the lower Synthesized section. Play a few notes to hear the “a” vowel sound that is imposed on the sawtooth wave.
6. Increase the Select knob value, and play a few notes. Note how the vowel sound morphs to an “e,” then an “i,” then finally a “u” at 100%.
7. Reduce the Select knob value, then modulate it with a new AHDSR envelope (AHDSR 2), and leave the depth at +100%.
8. Set Sustain to zero for the AHDSR 2 envelope. Note the “yeah” articulation this creates as you play each note.
9. Increase the Attack time for the AHDSR 2 envelope. Note the “aya” type articulation this creates as you play each note.

10. Adjust the Shift knob, the Size knob, and the Center knob, to explore the different timbres available.
11. Switch the order of vowels in the four pop-up menus, and also load different filter types such as Comb.

Alchemy granular element controls in Logic Pro for Mac

Source components are shown only in advanced view. Click the Advanced button to switch to advanced view, then click the A/B/C/D button to select the source you want to edit.

The parameters in this section are shown when the Granular button is active in a source subpage. The Granular section is available only when you import an audio sample using either granular or sampler mode. See Import browser.

Note: The sampler and granular engines are mutually exclusive: you can use one or the other within a single source, but not both together. You can, however, turn on further sources if both engines are required simultaneously.

Granular synthesis represents continuous sound as a stream of grains, or tiny pieces of sound. Alchemy generates grains by extracting 2- to 230-millisecond pieces from an audio file. The amplitude of each grain is shaped, along with any pitch and pan modifications, before the grain is sent to the output stream.

Grains can be reordered, time stretched, and pitch shifted. This provides an inexhaustible supply of potential raw material to use as the basis of your sounds.



Granular element parameters

In addition to the following controls, granular playback is affected by loop modes and by the settings and modulations of the Position and Speed knobs in each source subpage. See Source subpage controls. Modulations of the granular element update with each new grain. For an example of the impact this has, modulating the source Coarse Tune parameter with an LFO causes the stream of grains to rise and fall in pitch, but does not create pitch sweeps within each grain. If a large Size value is used in conjunction with a low Density value, modulations of source parameters such as pitch may sound *stepped*, rather than smooth.

- *Waveform overview display:* Shows real-time changes to the waveform.
- *On/Off button:* Turn on to use the granular element in the source.
- *Volume knob:* Set the volume of the granular element, independent of other source elements.
- *Size knob:* Adjust the duration of each grain from 2 msec to 230 msec.

- *Density knob:* Determine the number of potentially overlapping grains from 1 (no overlap) to 10.

  The Size and Density parameters interact with each other. When the Density value is 1, a single grain is sent to the output stream. As soon as one grain finishes, the next one is sent. A Size value of 100 msec sends a new grain every 100 msec.

  Increasing Density to 2 adds a second grain that is sent in between those of the first, resulting in a new grain every 50 msec, assuming a Size value of 100 msec. The first and second grains overlap each other. Higher Density values inject additional new grains into the output stream. These new grains occur more frequently and overlap more heavily.

  Setting Size to around 100 msec and Density to around 5 grains is often suitable for smooth pad sounds with no sharp transients. Setting Size between 40 and 80 msec and Density to around 2 grains is useful for drums and other sounds featuring sharp transients. Small Size values tend to produce a buzz that masks the original pitch of the sample. Large Size values tend to break up the sound. You can counteract both tendencies by increasing the Density.

  Note: Also important to the Size and Density parameters is the shape chosen in the Grain Shape pop-up menu. This can have a significant or subtle impact on sonic artifacts that may be introduced in the stream of grains.

- *RTime knob:* Add a small random offset to grain extraction positions in the sample. The default value is 3% because a small amount of randomization helps to smooth the output of the granular element.
- *RPan knob:* Add a random offset to the stereo position of each grain. The source Stereo button must be on for RPan to have an effect.
- *Num Taps knob:* Set the number of taps (up to 8). Taps retrigger the attack phase of the source.

  Note: Taps that fall within a looped area are retriggered on each loop cycle.

- *Tap Spacing knob:* Set the spacing between the specified number of taps. Values are shown as a percentage of the overall sound duration. Set to 100% to evenly space taps across the entire sound duration. Set to 50% to trigger more closely spaced taps that start at the halfway point of the sound. Set to zero to trigger taps in quick succession at the sound end point.
- *Stereo Offset knob:* Offset the stereo position to create a wider sound. The source Stereo button must be on for Stereo Offset to have an effect.
- *Grain Shape pop-up menu:* Choose the envelope shape that is applied to each grain. At a basic level, this applies a small fade-in and fade-out to each grain, but some shapes may have a more significant impact, depending on the current Size and Density values (and the source material). You can also step through the available grain shapes with the Previous and Next buttons (the arrows).

  This function is primarily intended to reduce or remove glitches, clicks, and crackles in the playback of a stream of grains, but it can introduce buzzy gaps between grains and can affect the tonality of grains. The effect of the chosen grain shape can also be quite subtle in many cases, particularly when you avoid extreme Size and/or Density values. There are no fixed rules when it comes to the choice of grain shape, given the infinite variety of source audio material. Therefore, you may want to experiment to achieve the required results.

Alchemy sampler element controls in Logic Pro for Mac

Source components are shown only in advanced view. Click the Advanced button to switch to advanced view, then click the A/B/C/D button to select the source you want to edit.

The parameters in this section are shown when the Sampler button is active in a source subpage. The sampler section is available only when you import an audio sample using either granular or sampler mode. See Import browser.

The sampler section allows audio files, known as *samples*, to be played directly. Samples played at a higher pitch than the original play back at a faster speed. Samples played at a lower pitch than the original play back at a slower speed. The sample waveform is displayed in the center. A progress bar indicates the current playback position for the most recently triggered note.

Note: The sampler and granular engines are mutually exclusive: you can use one or the other within a single source, but not both together. You can, however, turn on further sources if both engines are required simultaneously.



Sampler element parameters

- *Waveform overview display:* Shows real-time changes to the waveform.
- *On/Off button:* Turn the sampler element in the source on or off.
- *Volume knob:* Set the output level of the sampler element within the source. When multiple elements are used in a source, use this control to set the relative level of the sampled component.
- *Reverse button:* Turn on to play samples backwards. If the loop mode in the main source section is set to Forward/Back, turning on this parameter also reverses the loop playback order. See Source subpage controls.

Alchemy VA element controls in Logic Pro for Mac

Source components are shown only in advanced view. Click the Advanced button to switch to advanced view, then click the A/B/C/D button to select the source you want to edit.

The parameters in this section are shown when the VA (Virtual Analog) button is active in a source subpage.



Virtual Analog parameters

- *Waveform overview display:* Shows real-time changes to the waveform.
- *Oscillator On/Off button:* Turn the main oscillator on or off. When you click the Name bar File button, and choose Initialize Preset from the pop-up menu to initialize Alchemy to default settings, the VA element is automatically turned on.

  Important: Use of Initialize Preset automatically turns on wide unison mode. See Alchemy wide unison mode.
- *Wave pop-up menu:* Choose an oscillator waveform. Basic saw, sine, square, and triangle and many specialized waveforms are provided. You can also step through the available waveforms with the Previous and Next buttons (the arrows).
- *(Oscillator) Vol knob:* Set the output level of the oscillator. When multiple elements are used in a source, use this control to set the relative level of the oscillator component.
- *Sym knob:* Alter the symmetry, or shape, of the oscillator waveform. When a square wave is active, Symmetry acts as a pulsewidth control. The effective range is between 5% and 95% because extreme values may cause sonic artifacts.
- *Phase knob:* Set the oscillator start point (phase). Values from 0% to 99.9% set a fixed start point for the oscillator. A value of 100% causes random variations of the waveform start point each time the oscillator is triggered.
- *Sync knob:* Turn on oscillator sync and set the pitch that the main oscillator is synchronized with.
- *Num(ber) knob:* Set the number of unison voices.
- *Detune knob:* Set the amount of detuning and stereo width variance between unison voices or detuning between oscillators when wide unison mode is turned on. See Alchemy wide unison mode.
- *Noise On/Off button:* Turn the noise oscillator on or off.
- *Noise pop-up menu:* Choose a noise waveform. These have different spectral characteristics that can be further refined with filters. You can step through the available waveforms with the Previous and Next buttons (the arrows).
- *(Noise) Vol knob:* Set the output level of the noise oscillator. When multiple elements are used in a source, use this control to set the relative level of the noise component.

- *Low Cut knob:* Set a low cutoff frequency for the noise oscillator. All frequencies above this value are allowed to pass. All frequencies below are removed.
- *High Cut knob:* Set a high cutoff frequency for the noise oscillator. All frequencies below this value are allowed to pass. All frequencies above are removed.

  The Low Cut and High Cut parameters work in conjunction with each other to act as a bandpass filter, where the noise signal that falls within the two cutoff ranges is allowed to pass.

Alchemy wide unison mode in Logic Pro for Mac

Source components are shown only in advanced view. Click the Advanced button to switch to advanced view, then click the A/B/C/D button to select the source you want to edit.

The parameters in this section are shown when the VA (Virtual Analog) button is active in a source subpage.

Alchemy wide unison mode provides oscillator pan and detune options that enhance the perceived width of VA sources.

To turn on wide unison mode, choose the Wide Unison entry at the bottom of the Wave pop-up menu in the VA element controls.

Important: When you choose File > Initialize Preset, wide unison mode is automatically turned on.

All existing presets and patches load in the older, non-wide unison mode. Newly initialized settings and updated factory content will have the wide unison mode turned on.



Wide Unison mode parameters

In wide unison mode, the source Pan knob (in the source master controls to the left of the Waveform overview display) and Detune knob (in the VA element parameters) operate as outlined in this section.

- *Pan knob:* Set the oscillator output position in the stereo field. The behavior of the parameter changes as additional oscillators are added to the source. All odd-numbered oscillators (except the center panned oscillator) are hard panned as you increase the number of oscillators.
  - 1 oscillator: center panned
  - 2 oscillators: hard panned left/right
  - 3 oscillators: one oscillator is centered and the other two oscillators are panned hard left/right
  - 4 oscillators: when playing a C with full detune (2 semitones up/down), the Bb and Db are both hard right panned, and the remaining two oscillators are hard left panned

- *Detune knob:* Set the amount of detuning between wide mode oscillator voices. Oscillator pitches (in semitones) are spread in a linear fashion between the top and bottom values.

  Note: Detune knob scaling is *not* linear. Values closer to the centered position can be adjusted in smaller increments. Values further from the centered position are coarser.

  - 100% Detune value corresponds to 2 semitones up/down on each side
  - 50% Detune value plays a semi tone up/down on each side

Alchemy source modulations in Logic Pro for Mac

Source components are shown only in advanced view. Click the Advanced button to switch to advanced view.

Most Alchemy source parameters, such as Pan, Tune, and Position, can be modulated by modulation sources such as AHDSRs or LFOs. Parameters that have a modulation assignment are indicated by an orange arc around the control. See Alchemy modulation overview.

Note: Parameters that are morphed and have a modulation assignment show both an orange and green arc around the control.

This section focuses on Position, which is a modulation target. The principles discussed apply equally to other source parameter targets.

Position determines the playback position of audio data. When modulated, the playback path through the audio data is controlled by the selected modulation source. In sampler mode, the note on modulation value determines the initial offset for the play position within the audio data. Beginning at that position, the rest of the sound plays in a normal manner, although looped as if the Loop mode is set to All. In additive, spectral, or granular mode, Position can be continuously modulated forward or backward at any rate (including zero).

Create tempo-synced loops by modulating position

Synchronized playback of looped audio with the Logic Pro tempo is easy to achieve by modulating the Position parameter. This technique is possible with any synthesis method that permits continuous modulation of Position. This example uses the granular engine, but the same technique can be applied to the additive and spectral engines.

You can set Position and Speed to 0% and can modulate Position with a unipolar LFO, MSEG, or a performance control linked to the modulation wheel. When Position is modulated and Speed has a value greater than zero, the playback path is determined by a combination of modulation value (whenever this value changes) and the normal path (at a rate determined by Speed whenever the modulation value is static).

1. In Alchemy in Logic Pro, click the File button in the Name bar and choose Initialize Preset from the pop-up menu to reset all Alchemy parameters to default settings.

   Important: Use of Initialize Preset automatically turns on wide unison mode. See Alchemy wide unison mode.

2. In source A, perform a Granular import of a rhythmic or melodic sample that loops evenly.
3. Turn the source A Speed knob down to its minimum value of 0%, and play a few notes.

   You will hear that playback is frozen at the very beginning of the sample.

4. Do one of the following:

   - Control-click the Position knob, then choose Add Modulation > LFO > LFO 1 from the shortcut menu.

     Note the orange arc that appears around the Position knob. This indicates that the parameter has a modulation assignment.

   - Click the Position knob to select it as a modulation target, then click the first modulation rack slot, and choose LFO > LFO 1 from the modulation source pop-up menu. See Modulation rack controls.

5. In the modulation rack, turn the modulation rack slot 1 Depth knob clockwise. Set a value of 100%.

6. Click the LFO button in the modulation section if the LFO controls are not visible, then choose Basic > Ramp Up from the LFO Shape selection pop-up menu. Also in the LFO, turn off the Bipolar button.

   This routing increases Position smoothly so that the entire sample plays back from beginning to end, then jumps immediately back to the beginning and continues to loop.

7. Adjust the LFO Rate if loop playback is too fast.

8. Finally, adjust the Logic Pro tempo as you play additional notes to confirm that the loop is properly synchronized.

   Note: It is unnecessary to freeze playback by setting Speed to 0% (as performed in step 3) before modulating Position with the Ramp Up LFO shape. This is because this LFO shape is constantly changing, without any static/flat portions and thus is always controlling play position, making the Speed value irrelevant. If you use a different LFO shape with static/flat portions, however, setting Speed to a non-zero value causes play position to be controlled by a combination of the LFO shape and the normal playback behavior of the sample.

Alchemy morph controls in Logic Pro for Mac

Source components are shown only in advanced view. Click the Advanced button to switch to advanced view, then click the Morph button to view and use the morph controls.

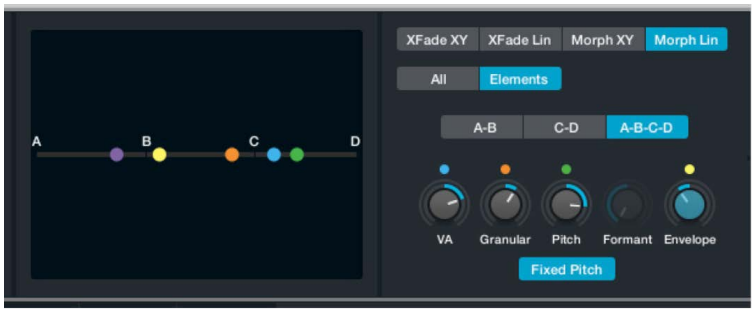
The morph controls determine how the four Alchemy sources interact. There are two basic types of interaction:

- *XFade:* In a crossfade, all active sources play simultaneously, and the X and Y knobs control the mix between them. This is equivalent to turning the Amp knobs in each source to attain the desired mix. If you crossfade from a source with a high Coarse Tune setting to a source with a low Coarse Tune setting, the high source fades out as the low source fades in. In the middle of the crossfade you hear both sources.
- *Morph:* In a morph, a single sound is generated with parameter values determined by interpolating between the settings of all four sources. If you morph from a source with a high Coarse Tune setting to a source with a low Coarse Tune setting, you hear a single sound during the morph. The sound tuning falls smoothly from the high value to the low one.

Morphing provides more scope than simple crossfades between sources. It also allows cross-synthesis, where you can combine different aspects of different sound elements. For example, you could apply the formants or other characteristics of an additive source to the spectral element of another source. See the tutorials found in Elemental morphs overview.

Only certain parameters, notably all parameters controlled with a knob, can be morphed. Morphed parameters are indicated by a green arc around the control. Parameters that are morphed and have a modulation assignment show both an orange and green arc around the control. Parameter settings are shared across all morphed sources, which means that changing a parameter in one source results in the corresponding parameter being changed for all morphed sources.

Note: Parameters that do not directly participate in the morph, including most buttons and pop-up menus, are indicated with a lock icon displayed at the top left of the control (the lock icons are shown only in source subpages). Where there is a parameter pairing of an On button and a pop-up menu, only the button shows the lock icon. Neither parameter participates in the morph.



Morph controls

- *Morph type buttons:* Choose one of four crossfade or morph modes.
  - *XFade XY:* This mode crossfades levels between all four sources.
    - Use the X knob to control the mix levels of sources A and C versus sources B and D.
    - Use the Y knob to control the mix levels of sources A and B versus sources C and D.
  - *XFade Linear:* This mode is useful for setting up crossfades based on Velocity or Key Follow.
    - Crossfade from source A to B to C to D as you increase the X knob value from 0% to 100%.
    - The Y knob is not shown in this mode.
  - *Morph XY:* This mode is like XFade XY, but all parameters of the sound are morphed. Regions of each source encompassed by corresponding warp markers are time-aligned in the morph. See Zone waveform editor.
    - *All/Elements buttons:* Use All to view and edit the X/Y values for all active parameters. Use Elements to view and edit the X/Y values of five parameters.
    - *X/Y knobs:* Edit the X/Y values for all active parameters. Y values are not shown in "linear" modes.
    - *Additive/VA knob X/Y:* Set the morph value for active additive sources. Also controls VA morph position if the VA element is active.
    - *Spectral/Granular knob X/Y:* Set the morph value for active spectral or granular sources. Also controls sampler morph position if the sampler element is active.
    - *Pitch knob X/Y:* Set the pitch morph value between active sources.

- *Formant knob X/Y:* Set the formant level morph value between active sources. Also controls morphing of the source filter knobs.
- *Envelope knob X/Y:* Set the envelope morph value between active sources. These controls morph the timing of the sound. In cases where source A has a short attack and source B has a long attack, for example, the length of the attack varies as you change the X knob.
- *Auto Gain button:* Turn on to match source levels, resulting in smoother morphs.
- *Fixed Pitch button:* Turn on to lock the root note to a pitch that doesn't change, regardless of morph position movements. Turn off for the best morphing quality.
- *Auto Align button:* Turn on to automatically align all morphed sources. As an example, Auto Align corrects the timing of words of four spoken voice samples saying the same phrase in each of the four morphed sources. Auto Align is automatically turned off when you set warp markers manually.

- *Morph Linear:* This mode is like XFade Linear, but all parameters of the sound are morphed. Regions of each source encompassed by corresponding warp markers are time-aligned in the morph. See Elemental morphs overview.
  - *All/Elements buttons:* Use All to view and edit the X value for all active parameters. Use Elements to view and edit the X values of five parameters.
  - *Source group buttons:* Choose source group A-B, C-D, or A-B-C-D. Morphs affect only the chosen group of sources.
  - *Additive/VA knob:* Set the morph value for active additive sources.
  - *Spectral/Granular knob:* Set the morph value for active spectral or granular sources. This also controls the sampler morphing position if the sampler element is active.
  - *Pitch knob:* Set the pitch morph value between active sources.
  - *Formant knob:* Set the formant level morph value between active sources.
  - *Envelope knob:* Set the envelope morph value between active sources.
  - *Auto Gain button:* Turn on to match source levels, resulting in smoother morphs.
  - *Fixed Pitch button:* Locks the root note to a pitch that does not change, regardless of morph position movements. Turn off for higher morphing quality.
  - *Auto Align button:* Turn on to automatically align all morphed sources. Auto Align is automatically turned off when you set warp markers manually. For example, Auto Align corrects the timing of words of four spoken voice samples saying the same phrase in each of the four morphed sources.

- *Morph square:* Display and edit the current X/Y morph position. Drag the point to change the X or Y value, or both.
  - *Morph XY mode:* When the Elements button is on, five points represent the following parameters: Additive (blue), Spec/Gran (orange), Pitch (green), Formant (purple), and Envelope (yellow). Drag each point to change the corresponding X and Y values.
  - *Morph Linear mode:* When the Elements button is on, five points represent the following parameters: Additive (blue), Spec/Gran (orange), Pitch (green), Formant (purple), and Envelope (yellow). Drag each point to change the corresponding X parameter value for the active source group: A-B, C-D, or A-B-C-D.

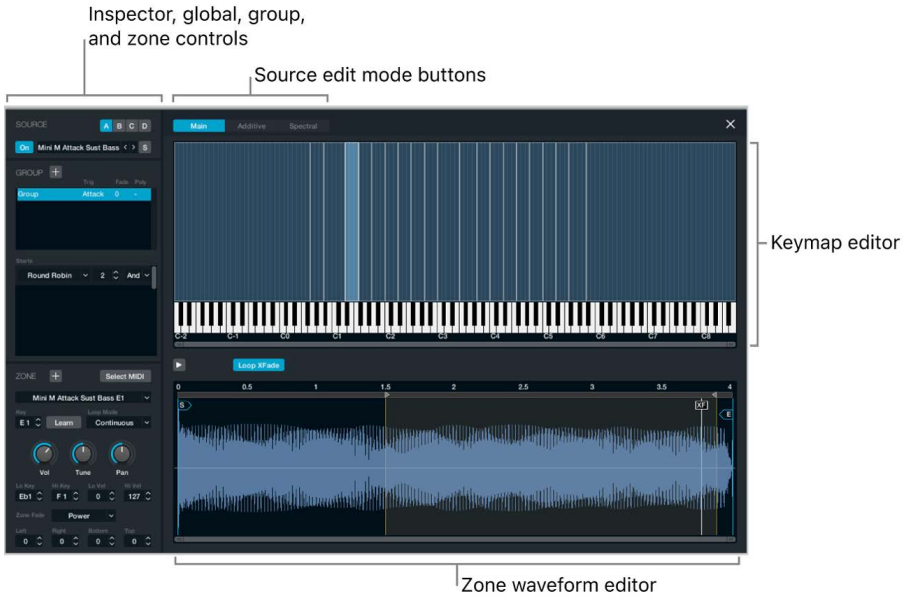
Alchemy source editors

Alchemy source edit window in Logic Pro for Mac

Source components are shown only in advanced view. Click the Advanced button to switch to advanced view, then click the A/B/C/D button to view detailed parameters for the selected source.

- *Edit button:* Open the source edit window. Each source subpage has its own Edit button.
- *Close window button:* Close the source edit window. Shown at the top right of all edit windows.

By default, the Main edit window is shown. Switch to Additive and Spectral edit windows by clicking the buttons at the top of the window. These windows provide additional parameters that let you precisely edit and sculpt your sounds. See Additive edit window overview and Spectral edit window.



The Main edit window is divided into three areas that interact with each other. You can edit parameters graphically in the keymap or waveform editor or can use corresponding fields and other parameters in the inspector.

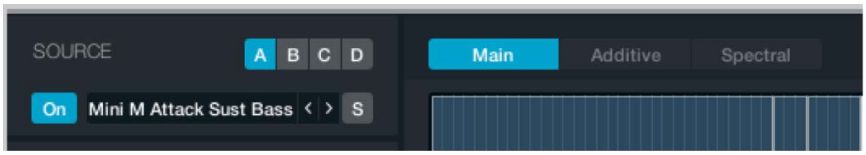
- *Inspector:* This area shows global parameters, group functions, and zone parameters. See Inspector global controls, Inspector group controls, and Inspector zone controls.
- *Keymap editor:* A graphical representation of, and editor for, sample zones. This area interacts with the zone parameters in the inspector. See Keymap editor.
- *Zone waveform editor:* A graphical representation of, and editor for, the waveform used in each sample zone. See Zone waveform editor.

Alchemy inspector global controls in Logic Pro for Mac

Source components are shown only in advanced view. Click the Advanced button to switch to advanced view, then click the A/B/C/D button to view detailed parameters for the selected source.

The source edit window is opened by clicking the Edit button on any source subpage. Click the close window icon (X) to close the source edit window.

The source inspector is divided into three main parameter groupings: global and source parameters, group parameters, and zone parameters. See Inspector group controls and Inspector zone controls.



Inspector global parameters

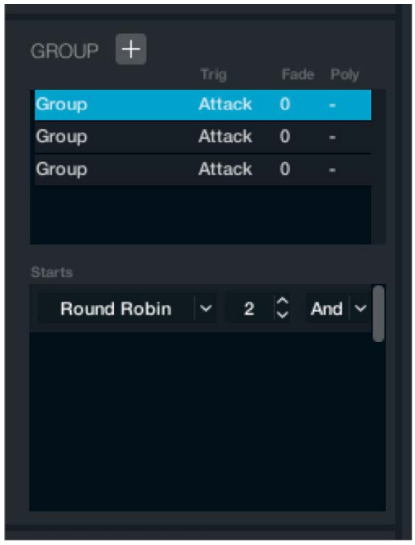
- *A/B/C/D buttons:* Select the corresponding source.
- *Main/Additive/Spectral buttons:* Show the Main, Additive, or Spectral edit window. Click the close button at the top right of the active window to close it.
- *On/Off button:* Turn the source selected with the A/B/C/D buttons on or off.
- *Source select pop-up menu:* Displays the name of the current source audio data. Click to open a pop-up menu with source content handling commands. You can also click the Previous and Next buttons (the arrows) to step through available waveform data. See Source master controls and Import browser.
- *Solo button:* Solo the source selected with the A/B/C/D buttons. All other sources are muted.

Alchemy inspector group controls in Logic Pro for Mac

Source components are shown only in advanced view. Click the Advanced button to switch to advanced view, then click the A/B/C/D button to view detailed parameters for the selected source.

The source edit window is opened by clicking the Edit button on any source subpage. Click the close window icon (X) at the top right to close the window.

The source inspector is divided into three main parameter groupings: global and source parameters, group parameters, and zone parameters. See Inspector global controls and Inspector zone controls.



Inspector group parameters

- *Group (+) pop-up menu:* Open a pop-up menu with two group creation commands.
  - *Empty:* Create a new empty group. The new group name is added below existing group names in the Group list shown under the Group pop-up menu. A sequentially assigned number is appended to the new group name.
  - *Import:* Open an import window where you can choose one or multiple samples to import as a group. See Import browser.
- *Keyswitch knob:* Use to switch between keyswitch groups that you have defined in the Starts pop-up menu.
- *Group list:* Double-click a group name to edit the name in a text field and change the Trigger, Fade, and Poly values. Control-click a group name to choose the Delete command.
  - *Trigger pop-up menu:* Set a trigger mode for all zones in the group. Attack triggers group zones when note on messages are received. Release triggers group zones for note off messages. This mode is useful for instruments with a distinctive end-of-note sound such as a key click or hammer thump. Such sounds can be included as a separate group with release triggering turned on. You can also use this feature creatively to add abstract reverb tails or to create a pad sound that changes dramatically during the release stage.

- *Fade pop-up menu:* Apply a fade *out* time value ranging from 0 to 100 for note off events when the *attack* trigger mode is active. Apply a fade *in* time value ranging from 0 to 100 for note off events when the *release* trigger mode is active. This parameter is primarily intended for use in conjunction with release triggering, to create a crossfade between the main body of the note and the release sample.
- *Poly pop-up menu:* Set a maximum polyphony value for the group. A common use of this feature is to create a group containing an open and a closed hi-hat sample. If you set group polyphony to one, either the closed or open hi-hat sample can play, but not both at the same time.

- *Starts pop-up menu:* Define when each group will trigger. You can create and combine multiple rules using boolean logic. The normal state for any newly created group is a single rule set to Always, unless that group was defined as a round robin group in the Import browser Dropzone. Click the field to choose a rule. This adds a new rule pop-up menu below the first and displays a condition pop-up menu to the right. Choose Delete from the pop-up menu to remove a rule.
  - *Always:* Zones in the group always trigger when a note on event arrives within the specified zone key and velocity ranges.
  - *Round Robin:* Choose to automatically switch between layers for each note on, so that striking a given note multiple times triggers variations of the sample rather than the same sample being played repeatedly.
    - *Order pop-up menus:* Set the playback order of round robin groups. At least two round robin groups must exist for this parameter to have an effect. Each group is assigned a different value. If two groups are assigned to the same value, both zones are triggered simultaneously. Each note-on sequentially triggers a round robin group from lowest to highest. Once the highest group number is triggered, the sequence starts again from the lowest group number. Drag vertically in the field or use the arrows to set a value.
  - *Random Round Robin:* Choose to automatically switch between layers for each note on, so that striking a given note multiple times triggers random variations of the sample rather than the same sample being played repeatedly. No Order pop-up menu is shown.
  - *Keyswitch:* Choose to switch a group on or off in accordance with either the Transform pad position or the Keyswitch parameter conditions. You can also assign keyswitching to a MIDI note or range of notes which prevents the group from triggering until one of these notes is played. Two pop-up menus are displayed, where you can choose keyswitch conditions.
    - *SNAP 1 - SNAP 8 pop-up menu:* These values represent Transform pad positions. For example, set the first field to SNAP 2 and the second field to SNAP 4, which results in the group being triggered only when the Transform pad is at position 2, 3, or 4. Assign multiple groups to different ranges to switch between up to eight different groups. .

- *KEYSW1 - KEYSW10 pop-up menu:* These values represent the position of the Keyswitch knob shown in the group list above. This knob is visible only when the KEYSW1-KEYSW10 options are chosen in these menus. For example, set the first field to KEYSW1 and the second to KEYSW5, which chooses the group being triggered when the Keyswitch knob is set to a value that falls within this range. Up to 10 Keyswitch knob positions are available, allowing you to switch between groups with the knob. Because the Keyswitch knob is available as a modulation target, this lets you create complex automated group switches.
- *C2 - G8 pop-up menu:* Set a specific note or range of notes as keyswitches. Play a note in this range to switch to a group, which remains active until another group is chosen.

- *Controller:* Switch groups on or off in accordance with specified performance controls in the Perform section, or in accordance with assigned MIDI controller values. Three pop-up menus are displayed where you can choose the controller type and set controller values.
  - *Type pop-up menu:* Choose Control 1 to Control 8 to control switching from any of the eight performance control knobs. Choose the XYPad options to control switching from the X or Y axes of either of the performance control X/Y pads. Choose the Env Attack, Env Decay, Env Sustain, or Env Release option to assign switching to the performance control envelope knobs. See Performance controls overview. You can also choose from any of the 128 MIDI controllers CC#0 to CC#127.
  - *Value pop-up menus:* Define the range of values that trigger the group. For example, set the type to Control 1, with field values of 30 and 50, which results in the group being triggered only when performance control 1 is set within the 30% to 50% range. You can set other groups with the same type but with a different control range to switch between groups with a single controller. Because the Control 1 knob used in the example is available as a modulation target, this lets you create complex automated group switches. Drag vertically in the fields or use the arrows to set a value.
- *Delete:* Remove the group rule.
- *Condition pop-up menus:* Define a condition for multiple group rules, using *And*, *Or*, and *Not* boolean statements. For example, set the first group rule to Controller and Control 1 with a range of 50% to 100%. Set a second group rule to Controller and Control 2 with a range of 50% to 100%. The different logic conditions result in different outcomes.
  - *And:* Both Control 1 and Control 2 must be set above 50% for the group to trigger. When either control is set below 50%, the group does not trigger.
  - *Or:* Either Control 1 or Control 2 must be set above 50% for the group to trigger. The group does not trigger when both controls are below 50%.
  - *Not:* The group triggers when Control 1 is set above 50%, and Control 2 is set below 50%. If both controls are set above or below 50%, the group does not trigger.

Crossfade to a different group of samples on note off

1. In Alchemy in Logic Pro, click the File button in the Name bar and choose Initialize Preset from the pop-up menu to reset a parameters to default settings.

   Important: Use of Initialize Preset automatically turns on wide unison mode. See Alchemy wide unison mode.

2. Switch to advanced view, and in the modulation section click the AHDSR tab. Set the Release time for the envelope to a suitable length for your release samples.
3. Switch to one of the sources with the A/B/C/D buttons, then click the Edit button to open the Main edit window.
4. Click the source select pop-up menu, and choose Import Audio.
5. Select a sample (or multiple samples) representing the main sustain portion of your sound, and import using any of the available import modes.

   Alchemy analyzes each sample to determine the root pitch (if not defined in the filename), set the root key, key range, and velocity range for each sample zone such that they span the entire keyboard and the entire dynamic range, and add all zones to a group named *Group 1*.

6. Click the "+" symbol at the top of the Group section, and choose Import.
7. Select a sample (or multiple samples) representing the release portion of your sound, and import (the import mode is automatically set to match the existing group).

   Alchemy again analyzes each sample and adds all zones to a group named *Group 2*.

8. Double-click Group 2, then click the Trig field and change it to Release.

   Zones in Group 2 will now trigger when you release each key, playing over zones in Group 1, which continue to sound until AHDSR 1 reaches the end of the release stage.

9. Double-click Group 1 in the list, then click the Fade field and change it to a value other than 0.

   Zones in Group 1 will now fade out when the note is releasing, allowing Group 2 to be heard during the release stage of the sound. Higher Fade values result in slower fades.

10. As an option, double-click Group 2 in the list, then click the Fade field and change it to a value other than 0.

    Zones in Group 2 will now fade in when the note is released, creating a crossfade between Groups 1 and 2 at note off. Higher values result in slower fades. Fading in the release group may be unnecessary if your release samples already have a natural fade in at the start, however, or undesirable if a percussive transient is required at note off. Try to set Fade values for Groups 1 and 2 to an identical small value to create a sudden, but click-free, crossfade at the end of each note.

Create random round robin variations

1. In Alchemy in Logic Pro, click the File button in the Name bar and choose Initialize Preset from the pop-up menu to reset all Alchemy parameters to default settings.

   Important: Use of Initialize Preset automatically turns on wide unison mode. See Alchemy wide unison mode.

2. Switch to advanced view, and set the Release time for AHDSR 1 to a suitable length for your release samples.
3. Switch to one of the sources with the A/B/C/D buttons, then click the Edit button to open the Main edit window.
4. Click the source select pop-up menu, and choose Import Audio.
5. Select a sample (or multiple samples) representing the main sustain portion of your sound, and import using any of the available import modes.

   Alchemy analyzes each sample to determine the root pitch (if not defined in the filename), set the root key, key range, and velocity range for each sample zone such that they span the entire keyboard and the entire dynamic range, and add all zones to a group named *Group 1*.

6. Click the Rule field, and change it from Always to Random Round Robin.
7. Click the "+" symbol at the top of the Group section, and choose Import.
8. Select a sample (or multiple samples) representing the second of your round robin variations, and import (the import mode is automatically set to match the existing group).

   Alchemy will again analyze each sample and add all zones to a group named *Group 2*.

9. Click the Rule field below Group 2, and change it from Always to Random Round Robin.

   Any notes you play will now randomly trigger either group 1 or group 2, but not both together. Note that if you play a chord, each individual note is randomly assigned to one of those groups.

10. Repeat steps 7 to 9 as needed to configure a group for each further variation you require.

Assign the source Keyswitch knob to switch between groups

1. In Alchemy in Logic Pro, click the File button in the Name bar and choose Initialize Preset from the pop-up menu to reset all Alchemy parameters to default settings.

   Important: Use of Initialize Preset automatically turns on wide unison mode. See Alchemy wide unison mode.

2. Switch to advanced view, and set the Release time for AHDSR 1 to a suitable length for your release samples.
3. Switch to one of the sources with the A/B/C/D buttons, then click the Edit button to open the Main edit window.
4. Click the source select pop-up menu, and choose Import Audio.
5. Select a sample (or multiple samples) representing the main sustain portion of your sound, and import using any of the available import modes.

   Alchemy analyzes each sample to determine the root pitch (if not defined in the filename), set the root key, key range, and velocity range for each sample zone such that they span the entire keyboard and the entire dynamic range, and add all zones to a group named *Group 1*.

6. Click the Rule field, and change it from Always to Keyswitch.
7. Click the first range field, and change it to Keysw1. The second range field also changes to the same value.
8. As an option, you can click the second range field and increase the value to specify a range of values that will trigger this group, instead of just a single value.
9. Click the "+" symbol at the top of the Group section, and choose Import.
10. Select a sample (or multiple samples) representing the second of your variations, and import (the import mode is automatically set to match the existing group).

    Alchemy will again analyze each sample and add all zones to a group named *Group 2*.
11. Click the Rule field below Group 2, and change it from Always to Keyswitch.
12. Click the first range field, and change it to the first unused Keysw value. If the second range field for Group 1 is set to Keysw3, choose Keysw4.
13. As an option, you can click the second range field and increase the value to specify a range of values that will trigger Group 2, instead of just a single value.
14. Repeat steps 9 to 12 for each new group of variations you require.
15. Rotate the Keysw knob to switch between the groups you created. Control-click this knob to add modulation routings from the shortcut menu.
16. Click the close button at the top right to close the source edit window.

Create round robin variations for just one Transform pad position

1. In Alchemy in Logic Pro, click the File button in the Name bar and choose Initialize Preset from the pop-up menu to reset all Alchemy parameters to default settings.

    Important: Use of Initialize Preset automatically turns on wide unison mode. See Alchemy wide unison mode.
2. Switch to advanced view, and set the Release time for AHDSR 1 to a suitable length for your release samples.
3. Switch to one of the sources with the A/B/C/D buttons, then click the Edit button to open the Main edit window.
4. Click the source select pop-up menu, and choose Import Audio.
5. Select a sample (or multiple samples) representing the main sustain portion of your sound, and import using any of the available import modes.

    Alchemy analyzes each sample to determine the root pitch (if not defined in the filename), set the root key, key range, and velocity range for each sample zone such that they span the entire keyboard and the entire dynamic range, and add all zones to a group named *Group 1*.
6. Click the Rule field, and change it from Always to Round Robin.

    It is automatically assigned the number 1 in the sequence, and a second rule (set to Always) is added below.
7. Click the Logic field for rule 1, and change it to Or.
8. Switch rule 2 from Always to Keyswitch.
9. Leave the first range field for rule 2 set to Snap1, and change the second range field to Snap7.

10. Click the "+" symbol at the top of the Group section, and choose Import.
11. Select a sample (or multiple samples) representing the second of your round robin variations, and import (the import mode is automatically set to match the existing group).

    Alchemy again analyzes each sample and adds all zones to a group named *Group 2*.
12. Click the Rule field below Group 2, and change it from Always to Round Robin.

    It is automatically assigned the number 2 in the sequence, and a second rule (set to Always) is added below.
13. Switch rule 2 from Always to Keyswitch.
14. Set the first Range field for rule 2 to Snap8. The second field is automatically set to the same value.

    Played notes only trigger zones in Group 1 unless the Transform pad is at position 8, in which case Groups 1 and 2 play alternately, in a round robin fashion.
15. Repeat steps 10 to 14 as needed to configure a group for each further variation you require.

Alchemy inspector zone controls in Logic Pro for Mac

Source components are shown only in advanced view. Click the Advanced button to switch to advanced view, then click the A/B/C/D button to view detailed parameters for the selected source.

The source edit window is opened by clicking the Edit button on any source subpage. Click the close window icon (X) at the top right to close the window.

The source inspector is divided into three main parameter groupings: global and source parameters, group parameters, and zone parameters. See Inspector global controls and Inspector group controls.



Inspector zone parameters

- *Add Zone (+) button:* Open the Import browser, where you can select one or multiple samples to import into zones. Newly imported samples are added to the current configuration. Each new sample results in a new zone. See Import browser.
- *Select MIDI button:* Turn on to automatically select zones by playing your MIDI keyboard.

- *Zone name field:* Displays the name of the current audio file in the selected zone. Click to open a pop-up menu where you can select other source zones.

  Note: The keymap editor lets you select multiple zones simultaneously. In this case the zone name field shows *Multiple* to indicate a multi-zone selection. Any parameter changes are applied to all selected zones. See Keymap editor.

- *Key field:* Set the root key for the selected zone with the up and down arrows. The chosen note triggers playback of the sound at its original pitch. A pitched sample should ideally be mapped to a matching root key. If the root note is defined as part of the filename, the Key parameter is set accordingly when the file is imported. If the root note is not in the filename, samples are analyzed on import to calculate their original pitch and a suitable root key setting. Drag vertically in the field or use the arrows to set a value.

- *Learn button:* Turn on to set the root key for a sample: the next MIDI note received defines the new root key. The button is automatically turned off once the root key has been learned.

- *Loop Mode field:* Open a pop-up menu with seven looping options.

  - *None:* Ignores the loop start and end points, and plays the entire sound once without looping.

  - *Continuous:* Plays from the beginning, enters the loop region, and loops continuously in a forward direction while a note is held and during the envelope release phase.

  - *Sustain:* Plays from the beginning, enters the loop region, loops continuously while a note is held, and exits the loop region to play the normal sound release phase.

  - *Forward/Back:* Like Continuous, but plays the loop region alternately forward and backward.

  - *Start/End:* In this mode, playback starts from the defined start point at 100% speed to the defined loop end position, then skips to the start point and repeats. These positions are set with the Loop Start and Loop End knobs in the source master controls.

    Note off messages have no impact on loop repetitions. If the loop end point is placed before the loop start point, playback is reversed (backward to the loop end point) when the loop start point is reached.

  - *Start/Length:* In this mode, playback starts from the defined start point at 100% speed to the defined loop end position, then skips to the start point and repeats. The length of the loop, controlled with the Loop Length knob in the source master controls, is set as a percentage of the overall length of the sound, such that the loop end point is equal to the loop start point plus the loop length.

    If the settings of Loop Start and Loop Length controls in the source master controls cause playback to go beyond the end of the underlying sample, silence is automatically inserted at the end of the sound.

  - *All:* Ignores the loop start and end points, and loops the entire sound continuously.

    Note: The loop start and loop end points can be edited in the source main edit page. See Zone waveform editor. In VA mode, a raw oscillator noise source is used rather than loopable data, so VA synthesis elements are not affected by the loop mode setting.

- *Volume knob:* Set the selected zone output level. This parameter is also useful for level-matching multiple selected zones.
- *Pan knob:* Determine the left/right pan position for each zone or the left/right balance for a stereo zone.
- *Coarse/Fine Tune knobs:* Tune the zone in semitones (Coarse) and cents (Fine=1/100 of a semitone).
- *Lo Key field:* Drag vertically or use the arrows to set the lowest MIDI note that triggers the zone. Alternatively, drag the left edge of the highlighted zone in the keymap editor. See Keymap editor.
- *Hi Key field:* Drag vertically or use the arrows to set the highest MIDI note that triggers the zone. Alternatively, drag the right edge of the highlighted zone in the keymap editor.
- *Lo Vel field:* Drag vertically or use the arrows to set the lowest MIDI note velocity that triggers the zone. Alternatively, drag the lower edge of the highlighted zone in the keymap editor.
- *Hi Vel field:* Drag vertically or use the arrows to set the highest MIDI note velocity that triggers the zone. Alternatively, drag the upper edge of the highlighted zone in the keymap editor.
- *Zone Fade pop-up menu:* Determine the method used to fade overlapping zones at the boundaries of key and velocity ranges. No more than two zones can be triggered simultaneously within one group, which means that crossfades can be set for zones with adjacent key ranges *or* for zones with adjacent velocity ranges, but not for both at the same time.
  - *Linear:* Create a linear ramp from full amplitude to zero. This typically results in a crossfade that sounds quieter at the halfway point. Linear mode can be useful when crossfading two highly similar sounds that may reinforce one another at the halfway point of an equal-power crossfade.
  - *Power:* Create an equal-power curved fade that is suitable for most sounds. When crossfading two sounds with a similar loudness, Power mode results in a smooth fade with the same apparent loudness at the halfway point.
- *Left field:* Drag vertically or use the arrows to set the range of the fade for the left edge of the zone, relative to the Lo Key parameter. If Lo Key is set to C2 and Left is set to 12, the zone fades in gradually for notes with pitches between C2 and C3.
- *Right field:* Drag vertically or use the arrows to set the range of the fade for the right edge of the zone, relative to the Hi Key parameter. If Hi Key is set to C4 and Right is set to 5, the zone fades out gradually for notes with pitches between G#4 and C5.
- *Bottom field:* Drag vertically or use the arrows to set the range of the fade for the lower edge of the zone, relative to the Lo Vel parameter. If Lo Vel is set to 20 and Bottom is set to 15, the zone fades in gradually for notes with velocities between 20 and 35. Velocity values below 20 do not trigger the zone.
- *Top field:* Drag vertically or use the arrows to set the range of the fade for the upper edge of the zone, relative to the Hi Vel parameter. If Hi Vel is set to 90 and Top is set to 30, the zone fades out gradually for notes with velocities between 60 and 90. Velocity values above 90 do not trigger the zone.

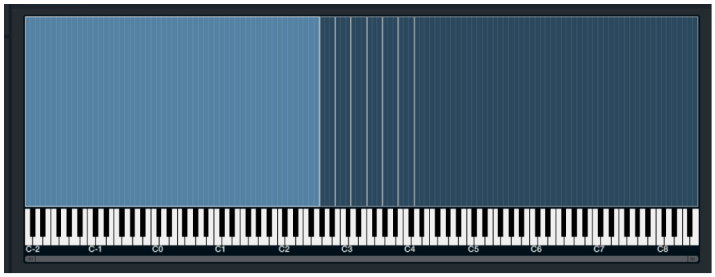
  Note: Zone fades and overlapping zones are not compatible with morphing sources. When morphing is turned on, only one zone is triggered at a time from each morphing source (the first zone in the list), and zone fades are turned off.

Alchemy keymap editor in Logic Pro for Mac

Source components are shown only in advanced view. Click the Advanced button to switch to advanced view, then click the A/B/C/D button to view detailed parameters for the selected source.

The source edit window is opened by clicking the Edit button on any source subpage. Click the close window icon (X) at the top right to close the window.

The keymap editor displays all sample zones in the currently selected group, with each zone represented as a rectangle. The size and location of the rectangle indicates the key and velocity range for the associated zone. The selected zone in the inspector is indicated by a white outline in the keymap editor. See Inspector zone controls.



- The left and right boundaries of the rectangle indicate the Lo Key and Hi Key parameters relative to the keyboard display below. Drag either boundary to set the Lo Key and Hi Key values.
- The bottom and top edges of the rectangle indicate the Lo Vel and Hi Vel parameters. Drag either the top or bottom edge to set the Lo Vel and Hi Vel values.
- Drag the middle of the scroll bar to view zones that are not visible in the display area. Horizontally drag the zoom controls at either end of the scroll bar to resize the contents of the visible display area.

Keymap editor menu commands

Control-click any zone to open a shortcut menu.

- *Cut:* Remove the selected zone or zones from the current group, and copy to the Clipboard.
- *Copy:* Copy the selected zone or zones to the Clipboard.
- *Paste to existing group:* Paste content from the Clipboard to an existing group chosen from the submenu.
- *Paste to new group:* Create a new group, and paste content from the Clipboard into it.
- *Delete:* Delete the selected zone or zones from the current group.
- *Exchange sample:* Open the Import browser where you can choose a replacement sample for the zone. The import mode used by the previous sample is retained, and the import mode buttons are dimmed. See Import browser.
- *Assign to existing group:* Move the selected zone or zones from the current group to an existing group chosen from the submenu.
- *Assign to new group:* Create a new group, and move the selected zone or zones from the current group into the new group.

Select and edit the values of one or more zones

1. In Alchemy in Logic Pro, select the zone by doing one of the following:
   - Click a rectangle in the keymap editor.
   - Choose the zone name from the pop-up menu in the zone section of the inspector.
2. Hold down Shift, then click multiple zones in the keymap editor to select them.

   All selected zones are highlighted with a white border, and Multiple is shown in the inspector zone name field.
3. Drag the left, right, top, or bottom edge of the zone. When multiple zones are selected, drag the edge of one selected zone to apply the same relative change to all other selected zones.

Delete one or more zones

1. In Alchemy in Logic Pro, select the zone (or zones) by doing one of the following:
   - Click a rectangle in the keymap editor.
   - Choose the zone name from the pop-up menu in the zone section of the inspector.
   - Hold down Shift, then click multiple zones in the keymap editor to select them.

     All selected zones are highlighted with a white border, and Multiple is shown in the inspector zone name field.
2. Control-click a zone, then choose Delete from the shortcut menu.

   The zone or zones are deleted. All associated sample, additive, or spectral data is removed from the source.

Audition a zone

Activate the > icon below the keymap editor to hear a preview whenever a zone is clicked. The sample is previewed, inclusive of loop settings but ignoring zone parameters such as Tune, Volume, and Panning.

In Alchemy in Logic Pro, select the zone by doing one of the following:

- Click a rectangle in the keymap editor.
- Click a key on the keyboard display below the zones.
  - Click near the top of a key to audition low velocities.
  - Click near the bottom of a key to audition high velocities.

    Audio is sent directly to the Alchemy outputs, bypassing any filters or effects that are turned on in other sections.

Create a multisample with key splits and crossfades for adjacent zones

1. In Alchemy in Logic Pro, click the File button in the Name bar and choose Initialize Preset from the pop-up menu to reset all Alchemy parameters to default settings.

   Important: Use of Initialize Preset automatically turns on wide unison mode. See Alchemy wide unison mode.
2. Click the Advanced button, then click the source A button.
3. Click the Edit button to open the Main source editor.

4. Click the source select field, and choose Import Audio from the pop-up menu.
5. In the Import browser, select two or more samples representing the same instrument played at different pitches, then click the Import button. Any import mode may be used.

   Alchemy analyzes each sample to determine the root pitch (if not defined in the filename) and sets the root key and key range for each sample zone accordingly.
6. Select a zone in the keymap editor, and drag its left or right edge so that it overlaps the adjacent zone. The amount of overlap determines the range of the crossfade.
7. Select the leftmost of the two overlapping zones, and adjust the Right parameter in the inspector zone section. If the overlap between the zones extends over four keys, set this parameter to 4.
8. Select the rightmost of the two overlapping zones, and set the Left parameter in the inspector zone section to the same value used in the previous step.

   These two zones will now crossfade from one to the other when you play notes in the overlapping region.
9. Import further samples, and repeat the previous three steps for each pair of zones you want to crossfade.

Add and adjust loop markers for a zone

1. In Alchemy in Logic Pro, select the zone you want to loop by doing one of the following:
   - Click a rectangle in the keymap editor.
   - Choose the zone name from the pop-up menu in the zone section of the inspector.
2. Click the Loop Mode field in the zone section of the inspector, then choose Continuous, Sustain, or Forward/Back.

   Loop start (right arrow) and end (left arrow) markers appear on the waveform display, with the loop region highlighted between them.
3. Drag the loop start marker handle (S) left or right to find a suitable loop start point.

   You can click and hold the loop start marker handle to zoom the waveform horizontally before dragging.
4. Drag the loop end marker handle (E) left or right to find a suitable loop end point.

   You can click and hold the loop end marker handle to zoom the waveform horizontally before dragging.
5. If required, turn on loop crossfading with the Loop XFade button, then drag the crossfade marker handle (XF) left to create a smooth loop. Crossfading the loop end with the loop start is used to avoid clicks and audible jumps.

   Note: The Loop XFade button is shown in sampler mode only. This parameter is not visible if other import modes are used. The sample must be imported in Sampler analysis mode, and the Sampler source element must be turned on if you want to use Loop XFade.

   Dragging the End marker handle (E) also moves the Xfade marker handle (XF).

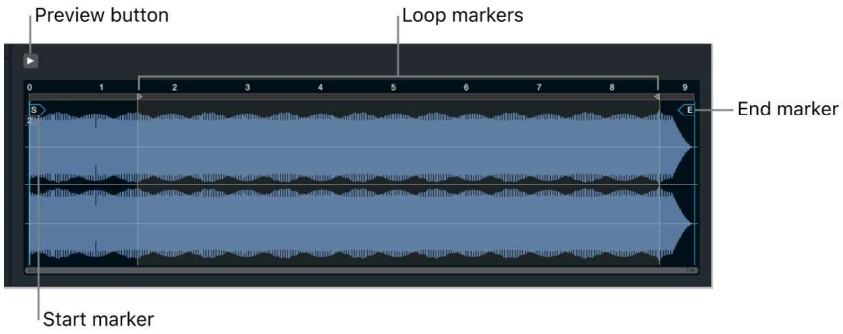
Alchemy zone waveform editor in Logic Pro for Mac

Source components are shown only in advanced view. Click the Advanced button to switch to advanced view, then click the A/B/C/D button to view detailed parameters for the selected source.

The source edit window is opened by clicking the Edit button on any source subpage. Click the close window icon (X) at the top right to close the window.

The waveform display of the audio associated with the currently selected zone is shown at bottom of the Main edit window. Zones can be selected in the keymap editor or zone parameters in the inspector. See Keymap editor and Inspector zone controls.

Note: If the additive or spectral engines are in use, the waveform display shows the resynthesized sound amplitude envelope rather than the original sample.



When morphing is turned on, vertical gray lines with numbered handles are displayed over the waveform. These are warp markers. For information on use of warp markers, see Manually time-aligning morphed sounds in Alchemy.

The bright vertical blue lines displayed over the waveform, labeled *S* and *E*, indicate playback start and end points. Drag the S or E handle right or left to trim unwanted audio playback from the start or the end of the sound.

Depending on the source material and active morph mode, light blue lines may appear on the background between the start and end markers. These can help with manual alignment of warp markers.

If the selected zone is set to one of the looping modes, loop start and loop end markers are displayed as faint orange lines, each with a small triangular handle shown in the ruler. The loop region is highlighted between these markers. The marker with the left-facing handle sets the loop end point. The marker with the right-facing handle sets the loop start point. Drag the triangular marker handles to adjust the loop start and end points. Click and hold the handle of a loop marker to zoom the waveform horizontally, allowing for more accurate placement.

Note: You cannot position the loop start point later than the loop end point. You can, however, drag both the loop start and loop end marker handles to the same position, creating a sustain point rather than a loop region. This option is available only in the additive or spectral engines, or with a sample loaded into the granular engine, but not in sampler mode.

Waveform display parameters

- *Preview (>) button:* Turn automatic audio previewing on or off when selecting zones in the keymap editor.
- *Loop XFade button:* Activate to turn on crossfades between loop start and end markers when Continuous or Sustain loop mode is chosen in the inspector.
  - A crossfade (XF) marker handle is shown at the loop end point. Drag this marker handle left to define a crossfade region, during which a portion of the sound from before the loop start marker is faded in, while the loop region is faded out.
  - In sampler mode, discontinuities between the loop start and end points can result in an audible click each time the loop restarts. Loops in additive, spectral, or granular modes do not click, but timbral or volume differences between the loop start and end points can often result in an obvious sounding loop. Unlike conventional samplers, the additive, spectral, and granular engines are capable of looping a single instant within the sound, with loop start and end markers at identical positions. This can work well with some sounds, but with others the loop stage may seem too static and obviously different from the start of the note.

  Tip: Looping clicks in sampler mode can usually be removed with minimal crossfades, while other timbral or volume discontinuities may need larger crossfades. Drag the XF marker handle while playing a note to find a suitable crossfade length.

  Note: The Loop XFade button is shown in sampler mode only. This parameter is not visible if other import modes are used. The sample must be imported in Sampler analysis mode, and the Sampler source element must be turned on if you want to use Loop XFade.
- *Time line:* Shows the overall length of the waveform in seconds. Fractional values (of the overall waveform length) are displayed in some situations, such as manual alignment of warp markers.
- *Scroll bar and zoom controls:* Drag the middle of the scroll bar to view waveform data that is not visible in the display area. Horizontally drag the zoom controls at either end of the scroll bar to resize the contents of the visible display area.

Alchemy additive edit window in Logic Pro for Mac

Source components are shown only in advanced view. Click the Advanced button to switch to advanced view, then click the A/B/C/D button to view detailed parameters for the selected source.

The source edit window is opened by clicking the Edit button on any source subpage. Click the close window icon (X) at the top right to close the window.

Use the Additive button at the top of the Main edit window to switch to the Additive edit window.

The Additive edit window allows detailed editing of additive resynthesis data and also lets you design sounds from scratch by creating the additive data yourself.

Additive synthesis represents each sound as a sum of individual partials (which are individual sine waves). The additive data describes each partial in terms of four parameters: amplitude, tune (pitch), pan, and phase, each of which changes over time.

You can think of the additive data as a series of snapshots, each of which captures the amplitude, pitch, pan, and phase of every partial at a particular point in time. In between snapshots, each parameter updates smoothly toward the following snapshot value. When played in succession and with the right timing, the series of snapshots describes a potentially complex and continuously evolving sound.

Note: Phase is not an independent parameter. The phase of a partial at any moment in the evolution of a sound is determined by the phase at the start point of the sound and by the (possibly changing) pitch of the partial. Because pitch information is captured in each snapshot, it isn't necessary for phase information to be captured in the same way. The phase of each partial is specified only at the absolute start point of the sound.



The Additive edit window shows additive data in two graphical displays.

- *Partial bar display:* Shows a series of vertical columns, or bars. Levels represent the amp, pitch, pan, or phase values of *individual partials* or *partial groups*, depending on the chosen display mode. See Partial bar display.
- *Partial envelope display:* Shows a number of points that operate in two ways, depending on the chosen display mode. See Partial envelope controls.

Alchemy partial bar display in Logic Pro for Mac

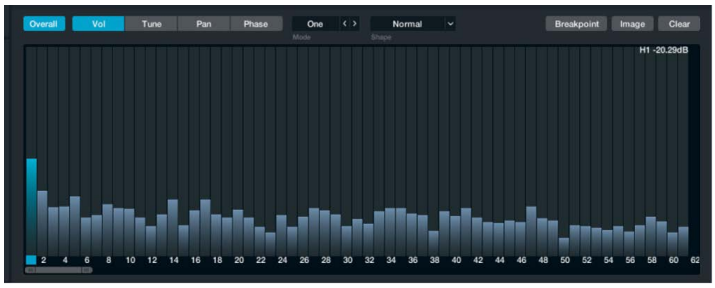
Source components are shown only in advanced view. Click the Advanced button to switch to advanced view, then click the A/B/C/D button to view detailed parameters for the selected source.

The source edit window is opened by clicking the Edit button on any source subpage. Click the close window icon (X) at the top right to close the window.

Use the Additive button at the top of the Main edit window to open the Additive edit window.

The Partial bar display shows up to 600 partials with independent amplitude, pitch, pan, and phase values at each Partial envelope point. You can edit individual partials or groups of related partials for each envelope point.

Note: Higher partials that are part of the additive data set may not be heard unless the Num Partials (number of partials) control on the respective source A/B/C/D subpage is set to a suitably high value. For example, raising the amplitude of partial number 72 has no effect when Num Partials is set to a value of 60. Conversely, setting a Num Partials value of 500 has no effect in additive mode unless partial data exists for 500 partials.



Partial bar display parameters

- *Overall button:* Use in conjunction with the Vol, Tune, Pan, and Phase buttons to display partial values averaged across all envelope points. When the Phase button is active, the Partial envelope only is updated.

  Turn on Overall to adjust the amplitude, pitch, and pan values of all partials across the entire sound, without the need to select and edit individual envelope points for each partial. Also see the Overall button information for the Partial envelope controls.

- *Partial display mode buttons:* Click Vol, Tune, Pan, or Phase to restrict the display to the chosen partial parameter type.
  - *Vol button:* Show partial amplitude values.
  - *Tune button:* Show partial pitch values. Also see Snap Pitch button information below.
  - *Pan button:* Show partial pan values.
  - *Phase button:* Show partial phase values.
- *Mode pop-up menu:* Choose a group of related partials for editing. You can also click the Previous and Next buttons (the arrows) to step through modes. Group adjustments retain the relative differences between partial values. Hold down Command while dragging to scale adjustments logarithmically. Note that the Shape pop-up menu affects editing behavior in each mode.
  - *One:* Adjustments affect the value of only the selected partial.
  - *All:* Adjustments affect the values of all partials.
  - *Even:* Adjustments affect the values of only even-numbered partials.
  - *Fifths:* Adjustments affect the values of only partials that are fifths apart from the selected partial and the selected partial itself.
  - *Octaves:* Adjustments affect the values of only partials that are octaves apart from the selected partial and the selected partial itself.
  - *Odd:* Adjustments affect the values of only odd-numbered partials.

- *Shape pop-up menu:* Choose a weighted group of related partials for editing.
  - *Normal:* Adjustments affect only the selected partial and grouped partials that precede it.
  - *Flat:* Adjustments evenly affect *all* grouped partials.
  - *Bright:* Adjustments are weighted toward higher grouped partials.
  - *Dark:* Adjustments are weighted toward lower grouped partials.
- *Snap Pitch button:* Visible only when the Tune button is active. Automatically limits partial pitch values to semitones when a bar is edited.
- *Breakpoint button:* Open a pop-up menu with a number of Partial envelope point-related commands. See Partial envelope controls.
  - *Copy Breakpoints:* Copy all partial data in the currently selected envelope point or points to the Clipboard.
  - *Paste Breakpoints:* Paste all partial data from the Clipboard to the currently selected envelope point or points.
  - *Paste Breakpoints Amp/Pitch/Pan Data:* Paste the currently visible parameter partial data from the Clipboard to the currently selected envelope point or points.
  - *Paste All Breakpoints Amp/Pitch/Pan Data:* Paste the currently visible parameter partial data from the Clipboard to every envelope point.
- *Image button:* Open an import dialog where you can choose a PNG format image file for conversion to additive synthesis data. See the task in this section.
- *Clear button:* Removes all partials and resets the Partial bar display and Partial envelope to default values.
- *Partial bar display:* Shows editable bars that represent partial values.
- *Partial number bar:* Indicates the selected partial. Click a partial select it without affecting any values.
- *Scroll bar and zoom controls:* Drag the middle of the scroll bar to view partials that are not visible in the display area. Horizontally drag the zoom controls at either end of the scroll bar to resize the contents of the visible display area.

Adjust the values of one or more partials

In Alchemy in Logic Pro, from the Partial bar display, do any of the following:

- Click directly at a position in a bar to set a value.
- Drag a bar vertically. Hold down Shift while dragging to set values with enhanced precision.
- Drag horizontally to create multiple new partials or to alter several existing partial values.
- Control-click a partial or drag a partial below the baseline to set it to a value of zero.
- Control-click in the Partial bar display, then drag left or right to set multiple partials to a value of zero.

Create e sound from scratch

This example introduces the basic principles of additive programming in Alchemy.

When you create a new, initialized preset by choosing Initialize Preset from the File button pop-up menu in the Name bar, source A is in VA mode.

Important: Use of Initialize Preset automatically turns on wide unison mode. See Alchemy wide unison mode.

For additive synthesis, a different default configuration is better suited. It is not necessary to change any settings from their initial values as soon as you enter the Additive edit window and start creating data; the configuration of source A switches, automatically, to a sensible set of defaults for additive programming.

1. In Alchemy in Logic Pro, open advanced view and click the A button to view the source A subpage, then click the source A Edit button to open the Main edit window. Click the Additive button at the top of the Main edit window to view the Additive edit window.
2. Set the Detail knob to 100% for an accurate view of all data.
3. Make sure the Partial bar display Vol button is on and the Overall button is off.
4. Drag from left to right in the Partial bar display to draw in bars that define the harmonic content at the beginning of the sound.

   Play a few notes on your MIDI controller to confirm that the sound begins with a bright timbre.
5. The default loop mode is Continuous which loops the sound indefinitely when you hold a note. If you do not want the sound to loop, change the Loop mode to None.
6. Select point 2 in the Partial envelope, then click the Breakpoint button and choose Copy Breakpoint from the pop-up menu.
7. Select point 3 in the Partial envelope, and drag it toward point 2. Release point 3 when you are close to point 2.
8. Click the Breakpoint button, and choose Paste Breakpoint from the pop-up menu. Make sure point 3 is selected.

   The data from point 2 is copied to point 3, smoothing out the levels between points.
9. Click the X icon at the top right to close the Additive edit window.

   The Additive element parameters are displayed.
10. Experiment with the Additive parameters to change the tone of your basic additive sound. Feel free to use other synthesis, filter, modulation, and effect parameters to make your additive sound more interesting.

Import an image file, and convert it to additive data

Alchemy is capable of importing an image in PNG (Portable Network Graphics) format and translating it into additive synthesis data. You can convert an existing image into sound, or you can design new sounds by drawing the image you want to import in an external graphics application.

1. In Alchemy in Logic Pro, open advanced view and click the A button to view the source A subpage, then click the source A Edit button to open the Main edit window. Click the Additive button at the top of the Main edit window to view the Additive edit window.
2. Click the Image button at the top right of the Additive edit window, then choose a PNG-format file in the file browser.

The selected file is translated into additive data using the criteria outlined below.

- Each column of pixels represents a snapshot in the additive data. The leftmost column describes snapshot 1, the next column to the right describes snapshot 2, and so on. Snapshots are timed at a steady rate of 20 per second.
- Within each column, there is a one-to-one correspondence between pixels and partials: the bottom row represents partial 1, the second row up represents partial 2, and so on. Thus the height of your image determines the number of partials. For example, an image 100 pixels tall translates to a 100-partial sound.
- The brightness of each pixel determines the amplitude of a particular partial in a particular snapshot. A black pixel corresponds to silence, while a white pixel corresponds to maximum amplitude. You can import both color and grayscale images. Color information other than brightness is ignored.

  Note: Importing an image with a lot of bright pixels results in additive data with numerous high-amplitude partials, which may cause clipping to occur.

3. If the results of an image import are unexpectedly noisy, you can reduce the overall amplitude by doing one or more of the following:
   - Reduce the Vol setting of the additive element.
   - Click the Overall button, then choose All from the Mode pop-up menu, and drag downward in the Partial bar display.
   - Use your graphics software to darken the image before importing it.

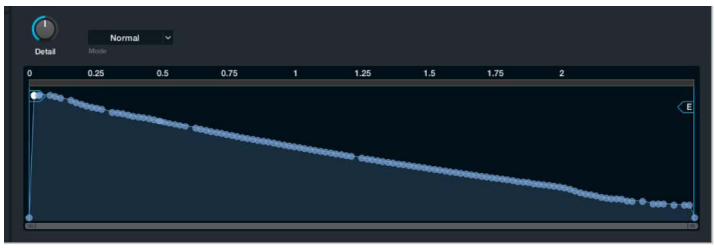
Alchemy partial envelope in Logic Pro for Mac

Source components are shown only in advanced view. Click the Advanced button to switch to advanced view, then click the A/B/C/D button to view detailed parameters for the selected source.

The source edit window is opened by clicking the Edit button on any source subpage. Click the close window icon (X) at the top right to close the window.

Use the Additive button at the top of the Main edit window to view the additive editor window.

The Partial envelope shows a number of points that indicate the temporal position of envelope snapshots (groups of partials) or individual partials, depending on the option chosen in the Mode pop-up menu (above the Partial envelope). These points also indicate the overall level or value of the visible parameter type in the selected snapshot or partial. Partials are shown in the Partial bar display. See Partial bar display.



Partial envelope parameters

- *Overall button:* Shown above the Partial bar display, sets two Partial envelope modes.
  - *Overall button on:* Each point represents a snapshot of summed amplitude, pitch, and pan values of *all partials* at a certain point in time. Typically, you choose snapshots in the envelope display and adjust individual or grouped partial values in the Partial bar display. You can, however, also adjust all partial values in a given snapshot by dragging vertically in the Partial envelope. The horizontal, or x-axis, indicates the temporal position of each snapshot. The vertical, or y-axis, indicates the summed value of each snapshot.
  - *Overall button off:* Each point in the Partial envelope has a one-to-one correlation with the selected partial in the Partial bar display. Adjusting a value in either the Partial bar display or the Partial envelope is immediately reflected in the other. The horizontal, or x-axis, indicates the temporal position of the selected partial. The vertical, or y-axis, indicates the value of the selected partial.

    Note: The time positions of envelope points may vary from one partial to the next as you choose different partials when the Overall button is off. This is one of the keys to the high quality of resynthesis in Alchemy: each partial can have an independent set of points. When the Overall button is on, Alchemy presents a series of envelope points linked to all partials, and adjustments to individual partial times and values are handled automatically.
- *Detail knob:* Set the resolution of the Partial envelope display. This affects the number of points shown.
  - A value of 25% reduces the total number of visible points. This may necessitate scrolling or zooming to view all points.
  - A value of 100% shows all points. The 500 or more visible points are unnecessary for anything other than the most precise edits, but you may find this level of detail useful when programming an additive sound from scratch, rather than editing resynthesis data.
- *Mode pop-up menu:* Choose a mode to determine the way the envelope responds when a point is dragged.
  - *Normal:* The dragged point only is moved. All other points remain stationary.
  - *Slide:* Dragging one point causes all subsequent points to move, retaining the relative distances between points.
  - *Stretch:* Dragging left compresses earlier points and stretches later points. Dragging right stretches earlier points and compresses later points. In either case, the total length of the envelope is preserved.

    Note: Point levels remain fixed in stretch mode, so you can only drag horizontally.
- *Partial envelope:* Display and edit envelope point positions and values.
  - The ruler indicates the current envelope position in seconds.
  - The Partial envelope shows only straight line segments between points, with no convex or concave lines.
  - The first and last points always have an amplitude of zero. In sounds with a fast attack phase, a second point with a nonzero amplitude is shown at a time position very close to the first point. You may need to increase the Detail setting to see the second point.

- Loop and warp markers can be superimposed on the Partial envelope. See Zone waveform editor.
- Warp marker movements automatically snap to existing Partial envelope points.
- Positioning the loop start and loop end markers at identical positions creates a sustain point which allows a particular segment of the sound (a snapshot) to be heard continuously. This method is useful for locating, auditioning, and editing a succession of snapshots.

- *Scroll bar and zoom controls:* Drag the middle of the scroll bar to view envelope points that are not visible in the display area. Horizontally drag the zoom controls at either end of the scroll bar to resize the contents of the visible display area.

Create or delete points in the partial envelope

In Alchemy in Logic Pro, do one of the following:

- Click anywhere in the Partial envelope to create a new point for the active partial (selected in the Partial number bar).
- Click on or near a line between existing points to create a new point.

  When the Partial envelope contains existing points, either method creates a snapshot of partial values that fall in between the partial values associated with the preceding and following points.

  If the Partial envelope is "blank" (only contains a single point), each newly-created point has a single associated partial value (for the partial selected in the Partial number bar). The partial (value) of the preceding point is also shown in the Partial bar display.

  Note: The Overall button affects the display of values shown in the Partial bar and related points in the Partial envelope.
- Double-click an envelope point to delete it.

Adjust partial pitch values with partial envelope points

When the additive editor displays pitch (the Tune button is on), point levels represent partial pitches. You can edit these levels to eliminate a pitch wobble or to create vibrato, for example.

1. In Alchemy in Logic Pro, click an envelope point to select it. You can also drag across multiple envelope points to select them.

   This selects a snapshot of multiple partials when the Overall button is on, or individual partials when the Overall button is off.

   When multiple points are selected, the Partial bar shows the average value of each partial across the selected envelope points. Edits of partial values are applied to all partials associated with the selected points.
2. Click the Tune button above the Partial bar display to view pitch values at the current envelope point.
3. If required, click the Mode pop-up menu above the Partial bar to choose how your edits are applied: to a single partial, all partials, or a group of related partials.
4. Drag the envelope point vertically to set the pitch value.

   The time and pitch of the selected point are displayed at the pointer position when clicked.

Fix unwanted pitch fluctuations with copied envelope points

In cases where a resynthesized sound has unwanted pitch fluctuations, you can resolve these artifacts by reducing the PVar (pitch variation) control on the source A/B/C/D subpage. The downside with this approach is that each partial is aligned with its harmonic tuning which may change the character of the sound significantly. Try this alternative method.

1. In Alchemy in Logic Pro, make sure the Overall and Tune buttons are active, then select a single Partial envelope point.
2. Click the Breakpoint button, and choose Copy from Breakpoint from the pop-up menu.
3. Click the Breakpoint button, and choose Paste All Breakpoints Pitch Data from the pop-up menu.

   The partial pitch values of the copied envelope point are applied to all other envelope points, thus retaining the inharmonic features of the copied point while simultaneously eliminating fluctuations from one point to the next.

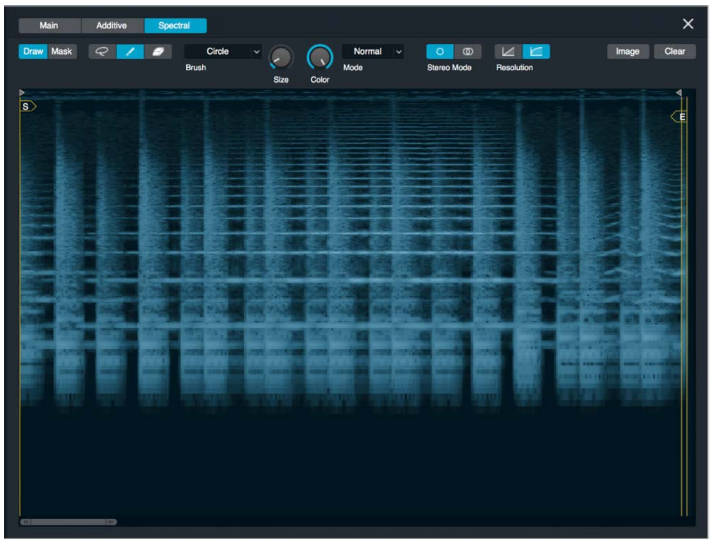
Alchemy spectral edit window in Logic Pro for Mac

Source components are shown only in advanced view. Click the Advanced button to switch to advanced view, then click the A/B/C/D button to view detailed parameters for the selected source.

The source edit window is opened by clicking the Edit button on any source subpage. Click the close window icon (X) at the top right to close the window.

Use the Spectral button at the top of the Main edit window to view the Spectral edit window.

Spectral edit window: Use to graphically edit spectral resynthesis data. You can also create new sounds by using simple paint tools directly in the display.



Spectral data is displayed as follows in the Spectral edit window, or “canvas.”

- Time, expressed in seconds, is represented along the x-axis from left to right. It is also shown at the pointer position.
- Frequency, in hertz, is represented along the y-axis from bottom to top. It is also shown at the pointer position.

- Amplitude is represented by brightness, using shades of blue. Dark blues represent lower amplitude values, and light blues represent higher amplitude values. Silence is black, and the loudest amplitude is white.
- A white play position indicator scrolls across the image for the most recently triggered voice.
- Warp markers are also shown, allowing you to change the start and end points.

Note: Accurate resynthesis requires a much finer frequency resolution than the spectral display can accommodate. Frequency information depicted in the display is a coarse representation of the underlying data. Graphical data creation and editing are performed at the resolution of the display. This means that you cannot paint conventional melodies and chords consisting of precise notes, for example.

Spectral editor parameters

- *Draw/Mask buttons:* Set a mode that affects the behavior of the paint tools.
  - *Draw:* The paint tools can be used to modify the image itself either by drawing new or deleting existing content or by increasing or decreasing the luminosity of specific parts of the spectrum.
  - *Mask:* The entire image is masked, as if covered with a layer of black wax. The paint tools can be used to selectively reveal parts of the underlying image, as if scraping away the wax to reveal the layer underneath.

    Note: The most recent edit made in the Spectral editor can be undone by pressing Command-Z. Multiple edits can be reversed by pressing Option-Command-Z, then choosing the step you want to revert to from the Undo History window.
- *Lasso, Brush, and Erase buttons:* Choose a select (Lasso) or brush mode. Erase uses the selected brush type to remove portions of the spectrogram.
  - *Lasso button:* Turn on to activate a select mode. Drag across the spectral canvas to select an area.
    - In mask mode, your selection only is played. The unselected portion of the canvas is silent.
    - In draw mode, your selection can be used as a brush shape.
  - *Brush button:* Turn on to activate brush mode. You can draw directly on the image when in draw mode, or scrape away the mask layer to reveal the underlying image when in mask mode. Any Lasso selection can be used as a brush shape that remains available until you select a different brush shape. See Brush pop-up menu.
    - Hold down Shift to limit painting to the vertical axis.
    - Hold down Command to limit painting to the horizontal axis.
  - *Erase button:* Turn on to selectively remove portions of the image with the selected brush type. See Brush pop-up menu.
- *Brush pop-up menu:* Choose a predefined brush shape. Visible only when the Brush or Erase button is active.
  - Blurred Circle, Circle, Horizontal Line, and Vertical Line produce the brush shapes described.
  - Transient produces a vertical edge that slopes to the right, typical of a percussive drum hit.
  - Custom uses the Lasso selection as a brush shape.

- *Size knob:* Scale the size of the brush chosen in the Brush pop-up menu.

  Note: When a custom brush is active, the Size knob is not available.

- *Color knob:* Set the amplitude scaling of the brush. At 100%, brush strokes are white. At 50%, brush strokes are mid-blue. At zero, brush strokes are black. This knob is visible only when both the Draw and Brush buttons are active.

  Note: When a custom brush is active, the maximum intensity is defined by the custom brush data.

- *Mode pop-up menu:* Choose a mode that determines how the brush interacts with the existing canvas image in draw mode. This menu is visible only when both the Draw and Brush buttons are active.
  - *Normal:* Each pixel on the canvas is set identically to the corresponding brush pixel.
  - *Add:* Brush pixel values are added to the existing canvas pixel values. In this mode, painting multiple coats of blue results in increasingly higher amplitudes with each coat.
  - *Multiply:* Brush pixel values are multiplied with the existing canvas pixel values. Use this to selectively brighten parts of the image.
- *Mono/Stereo buttons:* Switch the display mode between mono and stereo.

  Note: This does not change the sound. In mono mode, stereo sounds are represented as a single image that represents both channels. Edits affect the left and right channels equally. In stereo mode, stereo sounds are shown in left and right lanes. Edits affect only the channel they are made in.

- *Resolution buttons:* Switch the display between a linear and logarithmic representation of your spectral data on the vertical y-axis. Logarithmic (right button) is the default.
  - *Linear (left):* A simple linear scale with 11 kHz shown at the halfway point on the vertical y-axis. This mode is useful for detailed editing of very high frequency content.
  - *Logarithmic (right):* A logarithmic scale for frequency that corresponds more closely with human perception of pitch. Content at the halfway position on the y-axis is approximately 1 kHz. Log mode is useful for most vocal duties.
- *Image button:* Choose an import image option from the pop-up menu.
  - *Import Image:* Opens a dialog where you can select a file in PNG format that is placed directly on the spectral canvas. The imported image is positioned at the far left of the canvas. The height of the image is scaled to fit the entire vertical range of the canvas, and the width of the image is scaled by the same factor as the height, retaining the original image proportions. All color information is discarded, and image brightness information is mapped to amplitude. If the imported image width does not extend fully to the right edge of the canvas, existing data beyond the image edge remains in place. Images with a height of 256 lines result in a one-to-one map of pixels to spectral bins. The spectral editor display is actually 381 lines tall, so edits involve some blurring between bins.
  - *Import Image to Brush:* Opens a dialog where you can select a file in PNG format to be used as a brush for painting on the spectral canvas. The original dimensions of the image are preserved. You can place the image at any position with a click, or you can drag to paint with it.
- *Clear button:* all spectral data from the source, leaving only silence (a solid black image).
- *Scroll bar and zoom controls:* Drag the middle of the scroll bar to view spectral data that is not visible in the display area. Horizontally drag the zoom controls at either end of the scroll bar to resize the contents of the visible display area.

Alchemy main filters

Alchemy main filter controls in Logic Pro for Mac

Main filter components are shown only in advanced view. Click the Advanced button to switch to advanced view, then click the Global button.

There are two main filter modules that can operate in parallel or in series. These modules can be addressed as Filter 1 and Filter 2 from menus and as modulation targets.



The output from all four sources is sent to the main filters at levels determined by independent controls in each source. Both main filter modules provide multi-mode filters with identical controls.

Note: It is possible for each source to completely bypass the main filters, or to send a portion of the source signal to the main filters and another portion directly to the Effects section. Though you can use filters at multiple locations in the signal path, you can often attain the same or similar results by careful use of fewer filters, which helps to reduce CPU load.

Main filter parameters

- *On/Off buttons:* Turn each filter on or off.
- *Filter type pop-up menus:* Choose the filter. Use the descriptive names Clean, Edgy, Gritty, Rich, Sharp, Smooth or another filter type, such as Formant or Bitcrusher. You can step through the available filter types with the Previous and Next buttons (the arrows). See Filter types.

  Note: The chosen filter type can alter the names and functions of the default Cutoff, Resonance, and Drive knobs.
- *Cutoff knob:* Set the cutoff frequency for the chosen filter type.
- *Resonance knob:* Boost or cut frequencies above, below, or surrounding the value set with the Cutoff knob. Resonance behavior changes when different filter types are chosen.
- *Drive knob:* Overdrive the filter. This can cause distortions and aliasing, depending on filter type.

- *Level meters:* Indicate the current audio level received from all four sources. If the filter input level exceeds 0 dB, the meters indicate clipping by momentarily turning red. Clipping produces an undesirable digital audio artifact in the output stage. If clipping occurs, turn down the loudest source using the Vol knob or adjust the balance between sources.
- *Par/Ser knob:* Set the filter routing configuration to parallel or series. Note that this is pre-effects.
  - At 0% position, the total output of the first filter is sent to the main outputs of the filter module. This is parallel mode.
  - At 100% position, the total output of the first filter is sent to the the second filter input. This is series mode.
  - At 50% position, equal amounts of the the first filter output signal are sent to the second filter input and the main outputs of the filter module.

    Note: When the Par/Ser knob is set to 100%, a portion of the the first filter signal bypasses the second filter whenever the first filter FX Master knob is set above 0%.
- *FX master knobs:* Determine the balance between the filter module output sent to the Effects section and the filter module output sent directly to the Alchemy main output.
  - At 0% position, the total filter output is sent to the Alchemy main outputs, and none of it to the Effects section.
  - At 100% position, the total filter output is sent to the Effects section, and none of it to the Alchemy main outputs.
  - At 50% position, equal amounts of filter output signal are sent to the Effects section and the Alchemy main outputs.
- *Send destination pop-up menus:* Independently send the output of the main filter (1 or 2) module to the Main effects rack, or to the A/B/C/D effects rack. Choose FX Main, FX A, FX B, FX C, or FX D.

  Note: When the FX master knob for the first or second filter is at the full-left position, the total filter (1 or 2) output is sent to the Alchemy main outputs, and none of it to the Effects section.

Alchemy filter types in Logic Pro for Mac

Alchemy provides 39 multimode filter types. These can be used at the source level or as main filters that process all routed signals. Filters used at the source or main filter stages affect each voice independently. You can also insert filters in the Effects section, but they are monophonic when used at this point in the signal path and affect all voices (the overall signal) simultaneously.

- Two main filters. See Main filter controls.
- Four sets of three source filters. See Source filter controls.
- The MMFilter module in the Effects section. See Filter effects.

There are dozens of filter types to choose from in the main filters and the MMFilter effects module.

Alchemy filter types: Lowpass, bandpass, highpass

A lowpass (LP) filter passes the portion of a signal below a specified cutoff frequency and rolls off the portion above that frequency. A bandpass (BP) filter passes the portion of a signal occupying a band surrounding the cutoff frequency and rolls off the portions above and below that band. A highpass (HP) filter passes the portion of a signal above a specified cutoff frequency and rolls off the portion below that frequency.

There are multiple LP, BP, and HP filter designs in Alchemy, each with distinctive characteristics that you may prefer for a given purpose.

Decibel values (6 dB, 12 dB, 24 dB, and 60 dB) are shown for LP, BP, and HP filter types, indicating the steepness of the rolloff (frequency gain reduction).

The available LP, BP, and HP filter designs include:

- *Clean:* Bi-Quad filters. These filters have no Drive option.
- *Edgy:* Two-pole (12 dB slope) or four-pole (24 dB slope) analog-modeled filters. Four-pole variants provide a steeper roll-off of frequencies beyond the cutoff.
- *Gritty:* Two-pole filters designed to saturate heavily at higher Resonance and Drive settings.
- *Rich:* Two- or four-pole analog-modeled filters. Four-pole variants provide a steeper roll-off of frequencies beyond the cutoff.
- *Sharp:* Two-pole analog-modeled filters.
- *Smooth:* Two- or multi-pole state-variable filters.

The three principal filter controls have standard functions for all traditional filter types.

- *Cutoff knob:* Controls the filter cutoff frequency.
- *Resonance knob:* Boost or cut frequencies above, below, or surrounding the value set with the Cutoff knob. Resonance behavior changes when different filter types are chosen.
- *Drive knob:* Allows the filter to be overdriven; the precise effect varies with each filter design.

Alchemy filter types: Formant, notch, and peaking

A formant filter works like a bandpass filter with a potentially very narrow bandwidth. It is designed to mimic a formant, or fixed resonance, such as those produced by the hollow body of a guitar or by the shape of the vocal cavity (which is adjusted during speech to produce different vowel sounds).

- *Cutoff knob:* Controls the resonant frequency of the formant.
- *Amount knob:* Controls the width of the band surrounding the resonant frequency. Low values produce a narrow band, high values a wider one. Low resonance values may allow little or no sound to pass through the filter.

A notch filter cuts a narrow band around a resonant frequency. The remainder of the signal is not affected.

- *Center Frequency knob:* Sets the center frequency of the cut band.
- *Amount knob:* Controls the amount of attenuation.

A peaking filter boosts a narrow band around a resonant frequency. The remainder of the signal is affected minimally.

- *Center Frequency knob:* Sets the center frequency of the boosted band.
- *Gain knob:* Controls the amount of boost. Higher values are generally the most effective.
- *BW (Band Width) knob:* Sets the bandwidth around the cutoff frequency.

Alchemy filter types: Comb filters

A comb filter mixes the original signal with one or more copies of the signal which are delayed by a very short time interval. At some frequencies this mixture causes phase cancelations, while at other frequencies it causes reinforcements. The result is a spiky frequency spectrum with multiple resonant peaks. Graphically, these peaks resemble the teeth of a comb, which gives this filter type its name.

Alchemy offers three comb filter designs, each with its own character. The best choice is very much a question of your preference and the type of sound you are trying to create. That said, there are some distinguishing characteristics that may help guide you.

Comb Pos uses positive feedback on the delay lines, while Comb Neg uses negative feedback to produce less extreme effects, often with a hollow quality. These two are the less powerful combs and offer a much more gradual increase in resonance. They can be useful when you require either a less dramatic effect or you want to hear more of the exciter signal character in your sound. The latter point is noteworthy as this trait can be useful when you want a more naturalistic modeled sound.

For instrument modeling, it is best to assign combs as source filters, rather than in the master filter or effect sections. Although comb filters can be used in the master filter section, it is better to use the main filters for additional processing. Comb filters are also available in the Effects section but are not polyphonic, so you will achieve better results if they are placed earlier in the signal path.

- *Note knob:* Controls the delay time in the comb circuit. Lower values equate to a longer delay.
- *Feedback knob:* Controls the amount of feedback in the comb circuit.
- *Damp (Damping) knob:* Softens the frequency peaks in the comb-filtered spectrum.

  Note: Sending a percussive sound into a highly resonant comb filter causes it to ring at a frequency determined by the delay time you have set with the Cutoff knob.

Alchemy filter types: Ring modulation

Ring modulation is a process in which a modulator and a carrier signal are multiplied. Each frequency component of the modulator interacts with each frequency component of the carrier to produce two sidebands: a sum and a difference (carrier – modulator). When a filter is set to the RingMod type, the signal entering the filter acts as the modulator, while the carrier is supplied internally by the filter.

- *Note knob:* Controls the carrier frequency.
- *Mix knob:* Applies a constant offset to the carrier.
  - At 0%, the carrier wave varies between –1 and +1, resulting in classic ring modulation.
  - At 100%, the carrier wave varies between 0 and 1, resulting in classic amplitude modulation. In this case, the carrier signal itself is present alongside the sum and difference sidebands.

- *Weird knob:* Controls the carrier waveform, from pure and glassy at low values to noisy at higher values.
  - Set to 0% for a pure sine wave carrier. Use this setting to produce characteristic bell-like timbres.
  - Set above 0% to progressively truncate the carrier waveform. Higher settings produce additional sidebands, typically resulting in noisy timbres.

Alchemy filter types: Distortion

Alchemy filters offer several distortion effects.

- Bitcrusher is a harsher-sounding digital effect.
- Downsample is a lo-fi digital effect (similar to Bitcrusher but is frequency dependent).
- Mech is a more metallic-sounding distortion.
- Tube is a classic tube distortion effect.

Note: Distortion effects created in the source and main filter sections are polyphonic, with each voice distorted independently. This avoids unpleasant intermodulation effects when you play chords. In contrast, the Distortion module in the Effects section at the end of the signal path processes a mix of all voices.

The filter controls work as follows when the filter type is set to a distortion effect:

- *Amount knob:* Controls the intensity of the distortion effect.
- *Mix knob:* Controls the balance between clean and distorted signals. A value of 0% results in the clean signal only. A 50% value results in an equal mix of clean and distorted signal. A 100% value results in the distorted signal only.
- *PreGain knob:* Boosts the input gain.
- *Frequency knob:* Sets the frequency for the Downsample distortion type.

Alchemy filter types: Compressor and FM

These tools provide creative options at the filter stage of the signal path. Unlike dedicated FM synths, Alchemy does not have preconfigured algorithms or a mod matrix setup specifically for FM. It does, however, offer the option to run a huge number of FM filters in various series and parallel configurations. FM in Alchemy also has a different character to classic FM synths. FM in Alchemy is more like working with FM on analog synths where you are modulating oscillator frequency rather than phase. As a result, Alchemy's FM is great for adding different kinds of effects to your sound, as well as "organic" distorted textures.

The filter controls work as follows when the filter type is set to Compressor:

- *Threshold knob:* Set the threshold level. Signals above this threshold value are reduced in level.
- *Ratio knob:* Set the compression ratio (the ratio of signal reduction when the threshold is exceeded).
- *Gain knob:* Set the amount of gain applied to the compressed signal.

The filter controls work as follows when the filter type is set to FM:

- *Note knob:* Set the carrier oscillator frequency.

  Tip: For classic FM sounds, double-click the Note knob to set it to the center position, then Control-click the knob and choose Add Modulation > Note Property > Key Follow from the shortcut menu.

- *Mod (Modulation) knob:* Set the degree to which the modulator (source audio) can modulate the frequency of the carrier.
- *Feedback knob:* Set the amount that the output of the carrier is fed back into itself.

Alchemy master and voices section in Logic Pro for Mac

Master voice section controls are shown only in advanced view. Click the Advanced button to switch to advanced view, then click the Global button.

An individual voice generated in the source section is passed through the main filters, then into the master voice section where controls are provided for the amplitude, pan, and coarse and fine-tuning of the voice. The voice is then mixed with other voices at the input of the Effects section.

The master voice section also provides high-level control over various attributes of a preset such as polyphony, tuning, and glide.



Master voice controls

- *Vol knob:* Adjust the level of the voice. By default, AHDSR 1 modulates this parameter.

  Note: When an AHDSR envelope is assigned to modulate master volume, the modulation depth cannot be adjusted. This is because setting the depth below its default of 100% would cause notes to remain above zero amplitude indefinitely, resulting in hung notes. This rule applies to other modulators, such as an MSEG, where you also require a mod depth setting of 100% unless you actually want notes to sustain indefinitely.

- *Pan knob:* Adjust the stereo position of the voice. Acts as a pan control for mono sounds and as a balance control for stereo sounds.
- *Coarse/Fine Tune knobs:* Adjust the pitch of the voice in semitones (Coarse) and cents (Fine=1/100 of a semitone).

- *All/A/B/C/D buttons:* Select All to set global voice parameters, which override individual source settings. Select A/B/C/D to set individual source voice parameters. All of the following parameters can be set globally or per source.
- *Trigger Mode pop-up menu:* Determine how trigger events affect audio signals and glides (portamento). This parameter interacts with the Num and Glide controls and can be used globally or per source.

  Note: A trigger signal causes certain processes to execute from the beginning. These include: the playback of samples and modulators, including LFO, AHDSR, MSEG, and Sequencer. Individual modulators also have their own trigger options.

  - *Always:* If the Num value is 1, a trigger is generated at the start of each legato group, and portamento occurs at the start of every note. For all other Num values, a trigger is generated at the start of every note, and portamento occurs at the start of every note.
  - *Retrigger:* A trigger is generated at the start of every note, and portamento occurs at the start of every note.
  - *Legato:* If the Num value is 1, a trigger is generated at the start of each legato group, and portamento occurs at the start of each legato group. For all other Num values, behavior is unchanged when you play single notes. When you play a chord, each note of the chord is triggered individually.
- *Num(ber of voices) pop-up menu:* Set the maximum polyphony of a preset (up to 32 voices) when used globally, or maximum polyphony per source. A value of 1 results in a monophonic preset or source.

  Tip: Aim for the lowest possible polyphony value for the preset because this helps to reduce CPU load.
- *Voice Priority pop-up menu:* Choose the Newest (last played), Oldest (first played), Lowest or Highest (pitch) note to be prioritized when polyphony is exceeded. This parameter can be used globally or per source.
- *Rate/Time buttons:* The portamento (glide) mode can be set to Rate or Time. This parameter can be used globally or per source.
  - *Rate:* Alchemy glides from note to note at a fixed rate set with the Glide knob. Longer glides require more time. Shown as a percentage of the overall envelope length.
  - *Time:* Alchemy glides from note to note over a fixed amount of time (in msec) set with the Glide knob. Longer glides occur at a faster rate.
- *Glide knob:* Set the portamento rate or time. Glide causes slides from one note pitch to the next. This parameter can be used globally or per source.
- *Pitch bend Up/Down pop-up menus:* Set the maximum range for upward and downward pitch bend modulation, typically performed with your keyboard pitch wheel. Can be set globally or per source.

  Note: Individual sources can be set to respond to pitch bend or to ignore it. This is determined with the Keyscale parameter on each source subpage. Pitch bend is also available as a modulator, allowing further control over the pitch bend response of each source. See Source subpage controls.

Alchemy modulation section

Alchemy modulation overview in Logic Pro for Mac

Modulation components are shown only in advanced view. Click the Advanced button to switch to advanced view.

Alchemy features a modular modulation system that combines ease of use with extensive functionality. The modulation rack is shown at the left. See Modulation rack controls. The modulator control panel occupies the center portion of the display. This area updates dynamically to display modulation sources for the selected parameter.

- *Modulator panel type buttons:* Click to view settings and adjust the controls of each modulation source.

Note: Some modulation types, namely MIDI, Note property, and Perform control modulators, don't have a control panel in the center of the modulation section. These modulators are set with menu items in the modulation rack.



Almost every knob in the Alchemy interface is a modulation target, representing a parameter that can be modulated by as many as ten modulators. This not only includes voice parameters, but also effect, arpeggiator, and performance; even other modulation parameters can be modulation targets. Each mod rack entry may have its depth (amount it affects the target) modulated by up to five additional modulation sources. Each modulation depth mod entry may be further modulated by an additional modulation source.

Note: The additional layers of modulation are useful when the amount of modulation, such as vibrato, is controlled by an envelope, MSEG, or channel aftertouch, for example. The third layer of modulation is particularly useful for assigning velocity or the modwheel to add expressiveness to the preceding modulation layers.

When a parameter is the target of one or more modulators, an orange modulation arc is shown beside the blue value arc for the knob. This indicates that the knob is an *active* modulation target and shows the modulation range. Parameters that are morphed and have a modulation assignment show both an orange and a green arc around the control.

Note: The most recently clicked knob is highlighted in blue, making it easy to identify the current target. If you switch between source subpage A to source subpage B, for example, the highlight shifts to the corresponding knob on the new subpage. This behavior lets you quickly assign modulations to a parameter such as Fine Tune for multiple sources.

Each modulation type is described in the linked sections. The MIDI control, Note Property, and Perform control modulators do not have a graphical control panel.

- LFO. See LFO controls.
- AHDSR. See AHDSR envelope controls.
- MSEG. See Multiple segment envelope controls.
- Sequencer. See Sequencer.

- Envelope Follower. See Envelope follower parameters.
- ModMap. See ModMap parameters.
- MIDI control. See MIDI control modulators.
- Note Property. See Note property modulators.
- Perform. See Perform control modulators.

View or edit modulations assigned to a parameter

- In Alchemy in Logic Pro, click the parameter knob.

  The parameter name is shown in the Target pop-up menu at the top left of the modulation section. Directly below the Target pop-up menu is a modulation rack with up to ten slots (one is shown by default). Click a different knob, and the mod rack updates to reflect the new selection. See Modulation rack controls.

Use the modulation source shortcut menu

- In Alchemy in Logic Pro, Control-click a modulation target knob (filter Cutoff, for example), then choose a command from the shortcut menu.
  - *Add modulation:* Assign a new modulator by choosing it from a submenu. The new modulation assignment is shown in the first empty slot of the modulation rack.
  - *Clear modulation:* Removes all modulations from the knob, leaving the modulation rack empty.
  - *Copy modulation:* Copies information about all currently assigned modulators to the Clipboard.
  - *Paste modulation:* Applies all modulator information from the Clipboard. Use of the Copy and Paste commands lets you quickly assign the same modulations to multiple targets.

Assign a basic modulation with the modulation rack

As an alternative to the shortcut menu, you can assign modulation sources to targets in the modulation rack. This example assigns the LFO 1 modulation source to the source A Volume target. These steps apply equally to other modulation sources and targets. See Modulation rack controls.

1. In Alchemy in Logic Pro, click the File button in the Name bar and choose Initialize Preset from the pop-up menu to reset all Alchemy parameters to default settings.

   Play a note to hear the default sawtooth wave.

   Important: Use of Initialize Preset automatically turns on wide unison mode. See Alchemy wide unison mode.

2. In the source section, click the Vol knob for source A. In the modulation section, the Target pop-up menu changes to read Volume A.

3. Click in the top slot of the modulation rack, then choose LFO > LFO 1 from the pop-up menu. Click the LFO button to display the LFO controls in the area to the right of the modulation rack.

   Note: The pop-up menu displays the waveform shape used by the selected LFO in parentheses. For example, “LFO 1 (Sin)” indicates that LFO 1 is using a sine waveform. The waveform shape is updated in the modulation rack when changed in an LFO. See Alchemy LFO controls.

4. Rotate the Depth knob beside modulation slot 1 to set a value of 100%.
5. Hold a note to hear a tremolo effect, as the source A volume is modulated by LFO 1. Change the LFO Rate setting to adjust the speed of the effect.
6. Try adding another modulator to the rack in the slot beneath LFO 1.

   Note: Up to ten different modulators of any type can be assigned to each control in a preset.

Modify the basic modulation

In the preceding example, a tremolo effect is achieved by modulating the source A Vol knob with LFO 1. This example expands on the basic modulation by modulating the LFO 1 Rate with Key Follow, which makes the tremolo effect faster when you play higher notes. You can adapt this example to work with other modulator parameters, such as AHDSR Attack time.

1. In Alchemy in Logic Pro, Control-click the LFO 1 Rate knob, then choose Add Modulation > Note Property > Key Follow from the shortcut menu.

   LFO 1 Rate is shown in the Target pop-up menu, and Key Follow is listed in the first slot of the modulation rack.
2. Play a variety of higher and lower notes to hear the effect of Key Follow on the LFO Rate. If this effect is too extreme, you can reduce the value of the Depth knob shown to the right of the first slot in the modulation rack. See Modulation rack controls.

Adjust the modulation intensity

Each modulator has its own Depth control, and each Depth control can itself be assigned a modulator. A modulator can even be set to modulate its own depth. Up to five depth modulators may be assigned to any modulation source. See Modulation rack controls.

1. In Alchemy in Logic Pro, Control-click a Depth control in the modulation rack, then choose Show Modulation from the shortcut menu.

   The modulation rack updates to display the Depth control name in the Target pop-up menu. For example, LFO 1 Rate Depth 1 indicates the first modulator of the Depth target (which is modulating the LFO Rate 1 target).
2. Add one or more modulators to the rack.

Alchemy modulation rack controls in Logic Pro for Mac

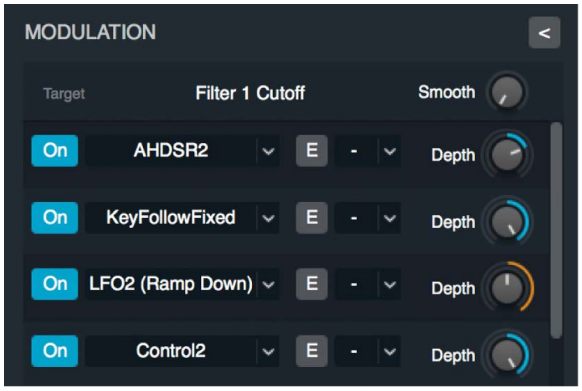
Modulation components are shown only in advanced view. Click the Advanced button to switch to advanced view.

Normally, two things happen when you click a knob. First, the modulation rack associated with the knob is displayed. Second, if a modulator has already been assigned to the first slot in the newly displayed rack, the corresponding control panel is shown to the right. Some modulators do not provide a control panel.

In two cases, however, Alchemy prevents these things from happening. If the knob you click is on a modulator control panel (if you click LFO Rate or AHDSR Attack Time, for example), Alchemy assumes you want to adjust the modulation of the current target instead of switching to a new target. Similarly, if you click a modulation Depth knob in the modulation rack, Alchemy assumes you want to adjust the depth, so the current rack view is retained.

You can still view and edit modulation assignments in these cases. If a knob is located on a modulator control panel or in the modulation rack, Control-click it to open a shortcut menu with several commands. Choose Show Modulation to display the modulation rack associated with the control.

Note: Some parameters cannot be modulated. Knobs in the Perform section are not modulation targets because they are designed for use as modulation sources. If a parameter is set with a menu command rather than a knob, you cannot modulate it, such as with Loop mode or Filter type.



Modulation rack parameters

- *Back button:* Shows the previously selected modulation target. The Back button is visible once you Control-click a modulation Depth knob and select Show Modulation from the shortcut menu.
- *Modulation Target pop-up menu:* Shows the selected modulation target and allows you to select any active modulation target. Up to ten modulation slots are shown below the Target pop-up menu. Each slot in the modulation rack has identical controls.
- *On/Off button:* Turn the modulation source on or off. Off deactivates the modulation assignment but doesn't remove it from the slot.
- *Modulation Source pop-up menus:* Choose the modulation source. Choose None to remove a source. Modulators are grouped by submenu. Choose New (modulator) to create additional modulation sources of the chosen type. Many modulator types allow up to 16 discrete modulation sources to be created. Each of these modulation sources can have independent parameter settings and can modulate different targets.

  When you assign a source, a new slot appears below the selected slot. Up to ten modulation slots are available for each target.
- *Edit (E) buttons:* Open the selected modulator control panel in the right half of the modulation section.

  Note: Not all modulator types have a control panel. Some are available only in the Modulation pop-up menu.
- *ModMap pop-up menus:* Assign, create, or delete a ModMap. A ModMap is used to process the output of other modulators. See ModMap parameters.
- *Depth knobs:* Set the modulation intensity in a range from -100% to 100%. Double-click to reset Depth to zero.

- *Smooth knobs:* Causes the target to respond more gradually to modulation at values above zero. If you find that rapid filter or amplitude modulation causes unwanted sonic artifacts, one solution is to add a small amount of smoothing. Larger values can also produce interesting effects.

  Note: The Smooth parameter is not available as a modulation target.

- *Scroll bar:* Drag vertically to view modulation rack routings (slots) that are not currently visible. The scroll bar is shown only when the number of assigned slots exceeds the modulation rack display area.

- *Mod Targets button:* Shows the currently viewed modulator targets. See Modulation target pop-up menu entry above, and the *Find targets for a modulator* and *Remove all unused targets* tasks below.

Select, create, or change a modulator in the modulation rack

- In Alchemy in Logic Pro, click a slot in the modulation rack, then choose an item from the pop-up menu.

Remove a modulation assignment from the modulation rack

- In Alchemy in Logic Pro, choose None from the modulation rack pop-up menu.

  The modulator has no effect on the currently selected target but remains available for assignment to other parameters.

  Note: Depending on the situation, you may prefer to use the On/Off button for a modulation slot to deactivate the routing temporarily. This retains all parameter settings of the modulation source.

Delete further modulation assignments from the modulation rack

Alchemy doesn't provide just one, but up to 16 modulators of certain types, such as LFOs or AHDSRs. That means you can have up to 16 LFOs, each with a different configuration. Any of these 16 LFOs can be used as a modulator for a modulation assignment.

- In Alchemy in Logic Pro, click a slot in the modulation rack, then choose Parameter group > Del parameter name (and number) from the pop-up menu.

  For example, you would delete LFO 2 by choosing LFO > Del LFO 2 from the pop-up menu.

Find targets for a modulator

The modulation rack shows all modulators (sources) assigned to a particular target. The Mod Targets button at the top right of the modulation section quickly displays all targets of a particular modulation source.

- In Alchemy in Logic Pro, click the Mod Targets button to open a pop-up menu that shows all modulators and all assigned targets for these modulators.

  If you initialize Alchemy by clicking the File button, and then choose Initialize Preset from the pop-up menu, you can browse the Target menu to see that:

  - AHDSR 1 modulates Master Volume.
  - Velocity, a Note Property, modulates Master Volume.
  - No other modulations are assigned.

Remove all unused targets

- In Alchemy in Logic Pro, click the Mod Targets button, then choose the Remove Unused Modulations entry from the pop-up menu.

  All currently unused modulations are handled as follows:

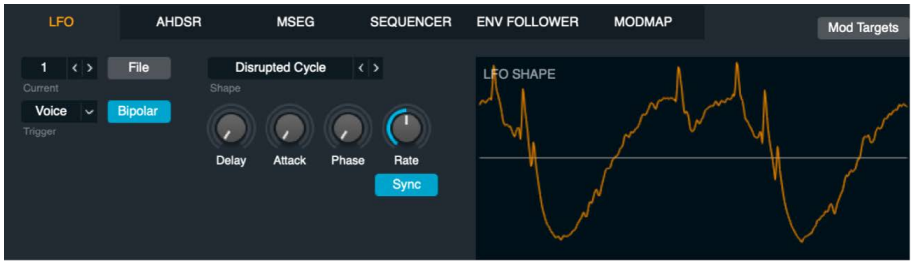
  - Any modulators assigned to a section which is turned off will be unassigned. For example, if Aftertouch assigned to Additive FX Harmonic A : Odd/Even, it will be unassigned when the additive module is turned off for A.
  - Any modulators (LFOs, AHDSRs, MSEGs, Sequencers, Modmaps) will be removed if they are not assigned.
  - Any 'Max' modulations will be removed if the same value can be achieved by changing the static value and removing the Max modulation.

Alchemy LFO controls in Logic Pro for Mac

Modulation components are shown only in advanced view. Click the Advanced button to switch to advanced view.

The LFO module provides a standard Low Frequency Oscillator, such as you might find in any conventional synthesizer or sampler. There are one or two unusual features, however.

Note: The LFO module has no depth control. The modulation depth is adjusted using the Depth knob alongside the slot in the modulation rack.



LFO controls

- *Current LFO pop-up menu:* Access each LFO control panel by selecting a number from the pop-up menu or with the Previous and Next arrows. Alchemy provides up to 16 LFOs—one by default, but more if you create them when assigning modulators.
- *File button:* Open a pop-up menu with a number of LFO-related commands.
  - *Preset submenu:* Choose a preset LFO shape. This can be used as is, or as a starting point for your own shapes.
  - *Save:* Save the current LFO settings. A dialog opens in which you can name and save the file (*.lfo). The new LFO name appears at the bottom of the Preset submenu.
  - *Copy/Paste:* Use these commands to copy parameter settings between LFOs.
  - *Clear:* Initialize the LFO.
- *Trigger pop-up menu:* Choose On to retrigger the LFO (start from zero) with each new played note. Off makes the LFO run freely (song start trigger). Voice On/Effects Off modulates voice parameters only, not effects. Also see the Delay, Attack, and Phase knob descriptions.

Note: Settings from earlier application versions may not behave as expected, because of the additional trigger option. Change to Voice On/Effects Off to match the former version behavior, then resave the setting.

- *Bipolar button:* When on, the LFO outputs negative and positive values per cycle (-50% to 50%). When off, only positive values are output (0% to 100%).
- *Shape pop-up menu:* Choose an LFO waveform from several categories. You can also step through the available waveforms with the Previous and Next buttons (the arrows). The chosen LFO waveform is shown in the LFO shape display and is updated in the modulation rack. For example, "LFO 1 (Sin)," which indicates that LFO 1 is using a sine waveform, is shown in the modulation rack. See Alchemy modulation rack controls.
  - *Basic:* Common waveforms such as Ramp Up, Ramp Down, Sine, Square, and Triangle, along with two random choices. Random Glide is a constantly fluctuating random modulator that moves smoothly between random values at a speed set with the Rate control. Random Hold is a stepped random modulator that jumps between random values at a speed set with the Rate control, holding each value until the next jump occurs.
  - *Complex:* A variety of complex cycles, sweeps, and patterns can be used to create regular (and irregular) patterns for your LFO modulations.
  - *Random Patterns:* A variety of complex patterns that can be used to create random LFO modulations.
  - *Serial-Angular:* A variety of stepped shapes.
  - *Serial-Smooth:* Provides complex rising-and-falling patterns.
  - *Ultra High Frequency:* The ultra-high frequency category waveform shapes contain multiple copies of a pattern. The modulation speed is a multiple of the speed set with the Rate control.
- *Delay knob:* Introduce a delay between the note on message and the first cycle of the LFO when Trigger is on. When Trigger is off, Delay has no effect. The delay time is adjustable over a range of 0.00 seconds to 20.00 seconds, or is set in tempo-synchronized divisions when Sync is turned on.
- *Attack knob:* Fade in the LFO output to increase modulation depth the longer a note is held (when Trigger is on). When Trigger is off, Attack has no effect. The attack time is adjustable over a range of 0.00 seconds to 20.00 seconds, or is so-synchronized divisions when Sync is turned on.
- *Phase knob:* Adjust the start point of the LFO from zero to later in the cycle (when Trigger is on). The available range is 0.00% to 100.00%.
- *Rate knob:* Set the LFO rate or frequency. When Sync is off, rate is adjustable from 0 Hz to 220 Hz. With Sync on, rate is set in beats and sub-beats.
- *Sync button:* Turn on to synchronize the LFO Rate, Attack, and Decay controls with the project tempo. Turn off to freely set the LFO speed, attack and decay times.
- *LFO shape display:* Shows real-time changes to the LFO waveform.

Alchemy AHDSR envelopes in Logic Pro for Mac

Modulation components are shown only in advanced view. Click the Advanced button to switch to advanced view.

The AHDSR module provides an envelope generator with attack, hold, decay, sustain, and release stages. The A, H, D, and R times can be set independently. You can set the level of the S stage which is maintained until a MIDI note off message is received.

The AHDSR display shows a graph of envelope generator output. The ruler along the top shows the time, calibrated in seconds or in beats when Sync is activated. The envelope appears as series of points joined by lines or curves representing the different envelope segments.

You can set AHDSR values with the controls outlined in this section or by dragging points on the envelope graphic.

You can adjust the curvature of segments between points. Drag the line between points upward to make a segment progressively more convex; drag it downward to make a segment progressively more concave. Convex, linear, and concave envelope segments produce characteristically different effects. Option-click the line to reset a segment to a linear slope.



Envelope controls

- *Current AHDSR pop-up menu:* Access each AHDSR control panel by selecting a number from the pop-up menu or with the Previous and Next arrows. Alchemy provides up to 16 envelopes—one by default, but more if you create them when assigning modulators.
- *File button:* Open a pop-up menu with a number of envelope handling commands.
  - *Preset submenu:* Choose a preset envelope shape. This can be used as is, or as a starting point for your own envelope shapes.
  - *Save:* Save the current envelope. A dialog opens in which you can name and save the envelope file (*.ahd). The new envelope name appears at the bottom of the Preset submenu.
  - *Copy/Paste:* Use these commands to copy settings between envelopes.
  - *Clear:* Initialize the AHDSR.
  - *Randomize:* Create a random envelope shape. This can be used as is, or as a starting point for your own envelope shapes.

- *Trigger pop-up menu:* Choose On to retrigger the envelope (start from zero) with each new played note. Off triggers the envelope for the first note only. Voice On/Effects Off modulates voice parameters only, not effects. The envelope retriggers only for notes received after all other notes have been released.

  Note: Settings from earlier application versions may not behave as expected because of the additional trigger option. Change to Voice On/Effects Off to match the former version behavior, then resave the setting.
- *Sync button:* Turn on to synchronize the envelope with the project tempo. Adjustment of AHDR knobs snaps the envelope to bars and beats. Turn off to set envelope stage times freely.
- *Envelope display:* The AHDSR display shows an editable graph of envelope generator output.
- *Attack knob:* Set the time required to reach the peak amplitude level after a note is played. Adjustable from 0.00 seconds to 20.00 seconds.
- *Hold knob:* Determine the time that the peak amplitude level is held before the decay stage of the envelope begins. Adjustable from 0.00 seconds to 20.00 seconds.
- *Decay knob:* Set the time required for the peak amplitude to fall to the sustain level. Adjustable from 0.00 seconds to 20.00 seconds.
- *Sustain knob:* Set the envelope sustain level as a percentage of peak amplitude. Adjustable from 0% to 100%.
- *Release knob:* Set the time required for the signal amplitude to fall from the sustain level to zero, after the key has been released (MIDI note off message). Adjustable from 0.00 seconds to 20.00 seconds.

  Tip: Where possible, try to use the minimum required envelope release times in order to reduce voice overlap, because this helps to reduce CPU load.

Move an envelope point

- In Alchemy in Logic Pro, drag a point to the required position.

  Note: Hold down the Shift key while dragging to adjust more precisely.

Alchemy multiple segment envelopes in Logic Pro for Mac

Modulation components are shown only in advanced view. Click the Advanced button to switch to advanced view.

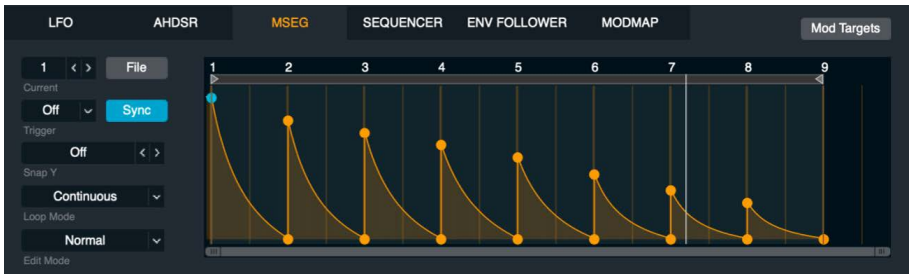
The Multiple Segment Envelope Generator (MSEG) allows complex modulation envelopes to be created and edited.

Any number of points can be added to an envelope. The envelope segments linking these points can be linear or curved. A Sync function allows envelopes to be linked to a grid derived from the Logic Pro tempo to create elaborate rhythmic patterns.

Envelopes are created and edited in two basic ways: by adding, moving, or removing points, and by adjusting the curve of the envelope segments between points. See the Edit Mode pop-up menu information.

Two pale gray vertical lines also appear in the MSEG display, each with a small triangular handle shown in the ruler. These are the envelope loop markers. The marker with the left-facing handle sets the loop end point. The marker with the right-facing handle sets the loop start point.

The loop markers are moved by dragging the handles horizontally. The loop start marker cannot be moved to the right of the loop end marker, and the loop end marker cannot be moved to the left of the loop start marker. Loop markers always snap to the nearest point. Also see the Loop Mode pop-up menu information.



MSEG controls

- *Current MSEG pop-up menu:* Access each MSEG control panel by choosing a number from the pop-up menu or with the Previous and Next arrows. Alchemy provides up to 16 envelopes—one by default, but more if you create them when assigning modulators.
- *File button:* Open a pop-up menu with a number of MSEG-related commands.
  - *Preset submenu:* Choose a preset envelope shape. This can be used as is, or as a starting point for your own envelope shapes.
  - *Save:* Save the current envelope. A dialog opens in which you can name and save the envelope file (*.mse). The new envelope name appears at the bottom of the Preset submenu.
  - *Copy/Paste:* Use these commands to copy settings between envelopes.
  - *Clear:* Initialize the MSEG.
  - *Randomize:* Create a random envelope shape. This can be used as is, or as a starting point for your own envelope shapes.

- *Trigger pop-up menu:* Choose On to retrigger the MSEG (start from zero) with each new played note. Off triggers the MSEG for the first note only. Voice On/Effects Off modulates voice parameters only, not effects. In Off mode, the MSEG retriggers only for notes received after all other notes have been released.

  Note: Settings from earlier application versions may not behave as expected because of the additional trigger option. Change to Voice On/Effects Off to match the former version behavior, then resave the setting.

- *Sync button:* Turn on to synchronize the MSEG with the project tempo. When you adjust or create envelope points, they snap to bars and beats. Turn off to set envelope point positions freely.

- *Snap Y pop-up menu:* Quantize point levels (or y values), limiting them to exact fractions of the available range. For example, a Snap Y setting of 1/3 snaps point levels to the values 0, 1/3, 2/3, and 1 when a point is dragged. Off turns quantization off and lets you set point levels freely. You can also step through Snap Y values with the Previous and Next buttons (the arrows).

  Note: The Snap Y setting doesn't move existing point levels into alignment with quantized positions; it only affects the response of points when created or dragged.

- *Loop Mode pop-up menu:* Choose one of four MSEG loop modes.

  - *None:* Looping is turned off. The loop markers remain visible in the MSEG display but have no effect in this mode.

  - *Continuous:* The loop section plays continuously in a forward direction while a note is held and continues after the note is released.

  - *Sustain:* The loop section is played while a note is held. When the note is released, the remainder (or release section) of the envelope plays.

  - *Forward/Back:* The loop section plays alternately in a forward/backward direction while a note is held and continues after the note is released.

- *Edit Mode pop-up menu:* Choose one of three modes that determine how the MSEG responds to edits.

  - *Normal:* Only the selected point or points are moved. Non-selected points remain in their current positions.

  - *Slide:* Dragging the selected point or points also moves all subsequent envelope points, retaining the relative distance between points.

  - *Stretch:* Dragging the selected point or points to the left compresses earlier points and stretches later points. Dragging to the right stretches earlier points and compresses later points. In either case, the total length of the envelope is preserved.

- *MSEG display:* Shows a graph of MSEG (Multiple Segment Envelope Generator) output, shown as points connected by lines.

- *Scroll bar and zoom controls:* Drag the middle of the scroll bar to view envelope points that aren't visible in the display area. Horizontally drag the zoom controls at either end of the scroll bar to resize the contents of the visible display area.

Add or remove an MSEG point

- In Alchemy in Logic Pro, to add a point, click on or near a line at the required position in the MSEG display.
- To remove a point, double-click it.

  Note: Make sure the appropriate Snap Y option is active before creating points. Also be mindful of the Sync button state.

Move one or more MSEG points

- In Alchemy in Logic Pro, drag a point to the required position.

  The x-axis value (time position) and y-axis value (level) are shown at the pointer position as you drag.

  Note: Hold down the Command key before start dragging a point to restrict movement to the horizontal direction. Hold down the Command key after you've started dragging to restrict movement to the vertical direction.

  Note: Hold down the Shift key while dragging to adjust more precisely.
- To move several adjacent points together as a group, first drag to select multiple points, then drag any point in the group to the required position.

  Note: Make sure the appropriate Edit Mode option is active before moving points. Also be mindful of the Sync button state.

Adjust MSEG envelope segment curvature

In Alchemy in Logic Pro, you can adjust the curvature of each segment between points. Convex, linear, and concave envelope segments produce characteristically different effects.

- Drag the line between points upward to make the segment progressively more convex.
- Drag the line between points downward to make the segment progressively more concave.
- Option-click the line to reset the segment to a linear slope.

Create an MSEG modulation routing for pitch control

Though this example is specific to pitch modulation, you can follow these steps to create a modulation routing for other parameters such as Pan or filter Cutoff.

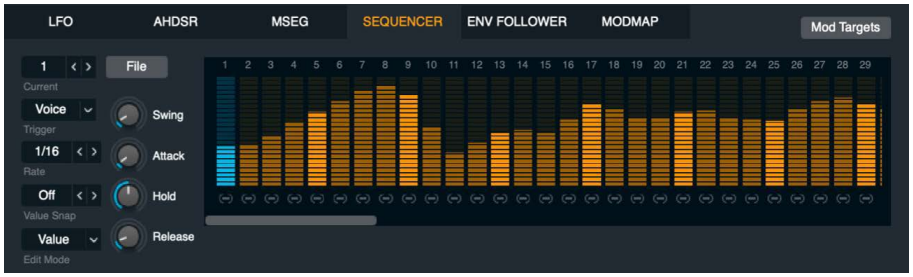
1. In Alchemy in Logic Pro, click the File button in the Name bar and choose Initialize Preset from the pop-up menu to reset all Alchemy parameters to default settings.

   Important: Use of Initialize Preset automatically turns on wide unison mode. See Alchemy wide unison mode.
2. In the master voice section, click the Coarse Tune knob, anSEG 1 as its modulator.
3. In the modulation rack, set the MSEG 1 Depth control to 24 semitones.
4. If the MSEG control panel is not visible, click the MSEG button at the top of the modulation section.
5. From the Snap Y pop-up menu, choose 1/24.
6. You can now use the MSEG envelope to control pitch in semitone increments, over a two-octave (a 24 semitone) range.

Alchemy sequencer in Logic Pro for Mac

Modulation components are shown only in advanced view. Click the Advanced button to switch to advanced view.

The sequencer module is a programmable step-based modulator that can play patterns of up to 128 steps that are synchronized with the project tempo. The level of each step, groove/swing, and envelope can be controlled globally for the pattern and per step.



Alchemy sequencer controls

- *Current sequencer pop-up menu:* Access each Sequencer control panel by choosing a number from the pop-up menu or with the Previous and Next arrows. Alchemy provides up to 16 sequencers—one by default, but more if you create them when assigning modulators.
- *Trigger pop-up menu:* Choose On to play the sequencer pattern from the beginning with each MIDI note on. Off makes the sequencer run continuously. Voice On/Effects Off modulates voice parameters only, not effects. The sequencer module is always synchronized with the Logic Pro tempo, regardless of the Trigger setting.

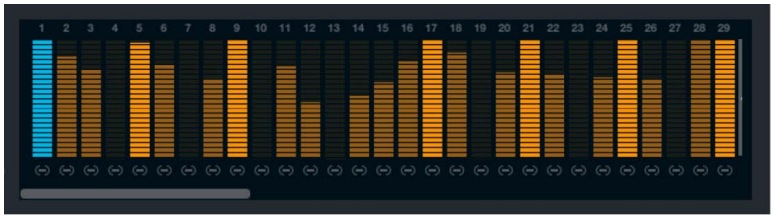
  Note: Settings from earlier application versions may not behave as expected because of the additional trigger option. Change to Voice On/Effects Off to match the former version behavior, then resave the setting.
- *Rate pop-up menu:* Set the duration of every step in the sequencer pattern, expressed as a fraction of a beat. For example, assuming a quarter-note beat, a value of 1/2 produces eighth note steps and a value of 1/4 produces sixteenth note steps. You can also step through Rate values with the Previous and Next buttons (the arrows).
- *Value Snap pop-up menu:* Quantize sequencer step values, limiting them to exact fractions of the available range. For example, a setting of 1/2 aligns bars to the values 0%, 50%, and 100% when dragged vertically. Off turns quantization off, which lets you set step values freely. You can also step through available values with the Previous and Next buttons (the arrows).

  Note: Value Snap doesn't move existing step values into alignment with quantized positions; it only affects the response of step values when you move or create them in the step editor.

- *Edit Mode pop-up menu:* Choose one of three edit modes. Your choice affects the appearance and behavior of the step editor to the right.
  - *Value:* View and edit the value of each step in the pattern.
  - *Length:* View and edit the length of each step in the pattern. Technically, you create a pattern of longer and shorter envelope shapes so that the attack, sustain (gate), and release stages fill a larger or smaller portion of the fixed step duration. These lengths are combined with the overall sequencer Attack/Gate/Release settings to determine the envelope shape of each step.
  - *Swing:* Create variations in the timing of steps. Each swing value ranges from 0 to 2; the middle value of 1 represents standard timing, while smaller values play earlier and larger values play later. These swing values are combined with the overall sequencer timing pattern set with the Swing knob.
- *File button:* Open a pop-up menu with sequencer-related commands.
  - *Preset submenu:* Choose a preset sequencer configuration. This can be used as is, or as a starting point for your own sequences.
  - *Save:* Store the current sequencer configuration (step values, lengths, and swing settings plus the sequencer control settings) in a new disk file (*.seq). The new sequence name appears at the bottom of the Preset submenu.
  - *Copy/Paste:* Store the current sequencer configuration in the Clipboard. Paste applies the configuration currently found on the Clipboard.
  - *Clear:* Reset the sequencer controls and pattern to default settings.
  - *Randomize:* Apply random offsets to the Swing, Attack, Gate, and Release parameter values.
  - *Import:* See the Import data from a MIDI file task in this section.
- *Swing knob:* Adjust timing to create swing effects. Values over 0% increase the duration of odd-numbered steps (1, 3, 5, and so on) while decreasing even-numbered step lengths.
- *Attack knob:* Set the time required for each step to reach its peak level.
- *Step editor:* Shows steps numbered from left to right across the top of the display. Steps are represented by vertical bars that you can edit directly. Tie buttons appear below each step. The appearance and behavior of the step editor is determined by the active Edit Mode menu choice.
- *Pattern Length control:* Drag the gray vertical bar < > left or right to increase or decrease the number of steps.
- *Scroll bar:* Drag the middle of the scroll bar to view steps that are not visible in the display area.

Use the Alchemy sequencer step editor

In Alchemy in Logic Pro, the step editor shows steps numbered from left to right across the top of the display. The appearance and behavior of the step editor is determined by the active Edit Mode menu choice. In the default Value mode, for example, the value of each step is represented by a vertA blank column is shown when the value is zero.



- Drag a bar vertically to adjust its value, or click directly at the required height.
- Set the values of multiple bars by dragging horizontally across them.

Import data from a MIDI file into the Alchemy sequencer

Alchemy can extract information from a short MIDI file and can apply this information to patterns in the step editor.

You can extract velocity data and set step values to match. If the MIDI file consists of notes of equal duration, such as a succession of eighth or sixteenth notes, every step in the resulting pattern will have an associated non-zero value. If the MIDI file consists mainly of notes of equal duration with occasional gaps, such as a succession of eighth notes with occasional eighth rests, the gaps are represented by step values of zero. If the MIDI file has irregular timing, or if it consists of chords rather than single notes, the results of this process are less predictable and usually less useful.

You can extract groove data (timing inflections) and set step swing values to match. If the MIDI file consists of nearly equal durations, such as eighth notes or sixteenth notes with timing inflections, this process yields useful results.

You can also extract note pitch data and set step values to match. The MIDI file should consist of equal durations with no gaps and should be limited to single pitches between a low C and a C two octaves higher. For example, a MIDI file could consist of notes C1 through C3, with the low C corresponding to a step value of zero. Because the pitch range is always two octaves, you should use the sequencer to modulate pitch with a depth of 24 semitones in order to reproduce the pattern of notes in the original MIDI file.

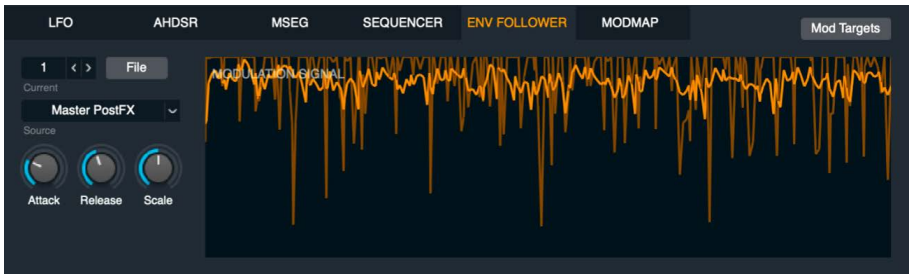
- In Alchemy in Logic Pro, choose one of the following Import commands from the Sequencer File pop-up menu.
  - *Import Velocity:* Set step values based on extracted velocity data and swing values based on extracted groove data.
  - *Import Note:* Set step values based on extracted note data and swing values based on extracted groove data.
  - *Import Groove:* Set swing values based on extracted groove data.

Alchemy envelope follower in Logic Pro for Mac

Modulation components are shown only in advanced view. Click the Advanced button to switch to advanced view.

The Env Follower module provides an envelope follower with Attack, Release, and Scale parameters. The Attack and Release times can be set independently. You can set the overall Scale, or amplitude, of the Env Follower output.

Adjustments to Attack and Release times cause a slowdown in the reaction time of the envelope follower to the input signal. The display shows changes in real time.



Env Follower parameters

- *Current Env Follower pop-up menu:* Access each Env Follower control panel by selecting a number from the pop-up menu or with the Previous and Next arrows. Alchemy provides up to 16 envelope followers—one by default, but more if you create them when assigning modulators.
- *File button:* Open a pop-up menu with a number of envelope follower handling commands.
  - *Preset submenu:* Choose a preset envelope follower setting. This can be used as is, or as a starting point for your own envelope follower settings.
  - *Save:* Save the current envelope follower settings. A dialog opens in which you can name and save the envelope file (*.ef). The new envelope name will appear in the Preset submenu.
  - *Copy/Paste:* Use these commands to copy settings between envelope followers.
  - *Clear:* Initialize the envelope follower.
- *Source pop-up menu:* Choose a source for the envelope follower from the Audio or Modulators submenu.
  - *Audio:* The output of any individual Alchemy source, master filters 1 or 2, master output pre- or post-effects, or the side chain input.
  - *Modulators:* Choose any active modulation source, Note Property, MIDI source, or Perform control.

    Note: The Source pop-up menu selection is saved with an envelope follower preset, but is only restored to the same modulator if the corresponding modulation source is active at the time of loading.
- *Attack knob:* Set the time required to reach the peak amplitude level of the audio or modulation source. Longer attack times cause the rise time to be lazy.

- *Release knob:* Set the time required to fall from the peak amplitude level reached during the attack phase to the current level of the audio or modulation source.

  Tip: Set the Attack time and Release time values to zero to closely follow the input source. When the envelope follower Source pop-up menu choice is audio, or a very fast modulation, this closely approximates the source envelope shape.

  Tip: Assign an LFO as the source and modulate the Attack, Release and Scale parameters with other modulation sources to create new and interesting LFO shapes not available from the LFO menu.

- *Scale knob:* Set the amplitude of the output signal, after the envelope follower has processed the incoming signal.

  Tip: Because an Env Follower modulation source may be assigned to multiple targets, the Scale parameter can be used to adjust the modulation depth of multiple target parameters simultaneously.

- *Env Follower display:* Shows the input source (dimmed), and the resulting signal after processing (bright). The display does not update if the Env Follower panel is inactive.

Alchemy ModMap in Logic Pro for Mac

Modulation components are shown only in advanced view. Click the Advanced button to switch to advanced view.

A ModMap is not a modulator. Instead, its purpose is to process the output of a modulator, mapping the original values to new ones before they are applied to a modulation target. ModMaps let you create curved velocity responses, scale the volume of each source across the keyboard, quantize the pitch response to a random-LFO modulation so it aligns with the steps of a scale, and much more.

Mapping is defined by the graphical shape of the ModMap, which represents a transfer function. The x (horizontal) axis represents the range of original modulation values, from 0.00 to 1.00. The y (vertical) axis represents the range of mapped modulation values, also ranging from 0.00 to 1.00. To see how a modulation value is affected by the ModMap, look at the original value along the x-axis; the corresponding y value determines the output of the mapping.

- A convex ModMap maps the middle range of inputs to values that are higher than the default output.
- A concave ModMap maps the same range of inputs to values that are lower than the default output.
- A horizontally flat ModMap maps a range of inputs to a single output.



ModMap parameters

- *Current ModMap pop-up menu:* Access each ModMap control panel by choosing a number from the pop-up menu or with the Previous and Next arrows.

  ModMaps are created or deleted with the ModMap pop-up menu commands in the modulation rack. The default ModMap does nothing because the output is identical to the input.

- *File button:* Open a pop-up menu with a number of ModMap-related commands.
  - *Preset submenu:* Choose a preset ModMap. This can be used as is, or as a starting point for your own ModMaps.
  - *Save:* Save the current ModMap. A dialog opens in which you can name and save the ModMap file (*.mma). The new ModMap name appears at the bottom of the Preset submenu.
  - *Copy/Paste:* Use these commands to copy settings between ModMaps.
  - *Clear:* Initialize the current ModMap.
- *Snap X pop-up menu:* Quantize the original point values, limiting them to exact fractions of the available range. For example, a Snap X setting of 1/3 snaps point values to 0, 1/3, 2/3, and 1 when a point is dragged. Off turns quantization off and lets you set point levels freely. You can also step through Snap X values with the Previous and Next buttons (the arrows).

  Choose Key mode to change the ModMap display to a familiar keyboard layout. Key mode works for any modulation source but is particularly useful for Key Follow sources, where a specific modulation amount can be assigned to individual notes.

- *Snap Y pop-up menu:* Quantize point levels (or y values), limiting them to exact fractions of the available range. For example, a Snap Y setting of 1/3 snaps point levels to the values 0, 1/3, 2/3, and 1 when a point is dragged. Off turns quantization off and lets you set point levels freely. You can also step through Snap Y values with the Previous and Next buttons (the arrows).

  Note: The Snap X and Snap Y settings do not move existing point levels into alignment with quantized positions. These parameters only affect the response of points when created or dragged.

- *ModMap display:* Shows a graphical representation of the ModMap, which represents a transfer function.

Add or remove a ModMap point

- In Alchemy in Logic Pro, to add a point, click on or near a line at the required position in the ModMap display.
- To remove a point, double-click it.

  Note: Make sure the appropriate Snap X/Y option is active before creating points.

Move a ModMap point

- In Alchemy in Logic Pro, drag a point to the required position.

  The x-axis and y-axis values are shown at the pointer position as you drag.

  Note: Hold down the Shift key while dragging to adjust more precisely without any snap restrictions.

Change the velocity curve with a ModMap

1. In Alchemy in Logic Pro, click the File button in the Name bar and choose Initialize Preset from the pop-up menu to reset all Alchemy parameters to default settings.

   Important: Use of Initialize Preset automatically turns on wide unison mode. See Alchemy wide unison mode.

2. Click the Master Vol knob to show the modulation rack in the modulation section. Note the two modulators loaded in the modulation rack: AHDSR 1 in the first slot, Velocity in the second.

3. Choose 1 from the pop-up menu to the right of the Velocity slot in the modulation rack.

   This applies the default ModMap to Velocity modulation of Master Vol.

4. In modulation rack slot 1, set the Depth control to 100%.

5. Play a few notes on your MIDI keyboard to confirm that the velocity response is unchanged by the default ModMap.

6. Drag the middle of the ModMap line upward to create a convex curve.

7. Play a few more notes on your MIDI keyboard, and listen to the effect of the convex velocity curve: notes you strike with medium force play louder than they did with the default curve.

8. Drag the middle of the ModMap line downward until the segment curves in the opposite direction, becoming concave.

9. Play a few more notes on your MIDI keyboard, and listen to the effect of the concave velocity curve: notes you strike with medium force play softer than they did with the default curve.

Use a ModMap to scale a single source level in a multi-source preset

This example covers a preset where two sources play at full level from the bottom of the keyboard up through approximately C4. Above this point, source A remains at full strength while source B gets softer as you play further up the keyboard.

1. In Alchemy in Logic Pro, click the File button in the Name bar and choose Initialize Preset from the pop-up menu to reset all Alchemy parameters to default settings.

   Important: Use of Initialize Preset automatically turns on wide unison mode. See Alchemy wide unison mode.

2. Click the Global button to view all sources, then load one sample (or multi-sample) into source A, and load a contrasting sample (or multi-sample) into source B.

   By default, there is a 50% crossfade between these sources, so you should hear a balanced mix of the two samples across the entire keyboard range.

3. Click the source B Vol knob to show the modulation rack in the modulation section.

4. In the first slot of the modulation rack, choose Note Property > Key Follow.

   The amplitude of source B is modulated according to MIDI note number, but this modulation does not yet have the required shape. Currently, the lowest notes are softest, notes in the middle of the keyboard are medium-loud, and notes at the top are loudest.

5. Choose 1 from the pop-up menu to the right of the source B slot in the modulation rack.

   This applies the default ModMap to Key Follow modulation of source B Vol.

6. In modulation rack slot 1, set the Depth control to 100%.

7. Play a few notes on your MIDI keyboard to confirm that the Key Follow response is unchanged by the default ModMap.
8. Edit the ModMap points as follows:
   - Drag the leftmost point up to a level of 1.00.
   - Click at a position along the line slightly to the right of the halfway mark to create a new point. Leave the level of this new point at 1.00.
   - Drag the rightmost point down to a lev.
9. Play across the keyboard range to hear the source B Vol response. If the response is not to your taste, make further adjustments to the point values or positions in the ModMap.

Use a ModMap to quantize pitch modulation to a pentatonic scale

1. In Alchemy in Logic Pro, click the File button in the Name bar and choose Initialize Preset from the pop-up menu to reset all Alchemy parameters to default settings.

   Important: Use of Initialize Preset automatically turns on wide unison mode. See Alchemy wide unison mode.
2. Click the master voice section Coarse Tune knob to show the modulation rack in the modulation section.
3. In the first slot of the modulation rack, choose LFO > LFO 1, and set the modulation Depth to –12.0 semitones.
4. Adjust LFO 1 settings as follows:
   - Shape = Basic > Random Hold
   - Rate = 1/2 beats
   - Bipolar = off
5. Play and hold a note to confirm that the pitch changes freely within the 12 semitone range, twice per beat.
6. Switch the view to Modmap, then set SnapX = 1/6 and SnapY = 1/12. These functions make it easier to create the required ModMap shape.
7. Click along the line to create ten new points between the first and last points.
8. Edit the ModMap points as follows:

X Positions	Y Positions
1/6	0/12
1/6	2/12
2/6	2/12
2/6	4/12
3/6	4/12
3/6	7/12
4/6	7/12
4/6	9/12
5/6	9/12
5/6	12/12

Alchemy MIDI control modulators in Logic Pro for Mac

Modulation components are shown only in advanced view. Click the Advanced button to switch to advanced view.

You can choose one of six different modulation sources from the MIDI submenu in the modulation rack. These are ideal for adding breath and foot controller modulations, for example.

Choose MIDI continuous controllers as modulators in the modulation rack

In Alchemy in Logic Pro, click a slot in the modulation rack, then choose one of the following MIDI control options from the MIDI submenu:

- *Pitchbend:* The pitch bend wheel.
- *ModWheel:* The modulation wheel.
- *Ctrl A (2 Breath):* MIDI continuous controller 2 (Breath).
- *Ctrl B (4 Foot):* MIDI continuous controller 4 (Footpedal).
- *Ctrl C (74 Bright):* MIDI continuous controller 74 (Brightness).
- *Ctrl D (75):* MIDI continuous controller 75 (General purpose).

  The CTRL A to D parameters are the defaults shown in the Alchemy extended parameters. Click the disclosure triangle at the lower left of Alchemy to view and change these parameter assignments.

Alchemy note property modulators in Logic Pro for Mac

Modulation components are shown only in advanced view. Click the Advanced button to switch to advanced view.

Several properties of incoming MIDI note data, as well as values generated per-note by Alchemy, are available as modulation sources.

Note: There is no Note Property control panel display.

Set note properties as modulators

In Alchemy in Logic Pro, click a slot in the modulation rack, then choose one of the following options from the Note Property submenu:

- *Velocity:* Modulation based on the velocity values of incoming MIDI note data.
- *Key Follow:* Modulation based on incoming MIDI note numbers. The modulation value glides to the value of higher-pitched notes. The Glide value is determined by the Glide parameter in the master voice section. See Alchemy master voice section.
- *Key Follow Fixed:* As per Key Follow, but the modulation value increases as you play higher pitches on your MIDI keyboard. This is a bipolar source, with C3 corresponding to zero.
- *Aftertouch:* Modulation based on channel or polyphonic aftertouch data.
- *Speed:* Modulation based on the elapsed time between notes. A progressively slower sequence of notes results in progressively greater modulation values.

- *Held:* A modulation signal that rises to full-scale immediately at note on and falls to zero immediately at note off.
- *Flip Flop:* A modulation signal that is alternately full-scale and zero on successive notes.
- *Flip Flop 2:* Like Flip Flop, but the value reverses every two notes: zero, zero, full, full, in a repeating pattern.

  Note: The Flip Flop modulators can be used together to create a round robin involving all four sources. To do this, set Morph mode to Morph XY or XFade XY, set the Morph X and Y knobs to 0%, then modulate X with Flip Flop and Y with Flip Flop 2 or vice versa. See Morph controls.
- *Stepped A2/3/4:* Stepped A4 behaves like Stepped 4, except it resynchronizes to zero when the Arp is at the start of a pattern. Stepped A2 behaves like Flip Flop, except it resynchronizes to zero when the Arp is at the start of a pattern. Stepped A3 is similar to Stepped 4, except with three steps.
- *Stepped 4/8/16:* Similar to Flip Flop, but cycles between the number of values (4, 8, or 16) at each subsequent note on. Steps are spaced at equal increments, starting from a value of zero. If you need further control, you can use a ModMap to define values for each step. See ModMap parameters.
- *Random 1-4:* Modulation based on a fixed random value per note. This is a unipolar source with values ranging from zero through full-scale. The four random sources are independent of each other.
- *Max:* Modulation based on a constant full-scale value.
- *Polyphony:* A modulation signal that is larger when there are more notes playing and smaller when there are fewer notes playing. Technically, the current number of played notes is divided by the total polyphony. This parameter may be useful for reducing volume when lots of notes are played. Polyphony is determined by the Num parameter in the master voice section. See Alchemy master voice section.
- *Playhead A/B/C/D:* Assign parameters for arpeggiator playhead position control.

  The assigned parameters are controlled by the arpeggiator playhead position of each source.
- *ArpMod A/B/C/D 1-2:* Assign parameters for arpeggiator control.

  The assigned parameters are controlled by the arpeggiator. The Poly Mod 1 and Poly Mod 2 pop-up menu assignments in the arpeggiator correspond to these parameters. When the arpeggiator is in All mode (denoted by the A), one arpeggiator is used for all sources. When you use multiple arpeggiators to control individual sources, Poly Mod options A/B/C/D are shown in the Note Properties pop-up menu. See Arpeggiator sequencer controls.

Alchemy perform control modulators in Logic Pro for Mac

Modulation components are shown only in advanced view. Click the Advanced button to switch to advanced view.

You can choose one of sixteen different modulation sources from the Perform submenu in the modulation rack. These correspond to the eight assignable knobs, two X/Y control squares, and four envelope knobs shown in the Perform section.

Set performance controls as modulators in the modulation rack

In Alchemy in Logic Pro, click a slot in the modulation rack, then choose one of the following from the Perform submenu:

- *Control 1-8:* Assignable knobs, numbered 1–8.
- *XYPad 1X:* The X (horizontal) axis of the left XY control square.
- *XYPad 1Y:* The Y (vertical) axis of the left XY control square.
- *XYPad 2X:* The X (horizontal) axis of the right XY control square.
- *XYPad 2Y:* The Y (vertical) axis of the right XY control square.
- *Mod Attack:* The Attack knob.
- *Mod Decay:* The Decay knob.
- *Mod Sustain:* The Sustain knob.
- *Mod Release:* The Release knob.

  Your choice is immediately mapped to the chosen Perform section control.

Alchemy Perform section

Alchemy performance controls in Logic Pro for Mac

When simple view is active, or if the Perform button is on in the Perform/Arp/Effects section in other views, the performance controls subpage is displayed.



The performance controls subpage provides eight knobs, two XY pads, and a set of ADSR envelope knobs for a total of sixteen controls. These can be assigned to modulate parameters in the same way as other modulation sources. In addition to these assignable controls, the Perform section has its own gain stage with a dedicated Snapshot Vol control. Also available is the Transform pad that lets you capture morphable snapshots of performance control settings. All performance controls are available as automation parameters for track and region automation.

Each performance control type is discussed in the linked sections.

- Transform pad. See Transform pad.
- Performance control knobs. See Performance control knobs.
- XY pads. See XY pad controls.
- Performance envelope controls. See Performance control envelopes.

Use performance controls effectively

Performance controls are a custom set of controls that modify the values of *key* sound parameters. You can use the Auto Assign feature, which seeks to find the most relevant parameters of the current preset to map to performance controls. Auto Assign can often provide a good starting point for assignments, but it is much better if you take the time to make manual assignments.

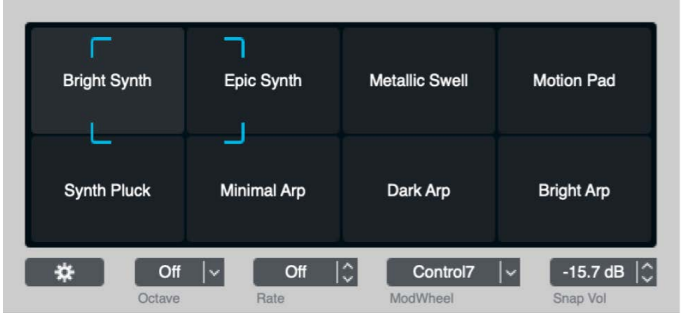
As you explore the included sound library, you may notice that many presets have a similar performance control layout. Some presets may require special performance controls, but the following guidelines work well for most presets.

- *Control knobs:* Provide quick access to a sound's main characteristics. You can assign knobs to other parameters in advanced view.
  - *Knobs 1 and 5:* Often used for Delay (1) and Reverb (5) level. If other effects are used, other effect parameters can be substituted here. Control of effects mix levels should take priority over other parameters, assigned to Knob 1.
  - *Knobs 2 and 6:* Often used for filter cutoff and resonance. If your preset doesn't use a filter, you might consider adding one in a fully open state in the Effects section.
  - *Knobs 3 and 7:* Often used to control rhythm or movement in a preset with LFO or arpeggiator parameter assignments.
  - *Knobs 4 and 8:* Often used for timbre, pitch, or other source parameters.
- *XY pad 1:* Generally used for morphing among sources. If more than one source is used, this XY control is the place to assign control of morphing or crossfading.
- *XY pad 2:* Can be used for control over additional morphing or effects parameters or for features unique to a particular preset.
- *Envelope knobs:* Normally control the corresponding parameters in AHDSR 1. You can reassign to other parameters in advanced view.
- *ModWheel:* Typically linked with the most playable performance control and should always be functional for the first snapshot. Take care with modwheel assignments to make sure settings of 50% or lower don't cause undesirable results, such as a dropout of the sound (from a very low minimum lowpass cutoff setting, for example). This is particularly important if your MIDI controller has a spring-loaded joystick.
- *Octave:* Turn off if you want to play your sound across the entire keyboard range.
- *Aftertouch:* Should always be assigned to something useful for the preset.

Alchemy Transform pad in Logic Pro for Mac

When simple view is active, or if the Perform button is on in the Perform/Arp/Effects section in other views, the performance controls subpage is displayed.

Alchemy presets contain eight snapshots of performance control settings, numbered 1 to 8. By default, snapshot 1 is selected. This is indicated by the framing box.



Transform pad controls

In addition to the parameters outlined, you can Control-click directly on snapshots in the Transform pad to open a shortcut menu with a number of snapshot and other commands. See the tasks in this section for details about use of these commands.

- *Transform pad:* Drag the framing box to different positions to morph between performance snapshots in real time. The performance control knobs update, modulating the associated parameters.
- *Action menu button:* Open a pop-up menu with a number of snapshot-related commands.
  - *Copy Snapshot:* Store all performance control settings of the selected snapshot in a buffer.
  - *Paste Snapshot:* Paste all performance control settings from the buffer to the selected snapshot.
  - *Copy 1 To All:* Copy all performance control settings of snapshot 1 into snapshots 2 to 8.
  - *Clear:* Capture the current performance control values and apply them directly to target parameters, preserving the current sound. Following use of the command, all performance control and Transform pad assignments are cleared, all controls are reset to zero, and the knob and pad name fields are wiped.
  - *Randomize Snapshots:* Create random performance control values for all snapshots except the currently active snapshot.
  - *Auto Assign All:* Automatically assign all performance controls to appropriate parameters in the current preset. This command also generates a random set of snapshots that are derived from the settings in snapshot 1. This is the quickest way to start with the performance controls.
  - *Auto Assign Empty:* Automatically assign all *unused* performance controls to appropriate parameters in the current preset. Existing performance control assignments are not affected, and no snapshot settings are created.

- *Snapshot Update:* Turn on to immediately write changes of performance control values to the nearest Transform pad snapshot. This prevents the performance knob from reverting to its previous position when you move a performance knob and then move the Transform pad framing box slightly.

  Tip: This parameter is on by default, but you can turn it on or off as preferred and then save the default preset in the plug-in header to recall this behavior in new presets.

- *Octave pop-up menu:* Define an octave range used to switch between Transform pad positions when you play notes within the specified range.
- *Rate field:* Drag vertically or use the arrows to set the speed of movement (interpolation) between snapshots when switching from a MIDI keyboard.
- *ModWheel pop-up menu:* Assign your keyboard modulation wheel to a performance control. This can be any knob, the x- or y-axis of pads, or the envelope controls. Set to None to turn off modwheel control of performance controls. A setting of None is remembered between presets. See Performance control knobs, XY pad controls, and Performance control envelopes.

  Each modwheel target offers an inverted option. This is useful when using a spring-loaded modulation wheel as it stops the target performance control from closing when the modulation wheel is released.

- *Snap Vol field:* Set the intensity of change from one snapshot to the next. In essence, this sets an initial level between snapshots.

  All snapshots should be approximately the same volume, with a peak level around –5 dB. You should also be mindful of relative volume levels between snapshots. For example, a soft pad and a distorted lead could both peak at –5 dB, but the lead may seem louder. In this scenario, lower the volume of the lead, rather than increase the pad level, so that relative volume levels seem similar between the pad and lead sounds.

Create and store a Transform pad snapshot

Unique variations should be created for each snapshot. Though the degree of variation is up to you, the changes should be significant, rather than simple filter or effect level changes. If your snapshots all sound like basic variations of the same sound, look at changing your performance control assignments to make them more flexible and complex. This will allow you to create more interesting and dynamic snapshots.

1. In Alchemy in Logic Pro, click the Transform pad snapshot you want to use.
2. Set each of the performance controls as required.
3. Control-click anywhere on the Transform pad, then choose Store Current Snapshot from the shortcut menu.

   Note: Store Current Snapshot only applies if Update Snapshot is turned off in the Transform pad File pop-up menu, if the framing box is not directly aligned with a Transform pad slot, or if you want to save the current settings to a different Transform pad slot.

Name or rename the selected Transform pad snapshot

Aim for a unique and descriptive name for each snapshot. Keep things simple and clear; if snapshot 1 is a bass, snapshot 2 is a pad, and snapshot 3 is arpeggiated, this should be represented in the name. If your variations are less straightforward to differentiate, use a name that describes the effect on the sound.

Note: Snapshot names are indexed and appear as text search results in the Browser. Snapshot names are also passed to Smart Controls.

1. In Alchemy in Logic Pro, click the Transform pad snapshot you want to name or rename.
2. Control-click anywhere on the Transform pad, then choose Rename Current from the shortcut menu.
3. Enter text in the highlighted text input field, then press Enter.

Replicate the selected Transform pad snapshot

1. In Alchemy in Logic Pro, click the Transform pad snapshot you want to replicate.
2. Control-click anywhere on the Transform pad, then choose Store Current Snapshot from the shortcut menu.
3. Control-click anywhere on the Transform pad, then choose Store Snapshot > (target snapshot) 1-8, or All from the shortcut menu.

   The stored snapshot is replicated in the target snapshot. If All is chosen, all eight snapshots are identical.

Swap a Transform pad snapshot

1. In Alchemy in Logic Pro, click the Transform pad snapshot you want to swap.
2. Control-click anywhere on the Transform pad, then choose Swap Current Snapshot from the shortcut menu.
3. Choose the target snapshot 1-8 from the submenu.

   The positions of the two snapshots are swapped.

Switch between Transform pad snapshots with your MIDI keyboard

1. In Alchemy in Logic Pro, select a MIDI note name, such as C4 from the Octave pop-up menu below the Transform pad.

   The first eight notes of the chosen octave are assigned to snapshots 1–8.
2. Press one of the assigned keys on your controller to select the corresponding snapshot.

   Note: MIDI notes assigned to snapshots do not trigger Alchemy voices. Choose Off from the Octave menu to make notes play across the entire range of MIDI note numbers.
3. Drag vertically in the Rate field to set the speed of movement between snapshots. You may also want to adjust the Snap Vol field value to set initial levels between snapshots.

Alchemy performance control knobs in Logic Pro for Mac

When simple view is active, or if the Perform button is on in the Perform/Arp/Effects section in other views, the performance controls subpage is displayed.

The eight knobs can each be named by double-clicking in the field below a knob and typing a name. When a custom name is assigned, auto-assign does not update the name. Delete the performance control text to turn auto-assigned names back on.

Control-click any perform knob to open a shortcut menu with a number of useful commands. Most of these commands are also available for the Perform section envelope controls. See Performance control envelopes.



Performance control knob shortcut menu items

- *Delete Modulation:* Shows a list of all targets receiving modulation from the knob. Select an item in this list to delete the corresponding modulation assignment.
- *Swap With:* Exchange all modulation assignments between the selected knob and any other performance control selected from a list.
- *Copy Setting to all Snapshots:* Updates all eight snapshots with the current value of the selected knob, which replaces the stored value of that knob in each snapshot.

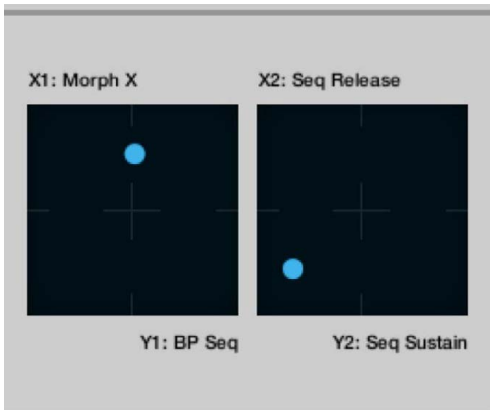
  For example, if you want to set the master Sustain knob to a value of 100% in every snapshot, turn the knob to the 100% position, Control-click it, then choose this command to apply the new setting to all eight snapshots.
- *Auto Assign:* Set a modulation target for the selected control and update the label.
- *Lock Control:* Turn on to prevent the Transform pad from affecting the assigned control. This is useful when you want to automate the assigned parameter independently of the Transform pad.
- *Invert Knob:* Swap the minimum and maximum values of the selected performance control and associated modulation targets. Technically, this parameter modifies the parameter range.

Alchemy XY pads in Logic Pro for Mac

When simple view is active, or if the Perform button is on in the Perform/Arp/Effects section in other views, the performance XY pads controls subpage is displayed.

XY pads: Drag the handles to modulate two parameters simultaneously. The x- and y-axes of each XY pad can be named independently by double-clicking the fields above or below and typing a name. When a custom name is assigned, auto-assign does not update the name. Delete the performance control text to turn auto-assigned names back on.

Control-click an XY pad to open a shortcut menu containing similar commands for both the x- and y-axis.



X/Y pad shortcut menu items

- *Delete Modulation X/Y:* Shows a list of all targets receiving modulation from the x- or y-axis. Select an item in this list to delete the corresponding modulation assignment.
- *Swap X/Y With:* Exchange modulation assignments between the x- or y-axis and any other performance control selected from a list.
- *Copy X/Y to All Snapshots:* Updates all eight snapshots with the current value of the selected x- or y-axis, which replaces the stored value of that axis in each snapshot.
- *Auto Assign:* Set a modulation target for the selected x- or y-axis and update the label.
- *Lock Control:* Turn on to prevent the Transform pad from affecting the assigned control. This is useful when you want to automate the assigned parameter independently of the Transform pad.

Alchemy performance control envelopes in Logic Pro for Mac

When simple view is active, or if the Perform button is on in the Perform/Arp/Effects section in other views, the performance controls subpage is displayed.

There are four envelope control knobs. By default, these are linked to the corresponding parameters in the main Alchemy level envelope (AHDSR 1), but you may reassign these knobs to other parameters. Control-click any knob to open a shortcut menu with a number of useful commands. See Performance control knob shortcut menu items.



Envelope controls

- *Attack knob:* Set the attack time for AHDSR 1, or adjust the value of assigned parameter or parameters.
- *Decay knob:* Set the decay time for AHDSR 1, or adjust the value of assigned parameter or parameters.
- *Sustain knob:* Set the sustain level for AHDSR 1, or adjust the value of assigned parameter or parameters.
- *Release knob:* Set the release time for AHDSR 1, or adjust the value of assigned parameter or parameters.

Alchemy arpeggiator

Alchemy arpeggiator overview in Logic Pro for Mac

Alchemy offers a flexible and powerful arpeggiator. The arpeggiator is shown only in browse or advanced view. Click the Browse or Advanced button to switch to either view, then click the Arp button in the Perform/Arp/Effects section.



The arpeggiator subpage consists of three main sets of controls.

- *Basic arpeggiator controls:* These buttons and knobs determine the basic behavior of the arpeggiator. See Basic arpeggiator controls.
- *Arpeggiator sequencer controls:* These buttons, pop-up menus, and knobs let you edit one or more arpeggiator sequencers. These sequencers are internal to the arpeggiator and are separate from the main Alchemy sequencers. See Arpeggiator sequencer controls.
- *Arpeggiator menu commands:* These provide a number of useful functions for managing arpeggiator patterns. See Arpeggiator menu commands.

If you're ready to begin some hands-on experimenting, see Use the arpeggiator.

Alchemy basic arpeggiator controls in Logic Pro for Mac

The arpeggiator is shown only in browse or advanced view. Click the Browse or Advanced button to switch to either view, then click the Arp button in the Perform/Arp/Effects section.

The basic arpeggiator controls consist of several buttons and nine knobs.



Basic arpeggiator controls

- *All/A/B/C/D buttons:* Choose the target source that is used to play the arpeggiator pattern. See the tasks in Use the arpeggiator.
  - When set to All, the arpeggiator pattern is played by all active sources.
  - You can restrict the arpeggiator pattern to a single target source by setting values for one of the A, B, C, or D arpeggiators. The other three sources, if active, respond to incoming MIDI data with no arpeggiation.
  - You can also activate and create different arpeggio patterns for any combination of the A, B, C, or D arpeggiators. When you play the keyboard, each active source is triggered by the corresponding active arpeggiator, resulting in up to four simultaneous (but different) arpeggiated sources.

  Note: Alchemy uses separate voices to play arpeggiated and non-arpeggiated output. Therefore, this feature requires the Num(ber) of voices to be set to at least 2 in the master voice section. See Alchemy master voice section.
- *Sync button:* Turn on to synchronize the arpeggiator with the project tempo. See Rate knob.
- *Key Trigger pop-up menu:* Choose an option to align the arpeggiator and host application grid values. Incoming and playing notes snap to this value when the host is in play mode.
  - Choose a value other than Off to restart the arpeggiator pattern when the first note is struck after a pause. This allows you to create pattern variations by playing chords on different beats in the bar and may feel more responsive. Playing legato does not retrigger the arpeggiator.
  - Choose Snap to Rate when you want to use a host application rate that differs from available menu options.
  - Choose Cycle Reset to turn on or off. This option starts the arpeggiation from the first note each time the pattern is repeated and when the host application is started (at step 1). Cycle Reset is turned on by default, and turned off for all existing presets for backward compatibility.
- *Mode knob:* Turn the arpeggiator on and off, and determine the order in which incoming notes are organized into a pattern. In addition to Off, you have the following choices:
  - *Up:* Plays the current notes from lowest to highest.
  - *Down:* Plays the current notes from highest to lowest.
  - *Up/Down:* Plays the current notes from lowest to highest and back again.
  - *Down/Up:* Plays the current notes from highest to lowest and back again.
  - *As Played:* Plays the current notes in the order they were originally played.
  - *Random:* Plays the current notes in a random, non-repeating order.
  - *Chord:* Plays all held notes simultaneously as a chord. The chords are retriggered and pulse in time with the arpeggiator rate and rhythm, as determined by the arpeggiator sequencer settings. See Arpeggiator sequencer controls.

    Note: Mode is a modulation target. This lets you set a Perform knob to modulate Mode, allowing you to turn the arpeggiator on and off during a performance.

- *Rate knob:* Set the duration of each arpeggiator step. When Sync is on the arpeggiator is synchronized with the project tempo, and Rate is set in bars/beats. When Sync is off, the Rate knob sets a constant length for each step which is unaffected by project tempo changes.
- *Octave knob:* Determine if the arpeggiator pattern is played only at its original pitch or across higher octaves when the pattern is repeated.
- *Length knob:* Set the length of each arpeggiated note. At the maximum setting of 100% the length of each note is a full step. Set the length to lower values to generate shorter notes for a more staccato effect (if triggering a sound with a fast release time).
- *Pattern knob:* Choose the active pattern in the step sequencer section to the right. Sixteen different step sequencer patterns can be created and edited for each arpeggiator. By default, all patterns are identical, so this parameter has no effect until your patterns are edited.
- *Swing knob:* Change the timing of the arpeggiator, moving even-numbered steps to later positions without changing the timing of odd-numbered steps. This can add a more relaxed feel or groove to the part. Swing generally works best when combined with a Rate knob setting of 1/4 of a beat (16th note swing feel) or a Rate knob setting of 1/2 a beat (8th note swing).
- *Note Vel knob:* Determine how the arpeggiator handles note velocity values. Arp sequencer 1 modulates arpeggiator Amp values by default. When Note Vel is set to a value of 0%, the step values of Arp sequencer 1 determine the velocities of notes in the arpeggiator pattern. When Note Vel is set to a value of 100%, arpeggiator pattern note velocities are determined by incoming MIDI velocities. Intermediate Note Vel values blend the Arp sequencer step values with incoming MIDI velocities.
- *Split knob:* Set the highest MIDI note number to be included in the arpeggiator pattern. Notes above this value are not arpeggiated. Set to the maximum value (G8) to arpeggiate all MIDI input.
- *Latch knob:* Hold the arpeggiator pattern when you lift your hands from the keyboard in Hold or Add mode. Set to Off to stop the arpeggiator pattern when you lift your hands.
  - *Hold:* Newly played notes are organized into a new pattern, which replaces the existing pattern.
  - *Add:* Newly played notes are added to the existing pattern.

    Tip: Try combining As Played mode with the Add latch setting to create an interactive step sequencer with up to 128 steps.

Alchemy arpeggiator sequencer in Logic Pro for Mac

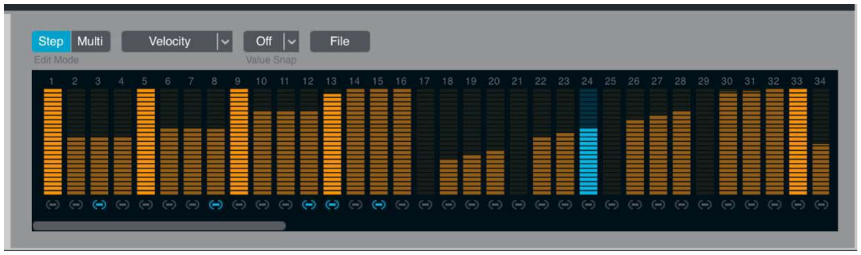
The arpeggiator is shown only in browse or advanced view. Click the Browse or Advanced button to switch to either view, then click the Arp button in the Perform/Arp/Effects section.

The arpeggiator sequencer section is similar in functionality to the step sequencer in the main modulation section, but with an important difference: the arpeggiator sequencer provides multiple sequences running in parallel, each hardwired to a different arpeggiator parameter. In addition, each arpeggiator provides a pair of Poly Mod sequencers, which can be used to modulate synthesizer parameters via the modulation matrixes. See Note property modulators.

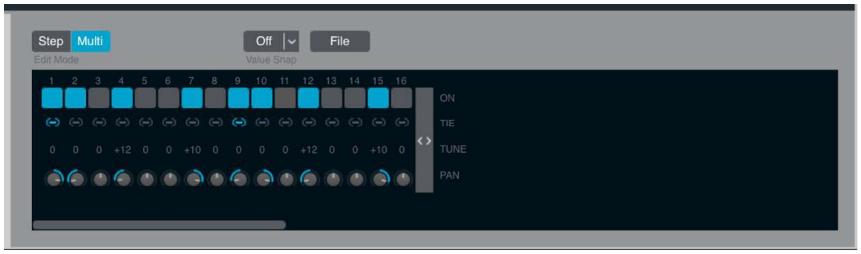
You can reduce the number of steps by dragging the gray end marker to the left, or you can increase the number of steps by dragging the marker to the right, up to a maximum of 128 steps. If the sequence is too long to fit in the display, you can scroll through all steps by dragging the gray scroll bar below.

Steps can be tied by clicking the link symbol below the step, which joins that step note to the next to make one longer note.

There are two different ways to view and edit sequencer data: Step and Multi edit mode. Click either button to choose the mode.



Step edit mode: In Step edit mode, you can view and edit the sequence one parameter at a time, with step values displayed as vertical bars. Drag vertically to edit step values. Option-click to reset a step to zero.



In Multi edit mode, you can edit common step parameters for multiple steps.

Alchemy Arpeggiator Step edit mode parameters

- *Edit Mode buttons:* Click the Step or Multi button to choose the sequencer view and edit mode.
- *Edit area:* Displays and allows you to edit arpeggiator steps and step parameters in Step or Multi edit mode.
- *Target pop-up menu:* Choose the parameter to view and edit.
  - *Velocity:* These values set the velocities of notes generated by the arpeggiator, assuming the Note Vel knob is not set to 100%. Set a step value to zero to create a rest. No note is played on that step.
  - *Tune:* These values determine tuning offsets for each step in semitones.
  - *Pan:* These values set the pan position of individual notes: drag up to pan the note left, or down to pan right.
  - *Swing:* These values determine timing offsets that make individual steps play early or late. The Swing sequence and the Swing knob can be used at the same time, allowing you to add a basic shuffle feel with the Swing knob, then fine-tune individual notes with the Swing sequence.
  - *Length:* These values set the note length for each step. Steps with zero length do not play. The Length knob globally scales the durations of all steps, but you can refine these lengths with the Length sequence. Try setting even-numbered steps shorter than odd-numbered steps, for a funky, groovy feel.
  - *Arp Mod 1-2:* Use these general purpose sequences to modulate any of the main synthesizer parameters: ArpMod 1-2 is available as a modulation source. See Note property modulators.
- *Value Snap pop-up menu:* Set a value that automatically snaps edits you make to sequence steps. For example, set to 1/3 to limit edits to 100%, 66%, 33%, or zero values. Off turns quantization off, which lets you set step values freely.

  Note: Value Snap doesn't move existing step values into alignment with quantized positions; it only affects the response of step values when you move or create them in the step editor.
- *Scroll bar:* Scroll to view steps that are not visible in the display area.

Alchemy Arpeggiator Multi edit mode parameters

- *Step On/Off buttons:* Turn individual steps on or off. These buttons actually display the Velocity sequence values, with any step value greater than zero displayed as on. Switching to Multi edit mode from Step edit mode does not change existing Velocity sequence values. However, if you turn off a step in Multi edit mode, then turn it back on, that step is set to 100% rather than its previous value.
- *Tie buttons:* Perform the same function as the tie buttons under each step in step mode: the step is tied to the next to make one longer note.

  Note: Any Velocity sequence value greater than zero is ignored for the subsequent tied step, as is the Pan sequence value. Pitch offsets work for the tied steps, however, so you can create pitch glides over tied steps by adjusting the Glide value in the master voice section, setting the Num(ber) of voices to 1, and setting the voice mode to Legato. See Alchemy master voice section.

- *Tune fields:* Drag vertically to set tuning offsets for each step in semitones. Double-click to reset the offset to zero.
- *Pan knobs:* Set panning for each step. Double-click to reset the offset to zero.
- *Scroll bar:* Scroll to view steps that are not visible in the display area.

Alchemy arpeggiator menu commands in Logic Pro for Mac

The arpeggiator is shown only in browse or advanced view. Click the Browse or Advanced button to switch to either view, then click the Arp button in the Perform/Arp/Effects section.

File button: Opens a pop-up menu with a number of useful arpeggiator pattern management commands.

Alchemy Arpeggiator file menu items

- *Preset:* Choose from a submenu of preset sequencer patterns.
- *Save:* Save the current sequencer pattern as a new preset.
- *Copy:* Copy the current sequencer pattern to the Clipboard.
- *Paste:* Paste a pattern from the Clipboard to the selected sequencer pattern.
- *Clear:* Reset the selected sequencer pattern to default values.
- *Randomize:* Fill the currently visible pattern with random values.
- *Import MIDI:* Import note, velocity, and timing information from a monophonic pattern, and set the Pitch, Velocity, and Swing sequence values accordingly.

Use the Alchemy arpeggiator in Logic Pro for Mac

The arpeggiator is shown only in browse or advanced view. Click the Browse or Advanced button to switch to either view, then click the Arp button in the Perform/Arp/Effects section.

Follow the steps outlined in the tasks to use the Alchemy arpeggiator.

Turn the selected arpeggiator on or off in Alchemy

- In Alchemy in Logic Pro, click the All/A/B/C/D button to select an arpeggiator in the Arpeggiator section, then do one of the following:
  - Set the Mode knob to any value other than Off to turn on the arpeggiator. Set the Mode knob to Off to stop the arpeggiator.
  - Set the Latch knob to any value other than Off to turn on the arpeggiator. Set the Latch knob to Off to stop the arpeggiator.
  - Stop the Logic Pro transport.

Create classic arpeggiator patterns in Alchemy

1. In Alchemy in Logic Pro, click the File button in the Name bar and choose Initialize Preset from the pop-up menu to reset all Alchemy parameters to default settings.

   Important: Use of Initialize Preset automatically turns on wide unison mode. See Alchemy wide unison mode.

2. Click the Arp button to show the arpeggiator section. All is selected by default.
3. Turn the Mode knob to select Up, Down, Up/Down, or Down/Up mode.
4. Play a chord and hold it: the individual notes of the chord are arpeggiated with regular 8th notes.
5. Try one or more of the following:
   - Set the Rate knob to 1/16 to create 16th note arpeggios instead of 8th notes.
   - Set the Octave knob to 2 or 3, so the pattern is transposed up by octaves before repeating at the original pitch.
   - Adjust the Swing knob to add a swing feel. Note that this knob is capable of extreme settings, so use lower values for a normal swing feel. Values around 25% result in a strong swing feel. Values of 10% or lower result in a subtle, relaxed feel to the groove.
   - Set the Latch knob to Hold to make the arpeggio continue cycling after you release the keys.
   - Experiment with Trigger mode. With Trigger turned on, the arpeggiator feels more responsive and allows you to deliberately restart the pattern with your playing. It also allows you to play out of time with the rest of the project.

Create an octave bouncing bassline in Alchemy

1. In Alchemy in Logic Pro, click the File button in the Name bar and choose Initialize Preset from the pop-up menu to reset all Alchemy parameters to default settings.

   Important: Use of Initialize Preset automatically turns on wide unison mode. See Alchemy wide unison mode.

2. Click the Arp button to show the arpeggiator section. All is selected by default.
3. Rotate the Mode knob to select Up, Down, Up/Down, or Down/Up mode.
4. Set the Octave knob to 2.
5. Play single notes in the lower registers: each note bounces up and down one octave, with an 8th note timing.
6. Rotate the Length knob to change note lengths. Higher settings result in a more legato feel, while lower settings create shorter staccato notes.

Arpeggiate multiple sources with different settings for each in Alchemy

1. In Alchemy in Logic Pro, click the File button in the Name bar and choose Initialize Preset from the pop-up menu to reset all Alchemy parameters to default settings.

   Important: Use of Initialize Preset automatically turns on wide unison mode. See Alchemy wide unison mode.

2. Click the Arp button to show the arpeggiator section. All is selected by default.

3. Click the A button at the top left of the arpeggiator.

   This switches the display to show settings for the source A arpeggiator. You can have a separate arpeggiator for each source instead of the single global arpeggiator available in All mode.

4. Rotate the Mode knob to turn on the source A arpeggiator, then play a chord, and adjust the Rate and Octave knobs to suit.

5. Turn on source B. Switch to advanced view if required.

6. Click the B button at the top left of the arpeggiator.

   This switches the display to show settings for the source B arpeggiator. The settings you made for the source A arpeggiator are not changed.

7. Rotate the Mode knob to turn on the source B arpeggiator, then play a chord, and adjust the Rate and Octave knobs to suit.

   Choosing a different mode results in source A and B alternately playing in unison or playing intervals from within the chord you play. Try setting one of the arpeggiators to Random mode to add a level of unpredictability. Choose different Rate settings to change the playback timing of sources. Try combining a straight timing for one source with a triplet or dotted timing for another to create complex polyrhythms.

8. Repeat steps 5-7 for sources C and D if you require additional arpeggiated sources.

   Note: You need at least four note polyphony to hear all four arpeggiated sources at once. See Alchemy master voice section.

Create a more complex arpeggiator pattern using the sequencer section in Alchemy

1. In Alchemy in Logic Pro, click the File button in the Name bar and choose Initialize Preset from the pop-up menu to reset all Alchemy parameters to default settings.

   Important: Use of Initialize Preset automatically turns on wide unison mode. See Alchemy wide unison mode.

2. Click the Arp button to show the arpeggiator section. All is selected by default.

3. Rotate the Mode knob to select Up, Down, Up/Down, or Down/Up mode.

4. Click the Step button to switch to Step edit mode, then choose Velocity from the pop-up menu.

5. Adjust step levels for the Velocity sequence to introduce volume changes. Try turning down all even numbered steps for a groovier feel, or Option-click specific steps to turn them off, thus creating a rest for that step.

6. Choose Length from the pop-up menu to display the length sequence. Experiment with shorter or longer notes for different steps: try lengthening odd numbered steps and shortening even numbered steps to enhance the groove.

7. Try tying a step to the next step by clicking the link symbol at the bottom. Experiment with tying different steps to generate different rhythms and with tying multiple subsequent steps to create a long note that spans several steps.

Create a step sequencer style pattern with pitch glides in Alchemy

1. In Alchemy in Logic Pro, click the File button in the Name bar and choose Initialize Preset from the pop-up menu to reset all Alchemy parameters to default settings.

   Important: Use of Initialize Preset automatically turns on wide unison mode. See Alchemy wide unison mode.

2. Click the Arp button to show the arpeggiator section. All is selected by default.
3. Turn the Mode knob to select Up, Down, Up/Down, or Down/Up mode.
4. Set the Rate knob to 1/4. You can also choose to use a small Swing knob value to create a looser feel.
5. Click the Multi button to switch to Multi edit mode.
6. Turn off several steps in the top Gate row to create a different rhythmic pattern. As an option, you can also adjust the Pan knobs to position steps left or right in the stereo field.
7. Click buttons in the Tie row to tie two or more steps.
8. Drag vertically in a Tune field for one of your tied steps. This results in a pitch change partway through the note. Values of +/- 12 semitones can work well to produce octave shifts up or down, but other values can also produce useful results: try +7 semitones for example, or -2.
9. Switch to advanced view if required, then click the Global button, and set the Num(ber) of voices to 1 to create a monophonic preset. See Alchemy master voice section.
10. Set the voice mode to Legato.
11. Click the Time button to set the Glide mode.
12. Set a small Glide knob value, such as 50ms or lower, to introduce pitch glides for tied and detuned steps. This results in smooth, rather than jarring, pitch transitions.

Create synchronized rhythmic pulsing chord patterns in Alchemy

1. In Alchemy in Logic Pro, click the File button in the Name bar and choose Initialize Preset from the pop-up menu to reset all Alchemy parameters to default settings.

   Important: Use of Initialize Preset automatically turns on wide unison mode. See Alchemy wide unison mode.

2. Click the Arp button to show the arpeggiator section. All is selected by default.
3. Turn the Mode knob to Chord.
4. Set the Rate knob to 1/4. You can also choose to use a small Swing knob value to create a looser feel.
5. Click the Multi button to switch to Multi edit mode.
6. Turn off a significant number of steps to make the pattern simpler. For example, turn on only steps 1, 4, 7, 10, 13, and 15 to create a triplet polyrhythm.
7. Choose Off in the Trigger pop-up menu to prevent the arpeggiator from retriggering when you play a chord.
8. You can choose to perform similar operations on further patterns, chosen with the Pattern knob. Use different values for each pattern to create complex polyrhythms.
9. As a further option, Control-click the Pattern knob, then choose Add Modulation from the shortcut menu, and link it to one of the performance controls. This lets you use the assigned performance control to switch between patterns while playing.

Alchemy effects section

Alchemy effects overview in Logic Pro for Mac

Effect modules are shown only in browse or advanced view. Click the Browse or Advanced button to switch to either view, then click the Effect button in the Perform/Arp/Effects section.

The Alchemy Effects section houses a powerful multi-effects processor that provides five independent effects racks: Main, A, B, C, and D. Each rack allows you to insert chains of effects. The signal flows through inserted effects from the top to the bottom of each rack. You can achieve different sonic outcomes by changing the order of effects in each rack.

- Individual sources (post source filters, if used) can be routed to the A/B/C/D racks. See Source master controls and Source filter controls.
- The main filters can be independently routed to the Main/A/B/C/D racks. See Main filter controls.

Output from the Effects section is mixed with the dry output signal from the main or source filter modules before being sent to the Alchemy main outputs.



For each effect loaded in each effects rack, there is a corresponding control panel shown in the area to the right. The top-to-bottom order of the selected rack is shown from left-to-right in the control panel area. Drag the gray scroll bar to view effects units or parameters that are not visible in the control panel area.

Effect control panels display a variable combination of buttons, knobs, displays, menus, and fields that are specific to the effect. Many effect control panels also include a File button that opens a pop-up menu with Load, Save, Copy, Paste, and Clear commands. These menu commands apply to the settings of the individual effect.

Note: The Clear command available in plug-in control panel menus restores default settings for the effect. It does not bypass the effect or remove it from the effects rack.

See the linked sections for information on individual effect parameters.

- Dynamic and utility effects
- Filter effects
- Modulation effects
- Reverb effects

Alchemy Effects rack parameters

- *Main/A/B/C/D rack buttons:* Use Main to view and set global effects settings for the overall instrument sound. Use A/B/C/D to set independent effect chains routed from each source or the main filters. It is possible to route both the main filter outputs and individual source outputs into the same effects rack.
- *Effect On/Off buttons:* Turn each effect slot in the selected effects rack on or off.
- *Effect slots:* Load an effect type from the pop-up menu. None removes an effect. Drag an effect name to move it to another slot.
  - A further effects slot is automatically created when an effect is inserted.
  - When you insert an effect in a slot that already contains an effect, the existing effect is replaced.
  - Choose None from the pop-up menu to remove an effect. The slot is not removed, it just remains unassigned.
  - When you drag the name of an inserted effect to a different slot to move it; preceding or following effects are moved up or down the list surrounding the target slot to preserve the order of the effects chain.
- *File pop-up menu:* Open a pop-up menu with several commands you can use for the selected effects rack (Main, A, B, C, or D). Commands do not apply to all racks simultaneously.
  - The Load, Save, Copy, and Paste commands apply to the entire contents of the selected effects rack and to the settings of each loaded effect.
  - The Clear command removes all loaded effects in the selected effects rack.
  - The Randomize command randomizes the settings of all currently loaded effects in the selected effects rack.
- *Scroll bar:* Drag vertically to view effects rack slots that are not currently visible. The scroll bar is shown only when the number of assigned slots exceeds the effects rack display area.

Alchemy dynamic and utility effects in Logic Pro for Mac

Effect modules are shown only in browse or advanced view. Click the Browse or Advanced button to switch to either view, then click the Effect button in the Perform/Arp/Effects section.

Effects in this group are used to control the level or stereo position of the signal.

Alchemy Amp effect parameters

Allows for a variety of auto-gate effects when modulated by an LFO or sequencer.

- *Gain knob:* Use a setting of 100% to preserve the input signal level. Lower values reduce this level, all the way to silence at 0%.

Alchemy Panner effect parameters

Allows for a variety of auto-pan effects when modulated by an LFO or sequencer.

- *Pan:* Low values boost the left and cut the right channel; high values boost the right and cut the left. At 50% the balance between left and right input channels is preserved.

Alchemy Phat Compressor effect parameters

Dynamics processor based on a soft-limiting compression algorithm.

- *Amount knob:* Set the amount of compression. Make-up gain is built in, so more compression results in greater apparent loudness.
- *Release knob:* Determine how quickly the compression effect subsides once the signal falls below the threshold.
- *Phat button:* Add a saturation effect, for a different compression color. Turn off for more neutral/transparent compression.

Alchemy Vintage Compressor effect parameters

Dynamics processor based on analog compression hardware.

- *File button:* Open a pop-up menu with several useful commands and to access presets.
  - *Presets:* Choose a preset from the submenu.
  - *Save:* Open a dialog where you can name and save the current settings to a new preset. The saved preset name is shown at the bottom of the Presets submenu list.
  - *Copy/Paste:* Copy the current settings to the Clipboard and paste the Clipboard contents.
  - *Clear:* Reset all parameter settings to default values.
- *Attack knob:* Set the time it takes for the compressor to react when the signal exceeds the threshold level.
- *Release knob:* Determine how quickly the compression effect subsides once the input signal falls below the threshold value.
- *Threshold knob:* Set the threshold level. Signals above this threshold value are reduced in level.
- *Input Gain knob:* Set the level of gain applied to the signal at the compressor input.
- *Ratio knob:* Set the ratio of compression when the signal exceeds the threshold value.
- *Makeup Gain knob:* Set the amount of gain applied to the compression signal at the compressor output.

Alchemy filter and distortion effects in Logic Pro for Mac

Effect modules are shown only in browse or advanced view. Click the Browse or Advanced button to switch to either view, then click the Effect button in the Perform/Arp/Effects section.

Effects in this group change the frequency content of the signal.

Alchemy Bandpass Filter effect parameters

An adjustable width bandpass filter. The rejected portion of the signal can be recovered at a later point in the signal path by using the Band Reject effect.

- *Low Cut/High Cut knobs:* Set the lower and upper boundaries of the passband.
- *Low Res/High Res knobs:* Add resonant emphasis to the passband boundaries.

Alchemy Band Reject effect parameters

Available only when the Bandpass Filter1/2 is inserted earlier in the signal path. Lets you mix in the portion of the signal that was rejected by the bandpass filter. A classic application of this feature is for multi-band distortion.

- *Mix knob:* Set the amount of band-rejected signal mixed back into the signal path, bypassing effects between the BP Filter and the Band Reject mix point.

Alchemy Multimode Filter effect parameters

Provides multiple filter types that process all voices routed through the Effects section.

- *Type pop-up menu:* Choose from dozens of different filter types. Each has a unique characteristic. See Filter types.

Note: The knobs described are common to most filter types but not all.

- *Cutoff knob:* Set the filter cutoff frequency.
- *Resonance knob:* Set the filter resonance or emphasis. Higher settings boost frequencies in the immediate vicinity of the cutoff frequency.
- *Drive knob:* Overdrive the filter. The precise effect varies with each filter design.
- *Mix knob:* Determine the wet/dry balance (0% = dry only; 50% = equal mix; 100% = wet only). Typically this is left at 100%.

Alchemy Three Band EQ effect parameters

Offers three bands of parametric EQ. Each band, shown as F1/F2/F3 (where F denotes frequency), has identical controls.

- *Gain knobs:* Set the amount of boost or cut to the three frequency bands.
- *Frequency knobs:* Set the center frequency for the three frequency bands.

  Note: The three frequency bands can each be set to values from 16 Hz through to 20 kHz, so it isn't necessary (although it may be useful) to tune F2 above F1, or F3 above F2.
- *Q knobs:* Set the bandwidth of the three frequency bands.

Alchemy Bass Enhancer effect parameters

Bass-boosting EQ, recommended for use with bass and low frequency percussion sounds.

- *Amount knob:* Set the intensity of the bass enhancement.
- *Tune knob:* Determine the frequency band to which the boost is applied.
- *Phat button:* Add a saturation effect, for a different color to the bass enhancement. For more neutral/transparent results, leave phat mode off.

Alchemy Distortion effect parameters

Distortion effect with multiple algorithms that can be used simultaneously.

- *Bitcrusher knob:* Use for a lo-fi digital distortion.
- *Mech knob:* Use to produce a nastier, more intense distortion.
- *Tube knob:* Use to simulate the warm distortion effect of an overdriven tube amp.
- *XCite knob:* Use to create distortions with an enhanced high-frequency range.
- *Pre Gain knob:* Set the amount of input gain applied before the distortion effect. This allows you to produce heavily saturated distortions with some distortion types, especially Tube.
- *Post Gain knob:* Set the output level to compensate for significant signal boosts caused by distortions such as Tube and Mech.

Alchemy Waveshaper effect parameters

An EQ unit that applies a predefined filter shape. Two sets of identical controls can be used to modify the pre-shaped and post-shaped signal.

- *Mix knob:* Set the balance between the dry input signal and the wet effect signal.
- *Gain knobs:* Set the amount of boost or cut to the pre- or post-filtered signal.
- *Shape knobs:* Set the pre- or post-filter shape toward a lowpass or highpass type. The center position has a balance of low and high filtering.
- *Frequency knobs:* Set the center frequency for the pre- or post-filtered bands.
- *Display:* View and edit the predefined filter curve. Drag the white handles to set the shape start, mid, and end points. Drag the line between points to create convex or concave curves. Option-click the line to reset it to a linear shape.
- *Bipolar button:* Turn on to use a bipolar filter curve. Turn off to use a unipolar filter curve.
- *File button:* Open a pop-up menu with several useful commands.
  - *Presets:* Choose a preset from the submenu.
  - *Save:* Open a dialog where you can name and save the current settings to a new preset. The saved preset name is shown at the bottom of the Presets submenu list.
  - *Copy/Paste:* Copy the current settings to the Clipboard and paste the Clipboard contents.
  - *Clear:* Reset all parameter settings to default values.

Alchemy delay and modulation effects in Logic Pro for Mac

Effect modules are shown only in browse or advanced view. Click the Browse or Advanced button to switch to either view, then click the Effect button in the Perform/Arp/Effects section.

This group includes delay, chorus, flanger, and phaser effects.

Alchemy Delay effect parameters

Stereo delay with dual-filtered feedback.

- *L/R Rate knobs:* Set the delay time in msec when the Sync button is off, or in beats when the Sync button is on. See Sync button parameter below.
- *L/R Offset knobs:* Add a small additional amount of delay, allowing you to adjust the feel of tempo-synced delays.
- *L/R Feedback knobs:* Determine the amount of delayed signal fed back to the input of the delay.
- *Filter A/B buttons:* Turn on to filter the delayed signal without affecting the dry signal. You can use one or both filters. Select a type for each filter via the selection fields. Most filters have Cutoff and Resonance controls or other filter type-specific controls.
- *Filter type pop-up menu:* Choose a type for each filter. See Filter types.
- *Cutoff knobs:* Set the filter cutoff frequency for filter A or B.
- *Resonance knobs:* Set the filter resonance or emphasis for filter A or B. Higher settings boost frequencies in the immediate vicinity of the cutoff frequency.
- *Sync button:* Turn on to synchronize delay rates with the project tempo. See L/R Rate.
- *Mono button:* Turn on to mix the left and right input channels down to mono. The result is fed to both the left and the right channels of the delay. The dry portion of the signal remains in stereo.
- *Crossover knob:* Determine stereo placement of the feedback signal. At 0%, left feeds left and right feeds right; at 50%, each channel is fed to both inputs; at 100%, left feeds right and right feeds left.
- *Initial Pan knob:* Determine stereo placement of the initial delayed signal (prior to Feedback). Typically, you would set Initial Pan to 0% or 100% when Crossover = 100%, and leave Initial Pan centered at 50% in most other cases.
- *Mix knob:* Determine the Wet/Dry balance (0% = dry only; 50% = equal mix; 100% = wet only).
- *File button:* Open a pop-up menu with several useful commands.
  - *Presets:* Choose a preset from the submenu.
  - *Save:* Open a dialog where you can name and save the current settings to a new preset. The saved preset name is shown at the bottom of the Presets submenu list.
  - *Copy/Paste:* Copy the current settings to the Clipboard and paste the Clipboard contents.
  - *Clear:* Reset all parameter settings to default values.

Alchemy Modulation FX parameters

A short delay with built-in LFO modulation, useful for chorus, flanging, and related effects.

- *File button:* Open a pop-up menu with several useful commands and to access presets.
  - *Presets:* Choose a preset from the submenu.
  - *Save:* Open a dialog where you can name and save the current settings to a new preset. The saved preset name is shown at the bottom of the Presets submenu list.
  - *Copy/Paste:* Copy the current settings to the Clipboard and paste the Clipboard contents.
  - *Clear:* Reset all parameter settings to default values.
- *Delay knob:* Set the base delay time. Short values are useful for flanging, values in the range of 10 to 40 msec are useful for chorus, and longer delay times can produce a variety of metallic and buzzing effects.
- *Rate knob:* Set the modulation speed. Faster rates and smaller depths are characteristic of chorus effects, while slower rates and greater depths are typical for flanging.
- *Mix knob:* Determine the wet/dry balance (0% = dry only; 50% = equal mix; 100% = wet only).
- *Depth knob:* Set the modulation intensity.
- *Feedback knob:* Set the level of the delayed signal that is fed back into the input. Medium to medium-high settings are common for flanging effects, while chorus tends to use little or no feedback.
- *Stereo knob:* Spread the delayed signal across the stereo field.

Alchemy Phaser effect parameters

A simple phasing effect with built-in LFO modulation.

- *File button:* Open a pop-up menu with several useful commands and to access presets.
  - *Presets:* Choose a preset from the submenu.
  - *Save:* Open a dialog where you can name and save the current settings to a new preset. The saved preset name is shown at the bottom of the Presets submenu list.
  - *Copy/Paste:* Copy the current settings to the Clipboard and paste the Clipboard contents.
  - *Clear:* Reset all parameter settings to default values.
- *Rate knob:* Set the modulation speed. Can be synchronized with the project tempo when Sync is turned on. See Sync button.
- *Depth knob:* Set the modulation intensity.
- *Spread knob:* Spread the delayed signal across the stereo field.
- *Frequency knob:* Set the filter cutoff frequency.
- *Poles knob:* Set the intensity of the phasing effect. Higher values result in a more cutting and harsh phasing sound.

  Tip: Where possible, try to use the minimum number of poles required because this helps to reduce CPU load.

- *Feedback knob:* Set the level of the delayed signal that is fed back into the input.
- *Mix knob:* Determine the wet/dry balance (0% = dry only; 50% = equal mix; 100% = wet only).
- *Sync button:* Turn on to synchronize delay rates with the project tempo. See Rate knob.

Alchemy reverb effects in Logic Pro for Mac

Effect modules are shown only in browse or advanced view. Click the Browse or Advanced button to switch to either view, then click the Effect button in the Perform/Arp/Effects section.

Effects in this group are used to add ambience to the signal.

Alchemy Acoustic Reverb effect parameters

High quality, full-featured reverb effect.

- *Time knob:* Set the length of the reverb tail. Values up to 20 sec are possible.
- *Predelay knob:* Set the amount of initial delay before diffuse reflections begin.
- *Size knob:* Determine the dimensions of the simulated space.
- *Width knob:* Determine how much reflections are spread across the stereo field.
- *Diffusion knob:* Determine the density of the early reflections.
- *Gate knob:* Set a threshold below which the reverb tail is gated. Attack shapes the beginning of the gated tail, while Decay shapes its end.
- *Attack knob:* Attack shapes the beginning of the gated reverb tail.
- *Decay knob:* Decay shapes the end of the gated reverb tail.
- *EQ knobs:* Modify the frequency content of the wet signal. Lo Freq sets the frequency of a low shelf that you can increase or decrease with Lo Gain. Hi Freq sets the frequency of a high shelf that you can increase or decrease with Hi Gain.
- *Damping knobs:* Determine the amount of frequency-specific losses (damping). Lo Freq sets the frequency of a low shelf that you can damp by increasing Lo Damp. Hi Freq sets the frequency of a high shelf that you can damp by increasing Hi Damp.
- *Variation knob:* Determine the reverb color.
- *Quality knob:* Determine the processing precision. Use low values to conserve CPU resources.
- *Mix knob:* Set the wet/dry balance (0% = dry only; 50% = equal mix; 100% = wet only).
- *File button:* Open a pop-up menu with several useful commands and to access presets.
  - *Presets:* Choose a preset from the submenu.
  - *Save:* Open a dialog where you can name and save the current settings to a new preset. The saved preset name is shown at the bottom of the Presets submenu list.
  - *Copy/Paste:* Copy the current settings to the Clipboard and paste the Clipboard contents.
  - *Clear:* Reset all parameter settings to default values.

Alchemy Classic Reverb effect parameters

CPU-efficient reverb.

- *File button:* Open a pop-up menu with several useful commands and to access presets.
  - *Presets:* Choose a preset from the submenu.
  - *Save:* Open a dialog where you can name and save the current settings to a new preset. The saved preset name is shown at the bottom of the Presets submenu list.
  - *Copy/Paste:* Copy the current settings to the Clipboard and paste the Clipboard contents.
  - *Clear:* Reset all parameter settings to default values.
- *Predelay knob:* Set the amount of initial delay before diffuse reflections begin.
- *Damping knob:* Determine the amount of high frequency loss in the reflections.
- *Hi/Lo Cut knobs:* Set high and low frequencies. The wet signal above and below is cut.
- *Size knob:* Determine the dimensions of the simulated space. Larger values equal longer reverb times.
- *Mix knob:* Determine the wet/dry balance (0% = dry only; 50% = equal mix; 100% = wet only).

Alchemy Convolution Reverb effect parameters

A high-quality reverb effect that uses recordings of actual acoustic spaces, such as caves, tunnels, halls, and so on. These recordings are known as impulse responses. You can apply an impulse response to your Alchemy preset which makes it sound like it is being played in the chosen acoustic space.

- *Dry knob:* Set the level of the dry signal sent to the main outputs.
- *Wet knob:* Set the level of the effect signal sent to the main outputs.
- *Pre-Delay knob:* Set the amount of initial delay before diffuse reflections begin.
- *Size knob:* Determine the dimensions of the simulated space. Larger values equal longer reverb times.
- *Stereo knob:* Set the output width of the convolution. When set to 100%, the output width is dictated by the impulse response sample. When set to 0%, it is equivalent to convolving with a mono version of the stereo impulse.
- *Start knob:* Determine the start point of the loaded impulse response.
- *End knob:* Determine the end point of the loaded impulse response.

  Tip: Where possible, try to use the shortest possible impulse response length required because this helps to reduce CPU load. If you change the IR length, you will need to adjust the envelope shown in the Impulse response display.
- *Impulse response display:* View the loaded impulse response waveform. An editable level envelope is superimposed over the waveform.
  - Drag the envelope points to set levels over the duration of the impulse response.
  - Click on or near the envelope line to create new points.
  - Double-click a point to delete it.

- *IR pop-up menu:* Load an impulse response (IR). Use the arrows to step forward and backward through impulse responses.
- *File button:* Open a pop-up menu with several useful commands and to access presets.
  - *Presets:* Choose a preset from the submenu.
  - *Save:* Open a dialog where you can name and save the current settings to a new preset. The saved preset name is shown at the bottom of the Presets submenu list.
  - *Copy/Paste:* Copy the current settings to the Clipboard and paste the Clipboard contents.
  - *Clear:* Reset all parameter settings to default values.

    Note: The impulse response file is not loaded, saved, or copied and pasted. File menu commands affect only the parameter settings.
- *Reverse button:* Reverse playback of the loaded impulse response.
- *On/Off buttons:* Turn integrated filter module on of off.
- *Filter type pop-up menus:* Choose the filter type. See Filter types.
- *Cutoff knobs:* t the cutoff frequency for each filter.
- *Resonance knobs:* Set the resonance or emphasis for each filter. Higher settings boost frequencies in the immediate vicinity of the cutoff frequency.

Alchemy extended parameters in Logic Pro for Mac

Alchemy provides additional parameters that are accessed by clicking the disclosure arrow 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. Per voice channels support pitchbend, aftertouch, modwheel, and Performance control assignment messages. See Performance controls overview. Controllers and MIDI messages sent on the base channel affect all voices.
- *Mono Mode Bend Range slider:* Set a value from 0 to 96.

  The chosen pitch bend range affects individual note pitchbend messages received on all but the common base channel. The default is 48 semitones, which is compatible with the GarageBand for iOS 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.
- *Ctrl A pop-up menu:* Choose a MIDI continuous controller as a source for use with the MIDI options in the modulation rack. CC#2 (Breath) is assigned by default.
- *Ctrl B pop-up menu:* Choose a MIDI continuous controller as a source for use with the MIDI options in the modulation rack. CC#4 (Footpedal) is assigned by default.
- *Ctrl C pop-up menu:* Choose a MIDI continuous controller as a source for use with the MIDI options in the modulation rack. CC#74 (Brightness) is assigned by default.
- *Ctrl D pop-up menu:* Choose a MIDI continuous controller as a source for use with the MIDI options in the modulation rack. CC#75 (General purpose) is assigned by default.

Alchemy tutorials

Alchemy tutorial introduction for Logic Pro for Mac

The following sections contain information to assist you as you start to explore sound creation in Alchemy.

Given the flexibility and variety of Alchemy synthesis engines, you can take a number of different approaches to sound design.

- If you prefer to tweak existing settings, it may be more suitable to use features that affect the entire instrument. At the simplest level this is achieved with performance controls such as the Transform pad. See Basic synthesis tweaks and Performance controls overview.
- If you prefer to build sounds from scratch—parameter by parameter—you ca and Create sounds from scratch.
- Alchemy also features advanced morphing capabilities that facilitate cross-synthesis to create new sounds. See Elemental morphs overview.

Whatever approach you favor, you can achieve new and interesting results. Experiment and familiarize yourself with each approach. Each has its strengths and weaknesses, and often a combination of methods may strike the best balance for your needs.

When programming a sound from scratch in Alchemy, the best approach is to work on each component of the sound in isolation.

To start, you need a plain vanilla setting. You can revert to this setting by choosing File > Initialize Preset in the Name bar. The default setting is sonically uninteresting, but provides a useful starting point for several examples.

Important: Use of Initialize Preset automatically turns on wide unison mode. See Alchemy wide unison mode.



Alchemy basic synthesis tweaks in Logic Pro for Mac

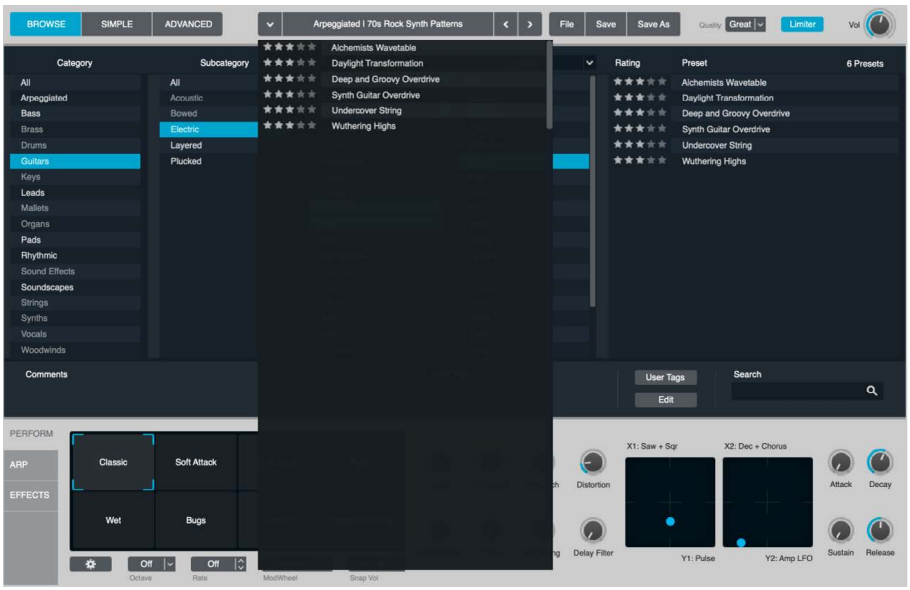
Alchemy provides thousands of ready-made presets for you to explore. These are not only a great resource for musical inspiration but can also provide numerous insights into how sound designers have created them. Even if you're not inclined to program your own sounds from scratch, some small tweaks to existing presets can provide that "perfect" sound for your projects or performances.

Note: Images shown in tutorials are not specific to presets used in tasks. They are included as a guide to help you find areas and parameters in the Alchemy interface.

Use performance and effect controls to tweak a sound in Alchemy

This task introduces you to basic operations in the performance and effect controls. This method of interaction works in the same way across presets. These controls can have a subtle or dramatic impact and can be used to very quickly create patch variations. If you happen across something that sounds great while making your tweaks, don't forget to click the Save As button at the top.

1. In Alchemy in Logic Pro, click the Preset name field in the Name bar and choose the *Analog Bass Sequence* preset. Make sure the Advanced view button is active.



2. Click the Perform button to view the performance controls, then drag the framing bracket in the Transform pad while playing your keyboard.

   The sound morphs between different sound variations. If you happen across something you like, you can use the Save As button in the Name bar to name and save a new preset.



3. Drag the framing bracket to the top left (“Snappy”), then set the Decay knob to zero and the Sustain knob to around 75%. Drag the Arp Mode knob to the leftmost Off position.

   When you hold a note on the keyboard you will hear a sustained sound with no sequenced pattern. Adjust the Cutoff, Resonance and Tube knobs to hear how they affect the sound.

4. Click the Effects button to view the Effects section, then click the Blue On buttons of all four effects to bypass them.

   Play a few notes to hear the impact this has, then click the gray On button to turn the Multimode Filter plug-in back on.



5. Choose the Formant option in the Multimode Filter Type menu.
6. Click the On button for the Delay plug-in and play a few notes to hear the effect.
7. Choose any preset from the File menu in the Delay effect interface. Drag the gray scroll bar at the bottom of the Effects section if the Delay effect isn't visible.
8. Return to the performance controls, and adjust the Resonance and Cutoff knobs while playing both staccato and held notes.
9. Drag the blue control points in both XY Pads while striking a note to hear their impact.
10. Double-click both the Resonance and Cutoff knobs to set them to the zero position.

Swap the source to tweak a sound in Alchemy

1. In Alchemy in Logic Pro, click the Global button toward the upper left to view sources A to D.

   You can see that the *Analog Bass Sequence* preset uses four sources: A, B, C and D.

2. Click the source select field for both source A and source B, then choose Load VA > Simple > Triangle in each.

   Play your keyboard to hear a more mellow synth type sound. Feel free to repeat this step with sources B and C, using different waveforms, or perhaps import a sample.



Mute a source to use a partial preset in Alchemy

1. In Alchemy in Logic Pro, click the File button in the Name bar and choose Initialize Preset, then choose the *Analog Bass Sequence* preset from the Preset name field. Make sure the Global button is active in advanced view.

   Important: Use of Initialize Preset automatically turns on wide unison mode. See Alchemy wide unison mode.

2. Click both the source A and source B buttons to mute these sources. A blue A, B, C, or D button indicates the source is active.

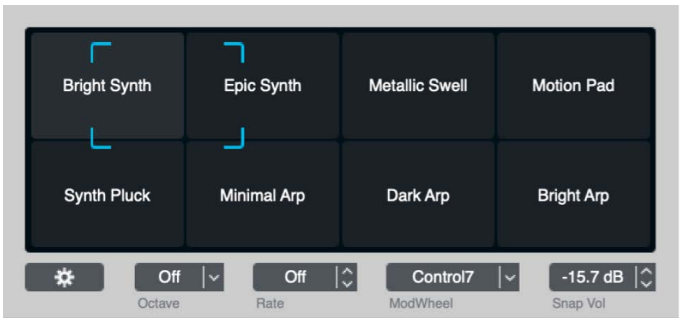
   Play the keyboard to hear a very different sequenced sound which emphasizes the interaction between Additive source C and the Saw used for source D. You can click the C and D source buttons, in turn, to hear them in isolation. It may surprise you to hear what they sound like individually.

   Tip: Muting sources in this way is also a good working practice when editing or adjusting source settings because it allows you to focus on the sonic characteristics you are trying to achieve for the source. You can unmute the source and adjust the Volume settings of each to set the balance between them once you've made your changes.



3. Click the Perform button to view the performance controls, then drag the framing bracket in the Transform pad while playing your keyboard.

   You will hear the sound morphing between different sound variations. If you happen across something you like, you can use the Save As button in the Name bar to name and save a new preset. Return the framing bracket to the “Snappy” position when done.



Adjust the tuning and envelope of a source in Alchemy

1. In Alchemy in Logic Pro, click the File button in the Name bar and choose Initialize Preset, then choose the *Analog Bass Sequence* preset from the Preset name field. Make sure the Global button is active in advanced view.

   Important: Use of Initialize Preset automatically turns on wide unison mode. See Alchemy wide unison mode.

2. Drag the source B Tune knob to -24 semis.
3. Click the source B Vol knob.

   The AHDSR envelope updates in the central modulation section.



4. Double-click the source B Pan knob to center it.

5. In the central modulation section, make sure the Attack, Hold, and Decay knobs are set to zero, set Sustain to around 80%, and Release to around 0.032. You may also want to experiment with adjustments of the Decay and other knobs in the performance controls section to hear their impact.



Adjust filters to tweak a sound in Alchemy

1. In Alchemy in Logic Pro, change the second filter Cutoff and Resonance settings in the main filter section to the right of the sources while playing.



2. Click the Filter type menu to experiment with different filters.
3. Turn on the first filter and try different settings.
4. Adjust the Par/Ser knob to hear how this affects the sound when the filters are used in series or in parallel.

   Drag the framing bracket in the performance controls to hear variations of your tweaked sound.

Recycle sources and sound settings from other presets in Alchemy

This is less of a tweak and more of a shortcut for sound creation. Alchemy provides thousands of presets that can be partially used for new sounds, sequences, modulation settings, arpeggios, effects settings, and more.

1. In any loaded Alchemy preset, click the source name field and choose Save Source from the pop-up menu.
2. Give the source a name in the file dialog, and click Save.

   This source is automatically saved in the Alchemy > Source subfolder.
3. In a new instance of Alchemy, click the Source select field and choose Load Source from the pop-up menu, then choose the saved source.

   The source (or sources) from up to four presets can be recycled in this way.
4. Steps 1-3 can be repeated for any Alchemy element that provides a File pop-up menu, such as modulation settings, arpeggiator settings, effects settings, and so on.

   Make use of existing presets as sources of raw material for your own sounds.

Create Alchemy sounds from scratch in Logic Pro for Mac

Alchemy provides a number of different synthesizer engines that can be used to resynthesize, or reconstruct, imported sampled sounds. It does this by analyzing sonic characteristics of the source, such as level, frequency, phase, and other components. This analysis is then mapped to values that are reconstructed by one or more of the synthesis engines. Details on importing and analysis of source material are covered in Import browser and in individual source sections. See Source elements overview.

Alchemy also allows you to use several of the built-in synthesizer engines to create sounds without sample importing and analysis. This section explores these pure synthesis options and other approaches to creating the basic building blocks of your sounds.

Once you have added the raw material to your sources, Alchemy's filters, modulation, and Effects sections are available to further refine your sound. See the information and tasks outlined in these sections: Main filter controls, Alchemy modulation overview, and Alchemy effects overview. Also look at the Performance controls overview and Alchemy arpeggiator overview sections.

Note: Images shown in tutorials are not specific to presets used in tasks. They are included as a guide to help you find areas and parameters in the Alchemy interface.

Explore pure additive synthesis in Alchemy

Pure additive synthesis allows you to create sounds by combining sine waves (referred to as harmonics, overtones, or partials) of different levels, phase, pitch, and pan position. If you're unfamiliar with this method of synthesis, see Additive edit window overview and Additive synthesis.

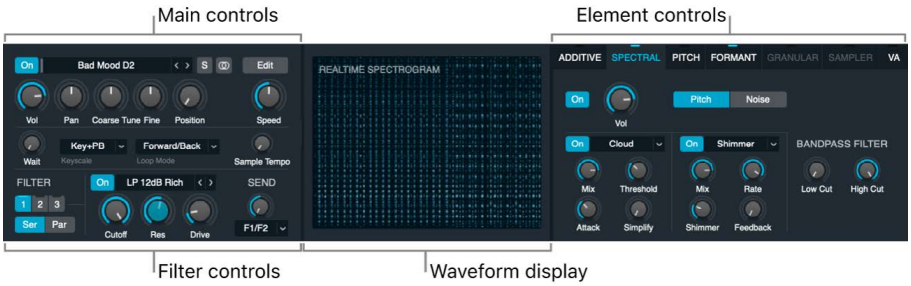
1. In Alchemy in Logic Pro, click the File button in the Name bar and choose Initialize Preset from the pop-up menu to reset all Alchemy parameters to default settings.

   Important: Use of Initialize Preset automatically turns on wide unison mode. See Alchemy wide unison mode.

2. Click the A button (below Global) to view source A parameters.

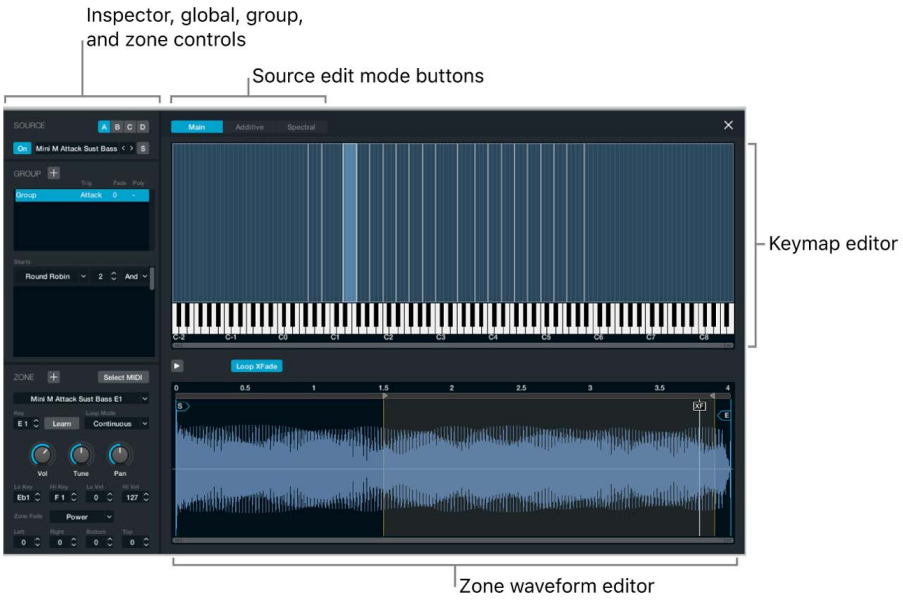
   This defaults to the VA (Virtual Analog) synthesis engine. Play a few notes to hear the sound.

3. Click the Additive button, just below the File button in the Name bar, then click the Additive On button to turn on the additive engine.



4. Click the Edit button to the left of the realtime spectrogram (waveform) display.

   The main edit window opens.



5. Click the Additive button to view the additive edit window, where the Vol button should be active and a single column is shown at the left of the Partial bar display.

   This is the first (and only) harmonic of your sound.

6. Drag vertically and horizontally across the Partial bar display to create a series of new harmonics of different levels. Create a pseudo waveform shape.

   Play your keyboard to hear how this sounds. Depending on what you've drawn, you should hear a harsh and edgy digital sound.



7. Click the Mode pop-up menu, and choose Fifth (harmonics).
8. Drag the first harmonic vertically to see how the fifth harmonics only are changed, noting that the change is relative to the first harmonic.

   This is how you adjust grouped harmonics. Adjusting groups of, rather than individual, harmonics is the best approach to take when creating additive sounds. Creating and adjusting individual harmonics is possible (using One in the Mode pop-up menu), but it is tedious and not an efficient approach. In general terms, even harmonics sound more musical and odd harmonics sound noisy or dissonant.
9. Click the Tune button, choose One (single harmonics) from the Mode pop-up menu, then draw horizontally and vertically (creating another pseudo waveform shape) in the Partial bar display.

   Play the keyboard to hear a deranged metallic sort of sound.
10. Choose Fifth from the Mode pop-up menu, then drag some points vertically and horizontally in the Partial envelope display.

    Play the keyboard to hear the fifth harmonics bend in pitch, controlled by the envelope associated with this group of harmonics.

    Harmonic envelopes control the level, pitch, pan, and phase of harmonics over time. This level of control over each harmonic (if required) lets you recreate almost any type of sound, but as mentioned, it is far more efficient to work with groups of harmonics. Individual editing of harmonics or their associated envelopes is only recommended for fixes to unwanted partials or for more impactful edits to the lower harmonics, such as the fundamental, second, fourth, and so on. Edits of upper harmonics are unlikely to be audible in many cases.

11. Close the edit window, and experiment with the Fundamental, Octaves, Odd/Even, and Fifths knobs in the additive effects section. Also adjust the PVar and Sym knobs to hear their impact.



12. Choose Stretch from the second additive effects pop-up menu, and experiment with the Stretch and String knobs.

13. As further steps, you may also want to Control-click different Additive controls to assign modulation routings. Try assigning a modulation sequencer to the String parameter in the additive effects section to create a stepped sample-and-hold type of sound. You can use a File menu preset for the sequencer or create your own sequence.

    If you want to take the modulation sequence concept further, you could create a sequence of individual harmonic pitches, resulting in a melody. Similarly, you could sweep a comb filter across the harmonic series, again resulting in a melody.

Explore VA synthesis in Alchemy

1. In Alchemy in Logic Pro, click the File button in the Name bar and choose Initialize Preset from the pop-up menu to reset all Alchemy parameters to default settings.

   Important: Use of Initialize Preset automatically turns on wide unison mode. See Alchemy wide unison mode.

2. Click the A button (below Global) to view source A parameters.

   This defaults to the VA (Virtual Analog) synthesis engine. Play a few notes to hear the sound.



3. Choose several waveforms from each category in the source select field, and play the keyboard each time to hear the sound.

   Alchemy has dozens of waveforms to choose from, providing a wealth of raw source material. Once you've explored the options, choose Complex > Soft Edge. This resembles a cheap electronic keyboard piano sound.

4. Turn on the first filter in the source parameters on the left, then double-click the Cutoff knob.

   Play to hear a much mellower sound evoking an electric piano. Feel free to try different filters, but LP 12 dB Smooth is a good option.

5. Click the Global button at the top left, then click the Master Vol knob.

   Adjust the Release knob to a value around 1.00 in the central AHDSR envelope section.



6. Click the Effects button at the lower left to view the Effects section.



7. Choose Modulation > Mod FX in the top effects slot, then click the Mod FX File button and choose Preset > Big Chorus.

   You may want to adjust the Rate to around 2.8 Hz.

8. Choose Modulation > Phaser in the second effects slot, then click the Phaser File button and choose Preset > Subtle Vibrato.

   Turn on the Sync button, then play to hear a simple, effected electric piano sound that is created from a single source. It goes without saying that this sound could be improved in many ways, but this primer should give you a feel for how quickly a basic preset can be created in Alchemy's VA engine. Keep the sound you've created for the next task.

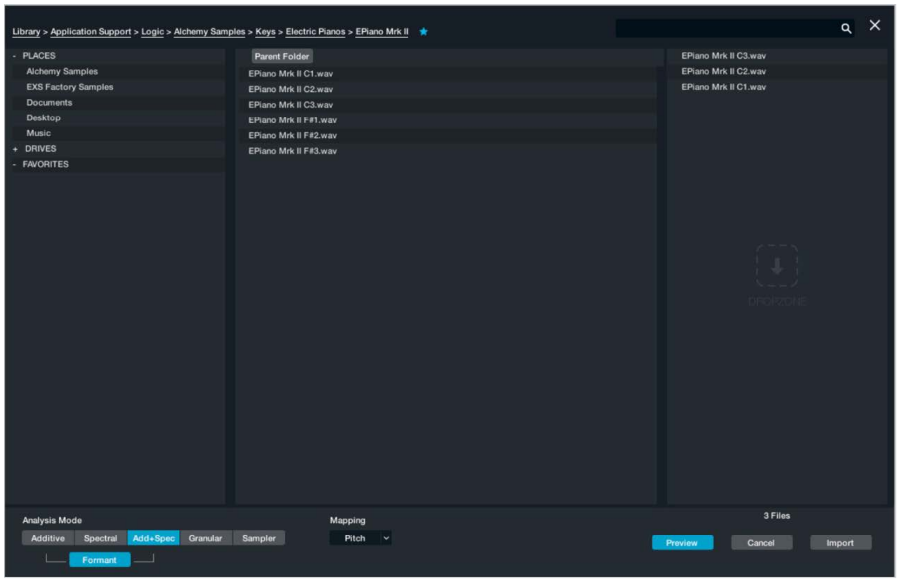
Combine synthesis methods to create sounds in Alchemy

Alchemy provides up to four sources in a preset. Each source has a full set of synthesis parameters associated with the chosen synthesis engine. This allows you to combine multiple synthesis types, such as virtual analog, additive, resynthesis, granular, or straight sample playback in a single preset. Because each synthesis method has different strengths and weaknesses, it is both easier and will deliver better results if you choose the most appropriate engine for the task.

Though there are areas of crossover between synthesis engines, the VA engine is strong for synthesizer type sounds, the additive engine is strong for digital and bell-like sounds, and the sampler/granular engines are best suited for sample manipulation, with the latter being particularly useful for loop mangling.

Alchemy also offers resynthesis which uses the additive or spectral synthesis engines, or a combination of the two, along with optional formant control. Resynthesis, put simply, analyzes a source audio file and reconstructs a facsimile of this sound using sine waves and noise signals. You can then manipulate these reconstructed elements to create new presets. The additive engine is well suited for single notes with a strong harmonic character. The spectral engine is suitable for chordal material and other complex signals. In many cases, you will achieve the best results by combining the two synthesis engines. This lets you manipulate the resynthesized source more precisely, playing to the strengths of each synthesis method.

1. In Alchemy in Logic Pro, choose advanced view and click the source B select field, then choose Import Audio from the pop-up menu to open the Import browser window.



2. Click the Preview button to turn automatic preview of selected files on or off.
3. Click the Sampler button to choose the import analysis mode.

   If the audio data is imported to the additive, spectral, add+spectral, or granular elements, analysis may take a moment or two. No analysis occurs when the Sampler analysis mode button is active.
4. Choose Alchemy Samples > Strings > Single Samples > Strings Mid, then click the Import button.

   The Import browser closes when the import is complete, and the previous window is displayed.
5. Click the Global button (if not active), then adjust the Vol knobs of source A and B to hear an electric piano/string layer.
6. Repeat for sources C and D, using different analysis modes and the same or different samples.

   An import progress dialog is shown when import methods other than Sample are used. The Import browser automatically closes when the import is complete.

7. Click the source select buttons at the left for sources B, C, and D, then click the Additive, Spectral, Pitch, Formant, Granular, and Sampler buttons at the upper right to view the parameters of each synthesis engine.



8. Listen to each source in isolation, and experiment with these synthesizer engine parameters to get a feel for what each engine provides for imported material.

   The use of different synthesis engines allows different kinds of manipulation options for each source, providing different sonic characteristics. Beyond the comparatively simple combining of sources and synthesis engines in this way, you can perform a sophisticated form of cross-synthesis with Alchemy morphing features. See Elemental morphs overview.

Alchemy elemental morphs overview in Logic Pro for Mac

Most people are familiar with the concept of morphing images and 3D graphics. We've all seen the videos of a human face morphing from one face to another, or transforming from a young to an older version of the same face. The idea of morphing between sounds commonly involves transitioning between the filter values, envelope shape, or amount of reverb for each sound. The Performance controls in Alchemy, particularly the Transform pad, allow this type of parameter morphing, often moving between multiple synth and effect settings simultaneously to produce dramatic changes to the sound.

Alchemy can also perform a deeper and more powerful form of morphing referred to as *elemental morphing*. Elemental morphing is an advanced form of morphing that operates at the sound generation level, making it possible to combine core characteristics of different sounds, technically referred to as cross-synthesis. It allows new sounds to be created by mixing and matching elements from multiple source sounds. Elemental morphing expands the possibilities for creative sound design well beyond what is possible with the parameter morphing described in the paragraph above.

The Alchemy additive, spectral, granular, and formant synthesis engines are key facilitators for this type of morphing. Each synthesis method has inherent strengths and weaknesses, making them more suitable for certain sound types than other synthesis engines. Having all of them available in Alchemy makes it a unique and powerful morphing tool. Another critical component for elemental morphing is Alchemy's ability to reconstruct audio using these synthesis engines; a process called resynthesis. Resynthesis makes it possible to perform elemental morphing on any audio because it creates a synthesized facsimile of the original sound that affords more manipulation options than a static sample. Without resynthesis, elemental morphing would be limited to sounds built from scratch within Alchemy.

Successful morphing depends on developing an understanding of which engine is best suited for the type of audio material you are working with. Your choice also affects the type and number of sound shaping options available. Morphing is a complex process with many variables that can impact on the results. As with any form of synthesis, the more you work with it and the more familiar you become with its strengths and weaknesses, the easier and faster it is to create sounds you have in mind.

See Morph use and tips, Morph drum beats, and Morph melodic sounds.

Note: Images shown in tutorials are not specific to presets used in tasks. They are included as a guide to help you find areas and parameters in the Alchemy interface.

Alchemy morph use and tips for Logic Pro for Mac

Source components are shown only in advanced view. Click the Advanced button to switch to advanced view, then click the Morph button to view and use the morph controls. See Morph controls.

It is important to choose the same analysis mode for all morphed sources. See Import browser.

Though you can morph with any synthesis type, the best morphing results are achieved with the spectral or additive engines. Spectral is the most suitable import method to use for the majority of audio material. Use Additive or Add + Spec import mode if dealing with pitched, monophonic sounds. Make sure the Formant button is turned on when importing sounds because this generally results in a higher quality morph.

If you are not creating a pitched instrument that requires the root note to remain at the default pitch, irrespective of the morph position, turn *off* the Fixed Pitch button, because this will result in a higher quality morph.

The time alignment of sounds is crucial to achieving a good morph. Selecting sounds with similar timing and loop characteristics tends to deliver the best results. Aim to have a similar loop duration and loop location within the selected samples. If the sounds you are morphing are comprised of more than single note samples or the timing of your sources doesn't align as you would like, adjust the warp marker positions. See the Manually time-aligning morphed sounds task in this section.

Morphs that move from 100% of all elements in a sound to 100% of another sound can often be suitable for sound effects but may be more difficult to shape into new playable instruments. You may find it better to mix the levels of individual elements for this type of use.

Note: Images shown in tutorials are not specific to presets used in tasks. They are included as a guide to help you find areas and parameters in the Alchemy interface.

Manually time-aligning morphed sounds in Alchemy

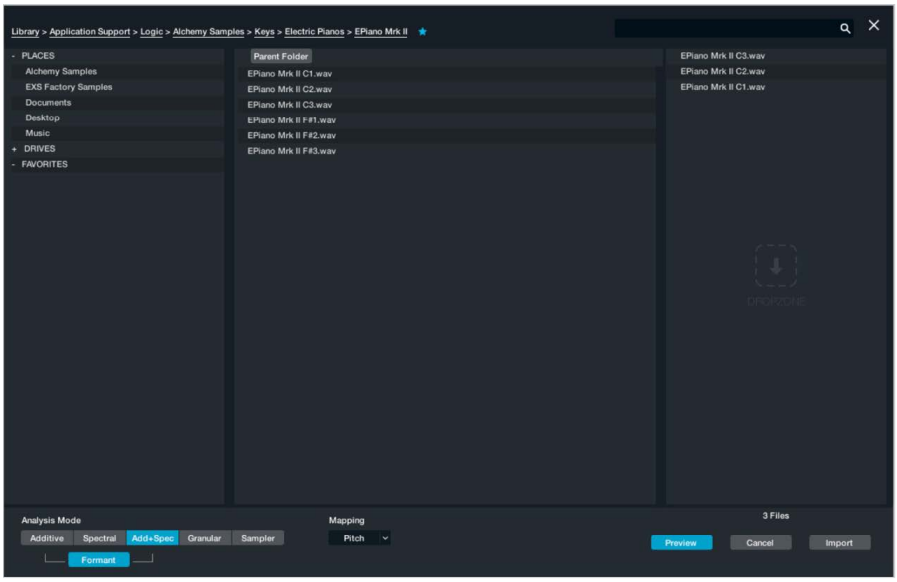
This task covers manual alignment of sources. Alchemy features an auto align function (see Morph controls) that can make this an automatic process, but in some cases you may need to manually intervene to achieve the best results.

When morphing is turned on for a source, warp markers are displayed in the zone waveform editor. See Zone waveform editor.

When you morph from a sound with a fast attack to one with a slow attack, Alchemy smoothly adjusts the attack time according to the morph all position, or the envelope position if in elements mode. Warp markers define the attack phase boundaries of each sound. More generally, warp markers define the boundaries of a series of time-aligned segments when two or more sounds are morphed together.

1. In Alchemy in Logic Pro, click the source select field for the active source, then choose Import Audio from the pop-up menu.

2. In the Import browser, click the Additive, Spectral, or Granular import mode button.



3. Select the files you want to use, then click the Import button.

   When you import additive, spectral, or granular data, Alchemy automatically sets warp markers at the following positions:

   - The absolute start point of the sound.
   - The end of the attack portion.
   - The loop start point. If a sample does not have loop points, loop points are automatically chosen.
   - The loop end point. If a sample does not have loop points, loop points are automatically chosen.
   - The absolute end point of the sound.

4. If you need to edit warp markers, shown as light gray lines, click the source Edit button to open the main edit window. Depending on the source material and active morph mode, light blue lines may appear on the background between the start and end markers. These can help with manual alignment of warp markers.

   Do any of the following in the zone waveform editor:

   - To move a warp marker, including the loop start and end, drag the marker handle left or right.

     Note: The loop start and loop end markers also double as warp markers. You cannot position the loop start later than the loop end. However, it is possible to drag both the loop start and loop end markers to the same position, creating a sustain point rather than an extended loop region.

   - To insert a new warp marker, double-click a position in the waveform editor.

     The number associated with each existing warp marker, inclusive of the loop markers, is updated accordingly.

- To delete an existing warp marker, double-click its handle.



If your sounds are single notes, the placement of the five default warp markers usually delivers good results when morphing. You may want to change the position of marker 2, placing it at the point where it sounds like the end of the attack for each sound. This can produce a more convincing morph between sounds with contrasting attack qualities.

If your sounds consist of multiple events such as musical phrases, drum loops, or a spoken sentence, you may find it useful to create additional warp markers. If you are morphing between two voices speaking the same sentence, placing a warp marker at the start of each word helps to preserve the integrity of each word during the morph. You must make sure the same numbered warp marker is at the start of the same word in each sentence.

If you are morphing between musical phrases or drum loops with different grooves or timing nuances, placing a warp marker on every beat, half beat, or quarter beat helps you to achieve a smoother morph.

Explore source morphing options and controls in Alchemy

This tutorial makes use of the Morph Example preset, which you should load from the Name bar. The preset contains acoustic guitar, kalimba, a sustained solo vocal, and synth brass samples. Additive and Formant was used as the import method because the content contains more pitched info than noise, and this also allows access to the additive harmonic controls. This basic material provides enough scope to go in a number of directions.

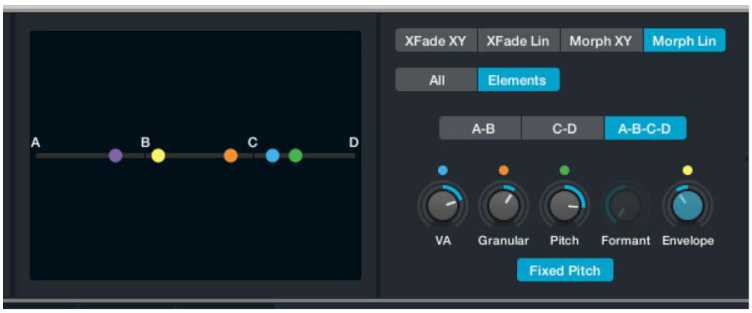
Play the keyboard while changing parameters to get a feel for what each does.

1. In Alchemy in Logic Pro, click the Advanced button to view all parameters. Click the A/B/C/D buttons, in turn, to look at parameter settings for all four sources.

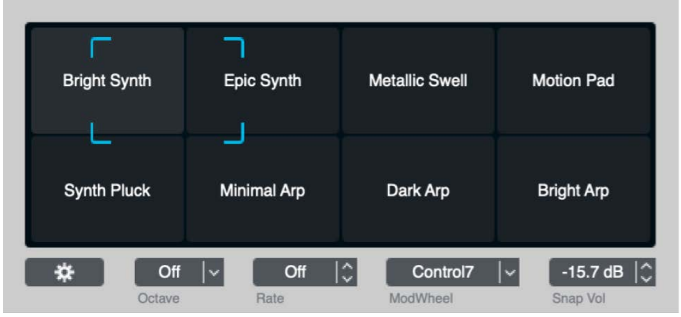
   Note that PVar is set to zero for all sources. This isn't always useful, but for this particular preset, no pitch variations are required.

2. Solo each source, in turn, and play a few notes, then unsolo before moving to the next source. This will give you a better understanding of the way the sound is constructed.

3. Click the Morph button at the left, and note the active Morph XY and Elements buttons. Experiment with the X and Y parameters, using either the points in the display or the X and Y knobs for the Additive, Pitch, Formant, and Envelope parameters.



4. Click the XFade XY, XFade Lin, and Morph Lin buttons to hear the impact each type of crossfade or morph has on the sound. Experiment with the All, Elements, X and Y, A-B, C-D, and A-B-C-D parameters available in each morph or crossfade mode.
5. In the Perform section, drag the Transform pad framing box to the “Bold Pluck,” “Glassy,” and other snapshots, repeating step 3 and playing as you audition each variation. You’ll see movements in the Transform pad update the other Perform section controls.

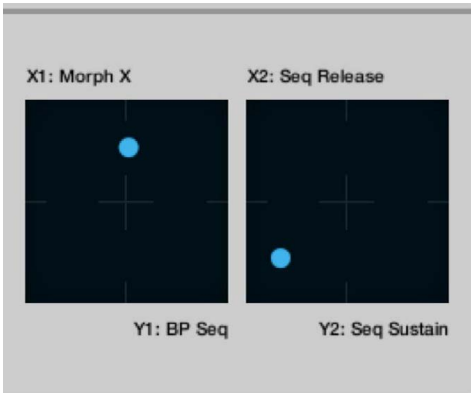


6. Click the Mod Targets button at the right of the modulation section and hover the pointer over the Perform menu item. Take a look at the number of, and the choice of, parameters assigned to each performance control.

   As you can see, the preset is far more complex than it appears on the surface. The morphing controls are not simply used to crossfade between sources, but between modulators, filter and effect parameters, mix settings, harmonic levels, and much more. Morphing can be viewed and used to blend the sounds of each element as well as aspects of each synthesis type into a hybrid of synthesis methods.

Crossfade all four sources using an X/Y pad performance control in Alchemy

Follow these steps to use the left X/Y pad in the performance controls section instead of the X/Y pad in the morph section. This allows you to control crossfades between sources in simple view and also allows you to include these modulations in Transform pad snapshots.



1. In Alchemy in Logic Pro, click the File button in the Name bar and choose Initialize Preset from the pop-up menu to reset all Alchemy parameters to default settings.

   Important: Use of Initialize Preset automatically turns on wide unison mode. See Alchemy wide unison mode.
2. Switch to advanced view, and set up each of the four sources with a different, but complementary sound.

   Feel free to use any of the synthesis engines for each source or to blend multiple engines within one source.

3. Click the Morph button at the left to view morphing parameters, then click the XFade XY button.
4. Make sure the X knob is set to zero, then Control-click the X knob and choose Add Modulation > Perform > XYPad 1X from the shortcut menu.
5. Make sure the Y knob is set to zero, then Control-click the Y knob and choose Add Modulation > Perform > XYPad 1Y from the shortcut menu.

Morph four spectral sources using an X/Y pad performance control in Alchemy

Follow these steps to use the left X/Y pad in the performance controls section instead of the X/Y pad in the morph section. This allows you to control crossfades between sources in simple view and also allows you to include these modulations in Transform pad snapshots.

1. In Alchemy in Logic Pro, click the File button in the Name bar and choose Initialize Preset from the pop-up menu to reset all Alchemy parameters to default settings.

   Important: Use of Initialize Preset automatically turns on wide unison mode. See Alchemy wide unison mode.
2. Switch to advanced view, then click the Morph button at the left to show morphing parameters.
3. Click the source A select field, and choose Import Audio.
4. Select Spectral mode in the Import browser, choose a suitable sample, and import it. Turn on Formant mode before importing.
5. Repeat steps 3 and 4 for each of the other three sources. Make sure Spectral mode is chosen and the Formant button is on when importing to sources B, C, and D.
6. Click the Morph button at the left to view morphing parameters, then click the Morph XY button.

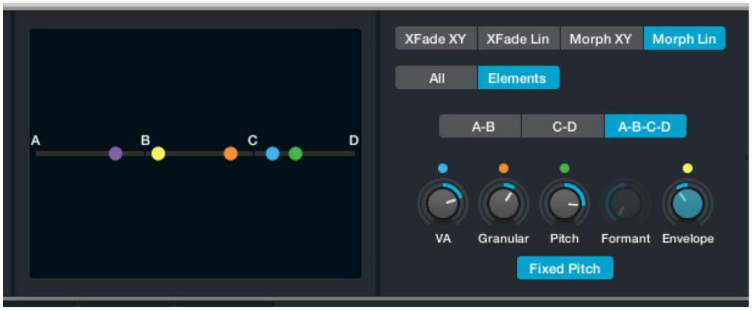
   You can now morph between the four sources using the X and Y knobs. Follow steps 7 and 8 to assign the X and Y knobs to the left X/Y pad in the Perform section.
7. Double-click the X knob to set it to zero, then Control-click the X knob and choose Add Modulation > Perform > XYPad 1X from the shortcut menu.
8. Double-click the Y knob to set it to zero, then Control-click the Y knob and choose Add Modulation > Perform > XYPad 1Y from the shortcut menu.

Create a linear morph between two speech samples in Alchemy

1. In Alchemy in Logic Pro, click the File button in the Name bar and choose Initialize Preset from the pop-up menu to reset all Alchemy parameters to default settings.

   Important: Use of Initialize Preset automatically turns on wide unison mode. See Alchemy wide unison mode.
2. Switch to advanced view, then click the Morph button to show morphing parameters.
3. Click the source A select field, and choose Import Audio.
4. Locate the first of your speech samples and import it, using Additive, Spectral, or Add+Spec mode. Turn on Formant mode before importing.
5. Repeat steps 3 and 4 for source B and your second speech sample. Use the same import mode as for the first sample, and make sure Formant mode is active.

6. Click the Morph button at the left to view morphing parameters, then click the Morph Lin button to view linear morphing parameters.



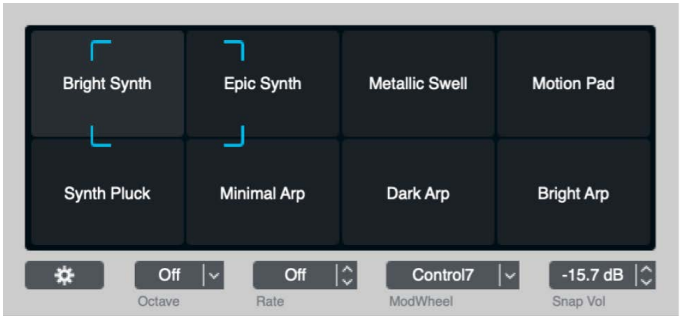
7. Click the A-B button to change the linear morphing mode.

   You can now use the X knob to morph between the two speech samples. Each resynthesis method creates different sounding results even before morphing, and the results halfway through the morph are dramatically different. Experiment with each sample import method to determine which delivers the best outcome.

8. Control-click the X knob, then choose Add Modulation > Perform > Control 1 from the shortcut menu.

9. Click the Perform button to view performance controls, then click the Wheel pop-up menu under the Transform pad, and choose Control 1.

   You can now use the mod wheel to move performance control 1.



10. As optional steps, click the source A button, then click the Edit button to view the main edit window for this source.

    - If necessary, drag the S and E handles (start and end point markers) in the waveform display to trim the file.
    - Double-click above the waveform display to place additional warp markers at important syllables in your speech samples.
    - Click the B button in the main edit window to view source B parameters.
    - Double-click to create additional warp markers. Existing warp markers already correspond with those created for source A. The syllable tagged with warp marker n in source B is time-aligned with the syllable tagged with warp marker n in source A at the halfway point of the morph.

Morph four additive sources using a pair of MSEG envelopes in Alchemy

1. In Alchemy in Logic Pro, click the File button in the Name bar and choose Initialize Preset from the pop-up menu to reset all Alchemy parameters to default settings.

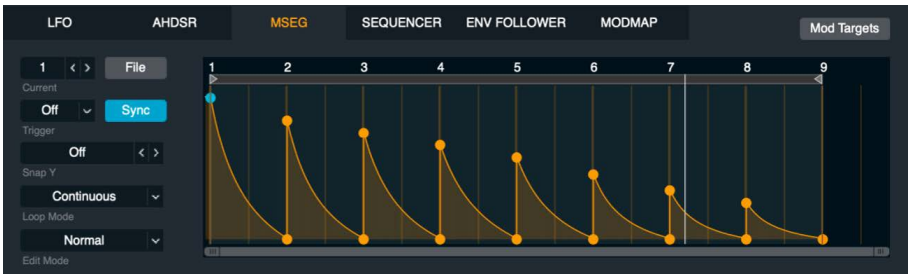
   Important: Use of Initialize Preset automatically turns on wide unison mode. See Alchemy wide unison mode.

2. Switch to advanced view, then click the Morph button to show morphing parameters.
3. Click the source A select field, and choose Import Audio.
4. Select Additive mode in the Import browser, choose a suitable sample, and import it. Turn on Formant mode.
5. Repeat steps 3 and 4 for each of the other three sources. Make sure Spectral mode is chosen and the Formant button is on when importing to sources B, C, and D.
6. Click the Morph button at the left to view morphing parameters, then click the Morph XY button.

   You can now morph between the four sources using the X and Y knobs. Follow the steps below to assign the X and Y knobs to MSEG 1 and MSEG 2, respectively.

7. Double-click the X knob to set it to zero, then Control-click the X knob and choose Add Modulation > MSEG > MSEG 1 from the shortcut menu.

   You can now define morphing on the X axis with the MSEG 1 envelope in the modulation section.



8. Double-click the Y knob to set it to zero, then Control-click the Y knob and choose Add Modulation > MSEG > MSEG 2 from the shortcut menu.

   You can now define morphing on the Y axis with the MSEG 2 envelope in the modulation section.

Morph drum beats in Alchemy in Logic Pro for Mac

This tutorial guides you through the process of importing two drum beats and then morphing between them. You will initially morph all elements collectively, then each individually.

Note: Images shown in tutorials are not specific to presets used in tasks. They are included as a guide to help you find areas and parameters in the Alchemy interface.

Import and morph multiple drum loops in Alchemy

The loops used for this example are from the Electro House and Tech House loop libraries. If you can't find the loops, you need to download these additional content packages. To do so, choose Logic Pro > Sound Library, then download the missing content in the Sound Library window.

1. In Alchemy in Logic Pro, click the File button in the Name bar and choose Initialize Preset from the pop-up menu to reset all Alchemy parameters to default settings.

   Important: Use of Initialize Preset automatically turns on wide unison mode. See Alchemy wide unison mode.

2. Click the source select field for source A and choose Import Audio.

   The Import window opens.

3. Open the Apple Loops browser, and type "Lockstep" in the search field. Drag the *Lockstep Beat* 02 loop to the Dropzone area of the Import window.

4. Click the Spectral and Formant analysis mode buttons at the lower left, then click the Import button.

   This determines the synthesis engine used for resynthesis of the audio sample. Spectral + Formant is a good general purpose option that works well for drum loops. It's also the default setting in a new instance of Alchemy. When analysis is complete, you can trigger the loop at its native pitch by playing C3 on your keyboard.

5. Drag the *Insider Groove Beat* loop to the controls area of source B and choose an import mode from the four options. Drop the loop to perform the import.

6. Play C3 to trigger both loops at once. The loops are 8 beats long and have a native tempo of 120, so playback is nicely synchronized. Although there are techniques for tempo matching imported loops, it is simpler to use matched samples and improves the potential for getting good results with minimal effort. If the loops have a very different groove, click the Edit button to open the main edit window, where you can do any of the following:

   - In the waveform display, create warp markers to divide the loop into equal length segments. The number of markers required depends on the length and complexity of the loop. Aim to have a segment for each significant beat in the loop.
   - Slide each warp marker forward or back as necessary to align it with the nearest drum hit transient. Hold a warp marker in the waveform display to zoom horizontally for more accurate adjustments.
   - Switch to the other drum loop source with the A/B/C/D buttons at the top left of the main edit window.
   - Slide each warp marker forward or back as before to align each with the nearest drum transient.

7. Click the A button (under Global), then choose Continuous from the source A Loop Mode pop-up menu. Repeat this step for the source B loop.

   This sets both beats to loop continuously while you hold down a note.

8. Turn off the Velocity On button in the modulation rack.

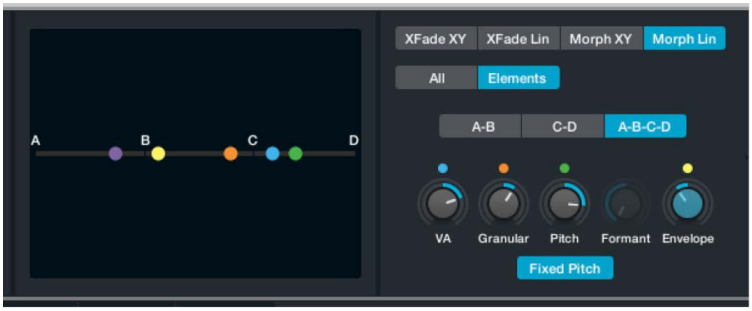
   When triggering loops, it can be helpful to turn off velocity control of the playback level so the loops trigger with the same loudness regardless of how hard you play. If the modulation area doesn't show this parameter, click the Global button, then click the center of the Master Volume knob shown at the upper right.



9. Click the Morph button at the left to show the morphing section at the upper right, then drag the handle in the middle of the morph control area from left to right to hear the level of each source crossfade.

   The default view shows XY set to crossfade, not morph, between the sources.

10. Select one of the two morph buttons in the morphing section to access the elemental morphing controls. Because there are only two sources, click the Morph Linear button, then click the A-B button to limit morphing to these two sources.



11. Drag the handle between the two sources while holding a note to create a fundamentally different effect than the crossfade between them.

    What you can hear is a true morphing between discrete components of each sound. The transition is not simply between the overall level of each sound, but between the levels of multiple narrow frequency bands and the formant characteristics of each sound.

    Drag the handle to the far left, then listen as you move slowly to the right to hear a phantom shaker-like rhythm that doesn't exist in either loop. Once you have finished exploring, move the handle back to the far left before looking at the next task.

Morph elements independently in Alchemy

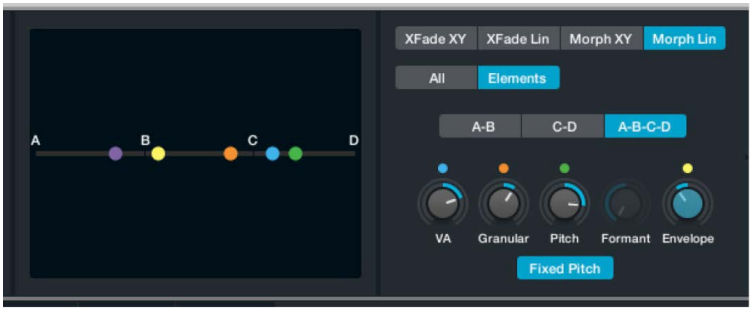
1. In Alchemy in Logic Pro, click the Morph XY button in the morph section, then click the Elements button to access controls that allow independent morphing between different sound elements, as opposed to morphing all sonic aspects simultaneously.
2. Play C3 on your keyboard, then rotate the Spectral/Granular knob to move the Spectral control handle (orange dot) from left to right. You can also drag the orange dot if not obscured by another dot.

   You can hear the spectral components of the first loop transition to those of the second loop. The change is similar to that heard in step 10 of the previous task, but the phantom shaker rhythm is clearer, and the kick sound from the Lockstep Beat remains intact longer. In fact, this combination causes the kick to change tone and become more pronounced at beat 1 of each bar. When the Spectral control is full right, the original kicks are replaced by a pronounced “thump” at the head of each bar.

Morph pitch and formants in Alchemy

1. In Alchemy in Logic Pro, move the Pitch handle (green dot) from left to right. You can also rotate the Pitch knob.

   Drum beats don’t generally contain much pitched information, so use of the Pitch control may have little impact when working with loops. In this case, however, setting Pitch to the far right position results in the prominent high tom of the Insider Groove Beat becoming louder near beat 2. Only the tail of the pitch pushes through however, creating an eighth note rhythm that starts near beat 3.

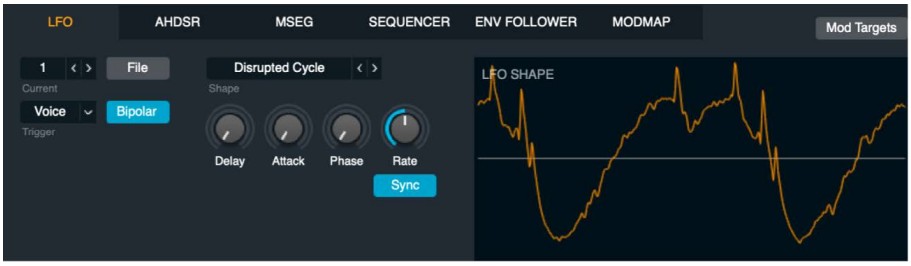


2. Set Formant (purple dot) to 75%-100% to make the onset of the tom pitch more clearly audible. You can also rotate the Formant knob to set this value.
3. Set the Spectral knob to zero.

   You can hear the arc of the tom hit, but without tonal characteristics because there’s little spectral material at this point in the Lockstep loop. This demonstrates the important role that resonances play in human perception of pitch information.

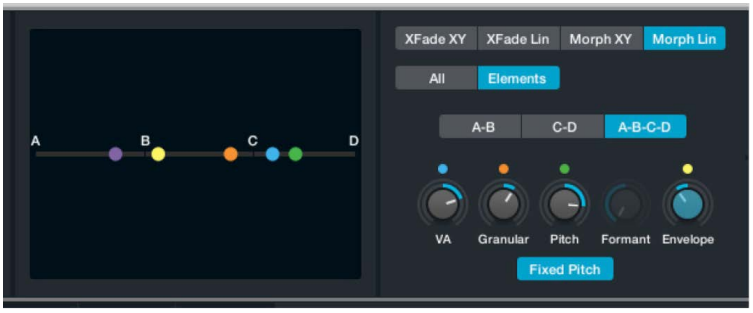
Assign performance controls to morphed drum loop elements in Alchemy

1. In Alchemy in Logic Pro, Control-click the Position knob for source A, then choose Add Modulation > LFO > LFO 1 from the shortcut menu.
2. In the modulation section, click the Shape pop-up menu for LFO 1 and choose Basic > RampUp.



3. Turn off Bipolar mode for the LFO, and set Rate to the same number of beats as the loop. For example, set the Rate knob to 4 beats for a one-bar loop, or to 8 beats for a two-bar loop, and so on.
4. Control-click the Position knob for source A, and choose Copy Modulation from the shortcut menu.
5. Click the source B button to view source B parameters.
6. Control-click the Position knob for source B, and choose Paste Modulation from the shortcut menu.

   The playback position of both loops is now controlled by LFO 1, and they play back in sync with the project tempo.
7. Click the Morph button to view morphing parameters, then click the Morph Lin button.
8. Click the A-B button to use the X knob to morph between the two time-aligned drum loops.
9. As an optional step, you can click the Elements button to use or assign element controls.



   - Control-click the Spect/Gran knob (orange dot), and choose Add Modulation > Perform > XYPad 1X from the shortcut menu.
   - Control-click the Pitch knob (green dot), and choose Add Modulation > Perform > XYPad 1Y from the shortcut menu.

     You can now use the left X/Y pad in the Perform section to morph the spectral data by moving the pointer horizontally. You can morph pitch data by moving the pointer vertically.

- Control-click the Formant knob (purple dot), and choose Add Modulation > Perform > XYPad 2X from the shortcut menu.
- Control-click the Envelope knob (yellow dot), and choose Add Modulation > Perform > XYPad 2Y from the shortcut menu.

  You can now use the right X/Y pad in the Perform section to morph the formant filters by moving the pointer horizontally. You can morph the envelope shape by moving the pointer vertically.

Morph melodic sounds in Logic Pro for Mac Alchemy

Sounds that contain melodic or harmonic material, such as those that are not purely drums or percussion, present more challenges when morphing. In these cases, it's especially important to understand some of the limitations and to know which synthesis options will deliver the best results. This group of tutorial tasks covers morphing between sounds that contain melodic material produced by monophonic instruments such as a solo voice, a synth lead, or an orchestral instrument like a flute or oboe. The example files have the same basic length, which makes results more predictable than if file lengths are quite different.

Note: Images shown in tutorials are not specific to presets used in tasks. They are included as a guide to help you find areas and parameters in the Alchemy interface.

Play a source melody with another source in Alchemy

1. In Alchemy in Logic Pro, click the File button in the Name bar and choose Initialize Preset from the pop-up menu to reset all Alchemy parameters to default settings.

   Important: Use of Initialize Preset automatically turns on wide unison mode. See Alchemy wide unison mode.

2. Click the source select field for source A, and choose Import Audio from the pop-up menu.

   The Import window opens.

3. Type "Amanda Aa Ee Ay Oh C3" in the search field, and drag the file to the Dropzone area of the Import window.

4. Click the Additive and Formant analysis mode buttons at the lower left, then click the Import button.

   This determines the synthesis engine used for resynthesis of the audio sample. This combination is generally best when you are working with melodic, monophonic audio files. When analysis is complete, you can trigger the audio at its native pitch by playing C3 on your keyboard. It's important to understand that you are hearing a resynthesized version of the original sound that is being generated by the additive synthesis engine and further shaped by the formant engine.

   Tip: It's best to select the same analysis mode for all audio you plan to morph between. If you don't do this, the primary synthesis element morphing control will be non functional when morphing between sources analyzed with different engines. For example, you can't perform additive morphing between a source generated by the additive engine and another source generated by the spectral engine.

5. Click the source B select field, and choose Import Audio from the pop-up menu.

6. Type “Velvet Rope Synth Lead” in the search field, and click to select the file or drag it to the Dropzone area of the Import window, then click the Import button.

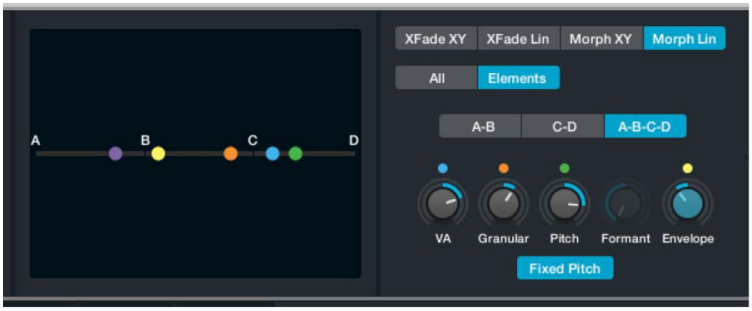
   Note: The previously selected import mode is retained, so there’s no need to change anything.

7. To set both files to loop continuously while you hold down the note, click the A button (under Global), then choose Continuous from the Loop Mode pop-up menu. Repeat this step for source B.

8. Click the Morph button at the left to show the morphing section at the upper right, then click the Morph Lin button and the A-B button. The All button should be active.

9. Drag the handle between the two sources while holding a note to hear the sound morph from the vocal (source A on the left) to the synth lead (source B on the right).

10. Click the Elements button to show the elemental morphing controls, then move the Pitch handle (green dot) to the far right while holding down C3 on your keyboard. You can also rotate the Pitch knob.



   When the note cycle restarts, you will hear the vocal sample “singing” the melody provided by the synth lead phrase in source B. The pitch morph is particularly effective because the source A material contains no chordal content and does not change pitch, and the Source B material has a very clear melodic phrase that the pitch analysis can easily track. In general, files that work well for Flex Pitch in Logic work well for pitch morphing applications in Alchemy. Keep this example loaded for the next task.

Explore additive morphs in Alchemy

Elemental morphing can sometimes lead to sudden and dramatic level changes. Before adjusting the Additive control, click the Global button, then double-click the Volume knobs for sources A and B. This sets them both to –18 dB.

1. In Alchemy in Logic Pro, click the Morph button at the left to show the morphing section at the upper right, then click the Morph Lin button, the A-B button, and the Elements button (if required).

2. Drag the Additive handle (blue dot) to the right while holding C3 on your keyboard. You can also rotate the Additive knob to set this value.

   As you move the control, the harmonic partials of the vocal sample transition to the partials of the synth lead. This doesn’t always produce pleasing results because partials in one sound may emphasize noise or artifacts in the other sound. The workflow is often one of sonic exploration in search of happy accidents rather than a predictable process. Play notes and chords above and below C3 as you experiment.

3. Set the Additive control to approximately 30%, Pitch to 100%, and Formant and Envelope to zero percent.

4. Click the A button at the left to view source A parameters, click the Additive button at the top right, then rotate the Fundamental knob toward the left.

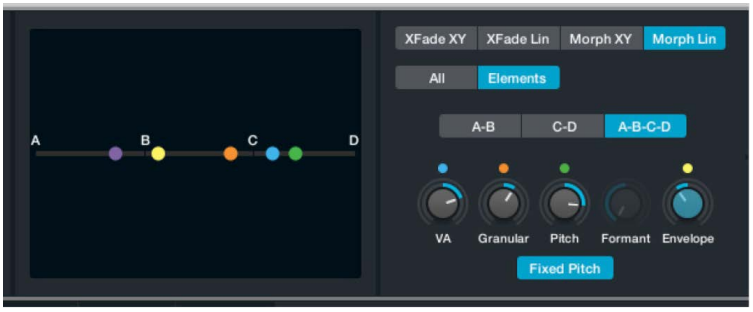
   This attenuates the fundamental partial which brightens the sound by making the higher frequency partials more pronounced.

   Turn down the Odd/Even knob to create a more hollowed out sound, emphasizing the partials of the synth lead. Experiment with other controls to get a feel for their effect on the sound. Play octaves and fifths as you experiment.

Create envelope morphs in Alchemy

The Envelope control in the morph section primarily morphs between the timing of sources. This has little effect when files are close in length, but this behavior changes when the playback length of files is altered.

1. In Alchemy in Logic Pro, click the Morph button at the left to show the morphing section at the upper right, then move the Envelope handle (yellow dot) to the far left. You can also use the Envelope knob to move the Envelope handle.

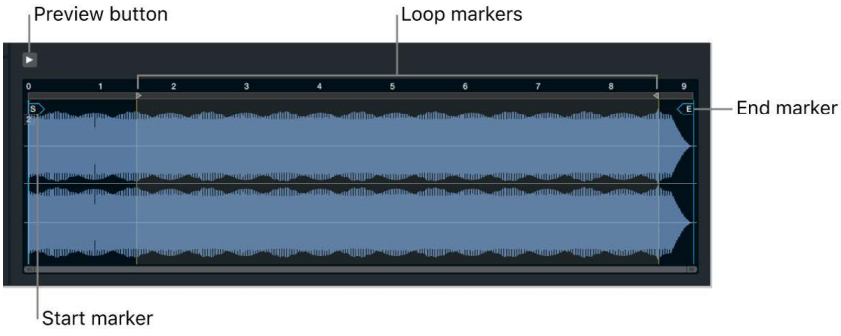


2. Click the A button at the left to view source A parameters, and click the Edit button at the right of the source filename.

   The source A edit window opens.

3. In the waveform editor at the bottom of the main edit window, drag the E handle to the very end of the audible part of the waveform.

   This effectively removes the silence at the end of the vocal phrase by reducing the playback length of the overall file.



4. Play C3 on your keyboard, and note that the phrase plays faster because the timing for the sound is currently based on the timing of the vocal sample.

   Also note that you can hear the entire synth lead phrase, which no longer fades away at the end. If you move the end marker further left, you can hear the entire phrase of the synth part played faster.

5. Click the X at the upper right to close the source A edit window.
6. Return to the morph controls, and adjust the Envelope knob to hear how it affects the behavior of samples with different lengths.

Use linked morph parameters in Alchemy

A green arc around a parameter indicates that it is morphed when you move dedicated morph controls. Setting general (non-morph area) parameters to different values for each source provides another technique that can influence morphing results.

1. In Alchemy in Logic Pro, click the source A button at the left, then click the Formant button at the upper right to view the Formant controls. Rotate the Smooth knob to the left until the Amanda vocal starts to break up and sound “fluttery.”



2. Click the source B button at the left, then click the Formant button at the upper right. Rotate the Smooth knob to the far left.
3. Click the Morph button at the left to show the morphing section at the upper right, then move the Formant handle (purple dot) horizontally while playing. You can also use the Formant knob to set this value.

   You can hear the “fluttery” effect as you move the Formant control to the left, but this disappears as you move it to the right.

Manipulate loops in Alchemy in Logic Pro for Mac

Logic Pro provides great options for working with loops and Apple Loops on tracks. If you want to trigger loops live, or shape them with synthesis tools, you can do so using Sampler, Ultrabeat, and Alchemy. Alchemy supports playback of Apple Loops as well as .aif, .caf, and .wav audio files, and you can use it to manipulate tempo and pitch in real time. Alchemy’s sophisticated synthesis engines can transform loops with multiple forms of resynthesis and extensive filter, modulation, and effects options.

Note: Images shown are not specific to the presets used in tasks. They are included as a guide to help you find areas and parameters in the Alchemy interface.

Time stretch and pitch shift samples in real time in Alchemy

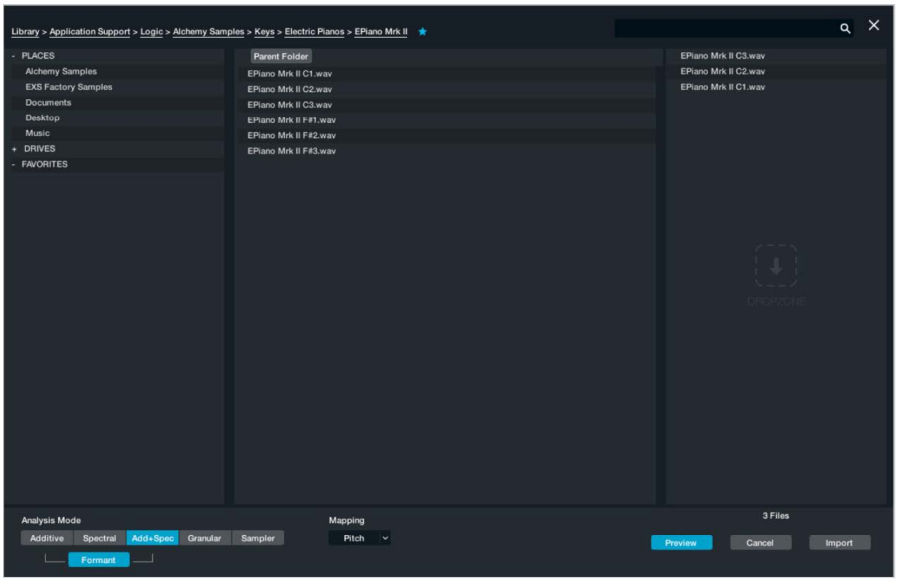
If you can’t find the loop used for this example, you need to download additional content packages. To do so, choose Logic Pro > Sound Library > Open Sound Library Manager from the Logic Pro menu, then download the missing content.

1. In Alchemy in Logic Pro, click the File button in the Name bar and choose Initialize Preset from the pop-up menu to reset all Alchemy parameters to default settings.

   Important: Use of Initialize Preset automatically turns on wide unison mode. See Alchemy wide unison mode.

2. Set the project tempo to 110 BPM.
3. Click the source select field for source A, and choose Import Audio from the pop-up menu.

   The Import window opens.



4. Open the Apple Loops browser, and type “Basic Vintage Break” in the search field. Drag the *Vintage Funk Loop 03* file from the Apple Loops browser to the Dropzone area of the Import window.

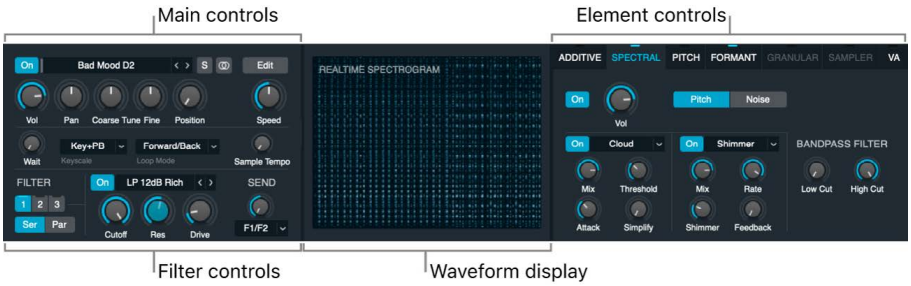
   Note that the original tempo of this file is 100 BPM and that it is 8 beats (2 bars) long.

5. Click the Spectral and Formant analysis mode buttons at the lower left, then click the Import button.

   This determines the synthesis engine used for resynthesis of the audio sample. Spectral + Formant is a good general purpose option that works well for drum loops. It’s also the default setting in a new instance of Alchemy. When analysis is complete, you can trigger the loop at its native pitch and speed by playing C3 on your keyboard. Playing other pitches transposes the loop, but retains the tempo.

6. Set the Speed knob in the source main controls section to zero percent (full left).

   This stops the loop from playing forward in time and repeats the first few samples at the current position.



7. Hold a note and rotate the Position knob to hear how playback is affected.

   The Position knob moves playback forward and backward through the loop. Set the value to zero percent (full left).

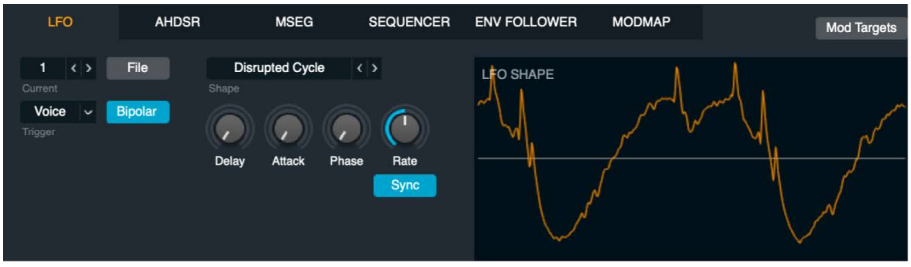
Modulate position with an LFO in Alchemy

1. In Alchemy in Logic Pro, Control-click the Position knob and choose Add Modulation > LFO > LFO 1 from the shortcut menu.

   This assigns LFO 1 to control the Position knob, and the central modulation section automatically updates to display the controls for LFO 1.

2. Switch the LFO shape from Sine to Ramp Up, and set the Rate to 2 (measures) to match the length of the Apple Loop.

   Tip: Changing the LFO shape to Ramp Down plays the loop backward by starting the Position at 100% and moving back to 0%.



3. Click the Bipolar button to turn it off.

   When Bipolar is active, the LFO outputs both negative and positive values in each cycle (from -50% to 50%). When off, only positive values are output (from 0% to 100%). If you play and hold a note on your keyboard, a small dot around the Position knob indicates its values changing from 0% to 100%. Bipolar is useful for controlling parameters like pan where 0 is in the midpoint of the knob's rotation and you want the value to oscillate around the center.

4. Play C3 to hear the loop at its native pitch but at a tempo that matches the project (110 BPM).

Explore further loop manipulation options in Alchemy

In Alchemy in Logic Pro, explore these additional options with the same loop to get a feel for Alchemy's loop manipulation capabilities.

- Turn off Trigger mode to hear the loop position continue to move through the file only when you play a note. This allows you to create gating and stuttering effects by rhythmically triggering notes. Try playing a series of very short 8th notes over this loop to change its feel.
- You can play notes of different pitches or can use pitch bend to add a chopped and screwed effect to your loops. The default pitch bend range is 2 semitones, but Alchemy supports bi-directional pitch bends of up to 36 semitones. To set this, click the Global view button and set the up and down pitch bend values independently in the master section.



Modulate position with an MSEG in Alchemy

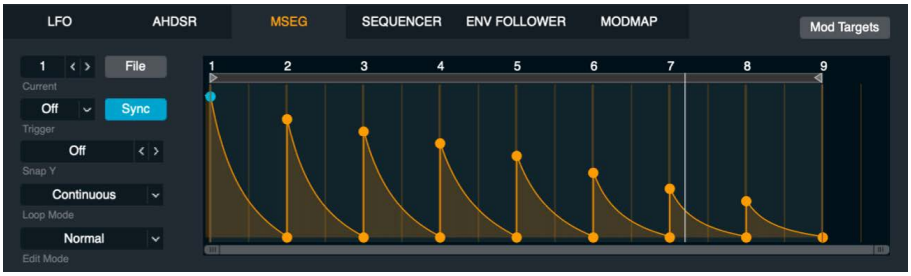
The MSEG envelope can duplicate the same kind of looping Ramp Up pattern used in the LFO task but is capable of longer, more complex patterns than a simple repeating ramp. MSEG envelopes are useful when you have specific ideas about how you want parameters to change over time. MSEG modulators are capable of creating a range of patterns from simple short ramps to very long, complex sequences.

1. In Alchemy in Logic Pro, click the Position knob, then click the button next to the modulation area to turn off the LFO.

   Play a few notes and you will hear that the loop no longer plays forward.

2. Control-click the Position knob, and choose Add Modulation > MSEG Env > MSEG 1 from the shortcut menu.

   This assigns MSEG 1 to control the Position knob. The central modulation section automatically updates to display the controls for MSEG 1.



3. Click the File button in the MSEG editor, and load the Ramp Up preset.

   This creates an 8 beat (2 bar) ramp from 0% to 100%. When you play a note, the result is identical to that of the LFO modulation used in the previous task.

4. Click along the graph line in the MSEG editor to add three points near bars 5, 7, and 9. Double-click a point to remove it.

   Note: The exact position of each point in the MSEG is important for keeping a consistent tempo.

5. Drag points horizontally to move them, noting that they snap to the tempo grid. Drag vertically to set the following values for each point.

   - Bar 1: Point value 0%
   - Bar 5: Point value 50%
   - Bar 7: Point value 25%
   - Bar 9: Point value 50%

6. Hold a note to hear the loop play forward, backward, and then forward again at the project tempo.

   More complex MSEG settings can cause the position to jump to specific points in the loop, thus creating variations of the original loop.

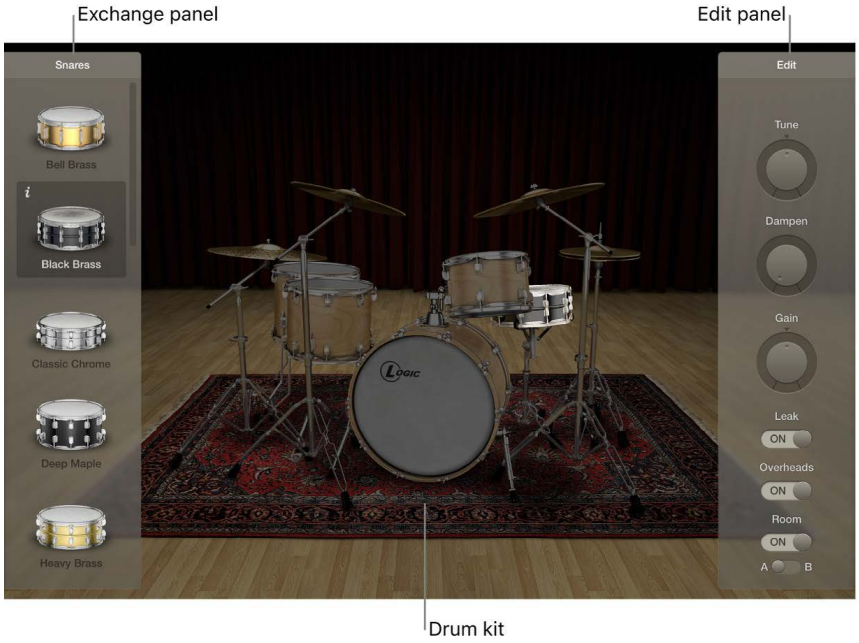
Drum Kit Designer

Drum Kit Designer in Logic Pro for Mac

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 multi-channel kits. Multi-channel kits are identified in the Library by a "+" at the end of the patch name. See "Work with Session Players" in Logic Pro Help for information on multi-channel kits.

If you're new to using plug-ins in Logic Pro, see Add, remove, move, and copy plug-ins.

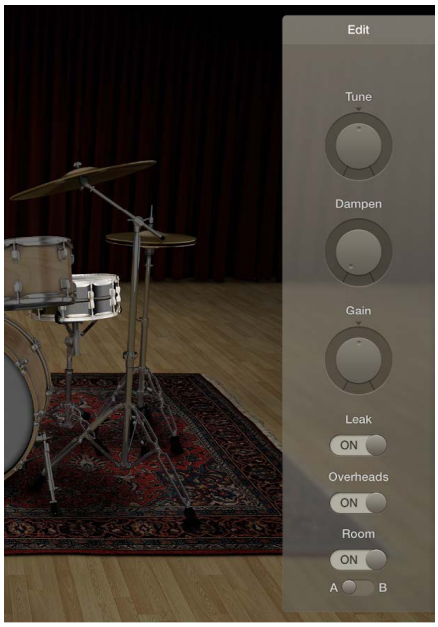


The Drum Kit Designer interface is divided into the following main areas.

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

Drum Kit Designer Edit pane in Logic Pro for Mac

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



Edit pane parameters

- *Tune knob and field:* Adjust the pitch of the selected kit piece.
- *Dampen knob and field:* Adjust the sustain of the selected kit piece.
- *Gain knob and field:* Adjust the volume of the selected kit piece.
- *Leak switch (multi-channel kits only):* Include the sound in the mic of the other kit pieces.
- *Overheads switch (multi-channel kits only):* Include the drum kit overhead mic in the sound.
- *Room switch (multi-channel kits only):* Choose between rooms A and B or to turn off the room emulation.

Use Drum Kit Designer in Logic Pro for Mac

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 multi-channel kits, you can additionally exchange toms, cymbals, and hi-hat. Multi-channel kits let you turn different microphones, such as overheads or room mics, on or off.

Note: Multi-channel 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

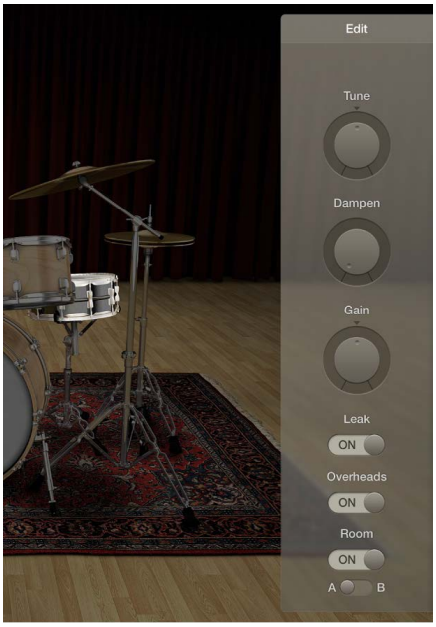
- In Logic Pro, click a drum or percussion piece.

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 pane to the left and can edit individual drum or percussion piece settings in the Edit pane to the right.

Adjust kit piece settings (all kits)

1. In Logic Pro, click a drum or percussion piece.

   The Edit pane 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 panes, click anywhere in the plug-in window background.

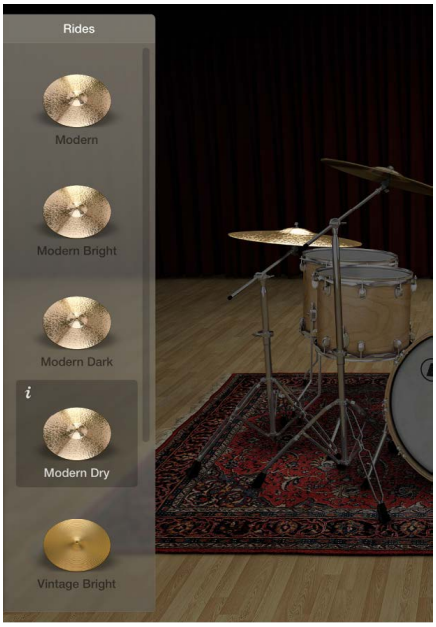
Exchange a kit piece

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

Note: Multi-channel kits and some drums are only available after you download additional content.

1. In Logic Pro, click a drum or percussion piece.

   The Exchange pane 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 pane. You may need to scroll 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 panes, click anywhere in the plug-in window background.

Adjust mic settings (multi-channel kits only)

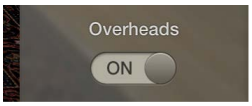
In the Edit pane, do any of the following:

- *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 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 in Logic Pro for Mac

Drum Kit Designer provides additional parameters that are accessed by clicking the disclosure arrow 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 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 in Logic Pro for Mac

Drum Kit Designer is compatible with the GM standard. You can also choose GM + ModWheel, which lets you use your keyboard modulation wheel to adjust the amount the hi-hat opens and closes during a drum performance.

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

The images show how drum sounds are remapped when different modes are chosen with the Input Mapping pop-up menu in the extended parameters, and when using kits that contain brush snares. Kits featuring brush snares provide a number of special features discussed below the images.



Brush snare special features

Kits that contain brush snares have preset behaviors that can enhance your performances and compositions.

Note: The following features apply only to keys that trigger a brush snare drum sound.

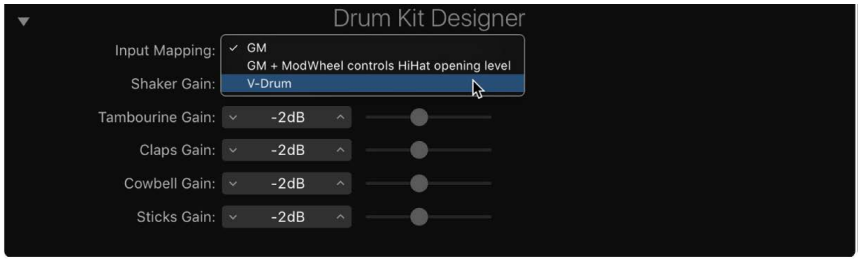
- Left and right hand playing is chosen automatically when using MIDI channel 1. This behavior is based on default settings such as the left hand being used for brush circles (if this hand isn't already in use), and playing situation. For example, snare taps will change from a default preference for the right hand to alternating hands if snare hits are played in rapid succession.

  Note: Hand usage can be forced by using MIDI channel 2 for the right hand and MIDI channel 3 for the left hand.

- Brush circles are played continuously while a note is held and loop randomly every bar, synchronized to the project downbeat, tempo, and meter. If a tap on the same hand as a held circle is played, the circle restarts immediately.
- There are two mute states that take brush force into account. These mute states can be active (caused by a brush hit) or passive (caused by the other brush resting on the snare), or can be a combination of both. The current mute state is automatically tracked, which results in a small sound variation each time a snare is played. For example, if a circle is playing, a snare tap will sound slightly different than when played from silence.
- The Snare Center Mute articulations (on keys C#0 and D#0) keep the brush on the snare head after the hit. These keys interrupt a playing circle on the same hand while held. The circle resumes when you release either key.

Choose mapping mode for Drum Kit Designer

1. In Logic Pro, click the disclosure arrow at the bottom-left corner of the plug-in window to show additional settings.
2. Choose one of the following from the Input Mapping pop-up menu:



   - GM
   - GM + ModWheel controls HiHat opening level
   - V-Drum

Drum Machine Designer

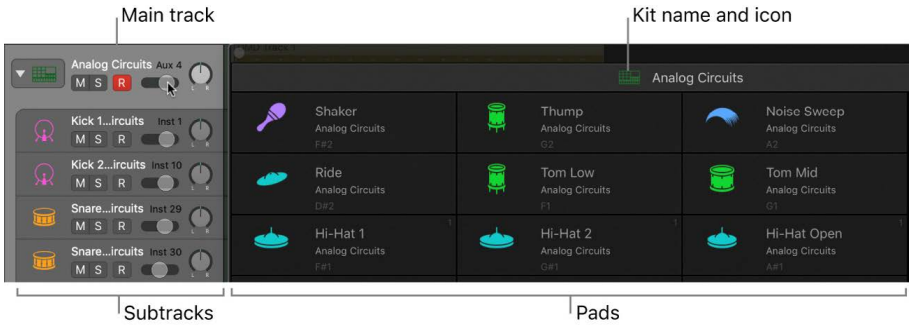
Drum Machine Designer overview in Logic Pro for Mac

Drum Machine Designer is the ideal way to create, organize, and interact with drum kits and kit pieces in Logic Pro for Mac. It is used for software instrument tracks but can also be played with a MIDI keyboard or pad controller on instrument channel strips.

When combined with Step Sequencer, it provides an incredibly flexible and inspiring platform for groove production.

Drum Machine Designer structure

Drum Machine Designer is *not* a plug-in, although it can be inserted into instrument channel strips. It is a track-based meta-instrument using a track stack, which combines a main track and a number of subtracks. Each track has a corresponding channel strip. For information on adding a Drum Machine Designer track to a project, see Use Step Sequencer with Drum Machine Designer.



- The main track channel strip is represented by the kit name and icon, shown in the kit controls bar at the top of the Drum Machine Designer window.
- Each subtrack channel strip is represented by a corresponding kit piece pad, shown in the grid.

Settings for the main track (kit) and all subtracks (kit pieces) are stored as a kit patch, which can hold multiple channel strips, each with its own instrument and effect plug-ins. Plug-in settings, by comparison, can hold only the plug-in settings of a single plug-in.

A single subtrack with its instrument and effect plug-ins is stored as a kit piece patch.

The icon shown on tracks and channel strips always matches the pad icon, and updates when a kit piece is exchanged or replaced. Double-click the track icon to open the Drum Machine Designer window. Click the arrow beside the main track icon to view or hide subtracks.

How Drum Machine Designer handles notes

MIDI notes received on the *main track* are distributed to subtracks, according to the input and output notes assigned to the pads. This is true for notes played by regions on the main track, and for notes played in real time when the main track is the focused track. For example, if the pad assigned to the first subtrack is set to input note C1 and to output note G2, a C1 played on the main track is converted to a G2. This note value is passed to the first subtrack and plays a G2 on the instrument inserted in the subtrack channel strip.

In contrast, notes played on a *subtrack* are not altered, and are passed directly to the instrument inserted in the channel strip, enabling you to play a subtrack instrument chromatically and polyphonically.

You can separate the notes in the main track MIDI region by doing one of the following:

- Select the main track MIDI region, then choose the Edit > Separate MIDI Events > By Note Pitch command.
- Control-click the main track MIDI region, then choose the Convert > Separate by Note Pitch command from the shortcut menu.

  Individual regions, containing note events, are created on each subtrack and can be handled and edited in the same way as any other MIDI region.

Drum Machine Designer sounds

You can access Drum Machine Designer kit patches and kit piece patches in the Library.

When the Library is open:

- Click the kit name to view and select kit patches in the Library.
- Click a kit piece pad icon or pad background to view and select kit piece patches in the Library.

Drum Machine Designer features an extensive collection of premapped kit patches and a large number of individual kit piece patches that you can add or use as replacement sounds for pads to create your own custom kits.

You can assign sounds to pads with patches from the Library, or drop audio material directly into Drum Machine Designer from the Browser, Loop Browser, or Audio Browser. You can also drop MIDI or audio regions onto Drum Machine Designer pads or to main track or subtrack headers.

Note: You can download additional sound packs in the Library to get more kits.

You edit the sound of the entire kit and individual kit pieces using Plug-ins. See Effects overview.

Drum Machine Designer interface in Logic Pro for Mac

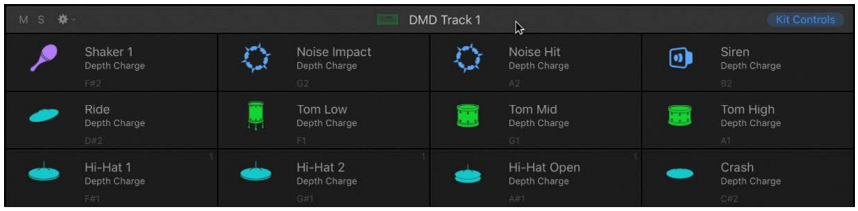
The Drum Machine Designer interface is divided into the following main areas and elements.



- *Kit controls bar:* A collection of kit pieces is known as a kit. Click the kit name or the Kit Controls button on the kit controls bar above the grid to interact with global effects and control settings for the entire kit in the Plug-in pane when visible. See Drum Machine Designer kit controls.
- *Grid:* Each pad offers its own kit piece or sound. Drum and percussion kit pieces assigned to pads can be either synthetically generated or sample-based. In fact, you can assign and use any instrument or plug-in you have at your disposal to a pad. Click the speaker icon on a pad to play the assigned sound. Click the icon or background of a pad to interact with its controls in the Plug-in pane. You can mute, solo, reorder, replace, and change the sound of each kit piece you have assigned to each pad. You can also assign each pad to different input and output MIDI note numbers and to different groups. See Drum Machine Designer kit controls and Use Drum Machine Designer pad controls.
- *Pad controls bar:* The Drum Machine Designer interface shows kit pieces laid out as pads in a grid that spans multiple pages. Click the page switcher controls (circles or arrows) on the pad controls bar below the grid to switch between pages. The pad controls bar also contains buttons that change the view shown in the Plug-in pane. See Drum Machine Designer grid pad controls bar.
- *Plug-in pane:* The area shows Quick Sampler parameters (Q-Sampler Main, Q-Sampler Details) if the kit piece uses the Quick Sampler plug-in. See Quick Sampler overview. This area also shows Smart Controls when you've inserted a plug-in on the track's channel strip. These buttons are visible only if the Plug-in pane is open. If the pane is not visible, click the disclosure arrow at the lower left of the grid.

Drum Machine Designer kit controls in Logic Pro for Mac

Drum Machine Designer shows a grid representation of the kit for the currently loaded patch. Each pad represents a kit piece. You can edit a number of settings for the entire kit, discussed in this section. To learn about interacting with kit pieces, see Drum Machine Designer Pad controls bar and Drum Machine Designer pad controls.



Kit controls bar parameters

To view kit Mute, Solo, and Action pop-up menu parameters, move the pointer over the kit controls bar.

- *Kit name and icon:* Click the kit name to view kit parameters in the Plug-in pane. Double-click the kit name to rename it (and the main track). Click the kit icon to stop all playing sounds. The Kit Controls button is highlighted when kit parameters are shown in the Plug-in pane.

  Note: The kit icon displays a muted speaker icon when you move the pointer over it during playback. This can be used to immediately stop playback of all kit pieces. This is useful when you want to silence a long One Shot sample that you are previewing in a Quick Sampler-based kit piece, for example.

- *Mute button:* Select to silence the kit (mute the main track channel strip).
- *Solo button:* Select to hear the kit in isolation. All other tracks and channel strips are muted, indicated by a flashing Mute button.
- *Action pop-up menu:* Choose a command that globally affects the appearance and behavior of the kit.
  - *Select Pad by Key:* Choose to turn on or turn off the automatic selection of pads with a MIDI keyboard. When on, the pad with an input note that corresponds to the incoming *played* MIDI note is selected.

    Note: MIDI notes played by Logic MIDI regions do not select pads, regardless of the setting of this menu item.
  - *Select Channel Strip by Pad:* Choose to turn on or turn off the selection of channel strips when you click a pad in the grid.

    Note: This option not only selects the corresponding subtrack and channel strip of the selected pad, but also sets focus to the main track. This ensures that notes played on the main track continue to be distributed to all subtracks. You can turn on both Select Pad by Key and Select Channel Strip by Pad to select pads, subtracks, and channel strips by playing a key on your MIDI keyboard.
  - *Open/Close Library:* Open or close the Library pane. Click a kit patch name in the Library to exchange the entire kit.

- *Use Empty Kit as Default:* Turn on this option to use an empty kit when you insert Drum Machine Designer on an instrument channel strip.
- *Hide/Show GM Drum Names:* Hide or show General MIDI (GM) drum names on each active pad. This is useful as a reference when mapping new sounds in a kit that you want to conform with the GM Drum standard. For example, when using Drum Machine Designer with Drummer.
- *Hide/Show Kit Piece Icons:* Hide or show the icons on each active pad.

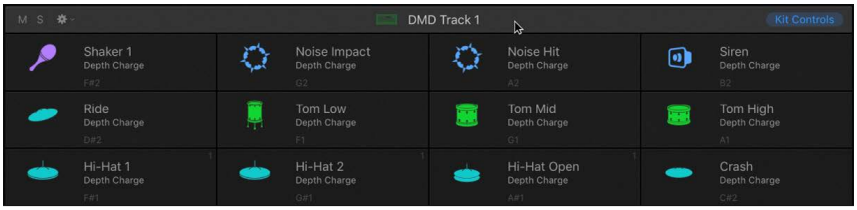
  Note: When you move the pointer over a pad, a small speaker icon is shown. Click this icon to play the sound.
- *Reorder Pads:* Choose the Change Sounds or Visual Only option to alter how pad layout changes affect assigned pad sounds. The pad layout can be changed by reordering pads. For example, when you insert a pad between other pads or drop one pad onto another to exchange them.
  - *Change Sounds:* Changing the pad layout moves the sounds of the pads, but pad input and output note settings are not changed: The sounds will appear in a different order when played via a MIDI keyboard or by a recorded region.
  - *Visual Only:* Changing the pad layout moves the pads and sounds together, so reordering will only have a visual effect: The pads will appear in a different order, but assigned sounds are still played by the same MIDI notes.
- *Reorder Pads by GM Drum Names:* Choose to automatically remap the order of pads to the General MIDI (GM) Drum standard, which is used by Drummer. This changes the visual order of pads, but does not remap sounds.

  Note: Regardless of the “Reorder Pads” setting, both “Sort Pads” options will sort the pads visually, in accordance with input note values.
- *Reorder Pads Chromatically:* Choose to automatically remap the order of pads chromatically from C1. This changes the visual order of pads, but does not remap sounds.

  Note: In order to play the instrument of a Drum Machine Designer subtrack chromatically and polyphonically, the instrument plug-in must react to different notes with different pitches, and its number of voices (polyphony) must be greater than one. If you have created a Drum Machine Designer kit and can’t play a subtrack instrument chromatically and polyphonically, please check that the instrument voices or polyphony parameter is set to a value greater than one. Also make sure the instrument key tracking parameter is turned on.
- *Update All Kit Names:* Choose to apply the current kit name to all kit pieces. This is useful when you add new kit pieces to an existing kit.
- *Assign Track Icon:* Choose to open a window where you can choose a kit icon from several categories. Select a category from the list and click an icon to assign it to the kit (and main track).
- *Clear all Pads:* Remove all assigned kit pieces from all pads.

Load and exchange kits

1. In Logic Pro, Click the Action pop-up menu in the kit controls bar in Drum Machine Designer, then choose Open Library.

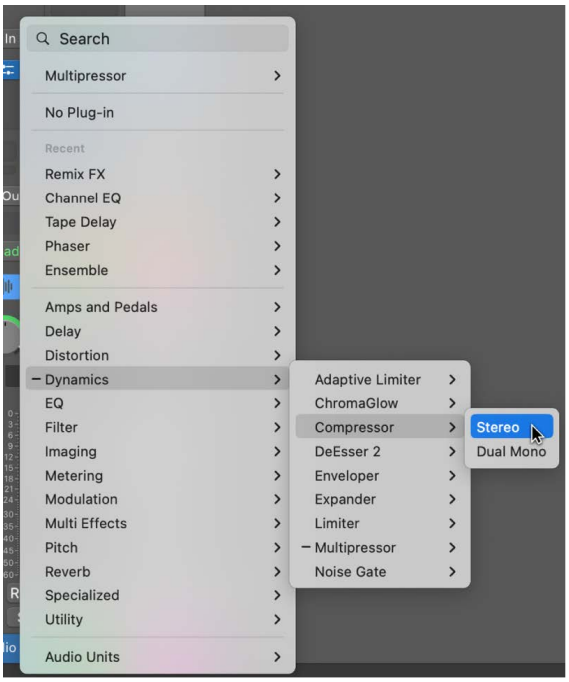


2. Click a kit patch in the Library to load that kit.

Edit the sound of the entire kit

You can add plug-ins to the main track of Drum Machine Designer to change the sound of the entire kit.

- In the Logic Pro channel strip for Drum Machine Designer, click an Audio Effect slot, then choose a plug-in from the pop-up menu.



See Add, remove, move, and copy plug-ins..

Save an edited kit as a kit patch

1. In Logic Pro, click the Save button at the bottom of the Library pane.

   If the Library is not visible, click the Action pop-up menu on the kit controls bar, then choose Open Library.

2. In the window that appears, navigate to the file location you want to use.

   The default folder location is User Kit Patches. If necessary, create a new folder.

3. Type the kit patch name, then press Save.

Pad controls

Drum Machine Designer pad controls bar in Logic Pro for Mac

The grid shows 16 pads on each of three available pages. A sound can be mapped to each pad. Use the Switcher controls in the pad controls bar *below* the grid to navigate between pages of pads.

Pads correspond to user-definable MIDI Input Note Numbers, typically starting from C1 at the lower left, and running to the upper right. When a pad play icon is clicked or it receives the corresponding Input Note Number, it sends a user-definable MIDI Output Note Number to the instrument plug-in in the channel strip of its subtrack.

Click either the Pad Controls button or a pad to view and edit kit piece parameters in the Plug-in pane. The Pad Controls button is highlighted when working with kit piece parameters in the Plug-in pane. Use the kit controls bar above the pads to view and edit kit parameters in the Plug-in pane when visible.

Note: These buttons are visible only if the Plug-in pane is open. If the pane is not visible, click the disclosure arrow at the lower left of the grid.



Pad controls bar parameters

If the subtrack of a Drum Machine Designer pad uses a Quick Sampler or Drum Synth plug-in, you will see Q-Sampler Main, Q-Sampler Details, or Drum Synth view buttons alongside the Pad Controls button below the grid. Use these buttons to access the respective instrument plug-in interface and parameters directly at the bottom of the Drum Machine Designer window.

- *Disclosure arrow:* Hide or show the plug-in editing area in the lower part of the Drum Machine Designer window.
- *Page switcher buttons:* Click the arrows or circle icons to choose pages in the grid.
- *Q-Sampler Main button:* View the Quick Sampler waveform display editor in the lower part of the Drum Machine Designer window. The button is highlighted when working with the Quick Sampler waveform display in the Plug-in pane, if visible. See Quick Sampler overview and Quick Sampler waveform display.

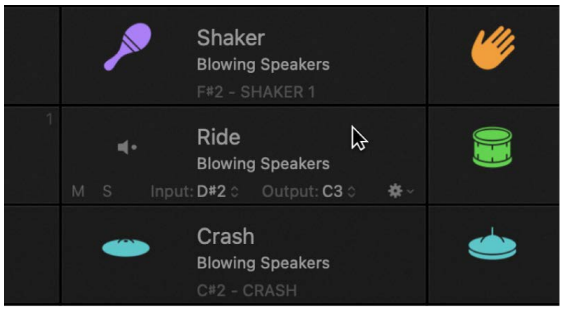
- *Q-Sampler Detail button:* View the Quick Sampler synthesis and other parameters in the lower part of the Drum Machine Designer window. The button is highlighted when working with Quick Sampler synthesis parameters in the Plug-in pane, if visible. This contains Quick Sampler LFO controls and a modulation matrix (Mod Matrix), along with pitch, filter, and amp controls.
- *Drum Synth button:* Show Drum Synth parameters in the lower part of the Drum Machine Designer window. The button is highlighted when working with Drum Synth parameters in the Plug-in pane, if visible. See Drum Synth overview.
- *Pad Controls button:* Show pad tone and effect parameters in the Plug-in pane. The button is highlighted when working with pad parameters in the Plug-in pane.

  Note: The Pad Controls button is visible only if the Plug-in pane is open.

Use Drum Machine Designer pad controls in Logic Pro for Mac

Move your pointer over, or click, the icon or pad background of any of the 16 pads shown on each grid page to access pad parameters in Drum Machine Designer. Use the pad controls bar below the grid to access editing options and to navigate between pad pages.

Use the kit controls bar above the grid to view and edit kit parameters in the Plug-in pane when visible.



Pad parameters

- *Pad:* Move the pointer over the pad to view Mute, Solo, Input, Output, and Action pop-up menu options. Click the speaker icon to play the sound. Drag the pad to reorder it in the grid. Double-click the pad to rename it (and the subtrack). You can also exchange the kit piece sound assigned to the pad.

  Note: When you click a pad icon or pad background, the Plug-in pane updates to show any plug-ins you inserted on the subtrack. Click the disclosure arrow at the lower left of the grid if the Plug-in pane is not visible.

  Note: The kit icon in the kit controls bar displays a muted speaker icon when you move the pointer over it during playback. This can be used to immediately stop playback of all kit pieces. This is useful when you want to silence a long One Shot sample that you are previewing in a Quick Sampler-based kit piece, for example.
- *Unassigned pad:* A pad without a corresponding subtrack is called an unassigned pad. You can recognize it by the missing icon, missing kit piece name and missing kit name, and by the "+" icon that appears when you move the pointer over the pad. Click the "+" icon to create a new, empty subtrack and channel strip, or click the pad and choose a kit piece in the Library. You can also drag an audio sample or a region onto the pad.

Note: In cases where you have used the Clear Pad command, the pad will have no assigned sound, but will have a corresponding subtrack, so the "+" icon is not displayed.

- *Mute button:* Use to silence the subtrack assigned to the pad. Multiple pads can be muted.
- *Solo button:* Use to hear the sound in isolation. This solos the subtrack. All other pads are muted, indicated by a flashing Mute button. Multiple pads can be soloed.
- *Input pop-up menu:* Choose, or learn, an input note number for the pad. You can assign multiple pads to the same input note, allowing you to create layered sounds consisting of multiple channel strips with different instruments.
- *Output pop-up menu:* Choose, or learn, an output note number for the pad.

  Note: MIDI notes received on the main track are distributed to subtracks, according to the input and output notes assigned to the pads. This is true for notes from regions on the main track, and for notes played in real time when the main track is the focused track. For example, if the pad assigned to the first subtrack is set to input note C1 and to output note G2, a C1 played on the main track is converted to a G2. This note value is passed to the first subtrack and plays a G2 on the instrument inserted in the subtrack channel strip.
- *Action pop-up menu:* Choose a command for the pad.
  - *Exclusive Group:* Choose a group for the pad. You can assign multiple pads to the same group. As soon as one drum sound in the group is triggered, all other sounds in that same group are stopped. For example, you could group three pads with open, semi-closed, and closed hi-hat sounds. Only one of these sounds can be played at a time, mirroring the behavior of real hi-hats.

    Note: The group number is shown at the top right of each pad in the group when any group member is selected.
  - *Open/Close Library:* Open or close the Library pane. Click a patch name in the Library to exchange the assigned kit piece.
  - *Resample Pad:* Choose to initiate a resampling process. This sends the input note of the selected pad to Drum Machine Designer, which plays the assigned instrument, inclusive of all channel strip and main track plug-ins. The resulting Drum Machine Designer output is resampled, and loaded into a new Quick Sampler instance on a new channel strip and subtrack, which is automatically assigned to the first available (lowest numbered) unassigned pad.

    Note: If multiple pads are assigned to the same input note, the mixed sound of all pads is resampled. This allows you to combine the sounds of multiple pads onto one pad.
  - *Update Kit Name for Kit Piece:* Choose to apply the current kit name to the selected kit piece.
  - *Assign Track Icon:* Choose to open a window where you can choose a kit piece icon from several categories. Select a category from the list and click an icon to assign it to the kit piece.
  - *Clear Pad:* Remove the kit piece assigned to the pad. This clears the channel strip of the pad subtrack.

Add kit pieces to unassigned pads

1. In Logic Pro, move the pointer over an unassigned pad.
2. Do one of the following:
   - Click a kit piece patch in the Library to add the kit piece.
   - Drag a MIDI or audio region, or an audio file from the Finder, Audio File Browser, or Loop Browser, onto the pad. You can use any audio file type that Logic Pro supports.

     A resampling process takes place when a region is dropped onto a pad and a new Quick Sampler instance is inserted on a new subtrack, which is automatically assigned to the first available (lowest numbered) unassigned pad.
   - Click the "+" icon that appears to create a new, empty channel strip and subtrack.

     Note: In cases where you have used the Clear Pad command, the pad will have no assigned sound, but will have a corresponding subtrack, so the "+" icon is not displayed. Click the pad background to assign a sound to the pad and subtrack.

Add samples to unused pads on a Drum Machine Designer main track or subtrack

You can quickly add samples to unused pads by dragging content to the main track header, or you can replace the sample for a pad by dragging content to the corresponding subtrack header.

1. In Logic Pro, drag one or more audio files, regions, or Apple Loops to a Drum Machine Designer main track or subtrack header.
   - When adding an audio file, the "Add Sample to" dialog appears in the track header. Drag the items to the Drum Machine Designer zone.

     The items are added to the first (lowest numbered) unassigned pads.
   - When adding an audio or MIDI region, a resampling process is triggered, with the region bounced offline as an audio file.
   - When adding a Step Sequencer region, a resampling process is triggered, with the region bounced offline as an audio file.

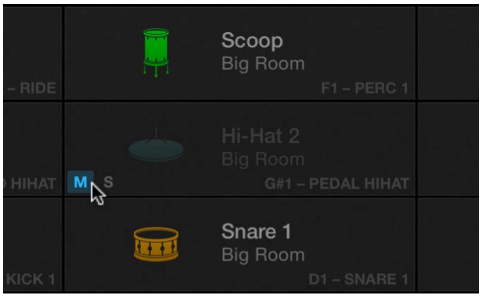
     Note: The resampling process that takes place when a region is dropped onto a pad inserts a new Quick Sampler instance on a new subtrack, which is mapped to the first available (lowest numbered) unassigned pad.

Play a drum or kit piece

- In Logic Pro, do one of the following:
  - Place the pointer over a pad, then click the speaker icon. The sound is played from start to end.
  - Place the pointer over a pad, then click-hold the speaker icon. The sound is played and stops immediately when you release the button.

Mute a kit piece

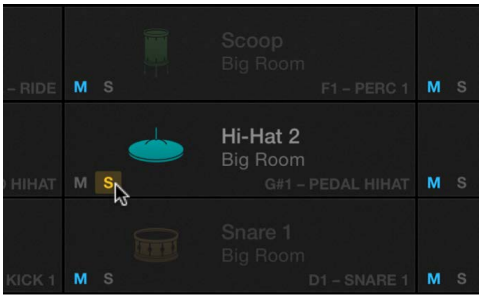
- In Logic Pro, place the pointer over the pad to view available options, then click the Mute button that appears.



  The muted pad is dimmed and the Mute button on the dimmed pad is highlighted. This silences the subtrack assigned to the pad.

Solo a kit piece

- In Logic Pro, place the pointer over the pad to view available options, then click the Solo button that appears.



  All other pads are dimmed, and the Solo button is highlighted. This solos the subtrack assigned to the pad.

Edit the sound of a kit piece

You can use Plug-in parameters to process and affect the overall kit or selected kit piece.

Note: If the Plug-in pane is not visible, click the disclosure arrow at the lower left of the grid.

1. In Logic Pro add a plug-in to the subtrack for the kit piece. See Add, remove, move, and copy plug-ins.
2. In Drum Machine Designer, click the icon or a pad background of the kit piece you want to edit.
3. Click the Pad Controls button in the pad controls bar.

Note: If a Drum Machine Designer subtrack or pad uses a Quick Sampler or Drum Synth plug-in, you will see additional Q-Sampler Main, Q-Sampler Details, or Drum Synth view buttons alongside the Pad Controls button below the grid. Use these to access the respective instrument plug-in interface and parameters directly at the bottom of the Drum Machine Designer window. See Quick Sampler overview and Drum Synth overview for details on editing parameters that you can view in the lower portion of the Drum Machine Designer interface.



Save an edited kit piece sound as a kit piece patch

1. In Logic Pro, click the icon or a pad background of the edited kit piece you want to save.
2. Click the Save button at the bottom of the Library pane. If the Library is not visible, click the Action pop-up menu on the pad, then choose “Open Library”.

   In the Finder window:
3. Browse to the file location you want to use. The default folder location is User Kit Piece Patches. If required, create a new folder.
4. Type the kit piece patch name, then press the Save button.

Reorder kit pieces

In Logic Pro, you can reorder kit pieces in the following ways:

- Drag a pad to an *empty* location in the grid.
- Drag a pad *between two pads* in the grid. A vertical blue line is shown to indicate the target location. The pad is placed between the target pads. Adjacent (and other) pads move to accommodate the dropped pad.
- Drag a pad *between pages* in the grid. An arrow is briefly shown at the left or right edge of the grid, which then moves to the target page. Drop the pad at the location you want to use on the page.
- Drag a pad *onto another pad* in the grid to swap their positions.

  Note: Dragging behavior is affected by the option you chose in the “Reorder Pads” submenu in the kit controls bar Action pop-up menu.

  - *Change Sounds:* Changing the pad layout moves the sounds of the pads, but pad input and output note settings are not changed: The sounds will appear in a different order when played via a MIDI keyboard or by a recorded region.
  - *Visual Only:* Changing the pad layout moves the pads and sounds together, so reordering will only have a visual effect: The pads will appear in a different order, but assigned sounds are still played by the same MIDI notes.

Replace kit pieces using pads

1. In Logic Pro, click a pad icon or pad background to select it.



2. Do one of the following:

   - Click a kit piece patch in the Library to exchange the kit piece. If the Library is not visible, choose Open Library from the Action or shortcut menu. Use Control-click to access the shortcut menu.
   - Drag a MIDI or audio region, or an audio file from the Finder, Audio File Browser, or Loop Browser, onto the pad. You can use any audio file type that Logic Pro supports.

     Note: A Replace with Quick Sampler message is shown on the pad.

Replace kit pieces using Drum Machine Designer subtracks

1. In Logic Pro, drag an audio file, region, or Apple Loop to a Drum Machine Designer subtrack.
2. When the Replace Sample dialog appears, drop the item.

Drum Synth

Drum Synth overview in Logic Pro for Mac

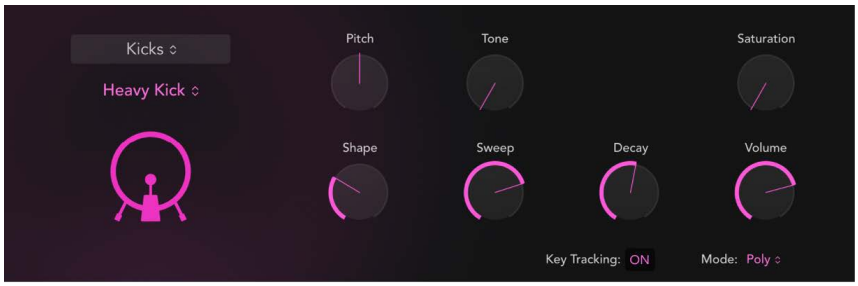
Drum Synth provides you with easy access to a powerful electronic drum synthesis engine. Depending on the sound type you've chosen, up to eight specially selected controls for pitch, filtering, effect, and envelopes allow you to quickly create the electronic drum sound you need. You can, of course, add further processing plug-ins in the instrument channel strip or busses.

Drum Synth is ideal for adding fills, flourishes, or individual hits to an existing Drummer groove, loops, and live or sequenced parts, and for drum replacement and doubling tasks.

You can use multiple Drum Synth instances, either as individual software instrument tracks or inside Drum Machine Designer, which allows you to build complete electronic drum sets, encompassing everything from vintage drum machine through to modern synthetic percussion sounds.

You can also directly load Drum Synth settings into Ultrabeat to access additional synthesis and processing parameters. Typically, you would replace Drum Synth with Ultrabeat on a channel strip, and the Drum Synth sound is automatically loaded.

If you're new to using plug-ins in Logic Pro, see Add, remove, move, and copy plug-ins.



Drum Synth global parameters

Drum Synth has four global parameters in the plug-in window.

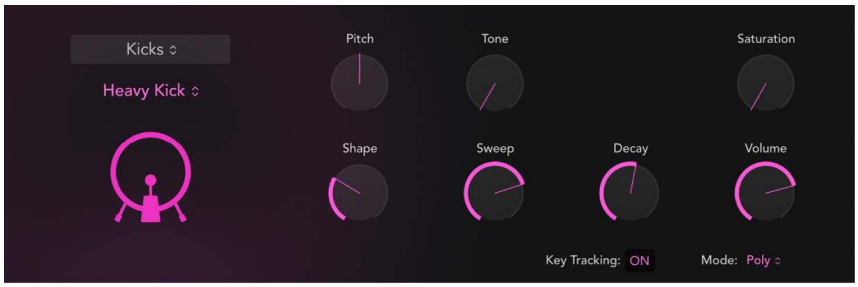
- *Group type pop-up menu:* Choose a drum or percussion instrument family: Kicks, Snares and Claps, Percussion, Hats and Cymbals.
- *Sound type pop-up menu:* Choose a drum or percussion sound type from the active instrument group. Back to Default reverts all parameter changes to their initial values.
- *Instrument icon:* Click to play the sound.
- *Key Tracking button:* Turn on to play the sound chromatically. Turn off to play the sound at the root pitch when any key is struck.

  The "Root Key" label is shown beside the keyboard in the Piano Roll Editor.
- *Mode pop-up menu:* Choose a playback mode.
  - *Mono:* One voice can be played. If you play another key, the newly played note cuts off the playing note.
  - *Poly:* You can play multiple notes simultaneously.
  - *Gate:* Like 'Mono', but the sound is heard only while you hold the key.

The parameters of individual types of drum sound are covered in other sections.

Drum Synth Kicks in Logic Pro for Mac

You can create kick drums that range from soft, classic electronic sounds to more modern and punchy sounds with the Drum Synth Kicks group instruments.



Drum Synth kick sound parameters

You can change the kick drum sound by adjusting its parameters. Not all parameters are available for each type of kick drum sound. Parameters interact with each other, changing the behavior and impact of adjustments you make, resulting in a broader range of potential sounds. Click the drum icon to play the sound.

- *Body knob:* Alter the depth of the sound, making it fuller and richer.
- *Decay knob:* Set the length of the decay phase of the sound.
- *Noise knob:* Add a noise signal to the sound.
- *Pitch knob:* Set the tuning of the sound in semitone steps.
- *Punch knob:* Adjust the tone of the attack phase of the sound.
- *Saturation knob:* Set the saturation amount. This adds a warm distortion.
- *Shape knob:* Adjust the shape of the kick drum.

- *Snap knob:* Set the amount of beater noise.
- *Sweep knob:* Adds heft to the attack phase of the sound.
- *Tone knob:* Set the overall tonal color of the sound.
- *Volume knob:* Set the output level of the sound.

Drum Synth Snares and Claps in Logic Pro for Mac

You can create synthetic and electronic snares, rimshots, and handclaps with the Drum Synth Snares and Claps group instruments.



Drum Synth snare and clap sound parameters

You can change the snare or clap sound by adjusting its parameters. Not all parameters are available for each type of snare or clap sound. Parameters interact with each other, changing the behavior and impact of adjustments you make, resulting in a broader range of potential sounds. Click the drum or clap icon to play the sound.

- *Body knob:* Alter the depth of the sound, making it fuller and richer.
- *Crush knob:* Add a lo-fi digital distortion to the sound.
- *Cycles knob:* Smear the sound. Set the rate with the Speed knob.
- *Decay knob:* Set the length of the decay phase of the sound.
- *Dirt knob:* Add grit and fizz to the sound.
- *Noise knob:* Add a noise signal to the sound.
- *Pitch knob:* Set the tuning of the sound in semitone steps.
- *Position knob:* Set the position of the drum strike.
- *Size knob:* Set the size of the drum shell.
- *Speed knob:* Set the rate of cycles. Use the Cycles knob to set intensity of smearing.
- *Tension knob:* Adjust snare tension, making the sound snap more or less.
- *Tone knob:* Set the overall tonal color of the sound.
- *Volume knob:* Set the output level of the sound.

Drum Synth Percussion in Logic Pro for Mac

You can create electronic tom sounds, synthetic percussion and wild metallic tones with the Drum Synth Percussion group instruments.



Drum Synth percussion sound parameters

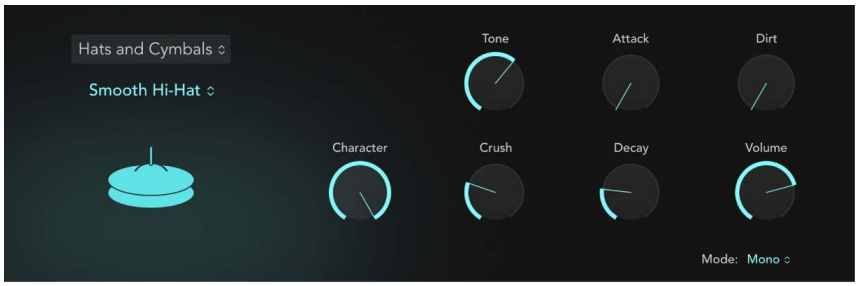
You can change the percussion sound by adjusting its parameters. Not all parameters are available for each type of percussion sound. Parameters interact with each other, changing the behavior and impact of adjustments you make, resulting in a broader range of potential sounds. Click the percussion instrument icon to play the sound.

- *Attack knob:* Set the length of the attack phase of the sound.
- *Body knob:* Alter the depth of the sound, making it fuller and richer.
- *Color knob:* Set the tonal color of the sound.
- *Crush knob:* Add a lo-fi digital distortion to the sound.
- *Decay knob:* Set the length of the decay phase of the sound.
- *Dirt knob:* Add grit and fizz to the sound.
- *Dissonance knob:* Alter tones by introducing inharmonic content.
- *Harmonic knob:* Alter metallic tones by introducing harmonic content.
- *Material knob:* Make the sound more woody, rubbery or bell-like. The Material knob interacts with the Tension knob.

  Note: The Material knob can also be impacted heavily by the Dissonance knob, if available as a parameter in the chosen sound.
- *Noise knob:* Add a noise signal to the sound.
- *Pitch knob:* Set the tuning of the sound in semitone steps.
- *Saturation knob:* Set the saturation amount. This adds a warm distortion.
- *Snap knob:* Set the tone of the attack phase.
- *Tension knob:* Adjust the rigidity of the core sound you set with the Material knob.
- *Tone knob:* Set the overall tonal color of the sound.
- *Volume knob:* Set the output level of the sound.

Drum Synth Hats and Cymbals in Logic Pro for Mac

You can create electronic crash and ride cymbal sounds, hi-hat sounds, and shaker sounds with the Drum Synth Hats and Cymbals group instruments.



Drum Synth hats and cymbals parameters

You can change the hat or cymbal sound by adjusting its parameters. Not all parameters are available for each type of hat or cymbal sound. Parameters interact with each other, changing the behavior and impact of adjustments you make, resulting in a broader range of potential sounds. Click the hat or cymbal icon to play the sound.

- *Attack knob:* Alter the attack phase of the sound.
- *Bell knob:* Introduce bell-like harmonics into the sound.
- *Body knob:* Alter the depth of the sound, making it fuller and richer.
- *Character knob:* Add a metallic edge to the sound with higher values.
- *Crush knob:* Add a lo-fi digital distortion to the sound.
- *Decay knob:* Set the length of the decay phase of the sound.
- *Dirt knob:* Add grit and fizz to the sound.
- *Dissonance knob:* Alter tones by introducing inharmonic content.
- *Metallic knob:* Alter tones by introducing harmonic content.
- *Noise knob:* Add a noise signal to the sound.
- *Pitch knob:* Set the tuning of the sound in semitone steps.
- *Tone knob:* Set the overall tonal color of the sound.
- *Volume knob:* Set the output level of the sound.

ES1

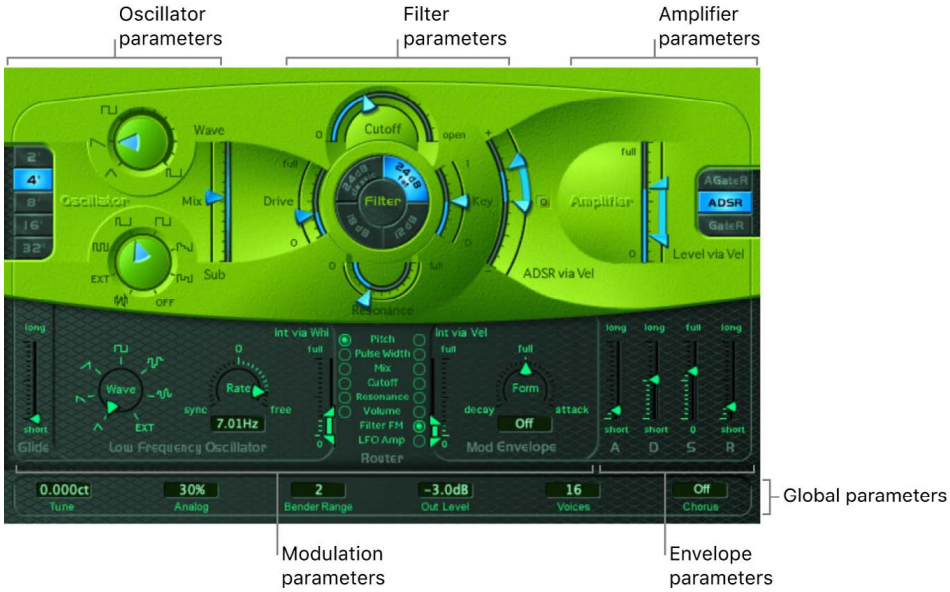
ES1 overview in Logic Pro for Mac

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 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, which will introduce you to the terminology and give you an overview of different synthesis systems and how they work.

If you're new to using plug-ins in Logic Pro, see Add, remove, move, and copy plug-ins.



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.
- *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.

- *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.
- *Amplifier parameters:* Located in the upper right, the amplifier parameters allow you to fine-tune sound level behavior. See ES1 amplifier parameters.
- *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.
- *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.

ES1 oscillator parameters

ES1 oscillators in Logic Pro for Mac

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:* Select the waveform of the primary oscillator, which is responsible for the basic color of the tone. See ES1 oscillator waveforms.
- *Mix slider:* 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:* 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.
- *2′, 4′, 8′, 16′, and 32′ buttons:* 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

- In Logic Pro, 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). Modulating the pulse width with a slowly cycling LFO, for example, allows periodically mutating, fat bass sounds.

ES1 oscillator waveforms in Logic Pro for Mac

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, and pads
Square	Hollow and “woody” sounding	Useful for basses, clarinets, and oboes
Pulse	“Nasal” sounding	Great for reed instruments, synth blips, and basses

Use the ES1 sub-oscillator in Logic Pro for Mac

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 turn off 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. In Logic Pro, 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 in Logic Pro for Mac

The global parameters affect the overall sound, or behavior, of 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:* 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:* Tune the instrument in cents. One cent is 1/100 of a semitone.
- *Analog field:* Change the pitch of each note and the cutoff frequency slightly and randomly. 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:* Set the sensitivity of the pitch bender, in semitone steps.
- *PB Range Down slider (Extended Parameters area):* Set the pitch bend range independently for upward and downward bends. Click the disclosure arrow at the lower left of the ES1 interface to access the Extended Parameters area.
  - *Linked:* The negative pitch band uses the value set in the global Bender Range field. Positive and negative bend ranges are the same.
  - *0:* No negative bend range at all, but the global value for positive bends is retained.
  - *1-24:* Independent amount of negative bend range to allow for different ranges. For example, +2 and -12.
- *Out Level field:* Set the ES1 master volume.
- *Voices field:* Set the maximum number of notes that can be played simultaneously.

  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 pitch bender, by choosing a high Glide parameter value when using the Legato setting.

  When set to *full* (polyphony), each played note has its own synth voice and an automatic release cutoff comes into effect. If you have set a long release time and play a non-legato chord progression, the chords won't smear into each other, which is useful for classic string synthesizer emulations.

- *Chorus field:* Choose a classic stereo chorus effect, an ensemble effect, or turn off the effects processor.
  - Off turns off 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.
- *Midi Mode pop-up menu (Extended Parameters area):* Determine how ES1 responds to MIDI controllers. Choose either Off or Full Remote.

ES1 filter parameters

ES1 filter parameters in Logic Pro for Mac

This section outlines the ES1 filter parameters.



Filter parameters

- *Cutoff slider:* Set the cutoff frequency of the lowpass filter.
- *Resonance slider:* 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 Overdrive the ES1 filter).

  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. Increase the Resonance parameter value to reduce 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 the Roland TB-303.

- *Drive slider:* 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:* 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.
- *Filter Boost button (Extended Parameters area):* 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 Overdrive the ES1 filter.

Overdrive the ES1 filter in Logic Pro for Mac

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.

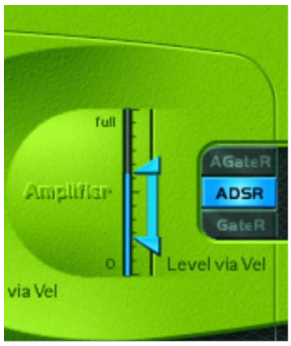
Output a sine wave from the filter

1. In Logic Pro, 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 arrow 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.

Logic Pro for Mac ES1 amplifier parameters

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



Amplifier parameters

- *Level via Vel slider:* 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:* Determine the ADSR envelope generator used for control of the amplifier envelope. See ES1 envelope parameters overview.

ES1 envelope parameters

ES1 envelope parameters in Logic Pro for Mac

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

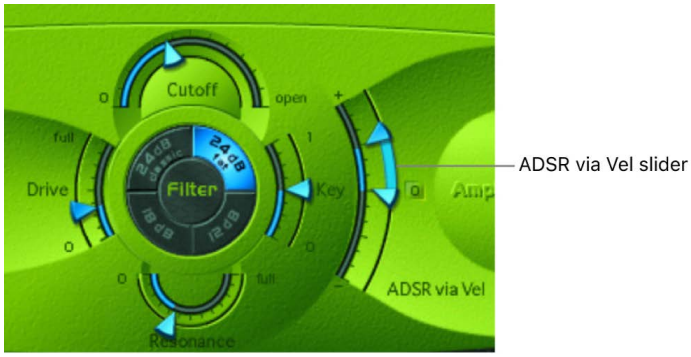


Envelope parameters

- *Attack slider:* Set the time it takes for the envelope to reach the initial desired level.
- *Decay slider:* Set the time it takes for the envelope to fall to the sustain level, following the initial attack time.
- *Sustain slider:* Set the sustain level, which is held until the key is released.
- *Release slider:* Set the time it takes the envelope to fall from the sustain level to a level of 0.

ES1 cutoff envelope modulation in Logic Pro for Mac

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



Set the modulation range with the two arrows on the ADSR via Vel slider.

- 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.

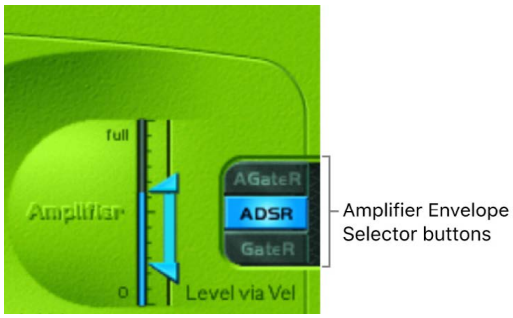
Tip: If you're unfamiliar with these parameters, set the Cutoff parameter to a low value, Resonance to a high value, and move both ADSR via Vel arrows upward. Constantly strike a note on the keyboard while changing the arrows to learn how these parameters work.

ES1 amplifier envelope modulation in Logic Pro for Mac

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).

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.



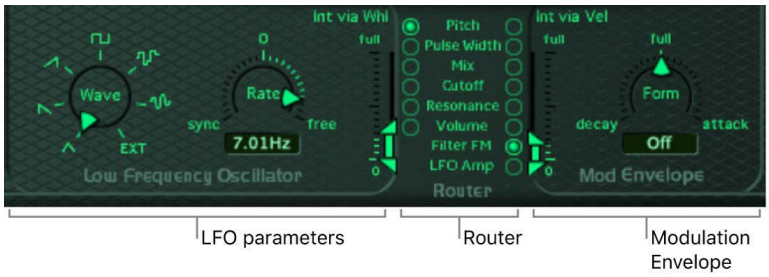
The ES1 amplifier envelope mode buttons affect played notes in different ways.

- *AGateR:* In this mode, move the Attack and Release sliders of the ADSR envelope to set these phases of the sound. Between these phases, the Gate control signal maintains a constant level while a note is held. When you release the key, the release phase begins immediately. The Decay and Sustain sliders of the ADSR Envelope have no impact on the sound level.
- *ADSR:* Set this mode to control the level of the sound over time with the ADSR envelope.
- *GateR:* In this mode, the Gate control signal maintains 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 level.

ES1 modulation

ES1 modulation overview in Logic Pro for Mac

ES1 offers a number of 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.

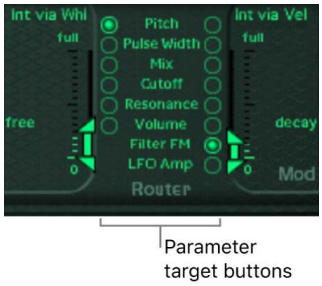


Modulation parameters

- *LFO parameters:* Used to modulate other ES1 parameters. See Use the ES1 LFO.
- *Router:* Used to choose which ES1 parameters are modulated. See Use the ES1 router.
- *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.

Use the ES1 router in Logic Pro for Mac

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.

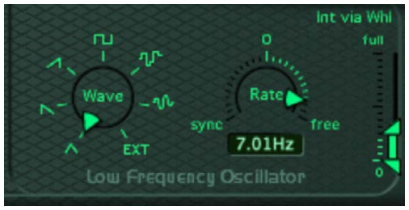


Router parameters

- *Pitch buttons:* Modulate the pitch—the frequency—of the oscillators.
- *Pulse Width buttons:* Modulate the pulse width of the pulse wave.
- *Mix buttons:* Modulate the mix between the primary oscillator and the sub-oscillator.
- *Cutoff buttons:* Modulate the cutoff frequency of the filter.
- *Resonance buttons:* Modulate the resonance of the filter.
- *Volume buttons:* Modulate the main volume.
- *Filter FM button (mod env only):* Use the triangle waveform 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 Overdrive the ES1 filter).
- *LFO Amp (mod env only):* Modulate the overall amount of LFO modulation.

Use the ES1 LFO in Logic Pro for Mac

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



LFO parameters

- *Wave knob:* 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 dial and field:* Set the speed, or 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 Logic Pro—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 modulation wheel. For example, to change the pulse width by moving your keyboard modulation wheel, choose pulse width as the LFO modulation target and set the modulation intensity range using the Int via Whl slider.
- *Int via Whl slider:* Move the upper arrow to set LFO intensity at the maximum modwheel position. The lower arrow defines LFO intensity when the modwheel is set to zero. The distance between the arrows—shown as a green bar—indicates keyboard modulation wheel range.

  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 in Logic Pro for Mac

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 dial and field:* 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:* Set the top arrow to define the upper modulation envelope limit 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. In Logic Pro, 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

- In Logic Pro, drag the Form slider to a positive value—toward attack to fade in the LFO modulation. The higher the value, the longer it takes for you to hear the modulation.
- Drag the Form slider to a negative value—toward decay to fade out the LFO modulation. 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. In Logic Pro, 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 controller assignments in Logic Pro for Mac

The table outlines the way ES1 responds to 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

Controller number	Parameter name
23	Decay slider
24	Sustain slider
25	Sustain 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

ES2 overview in Logic Pro for Mac

ES2 combines subtractive synthesis with elements of FM and wavetable synthesis to 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, which introduces you to the fundamentals and terminology of different synthesis systems.

If you're new to using plug-ins in Logic Pro, see Add, remove, move, and copy plug-ins.

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.

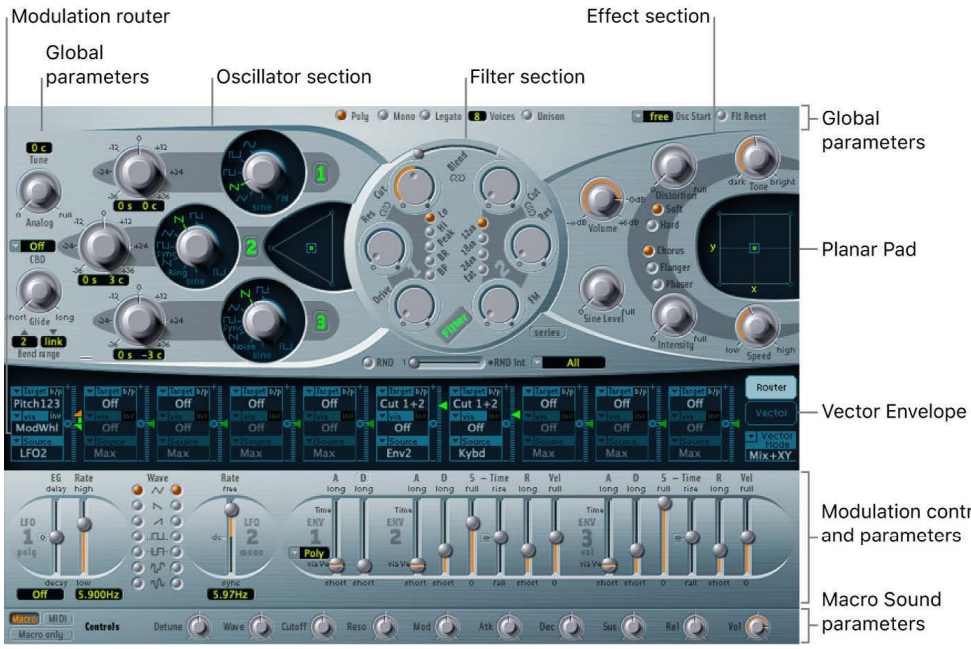
Lastly, Distortion, Chorus, Phaser, and Flanger effects are built into ES2.

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 ES2. See ES2 sound design from scratch and ES2 sound design with templates.

Note: You can find tasks that cover the use of parameters as modulation targets or sources throughout the ES2 Help. This underlines one of the strengths of ES2—the vast modulation possibilities it offers. Follow the steps in these tasks to create expressive, evolving sounds. See ES2 modulation overview.

ES2 interface overview in Logic Pro for Mac

The ES2 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.
- *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 ES2 global parameters overview.
- *Filter section:* The circular area houses the filter section, including the Drive and Filter FM parameters. See ES2 filter overview.
- *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 the ES2 dynamic stage.
- *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 ES2 modulation router.
  - The Vector Envelope is a flexible, powerful envelope generator that provides extensive control over your sound. See Use the ES2 Vector Envelope.
- *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.

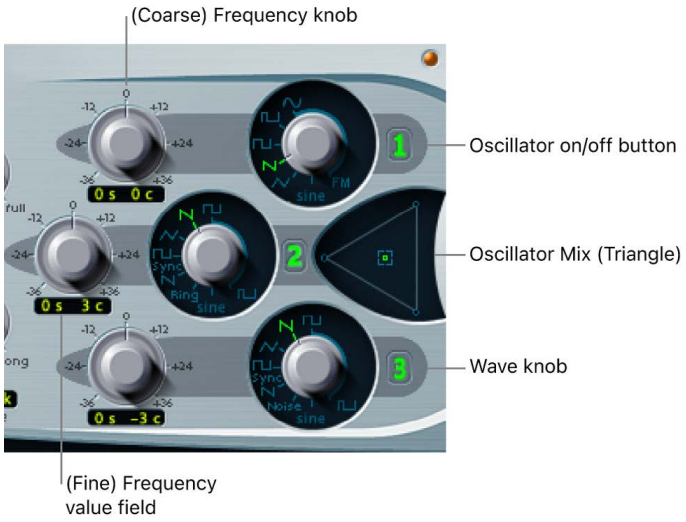
- *Planar Pad:* The square area at the top right is a two-dimensional controller known as the Planar Pad. The Planar Pad facilitates the simultaneous manipulation of two assignable parameters, and can be controlled with the mouse, another controller, or the Vector Envelope. See Use the ES2 Planar Pad.
- *Effect section:* The built-in effect-processing options are found to the right of the output parameters. See ES2 integrated effects processor.
- *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 sound (and that of ES2-based GarageBand instruments). You can reassign MIDI control numbers for these parameters. See ES2 macros and controllers overview.

ES2 sound sources

ES2 oscillators overview in Logic Pro for Mac

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:* Click the number to the right of each oscillator to independently turn each on or off. 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:* Choose the waveform that an oscillator generates. The waveform is responsible for the basic tonal color. See ES2 basic oscillator waveforms.
- *Coarse Frequency knobs:* Set the oscillator 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 = 1/100 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 crossfade (set the level relationships) among the three oscillators. See Balance ES2 oscillator levels.

ES2 basic oscillator waveforms in Logic Pro for Mac

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). The table outlines 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).

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
- Noise for oscillator 3. See Use the ES2 noise generator.

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. Ring modulation principles are described in Use ES2 ring modulation.

Use ES2 pulse width modulation in Logic Pro for Mac

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 the oscillator 2 ring modulator) and oscillator 3.



Set a basic pulse width in oscillator 2 or 3

- In Logic Pro, 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. In Logic Pro, 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 ES2 frequency modulation in Logic Pro for Mac

The principle of frequency modulation (FM) synthesis was developed in the late 1960s and early 1970s by John Chowning. It was popularized by the Yamaha 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 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. In Logic Pro, 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 ES2 ring modulation in Logic Pro for Mac

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. In Logic Pro, 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 ES2 pulse width modulation).



Use ES2 Digiwaves in Logic Pro for Mac

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

- In Logic Pro, 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 in Logic Pro for Mac

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, the oscillator 3 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. In Logic Pro, 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.

Detune analog oscillators in ES2 in Logic Pro for Mac

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 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 drift 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.

Stretch tuning in ES2 in Logic Pro for Mac

The (coarse) Frequency knob of each oscillator lets you tune oscillators 1, 2, and 3 in semitones or octaves. The (fine tune) Frequency parameter lets you fine-tune each oscillator in cents (1/100 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.



Detuned oscillators periodically beat against each other at a certain frequency. CBD sets the beating frequency between low and high notes or retains a constant beating frequency across the keyboard range. Choose a CBD value to detune the harmonics/beating of low note frequencies in a ratio proportionate to 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 in Logic Pro for Mac

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 crossfade—set the level relationships—between the three oscillators. This is self-evident in use. If you move the pointer along one side of the Triangle, it crossfades 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 ES2 Vector Envelope.

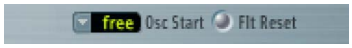
Modulate triangle coordinates with the modulation wheel

1. In Logic Pro, 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.

Set ES2 oscillator start points in Logic Pro for Mac

The oscillators can run freely or can begin at the same phase position of their respective waveform cycles each time a note is struck. This behavior is chosen in the Osc(illator) Start pop-up menu.

Choose “free,” “soft,” or “hard” to change the waveform phase start position.



- *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 in Logic Pro for Mac

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. In Logic Pro, 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

ES2 global parameters in Logic Pro for Mac

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. The Volume knob is at the lower right.



Global parameters

- *Keyboard Mode buttons:* Switch ES2 between polyphonic, monophonic, and legato behaviors. See Set the ES2 keyboard mode.
- *Unison button:* Turn unison mode on or off. See Use unison and voices in ES2.
- *Voices field:* Set the maximum number of notes that can be played simultaneously.
- *Glide knob:* 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.
- *Bend Range fields:* Define the upward and downward pitch bend range. See Set the ES2 pitch bend range.
- *Tune field:* 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 Detune analog oscillators in ES2.
- *Constant Beat Detuning pop-up menu:* Detuned oscillators periodically beat against each other at a certain frequency. Use CBD to set the beating frequency between low and high notes or to retain a constant beating.
- *Oscillator Start pop-up menu:* Choose “free,” "soft," or "hard" from the Osc(illator) Start pop-up menu. See Set ES2 oscillator start points.
- *Volume knob:* Set the overall instrument output level.

Set the ES2 keyboard mode in Logic Pro for Mac

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

- In Logic Pro, 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 in Logic Pro for Mac

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. In Logic Pro, 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 ES2 global parameters overview.
   - The intensity of detuning (voice deviation) is set with the Analog parameter. See Detune analog oscillators.

Use polyphonic unison mode

- In Logic Pro, 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 in Logic Pro for Mac

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

- In Logic Pro, 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 in Logic Pro for Mac

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



Set independent upward and downward bend ranges

- In Logic Pro, drag in either field to set a bend range.

Set an identical upward and downward bend range

1. In Logic Pro, 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.

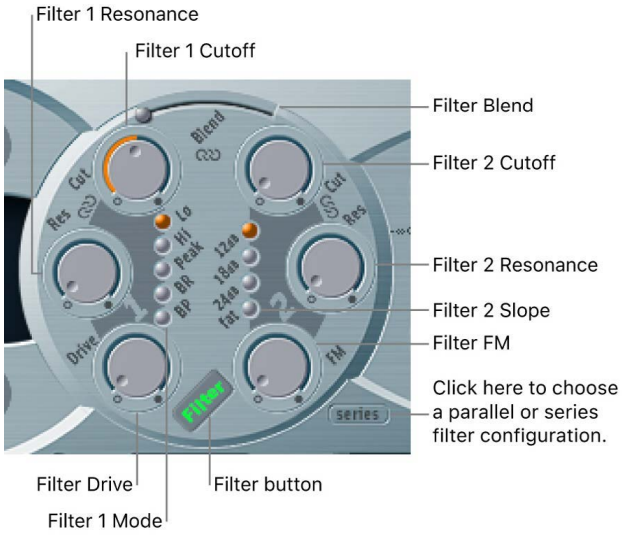
   Note: A downward bend of 4 semitones results in a combined bend range of 8 semitones—9 if you include the standard pitch, or “no bend” position.

ES2 filter parameters

ES2 filter overview in Logic Pro for Mac

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

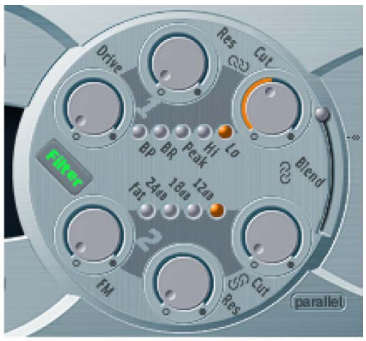
- *Filter button:* Turn the entire filter section on or off. Deactivating the filter section makes it easier to hear adjustments to other sound parameters, because the filters always heavily affect the sound. Turning off the filters also reduces processor load.
- *Filter Configuration button:* Switch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

ES2 filter configuration in Logic Pro for Mac

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.



Series filter signal flow



Parallel filter signal flow

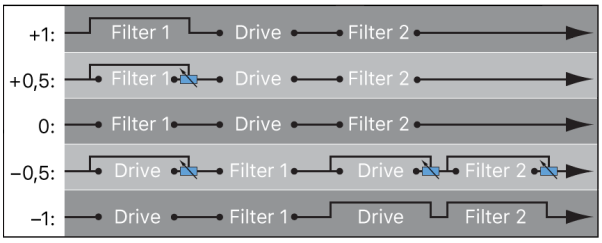
In the figure to the left, the filters are cabled in series. This means that the signal of all oscillators (combined at the Oscillator Mix Triangle) passes through the first filter, then this filtered signal passes through Filter 2, if Filter Blend is set to 0, the middle position. The output signal 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 Crossfade between ES2 filters.

Crossfade between ES2 filters in Logic Pro for Mac

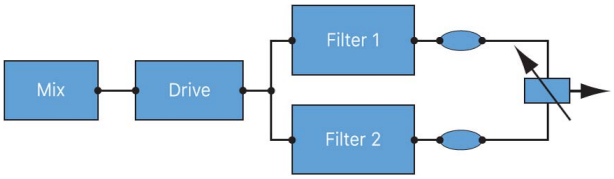
Filter Blend can have a significant effect on the ES2 signal flow. Regardless of whether parallel or series filter configurations are chosen, a Filter Blend setting of –1 results in only Filter 1 being audible. A Filter Blend setting of +1 limits audibility to Filter 2. The figures illustrate the signal 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.

Crossfade between ES2 filters

- In Logic Pro, drag the Filter Blend slider to crossfade 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 crossfaded. You hear the effect of both filters.

  You can also crossfade 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 in ES2

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

ES2 Filter 1 modes in Logic Pro for Mac

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):* Allow frequencies that fall below the cutoff frequency to pass. The slope of Filter 1 is fixed at 12 dB/octave.
- *Hi (highpass):* Allow 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 center frequency.

ES2 Filter 2 slopes in Logic Pro for Mac

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: 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: Adds 24 dB per octave of rejection. Fat mode has a built-in compensation circuit that retains the sound 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.



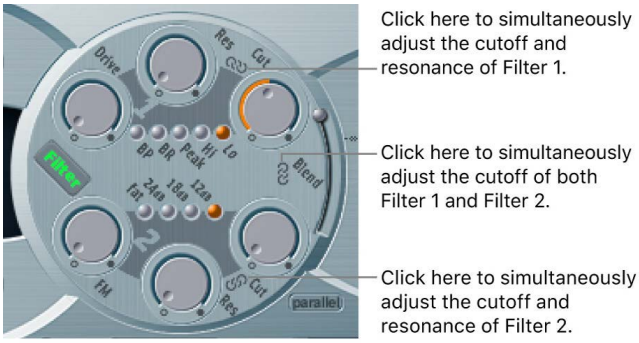
Cutoff and resonance parameters

- *Cutoff Frequency knob:* 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:* 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 ES2 filter parameters simultaneously

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

- In Logic Pro, 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 in Logic Pro for Mac

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 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. Turn on Filter Reset to provide 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

- In Logic Pro, 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 ES2 Fat(ness) parameter

- In Logic Pro, 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 in Logic Pro for Mac

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. This is because analog filters can behave uniquely when overdriven, forming an essential part of the sonic character of a synthesizer. 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 Crossfade 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 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 Filter 2 frequency in Logic Pro for Mac

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 Enhance ES2 sounds with Sine Level).



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 ES2 frequency modulation.

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 in ES2

1. In Logic Pro, 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 frequency, you can set up a second routing to modulate Oscillator 1 pitch.

2. Set up a modulation routing as follows: modulation target Pitch 1, source LFO1.
3. Adjust LFO settings.

ES2 amplifier parameters

Use the ES2 dynamic stage in Logic Pro for Mac

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.

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 ES2 dynamic stage (Amp)

1. In Logic Pro, 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.

Enhance ES2 sounds with Sine Level in Logic Pro for Mac

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.



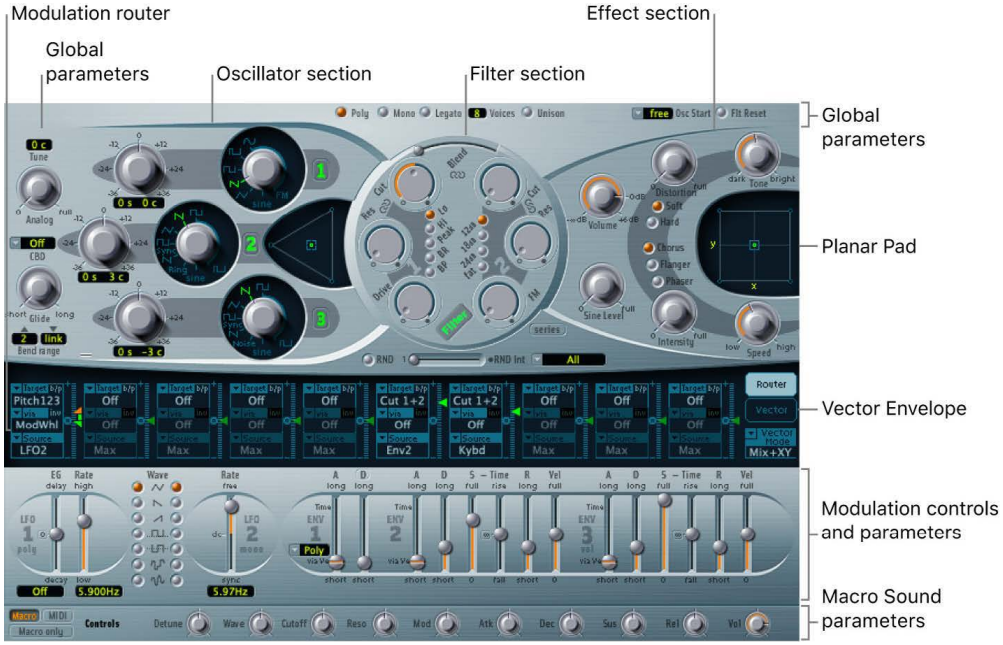
- In cases where oscillator 1 is frequency-modulated by oscillator 2 (if you have turned up FM with the waveform selector), *only* the pure sine wave is mixed into the dynamic section, not the distorted FM waveform.
- Any modulation of oscillator 1 pitch set in the router affects the frequency of the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

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.

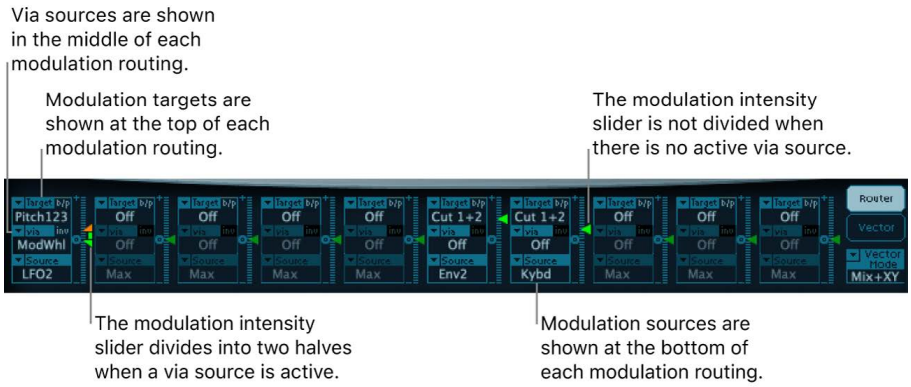


- *Modulation router:* The modulation router—or router, for short—links modulation sources, such as the envelope, to modulation targets, such as the oscillators and filters. The router features ten modulation routings, arranged into columns. See Use the ES2 modulation router.
- *Modulation sources:* The modulation sources include the LFOs and envelopes. See ES2 LFO overview and ES2 envelopes overview.
- *Vector Envelope:* The Vector Envelope is an extremely sophisticated, loop-capable, multipoint envelope that can control the Planar Pad and Triangle (oscillator mix parameter). Click the Vector Envelope button to view it in place of the modulation router. See Use the ES2 Vector Envelope.
- *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 ES2 Planar Pad.

ES2 modulation router

Use the ES2 modulation router in Logic Pro for Mac

The modulation router—or router—spans the center of the ES2 interface. Click the Router button to view the modulation router in place of the Vector Envelope. You can hide or show the router/Vector Envelope by clicking the disclosure arrow at the lower left of the section. If you're new to synthesizer modulation routings, see Modulation overview.



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

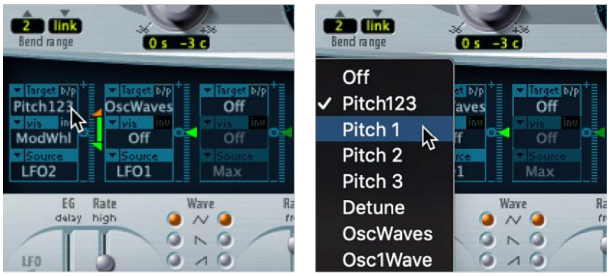
The modulation intensity—how strongly the target is influenced by the source—is set with the vertical slider to the right of the modulation routing.

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 in ES2

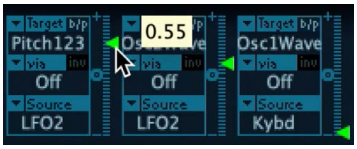
1. In Logic Pro, Choose the Target parameter you want to modulate.



2. Choose the Source parameter you want to use for modulation of the target.

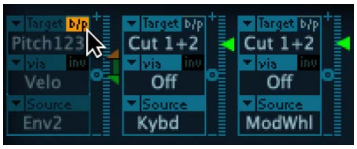


3. Drag the Intensity slider to set a fixed modulation value. When via is active, this slider sets the minimum modulation intensity.



Bypass a modulation routing in ES2

- In Logic Pro, click the Bypass (b/p) button at the top right of each modulation routing.



  Turn on or turn off individual modulation routings, without losing settings.

Copy the entire ES2 modulation matrix

You can copy the entire set of router channels and can paste these into another preset.

1. In Logic Pro, Control-click the Router button, then choose Copy Matrix from the pop-up menu.
2. Choose a new preset from the plug-in header menu.
3. Control-click the Router button, then choose Paste Matrix from the pop-up menu.

Clear the entire ES2 modulation matrix

- In Logic Pro, Control-click the Router button, then choose Clear Matrix from the pop-up menu.

  All router channels are initialized and turned off.

Control ES2 modulation intensity in Logic Pro for Mac

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.

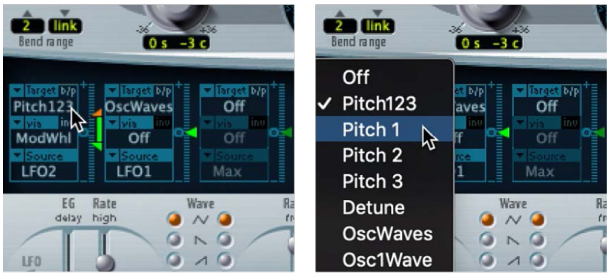
You can control modulation intensity by choosing an additional modulation source from the via pop-up menu.

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

- Move the upper half of the slider to set the maximum modulation intensity when the via controller is at 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 in ES2

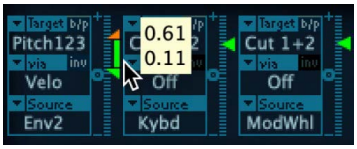
1. In Logic Pro, choose a modulation target from the Target pop-up menu.



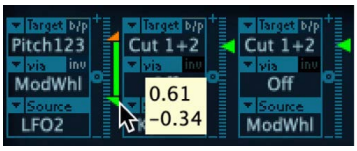
2. Choose a modulation source from the Source pop-up menu.
3. Choose the modulation source that you want to use for control of modulation intensity from the via pop-up menu.



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

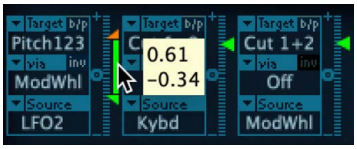


5. Vertically drag the lower arrowhead of the Intensity slider to set the minimum modulation intensity.



Move the entire via range in ES2

- In Logic Pro, vertically drag the range (the area between the two slider halves).

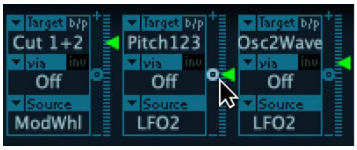


Both arrowheads move simultaneously.

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 in ES2

- In Logic Pro, click the zero symbol beside the via parameter.



Invert the effect of the via modulation source in ES2

- In Logic Pro, click the via invert (inv) parameter to the right of the via parameter.

ES2 LFOs

ES2 LFO overview in Logic Pro for Mac

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 are *not* 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 and field:* 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 off the LFO 1 envelope generator.
- *LFO 1 Rate slider and field:* 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.
- *LFO 2 Rate slider and field:* Set the frequency of LFO 2 modulation. LFO 2 can be synchronized with the Logic Pro tempo.

ES2 LFO waveforms in Logic Pro for Mac

Choose a waveform for LFO 1 or LFO 2 with the LFO Wave buttons. The table 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.

Waveform	Comments
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 in Logic Pro for Mac

ES2 LFOs can be used to create delayed modulations, free modulations, and modulations that are synchronized with the Logic Pro tempo.

Set the LFO 1 modulation fade time in ES2

- In Logic Pro, set a positive LFO 1 EG value to fade in the modulation.

  The higher the value, the longer the delay time.

- Set a negative LFO 1 EG value to fade out the modulation.

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

Set up a delayed vibrato in ES2

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

1. In Logic Pro, 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 in ES2

- In Logic Pro, 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 ES2 LFO 2 rate with the song tempo

- In Logic Pro, choose a value in the lower half of the LFO 2 Rate slider range to synchronize LFO 2 with the application tempo.

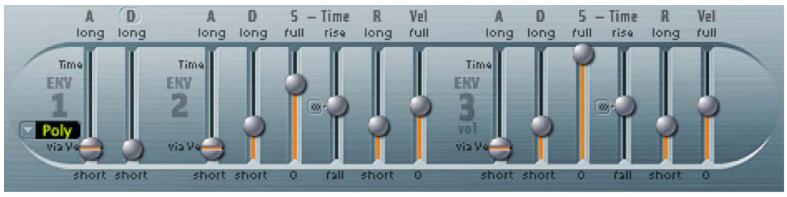
  The rate is displayed in rhythmic values (when project tempo synchronization is active).

  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 in Logic Pro for Mac

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 ES2 Vector Envelope.



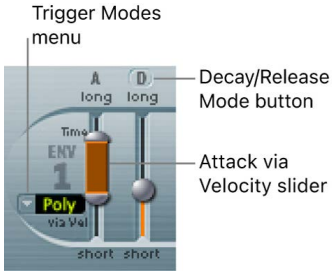
To learn more about the roots of the term “envelope generator” and its basic function, see Amplifier envelope overview.

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 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 in Logic Pro for Mac

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



ES2 Envelope 1 parameters

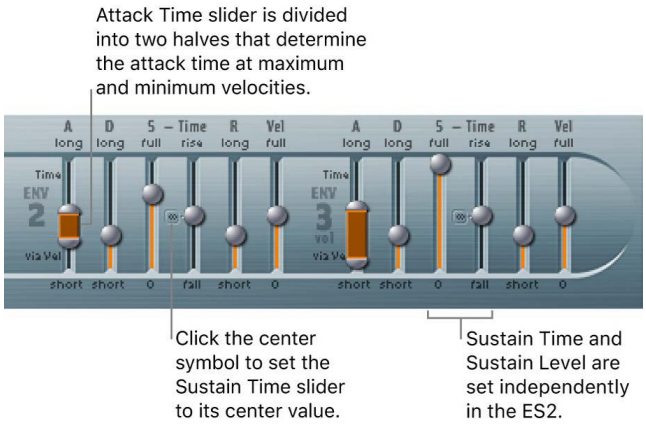
- *Trigger Modes pop-up menu:* Choose a mode to define the trigger behavior of ENV 1.
  - *Poly:* The envelope generator behaves as you would expect on any polyphonic synthesizer: every voice has its own envelope.
  - *Mono:* A single envelope generator modulates all voices in the same way. All notes 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. Move the lower slider to set 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:* Switch 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. Set the decay time with the Decay time slider. Define the time required for the level to decrease to zero with the Release time slider, once the key is released.
  - *In Attack/Release mode:* The envelope level remains at its maximum after the attack phase is over, while the key remains depressed.

ES2 Envelope 2 and 3 in Logic Pro for Mac

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.



ES2 Envelope 2 and 3 parameters

- *Attack slider:* Define the time required for the level of a note to rise from an 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:* Set 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/Sustain Time sliders:* Set the sustain level and the sustain time. See Use ES2 Envelope 2 and 3 sustain.
- *Release Time slider:* Define the time required for the sustain level to fall to zero, after the key is released.
- *Velocity Sensitivity slider:* Determine 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 in Logic Pro for Mac

When the Sustain Time (rise) slider is set to its center value, the Sustain (S) Level slider behaves like the sustain parameter of a standard 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 in ES2

1. In Logic Pro, 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.

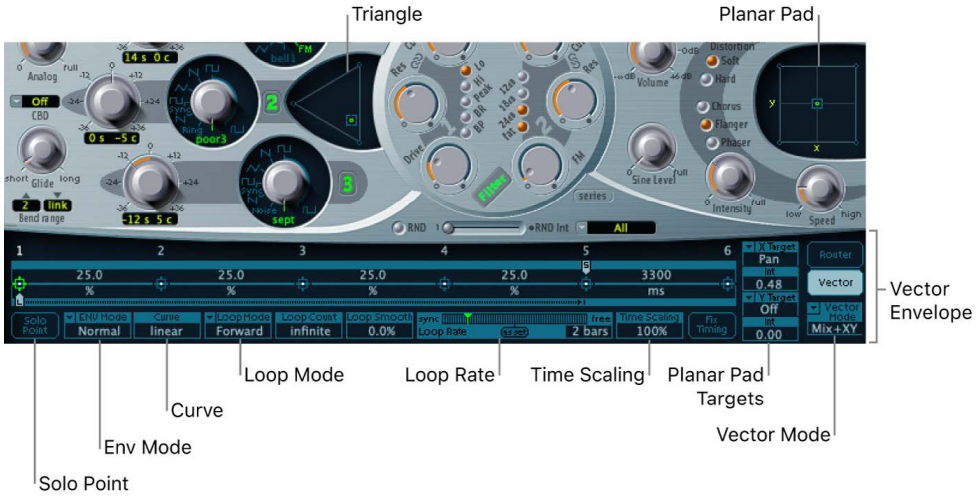
   This routing simulates the behavior of pianos and plucked instruments, where high notes decay faster than low notes.

Use the ES2 Vector Envelope in Logic Pro for Mac

The Vector Envelope is a multipoint, loop-capable modulation source. It provides 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 reveals how easy these features are to use. Combine these facilities with other synthesis options to create truly unique sounds that are—quite literally—moving.



Display the Vector Envelope

- In Logic Pro, click the Vector Envelope button to the right of the router to display the Vector Envelope.

Turn the Vector Envelope on or off

- In Logic Pro, turn off the Solo Point button to make the Vector Envelope active.
- Turn on the Solo Point button to deactivate the Vector Envelope.

  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

In Logic Pro, the Vector Mode pop-up menu is used to set a Vector Envelope target.

- Choose the Vector Envelope target—the Planar Pad, Triangle, or both.
  - *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

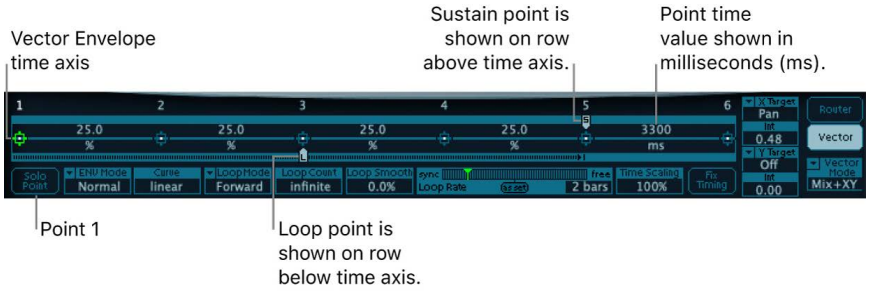
1. In Logic Pro, Control-click anywhere in the Vector Envelope to open a shortcut menu of commands and functions.
2. Choose any item in the menu to perform the operation.

   See ES2 Vector Envelope shortcut menu.

Vector Envelope points, times, and loops

Use ES2 Vector Envelope points in Logic Pro for Mac

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 no loop is 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 Vector Envelope point

- In Logic Pro, click the point to select it.

  Once selected, you can edit the point.

Create a new Vector Envelope point

- In Logic Pro, 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 Vector Envelope point

- In Logic Pro, Control-click the point.

Revert to the default value for a Vector Envelope point

In Logic Pro, 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.

ES2 Vector Envelope solo and sustain in Logic Pro for Mac

The Solo Point button turns the Vector Envelope on or off. If active, 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.

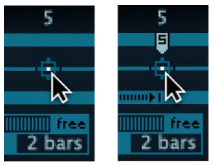
Note: You can independently turn off Vector Envelope modulation of the Planar Pad by setting Vector Mode to off. See Use the ES2 Vector Envelope.

Any point can be declared the Sustain point. Assuming that the played note is held long enough and no loop is 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 Vector Envelope point as the Sustain point

- In Logic Pro, 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 ES2 Vector Envelope loops in Logic Pro for Mac

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 with 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.

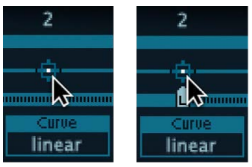
Any point can be declared the Loop point. Provided that the note is held for a suitable length of time, portions of the envelope can be repeated, or looped.

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 Vector Envelope point as the Loop point

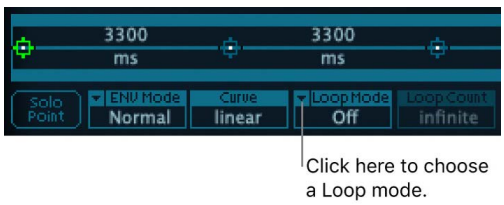
- In Logic Pro, click in the turquoise strip below the chosen point.

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



Set a Vector Envelope Loop mode

- In Logic Pro, Choose a Vector Envelope Loop mode.
  - *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 not available.
  - *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

In Logic Pro, 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).

  Set a defined speed for the Vector Envelope loop cycle. You can also synchronize the loop speed 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

- In Logic Pro, Drag to set a value for Loop Smooth. This evens out the transition from the Sustain point to the Loop point in Forward or Backward loop mode.
  - 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

- In Logic Pro, Set the Vector Envelope loop cycle to repeat a specified number of times. Following the defined repetitions, the Vector Envelope runs from the Sustain point onward. Possible values are 1 to 10 and “infinite.”



ES2 Vector Envelope release behavior in Logic Pro for Mac

Set one of two release phase options in the Env Mode menu: Normal or 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.

ES2 Vector Envelope point transitions in Logic Pro for Mac

Drag vertically in the Curve field to define the shape of transitions from point to point. Choose from nine convex and nine concave shapes. You can also choose “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 ES2 Vector Envelope times in Logic Pro for Mac

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 Vector Envelope time value

- In Logic Pro, drag the numerical value vertically.



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

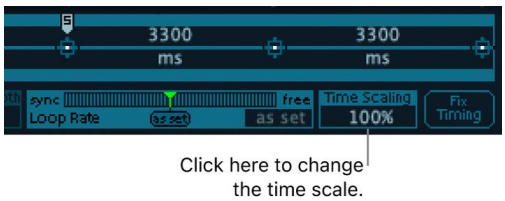
Set a time value without affecting the absolute time positions of later Vector Envelope points

- In Logic Pro, 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.

ES2 Vector Envelope time scaling in Logic Pro for Mac

You can stretch and compress the entire Vector Envelope, which is faster than modifying each point. To double the Vector Envelope speed, set Time Scaling to 50%. Set to 200% to halve the speed.



- 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 Vector Envelope time scaling and the loop rate with Fix Timing

- In Logic Pro, 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.

ES2 Vector Envelope shortcut menu in Logic Pro for Mac

Control-click anywhere in the Vector Envelope to open a shortcut menu that contains a number of commands that can accelerate your workflow.

Vector Envelope shortcut menu commands

- *Delete Selected Point:* Delete the currently selected point.
- *Insert Point to the Right of Selected Point:* Create a point to the right of the currently selected point. A new point is created and is automatically selected.

  In cases where you create a new point between two existing points, the segment that previously existed is divided into two equal-sized segments. 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 control point positions in the Triangle and the Planar Pad are fixed, thus ensuring that newly created points don't affect any previously defined movements.
- *Copy Point:* Copy the currently selected point to the Clipboard.
- *Paste Point:* Paste the copied point from the Clipboard to the currently selected point.
- *Paste to All Points:* Paste the copied point from the Clipboard to all points.

  This command is useful for duplicating segment times and XY positions.
- *Set Segments to Average Time:* Change the segment time values of all points to a value that is averaged over the entire Vector Envelope length.
- *Init to 8/8 Loop:* Create 8 new points. The entire timeline is divided into segments of equal length.

  WARNING: In cases where you have previously created several points, use of this command will overwrite your existing segments.
- *Init to 16/16 Loop:* Create 16 new points. The entire timeline is divided into segments of equal length.

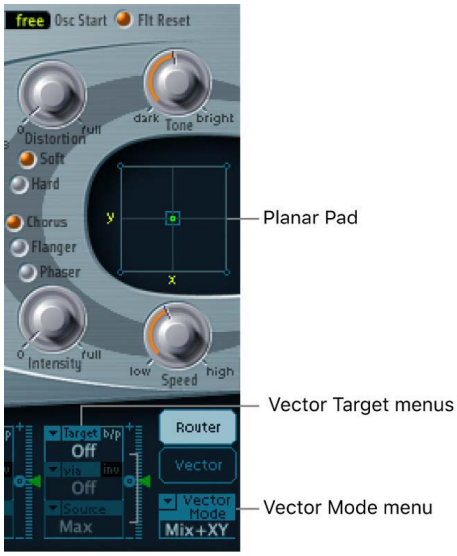
  WARNING: In cases where you have previously created several points, use of this command will overwrite your existing segments.
- *Clear Envelope:* Reset the Vector Envelope.

  Note: Only three points will remain when this command is used: Start, Sustain, End.
- *Copy Envelope:* Copy the entire Vector Envelope to the Clipboard.
- *Paste Envelope:* Paste the copied Vector Envelope from the Clipboard.

Use the ES2 Planar Pad in Logic Pro for Mac

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, 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 and Control ES2 modulation intensity.

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

Set the ES2 modulation intensity

- In Logic Pro, 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 in Logic Pro for Mac

The table outlines 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).
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 ES2 pulse width modulation, Use ES2 frequency modulation, Use the ES2 noise generator, and Use ES2 Digiwaves.
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.

Target	Comments
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 Enhance ES2 sounds with Sine Level.
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 level.
Osc2Levl	(Osc 2 Level) allows modulation of oscillator 2 level.
Osc3Levl	(Osc 3 Level) allows modulation of oscillator 3 level.

ES2 filter modulation targets in Logic Pro for Mac

The table outlines all filter-related modulation targets.

Target	Comments
Cutoff 1	Modulates the Cutoff Frequency parameter of Filter 1. See Filter cutoff and resonance overview.
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 Filter 2 frequency.
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*.

Target	Comments
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 Filter 1 cutoff frequency is rising, the cutoff of Filter 2 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 bandwidth.
Filter Blend	(FltBlend) modulates the Filter Blend parameter. See Crossfade between ES2 filters.

Other ES2 modulation targets in Logic Pro for Mac

The table outlines 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 panorama 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) lets you 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 in Logic Pro for Mac

All modulation targets outlined in this table 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 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* ENV2 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 in Logic Pro for Mac

The table outlines all modulation sources.

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 ES2 Planar Pad and Use the ES2 Vector Envelope.
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 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 macros and controllers overview.
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.

Source	Comment
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 in Logic Pro for Mac

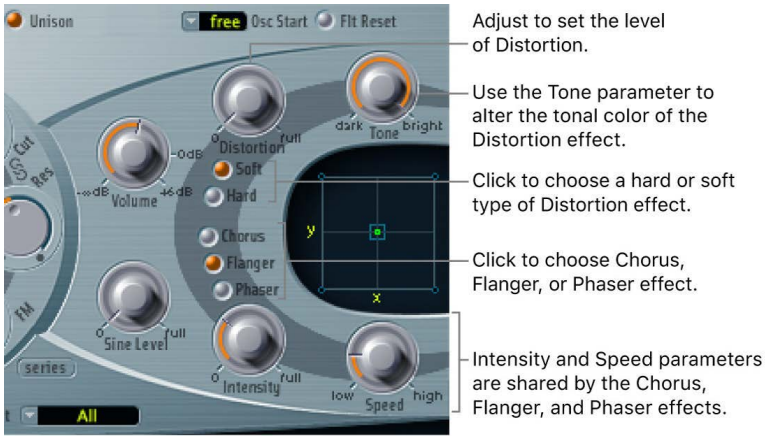
The outlined sources can be used to control the modulation intensity.

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 modwheel.
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 in Logic Pro for Mac

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 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.

ES2 Distortion effect parameters

- *Soft button:* Activate the Distortion effect Soft mode. The distortion circuit sounds somewhat like a tube overdrive.
- *Hard button:* Activate the Distortion effect Hard mode. The distortion effect sounds like a fully transistorized fuzz box.
- *Distortion knob:* Set the amount of distortion. Turn this knob to zero to effectively turn off the effect.
- *Tone knob:* Control the treble portion of the distortion signal.

ES2 Chorus, Flanger, or Phaser effect parameters

- When Chorus is on:
  - *Intensity knob:* Set the depth of the effect—how rich the modulation is. Turn this knob to zero to turn off the effect.
  - *Speed knob:* 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 macros and controllers in Logic Pro for Mac

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



- *Macro button:* Show a number of macro controls that affect groups of other parameters.
- *MIDI button:* Assign MIDI controllers to particular modulation routings. See ES2 via modulation source reference.
- *Macro only button:* Replace the ES2 interface with a smaller view that is limited to the macro controls.

Use ES2 macro controls in Logic Pro for Mac

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

When 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.

ES2 controller assignments in Logic Pro for Mac

The Controller Assignments area lets you assign MIDI keyboard 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 in ES2

1. In Logic Pro, click the MIDI button in the lower-left corner to display the Controller Assignments.
2. Choose the controller name or number that you want to use from any Ctrl A to Ctrl F pop-up menu.

Learn a MIDI controller assignment in ES2

1. In Logic Pro, 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 Logic Pro for Mac

In surround ES2 instances, additional parameters are shown in the slide out Extended Parameters section at the bottom of the plug-in window.



ES2 Surround parameters

- *Surround Range slider and field:* Drag to set the range of the surround angle (0 to 360 degrees). In other words, 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 slider and field:* Drag to set 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 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.
- *Legacy Surr(ound) Level Mode checkbox:* Select to compensate for changes in surround level handling in projects created with earlier Logic Pro versions, if required. This parameter should be turned off for new projects.

ES2 extended parameters in Logic Pro for Mac

ES2 provides additional parameters that are accessed by clicking the disclosure arrow 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. Per-voice channels support pitchbend, aftertouch, modwheel, and Ctrl A-F assignment messages. See Set ES2 controller assignments. Controllers and MIDI messages sent on the base channel affect all voices.

- *Mono Mode Bend Range slider:* Set a value from 0 to 96.

  The chosen pitch bend range affects individual note pitchbend messages received on all but the common base channel. The default is 48 semitones, which is compatible with the GarageBand for iOS 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

ES2 randomization parameters in Logic Pro for Mac

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



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 in ES2

- In Logic Pro, 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 in ES2

- In Logic Pro, move the Random Intensity slider farther to the right.

Create several slight variations of the current setting in ES2

- In Logic Pro, reload the original setting after each random alteration, saving each with a new name.

Restrict ES2 randomization in Logic Pro for Mac

You can restrict randomization to particular groups of parameters with 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 in Logic Pro for Mac

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 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 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 in Logic Pro for Mac

"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 supplied 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.

In Logic Pro, 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 spectrum.

In many supplied settings, Unison mode is active. If your computer can't cope with the processing demands, switch off Unison mode and insert an Ensemble effect in a bus, for use with several plug-ins. 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.

In Logic Pro, 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 in Logic Pro for Mac

Not every sound needs to consist of several oscillators. There are many 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 ES2

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.

In Logic Pro, 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 ES2

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.

In Logic Pro, 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 Logic Pro for Mac

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 in ES2

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.

In Logic Pro, 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 will hear a broad spectrum of FM sounds. Some settings may 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 can 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.

In Logic Pro, 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 in ES2

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.

In Logic Pro, 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 ES2

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.

In Logic Pro, do the following:

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

Create FM sounds with wavetables in ES2

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 Logic Pro tempo—here, two bars.

In Logic Pro, 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 in ES2

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!

In Logic Pro, 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 in ES2

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.

In Logic Pro, 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 in Logic Pro for Mac

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

Set up a basic PWM sound in ES2

- In Logic Pro, 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 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 basic PWM sound fatter in ES2

- In Logic Pro, 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 in Logic Pro for Mac

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 the oscillator 2 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 in ES2

In Logic Pro, 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 in Logic Pro for Mac

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 in ES2

In Logic Pro, 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 ES2 in Logic Pro for Mac

This tutorial provides hints for Vector Envelope programming.

Familiarize yourself with the ES2 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.

- In Logic Pro, 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 the oscillator 1 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 the oscillator 1 sawtooth wave, following the release of the key.

Vector synthesis with the ES2 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.

In Logic Pro, do the following:

- Switch on Solo Point, to more easily listen to the settings for the single points.
- Click point 1 to hear only the oscillator 1 sawtooth wave.
- Drag the pointer in the Planar Pad to the far left, which results in a low cutoff frequency for oscillator 2.

- Click Point 2 to hear only the oscillator 2 rectangular wave.
- Drag the pointer in the Planar Pad all the way down, which results in the rightmost panorama position.
- Click Point 3 to hear only the oscillator 3 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 in ES2

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 the oscillator 2 wavetable.
- Oscillator 2 outputs crossfaded 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.

In Logic Pro, 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 in ES2

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.

In Logic Pro, 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 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 in ES2

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.

In Logic Pro, 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 in Logic Pro for Mac

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 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 in Logic Pro for Mac

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. This might be a useful template for emulations of fretted instruments, harpsichords, clavinets, and so on.

Take a look at the sound 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 to attain 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 window—or all wavetables together, where it makes sense.

Set Envelope 1 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 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 aftertouch. Feel free to change the wheel and aftertouch 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 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 in Logic Pro for Mac

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 modwheel 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 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 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. If you change Pitch 2 maximum to a lower minus value, remember to set Pitch 3 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 pitch and pan movements. Just exchange it for LFO 1 on modulation routings 2 and 3. Note that there is no modulation source for the Leslie acceleration, so you need to use it in a static way by fading it in. Alternatively, you 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 in Logic Pro for Mac

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 the oscillator 2 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, aftertouch, 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 the oscillator 1 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. This configuration is set up in modulation routing 3 (oscillator 3 Wave moved by Envelope 1 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 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 aftertouch—which closes the filters—has been programmed. This allows you to play with an additional decrescendo, remotely controlled by aftertouch. Get a feel for the way the patch responds. You’ll find that it offers quite a few controls for expression—velocity, aftertouch after note-on, and aftertouch 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 in Logic Pro for Mac

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).

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. You can simultaneously scroll through the oscillator 1 and oscillator 3 wavetables, and change oscillator 2 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 can 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 FM parameter, which you can redraw (set a negative modulation amount for modulation routing 4 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 aftertouch 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 in Logic Pro for Mac

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 pitch or tuning, the overall pitch of the sound goes out of tune or is 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 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).

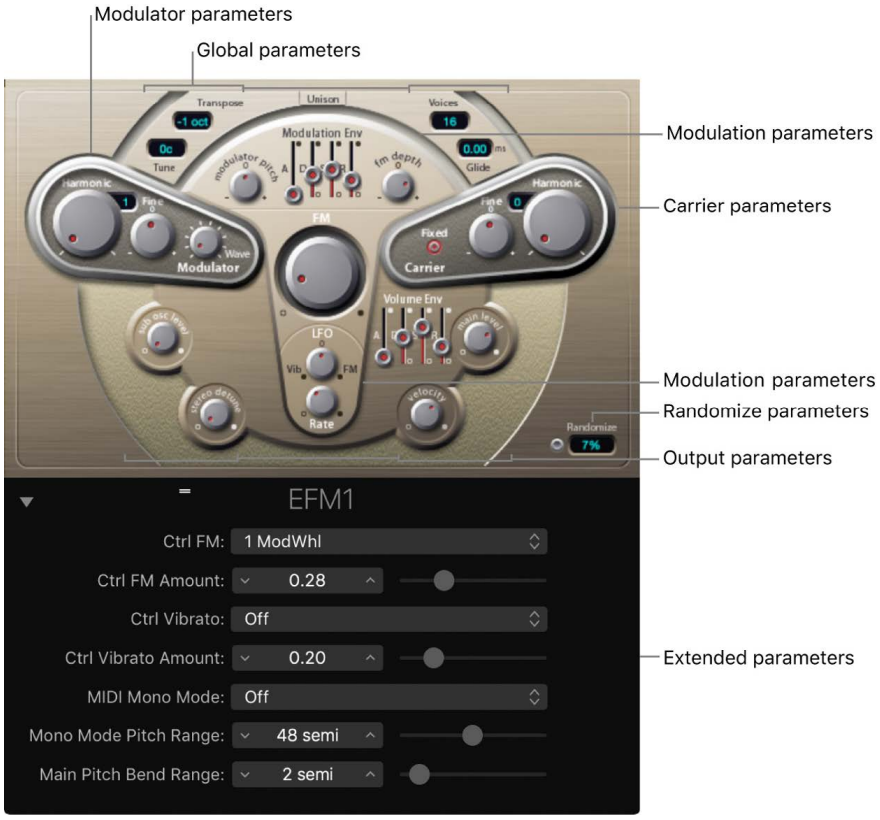
EFM1

EFM1 overview in Logic Pro for Mac

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, which will introduce you to the terminology and give you an overview of various synthesis systems and how they work.

If you're new to using plug-ins in Logic Pro, see Add, remove, move, and copy plug-ins.



EFM1 is divided into several areas.

- *Global parameters:* The top section contains parameters that set the overall tuning of EFM1. Further controls let you set the Glide (portamento) time, limit the number of voices, and thicken the sound with Unison. See EFM1 global parameters.
- *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 EFM1 Modulator and carrier overview.
- *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.
- *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.
- *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.
- *Extended parameters:* Accessed by clicking the disclosure arrow 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.

EFM1 modulator and carrier parameters

EFM1 Modulator and carrier in Logic Pro for Mac

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:* Set the tuning ratio between the modulator (left) and carrier (right) oscillators. See Set the EFM1 tuning ratio.
- *Fine knob:* 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:* Choose a different waveform for the modulator oscillator. See Choose an EFM1 modulator waveform.
- *FM (Intensity) knob:* Set the amount of carrier oscillator frequency modulation by the modulator oscillator. Adjust the FM knob to increase the intensity and number of overtones, 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:* Set the tuning ratio between the modulator and carrier oscillators. See Set the EFM1 tuning ratio.
- *Fine knob:* 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:* Disconnect the carrier frequency from keyboard, pitch bend, and LFO modulations, resulting in a carrier tone that is free of these modulation sources.

Set the EFM1 tuning ratio in Logic Pro for Mac

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 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 EFM1 (see EFM1 global parameters).

Experiment with basic tuning ratios

In Logic Pro, 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 an EFM1 modulator waveform in Logic Pro for Mac

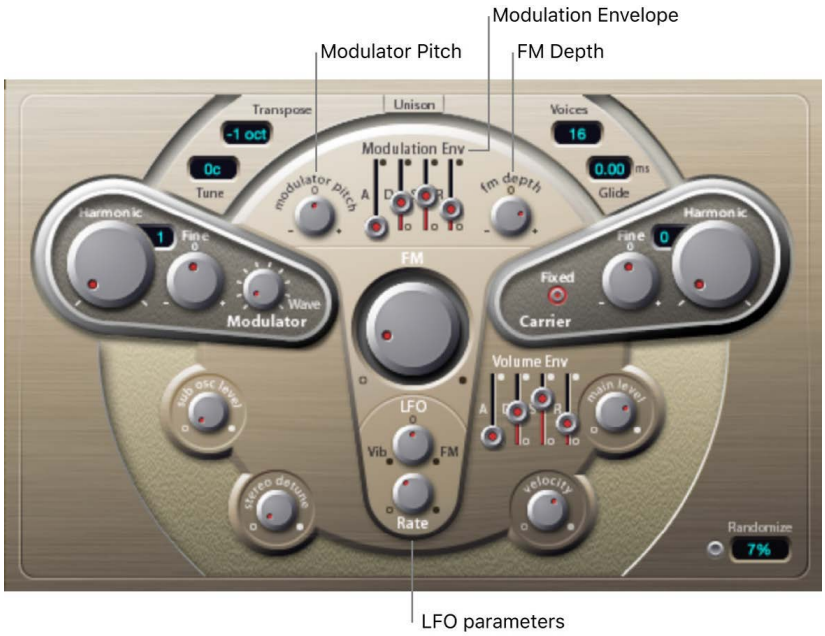
In classic FM synthesis, sine waves are used as modulator and carrier waveforms. The EFM1 modulator oscillator provides a number of additional digital waveforms, which extend its sonic capabilities significantly. These waveforms contain additional harmonics that add a new level of richness to the resulting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

Choose a different waveform

- In Logic Pro, 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 controls in Logic Pro for Mac

FM synthesis is, at its core, caused by the intensity and type of modulations that take place in the signal path. 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 Envelope sliders:* Control both the FM (Intensity) and Modulator pitch parameters over time. The envelope is triggered every time a MIDI note is received.
  - *Attack slider:* Set the time it takes to reach the maximum envelope level.
  - *Decay slider:* Set the time it takes to reach the sustain level.
  - *Sustain slider:* Set a level that is held until the MIDI note is released.
  - *Release slider:* Set the time it takes to reach a level of 0, after the MIDI note has been released.
- *Modulator Pitch knob:* 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:* 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 knob:* 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:* Set the speed of the LFO.

EFM1 global parameters in Logic Pro for Mac

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



Global parameters

- *Transpose pop-up menu:* Choose the base pitch. You can transpose by semitones or octaves.
- *Tune field:* Fine-tune the pitch in cents. One cent is 1/100 of a semitone.
- *Voices pop-up menu:* Choose the number of simultaneously playable voices. Choose from: 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:* Layer two complete voices, thus making the sound richer. EFM1 can be played with up to eight-voice polyphony when in unison mode.
- *Glide field:* Introduce a continuous pitch bend between two consecutively played notes. Adjust the value to set the time required 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 in Logic Pro for Mac

EFM1 provides the following level controls.



Output parameters

- *Sub Osc Level knob:* 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. Increase the Sub Osc Level control to mix the sub-oscillator sine wave with the FM engine output.
- *Stereo Detune knob:* 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 Envelope:* Shapes the level of the sound over time. The volume envelope is triggered every time a MIDI note is received.
  - *Attack slider:* Set the time it takes to reach the maximum volume level.
  - *Decay slider:* Set the time it takes to reach the sustain level.
  - *Sustain slider:* Set a level that is held until the MIDI note is released.
  - *Release slider:* Set the time required to reach a level of zero, after the MIDI note has been released.
- *Main Level knob:* Set the overall output level.
- *Velocity knob:* Determine the sensitivity to incoming MIDI velocity messages. EFM1 dynamically reacts to MIDI velocity—harder playing results 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 in Logic Pro for Mac

The Randomize feature generates new sounds by randomly altering a number of key 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:* Create a new sound by randomizing multiple parameters.
- *Randomize field:* Determine the amount of randomization—variance from the original sound.

Create a random sound

- In Logic Pro EFM1, 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

- In Logic Pro EFM1, 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 in Logic Pro for Mac

Click the disclosure arrow 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 Bend Range slider:* Set a range from 0 to 96 semitones.

  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 the GarageBand for iOS 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.
- *Main Pitch Bend Range slider:* Set a range from 0 to 24 semitones.

Assign EFM1 MIDI controllers in Logic Pro for Mac

EFM1 allows remote control 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. In Logic Pro, click the disclosure arrow at the lower left to view the extended parameters.
2. Choose the controller name or number from the Ctrl FM or Ctrl Vibrato pop-up menu.
3. Set the FM or vibrato amount using the slider below the pop-up menu.

ES E

ES E overview in Logic Pro for Mac

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, which will introduce you to the terminology and give you an overview of different synthesis methods and how they work.

If you're new to using plug-ins in Logic Pro, see Add, remove, move, and copy plug-ins.



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.
- *LFO parameters:* The LFO parameters (below the Wave knob) are used to modulate the sound. See ES E LFO parameters.
- *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.
- *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.

- *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.
- *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 ES E extended parameters.

ES E oscillator parameters in Logic Pro for Mac

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



Oscillator parameters

- *Wave knob:* 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 in Logic Pro for Mac

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 pulse width—pulse width modulation (PWM).



LFO parameters

- *Vib/PWM knob:* Define the intensity of LFO modulation.
- *Speed knob:* 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. Set the Wave parameter to the 12 o'clock position (50% rectangular) for pulse width to attain the maximum modulation range.

ES E filter parameters in Logic Pro for Mac

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



Filter parameters

- *Cutoff knob:* Control the cutoff frequency of the filter.
- *Resonance knob:* 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:* 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).
- *Velo Filter knob:* 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 in Logic Pro for Mac

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:* Set the time required for the signal to reach the initial signal level, known as the sustain level.
- *Release slider:* Set the time it takes for the signal to fall from the sustain level to a level of zero.

ES E output parameters in Logic Pro for Mac

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



Output Parameters

- *Volume knob:* Set the overall output level.
- *Velo Volume knob:* 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 and II/Ensemble buttons:* Turn 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.

ES E extended parameters in Logic Pro for Mac

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

Extended parameters

- *Bend Up slider:* 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.
- *Bend Down slider:* Set the negative—downward—pitch bend range in semitone steps, by up to 2 octaves—a value of 24.
- *Tune slider:* Tune the instrument sound in cents. A cent is 1/100 of a semitone.

ES M

ES M overview in Logic Pro for Mac

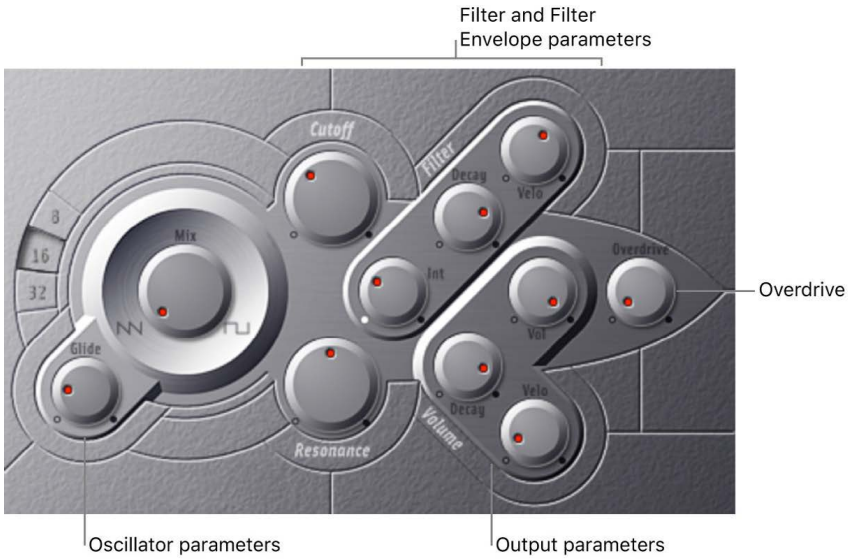
The monophonic ES M (ES Mono) synthesizer is a good starting point if you're looking for bass sounds that punch through your mix.

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, which will introduce you to the terminology and give you an overview of different synthesis methods and how they work.

If you're new to using plug-ins in Logic Pro, see Add, remove, move, and copy plug-ins.



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.
- *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.
- *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 envelope and output controls.
- *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 ES M extended parameters.

ES M oscillators in Logic Pro for Mac

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



Oscillator parameters

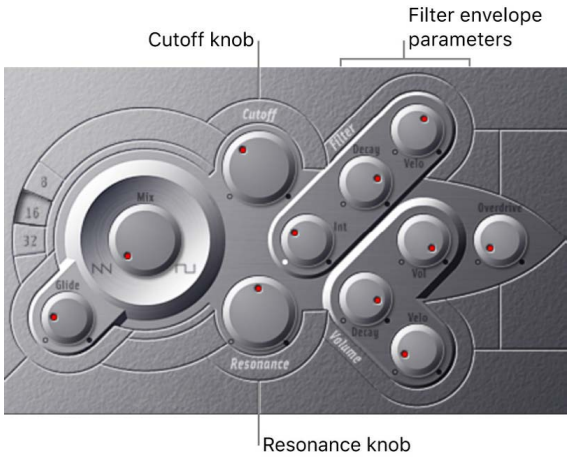
- *Mix knob:* Set the waveform of the oscillator, which is responsible for the basic color of the tone.
  - Setting the Wave parameter all the way to the left causes the oscillator to output sawtooth signals.
  - Setting the Wave parameter all the way to the right outputs a 50% rectangular wave, which 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:* Introduce a continuous pitch bend between two consecutively played notes. Adjust the value to set the time required 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 in Logic Pro for Mac

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

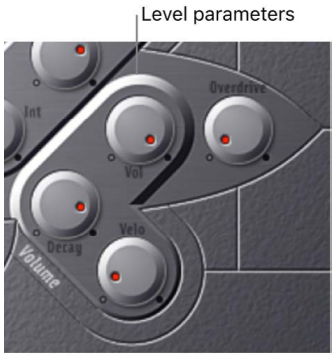
- *Cutoff knob:* Set the cutoff frequency of the filter. Its slope is 24 dB/octave.
- *Resonance knob:* Boost or cut portions of the signal that surround the frequency defined by the cutoff parameter.

  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:* Define the amount (depth) of cutoff frequency modulation applied by the envelope generator.
- *Decay knob:* Set the decay time of the filter envelope.
- *Velo knob:* 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 envelope and output controls in Logic Pro for Mac

The ES M output stage offers the following parameters.



Envelope and output parameters

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

ES M extended parameters in Logic Pro for Mac

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

Extended parameters

- *Pitch Bend Up slider:* Set the upward pitch bend range in semitone steps. This lets you use the pitch bend controller of your keyboard to bend the ES M pitch.
- *Pitch Bend Down slider:* Set the downward pitch bend range in semitone steps by up to 2 octaves (a value of 24).
- *Tune slider:* Tune the instrument in cents. One cent is 1/100 of a semitone.

ES P

ES P overview in Logic Pro for Mac

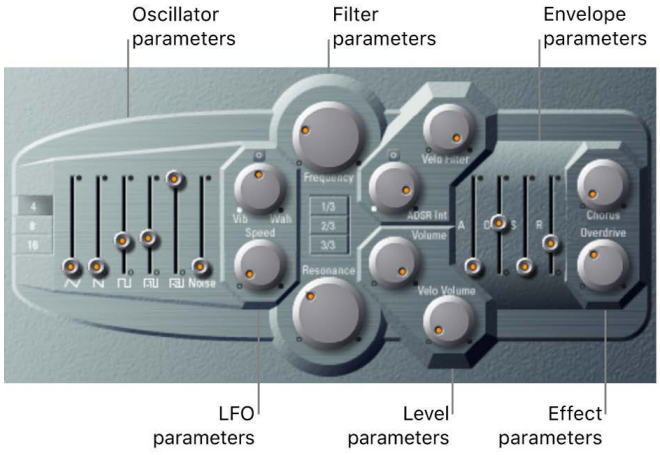
The eight-voice ES P (ES Poly) emulates classic polyphonic synthesizers of the 1980s.

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, which will introduce you to the terminology and give you an overview of different synthesis methods and how they work.

If you're new to using plug-ins in Logic Pro, see Add, remove, move, and copy plug-ins.



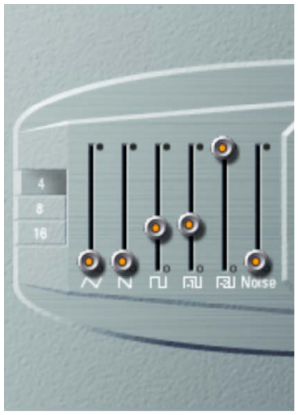
ES P is divided into several areas.

- *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.
- *LFO parameters:* The LFO parameters (to the right of the oscillator parameters) are used to modulate the sound. See ES P LFO parameters.
- *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 filter parameters.

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

ES P oscillator parameters in Logic Pro for Mac

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:* Set the level of the triangle waveform output by the oscillators.
- *Sawtooth oscillator slider:* Set the level of the sawtooth waveform output by the oscillators.
- *Rectangle oscillator slider:* Set the level of the rectangle waveform output by the oscillators. The pulse width is fixed at 50%.
- *Sub-oscillator -1 octave slider:* Set the level of the (rectangular) sub-oscillator waveform, which is one octave lower than the main oscillators. Use this to thicken the sound. The pulse width is fixed at 50%.
- *Sub-oscillator - 2 octaves slider:* Set the level of the (rectangular) sub-oscillator waveform, which is two octaves lower than the main oscillators. Use this to thicken the sound. The pulse width is fixed at 50%.

- *Noise generator slider:* Set the level of white noise. This is the raw material for classic synthesizer sound effects, such as waves, wind, and helicopters.
- *4, 8, and 16 buttons:* Set 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 in Logic Pro for Mac

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:* Set the rate of the vibrato or cutoff frequency modulation.

ES P filter parameters in Logic Pro for Mac

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



Filter parameters

- *Frequency knob:* Set the cutoff frequency of the lowpass filter.
- *Resonance knob:* 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 buttons:* The cutoff frequency can be modulated by MIDI note number (keyboard position); you may know this parameter as *keyboard follow* on other synthesizers. 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:* Define the amount (depth) of cutoff frequency modulation applied by the envelope generator. (See ES P envelope and level controls.)
- *Velo Filter knob:* 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 in Logic Pro for Mac

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:* Set the time required for the signal to reach the initial signal level (the sustain level).
- *Decay slider:* Set the time it takes for the signal to fall from the attack level to the sustain level.
- *Sustain slider:* Set the signal level (the sustain level).
- *Release slider:* Set the time it takes for the signal to fall from the sustain level to a level of zero.
- *Volume knob:* Set the overall output level.
- *Velo Volume knob:* Set the amount (depth) of velocity sensitivity to incoming MIDI note events. At 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 in Logic Pro for Mac

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



ES P effect parameters

- *Chorus knob:* Set the intensity (depth) of the integrated chorus effect.
- *Overdrive knob:* Set the overdrive/distortion level of the synthesizer 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.

ES P extended parameters in Logic Pro for Mac

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

- *Bend Up slider:* Set the 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.
- *Bend Down slider:* Set the downward pitch bend range in semitone steps, by up to 2 octaves (a value of 24).
- *Tune slider:* Tune the instrument in cents. One cent is 1/100 of a semitone.

EVOC 20 PolySynth

EVOC 20 PS and vocoding

EVOC 20 PS overview in Logic Pro for Mac

EVOC 20 PS 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 PS “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 PS 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 PS, 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.

If you’re new to using plug-ins in Logic Pro for Mac, see Add, remove, move, and copy plug-ins.

Set up EVOC 20 PS Logic Pro

1. In Logic Pro, insert EVOC 20 PS into the Instrument slot of an instrument channel strip.
2. Choose an input source from the Analysis Input pop-up menu in the plug-in header. This can be an audio track, live input, or bus, depending on the host application.

   EVOC 20 PS 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 PS 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 synthesizer 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.

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.

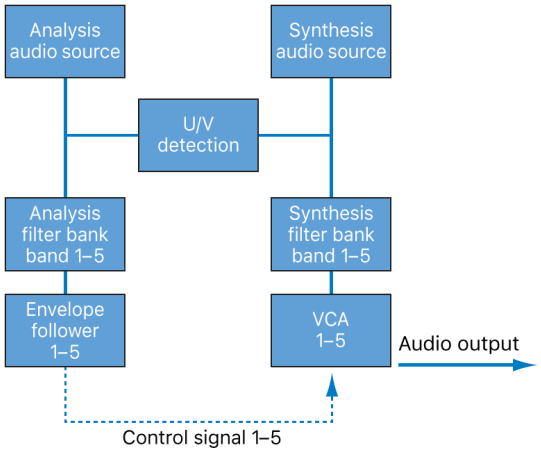
The speech *analyzer* and *synthesizer* features of a vocoder are two bandpass *filter banks*. Bandpass filters allow a frequency band—a slice in the overall frequency spectrum—to pass through unchanged. Frequencies that fall outside the band are cut.

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 is 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.

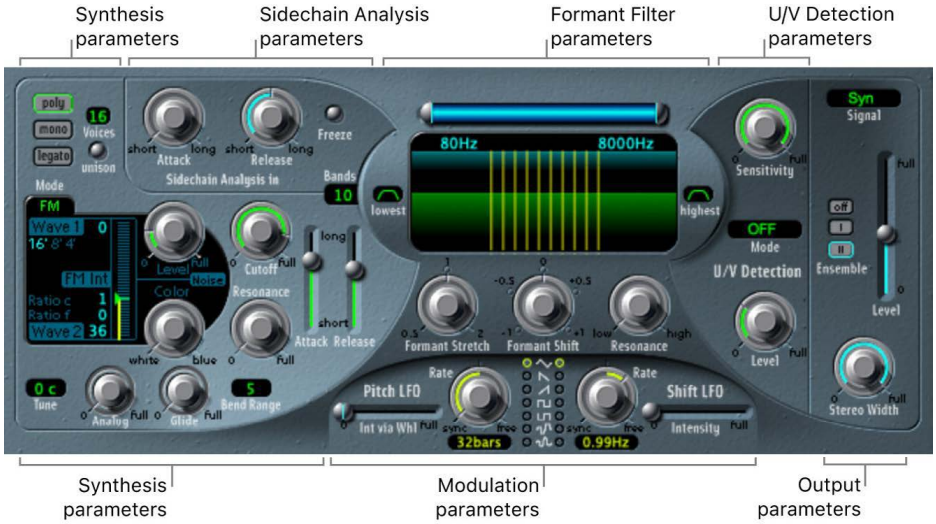
These control signals are then sent to the synthesis filter bank—where they control the levels of the corresponding synthesis filter bands. This is done with voltage-controlled amplifiers (VCAs) in analog vocoders. Volume changes to the bands in the analysis filter bank are imposed on the matching bands 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 character is reproduced by the synthesis filter bank. EVOC 20 PS provides up to 20 bands per bank. See EVOC 20 block diagram for a detailed image of the EVOC 20 PS signal path.

EVOC 20 PS interface in Logic Pro for Mac

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



- *Sidechain Analysis parameters:* Determine how the input signal is analyzed and used by the EVOC 20 PS. See EVOC 20 PS analysis controls.
- *U/V Detection parameters:* Detect the unvoiced portions of the sound in the analysis signal, improving speech intelligibility. See EVOC 20 PS (U/V) detection.
- *Synthesis parameters:* Control the polyphonic synthesizer of the EVOC 20 PS. See EVOC 20 PS synthesis overview.

- *Formant Filter parameters:* Configure the analysis and synthesis filter banks. See EVOC 20 PS formant filter.
- *Modulation parameters:* Modulate the synthesizer and filter banks—through two LFOs. See EVOC 20 PS modulation.
- *Output parameters:* Configure the output signal of the EVOC 20 PS. See EVOC 20 PS output parameters.

EVOC 20 PS analysis controls in Logic Pro for Mac

The parameters in the Sidechain Analysis section control how EVOC 20 PS 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:* 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:* 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 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 active, the analysis filter bank ignores the input source, and the Attack and Release knobs have no effect.
- *Bands field:* Drag to set the number of frequency bands (up to 20) 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.
- In Logic Pro, click the Freeze button to hold, or sustain, the sound spectrum of the analysis input signal.



Set the number of filter bank bands

- In Logic Pro, drag vertically on the Bands field to set the number of frequency bands used by the EVOC 20 PS filter bank.



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 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 PS (U/V) detection in Logic Pro for Mac

Human speech consists of a series of voiced sounds—tonal sounds or formants—and unvoiced (U/V) 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 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.

EVOC 20 PS 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 allows 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:* Determine how responsive U/V detection is. Use higher settings to recognize more of the individual, unvoiced portions of the input signal. 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 Mode pop-up menu.
- *Mode pop-up menu:* Choose the sound source used to replace the unvoiced content in 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.

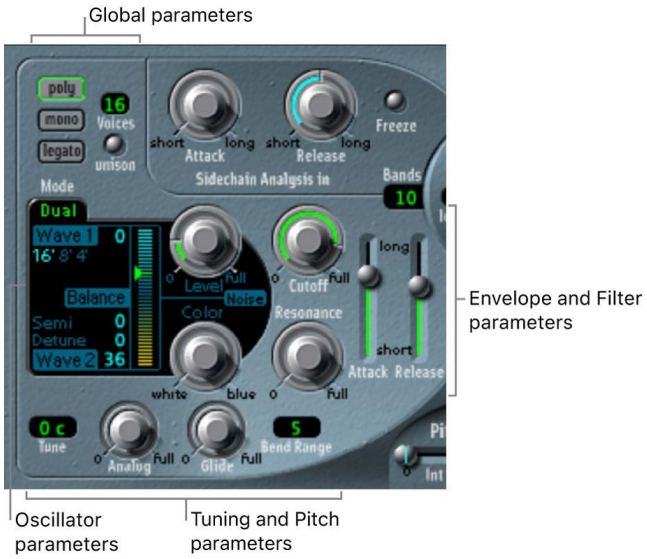
- *U/V Level knob:* Set the volume of the signal used to replace the unvoiced content in the input signal.

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

EVOC 20 PS synthesis parameters

EVOC 20 PS synthesis overview in Logic Pro for Mac

EVOC 20 PS is equipped with a polyphonic synthesizer that accepts MIDI note input. Synthesis section parameters are outlined below.



Synthesis parameters

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

EVOC 20 PS oscillator parameters

EVOC 20 PS oscillators overview in Logic Pro for Mac

You can switch EVOC 20 PS oscillators between dual mode and FM mode. The Synthesis section also incorporates a noise generator that can add a further color to your sound.



- *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).

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

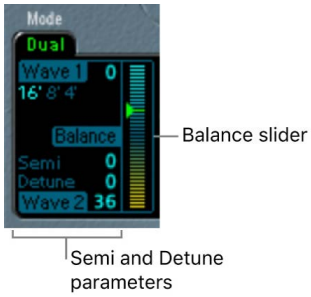
Both Dual and FM mode provide the following common parameters.

Common oscillator parameters

- *16′, 8′, 4′ value buttons:* 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 PS oscillator dual mode in Logic Pro for Mac

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

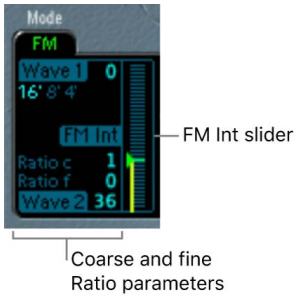


Dual mode oscillator parameters

- *Semi field:* Adjust the tuning of oscillator 2 in semitone steps.
- *Detune field:* Fine-tune both oscillators in cents. One hundred cents equals one semitone step.
- *Balance slider:* Set the level balance between the two oscillator signals.

EVOC 20 PS oscillator FM mode in Logic Pro for Mac

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



FM mode oscillator parameters

- *Ratio coarse field:* Drag to adjust the frequency ratio between oscillator 2 and oscillator 1 in semitone steps.
- *Ratio fine 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 PS noise generator in Logic Pro for Mac

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 PS (U/V) detection.

Noise generator parameters

- *Level knob:* Control the amount of noise added to the signals of the two oscillators.
- *Color knob:* 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 PS tuning and pitch in Logic Pro for Mac

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



Tuning and pitch parameters

- *Tune field:* Set the overall tuning in cents. One hundred cents equals one semitone step.
- *Analog knob:* 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:* Determine the time it takes for the pitch to slide from one note to another—portamento. Also see EVOC 20 PS global parameters for information about mono and legato mode.
- *Bend Range field:* Determine the pitch bend modulation range, in semitone steps.

EVOC 20 PS filters in Logic Pro for Mac

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



Filter parameters

- *Cutoff knob:* Set the cutoff frequency of the lowpass filter. Turn to the left to remove high-frequency content from the synthesizer signal.
- *Resonance knob:* 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 PS envelopes in Logic Pro for Mac

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



Envelope parameters

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

EVOC 20 PS global parameters in Logic Pro for Mac

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



Global parameters

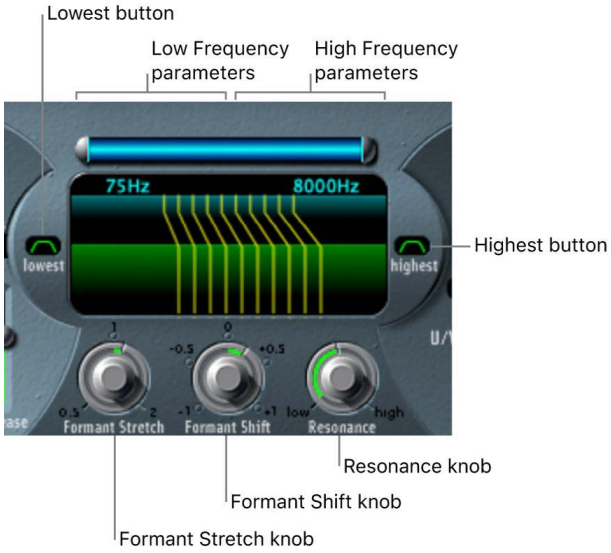
- *Poly/Mono/Legato buttons:* Determine the keyboard mode.
  - When Poly is on, you can set the maximum number of voices in the Voices field. (When Mono or Legato is on, a single voice is heard.)
  - When Mono is on, Glide is always active and the envelopes are retriggered by every note played (multi 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 PS tuning and pitch controls.
- *Voices field:* Set the maximum number of voices in the numeric field (only when Poly is turned on).
- *Unison button:* 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 PS output parameters.

EVOC 20 PS formant filter in Logic Pro for Mac

EVOC 20 PS 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/High Frequency parameters:* 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:* Determine 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:* Determine 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:* 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:* 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:* 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 PS modulation in Logic Pro for Mac

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

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



Modulation parameters

- *Int via Whl slider:* 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 and fields:* 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:* 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:* Define the amount of formant shift modulation by the Shift LFO.

EVOC 20 PS output controls in Logic Pro for Mac

The Output section controls the type of signal, stereo width, and level of signal that is sent from EVOC 20 PS. The Output section also has an integrated ensemble effect processor.



Output parameters

- *Signal pop-up menu:* Choose the signal that is sent to the 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:* Set the overall volume of the output signal.
- *Stereo Width knob:* 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 PS performance tips

EVOC 20 PS level and frequency tips in Logic Pro for Mac

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 can 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.

Avoid sonic artifacts in EVOC 20 PS in Logic Pro for Mac

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 rises. 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 Noise Gate plug-in, use Threshold to define the level above which the gate opens, and use Hysteresis to define a lower Threshold level below which the gate closes. The Hysteresis value is relative to the Threshold level.



Unwanted triggering by low or high frequency noise is avoided by the dedicated sidechain filters of the Noise Gate plug-in.

Enhance speech intelligibility in Logic Pro for Mac EVOC 20 PS

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 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 can 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.

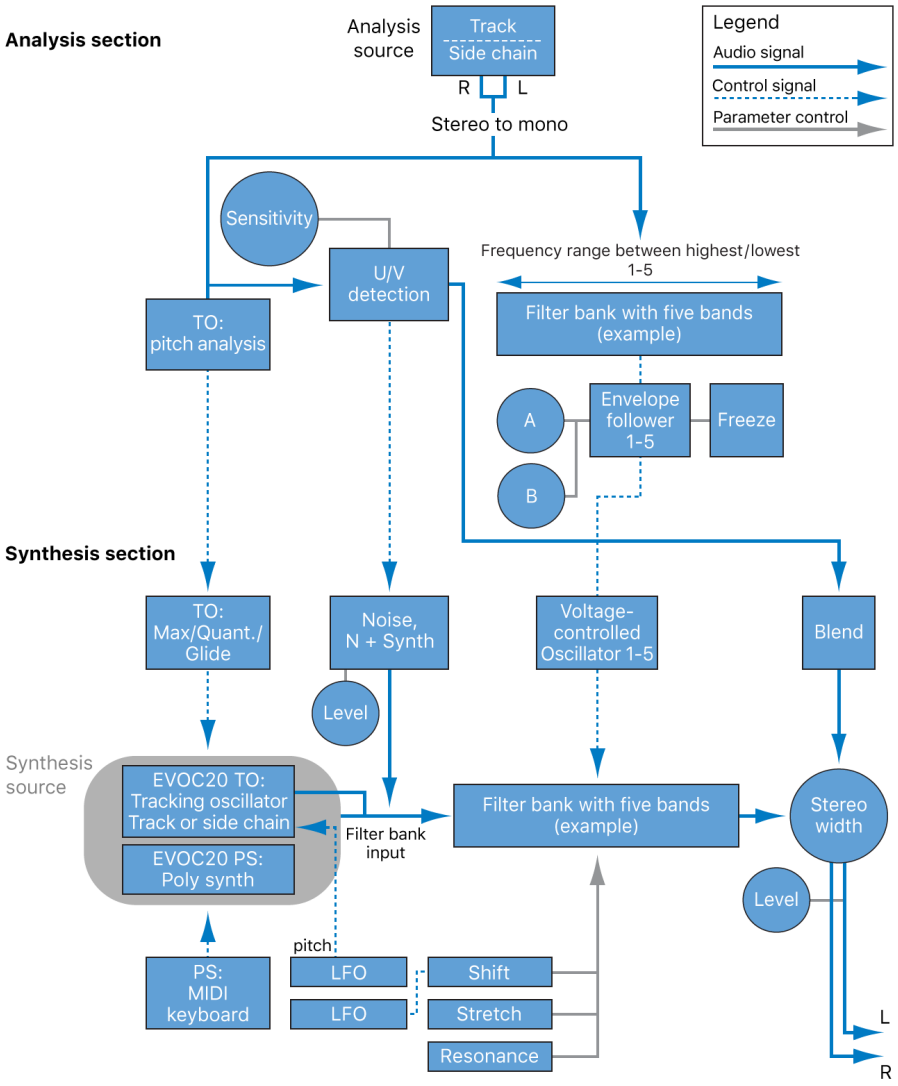
From 1980 to the present, EMS in the UK, Synton in Holland, and PAiA in the USA have been—and remain—the main flyers of the vocoding flag.

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 for Logic Pro for Mac

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



Retro Synth

Retro Synth overview in Logic Pro for Mac

Retro Synth is a flexible 16-voice synthesizer that can produce a wide variety of sounds.

Retro Synth provides four different synthesizer engines—Analog, Sync, Wavetable, and FM. 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.

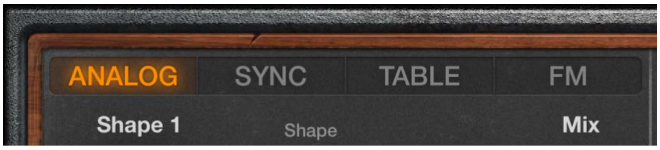
If you're new to synthesizers, it might be best to start off with Synthesizer basics overview, which will introduce you to the fundamentals and terminology of different synthesis systems.

If you're new to using plug-ins in Logic Pro, see Add, remove, move, and copy plug-ins.

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

- In Logic Pro, click the name 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.
  - *Sync:* Use for aggressive synthesizer sounds, particularly leads and basses. See Retro Synth Sync oscillator.
  - *Table:* Use for clean digital synthesizer sounds, such as pads and basses, and evolving effect sounds. See Retro Synth Table oscillator.
  - *FM:* Use for classic digital synthesizer sounds. Of note are bells, electric piano, clavinet, and spiky bass sounds. See Retro Synth FM oscillator.

Your choice of synthesizer engine changes the controls available. Most of these changes are seen in the Oscillator section.

Retro Synth Analog oscillator in Logic Pro for Mac

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 filter controls, amp and effect controls, modulation, and 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 2 knobs:* 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.
- *Shape Modulation knob:* Choose a waveform shape modulation source, and set the modulation intensity. The centered (off) position prevents all waveform shape modulation with the LFO or filter envelope.
- *Vibrato knob:* Set the amount of vibrato (pitch modulation).
- *Semitones knob:* Set the pitch of oscillator 2 in semitone steps, over a range of ±2 octaves.
- *Cents knob:* Precisely adjust the frequency of oscillator 2 in cents (1 cent = 1/100 semitone).
- *Mix slider:* Crossfade (set the level relationships) between the oscillators (Shape 1 and 2).

Retro Synth Sync oscillator in Logic Pro for Mac

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 filter controls, amp and effect controls, modulation, and 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:* Choose a sync modulation source, and set the modulation intensity. The centered (off) position prevents all waveform modulation with the LFO or filter envelope.
- *Vibrato knob:* Rotate to set the amount of vibrato (pitch modulation).
- *Sync knob:* 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

Retro Synth Table oscillator in Logic Pro for Mac

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 filter controls, amp and effect controls, modulation, and 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. Despite the clean tone, wavetable synthesis can also sound warm when combined with the right filter type.

Retro Synth also allows you to create your own custom wavetables.



Table oscillator parameters

- *Shape 1 and Shape 2 knobs:* Rotate to choose from 100 waveforms, known as Digiwaves. Digiwaves are very short samples of the attack transients of various sounds and instruments. You can also choose the waves in custom wavetables.
- *Wavetable pop-up menu:* Choose a wavetable, reverse the loaded wavetable, or create, save, and delete custom wavetables. See Use the Wavetable menu.
- *Shape/Formant button:* Switch between shape and formant modes.
  - *Shape mode:* Choose waveforms from default or custom wavetables with the Shape knobs.
  - *Formant mode:* Use the Shape Modulation knob to stretch or compress the formant spectrum—a series of fixed frequency peaks—in the active waveform (chosen with the Shape knobs). This is similar to grain compression in a granular synth.
- *Shape Modulation knob:* In shape mode, choose a modulation source and set the modulation intensity. In formant mode, stretch or compress the formant spectrum of the active wave. The centered (off) position prevents all waveform shape modulation with the LFO or filter envelope.

  This parameter is also available as the Wave Variation (Formant) real-time modulation target. See Retro Synth global and controller settings.
- *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 = 1/100 semitone).
- *Mix slider:* Move to crossfade (set the level relationships) between the oscillators (Shape 1 and Shape 2).

Use the Retro Synth Wavetable menu in Logic Pro for Mac

The Retro Synth Wavetable menu lets you create and manage custom wavetables.

Wavetable menu

- *Create Wavetable from Audiofile:* Opens a file browser where you can choose an audio file for use as a custom wavetable.
- *Save Wavetable as:* Opens a file browser where you can name and save your custom wavetable.
- *Delete Selected User Wavetable:* Removes the loaded user wavetable.
- *Custom Wavetable:* Loads the custom wavetable in memory. You can switch between this and a supplied wavetable.
- *App Wavetables pop-up menu:* Load a supplied wavetable. You can switch between a supplied wavetable and the custom wavetable.
- *Reverse Wavetable Order:* Reverses playback of the loaded wavetable. For example, a wavetable with 10 waveforms would sequentially play waveform 10 through to 1 when reversed.

Create a custom wavetable in Retro Synth

You can create a custom wavetable from any audio file that contains one or more sections of constant pitch. Where multiple sections are used, a section of silence should delineate them.

Timbral changes within sections are automatically detected with an adjustable parameter, resulting in a wavetable entry for each timbral shift. You should limit the speed of timbral changes within sections as very fast shifts can deliver unpredictable results.

If possible, use multiple audio sections with a constant pitch to create your source audio file. A file with a constant pitch does not require the insertion of silence between sections.

1. In Retro Synth in Logic Pro, open Table mode and choose Create Wavetable from Audiofile from the Wavetable pop-up menu.

   Note: You can also drag a suitable audio file onto the Shape 1 knob.

2. In the File browser, choose the audio file.
3. Click the disclosure arrow at the lower left of the plug-in window to show the extended parameters, then choose an item from the Audio File Analysis pop-up menu.

   This parameter changes the sensitivity of detection within contiguous audio sections. Choose the algorithm that works best for the audio material.

4. Choose “Save Wavetable as” from the Wavetable pop-up menu, then name your new wavetable in the File browser.

   The new wavetable name appears in the Wavetable pop-up menu. Choose the name to load it.

Retro Synth FM oscillator in Logic Pro for Mac

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 filter controls, amp and effect controls, modulation, and 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 the Retro Synth 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 synthesis is noted for synthetic brass, bell-like, electric piano, and spiky bass sounds.



FM oscillator parameters

- *Vibrato knob:* Rotate to set the amount of vibrato (pitch modulation).
- *Modulation knob:* Choose a modulation source (LFO or Filter Envelope), and set the modulation intensity. This modulates the target chosen with the FM/Harmonic switch.
- *FM (Amount) slider:* The carrier waveform is a simple sine wave. Drag to adjust the level of this basic tone.
- *FM/Harmonic target switch:* Choose a modulation target—FM (Amount), Harmonic, or both—for 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 right switch position lets you use the LFO or Filter Envelope to modulate the Harmonic content.
- *Harmonic/Inharmonic sliders:* 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. This parameter is also available as the Wave Variation (FM Harmonic) real-time modulation target. See global and controller settings.

  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.
- *Shape slider:* 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:* Crossfade (set the level relationships) between the modulator and carrier oscillators.

Retro Synth filter in Logic Pro for Mac

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 one of the available filter types and 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 modulation section controls. See modulation and global and controller settings.



Filter parameters

- *On/off button:* Turn the filter section on or off. The filter is turned on by default (indicated by the highlighted button at the top left of the filter section). Turn off 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. This changes the sonic character and behavior of other filter parameters. There are eight lowpass filters with different slopes, four highpass, four bandpass, a band reject, and a peak filter available. Use the descriptive names—Creamy, Edgy, Gritty, Lush, Lush (Fat), and Sharp—to make a choice that's right for your sound. See Retro Synth filter types.
  - *LP (lowpass):* Allows frequencies that fall below the cutoff frequency to pass. You can choose one of four slopes from the eight models 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. There are three slopes to choose from: 6, 12, and 24 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 either 6 or 12 dB/octave on each side of the 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. Band reject has a slope of 6 dB/octave.
    - *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 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 vertically to boost or cut signal portions above, below, or surrounding the cutoff frequency.
- *Key (Follow) slider:* 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/Drive knob:* Depending on the chosen filter type, set the intensity of filter cutoff modulation with the Osc1 sine wave generator or increase filter input gain. When the Filter FM label is shown, 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 prevents filter frequency modulation by the Oscillator 1 sine wave generator. When the Drive label is shown, you can alter the signal level at the filter input. This is typically used to overdrive the signal, which distorts the waveform and changes the filter response, leading to more aggressive tones.

  Note: The Oscillator 1 sine wave generator always generates a sine signal at the frequency of Oscillator 1.
- *LFO knob:* 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:* 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 filter types in Logic Pro for Mac

Retro Synth provides multiple filter types.

Retro Synth filter types: lowpass, bandpass, highpass

A lowpass (LP) filter passes the portion of a signal below a specified cutoff frequency and rolls off the portion above that frequency. A bandpass (BP) filter passes the portion of a signal occupying a band surrounding the cutoff frequency and rolls off the portions above and below that band. A highpass (HP) filter passes the portion of a signal above a specified cutoff frequency and rolls off the portion below that frequency.

There are multiple two-pole, four-pole, multi-pole state-variable and analog-modeled LP, BP, and HP filter designs in Retro Synth, each with distinctive characteristics that you may prefer for a given purpose. The available LP, BP, and HP filter designs include Creamy, Edgy, Gritty, Lush, Lush (Fat), and Sharp variants.

- Creamy: Multi-pole state-variable filter.
- Edgy: Two- or four-pole analog-modeled filters. Four-pole variants provide a steeper roll-off of frequencies beyond the cutoff.
- Gritty: Two-pole filters designed to saturate heavily at higher Resonance and Drive settings.
- Lush: Four-pole lowpass analog-modeled filters.
- Lush (Fat): Four-pole analog-modeled filters.
- Sharp: Two-pole analog-modeled filters.

The three principal filter controls have standard functions for all filter types.

- *Cutoff:* Controls the filter cutoff frequency.
- *Resonance:* Controls the filter resonance or emphasis. Higher settings boost frequencies in the immediate vicinity of the cutoff frequency.
- *Drive:* Allows the filter to be overdriven; the precise effect varies with each filter design.

Retro Synth filter types: band reject and peaking

A band reject filter cuts a narrow band around a resonant frequency. The remainder of the signal is affected minimally.

- *Cutoff:* Sets the frequency of the cut band.
- *Amount:* Controls the amount of attenuation.

A peaking filter boosts a narrow band around a resonant frequency. The remainder of the signal is affected minimally.

- *Cutoff:* Sets the frequency of the boosted band.
- *Gain:* Controls the amount of boost. Higher values are generally the most effective.
- *BW:* Sets the bandwidth around the cutoff frequency.

Retro Synth amp and effect controls in Logic Pro for Mac

Retro Synth 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 the integrated Chorus effect or can 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:* Set the overall output level.
- *Sine Level knob:* Mix a sine wave at the frequency of oscillator 1 (Shape 1) directly into the output stage. This sine signal is not processed by the filter.

Effect parameters

- *On/off button:* Turn 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 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:* Set the balance between the original and effect signals. High values result in stronger effect processing.
- *Rate knob:* Set the modulation speed of the effect.

Tip: Use Logic Pro 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 in Logic Pro for Mac

Retro Synth 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 the sound level over time. The Filter Envelope controls the filter over time. See Retro Synth envelopes.

The 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 can also use your keyboard modulation wheel, aftertouch, and velocity as real time control sources. See Retro Synth global and controller settings.

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. In Logic Pro, 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 pane is visible) and Filter Env sections. See LFO and Vibrato and 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 modulation wheel to change the LFO speed. If you don't want to use your keyboard 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 sensitivity of envelope modulation by velocity.

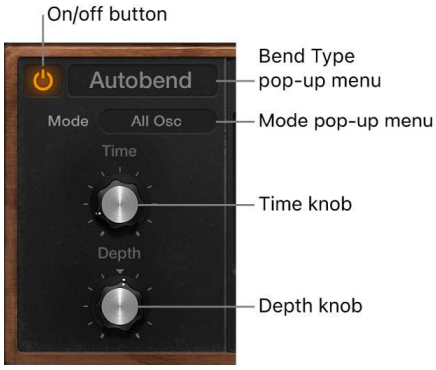
Use vibrato to modulate oscillator pitch

1. In Logic Pro, rotate the Vibrato knob to set the amount of vibrato (pitch modulation).
2. Click the Vibrato tab if the LFO pane 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 Logic Pro tempo. See 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 modulation wheel to change the vibrato speed. If you don't want to use your keyboard modulation wheel, click the Via pop-up menu to assign a different MIDI controller.

Retro Synth Glide and Autobend in Logic Pro for Mac

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 Polyphony control. See Retro Synth global and controller settings.



Glide/Autobend parameters

- *On/off button:* Turn the Glide/Autobend section on or off.
- *Bend Type pop-up menu:* Choose Glide or Autobend mode.
- *Mode pop-up menu:* Choose the parameter that you want to bend. Choose from: Oscillator 1 + Sine, Oscillator 2, All Oscillators, Opposed (with one oscillator bending up, while the other bends down by an equal amount), and Oscillators + Filter.
- *Time knob:* 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) Set the pitch bend range (over a range of ±3 octaves).

Retro Synth LFO and Vibrato in Logic Pro for Mac

The Retro Synth 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

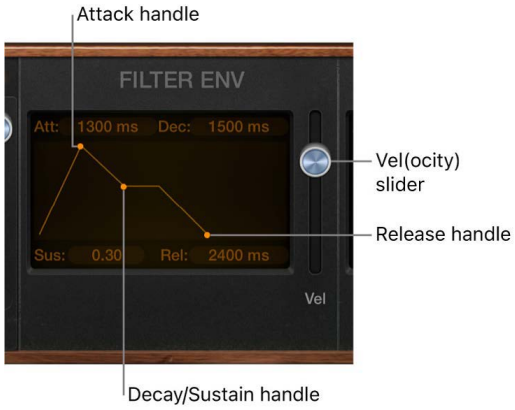
- *LFO/Vibrato tabs:* Change between LFO and Vibrato parameter panes.
- *Waveform display:* Click the buttons at the top of the display to independently choose an LFO or Vibrato waveform.
  - The sawtooth waves are suitable for bubbling, rhythmic effects.
  - The triangle wave is suitable for vibrato 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 switch:* Turn on to synchronize the modulation speed with the host tempo. Turn off to control the modulation speed manually.
- *Rate slider:* Independently set the maximum LFO or Vibrato speed.
- *Source pop-up menu:* Choose your MIDI keyboard Mod Wheel, Aftertouch, or Modwheel + Aftertouch as the control source for the LFO or Vibrato output level.

Retro Synth envelopes in Logic Pro for Mac

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 envelopes.



Envelope parameters

- *Attack handle:* Drag horizontally to set the time it takes for the envelope to reach the initial level.

  Note: Oscillators automatically switch from free mode to synced mode when the Attack time is set to values below 0.50 ms. This has a pronounced effect on stacked voices, in particular. See global and controller settings.

- *Decay handle:* Drag horizontally to set the time it takes for the envelope to fall to the sustain level, following the initial attack time.
- *Sustain handle:* 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 for the envelope to fall from the sustain level to a level of zero.
- *Velocity slider:* Determine how sensitive the envelope is to incoming velocity.
  - If the Vel slider is set to zero, the envelope outputs its maximum level when you strike the keys at any velocity.
  - At a Vel slider value of 100%, the entire dynamic range is under velocity control.

    To explain, raising the slider value reduces the envelope minimum amplitude, with the difference being dynamically controlled by keyboard velocity. For example, when you set the Vel slider to 25%, the minimum envelope amplitude is reduced to 75%. The remaining 25% is added in response to the velocities of keys you play. So, a key played with a zero velocity results in an envelope amplitude of 75%. A key played with a MIDI velocity value of 127 will result in an envelope amplitude of 100%. When you raise the Vel slider value, the minimum amplitude decreases even further.

Retro Synth global and controller settings in Logic Pro for Mac

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 Filter Cutoff, Wave Shape (Pulse Width), or LFO/Vibrato Rate controls. Multiple MIDI controllers can be assigned to the same 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.
- *Polyphony pop-up menu:* Choose the maximum number of notes that can be played simultaneously (up to 16) or run as a monophonic 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 = 1/100 semitone).
- *Stereo Spread field:* Click the arrows or drag vertically to set the amount of voice panning, relative 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.
- *Unison pop-up menu:* Set the number of voices played in unison mode. Behavior in unison mode depends on the Polyphony parameter value. 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 in the Polyphony pop-up menu, voices are stacked, but you can play polyphonically.
  - Monophonic unison mode: When Mono or Legato is chosen in the Polyphony pop-up menu, all voices are stacked, but you can only play monophonically or in a legato style.

Controller parameters

The modulation targets available in the pop-up menus listed below change when different synthesizer engines are active.

- *Mod Wheel to pop-up menu and slider:* Choose a modulation target for your keyboard modwheel. The slider sets the maximum modulation amount.
- *Velocity to pop-up menu and slider:* Choose a target for modulation with keyboard velocity. The slider sets the maximum modulation amount.
- *Aftertouch to pop-up menu and slider:* Choose a target for modulation with keyboard aftertouch. The slider sets the maximum modulation amount.
- *Assignable CC pop-up menu and slider:* Choose a target for modulation with a MIDI continuous controller. The slider sets the maximum modulation amount.

  Note: This modulation source is ideal for use with MIDI CC#4 Foot Controller that’s often used in conjunction with aftertouch. It’s also useful for MPE devices that offer enhanced controller functions.

Retro Synth extended parameters in Logic Pro for Mac

Retro Synth provides additional parameters that can be accessed by clicking the disclosure arrow 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. Per-voice channels support pitchbend, aftertouch, modwheel, and CC messages. Controllers and MIDI messages sent on the base channel affect all voices.

- *Mono Mode Bend Range slider:* Set a value from 0 to 96.

  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 the GarageBand for iOS 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.

- *Audio File Analysis pop-up menu:* Change the sensitivity of detection within contiguous audio sections when creating custom wavetables. Choose the algorithm that is best-suited to the audio material. See Create a custom wavetable in Retro Synth.

Sample Alchemy

Sample Alchemy in Logic Pro for Mac

Sample Alchemy is designed to allow you to quickly resynthesize a single sample and transform it into a unique playable instrument. It provides a variety of synthesis techniques, such as granular, additive, and spectral synthesis, which can be combined to create a wide range of sounds, including unique sound beds, pads, effects, and rhythms.

You can control up to four independent sound generation sources using handles. Labeled A, B, C, and D, the handles control separate layers and can be placed at precise points on a waveform.

The five flexible play modes offer different ways to play back a sample and interact with it, from Classic mode, which lets you play the sample from start to finish, to Loop mode, where you place the handles on the waveform to create up to two snippets of looped audio. In addition, Scrub mode lets you move the handles across the waveform to trigger it as if scrubbing through tape, and Bow mode replicates the bow action used to play stringed instruments. Finally, Arp (arpeggiator) mode can generate repeating patterns of notes that cycle through different sections of the sample, creating intricate sequences based on MIDI notes.

Sample Alchemy is a sample-based instrument, so you can quickly begin creating sounds by loading samples and loops directly from the Loop Browser, Finder, or regions in the Tracks area. Sample Alchemy works best with monophonic instruments, vocals, or found sounds. When you record your instruments or vocal samples directly in Logic Pro for Mac, you can conveniently drag audio regions straight into an instance of Sample Alchemy.

You can use Sample Alchemy in a number of ways in your projects, such as building unique playable instruments or recording automation of the handles to add unique sound effects and synth elements. Here are a few examples of the types of sounds that you can create with Sample Alchemy:

- Rich and complex synth pads
- Evocative keys
- Rhythmic and evolving arpeggiator patterns
- Unique and playable effects
- Dynamic and evolving soundscapes
- Big fat basses
- Vocal-like formants

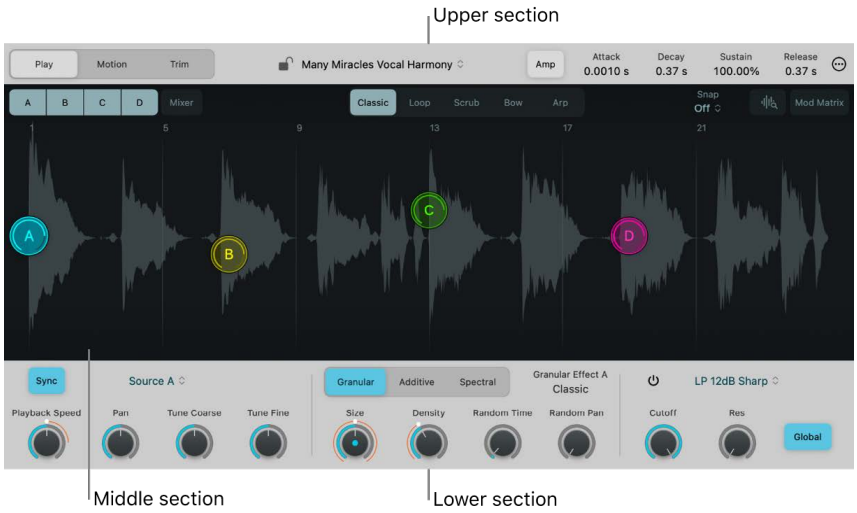
Additionally, with a library of predesigned sounds and a wide variety of sound-manipulation tools such as filters, synthesis effects, and modulators, Sample Alchemy allows you to create sounds tailored to your needs.

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

If you're new to using plug-ins in Logic Pro, see Add, remove, move, and copy plug-ins.

Sample Alchemy interface in Logic Pro for Mac

The Sample Alchemy interface consists of three primary sections:



- The upper section contains the Edit Mode buttons, the sample name and lock, the Amp/Mod button for the envelope generators, and the More button for settings.
- The middle section contains the source buttons, Mixer button, five play mode buttons, Snap menu, Zoom button, and Mod Matrix. The central part of this view includes the sample waveform display and the source handles. Here you can interact with the sound by setting a play mode and moving the handles. This view changes depending on the selected mode. See play modes, Motion mode, and Trim mode.
- The lower section contains the source controls on the left, the synthesis modes and their parameters in the middle, and the Filter controls on the right. This view changes depending on the selected mode.

Import or export source material via Sample Alchemy in Logic Pro for Mac

Sample Alchemy provides various ways to add an audio file, Apple Loop, or region. Once you've added the file, you can choose different modes to change playback behavior, you can directly edit waveform markers, and you can alter the tonal color and performance characteristics with the synthesis and modulation parameters. You can also export the source sample for use elsewhere in your project.

Drag a sample or loop from the Loop Browser into Sample Alchemy

- Drag a sample or loop from the Loop Browser into the Sample Alchemy waveform display.

  Tip: The Tonal filter accessible in the Loop Browser under Descriptors is a good way to find sounds that work well in Sample Alchemy.

Drag an audio region from the arrangement into Sample Alchemy

- Drag an audio region from the arrangement in your project into the Sample Alchemy waveform display.

Drag an audio file onto a track header

1. Drag a sample to a software instrument track header.
2. In the Create New Track using dialog, select Sample Alchemy.

Add an audio file using the Sample pop-up menu

1. Click the Sample pop-up menu at the top of Sample Alchemy.
2. Click Load Audio to open the Finder.
3. In the Finder, browse to the file you want to import, then click Open.

Export an audio file using the Sample pop-up menu

1. Click the Sample pop-up menu at the top of Sample Alchemy.
2. Click Export Audio.
3. In the Finder, name the file you want to save, then click Save.

Sample Alchemy edit modes in Logic Pro for Mac

There are three edit modes. Choose Play, Motion, and Trim buttons to determine the edit mode.

- *Edit mode buttons:* Select an edit mode.
  - *Play button:* Select to choose one of the five play modes just above the center of the waveform display, each providing a different way to play back the sample. See Play modes.
  - *Motion button:* Select Motion mode. Motion mode lets you record the movement of the source handles. After recording, the movement plays back in sync with the tempo of your project when the sound is triggered. See Motion mode.
  - *Trim button:* Select Trim mode. This mode lets you crop the sample. See Trim mode.

Sample Alchemy play modes in Logic Pro for Mac

Each of the five play modes provides a different way to play back the sample. After you select the Play button at the upper left, select from Classic, Loop, Scrub, Bow, and Arp buttons to determine the play mode.

- *Classic button:* Select Classic mode to play a sample from beginning to end. Sample playback begins from the handle point and finishes at the end of the sample. This mode is great for playing a sample in a conventional manner. In this mode, the first source parameter (Playback Speed) determines the sample playback speed.
- *Loop button:* Select Loop mode to loop between two points within a sample. In this mode, playback starts from the defined start point, playing to the defined loop end position, then skips to the start point and repeats. These positions are set on the sample using the source handle and source handle end point. You can also place the handle after the loop end point to play the sample in reverse, and move the entire audio section between these markers by dragging the bar between the loop start and end points. In this mode, the first source parameter (Loop Speed) determines the sample playback speed. Set the playback speed of each loop on each source independently to create more dynamic sounds. This mode is ideal for manipulating musical phrases and looped, rhythmic material. For instance, it lets you play particular sections of a waveform and repeat them as the basis for a new rhythmic sound.
- *Scrub button:* Select Scrub mode to play the sample at the touch point and in unison with up to four sources. Play the sample at the position you touch on the waveform, or scrub through the waveform as you touch and move source handles. Up to four sources can be heard simultaneously, depending on how many have been turned on. Scrub mode is great for selecting precise points of the sample and playing all sources in unison for VA synthesizer–style sounds or scrubbing through a sample waveform like a tape machine. In this mode, the first source parameter (Scrub Jitter) determines how much random movement is applied to the handle position for playback. This can be useful for creating a unison-like effect.

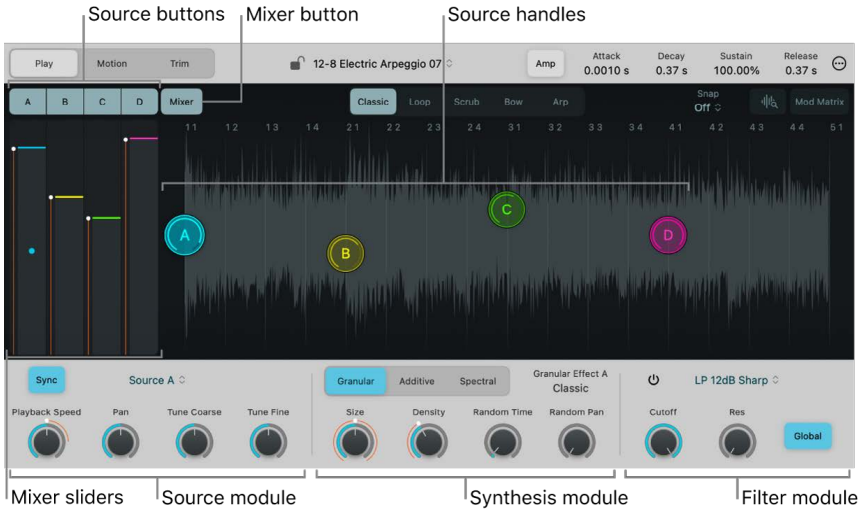
- *Bow button:* Select Bow mode to play the sample at each handle position with the bow action mechanism used to play stringed instruments. The up-and-down motion of a bow is replicated to play the sample sources. The up motion plays the sample normally, and the down bow plays the sample in reverse. Up to four sources can be heard simultaneously, depending on how many have been turned on. Bow mode is great for selecting precise points of the sample and playing them back and forth in unison. In this mode, the first source parameter (Bow rate) determines the speed of the bow action. The speed of the bow action can also be synced to the tempo of your project using the Sync button.
- *Arp button:* Select Arp mode to generate repeating patterns of notes that trigger different sections of the sample to create intricate sequences based on the notes played in real time or from a MIDI region. In this mode, the first source parameter (Arp Speed) determines the rate of the arpeggiator. The rate of the arpeggiation is automatically synced to the tempo of your project.

Sample Alchemy sources in Logic Pro for Mac

Sample Alchemy sound-generating modules are called *sources*. Four independent sources (labeled A/B/C/D) are available in each preset, and each source has an identical set of controls.

Source components are shown when you click the A, B, C, and D buttons to select the source you want to edit. Some parameters can be set globally or per source. Global parameters override individual source settings.

The components in each source are displayed in three modules: source, synthesis, and filter.



- *Sample Lock button:* Click the lock to keep the current sample when changing presets.
- *Sample pop-up menu:* Load a new sample, export the current sample, or show the current sample in the Finder.
- *Source buttons:* Turn a source on or off. When turned on, a source is a sound-generating module indicated by a handle labeled A/B/C/D that is positioned on the waveform.

- *Mixer button:* Show or hide the mixer pane. The mixer lets you adjust the volume for each source.
- *Mixer sliders:* Adjust the volume for each source. A dot indicates which source you're currently adjusting.
- *Snap pop-up menu:* Choose a Snap value. If you select the Play or Motion edit mode, the Snap value affects the movement of source handles. In Trim mode, the Snap value affects edits to loop start and end markers in the waveform display. The following options are available in the Snap pop-up menu:
  - *Auto:* The Snap value is automatically set to the best value based on the selected mode and the sample.
  - *Off:* Turn snap off.
  - *Transients:* Snap to transients in the sample.
  - *Beat:* Snap to beat values in the sample.
  - *Fixed:* Set the Snap value to allow source handles to move only on the Y axis (vertically) in play modes and Motion mode.
- *Zoom button:* Tap Zoom to access the horizontal zoom slider. Drag left or right to view a specific waveform section, or adjust the slider's handles for more or less detail.
- *Mod Matrix button:* Show or hide the Mod Matrix pane, which you can use to assign up to four independent routings of modulation to a target.
- *Source handles:* Click the handles labeled A/B/C/D to hear the sample preview. Drag left or right to adjust the playback position, and drag up or down to adjust the first parameter in the source module. When preview mode is turned on in the More menu settings and you click a handle, Sample Alchemy plays a single MIDI note that you can record to a MIDI region in Logic Pro. See More menu. You can record handle movements using automation or directly in Sample Alchemy using Motion mode. See Record handle movements to a MIDI region and Motion mode.
- *Source module:* Adjust the parameters for the selected source. The first parameter available here changes depending on the selected play mode. For example, in Classic mode you can control the speed at which the sound is played, and in Loop mode you can control the loop speed. The other parameters in this column let you set the panning and tuning for each source.
- *Source pop-up menu:* Choose the source (A/B/C/D) you want to edit, or choose Select All to edit all sources.
- *Synthesis module:* Choose from multiple sound-generating modes based on different methods of synthesis. Here you can set a synthesis mode for each source. See Sample Alchemy synthesis modes.
- *Filter module:* Set the filter for each source or globally. See Sample Alchemy filters.

Source module parameters

- *Playback Speed knob:* Set the sample playback speed of the sample. Set low speed values to create gradually evolving textures.
- *Playback Speed Sync button:* Turn on to synchronize the playback speed with the project tempo. Turn off to freely set the playback speed. This button is active only when the loaded sample contains tempo information.
- *Loop Speed knob:* Set the sample loop speed between the start point and end point.
- *Loop Speed Sync button:* Turn on to synchronize the loop speed with the project tempo. Turn off to freely set the loop speed. This button is active only when the loaded sample contains tempo information.
- *Scrub Jitter knob:* Set the depth of the jitter action.
- *Bow Rate knob:* Set the rate of the bow action.
- *Bow Rate Sync button:* Turn on to set the rate in tempo-synced note values instead of Hz. Turn off to freely set the bow rate.
- *Arp Rate knob:* Set the duration of the arpeggiator steps. The arpeggiator is synchronized with the project tempo, and the Rate is set from whole notes (1) to 1/32 notes. Dotted and triplet subdivisions provide more complex and expressive rhythmic patterns.
- *Pan knob:* Set the source output position in the stereo field.
- *Tune Coarse knob:* Tune the selected source in semitone increments.
- *Tune Fine knob:* Tune the selected source pitch in cent increments.

Turn on a source in Sample Alchemy

- Click a source button (A/B/C/D) to turn on a source.

Turn off a source in Sample Alchemy

- Click a highlighted source button (A/B/C/D) to turn off a source.

Select a source in Sample Alchemy

- Click a source handle (A/B/C/D) to select a source.

Select multiple sources in Sample Alchemy

- Click the Source menu on the lower left side of Sample Alchemy, then choose Select All.

Mix sources in Sample Alchemy

- Click the Mixer button on the upper-left side of Sample Alchemy, then drag a volume level up or down.

Record handle movements to a MIDI region

1. Select the track that has the Sample Alchemy plug-in loaded.
2. Open the Sample Alchemy plug-in window with a sample loaded.
3. In the Track inspector, ensure that the Internal MIDI In > Instrument Output submenu has this track selected.
4. Start recording your project and move the handles in Sample Alchemy.

The MIDI region records a MIDI note every time you click a handle, and the movement of the handle is recorded as region automation. For information about routing MIDI internally in Logic Pro, see Route MIDI internally to software instrument tracks.

Sample Alchemy synthesis modes in Logic Pro for Mac

Each source has an independent set of parameters and therefore can be set to a different synthesis type. For example, you can combine granular synthesis with additive synthesis or spectral synthesis. Each synthesis method has inherent strengths and weaknesses that make it more suitable for certain sound types than other synthesis engines.



The synthesis module in the lower section has three synth mode buttons you can use to view the synthesis engine controls and an effect menu:

- *Synth mode buttons:* Choose a granular, additive, or spectral synth mode. Each mode provides a different type of synthesis.
  - *Granular button:* Granular mode splits sounds into small “grains” to create unique textures, from smooth pads to glitchy effects and evolving, fragmented sounds. For more information about Granular mode, see Granular effect controls.
  - *Additive button:* Additive mode in Sample Alchemy builds sounds by combining sine waves at different frequencies and amplitudes, providing precise control over harmonics for complex, evolving tones. For more information about Additive mode, see Additive effects.
  - *Spectral button:* Spectral mode analyzes a sound’s frequency spectrum, letting you adjust individual frequencies to reshape its timbre and create unique tonal changes. For more information about Spectral mode, see Spectral effects.
- *Source Effect pop-up menu:* Choose a synthesis effect type for the Additive or Spectral synth engines and change related parameters in the area below.

Sample Alchemy granular effect controls in Logic Pro for Mac

Granular synthesis is a technique used to create and manipulate sounds by breaking them down into small pieces, or “grains,” and manipulating these grains independently.

In granular synthesis, a sound is typically broken down into a series of short, overlapping grains, each of which is a few milliseconds in length. These grains are then played back in sequence, either as a continuous stream or in a randomized or more structured pattern.

You can create a wide range of sounds and timbres by manipulating various parameters of the grains, such as their pitch, duration, and position in the original sound waveform. Granular synthesis can be used to create everything from subtle changes in a sound to radical transformations, and it is often used to create unique and unconventional sounds that are difficult to achieve using other synthesis methods.

The parameters in this section are shown when Granular is selected in a synthesis module.

Click one of the handles labeled A/B/C/D to select the source you want to edit.



Granular module parameters

- *Size knob:* Adjust the duration of each grain from 2 ms to 230 ms.
- *Density knob:* Determine the number of potentially overlapping grains from 1 (no overlap) to 10.

  The Size and Density parameters interact with each other. When the Density value is 1, a single grain is sent to the output stream. As soon as one grain finishes, the next one is sent. A Size value of 100 ms sends a new grain every 100 ms.

  Increasing Density to 2 adds a second grain that is sent in between those of the first, resulting in a new grain every 50 ms, assuming a Size value of 100 ms. The first and second grains overlap each other. Higher Density values inject additional new grains into the output stream. These new grains occur more frequently and overlap more heavily.

  Setting Size to around 100 ms and Density to around 5 grains is often suitable for smooth pad sounds with no sharp transients. Setting Size between 40 and 80 ms and Density to around 2 grains is useful for drums and other sounds featuring sharp transients. Small Size values tend to produce a buzz that masks the original pitch of the sample. Large Size values tend to break up the sound. You can counteract both tendencies by increasing the Density.
- *Random Time knob:* Add a small random offset to grain extraction positions in the sample. The default value is 3% because a small amount of randomization helps to smooth the output of the granular element.
- *Random Pan knob:* Add a random offset to the stereo position of each grain.

Sample Alchemy additive effects in Logic Pro for Mac

Additive synthesis represents each sound as a sum of individual partials. Each partial is synthesized using a sine wave and is represented in terms of four parameters: amplitude, pitch, pan, and phase, each of which changes over time.

You can think of the additive data as a series of snapshots, each of which captures the amplitude, pitch, pan, and phase of every partial at a particular point in time. In between snapshots, each parameter updates smoothly toward the following snapshot value. When played in succession and with the right timing, the series of snapshots describes a potentially complex and continuously evolving sound.



Sample Alchemy additive effect controls in Logic Pro for Mac

The parameters in this section are shown when Additive is selected in a synthesis module. The controls also vary depending on the selected additive effect.

Click one of the handles labeled A/B/C/D to select the source you want to edit.



Harmonic

The Harmonic effect allows you to control groups of partial levels, with controls relating to harmonic intervals.

- *Odd/Even knob:* Set the balance between odd or even harmonics. Low values increase the level of odd-numbered partials (1, 3, 5, 7, and so on), making the sound more hollow. High values increase the level of even harmonics (retaining the fundamental tone: harmonic/partial 1) to make the sound brighter and sweeter.
- *Fifths knob:* Set the level of the fundamental tone and all partials at fifth intervals (7 semitones) above it. Higher values boost harmonics 1, 3, 9, 27, and so on, with a corresponding reduction in the levels of other harmonics. Low values have the reverse effect and can make the sound more cutting and edgy.

- *Octaves knob:* Set the level of the fundamental tone and all partials at whole octave intervals above it. Set to zero to completely remove harmonics 1, 2, 4, 8, 16, and so on, while boosting the levels of non-octave harmonics. Set to higher values to boost the levels of octave harmonics while reducing the levels of other harmonics.
- *Fundamental knob:* Set the level of the fundamental tone and all partials above it. Set to zero to completely remove the fundamental tone. Set to 100% to hear the fundamental tone in isolation. Higher values tend to make the sound thicker.

Partials Lock

Partials Lock is an effect that sets all of the partials to the same zero phase value at the start. It sets the partials to have no variation in pitch over the course of the sound, resulting in a clean, artificial sound.

- *Symmetry knob:* Alter the symmetry, or shape, of the sine waves by lengthening the first half of the waveform while shortening the second, or the reverse. The audible effect is similar when the knob is turned in either direction. Symmetry alters waveforms until they are no longer pure sine waves in shape, resulting in each partial developing independent harmonics and making the sound brighter.
- *Odd/Even knob:* Set the balance between odd or even harmonics. Low values increase the level of odd-numbered partials (1, 3, 5, 7, and so on), making the sound more hollow. High values increase the level of even harmonics (retaining the fundamental tone: harmonic/partial 1) to make the sound brighter and sweeter.
- *Pitch Var knob:* Tune all partials simultaneously. This occurs before processing by the additive effect modules that stretch or shift partial tunings. Set to 0% to tune all partials in a perfect harmonic series. Set to 100% to make each partial follow the pitch fluctuations detected in the original audio file. The sonic impact of this parameter is highly dependent on the audio material: Sounds with strong inharmonic content, such as bells, are dramatically changed by reducing pitch variations. If all partials are tuned to the harmonic series, however, turning the knob has no influence on the sound.
- *Num Partials knob:* Set the number of additive partials that are generated. The number of oscillators required depends on the sound. For example, a flute has a limited number of harmonics and requires fewer partials than a cello or a violin. The playable register can also affect the number of oscillators required: High notes will accommodate only a small number of higher harmonics before reaching the limits of audibility, whereas low bass notes may have hundreds of harmonics without reaching the limit. Always set Num Partials to the lowest number of partials that are required by the sound because this helps reduce CPU load.

Partials

Partials is an effect that allows you to manipulate the spectral content of a sound by modifying individual partials within the sound.

- *Odd/Even knob:* Set the balance between odd and even harmonics. Low values increase the level of odd-numbered partials (1, 3, 5, 7, and so on), making the sound more hollow. High values increase the level of even harmonics (retaining the fundamental tone: harmonic/partial 1) to make the sound brighter and sweeter.
- *Symmetry knob:* Alter the symmetry, or shape, of sine waves by lengthening the first half of the waveform while shortening the second, or the reverse. The audible effect is similar when the knob is turned in either direction. Symmetry alters waveforms until they are no longer pure sine waves in shape, resulting in each partial developing independent harmonics and making the sound brighter.
- *Pitch Var knob:* Tune all partials simultaneously. This occurs before processing by the additive effect modules that stretch/shift partial tunings. Set to 0% to tune all partials in a perfect harmonic series. Set to 100% to make each partial follow the pitch fluctuations detected in the original audio file. The sonic impact of this parameter is highly dependent on the audio material: sounds with strong inharmonic content, such as bells, are dramatically changed by reducing pitch variations. If all partials are tuned to the harmonic series, however, turning the knob has no influence on the sound.
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Formant

Formant is an effect that allows you to shape the spectral content of a sound by emphasizing or attenuating specific frequency bands.

- *Shift knob:* Shift all synthesized formant filters up or down in semitones. Higher values can make sounds seem brighter or thinner. Lower values can create a darker, thicker character.
- *Key Track knob:* Determine how the formant filter tracks notes on the keyboard. At 100%, filter resonances shift up or down in pitch with the note. Set to lower values to reduce key tracking, which may make some sounds playable over a wider keyboard range.
- *Size knob:* Stretch the formant filter to alter the perceived size of the resonant chamber. Size works in conjunction with the Center knob.
- *Center knob:* Set the center frequency for the formant stretch that you set with the Size knob. Resonances below the center frequency are shifted upward as the Size value is increased. A corresponding downward shift occurs to resonances above the center frequency.

  Note: The Center knob has no effect when the Size knob is set to 100%.

Formant Synth

Formant Synth is an effect that applies a vocal formant effect to shape the timbre of the synthesized sound.

- *Shift knob:* Shift the formants up or down in semitones. Higher values can make sounds seem brighter or thinner. Lower values can create a darker, thicker character.
- *Size knob:* Stretch the formant filter to alter the perceived size of the resonant chamber. Size works in conjunction with the Center knob.
- *Center knob:* Set the center frequency for the formant stretch that you set with the Size knob. Resonances below the center frequency are shifted upward as the Size value is increased. A corresponding downward shift occurs to resonances above the center frequency. Note: The Center knob has no effect when the Size knob is set to 100%.
- *Vowel knob:* Morph smoothly through the four filter shapes: A, E, I, and O. The displayed value indicates position. Whole numbers indicate a particular filter unit, and fractional values indicate a position between filters.

Sample Alchemy spectral effects in Logic Pro for Mac

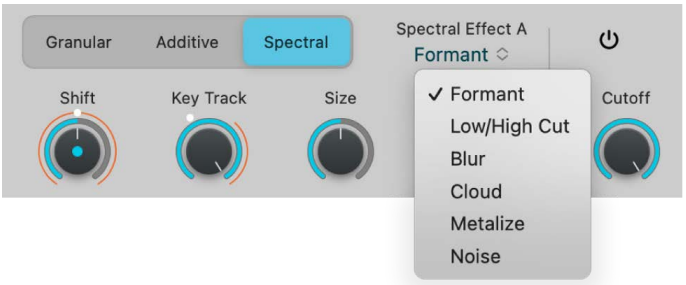
Spectral resynthesis analyzes the changing frequency spectrum of a signal and attempts to recreate these spectral characteristics. In Sample Alchemy, the audible spectrum of a signal is split into a large number of spectral bins. Energy distribution across these bins is analyzed and the sound is recreated by filling each spectral bin with the required amount of signal, using either sine waves or filtered noise. The results are then summed.



Sample Alchemy spectral effect controls in Logic Pro for Mac

The parameters in this section are shown when Spectral is selected in a synthesis module. The controls also vary depending on the selected spectral effect.

Click one of the handles labeled A/B/C/D to select the source you want to edit.



Formant

Formant is an effect that allows you to shape the spectral content of a sound by emphasizing or attenuating specific frequency bands.

- *Shift knob:* Shift the formant filter up or down in semitones. Higher values can make sounds seem brighter or thinner. Lower values can create a darker, thicker character.
- *Key Track knob:* Determine how the formant filter tracks notes on the keyboard. At 100%, filter resonances shift up or down in pitch with the note. Set to lower values to reduce key tracking, which may make some sounds playable over a wider keyboard range.
- *Size knob:* Stretch the formant filter to change the perceived size of the resonant chamber. Use this parameter to alter the size of a guitar body or to make a child's vocal sample sound like that of a giant, for example. The Size knob works in conjunction with the Center parameter.
- *Center knob:* Set the center frequency for formant stretching (controlled with the Size knob). Resonances below the center frequency are shifted upward as the Size knob value is increased. A corresponding downward shift occurs to resonances above the center frequency.

  Note: The Center knob has no effect when the Size knob is set to 100%.

Low/High Cut

Use the Low and High Cut knobs to shape the tonal characteristics of the sound. The Low Cut and High Cut parameters work in conjunction with each other to act as a bandpass filter, where signals that fall within the two cutoff ranges are allowed to pass.

- *Low Cut knob:* Set a cutoff frequency. All signals above this frequency are allowed to pass. Signals below the frequency are cut.
- *High Cut knob:* Set a cutoff frequency. All signals below this frequency are allowed to pass. Signals above the frequency are cut.

Blur

Blur produces a frequency-blurring effect.

Tip: Try a melodic loop with pitch variations to best hear the impact of this effect.

- *Mix knob:* Set the balance between the original signal and the processed sound.
- *Length knob:* Set the time period that frequencies are sustained (blurred over time).
- *Variance knob:* Set the degree of variation for frequency selection (frequencies that are blurred).
- *Gate knob:* Adjust the volume threshold so that only frequency bands louder than the threshold are played back. It's similar to a noise gate but applied to each band individually.

Cloud

Cloud produces what might best be described as a cloud of frequency grains, resulting in a textured chorus effect.

Tip: Try this effect on vocal samples.

- *Mix knob:* Set the balance between the original signal and the processed sound.
- *Threshold knob:* Set how many of the frequency bands have the effect applied to them—the higher this value, the more bands are affected.
- *Attack knob:* Set the time it takes for frequencies emphasized by the Threshold setting to fade in.
- *Simplify knob:* Reduce lower amplitude frequencies to enhance prominent frequencies, lessening detail in favor of an increasingly pure tone.

Metallize

Metallize produces classic comb filter–style effects.

Tip: Experiment with drum loops to clearly hear the impact of the controls.

- *Mix knob:* Set the balance between the original signal and the processed sound.
- *Interval knob:* Set the number of semitones that the affected bands are shifted up.
- *Simplify knob:* Reduce the number of effect-generated frequencies, resulting in a simplified sound.
- *Feedback knob:* Set the intensity of the effect. Higher settings emphasize harmonics, creating metallic resonances.

Noise

Use Noise to fill spectral bins with filtered noise. The Low Cut and High Cut parameters work in conjunction to act as a bandpass filter, where signals that fall within the two cutoff ranges are allowed to pass.

- *Low Cut knob:* Set a cutoff frequency. All signals above this frequency are allowed to pass. Signals below the frequency are cut.
- *High Cut knob:* Set a cutoff frequency. All signals below this frequency are allowed to pass. Signals above the frequency are cut.

Sample Alchemy filters in Logic Pro for Mac

In a Sample Alchemy, the filter module modifies the harmonic content of the sound being produced. It can shape the tonal characteristics of the sound by removing or reducing certain frequencies while allowing other frequencies to pass through.

Sample Alchemy provides an extensive range of filter types, including lowpass, highpass, comb, downsampler, and FM filters. Each filter type has unique sonic characteristics and responds differently to incoming signals.

The filters in Sample Alchemy can also be modulated by an envelope, an LFO (low frequency oscillator), or other modulation sources which allows them to be controlled over time to create dynamic and evolving sounds. Modulation can be assigned in the Mod Matrix pane. See Sample Alchemy Mod Matrix.



Sample Alchemy filter types in Logic Pro for Mac

This flexible collection of filters allows you to subtly or dramatically sculpt the tone of your instruments:

- *Lowpass (LP):* This filter type passes the portion of a signal below a specified cutoff frequency and rolls off the portion above that frequency. See Sample Alchemy lowpass, bandpass, and highpass filters.
- *Bandpass (BP):* This filter type passes a certain band of frequencies and rolls off frequencies on either side of the band. See Sample Alchemy lowpass, bandpass, and highpass filters.
- *Highpass (HP):* This filter type passes the portion of a signal above a specified cutoff frequency and rolls off the portion below that frequency. See Sample Alchemy lowpass, bandpass, and highpass filters.
- *Comb PM:* This filter mixes the original signal with one or more copies of the signal, which are delayed by a very short time interval. See Sample Alchemy Comb PM filter.
- *Downsampler:* This filter creates a lo-fi digital effect (similar to a bitcrusher). See Sample Alchemy downsampler filter.
- *FM:* This filter uses a modulator oscillator and a sine wave carrier oscillator. The modulator oscillator modulates the frequency of the waveform generated by the carrier oscillator within the audio range, thus producing new harmonics. See Sample Alchemy FM filter.

Sample Alchemy filter general parameters

- *Filter On/Off button:* Turn the filter module on or off.
- *Filter Type pop-up menu:* Choose a filter type.
- *Global button:* Turn on the Global button and choose a filter type and configuration to use for all sources, overriding their individual filter type settings.

Sample Alchemy lowpass, bandpass, and highpass filters in Logic Pro for Mac

A lowpass (LP) filter passes the portion of a signal below a specified cutoff frequency and rolls off the portion above that frequency. A bandpass (BP) filter passes the portion of a signal occupying a band surrounding the cutoff frequency and rolls off the portions above and below that band. A highpass (HP) filter passes the portion of a signal above a specified cutoff frequency and rolls off the portion below that frequency.

There are multiple two-pole, four-pole, multipole state-variable, and analog-modeled lowpass (LP) and highpass (HP) filter designs in Sample Alchemy, each with distinctive characteristics that you may prefer for a given purpose. The available LP and HP filter designs include Sharp, Clean, Gritty, and Edgy variants.

- *Sharp:* Two-pole analog-modeled filters.
- *Edgy:* Two- or four-pole analog-modeled filters. Four-pole variants provide a steeper roll-off of frequencies beyond the cutoff.
- *Clean:* Bi-Quad filters.
- *Gritty:* Two-pole filters designed to saturate heavily at higher resonance (Res) settings.

Sample Alchemy lowpass, bandpass, and highpass filter parameters

The filter controls have standard functions for all lowpass (LP) bandpass (BP), and highpass (HP) filter types.

- *Cutoff knob:* Controls the filter cutoff frequency.
- *Res(onance) knob:* Controls the filter resonance or emphasis. Higher settings boost frequencies in the immediate vicinity of the cutoff frequency.

Sample Alchemy Comb PM filter in Logic Pro for Mac

A comb filter is a type of digital audio effect that creates a characteristic “comb” pattern of peaks and troughs in the frequency spectrum of an audio signal. It is named after the shape of the frequency response curve, which looks like the teeth of a comb.

The Comb PM filter is useful for classic bright Karplus-Strong style sounds, where the exciter impulse is not easily heard and the comb is more prominent. Be careful with your resonance level because it is capable of quickly going to extremes, which can lead to feedback. Start with a resonance level of zero, and increase (or decrease) slowly to find a suitable effect strength.

Comb PM filter parameters

- *Note:* Set the modulation frequency.
- *Feedback:* Controls the amount of feedback in the comb circuit.

Sample Alchemy downsampler filter in Logic Pro for Mac

The downsampler filter is a type of digital audio effect that reduces the sampling rate of a digital audio signal. This can be used to create a range of different effects, such as creating a lo-fi or vintage sound.

Downsampling works by reducing the number of samples that are used to represent the audio signal. This can be done by discarding some of the samples. The resulting audio signal has a lower sample rate and a lower frequency range, which can give it a different character or timbre.

Downsampler filter parameters

- *Freq:* Set the downsample frequency.
- *Mix:* Set the balance between the original and filtered signals. High values result in stronger downsampling.

Sample Alchemy FM filter in Logic Pro for Mac

Frequency modulation (FM) in Sample Alchemy has a different character from classic FM synths. Unlike dedicated FM synths, Sample Alchemy does not have preconfigured algorithms or a mod matrix setup specifically for FM.

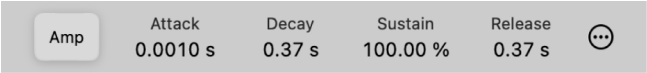
FM in Sample Alchemy is more like working with FM on analog synths, where you are modulating oscillator frequency rather than phase. As a result, Sample Alchemy's FM is great for adding different kinds of effects to your sound, as well as "organic" distorted textures.

FM filter parameters

- *Note:* Set the carrier oscillator frequency.
- *Mod:* Set the degree to which the modulator (source audio) can modulate the frequency of the carrier.

Sample Alchemy envelope generators in Logic Pro for Mac

You can control the level of the amplitude or modulation over time with dedicated envelope generators. Use the Amp/Mod button to access each envelope. The parameters for each envelope are the same.



Amp and Mod Envelope parameters

- *Amp/Mod button:* Choose an envelope type. You can choose between the Amp (amplitude) envelope or Mod (modulation) envelope.
- *Attack value:* Drag up or down to set the time it takes for the envelope to reach the initial level.

- *Decay value:* Drag up or down to set the time it takes for the envelope to fall to the sustain level, following the hold phase or the initial attack time.
- *Sustain value:* Drag up or down to set the sustain level, which is held until you release the note.
- *Release value:* Drag up or down to set the time it takes for the envelope to fall from the sustain level to a level of zero after you release the note.

Sample Alchemy Mod Matrix in Logic Pro for Mac

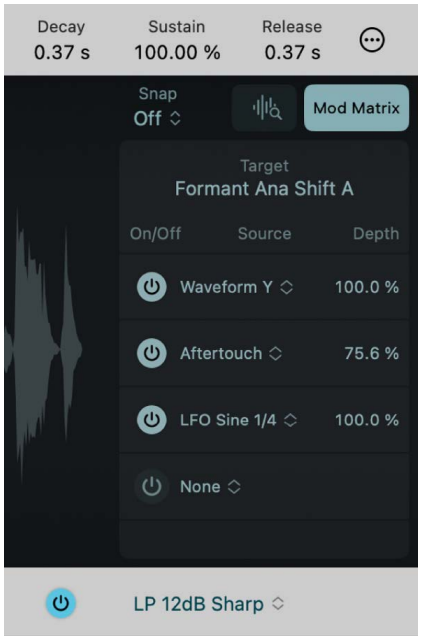
Sample Alchemy LFOs and envelopes are known as *modulation generators*. These modulation sources are used to control modulation targets, such as filter cutoff, source synthesis parameters, and source controls. You can assign up to four independent routings of modulation sources and targets in the Mod Matrix pane.

Sample Alchemy provides four LFOs and dedicated modulation and amplitude envelopes that can also be assigned as modulation sources or targets in the Mod Matrix pane.

Use the Mod Matrix to control how the sound changes as you move handles up and down the waveform using the Waveform Y modulator. Your keyboard modulation wheel, aftertouch, pitch bend, velocity, and MIDI continuous controller features can also be assigned as real-time control sources for Sample Alchemy parameters in the Mod Matrix pane.

When a parameter is the target of one or more modulators, an orange modulation arc is shown beside the blue value arc for the knob. This indicates that the knob is an active modulation target and shows the modulation range. The amount of depth you assign in the Mod Matrix pane determines how much a modulation source affects its target.

The Mod Matrix pane shows the modulators applied to the currently selected control in Sample Alchemy. This is shown via the Target field at the top of the Mod Matrix pane as well as by the dot that appears in the center of the currently selected knob.



Mod Matrix parameters

- *Mod Matrix button:* Show or hide the Mod Matrix pane to assign up to four independent routings of modulation to a target.
- *Target pop-up menu:* Displays the selected target. Select a target from the menu or click a parameter in the main Sample Alchemy interface to select it as new modulation target.
- *On/Off button:* Turn the row of source modulation for the selected target on or off.
- *Source pop-up menu:* Choose a modulation source.
- *Depth field:* Drag vertically to set the maximum value or intensity of modulation. You can also double-click the field to enter a numerical value.

Set up modulation routing for Sample Alchemy in Logic Pro for Mac

You can assign Sample Alchemy LFOs and envelopes as modulation sources or targets in the Mod Matrix pane. You can also assign Waveform Y and your keyboard modulation wheel, aftertouch, pitch bend, velocity, and MIDI continuous controllers as real-time control sources for Sample Alchemy target parameters in the Mod Matrix pane.

When a parameter is the target of one or more modulators, an orange modulation arc is shown beside the blue value arc for the knob. This indicates that the knob is an active modulation target and shows the modulation range.

Route the source handles to control volume

By default, the movement of the source handles (A/B/C/D) along the y-axis (vertcal axis) of the sample waveform is mapped to modulate the first parameter in the synthesis module. To map the source handles to control volume instead, follow these steps:

1. In the Source module in Logic Pro, click the Source pop-up menu and choose All to edit all sources.
2. Click the Mod Matrix button to show the Mod Matrix.
3. Click the first synth knob in the Synthesis module to select it.
4. In the Mod Matrix, click Waveform Y to open the Source pop-up menu, then choose None to remove the current mapping.
5. Click the Mixer button to open the mixer, then click slider A to select it as the modulation target in the Mod Matrix.
6. In the Mod Matrix, click the Source pop-menu and choose Waveform Y.
7. Adjust the Depth value in the Mod Matrix, and adjust the level of the mixer sliders to fine-tune your sound.

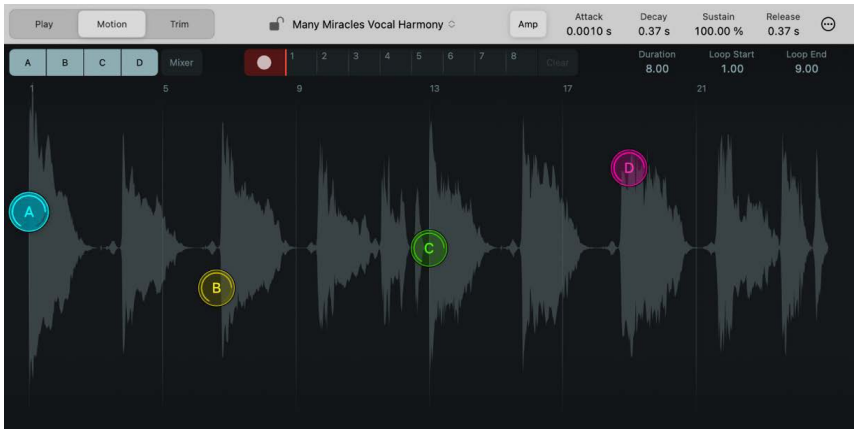
Sample Alchemy Motion mode in Logic Pro for Mac

Using Motion mode, you can record the movement of source handles to automate various parameters so that they change over time, producing dynamic and evolving sounds.

After you record a motion sequence, it is automatically looped and remains tempo-synchronized to your project. When you play back the sound, the Motion sequence you recorded is visually replicated on the screen.

Add to your recording by making overdubs or manually setting the Loop Start and Loop End points. If you're unsatisfied with your recording, you can use the Clear button to erase the motion recording and start over.

Tip: Instead of using Motion mode to record the movement of the handles, you can record it directly to a region. After you record your performance, you can edit the automation captured in the region to suit your project.



Motion mode parameters

- *Record button:* Click the Record button to start recording motion. The Record button flashes until you touch the handles or a MIDI input is received. Recording when a recording already exists will cause any additional touches to override that particular section of the recording, in a similar manner to Touch automation.
- *Timeline:* Displays the playback position of the latest note, with beat markers as the axis.
- *Clear button:* Delete the motion recording. This button only appears after a motion recording has been made.
- *Duration:* Set the duration of the recording in beats. Changing this value will stretch the motion recording and loop start and end positions accordingly.
- *Loop Start:* Set the motion playback loop start point in beats.
- *Loop End:* Set the motion playback loop end point in beats.

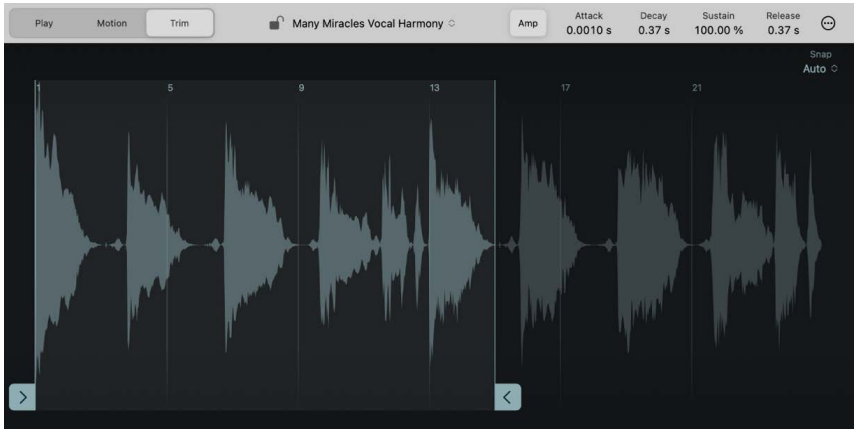
Record a motion sequence

1. Click the Record button to turn on the motion recorder (if necessary). The Record button flashes until you touch the handles or a MIDI input is received.
2. Move the source handles as required for the duration of the recording.
3. Release the handles to automatically stop the recording, or click the Stop button.
4. Click Record again if you want to overdub the motion recording.
5. Play a MIDI note to listen to and visualize your motion recording. You can now switch to Play mode if you want. Any movement of the handles shifts your motion recording up/down/left/right.

Sample Alchemy Trim mode in Logic Pro for Mac

Trim mode lets you select the exact part of the sample you want to use. You can see the current audio file in the waveform display and directly trim the sample using markers. You can zoom in or out with pinch gestures or scroll using single or two-finger swipes. When you zoom in, the scroll bar indicates the current position within the waveform.

When you click or drag markers and handles in the waveform display, you can trim the waveform more accurately by choosing a Snap value, which allows you to snap the markers to a transient or beat.



Trim mode parameters

- *Start and end markers:* Drag the start and end markers to set the sample start and end points. Playback occurs between these markers.
- *Zoom in/out:* Use the pinch gesture on your Trackpad to zoom in and out of the waveform.

Set trim start and end markers

To set the start and end markers in Trim mode, do any of the following:

- Click the Snap menu and choose a Snap value. See Source overview.
- Click the start marker then, drag it to the left or right to set the start position of the sample.
- Click the end marker then, drag it to the left or right to set the end position of the sample.

  Note: You can use the pinch gesture on your Trackpad to zoom in and out of the waveform for more precise editing. If you zoom in close to the waveform, swipe to the left or right to see the start or end markers.

Sample Alchemy More menu in Logic Pro for Mac

The More menu at the upper right contains a number of global, sample, and MIDI handling options.

- *More button:* Open or close the More menu to choose a number of global, sample, and MIDI handling options.

Global settings

- *Volume:* Set the volume level of the preset.
- *Polyphony:* Set the maximum polyphony of a preset (up to 16 voices). A value of 1 results in a monophonic preset or source.
- *Glide:* Set the portamento rate. Glide causes slides from one note pitch to the next.
- *Play Mode:* Determine how new notes are treated. This parameter interacts with the Polyphony and Glide controls.
  - *Always:* If the Polyphony value is 1, a trigger is generated at the start of each legato group, and portamento occurs at the start of every note. For all other Polyphony values, a trigger is generated at the start of every note, and portamento occurs at the start of every note.
  - *Retrigger:* A trigger is generated at the start of every note, and portamento occurs at the start of every note.
  - *Legato:* If the Polyphony value is 1, a trigger is generated at the start of each legato group, and portamento occurs at the start of each legato group. For all other Polyphony values, behavior is unchanged when you play single notes. When you play a chord, each note of the chord is triggered individually.
- *Bend:* Set the maximum range for upward and downward pitch bend modulation, typically performed with your keyboard pitch bend wheel.
- *Output Limiter:* Turn the output limiter on or off. The output limiter manages the maximum output level of Sample Alchemy, preventing distortion and clipping.

Sample settings

- *Preview Mode:* Turn preview mode off or on (it's on by default). In preview mode, a MIDI note is generated when you click a handle. When preview mode is off, there is no sound when you click the handles, and the preview MIDI note can't be recorded in Logic Pro.
- *Preview Key:* Sets the note used to preview samples when preview mode is on.
- *Pitch Lock:* Lock the pitch of the sample to the root key. If you import an audio file containing notes with various pitches, Pitch Lock allows you to lock them all to a single note. After this, you can play Sample Alchemy (for example, with a MIDI keyboard), and it will play the notes you hold down, not the differing pitches within the sample.
- *Reverse:* Set the sample to play in reverse.
- *Root Key:* Adjust the pitch of the sound in semitones.
- *Root Cents:* Adjust the pitch of the sound in cents.
- *Tempo:* Set the tempo (shown in beats per minute) of the sample, or choose None if the sample is not rhythmic.
- *Derive Tempo from Loop Length:* Calculate a tempo for the audio file based on the length of the sample between trim handles.

MIDI settings

- *MIDI Mono Mode (MPE):* You can choose the settings for MIDI Mono Mode:
  - *MIDI Mono Mode:* Choose Off, On (Common Base Channel 1), or On (Common Base Channel 16). In either mode, each voice receives on a different MIDI channel. Per-voice channels support pitch bend, aftertouch, modulation wheel, and controller assignment messages.
  - *Pitch Bend Range:* Set a value from 0 to 96. The chosen pitch bend range affects individual note pitch bend messages received on all but the Common Base Channel. The default is 48 semitones. When using a MIDI guitar, 24 semitones is the preferable setting because most guitar-to-MIDI converters use this range by default.
- *MIDI Assign:* You can choose one of four different modulation sources from the MIDI submenu. Ctrl A/B/C/D can be assigned to modulation targets in the Mod Matrix pane. These are ideal for adding breath and foot controller modulations.
  - *Ctrl A–D:* Set the MIDI continuous controller that is assigned to Ctrl A–D.

Create Alchemy track

- *Create Alchemy track:* Creates a new Alchemy track that contains the current Sample Alchemy audio material.

Sampler

Sampler overview in Logic Pro for Mac

Sampler is a sophisticated software sampler with which you can quickly create multisample instruments using drag-and-drop workflows.

Audio files, called *samples*, are combined into tuned, organized collections called *sampler instruments*. A sampler instrument is the file type that is loaded into Sampler with the plug-in Settings pop-up menu. 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.

Because sampler instruments are based on audio recordings, they are ideally suited to emulating real instruments such as guitars, pianos, and drums. Sampler provides an extensive library of sampler instruments that includes these and many other sounds. Sampler is fully compatible with EXS instrument libraries.

You can also use Sampler to edit and create your own sampler instruments. You can quickly assign samples to specific key and velocity ranges in existing or new sampler instruments by dragging them into Sampler. You can graphically resize and move individual samples or groups of samples, as well as assign a number of playback parameters to a single sample or a group of samples.

Sampler is a flexible synthesizer in its own right. You can create expressive sounds by using any sample as a basic synthesizer waveform, which you can then process with an extensive collection of filters and modulators. If you're new to the concepts and use of synthesizers and samplers, see Synthesizer basics overview.

You can use Sampler as a mono, stereo, or surround instrument, and can route loaded samples to multiple audio outputs. This lets you independently process individual drum sounds in a drum kit, for example. If you want to play and edit a single sample instrument, try using Quick Sampler.

If you're new to using plug-ins in Logic Pro for Mac, see Add, remove, move, and copy plug-ins and Use multi-output instruments.

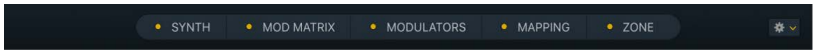


The Sampler interface is contained in a single scrolling window, with shortcut buttons shown in the Navigation bar at the top. You can show or hide sections (panes) of the interface and can use the Navigation bar buttons to resize and quickly navigate between panes.

- *Synth pane:* This pane contains synthesis options that you use to set global pitch, filter, pan and volume settings for your sounds. See Sampler Synth pane and Sampler Synth Details.
- *Mod Matrix pane:* This pane contains modulation routing options that you can configure to affect the playback of your sounds. See Use the Sampler Mod Matrix.
- *Modulators pane:* This pane contains LFO and envelope modulators that you use to control the pitch, amplitude, and filter settings of your sounds. These modulators can be routed to parameters in the Mod Matrix. See Sampler modulation overview.
- *Mapping pane:* You use this pane to create and edit sampler instruments, and to set and control group and zone parameters. See Sampler Mapping and Zone pane overview.
- *Zone pane:* You use this pane to edit individual samples (zones) within sampler instruments. See Sampler Zone pane.

Use the Navigation bar shortcut buttons

In Logic Pro, you can use the Navigation bar shortcut buttons to hide or show panes within the Sampler interface, to quickly navigate between visible panes, and to resize panes within the visible area. The Navigation bar Action pop-up menu ⚙ provides commands used to manage synthesizer parameter and mapping data.



Navigation bar parameters

- *Navigation buttons:* Use to view, and scroll to panes. Double-click to switch pane zoom levels. Click the LED to hide panes. Drag samples here to add them. You can do the following:
  - Click a navigation button to scroll the main UI to the corresponding pane.
  - Click the yellow LED inside a navigation button to hide the corresponding pane.
  - Click the navigation button of a hidden pane to show the pane and automatically scroll to it.
  - Double-click a navigation button to expand the corresponding pane vertically. The pane expands to the maximum useful pane height, in proportion to the height of other open panes and the overall height of the plug-in window.
  - Double-click the navigation button of a vertically expanded pane to reduce the pane to its default height or to a height that you have set.

    Tip: You can resize panes vertically by moving the pointer over the boundary between panes until it becomes a handle, then dragging the handle.
  - Option-click a navigation button to expand the corresponding pane vertically and hide all other panes.
- *Action pop-up menu:* Choose commands used to manage synthesizer parameter and mapping data.

  Note: You can directly drag one or more samples onto the Navigation bar to create new groups and zones. See Create zones with drag and drop.

Use the Navigation bar Action pop-up menu

In Logic Pro, you can use Navigation bar Action pop-up menu ⚙ items to perform synth parameter handling and import commands.

These commands include functions that were used in the precursor to Sampler, the EXS24 mkII. To explain, the former *EXS Instrument* is now known as a *mapping*. The former *EXS Setting* is now referred to as *synth parameters*. If you prefer to work with these elements, you can use the Navigation bar Action pop-up menu ⚙ commands.



You can choose any of the listed commands from the Navigation bar Action pop-up menu ⚙ to perform these operations:

- *Initialize Synth Parameters:* Recalls a neutral setting for all parameters in the Synth, Mod Matrix, and Modulators panes. This provides a clean slate when you are adjusting the parameters of your sampler instrument.

  Note: This does not affect the existing mapping.
- *Copy Synth Parameters without Mapping:* Copies all current parameter values from the Synth, Mod Matrix, and Modulators panes to the Clipboard.
- *Paste Synth Parameters without Mapping:* Pastes all parameter values of the Synth, Mod Matrix, and Modulators panes stored in the Clipboard into another sampler instrument.

  Note: This does not affect the mapping of the target instrument.

- *Import Synth Parameters without Mapping:* Choose an instrument to use as a source for Synth, Mod Matrix, and Modulators pane parameter values.

  Note: This does not affect the mapping of the target instrument.

- *New Mapping:* Creates a new, empty mapping that can be used as a source (template) for your sampler instruments.

  Note: This does not affect existing Synth, Mod Matrix, and Modulators pane parameter values.

- *Import Mapping without Synth Parameters:* Choose an instrument as a sample mapping data source. Synth, Mod Matrix, and Modulators pane parameter values are ignored.

  Note: This does not affect the Synth, Mod Matrix, and Modulators pane parameter values of the target instrument.

Scroll, zoom, and resize in Sampler

In Logic Pro, the overall Sampler window and panes within it can be resized. You can scroll panes vertically, and horizontally in some cases, and can zoom the Mapping and Zone panes. Combine these features and those of the Navigation bar to quickly move in, and adjust, the Sampler interface.

Important: To use gestures, such as swiping or pinching, you must turn on these mouse or trackpad features in System Settings.

Do any of the following to scroll:

- Drag vertically on the scrollbar at the right edge of the Sampler window to move between panes.

  Each pane provides one or more transparent scrollbars that you can drag to move vertically or horizontally *within* the pane.

  Note: Scrollbars are shown only when the pane or Sampler window content exceeds the display area size.

- Use a two-finger vertical swipe to scroll with the trackpad, or a single-finger vertical swipe with a Magic Mouse to vertically scroll a pane (except the Zone pane) or to scroll the Sampler window.

- In the *Mapping or Zone* panes, use a two-finger horizontal swipe with the trackpad, or a single-finger horizontal swipe with a Magic Mouse to horizontally scroll.

Do any of the following to zoom:

- In the *Mapping or Zone* panes, use a pinch gesture to zoom.

- In the *Mapping* pane, use the zoom slider.

- In the *Zone* pane, use a two-finger vertical swipe to zoom with the trackpad, or a single-finger vertical swipe with a Magic Mouse to zoom the waveform.

- In the *Zone* pane, use the zoom buttons.

  Tip: You can double-click any pane button in the Navigation bar to quickly switch between a maximized view and the current view of that pane.

Do any of the following to resize:

- Drag vertically on the top or bottom edge of the Sampler window to change the window height.
- Drag horizontally on the left or right edge of the Sampler window to change the window width.
- Drag diagonally on any corner of the Sampler window to change both the window height and width.

  A resize handle is shown when you move the pointer over any edge or corner of the Sampler window.
- Double-click the name at the top of any pane to change its height.
- Drag vertically on the top or bottom edge of a pane to change the pane height.

  A resize handle is shown when you move the pointer over the boundary between panes. The adjacent pane is also resized.

  Tip: You can double-click any pane button in the Navigation bar to quickly switch between a maximized view and the current view of that pane.

Load and save Sampler instruments in Logic Pro for Mac

A sampler instrument is the file type that is loaded into Sampler with the plug-in Settings pop-up menu. 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 can also save patches, comprising a sampler instrument and associated plug-in settings and routings, in the Library.

Load an instrument from the plug-in Settings pop-up menu

1. In the Logic Pro Sampler plug-in header, click the plug-in Settings pop-up menu.
2. Browse to the instrument family, App Presets, or other folder, then choose the sampler instrument you want to play or edit.

Load the next or previous instrument in your sampler instrument library

In Logic Pro, do one of the following:

- In the plug-in header, click the Previous and Next buttons (the arrows).
- Use the Previous Instrument or Next Instrument key command.

  If Sampler is in focus, you can also use the following key commands:

  - Previous Plug-in Setting or Instrument
  - Previous Channel Strip or Plug-in Setting or Instrument
  - Next Plug-in Setting or Instrument
  - Next Channel Strip or Plug-in Setting or Instrument

Tip: You can also browse through your sampler instruments by using your MIDI keyboard. You can assign a MIDI event, such as a MIDI note, control change, or program change to select the previous or next sampler instrument in Sampler settings.

See Sampler settings.

Filter the instrument display in the plug-in Settings pop-up menu

You can use a keyword filter to restrict the plug-in Settings pop-up menu to show only instruments that contain your search term.

1. In the Logic Pro Sampler plug-in header, click the plug-in Settings pop-up menu.
2. Choose Search Filter…

   A text entry dialog opens.
3. Enter your search term, then click OK or press Return. Enter “bass” to try this.

   The Settings pop-up menu now displays only instruments that contain your search term.
4. Click the Settings pop-up menu, then browse to the instrument family, App Presets, or other folder, and choose the sampler instrument you want to play or edit.
5. In the Settings pop-up menu, click the Enable Search Filter menu item to show all instruments.

   This retains your search term, but does not filter results.
6. Click the Enable Search Filter menu item again to turn on the search filter and find the sampler instrument.

Open Quick Sampler settings with Sampler

In Logic Pro, do one of the following:

- Choose the Quick Sampler setting name from the Sampler plug-in Settings pop-up menu. Use the Load menu item to browse to the default location: ~/Music/Audio Music Apps/Plug-In Settings/Quick Sampler/.

  The Quick Sampler sound is loaded as a group containing one or more zones. You can now edit, process, and handle the content as you would with any group or zone.
- Choose Sampler from the Instrument pop-up menu on an instrument channel strip that contains a Quick Sampler instance.

  This replaces Quick Sampler with Sampler on the channel strip, and the sound is automatically loaded as a group containing one or more zones.

  Important: You *cannot* open a Sampler setting with Quick Sampler.

Save a sampler instrument

In Logic Pro, you can access all basic sampler instrument file operations using the commands in the plug-in Settings pop-up 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, Sampler uses the existing filename and overwrites the original instrument. 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. Like the Save A Copy As command, this command may be useful when you want to save a Sampler setting (including audio data) that is unique to a specific project. Storing this in a location outside of the App Presets library or user folders may also be practical for sharing a copy of your Sampler setting (with or without audio data) with a colleague or friend.

In the Finder window, you can:

- Browse to any folder location or create a new folder.
- Click the “Save with audio data” checkbox to duplicate the samples and place them in a folder named after the Sampler setting file.

  For example, the save location is the default User Library, the instrument files (.exs) are placed in the Sampler Instruments folder. The audio files are placed in a subfolder named after the plug-in setting. This subfolder is placed in a Samples folder. The original audio files are not touched or moved.
- Click the Consolidated checkbox to consolidate the saved samples into a minimum number of audio files of the CAF file type. This option is available only when the “Save with audio data” checkbox is selected.

- *Save A Copy As:* Saves a copy of the currently loaded sampler instrument. 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.

  In the Finder window, you can:

  - Browse to any folder location or create a new folder.
  - Click the “Save A Copy with audio data” checkbox to duplicate the samples and place them in a folder named after the Sampler setting file.

    For example, the save location is the default User Library, the instrument files (.exs) are placed in the Sampler Instruments folder. The audio files are placed in a subfolder named after the plug-in setting. This subfolder is placed in a Samples folder. The original audio files are not touched or moved.
  - Click the Consolidated checkbox to consolidate the saved samples into a minimum number of audio files of the CAF file type. This option is available only when the “Save A Copy with audio data” checkbox is selected.
- *Save As Default:* Saves the currently loaded sampler instrument as the default instrument. This instrument serves as a template for future sampler instruments and is used when you create a new instrument. It is also used when you choose the Recall Default command in the plug-in Settings pop-up menu.

Save a Sampler patch in the Library

You can also save patches, comprised of a sampler instrument and associated plug-in settings and routings, in the Library.

1. In Logic Pro, click the Save button at the bottom of the Library pane. If the Library is not visible, click the Library button on the menu bar or use the default key command: Y.
2. In the Finder window, browse to the file location you want to use. The default folder location is User Patches. If you wish, create a new folder.
3. Enter the patch name, then click the Save button.

Load a Sampler patch from the Library

You can load your saved Sampler patches from the Library for the selected instrument channel.

1. In Logic Pro, open the Library pane. Default key command: Y
2. Browse to the folder, then choose the sampler patch you want to play or edit. For example, browse to User Patches > Orchestral > Strings > Dark Filter Strings.

Sampler Synthesizer parameters

Sampler Synth pane in Logic Pro for Mac

You can use the Synth parameters to control Sampler global and filter settings for the entire loaded sampler instrument. Click the Details button at the top right of the Synth pane to view or hide the Synth Details slide-out pane which provides additional parameters. Several synthesizer parameter handling commands are also found in the Navigation bar pop-up menu. See Sampler overview.

If you're new to the concepts and use of synthesizers, see Synthesizer basics overview.

You can use the Modulators pane to assign and adjust the LFOs and envelopes which control synthesizer and other parameters and can use the Mod Matrix pane to set up modulation routings. See Sampler modulation overview.

You can adjust and control individual samples (zones), or grouped samples, in the Mapping and Zone panes. See Sampler Mapping and Zone pane overview.



Synth pane parameters

Double-click a parameter knob to reset it to the default value.

Double-click a parameter value field to enter in a new value. Press Return to complete the operation.

- *Details button:* View or hide a slide-out pane which provides additional synthesis parameters. See Sampler Synth Details.
- *Tune knob and field:* Raise or lower the pitch of the sampler instrument in semitone increments. At the default (zero) position, no pitch change occurs.
- *Fine knob and field:* Tune the sampler instrument in cent increments—1/100 of a semitone. Use this parameter to correct samples that are slightly out of tune.
- *Filter On/Off buttons:* Turn each filter section on or off. Each filter can have an independent or shared envelope, defined in the Mod Matrix pane. Turning off the filter section makes it easier to hear adjustments to other sound parameters because the filter always heavily affects the sound.
- *Filter Type pop-up menu:* Choose the type and slope of the filter. See Sampler filter types.

- *Cutoff knob:* 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:* 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 the frequencies at the cutoff frequency.
  - In a highpass filter, resonance emphasizes the frequencies at the cutoff frequency.
  - In bandpass filters, resonance emphasizes the portions of the signal—the frequency band—that surround the defined frequency, set with the Cutoff knob. In Sampler, Resonance defines the width of the frequency band.
- *Drive knob:* 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. Sampler is flexible in this regard, allowing tonal colors that range from a subtle fuzz to a hard distortion.
- *Filter configuration buttons:* Switch between a series or parallel filter configuration.
  - *Series button:* Turn on to pass the signal through the first filter, which forwards this filtered signal through Filter 2. The output signal of Filter 2 is then sent to the input of the dynamic stage (Amplifier section).
  - *Parallel button:* Turn on to pass the signal through both filters. If the Filter Blend knob is set to the center position, you'll hear a 50/50 mix of the signal, routed via Filter 1 and Filter 2. The output signals of the two filters are mixed, then sent to the input of the dynamic stage (Amplifier section).

    Tip: If one of the two filters is turned off (bypassed), you can mix the original signal and the filtered signal to create a simple phaser effect, for example.
- *Filter Blend knob:* Set the balance between Filter 1 and Filter 2 in parallel mode. See Crossfade between Sampler filters.
- *Volume knob:* Set the overall output signal level of Sampler. The levels of individual samples or groups of samples are set in the Mapping and Zone panes.
- *Pan knob:* Set the pan position—or balance—of the Sampler output signal.

Create or view a modulation routing with the Synth pane shortcut menu

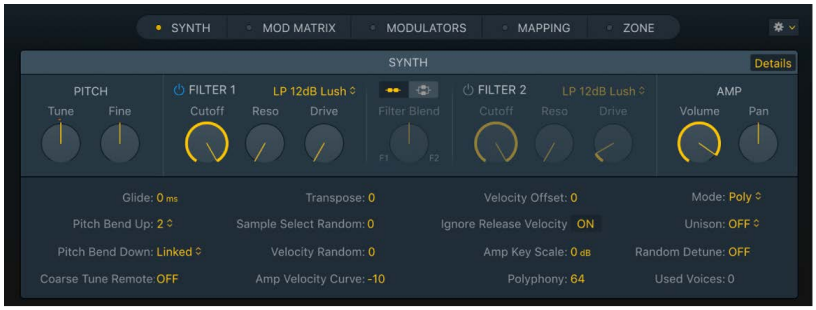
- In Logic Pro Sampler, Control-click on any parameter in the Synth pane that is available as a modulation target to open a shortcut menu, then choose one of the following:
  - *Add Modulation:* Use to create a new modulation routing in the Mod Matrix pane and choose the Source parameter from the pop-up menu.
  - *Show Modulation:* Use to highlight all existing modulation routings that use the parameter by activating the view filter in the Mod Matrix pane. See Use the Sampler Mod Matrix.

Sampler Synth Details in Logic Pro for Mac

You can use the Synth and Synth Details parameters to affect the overall behavior of the loaded instrument. If you're new to the concepts and use of synthesizers and samplers, see Synthesizer basics overview.

Click the Details button to view or hide the Synth Details slide-out pane.

- Drag vertically in parameter value fields to change them.
- Option-click a parameter value to reset it to the default value.
- Double-click a parameter value field to enter a new value. Press Enter to complete the operation.



Synth Details parameters

- *Glide field:* Set the time it takes to slide from one note pitch to another—the portamento time.

  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. See Use Sampler Unison, Mode, and Polyphony parameters.

- *Pitch Bend Up/Down pop-up menus:* Set the upper and lower limit of pitch bends in semitones. A value of 0 prevents 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).

- *Coarse Tune Remote pop-up menu:* Choose a note value that sets the center key for keyboard-controlled transposition. The range is an octave above and below this key. Play a note in this range to transpose the mapping in semitones. This is much like using the Tune parameter.
- *Transpose field:* Use to transpose the MIDI input value in semitone increments.
- *Sample Select Random field:* Set the range of randomization values for Sample Select modulation. This is useful for instruments with multiple layered zones as it can emulate the subtle tonal fluctuations caused by changes in playing technique. You can also use it to switch wildly between samples for more extreme sonic results.
- *Velocity Random field:* Set the range of randomization values for velocity modulation. This is useful for emulating tonal variations in instruments when played, struck, or blown at different levels.
- *Amp Velocity Curve field:* Use to determine how Sampler responds to incoming velocity values. Negative values increase the responsiveness to soft key strikes, and positive values decrease responsiveness.
- *Velocity Offset field:* Use to increase or decrease incoming MIDI note velocity values, expanding or limiting the dynamic response to MIDI note events.
- *Ignore Release Velocity button:* Turn on to play samples at a note-off velocity level that is equal to the note-on velocity level. Turn off to control the note-off velocity level with a suitably-equipped MIDI keyboard. If your keyboard cannot send note-off velocity data, Logic Pro ignores the Ignore Release Velocity setting, and the note-off velocity level is *always* equal to the note-on velocity level.

  Note: This parameter applies to the release trigger function of the Gigasampler format, and should be turned on for this purpose.
- *Amp Key Scale field:* Set the amount of level modulation by keyboard position. 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 lower-pitched notes are often louder than high notes.
- *Polyphony field:* Set the maximum number of keys that can be played simultaneously. See Use Sampler Unison, Mode, and Polyphony parameters.
- *Mode pop-up menu:* Switch between legato, monophonic, and polyphonic behaviors.
- *Unison pop-up menu:* Set the number of unison voices or turn unison mode off.
- *Random Detune field:* 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 Detune is also effective when you are emulating various stringed instruments.
- *Used Voices field:* Shows the number of playing voices in real time.

Sampler Unison, Mode, and Polyphony in Logic Pro for Mac

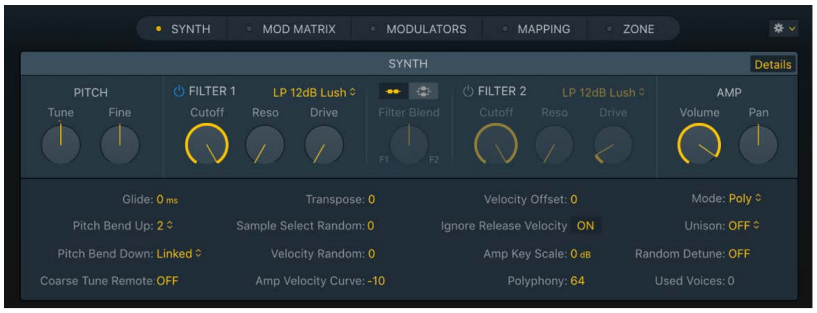
You will find all parameters discussed in this section in the Synth Details slide-out pane. Click the Details button at the top-right of the Synth pane if it is not visible.

In Unison mode, multiple voices are played when you play a key. You can use this to achieve a richer sound, particularly when you also use the Random Detune parameter to slightly detune each voice. This is ideal when you are emulating classic analog synthesizers, and is also useful for stringed instruments.

You can use the Mode pop-up menu to determine the keyboard mode. This changes the way Sampler responds when you play one or multiple notes on your keyboard.

Voices are equally distributed in the panorama field and are evenly detuned. You can use the Random Detune field to determine the amount of tuning deviation between voices.

Note: The number of voices actually used for each note you play increases with the number of layered sample zones. For example, a sampler instrument that contains four layered zones will use four voices when you play a key. This is shown in the Used Voices field in real time.



Change the keyboard mode

A polyphonic instrument, such as an organ or piano, lets you play several notes simultaneously. Brass or reed instruments are monophonic, which means that you can play only one note at a time. You can choose an appropriate keyboard mode for the type of instrument that is loaded. You are also free to use a monophonic mode for polyphonic instruments such as pianos or guitars, which allows playing styles that are not possible with these polyphonic instruments.

In the Synth Details slide-out pane of Logic Pro Sampler:

- Choose Legato, Mono, or Poly from the Mode pop-up menu.
  - 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, sample playback and envelopes are always retriggered regardless if you play staccato, or in a legato style—play a new key while holding another.
  - In Poly mode, you can play several notes 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 monophonic unison mode

In the Synth Details slide-out pane of Logic Pro Sampler:

1. Choose Mono or Legato from the Mode pop-up menu, depending on the keyboard mode you want to use.
2. Set the number of unison voices with the Unison pop-up menu.

   The intensity of detuning—voice deviation—depends on the value you set with the Random Detune field. See Sampler Synth Details.

Use polyphonic unison mode

In the Synth Details slide-out pane of Logic Pro Sampler:

1. Choose Poly from the Mode pop-up menu.
2. Set the number of unison voices with the Unison pop-up menu.

   In Poly/Unison mode, each note you play is effectively doubled. These two voices are heard when you play each note.

Set the number of keys that can be played simultaneously

In the Synth Details slide-out pane of Logic Pro Sampler:

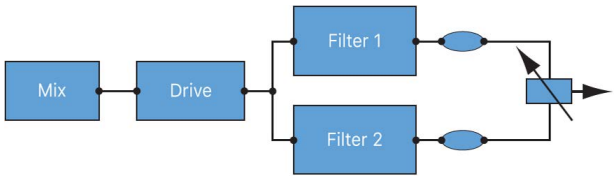
- Drag vertically in the Polyphony field to determine the maximum number of notes that you can play.

  When you play the keyboard, the Used Voices field shows the number of voices in use. The number of voices actually used for each note you play increases with the number of layered sample zones.

Crossfade between Sampler filters in Logic Pro for Mac

You can use Filter Blend to affect Sampler signal flow. The figure illustrates the signal flow between the signal mix stage (pre-Filter) 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. See Sampler Synth pane.

In a parallel configuration, the overdrive/distortion circuit—the Drive parameter—is always wired *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 Mod Matrix pane. 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, using an LFO or velocity, or both, as a modulation source.

Crossfade between active parallel filters

- In the Synth pane of Logic Pro Sampler, rotate the Filter Blend knob to crossfade between the two filters when they are cabled in parallel.

  Note: Both filters must be turned on. If only one filter is active, you will hear a crossfaded mix of the original signal and the filtered signal.

  - Set Filter Blend to the leftmost position to hear only the effect of Filter 1.
  - Set Filter Blend to the rightmost position to hear only the effect of Filter 2.
  - In between these positions, the filters are crossfaded. You hear the effect of both filters.

Modulate Filter Blend with an LFO

1. In the Mod Matrix pane of Logic Pro Sampler, click the Plus button (+) to create a new modulation routing.
2. In the new modulation routing, choose Filter Blend in the Target pop-up menu, and LFO 2 in the Source pop-up menu.
3. Adjust the settings of LFO 2.

   Note: Any LFO can be used as the modulation source.

Sampler filter types in Logic Pro for Mac

Sampler provides multiple filter types you can choose in the Synth pane. Each filter type has unique sonic characteristics and responds differently to incoming signals. You can use this flexible collection of filters to subtly or dramatically sculpt the tone of your instruments. If you're new to the concepts and use of synthesizer filters, see Filters overview.

Sampler filters

- *Lowpass (LP):* This filter type passes the portion of a signal below a specified cutoff frequency and rolls off the portion above that frequency.
- *Bandpass (BP):* This filter type passes the portion of a signal occupying a band surrounding the cutoff frequency and rolls off the portions above and below that band.
- *Highpass (HP):* This filter type passes the portion of a signal above a specified cutoff frequency and rolls off the portion below that frequency.
- *Band reject (BR):* This filter type cuts a narrow band around a resonant frequency. The remainder of the signal is affected minimally.
- *Peaking:* This filter type boosts a narrow band around a resonant frequency. The remainder of the signal is affected minimally.

There are multiple two-pole, four-pole, multi-pole state-variable and analog-modeled LP, BP, and HP filter designs in Sampler, each with distinctive characteristics that you may prefer for a given purpose. The available LP, BP, and HP filter designs include Creamy, Edgy, Gritty, Lush, Lush (Fat), and Sharp variants.

- *Creamy:* Multi-pole state-variable filter.
- *Edgy:* Two- or four-pole analog-modeled filters. Four-pole variants provide a steeper roll-off of frequencies beyond the cutoff.
- *Gritty:* Two-pole filters designed to saturate heavily at higher Resonance and Drive settings.

- *Lush:* Four-pole lowpass analog-modeled filters.
- *Lush (Fat):* Four-pole analog-modeled filters.
- *Sharp:* Two-pole analog-modeled filters.

The three principal filter controls have standard functions for all filter types.

- *Cutoff:* Controls the filter cutoff frequency.
- *Resonance:* Controls the filter resonance or emphasis. Higher settings boost frequencies in the immediate vicinity of the cutoff frequency.
- *Drive:* Allows the filter to be overdriven; the precise effect varies with each filter design.

Sampler Modulation

Sampler modulation overview in Logic Pro for Mac

In Sampler, you can use the extensive range of modulation Sources and Targets to generate extraordinary sounds that constantly evolve, or are just plain expressive to play. You can use modulators to more accurately emulate different playing techniques of various instruments. As examples, emulating the change in embouchure of a brass player, or introducing a delayed vibrato for a stringed instrument.

If you're new to the concepts and use of synthesizer modulators, see Modulation overview.



You can access all Sampler modulation parameters and functions in two panes.

- *Mod Matrix pane:* Use to create a modulation routing where you can link a modulation Source, such as an envelope, to a modulation Target, such as the filter. You can use up to twenty modulation routings in your sampler instruments. See Use the Sampler Mod Matrix.
- *Modulators pane:* You can use the modulators, which include the LFOs and envelopes, to control your sounds. See Sampler LFO overview and Sampler envelopes.

Sampler Mod Matrix pane

Use the Sampler Mod Matrix in Logic Pro for Mac

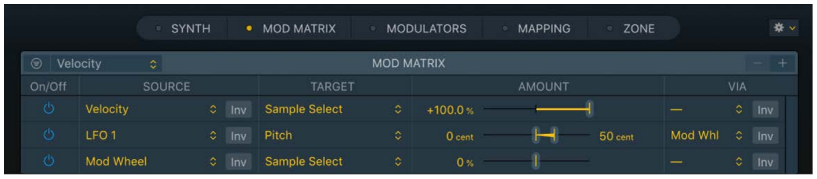
You can use the Mod Matrix pane to route any modulation Source to any modulation Target—much like the connections made in an old-fashioned telephone exchange or a studio patch bay. If you're new to synthesizer modulation routings, see Modulation basics.

You can use up to twenty modulation routings of Source, Via, and Target—arranged in horizontal rows—simultaneously. You can select the same target in several parallel modulation routings, and you can also use the same Sources and the same Via controllers in multiple modulation routings of different Targets.

You can set the modulation intensity—how strongly the target parameter is influenced by the source parameter—with the Amount slider to the right of the modulation routing Source and Target.

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

Details on available Sources, Targets, and Via sources are found in Sampler modulation Sources, Sampler modulation Targets, and Sampler modulation Via sources.



Mod Matrix parameters

Double-click a parameter value field to enter in a new value. Press Return to complete the operation.

- *Filter On/Off button:* Turn on the Mod Matrix view filter to restrict the display of modulation routings to match the criteria set in the Filter pop-up menu.

  Note: This is a visual filter only. It does *not* remove or block non-matching modulation routings.

- *Filter pop-up menu:* Choose a filter criteria to limit the display of modulation routings. Only routings that match this criteria are shown when the (view) Filter button is active.
  - *Filter by Source/Via:* Only routings that contain the matching Source or Via source are shown.
  - *Filter by Target:* Only routings that contain the matching Target are shown.
  - *Always select last clicked:* The filter criteria is automatically set to show only routings that contain the matching Source, Via source, or Target of the most recently clicked parameter.

    Note: Only parameters that can be selected as Sources, Via sources, or Targets in the Mod Matrix pane can be clicked.
- *-/+ buttons:* Delete or add a modulation routing.
- *Column sort buttons:* Use to alphabetically sort all visible Source, Target, or Via routings. Click again to reverse the sort order.

- *On/Off button:* Turn on the modulation routing, or turn off to bypass it, without losing settings.
- *Source pop-up menu:* Choose the parameter you want to use as the modulator of the Target parameter.
- *Inv button:* Invert the effect of the modulation Source. A negative value becomes positive, and vice versa.
- *Target pop-up menu:* Choose the parameter you want to modulate.
- *(Max) Amount parameter:* Drag to set the maximum modulation intensity when a Via source is active.
- *(Min) Amount parameter:* Set the modulation intensity. When a Via source is active, this field sets the minimum modulation intensity with the Via controller set to its minimum value.
- *Amount parameter:* Drag to set the modulation intensity. Drag either handle when a Via source is active. Drag the area between handles to move both.
- *Via pop-up menu:* Via defines a further modulation source, which is used to control the modulation intensity. See Use Via sources in Sampler.
- *Inv button:* Invert the effect of the Via modulation source. A negative value becomes positive, and vice versa.

The following tasks apply to all modulation routings.

Create a basic modulation routing

1. In the Logic Pro Sampler Mod Matrix pane, click the Plus button (+) at the top right to create a new modulation routing.

   Tip: You can Control-click any target parameter in the Synth pane to open a shortcut menu where you can directly add a modulation source. This creates a new modulation routing with the chosen source and target parameters in the Mod Matrix.

2. Choose the parameter you want to modulate from the Target pop-up menu.
3. Choose the parameter you want to use as the modulator of the target from the Source pop-up menu.

   Note: Sampler will automatically create a corresponding Envelope or LFO if it does not exist in the Modulators pane. For example, choosing LFO 3 from the Source or Target pop-up menu will create LFO 3 in the Modulators pane.

4. Drag the Amount slider handle to set a fixed modulation intensity.

Bypass a modulation routing

- In the Logic Pro Sampler Mod Matrix pane, click the On/Off button to turn off a modulation routing.

  This bypasses individual modulation routings without losing settings.

Remove a modulation routing

1. In the Logic Pro Sampler Mod Matrix pane, click the modulation routing you want to remove.
2. Click the Minus button (—) at the top right of the Mod Matrix pane to delete the modulation routing.

   This completely removes the routing from the Mod Matrix pane.

Use the Mod Matrix view filter

The Mod Matrix view filter is useful when setting up new modulation routings and when dealing with complex instrument routings. You can use it to quickly find all routings that use an LFO, Velocity, or other criteria.

1. In Logic Pro Sampler, click the Filter On/Off button at the top left of the Mod Matrix pane to activate view filtering.
2. Choose the modulation routing criteria from the Filter pop-up menu at the top left of the Mod Matrix pane.
   - Only routings which contain Targets or Sources that match the criteria are shown in the Mod Matrix pane.
   - If you choose the Select last clicked entry, the view filter follows the most recently edited or clicked source or target parameter, and displays routings which use this Source or Target in the Mod Matrix pane.
   - When the view filter is active, adding a modulation routing automatically adds a Source or Target that matches the current view filter criteria. For example, if the view filter criteria is LFO 1, creation of a new modulation routing will automatically add LFO 1 as a source.

   Tip: You can Control-click any target parameter in the Synth pane to open a shortcut menu where you can choose the View Modulation command. This chooses the parameter as the modulation routing criteria from the Filter pop-up menu and turns on the view filter in the Mod Matrix pane.

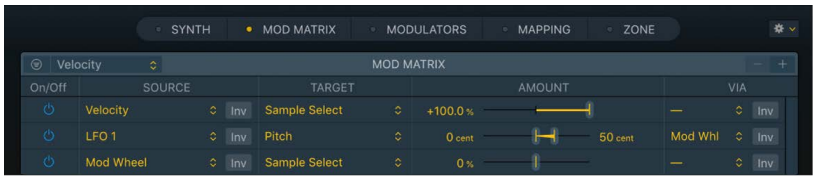
Use Sampler Mod Matrix Via sources in Logic Pro for Mac

You can set a fixed modulation intensity by horizontally dragging the Amount slider handle in a basic modulation routing—consisting of a Target and Source. The Amount slider value always defines a constant modulation intensity.

For further control, you can modulate the amount of modulation by using the Via pop-up menu to define a further modulation source that controls the modulation intensity.

If you choose a value other than None from the Via pop-up menu, the Amount slider displays two handles.

- The left Amount slider handle (Min Amount) defines the minimum modulation intensity, when the assigned Via controller—the modulation wheel, for example—is set to its minimum value.
- The right Amount slider handle (Max Amount) defines the maximum modulation intensity when the assigned Via controller is set to its maximum value.
- The area between the two Amount slider handles defines the modulation range of the Via controller.



Create a modulation routing that includes a Via source

1. In the Logic Pro Sampler Mod Matrix pane, click the + button to create a new modulation routing.
2. Choose the target parameter you want to modulate from the Target pop-up menu.
3. Choose a modulation source from the Source pop-up menu.

   Note: Sampler will automatically create a corresponding Envelope or LFO if it does not exist in the Modulators pane. For example, choosing LFO 3 from the Source or Target pop-up menu will create LFO 3 in the Modulators pane.
4. Choose the modulation source that you want to use for control of modulation intensity from the Via pop-up menu.

   A second slider handle is created and is placed on top of the Amount slider handle.
5. Move the pointer to the right of the overlapped slider handles, then drag toward the right to set the maximum modulation intensity.

   You will see the second Amount slider handle (Max Amount) and a range appear as you drag.
6. Drag the left Amount slider handle (Min Amount) to set the minimum modulation intensity.

Move the entire modulation range

- In the Logic Pro Sampler Mod Matrix pane, horizontally drag the area between the Min and Max Amount slider handles (the range) in a modulation routing that contains an active Via source.

  Both handles move simultaneously.

  Tip: If the area between the handles is too small to grab, adjust the Amount and Via Amount slider value fields by the same number to move the area.

Set the modulation amount to zero

In the Logic Pro Sampler Mod Matrix pane, you can set the modulation amount to zero in the following ways.

- When using a modulation routing with no active Via source in the Mod Matrix pane:
  - Double-click anywhere on the Amount slider.
- When using a modulation routing with a Via source:
  - Double-click to the left of the Amount slider handle (Min Amount) to set the minimum modulation amount value to zero.
  - Double-click to the right of the second Amount slider handle (Max Amount) to set the maximum modulation amount value to zero.

Invert the effect of the Via modulation source

- In the Logic Pro Sampler Mod Matrix pane, click the Via Inv button to invert the effect of the Via modulation source.

  A negative value becomes positive, and vice versa.

Sampler LFOs

Sampler LFO overview in Logic Pro for Mac

You can use any of the four identical LFOs (low frequency oscillators) shown in the Modulators pane to control and add animation to your instruments. All LFOs are available as sources and targets in the Mod Matrix pane. If you're new to synthesizers and the concept behind LFOs, see Common modulation sources.

You can use all Sampler LFOs polyphonically or monophonically. When used polyphonically, this means that modulation of multiple voices is *not* phase-locked. The LFOs can also be key-synced—each time you play a key, LFO modulation of this voice is started from zero.

To understand the non-phase-locked characteristic more fully, imagine playing a chord on your keyboard. If an LFO 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 you play.

When you use LFOs monophonically, modulation is identical for all voices. For example, imagine a chord you are playing on the keyboard is using LFO 2 to modulate pitch, which you set up in the Mod Matrix. In this situation, the pitch of all voices in your played chord rise and fall synchronously.

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

You can set all LFOs to either oscillate freely or to be synchronized with the Logic Pro tempo, in values ranging between 32 bars and 1/128th triplets.

You can fade LFOs in or out automatically with the built-in ramp generator available in each LFO.



LFO parameters

Double-click a parameter value field to enter a new value. Press Return to complete the operation.

- *Minus button (—):* Highlight the modulator you want to remove, then click the button to delete the modulator from the Modulators pane.
- *+ LFO button:* Add an LFO to the Modulators pane. Up to four LFOs can be used.
- *Waveform pop-up menu:* Choose the LFO waveform. For details about how to use them, see Sampler LFO waveforms.

- *Sync button:* Turn on to synchronize the LFO with the project tempo. The note icon is illuminated when active.
- *Rate field:* Use to set the frequency, or speed, of LFO modulation. The value is shown in note values when the Sync button is active. See Set the Sampler LFO rate.
- *Fade field:* Use to set the time it takes for LFO modulation to fade in or fade out, depending on which Fade button is active. See Use the Sampler LFO ramp generator.
- *Fade In or Fade Out button:* Click to set an LFO fade-in or fade-out. Use the Fade field to set the fade duration.
- *Waveform display:* Shows the current LFO waveform. Drag horizontally to set the frequency, or speed, of LFO modulation.
- *Phase field:* Drag vertically to offset the waveform start point.
- *Mono or Poly button:* Set the LFO to run monophonically or polyphonically. Mono results in identical modulation of all voices. Poly provides independent modulation of each voice.
- *Unipolar or Bipolar button:* Run the LFO waveform in one pole (positive) or two pole (positive and negative) mode.
- *Key Trigger button:* Turn on key sync for the LFO. Each time you play a key, the LFO waveform is restarted from the start point.

Sampler LFO waveforms in Logic Pro for Mac

You can use the LFO Waveform pop-up menu to choose different LFO waveforms. The table outlines how these waveforms can affect your sounds. See Sampler LFO overview.

Tip: Try different waveforms while a modulation routing of Pitch is engaged and running to better understand how they affect the sound.

Waveform	Comments
Sine	Well-suited for vibrato effects.
Triangle	Well-suited for subtle modulations, particularly on pad, string, and other sustained sounds.
Sawtooth	Well-suited for helicopter and space gun sounds. Intense modulations of pitch with a bipolar sawtooth wave lead to bubbling sounds. Intense sawtooth modulations of lowpass filter cutoff and resonance create rhythmic effects. Interesting results can be generated by changing the start point of the modulation cycle with the Phase field.
Square	Use of square/rectangle waves periodically switches the LFO between two values. The upper wave cycle switches between a positive value and 0. The lower wave cycle 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 target with a suitable modulation intensity that leads to an interval of a fifth.

Waveform	Comments
Random	The two Random waveforms output random values. A random value is selected at regular intervals, as defined by the LFO Rate parameter. The Random waveform steps between randomized values (rapid switches between values). The Smoothed Random wave moves more fluidly between values. *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.

Sampler LFO ramp generator in Logic Pro for Mac

You can use the Sampler LFO ramp generator to set the time it takes for the LFO modulation to fade in or fade out. When you set the Fade field to a value of zero no fade in or fade out occurs.

Set the LFO modulation fade time

In Logic Pro Sampler:

- *To fade in the modulation:* Click the Fade In button, then vertically drag the Fade field to a value greater than zero.

  The higher the value, the longer the delay time before the LFO modulation starts.

- *To fade out the modulation:* Click the Fade Out button, then vertically drag the Fade field to a value greater than zero.

  The higher the value, the longer the fade out time for the LFO modulation.

Set up a delayed vibrato

You may often use the LFO ramp generators to introduce a delayed vibrato; many instrumentalists and singers intonate longer notes this way.

1. In Logic Pro Sampler, click the Mod Matrix button in the Navigation bar to view the Mod Matrix pane.
2. Click the + button to create a new modulation routing.
3. In the modulation routing, choose Pitch in the Target pop-up menu and choose LFO 1 in the Source pop-up menu.
4. Set a slight modulation intensity with the Amount slider.
5. In the Modulators pane, set an LFO 1 Rate of about 5 Hz.
6. Choose Triangle in the LFO 1 Waveform pop-up menu.

   Tip: Chaotic and fast modulations of frequencies (choose Pitch from the Target pop-up menu) by any LFO source—with a delayed Random waveform, a high Rate, and short fade-out time—are ideal for emulating the attack phase of brass instruments.

7. Click the Fade In button, then vertically drag the Fade field to a value greater than zero.

Set the Sampler LFO rate in Logic Pro for Mac

LFOs, particularly when you use them monophonically, are well suited for creating rhythmic modulation effects—effects that retain perfect synchronicity even during project tempo changes. You will find that the triangle waveform is most suitable for adding vibrato to a sound, and when using an LFO as a modulation source for other LFOs. You are, of course, free to use any LFO waveform. See Sampler LFO waveforms.

Use the Rate parameter to set the respective LFO to run freely at any speed or to synchronize it with the project tempo when the Sync button is turned on.

Set a free LFO rate

In Logic Pro Sampler, click the Modulators button in the Navigation bar to view the Modulators pane.

- In the LFO you want to use, make sure the Sync button is turned off, then drag the Rate field vertically or the Waveform display horizontally to set a value.

  The rate is displayed in hertz.

Synchronize the LFO rate with the Logic Pro tempo

In Logic Pro Sampler, click the Modulators button in the Navigation bar to view the Modulators pane.

- In the LFO you want to use, make sure the Sync button is turned on, then drag the Rate field vertically or the Waveform display horizontally to set a value.

  The rate is displayed in rhythmic values. Synchronized rates range from speeds of 1/128th triplets to a periodic duration of 32 bars. Punctuated values are also available.

Use Sampler envelopes in Logic Pro for Mac

You can use up to five of Sampler's identical envelope generators per voice. These envelopes are shown in the Modulators pane and are available as targets and sources in the Mod Matrix pane. For details on the roots of the term *envelope generator* and its basic function, see Amplifier envelope overview.

ENV 1 is hard-wired to Amplitude and defines changes in level over time—for each note that you play. You can also use ENV 1 as a source for other modulation targets in the Mod Matrix pane.

Envelope time parameters—attack, decay, release, and so on—are also available as modulation targets in the Mod Matrix pane. See Sampler modulation Targets.



Envelope parameters

Double-click a parameter value field to enter a new value. Press Return to complete the operation.

- *Minus button (—):* Highlight the modulator you want to remove, then click the button to delete the modulator from the Modulators pane.

  Note: ENV 1 can *not* be removed from the Modulators pane.

- *+ ENV button:* Add an envelope to the Modulators pane. Up to five envelopes can be used. ENV 1 is hard wired to amplitude and is always visible.
- *Envelope display:* Use to control your instruments over time. Drag points (handles) or lines in the display, or drag vertically in fields, to adjust envelope parameter values.
- *Envelope Type pop-up menu:* Choose the envelope shape that you want to use. This changes the appearance and available parameters shown in the envelope display. The envelope types shown are abbreviations for the controls in the envelope shape: AR is an envelope that provides Attack and Release controls. AHDSR provides Attack, Hold, Decay, Sustain, and Release parameters.
- *Delay field:* Drag vertically to set the time before the envelope attack phase begins. This setting is useful for sounds such as tambourines which have a noise element that precedes the shake or hit.
- *Attack handle and field:* Drag the handle horizontally to set the time it takes for the envelope to reach the initial level. Drag the field vertically.
- *Curve and field:* Drag to set the shape of the envelope attack curve. Move the pointer over the attack portion (the line) of the graphical envelope display, then drag to adjust the curve.
- *Hold handle and field:* Drag the handle horizontally to set the time that the full level is held, following the attack phase, before the decay phase begins. Drag the field vertically.
- *Decay field:* Drag vertically to set the time it takes for the envelope to fall to the sustain level, following the hold phase or the initial attack time.
  - Drag the Decay/Sustain handle horizontally to set the time it takes for the envelope to fall to the sustain level, following the attack/hold phase. Drag vertically to set the sustain level, which is held until you release the key.
  - 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.

    Note: If you release the key before the decay phase has finished, the envelope moves immediately to the release phase.
- *Sustain field:* Drag vertically to set the sustain level, which is held until you release the key.
- *Release handle and field:* Drag the handle horizontally to set the time it takes for the envelope to fall from the sustain level to a level of zero. Drag the field vertically.
- *Vel slider:* Set the sensitivity of amplitude envelope modulation in response to incoming velocity data.

- If the Vel slider is set to zero, the envelope outputs its maximum level when you strike the keys at any velocity.
- At a Vel slider value of 100%, the entire dynamic range is under velocity control.

  To explain, raising the slider value reduces the envelope minimum amplitude, with the difference being dynamically controlled by keyboard velocity. For example, when you set the Vel slider to 25%, the minimum envelope amplitude is reduced to 75%. The remaining 25% is added in response to the velocities of keys you play. So, a key played with a zero velocity results in an envelope amplitude of 75%. A key played with a MIDI velocity value of 127 will result in an envelope amplitude of 100%. When you raise the Vel slider value, the minimum amplitude decreases even further.

Sampler modulation reference

Sampler modulation Targets in Logic Pro for Mac

The table outlines targets you can choose in the Target pop-up menus of the Mod Matrix pane which can be modulated in real time. Also see Sampler modulation Sources and Sampler modulation Via sources.

Target	Comments
None	No target is chosen.
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 target 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 in the Mapping and Zone panes 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. In cases where you're using multiple Xfades, Sampler needs to be prepared to play any velocity layer in parallel, in order to perform a real time crossfade. To allow for smooth crossfades, this means that all velocity layers need to be active.
Bit Resolution	Modulates the bit depth. This lets you create lo-fi sound effects. For example, using an envelope to make a sustained sound break up over time.
Sample Start/End	Modulates the sample start or end time. This allows you to trigger a drum loop partway through, for example. Modulation of both could create interesting chopping or semi gated effects.
Loop Start/End	Modulates the loop start or end time. This allows you to cycle different sized portions of a drum loop, for example.
Loop Position	Modulates the loop position. This moves the entire looped portion of the sample without changing the loop length.

Target	Comments
Flex Speed	Modulates the Flex Speed parameter value, when Flex is turned on. You could use this to double or halve the synchronized playback speed of audio, allowing creative rhythmic effects.
Pitch	Modulates the frequency (pitch) of the loaded sampler instrument. 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. Slight envelope modulations can make the amount of detuning change over time, which can be particularly useful for brass sounds.
Glide Time	Modulates the duration of the Glide (portamento) effect. If you modulate Glide with Velocity, keystrike speed sets the time required for played notes to reach the target pitch.
Filter Cutoff	Modulates the Cutoff Frequency parameter independently for each filter.
Filter Resonance	Modulates the Resonance parameter independently for each filter.
Filter Drive	Modulates the filter Drive parameter independently for each filter.
Filter Blend	Modulates the Filter Blend parameter which changes the balance between the filter units. See Crossfade between Sampler filters.
Volume	Controls the main output level.
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.
LFO Rate	Modulates the LFO frequency (rate). For example, you can automatically accelerate or slow down the LFO 1 Rate by modulating this target with one of the envelope generators or another LFO.
LFO Fade	Controls the LFO Ramp Generator parameter. See Use the Sampler LFO ramp generator.
LFO Phase	Controls the LFO waveform start position.
Env Delay	Modulates the Delay (pre Attack phase) time of the envelope.
Env Attack	Modulates the Attack time of the envelope.
Env Attack Curve	Modulates the Attack curve shape of the envelope.
Env Hold	Modulates the Hold time of the envelope.
Env Decay	Modulates the Decay time of the envelope.
Env Release	Modulates the Release time of the envelope.
Time	Scale (lengthen or shorten) the envelope time intervals, based on key position. Note position C3 is the center point. Time intervals for zones assigned to keys above C3 can be made shorter with the Amount slider. Time intervals for zones assigned to keys below C3 can be made longer.
All Env Time Stages	Simultaneously modulate all envelope time parameters, as applicable to the chosen envelope type: Delay, Attack, Hold, Decay, Release.
Hold	Modulates the controller assigned to the sustain pedal function.
Articulation ID	Modulates the Articulation ID. See Sampler articulation handling.

Sampler modulation Sources in Logic Pro for Mac

The table outlines sources you can choose in the Source pop-up menus of the Mod Matrix pane that can be used for real-time modulation of Targets. Also see Sampler modulation Targets and Sampler modulation Via sources.

Source	Comments
None	No source is chosen.
Side Chain	Side Chain modulation uses the side-chain signal level as a modulation signal. The side-chain source is chosen from the Side Chain pop-up menu in the plug-in window header. 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.
Random	A random value generator is used as a source. *Note:* The random value updates each time a MIDI note-on message is received.
LFO 1 - 4	The LFO you choose is used as a source.
Envelope 1 - 5	The Envelope Generator you choose is used as a source.
Key	This source outputs keyboard position (MIDI note number). The default center point is C3. A value of –100 indicates five octaves below C3. A value of +100 indicates five octaves above. Modulate the Cutoff target 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 Amount value of zero proportionately scales cutoff frequencies with keyboard note pitches.
Velocity	Velocity serves as a modulation source.
Release Velocity	The modulation occurs when you release a key (this requires a keyboard that sends release velocity information).
Pitch Bend	The pitch bend wheel is used as a modulation source.
Aftertouch	Aftertouch (also known as keyboard pressure) serves as a modulation source. Sampler reacts to mono and poly aftertouch. *Note:* If you set the target 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.
Mod Wheel	Your keyboard Modulation Wheel controller is used as a modulation source.
MIDI Controllers 2–119	The MIDI controller you choose serves as a modulation source. Logic Pro uses controllers 7 and 10 for volume and pan control 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.

Sampler modulation Via sources in Logic Pro for Mac

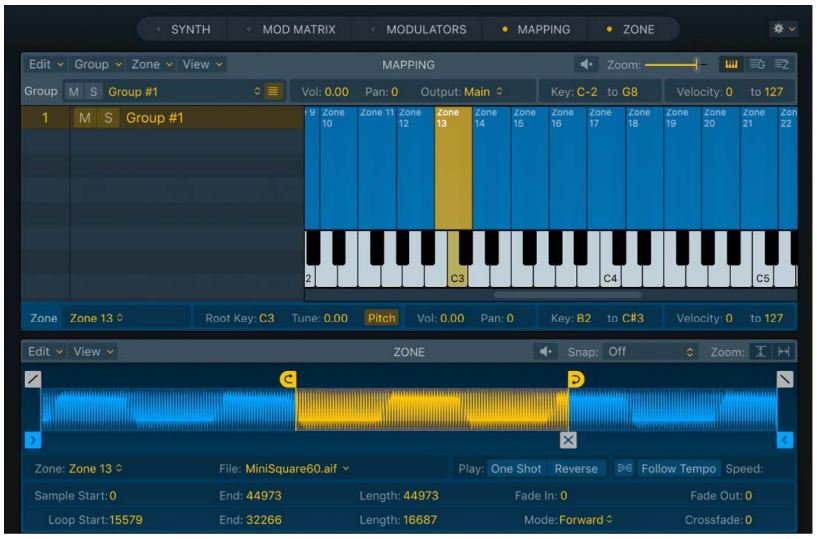
The table outlines sources you can choose in the Via pop-up menus of the Mod Matrix pane that can be used to control the modulation intensity. Also see Sampler modulation Sources and Sampler modulation Targets.

Via Source	Comments
None	No via source is chosen.
Side Chain	Uses the side-chain signal as a modulation signal. Choose the side-chain source from the Side Chain pop-up menu in the plug-in window header. 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 via 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.
Random	A random value is used as a via source.
LFO 1 - 4	Choose the LFO you want to use as a via source.
Envelope 1 - 5	Choose the Envelope Generator you want to use as a via source.
Key	Outputs the keyboard position (the MIDI note number). The default center point is C3. A value of -100 indicates five octaves below C3. A value of +100 indicates five octaves above. Modulate the Cutoff target 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 Amount value of 50 proportionately scales cutoff frequencies (by half) with keyboard note pitches.
Velocity	Velocity serves as a via modulation source.
Release Velocity	The via modulation occurs when you release a key (this requires a keyboard that sends release velocity information).
Pitch Bend	The pitch bend wheel is used as a via modulation source.
Aftertouch	Aftertouch (also known as keyboard pressure) serves as a via modulation source. Sampler reacts to mono and poly aftertouch. *Note:* If you set the target 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.
Mod Wheel	Your keyboard Modulation Wheel controller is used as a via modulation source.
MIDI Controllers 2–119	The MIDI controller you choose serves as a via modulation source. Controllers 7 and 10 are used for volume and pan control 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.

Sampler Edit panes

Sampler Mapping and Zone panes in Logic Pro for Mac

You can use the Mapping and Zone panes to create and edit sampler instruments. The two panes are designed for different tasks, but work together to provide full control of individual samples, or *zones*, and their use within one or more collections of zones, known as *groups*.



A sampler instrument consists of zones and groups:

- A *zone* is a location into which a single sample (an audio file) is loaded from hard disk. You can numerically edit *all* zone parameters in the Zone view. You can also edit basic zone parameters such as start, fade, and loop marker positions directly in the waveform display of the Zone pane.
- Zones are assigned to *groups*, which provide parameters that you can use to simultaneously affect all zones contained in the group. You can define as many groups as required. You can numerically edit group parameters in the Group view.
- You can easily assign zones to keyboard notes or note ranges in the Key Mapping Editor. In addition, you can graphically assign zones to velocity ranges in the Key Mapping Editor, making it easy to create multi layered sampler instruments.

See Create instruments with Sampler, Create Sampler zones, Create Sampler groups, and Sampler zone and group edits.

Switch between Mapping pane editors

The Mapping pane has three edit windows or views you can choose by clicking the buttons at the top right.

- *Key Mapping Editor button:* Shows groups to the left, with zones arranged graphically across a keyboard in the Key Mapping Editor. See Use Sampler Key Mapping Editor.
- *Group view button:* Shows groups and associated parameters in a column layout. See Use Sampler Group view.
- *Zone view button:* Shows groups to the left, with zones and associated parameters in a column layout. See Use Sampler Zone view.
- *Menu bar:* The menu bar at the top of the Mapping pane contains elements that are common to all editor views. See Mapping pane menu bar.

Sampler Key Mapping Editor in Logic Pro for Mac

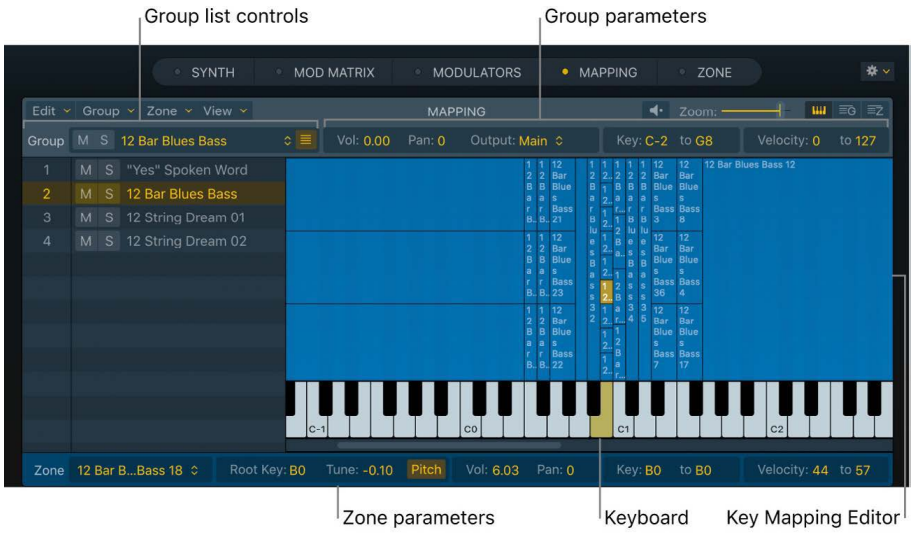
You can choose from three editor views in the Mapping pane: Key Mapping Editor, Group view, and Zone view. Click the buttons at the top right to switch between these edit panes. The menu bar at the top of the Mapping pane contains elements that are common to all editors. See Mapping pane menu bar.

The Key Mapping Editor provides you with a graphical, grid-like representation of zones, mapped across a keyboard. You can view one or more selected groups, and move and resize all zones horizontally and vertically. You can also use the parameter fields above and below the graphical display area to adjust a number of group and zone parameters. The group list at the left can be hidden or shown. See Graphically edit Sampler zones and groups.

Within each group mapping area (notes and velocities), zones can't overlap. When you force an overlap by dragging zones one above the other within the same group, the Key Mapping Editor automatically cuts zones in order to make space for other zones. This allows you to add zones or delete mapped areas without having to edit all affected zones directly. You can protect either the selected or unselected zones when dragging zones.

Note: Changes to parameter values and graphical edits you make affect all selected zones. The Group and Zone parameters found in the Key Mapping Editor are a subset (duplicates) of the parameters found in the Zone view. A change to a parameter value in one editor view will be reflected in the other views.

Tip: The Key Mapping Editor works hand in hand with the Zone pane, which shows the selected zone waveform and associated parameters.



The Key Mapping Editor consists of three main areas. You can hide or view the group list by clicking the Group List Display button in the Group list controls.

- *Key Mapping Editor:* Drag zones into the Key Mapping Editor to add them. Zones are represented by rectangles. Zone key range is indicated by the width. Zone velocity range is indicated by the height. Zone parameters are shown below the keyboard.
- *Keyboard:* Drag zones onto the keyboard to add them to a key. Click a key to play the mapped zone. The root key of each zone is indicated in gold.

- *Group list controls:* Use to choose a group, mute, solo, or hide and view the group list. Group parameters are shown above the Key Mapping Editor.
  - *Group number:* Click a group number to move focus to that specific group in the group list, when multiple groups are selected. The focus feature makes it easier for you to graphically edit complex mappings that contain multiple overlapping groups. New groups are automatically assigned a consecutive number. See Select groups or zones in the Key Mapping Editor.
  - *Mute button:* Turn on to silence the selected group. Turn off to make the selected group audible.
  - *Solo button:* Turn on to hear the selected group in isolation. Turn off to make all groups audible.
  - *Group Name pop-up menu:* Displays the group name. Click to choose a different group.
  - *Group List Display button:* Hide or view the group list. When hidden, the Key Mapping Editor is maximized. You can mute, solo, and choose a group from the Group Name pop-up menu, when the group list is hidden.

Key Mapping Editor group parameters

Group parameters are shown above the Key Mapping Editor. Drag vertically in parameter value fields to change them. For example, vertically drag the group Pan field value to change it. Alternatively, you can double-click in a field and enter a value, then press the Return key.

- *Group Mixer parameters:* Globally adjust output settings for the selected group.
  - *Volume:* Use to set 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:* Use to set the pan position of the group—stereo balance for stereo samples—and the pan position of all assigned zones simultaneously.
  - *Output pop-up menu:* Choose the outputs used by the group: main outputs, paired channels, or individual outputs. This allows individual groups to be routed independently to aux channels in a multi-output Sampler instance.

    Note: This has an impact on any individual zone output assignments.
- *Group Key Range fields:* Use to define a key range for the selected group.
  - Use the left field to set the lowest note for the group.
  - Use the right field to set the highest note for the group. When you play notes outside this range, the zones assigned to this group are not triggered.

    Note: Take care with these parameters because they override zone range settings, possibly making some zones inaudible.
- *Group Velocity Range fields:* Use to define a velocity range for the selected 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.
  - Use the left field to set the lowest velocity that triggers the group.
  - Use the right field to set the highest velocity that triggers the group. When you play notes outside this range, the zones assigned to this group are not triggered.

Key Mapping Editor zone parameters

Zone parameters are shown below the keyboard. Drag vertically in parameter value fields to change them. For example, vertically drag the zone Tune field value to change it. Alternatively, you can double-click in a field and enter a value, then press the Return key.

- *Zone Name pop-up menu:* Displays the current zone name. Select a different zone to display and edit its parameters.

  Tip: You can also display and select a different zone in the Mapping pane Key Mapping Editor or Zone view by playing a note. To make use of this behavior, turn on the Select from Last Played Keys > Groups and Zones menu option in the Mapping pane Edit menu. You can automatically preview a zone when you play a note by turning on the Preview Selected Zone option in the Zone menu.

- *Zone Pitch fields:* Use to set the tuning of the selected zone.
  - *Root Key:* Use to set the root note of the zone—in other words, the note at which the sample is heard at its original pitch.
  - *Tune:* Drag vertically to tune the zone in semitone/cent increments.
  - *Pitch button:* Turn on to change the sample pitch when triggered by different keys. When turned off, the sample is always played at its original pitch, regardless of the note played.

    Note: When the Pitch button is turned off, the value set with the Tune parameter is added to, or subtracted from, the original pitch.
- *Zone Mixer fields:* Use these parameters to define output level and pan position of the selected zone.
  - *Volume:* Use to set the output level of the zone.
  - *Pan:* Use to set the pan position of the zone. This parameter works only when Sampler is used in stereo.
- *Zone Key Range fields:* Use these parameters to define a key range for the zone. When you play notes outside this key range, the sample assigned to this zone is not triggered.
  - Use the left field to set the lowest note for the zone.
  - Use the right field to set the highest note for the zone.
- *Zone Velocity Range fields:* Use to define a velocity range for the zone. When you play notes outside this velocity range, the sample assigned to this zone is not triggered.
  - Use the left field to set the lowest velocity that triggers the zone.
  - Use the right field to set the highest velocity that triggers the zone.

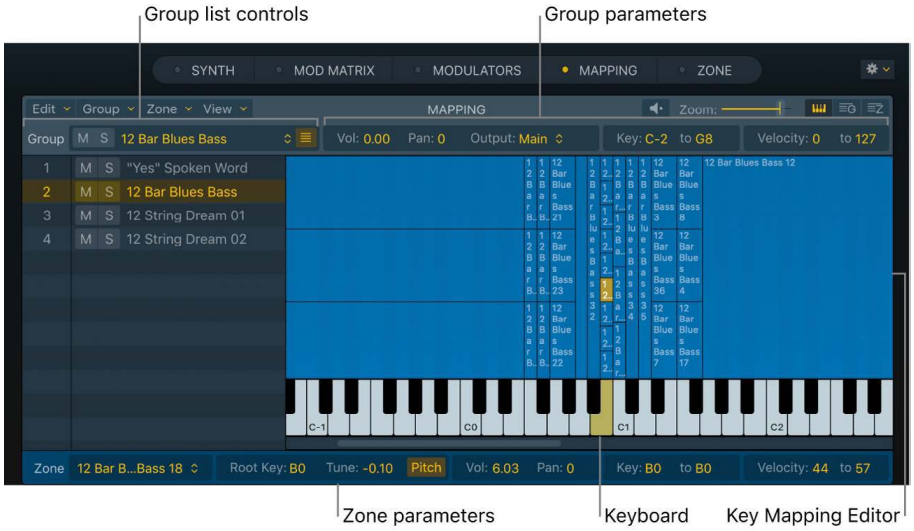
Navigate the Key Mapping Editor

- In Logic Pro Sampler, drag the scrollbars to navigate to non visible zones or groups. You can also use a two-finger swipe to scroll vertically or horizontally. If you are using a Magic Mouse, use a single-finger swipe to scroll.
- Use a pinch gesture to zoom the keyboard display. Alternatively, drag the Zoom slider.

  Tip: You can double-click the Mapping button in the Navigation bar to quickly switch between a maximized and the current view level.

  Important: To use gestures, such as swiping or pinching, you must turn on these mouse or trackpad features in System Settings.

Select groups or zones in the Key Mapping Editor



- In Logic Pro Sampler, when the group list is visible: Click a group name or number to select it and to move focus to the group. Shift-click or drag (from outside any group) to select multiple groups. The Key Mapping Editor area above the keyboard shows only zones that belong to the selected group. You can also remotely select groups with your MIDI keyboard.

  Note: To activate selection of groups from a MIDI keyboard, turn on the Select from Last Played Keys > Groups or Groups and Zones menu option in the Mapping pane Edit menu.

  Note: When multiple groups are selected, you will see the zones of all selected groups in the Key Mapping Editor. Each selected group has a number that you can click to move focus to that specific group. The focus feature makes it easier for you to graphically edit complex mappings that contain multiple overlapping groups.

- When the group list is *not* visible: Choose a group in the Group Name pop-up menu. The Key Mapping Editor area above the keyboard shows only zones that belong to the selected group. You can also remotely select groups with your MIDI keyboard.

- Click a zone in the Key Mapping Editor area above the keyboard to select it. Shift-click or drag (from an empty area outside any zone) to select multiple zones. You can adjust zone parameters shown below the keyboard and directly edit zones graphically. See Graphically edit Sampler zones and groups.

Use Sampler Group view in Logic Pro for Mac

You can choose from three editor views in the Mapping pane: Key Mapping Editor, Group view, and Zone view. Click the buttons at the top right to switch between these edit panes. The menu bar at the top of the Mapping pane contains elements that are common to all editor views. See Mapping pane menu bar.

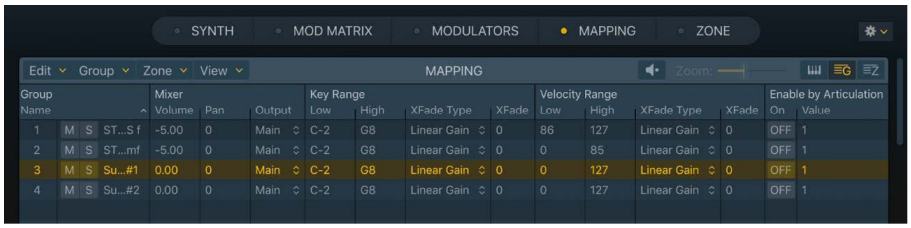
In Group view, you can: View and edit all group parameters, organized into columns, with related parameters shown in subcolumns. Group parameters provide simultaneous control of all zones assigned to the group.

- Click a group name or number to select it, or Shift-click or drag (from an empty area outside any group) to select multiple groups. You can also select groups with your MIDI keyboard.

  Note: To activate the selection of groups with a MIDI keyboard, turn on the Select from Last Played Keys > Groups or Groups and Zones menu option in the Mapping pane Edit menu.

  Note: When multiple groups are selected, changes to parameter values affect *all selected groups*. Each group has a number that you can click to move focus to that specific group, when multiple groups are selected. The focus feature makes it easier for you to edit complex mappings that contain multiple overlapping groups.

- Value changes are relative. For example, in two selected groups with High Velocity values of 12 and 27, reducing the value of the first group to 10 reduces the second group value to 25.
- Click parameter subcolumn headers to sort groups. For example, click the Low Key Range subcolumn header to sort zones by the lowest note.
- Control-click a parameter column header to open a shortcut menu where you can hide or show individual group parameters. You can also quickly show all columns, restore the default column view, and restore the default column width.
- Horizontally drag parameter columns to resize.
- Drag the scrollbars to navigate to non visible group parameters. You can also use a two-finger swipe on your trackpad to scroll vertically or horizontally. If you are using a Magic Mouse, use single-finger swipes.



Group view parameters

Drag vertically in parameter value fields to change them. For example, vertically drag the value shown for a group in the Pan column to change it. Alternatively, you can double-click in a field and enter a value, then press the Return key.

- *Group parameters:* New groups are automatically assigned a consecutive number.
  - *Mute button:* Mute or unmute a group.
  - *Solo button:* Solo or unsolo a group. All other groups are silenced when active. You can solo multiple groups.
  - *Name field:* Displays the group name. Double-click to enter a new group name.
- *Mixer parameters:* Globally adjust group settings.
  - *Volume field:* Adjust 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 field:* Adjust the pan position of the group—stereo balance for stereo samples—and the pan position of all assigned zones simultaneously.
  - *Output field:* Use to define the outputs used by the group. Choices include the main outputs, paired channels, or individual outputs. This allows individual groups to be routed independently to aux channels in a multi-output Sampler instance.
- *Key Range parameters:* Use to define a key range for the group. When you play notes outside this range, zones assigned to this group will not be heard.
  - *Low field:* Set the lowest note for the group.
  - *High field:* Set the highest note for the group.
  - *XFade Type pop-up menu:* Choose the crossfade type that best suits the group audio material. The Linear dB and Linear Gain options scale the amplitude for the group crossfade. Equal Power applies an exponential crossfade curve that causes a volume boost of 3 dB in the middle of the crossfade range. This fades out/fades in at an equal volume level. See Fade between sample groups in Sampler.
  - *XFade field:* Set the crossfade range for zones grouped by the specified key range. Only zones within the specified key range are crossfaded.
- *Velocity Range parameters:* Use to define a velocity range for the group. When you play notes outside this velocity range, you will not hear zones assigned to this group. This feature is useful when you want to dynamically mix—or switch between—grouped zones (samples) by playing your MIDI keyboard harder or softer. This is ideal for layered sounds, such as a piano/string layer, or when switching between different percussion samples, as examples.
  - *Low field:* Set the lowest velocity that triggers the group.
  - *High field:* Set the highest velocity that triggers the group.
  - *XFade Type pop-up menu:* Choose the crossfade type that best suits the group audio material. The Linear dB and Linear Gain options scale the amplitude for the group crossfade. Equal Power applies an exponential crossfade curve that causes a volume boost of 3 dB in the middle of the crossfade range. This fades out/fades in at an equal volume level.
  - *XFade field:* Set the crossfade range for zones grouped by the specified velocity range. Only zones within the specified velocity range are crossfaded.

- *Round Robin parameters:* Control groups in a round robin cycle. Round robin cycles are used to switch between groups of samples assigned to a single key. See Make advanced group selections.
  - *ON/OFF button:* Turn on to use the group in a round robin cycle.
  - *Cycle pop-up menu:* Choose to add or remove groups from the round robin cycle, or move the selected group to the end of the cycle, or to a new cycle.
- *Enable by Articulation parameters:* Use to define an Articulation ID for the group. This parameter is available as a modulation target, so you can switch between sample groups with a MIDI controller. For example, you could use your keyboard modulation wheel to switch between several hi-hat groups with different opening degrees. See Sampler articulation handling.
  - *ON/OFF button:* Turn on to use an Articulation ID for the group.
  - *Value field:* Set the Articulation ID number used by the group.
- *Enable by Bend parameters:* Use to define a bend range for the group. Values outside the specified range do not affect the group. See Make advanced group selections for information on the use of all *Enable by…* options.
  - *ON/OFF button:* Turn on to use a defined bend range for the group.
  - *Low field:* Set the minimum bend range value.
  - *High field:* Set the maximum bend range value.
- *Enable by Channel parameters:* Use to define a MIDI channel for the group. Only this MIDI channel affects the group.
  - *ON/OFF button:* Turn on to use a defined MIDI channel for the group.
  - *Value field:* Set the MIDI channel.
- *Enable by Control parameters:* Use to define a MIDI controller and range for the group. If you set up controller 64 (Sustain), for example, the group will be held (sustained) when an incoming controller 64 message (within the defined range) is received. Controller values outside the specified range do not affect the group.
  - *ON/OFF button:* Turn on to use a controller for the group.
  - *Value field:* Set the MIDI controller number.
  - *Low field:* Set the minimum controller value.
  - *High field:* Set the maximum controller value.
- *Enable by Note parameters:* Use to define a MIDI note number for the group. Only this MIDI note number affects the group.
  - *ON/OFF button:* Turn on to use a defined MIDI note number for the group.
  - *Value field:* Set the MIDI note value.
- *Enable by Tempo parameters:* Use to define a tempo range for the group. Values outside the specified range do not affect the group.
  - *ON/OFF button:* Turn on to use a defined tempo range for the group.
  - *Low field:* Set the minimum tempo range value.
  - *High field:* Set the maximum tempo range value.

- *Release Trigger parameters:* Use to determine if zones pointing to this group are triggered on key down or on key release. 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.
  - *ON/OFF button:* Turn on to turn on key release triggers. If turned off, zones in the group are triggered only on key down.
  - *Decay button:* Turn on to turn on the (Decay) Time field.
  - *Time field:* Use to set the time it takes for the level of a sample (triggered by key release) to decay.

    Note: The Decay button and Time parameters function only when the Release Trigger parameter is turned on (set to key release).
- *Playback parameters:* Use these parameters to determine playback behavior for the group.
  - *Voices pop-up menu:* Use to set the number of voices that the group can play.
  - *Exclusive pop-up menu:* Assign multiple groups to the same Exclusive Class. Groups in the same class can only be used independently. As soon as one group in the class is triggered, no other group in that same class can be triggered.

    Tip: A practical use of the Playback parameters 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 an exclusive 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.
- *Envelope 1 and 2 Offsets parameters:* Independently offset the envelope parameter settings for each group. This is useful if you want Envelope 1, for example, to uniquely affect the zones in a specific group—after the initial impact of the envelope on other played groups. See Sampler envelopes for details on envelope parameters.
- *Filter Offsets parameters:* 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. See Sampler Synth pane for details on filter parameters.
- *Playback Details parameters:* Use to define the sample select behavior for the group.
  - *Sample Select Random Offset field:* Set a randomization range offset value for Sample Select modulation, which you define with the Sample Select Random parameter in the Synth Details slide-out panel.

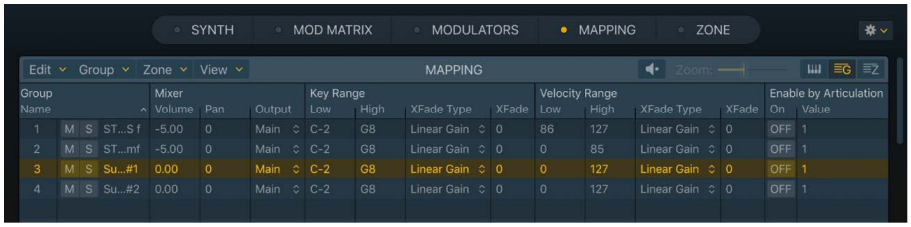
Advanced Sampler group selection in Logic Pro for Mac

You can define a specific event to use as a group selection switch. Whenever the defined selection event is triggered, zones in 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.

Group view provides two options for group selection: *Round Robin* and *Enable by…*

Control-click the top of any column to open a shortcut menu where you can hide or show any of the group selection filters.

You can use multiple group selection filters to refine your group selections. For example, you could specify that only a particular range of specified controller message values switches between different articulations. This could be further refined with a second criterion, such as Enable by Channel specified as a group selection filter.



Define a base group and switch between groups with MIDI notes

In Logic Pro Sampler, if you want Sampler to automatically switch between two string sample groups, for example—one for staccato samples and one for legato samples—you could set the group selection filter to Enable by Note, and assign a different MIDI note number to activate each group. You can then use a note that is not assigned to a zone as a remote group switch.

The following method assumes that several groups exist. See Create Sampler groups.

Note: This method can be applied to any of the Enable by… group selection options. You are free to combine multiple Enable by… options.

1. In Group view, Control-click the top of any column to open a shortcut menu where you can hide or show any of the Enable by… filters and other group parameter settings.
2. Choose Show all columns in the shortcut menu.

   Note: You can choose to hide or show individual group parameter settings, instead of Show all columns.
3. Click the OFF button in the Enable by Note On subcolumn.

   This is the base, or first, group you want to switch.
4. Drag vertically in the Enable by Note Value field to change the note number of the base group.

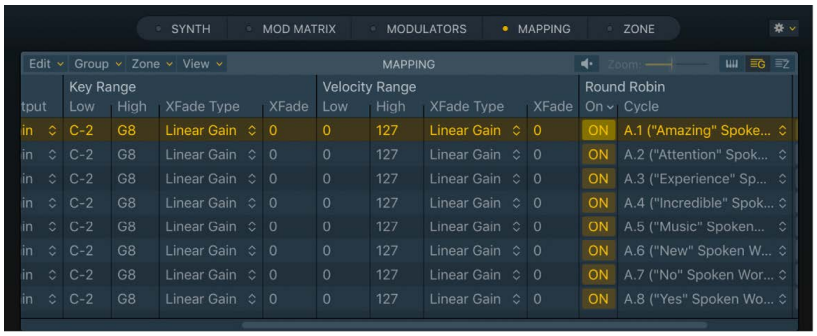
   This should be a note that has no assigned zone. When you play this note, this group is turned on—all other groups are turned off.
5. Click the OFF button in the Enable by Note On subcolumn for the second group you want to switch.
6. Drag vertically in the Enable by Note Value field to change the note number of the second group.

   This should be a note that has no assigned zone. When you play this note, the second group is turned on—all other groups are turned off.

Set up a round robin in Sampler

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 Logic Pro Sampler, you can use a round robin to step through groups, one after the other. For example, you could layer several hi-hat zones on a single key, each in its own group. Group 1 could contain an open hi-hat, and Group 8 a closed hi-hat, with Groups 2 to 7 containing different partially closed hi-hat sounds. When you repeatedly strike the key, Group 1 is played, then Group 2, and so on, in sequence. A further key strike restarts the cycle from Group 1.



This example uses one method to quickly add multiple groups and zones. You are free to create these using any method. See Create groups and Create zones.

Tip: You can also use the Group menu Create Round Robin command to create round robins. To use this command, select multiple groups in the Mapping pane and choose the command. All selected groups are chained from top to bottom.

1. Insert Sampler on an instrument channel strip.

   The Mapping and Zone panes are displayed.

2. Open the Loop Browser. Default key command: O
3. Select and drag several Apple Loop files to the Navigation bar. Drop them on the Optimized: Zone per Note dropzone.

   Tip: Choose a number of distinctly different Apple Loops. Press and hold Command, then click to select each file.

4. Click the Group view button.

   You can see multiple groups, named after the Apple Loops you just added.

5. Play any key on your keyboard to trigger, and select, all groups.
6. Control-click the top of any column header in Group view to open a shortcut menu, then choose Round Robin.

   The Round Robin column and parameter subcolumns are displayed in Group view. If you don’t see them, scroll horizontally until they are visible.

7. Click the OFF button of any group in the On subcolumn.

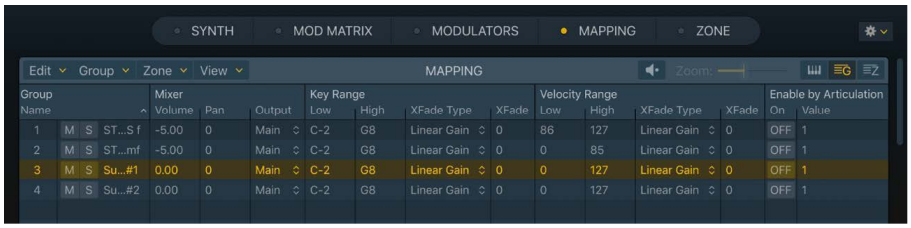
   You will see that the Cycle subcolumn is automatically populated with A.1 (Filename 1), A.2 (Filename 2), A.3 (Filename 3), and ensuing entries.

8. Now repeatedly strike a key on your keyboard.

   You will hear, and see, each group being played sequentially.

Crossfade sample groups in Logic Pro for Mac Sampler

You can use the crossfade (Xfade) parameters to crossfade between layered groups—by velocity or by key range.



Crossfade between groups by velocity

When you assign a zone to a group, you can set the lowest and highest MIDI note velocity that triggers the group—and consequently all zones within that group. You can layer grouped zones on the same keyboard note and trigger them individually by playing at different velocities. When you have distinctly different audio samples in adjacent groups, crossfading is very helpful in creating realistic-sounding sampler instruments.



For example in the image above, see the two layered samples, zone 121 and zone 122, on MIDI notes C1 to E1:

In the Sustain pedal #1 group:

- Zone 121 is a piano sample with minimal sustain pedal noise, set to a MIDI note velocity range of 0 to 51.
- Zone 122 is a piano sample with stronger sustain pedal noise, set to a MIDI note velocity range of 52 to 80.

- The crossfade value for these two zones is set to zero. There is no crossfade.

  In this example, the maximum velocity range value of zone 121 and the minimum velocity range value of zone 122 are adjacent. If you play note D1 at velocities above or below a value of 51, you can clearly hear each sample being triggered.

  To make this transition less abrupt, you can expand the velocity range of these zones by applying a crossfade value to the Sustain pedal #1 group. Each layered zone within this group will be crossfaded by the Xfade amount you set. The crossfade takes place in the extended velocity range area.

  In Group view, do the following for the Sustain pedal #1 group:

  - Click the Mapping button in the Navigation bar, then click the Group view button.
  - Click the Sustain pedal #1 group in the Name column.
  - Scroll to the Velocity Range column, and set values for these parameter subcolumns.
    - *Low field:* Set the lowest velocity that triggers the group. For zone 121 this would be a value of zero.
    - *High field:* Set the highest velocity that triggers the group. For zone 122 this would be a value of 80.
    - *XFade field:* Set the crossfade range for zones grouped by the specified velocity range. Only zones within the specified velocity range are crossfaded.
    - *XFade Type pop-up menu:* Choose the crossfade type that best suits the group audio material. The Linear dB and Linear Gain options scale the amplitude for the group crossfade. Equal Power applies an exponential crossfade curve that causes a volume boost of 3 dB in the middle of the crossfade range. This fades out/fades in at an equal volume level.

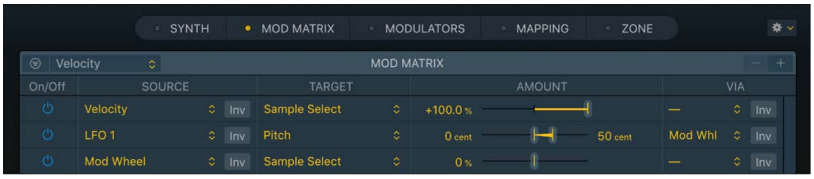
    Note: The settings made here override zone settings. When a zone velocity range is larger than the group setting, the zone velocity range is limited by the group velocity range setting.

Smoothly crossfade between groups in real time with a modulator

In Logic Pro Sampler, you can use a continuous modulation source, such as the modulation wheel, assigned to the Sample Select target to switch between layered samples—in groups with different velocity or key ranges. Adjust the crossfade amount to achieve smooth transitions between groups.

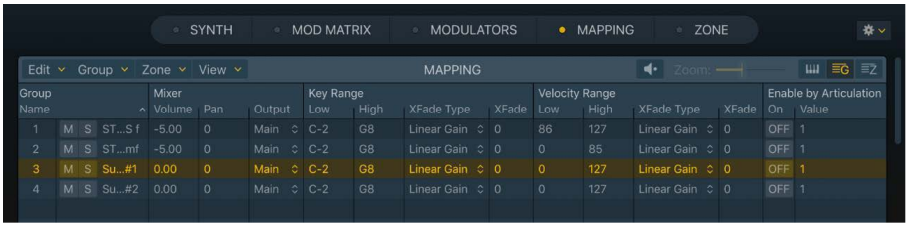
In an instrument that contains different samples mapped to several groups with different velocity (or key range) layers:

1. In the Mod Matrix pane, choose Sample Select from the Target pop-up menu in a modulation routing.
2. In the Mod Matrix pane, choose a modulation source from the Source pop-up menu, such as MIDI controller 1, the modulation wheel.



3. In the Mapping pane, click the Group view button at the top right, then set the Key Range or Velocity Range crossfade (Xfade) parameter to a value that provides the smoothest transitions between layers. Alternatively, you can choose a value that abruptly switches between sample layers.

   If the Xfade parameters are not shown, Control-click the column header to open a shortcut menu, where you can choose to view the Key Range and Velocity Range parameters. If you don't see the Xfade parameters, scroll horizontally until they are visible.



4. Adjust the crossfade (Xfade) parameter in multiple groups, as you wish.
5. Play your keyboard and adjust the modulation wheel to crossfade between groups containing your layered zones.

Use Sampler Zone view in Logic Pro for Mac

You can choose from three editor views in the Mapping pane: Key Mapping Editor, Group view, and Zone view. Click the buttons at the top right to switch between these editor views. The menu bar at the top of the Mapping pane contains elements that are common to all views. See Mapping pane menu bar.

You can use Zone view to: View and edit all zone parameters, organized in columns, with related parameters shown in subcolumns. The zones from one or more selected groups can be shown.

Note: The Group and Zone parameters found in the Key Mapping Editor are a subset (duplicates) of the parameters found in Zone view. A change to a parameter value in one editor view will be reflected in the other views.

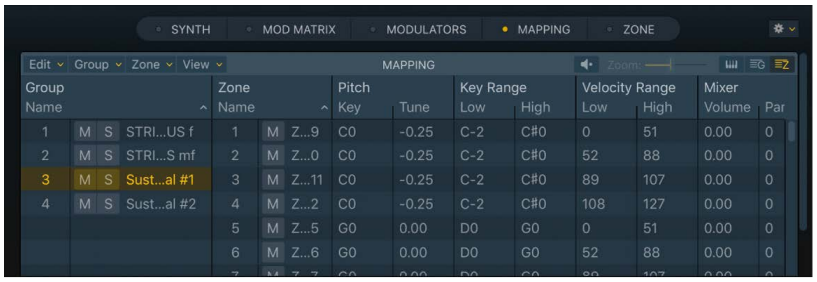
In Zone view, you can do the following:

- Click a zone or group to select it. Shift-click to select multiple zones or groups. You can also select zones or groups with Edit menu commands or with your MIDI keyboard. When multiple zones are selected, you can click the number to the left of the zone name to move focus to that specific zone. This limits previewing and edits to the focused zone, while retaining your multi-zone selection.

  Note: Changes to parameter values affect all selected zones. Value changes are relative. For example, in two selected zones with High Velocity values of 12 and 27, reducing the value of the first zone to 10 reduces the second zone value to 25.

- Click parameter subcolumn headers to sort zones. For example, click the Key subcolumn header to sort zones by note names.
- Horizontally drag parameter columns to resize.
- Control-click a parameter column header to open a shortcut menu where you can hide or show individual zone parameters. You can also quickly show all columns, restore the default column view, and restore the default column width.

- Drag the scrollbars to navigate to non-visible zone parameters. You can also use a two-finger swipe on your trackpad to scroll vertically or horizontally. If you are using a Magic Mouse, use single-finger swipes.



Zone view parameters

Drag vertically in parameter value fields to change them. For example, vertically drag the value shown for a zone in the Tune column to change it. Alternatively, you can double-click in a field and enter a value, then press the Return key.

- *Zone parameters:* New zones are automatically assigned a consecutive number.
  - *Zone Mute button:* Mute or unmute a zone.
  - *Zone Name field:* Displays the zone name. Double-click to enter a new zone name.
- *Audio filename pop-up menu:* Displays the audio filename. Click to open a shortcut menu that contains the following commands:

  Note: If at least one audio file is missing an extra column is shown to the left of the audio file column. Each missing file is marked with an exclamation mark icon. You can sort this column by clicking on the column header (exclamation mark icon). The column is visible only if there are missing audio files.

  - *Load Audio File:* Opens a dialog where you can select an audio file. Default key command: Control-F.
  - *Rename Current File:* Enter a new name for the audio file.
  - *Detach Current File:* Remove the audio file from the zone.
  - *Show in Finder:* Opens a Finder window, showing the location of the audio file.
  - *Open in Audio File Editor:* Opens the selected sample in the Logic Pro Audio File Editor (default key command: Control-W), or the sample editor chosen in Logic Pro > Settings > Audio > File Editor > External Sample Editor.
- *Pitch parameters:* Use to determine the tuning of the zone.
  - *Key field:* Set the root note of the zone—in other words, the note at which the sample assigned to the zone is heard at its original pitch.
  - *Tune field:* Use to fine-tune the zone.
- *Key Range parameters:* Use these parameters to define a key range for the zone. When you play notes outside this range, you will not trigger or hear the sample assigned to this zone.
  - *Low field:* Set the lowest note for the zone.
  - *High field:* Set the highest note for the zone.

- *Velocity Range parameters:* Use these parameters to define a velocity range for the zone. When you play notes outside this range, you will not trigger or hear the sample assigned to this zone.
  - *Low field:* Set the lowest velocity that triggers the zone.
  - *High field:* Set the highest velocity that triggers the zone.
- *Mixer parameters:* Use these parameters to define zone output handling.
  - *Volume field:* Set the overall output level of the zone.
  - *Pan field:* Set the pan position of the zone. This parameter works only when Sampler is used in stereo.
  - *Scale field:* Balance the level of a zone (sample) across the defined key range. A negative value makes lower notes softer than higher notes; positive values have the opposite effect.
  - *Output pop-up menu:* Set the outputs used by the zone. Choices include the main outputs, paired channels, or individual outputs. This allows individual zones to be routed independently to aux channel strips (in a multi-output Sampler instance).
- *Playback parameters:* Use these options to determine zone playback behavior.
  - *Pitch button:* Turn on to change the sample pitch when you play different keys on your keyboard. When turned off, the sample is always played at its original pitch, regardless of which note you play.
  - *1 Shot button:* Turn on to make the zone ignore the length of incoming MIDI note events—resulting in the sample assigned to the zone always being played from beginning to end whenever you play a note (or 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 button:* Turn on to play the sample from the sample end marker to the sample start marker.
- *Group Assignment pop-up menu:* Shows the group assignment of a zone. You can choose another group to reassign the selected zone or zones. For more information, see Create Sampler groups and Use Sampler Group view.
- *Loop parameters:* You can control all aspects of zone loops with these options.
  - *Loop ON/OFF button:* Turn on to use looping and to set other loop parameters.
  - *Loop Mode pop-up menu:* Choose a looping mode.
    - *Forward:* Playback cycles from the loop start point to the loop end point while you hold a key.
    - *Reverse:* Playback cycles from the loop end point to the loop start point while you hold a key.
    - *Alternate:* Playback continuously cycles from the loop start point to the loop end point, then switches from the loop end point to the loop start point, while you hold a key.
  - *Play to End on Release ON/OFF button:* Turn on to continue playback to the end marker position after you have released the key—provided that the amp release time is long enough for the audio portion after the loop to be audible. This feature is useful for allowing the natural decay of a sampled acoustic instrument to be heard during the envelope release phase, for example.

- *(Loop) Start and End fields:* Use to define the loop start and end points, allowing you to loop a portion of the audio file.
- *Tune field:* Change the tuning of the looped portion of the audio file in cent increments.
- *Xfade (Crossfade) field:* Use to define the crossfade time between the loop end and loop start points. 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 would normally produce clicks at the transition point—the join in the loop.
- *Eq. Pwr (Equal Power) ON/OFF button:* Turn on to set an exponential crossfade curve that causes a volume boost of 3 dB in the middle of the crossfade range. This fades 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.
- *Audio File Tail parameters:* Use these options to add and control a sample that is added to the end of a zone. This is typically used to add a reverb tail, or a section of reversed audio, but you are free to use any audio material.
  - *Name pop-up menu:* Displays the audio file tail filename. Missing files are indicated with an icon in a column to the left. Click to open a shortcut menu that contains file handling commands.
  - *Tune field:* Tune the audio file tail.
  - *Volume field:* Set the overall output level of the audio file tail.
  - *(Audio File Tail) Start and End fields:* Set the audio file tail start and end points, respectively.

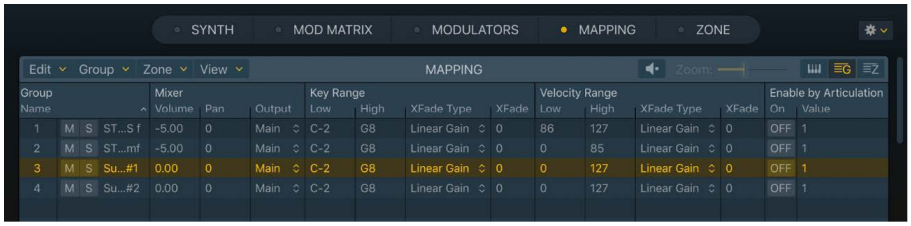
- *Sample parameters:* Use to define zone sample parameters. Sample parameters can be graphically edited in the Zone pane.
  - *(Sample) Start and End fields:* Set the sample start and end points, respectively.
  - *Fade In and Fade Out fields:* Use to set the fade times for a zone. Values are shown in samples. The higher the value, the longer it takes for the fade.

    Note: This parameter defaults to a value of 0, except when the sampler instrument is created with the Track > Convert Regions to New Sampler Track command in Logic Pro. This feature uses transient markers and results in a default Fade field value that matches the slicing offset of the following transient marker. See Create Sampler instruments from Logic Pro for Mac audio regions.
  - *Anchor field:* Set a value to trigger playback of the sample earlier in the project. To explain further, on playback, Logic looks into the future and advances sample playback by the amount set in the anchor field. The sample is played earlier, so that the anchor sounds exactly on the downbeat of the note. The sample anchor is useful for instruments such as shakers that have a rhythmic anchor at a position after the beginning of the sample.

    Note: The Anchor field value affects only sequenced playback of the sample. Live keyboard playing triggers the sample at the anchor point, so any sample data that precedes this is not played.

Sampler Mapping pane menu in Logic Pro for Mac

You can use Mapping pane menu commands and functions at any time. The menu bar displays the same menus and elements in the Key Mapping Editor, Group view, and Zone view.



Mapping pane menu parameters

See the tasks in this section to learn more about the commands and functions in each pop-up menu.

- *Edit pop-up menu:* Choose an editing or selection command or function.
- *Group pop-up menu:* Choose a group-related command or function.
- *Zone pop-up menu:* Choose a zone-related command or function. This menu contains Split, Automap, Remap, Loop, and Audio File submenus.
- *View pop-up menu:* Choose an option that affects the appearance and behavior of the display.
- *Preview button:* Play the selected zone.
- *Zoom slider:* In the Key Mapping Editor, drag to zoom the view in or out. You can also use pinch gestures to zoom directly in the graphical editor with your trackpad. If you are using a Magic Mouse, use a single-finger vertical swipe to zoom.
- *Editor view buttons:* Use to switch to the Key Mapping Editor, Group view, or Zone view.

Use Mapping pane Edit menu commands

In Logic Pro Sampler, choose any of the following commands or functions from the Edit menu:

- *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 *between two Sampler instances*, associated zones are also copied and group assignments of the zones are retained.
  - When you copy groups *within a single Sampler instance*, only the groups themselves are copied, not the associated zones. This behaves like the Group menu Duplicate command.
- *Delete:* Deletes the currently selected zone or group.

  WARNING: Take care when deleting a group because all zones associated with the group are also deleted.
- *Select All:* Selects all zones and groups in the loaded sampler instrument.
- *Deselect All:* Selects none of the zones and groups in the loaded sampler instrument.

- *Select Unused:* Selects all unused zones and groups in the loaded sampler instrument. These commands are available only when the Key Mapping Editor or Zone view is active.
  - *Unused Group:* A group that doesn't contain at least one zone, is not part of a round robin, and has no keyswitch is considered unused.
  - *Unused Zone:* A zone without a reference to an audio file is considered empty or unused.

    Note: A zone without a reference can be created manually by using the New command in the Zone menu. You can manually remove the zone audio reference by using the Detach command in the Zone > Audio files menu.
- *Invert Selection:* Switches the current selection. For example, if you have selected all unused zones and groups, this command will select all used zones and groups.
- *Select from Last Played Keys:* Selects groups, zones and groups, or neither when you play keys on your MIDI keyboard.
  - *None:* No zone or group is selected when you play your MIDI keyboard.
  - *Groups:* One or more groups is selected when you play your MIDI keyboard.
  - *Groups and Zones:* Both zones and groups are selected when you play your MIDI keyboard.
- *Edit Key Labels:* Open a dialog where you can name keys. This is ideal for GM Drum kit mapping, for example.
- *Edit Output Labels:* Open a dialog where you can name audio outputs.

Use Mapping pane Group menu commands

In Logic Pro Sampler, choose any of the following commands or functions from the Group menu:

- *New:* Create a new, empty group.
- *Duplicate:* Create a copy of the selected group (or groups), inclusive of zones and all group parameter settings.
- *Create Round Robin:* Creates a round robin of multiple selected groups. Repeatedly striking a key will sequentially step through these groups in a continuous cycle. See the task in Make advanced Sampler group selections.
- *Merge:* Merges multiple selected groups.

Use Mapping pane Zone menu commands

In Logic Pro Sampler, choose any of the following commands or functions from the Zone menu:

- *New:* Create a new empty zone (a zone with no reference to an audio file).
- *Load Audio Files:* Open a browser window where you can choose one or more audio files.
- *Normalize Loudness:* Analyze the selected zone to find the perceived loudness and raise or lower the volume of the zone to achieve a perceived loudness of -12 LUFs. The dynamic relationships of sample levels within the zone remain unaltered. When multiple zones are selected, all will play at the same loudness and the dynamic relationships of sample levels between zones will change.

  Tip: If you want to maintain the same relative gain relationship between zones, raise or lower the group volume.

- *Retune:* Analyzes the audio content and sets the root key and tuning value automatically.
- *Assign to New Group:* Create a new empty group and move the selected zone(s) to this group.
- *Always Move Root Key with Zone:* Turn on to automatically move the root key when you drag one or more zones in the Key Mapping Editor.

  Tip: This is useful when you want to change a zone's keyboard position without transposing the zone pitch, when moving it up or down the keyboard.
- *Prefer Splitting Zones by Velocity:* Turn on to change the appearance and behavior of zones in the Key Mapping Editor when you drag the boundary between two zones stacked vertically.
  - Turn on to cut zones *horizontally* (by velocity) to resolve a zone overlap within the same group.
  - Turn off to cut zones *vertically* (by note) to resolve a zone overlap within the same group.
- *Slide Zones Over:* Turn on to change the appearance and behavior of zones in the Key Mapping Editor when you drag them horizontally to positions that overlap other zones.
  - Turn on to cut *unselected* zones to resolve a zone overlap within the same group.
  - Turn off to cut *selected* zones to resolve a zone overlap within the same group.
- *Protect Unselected Zones in Zone List:* Turn on to prevent accidental overlaps, and therefore modifications, of unselected zones when you edit notes and velocities in Zone view.
- *Preview Selected Zone:* Automatically plays the selected zone when selected.

In Logic Pro Sampler, choose any of the following commands or functions from the Zone > Split menu:

- *Create Zones Split at Transients:* Use to analyze the selected zone for transients. A new zone is created for each transient detected in the audio file. This feature is most commonly used to split loops that contain drum, percussion, or other rhythmic patterns.
- *Create Zones Split at Silence:* Use to analyze the selected zone for points with extended periods of silence. New zones are created for the non-silent sections of the zone. This feature is most commonly used to split custom recordings of instruments, where notes of various pitches and loudness levels are played, separated by a second or two of silence.
- *Create Zones Split at Notes:* Use to analyze the selected zone for clear, pitched notes. A new zone is created for each pitched note, of sufficient duration and harmonicity, detected in the audio file. This feature is most commonly used to split loops that contain musical phrases.

  Tip: For all of these "Create" functions, once a zone is split, you can use Automap menu commands with the resulting zones to build a playable instrument.

Use Mapping pane Zone menu Automap commands

Automapping is a process that repositions zones across keyboard and velocity ranges, based on the audio content of each zone. As examples: note pitch or loudness, or a filename.

In Logic Pro Sampler, choose any of the following commands or functions from the Zone > Automap menu:

- *Automap using Current Root Note:* Selected zones are extended horizontally to fill the entire key range without gaps between zones, relative to the current root note position. If multiple selected zones have the same root note, they will be sorted by velocity, based on audio content loudness levels.
- *Automap using Pitch Detection:* Selected zones are repositioned and extended horizontally to fill the entire key range without gaps between zones, in accordance with their pitch. If multiple selected zones have the same pitch, they will be sorted by velocity, based on audio content loudness levels.
- *Automap using Mapping Data from Audio Files:* Selected zones are mapped using any recognized mapping data contained within the audio file header. Should the content of audio files overlap, new groups are created and zones are automatically moved to avoid overlaps.
- *Automap using Root Note from Audio File Names:* Selected zones are repositioned and extended horizontally to fill the entire key range without gaps between zones based on root note data in the audio filename. If multiple selected zones have the same root note in the filename, they will be sorted by velocity, based on audio content loudness levels.
- *Automap using Root Note from Audio File:* Selected zones are repositioned and extended horizontally to fill the entire key range without gaps between zones based on root note metadata in the audio file. If multiple selected zones have the same root note metadata in the file, they will be sorted by velocity, based on audio content loudness levels.
- *Automap Velocities Only:* Selected zones are repositioned and extended vertically to fill the entire velocity range without gaps between zones based on audio content loudness levels.

Use Mapping pane Zone menu Remap commands

Remapping is a process that repositions zones across keyboard and velocity ranges, with no analysis of audio file content.

In Logic Pro Sampler, choose any of the following commands or functions from the Zone > Remap menu:

- *Remap Notes:* Use to open a dialog where you can specify a start note and width that is applied to all selected zones.
- *Remap Velocities:* Use to open a dialog where you can specify a velocity range that is applied to all selected zones.
- *Remap White Notes:* Use to remap all selected zones to white notes only.
- *Remap Black Notes:* Use to remap all selected zones to black notes only.
- *Remap All Notes:* Use to remap all selected zones to fill all free keys, starting from the position/key of the lowest selected zone.

- *Remap to Root Notes:* Use to remap all selected zones to their respective root key positions.
- *Remap Evenly:* Use to evenly remap all selected zones across the entire keyboard range.
- *Remap to Round Robin:* Use to remap all selected zones to a new round robin group. Different round robin numbers are automatically assigned to each zone.
- *Pivot on Corner:* Use to remap selected zones to the position/key of the lowest selected note. Selected zones are stacked vertically on the key, in order from lowest (at the bottom) to highest numbered. The velocity range of each stacked zone is automatically adjusted.

  Note: In cases where selected zones are already stacked vertically, Pivot on Corner will perform the inverse function and will place each zone on a different key, using the full velocity range available.
- *Mirror Velocities:* Use to change the velocity range positions of all selected zones. Selected zone positions are mirrored vertically on the key.
- *Fill Gaps:* Use to expand the velocity and key ranges of all selected zones that have gaps between them. Expansion of these ranges is balanced between selected zones.
  - *Fill Gaps (Notes Only):* Expands the key ranges of all selected zones that have gaps between them.
  - *Fill Gaps (Vel Only):* Expands the velocity ranges of all selected zones that have gaps between them.
- *Swap:* Use to swap the positions of two selected zones.

Use Mapping pane Zone menu Root Key commands

In Logic Pro Sampler, choose any of the following commands or functions from the Zone > Root Key menu:

- *Read Root Key from Audio Analysis:* The selected zone is analyzed for the root key.
- *Read Root Key from Audio File Name:* The selected audio file name is scanned for root key data.
- *Read Root Key from Audio File Chunk:* The selected audio file chunk is scanned for root key data. A chunk can be specified in the metadata of a WAV file.
- *Read Root Key from Audio File SMPL Chunk:* The selected audio file SMPL chunk is scanned for root key data. The SMPL chunk can be specified in the metadata of a WAV file.

Use Mapping pane Zone menu Loop commands

In Logic Pro Sampler, choose any of the following commands or functions from the Zone > Loop menu:

- *Auto-Loop:* Use to analyze the selected zone and to automatically set loop start and loop end marker positions.
- *Set Start to Loop Start:* Use to move the (sample) start marker to the current loop start marker position.
- *Set End to Loop End:* Use to move the (sample) end marker to the current loop end position.

- *Set Loop from Start to End:* Use to move the loop start and end markers to the (sample) start and end marker positions.
- *Optimize Loop Start:* Use to automatically adjust the loop start point to create a smooth loop cycle.
- *Optimize Loop Crossfade/End:* Use to automatically set crossfade values at the point in the loop cycle where the loop end and loop start markers cross over. The loop length is not affected.

These Loop commands are also available in the Edit menu in the Zone pane.

Use Mapping pane Zone menu Audio Files commands

In Logic Pro Sampler, choose any of the following commands or functions from the Zone > Audio Files menu:

- *Detach:* Use to remove the audio file association from the selected zone. This results in an empty zone.
- *Write Loop:* Use to write loop start and end data to the header of the audio file associated with the selected zone.
- *Write Mapping:* Use to write mapping data such as: the root key, key range, and velocity range to the header of the audio file associated with the selected zone.
- *Update Info from Audio File:* Use to retrieve and refresh mapping data from the header of the audio file associated with the selected zone.

Use Mapping pane Zone menu Rename commands

In Logic Pro Sampler, choose any of the following commands or functions from the Zone > Rename menu:

- *Rename using Audio File Name:* Use to rename the selected zone, based on the audio filename.
- *Rename using Audio File Name and Root:* Use to rename the selected zone, based on the audio filename and root note.
- *Rename using Audio File Name, Root and Velocity:* Use to rename the selected zone, based on the audio filename, root note, and velocity.
- *Add Root and Velocity to Existing Name:* Use to add audio file root note and velocity data to the selected zone name.

Use Mapping pane View menu commands

In Logic Pro Sampler, choose any of the following display functions from the View menu to change the layout and appearance of the Key Mapping Editor, Group view, and Zone view:

- *Show Zone Names:* Turn on to display zone names on each zone in the Key Mapping Editor.
- *Show Zones from all Selected Groups:* Turn on to restrict the display of zones to selected groups in the Key Mapping Editor and Zone view.
- *Show Group Column in Mapping Editor:* Turn on to display the group list in the Key Mapping Editor.
- *Show Group Column in Zone List:* Turn on to display the group list in Zone view.

- *Visible Group List Columns:* Choose each item to display the corresponding parameter column in Group view. A checkmark is shown beside active items. Choose an active item to hide the column in Group view.
  - *Show all/used columns:* Choose to show all or used columns only in Group or Zone view.
  - *Restore column options:* Choose a Restore item to reset the current Group or Zone view column layout to default settings.
  - *Use column layout as default:* Choose to set the current Group or Zone view column layout as the default column layout.
- *Visible Zone List Columns:* Choose each item to display the corresponding parameter column in Zone view. A checkmark is shown beside active items. Choose an active item to hide the column in Zone view.
  - *Show all/used columns:* Choose to show all or used columns only in Group or Zone view.
  - *Restore column options:* Choose a Restore item to reset the current Group or Zone view column layout to default settings.
  - *Use column layout as default:* Choose to set the current Group or Zone view column layout as the default column layout.

Use the Sampler Zone pane in Logic Pro for Mac

You can see the current waveform of the selected zone and can directly edit a number of sample playback parameters in the Zone pane waveform display. You can zoom in or out with pinch gestures, or scroll the waveform display using two-finger swipes or by dragging the scroll bar. You can use single-finger swipes if you are using a Magic Mouse.

Important: To use gestures, such as swiping or pinching, you must turn on these mouse or trackpad features in System Settings.

As you move the pointer across the waveform display, it changes in appearance and function. For example, the pointer will change to indicate that you can move a slice marker or fade marker. In addition, parameters and values related to the current pointer tool function are shown below the waveform display (and in Zone view).

Click or drag markers and handles in the waveform display to change values. Drag vertically in parameter value fields to change them. For example, vertically drag the value shown for Loop Start to change it. Alternatively, you can double-click in a field and enter a value, then press the Return key.

You can also Control-click the waveform display to open a shortcut menu that allows you to open the sample in the Logic Pro Audio File Editor (or an external editor), if you prefer. See Edit in the Audio File Editor in Logic Pro for Mac.

You can specify which editor to open in Sampler settings.

Tip: The Zone pane is best used with the Mapping pane also visible. See Mapping and Zone pane overview.



Zone pane parameters

Double-click a parameter value field to enter a new value. Press Return to complete the operation.

- *Waveform display:* Shows the selected zone waveform, marker handles, and shaded marker areas. You can drag the marker handles to adjust zone playback.
- *Edit menu:* Choose an editing command or function. See the task in this section to learn more about the commands and functions in this pop-up menu.

  The Zone pane Edit menu also includes the Loop commands available in the Mapping pane Zone menu.
- *View menu:* Choose an option that affects the appearance of the display. See the task in this section to learn more about the commands and functions in this pop-up menu.
- *Preview button:* Play the selected zone.
- *Snap pop-up menu:* Choose a value. Edits to crossfade, sample, fade, or loop markers in the waveform display automatically snap to the nearest possible value.
- *Zoom vertical button:* Switch between the maximum vertical zoom view for the currently displayed waveform section and the default zoom view.
- *Zoom horizontal button:* Switch between the last manually set zoom level, if applicable, and an optimized zoom level that shows the area between the sample start and end markers. When no manual zoom level is set, the entire audio file is shown.

  Tip: You can use your trackpad to zoom in or out on the waveform display with pinch gestures, or scroll using two-finger swipes or by dragging the scroll bar. You can use a Magic Mouse to perform the following gestures: one finger horizontal swipe to scroll, or use a one finger vertical swipe to zoom.
- *Zone pop-up menu:* Displays the zone name. Click to choose other zones.
- *File pop-up menu:* Displays the audio filename. Click to choose file handling commands.
  - *Load Audio File:* Opens a dialog where you can select an audio file.
  - *Rename Current File:* Opens a dialog where you can rename the selected audio file.
  - *Detach Current File:* Detach the selected audio file from the zone. You can load and assign another audio file to this zone.
  - *Show in Finder:* Shows the full path of the loaded audio file in the Finder.
  - *Open in Audio File Editor:* Opens the selected sample in the Logic Pro Audio File Editor, or the sample editor chosen in the External Sample Editor preference.

- *Playback buttons:* Use these options to determine zone playback behavior.
  - *One Shot button:* Turn on to make the zone ignore the length of incoming MIDI note events, resulting in the sample always being played from start 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 parameters.

    Loop and crossfade parameters are dimmed when the One Shot button is active.
  - *Reverse button:* Turn on to play the sample from the sample end marker to the sample start marker.
- *Flex On/Off button:* Turn Flex mode on or off. If Flex mode is turned on, an audio sample is played at its original speed for all note pitches. See the task in this section to learn more about Flex use.
- *Follow Tempo button:* When Flex mode is active, turn on to follow the project tempo.
- *(Flex) Speed pop-up menu:* When Flex mode is active, choose a playback speed division or multiplication value.

  Tip: Flex Speed is also available as a modulation target.
- *Sample Start/End handles and fields:* Set the sample start and end points. Drag to reposition. Option-click, then drag either handle to move both markers.
- *Sample Length field:* Displays the length between the sample start and end marker positions. Drag vertically to move the end and fade-out marker.
- *Fade In/Fade Out handles and fields:* Set the fade-in or fade-out time for the zone. Drag to reposition. Option-click, then drag either handle to move both markers. The fade markers cannot be positioned before or after the sample start and end markers.
- *Loop Start/Loop End handles and fields:* Set the loop start and end points. Drag to reposition. Drag the shaded loop area to move both markers and the crossfade marker. Alternatively, you can Option-click, then drag either handle to move both markers and the crossfade marker.
- *Loop Length field:* Displays the length between the loop start and end marker positions. Drag vertically to move the loop end and crossfade marker.
- *Loop Mode pop-up menu:* Choose a looping mode. Set to No Loop to turn off looping.
  - *No Loop:* Turn off looped playback. This hides loop markers and crossfades from the waveform display.
  - *Forward:* Playback cycles from the loop start point to the loop end point while you hold a key.
  - *Reverse:* Playback cycles from the loop end point to the loop start point while you hold a key.
  - *Alternate:* Playback continuously cycles from the loop start point to the loop end point, then switches from the loop end point to the loop start point, while you hold a key.
  - *Play to End on Release:* Turn on to continue playback to the end marker position after you have released the key—provided that the amp release time is long enough for the audio portion after the loop to be audible. This feature is useful for allowing the natural decay of a sampled acoustic instrument to be heard during the envelope release phase, for example.

- *Crossfade handle and field:* Drag to set the crossfade time between the end and start of a looped sample. The crossfade marker handle is indicated with an X in the waveform display. Option-click, then drag the handle to move both crossfades.

  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.

Use Sampler Zone pane start, end, loop, fade, anchor, and crossfade markers

You can use Sampler Zone pane markers to alter audio playback in Logic Pro. To define the possible positions for markers, use the Snap menu.



Do any of the following:

- Drag the blue start or end marker to set the sample start and end point. Playback occurs between these markers.
- Press and hold Option, then drag either the start or end marker to move both the sample start and end point.
- Drag the gray fade in or fade out marker to change the length of the fade at the beginning and end of the audio file segment between the start and end markers.
- Press and hold Option, then drag either the fade in or fade out marker to move both fade markers.
- Drag the yellow loop start or end marker to set loop boundaries. Playback cycles between these markers when you hold a note.
- Drag the yellow shaded area between the loop start and end markers to move the entire loop. Alternatively, press and hold Option, then drag either the loop start or end marker to move the entire loop
- Drag the gray crossfade marker (indicated with X) to set the length of the crossfade at the beginning and end of loop boundaries, smoothing out audible glitches as the loop cycles across the loop end and start points.
- Drag the light blue anchor marker to set the absolute start point of the audio file.

  Note: The anchor marker is visible only when Show Anchor is selected in the Zone pane View menu.

Quickly zoom a portion of the Zone pane waveform display

In Logic Pro Sampler, the zoom buttons and pinch gestures can be used to zoom the waveform display shown in the Zone pane, but you can also do the following to quickly zoom a portion of the waveform:

- Press and hold Control and Option, then drag in the waveform display.

  The selected area is zoomed to fill the entire waveform display.

  Tip: Double-click the Zone button in the Navigation bar to quickly switch between a maximized view of the Zone pane and the current view.

Use the Sampler Zone pane shortcut menu

In Logic Pro, you can use the shortcut menu to choose a number of sample-handling commands. Control-click in the waveform display to access available options from the shortcut menu.

Important: The shortcut menu is context-sensitive. The commands shown in the Zone pane shortcut menu change when you click on different portions of the waveform display. For example, you will see different Auto-Loop commands when you Control-click directly on the yellow shaded loop area in the waveform display.

- *Auto-Loop:* Analyzes the audio content and automatically sets a loop. You can use the command multiple times to try different automatic loops.
- *Auto-Loop within Loop Area:* Analyzes the audio content and automatically sets a loop within the area defined by the loop start and end markers. You can use the command multiple times to try shorter automatic loops.
- *Retune:* Analyzes the audio content and sets the root key and tuning value automatically.
- *Crop Sample:* Crops (cuts) the portion(s) of the sample outside the start and end markers.
- *Crop Loop:* Crops (cuts) the portion(s) of the sample outside the loop start and end markers.

  Note: The “Crop” commands don’t modify or cut existing files. New files are created and are used in place of the original files.
- *Write Sample Loop to Audio File:* Saves the loop data to the file header of the audio file.
- *Open in Audio File Editor:* Opens the selected sample in the Logic Pro Audio File Editor, or the sample editor chosen in the External Sample Editor preference. See Logic Pro Audio File Editor.
- *Linear or Equal Power Crossfade:* Set crossfade gain behavior. Linear scales amplitude for the crossfade. Equal Power boosts the middle of the crossfade range and fades out/in at an equal volume level.

Use the Sampler Zone pane Edit menu

In Logic Pro, you can use the Zone pane Edit menu to choose a number of sample-handling commands. See also Use Sampler Zone pane Loop submenu commands. Choose any of the following:

- *New:* Create a new, empty zone. You can add audio to this zone from the File menu or with a drag and drop operation.
- *Retune:* Analyzes the audio content and sets the root key and tuning value automatically.
- *Re-Analyze Transients and Tempo:* Re-analyzes the current audio for transient and tempo changes, following edits you have made. This corrects transient or tempo changes caused by incorrect tempo data written in the file header, unexpected results from Flex use, or other tempo and transient manipulations, for example.
- *Derive Tempo from Loop Length:* Calculates a tempo for the audio file based on the Loop Length. If using the Follow Tempo button does not yield the desired results, you can choose this option to calculate a revised tempo at which the loop is aligned to the beat.
- *Crop Sample:* Crops (cuts) the portion(s) of the sample outside the start and end markers.
- *Crop Loop:* Crops (cuts) the portion(s) of the sample outside the loop start and end markers.
- *Open in Audio File Editor:* Opens the selected zone in the Logic Pro Audio File Editor.
- *Optimize Sample Gain:* Analyzes the audio content and sets automatic gain values.
- *Linear or Equal Power Crossfade:* Set crossfade gain behavior. Linear scales amplitude for the crossfade. Equal Power boosts the middle of the crossfade range and fades out/in at an equal volume level.

Use Sampler Zone pane Loop submenu commands

In Logic Pro Sampler, choose any of the following commands or functions from the Edit > Loop submenu in the Zone pane:

- *Auto-Loop:* Analyzes the audio content and automatically sets a loop. You can use the command multiple times to try different automatic loops.
- *Auto-Loop within Loop Area:* Analyzes the audio content and automatically sets a loop within the area defined by the loop start and end markers. You can use the command multiple times to try shorter automatic loops.
- *Set Start to Loop Start:* Use to move the (sample) start marker to the current loop start marker position.
- *Set End to Loop End:* Use to move the (sample) end marker to the current loop end marker position.
- *Set Loop from Start to End:* Use to move the loop start and end markers to the (sample) start and end marker positions.
- *Optimize Loop Start:* Use to automatically adjust the loop start point to create a smooth loop cycle.

- *Optimize Loop Crossfade/End:* Use to automatically set crossfade values at the point in the loop cycle where the loop end and loop start markers cross over. Loop length is not affected.
- *Write Sample Loop to Audio File:* Saves the loop data to the file header of the audio file. To save the new audio file with a different name, click the Name field and choose the Rename Current Audio command.

Use the Sampler Zone pane View menu

In Logic Pro, you can use the Zone pane View menu to choose a command that changes the appearance of the Zone pane. Choose any of the following:

- *Display Seconds/Sample/Beats Units:* Shows time, sample, or bar/beat values in the waveform display.
- *Display Stereo Channels/Mono Sum:* Shows a stereo or summed mono waveform view in the waveform display.
- *Show Anchor:* Shows the absolute start point of the audio file in the waveform display. This can be useful for noises that precede instrument sounds such as the intake of breath of a horn player or tambourine shakes, as examples.

Synchronize Sampler audio with Flex in Logic Pro for Mac

In Sampler, you can use the Flex parameters to synchronize the audio playback tempo. This is ideal for synchronously playing an Apple Loop in a key that matches the key of other tracks, for example. Flex works for files with tempo information, including:

- Apple Loops
- Regions from the Tracks area
- Bounced audio files
- Files analyzed in the Smart Tempo Editor

Use Flex to synchronize audio in the Sampler Zone pane

Follow these steps to use Flex to play an Apple Loop synchronously with other tracks in your project.

1. In Logic Pro, load a project with at least a few tracks.
2. Open the Loop Browser. Default key command: O.
3. Drag a melodic Apple audio loop, such as a rhythmic acoustic or electric guitar part, into the Zone pane waveform display. Drop it on the Original dropzone.
4. Play some notes on your keyboard over a few octaves.

   You will hear that the pitch of the loop changes, along with the playback speed.
5. Click the Flex button below the waveform display and play some notes on your keyboard over a few octaves.

   You will hear that the pitch of the loop changes, but the playback speed doesn’t.
6. Start playback of your project and play a few notes on your keyboard.

   You will hear that the pitch of the loop changes, but the playback speed isn’t synchronized with the project tempo.

7. While the project is playing, click the Follow Tempo button below the waveform display and play some notes on your keyboard.

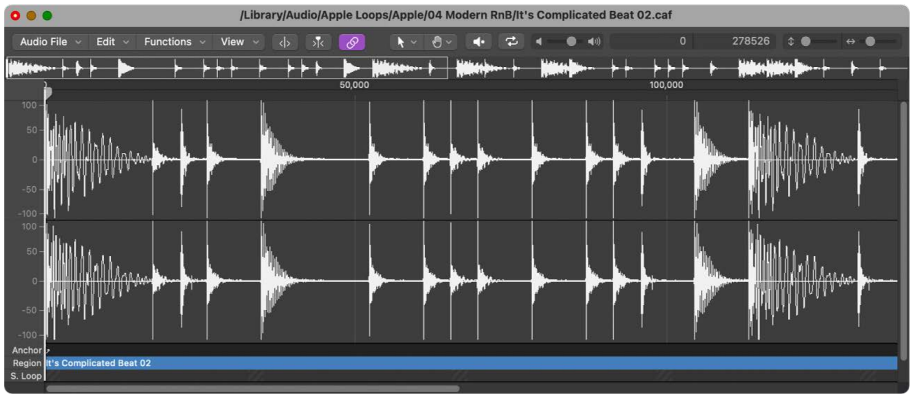
   You will hear that the pitch of the loop changes, and the playback speed is perfectly synchronized with the project tempo.

8. As an option, you can choose a value in the Speed pop-up menu to divide or multiply the synchronized playback speed of the Apple Loop. Try this while the project is playing.

Edit in the Audio File Editor in Logic Pro for Mac

Sampler and the Logic Pro Audio File Editor work seamlessly together. While you will typically perform most sample editing tasks in the Zone pane, you have the option of performing edits and adjustments in the Audio File Editor or an external audio editing application.

See Audio File Editor overview and Sampler settings.



Choose an external sample editing application

By default, the Logic Pro Audio File Editor is opened when you use the Open in Audio File Editor shortcut menu command in the Zone pane. You can override this behavior by choosing a different audio file editing application as the default.

1. In the Logic Pro Audio Settings window, click the File Editor tab.
2. Click the External Sample Editor field to open a Finder window.
3. Browse, and select, the audio editing application you want to use, then click the Choose button.

   The path and name of the application is shown in the External Sample Editor field.

4. Click the Remove button below the External Sample Editor field to restore the Logic Pro Audio File Editor as the default.

Open the Logic Pro Audio File Editor from the Sampler Zone pane

- In Logic Pro Sampler, Control-click the waveform display in the Zone pane, then choose Open in Audio File Editor from the shortcut menu.

  Tip: You can also use this command to open an external sample editor that you have set in the settings.

Edit sample borders and loop points in the Audio File Editor

1. In Logic Pro, after the sample is opened in the Audio File Editor, drag the sample (region) and loop borders graphically in the Region and S. Loop rows below the waveform. The region length is represented as a red bar. The loop length is represented by a yellow bar.

   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 Sampler.

   Note: Edited samples may have values that are not accurately shown in the Zone pane.

Create instruments, zones, and groups

Sampler instruments overview in Logic Pro for Mac

A *sampler instrument* is the file type that is loaded into Sampler. You load sampler instruments with the plug-in Settings pop-up menu, or you can load a patch from the Library.

When you load a sampler instrument, the associated audio files are automatically located on the hard disk (or disks) and are loaded into computer RAM. You play and record the loaded sampler instrument in the same way as any software instrument.

Sampler instruments can use keyboard mappings, articulation settings, and other options to emulate natural elements of real instruments or to optimize them for live performances. For example, some percussion instruments in the App Presets library contain special performance patches that make it easy to play these instruments with a MIDI keyboard.

A sampler instrument tells Sampler 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 hard disk.
- Zones are assigned to *groups*, which provide parameters that you use to affect all zones contained in the group. You can define as many groups as you like.

Sampler is compatible with the following audio file formats: AIFF, WAV, and CAF.

Sampler is also compatible with the SoundFont2, DLS, and Gigasampler sample formats. See Add SoundFont2, DLS, and Gigasampler files.

Each audio file is loaded into Sampler as a separate sample and is automatically assigned to a zone in the Zone and Mapping panes. You can edit and organize these zones into sampler instruments. See Sampler Mapping and Zone pane overview.

You can also create sampler instruments with the Auto Sampler utility plug-in.

Important: Audio files are *not* contained within a sampler instrument. The sampler instrument only stores information about the files, including 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 be unable to find it. You can move audio files to another location on your system, however, because Sampler automatically searches for files when you load a sampler instrument. The search mechanism uses Spotlight, so you should make sure that Spotlight is running and updated for all your drives.

See Sampler storage locations and Sampler instrument management.

Create instruments with Logic Pro for Mac Sampler

You can create your own sampler instruments in the way that suits you best. The simplest and fastest way is with drag and drop.

- You can drag one or more audio files directly onto the Sampler Navigation bar.
- You can drag one or more audio files directly onto the Sampler Mapping pane. This is generally done in the Key Mapping Editor, but you can also drag and drop audio into Zone and Group views.
- You can drag a single audio file directly into the Sampler Zone pane.

You can also methodically create empty zones and groups using menu commands, then fill them with audio content, using either menu options or drag and drop.

You can add zones and groups to loaded instruments, or you can create a new instrument from scratch, and fill it with zones and groups.

Editing of many zone and group parameters can also be performed graphically, or with numerical fields and menus. Creation and editing of sampler instrument zones and groups is performed in the Mapping and Zone panes.

Important: Sampler cannot directly record samples as you would with a hardware sampler. You can record samples in the Track area in Logic Pro, then drag them into Sampler.

For information about creating zones and groups, see Create zones and Create groups.

Quick instrument creation in Sampler in Logic Pro for Mac

To quickly create an instrument in Sampler, just drag one or more samples onto the Navigation bar. Sampler analyzes the samples, creates zones containing the samples, and then maps the zones to keys.

To simplify instrument creation, the Navigation bar provides Chromatic and Optimized dropzones:

- When you drag one or more samples onto the Chromatic dropzone, Sampler chromatically maps samples as zones across the keyboard range, starting from C2. Each zone is mapped to a single key on the keyboard. The original file length, tuning, and volume are used. Looping data is read from the file header, if present. The root key can also be read from the filename.

  Note: The Read Root Key from preference has a significant impact on chromatic import behavior. For details, see the Sampler settings section in the Logic Pro User Guide.

- When you drag one or more samples onto the Optimized dropzone, Sampler automatically determines the root notes of the zones and maps the zones across the keyboard. If there are multiple zones with the same root note, Sampler creates velocity layers—a different sample is triggered depending on how hard you strike the key—based on the perceived loudness of the samples. Loop point data in the file header is used, if present. If the file header does not contain looping data a search is performed to find the best loop points, and the crossfade length and type are automatically set. The file is also analyzed to find the perceived loudness, and the volume is raised or lowered to achieve a perceived loudness of -12 LUFs. Start and end markers are also automatically set to crop silence that precedes or follows the analyzed audio.

  Follow the steps in the task to create your own instruments in record time.

Create groups and zones by dragging audio content onto the Navigation bar

1. Choose Sampler from the Instrument slot on a software instrument channel strip.
2. Drag samples from the Logic Pro browsers, the Finder, a region or cell, or even from a marquee selection on a region to one of the import options in the Navigation bar: Chromatic or Optimized.

   - To make an instrument from multiple samples of pitched instrument notes, such as the different notes of a piano, or a single sample that contains multiple note pitches like a vocal sample or a bass line, choose an Optimized dropzone.
   - If you have multiple samples of sounds that don't have a particular pitch that you want assigned to MIDI notes, such as multiple drums in a kit, choose a Chromatic dropzone.

     Tip: You can add any type of loop from the Loop Browser, including MIDI loops, Step Sequencer loops, and Session Player loops.

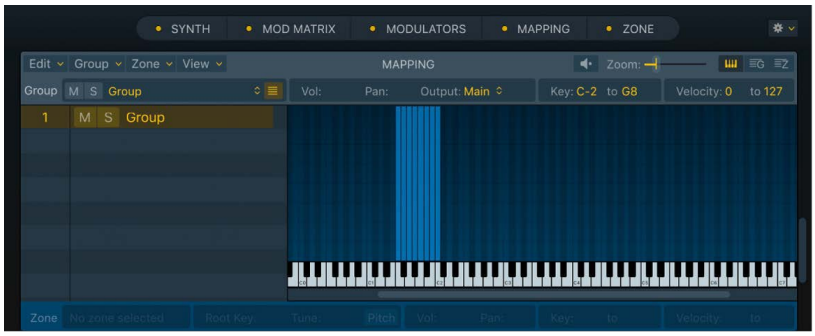


3. The Chromatic and Optimized import options in the Navigation bar divide into two further dropzones when you drag audio content onto them.

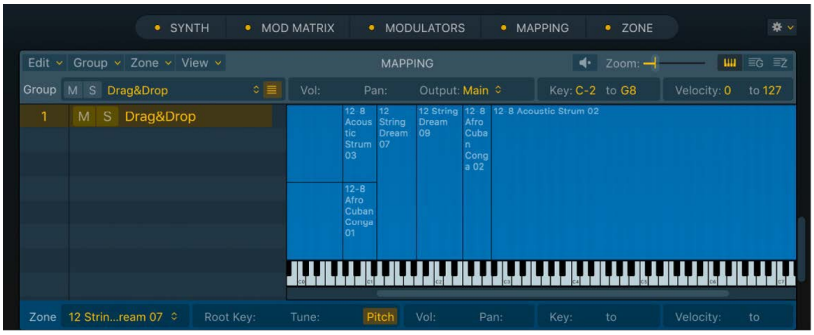
   - *Zone per File:* Creates a group containing one zone for each file dropped on the dropzone.
   - *Split at Silence:* Creates a new group for each file dropped on the dropzone. Each file is split into segments at extended periods of silence, and a new zone is created in the associated group for each segment. These segments are mapped chromatically from C2.

     If you import more samples than the number of available keys, Sampler automatically creates additional groups, which start mapping samples chromatically from C2 in each new group.

     Depending on the audio material and existing zones, using the Zone per File option to import several audio files may result in an onscreen view similar to the following image:



Depending on the audio material and existing zones, using the Zone per Note option to import a single audio file may result in an onscreen view similar to the following image:



Depending on the audio material and existing zones, using the Zone per Note option to import multiple audio files may result in an onscreen view similar to the following image:



Create Sampler instruments from Logic Pro for Mac audio regions

In Logic Pro, you can convert one or more 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.

In the case of a single selected region, slices of the audio are sequentially mapped—in accordance with analyzed transient positions—across the key range, starting with the first transient.

This operation also creates a new track and MIDI region.

- When you convert multiple audio regions, the MIDI region contains automatically created trigger notes that match the time positions of the source audio regions. You can reposition or edit these trigger notes in the Piano Roll Editor.
- When you convert a single audio region, the MIDI region contains automatically created trigger notes that match analyzed transient positions in the source audio region. You can reposition or edit these trigger notes in the Piano Roll Editor. You can also change zone parameters in the Sampler Mapping and Zone panes.

Convert audio regions to sampler instruments and tracks

1. In the main window, select the audio regions you want to convert.
2. Control-click any selected region, then choose Convert > Convert to new Sampler Track (or use the default key command: Control-E).
3. In the dialog, choose Create Zones From Regions or Create Zones From Transient Markers.
4. In the Instrument Name field, enter your text.
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.

Create zones with drag and drop in Logic Pro for Mac

A *zone* is a location into which a single audio file—or sample, if you prefer this term—can be loaded. The sample loaded into the zone is memory resident—it uses the RAM of your computer. You can define as many zones as needed.

A zone offers parameters that control sample playback. You can set 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 for each zone. In addition, you can adjust sample start, end, and loop points, volume, and several other parameters for the zone.

Sampler provides several methods you can use to create and add zones. You can quickly add a single zone to Sampler by dragging an audio file, audio or software instrument region, or Apple Loop to the Mapping or Zone panes or the instrument track header. You can create multiple zones by dragging multiple audio files, regions, or loops onto the Sampler Mapping pane or the Navigation bar. You can also manually create a zone and can add an audio file by using menu options. In addition, you can choose whether the new zone maps samples chromatically or based on analysis of the material.

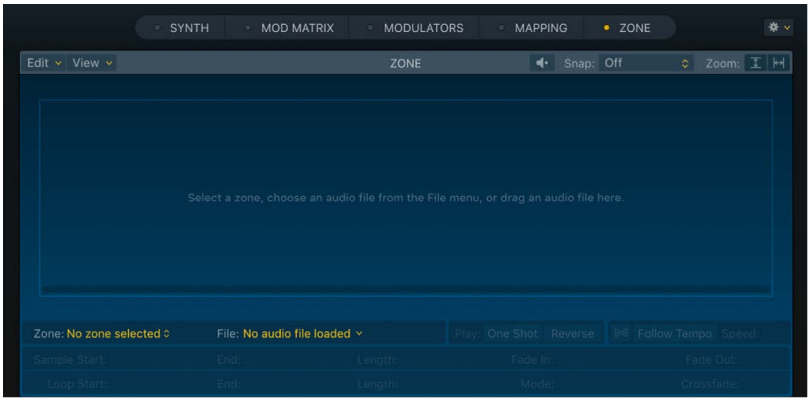
Tip: Any type of loop can be added from the Loop Browser, including MIDI loops, Step Sequencer loops, and Session Player loops.

Important: When you drag a region to the track header area or to another drag zone to create a sample-based software instrument using the region, the region is bounced through the plug-ins on the track. For software instrument tracks, this includes any MIDI plug-ins, the track instrument, and any audio plug-ins. For audio tracks, this includes any audio plug-ins and other processing, such as Flex. The resulting audio file is used in the sample-based instrument. This is different than dragging an audio file to a drag zone, which does not trigger a bounce.

Create a zone by dragging content into the zone pane

1. In Logic Pro Sampler, click the Zone button in the Navigation bar to view an empty Zone pane.
2. To add material from the Main window, Finder, Loop Browser, or File Browser: Drag an audio file, audio or MIDI region, or Apple Loop into the Zone pane.

   The root key for the zone is the key at which the sample is played at its recorded pitch.



Create a zone by dragging content to one or more keys

1. In Logic Pro Sampler, click the Mapping button in the Navigation bar to view an empty Mapping pane. Click the Key Mapping Editor button if not already shown.
2. From the Main window, Finder, Loop Browser, or File Browser: Drag an audio file, audio or MIDI region, or Apple Loop directly onto a key on the Mapping pane keyboard.

   The root key is shown in gold on the keyboard. The start key, end key, and root key are all set to the note that the file is dragged to.

   Note: You can drag multiple audio files onto a key in the Key Mapping Editor. This automatically creates a stack of layered zones, split by velocities.



   The drop location changes the import behavior:

   - When you drop an audio file onto the keyboard, it is mapped to that key.
   - When you drop an audio file onto the Key Mapping Editor above the keyboard, it is mapped to a range of keys. As you drag toward the top of the Key Mapping Editor, the key range expands.

     The root key is shown in gold, and the key range (containing the start key and end key) is shown across the keyboard.



Add multiple files to the Key Mapping Editor

In Logic Pro Sampler, you can add audio files to the Mapping pane in several ways and in different editor views.

1. Click the Mapping button in the Navigation bar to view an empty Mapping pane. Click the Key Mapping Editor button if not already shown.
2. From the Main window, Finder, Loop Browser, or File Browser: Drag one or more files into the Mapping pane, either directly onto the keyboard or the Key Mapping Editor above the keyboard. Shift-click or Command-click to select multiple files.



The drop location changes the import behavior:

- When you drop an audio file onto the keyboard, it is mapped to that key. When you drop multiple audio files onto the keyboard, each is mapped to its own key.

  The root key is shown in gold on the keyboard. Multiple gold keys are shown when you drag more than one audio file.

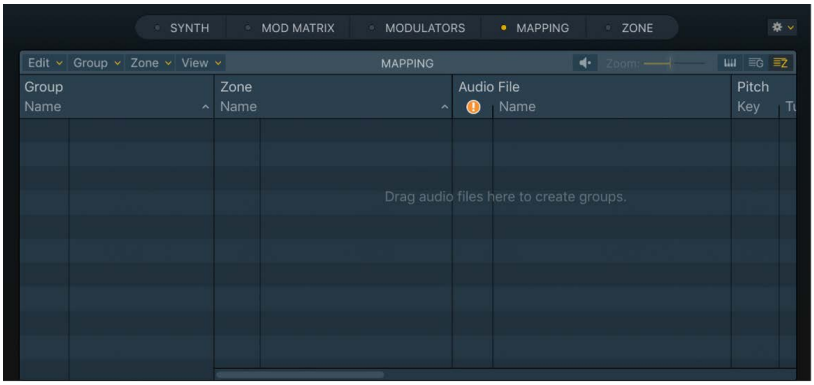
- When you drop an audio file onto the Key Mapping Editor above the keyboard, it is mapped to a range of keys. As you drag toward the top of the Key Mapping Editor, the key range expands.

  The root key is shown in gold, and the key range is shown across the keyboard. Multiple gold keys and key ranges are shown when you drag more than one audio file.

Add multiple files to the Zone view and create a new group

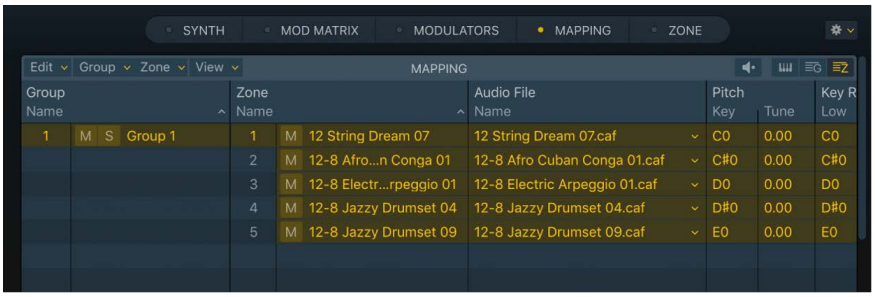
In Logic Pro Sampler, you can add audio files to the Mapping pane in several ways and in different editor views.

1. Click the Mapping button in the Navigation bar to view an empty Mapping pane. Click the Zone view button if not already shown.



2. From the Main window, Finder, Loop Browser, or File Browser: Drag one or more files onto the Zone view. Shift-click or Command-click to select multiple files.

   A group is automatically created and the audio file is added to the group. When you drop multiple audio files, all are added to the group.

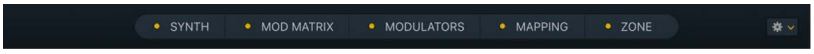


Add zones by dragging audio files, regions, or loops to a software instrument track

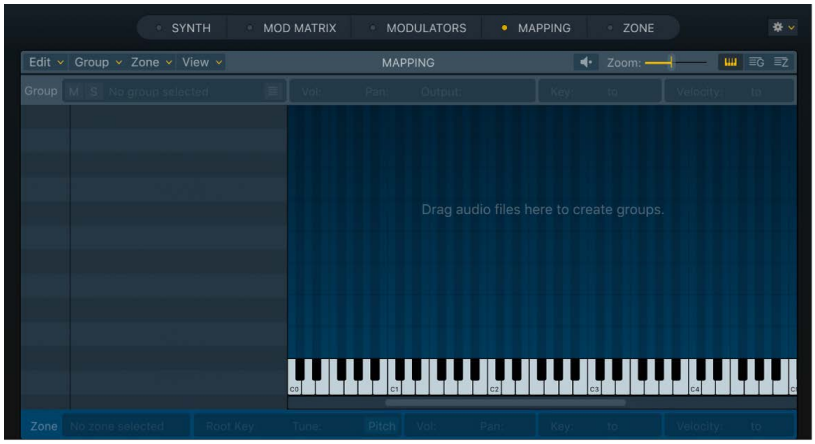
1. In Logic Pro Sampler, drag one or more audio files, regions, or Apple Loops to a software instrument track header with a Sampler instance inserted. Shift-click or Command-click to select multiple files.
2. When the “Add new zones” dialog appears, drag the items to one of the following:
   - Sampler (Chromatic Map) maps samples to successive chromatic notes.
   - Sampler (Optimized Map) maps samples according to analysis of the material.

Create a new, empty instrument

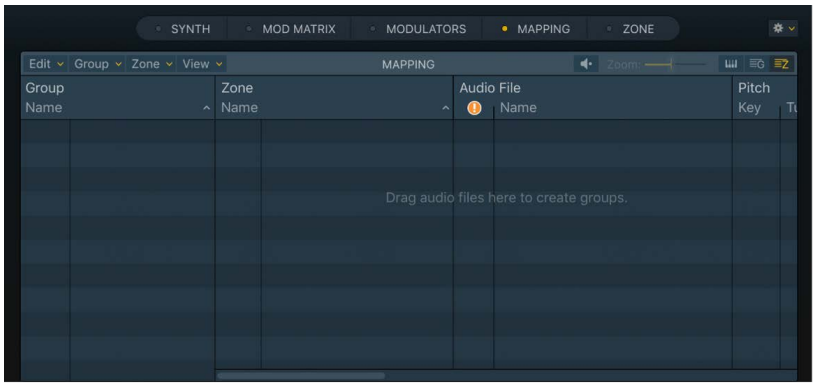
1. In Logic Pro, insert Sampler into an instrument channel.
2. Click the Mapping or Zone buttons (or both) in the Navigation bar to turn on these panes, if they are not shown.



3. Click the Mapping button in the Navigation bar to view an empty Mapping pane. You will typically add audio files in either the Key Mapping Editor, Navigation bar, or Zone view, but you can also drop files into the Group view or Zone pane.
   - Click the Key Mapping Editor button to add audio files in the Key Mapping Editor.



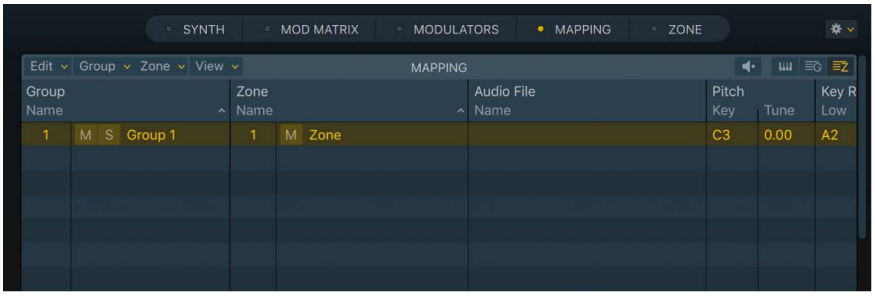
   - Click the Zone view button to add audio files in this view.



Create an empty zone with a menu command, and assign a sample to it

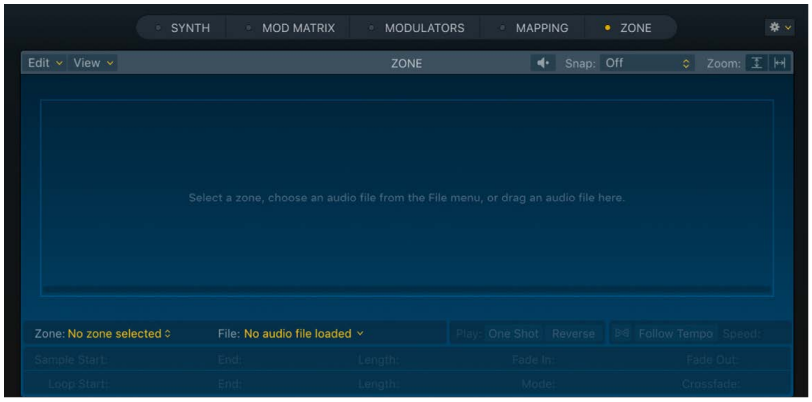
1. In the Logic Pro Sampler Mapping pane, choose Zone > New.

   A new group is shown in the Mapping pane, and a zone is automatically added to the group.

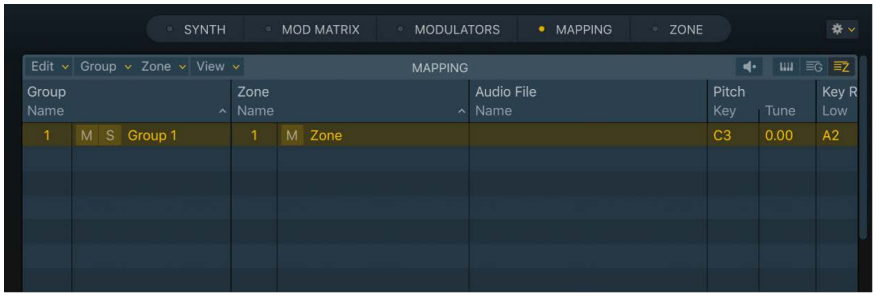


2. Do one of the following:

   - In the Zone pane, drag an audio file into the waveform display area.

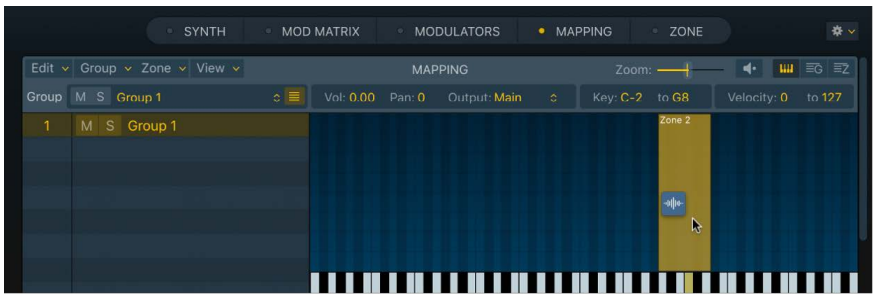


   - In Zone view, click the File column, then choose Load Audio File from the pop-up menu.
   - In Zone view, drag an audio file into the empty Name field of the zone in the Audio File column.

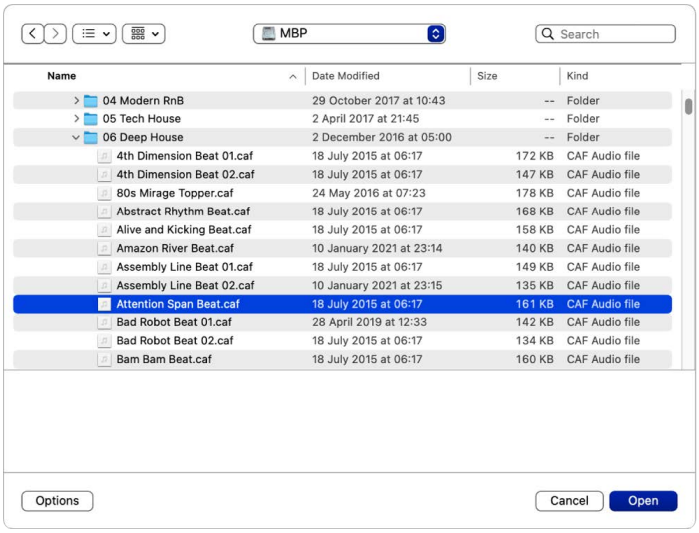


   - In the Key Mapping Editor, press and hold Command-Shift, then drag the pointer to create a new, empty zone. You can add a sample to the zone using any of the methods discussed.

- In the Key Mapping Editor, drag an audio file directly onto the zone.



3. If you chose the Load Audio File option in step 2, locate the audio file you want and select it in the File Selector window.



- Click the Options button at the lower left of the File Selector window to show or hide the checkboxes and Play button.
- Select “Hide used audio files” to dim the names of files used in the currently loaded sampler instrument.
- Select “Preview audio file in Sampler Instrument” to temporarily replace the sample files in the currently selected zone. The zone is not directly triggered by selecting this option, but it can be triggered by playing MIDI notes while the File Selector window 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 preview looped playback of the currently selected audio file, click the Play button at the lower right of the File Selector window. The Play button label changes to Stop during playback.
   - You can step through files by using the Down Arrow key, or by clicking them, to audition each file in turn.
   - Click the Stop button to stop playback.
5. When you find an audio file you want to use, click the Open button to add it to the zone.

   When the audio file is loaded, the sample name is displayed in the Name field for the zone in the Audio File column of Zone view.

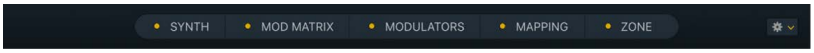
Create zones with menu commands in Logic Pro for Mac

You can manually create a zone and can add an audio file by using menu options. You can also choose whether the new zone maps your samples chromatically or based on analysis of the audio material.

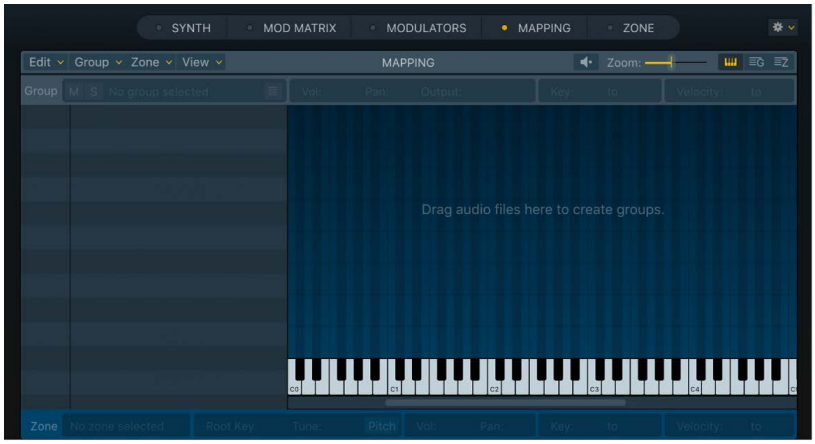
Create a new, empty instrument

1. In Logic Pro, insert Sampler into an instrument channel.
2. Click the Mapping button in the Navigation bar to view the Mapping pane.

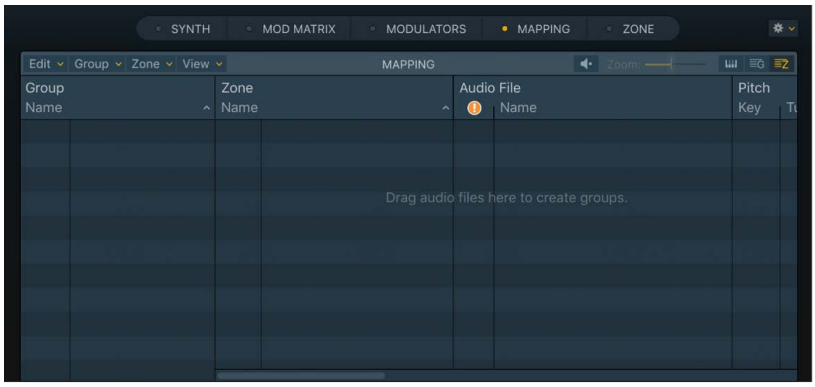
   You can also click the Zone button if you want to see the Zone pane.



3. The empty Mapping pane is ready for you to add audio material. You have a choice of adding audio files in either the Key Mapping Editor or Zone view. You can also add files by dragging them to the Navigation bar, Group view, or Zone pane.
   - Click the Key Mapping Editor button to add audio files in the Key Mapping Editor.



   - Click the Zone view button to add audio files in this view.

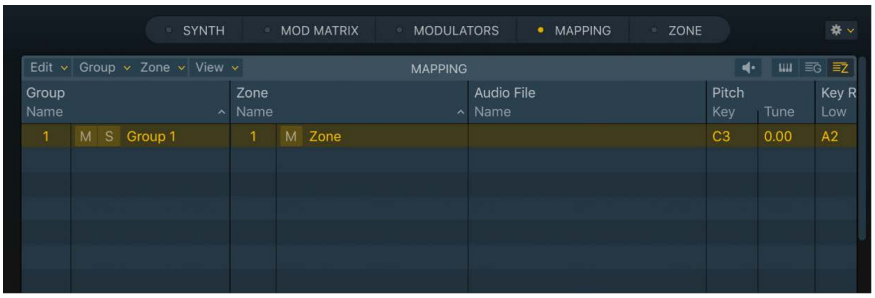


Create an empty zone, and assign a sample to it

1. In Logic Pro Sampler, do one of the following:

   - In Zone view, choose Zone > New to create a new, empty zone.
   - In the Key Mapping Editor, press and hold Command-Shift, then drag the pointer to create a new, empty zone.

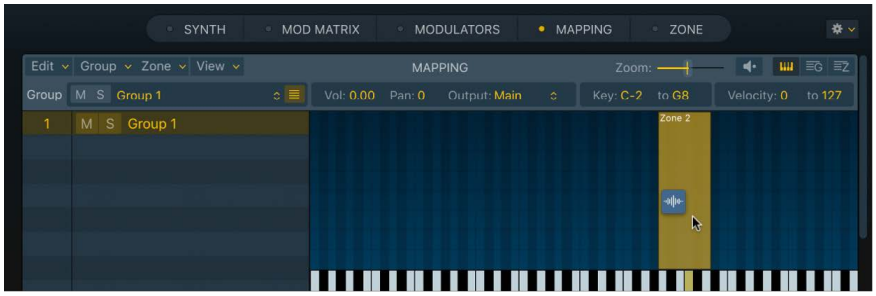
   A new group is shown in the Mapping pane and a zone is automatically added to the group.



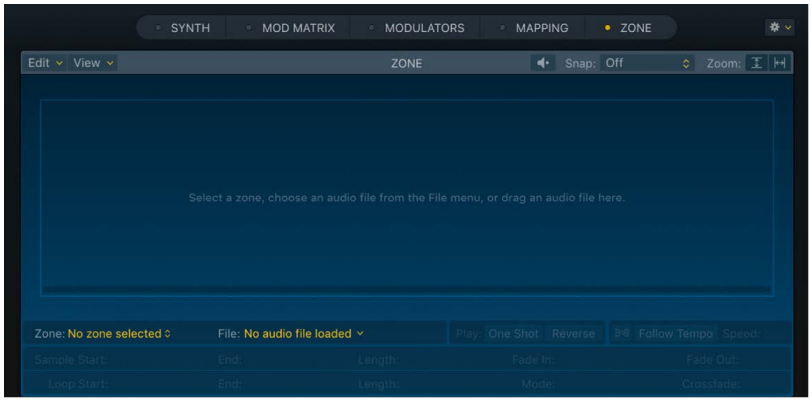
2. Do one of the following:

   - In Zone view, click the empty Name field for the zone in the Audio File column, then choose Load Audio File from the pop-up menu.
   - In Zone view, drag an audio file into the empty Name field of the zone in the Audio File column.
   - In the Key Mapping Editor, click a zone, then choose Zone > Load Audio Files (or use the Load Audio Files key command: Control-F).

     This method associates the audio file with the selected zone.
   - In the Key Mapping Editor, drag an audio file directly onto the zone.



   - In the Zone pane, if open, drag an audio file into the waveform display area.

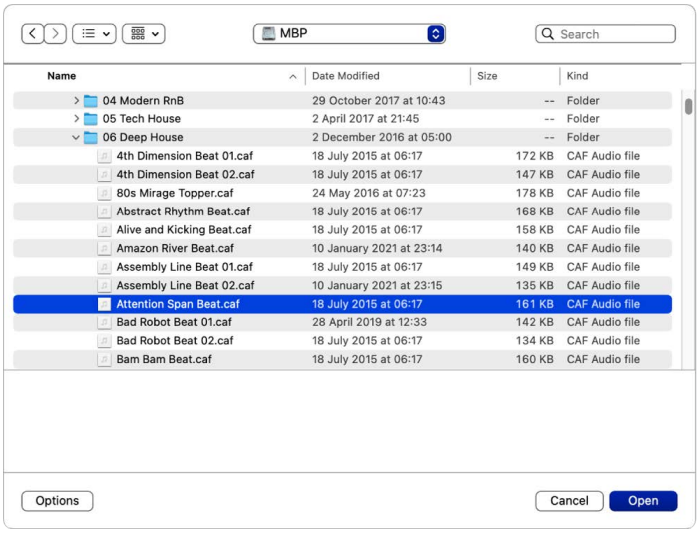


Create an assigned zone with menu commands

1. In Logic Pro Sampler, do one of the following:
   - In Zone view, Choose Zone > Load Audio Files (or use the Load Audio Files key command: Control-F).

     This method creates a new zone with the selected audio file.
   - In the Key Mapping Editor, click a zone, then choose Zone > Load Audio Files (or use the Load Audio Files key command: Control-F).

     This method associates the audio file with the selected zone.
2. Locate the audio file you want and select it in the File Selector window.



   - Click the Options button at the lower left of the File Selector window to show or hide the checkboxes and Play button.
   - Select “Hide used audio files” to dim the names of files used in the currently loaded sampler instrument.
   - Select “Preview audio file in Sampler Instrument” to temporarily replace the sample files in the currently selected zone. The zone is not directly triggered by selecting this option, but it can be triggered by playing MIDI notes while the File Selector window 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).
3. To preview looped playback of the currently selected audio file, click the Play button at the lower right of the File Selector window. The Play button label changes to Stop during playback.
   - You can step through files by using the Down Arrow key, or by clicking them, to audition each file in turn.
   - Click the Stop button to stop playback.
4. When you find an audio file you want to use, click the Open button to add it to the zone.

   When the audio file is loaded, the sample name is displayed in the Name field for the zone in the Audio File column in Zone view.

Create groups in Logic Pro for Mac Sampler

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 all hi-hat samples, for example.

In this context, the Sampler 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.

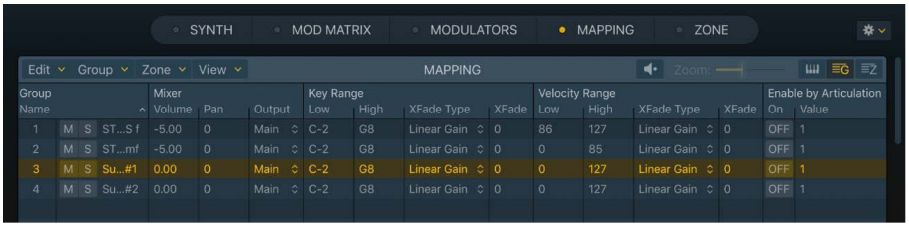
A group makes it possible to define a velocity range for all assigned zones, allowing you to specify a velocity window in which the grouped zones are triggered, for example. Each group also features offset parameters for envelope 1 (amplitude), envelope 2 (assignable), and filter settings made in the Synth pane. See Use Sampler Group view for information on parameters and options.

You can also play all zones without defining and assigning multiple groups—at least one group will always exist as the default location for all zones—in this case, changes to Synth and other parameter settings affect all samples (in all zones) equally.

Create a new group

1. In Logic Pro Sampler, click the Mapping button in the Navigation bar to view the Mapping pane, then click the Group view button if Group view is not shown.
2. Choose New from the Group menu.

   A new group appears in the Name column at the left side.



Add a zone to a group

In any of the Logic Pro Sampler Mapping pane views, do one of the following:

- Select an audio file or loop—then drag it onto a group name displayed in the Mapping pane.

  A new zone, containing an audio file or loop, is created. Shift-click or Command-click to select multiple files.

  Tip: You can add any type of loop from the Loop Browser, including MIDI loops, Step Sequencer loops, and Session Player loops.

- Select an audio file or loop—then drag it into the empty area below the last group name displayed in the Mapping pane. This creates a new group, containing the dragged zone, or zones.
- Choose New from the Zone menu.

  This creates a new zone in the selected group. If no group is selected, a new group is created—containing the new, empty zone.
- Choose Zone > Load Audio Files (or use the Load Audio Files key command: Control-F).

  This method creates a new zone with the selected audio file.

Assign an existing zone to a group

1. In Logic Pro Sampler, select a zone (or multiple zones) in the Key Mapping Editor or Zone view.

   Shift-click or Command-click to select multiple zones.

2. Choose Cut or Copy from the Edit menu.
3. Click the target group, then choose Paste from the Edit menu.

Delete a zone from a group

1. In Logic Pro Sampler, select a zone (or multiple zones) in the Key Mapping Editor or Zone view.

   Shift-click or Command-click to select multiple zones.

2. Press the Delete key or choose the Delete command from the Edit menu.

Delete all unused groups

1. In the Logic Pro Sampler Mapping pane, go to the Edit menu, then choose Select Unused.

   A group that doesn't contain at least one zone, is not part of a round robin, and has no keyswitch is considered unused, and is selected.

2. Press the Delete key or choose the Delete command from the Edit menu.

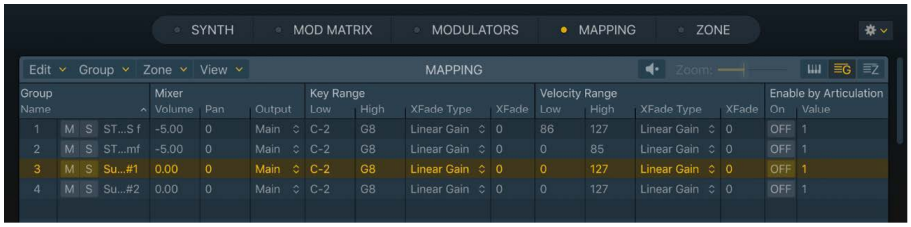
Edit Sampler zones and groups

Edit Sampler zones and groups in Logic Pro for Mac

You can extensively customize your instruments with the unique parameters offered by zones and groups.

You can use zone parameters to edit the pitch, velocity range, panorama, looping parameters, and other aspects of zones. See Use Sampler Zone view and Zone pane.

You can use group parameters to adjust the velocity and output, and to offset envelopes and filters for a group of zones, for example. See Use Sampler Group view.



Editing techniques, menu selection commands, and other parameter interactions that are shared by zones and groups are discussed in these sections:

- Select a zone or group in Sampler
- Graphically edit zones and groups in Sampler

Select a Sampler zone or group in Logic Pro for Mac

You can select and edit zones and groups in a number of ways. To make selection easier, you can first use the sort options.

Sort zones or groups

In Logic Pro Sampler, you can easily sort zones and groups in the Mapping pane Group view or Zone view. For example, to sort your zones alphabetically, click the Name subcolumn header in Zone view. If you want to sort groups by lowest to highest starting velocity, click the Low subcolumn heading in the Velocity Range column of Group view. The group with the lowest starting velocity range is displayed at the top of the list.

Do one of the following in Zone view or Group view:

- Click the subcolumn header that you want to sort by.
- Click the triangle in the subcolumn header to invert, or reverse, the sort order.

Select zones or groups in the Key Mapping Editor

In Logic Pro Sampler, you can select one or more zones or groups in the Key Mapping Editor.

When multiple groups are selected, you will see the zones of all selected groups in the Key Mapping Editor. Each selected group has a number that you can click to move focus to that specific group. The focus feature makes it easier for you to graphically edit complex mappings that contain multiple overlapping groups.

- The zones of the focused group are displayed in the foreground and can be graphically edited.
- The zones of the other selected groups are still visible but dimmed.
- Only zones in the focused group can be selected and edited or previewed.



- Click a group name to select it and to move focus to the group. Shift-click or drag (from an empty area outside any group) to select multiple groups. The Key Mapping Editor above the keyboard shows only zones that belong to the selected group, or groups. You can also remotely select groups with your MIDI keyboard.

  Note: To allow group selection with your MIDI keyboard, turn on the Select from Last Played Keys > Groups or Groups and Zones menu option in the Mapping pane Edit menu.

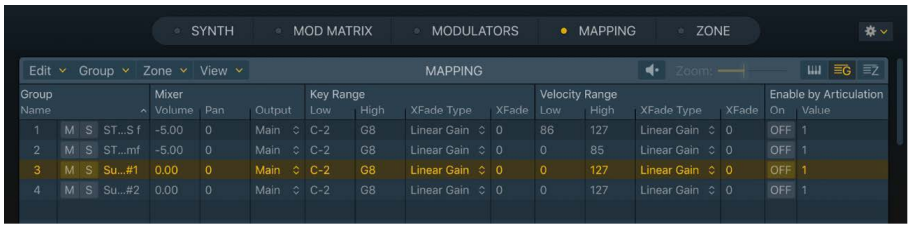
- Click a group number to move focus to that specific group when multiple groups are selected.
- Click a zone to select it. Shift-click or drag (from an empty area outside any zone) to select multiple zones. You can adjust zone parameters shown below the keyboard.

Select zones or groups using Edit menu commands

- In Logic Pro Sampler, 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.
  - *Deselect All:* Selects none of the zones and groups of the loaded sampler instrument.
  - *Select Unused:* Selects all unused zones and groups of the loaded sampler instrument.

    Note: If a group has focus in Zone view or the Key Mapping Editor, this will affect the criteria used by this command.
  - *Invert Selection:* Switches the current selection. For example, if you have selected all *unused* zones and groups of the loaded sampler instrument, choosing this command will select all *used* zones and groups.

Select zones or groups by clicking them or pressing arrow keys



- In the Logic Pro Sampler Mapping pane Group or Zone view, click zones and groups 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.

Use a MIDI keyboard to switch between Sampler zones or groups

In Logic Pro Sampler, choose Select from Last Played Keys > Groups and Zones menu option in the Mapping pane Edit menu.

- To switch between zones, press a key on a connected MIDI keyboard. You can continue to select zones by clicking them in the Mapping pane when this feature is turned on.
- To switch between groups, press a key on a connected MIDI keyboard. This is useful when you want to adjust the velocity of instrument groups, for example.

Graphically edit Sampler zones and groups in Logic Pro for Mac

You can edit zone and group parameters numerically in the Mapping pane Zone view and Group view. You can also graphically edit a number of zone and group parameters in the Key Mapping Editor and the Zone pane.

In addition, you can edit the audio file of a zone in the Logic Pro Audio File Editor.

Within each group mapping area (notes and velocities), zones can't overlap. When you force an overlap by dragging zones one above the other within the same group, the Key Mapping Editor automatically cuts zones in order to make space for other zones. This allows you to add zones or delete mapped areas without having to edit all affected zones directly. You can protect either the selected or unselected zones when dragging zones.



Move one or more zones by dragging

In the Logic Pro Sampler Key Mapping Editor:

- Drag the zone to the target Key Mapping Editor position.
- Shift-click or drag (from an empty area outside any zone) to select multiple zones, and drag them to the target Key Mapping Editor position.

Move one or more selected zones to the left or right with a key command

In the Logic Pro Sampler Key Mapping Editor:

- Select one or more zones, then 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

Restrict zone dragging to one direction

In the Logic Pro Sampler Key Mapping Editor:

- Press and hold the Command key to restrict movement to the initial horizontal or vertical drag direction.

Change the root key when moving a zone

The default behavior when dragging a zone to a key in the Key Mapping Editor is set with the “Always Move Root Key with Zone” option in the Zone menu.

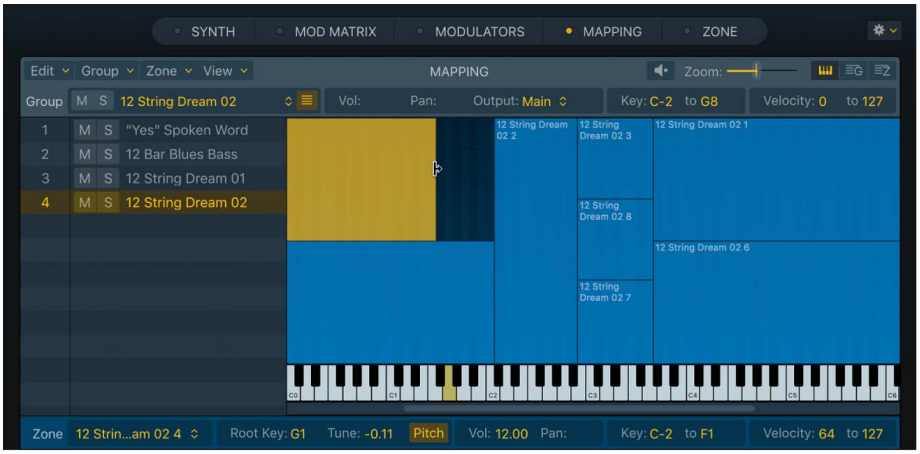
In the Logic Pro Sampler Key Mapping Editor do one of the following:

- Drag the zone. If the “Always Move Root Key with Zone” option in the Zone menu is *not* active, press and hold Option-Command while dragging the zone.
- Select a zone, then use one of the following key commands:
  - *Shift Selected Zone(s)/Group(s) Left (Zones incl. Root Key):* Option-Shift-Left Arrow key
  - *Shift Selected Zone(s)/Group(s) Right (Zones incl. Root Key:* Option-Shift-Right Arrow key

Change the start or end note of a zone

In the Logic Pro Sampler Key Mapping Editor:

- Drag the start or end point of the zone or group to the target position.



Change the velocity range of a zone

In the Logic Pro Sampler Key Mapping Editor:

1. Move the pointer to the top or bottom of a zone (the pointer changes to the resize icon).

   Shift-click to select multiple zones if you want to simultaneously adjust their velocity range.



2. Drag the resize icon upward to raise the value, or downward to reduce the value.

   If you drag over (overlap) another zone assigned to the same key range in the group, its velocity range will be resized. If you want to layer zones, use different groups.

Divide a zone vertically

In the Logic Pro Sampler Key Mapping Editor:

1. Select a zone.
2. Press and hold Command-Option to display a vertical cut icon at the pointer position, then click the zone.

   The zone is divided vertically and a new zone is created. Each zone can be independently edited.



Divide a zone horizontally

In the Logic Pro Sampler Key Mapping Editor:

1. Select a zone.
2. Press and hold Shift-Option to display a horizontal cut icon at the pointer position, then click the zone.

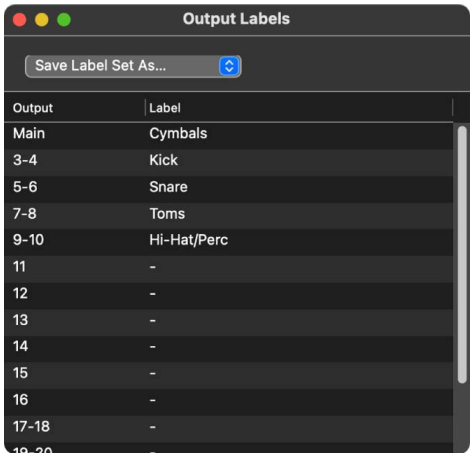
   The zone is divided horizontally and a new zone is created. Each zone can be independently edited.



Sampler zone and group output labels in Logic Pro for Mac

You can name the outputs used by instrument zones or groups and can save these names as a label set.

Output label names are shown in the Mapping pane Group and Zone views and on Aux channel strips in the Mixer window when using a multi-output instance of Sampler.



Set output labels for zones and groups

In the Logic Pro Sampler Mapping pane:

1. To open the Output Labels window, choose Edit > Edit Output Labels.
2. Click the field in the Label column to the right of the Output that you want to rename, and enter the new name.
3. Press Return, or click outside the field to complete text entry.
4. Select the new name from the Output column pop-up menu for the zone or group in the Mapping pane Group or Zone view.

Save an output label set

- In Logic Pro Sampler, choose Save Label Set As from the pop-up menu in the Output Labels window.

Load an output label set

- In Logic Pro Sampler, choose the label set name from the pop-up menu in the Output Labels window.

Reset an output label set

- In Logic Pro Sampler, 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.

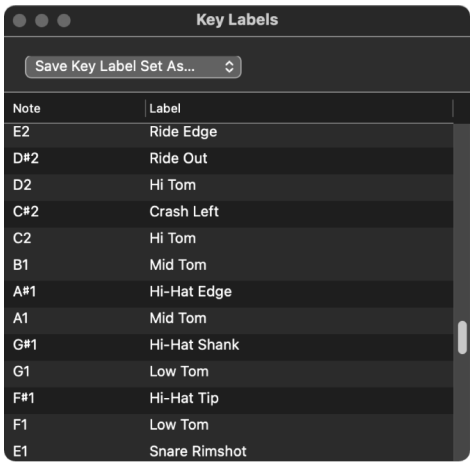
Delete an output label set

1. In Logic Pro Sampler, 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.

Use key labels in Sampler in Logic Pro for Mac

You can add names to keys used by instruments and save these names as a key label set. This is ideal for drum kits.

Key label names are shown in the Logic Pro Piano Roll and Step Sequencer edit windows.



Set key labels for zones and groups

In the Logic Pro Sampler Mapping pane:

1. To open the Key Labels window, choose Edit > Edit Key Labels.
2. Click the field in the Label column to the right of the note that you want to add a name to, and enter the new name.
3. Press Return, or click outside the field to complete text entry.

Save a key label set

1. In Logic Pro Sampler, choose Save Key Label Set As from the pop-up menu in the Key Labels window.
2. Enter a name in the Key Label Set Name field, then click Save.

   The key label set name is shown in the pop-up menu in the Key Labels window.

Load a key label set

- In Logic Pro Sampler, choose the key label set name from the pop-up menu in the Key Labels window.

Delete a key label set

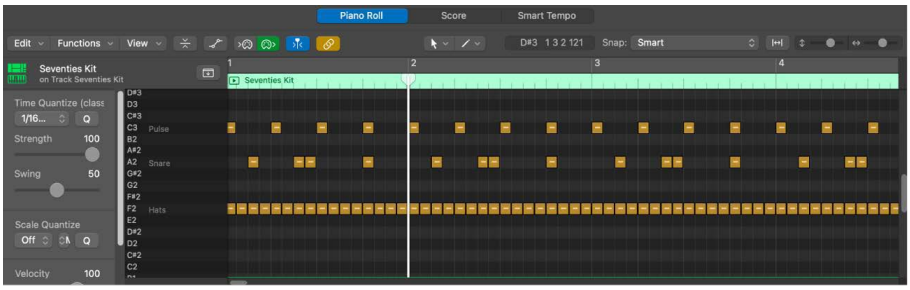
1. In Logic Pro Sampler, choose the key label set name from the pop-up menu in the Key Labels window.
2. Choose Delete “key label set name” from the pop-up menu in the Key Labels window.

Use key labels in a MIDI region in Logic Pro

In the Logic Pro Sampler Mapping pane:

1. To open the Key Labels window, choose Edit > Edit Key Labels.
2. Click the field in the Label column to the right of the note that you want to add a name to, and enter the new name.
3. Press Return, or click outside the field to complete text entry.
4. Open the Piano Roll editor to view the contents of an existing MIDI region used for Sampler. If no region exists, Control-click on the Sampler track then choose Create MIDI Region from the shortcut menu.

   Your edited key label names are shown to the right of the Piano Roll editor keyboard.

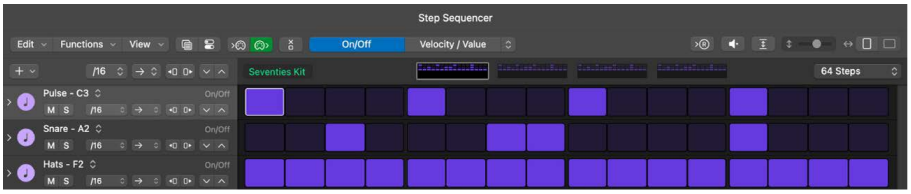


Use key labels in a Pattern region in Logic Pro

In the Logic Pro Sampler Mapping pane:

1. To open the Key Labels window, choose Edit > Edit Key Labels.
2. Click the field in the Label column to the right of the note that you want to add a name to, and enter the new name.
3. Press Return, or click outside the field to complete text entry.
4. Open the Step Sequencer to view the contents of an existing Pattern region used for Sampler. If no region exists, Control-click on the Sampler track then choose Create Pattern Region from the shortcut menu.

   Your edited key label names are shown in Step Sequencer row headers, in place of note names.



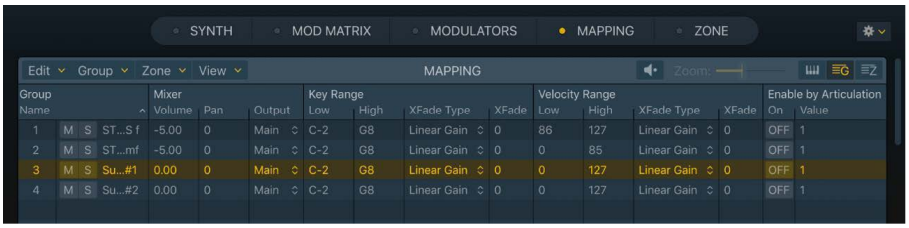
Sampler articulation handling in Logic Pro for Mac

Sampler includes articulation functions you can use to directly modulate or switch between grouped or individual samples. This allows a number of expressive playing and playback possibilities, such as switching between pizzicato and bowed strings, or triggering instruments with different articulations such as slides, scoops, and falls.

In Sampler, each group can be assigned an identifier called an *articulation ID*. The articulation ID is a property of a note event, and can be added and edited in the Piano Roll Editor, Event List, or Score Editor, or used within MIDI scripts. If Sampler receives a matching identifier, it selects the group or groups with this assigned articulation ID for playback. Non-matching groups are silent. See Make advanced group selections in Sampler.

View and edit articulations in Sampler

1. In Logic Pro Sampler, click the Navigation bar Mapping button to view the Mapping pane.
2. Click the Group view button at the top right of the Mapping pane.



3. Click the ON button in the Enable by Articulation column.

   The articulation ID number appears in the Value column. For included software instruments that use articulations, the group name usually reflects the articulation associated with the ID number. For example, *Brass Falls*.

   An instrument may have a series of groups set to the same note range, each set to use a different articulation. When a note event in the specified range occurs, only the group whose articulation Value corresponds with the articulation ID number for the note event plays.

4. Vertically drag the numeral in the Value column to set the articulation ID number for the selected group.

Create a modulation routing to dynamically change articulations

You can route a modulation source, such as a MIDI controller, to an Articulation ID target parameter. This lets you dynamically change or switch between articulations using the source MIDI controller.

1. In Logic Pro Sampler, click the Navigation bar Mod Matrix button to view the Mod Matrix pane.
2. Click the Plus button (+) to create a new modulation routing.
3. In the modulation routing, choose the controller you want to use from the Source pop-up menu.

   For example, you could assign a MIDI foot pedal to switch between two layered groups: one containing a bowed string section and the other containing a plucked string section.

4. Choose Articulation ID from the Target pop-up menu.
5. Drag the Amount slider all the way to the right, so it is set to 100%.

When you manipulate the controller, the articulation ID changes through the range of used values.

Manage sampler instruments

Manage Sampler instruments in Logic Pro for Mac

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

It is recommended that you copy any sampler instruments and all associated audio files to a local or networked hard drive. This provides fast access to your sampler instruments and makes it easier to organize your instrument library to meet your needs. You can play back samples that exceed the size of your computer RAM by streaming them from hard disk. This feature is not practical for optical drives.

In Logic Pro, you can save sampler instruments and associated audio files with the project bundle. This makes it easy to keep all sampler instruments and audio data used in your project in one location.

Organize your sampler instruments into a preferred hierarchy

1. Create a folder on the desktop—named Basses, for example—and drag it into the ~/Music/Audio Music Apps/Sampler Instruments/ folder.
2. Drag the sampler instruments you want to move into this newly created folder.

   The modified menu structure is reflected in the Settings pop-up menu.

   The Settings pop-up menu displays submenus only for folders that actually contain sampler instrument files. Other folders are not added to the Settings pop-up menu. Aliases pointing to folders that contain sampler instrument files outside the ~/Music/Audio Music Apps/Sampler Instruments/ folder can also be added to the Settings pop-up 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. Copy the sampler instrument file into the ~/Music/Audio Music Apps/Sampler Instruments/ folder.
2. Copy the associated samples into ~/Music/Audio Music Apps/Samples/*Instrument Name*/ folder.

Sampler storage locations in Logic Pro for Mac

Sampler instruments and samples are stored in specific locations.

Sampler instrument storage locations

To be visible in the Settings pop-up menu, instruments must be stored in the following folder locations:

- *~/Music/Audio Music Apps/Sampler Instruments/:* User-defined or edited instruments are stored here.
- *~/Music/Audio Music Apps/Sampler Instruments/Auto Sampled/:* Instruments created with the Auto Sampler utility plug-in are stored here.
- *~/Music/Logic Pro Library.bundle/Plug-In Settings/Sampler/:* Supplied sampler instruments are installed here.

  Note: You can move the Logic Pro Library.bundle to any location on your local or external drive. See Logic Pro Sound Library settings.
- *…/ProjectName:* Logic Pro also searches for sampler instruments in the project folder.
  - If you use the Save As command, you have the choice to save older project files as a project bundle or as a folder.
  - If you save a new project, sampler instruments and samples can be saved as Assets with the project bundle. See Project settings in the Logic Pro User Guide.

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), and they are shown in the Settings pop-up menu.

Sample storage locations

Samples are stored in the following folders:

- *~/Music/Audio Music Apps/Samples/:* User-defined or edited samples are stored here.
- *~//Music/Audio Music Apps/Samples/Auto Sampled/:* Samples created with the Auto Sampler utility plug-in are stored here.
- *~/Music/Logic Pro Library.bundle/Samples/:* Supplied samples are installed here.

  Note: You can move the Logic Pro Library.bundle to any location on your local or external drive. See Logic Pro Sound Library settings.

SoundFont2, DLS, and Gigasampler files in Logic Pro for Mac

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

You can store your imported sampler instruments in any folder on any of your computer hard drives. You can access these instruments from the Settings pop-up menu of a Mixer's channel strip if stored in the default location: ~/Music/Audio Music Apps/Sampler Instruments folder.

If your files are stored elsewhere, you will need to create an alias in the ~/Music/Audio Music Apps/Sampler Instruments folder that points to the target folder.

Add SoundFont2, DLS, or Gigasampler files to your Sampler library

1. Open Finder.
2. Copy or move your SoundFont2, DLS, or Gigasampler files into the ~/Music/Audio Music Apps/Sampler Instruments/ folder.

   A sampler instrument file is created in the Sampler Instruments folder.
3. The raw samples associated with the sampler instrument are placed automatically in a folder named after their original format in ~/Music/Audio Music Apps/.
4. To use the added instrument, choose it from the format submenu in the Settings pop-up menu.

SoundFont2 files can contain multiple sounds, stored in "banks," in addition to single-instrument files.

When you load a SoundFont2 bank into Sampler, it creates a bank folder (.bk) and a Samples folder, named after the SoundFont2 bank file. The .bk suffix is appended to each folder name. A sampler instrument file is automatically created for all sounds in the bank and is placed in the new bank folder.

The Settings 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 dozens of individual drum kits from several different vintage drum machines, into Sampler, the following happens:

- A new folder named Vintage Drums.bk is created in the ~/Music/Audio Music Apps/Sampler Instruments/E-MU/ folder.
- A second folder named Vintage Drums.sf2.Samples is created in the ~/Music/Audio Music Apps/SoundFont Samples/ folder.
- The Settings pop-up menu hierarchy is updated and the original Vintage Drums entry is replaced with a Vintage Drums.bk entry. This new entry is a folder that contains the individual sampler instruments, which can be selected and loaded as usual.

Sampler memory management in Logic Pro for Mac

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 random-access memory (RAM). To use these huge sampler instruments, Sampler can use a portion of your hard drive as virtual memory. When you turn on Sampler virtual memory, only the initial attacks of audio samples are loaded into the computer 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 Sampler 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 varies in accordance with the number and size of audio files in the project, and other plug-ins used. Sample playback instrument plug-ins not made by Apple can significantly affect the amount of RAM that Logic Pro uses.

Sampler extended parameters in Logic Pro for Mac

You can use Sampler extended parameters to allow use with MPE and guitar controllers, providing enhanced expression capabilities. You can access these parameters by clicking the disclosure arrow 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 active mode, each voice receives on a different MIDI channel. Per-voice channels support pitchbend, aftertouch, modwheel, and CC messages. Controllers and MIDI messages sent on the base channel affect all voices.

- *Mono Mode Bend Range slider:* Set a value from 0 to 96.

  The chosen pitch bend range affects individual note pitchbend messages received on all but the common base channel. The default is 48 semitones, which is compatible with the GarageBand for iOS 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

Sculpture overview in Logic Pro for Mac

This section contains key information and concepts that you need to understand before looking at Sculpture features and parameters. If you're new to synthesizers, it might be best to start off with Synthesizer basics overview, which will introduce you to the terminology and give you an overview of different synthesis methods and how they work.

If you're new to using plug-ins in Logic Pro, see Add, remove, move, and copy plug-ins.

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 lets you 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 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.

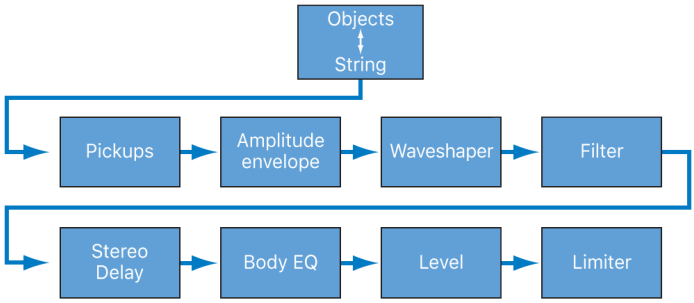
Like a real instrument, Sculpture generates sounds by using an object, such as a fingertip, wind, drumstick, or violin bow, to stimulate another object, such as a guitar string or reed.

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 lets you virtually model the physical consistency and behavior of all components involved—hence *component modeling* synthesis.



This figure shows the signal flow of the core Sculpture 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 sent to an EQ-like module (the Body EQ), which simulates the spectral shape/body response of your instrument, and then processed by an integrated delay effect. 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
- Basic Sculpture sound programming
- Electric bass programming in Sculpture
- Synthetic sound programming in Sculpture
- Explore Sculpture modulations

Sculpture is an instrument that requires some investment of your time, but it can 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 in Logic Pro for Mac



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. See Sculpture string overview.
  - *Object parameters:* The area at the top left contains the objects that are used to stimulate or otherwise affect the string in some way. See Sculpture objects overview.
  - *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 effect, Body EQ, and output 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 lets you 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 in Logic Pro for Mac

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.

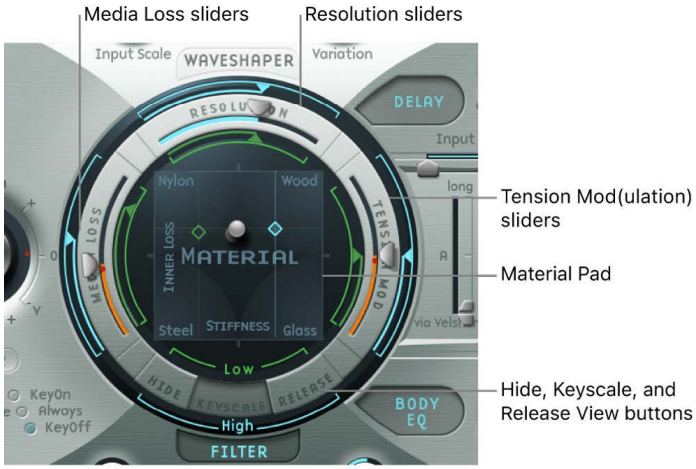
Sculpture's string and the excite/disturb objects are similar to the oscillators in traditional synthesizers. The string is considerably more sophisticated in concept than simple oscillators, however.

In essence, you are creating the waveform, or base timbre, by mathematically describing the string 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 the Sculpture 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 the Sculpture string is still vibrating for a specific note, retriggering that same note interacts 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 alters 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. A number of parameters can be morphed between up to five morph points. These are indicated in the parameter descriptions. See Sculpture morph overview.



Material Pad in Keyscale view

String parameters

- *Hide, Keyscale, and Release view buttons:* Show or hide different groups of parameters.
- *Material Pad:* Determine 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 view modes in Logic Pro for Mac

Click the Hide, Keyscale, or Release button to show or hide the corresponding parameters in the ring surrounding the Material Pad.



Keyscale view



Release view



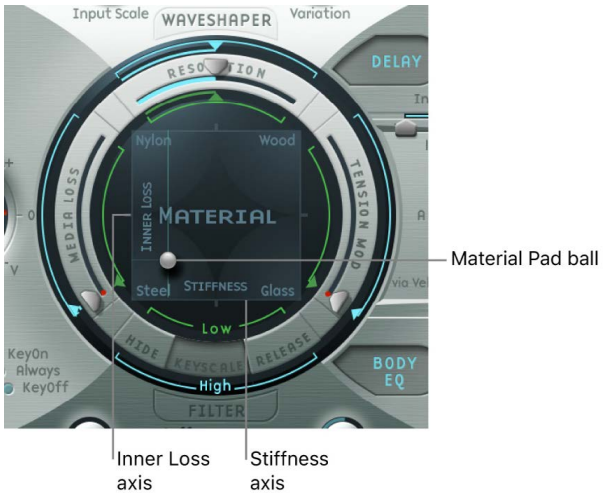
Hide view

Hide, Keyscale, and Release parameters

- *Hide button:* Hide several elements, simplifying the interface and making it impossible to accidentally change Key Scale or Release parameters.
- *Keyscale button:* Set 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:* Set string Release parameters, which affect the vibrations of the string after the key is released.

Sculpture basic Material Pad controls in Logic Pro for Mac

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.

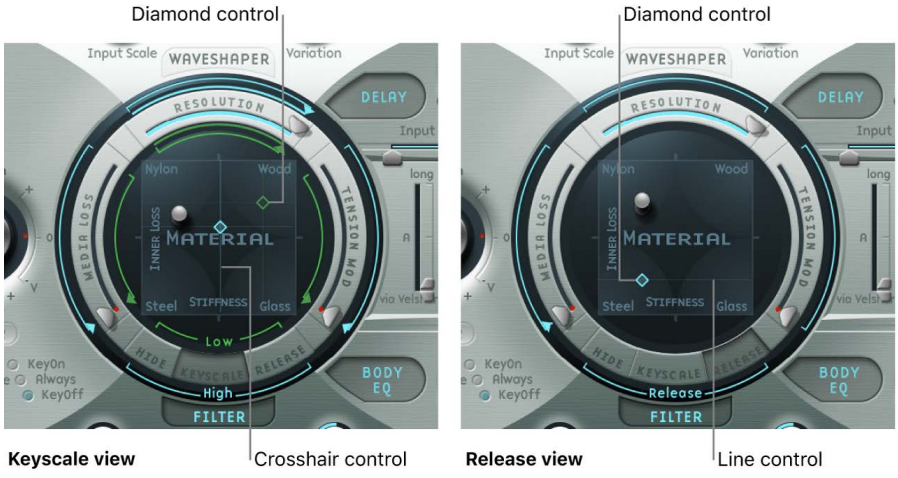
Material Pad parameters

- *Inner Loss:* Emulate 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:* Set 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:* Drag to control both the Inner Loss and Stiffness parameters simultaneously. 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 Sculpture string parameter sliders.

Sculpture Material Pad Keyscale or Release view in Logic Pro for Mac

In Keyscale or Release view, the Material Pad shows additional Keyscale and Release parameter controls.



Material Pad Keyscale and Release parameters

- *Diamond controls:* 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 lets you 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. In Logic Pro, 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. In Logic Pro, 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.

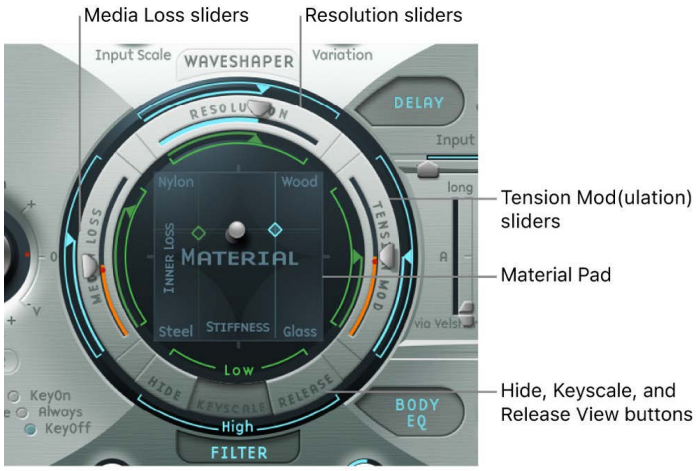
Adjust Inner Loss release scaling in the Material Pad

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 Sculpture string parameter sliders.

1. In Logic Pro, click the Release button.
2. Vertically drag the blue Release line.

Sculpture string parameter sliders in Logic Pro for Mac

The sliders on the outer ring of the Material Pad refine the properties and behavior of the string. High Scaling sliders are blue. Low Scaling sliders are green.



Material Pad in Keyscale view

String parameter sliders

- *Resolution slider:* 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. Resolution behavior can be changed with the Render Mode pop-up menu in the extended parameters. In Keyscale view, the Resolution High and Low Scaling sliders are shown:
  - *Resolution High Scaling slider:* Set the key tracking resolution—the accuracy of key tracking—for notes above middle C (C3).
  - *Resolution Low Scaling slider:* Set the key tracking resolution for notes below middle C.
- *Media Loss slider:* 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:* In Keyscale view, set the key tracking resolution for notes above middle C (C3). In Release view, set media loss behavior when the key is released.
  - *Media Loss Low Scaling slider:* In Keyscale view, set the key tracking resolution for notes below middle C (C3). In Release view, set media loss behavior when the key is released.

- *Tension Mod slider:* Set the momentary detuning of the string.
  - *Tension Mod High Scaling slider:* Set the tension modulation behavior for notes above middle C.
  - *Tension Mod Resolution Low Scaling slider:* 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. In Logic Pro, 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. In Logic Pro, 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. In Logic Pro, 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. In Logic Pro, 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 in Logic Pro for Mac

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. A number of parameters can be morphed between up to five morph points. These are indicated in the parameter descriptions. For more information, see Sculpture morph overview.

Important: At least one object must be used to excite or disturb the string, because the string itself does not make any sound.

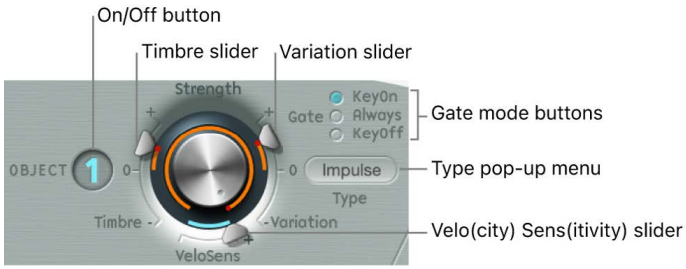
There are a number of different string *excite/disturb/damp* models available, such as blow, pluck, bow, and so on. These can radically alter the general timbre of the string attack phase, making it possible to create bowed or plucked flute and bell sounds, or guitars with a flute-like attack, for example.

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 affects the string. This in turn alters the interaction of *any* other active object with the string, often resulting in a completely different sonic character.

The goal of changing the sound 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 affects 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, 3):* Turn the object on or off.
- *Type pop-up menus:* Choose the excite, disturb, or damp type. Your choice affects string interaction and changes the behavior of the Timbre and Variation controls. See Sculpture excite objects 1 and 2 and Sculpture disturb and damp objects 2 and 3.
- *Gate mode buttons:* 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:* Set the intensity of the excitation/disturbance (depending on type). A value of 0.0 effectively stops excitation/disturbance. In contrast to the On/Off button of each object, you can fade in the Strength parameter with modulation or morphing options. This parameter can be morphed.
- *Timbre slider:* Determine the timbre (tonal color) of the chosen excitation/disturbance type. Behavior varies between object types. Zero (0.0) is the default value for the object. Positive values make the sound brighter. Negative values lead to a more mellow sound. This parameter can be morphed.
- *Variation slider:* Adjust this additional timbre parameter. Behavior varies between object types. This parameter can be morphed.
- *VeloSens slider:* 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 objects 1 and 2 in Logic Pro for Mac

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 objects 2 and 3.

The table outlines all excite types available for Objects 1 and 2, and provides information on 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 objects 2 and 3 in Logic Pro for Mac

The table outlines 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 objects 1 and 2.

Name	Description	Strength controls	Timbre controls	Variation controls
Disturb	A disturb object that is placed at a fixed distance from the string 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 string 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 is 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
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 resting position	The slope (steepness) of the boundary. A value of 0.0 places the boundary parallel to the string. Other values move the boundary closer to the string on one end and farther away on the other.	The amount of reflection at the boundary limits

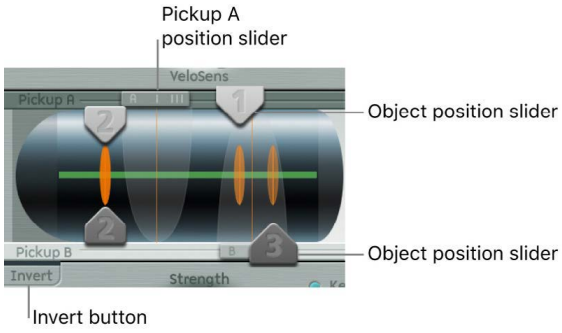
Name	Description	Strength controls	Timbre controls	Variation controls
Mass	Used to model an additional mass attached to the string. This can lead to inharmonic sounds, and very interesting results, if the position of this mass is modulated along the string.	The mass size/ weight	No effect	No effect
Damp	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

Sculpture pickup parameters in Logic Pro for Mac

The pickup parameters discussed in this section apply on a per-voice basis. A number of parameters can be morphed between up to five morph points. These are indicated in the parameter descriptions. See Sculpture morph overview.

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 can show the range of string 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:* Drag sliders 1, 2, and 3 to determine the respective position of each (excite/disturb/damp) object along the string. This parameter can be morphed.
- *Pickup A position slider:* 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. This parameter can be morphed.
- *Pickup B position slider:* Set the position of Pickup B along the string. This parameter can be morphed.
- *Invert button:* Invert the phase of Pickup B. Options are: normal or inverted.

  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.

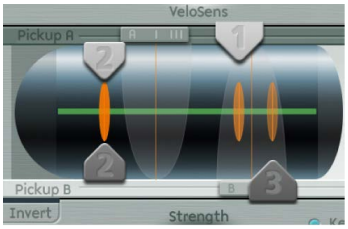
Adjust the Pickup A or Pickup B position

- In Logic Pro, drag the slider handle at the top or bottom of the Pickup display.

Adjust an object pickup position

- In Logic Pro, 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

- In Logic Pro, Control-click the green horizontal line—the string—to turn string animation on or off.

  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 turn it off if your computer is struggling to process all data in real time.

Sculpture spread controls in Logic Pro for Mac

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.

Tip: You can create animated width and chorus effects by modulating the Pickup Position parameters with an LFO or other modulator.

Sculpture global parameters in Logic Pro for Mac

These are found across the top of the Sculpture interface, unless otherwise specified.



Global parameters

- *Glide Time field:* 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:* Fine-tune the entire instrument, in cents. A cent is 1/100 of a semitone.
- *Warmth field:* Slightly detune each voice to warm or thicken the sound. This parameter emulates the random fluctuations caused by the components and circuitry of analog synthesizers.

- *Transpose field:* 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:* Specify the number of voices that can be played at any one time. Sixteen voices is the maximum polyphony of Sculpture.
- *Keyboard Mode buttons:* 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 the 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 happens 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.
- *Bender Range Up/Down fields:* Set the upward/downward pitch bend range.
  - Separate settings are available for upward and downward pitch bends—using your MIDI keyboard 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, alters the shape of the modeled string, rather than acting as a simple pitch bend.

Sculpture amplitude envelope in Logic Pro for Mac

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

- *Attack Soft/Hard sliders:* Set the attack time for a note played at maximum velocity with the lower slider (Soft). Set the attack time for a note played at minimum velocity with the top slider (Hard). 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.

- *Decay slider:* Set the time it takes for the signal to fall to the sustain level, following the initial strike/attack time.
- *Sustain slider:* Set the sustain level. The sustain level is held until the key is released.
- *Release slider:* 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 Waveshaper in Logic Pro for Mac

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 the Korg O1/W.



Waveshaper parameters

- *Waveshaper On/Off button:* Turn the Waveshaper on or off.
- *Type pop-up menu:* Choose one of four waveshaping curves. See the table.
- *Input Scale knob:* 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. This parameter can be morphed.

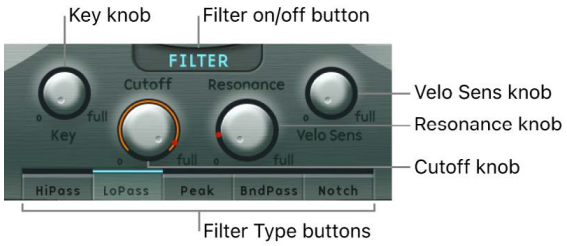
  Note: Given its impact on the harmonic spectrum, Input Scale should be viewed and used as a timbral control, rather than a level control. Also note that extreme Input Scale values can introduce processing noise at the Waveshaper output.
- *Variation knob:* Set the value of the preassigned parameter for the selected Waveshaper curve. Behavior varies between curve types. This parameter can be morphed. 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 in Logic Pro for Mac

The parameters discussed in this section apply on a per-voice basis. A number of parameters can be morphed between as many as five morph points. These are indicated in the parameter descriptions. For more information, see Sculpture morph overview.

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.



Filter parameters

- *Filter On/Off button:* Turn the filter section on or off.
- *Filter type buttons:* Click to choose the filter mode.
  - *HiPass:* Allow 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:* Allow 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:* Choose to specify the center of a frequency band with Cutoff. Control bandwidth and gain with Resonance. Frequencies outside the band remain at their current level. Peak filters are generally used to enhance a frequency range.
  - *BandPass:* Allow the frequency band surrounding the center frequency 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:* Cut the frequency band surrounding the center frequency. 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:* 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. This parameter can be morphed.
- *Resonance knob:* Set the filter resonance value. This parameter can be morphed.
  - 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 knob:* 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 turns off 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:* 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 turns off velocity sensitivity. A value of 1.0 results in maximum velocity sensitivity.

Sculpture Body EQ parameters

Sculpture Body EQ overview in Logic Pro for Mac

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, allowing you to alter these properties separately.

The Body EQ affects the summed signal of *all* voices, rather than each voice independently.



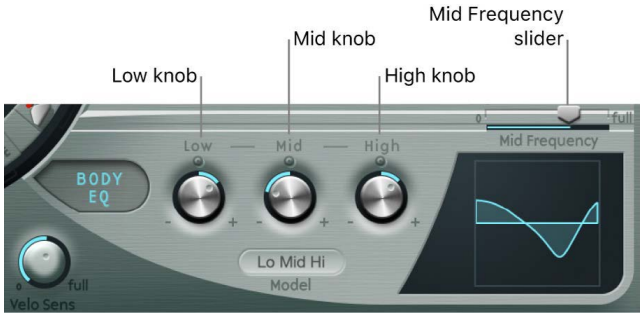
Body EQ global parameters

- *Body EQ On/Off button:* Turn 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 Sculpture Basic EQ model, and Sculpture Body EQ models.

Sculpture Basic EQ model in Logic Pro for Mac

The Basic EQ parameters differ from other EQ models.



Basic EQ parameters

- *Low/Int knob:* Set the gain of a low shelving filter (or scale the intensity of model formants).
- *Mid/Shift knob:* Set the gain of a peak filter (or shift the formants logarithmically). This is sweepable—see “Mid Frequency/Fine Structure slider” below.
- *High/Stretch knob:* Set the gain of a high shelving filter (or stretch the formant frequencies, relative to each other).
- *Mid Frequency/Fine Structure slider:* Sweep the center frequency of the mid band between 100 Hz and 10 kHz (or enhance the spectral (harmonic) structure).

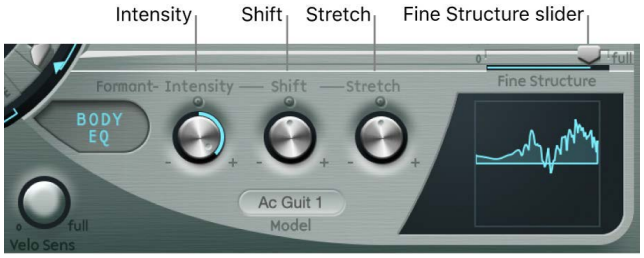
Adjust the Basic EQ (Lo Mid Hi model) graphically

In Logic Pro, do any of the following:

- *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.

Sculpture Body EQ models in Logic Pro for Mac

All other Body EQ models provide the following parameters:



Body EQ parameters

- *Formant–Intensity knob:* Rotate to scale the intensity of model 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

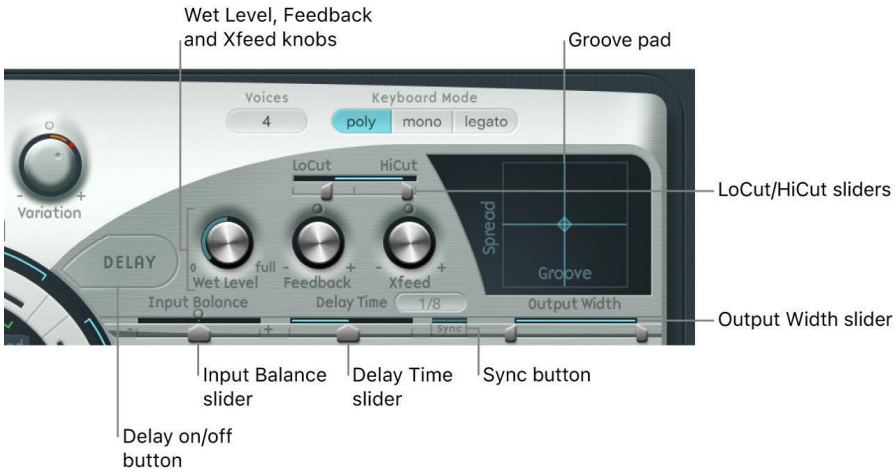
In Logic Pro, do the following:

- *To control the Formant Intensity parameter:* Drag the graphic vertically.
- *To control the Formant Shift parameter:* Drag the graphic horizontally.

Sculpture delay effect parameters

Sculpture delay effect overview in Logic Pro for Mac

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.



Sculpture Delay effect parameters

- *Delay On/Off button:* Turn the Delay effect on or off.
- *Wet Level knob:* Set the Delay output level.
- *Feedback knob:* 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 knob:* Set the amount of delay signal that is fed from the delay unit 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 the Extended Parameters area.
- *LoCut slider:* Determine the cutoff frequency of the highpass filter at the delay line output/feedback loop.
- *HiCut slider:* Determine the cutoff frequency of the lowpass filter at the delay line output/feedback loop.
- *Groove Pad:* Graphically adjust delay times in stereo or surround instances. See Sculpture Groove Pad (stereo) and Sculpture Groove Pad (surround).
- *Input Balance slider:* 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:* 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:* 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 Groove Pad (stereo) in Logic Pro for Mac

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:* Distribute delay taps to the left/right channels rather than smearing them across channels, 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 Groove Pad (surround) in Logic Pro for Mac

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 output parameters in Logic Pro for Mac

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.

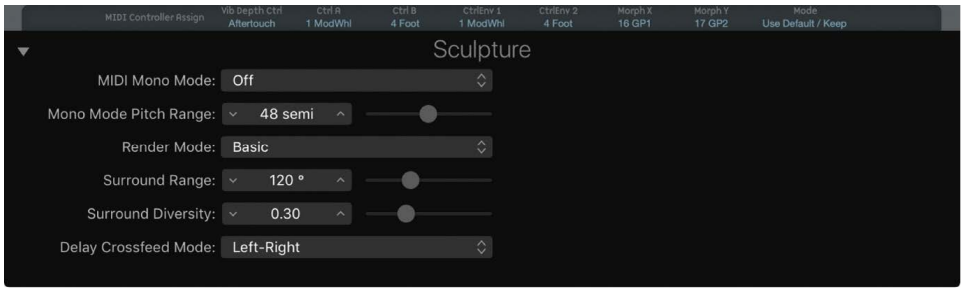


Output parameters

- *Level knob:* Set the overall output level of Sculpture.
- *Level Limiter mode buttons:*
  - *Off button:* Turn the Level Limiter off.
  - *Mono button:* Turn on a monophonic limiter that processes the summed signal of all voices.
  - *Poly button:* Turn on a polyphonic limiter that processes each voice independently.
  - *Both button:* Turn on a combination of both limiter types.

Sculpture surround range and diversity in Logic Pro for Mac

In surround instances, Sculpture Extended Parameters display 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 pickup 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 position carry the signal. A diversity of 1 means that all speakers carry an identical amount of the signal.

Sculpture modulation controls

Sculpture modulation overview in Logic Pro for Mac

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 in Sculpture

- In Logic Pro: 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 in Logic Pro for Mac

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 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 are *not* phase-locked.

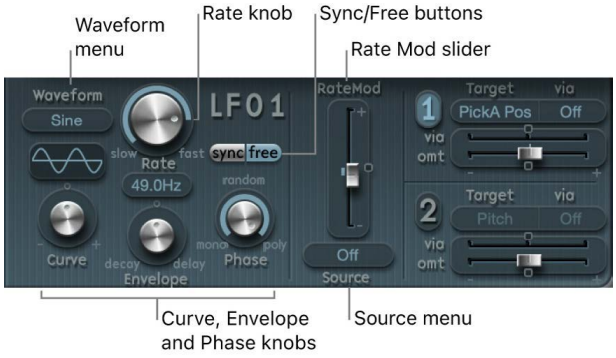
If a random (in-between) value is used, some notes are modulated synchronously, and others are not modulated synchronously.

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.
- *Waveform display:* Shows the results of changes to the Waveform pop-up menu and Curve knob parameter settings.
- *Curve knob:* 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 and field:* 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:* Choose either synchronized or free-running LFO rates. These buttons interact with the Rate knob. The synchronized value is derived from the Logic Pro for Mac tempo and meter.
- *Envelope knob:* Set the time it takes for the LFO modulation to fade in or fade out. See Modulate Sculpture LFOs.
- *Phase knob:* 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:* Set the intensity—the amount—of LFO rate modulation.

Sculpture LFO waveforms in Logic Pro for Mac

The LFO Waveform pop-up menus set different waveforms for the LFOs. The table 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 three 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 S & H from CtrlA setting is controlled by the CtrlA modulation source. For example, you could assign aftertouch as the CtrlA source to trigger S & H modulation of pitch. 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 in Logic Pro for Mac

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 occurs.



LFO target and source parameters

- *LFO 1/2 On/Off buttons:* Turn each LFO on or off, independently.
- *Modulation Target pop-up menus:* Choose the modulation targets.
- *Via source pop-up menus:* Choose (or turn off) 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:* Set the maximum via modulation amount, such as when the Modulation wheel is at the maximum position, for example.

Set the LFO modulation fade time in Sculpture

In Logic Pro, do the following:

- *To fade in the modulation:* Rotate the Envelope knob to a positive value.

  The higher the value, the longer 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 the fade-out time.

Set up a delayed vibrato in Sculpture

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

1. In Logic Pro, 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 in Logic Pro for Mac

One LFO is hard-wired to pitch, for vibrato effects, periodic pitch modulations. You can adjust the strength of the vibrato effect with 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.



Vibrato parameters

- *Waveform pop-up menu:* Choose the waveform used for vibrato. See Sculpture LFO waveforms.

  Note: There are two special rectangular waves, Rectangle unipolar (shown as Rect01 when active) and Rectangle bipolar (Rect11 when active)—the former switching between values of 0.0 and 1.0, and the latter switching between values of –1.0 and +1.0.

- *Curve knob:* 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:* 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.
- *Rate knob and field:* Set the rate of vibrato, which can be either synced to the host application tempo or set independently in Hz 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 Logic Pro for Mac tempo and meter.
- *DepthviaVibCtrl sliders:* Drag to define the impact of the controller assigned to Vib Depth Ctrl (see Define Sculpture MIDI controllers).
  - *Via slider:* Determine the modulation intensity of the controller assigned to Vib Depth Ctrl.
  - *Amt slider:* Determine the maximum modulation amount.

Sculpture Jitter generators in Logic Pro for Mac

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.

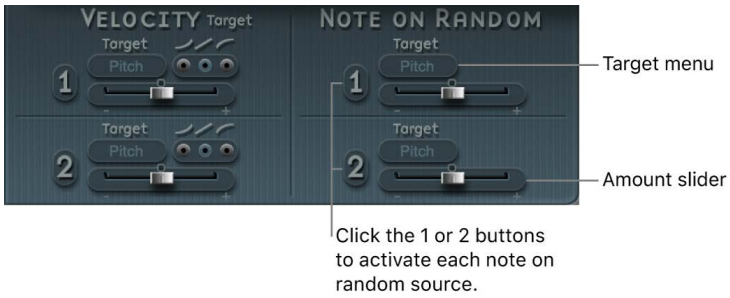


Jitter generator parameters

- *Rate knobs and fields:* Set the speed of the modulation (jitter) signal for each jitter generator.
- *Jitter 1/2 On/Off buttons:* Turn each jitter generator on or off, independently.
- *Target 1/2 pop-up menus:* Choose modulation targets 1 and 2, for each jitter generator.
- *Amount 1/2 sliders:* Determine the amount of modulation for each jitter source.

Sculpture note-on random modulators in Logic Pro for Mac

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.



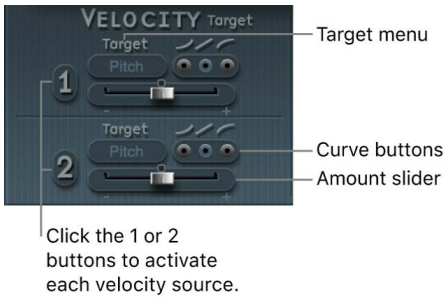
Note-on Random parameters

- *On/Off buttons (1 and 2):* Turn note-on random modulation on or off.
- *Target pop-up menus:* Choose the modulation target—the parameter that is randomly modulated when a note is played.
- *Amount sliders:* Set the amount, or strength, of modulation.

Sculpture velocity modulators in Logic Pro for Mac

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.

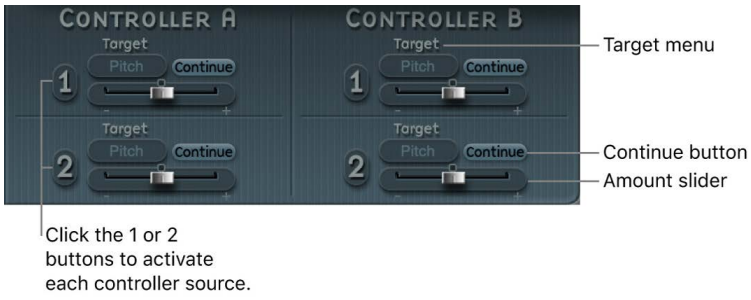


Velocity modulator parameters

- *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:* Set the amount, or strength, of modulation.
- *Curve buttons:* Choose from concave, linear, and convex velocity curves.

Use Controller A and B in Sculpture in Logic Pro for Mac

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):* Set the amount, or strength, of modulation.

Sculpture envelope parameters

Sculpture envelope overview in Logic Pro for Mac

Sculpture control sources include two envelopes. In traditional synthesizer designs, envelopes are used to control amplitude and filter levels over time. By comparison, Sculpture 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 1/2 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:* Set the modulation amount when the incoming via signal is 0, such as when the modulation wheel is at its minimum position, for example.
  - *Via amount slider:* Set the modulation amount when the incoming via signal is at full level, such as when the modulation wheel is at its maximum position, for example.
- *VariMod source pop-up menu:* Choose a modulation source. (VariMod is available only for recorded envelopes.)
- *VariMod slider:* Set the amount of envelope variation.
- *Ctrl/Env mode buttons:* 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 allows you to record and edit envelopes. See Sculpture envelope display and Sculpture active envelope controls.

Sculpture active envelope controls in Logic Pro for Mac

The following parameters are active only if the envelope is engaged (Mode buttons set to either Env or Ctrl+Env).

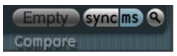
Envelope parameters

- *A-Time Velosens slider:* 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:* 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.

- *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.

- *Compare button:* Toggle 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).

Sculpture envelope display in Logic Pro for Mac

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 turn this feature on and off by clicking the Autozoom button—the small magnifying glass.
- Autozoom is automatically turned off 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.

- In Logic Pro, Control-click the envelope buttons or the envelope display background, then choose Copy, Paste, or Clear from the shortcut menu.

Use Sculpture envelope nodes in Logic Pro for Mac

When an envelope is first opened, a default envelope curve is automatically created. 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.

Adjust the time between nodes in Sculpture

- In Logic Pro, 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 in Sculpture

- In Logic Pro, drag a handle up or down.

Change the shape of the curve between nodes in Sculpture

- In Logic Pro, for simple (non-recorded) 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 in Logic Pro for Mac

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 turned off.
- 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 starts after the key is released.

Record Sculpture envelopes in Logic Pro for Mac

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).

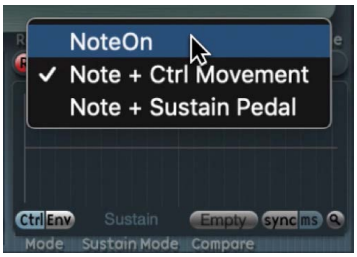


Envelope recording parameters

- *R(ecord) button:* Start or stop 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 in Sculpture

1. In Logic Pro, 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 in Sculpture

In Logic Pro, 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 in Sculpture

- In Logic Pro, 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 lets you 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 in Sculpture

- In Logic Pro, the envelope segments and handles are set automatically after recording. Drag the vertical lines that intersect the handles to allow editing.

Note: This does not change the shape of the envelope.

Sculpture morph parameters

Sculpture morph overview in Logic Pro for Mac

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 with 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.

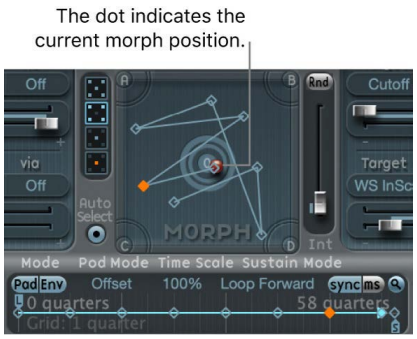
The Morph Pad and Morph Envelope let you create, and precisely control, the movements and blending between morph points.



The Morph section consists of two parts:

- *Morph Pad:* Use to display and edit, or draw, morph point paths. There are five morph points—the four corners and the center. A number of menu options are also available for randomizing, copying, and pasting morph points or Morph Pad states.
- *Morph Envelope:* Use 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.



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.

Sculpture Morph Pad

Morph points in the Sculpture Morph Pad in Logic Pro for Mac

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 is 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 in Logic Pro for Mac

You can open the shortcut menu by Control-clicking the Morph Pad. The menu contains Copy, Paste, and Exchange commands.



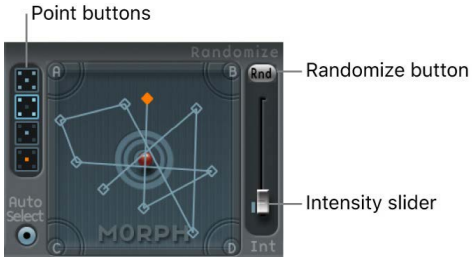
- *Copy selected Point:* Copies the current morph point into the Clipboard.
- *Copy current Pad Position:* Copies the current morph state into the Clipboard.
- *Paste to selected Point:* Pastes the Clipboard content to the selected point.
- *Exchange selected Point:* Swaps previously copied data with the selected point.
- *Paste to all Points:* Pastes the Clipboard content to all selected points.

Randomize points in the Sculpture Morph Pad in Logic Pro for Mac

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:* Define the number of morph points you want to use for randomization. The active button indicates which points are randomized.

  The bottom button, when selected, limits randomization to the currently selected morph point.

- *Auto Select button:* Automatically select the closest morph point.
- *Randomize button:* Create randomized values for all parameters of the chosen morph points.
- *Intensity slider:* Set the amount of randomization from 1% (slight deviation) to 100% (completely random values).

Randomize morph points in Sculpture

This example provides a general approach that you can follow for morph point randomization.

1. In Logic Pro, 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.

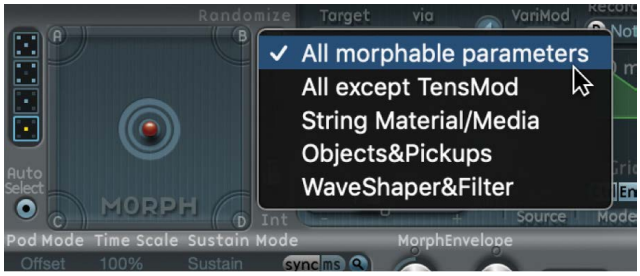
   Note: The morph ball is visible only when the Record Trigger button is active.

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 in Logic Pro for Mac

You can Control-click the Rnd button to open a shortcut menu which contains commands that determine the parameters 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 points in the Sculpture 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.

Sculpture Morph Envelope

Sculpture Morph Envelope display in Logic Pro for Mac

The Morph Envelope contains nine points and eight segments. It has recording behavior that is similar to the controller envelopes.



- 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.
- During Morph Envelope playback, the current position is indicated by a moving red dot and line. When multiple voices are played, this indicator is shown for the most recently triggered voice.

Adjust the time between nodes in the Sculpture Morph Envelope

- In Logic Pro, 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 in Logic Pro for Mac

This section describes Morph Envelope parameters.



Morph Envelope parameters

- *Record button:* Arm the envelope for recording. See Record Morph Envelopes.
- *Trigger mode pop-up menu:* Choose the event type that triggers recording.
- *Mode buttons:* Turn on the Morph Envelope. See Sculpture Morph Envelope mode.
- *Pad Mode pop-up menu:* Choose a Morph Envelope mode.
- *Timescale field:* 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 Morph Envelope Sustain and loop.
- *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:* 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:* 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:* 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 Morph Envelope mode in Logic Pro for Mac

The Mode buttons activate the Morph Envelope and provide a choice of several modes.

- *Both buttons off:* Morph function is turned off.
- *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:



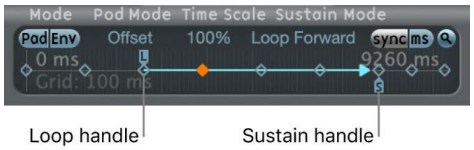
  - *Offset:* The default mode. Behavior is the same as Env+Pad mode: Envelope is running, and the position of the morph ball or X/Y MIDI controllers is used as an offset for any envelope movements.
  - *Point Set:* Envelope is running. The selected envelope point can be edited by moving the morph ball 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 in Logic Pro for Mac

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 turned off.
- *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, allowing 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 in Logic Pro for Mac

This section outlines the steps needed to record a Morph Envelope.



Choose a record trigger mode in the Sculpture Morph Envelope

- In Logic Pro, click the Trigger mode pop-up menu to the right of the R button, then choose one of the following trigger modes to 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 in Sculpture

1. In Logic Pro, 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 activate 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).

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 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.

Stop a morph envelope recording in Sculpture

In Logic Pro, 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 in Logic Pro for Mac

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/B pop-up menus:* Choose two controllers that can be used for side chain modulations or as via modulation sources, as set in the CtrlA and CtrlB modulation routing panes.
- *CtrlEnv 1/2 pop-up menus:* Choose controller assignments for the two control envelopes, which can be used as a modulation signal or an offset. Offsets occur in cases where the control envelope is set to Ctrl only or Ctrl+Env modes. These assignment pop-up menus also define the source for recording controller movements.
- *Morph X/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 either the default MIDI controller assignments or controller assignments loaded from the setting. If you choose Use Default, assignments remain unchanged. If you choose Load From Setting, you use the controller 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 in Sculpture

1. In Logic Pro, 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 extended parameters in Logic Pro for Mac

Sculpture provides additional parameters that can be accessed by clicking the disclosure arrow 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. Per-voice channels support pitchbend, aftertouch, modwheel, Vibrato Depth Ctrl, and Ctrl A and Ctrl B assignment messages. See Define Sculpture MIDI controllers. Controllers and MIDI messages sent on the base channel affect all voices.

- *Mono Mode Bend Range slider:* Set a value from 0 to 96.

  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 the GarageBand for iOS 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.

  In certain situations, such as when sliding over a wider range in Mono Mode or applying pitch bend upwards after already dynamically modulating string stiffness to higher values, the resulting pitch may not precisely track the incoming pitch bend beyond a specific threshold. This behavior is intentional, aimed at maintaining model stability while still permitting extreme adjustments of model parameters for innovative sound design purposes. Unlike the Mono Mode Bend Range, the standard Bend Range Up is actually used to determine an appropriate safety margin so that notes can still be precisely pitched up by the corresponding amount. As a result, the standard Bend Range Up setting can be used to ensure a specific upward bend range even when operating in MIDI Mono Mode.

- *Render Mode pop-up menu:* Choose Basic, Extended, or High Definition mode.

  Sculpture's string model can roughly be seen as a chain of springs and masses. The maximum number of elements (masses) is set with the Resolution slider. The Resolution slider interacts with the pitch, stiffness and pitch bend range settings to automatically set the number of overtones that are actually used. This Sculpture string model affords advantages over the more common waveguide models, especially when it comes to interactions with the string with multiple or larger objects. An inherent property of this flexible string model is that higher overtones are slightly below integer multiples of the fundamental, leading to increasingly inharmonic higher overtones for a string without any stiffness.

  Each Render Mode option changes the number of available elements and/or processing.

  - *Basic:* Choose to set a maximum of 100 elements, with some internal headroom for extreme key scaling. Suitable for many sound types, and has a lower processing load than other render modes. There is a direct correlation between the number of vibration modes and elements, resulting in a maximum of 99 overtones in the spectrum of a single voice. For example, a 100 Hz bass note (with no string stiffness) has a highest overtone of around 10 kHz, making the more audibly inharmonic upper third of the harmonic series fall into the audible range.

- *Extended:* Choose to set a maximum of 1000 elements. The Resolution slider, coupled with pitch, stiffness and pitch bend range settings affect the number of elements (overtones) used. This mode automatically reduces the number of audible inharmonic overtones until none of the overtones happen at rates above half the sampling frequency.
- *High Definition:* Choose to set a maximum of 1000 elements and to also turn on internal 2x oversampling, which adds frequency headroom for more overtones. The Resolution slider, coupled with pitch, stiffness and pitch bend range settings affect the number of elements (overtones) used. This mode is significantly more processor-intensive than Extended mode as it runs at double the frequency and allows more elements per string for the same note pitch.

Sculpture tutorials

Explore Sculpture

Explore Sculpture overview in Logic Pro for Mac

The following sections contain information to assist you as you start to explore sound creation in Sculpture. See Explore the Sculpture string, Explore Sculpture objects, Explore Sculpture pickups, and Explore other processing options.

The creation of basic instrument sounds is discussed in Basic sound programming in Sculpture. For a more detailed look at programming particular types of sounds, see Electric bass programming in Sculpture and Synthetic sound programming in Sculpture.

Given the flexibility of the Sculpture 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 morphing capabilities to create new sounds, you can. See Randomize points in the Sculpture Morph Pad.
- 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 Waveshaper, and Sculpture modulation overview.

Whatever approach you favor, you can achieve new and interesting results. Experiment and familiarize yourself with each approach. Each has its strengths and weaknesses, and often 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 interplay in Sculpture.

To start, you 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 in Sculpture

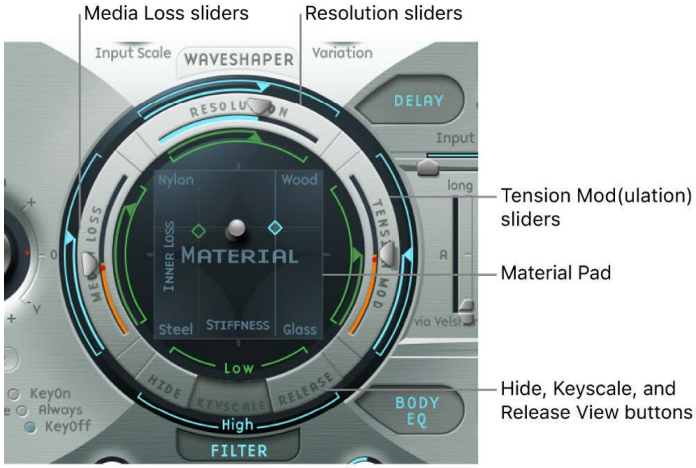
- In Logic Pro, choose Save Settings As from the Settings pop-up menu, then enter the name you want—maybe “neutral” or “vanilla”—and click Save.

  This setting can be reloaded as you work through the examples.

Explore the Sculpture string in Logic Pro for Mac

The string is the central synthesis element of Sculpture and is responsible for the basic tone. It offers parameters that let you adjust its material—what it is made of, in other words—and define the environment that it is 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.



Material Pad in Keyscale view

Set a basic tone in Sculpture

1. In Logic Pro, 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.

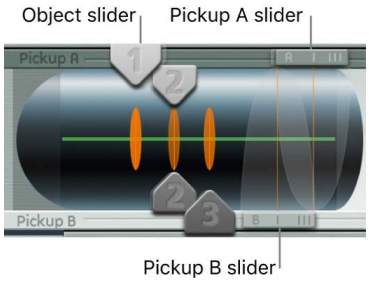
Change the string properties in Sculpture

1. In Logic Pro, 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 objects in Logic Pro for Mac

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.



Explore the objects in Sculpture

1. In Logic Pro, 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 the Object 1 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 interplay in Sculpture in Logic Pro for Mac

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.

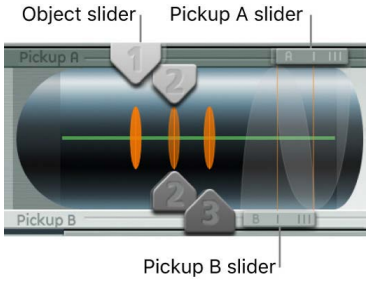
Generally, such adjustments won't need to be radical, and may involve only a small tweak to the Strength parameters, or perhaps to the pickup positions of each object, for example. These parameters have the greatest impact on the tone and level of the objects and should be the first things you look at if turning on 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 pickups in Logic Pro for Mac

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 in Sculpture

1. In Logic Pro, 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 the Object 1 Strength control to hear things better, or to adjust the tone. You can also use Object 1 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 in Sculpture

- In Logic Pro, 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, directly opposite the Pickup section.

Explore other processing options in Sculpture in Logic Pro for Mac

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 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 in Logic Pro for Mac

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 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 Logic Pro for Mac.

Breath control is available when you use Sculpture, even if you don’t own a breath controller.

Use breath control without a breath controller in Sculpture

1. In Logic Pro, 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 trigger the CtrlEnv 1 and 2 parameters.

Create basic sounds in Sculpture

Basic sound programming in Sculpture in Logic Pro for Mac

This section covers the creation of basic types of sounds, such as organs, basses, guitars, and so on. See Create acoustic instruments in Sculpture, Sculpture stringed instrument programming, and Classic synthesizer programming in Sculpture.

For a detailed look at programming particular types of sounds, see Electric bass programming in Sculpture and Sculpture synthetic sound programming.

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 interacts 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 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 supplied 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. Turn different parameters on or off to see what each does.

  Have fun and take risks—you can't break anything.

Program acoustic instruments in Sculpture in Logic Pro for Mac

The tasks outlined in this section provide programming guidelines, tips, tricks, and information to assist you in creating particular types of acoustic instrument sounds in Sculpture.

Create a bell sound in Sculpture

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. In Logic Pro, load the #default (or your vanilla) setting file.
2. Choose Strike from the Object 1 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 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 in Sculpture

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. In Logic Pro, load the #default (or your vanilla) setting file.
2. Set Object 1 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 "l" of Inner Loss, and the "l" 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 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 in Sculpture

In Logic Pro, 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 the Object 2 type, then experiment with the Object 1 and 2 positions. This can also result in different brass sounds.

Create flute-like sounds in Sculpture

Use this approach as the basis for instruments in the wind family, including flutes, clarinets, shakuhachis, pan pipes, and so on.

1. In Logic Pro, 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 type to Blow.
4. Set Object 2 type to Noise.
5. Set the Gate of both objects to Always.
6. Adjust Object 2 Strength to a value of around 0.25.
7. Adjust the Object 1 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 pickup position to around 0.27.
12. Set Object 2 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 in Sculpture

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. In Logic Pro, load the #default (or your vanilla) setting file. (Object 1 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 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 the Object 2 Timbre parameter upward.
12. Carefully adjust the Object 2 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 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 in Sculpture

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 varies, 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. In Logic Pro, load the #default (or your vanilla) setting file.
2. Set Object 1 type to Strike.
3. Activate Object 2, and set its type to Disturb 2-sided.
4. Set Object 2 Gate mode to Always.
5. Object 1 Strength should be about 0.84.
6. Object 2 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.

Program stringed instruments in Sculpture in Logic Pro for Mac

The tasks outlined in this section provide programming guidelines, tips, tricks, and information to assist you in creating stringed instrument sounds in Sculpture.

Create a bass sound in Sculpture

1. In Logic Pro, 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 the Object 1 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 want to adjust the amplitude envelope 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 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 create.

Create a guitar sound in Sculpture

Guitar, lute, mandolin, and other plucked-type instruments, including harps, can be created from this basic setting.

1. In Logic Pro, 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 type to Impulse, if not already chosen.
4. Activate Object 2 and set its type to Pick.
5. Move Pickup A position to the extreme right.
6. Move Object 2 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 in Sculpture

In Logic Pro, 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 in Sculpture

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. In Logic Pro, load the #default (or your vanilla) setting file.
2. Set Transpose to –1 Oct.
3. Set Object 1 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 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 in Sculpture

In Logic Pro, 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 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.

Program classic synthesizer sounds in Sculpture in Logic Pro for Mac

One strength of Sculpture is the ability to create endlessly evolving pad and atmospheric sounds. It can also generate 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 in Sculpture

1. In Logic Pro, load the #default (or your vanilla) setting file.
2. Set the Voices parameter to 16.
3. Set Object 1 type to Bow.
4. Set Object 2 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 is a little sweeter.
8. Move Object 1 position to a value of 0.84.
9. Move Object 2 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 in Sculpture

1. In Logic Pro, load the #default (or your “vanilla pad”) setting file.
2. Click the LFO 1 button at the bottom left of the interface.
3. Click the 1 button, and play the keyboard.

   The difference you 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, little is 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 in Sculpture

1. In Logic Pro, 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 in Sculpture in Logic Pro for Mac

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 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 non-wound 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 in 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 in 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 in 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 becomes and the more hollow it sounds. The farther you move the pickup toward the end of the string, the more overtone content is heard, becoming more dense and compact. The sound has 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 in Sculpture in Logic Pro for Mac

This section covers programming of a basic bass sound which serves as the foundation for a number of different bass sound examples. See Program a picked bass sound in Sculpture, Program a slap bass sound in Sculpture, and Program a fretless bass sound in Sculpture.

Sequentially follow the tasks in this section and Refine the basic Sculpture 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 in Sculpture

1. In Logic Pro, 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 in Sculpture

1. In Logic Pro, 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 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.

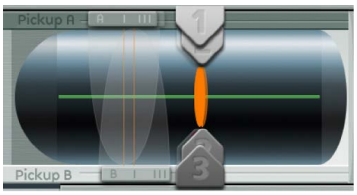
   Play some notes in the lower range. You'll hear that the sound is very muffled, hollow, and distorted. Before you adjust further parameters in Object 1, you need to set the position of the pickup.

   This is accomplished in the Pickup display 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.

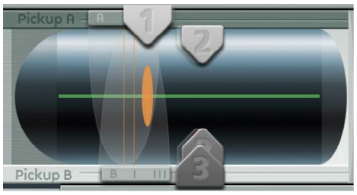
5. Drag Pickup B to the exact position of Pickup A while keeping an eye on the help tag. The two thin orange lines should overlap perfectly. As a suitable value for the example, set both pickups to 0.10.



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 in Sculpture

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. In Logic Pro, 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 may benefit from a small amount of this effect.

Splay overtones in Sculpture

1. In Logic Pro, 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 in Sculpture

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. In Logic Pro, 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 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 in Sculpture

1. In Logic Pro, 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 in Sculpture

To more realistically replicate the different tonal ranges of the bass, use the 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.

- In Logic Pro, 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 in Sculpture in Logic Pro for Mac

This section covers modifications to the basic bass sound. See Program a picked bass sound in Sculpture, Program a slap bass sound in Sculpture, and Program a fretless bass sound in Sculpture.

Sequentially follow the tasks in this section after reading Program a basic bass sound in 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 in Sculpture

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. In Logic Pro, play some notes at the higher end of the bass 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 in Sculpture

1. In Logic Pro, 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 Sculpture

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. In Logic Pro, 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 has a similar effect. Try different combinations here as well.

Alter the frequency spectrum of your basic bass with the Body EQ in Sculpture

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. In Logic Pro, 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 in Sculpture in Logic Pro for Mac

The basic bass is played with the fingers. In this example, you will simulate playing the strings with a pick, using the Pick object type. The Timbre parameter is used to adjust the relationship between the speed and intensity at which the string is struck. The Variation parameter is 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 in Sculpture

1. In Logic Pro, 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 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 in Sculpture

1. In Logic Pro, 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 in Sculpture

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 is 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. In Logic Pro, set the Object 3 type to Damp.
2. Set Object 3 Strength to 0.50.
3. Move Object 3 a little 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 in Sculpture

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. In Logic Pro, Object 3 is used as a damper. Choose the Damp type.
2. Adjust Object 3 Timbre to a 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 in Sculpture

1. In Logic Pro, 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 in Sculpture

1. In Logic Pro, 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 in Sculpture in Logic Pro for Mac

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 in Sculpture

1. In Logic Pro, 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 does 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 in Sculpture

1. In Logic Pro, 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 in Sculpture

1. In Logic Pro, 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 in Sculpture in Logic Pro for Mac

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 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 in Sculpture

1. In Logic Pro, 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 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 can 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 bass sound in Sculpture in Logic Pro for Mac

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 the Delay section to emulate this. Heavy delays are, however, used on basses in "dub" style reggae.

Emulate a chorus effect by modulating pickup positions in Sculpture

1. In Logic Pro, 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 in Sculpture

1. In Logic Pro, 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 in Sculpture

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 output with a reverb plug-in.

1. In Logic Pro, 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 in Sculpture

1. In Logic Pro, 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 in Sculpture in Logic Pro for Mac

The Electric bass programming in Sculpture 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 the Sculpture 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 in Sculpture.

Program a sustained synthesizer sound in Sculpture in Logic Pro for Mac

First, load the 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.

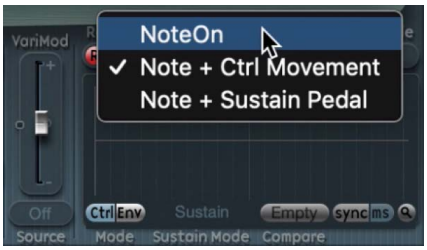
Record an envelope in Sculpture

1. In Logic Pro, 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 in Sculpture

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. In Logic Pro, 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 can 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 in Sculpture

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. In Logic Pro, 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.

Program morphed sounds in Sculpture in Logic Pro for Mac

The Morph Pad is in the middle of the lower part of the Sculpture window. Each corner of the Morph Pad can contain a different setting for a diverse number of parameters. You can crossfade between these settings and can morph the sound by dragging the red ball in the center of the Morph Pad.



Vary the sound with the Morph Pad in Sculpture

1. In Logic Pro, 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. You can also automate the Morph Pad using a recorded envelope (for more information, see Record Morph Envelopes in Sculpture).

Use the randomizing function in Sculpture

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. In Logic Pro, 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.

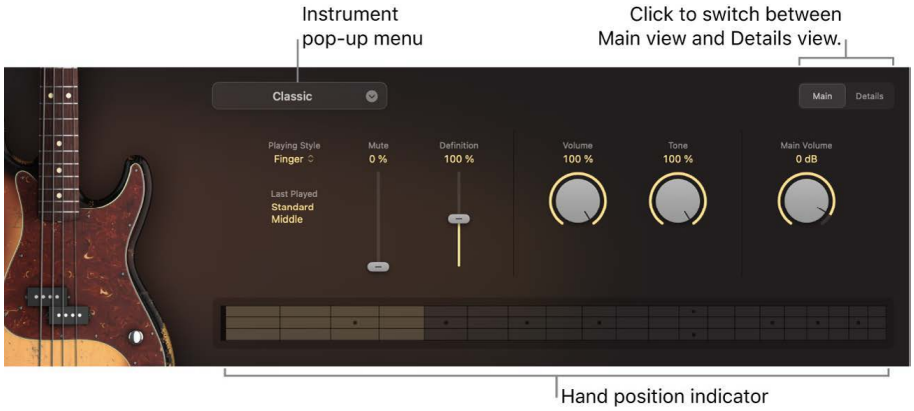
   When you next move the morph ball, you'll hear the variations you just created.

Studio Bass

Studio Bass in Logic Pro for Mac

The studio instruments are high-quality multisampled players that offer a collection of solo and grouped instrument presets. See Studio Horns, Studio Piano, and Studio Strings.

Studio Bass is a highly expressive sample-based software instrument that emulates the sound of several electric basses and an upright acoustic bass, each with their own sonic characteristics. It's optimized for use with the Bass Player, but it also works with MIDI and pattern regions. You can also use a MIDI keyboard to play and record with Studio Bass.



If you're new to using plug-ins in Logic Pro, see Add, remove, move, and copy plug-ins.

Studio Bass contains parameters for volume, tone, pickup controls, and other sonic characteristics typical of the selected bass instrument. See Main view parameters and Details view parameters.

As with a real bass instrument, notes played with Studio Bass can have different tonal qualities depending on their neck position. When a note is played, Studio Bass provides visual feedback by indicating the positions of the hand and notes in the area resembling a fretboard at the bottom of the plug-in window. You can influence these positions through the use of MIDI Out channels. See String and position MIDI channel assignments.

You can use articulations with Studio Bass to create expressive, nuanced performances and recordings. When you load a Studio Bass patch from the Library, an articulation set is automatically loaded. There are separate articulation sets for each playing style of the electric bass instruments, and one for the American Upright instrument. See Choose patches in the Library and Articulation editing overview in the Logic Pro User Guide.

When using a Studio Bass patch from the Library, you can use a keyswitch map to switch between different articulations on the fly. See Studio Bass keyswitch mapping.

You can also edit the articulations for each note in a MIDI region or step in a pattern region. See the topic Manage software instrument articulations in the Logic Pro for Mac User Guide.

Choose a bass instrument

- Click the Instrument pop-up menu in Studio Bass and choose one of the following:
  - *Classic:* A vintage electric bass heard on some of the earliest country and rock recordings.
  - *Sixties:* A hollow-bodied bass with flat-wound strings and rich, round tone.
  - *Rock:* A solid bass with a punchy and aggressive sound with rich overtones.
  - *Session:* A studio workhorse with a wide variety of tones thanks to its ability to blend both pickups.
  - *Modern:* A combination of a humbucker pickup and a two-band EQ providing a tight and versatile sound.
  - *American Upright:* A timeless acoustic sound full of rich and natural tones.

Switch between the Main and Details view

- In the upper-right corner of the Studio Bass plug-in window, click Main or Details.

Studio Bass Main view in Logic Pro for Mac

The Main view for Studio Bass in Logic Pro for Mac contains parameters for the playing style and various controls for tone and volume. With the exception of the American Upright, each bass instrument has its own set of pickup controls.



Common parameters

- *Playing Style pop-up menu:* Choose the hand playing style for the bass instrument.
  - *Finger:* Play the bass with fingers.
  - *Pick:* Play the bass with a pick.
  - *Slap:* Play the bass in a funky slap style. The Slap playing style is available only for the Classic and Modern bass instruments.

  The Playing Style parameter is not available for the American Upright bass.
- *Mute slider:* Control the degree to which the bass strings are muted.

  The Mute slider is not available for the American Upright bass.

- *Definition slider:* Control level of transient attacks on the strings.

  The Definition slider is not available for the American Upright bass.

- *Last Played field:* Displays the most recently used articulation. This may be different from the articulation shown in the plug-in header. See Manage articulations for software instruments.
- *Main Volume:* Set the output level of the instrument.

Classic bass parameters

- *Volume knob:* Set the pickup volume.
- *Tone knob:* Set the tone of the pickup.

Sixties bass parameters

- *Neck Volume knob:* Set the volume of the neck pickup.
- *Bridge Volume knob:* Set the volume of the bridge pickup.
- *Rhythm/Solo switch:* In the Rhythm position, lower frequencies are boosted. In the Solo position, higher frequencies are boosted.
- *Bass switch:* Turn the bridge pickup on or off.
- *Treble switch:* Turn the neck pickup on or off.

Rock bass parameters

- *Neck Volume knob:* Set the volume of the neck pickup.
- *Bridge Volume knob:* Set the volume of the bridge pickup.
- *Neck Tone knob:* Set the tone of the neck pickup.
- *Bridge Tone knob:* Set the tone of the bridge pickup.
- *Pickup pop-up menu:* Choose a pickup configuration.
  - *Neck:* Only the output of the neck pickup is heard.
  - *Both:* The outputs of the bridge and the neck pickups are blended together.
  - *Bridge:* Only the output of the bridge pickup is heard.
  - *Stereo:* The output of the bridge pickup is in the right channel, and the output of the neck pickup is in the left channel. You can use plug-ins in dual mono mode to apply different effect settings to each channel.

Session bass parameters

- *Neck Volume knob:* Set the volume of the neck pickup.
- *Bridge Volume knob:* Set the volume of the bridge pickup.
- *Tone knob:* Set the tone of the pickups.

Modern bass parameters

- *Volume knob:* Set the pickup volume.
- *Bass knob:* Raise or lower the bass frequencies in the tone.
- *Treble knob:* Raise or lower the treble frequencies in the tone.

American Upright parameters

- *Growl:* Control the volume of upper harmonics created by the string resonating against the fingerboard.
- *Growl Attack:* Control the time it takes for the growl to become audible. Longer notes have longer growl attacks, and higher notes have faster growl attacks.
- *Noises:* Increase the probability of various instrument noises, such as, rattles, buzzes, and crackles, based on velocity and on what note you play. This parameter is a submix of the Rattles, Releases, and Handling parameters in the Details view.

Studio Bass Details view in Logic Pro for Mac

In the Details view of Studio Bass, you can change the timing of portamento slides between notes and adjust sonic qualities of the bass, such as the noise of fingers, strings, and electric circuits. You can also choose between mono and polyphonic voice settings and set the prevalence of open string tones in your performance.



Portamento parameters

- *Portamento knob:* Set the time it takes to slide from one note to another.
- *Tempo Sync button:* When turned on, the portamento time is set to note values instead of milliseconds.

Noise Parameters

- Electric basses:
  - *Scrapes slider:* Set the volume of the sound of the pick scraping on the strings. This control is only available when the Playing Style parameter in the Main view is set to Pick.
  - *String Noise slider:* Set the volume of the sound of the hand moving on the strings.
  - *Hum & Hiss slider:* Set the volume of the electronic circuit that is characteristic of each bass instrument.
  - *Release Noise slider:* Set the volume of the sound that occurs when the finger dampens the note.
- American Upright bass:
  - *Rattles slider:* Set the volume of resonant vibrations characteristic of upright bass instruments.
  - *Releases slider:* Set the volume of the sound that occurs when the finger dampens the note.
  - *Handling slider:* Set the volume of the sound of hands moving and changing their position on the instrument.

  The output of these parameters are summed together and controlled by the Noises parameter in the Main view.

Pitch, Voice Mode, and Open String parameters

- *Pitchbend Up value:* Set the upward pitch bend range in semitone steps.
- *Pitchbend Down value:* Set the downward pitch bend range in semitone steps.
- *Voice Mode pop-up menu:* Choose polyphonic or monophonic behaviors.
  - *Polyphonic:* Multiple notes can be played at the same time.
  - *Monophonic:* Only one note can play at a time.
  - *Mono per String:* Only one note per string can play at a time, limited to four notes.
- *Open Strings pop-up menu:* Choose how or whether notes on open strings are played.
  - *Use:* Play notes on open strings if they are near the current playing position.
  - *Avoid:* If possible, don’t play notes on open strings.
  - *Prefer:* Play notes on open strings if available.

Mute Controller

- *Mute Controller:* Choose a MIDI controller to adjust the Mute parameter in the Main view.

  The Mute Controller parameter is not available for the American Upright bass.

Studio Bass keyswitch mapping in Logic Pro for Mac

This section outlines the default keyswitch mapping used in of the Studio Bass instruments.

When you load a Studio Bass patch from the Library, you can use a keyswitch map in the Smart Controls to switch between articulations while performing or recording. This lets you add legato or slides, for example, to a bass line in real time. See the Manage software instrument articulations topic in the Logic Pro for Mac User Guide.

The table shows the default keyswitch mapping, starting from C-1 (MIDI note number 12). You can, of course, transpose the octave of notes used as keyswitches.

Note: Some articulation IDs differ for the American Upright. In such cases, use the second number in the Articulation ID field.

Key	MIDI Note #	Articulation ID	Finger	Pick	Slap	American Upright
C-1	12	1	All (Auto)	All (Auto)	All (Auto)	All (Auto)
C#-1	13	2	Index/Middle (Auto)	Down/Up (Auto)	Slap/Pop (Auto)	Standard
D-1	14	44/52	Index Only	Down Only	Slap Only	Pickup Note
Eb-1	15	3	Legato (HammerOn/ PullOff)	Legato (HammerOn/ PullOff)	Legato (HammerOn/ PullOff)	Legato (HammerOn/ PullOff)
E-1	16	45/57	Middle Only	Up Only	Pop Only	Slap
F-1	17	4	Dead Note	Dead Note	Dead Note	Dead Note
F#-1	18	6	Dead Note Harmonic	Dead Note Harmonic	Dead Note Harmonic	Dead Note Harmonic
G-1	19	5	Harmonic	Harmonic	Harmonic	Harmonic
G#-1	20	56/58	Pickup Hit Single	Pickup Hit Single	Pickup Hit Single	Left Hand Pat
A-1	21	12	Slide by Velocity	Slide by Velocity	Slide by Velocity	Slide by Velocity
Bb-1	22	57/59	Pickup Hit All	Pickup Hit All	Pickup Hit All	Body Tap
B-1	23	37	Slide In +2	Slide In +2	Slide In +2 Pop	Slide In +2

Studio Bass neck position and string MIDI Out channels in Logic Pro for Mac

Just like with a real bass instrument, notes you play with Studio Bass have different tonal qualities when played on different strings and at different positions on the neck. When playing the Studio Bass in Logic Pro for Mac, you can use different MIDI Out channels to influence the preferred neck position and string of the notes you play. You can set the MIDI Out channel in the Track inspector for the track where Studio Bass is loaded. See the Edit track parameters topic in the Logic Pro for Mac User Guide.

MIDI Out Channel	Function	Description
1	Auto	This chooses the most natural-sounding neck position for the played note.
2	Prefer the E string.	Notes are played on E-string if available. If a note is not playable on the preferred string, it is played on another one based on the current position.
3	Prefer the A string.	Notes are played on A-string if available. If a note is not playable on the preferred string, it is played on another one based on the current position.
4	Prefer the D string.	Notes are played on D-string if available. If a note is not playable on the preferred string, it is played on another one based on the current position.
5	Prefer the G string.	Notes are played on G-string if available. If a note is not playable on the preferred string, it is played on another one based on the current position.
7	Move the hand position to the bottom of the neck.	If a note is not playable in this position on any string, the position is moved to the closest alternative.
8	Move the hand position to the 3rd fret.	If a note is not playable in this position on any string, the position is moved to the closest alternative.
9	Move the hand position to the 5th fret.	If a note is not playable in this position on any string, the position is moved to the closest alternative.
10	Move the hand position to the 7th fret.	If a note is not playable in this position on any string, the position is moved to the closest alternative.
11	Move the hand position to the 9th fret.	If a note is not playable in this position on any string, the position is moved to the closest alternative.

MIDI Out Channel	Function	Description
12	Move the hand position to the 12th fret.	If a note is not playable in this position on any string, the position is moved to the closest alternative.
13	Prefer notes played on the E and A strings.	If a note is not playable on either of the preferred strings, the position is moved to the closest alternative.
14	Prefer notes played on the D and G strings.	If a note is not playable on either of the preferred strings, the position is moved to the closest alternative.

Studio Horns

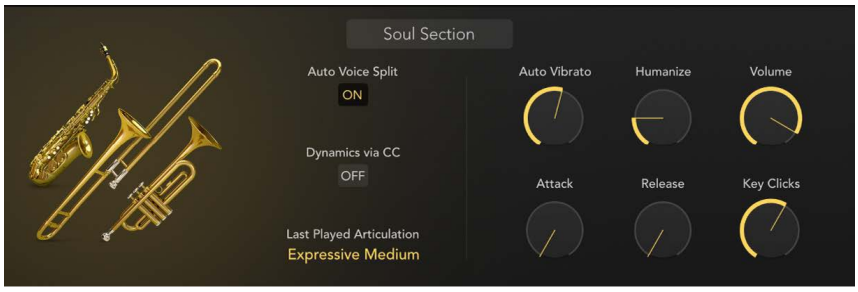
Studio Horns in Logic Pro for Mac

The studio instruments are high-quality multisampled players that offer a collection of solo and grouped instrument presets. See Studio Bass, Studio Piano, and Studio Strings.

Studio Horns provides a collection of highly expressive single and section brass instruments. It has a simple, clean interface, with all main controls at hand. Additional controls let you customize Studio Horns to match your preferences and to work best with your controllers. See Studio Horns extended parameters.

Default section and single instrument keyswitches are covered in Studio Horns keyswitch mapping.

If you're new to using plug-ins in Logic Pro for Mac, see Add, remove, move, and copy plug-ins.



Studio Horns parameters

- *Preset pop-up menu:* Choose a solo or section instrument.

  You can use MIDI channel events to control section instruments. If you want to play specific instruments in a section, you can send MIDI messages on their respective channels, then add a list of channel assignments. See Studio horns section MIDI channels.

- *Monophonic button:* Turn on monophonic mode for solo instruments.

- *Auto Voice Split button:* When on, chords played on the keyboard are automatically assigned to different instruments or instrument groups, or "voices" in the section.

  Depending on the size and type of section, a voice can consist of one or multiple instruments. This mimics the distribution of notes (voicings) among players, based on what instrument group is best suited to a particular note range, for example. See Studio Horns extended parameters for details about additional Auto Voice Split functions.

  Note: When Auto Voice Split is off, Horn section instruments that share a key range are layered, with all horns assigned to a key range played simultaneously.

- *Dynamics via CC button:* Turn on to allow control of instrument dynamics (soft-loud) and timbre in real time. Additional dynamic control and mapping functions are available in the extended parameters.
- *Last Played Articulation field:* Displays the most recently used articulation. This may be different to the articulation shown in the plug-in header.
- *Auto Vibrato knob:* Set the intensity of vibrato in the chosen sound. Further vibrato control is available in the extended parameters.
- *Humanize knob:* Set the amount of random variation in the chosen sound. This emulates changes in embouchure by introducing small pitch and level fluctuations.
- *Volume knob:* Set the output level of the instrument.
- *Attack knob:* Set the amount of time it takes for the instrument to fade in, following a MIDI note on message.
- *Release knob:* Set the amount of time it takes for the instrument to fade out, following a MIDI note off message.
- *Key Clicks knob:* Set the level of key and pad noise for sections that contain saxophones and solo saxophone instruments.

  Note: The Key Clicks knob isn't shown for trumpet or trombone instruments.

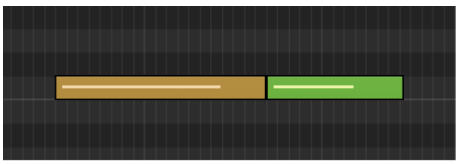
Choose a section or single instrument

1. In Logic Pro, click the Preset pop-up menu to open a window that shows available presets, then click or double-click a section or a single instrument icon to load the sound.
2. Click outside the window to close it.

Use advanced articulation features

Studio Horns provides several articulations for falls and doits. These can be played on your keyboard like any other articulation. Horn players often attach a fall or doit directly to a note end, so the fall or doit happens with no gap after the previous note. Studio Horns features a special mode which allows you to smoothly connect a fall or doit articulation to the previous note at any position "on-the-fly", as a real musician would play it.

- In Logic Pro, to create a smooth fall or doit, add a second (Fall or Doit articulation) note of the same pitch directly following any note. There should be no, or a minimal, gap from the previous note.

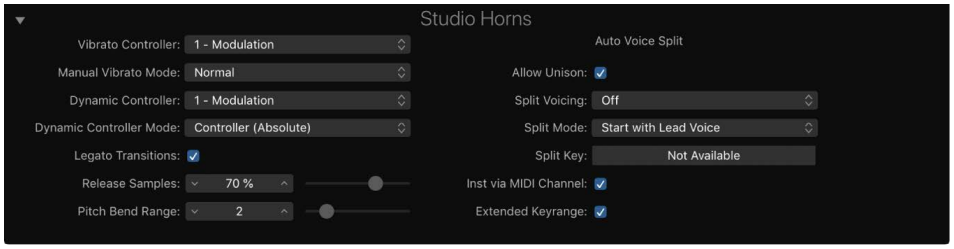
  Note: Velocity of the Fall or Doit note can be different to the previous note. You can also vary the length of the Fall or Doit note. This allows you to create very subtle falls or doits at the end of any note.
- To perform falls and doits live on a keyboard, you can use the predefined keyswitches for Falls and Doits. These operate in Trigger mode, which automatically creates the Fall or Doit event and places it immediately after the previous note. Try this by holding a note with a Sustain articulation. While the note is playing, press the keyswitch for the Fall Long articulation. The sustained note is then stopped with a long fall.

Studio Horns extended parameters in Logic Pro for Mac

The extended parameters provide advanced control of performance and controller functions.

Click the disclosure arrow at the lower left to view the extended parameters.

Note: The parameters are different for solo and section instruments.



Studio Horns common extended parameters

The following parameters apply to solo or section instruments.

- *Vibrato Controller pop-up menu:* Choose a MIDI controller to adjust vibrato.
- *Manual Vibrato Mode pop-up menu:* Choose a vibrato response curve for incoming MIDI controller data. Choices are Off, Normal, Slow, Fast, Gentle, and Expressive.

  Note: This affects manually controlled vibrato only.
- *Dynamic Controller pop-up menu:* Choose the MIDI controller used to adjust an instrument's dynamics (soft-loud) while a note is playing. This function requires the Dynamics via CC button in the main interface to be turned on.
- *Dynamic Controller Mode pop-up menu:* Choose a mode to determine assigned MIDI controller behavior and instrument response to incoming velocity data.
  - *Controller (Absolute):* Dynamics are controlled only by the absolute value of the assigned MIDI controller.
  - *Velocity & Controller (Catch):* Dynamics are controlled both by incoming velocity data and by the value of the assigned MIDI controller when it matches any existing controller value. For example, a MIDI control knob with a value of 15 has no impact on dynamic behavior until it reaches a velocity value of 47 in an existing region. Once this controller value is reached, the assigned MIDI control knob will have an impact on instrument dynamics.
  - *Velocity & Controller (Relative):* Dynamics are controlled by both incoming velocity data and by the value of the assigned MIDI controller, relative to any existing controller value. For example, a MIDI control knob with a value of 15 will immediately impact dynamic behavior, relative to the existing velocity value. Adjusting a knob with a value of 15 to a value of 18 will adjust an existing velocity value of 47 to a value of 50.
- *Legato Transitions checkbox:* Turn on to allow legato transition samples for both monophonic and polyphonic modes, enhancing the realism of instruments played in a legato style.

- *Release Samples slider and field:* Set the volume of release samples. Release samples, as the name suggests, are triggered when you release a key. These allow for a natural-sounding decay in sustained articulations, for example. If a low release sample value is set, the sound may cut off abruptly, which can sound unnatural.
- *Pitch Bend Range slider and field:* Determine the range for pitch bend modulation. This is bipolar, with a range of ±12 semitones.
- *Extended Key Range checkbox:* Turn on to allow the instrument (or instruments in a section) to play beyond its normal key range.
  - When off, the instrument (or instruments in a section) plays in the original key range. This is the default, and should be used if you are aiming for authentic-sounding performances.
  - When on, the instrument uses the full keyboard range, allowing you to create parts that extend beyond the realistic range of the instrument (or instruments in a section).

Studio Horns Auto Voice Split extended parameters

The following parameters apply only to section instruments when the Auto Voice Split button is active in the main window.

- *Allow Unison checkbox:* Turn on to activate unison mode for split voices.
  - When turned on, multiple instruments within a section can play the same note simultaneously when a single or multiple keys are pressed.
  - When turned off, one instrument within the section is played per key. To have all instruments of a seven piece section play, you need to press seven keys.
- *Split Voicing pop-up menu:* Determine the octaves used by the respective voices/instrument(s) in the section. Results vary based on your Split Mode pop-up menu choice.
  - *8va:* Double the top part an octave up. This is a common technique used to make melody lines more prominent.
  - *Drop 2:* Transpose the second note from the top of the chord down an octave. This technique is used to make an arrangement sound thicker and richer.
  - *Drop 2+4:* Transpose the second and fourth note from the top of the chord down an octave. This technique is used to thicken an arrangement. It is more suitable for sections with five or more instrument “voices,” but can be useful with smaller sections.
- *Split Mode pop-up menu:* Choose the mode used to split individual or grouped ensemble instruments.
  - *Start with Lead Voice:* The first incoming note is assigned to the Lead Voice, which is usually a higher-range instrument such as a trumpet. All subsequent notes are assigned, in descending order, to the Middle and Bass Voices.
  - *Start with Bass Voice:* The first incoming note is assigned to the Bass Voice. All subsequent notes are assigned, in ascending order, to the Middle and Lead Voices.
  - *Start by Key Split:* Incoming notes below the defined split key (set with the Split Key pop-up menu) are assigned to the Bass Voice and the lower Middle Voices. Notes above the split key are assigned to the Lead Voice and the higher Middle Voices.

- *Split Key pop-up menu:* Set the MIDI note number used as the split point for individual ensemble instruments.

  Note: This parameter applies only when *Start By Key Split* is chosen in the Split Mode pop-up menu.

- *Inst via MIDI Channel checkbox:* Turn off to play the full section on *any* incoming MIDI channel. Active Unison / Auto Voice Split / MIDI Split settings apply. When on (default), the following applies:
  - MIDI Channel 1: Full section is played with the active Unison / Auto Voice Split / MIDI Split settings.
  - MIDI Channels 2-16: Access individual instruments within the section. See Studio horns section MIDI channels.

Studio Horns keyswitch mapping in Logic Pro for Mac

This section outlines the default keyswitch mapping of the Studio Horns instruments.

Articulations are accessed with the Articulation pop-up menu in the plug-in header. Articulations can be remotely switched while performing or recording, allowing you to add a fall or trill to a solo saxophone part, for example.

The table shows the default keyswitch mapping, starting from C0 (MIDI note number 24). You can, of course, transpose the octave of notes used as keyswitches.

X indicates available articulation.

* Trumpet 1, Trombone 1, Alto Sax 1, and Tenor Sax 1.

Key	MIDI Note #	Articulation ID	Articulation	Sections	Lead Instruments*
C0	24	1	Sustain	X	X
C#0	25	14	Growl		X
D0	26	6	Expressive Medium	X	X
Eb0	27	7	Expressive Long		X
E0	28	8	Expressive Short	X	X
F0	29	2	Staccato	X	X
F#0	30	4	Marcato Short		X
G0	31	10	Fall Long	X	X
G#0	32	5	Marcato Long		X
A0	33	11	Fall Medium	X	X
Bb0	34	9	Passionate		X
B0	35	12	Fall Short	X	X
C1	36	13	Doit	X	X

Key	MIDI Note #	Articulation ID	Articulation	Sections	Lead Instruments*
C#1	37	16	Scoop		X
D1	38	15	Shake (Horns) / Trill (Saxes)	X	X

Studio horns section MIDI channels in Logic Pro for Mac

In Studio Horns section presets, each instrument uses a dedicated MIDI channel. This lets you play specific instruments in a section by sending MIDI messages on their respective channels. Similarly, if you create a region for a section instrument, you can assign each MIDI event a specific MIDI channel to ensure that it is played only by a particular instrument within the section.

- MIDI channel 1 is always the “section” channel, meaning that it is used to play the entire section, either via Auto Voice Split, or as a Layer/Key Split, depending on the plug-in setting.
- MIDI channel 2 and above are used to access the different instruments of a section directly.

Note: When the *Inst via MIDI Channel checkbox* in the Extended parameters is turned off, the full section is played on *any* incoming MIDI channel. Active Unison / Auto Voice Split / MIDI Split settings apply. Full section playback is also triggered by incoming MIDI channel data that is not used in the section.

MIDI Channel	Horn Sections
1	Full Section
2	Alto Sax 1
3	Alto Sax 2
4	Tenor Sax 1
5	Tenor Sax 2
6	Baritone Sax
7	Trumpet 1
8	Trumpet 2
9	Trumpet 3
10	Trumpet 4
11	Trombone 1
12	Trombone 2
13	Trombone 3
14	Bass Trombone

Studio Piano

Studio Piano in Logic Pro for Mac

The studio instruments are high-quality multisampled players that offer a collection of solo and grouped instrument presets. See Studio Bass, Studio Horns, and Studio Strings.

Studio Piano is a sample-based software instrument that accurately reproduces the sounds of various pianos and simulates many of their physical characteristics, such as action movement, dampers, and resonance.



It's optimized for use with the Keyboard Player, but you can also use Studio Piano with a MIDI keyboard to play and record, and you can use it with MIDI and pattern regions. See the Edit a Keyboard Player performance topic in the Logic Pro User Guide.

If you're new to using plug-ins in Logic Pro, see Add, remove, move, and copy plug-ins.

Studio Piano parameters

- *Instrument pop-up menu:* Choose a piano. You can choose from the following:
  - Studio Grand
  - Concert Grand
  - Vintage Upright
  - Studio Grand (Mono Mic)
- *Stereo Mic A slider:* Set the volume of a stereo pair of condenser microphones.
- *Stereo Mic B slider:* Set the volume of a stereo pair of ribbon microphones.
- *Mono Mic slider:* Set the volume of a ribbon microphone.
- *Stereo Mic A On/Off button:* Turn stereo microphone pair A on or off.
- *Stereo Mic B On/Off button:* Turn stereo microphone pair B on or off.

- *Mono Mic On/Off button:* Turn the mono microphone on or off.
- *Main Volume:* Set the output level of the instrument.
- *Pedal Noise value:* Set the volume of the sound of pressing and releasing the piano pedals.
- *Key Noise value:* Set the volume of the sound of pressing and releasing the keys.
- *Release Samples value:* Set the volume of the damper returning to the string when you release a key. Release samples allow for a natural-sounding decay. If a low release sample value is set, the sound may cut off abruptly, which can sound unnatural.
- *Sympathetic Resonance value:* Set the volume of the resonance of undamped strings.

Studio Strings

Studio Strings in Logic Pro for Mac

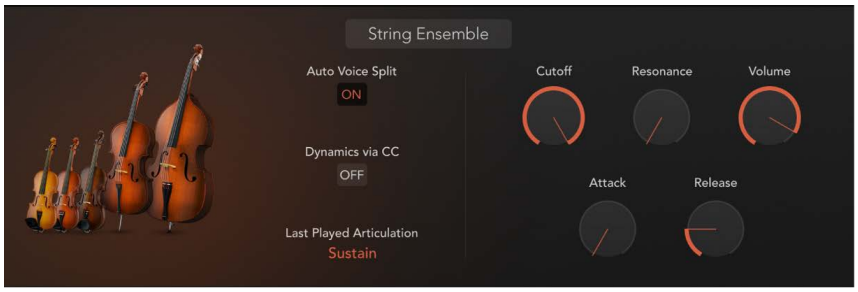
The studio instruments are high-quality multisampled players that offer a collection of solo and grouped instrument presets. See Studio Bass, Studio Horns, and Studio Piano.

Studio Strings provides a collection of highly expressive single and section stringed instruments. It has a simple, clean interface, with all main controls at hand. Additional controls let you customize Studio Strings to match your preferences and to work best with your controllers. See Studio Strings extended parameters.

Studio Strings instruments feature a large collection of samples that cover different playing styles, such as sustain, staccato, or falls. These playing styles, called articulations, are chosen in the plug-in header. You can also switch articulations remotely with keyswitches.

Default section and single instrument keyswitches are covered in Studio Strings keyswitch mapping.

If you're new to using plug-ins in Logic Pro, see Add, remove, move, and copy plug-ins.



Studio Strings parameters

- *Preset pop-up menu:* Choose a solo or section instrument.

  You can use MIDI channel events to control section instruments. If you want to play specific instruments in a section, you can send MIDI messages on their respective channels, then add a list of channel assignments. See Studio strings section MIDI channels.

- *Monophonic button:* Turn on monophonic mode for solo instruments.
- *Auto Voice Split button:* When on, chords played on the keyboard are automatically assigned to different instruments or instrument groups, or "voices" in the section.

Depending on the size and type of section, a voice can consist of one or multiple instruments. This mimics the distribution of notes (voicings) among players, based on what instrument group is best suited to a particular note range, for example. See Studio Strings extended parameters for details about additional Auto Voice Split functions.

Note: When Auto Voice Split is off, String sections are split by key ranges, with each string instrument within the section spanning a specific key range, and no layered instruments.

- *Dynamics via CC button:* Turn on to allow control of instrument dynamics (soft-loud) and timbre in real time. Additional dynamic control and mapping functions are available in the extended parameters.
- *Last Played Articulation field:* Displays the most recently used articulation. This may be different to the articulation shown in the plug-in header.
- *Cutoff knob:* Set the cutoff frequency of the sound.
- *Resonance knob:* Set the amount of cut or boost of the frequency range surrounding the defined cutoff frequency.
- *Volume knob:* Set the output level of the instrument.
- *Attack knob:* Set the amount of time it takes for the instrument to fade in, following a MIDI note on message.
- *Release knob:* Set the amount of time it takes for the instrument to fade out, following a MIDI note off message.

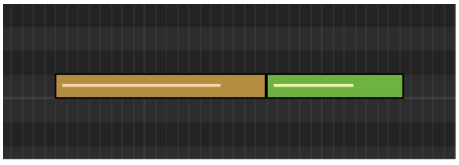
Choose a section or single instrument

1. In Logic Pro, click the Preset pop-up menu to open a window that shows available presets, then click or double-click a section or a single instrument icon to load the sound.
2. Click outside the window to close it.

Use advanced articulation features

Studio Strings provides several articulations for falls. These can be played on your keyboard like any other articulation. String players often attach a fall directly to a note end, so the fall happens with no gap after the previous note. Studio Strings features a special mode which allows you to smoothly connect a fall articulation to the previous note at any position “on-the-fly”, as a real musician would play it.

- In Logic Pro, to create a smooth fall, add a second (Fall articulation) note of the same pitch directly following any note. There should be no, or a minimal, gap from the previous note.



Note: Velocity of the Fall note can be different to the previous note. You can also vary the length of the Fall note. This allows you to create very subtle falls at the end of any note.

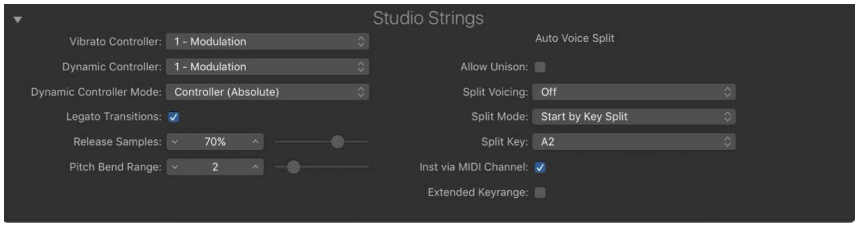
- To perform falls live on a keyboard, you can use the predefined keyswitches for Falls. These operate in Trigger mode, which automatically creates the Fall event and places it immediately after the previous note. Try this by holding a note with a Sustain articulation. While the note is playing, press the keyswitch for the Fall Long articulation. The sustained note is then stopped with a long fall.

Studio Strings extended parameters in Logic Pro for Mac

The extended parameters provide advanced control of performance and controller functions.

Click the disclosure arrow at the lower left to view the extended parameters.

Note: The parameters are different for solo and section instruments.



Studio Strings common extended parameters

The following parameters apply to solo or section instruments.

- *Vibrato Controller pop-up menu:* Choose a MIDI controller to adjust vibrato.
- *Dynamic Controller pop-up menu:* Choose the MIDI controller used to adjust an instrument's dynamics (soft-loud) while a note is playing. This function requires the Dynamics via CC button in the main interface to be turned on.
- *Dynamic Controller Mode pop-up menu:* Choose a mode that determines assigned MIDI controller behavior and instrument response to incoming velocity data.
  - *Controller (Absolute):* Dynamics are controlled only by the absolute value of the assigned MIDI controller.
  - *Velocity & Controller (Catch):* Dynamics are controlled by both incoming velocity data and by the value of the assigned MIDI controller when it matches any existing controller value. For example, a MIDI control knob with a value of 15 has no impact on dynamic behavior until it reaches a velocity value of 47 in an existing region. Once this controller value is reached, the assigned MIDI control knob will have an impact on instrument dynamics.
  - *Velocity & Controller (Relative):* Dynamics are controlled by both incoming velocity data and by the value of the assigned MIDI controller, relative to any existing controller value. For example, a MIDI control knob with a value of 15 will immediately impact dynamic behavior, relative to the existing velocity value. Adjusting a knob with a value of 15 to a value of 18 will adjust an existing velocity value of 47 to a value of 50.
- *Legato Transitions checkbox:* Turn on to allow legato transition samples for both monophonic and polyphonic modes, enhancing the realism of instruments played in a legato style.
- *Release Samples slider and field:* Set the volume of release samples. Release samples, as the name suggests, are triggered when you release a key. These allow for a natural-sounding decay in sustained articulations, for example. If a low release sample value is set the sound may cut off abruptly, which can sound unnatural.

- *Pitch Bend Range slider and field:* Determine the range for pitch bend modulation. This is bipolar, with a range of ±12 semitones.
- *Extended Key Range checkbox:* Turn on to allow the instrument (or instruments in a section) to play beyond its normal key range.
  - When off, the instrument (or instruments in a section) plays in the original key range. This is the default, and should be used if you are aiming for authentic-sounding performances.
  - When on, the instrument uses the full keyboard range, allowing you to create parts that extend beyond the realistic range of the instrument (or instruments in a section).

Studio Strings Auto Voice Split extended parameters

The following parameters apply only to section instruments when the Auto Voice Split button is active in the main window.

- *Allow Unison checkbox:* Turn on to activate unison mode for split voices.
  - When turned on, multiple instruments within a section can play the same note simultaneously when a single or multiple keys are pressed.
  - When turned off, one instrument within the section is played per key. To have all instruments of a seven piece section play, you need to press seven keys.
- *Split Voicing pop-up menu:* Determine the octaves used by the respective voices/instrument(s) in the section. Results vary based on your Split Mode pop-up menu choice.
  - *8va:* Double the top part an octave up. This is a common technique used to make melody lines more prominent.
  - *Drop 2:* Transpose the second note from the top of the chord down an octave. This technique is used to make an arrangement sound thicker and richer.
  - *Drop 2+4:* Transpose the second and fourth note from the top of the chord down an octave. This technique is used to thicken an arrangement. It is more suitable for sections with five or more instrument “voices,” but can be useful with smaller sections.
- *Split Mode pop-up menu:* Choose the mode used to split individual or grouped ensemble instruments.
  - *Start with Lead Voice:* The first incoming note is assigned to the Lead Voice, which is usually a higher-range instrument such as a violin. All subsequent notes are assigned, in descending order, to the Middle and Bass Voices.
  - *Start with Bass Voice:* The first incoming note is assigned to the Bass Voice. All subsequent notes are assigned, in ascending order, to the Middle and Lead Voices.
  - *Start by Key Split:* Incoming notes below the defined split key (set with the Split Key pop-up menu) are assigned to the Bass Voice and the lower Middle Voices. Notes above the split key are assigned to the Lead Voice and the higher Middle Voices.
- *Split Key pop-up menu:* Set the MIDI note number used as the split point for individual ensemble instruments.

  Note: This parameter applies only when *Start By Key Split* is chosen in the Split Mode pop-up menu.

- *Inst via MIDI Channel checkbox:* Turn off to play the full section on *any* incoming MIDI channel. Active Unison / Auto Voice Split / MIDI Split settings apply. When on (default), the following applies:
  - MIDI Channel 1: Full section is played with the active Unison / Auto Voice Split / MIDI Split settings.
  - MIDI Channels 2-16: Access individual instruments within the section. See Studio strings section MIDI channels.

Studio Strings keyswitch mapping in Logic Pro for Mac

This section outlines the default keyswitch mapping of the Studio Strings instruments.

Articulations are accessed with the Articulation pop-up menu in the plug-in header. Articulations can be remotely switched while performing or recording, allowing you to instantly switch between a bowed or staccato string section, for example.

The table shows the default keyswitch mapping, starting from C0 (MIDI note number 24). You can, of course, transpose the octave of notes used as keyswitches.

X indicates available articulation.

Key	MIDI Note #	Articulation ID	Articulation	Sections, Double Bass	Violins, Violas, Cellos
C0	24	1	Sustain	X	X
C#0	25	31	Tremolo	X	X
D0	26	19	Accented Sustain	X	X
Eb0	27	25	Fortepiano Slow	X	X
E0	28	23	Fortepiano Fast	X	X
F0	29	3	Spiccato	X	X
F#0	30	2	Staccato	X	X
G0	31	28	Pizzicato	X	X
G#0	32	29	Trill Semi	X	X
A0	33	30	Trill Whole	X	X
Bb0	34	10	Fall Long		X
B0	35	12	Fall Short		X
C1	36	17	Scoop Slow		X
C#1	37	16	Scoop Fast		X
D1	38	21	Crescendo Slow		X
Eb1	39	20	Crescendo Fast		X
E1	40	27	Glissando Up		X
F1	41	26	Glissando Down		X

Studio strings section MIDI channels in Logic Pro for Mac

In Studio Strings section presets, each instrument uses a dedicated MIDI channel. This lets you play specific instruments in a section by sending MIDI messages on their respective channels. Similarly, if you create a region for a section instrument, you can assign each MIDI event a specific MIDI channel to ensure that it is played only by a particular instrument within the section.

- MIDI channel 1 is always the “section” channel, meaning that it is used to play the entire section, either via Auto Voice Split, or as a Layer/Key Split, depending on the plug-in setting.
- MIDI channel 2 and above are used to access the different instruments of a section directly.

Note: When the *Inst via MIDI Channel checkbox* in the Extended parameters is turned off, the full section is played on *any* incoming MIDI channel. Active Unison / Auto Voice Split / MIDI Split settings apply. Full section playback is also triggered by incoming MIDI channel data that is not used in the section.

MIDI Channel	String Sections
1	Full Section
2	Violins 1
3	Violins 2
4	Violas
5	Cellos
6	Double Basses

Ultrabeat

Ultrabeat overview in Logic Pro for Mac

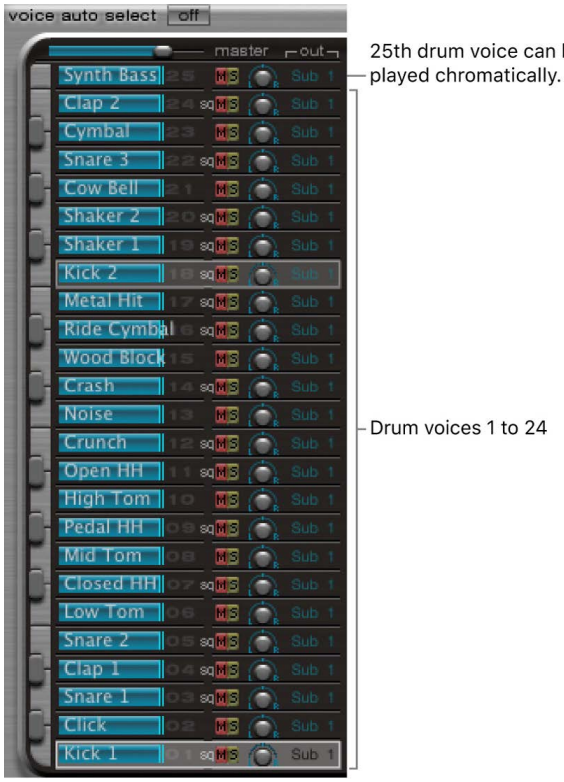
Ultrabeat is a synthesizer that is designed to create percussive sounds and polyphonic rhythms. It also incorporates a powerful step sequencer which you can use to create polyphonic rhythmic sequences and patterns. See Ultrabeat step sequencer overview.

If you're new to using plug-ins in Logic Pro, see Add, remove, move, and copy plug-ins and Use multi-output instruments.

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 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, allowing you to play this sound chromatically.

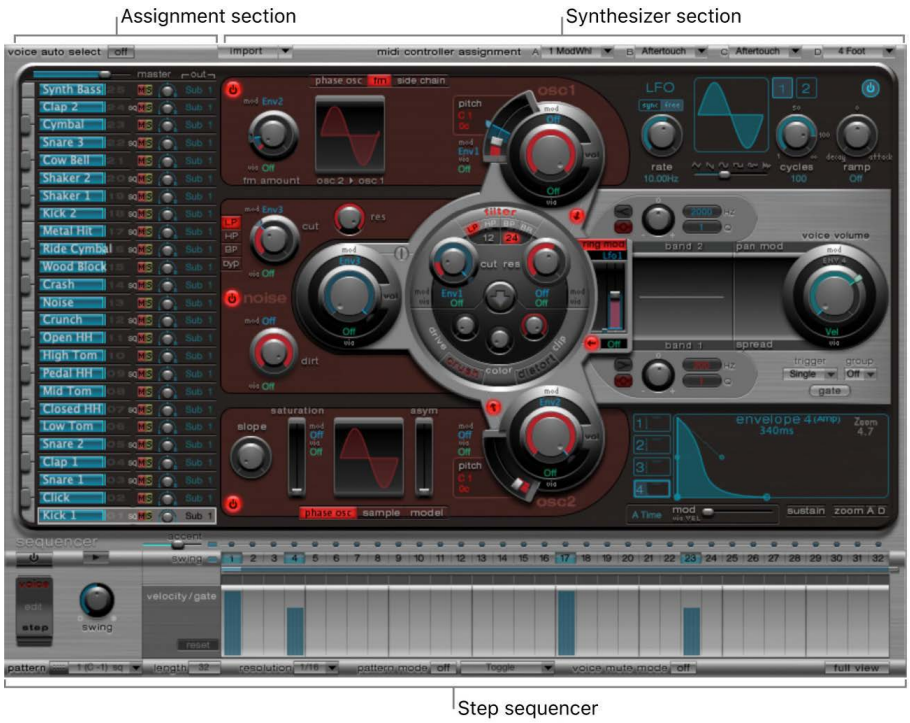


The 24-drum pad assignment is compatible with the widely adopted GM (General MIDI) MIDI Drum note mapping standard. If your MIDI keyboard is limited to two octaves or does not support transposition, use the Transpose options of Logic Pro to shift incoming MIDI notes up or down one or more octaves.

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 in Logic Pro for Mac

The Ultrabeat user interface is divided into three main sections.



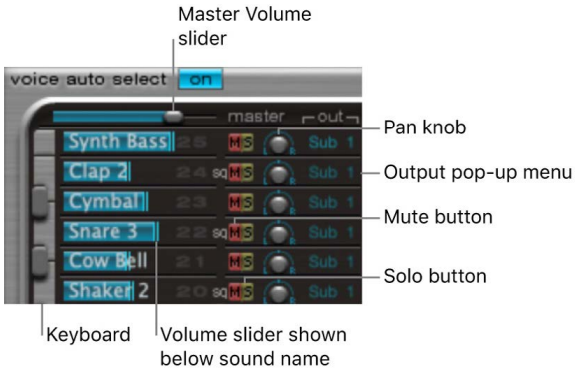
- *Assignment section:* Displays all drum sounds in a drum kit. You can select, name, and organize drum sounds here. This section also includes a small mixer, used to adjust the level and pan position of each sound. See Ultrabeat Assignment section overview.
- *Synthesizer section:* Used to create and 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.
- *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 Logic Pro or a keyboard. See Ultrabeat step sequencer overview.

Ultrabeat Assignment section

Ultrabeat Assignment section in Logic Pro for Mac

The Assignment section displays all sounds in a drum kit. You can:

- Select, organize, and name sounds
- Import sounds from other Ultrabeat settings or sampler instruments
- Set relative levels and pan positions for each sound
- Mute or solo sounds in the drum kit



Assignment section parameters

- *Master Volume slider:* Control 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.
- *Mute buttons:* Silence one or multiple sounds in the drum kit.
- *Solo buttons:* Hear one or multiple drum sounds in isolation. All other drum sounds (unsoloed) are muted.
- *Pan knobs:* Position drum sounds in the stereo field (panorama).

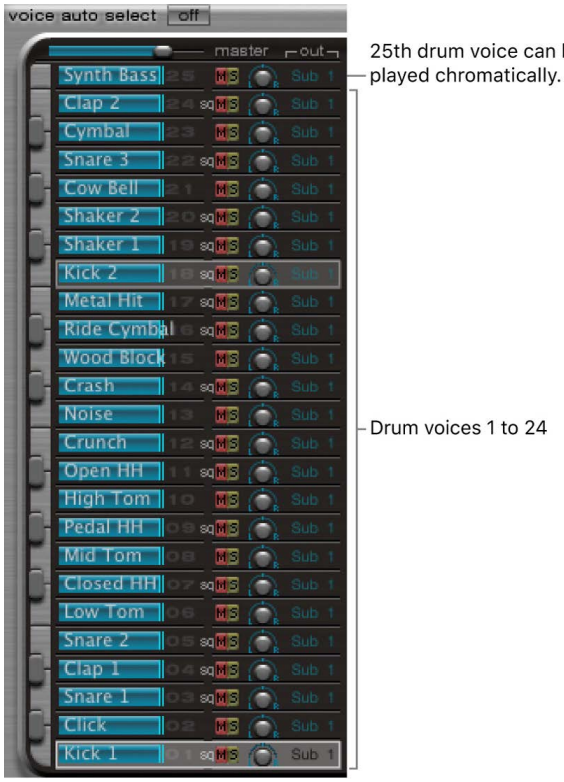
- *Output pop-up menus:* Use to independently route each drum sound to individual output subgroups or output (subgroup) pairs. Ultrabeat features eight separate stereo and eight separate mono outputs when inserted as a multi-output instrument. There are also 25 subgroups available–one for each sound in the kit, if required.

  See Use multi-output instruments.

  - Drum sounds routed to a subgroup output or output pair other than Main (1–2) are automatically removed from the main output channel strips.
  - Choosing a subgroup output or output pair other than Main (1–2) routes the sound to an aux channel strip.
  - Each sound can be assigned to an individual subgroup or you can assign multiple sounds to the same subgroup. Each subgroup is routed to a corresponding aux channel strip. This lets you individually process sounds, or process grouped sounds (all toms, for example), through effects inserted in an aux channel strip.

Play and select Ultrabeat sounds in Logic Pro for Mac

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

In Logic Pro, 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.

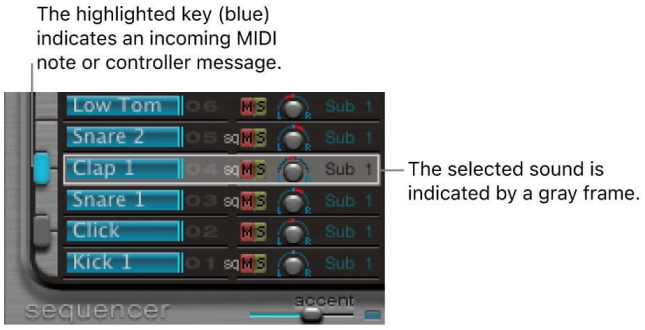
The corresponding key of the keyboard beside the drum sound name turns blue when it is clicked or when it receives MIDI information.

Select a drum sound

- In Logic Pro, click the name of the sound in the Assignment section.

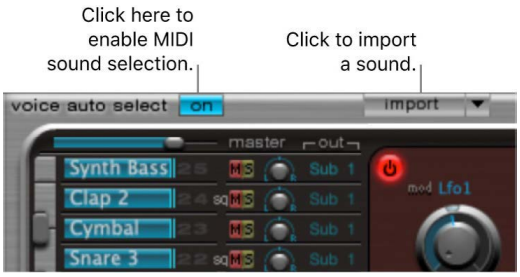
The selected sound is indicated by the gray frame around the assignment row. The parameters of the selected sound are shown in the Synthesizer section to the right. See Ultrabeat Synthesizer section overview.

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

- In Logic Pro: Turn on Voice Auto Select to select sounds by playing your MIDI keyboard.



Name, swap, and copy Ultrabeat sounds in Logic Pro for Mac

This section outlines the steps required to rename and rearrange the positions of drum sounds in a drum kit.

Rename a sound

1. In Logic Pro, 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 dragging or with a shortcut menu operation.

Drag to swap or copy drum sounds

- In Logic Pro, 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 operation *swaps* the two drum sounds (including Mixer settings: volume, pan, mute, solo, and output configuration). Sequences are *not* swapped.
  - Option-drag to *copy* the sound. Sequences are *not* copied.

Swap or copy drum sounds using a shortcut menu command

1. In Logic Pro, Control-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):* Allows 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 lets you 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.

Change the attributes of all voices

You can change the attributes of all voices with the following operations.

- In Logic Pro, Command-drag the Volume of a voice to set this volume value for all voices.
- Command-drag the Output of a voice to set this output assignment for all voices.

Import sounds into Ultrabeat in Logic Pro for Mac

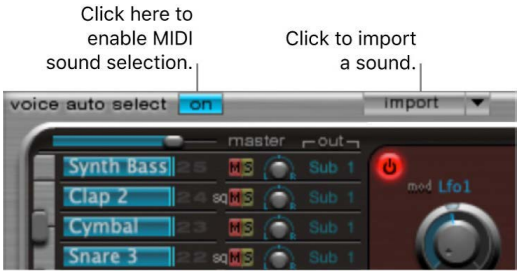
Ultrabeat can directly import sampler instruments, in addition to loading audio samples and its own settings, or Drum Synth settings. This provides enhanced sound design and rhythm processing options for sampler instruments, along with the ability to play and control Sampler drum kits with the intuitive drum mixer layout and Ultrabeat sequencer.

There are two key elements that you use to import sounds and sequences into Ultrabeat:

- *Import button:* Choose a kit that contains sounds or sequences you want to import.
- *Import list button:* Hide or show the import list, then add drum sounds or sequences from Ultrabeat settings or sampler instruments to the active kit.

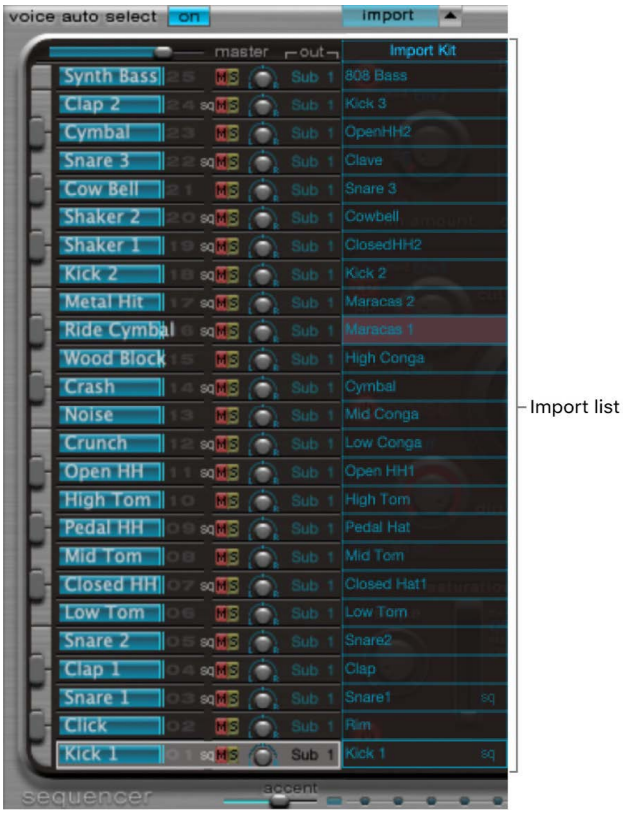
Open an Ultrabeat setting or sampler instrument with the import list

1. In Logic Pro, click the Import button.



2. Select the source Ultrabeat setting or sampler instrument in the dialog, then click Open.

   A list of all sounds found in the selected setting, or samples in the sampler instrument, is shown to the right of the Assignment section Mixer.



Drag sounds into your drum kit

- In Logic Pro, 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. In Logic Pro, 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 lets you 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 a sampler instrument into Ultrabeat

- In Logic Pro, drag a sampler instrument filename from the Finder directly onto the Assignment section.

  Ultrabeat reproduces the Sampler layout as closely as possible. Layered Sampler zones are set up as layered drum sounds, using the sample playback mode of oscillator 2. See Use Ultrabeat oscillator 2 sample mode.

  Note: This method does not allow paging through the sampler instrument if it contains more than 25 sounds (samples). Ultrabeat maps only sample zones and layers that fall within the Ultrabeat drum sound range of C1 to C3. All other samples (zones) are ignored.

Open Drum Synth settings with Ultrabeat

In Logic Pro, do one of the following:

- Choose the Drum Synth setting name from the Ultrabeat Setting menu. Use the Load menu item to browse to the default location: ~/Music/Audio Music Apps/Plug-In Settings/Drum Synth.

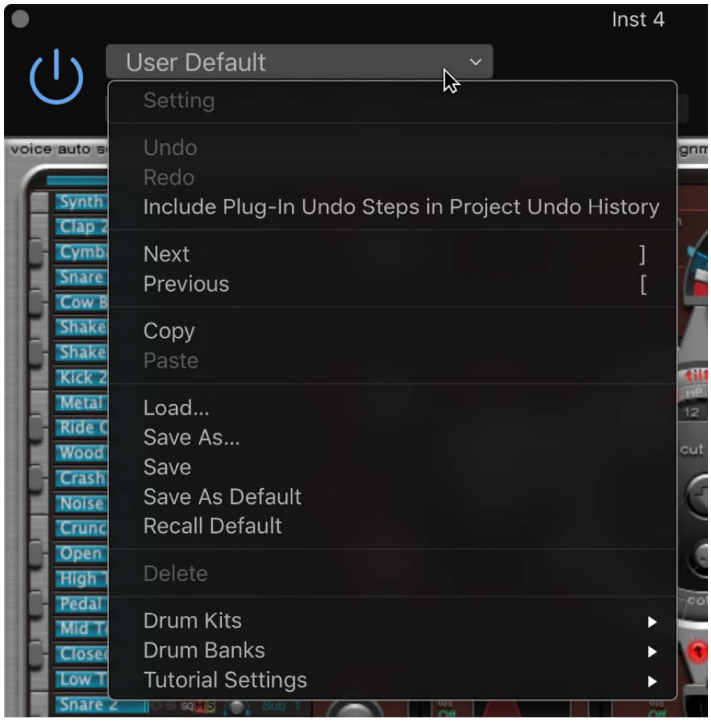
  The Drum Synth sound is loaded into the Assignment section. You can edit, process, and handle it as you would with any Ultrabeat sound.

- Choose Ultrabeat from the Instrument pop-up menu on an instrument channel strip that contains a Drum Synth instance.

  Drum Synth is replaced with Ultrabeat on the channel strip and the Drum Synth sound is automatically loaded into the Assignment section.

Ultrabeat settings in Logic Pro for Mac

Ultrabeat settings are saved and loaded in the same way as other instruments. You can also use the Setting pop-up menu to load Drum Synth settings, allowing you to leverage Ultrabeat synthesis functions for Drum Synth sounds.



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 Setting 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 in Logic Pro for Mac

The Ultrabeat 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, allowing 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 the Ultrabeat synthesis engine is based on classic synthesizer designs. If you're new to synthesizers, it might be best to start with Synthesizer basics overview, 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–envelope.

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 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 about and exploring the Ultrabeat 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 Ultrabeat filter section.

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.

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 the 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.

The output of the drum sound is then sent to the Assignment section mixer. See Ultrabeat Assignment section overview.

Ultrabeat sound sources

Ultrabeat oscillator overview in Logic Pro for Mac

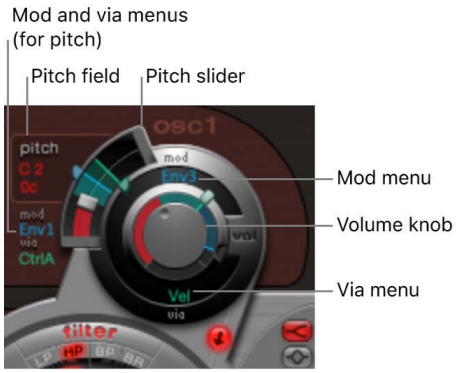
Ultrabeat oscillators are used to generate waveforms. Oscillator 2 can use a sample in place of a waveform. The signal of one or both oscillators is then sent to other portions of the synthesizer engine for shaping, processing, or manipulation.

- 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 and Ultrabeat noise generator.

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:* Set 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 = 1/100 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 highlighted 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.

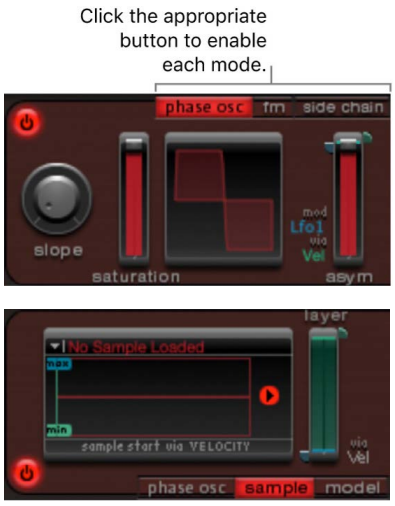
Switch between Ultrabeat oscillator synthesis modes

Oscillator 1 can switch among three synthesis engine types: phase oscillator, fm, and side chain (external audio input). Each mode provides different parameters and features.

Oscillator 2 can switch among three synthesis engine types: phase oscillator, sample, and model. Each mode offers different parameters and features.

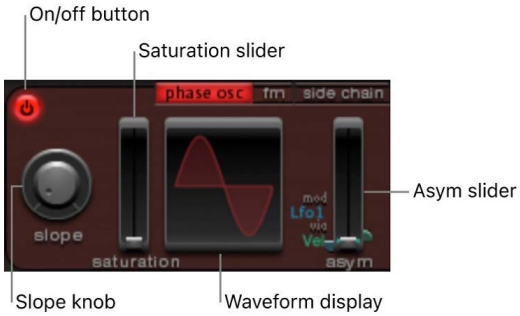
- In Logic Pro, click a button on the upper edge of oscillator 1 to select a mode (synthesizer engine).

- Click a button on the lower edge of oscillator 2 to select a mode (synthesizer engine).



Ultrabeat oscillator 1 phase osc mode in Logic Pro for Mac

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 Ultrabeat oscillator 2 phase osc 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 gain, eventually causing the waveform to clip. Higher values make the waveform shape more rectangular. This results in a corresponding increase in odd-numbered overtones.
- *Asymmetry 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, allowing dynamic sound changes at the oscillator level. See Ultrabeat modulation overview.

Note: The oscillator 2 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 in Logic Pro for Mac

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 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.

In FM synthesis mode, oscillator 1 (the carrier) generates a sine wave. The frequency of the oscillator 1 sine wave is modulated by the waveform of oscillator 2 (the modulator).

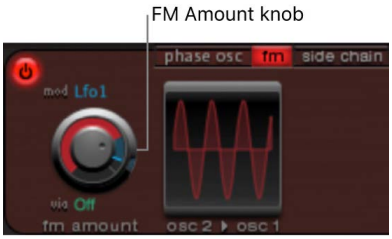
- When oscillator 2 outputs a positive (or higher) frequency signal, the frequency of oscillator 1 increases.
- When oscillator 2 outputs a negative (or lower) frequency signal, the frequency of oscillator 1 decreases.

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. In Logic Pro, 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.

Ultrabeat oscillator 1 side chain in Logic Pro for Mac

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 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 lets you 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:

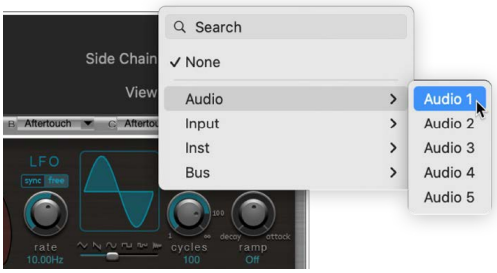
- The side chain affects only the selected drum sound—other Ultrabeat drum sounds and sequences are not altered.
- A side chain audio signal alone is not enough to trigger Ultrabeat. To hear the side chained audio signal, you need to make sure that Ultrabeat is triggered by MIDI or the internal step sequencer.

Use an Ultrabeat side chain

1. In Logic Pro, 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.

Ultrabeat oscillator 2 phase osc mode in Logic Pro for Mac

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, allowing dynamic sound changes at the oscillator level. See Ultrabeat modulation overview.

Create classic waveforms with the Ultrabeat phase oscillator

The basic waveforms of classic analog synthesizers can be reproduced with the phase oscillator: sine, rectangular, and sawtooth waves result from different Slope, Saturation, and Asym parameter value combinations.

InLogic Pro, 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.

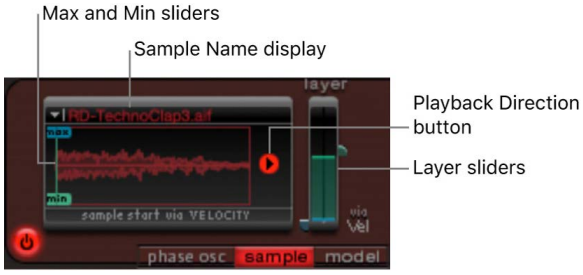
Ultrabeat waveform characteristics in Logic Pro for Mac

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 Logic Pro for Mac

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:* Change the playback direction of the sample (forward or backward).
- *Sample Layer sliders:* Ultrabeat samples and sounds imported from sampler instruments often have 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 Min slider determines which sample layer is triggered at a MIDI note velocity = 1.
  - The Max slider determines which sample layer is triggered at a MIDI note velocity = 127.

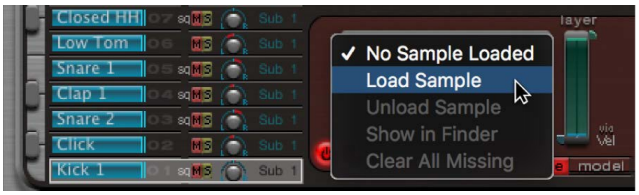
  Note: If you have loaded a single sample that does not have multiple layers, the Vel Layer slider has no effect.

Load a sample into Ultrabeat oscillator 2

Ultrabeat supplied settings include a selection of multilayer drum and percussion samples. You can also load your own samples in AIFF, WAV, or CAF format.

Note: The velocity layering function is not available for user-loaded samples.

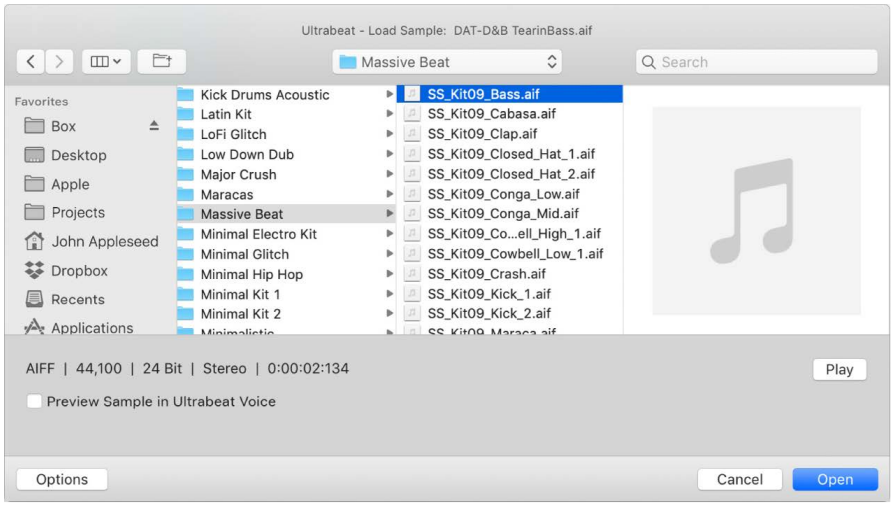
1. In Logic Pro, choose Load Sample from the pop-up menu in the upper-left corner of the waveform display.



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 the Ultrabeat load sample preview

You can audition samples before importing them into Ultrabeat.



In Logic Pro, do one of the following:

- To preview audio files (AIFF, WAV, 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.

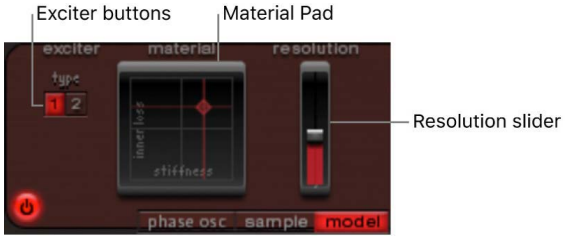
Ultrabeat oscillator 2 model mode in Logic Pro for Mac

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 lets you 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 lets you 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:* Select one of the two contrasting exciters. Each offers different sound characteristics.

  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:* Determine the precision of the calculation. High values produce extra harmonics. Low values produce fewer harmonics, or inharmonic spectra.

Use the Ultrabeat 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).

- In Logic Pro, you can simultaneously adjust the Inner Loss and Stiffness parameter positions by dragging 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.

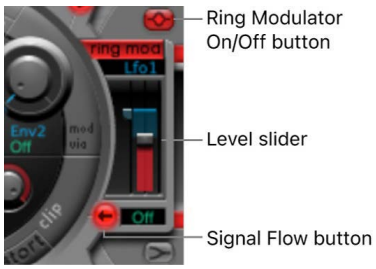
  Note: Option-click the ball to reset all string parameters to their default values.

Ultrabeat ring modulator in Logic Pro for Mac

Ring modulation is a powerful tool for the creation of inharmonic, metallic, bell-like sounds. The Ultrabeat 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 turned on 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:* Turn the ring modulator on or off.

  Note: If you want to hear the ring modulator signal in isolation (to better judge your settings), temporarily set the volume of both oscillators to a value of 0.

- *Level slider:* 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.

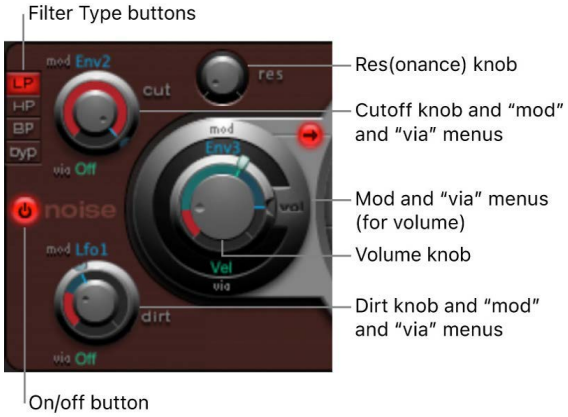
- *Signal Flow button:* Determine the routing of the ring modulator signal to the filter (red) or to the EQ section (gray). The direction of the arrow on the Signal Flow button illustrates the routing.

  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 in Logic Pro for Mac

The Ultrabeat noise generator lets you 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

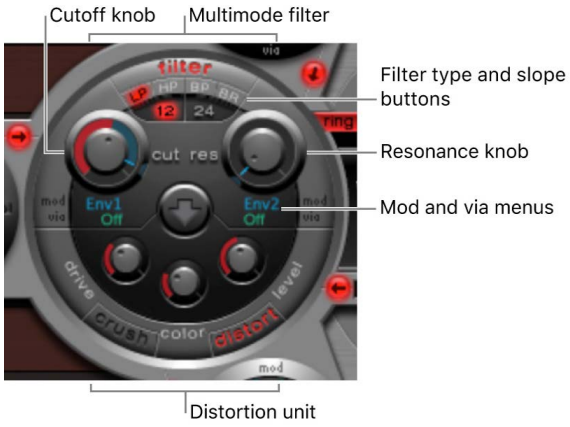
- *On/Off button:* Turn the noise generator on or off. When programming a drum sound, you can turn the individual sound sources on or off. You can also listen to, or remove, individual components of the sound separately in this way.
- *Filter Type buttons:* Switch the noise generator integrated filter 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):* Bypass the integrated filter.
- *Cutoff and Resonance knobs:* Rotate to set the cutoff/center frequency and resonance/bandwidth behavior of the noise generator 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:* Set higher values to 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:* Use to route the noise generator signal through the main Ultrabeat filter or directly to the EQ (Output) section. When turned on, the button is highlighted 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.

Ultrabeat filter section in Logic Pro for Mac

The Ultrabeat multimode filter can dramatically, or subtly, alter the timbre of your drum sounds.



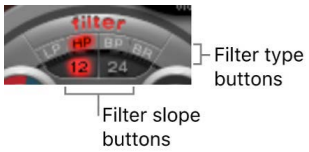
Filter parameters

- *Filter (On/Off) button:* Turn 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 turned off.
- *Filter Type buttons:* Switch the filter between lowpass, highpass, bandpass, or band-rejection filter types.
- *Filter Slope buttons:* Switch the filter between different slopes.
- *Cutoff and Resonance 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 modulation.

Set the Ultrabeat filter type

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

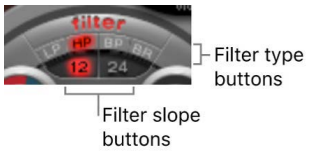
- In Logic Pro, to select a filter type, click one of the following buttons:
  - *LP (lowpass):* This filter type allows frequencies that fall below the cutoff frequency to pass. When set to LP, the filter operates as a lowpass filter. The slope of the filter can be set to 12 or 24 dB/octave in LP mode.
  - *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 the Ultrabeat 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.

- In Logic Pro, 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 the Ultrabeat filter Cutoff parameter

- In Logic Pro, 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.

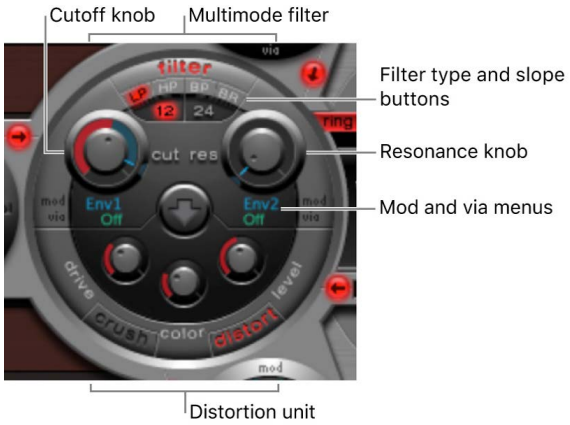
Use the Ultrabeat filter Resonance parameter

- In Logic Pro, 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 signal flow order through the Ultrabeat filter and distortion unit

The output signals of both oscillators, the ring modulator, and the noise generator are sent to the central Filter section (if not bypassed with the various Signal Flow buttons). The Filter section offers a multimode filter and a distortion unit. Sounds are passed through the filter and distortion unit as determined by the direction of the arrow found in the center of the filter section.

- In Logic Pro, 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 in Logic Pro for Mac

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 signal flow order through the Ultrabeat filter and distortion unit).



Distortion parameters

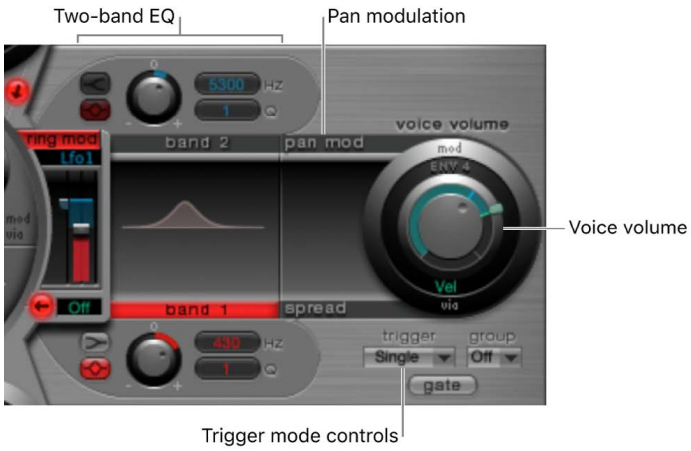
- *Crush and Distort buttons:* Activate the mode you want to use. The name of the active effect type is indicated in red. If neither button is active, the Distortion circuit is bypassed.
- *Drive knob:* Control the amount of distortion.
- *Color knob:* Determine the basic tone of the distortion. Higher values result in a brighter sound. Lower values lead to a darker, warmer tone.
- *Level knob:* Set the output level of the Distortion effect in Distortion mode. In Bit Crusher mode, set a threshold level for incoming signals. Sound sources must reach this threshold before distortion (bit crushing) begins.

Ultrabeat Output section

Ultrabeat output section overview in Logic Pro for Mac

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.

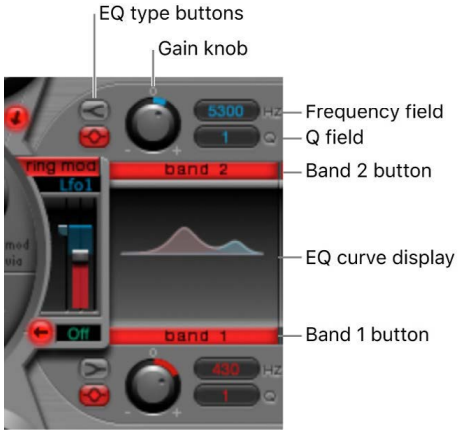
The Output section passes signals through both equalizers (EQ), then on to the Pan Modulation/Stereo Spread section (in a preconfigured order) before the final level is set for the selected sound and the trigger behavior is adjusted.



- *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.

Ultrabeat two-band EQ in Logic Pro for Mac

Both equalizer bands have almost identical features. You can adjust each band separately.



Two-band EQ parameters

- *Band 1/2 buttons:* Turn each band on or off. Band 1 is a low shelving EQ. Band 2 is a high shelving EQ. If neither EQ is activated, the signal passes through unaffected.
- *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:* Boost or cut a certain frequency range as determined by the EQ type and Hz settings. 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 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 EQ curve

In Logic Pro, 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.

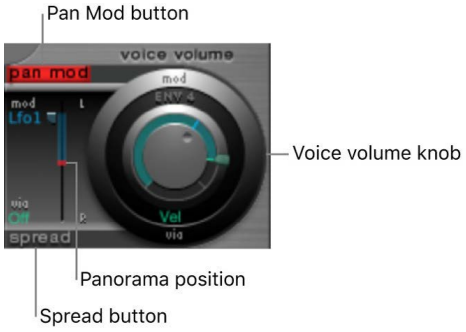
Ultrabeat pan and stereo spread

Use Ultrabeat pan modulation in Logic Pro for Mac

The EQ output signal is passed 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 pan position set in the Assignment section mixer.



Pan modulation parameters

- *Pan Mod button:* 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:* 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.

Use Ultrabeat stereo spread in Logic Pro for Mac

The EQ output signal is passed 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.



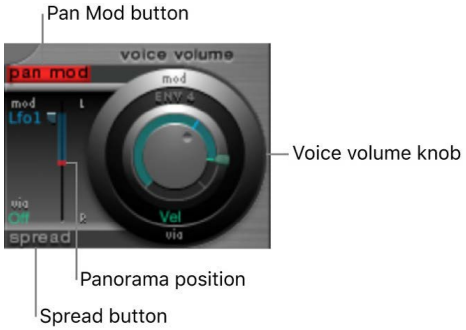
Stereo spread parameters

- *Spread button:* Turn on stereo spread mode. If neither mode is activated, the signal passes through unaffected.
- *Lo Frequency slider:* Move to set the width of bass frequencies.
- *Hi Frequency slider:* Move to set the width of high frequencies.

Ultrabeat voice volume control in Logic Pro for Mac

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.

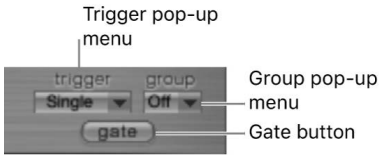


The intensity of Envelope 4 impact on Voice volume can also be modulated with a via source.

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 mixer. See Ultrabeat Assignment section overview.

Ultrabeat trigger mode in Logic Pro for Mac

The way Ultrabeat reacts to a succession of incoming notes is set independently for each sound. Parameters that control this aspect of Ultrabeat 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 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:* Turn the Gate function on or off. When active, the sound immediately cuts off when the MIDI note is released, 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 the Ultrabeat internal sequencer. Note length can be an important creative element when programming rhythm tracks.

Ultrabeat modulation

Ultrabeat modulation overview in Logic Pro for Mac

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 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.

Ultrabeat Mod and via modulation in Logic Pro for Mac

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 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 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 an Ultrabeat mod routing in Logic Pro for Mac

This section outlines the creation of mod (and via) modulation routings in Ultrabeat. The following applies to all parameters that offer mod (and via) modulation options.

Create a modulation routing

1. In Logic Pro, 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. 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 Set Ultrabeat MIDI controllers A–D.



4. Adjust the mod and via controls.

Set Ultrabeat MIDI controllers A-D in Logic Pro for Mac

The MIDI Controller Assignments area lets you assign any MIDI controller shown in the menus to each of the four controller slots—Ctrl A, B, C, or D.



These assignments allow 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

- In Logic Pro: Click a Ctrl A–D pop-up menu, then choose the controller name or number that you want to use.

Learn a MIDI controller assignment

1. In Logic Pro, 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 in Logic Pro for Mac

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 1/2 buttons:* Select the corresponding LFO, allowing independent parameter adjustments.
- *On/off button:* Turn the selected LFO on or off.
- *Sync/free buttons:* The LFO speed can be synchronized with the Logic Pro for Mac tempo or set independently. Click either button to activate the corresponding mode.
- *Rate knob and field:* 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:* Drag to determine the shape of the LFO waveform. The shape updates in the display.
- *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

- In Logic Pro, 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 lets you 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 outputs *random* values. A random value is selected at regular intervals, as defined by the LFO rate. Use a random modulation of oscillator pitch to generate an effect commonly referred to as a *random pitch pattern generator* or *sample and hold*. Play very high notes, at high LFO rates and high intensity—you’ll recognize this well-known effect from hundreds of science fiction movies. 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.

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 lets you set the number of LFO cycles with the Cycles parameter. After completing the defined number of cycles, the LFO stops oscillating.

- In Logic Pro, 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 in Logic Pro for Mac

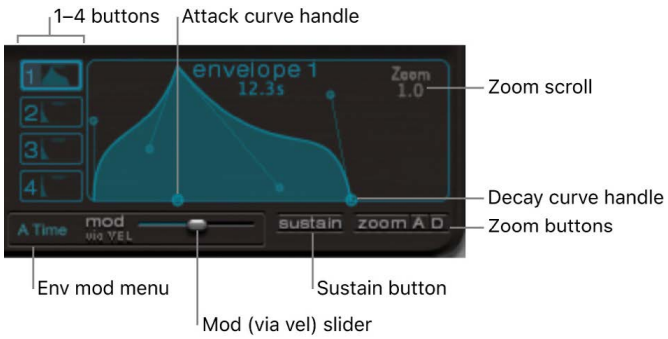
Ultrabeat features four identically specified envelope generators per voice. They are abbreviated as Env 1 to Env 4. In addition to potential use as a modulation source (in the mod pop-up menus of various sound parameters), Env 4 is permanently connected to the Voice Volume parameter. In other words, each Ultrabeat drum sound has a hard-wired volume envelope generator—Env 4.

See Attack, decay, sustain, and release 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.

The 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.

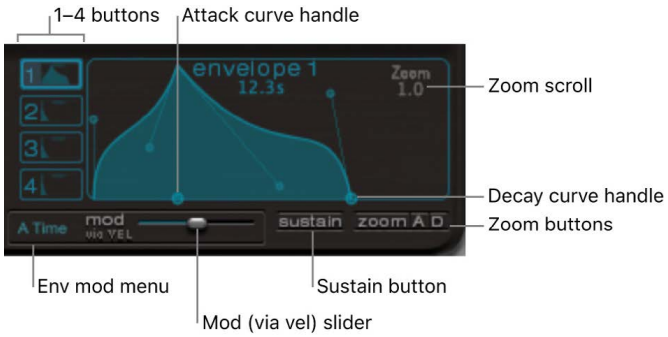
Both of the larger handles on the x-axis (the horizontal, or time axis, at the bottom) control the attack and decay times, respectively. A vertical line extends up from the first of the two handles (attack), and divides the envelope into an attack and decay phase.

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 in Logic Pro for Mac

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:* 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 (set 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:* Turn on to display a red handle and line on the x-axis which can be dragged horizontally 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.

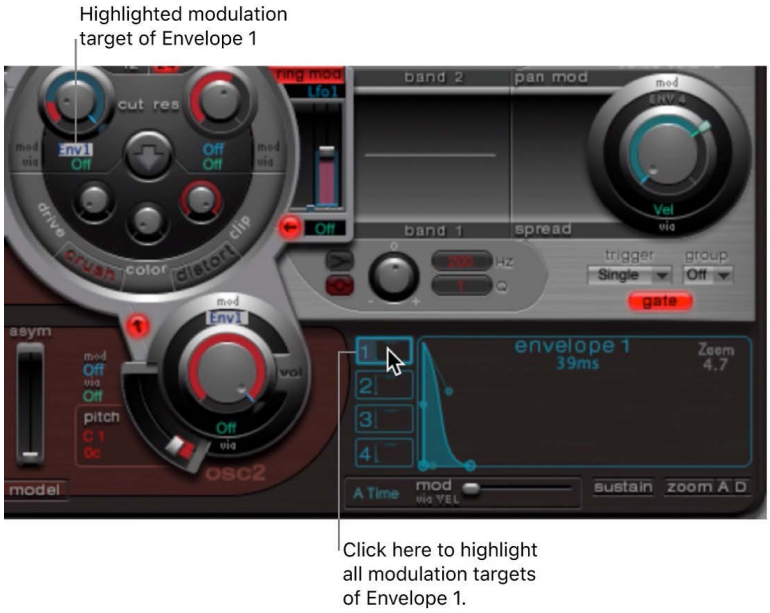
  Note: When the Zoom function is turned on, the decay handle can be dragged beyond the right edge of the envelope display area, allowing you to lengthen the decay time. After you release the mouse button, the envelope graphic is automatically resized to fit the display area.
- *Zoom A/D buttons:* Click to show only the attack (A) or decay (D) phase across the entire width of the envelope display. This lets you perform more accurate edits to envelope shapes (down to millisecond values).

Ultrabeat modulation target display in Logic Pro for Mac

Ultrabeat has a feature that makes it easy to find the modulation targets of LFOs and envelopes.

Find modulation targets of LFOs and envelopes

- In Logic Pro, click the numerical field of the modulation source to highlight all of its modulation targets.



Ultrabeat step sequencer

Ultrabeat step sequencer overview in Logic Pro for Mac

Ultrabeat incorporates a step sequencer which you can use to create polyphonic rhythmic sequences and patterns. The sequencer displays running light-style controls like those found on classic drum machines and shares many of the sequence and pattern creation methods employed in these devices.

The Ultrabeat step sequencer expands on the features of hardware drum machines by providing extensive automation and editing features. These let you 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 lets you 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 the Ultrabeat internal step sequencer.

If you're unfamiliar with the concept of step sequencing, see Ultrabeat step sequencer basics.

Ultrabeat step sequencer basics in Logic Pro for Mac

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 sound 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.

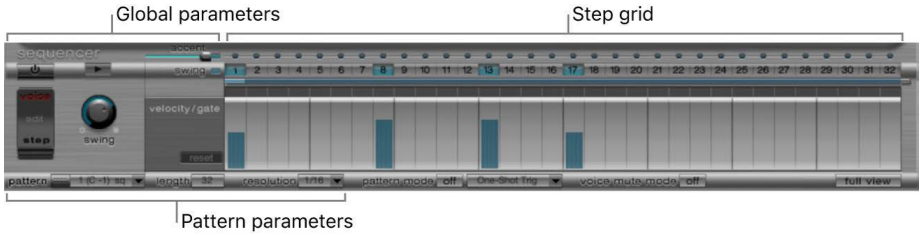
Despite these technological advances, step sequencers haven't disappeared completely. Hardware groove boxes have experienced a renaissance in recent years due to their intuitive nature, which has made them a favored tool for rhythm programming.

The Ultrabeat 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 in Logic Pro for Mac

The Ultrabeat 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.



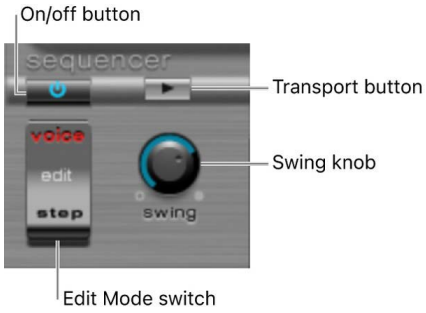
The step sequencer is divided into three sections.

- *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 Use Ultrabeat Step grid full view.

Ultrabeat global sequencer controls in Logic Pro for Mac

The following parameters apply globally to all patterns.



Global sequencer parameters

- *On/Off button:* Turn the step sequencer on or off.
- *Edit mode switch:* Select either Voice or Step mode.
  - *Voice mode (default):* In Voice mode, editing the parameters of a drum sound changes the drum sound itself.
  - *Step mode:* In Step mode, you can automate a the parameters of a sound from one step to the next. See Ultrabeat step automation overview.

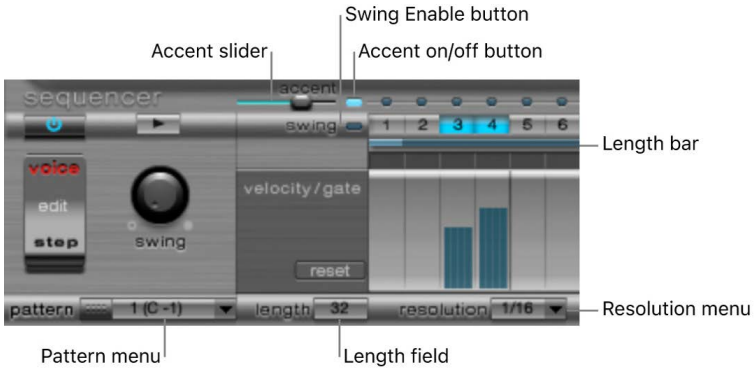
- *Transport button:* Start and stop the sequencer pattern. The step sequencer is always synchronized with the host application tempo.

  Note: If the Transport button is blue, the step sequencer interprets incoming MIDI notes between C-1 and B0 as performance information. See Ultrabeat sequencer MIDI control.

- *Swing knob:* Rotate to set the swing intensity for all sounds that have the Swing function turned on. See Use the Ultrabeat swing function.

Ultrabeat pattern controls in Logic Pro for Mac

A pattern contains all events, stored in sequences, for all 25 sounds. You can select one of 24 patterns and can 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:* Adjust the length of the grid (and therefore the pattern) by dragging in the field or the bar beneath the swing buttons.
- *Resolution pop-up menu:* 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 lets you 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 accents.
- *Swing Enable button:* Turn on to play the grid of the currently selected sound in accordance with the Swing knob setting. See Use the Ultrabeat swing function.

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. In Logic Pro, 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. In Logic Pro, 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. In Logic Pro, 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 the Ultrabeat swing function in Logic Pro for Mac

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.

Use the swing function

1. In Logic Pro, 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.

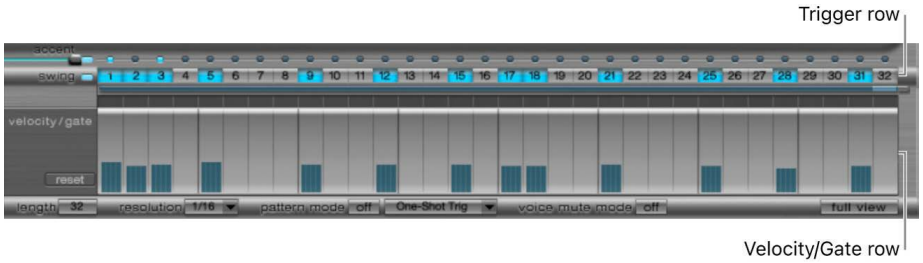
   At a zero setting (full left position), the Swing function is turned off. Rotate the knob to the right to move affected notes toward the following note.

Ultrabeat Step grid

Ultrabeat Step grid overview in Logic Pro for Mac

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 height represents the velocity.
  - The bar length indicates the note length.

Create or remove Ultrabeat steps in Logic Pro for Mac

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. In Logic Pro, 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.
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 Use Ultrabeat Step grid full view.

Remove a step

1. In Logic Pro, 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 in Logic Pro for Mac

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

In Logic Pro:

- *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 Ultrabeat step length and velocity in Logic Pro for Mac

The Velocity/Gate row lets you set the length (gate time) and velocity of notes in the Trigger row. Both parameters are displayed as a single graphical bar.

- The bar height represents note velocity.
- The bar width indicates the note length (gate time).



Change the length and velocity values for each step

In Logic Pro, 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 Ultrabeat trigger mode) or use envelopes in sustain mode (see Ultrabeat envelope parameters), in conjunction with rhythmically useful (short) decay times.

Reset all velocity and gate values to default settings

- In Logic Pro, the Reset button is shown at the left of the velocity/gate row. Click to reset all velocity and gate values.

  The default velocity setting is 75 percent. The default gate value is all four sections active.

Use the Ultrabeat velocity/gate shortcut menu

- In Logic Pro, Control-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 accents in Logic Pro for Mac

The Accent setting can be switched on or off for each drum sound. This lets you turn accents on for cymbals but turn accents off for the kick drum, for example.

Turn on accents and set the accent level

1. In Logic Pro, 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

- In Logic Pro, click the blue LED above the step (steps 1 and 3 in the figure).

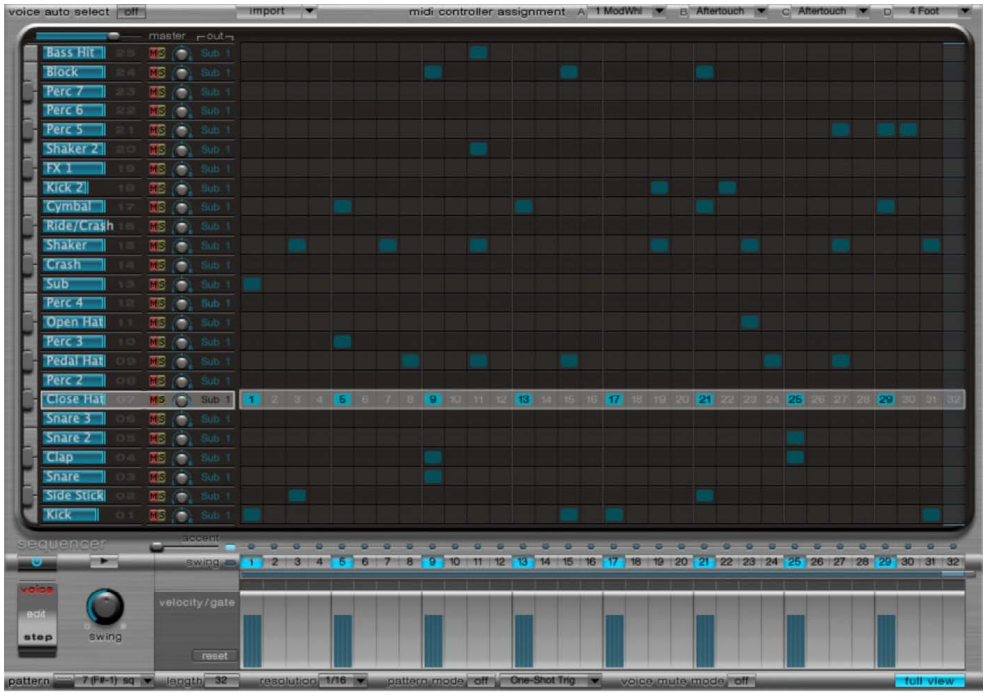
  The sound at this step position is accentuated (played louder).



Use Ultrabeat Step grid full view in Logic Pro for Mac

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), 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 Ultrabeat step sequencer parameters

Ultrabeat step automation overview in Logic Pro for Mac

Click the Edit Mode switch to turn on Ultrabeat step automation. Step automation lets you 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, the Ultrabeat 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 inactive.
- 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 the Ultrabeat offset row in Logic Pro for Mac

The offset row lets you 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 are 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 adds the parameter to the Parameter Offset pop-up menu and creates an offset, so take care.

Create a parameter offset

1. In Logic Pro, 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 is 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

- In Logic Pro, Control-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

- In Logic Pro, 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):* Allows 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 in Logic Pro for Mac

Patterns programmed in the Ultrabeat step sequencer can be exported as MIDI regions into Logic Pro.

Export an Ultrabeat pattern into Logic Pro

1. In Logic Pro, choose the pattern that you want to export from the Pattern pop-up menu.
2. Select the area to the left of the Pattern pop-up menu.



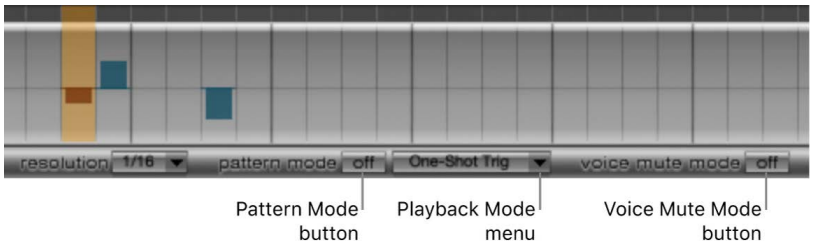
3. Drag the pattern to the target position on the respective Ultrabeat track.

   A region is created containing all MIDI events, including Swing and Accent parameter settings. Accents are interpreted as polyphonic aftertouch 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 the Ultrabeat step sequencer.

Ultrabeat sequencer MIDI control in Logic Pro for Mac

Pattern performance can be influenced by incoming MIDI notes. This allows 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 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:* Determine 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 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 in Logic Pro for Mac

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 will 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 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: In the plug-in Settings menu, choose Tutorial Kit from the Tutorial Settings subfolder. 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
- Create Ultrabeat snare drums
- Create Ultrabeat tonal percussion
- Create Ultrabeat hi-hats and cymbals
- Create metallic Ultrabeat sounds
- Tips for extreme Ultrabeat sounds

Create Ultrabeat kick drums in Logic Pro for Mac

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. In Logic Pro, choose Settings > 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 may want a longer decay phase, whereas at faster tempos a shorter decay time is more appropriate. The attack time of Env 4 should be very short (zero, in most cases) or the sound loses 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. In Logic Pro, make sure that Env 1 is chosen from the mod pop-up menu of the oscillator 1 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 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. In Logic Pro, 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. In Logic Pro, 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

In Logic Pro, 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 envelope 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. In Logic Pro, 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 lets you 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 in Logic Pro for Mac

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. In Logic Pro, 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. In Logic Pro, 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. In Logic Pro, 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 the 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 TR-808 resonating filters. Shifting the pitch of both oscillators simulates the behavior of the TR-808 Tone control.

Complete the 808 emulation by adding some noise

1. In Logic Pro, 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. In Logic Pro, select the “snare 3-808” sound.
2. Choose 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. In Logic Pro Ultrabeat, 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 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. In Logic Pro, 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 in Logic Pro for Mac

Tonal percussion sounds such as toms or congas are relatively easy to emulate electronically with sine or triangular wave oscillators. The Ultrabeat phase oscillator offers you a broad spectrum of suitable basic sounds to start with. 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

- In Logic Pro, 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 in Logic Pro for Mac

Electronic hi-hat sounds are easy to create in Ultrabeat.

Create a hi-hat in Ultrabeat

1. In Logic Pro, 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.

- In Logic Pro, 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.

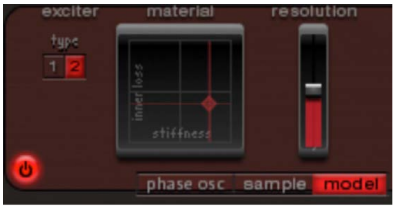
Tip: Certain electronic cymbal sounds, such as the TR-808 cymbal, can't be perfectly replicated using the Ultrabeat synthesis engine. In these cases, your best option is to use a sample. A number of these samples are included in the Ultrabeat sample library for your use.

Create metallic Ultrabeat sounds in Logic Pro for Mac

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. In Logic Pro, 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 in Logic Pro for Mac

Ultrabeat features very fast envelopes and advanced LFOs which can be used for 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.

Utilities

Utility instruments in Logic Pro for Mac

The Utilities found in the instrument plug-in menu include:

- External Instrument lets you route hardware MIDI sound generators through the Mixer, which you can then process with Logic Pro for Mac effects.
- Klopfgeist is an instrument that provides a metronome click.
- Test Oscillator can be used as an instrument or effect. It is used to calibrate tuning of studio equipment and instruments.

If you're new to using plug-ins in Logic Pro, see Add, remove, move, and copy plug-ins.

External Instrument in Logic Pro for Mac

You can use External Instrument to route hardware MIDI sound generators through the Mixer, which you can then process with Logic Pro effects. The plug-in is inserted into instrument channel strips in place of a software instrument. External Instrument supports MIDI communication through the instrument channel strip, allowing you to control both MIDI and audio routing of an external sound module with a single plug-in.

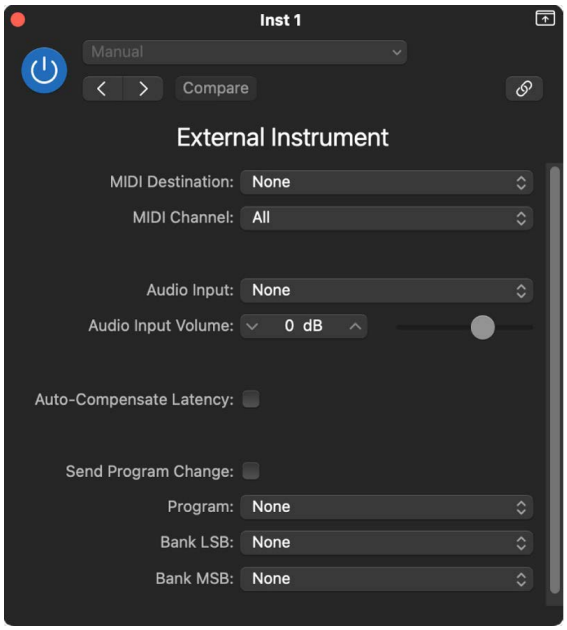
If you're new to using plug-ins in Logic Pro, see Add, remove, move, and copy plug-ins.

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 lets you 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 the computer CPU—apart from effects used in the channel strip (or destination channels).
- 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 (bouncing cannot happen faster because MIDI hardware is involved). This makes the creation of a mix, including all internal and external devices and tracks, a one-step process.

Tip: To avoid constant repatching of devices, it is best to use an audio interface that supports multiple inputs and outputs.

Tip: When using multi-timbral MIDI sound sources, you can gain maximum flexibility with 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 an individual External Instrument instance.



External Instrument parameters

- *MIDI Destination pop-up menu:* Choose the target MIDI port.
- *MIDI Channel pop-up menu:* Choose the target MIDI 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:* Set the incoming signal level.
- *Auto-Compensate Latency checkbox:* Turn on to automatically compensate for small timing differences between MIDI and audio playback.

  Logic pre-buffers playback of audio and MIDI data when this feature is turned on, aligning audio and MIDI playback timing anomalies that can potentially occur with some audio and MIDI hardware combinations. Latency figures reported by connected audio and MIDI hardware are used to determine the buffer amount. In general, you should not need to turn on this feature.

- *Send Program Change checkbox:* When selected, sends the MIDI program change (and bank select, if applicable) message, and sends these messages when the project, patch, or preset is loaded.

  Note: Program/bank change messages are stored when you save a plug-in setting and are automatically sent when you load the setting.

- *Program pop-up menu:* Choose a MIDI program change number.
- *Bank LSB pop-up menu:* Choose a MIDI bank LSB (Least Significant Byte) value.
- *Bank MSB pop-up menu:* Choose a MIDI bank MSB (Most Significant Byte) value.

  The Bank LSB and MSB messages can be combined to access presets (patches) in MIDI devices that offer more than 128 presets. Some devices only require the MSB message and a program change message to access presets above 128. Other devices may require the use of all three values. Consult the documentation for your device to determine the required values for these parameters.

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.

   Tip: You can create one or more External Instrument tracks in the New tracks dialog.

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 effect slots of the channel strip (or channel strips, if you are using multiple External Instrument instances with a multi-timbral sound source).

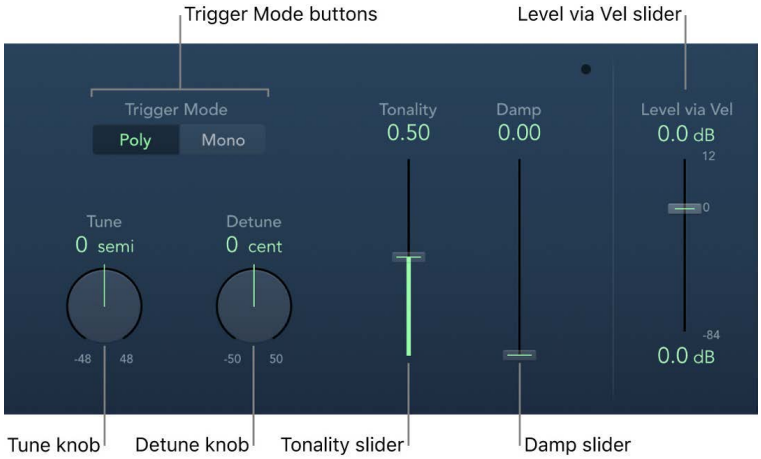
   You can also route the instrument channel strip to aux channel strips, if you want to use send effects.

Klopfgeist in Logic Pro for Mac

Klopfgeist is an instrument that provides a metronome click. It is inserted in the channel strip adjacent to the highest numbered channel strip by default.

Klopfgeist can also be inserted in any instrument channel strip for use as an instrument.

If you're new to using plug-ins in Logic Pro, see Add, remove, move, and copy plug-ins.



Klopfgeist has a number of synthesizer parameters that you can use to quickly create metronome click sounds.

Klopfgeist parameters

- *Trigger Mode buttons:* Activate Poly to use Klopfgeist as a four-voice polyphonic instrument. Activate Mono to use it as a monophonic instrument.
- *Tune knob and field:* Tune Klopfgeist in semitone steps.
- *Detune knob and field:* Rotate to fine-tune in cents (one cent equals 1/100 of a semitone).
- *Tonality slider and field:* Change the sound from a short click to a pitched percussion sound, similar to a wood block or claves.
- *Damp slider and field:* Set the release time. The shortest release time is attained when Damp is at its maximum value.
- *Level via Vel slider and fields:* 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.

Vintage B3 Organ

Vintage B3 Organ in Logic Pro for Mac

Vintage B3 Organ (abbreviated to Vintage B3 in the guide) 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. The Vintage B3 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 enormous level of crosstalk and the scratchiness of the key contacts. You can adjust the intensity of these characteristics, generating 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.

If you're new to using plug-ins in Logic Pro, see Add, remove, move, and copy plug-ins.

The Vintage B3 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.



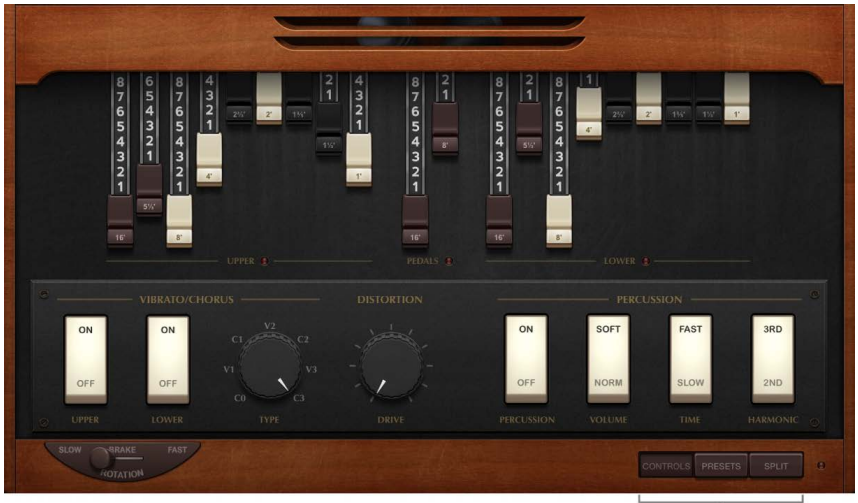
- *Main button:* Shows the draw bars, which are used to make changes to the basic organ sound in real time. Additional performance and setup controls can be accessed with the Control, Preset, and Split buttons in the lower-right corner. See Vintage B3 Main window overview.
- *Rotor Cabinet button:* Shows the Leslie speaker cabinet model and control parameters. See Vintage B3 Rotor Cabinet window.
- *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 in Logic Pro for Mac

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.



- 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 in Logic Pro for Mac



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.

Drag down the draw bars to make the selected sine choirs louder. You should note that 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. 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 B3 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. In controls view, choose “all the way down” from the Bass pop-up menu to simulate the Model A.

Vintage B3 Scanner Vibrato and Chorus in Logic Pro for Mac

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 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 (Main window)

- *Upper/Lower switches:* 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 scanner vibrato settings.
- *Type knob and field:* 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.

Scanner Vibrato and Chorus parameters (Options window)

- *Mode switch:* Set to Model to approximate the sound of the scanner vibrato. Set to Real to simulate the original circuitry, comprised of a delay line and a multi-section capacitor, which blends between delay line taps.
- *Rate knob and field:* Set the vibrato or chorus speed.
- *Depth knob and field:* Mix the dry signal with the chorus signal. This parameter is active only when a chorus setting is engaged (C1, C2, or C3).

Vintage B3 Percussion effect in Logic Pro for Mac

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 (Main window)

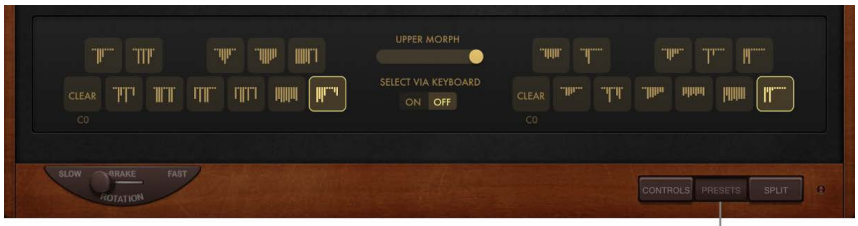
- *Percussion switch:* Activate percussion for the upper manual.
- *Harmonic switch:* Determine which harmonic is heard (the button toggles between the 2nd and 3rd harmonic).
- *Time switch:* Switch between a slow or fast decay. The time is set with the Time knobs in the Options window.
- *Volume switch:* Switch between a low or high decay level. The level is set with the Volume knobs in the Options window.

Percussion effect parameters (Options window)

- *Mode switch:* Mono triggers the percussion effect with each key press. Poly engages a polyphonic percussion effect that only (re)triggers after all keys are 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.
- *Perc on Preset switch:* Set to B0 to simulate the B preset key restriction. Choose All if you want percussion to always be available.
- *Time knobs and fields:* Set independent percussion decay times for the slow and fast Time switch settings. No percussion decay occurs when Slow is set to maximum.
- *Volume knobs and fields:* Set independent low and high decay levels 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 and field:* 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 and field:* 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 in Logic Pro for Mac

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). Vintage B3 replicates these features.



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. Vertical lines on each key indicate draw bar positions. 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 Use Vintage B3 Morph controls.

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.

The 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 Logic Pro 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 Use a single-channel controller with Vintage B3.

Choose a registration

1. In Logic Pro, 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. In Logic Pro, click Main in the control bar, then click the Preset button at the lower right.
2. Click the lowest preset key (shown as “C”) for the upper or lower manual. The other 11 keys, from C# to B, recall registrations for the upper or lower manuals.
3. Play MIDI note number 24.

Switch Vintage B3 registrations while playing (organ gate effect)

1. In Logic Pro, 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.

Turn off MIDI preset key switching

Turn off preset switching with MIDI notes 24 to 35, thus eliminating problems that may arise from transpositions.

1. In Logic Pro, 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. Use the “draw bar affects” feature to prepare a new registration while playing, then switch to the new registration later.

1. In Logic Pro, 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 in Logic Pro for Mac

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 with Vintage B3 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 with Vintage B3 for more information.

Vintage B3 also emulates B3 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 Set a Vintage B3 MIDI control mode.

Use multiple or multichannel controllers with Vintage B3 in Logic Pro for Mac

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 Set 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. In Logic Pro, 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. In Logic Pro, 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. In Logic Pro, 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 Logic Pro. 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. In Logic Pro, 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 with Vintage B3 in Logic Pro for Mac

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. In Logic Pro, 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. In Logic Pro, 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. In Logic Pro, 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. In Logic Pro, 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 Logic Pro. 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 in Logic Pro for Mac

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.

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 Vintage B3 Cabinet parameters, Vintage B3 Motor parameters, and Vintage B3 Brake parameters.

For information about microphone parameters, see Vintage B3 Microphone parameters.



Basic rotor speaker controls

Basic rotor speaker parameter

This parameter is shown at the lower left in the Main and Rotor Cabinet windows.

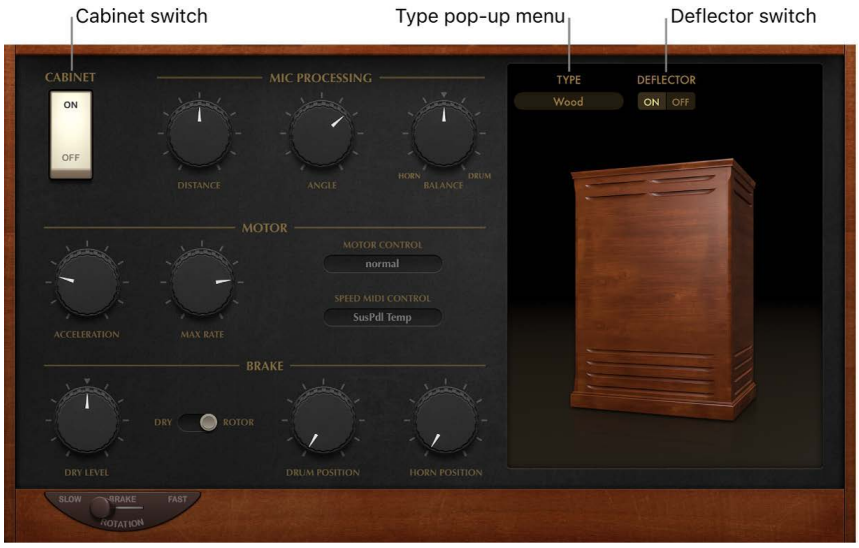
- *Rotation switch:* Switch the rotor speed between Slow, Brake, and Fast modes.
  - *Slow:* Slow rotor movement.
  - *Brake:* Stops the rotor.
  - *Fast:* Fast rotor movement.

  Note: When using a pedal to control rotor speed, you can hold down the pedal for a second or so, then release it to activate braking. Repeat to switch to the previously active speed mode: Fast or Slow.

Vintage B3 Cabinet parameters in Logic Pro for Mac

The advanced cabinet parameters are divided into three groups: Cabinet, Motor, and Brake. Also see Vintage B3 Motor parameters and Vintage B3 Brake parameters.

The microphone parameters are described in Vintage B3 Microphone parameters.



Cabinet parameters

- *Cabinet switch:* Turn 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 signal is routed slightly to the left, and the treble rotor 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:* Emulate 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.

Vintage B3 Motor parameters in Logic Pro for Mac

The advanced cabinet parameters are divided into three groups: Cabinet, Motor, and Brake. See also Vintage B3 Cabinet parameters and Vintage B3 Brake parameters.

The microphone parameters are described in Vintage B3 Microphone parameters.



Motor parameters

- *Acceleration knob and field:* Set the time required for the rotors to attain the speed set with the Max Rate knob and also the time required 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 and field:* Set the maximum possible rotor speed.
- *Motor Control pop-up menu:* Choose different speeds for the bass and treble rotors. Use the Rotation switch to choose slow, brake, or fast mode. See Vintage B3 Rotor Cabinet window.
  - *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.

  Note: When using a pedal to control rotor speed, you can hold down the pedal for a second or so, then release it to activate braking. Repeat to switch to the previously active speed mode: Fast or Slow.

  - *Modwheel:* Assigns the modulation wheel to switch between all three speed settings. Brake is selected around the modulation wheel center position, slow is selected in the lower third, and fast in the upper third of the modulation wheel range.
  - *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.

Vintage B3 Brake parameters in Logic Pro for Mac

The advanced cabinet parameters are divided into three groups: Cabinet, Motor, and Brake. See also Vintage B3 Cabinet parameters and Vintage B3 Motor parameters.

The microphone parameters are described in Vintage B3 Microphone parameters.



Brake parameters

- *Dry Level knob and field:* Set the level of the dry signal. This can also be useful if Dry is active for the Output switch.
- *Output switch:* Choose either switch position to modify the 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 and fields:* Set an exact stop position for the Leslie horn or drum (bass) rotator. 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 in Logic Pro for Mac

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 Vintage B3 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 in Logic Pro for Mac

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 and field:* Define the stereo width of the Horn deflector microphone.
  - *Drum knob and field:* Define the stereo width of the Drum deflector microphone.
- *When other cabinets are chosen in the Type pop-up menu:*
  - *Distance knob and field:* 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 and field:* Define the stereo image by changing the angle of the simulated microphones between 0 and 180 degrees.
- *Balance knob and field:* Set the balance between the horn and drum microphone signals.

Vintage B3 Options window

Vintage B3 Options window in Logic Pro for Mac

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 Percussion effect, Scanner Vibrato and Chorus, and Vintage B3 Morph parameter.

The Edit Preset Key parameter is discussed in Switch Vintage B3 registrations with a two-draw bar controller.

Vintage B3 Master and Click controls in Logic Pro for Mac



Master parameters

- *Tune knob and field:* Change Vintage B3 tuning in cents. A cent is 1/100 of a semitone. At a value of 0 c (zero cents), the central A key is tuned to 440 Hz, or concert pitch.
- *Volume knob and field:* 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 and field:* 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/Off knobs and fields:* Set the level of the key click sound heard during note on or note off messages.
- *Pedal knob and field:* Set the level of the key click sound heard during note on and note off messages for the pedal register.
- *Velocity knob and field:* Set the velocity sensitivity of the click parameters.

Vintage B3 Morph parameters in Logic Pro for Mac

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 (Main window)

- *Upper Morph slider:* 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:* 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 bypass the morph range that is set up.

Morph parameters (Options window)

- *Mode switch:* Click Step to abruptly switch, or click Linear to smoothly morph (crossfade), between presets.
- *Range pop-up menu:* Choose a range of keys for morph presets.
- *MIDI Controller pop-up menu:* Assign 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 MIDI message.

Use Vintage B3 Morph controls in Logic Pro for Mac

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. In Logic Pro, 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. In Logic Pro, 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.

- In Logic Pro, 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 in Logic Pro for Mac

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 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

In Logic Pro, do one of the following:

- Use the Master switch to turn on or turn off the entire Vintage B3 effects section.
- Use the On/Off switches to independently activate or deactivate the Reverb, EQ, Wah, and Distortion effects.

Change the routing of the EQ, Wah, and Distortion effects.

- In Logic Pro, 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

- In Logic Pro: Set the Pedal switch to Bypass. 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 controls in Logic Pro for Mac

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 and field:* Adjust the overall EQ level.
- *Low knob and field:* Adjust the level of the low frequency range.
- *Mid knob and field:* Adjust the level of the mid frequency range.
- *High knob and field:* Adjust the level of the high frequency range.

Vintage B3 Wah effect in Logic Pro for Mac

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:* 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 and field:* Determine the sensitivity of the Wah effect to incoming MIDI controller data.
- *Bite knob and field:* Boost the levels of signals surrounding the cutoff frequency. Bite is effectively a filter resonance parameter, where high values make the Wah effect sound more aggressive.

Use an expression pedal to control the Wah effect

1. In Logic Pro, 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 allows control of Wah 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. In Logic Pro, 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 in Logic Pro for Mac

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 and field:* 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 and field:* 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 in Logic Pro for Mac

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:* Patch 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 and field:* Set the balance between the reverb and original signal levels.

Vintage B3 Expert window

Vintage B3 Expert window in Logic Pro for Mac

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, Sustain controls, Condition controls, and Organ Model controls.

See Set 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 and Tonewheel sound generation.



Vintage B3 Pitch controls in Logic Pro for Mac

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:* 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:* 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:* 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:* Independently set the upward/downward pitch bend 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 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 in Logic Pro for Mac

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:* Control the sustain (release) phase of the upper register.
- *Lower Manual slider:* Control the sustain (release) phase of the lower register.
- *Pedals slider:* 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 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 in Logic Pro for Mac

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 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:* 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) random click color variations.

- *Filter Age slider:* Set the center frequencies of the filters to emulate aging capacitors. The high frequency output signals of B3 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:* Add a “breathy” quality to your sound—the result of crosstalk between all tonewheels—including the tonewheels of notes that you don’t play.

- *Drawbar Leak slider:* 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:* 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. Because crosstalk is audible only on certain B3 tonewheels, any “rumble” when chords are played is avoided.

- *Random FM slider:* 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 in Logic Pro for Mac

The Organ Model controls change the basic tonal quality.

Organ Model parameters

- *Maximum Wheels slider:* Set the number of tonewheels that are emulated. Reduce the value to minimize processing load. Reducing the value diminishes some overtones, so keep the number high if you're after an ultra-realistic simulation.
- *Tonal Balance slider:* Change the mix relationship of the higher and lower tonewheels. Use positive values for a lighter and brighter sound. Experiment with different tonal balance and equalizer settings. See Vintage B3 EQ controls, for further information.
- *Lower Manual Volume slider:* Set the relative level between the upper and lower manuals (and the pedalboard).
- *Pedals Volume slider:* Set the relative level between the upper (and lower) manual and the pedalboard.
- *Shape slider:* Alter the waveforms of the tonewheel generator to produce sounds that resemble the tones of Farfisa, Solina, or Yamaha organs. The Hammond 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:* Adjust to suppress the treble of the pedal draw bars which can sound bright within the context of the combined upper/lower/pedal sound. At the maximum position you hear only a solid bass organ fundamental in the bass register.
- *Extended bass switch:* 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, are not available on the original B3.
- *Ultrabass switch:* Turn on to prevent duplication of the 16″ drawbar in the lowest octave. Turn off to mirror the original B3 behavior. The on position results in a sound that resembles that of early tone wheel organs like the B, BV, and BCV, which had no dedicated tone wheels with a complex waveform for the pedal. This is also known as "Bass all the way down". See Use multiple or multichannel controllers with Vintage B3.

  Note: Ultrabass and Extended Bass can be combined, but Ultrabass has no effect on the added lowest octave. This octave always has a duplication of the 16″ drawbar tone.

Use a MIDI controller with Vintage B3

Set a Vintage B3 MIDI control mode in Logic Pro for Mac

Choose a supported external MIDI controller to use with Vintage B3.

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. In Logic Pro Vintage B3, 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 Roland VK or Korg CX mode if you use a Roland VK series or Korg CX-3 draw bar organ as a remote controller for Vintage B3.
   - Choose Hammond Suzuki mode if you use a Hammond XB series organ as a remote controller for Vintage B3.
   - Choose Native Instruments B4D mode if you use a Native Instruments B4D remote controller for Vintage B3.
   - Choose Nord Electro mode if you use a Clavia Nord Electro 2 as a remote controller for Vintage B3.

Vintage B3 Roland VK or Korg CX mode in Logic Pro for Mac

The table outlines 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′

Controller number	MIDI mode VK or CX: assigned parameter
Rotor Cabinet	
80, 92	Slow/Brake/Fast
81	Slow/Brake
Reverb	
82	Reverb Level
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

Controller number	MIDI mode VK or CX: assigned parameter
117	Pedal Volume
Rotor Fast Rate	
118	Rotor Fast Rate

Vintage B3 Hammond Suzuki mode in Logic Pro for Mac

The table outlines 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 Native Instruments B4D mode in Logic Pro for Mac

The table outlines 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

Controller number	MIDI mode Native Instruments B4D: assigned parameter
Equalizer	
90	EQ Low
70	EQ Mid
5	EQ High
Distortion/Click	
76	Distortion Drive
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 Nord Electro mode in Logic Pro for Mac

The table outlines 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′

Controller number	MIDI mode Nord Electro: assigned parameter
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

Vintage B3 Crumar mode in Logic Pro for Mac

The table outlines MIDI Control Change Message number assignment when MIDI mode is set to Crumar. This setting matches the controller mapping of the Crumar Mojo Classic.

Controller number	MIDI mode Crumar: assigned parameter
12 to 20	Upper draw bars 1 to 9
21 to 29	Lower draw bars 1 to 9
33 and 35	Pedal 1 and 2
55	Pedals to Lower
56	Pedalboard Release Time
Chorus/Vibrato	
31	Upper Vibrato on/off
30	Lower Vibrato on/off
73	Vibrato mode (selection goes from V1 to C3, C0 is excluded)
Percussion	
66	Percussion on/off
70 and 71	Percussion Volume (soft/normal) and Time (short/long)
89	Percussion Level
72	Percussion Harmonic (2nd/3rd)
Equalizer	
10	EQ High
9	EQ Mid
8	EQ Low
Reverb	
91	Reverb Amount
Distortion/Click	
76	Distortion Drive
75	Key Click
Leakage/Crosstalk	
86	Leakage
87	Crosstalk
88	Crosstalk Shape
Leslie	
85	on/off

Controller number	MIDI mode Crumar: assigned parameter
1	Leslie Speed
68	Controls Brake function
Speaker/Mic	
90	Balance
93	Mic. Distance
94	Horn EQ

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 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. Humans 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 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.

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

Vintage Clav overview in Logic Pro for Mac

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, which allows 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.

If you're new to using plug-ins in Logic Pro, see Add, remove, move, and copy plug-ins.

Vintage Clav interface in Logic Pro for Mac

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 has 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.
- *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.
- *Extended parameters:* Click the disclosure arrow at the lower left to access Vintage Clav extended parameters.

Vintage Clav Main window

Vintage Clav Main window in Logic Pro for Mac

The Main window provides access to the most commonly used parameters.



Main window parameters

- *Pickup Position display:* The two dark bars 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 and field:* 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 and field:* 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:* Make the sound nasal with a strong bass cut.
  - *Treble:* Make the sound sharper with a gentle bass cut.
  - *Medium:* Make the sound thinner with a slight bass reduction.
  - *Soft:* Make the sound softer or more muted.
- *Pickup switches:* Use the AB and CD switches to alter the “wiring” of the virtual pickups, thus changing the tone. See Use Vintage Clav Pickup parameters.
- *Volume knob and field:* Set the overall output level.

  Note: MIDI controller 11 scales the output level—unless it is assigned to control Wah or Damper parameters.
- *Damper slider and field:* 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 in Logic Pro for Mac

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.

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 sound of the original instrument but not the mechanical timbre jumps, try the Mellotone model.

Vintage Clav model characteristics in Logic Pro for Mac

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 Pickups in Logic Pro for Mac

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 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

- In Logic Pro, drag the handle at either end of a pickup to another position.

Reposition a pickup

- In Logic Pro, drag the handle in the middle of the pickup to a new position along the strings.

Change the Vintage Clav pickup mode

- In Logic Pro, 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 in Logic Pro for Mac

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.



Drag the Key parameter to set a key scale modulation of the panning position. In other words, the played keyboard note position determines the panning position.

Use the Pickup parameter to spread the two pickup signals across the stereo spectrum when both pickups are active (Upper+Lower or Upper-Lower 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.

Set panning by keyboard position

- In Logic Pro, 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

- In Logic Pro, 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 further to the right and the other further to the left.

  Set this parameter to the maximum value for extreme left/right panning.

Vintage Clav Effects window

Vintage Clav Effects window in Logic Pro for Mac

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.

Use 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 in Logic Pro for Mac

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 turn it off completely.



Compressor effect parameters

- *On/off button:* Turn the Compressor effect on or off.
- *Ratio knob and field:* 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 in Logic Pro for Mac

The Distortion effect can provide warm overdrive or aggressive distortion.



Distortion effect parameters

- *On/off button:* Turn the Distortion effect on or off.
- *Gain knob and field:* Set the level of the Distortion effect. If Gain is at the minimum value, no distortion is heard.
- *Tone knob and field:* 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 in Logic Pro for Mac

Vintage Clav features a choice of three modulation effect types Phaser, Flanger, or Chorus.



Modulation effect parameters

- *On/off button:* Turn the Modulation effect on or off.
- *Mode pop-up menu:* Choose Phaser, Flanger, or Chorus as the modulation effect.
- *Intensity knob and field:* 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:* Synchronize the Phaser or Flanger effect to the host application tempo. The Rate knob sets bar and beat values, including triplets.
- *Rate knob and field:* 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 in Logic Pro for Mac

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 Misc parameters.



Wah effect parameters

- *On/off button:* Turn 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 and field:* Set the cutoff frequency of the filter. At the extreme left position, Range limits cutoff modulation to a narrow frequency range. Turn the Range knob to the right to set a wider control range.
- *Envelope knob and field:* 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 in Logic Pro for Mac

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 in Logic Pro for Mac

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 and field:* 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 and field:* 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 and field:* Control the amount of click level variance across the keyboard. This slider simulates the wearing of some hammers, but not others, emulating the “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 and field:* 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:* Turn 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 in Logic Pro for Mac

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 and field:* 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 and field:* 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/Inharmonicity sliders and fields:* Intensify or reduce inharmonic overtones. Different values let you create metallic, bell-like sounds or DX-style electric pianos. 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 and field:* Add a slight upward pitch bend effect immediately after a string is 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, allowing 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 and field:* 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 in Logic Pro for Mac

The Pitch parameters affect the tuning of the selected model.



Pitch parameters

- *Tune slider and field:* Adjust tuning in one-cent intervals. At a value of 0 c (zero cents), the central A key is tuned to 440 Hz, or concert pitch.
- *Stretch Tuning slider and field:* Use to deviate from the default equal-tempered scale by altering 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 and field:* 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 and field:* Determine the pitch bend range in semitone steps. Use your keyboard pitch bend wheel to control pitch bends.
- *Pitch Pressure slider and field:* Adjust aftertouch sensitivity. On an original D6, applying pressure to a depressed key raises the pitch slightly. Pitch Pressure emulates this behavior. 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 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 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 in Logic Pro for Mac

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.



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 prevents 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 prevents 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) 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 in Logic Pro for Mac

Vintage Clav provides two parameters that can be accessed by clicking the disclosure arrow 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 Bend Range slider:* Set a value from 0 to 96.

  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 the GarageBand for iOS 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 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 resulted in limited keyboard dynamics and also damped all reeds 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 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.

Vintage Electric Piano

Vintage Electric Piano overview in Logic Pro for Mac

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 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.

The Vintage Electric Piano sound engine uses component modeling synthesis techniques to generate ultra-realistic 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.

The integrated effects include classic equalizer, overdrive, stereo phaser, stereo tremolo, and stereo chorus effects that are commonly used with electric piano sounds.

If you're new to using plug-ins in Logic Pro, see Add, remove, move, and copy plug-ins.

Vintage Electric Piano interface in Logic Pro for Mac

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 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 allow you to alter the tone and playing behavior of the selected instrument model.

- *Bass Boost knob and field:* Enhance the low end of the sound. This parameter emulates the behavior of the control found on the original Rhodes piano.

- *Volume knob and field:* Set the overall output level of Vintage Electric Piano.

- *Extended parameters:* Click the disclosure arrow at the lower left to access Vintage Electric Piano extended parameters if needed.

Vintage Electric Piano Effects window

Vintage Electric Piano EQ controls in Logic Pro for Mac

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:* Turn the equalizer on or off.
- *Bass knob and field:* 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 and field:* 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 in Logic Pro for Mac

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:* Turn the Drive effect on or off.
- *Drive mode switch:* Switch between two types of distortion effect.
- *Gain knob and field:* Set the amount of harmonic distortion.
- *Tone knob and field:* Equalize the sound before amplification or distortion 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 in Logic Pro for Mac

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:* Turn the Chorus effect on or off.
- *Rate knob and field:* Set the speed of the Chorus effect, in Hz. High values may result in the piano sounding detuned.
- *Intensity knob and field:* Set the intensity of the Chorus effect. Technically, this sets the amount of delay time deviation.

Vintage Electric Piano Phaser in Logic Pro for Mac

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:* Turn the Phaser effect on or off.
- *Rate knob and field:* 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:* Synchronize the Phaser effect to the host application tempo. The Rate knob sets bar and beat values, including triplets.
- *Color knob and field:* Set the amount of Phaser output signal that is fed back to the effect input. This changes the tonal color of the phasing effect.
- *Stereo knob and field:* 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 in Logic Pro for Mac

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:* Turn the Tremolo effect on or off.
- *Rate knob and field:* 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:* Synchronize the Tremolo effect to the host application tempo. The Rate knob sets bar and beat values, including triplets.
- *Intensity knob and field:* Set the amount of amplitude modulation.
- *Stereo knob and field:* 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 controls in Logic Pro for Mac

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 and field:* 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 and field:* 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 and field:* 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 and field:* Adjust the 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 and field:* Set the level of the (inharmonic) treble portion of the tone. This is useful for emulating classic electric piano sounds.
- *Damper Noise knob and field:* Set the level of damper noise. This emulates the damping felt hitting the vibrating tine in the original instruments.

Vintage Electric Piano pitch controls in Logic Pro for Mac

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 and field:* Tune Vintage Electric Piano in one-cent increments. At a value of 0 c (zero cents), the central A key is tuned to 440 Hz, or concert pitch. The range is plus or minus half a semitone.
- *Down/Up knobs and fields:* Set the pitch bend range in semitone steps.
- *Warmth knob and field:* 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.

  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 knob and field:* 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 knob and field:* 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

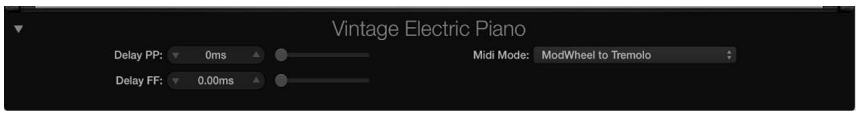
The tones of upright pianos, and to a lesser extent grand pianos (due to their longer strings), have inharmonicities in their harmonic structure. This also applies to other stringed instruments, but it particularly affects pianos due to the length, density, and tension of the strings. If a piano is perfectly tuned to equal temperament across the keyboard range, the overtones of the low strings and the fundamentals of the high strings 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 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 in Logic Pro for Mac

Click the disclosure arrow 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. 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 a pre-amplifier and two-channel combo amplifier) and as a stage piano, without an 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 and more versatile 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 goals that Rhodes technicians may have envisaged when preparing their keyboards.

Hohner and Wurlitzer models

The extremely rare Hohner Electra Piano (not the similarly named all-electronic Electrapiano from RMI) 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 zero 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 useful sound nonetheless.

Vintage Electric Piano MIDI controllers in Logic Pro for Mac

Vintage Electric Piano responds to the MIDI continuous controller numbers (CC) outlined in the table.

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

Vintage Mellotron

Vintage Mellotron in Logic Pro for Mac

Vintage Mellotron recreates the features and a number of sounds from this famous keyboard instrument that has featured on dozens of classic rock and pop songs from the 1960s, ‘70s, and beyond.

The Mellotron is considered a precursor to modern sample-playback instruments. It uses a number of magnetic tapes that are drawn across a tape head when a key is pressed. Each key has an individual tape playhead and pressure pads on a spring-loaded mechanism. Rollers for each key are driven by a constantly turning capstan that spans the keyboard range.

Each key has a tape strip with up to three different sounds running in parallel. Sound length is limited to eight seconds, at which point the sound abruptly stops. Tapes return to their start position when the corresponding key is released. By offsetting the playheads with the racks that hold the tapes, a musician can switch the entire keyboard between a string and choir sound, for example. Partial offsets of the tape heads result in a layered blend of two adjacent sounds on each tape strip.

More advanced Mellotron models can use longer tapes, with different sounds allocated at precise positions along their length. This is similar to switching between banks of presets on a modern synth. Even then, a maximum of around 24 sounds is possible. If you require different sounds, the machine needs to be dismantled, and a new tape rack is used to replace one already in place. Not ideal, and certainly tough to accomplish during a live performance.

The original library sounds were recorded note by note, with varied performances and pitch fluctuations. This makes Mellotron instrument mapping somewhat inconsistent across the keyboard range, which is an essential part of its sonic character and charm.

The mechanical complexity of early Mellotrons—tape stretching, head alignment problems, and environmental factors, such as temperature and humidity fluctuations, smoke, dust, and magnetic fields—made them difficult to maintain. Even when new, no two Mellotrons sounded exactly the same because of these variables.

Vintage Mellotron features painstakingly sampled versions of each note from the original Mellotron sound library tapes, capturing the full sound length and performance quirks. Unlike the originals, Vintage Mellotron sounds are looped, allowing you to indefinitely sustain notes. Looping isn’t static, so sounds retain their “organic” flavor and mirror the continuous sonic movement of the original instruments.

Vintage Mellotron also allows you to blend any two sounds together, including combinations that weren't available in the original tape library. A further bonus is control of the balance between sounds, a feature not found on the original machines.

A tape speed control mimics the tonal fluctuations caused by this feature on the original instruments. Also included are octave transposition and tone controls. See Vintage Mellotron controls.

As with other instrument plug-ins, you can fully automate Vintage Mellotron parameters.

If you're new to using plug-ins in Logic Pro, see Add, remove, move, and copy plug-ins.

Vintage Mellotron controls in Logic Pro for Mac

Vintage Mellotron is divided into two sections. The upper section is used to choose one or two sounds, set the playback octave for each, and to adjust the mix between them. The lower section houses global controls for the entire instrument.



Vintage Mellotron parameters

- *Sound A/B pop-up menus:* Choose an A or B instrument sound.
- *Sound A/B Transpose buttons:* Set an independent playback octave for instrument sound A or B.

  This mimics the behavior of the half speed or double speed tape switches found on some Mellotron models, but enhances these facilities by allowing independent octave control for each sound.
- *Blend A/B knob:* Set the level balance between instrument sound A and B. Set to the full left or right position to hear sound A or B in isolation.
- *Tape Speed knob:* Set the tape speed for all notes. This mimics the tonal fluctuations caused by this control on the original instrument.
- *Tone knob:* Rotate to the right to reduce bass and to make the sound brighter and more nasal. Rotate to the left to reduce brightness, making the sound warmer and more mellow.
- *Volume knob:* Set the overall output level of Vintage Mellotron.

Vintage Mellotron extended parameters

The extended parameters provide three additional controls that were not available on the original instruments. These provide flexibility, but their use isn't true to the source hardware.

Click the disclosure arrow at the lower left to view the extended parameters.

- *Attack slider:* Set the time required for the signal to reach the initial signal level, known as the sustain level.
- *Release slider:* Set the time it takes for the signal to fall from the sustain level to a level of zero, after releasing a key.
- *Pitch Bend Range slider:* Set the pitch bend range in semitone steps. This allows you to use the pitch bend controller of your keyboard to bend Vintage Mellotron pitch.
- *Fixed Velocity checkbox:* Click to prevent your MIDI keyboard velocity data from affecting Vintage Mellotron dynamics. This mirrors the behavior of the original instruments that did not have velocity-sensitive keyboards.

Legacy instruments

Legacy instruments in Logic Pro for Mac

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 you open a GarageBand project, a Logic Pro project, or a MainStage concert that uses one or more of these instruments.

You can use these plug-ins or you can replace them with other instrument plug-ins available in Logic Pro.

You cannot directly insert these plug-ins in Logic Pro unless you override the Instruments plug-in menu.

If you're new to using plug-ins in Logic Pro, see Add, remove, move, and copy plug-ins.

Display and insert legacy instrument plug-ins in Logic Pro

1. In Logic Pro, press Option, then click the Instrument slot on a channel strip.

   The plug-in menu opens, with a Legacy submenu shown below Vintage Mellotron.

2. Choose the legacy instrument plug-in that you want to insert from the Legacy submenu.

Emulated instruments

Legacy Bass in Logic Pro for Mac

Bass emulates electric and acoustic basses.

If you're new to using plug-ins in Logic Pro, see Add, remove, move, and copy plug-ins.

- *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.

Legacy Church Organ in Logic Pro for Mac

Church Organ emulates a pipe organ.

If you're new to using plug-ins in Logic Pro, see Add, remove, move, and copy plug-ins.

- *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 turning specific pipes on or off, 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.

Legacy Drum Kits in Logic Pro for Mac

Drum Kits includes: rock, pop, jazz, electronic, orchestral, and Latin kits, among others.

If you're new to using plug-ins in Logic Pro, see Add, remove, move, and copy plug-ins.

- *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.

Legacy Electric Clav(inet) in Logic Pro for Mac

Electric Clavinet, which is based on Vintage Clav, emulates the Hohner D6 clavinet.

If you're new to using plug-ins in Logic Pro, see Add, remove, move, and copy plug-ins.

- *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.

Legacy Electric Piano in Logic Pro for Mac

Electric Piano, which is based on Vintage Electric Piano, sounds like the Fender Rhodes and Wurlitzer electric pianos.

If you're new to using plug-ins in Logic Pro, see Add, remove, move, and copy plug-ins.

- *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.

Legacy Guitar in Logic Pro for Mac

Guitar emulates a number of acoustic and electric guitar sounds.

If you're new to using plug-ins in Logic Pro, see Add, remove, move, and copy plug-ins.

- *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.

Legacy Horns in Logic Pro for Mac

Horns emulates several brass sections and a number of individual brass instruments.

If you're new to using plug-ins in Logic Pro, see Add, remove, move, and copy plug-ins.

- *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.

Legacy Piano in Logic Pro for Mac

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.

If you're new to using plug-ins in Logic Pro, see Add, remove, move, and copy plug-ins.

- *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.

Legacy Sound Effects in Logic Pro for Mac

Sound Effects provides a number of nature sounds, laughter, applause, and so on.

If you're new to using plug-ins in Logic Pro, see Add, remove, move, and copy plug-ins.

- *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.

Legacy Strings in Logic Pro for Mac

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.

If you're new to using plug-ins in Logic Pro, see Add, remove, move, and copy plug-ins.

- *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.

Legacy Tuned Percussion in Logic Pro for Mac

Tuned Percussion emulates a vibraphone, xylophone, timpani, steel drums, and other tuned percussion instruments.

If you're new to using plug-ins in Logic Pro, see Add, remove, move, and copy plug-ins.

- *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.

Legacy Voice in Logic Pro for Mac

Voice emulates a mixed choir.

If you're new to using plug-ins in Logic Pro, see Add, remove, move, and copy plug-ins.

- *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.

Legacy Woodwind in Logic Pro for Mac

Woodwind emulates the sound of wind instruments such as; flutes, clarinets, saxophones, and several other instruments from various world cultures.

If you're new to using plug-ins in Logic Pro, see Add, remove, move, and copy plug-ins.

- *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.

Legacy Tonewheel Organ in Logic Pro for Mac

Tonewheel Organ, which is based on Vintage B3, emulates the Hammond B3 organ but it can also emulate Farfisa, Wurlitzer, and other electric organs.

If you're new to using plug-ins in Logic Pro, see Add, remove, move, and copy plug-ins.

- *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

Legacy Analog Basic in Logic Pro for Mac

Analog Basic, which is based on the ES2, is a simple analog synthesizer that is useful for a range of musical styles.

If you're new to using plug-ins in Logic Pro, see Add, remove, move, and copy plug-ins.

- *Volume slider:* Set the overall volume level of the instrument.
- *Mix slider:* Determine the balance between the oscillator signals.
- *Tuning slider:* Set the overall pitch of the instrument.
- *Cutoff slider:* Allow less sound through at low values and more at high values, damping the sound or making it brighter.
- *Resonance slider:* Emphasize the frequency range around the point determined by the Cutoff parameter.
- *Attack slider:* Make 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:* Make 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:* Determine the level of the sound after the Attack or Decay phase has completed.

Legacy Analog Mono in Logic Pro for Mac

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.

If you're new to using plug-ins in Logic Pro, see Add, remove, move, and copy plug-ins.

- *Volume slider:* Set the overall volume level of the instrument.
- *Glide slider:* Determine the time it takes a note pitch to change, or slide, to another note pitch.
- *Mix slider:* Determine the balance between the oscillator signals.
- *Cutoff slider:* Allow less sound through at low values and more at high values, damping the sound or making it brighter.
- *Resonance slider:* Emphasize the frequency range around the point determined by the Cutoff parameter.
- *Attack slider:* Make 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:* Make 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:* Determine the complexity of the sound texture, making the sound fuller.

Legacy Analog Pad in Logic Pro for Mac

Analog Pad, which is based on the ES2, can generate warm analog synthesizer pad sounds that are useful for a range of musical styles.

If you're new to using plug-ins in Logic Pro, see Add, remove, move, and copy plug-ins.

- *Volume slider:* Set the overall volume level of the instrument.
- *Modulation slider:* Make the sweeping movement of the pad faster or slower.
- *Character slider:* Determine whether the sound is soft or sharp.
- *Cutoff slider:* Allow less sound through at low values and more at high values, damping the sound or making it brighter.
- *Resonance slider:* Emphasize the frequency range around the point determined by the Cutoff parameter.
- *Cutoff Envelope slider:* Determine the strength of the sweeping movement.
- *Duration slider:* Determine the duration of the sweeping movement.
- *Animation slider:* Determine the envelope effect on the pad sound.

Legacy Analog Swirl in Logic Pro for Mac

Analog Swirl, which is based on the ES2, can generate chorused, swirling analog synthesizer pad sounds.

If you're new to using plug-ins in Logic Pro, see Add, remove, move, and copy plug-ins.

- *Volume slider:* Set the overall volume level of the instrument.
- *Modulation slider:* Make the sweeping movement of the pad faster or slower.
- *Cutoff slider:* Allow less sound through at low values and more at high values, damping the sound or making it brighter.
- *Resonance slider:* Emphasize the frequency range around the point determined by the Cutoff parameter.
- *Attack slider:* Make 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:* Make 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:* Determine the level of the sound after the Attack or Decay phase has completed.
- *Release slider:* Determine the time it takes for notes to fade out after you let go of the keys on your keyboard.

Legacy Analog Sync in Logic Pro for Mac

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.

If you're new to using plug-ins in Logic Pro, see Add, remove, move, and copy plug-ins.

- *Volume slider:* Set the overall volume level of the instrument.
- *Sync slider:* Determine the synchronization (or lack of it) between the two oscillators, and therefore the harshness of the sound.
- *Sync Modulation slider:* Determine how much the synchronization of the two oscillators is modulated, resulting in more complex (and harder) tones.
- *Sync Envelope slider:* Determine the amount that envelope parameters affect the sound.
- *Cutoff slider:* Allow less sound through at low values and more at high values, damping the sound or making it brighter.
- *Attack slider:* Make 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:* Make 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:* Determine the level of the sound after the Attack or Decay phase has completed.

Legacy Digital Basic in Logic Pro for Mac

Digital Basic, which is based on the ES2, can generate simple digital synthesizer sounds that are useful for a range of musical styles.

If you're new to using plug-ins in Logic Pro, see Add, remove, move, and copy plug-ins.

- *Volume slider:* Set the overall volume level of the instrument.
- *Mix slider:* Mix two tones together.
- *Tuning slider:* Set the overall pitch of the instrument.
- *Harmonics slider:* Increase or decrease the number of harmonics, or overtones, in the sound. This can change the sound dramatically or subtly, so feel free to experiment.
- *Timbre slider:* Change the color of the sound from dark to bright.
- *Attack slider:* Make 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:* Make 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:* Determine the time it takes for notes to fade out after you let go of the keys on your keyboard.

Legacy Digital Mono in Logic Pro for Mac

Digital Mono, which is based on the ES2, is ideal for monophonic digital synthesizer lead sounds.

If you're new to using plug-ins in Logic Pro, see Add, remove, move, and copy plug-ins.

- *Volume slider:* Set the overall volume level of the instrument.
- *Tuning slider:* Set the overall pitch of the instrument.
- *Harmonics slider:* Increase or decrease 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:* Change the color of the sound from dark to bright.
- *Timbre Envelope slider:* Dynamically change 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:* Make 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 detune each played note from the others, making the sound a little thicker, particularly when *high* parameter values are used.
- *Distortion slider:* Distort 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.

Legacy Digital Stepper in Logic Pro for Mac

Digital Stepper, which is based on the ES2, is a digital synthesizer that can step through a number of tones, creating a rhythmic pattern.

If you're new to using plug-ins in Logic Pro, see Add, remove, move, and copy plug-ins.

- *Volume slider:* Set the overall volume level of the instrument.
- *Balance slider:* Set the balance between a harder and more spiky sound (digital) and a warmer, softer sound (analog).
- *Modulation slider:* Apply more or less modulation, making the sound more lively when high settings are used.
- *Harmonics slider:* Increase or decrease 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:* Determine how noticeable the tonal steps are. Large values make them more noticeable and small values less noticeable.
- *Cutoff slider:* Allow less sound through at low values and more at high values, damping the sound or making it brighter.
- *Cutoff Steps slider:* Set the amount of cutoff applied to each step. Large values make the cutoff effect more pronounced and small values less pronounced.
- *Duration slider:* Set the length of the steps.

Legacy External Instrument in Logic Pro for Mac

You can use External Instrument Legacy to route external MIDI sound generators through the Mixer, which you can then process with effects.

You can also use External Instrument Legacy to transmit and receive MIDI information through the instrument channel strip that it is inserted into. This lets you 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 plug-in is inserted into instrument channel strips in place of a software instrument.

If you're new to using plug-ins in Logic Pro, see Add, remove, move, and copy plug-ins.

Note: It is recommended that you swap External Instrument Legacy instances with the updated and more flexible External Instrument plug-in immediately after importing your project.

External Instrument parameters

- *Environment Destination pop-up menu:* Choose the target environment object (often a 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.

Legacy Hybrid Basic in Logic Pro for Mac

Hybrid Basic is a sample-based synthesizer that can create spectacular sounds.

If you're new to using plug-ins in Logic Pro, see Add, remove, move, and copy plug-ins.

- *Volume slider:* Set the overall volume level of the instrument.
- *Waveform pop-up menu:* Choose the sample set used to generate the basic synthesizer sound.
- *Glide slider:* Determine the time it takes a note pitch to change, or slide, to another note pitch.
- *Wheel to Vibrato slider:* Determine the amount of pitch modulation by your keyboard modulation wheel.
- *Wheel to Cutoff slider:* Determine the depth of Cutoff modulation by your keyboard modulation wheel.
- *Cutoff slider:* Allow 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:* Emphasize the frequency range around the point determined by the Cutoff parameter.
- *Cutoff Attack slider:* Determine the time it takes before the Cutoff parameter begins to affect the sound.
- *Attack slider:* Make 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:* Make 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:* Determine the level of the sound after the Attack or Decay phase has completed.
- *Release slider:* Determine the time it takes for notes to fade out after you let go of the keys on your keyboard.

Legacy Hybrid Morph in Logic Pro for Mac

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.

If you're new to using plug-ins in Logic Pro, see Add, remove, move, and copy plug-ins.

- *Volume slider:* Set the overall volume level of the instrument.
- *Waveform pop-up menu:* Choose the sample set used to generate the basic synthesizer sound.
- *Morph slider:* Control crossfades between the two sample layers.
- *Morph Envelope slider:* Control 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 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:* Allow 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:* Emphasize the frequency range around the point determined by the Cutoff parameter.
- *Cutoff Envelope slider:* Determine the strength of the envelope shaping applied to the Cutoff parameter.
- *Attack slider:* Make 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:* Make 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:* Determine the level of the sound after the Attack or Decay phase has completed.
- *Release slider:* Determine the time it takes for notes to fade out after you let go of the keys on your keyboard.

Synthesizer basics

Synthesizer basics overview

If you're 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. See Instruments introduction.

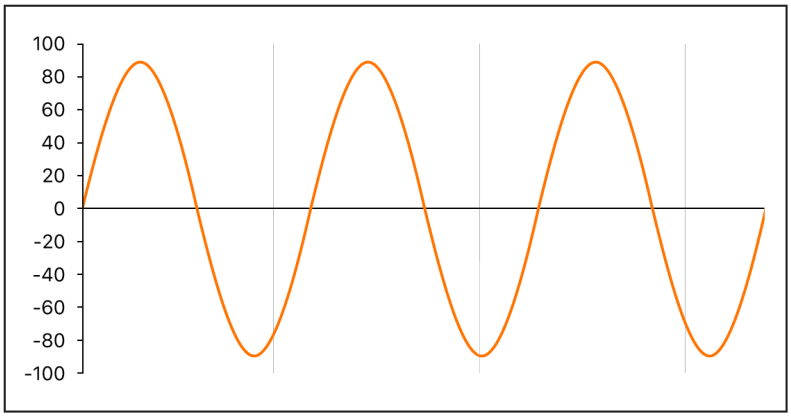
Sound basics

Sound basics overview

Prior to considering any of the sound-generating components found in a synthesizer, it's important that you understand sound itself.

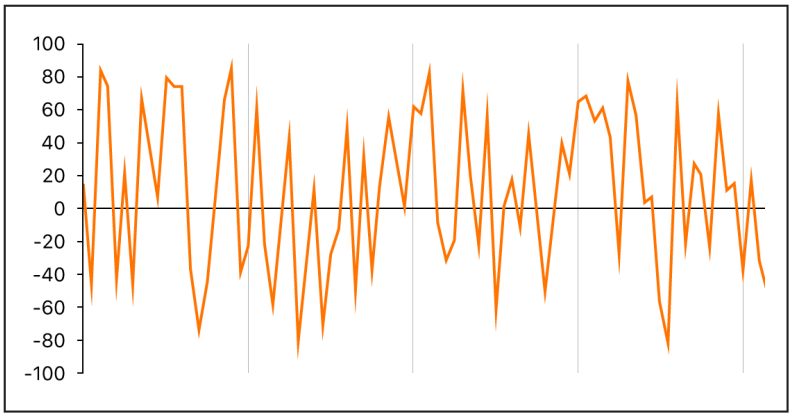
Technically, sound is the conversion of physical energy—such as a hand clap—to an air pressure disturbance. This change in air pressure is transmitted as a series of vibrations—a sound wave—through the air. Sound vibrations can also be transmitted through other matter, such as a wall or floor.

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.

Non-fundamental 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."

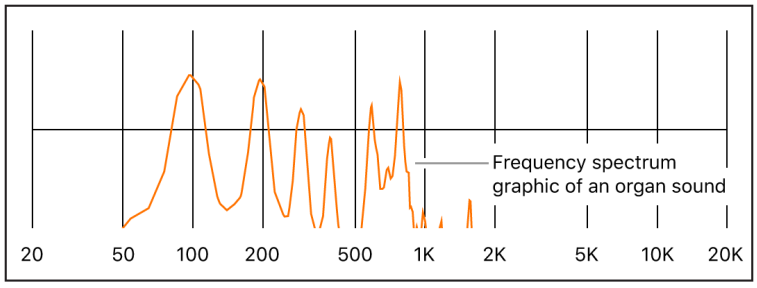
Tones that cannot be multiplied or divided by a whole number are known as *inharmonic overtones,* or *partial tones*. When you combine a number of these inharmonic overtones, it tends to sound "noisy."

Non-fundamental 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*.

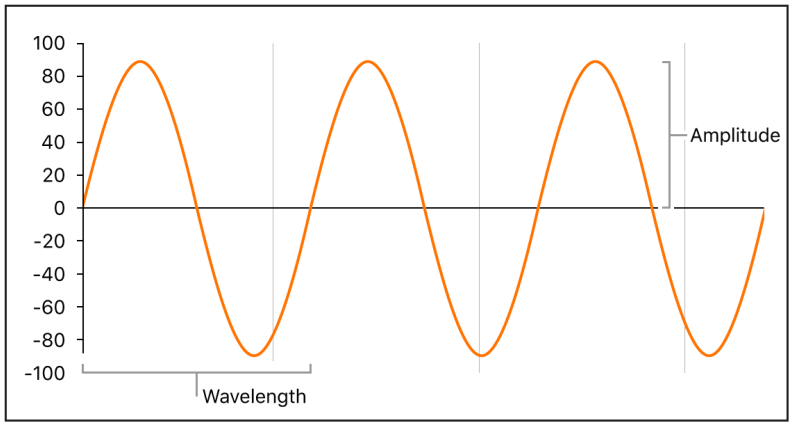
The frequency spectrum shows all individual sonic elements in a sound. It is shown low to high and runs from left to right over time. The respective levels of all harmonics are reflected vertically, with taller spikes indicating higher levels.



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*.

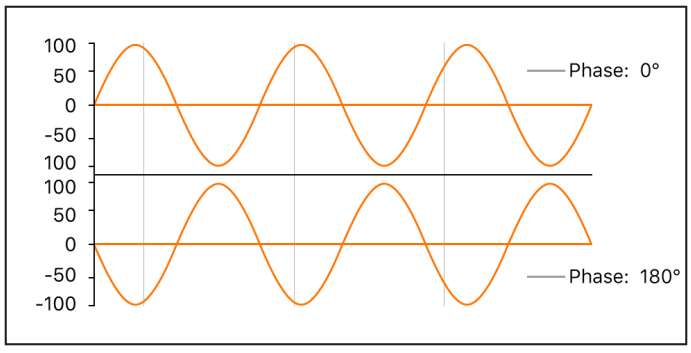


- *Amplitude:* The amplitude of a waveform indicates the amount of air pressure change. It can be measured as the maximum vertical distance from zero air pressure, or "silence" (shown as a horizontal line at 0 dB in the illustration). Put another way, amplitude is the distance between the horizontal 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 becomes 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.

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.

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. See ES1 overview.

Virtual analog synthesizers have other advantages over their analog counterparts as well. They are 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 multi-timbral, 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. 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.

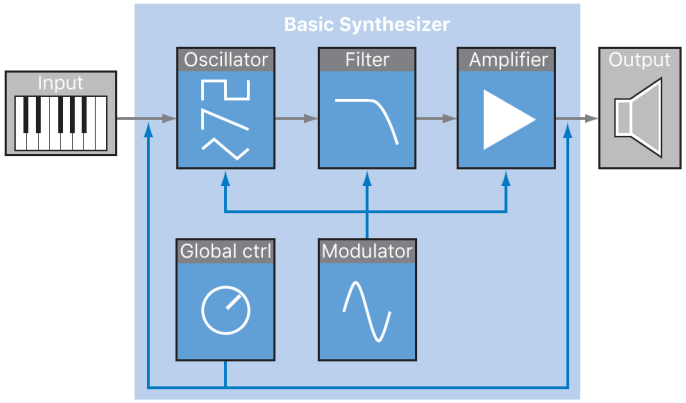
The distinctive timbre, intonation, and volume characteristics of a real instrument can theoretically be recreated by combining these components in a way that resembles the natural behavior of the instrument you are trying to emulate.

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 with sample playback instruments like Alchemy or Sampler, which are able to recreate real instruments far more convincingly by using multi-gigabyte 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 signal flow.



Signal-generating and processing components

- *Oscillators:* Generate the basic signal. This is usually a waveform that is rich in harmonics. See Oscillators. 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.
- *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.

Modulation and control components

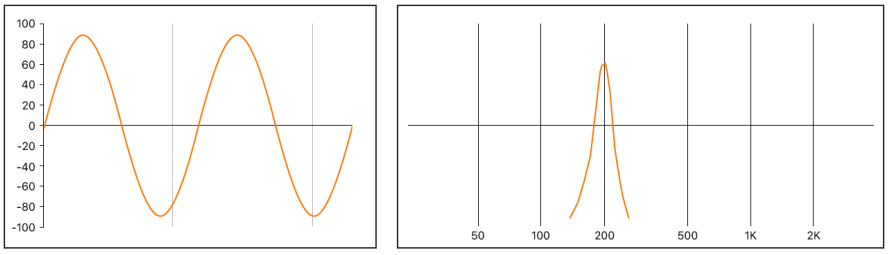
- *Modulators:* Used to modulate the signal-generating and processing components. Modulations can be machine-based—automatically generated by a synthesizer component—or can be manually activated by using the modulation wheel, for example. Most synthesizers have a component called an *LFO* (low frequency oscillator) to provide a waveform that modulates the signal. See Modulation overview.
- *Global controls:* Set the overall characteristics of your synthesizer sound, such as tuning, glides between notes, pitch bends, and monophonic or polyphonic playback. See Global controls.

Oscillators

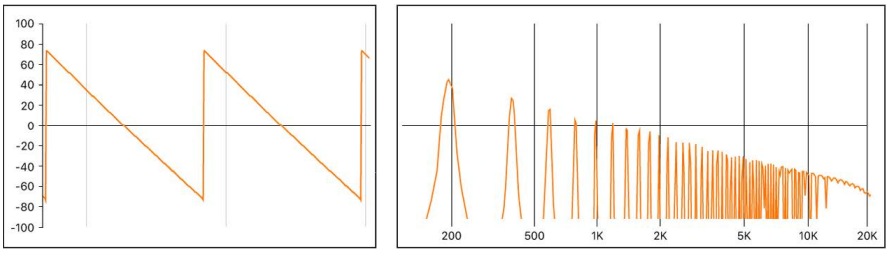
The audio signal of a synthesizer is generated by the oscillator. You can choose from a selection of waveforms that contain various types and amounts of harmonics. The level relationships between the fundamental tone and the harmonics of the chosen waveform are responsible for the basic sound color or timbre.

Waveform types

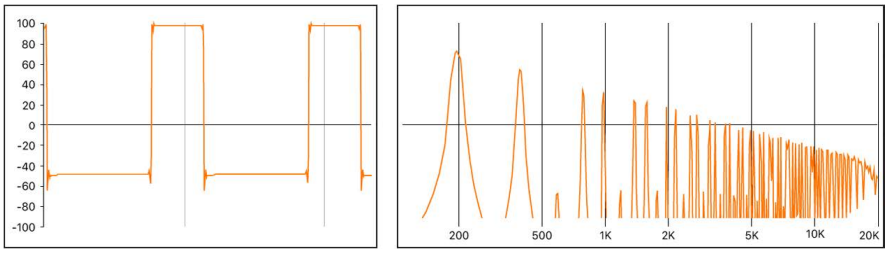
- *Sine wave:* Clean and clear-sounding, a sine wave contains only the first harmonic; in other words, it is the fundamental tone. The sine wave, used alone, can create "pure" sounds like whistles, the sound of wet fingers on the rim of a glass, tuning forks, and so on.



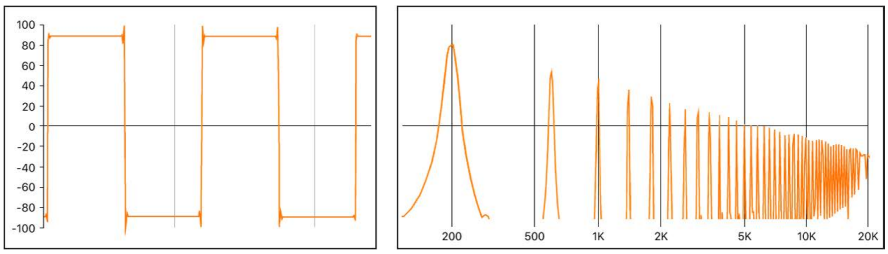
- *Sawtooth wave:* Clear and bright-sounding, a sawtooth wave contains both odd and even harmonics, as well as the fundamental tone. It is ideal for creating string, pad, bass, and brass sounds.



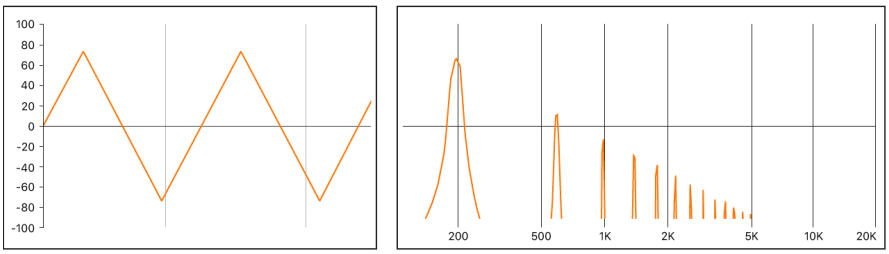
- *Square and pulse waves:* Hollow and woody-sounding, a square wave can contain a wide range of odd harmonics, as well as the fundamental tone. It is useful for creating reed instruments, pads, and basses. It can also be used to emulate kick drums, congas, tom-toms, and other percussive instruments—often 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.

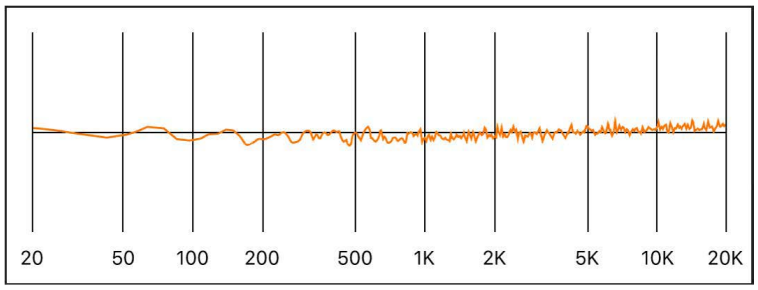


- *Triangle wave:* A triangle wave contains only odd harmonics, as well as the fundamental tone. The higher harmonics of the triangle wave roll off faster than those of a square wave, making the triangle wave sound softer. It is ideal for creating flute sounds, pads, and vocal “oohs.”

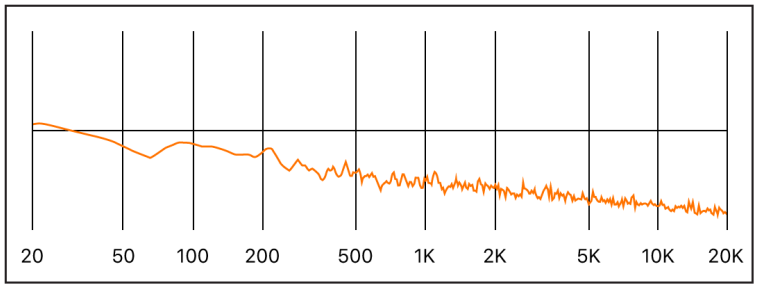


- *Noise: white, pink and red, blue:* Noise is useful for emulating percussive sounds, such as snare drums, or wind and surf sounds. There are more noise wave colors than those listed, but they are rarely found in synthesizers.

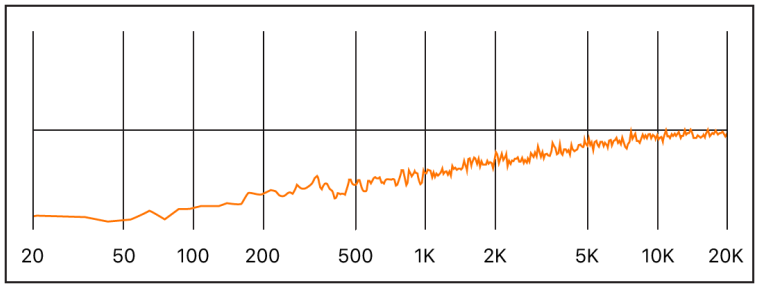
- *White noise:* The most common noise waveform found on synthesizers. White noise contains all frequencies—at full level—around a center frequency.



- *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

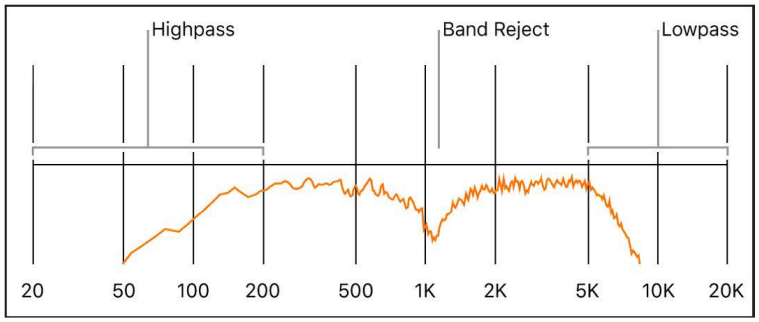
Filters overview

The purpose of the filter in a subtractive synthesizer is to remove portions of the signal—the frequency spectrum—sent from the oscillators. After filtering, 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. See Modulation overview.

Basic filter types

There are several basic filter types. Each has a different effect on various portions of the frequency spectrum.

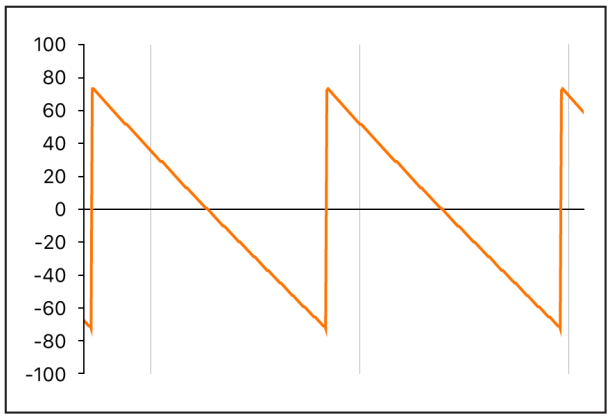


- *Highpass filter:* High frequencies are passed; low frequencies are attenuated.
- *Lowpass filter:* Low frequencies are passed; high frequencies are attenuated.
- *Bandpass filter:* Only frequencies within a frequency band are passed.
- *Band reject filter:* Only frequencies within a frequency band are attenuated. This filter type is also known as a *notch* filter.
- *Allpass filter:* All frequencies in the spectrum are passed, but the phase of the output is modified.

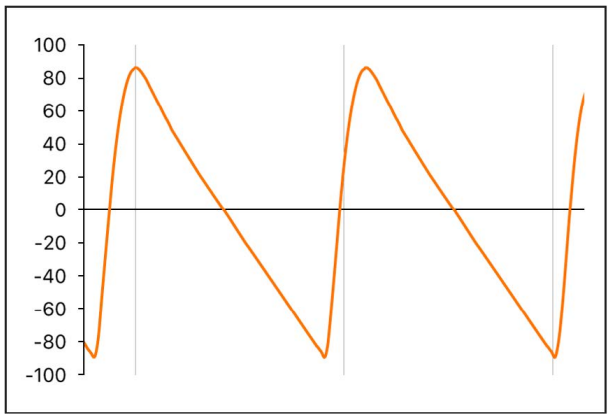
Cutoff frequency

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.”

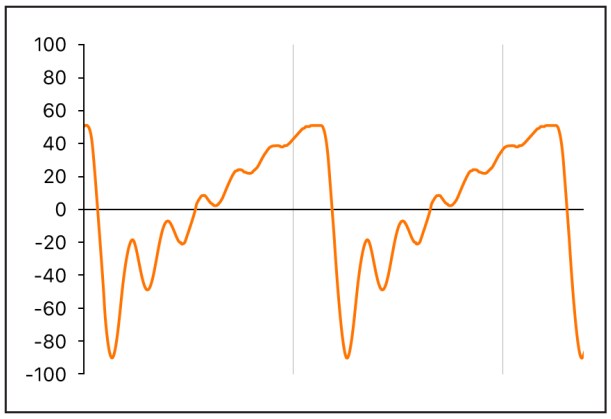


This example illustrates how using a filter to cut away portions of the frequency spectrum alters the waveform shape, thus changing the timbre of the sound.

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.



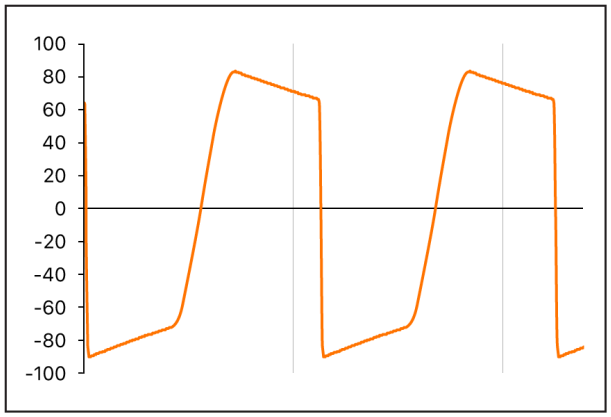
This resonant filter setting results in much brighter and harsher signals close to the cutoff frequency. Frequencies below the cutoff point are not affected.

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

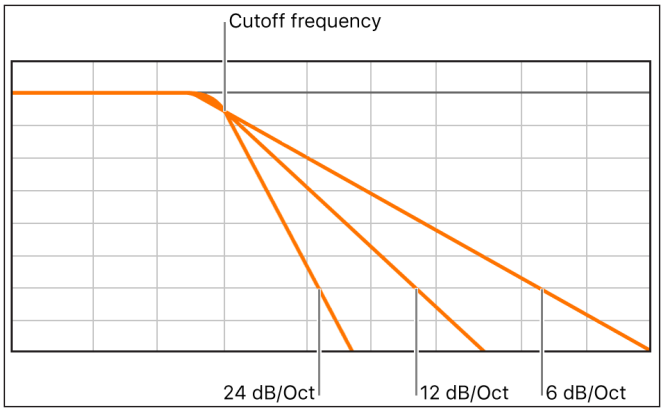
Filter drive adds an amount of gain to the waveform as it enters the filter—an input gain control—overdriving the filter and distorting the waveform. This waveform distortion changes the timbre of the sound, making it harsher.



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 dynamic range.

Filter slope

A filter cuts off the signal at the specified cutoff frequency. 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 is some *decay*, the time it takes to fall from the peak level.

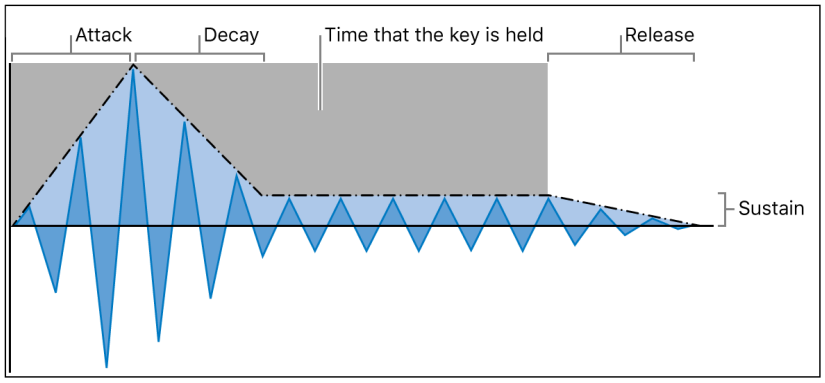
These two sounds clearly have different characteristics over time.

Synthesizers emulate these sonic characteristics by providing control over different parts—the beginning, middle, and end—of the sound 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 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 Sampler, provide a modulation router. Alchemy and Sculpture provide further unique modulation options.

The router lets you 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). Keyscale modulation is often used to control filter cutoff, resonance, or both; higher notes sound brighter than low notes. This emulates the behavior of many acoustic instruments.
- *Controls:* You can use controls such as the modulation wheel, ribbon controllers, or pedals attached to your keyboard. The modulation wheel is most commonly used for pitch bends during performance.
- *Automatic modulation:* You can use envelope generators or LFOs to modulate signals automatically. The most common LFO modulations are control of the pitch or level of a sustained note, resulting in a vibrato or tremolo.

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.

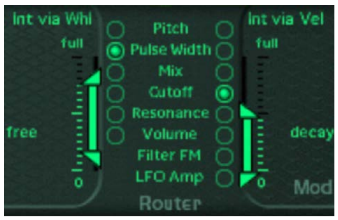
The modulation wheel, pitch bend ribbons, foot pedals, keyboard, and other input options are referred to as *modulation controllers, MIDI controllers,* or just *controllers*.

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 dynamically responds 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, vertically drag the upper or lower arrows, or both, 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).

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 the Logic Pro project or concert tempo.
- *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), 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 or Volume:* Sets the overall loudness of your sound. This control is the master output volume control of your synthesizer.
- *Tune:* Sets the overall pitch of your sound—typically in semitone steps. Many Logic Pro instruments provide additional fine-tuning in cents; a hundredth of a semitone.
- *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. It 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, the played note is 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 synthesizers, such as Alchemy.

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 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.

- Sample-based synthesis
- Frequency modulation (FM) synthesis
- Component modeling synthesis
- Wavetable, Vector, and LA synthesis
- Additive synthesis
- Spectral synthesis
- Resynthesis
- Phase distortion synthesis
- Granular synthesis

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.

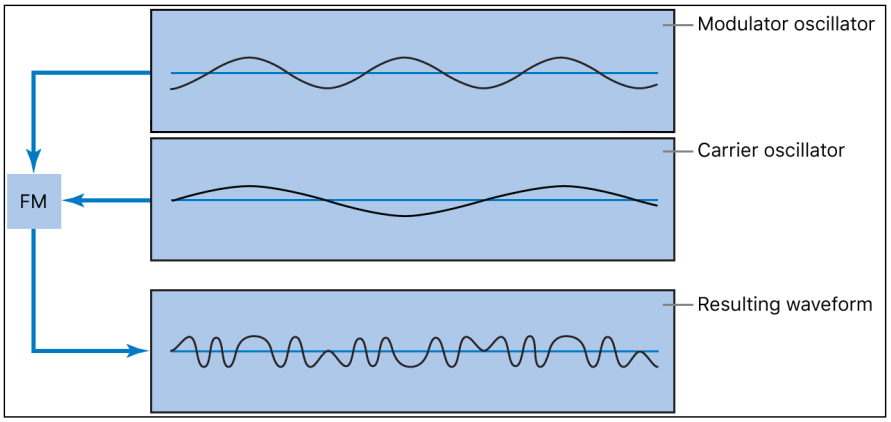
Sampler is a sample player that can be used much like a sample-based synthesizer, due to the subtractive synthesis features that it offers. Alchemy can also be used in this way, but adds additive and spectral resynthesis features that can result in very different sounds than are possible with subtractive synthesis techniques. Alchemy also provides a granular synthesis engine that offers further sample manipulation options, again extending the potential sonic outcome. See Resynthesis and Granular synthesis.

Popular instruments that use the sample-based synthesis approach include the Korg M1, O1/W, and Triton; the Roland JV/XP instruments; the Yamaha 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 waveform generated by the carrier oscillator within the audio range, thus producing new harmonics. These harmonics are known as *sidebands*.

Where there is a mathematical relationship between the carrier and modulator waveforms, the sound produced is harmonic. Where the modulator is a non-integer multiple of the carrier waveform, inharmonic sidebands are produced, resulting in an inharmonic sound.



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 the Yamaha DX series of synthesizers. The DX7, sold from 1983 to 1986, remains the most commercially successful professional-level hardware synthesizer ever made. The Retro Synth FM synthesizer adds a filter section and other features to the FM engine, opening up a much broader range of potential sounds.

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 the physical characteristics of an instrument, 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 blowing; 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 LA 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. ES2 and Retro Synth include a number of wavetable features. Alchemy takes this a step further with granular synthesis which shares some aspects with wavetable synthesis. See Granular synthesis.

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 that were played with filtered sawtooth or pulse waves.

Vector synthesis—used in the Sequential Circuits Prophet-VS and Korg 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 ES2 to perform vector synthesis by modulating the Oscillator Mix (Triangle) parameter with the Vector Envelope.

Additive synthesis

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 method, 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 the level and pitch of 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. In practical terms, working with groups of related harmonics is the best approach due to the mathematical relationships between them and the impact this has on the overall tone when adjusting them en masse, rather than individually.

Alchemy can be used as a true additive synthesizer, where you create sounds from scratch with sine waves, with full control of the level, pitch, and pan position of each harmonic. Alchemy also allows you to resynthesize imported samples with additive (and spectral) synthesis techniques. See Resynthesis.

Some aspects of the additive synthesis approach are also used in Vintage B3 and 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. Because Vintage B3 doesn't provide envelope control over each harmonic, it is limited to organ emulations.

Spectral synthesis

Spectral (modeling) synthesis lets you build a sound by combining multiple (sine wave) harmonics and filtered noise signals. This synthesis method shares many underlying principles with vocoders, but tracks peaks in the overall spectrum, rather than individual amplitudes and frequencies in the signal.

Alchemy provides a flexible spectral synthesis implementation, known technically as multiresolution sinusoidal modeling. In other words, a custom filter bank is used to analyze peaks (and other elements) in the frequency spectrum of the signal. Harmonic components, based on the spectral analysis, are modeled as a combination of sine waves and white noise passed through a filter that changes over time. The noise components are typically used to model "percussive" elements such as a piano strike or a speech fricative in a vocal sample, for example. The (sine wave) harmonic components are used to model the piano note or remainder of the vocal sound. The output of the modeled sound is a combination of the frequencies and levels of the detected harmonic components and the noise signal passed through a time-variable filter.

The spectral synthesis engine in Alchemy can be used to create sounds from scratch, by drawing or painting in the spectral edit window. You can also import and convert an image file into a spectrogram (an image of the frequency spectrum) in the spectral edit window. You can then edit this converted image with the drawing and painting tools. Alchemy analyzes the spectrogram and replaces peaks and percussive components with sine harmonics and filtered noise elements to create a sound.

Alchemy can also break imported samples down into "spectral bins," with each bin storing the amplitude and phase values in the given frequency band. These bins are used to resynthesize (or reconstruct an approximation of) the original sound. See Resynthesis. In noise mode, the amplitude values are used to generate filtered noise for each bin. In pitch mode, the amplitude and phase values are used to synthesize a sine wave for each bin. The signals associated with each bin are then summed and sent to other parts of the Alchemy synthesis engine.

Resynthesis

You can analyze the frequency components of a recorded sound and then resynthesize—reconstruct—a representation of the sound using additive or spectral techniques. See Additive synthesis and Spectral synthesis.

At a basic conceptual level, additive synthesis and spectral (modeling) synthesis are similar in that both techniques can recreate complex sounds by adding together many simpler signals. In practice, however, the two methods are very different. Additive resynthesis attempts to understand the harmonic structure of an audio file and then recreates each harmonic partial with a separate sine wave. Spectral resynthesis instead analyzes the changing frequency spectrum of the signal and attempts to recreate these spectral characteristics in the resynthesized signal.

An additive resynthesis system generates a series of sine waves, with appropriate pitches and levels over time, for each harmonic. It does this by calculating the frequency and amplitude of each harmonic in the overall frequency spectrum of the analyzed sound. After the sound has been resynthesized in this fashion, you can adjust the frequency and amplitude of any harmonic (sine wave). Theoretically, you could even restructure a harmonic sound to make it inharmonic, for example.

In a spectral resynthesis system, the audible spectrum is split into a large number of "spectral bins," and the distribution of energy across these bins is analyzed. The sound is recreated by filling each spectral bin with the required amount of signal, either using sine waves or filtered noise, and the results are then summed.

The difference in approach means the two techniques are suited to different types of sounds. Additive resynthesis excels at recreating single notes with a clear harmonic structure. Spectral resynthesis is better suited for complex inharmonic sounds such as drums, or polyphonic material containing chords rather than individual notes.

Alchemy can resynthesize sounds using additive or spectral methods. It can also perform resynthesis using a combination of the two techniques, which is useful for sounds that feature both a clear pitch component and a noisy component. Examples of such sounds are the hammer strike of a piano and the string tone, or the breath noise of a flute and the flute tone.

Phase distortion synthesis

Phase distortion synthesis creates different waveforms by modifying the phase angle of a sine wave with a second modulator wave. In some respects, this is similar to FM synthesis. The principal difference between the two approaches is that the two waveforms are synchronized each cycle in phase distortion synthesis, resulting in the creation of more harmonic overtones.

In essence, you can bend a sine wave until it becomes a sawtooth wave, a triangle wave, a square wave, and so on. The synthesis engine beyond the waveform generators typically follows a subtractive synthesizer design.

Phase distortion synthesis was commercially introduced in the 1984 Casio CZ series synthesizers.

Several Logic Pro synthesizers allow you to reshape the source waveform, but you are not restricted to sine waves as the raw material.

Granular synthesis

The basic premise behind granular synthesis is that a sound can be broken down into tiny particles, or grains. In many respects, granular synthesis is similar to wavetable synthesis, but it works on a much finer scale. This method is ideal for creating constantly evolving sounds and truly unique tones.

Alchemy extracts sampled grains—2- to 230-milliseconds long—which can be manipulated, reorganized, or combined with grains from other sounds to create new timbres. The effects that result from this technique depend on factors such as the duration and amplitude envelope of each grain, the degree of overlap or separation from one grain to the next, and the pitch and pan position of each piece of sound.

If each new grain is extracted from slightly further into the sample than its predecessor, playback of the resulting stream of grains in their original sequence, at the original speed, essentially puts the pieces of sound back together to closely resemble the source audio material. If you play back the stream of grains at a slower speed, separation (a small gap) occurs between grains. If you play back the stream of grains at a faster rate, each grain overlaps with the next one.

A number of interesting manipulations are possible using this synthesis method.

- Time-stretching. Grains can be sent out at a faster or slower rate than their counterparts in the original sample, allowing faster or slower playback—without the changes to pitch that occur with traditional sample playback. You can even “freeze” a sample at a certain position by extracting multiple grains from a single point. On this latter point, you could repeat a drum hit “grain” multiple times in a time-stretched loop to create a different drum pattern, for example.
- Pitch-shifting. Modifications to the pitch of each grain allow you to vary the pitch of a sample without affecting its timing. By modulating the pitch or pan position of each grain, you can also create spatial and “blurring” effects.
- You can also scramble the order in which grains are played back to produce effects ranging from mild fuzziness to extreme mangling.

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 allowed 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 aftertouch 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, allowing 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 workshop. 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 Mathews. 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 preset storage 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.

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