Performer® Lite
Plug-in Guide

MOTU®

1280 Massachusetts Avenue
Cambridge, MA 02138
Business voice: (617) 576-2760
Business fax: (617) 576-3609
Technical support: (617) 576-3066
Tech support web: www.motu.com/support
Web site: www.motu.com

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Part 1

Plug-ins

CHAPTER 1 Audio Effects Plug-ins

OVERVIEW

Performer Lite includes a variety of audio effects plug-ins. The following sections discuss the settings for each individual effect.

For general information about audio effects plug-ins, see chapter 57, “Audio Effects Plug-ins” (page 510) in the *Performer User Guide*.

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CHANNEL CONFIGURATIONS

Performer Lite allows you to configure your system with a combination of mono, stereo and *n*-channel signal paths. An *n*-channel signal path is a path where *n* equals the number of channels.

A signal can begin as mono and through a panner or processor end up as a stereo or multichannel signal. MOTU Audio System effects are designed to accommodate a wide variety of inputs and outputs.

The following shorthand is used to describe the available configurations of MOTU Audio System effects:

mono to mono	**M-M**
mono to stereo	**M-S**
mono to n-channels	**M-n**
stereo to stereo	**S-S**
stereo to n-channels	**S-n**
n-channels to n-channels	**n-n**

☛ If a plug-in does not support the channel configuration for a track, it will not appear in the track’s plug-in menu. For example, plug-ins that do not support a stereo-to-stereo or stereo-to-n configuration will not appear in the plug-in menu for stereo tracks.

COMMON SETTINGS

This section describes some controls which are common between a number of plug-ins.

Expand buttons

Several plug-ins have sections that can be shown or hidden as desired using an *expand* button.

The Mix setting

For effects that have it, the *Mix* setting controls how much of the effected signal is included. In most of the effect presets that are included with Performer Lite, Mix is set to 100% for in-line use; if you are applying the effect to a dedicated bus/aux track, you should set the mix to 100%.

MIDI control of plug-in parameters

Several audio plug-ins allow you to control their settings from MIDI note data, either from a MIDI track or from your controller keyboard.

Tempo lock

Many of Performer Lite's included plug-ins allow you to lock certain parameters, such as their LFOs, to the tempo of your sequence. This allows the effect to stay in sync with the beat of your music, even if there are tempo changes.

Any plug-in that supports tempo-locked parameters will display the *Tempo Lock* menu. The Tempo Lock menu provides several different ways of synchronizing the plug-in parameter to the tempo, as demonstrated below in the Echo plug-in.

Figure 1-1: Choosing what type of tempo lock you would like.

The choices for tempo lock shown above in Figure 1-1 are explained below.

Real time

Lock to *real time* if you don't need to synchronize the plug-in's parameters to the tempo of your sequence and instead need to work with them in a real time format such as milliseconds.

Beats

Lock to *beats* when you want the effect to follow the "pulse" of your music. Use this mode for 4/4-based dance music (or similar meters like 3/4, 2/4, etc.)

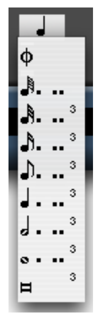


Figure 1-2: The beats menu displays note durations that are referenced to 4/4 time. A quarter note represents 1 beat; an eighth note represents a half a beat, and so on.

This mode is also ideal for tempo-based effects over music with meter changes because the automation will always follow the beat — even through your meter changes, as determined by the *click value* of each meter change. Remember, when you insert meter changes, you also choose what 'gets the beat'. This is what we refer to as the *click value*. Beat-based automation always follows the click value of each meter change.

Here's an example: let's say that your sequence changes from 4/4 time to 6/8 time, and in the 6/8 section, a dotted quarter-note gets the beat. You then choose a quarter note from the *Beats* tempo lock menu. In this case, plug-in automation will follow "the beat" which, in 4/4 time is a quarter note pulse. In the 6/8 section, the automation will follow dotted quarter notes because the dotted quarter note is getting the beat (as prescribed by the 6/8 meter change click value).

☛ The rule of thumb when using Beats mode is this: a quarter equals one beat, whatever the beat happens to be (as determined by the meter). It could be a dotted quarter (in 6/8 time) or a half note (in 4/2 time) and so on. Use Beats mode when you want automation to follow 'the beat', and the beat is changing from meter change to meter change.

Note value
Lock to *Note value* when you want the plug-in to pulse at a particular note duration value, regardless of meter. For example, if you choose a 16th note, the effect will pulse to a 16th note pattern (120 ticks at 480 PPQ) regardless of any meter changes in the sequence.

Bars
Lock to *Bars* when you want an effect to pulse according to measure (bar) boundaries. This is a convenient way to align effects automation on a slightly larger musical scale than beats. For example, you might program a filter sweep to finish on the downbeat of every measure. This is particularly useful when you have meter changes because automation will speed up and slow down dynamically to maintain the measure-based relationship you specify.

The Bars menu has standard settings you'd expect, like 1 bar, half a bar, and two bars. But it also has fractional bar lengths that can produce very interesting syncopated and poly-rhythmic effects.

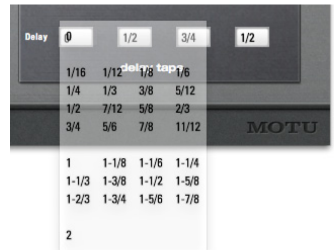


Figure 1-3: Specifying tempo-based automation by a number of bars, such as 1 bar, 2 bars or half a bar. Experiment with the fractional measure lengths for interesting effects.

ACE-30

M-M	M-S	M-n	S-S	S-n	n-n
yes	yes	no	yes	no	no

The ACE-30 is an amp modeling plug-in that allows you to select various preamp circuits from both the vintage-style Vox® AC30/6®, as well as the more modern Vox ®AC30 CC2X®.

Input: the input jack array is similar to the Fender® Bassman®. The input matrix provides choices for the channel (horizontal) and input attenuation (vertical).

The vintage-style Vox AC30/6 originally had three different preamps and two impedance levels the amp drew from: Normal, Top Boost (Bright), and Tremolo, and High and Low impedance (thus the AC30/6 name).

In Performer Lite's ACE-30 model, we have chosen to separate out the Tremolo circuit for more independent control and versatility of tone. This model is more similar to Vox (Korg's later model (the CC2X).

Normal: controls volume when the normal channel is selected (no tone stack).

Top Boost: provides an additional "bright" gain stage with tone control features.

Top Boost Setting	Description
Volume	Volume of Top Boost channel
Treble	Treble tone stack control
Bass	Bass tone stack control

Tone Stack: choose *Vintage* for a Gibson tone stack sound (including the notorious schematic error made by Gibson, but none-the-less copied by Vox). The Vintage setting has a much sharper EQ notch — at approximately 2k-4k — when the volume is turned up. The *Modern* setting is more similar to a Fender-style tone stack.

Vox's later (most popular) AC30 models (from the mid 1960s) were copied from a Gibson GA-70/ GA-77 schematic. The ACE-30 plug-in models incorporate even the smallest details of these amp designs.

Tremolo: use the switch to enable/disable tremolo.

Speed: controls the LFO frequency (2-8 Hz).

Depth: adjusts the depth of the tremolo (volume variance), where 0 is completely off.

Master: the master section provides final output stage controls.

Tone Cut: similar to the presence control of a Fender-style power amp, but operates backwards.

Volume: controls the master volume (the final output after the sound has been run through all the circuits).

ANALOG CHORUS

M-M	M-S	M-n	S-S	S-n	n-n
yes	yes	no	yes	no	no

Analog Chorus emulates the popular Boss CE™ series of effect pedals from the early ’80s.

Rate: sets the rate of the LFO, in Hertz (Hz).

Depth: sets the depth of the LFO.

Mix (mono and stereo-to-stereo only): controls the wet signal level.

Mode (mono-to-stereo only): selects between sum+difference (I) or split wet+dry (II).

Status light: displays the bypass/enabled state; when lit, the effect is active.

Pedal: bypasses/enables the effect. This works the same as the Effect window’s Bypass button.

ANALOG DELAY

M-M	M-S	M-n	S-S	S-n	n-n
yes	yes	no	yes	no	no

Analog Delay is a companding delay pedal model based on 18v bucket-brigade pedals from the early 80’s. It emulates the DOD 585, which utilizes the same chips as the Electro-Harmonix™ Memory Man™ for the compander and bucket brigade device.

Mix: this is essentially a wet/dry crossfader.

Repeat: controls the amount of feedback. Low settings have few echoes, high settings self-oscillate. Self-oscillation allows the delay to maintain its signal indefinitely.

Delay Time: sets the clock frequency of the bucket brigade model, which determines the delay time between samples and reflection frequency for aliases. In other words, turning the knob clockwise causes the bucket brigade to cycle less frequently. Samples are moved from the input toward the output less often, and as a result the audio delay increases.

Conversely, turning the Delay Time knob counter-clockwise corresponds to increasing the speed at which the “buckets” are transferred. þThe signal samples in the buckets take less time to be transferred from input to output, thus the delay time goes down.

X-Feed: this controls the stereo routing of the inputs and feedback loop within the delay circuit.

For a mono-stereo signal, 5 is equivalent to parallel mono, and 0 or 10 creates a ping-pong delay whose first echo follows the direction of the knob’s pointer.

For a stereo-stereo signal, the mono behavior is preserved for the left channel and mirrored by the right channel.

Pedal switch: the footswitch works like the STNDBY button on the MasterWorks Leveler. This forces the mix control to 0 internally while leaving the delay circuit alive, which allows self-oscillation to build up without being heard.

BYP: this switch controls the *Bypass* parameter, causing all model elements to be cleared and removed from the FX chain.

1X/4X: this switch selects a delay time multiplier that modifies the model delay time without affecting the clock effects. On 4X, the delay time is quadrupled. On 1X, the delay time is unaffected.

ANALOG PHASER

M-M	M-S	M-n	S-S	S-n	n-n
yes	yes	no	yes	no	no

Analog Phaser is a model of the *MXR '74 Vintage Phase 90* phaser. This pedal model produces the “dirty phaser” sound popularized by Eddie Van Halen.

Pedal switch: bypasses/enables the effect. This works the same as the Effect window’s Bypass button.

Speed: controls the LFO frequency.

CUSTOM '59

M-M	M-S	M-n	S-S	S-n	n-n
yes	yes	no	yes	no	no

Custom '59 is an amp modeling plug-in that lets you mix and match preamp tubes and circuits on-the-fly, with complete automation of all parameters.

Modeled amps

Custom '59 models three renowned guitar amps: the original Fender® Bassman®, the Marshall® JTM45® and the Marshall® JCM800®.

Fender® Bassman®

Originally designed for the Fender Precision Bass®, the Fender Bassman amp was quickly adopted by guitarists and eventually became perhaps the most famous and sought after guitar amp of all time. With its classic 4x10 design (four ten-inch speakers) and classic lacquered tweed cover, the Bassman sound is a bona fide icon among guitar tones and a must-have for any tone aficionado.

Marshall® JTM45®

Introduced in the early 1960's, the Marshall JTM45 was essentially a clone of the Fender Bassman. Made popular by Eric Clapton, the so-called "Bluesbreaker" amp is named after Clapton's band at the time, in which he popularized the now signature sound of playing a Les Paul through the heavily distorted JTM45. Ever since, the JTM45 has been ensconced in the pantheon of world-famous guitar amplifiers.

Marshall® JCM800-1987®

By the early '80s, Marshall had developed the JCM800, with higher power tubes and a power boost from 50 to 100 watts. This amp produces perhaps the most widely recognized guitar tones of all time.

Create your own amp

Custom '59™ lets you play an extremely accurate reproduction of the sound of each of these three famous amps. But you can also mix and match the preamp tube, preamp circuit and tone stack from each model to create your own custom amp.

Input jacks: selects the input channel and impedance. Channel I and Channel II each have a high-Z input (1) and a low-Z input (2).

Vol I, Vol II: volume controls for each channel.

3-band EQ: cuts or boosts for low, mid, and high frequencies.

Master: output level.

Input Tube: selects a tube for the input stage. This determines headroom, first-stage gain, distortion characteristics and to some extent the frequency response of the volume control circuits.

Preamp Circuit: selects the volume control circuit model for the indicated amp.

Tone Stack: selects the tone control circuit model for the indicated amp.

Power Amp

When *Preamp* is selected in the Power Amp menu, only the pre-amp stage of Custom '59 is activated. When one of the other settings is chosen, both the pre-amp stage and the power amp stage are activated. Additionally, the *Presence* control appears, which controls a progressive high frequency shelf (Figure 1-4).

Figure 1-4: Custom '59 Power Amp settings.

Power Amp models

Each Power Amp model has different characteristics.

Preamp: clean, high-fidelity solid-state power stage (no post-processing of pre-amp model beyond a simple gain control).

Vintage: spongy, touch-sensitive and loose to the point of sounding "flabby" at high distortion levels.

Classic: still touch-sensitive but with a more defined overdrive character.

Modern: tighter, sacrificing some touch-sensitivity for increased definition at maximum drive levels.

DELAY

The delay plug-in produces classic delay effects. With mono-to-stereo processing, stereo-to-stereo processing, and separate left/right channel controls, you can create complex stereo and 'ping-pong' delay effects.

Figure 1-5: Delay.

M-M	M-S	M-n	S-S	S-n	n-n
no	yes	yes	yes	yes	yes

☛ To program multiple channels at a time, hold down the Option key.

Tempo lock

This menu lets you choose the format for the delay time. It can be milliseconds, or one of several tempo-based modes, which lock the delay taps to the tempo of your sequence, even if there are tempo changes. Accordingly, the delay section of the plug-in displays a note value menu (or bar length menu). The tempo-based modes are Beats, Note Values, and Bars. For details about these tempo-based modes, see “Tempo lock” on page 8.

☛ If the transport is not moving, Delay assumes the last tempo you played at or 120BPM if you have not played at all.

Figure 1-6: Choose real-time from the Tempo lock menu when you want to specify the delay time in milliseconds. Choose Beats, Note Values, or Bars to specify the delay time in musical values such as note duration or measure length. Beat and bar delay times stay locked to the tempo of your sequence, even through tempo changes.

Delay

This controls the length of the delay. As shown above in Figure 1-6, you can specify delay time in milliseconds (real time) or by specifying note duration (or bar length). Just choose the desired time format from the Tempo Lock menu.

To generate complex, poly rhythmic effects, try choosing different note values for the left and right channel delays.

To generate ping-pong effects, use cross-feedback without normal feedback.

Out of range

The *Out of range* light only applies to beat-based automation modes (beats, bars or note values). If you’ve chosen one of these modes, the *Out of range* light illuminates when the length you’ve specified for the delay makes the total delay time longer than 2 seconds (the maximum time allowed by this plug-in).

Note that the current sequence tempo factors into the delay time for beat-based modes. For example, a quarter note is one second long at 60 bpm but only a half a second long at 120 bpm. So if you specify a whole note delay, and the sequence tempo is 60 bpm, the total delay time you’ve specified is 4 seconds, which is longer than the 2-second maximum allowed by the plug-in. In this case, the *Out of range* light illuminates to alert you to this fact. To turn off the light, choose a shorter note, beat or bar value, or increase the tempo of the sequence.

Mix

Controls the overall level of the delay echoes mixed with the original signal.

Filter

This is a standard EQ filter that you can apply to the signal before it is fed into the feedback and cross feedback processors. Filter types include low pass, high pass, notch and bandpass filters with appropriate frequency and width settings, where applicable. This is a great way to apply an ‘effected’ sound to the delay taps, which can add more interest and dimension to the overall delay effect.

Feedback and cross-feedback

☛ Warning! Be very careful when working with these controls, as they can quickly generate ear-splitting, speaker blowing feedback paths if you are not careful.

☛ If you accidentally generate a feedback loop that is getting out of control, *go for the panic button, and then stop playback or adjust the delay settings*. Simply stopping playback won't stop the feedback!

The feedback control adds feedback to the delay processor on the same channel. The cross-feedback control adds feedback to the opposite channel's delay processor. Used sparingly, these controls can greatly add to the complexity of the delay effect.

Delay - Surround Versions

The Delay produces a wide variety of delay effects using a surround speaker matrix.

How it works

The Feedback delay provides an independently programmable signal path for each (non-LFE) channel in your surround matrix. Therefore, if using a 5.1 surround matrix, you have a total of five channels of delay. The interface displays the parameters of a single channel of delay at a time. The display window in the upper left hand corner and the speaker radio buttons tell you which channel you are currently programming. Holding down the Option key programs all channels simultaneously.

To select a channel to edit, click a speaker icon on the circular display in the lower right hand corner of the display. The current channel name will be displayed in the upper left hand corner and the speaker icon will highlight.

Figure 1-7: The surround delay.

Panic: because it is possible to send 100% of the signal to multiple destinations, which in turn are feeding back into a number of other destinations, delays can spiral out of control rather quickly. If this happens, the panic button will zero out all of the delay lines giving you enough time to stop playback or reduce the feedback gains. Remember, stopping playback will not stop DSP processing.

Input

The number of source channels determines the input behavior of the n-channel version of Delay.

Mono to n - One input gain knob is provided to control the amount of input signal sent to current delay channel.

Stereo to n - Two input knobs are provided representing the left and right sides of the source signal. The left and right inputs can be used independently or mixed to send signal to the current delay channel.

N to N - each input is hard-wired to its corresponding channel output. A knob is provided to trim the input.

Feedback Controls

The Delay provides a feedback path from each non-LFE channel to every other non-LFE channel For example: if using a 7.1 surround matrix, a total of 49 independent feedback paths are available. In 10.2, there is a total of 100 feedback paths.

Feedback: controls how much post-delay, post-filtered signal is recirculated in the currently selected channel.

X Feedback: controls how much post-delay, post-filtered signal is sent to each other delay channel from the currently selected channel.

DELTA FUZZ

M-M	M-S	M-n	S-S	S-n	n-n
yes	yes	no	yes	no	no

Delta Fuzz emulates the Electro-Harmonix Big Muff ¼™ pedal, used by many ’80s and ’90s bands including the Smashing Pumpkins, Dinosaur Jr., and Mudhoney.

Volume: output gain.

Tone: variable high-frequency boost, with shallow midrange notch.

Sustain: amount of distortion.

Status light: displays the bypass/enabled state; when lit, the effect is active.

Enable switch: bypasses/enables the effect. This works the same as the Effect window’s Bypass button.

DIAMOND DRIVE

M-M	M-S	M-n	S-S	S-n	n-n
yes	yes	no	yes	no	no

Diamond Drive emulates the Voodoo Lab Sparkle Drive™ pedal, which combines an Ibanez TS9™ clone with a dirty/clean crossfader.

Gain: amount of distortion.

Tone: variable high-frequency roll-off, with a peak at high settings.

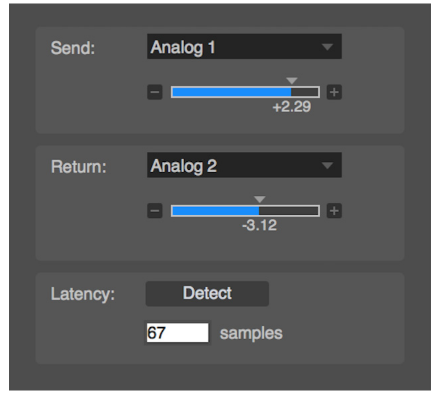
Clean: crossfades between distorted signal and clean signal.

Volume: output gain.

Status light: displays the bypass/enabled state; when lit, the effect is active.

Enable switch: bypasses/enables the effect. This works the same as the Effect window's Bypass button.

HARDWARE INSERT



M-M	M-S	M-n	S-S	S-n	n-n
yes	yes	no	yes	no	no

The Hardware Insert plug-in functions just like other DP effects plug-ins, but loops the channel's audio signal to a piece of outboard hardware for external processing by the hardware. Audio is then returned to the plug-in, with latency compensation, if desired. It can be instantiated in line with other software plug-ins and even saved as part of an effect chain clipping. The hardware Insert plug-in allows you to easily incorporate outboard gear into your DP effects chains.

Send: Choose the output (or output pair) on your audio interface that is connected the input on your outboard gear. The choices in this menu are set up in DP's Bundles window Outputs tab.

Use the volume slider to boost or attenuate the output signal.

Return: Choose the input (or input pair) on your audio interface connected to the output from your outboard gear. The choices in this menu are set up in DP's Bundles window Inputs tab.

Use the volume slider to boost or attenuate the input signal.

Latency: Click the Detect button to make the Hardware Insert plug-in detect the round trip latency to and from the outboard gear. The detected latency is displayed in the text box below (in samples). You can also enter the number of samples manually. When the samples text box has anything other than zero in it, latency compensation is employed to ensure that the audio signal returned from the outboard gear remains in phase with the rest of the channels in the mix.

INTELLIGENT NOISE GATE

M-M	M-S	M-n	S-S	S-n	n-n
yes	yes	no	yes	no	no

Intelligent Noise Gate is a noise gate designed specifically for recording instruments that are prone to AC mains interference.

☛ In the stereo-to-stereo variant, the stereo channels are processed independently.

Threshold: trigger level that opens the gate.

Attack: rise time constant of the gate, in microseconds (µs).

Hold: minimum amount of time the gate will stay open once triggered, in milliseconds (ms). Decrease if noise overhanging the note is a problem.

Release: fall time constant of the gate, in milliseconds (ms).

Mains Frequency: tune this to your national power grid.

Noise Type: AC noise comes in two flavors: "Hum" is what gets picked up inside an amp using AC heated preamp tubes, and "Buzz" is due to ripple in poorly-regulated DC power supplies such as those for old stomp boxes. In North America, these correspond to approximately 60 Hz and 120 Hz fundamental frequencies, respectively. The switch should be set to match the dominant noise source in the input.

Status light: displays the bypass/enabled state; when lit, the effect is active.

Pedal: bypasses/enables the effect. This works the same as the Effect window's Bypass button.

INVERT PHASE

M-M	M-S	M-n	S-S	S-n	n-n
yes	no	no	yes	no	no

Invert Phase flips the phase of the input signal. This is useful when you have recorded two out-of-phase signals, such as the top and bottom of a snare drum. Place the Invert Phase plug-in on one of the out-of-phase tracks to bring both tracks into phase.

LIVE ROOM B

M-M	M-S	M-n	S-S	S-n	n-n
yes	yes	no	yes	no	no

Live Room B models a loudspeaker cabinet in a physical space. As the name implies, the modeled characteristics of this plug-in are intended for bass guitar, although it can be successfully applied to many kinds of audio material.

Live Room B requires a library file containing data about these physical models. This file is placed by the Performer Lite installer here:

/Library/Application Support/MOTU/LiveRoomG/LiveRoomB Data.bundle

Controls

Cabinet Drive: amount of distortion provided by the cabinet.

Output Gain: output level.

Cab: selects the cabinet model. See “Cabinet models” on page 22.

Decay: applies a down-fade to the reverb portion of the response at the indicated rate, which simulates decreasing the reverb time of the room.

Damping: controls the high-frequency roll-off of the room, similar to hanging curtains or setting up gobos.

Display area: graphical representation of the cabinet selection, mic types, and mic positions. This is for display purposes only; the graphic cannot be edited.

Microphone mixer

There are four microphone channels: two mono (channels 1 and 2) and one stereo (channels 3 and 4), each with their own set of the following controls:

Mic: selects the microphone type and position. Several microphone models are provided:

Close-mics:

- AKG D 112
- Sennheiser e602-II
- Shure SM7B
- Yamaha SubKick
- M-Audio Sputnik

Matched pair:

- Schoeps small diaphragm condensers

Channels 1 and 2 can be set to one of six mics with set positions: *DYN112, DYN602, DYN7B, SubKick, Condenser*, or *Far Omni*.

Channels 3 and 4 can be set to one of four mic options: *XY, ORTF, Blumlein*, or *Wide Omni*.

To disable a channel (or pair), choose *None*.

Pre-delay: advances or delays the signal to compensate for the time it takes for a signal to reach a microphone. The range corresponds to about +/- 10 feet, and the plug-in automatically adjusts the relative delay to prevent chopping off the direct sound and early reflections in close-mic responses.

3-band EQ: cuts or boosts up to 15 dB for low, mid, and high frequencies.

Pan / Width: (mono-to-stereo only) on channels 1 and 2, *Pan* controls the placement of the mono source in the stereo output signal; on channels 3 and 4, *Width* controls the width of the stereo signal.

Solo and Mute: solos or mutes the channel.

Fader: controls the output level.

Side chain outputs

Live Room B provides a side chain output for each of its four channels. The side chain output signals are split before Live Room B’s EQ, solo/mute, pan, fader, and output gain controls. This enables you to take advantage of Performer Lite’s full mixing environment for each microphone channel, including plug-ins and automation.

These side chain outputs are configured in the same manner as multiple outputs from a virtual instrument. They can be accessed in the same way as any other audio input: in the *Input* menu in the Tracks window, from the *Audio Input* menu, underneath the fader in the Mixing Board (Figure 1-19), or in the Bundles window.

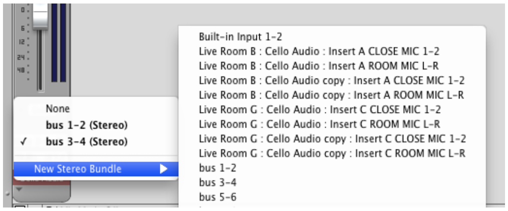


Figure 1-8: Accessing the Live Room B side chain inputs

Cabinet models

The following cabinet models are provided:

8x10 Fridge: modeled after a sealed Ampeg SVT-810.

4x10 Giant II: modeled after the rear-ported SWT Goliath II with the HF driver fully attenuated.

1x15 Ported: modeled after the front-ported Ampeg SVT-15.

1x18 Big Bear: modeled after the rear-ported SWR Big Ben.

LIVE ROOM G

M-M	M-S	M-n	S-S	S-n	n-n
yes	yes	no	yes	no	no

Live Room G models a loudspeaker cabinet in a physical space. As the name implies, the mics and cabinets modeled by this plug-in are intended for guitar, although they can be successfully applied to many kinds of audio material.

Live Room G requires a library file containing data about these physical models. This file is placed by the Performer Lite installer here:

/Library/Application Support/MOTU/
LiveRoomG/LiveRoomG Data.bundle

Controls

Cabinet Drive: amount of distortion provided by the cabinet.

Output Gain: output level.

Cab: selects the cabinet model. See “Cabinet models” on page 23.

Decay: applies a down-fade to the reverb portion of the response at the indicated rate, which simulates decreasing the reverb time of the room.

Damping: controls the high-frequency roll-off of the room, similar to hanging curtains or setting up gobos.

Display area: graphical representation of the cabinet selection, mic types, and mic positions. This is for display purposes only; the graphic cannot be edited directly.

Microphone mixer

There are four microphone channels, two mono (channels 1 and 2) and one stereo (channels 3 and 4), each with their own set of the following controls:

Mic: selects the microphone type and position. Channels 1 and 2 can be set to one of five options: On Axis, Off Axis, Near, Rear, Far Omni; channels 3 and 4 can be set to one of four mic options: XY, ORTF, Blumlein, Wide Omni. To disable a channel (or pair), choose *None*.

Pre-delay: advances or delays the signal to compensate for the time it takes for a signal to reach a microphone.

3-band EQ: cuts or boosts up to 15 dB for low, mid, and high frequencies.

Pan / Width (mono-to-stereo only): on channels 1 and 2, Pan controls the placement of the mono source in the stereo output signal; on channels 3 and 4, Width controls the width of the stereo signal.

Solo and Mute: solos or mutes the channel.

Fader: output level.

Side chain outputs

Live Room G provides a side chain output for each of its four channels. The side chain output signals are split before Live Room G's EQ, solo/mute, pan, fader, and output gain controls. This enables you to take advantage of Performer Lite's full mixing environment for each microphone channel, including plug-ins and automation.

These side chain outputs are configured in the same manner as multiple outputs from a virtual instrument — see "Multiple audio outputs" on page 77 in the *DP4 User Guide*.

Cabinet models

The following cabinet models are provided:

4x12 Modern: Intended for ultra-distorted chunks and sludge, yet versatile enough to handle smooth Santana-style leads.

4x12 Vintage: Based upon an aging, road-worn British monster held together with gaffer's tape and AquaNet.   Perfect for those '80s hair-band tributes and '70s proto-metal.

2x12 ACE-30: Based on an AC30 with carefully-aged Celestion "Alnico Blue" drivers for guitar tones that never go out of style.

2x12 Combo: For Muscle Shoals-style southern-rock and country guitar tones.

1x12 Combo: Based on a first-generation Boogie Mark IV with the original 12-inch Black Shadow driver in the finished walnut version of the open-back cabinet. This cabinet is a versatile tone machine.

1x12 Citrus: Based on an Orange 1x12-inch closed-back extension cabinet loaded with a Celestion Vintage 30 driver.

4x10 Combo: Tuned for blues, jazz, rock and country. Based on a classic.

2x10 Tilted: Based on a Fender Vibrolux 2x10-inch in the tilted-back configuration. This cabinet pairs well with single coil pickups.

1x8 Junior: This one was set up to record distorted rhythm guitar tones for rock and pop. If you like Eddie Money then this is your cabinet.

LIVE STAGE

M-M	M-S	M-n	S-S	S-n	n-n
yes	yes	no	yes	no	no

Live Stage is a model of recording a signal reproduced by a loudspeaker cabinet in a studio. The cabinet and microphone settings offered by the plug-in are ideal for guitar and bass guitar tones. This is a great low-CPU direct box for live performance use: just choose a cabinet and a mic, adjust damping and decay, set your levels, and rock out!

Gain: adjusts level of the signal coming in and being fed through the effect.

Damping: controls the high-frequency roll off of the room, similar to using curtains or gobos.

Decay: applies a down-fade to the reverb portion of the response at the indicated rate, which effectively decreases the reverb time of the room.

MASTERWORKS COMPRESSOR

The MasterWorks Compressor offers precise level control of specific frequency bands of a digital audio signal. This plug-in offers three separate

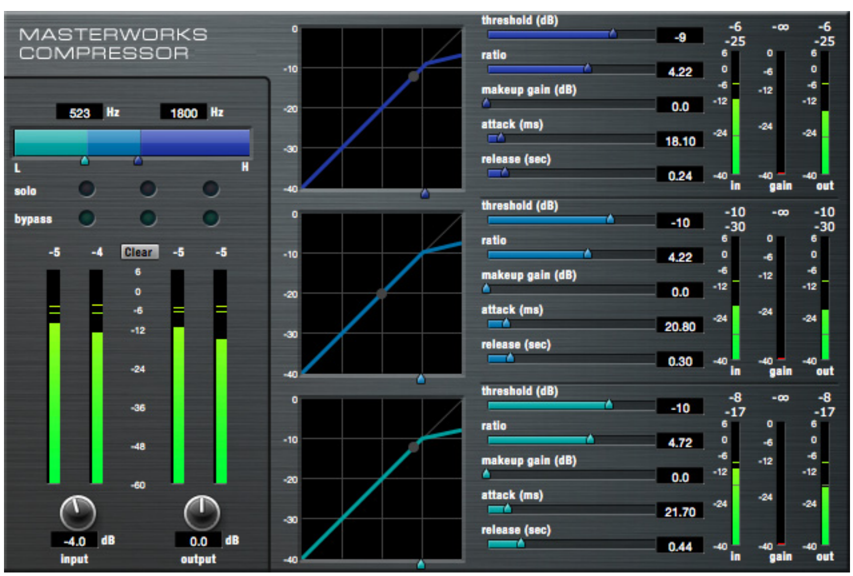


Figure 1-9: The MasterWorks MultiBand Compressor plug-in.

compressors with crossover controls, allowing you to set the frequency range and compression characteristics of each band.

M-M	M-S	M-n	S-S	S-n	n-n
yes	no	no	yes	no	no

The MasterWorks Compressor can be used to subtly control bass, mid-range or treble frequencies independently of each other. A wide band compressor will respond to the loudest parts of a signal regardless of its frequency. This can cause the kick drum to attenuate cymbal hits, for example. A multi-band compressor gives separate control of the three frequency bands, so — continuing with the same example — a loud, low-frequency signal will not trigger high-frequency compression. Another common use of frequency-specific compression is de-essing. You may want to control the sibilants of an audio signal without affecting lower frequencies. The MasterWorks compressor will allow you to get maximum punch on an individual track or the entire mix, while maintaining sonic clarity.

Signal flow

The MasterWorks Compressor plug-in has the following signal flow:

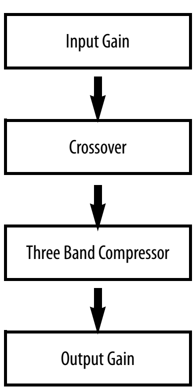


Figure 1-10: MasterWorks Compressor signal flow.

How it works

The MasterWorks Compressor works very much like a traditional wide band compressor. The difference is that the MasterWorks Compressor is actually three independent compressors that act upon differing frequency ranges of the incoming signal. These frequency ranges are set by the crossover controls.

Each compressor has a make-up gain control for adding gain to that frequency band to compensate for compressor attenuation. Additionally, there are global input and output gain controls for affecting the entire frequency range of the signal.

Soloing/muting

Each frequency band can be soloed or muted, allowing you to audition only the band you are working with. This can be very useful for tuning in on the exact frequencies you want to control.

Many instruments create a wide frequency range of sound, and the MasterWorks Compressor allows you to control each part of that sound with great precision. A good example of this is an electric bass guitar. The primary frequencies that define the notes of this instrument are from around 250 Hertz (Hz) to 1500 Hz. Frequencies below 250 Hz are “felt” more than heard, and frequencies around 200 Hz are generally finger noise and string buzz. The MasterWorks Compressor allows you to control each aspect of the sound for a very even, easy-to-mix bass track.

De-essing

Basic de-essing is easily achieved by compressing only the high frequency content of the mic signal. Use fast attack and release times.

Controls

Input Gain: Boosts or attenuates the incoming signal.

Crossover: Two crossover points separate the frequency spectrum into three bands for separate compression.

Output Gain: Boosts or attenuates the global output of the plug-in.

Lo Band, Mid Band, Hi Band: Select among frequency bands for three separate compressors. The following controls are available for each compressor band.

Solo: Mutes the non-soloed frequency bands. Very useful for hearing what a specific frequency compressor is doing.

Bypass: Disables compression for the selected band.

Threshold: Determines the level above which the compressor will have an effect. Threshold can be set with the knob, slider that appears on the band input meter, and in the graph display.

Ratio: Determines the amount of compression on the signal over the threshold.

Make up: Adds gain to the signal after compression. Because compression attenuates a signal, it can be desirable to add gain to the signal after compression.

Attack: Determines the time it takes for the compressor to react after the signal has exceeded the threshold.

Release: Determines the time it takes for the compressor attenuation to return to zero after the input signal has dropped below the threshold level.

Clear: Clears peak meters for the chosen band.

RMS: Shows the Root Mean Square, or average level. Since the average level of a mix is usually well below any maximum peaks, it is useful to know what the average level is, as well as the loudest points. This is very helpful when matching the apparent levels of different frequency bands.

Compression Graph: Gives a graphic display of compression parameters and the actual signal level for each band. Any combination of bands can be viewed at the same time.

MasterWorks Compressor presets

The MasterWorks Compressor presets (found in the MasterWorks window Preset menu) are designed for many common applications. Compression is a level-dependent effect. As a result, input level — both broadband and frequency specific — is critical to what the final output is. The presets will do different things depending on input level and bandwidth of the audio signal.

Compression can be used to make a signal more even or natural-sounding, or it can be used as an obvious special effect. Try putting a drum mix through the MasterWorks compressor, and listen to how the cymbals get “squeezed”. Add heavy compression to the bottom end of a full mix, then increase the make up of that frequency for a tightly controlled kick drum and bass. Run a solo acoustic guitar or piano through even multi-band compression and hear how each part of the instrument is more present, without mix-ruining peaks.

You may only want to compress specific frequency bands, such as when you are de-essing, or you may want to compress the entire frequency spectrum of the audio signal. A major advantage of multiband compression is that frequency-specific peaks do not cause broad band compression. A good example of this is a slapped electric bass. The high frequency portion of the signal will not cause the low frequency to be limited. On a full mix, it is possible to have smooth control over lows, mids, and highs without "hole-punching".

As in the case of any effect preset, these are starting points. Experiment! Your ears will be the final judge of the usefulness of the effect.

MASTERWORKS EQ

M-M	M-S	M-n	S-S	S-n	n-n
yes	no	no	yes	no	yes

Inspired by legendary British large console EQs, the MasterWorks EQ gives you the look, feel and sound of the most sought-after classic equalizers.

Five bands of EQ filtering are provided, each with four EQ types that provide current popular EQ styles and vintage analog EQ styles alike. Two mid bands (LMF and HMF) include shelf filtering. Two additional bands of variable slope low pass and high pass filtering are provided. The filter response display provides comprehensive control and visual feedback of the EQ curve being applied. The MasterWorks EQ has been carefully crafted and meticulously engineered to produce musical results in a wide variety of applications.

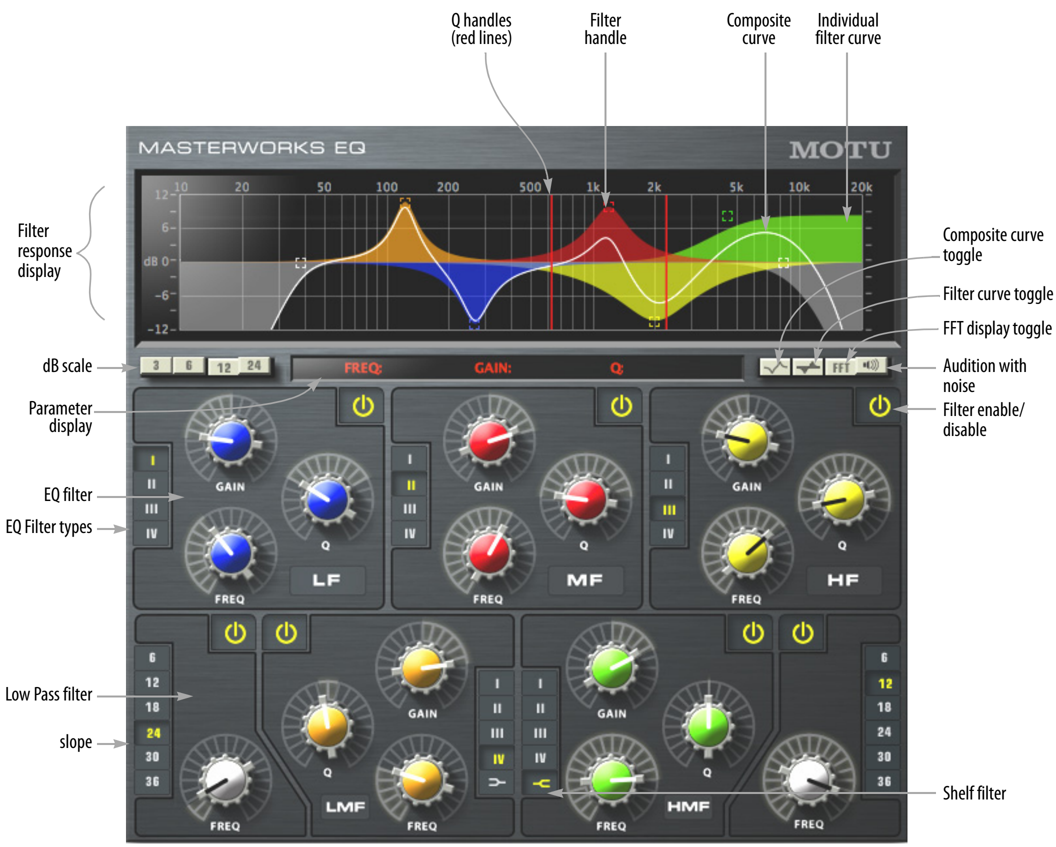


Figure 1-11: MasterWorks EQ.

Quick reference

Filter response display: Shows the response curve and frequency analysis for the current settings.

dB scale: Lets you specify the vertical scale (in dB) of the filter response display.

Parameter display: Shows the precise numbers of the parameter you are adjusting (or hovering over with the arrow cursor). The labels (*frequency, gain*, etc.) match the color of the filter being displayed.

EQ filter: one of five bands of EQ that can be independently enabled and programmed.

Filter type: Lets you choose from one of four or five EQ styles for each independent band of EQ.

Low Pass filter: Both a low pass and high pass filter are supplied with six different slope settings.

Slope: Lets you choose the slope (fall off) characteristics of the low pass and high pass filter.

Q handle: Drag the Q handle lines to graphically adjust the Q setting for the currently selected filter. To select the filter, click its filter handle.

Filter handle: Drag this handle to graphically adjust the filter's boost/cut and/or frequency.

Composite curve: shows the overall response curve of the current settings in the window.

Individual filter curve: Each filter has a color (indicated by its knobs). When filter curves are being displayed (the filter curve option is turned on), each individual filter's response curve is displayed in the filter's color.

Composite curve toggle: Turns the composite curve display on or off.

Filter curve toggle: Turns the display of the filter curves on or off.

FFT display toggle: The filter display area can also produce an FFT display of the processed signal. Use this button to turn it on or off.

Audition with noise: When this option is enabled, and you adjust a parameter, you will hear a soft bit of pink noise so that you can hear the effect of the adjustment you are making. You only hear pink noise while actually dragging a knob or control point in the filter response display.

Filter enable/disable: Turns the filter on or off.

How it works

The MasterWorks EQ operates like a standard EQ filter, but with much more sophisticated processing algorithms "under the hood". There are five bands of EQ, each with their own unique knob color, plus additional low pass and high pass filters. The five bands are labeled as follows:

Filter	Label
Low frequency	LF
Mid frequency	MF
High frequency	HF
Low-mid frequency	LMF
High-mid frequency	HMF

These labels, along with the position of the filters in the window, are merely conceptual guides. In fact, each filter can be set to any center frequency you wish.

Each filter can be independently turned on or off with the enable/disable button shown in Figure 1-11 on page 27. Each filter can be set to one of four different filter types (I, II, III or IV). The LMF and HMF filters provide an extra low and high shelf setting, in addition to the four standard band settings.

The additional low pass and high pass filters have gray cutoff frequency knobs and six settings for slope (in octaves/dB).

Frequency response display

The frequency response display at the top of the window displays the response curve of the current settings in the window. The (horizontal) frequency range is from 10 hertz to 20 KHz. The (vertical) amplitude scale is in dB and is adjustable between 3 and 24 dB using the four *dB scale* buttons (Figure 1-11 on page 27).

Showing and hiding filter curves

To view a filter in the display, turn on the filter. The shape of the filter, according to its current settings, is shaded in the same color as the filter's knob(s). For example, the MF filter is shaded in red, and the high pass filter is shaded in gray.

Use the *composite curve* and *filter curve* buttons (Figure 1-11 on page 27) to show or hide them in the display.

Adjusting filters in the display

Each filter has a handle, displayed as shown below in Figure 1-12 (in the filter's color), for adjusting its boost/cut and/or frequency:

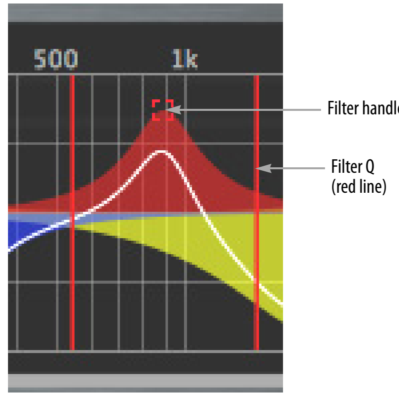


Figure 1-12: Drag the filter handle to adjust its frequency and/or boost/cut. Drag the Filter Q handles to adjust the Q.

For the EQ filters, when you click the handle, you'll also see lines on either side for adjusting the Q parameter, as shown above.

The FFT display

Use the FFT display button to show or hide the FFT display (Figure 1-13 below), which dynamically updates when signal is passed through the MasterWorks EQ plug-in. The shape of the FFT curve reflects any filtering being applied to the signal by MasterWorks EQ.

EQ filters

The EQ filters have three parameters:

Control	unit	range
Gain	dB	-20.00 to +20.00
Frequency	Hertz	10 Hz to 20 kHz
Q	n/a - see note below	0.64 to 16.00

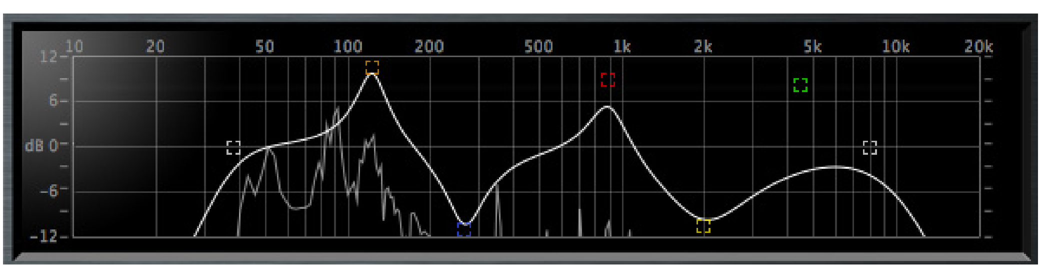


Figure 1-13: FFT display. This example is also showing the composite curve.

Q

The Q setting does not have a unit of measurement. Rather, it is the ratio of the filter's center frequency to the bandwidth of the filter. In addition, the actual Q value for the EQ curve being applied is dependent on three factors: the gain setting, the filter style, and the Q setting.

Filter types

Each filter can be independently set to one of four different filter types: I, II, III and IV. These, and the additional shelf filters for the LMF and HMF band, are discussed in the section "EQ filter styles".

Fine-tune adjustment

Hold down the Command key when turning a knob for fine-tuned adjustment of each parameter.

Returning to zero (or nominal frequency)

To return a knob to zero, or it's nominal frequency, double-click it.

EQ filter styles

EQ is one of the most widely used processing tools and can be applied to many different situations, from minor corrective tasks to highly creative applications. Over the years, many EQs have been engineered for specific applications or to achieve a certain sound. The MasterWorks EQ has been designed to be flexible enough to cover a broad range of applications. To that end, several different filter types are supplied, varying mostly in the way they handle the dynamic interaction between Gain and Q. This crucial relationship has been modeled to emulate the smooth and musical character of classic analog EQ circuits, in which the Gain/Q dependency was dictated by the actual circuit design and electrical components used.

The following sections describe the character of each type of EQ filter and their suggested applications. In the illustrations for each filter style (Figure 1-14 through Figure 1-17), the settings for the three example curves are the same for the purpose of comparison:

- Frequency = 1.00 kHz
- Q = 1
- Gain = +3, +10 and +20 dB

Type I

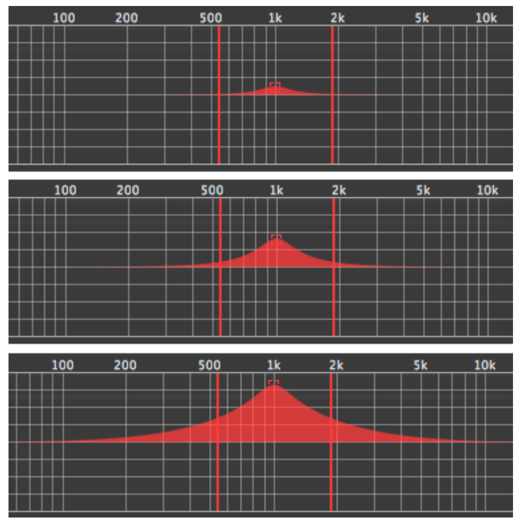


Figure 1-14: Type I EQ filter style.

The *Type I* EQ filter has the least amount of Gain/Q interaction, providing the most precision and control of all the EQ filter types. Even small adjustments in gain or reduction produce relatively high Q. This EQ style is best for situations that call for precise EQ adjustments requiring the maximum amount of individual parameter control. For more general shaping (e.g. full mixes) or subtle control (e.g. vocals), the other styles discussed in the following sections might be more appropriate. This filter type is the most similar to Performer Lite's standard parametric EQ.

Type II

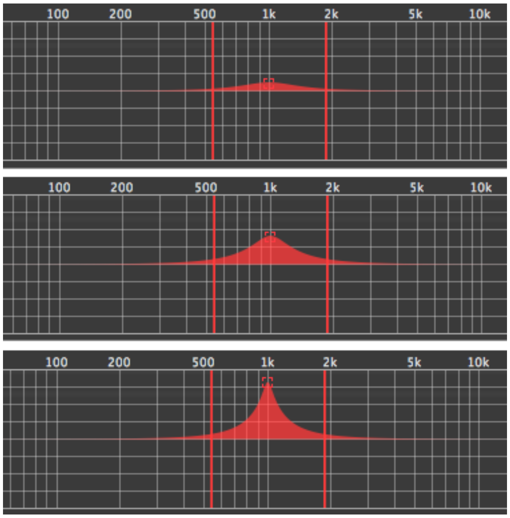


Figure 1-15: Type II EQ filter style.

The *Type II* EQ filter produces constant Q response during boost or cut. The Type II style emulates several classic legacy EQs and produces good results for resonance control on drums and percussion because it provides relatively high Q values with more extreme gain or cut settings.

Type III

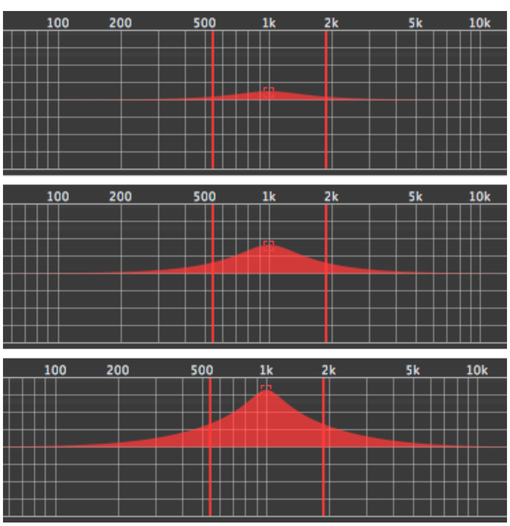


Figure 1-16: Type III EQ filter style.

The *Type III* EQ filter increases Q as boost is applied. Therefore, lower amounts of boost provide a softer, “wider” EQ effect (since the affected frequency range widens), whereas higher boost tends to sound louder and more “up front”, due to the increase in Q as the gain is increased. The more gentle Q curve at lower settings is well suited for overall EQ fills and more subtle corrections in instrument and vocal sources. Boosting or cutting by small amounts will seem to produce the effect that your ear expects, without the need to adjust Q. As a result, this filter style, and similar EQs with this characteristic behavior, are often referred to as being more “musical”. More specifically, this style emulates the classic Neve EQs, their modern derivatives and later SSL G series EQs. Many current popular outboard “boutique” EQs exhibit this same gain/Q relationship.

Type IV

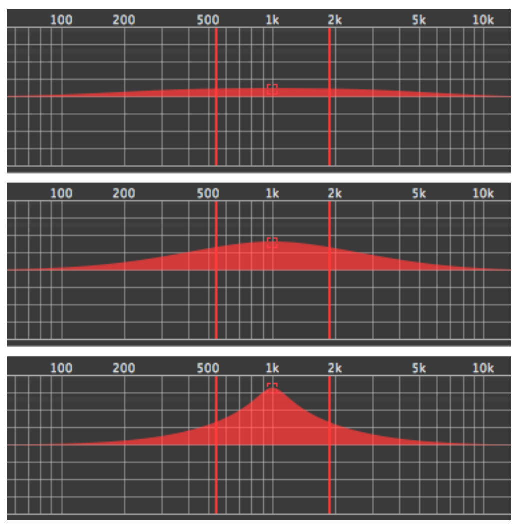


Figure 1-17: Type IV EQ filter style.

The *Type IV* EQ filter is a more extreme form of the Type III filter. It exhibits a high degree of interaction between Q and gain in order to maintain as closely as possible an equal amount of area under the response curve as gain is adjusted.

Type IV is the most gentle of the four EQ styles and is ideal for large scale EQ adjustments, especially on sub-mixes and complete mixes. This EQ style is also ideal for any applications where subtle changes in the overall character of the sound are desired. For example, it can be used for mastering applications, such as the overall adjustments that must often be applied to entire tracks to match other tracks on the album.

Shelf filters

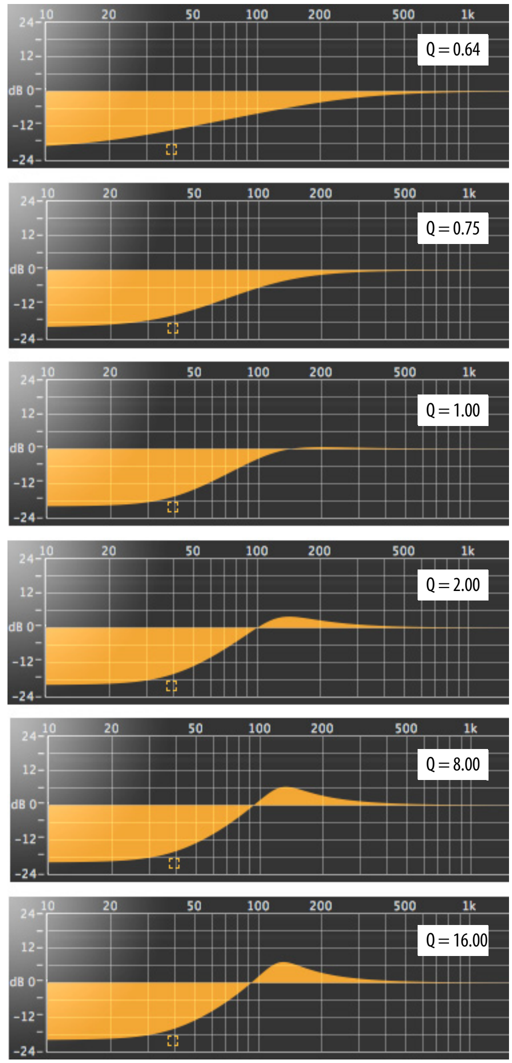


Figure 1-18: Shelf filter Q parameter overshoot.

When the LMF and HMF bands are set to their *shelf filter* setting (Figure 1-11 on page 27), the Q parameter controls the amount of overshoot applied to the response curve, as illustrated in Figure 1-18. When Q = 0.64 (the lowest setting), normal shelving is applied with no overshoot. This produces the response provided by a first order shelf. When Q = 0.83 (the default setting, not shown), the response corresponds to a second order shelf, still with no overshoot. This is the same response as Performer Lite's parametric EQ and many hardware EQs. In some situations, this form of accurate, clean shelving can sound harsh, especially when compared to legacy analog EQs. To soften the results, the overshoot is increased as Q is increased, as shown Figure 1-18 for Q values of 1.00, 2.00, 8.00 and 16.00. This overshoot region produces a boost in frequencies just above the cutoff, which compensates in a smooth, more pleasing fashion for the perceived drop in low frequencies being cut.

Conversely, when shelving boost is being applied, overshoot cuts frequencies just above the cutoff to again compensate in a smooth and pleasing fashion for the perceived boost in low frequencies:

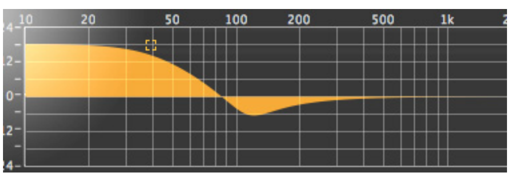


Figure 1-19: Overshoot when low shelf boost is applied.

Overshoot is also applied to high shelf boost and cut:

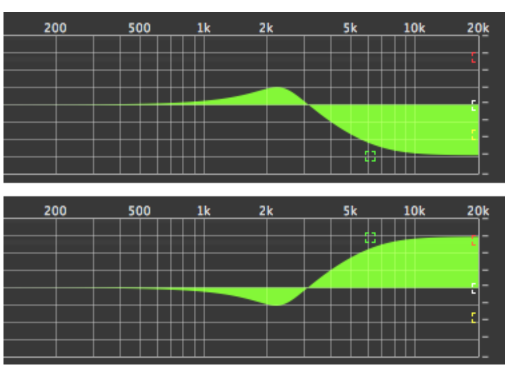


Figure 1-20: Overshoot when high shelf cut and boost is applied.

Overshoot tends to produce more of what one would expect to hear when applying shelving and is therefore considered to be more musical than shelving without overshoot. This effect, which has gained tremendous popularity among audio engineers, was first made popular in original Neve series EQs and later in the SSL G series.

At the maximum Q setting of 16.00, the overshoot region consists of half the total boosted (or cut) gain. For example, with a maximum gain setting of +20dB, the loss in the overshoot region is -10 dB.

Overshoot curves are symmetrical for both cut and boost.

Low pass and high pass filters

MasterWork EQ's low pass and high pass filters are similar to those found in Performer Lite's parametric EQ (which has a fixed slope of 12 dB per octave), except that MasterWorks EQ provides six different slope (roll off) settings: 6, 12, 18, 24, 30 and 36 dB per octave. This control over the shape of the "knee" gives you a great deal of flexibility and control for a wide variety of applications.

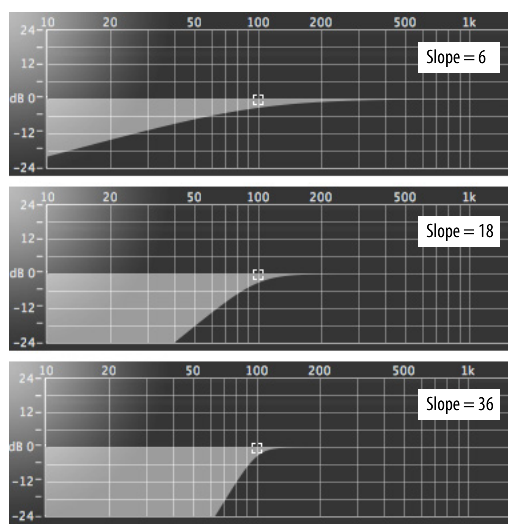


Figure 1-21: The low pass filter with three example slope settings.

MASTERWORKS GATE

The MasterWorks Gate provides sophisticated gating effects with simple to use graphic controls.

M-M	M-S	M-n	S-S	S-n	n-n
yes	no	no	yes	no	no

☛ Hold down the Option key while adjusting a parameter to program both sides of the stereo gate.

Controls

Input: boosts or attenuates the signal before going into the gate.

Threshold: sets the point at which the gate is triggered. Lowering the threshold causes the gate to be opened by a lower level signal. Threshold can also be set graphically by dragging the red control point at the bottom of the ratio graph.

Range: sets the range of attenuation applied to the signal when the gate is closed. A setting of -inf causes the gate to close completely, which results in no signal being passed. Higher settings attenuate or 'duck' the signal. Range is displayed in the ratio graph as a diagonal line below the threshold point.

Attack: determines how quickly the gate opens after the signal has crossed the threshold.

Hold: determines how long the gate stays open after the signal has descended below the threshold.

Decay: determines how quickly the gate will close down after the hold period has elapsed.

Status lights: indicate current gate envelope status. Red=closed, green=open and yellow indicates the hold segment of the envelope is active.

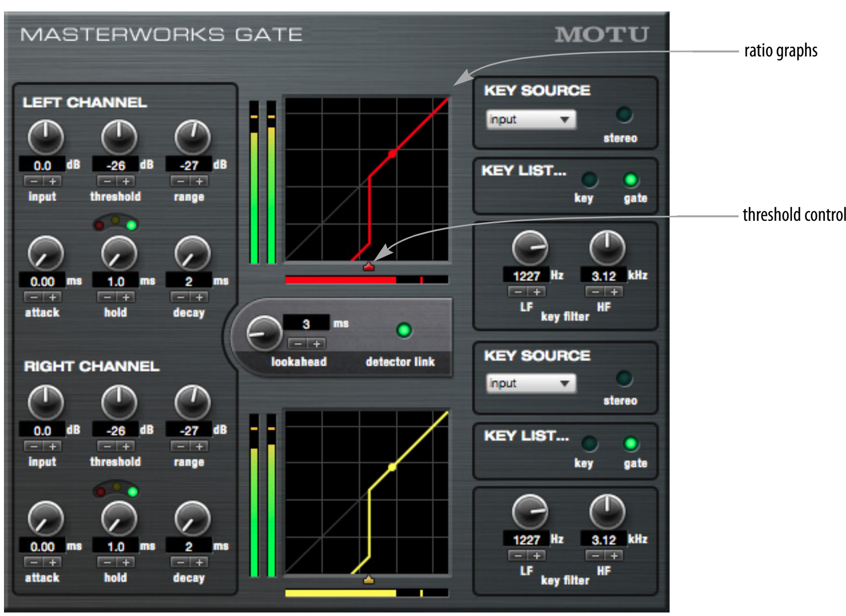


Figure 1-22: The MasterWorks Gate plug-in.

Look-ahead: by looking for transients before they occur, the MasterWorks Gate can open the gate before the signal, thus preserving the attack transient.

The look-ahead parameter introduces a delay into the signal path for analysis purposes. Look-ahead analyzes the non-delayed signal and it tells the gate when to open. You can avoid the look-ahead delay by setting look-ahead to zero or compensate by nudging the track ahead in the sequence editor by the exact amount of look-ahead you specify.

Keying

MasterWorks Gate can be keyed three ways: from the input signal, from a sidechain bus or by a MIDI signal.

The MasterWorks Gate publishes itself as a MIDI device to Performer Lite (and other CoreMIDI-compatible software). In Performer Lite, you can trigger MasterWorks Gate with MIDI notes. With automatable envelope parameters and MIDI triggering, the MasterWorks Gate works like a VCA on a modular synthesizer. When teamed with Multimode Filter, you now have a very flexible signal processing environment.

Key Filters

The key filters can be used to isolate a specific frequency range of the key signal. This can be useful for preventing false triggering. The default setting of the key filter permits all frequencies to pass.

Key Listen

In Key Listen mode, MasterWorks Gate outputs the key signal instead of the gated input signal.

Key listening is a valuable tool for tuning the gate. If your key signal is also the input signal, key listen allows you to audition the key filters to isolate a frequency range. If your key signal is a side chain bus, key listen will allow you to hear the sidechain input. If you are keying off a MIDI signal, MasterWorks Gate will play a tone representing the MIDI trigger.

Stereo: when stereo is enabled, the key source is a stereo bus.

Figure 1-23: The MasterWorks Gate mono version.

MASTERWORKS LIMITER

The MasterWorks Limiter maximizes the apparent volume within the available dynamic range by reducing the distance between average program levels and their associated peaks.

M-M	M-S	M-n	S-S	S-n	n-n
yes	no	no	yes	no	yes

MasterWorks Limiter allows for precise control of dynamics and output level of digital audio signals. It also includes quantization and dithering to preserve fidelity when changing bit resolution of digital audio samples.

MasterWorks Limiter processes audio at 64-bit floating point resolution, which provides the highest possible fidelity with no added noise. This means you can preserve the fidelity of your 24-bit audio files. In addition, MasterWorks will dither your final mix to any resolution you require, from 24-bit resolution for DVD to 8-bit resolution for internet and multimedia material. You can also apply unconventional bit depths for Special effects. (Ever tried 5-bit audio?) MasterWorks will maximize a signal for any available output resolution.

MasterWorks is designed primarily to process your final mix output, but it is also very useful for submixes. For example, use MasterWorks on your drum submix to fatten up its sound.

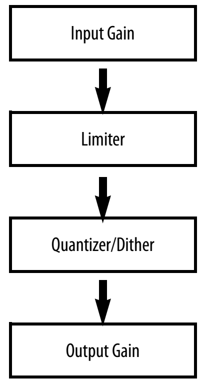


Figure 1-25: MasterWorks Limiter signal flow.

How it works

The MasterWorks Limiter is a two part plug-in. It starts with a sophisticated level control section, then adds the ability to quantize and dither samples for the purpose of changing bit resolution.

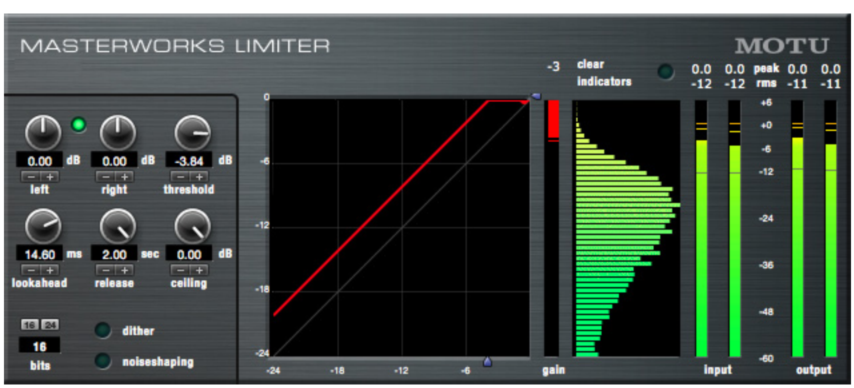


Figure 1-24: The MasterWorks Limiter plug-in.

Digital audio samples have a fixed dynamic range. Unlike analog media such as tape or vinyl, digital audio cannot be any louder than "0" VU. When mixing for digital media, it is important to keep the program material within the available dynamic range and below digital zero.

A related process, called *normalization*, finds the loudest sample in a region and increases its gain to the maximum available amplitude. All other samples in the region are increased by the same amount. The problem with normalization is that one loud spike in the mix will cause the rest of the mix to be at an apparently low level. Limiting allows you to increase the overall gain of the signal while limiting only signals that exceed "digital zero". Limiting is "brick wall", in that signals cannot be any louder than the set ceiling. As the average signal level is increased, and peaks are limited, there is less and less dynamic range to the program material. Too much limiting will sound very unnatural. Just the right amount of maximization will make the program material seem loud without ever exceeding the available dynamic range.

Changing bit depth with dither

Bit depth or *resolution* is the number of ones and zeros used to describe a digital audio sample. Commercial CDs use 16-bit samples. MOTU software has the ability to record 16 bit, 24 bit, or 32 bit floating point samples. Under some circumstances, it may be desirable to have lower sample resolution, such as 8 bit web audio or 12 bit sampler playback.

It is possible to change the bit depth of digital audio. Reducing the bit depth adds quantization distortion. To overcome this, *dither* can be applied. Dither is a small bit of noise that is applied the digital audio as it is quantized. This actually reduces the amount of distortion introduced by quantization, and has a more pleasing sound, particularly for softer passages in material with a

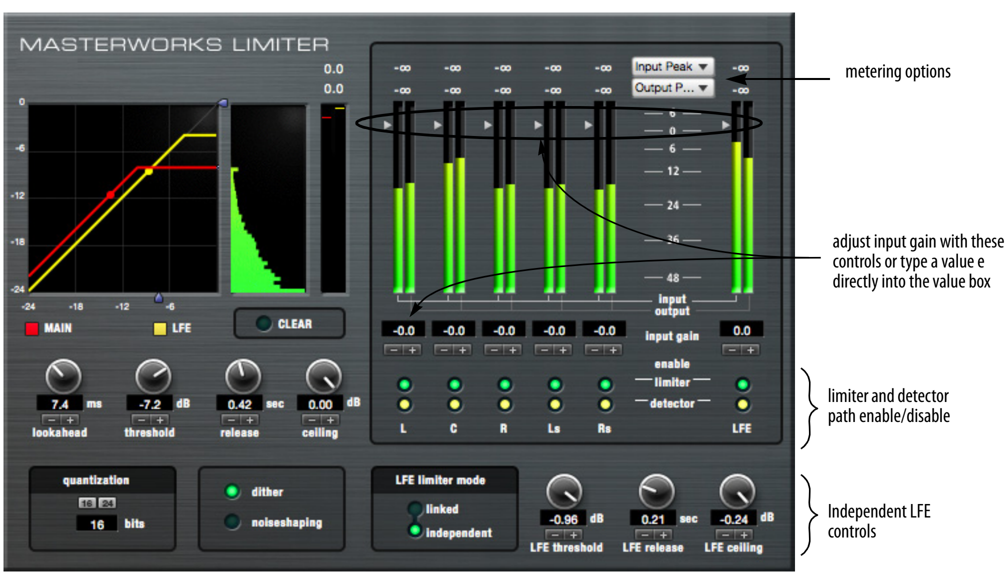


Figure 1-26: MasterWorks Limiter Surround Edition.

wide dynamic range. The disadvantage of adding dither is an increased noise floor. *Noise-shaping* spectrally shapes the dither noise so that it is less noticeable.

Controls

Input Gain L/R: Boosts or attenuates the signal going into the limiter.

Link: Links left and right inputs. If left and right gain controls are set to different values, and then the link button is engaged, the relative difference between left and right is maintained while linked.

Threshold: Sets the point at which the limiter starts to affect the input signal. Lowering the threshold setting has the apparent effect of raising the output gain of the signal. What is actually happening is that the average level of the signal is increased while the limiter assures that the signal will not exceed the ceiling.

Threshold can be set three ways:

1 with the Threshold knob (and associated value box directly beneath)

2 with the slider on the compressor's input meter

3 with the handle located at the bottom of the input/output ratio graph.

Clear: Clears peak settings from the meters. Peak meters show the loudest that the signal has been detected.

RMS: Shows the average level of a signal. Since the average level of a mix is usually well below any maximum peaks, it is useful to know what the average level is as well as the loudest points. This is very helpful when matching the apparent levels of more than one mix.

Lookahead: Senses program material that exceeds the threshold before it actually occurs. By looking ahead for peaks, the Limiter can be ready for attenuation before the peak, giving a very smooth limiter attack.

The lookahead parameter introduces a delay into the signal path for analysis purposes. Lookahead analyzes the non-delayed signal and it tells the limiter how to process the delayed signal. This will help provide a more natural, smooth sounding final output. The more lookahead you specify, the smoother the limiter will sound. Be aware that if Performer Lite's automatic plug-in delay compensation is not enabled and you combine the output of the lookahead limiter with the original signal, you will perceive a delay that directly corresponds to the amount you specified in the lookahead parameter box. In general, the lookahead should be short compared to the release.

Release: Controls how long it takes for the limiter to recover from attenuating the signal after the signal has dropped below the threshold.

Ceiling: Sets the maximum output of the limiter

Quantizer/16/24 Buttons: Easily sets output quantization. Other bit resolutions can be set by typing the number into the value box.

Dither: The final output can be dithered to any resolution from 24 bits to 1 bit. Dithering applies very small amounts of noise to the signal to reduce the quantization distortion that occurs at low levels.

Noiseshaping: Changes the characteristic of the dither noise, making it less noticeable. But don't use noiseshaping if you plan to process your output further, such as in a sampler.

Presets

The MasterWorks Limiter presets represent common applications. If you are changing bit depth, you'll see presets to go from 16 bits to 24 bits. For the bit depth changes, dither is part of the preset. If you are not changing bit resolution, don't add dither.

Compression and limiting are subjective effects. Sometimes you may want to hear the effect of the limiter. Other times, you may want a more subtle effect that is “invisible” and just catches the audio peaks. Input gain will have a strong bearing on what the MasterWorks Limiter actually does to the signal. Experiment with input gain and threshold to get the sounds that are most appropriate.

Masterworks Limiter (n-n channels)

For surround mastering, the Masterworks Limiter Surround Edition (Figure 1-26 on page 37) provides an independent LFE channel and comprehensive control over the detector path.

Metering

There are two slots for displaying peak and RMS values on input or output on every channel. You can set which values you are looking at by using the menus above the meter value bar.

The detector path

In most limiters, the detector path and the input are the same thing. How the limiter reacts depends entirely on the input signal. Surround applications may require a bit more flexibility and that is why the detector path has been decoupled from the input. MasterWorks Limiter allows you to determine which inputs get sent to the detector path. In addition, you can exempt channels from the action of the limiter. For example, if you want to limit the left and right channels based on dialog in the center channel, you would enable only the center channel to the detector path and enable the limiter on the left and right channels.

LFE limiter mode

In linked mode, the LFE channel is treated like any surround channel, that is, the limiting of the limiter will be determined by the main detector path. In independent mode, the detector path for the LFE is unlinked from the other channels and operates independently on its own input. Three additional controls appear in independent mode: LFE Threshold, LFE Release and LFE ceiling. These controls work exactly like their counterparts in the main section of the limiter, but operate only on the LFE signal. The independent LFE controls have their own color coded display in the ratio graph and limiting display. Look-ahead still determines the Look-ahead (if any) on all channels regardless of the status of the LFE limiter mode.

PATTERN GATE

M-M	M-S	M-n	S-S	S-n	n-n
yes	no	no	yes	no	yes

The *Pattern Gate* (Figure 1-27) slices up the audio signal passing through it into pulses determined by the *Speed* menu, which displays metric divisions locked to the tempo of the sequence.

The pattern gate can be applied to just about any sound that sustains. Remember, however, that the tempo of the sequence plays an important role in the results.

Pulse shape

The shape of each pulse is determined by the *Pulse Shape* graph (Figure 1-27), which represents 100% of the length of each pulse. Drag the handles to modify pulse *Depth*, *Attack*, *Sustain* and *Decay*. Or edit the numeric values below the graph. Drag the *Depth* handle vertically to soften the gate, such as for a tremolo effect.

Pattern and Length

The pattern itself is determined by the *Pattern* LED strip (Figure 1-27): click each pulse to toggle it on or off. Set the *Length* of the pattern (from 1-16 pulses) by dragging the *Length* handle.

Pattern Gate LFO

The Pattern Gate *LFO* (Figure 1-27) can be used to modulate the four pulse shape parameters (*Depth*, *Attack*, *Sustain* and *Decay*). Choose the desired LFO waveform (sine, sawtooth or rectangle) from the menu provided and set the desired *Symmetry* between 0 and 100, where 50 is normal symmetry. Then choose the desired *Period*, which is expressed in a number of pattern gate steps. The range is 0-256 steps. For example, if you choose a period of 110 steps, then the LFO will complete one cycle in 110 pattern gate steps.

After you've set up the Pattern Gate as desired, apply it to the desired pulse shape parameters using the LFO menus below each parameter. The range for the LFO setting is -100 to +100, where zero applies no LFO effect at all. +100 modulates the setting from its current value all the way to the maximum possible setting. -100 modulates the setting from its current value all the way to the minimum possible setting.

Swing

When the *Swing* parameter (Figure 1-27) is set to zero, the pattern gate plays in straight time (no swing). Other settings are as follows:

Swing amount	Ratio	Feel at 8th note speed
0	1 to 1	Straight 8ths
100	2 to 1	Triplet 8ths
125	2.5 to 1	Hard 8th swing
150	3 to 1	Hard 8th shuffle

Negative values invert the ratio, which moves the swung note closer to the base note.

Figure 1-27: The Pattern Gate plug-in.

PROVERB

M-M	M-S	M-n	S-S	S-n	n-n
yes	yes	yes	yes	yes	yes

ProVerb (Figure 1-28) is a convolution processor designed to be used as a reverb unit. Many factory-supplied *impulse responses* (sampled spaces) are provided, and you can load any audio file as an impulse response from your computer desktop, as long as it is in a standard audio file format (such as WAVE, AIFF, SDII, etc.).

The processing units model a matrix of mono reverb units on a dedicated bus, along with a highly configurable signal routing matrix designed to meet the needs of music production, foley and dialogue replacement. Additional features include a four-band graphic EQ and a dynamic mixing feature (ducker).

Choosing an impulse response

Choose the desired impulse response (acoustic space) from the Impulse Response menu.

This menu lets you choose both factory-supplied impulse responses and ones you have added to your library, and it operates similarly to Model 12's Instrument menu. See "Instrument menu" on page 74.

Only impulse responses that match the current channel configuration will be shown. For example, when using the stereo-to-stereo variant of ProVerb, mono-to-mono and 5.1 surround impulse responses will not be shown.

Next/previous IR buttons

Next/previous buttons appear to the right of the currently loaded impulse response. When the last or first impulse response is reached, the next/previous buttons will wrap around to the other end of the list.

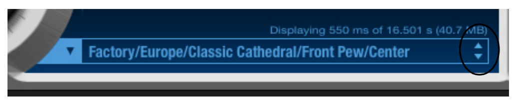


Figure 1-29: Next/previous impulse response arrows

Figure 1-28: ProVerb.

Shortcut menu

Right-click on the path of the current impulse response to display a shortcut menu containing the other impulse responses in the same submenu.

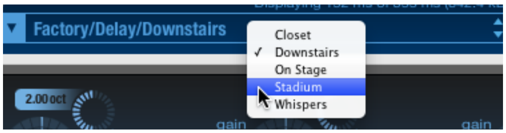


Figure 1-30: Impulse response shortcut menu

Importing your own impulse responses

ProVerb can use any standard audio file as an impulse response. Most standard audio file formats are supported. Simply drag the audio file, or a folder of audio files, from the computer desktop and drop it into ProVerb's waveform display (Figure 1-28).

You will then be prompted to select the installation location for the files. The files being imported can be copied to Shared, User, or Project, or any subfolder in one of those locations.

Figure 1-31: Importing impulse responses

While an import is in progress, a progress bar is shown . The import can be cancelled at any time with the Cancel button.

ProVerb impulse response management

ProVerb organizes impulse responses into four categories:

Category	Location on Mac
Factory	In the MOTU application support folder: /Library/Application Support/MOTU/ProVerb/ProVerb Data.bundle
User	In your user directory application support folder: User/Library/Application Support/MOTU/ProVerb/User
Shared	In the shared user directory application support folder: /Users/Shared/Library/Application Support/MOTU/ Plug-ins/ProVerb/Shared
Project	In the Performer Lite project folder: Project/Plug-in Data/ProVerb/Project/

Category	Location on Windows
Factory	C:\Program Data\MOTU\Plug-ins\ProVerb\ProVerb Data.bundle
User	C:\Users\[user name]\AppData\Roaming\MOTU\Plug-ins\ProVerb\User
Shared	C:\Program Data\MOTU\Plug-ins\ProVerb\Shared
Project	[DP project name]\Plug-in Data\ProVerb\Project

Mac OS X

The *Factory* impulse responses are always available. *User* impulse responses are stored in your user directory and are therefore available only when you are the currently logged in user. You can use the user directory to protect your own impulse response content. *Shared* impulse responses are stored in the system application support folder and can therefore be made available to any users that you wish to share them with. *Project* impulse responses are stored with the host software project itself. By keeping them with the rest of the files associated with the project, you don't have to worry about losing them when exchanging the project with a colleague, archiving the project, transferring it, etc.

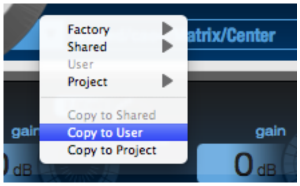


Figure 1-32: The Impulse Response menu.

To remove impulse responses from your ProVerb library:

1 Locate the desired category (Shared, User, or Project) in the Mac OS Finder or Windows desktop.

2 For Mac OS X: If you would like to remove the entire category, move the bundle to the Trash. If you would like to remove individual files or folders from that category, right-click the bundle file and choose *Show Package Contents* from the contextual menu. Select the desired folders and move them to the Trash.

For Windows: Go to Windows Explorer> Tools> Folder options > View tab and choose the *Show hidden files, folders and drivers* option under the Hidden files and folders category. To remove the entire category, move the folder to the Trash. To remove individual files or folders, browse and select the desired items and move them to the trash.

3 Choose *Reload All Trees* from ProVerb's Impulse Response Menu, and the impulse response library will update to reflect the changes.

Non-automated settings

The *Predelay*, *Damping*, and *Length* knobs (Figure 1-28) modify the impulse response during the loading process. Any change to these parameters necessitates a re-load of the impulse response. In conventional convolution processors, changing these parameters might take as long as 15-20 seconds to recalculate the impulse response. In ProVerb, however, these parameters have been optimized so that they can be adjusted smoothly in real time. They cannot, however, be automated.

Predelay

Predelay time-shifts the response function ± 99 milliseconds. Although this control is primarily intended to align the early reflections with the dry signal to eliminate comb filtering, it can also be used for artistic and sound design purposes.

Length

Length time-stretches the impulse response by resampling it. Use the length parameter to adjust the perceived size of the original space. Since resampling is applied, changing the length is also quite useful if you have a room response with a resonant frequency aligned with a frequency in the input signal: just resize the room slightly and the resonance will disappear — a technique that can be especially useful for kick drums and toms. This can be adjusted ± 4 times the original length.

Damping

Most impulse responses are recorded in empty rooms. The *Damping* parameter simulates the effects of hanging curtains, changing atmospheric conditions, or adding people. It is simply a low-pass filter that simulates the effect of distance on high frequencies, and the Damping setting controls the rate at which the cutoff frequency decreases with distance.

Automated Parameters

The rest of ProVerb's settings (below) can be fully automated.

Input/output routing

The input routing matrix is located just to the left of the Length knob. The *Input gain* and *Output gain* behave the same as those found on the front of a rackmount reverb processor patched into an effects loop. Use them to control the level of the signal before it enters — and after it exits — ProVerb, respectively.

Width

Width controls the stereo distribution of the input signal, and is thus disabled for the mono-mono variant.

In stereo-to-stereo mode, "hard left" maintains stereo routing and "hard right" feeds the left channel into the right input and the right input into the left channel. The dry signal is unaffected for the stereo variant.

In mono-to-stereo mode, *Width is replaced by Pan,* which sets the dry signal's lateral placement within the mix. The wet signals are unaffected.

Mix

Mix controls the ratio of wet signal to dry signal. Click the label above the knob readout to reveal a menu that lets you choose one of four options:

Mix option	What it does
3 dB cross-fader	Performs the mix using an equal-loudness cross-fader.
6 dB cross-fader	Same as above, but with an equal-amplitude cross-fader. This option produces the "hole - in-the-middle" effect, but it is not as likely to generate overs.
Effect level	Operates like the effect return level knob on a mixer. The dry signal gain remains constant, while the effected signal is fully attenuated at 0, not attenuated at 100%.
Dry level	The effect return remains constant, while the dry signal level is attenuated.

Four-band EQ

The four-band EQ is derived from MasterWorks EQ (page 27). The first and fourth knob sets provide low and high shelving filters, respectively. The two center knob sets are typical boost/cut parametric EQ bands. The EQ section follows the reverb unit in the signal chain, and does not affect the dry signal.

Dynamic Mixing (ducker)

The *Dynamic Mixing* section (Figure 1-28) follows the EQ in the signal chain and when enabled lowers the level of the reverb in the mix based on the level of the dry signal. Properly tuned, it allows a "wetter" mix while retaining intelligibility of the input signal. The signal chain model can be described as follows: a compressor after the reverb, but before the return, with its side chain input driven by the dry signal. The *Threshold* setting (*Thresh*) determines when the compressor kicks in, and *Sensitivity* (*Sens*) determines how strongly the compressor responds. The *Comp* knob defines the amount of gain reduction applied to the wet signal.

ProVerb metering

The *In* meters read the input to the convolution unit after the *Input gain* is applied. The *Out* meter is computed before the *Mix* stage. These meters indicate signal energy or loudness, with a peak indicator that reads the true peak value of the signal. A significant difference in the peak indicator position and meter level is therefore normal. Clip warning indicators will illuminate if any overs occur.

RXT

M-M	M-S	M-n	S-S	S-n	n-n
yes	yes	no	yes	no	no

RXT is an emulation of ProCo's The Rat™ distortion pedal.

Distortion: amount of distortion.

Filter: variable lowpass filter.

☛ This knob operates in reverse when compared to the other pedal plug-ins.

Volume: output gain.

Status light: displays the bypass/enabled state; when lit, the effect is active.

Enable switch: bypasses/enables the effect. This works the same as the Effect window's Bypass button.

SOLOIST

M-M	M-S	M-n	S-S	S-n	n-n
yes	yes	no	yes	no	no

The Soloist is a model of the two-channel Mesa™ *Dual Rectifier Solo Head*™ amplifier of the 1990's. It was modeled with the silicon diode rectifiers (which is characteristic of tighter attacks, added brightness, and substantially more headroom) and bold power settings enabled (which supplies full design voltage to all components).

Orange versus Red channels

The *Orange/Red* switch toggles between two distinctly different signal paths.

Orange channel gain

Normal: a vintage, high gain setting that produces a fatter distortion channel with emphasis in the low/low-mid frequencies.

Clean: a cleaner incarnation of the Orange channel that produces sweet, thick tones with only a mild distortion.

Channel cloning

Channel Cloning is a distinct design characteristic of Mesa engineering that allows you clone certain features of one channel and apply them to the other channel. The green middle light on the three-way cloning switch indicates that no channel cloning is taking place. In the original Mesa amp, when Channel Cloning is engaged, it switches on an alternate circuit within the channel it is applied to.

Org to modern: switches the Orange channel to the alternate circuit, which changes a few key features (such as *Presence*) to make the Orange channel sound more like the Red channel. Essentially, this is adding additional gain and higher frequencies to the Orange channel.

Red to vintage: switches the Red channel to the alternate circuit, which changes a few key features to make the Red channel sound more like the Orange channel. This rounds off some of the higher frequencies and additionally reduces gain, making it more of a "hot rhythm guitar" and less of an "extremely high gain lead guitar."

Controls

Master: controls the output level of each channel.

Presence: controls attack and brightness of the selected channel. This knob works in conjunction with the Bass, Mid, and Treble settings.

☛ In the Red vintage configuration, the Orange channel Presence knob is fully active. This is an undocumented feature of the original Mesa Dual Rectifier Solo Head.

Bass: controls the amount of low frequencies factored into the final output of the signal.

Mid: determines the amount of midrange mixed into the final output of the signal.

Treble: this is an extremely powerful control, as the sound leaving the Treble stage is what feeds the Mid and Bass stages.

Gain: the ultimate tone shaping knob (and first stage of the amp). This determines how hot your signal is before traveling to the rest of the circuitry.

TRIM

Trim (Figure 1-33 below) provides simple gain and attenuation. but it offers a number of useful additional features, including long-throw, precise metering with an adjustable range, a phase invert, and left/right panning in the stereo version.

M-M	M-S	M-n	S-S	S-n	n-n
yes	no	no	yes	no	yes

Trim uses very little processing power. Its main purpose is to be used at the end of a chain of effects or as part of the output stage for an auxiliary track, subgroup or master fader. Trim will help you control and monitor levels at all points of the mix.

Signal flow

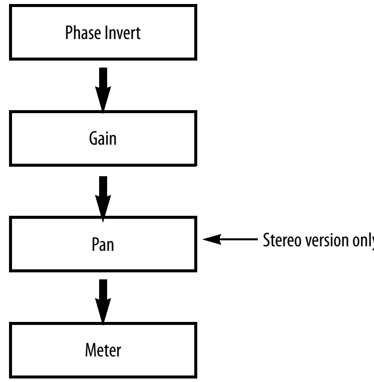


Figure 1-34: Trim plug-in signal flow.

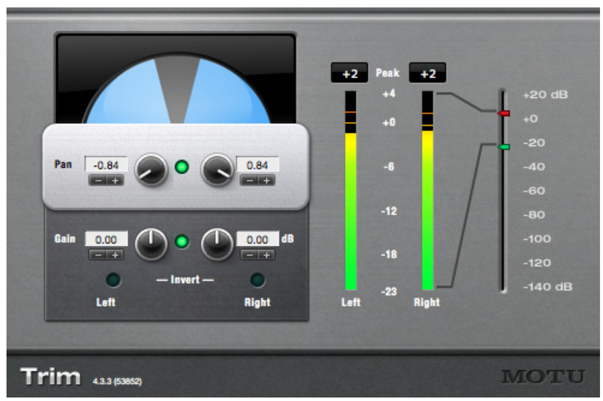


Figure 1-33: The mono, stereo and n-channel Trim plug-ins.

How it works

Trim can add up to 40 dB of gain or attenuation to a digital audio signal. The metering can be zoomed to show a close-up of any part of the signal range. The meter shows up to 20 dB over digital zero so you can see exactly how far you have exceeded the available dynamic range of the A/D converters or final 16 bit or 24 bit file. The meter shows real-time, average level, and peak/hold (click to clear).

The invert button inverts the phase of the signal by 180°. This can be useful if you have two mics recording a stereo image of a sound source. If the mics are out of phase or wired incorrectly, inverting the phase of one of the signals may improve the sound.

Controls

Gain: Provides -/+ 40db of gain.

Link: (Stereo only) Links the left and right gain controls. If left and right are set to different levels and then Link is engaged, the difference between left and right is maintained when either is changed.

Pan controls: (Stereo only) The pan controls give you separate pan control of the left and right channels, allowing you to restrict — or even completely swap — the left/right channels. Zero (0.00) is hard left, 1.00 is hard right and 0.5 is mono.

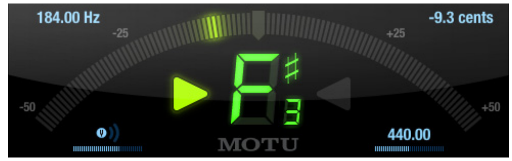
By default, the pan controls are linked (as indicated by the green link light between them). Use the button between them to disable linking for independent pan control of the left and right signals.

Invert: Inverts the phase of the input signal by 180°.

Peak: Indicates the highest peak since the peak indicator has been cleared. Click it to clear it.

Range Adjust: Allows you to zoom in on any part of the meter range.

TUNER



M-M	M-S	M-n	S-S	S-n	n-n
yes	yes	no	yes	no	no

The Tuner plug-in is an accurate and easy to use tuner.

Detected frequency: fundamental frequency of the incoming signal, in Hertz (Hz).

Detected note: note name and octave that correspond to the detected fundamental frequency.

Meter: representation of the pitch difference between the detected note and the detected fundamental frequency. The horizontal position of the illuminated segments indicates how far the detected frequency is from the detected note. The number of illuminated segments indicates uncertainty or inharmonicity in the signal; a greater number of illuminated segments represents greater uncertainty. The color of the segments changes gradually from green (in tune) to yellow, orange, and red (progressively further out of tune).

Meter value: difference between the detected note and the detected frequency, in cents.

Arrows: the direction in which the detected frequency needs to move to match the frequency of the detected note. The color of the arrows changes progressively in the same manner as the meter segments. When the detected fundamental frequency matches the detected note within three cents, both arrows will be illuminated.

Reference frequency: sets the frequency reference for the pitch A4, between 400 and 480 Hz. The default frequency is 440 Hz. The reference frequency can be adjusted in increments as small as 0.01 Hz when Command-dragging on the bar below the number.

Reference generator: generates a tone at the reference frequency. Click the tuning fork button to toggle the generator. Click and drag on the bar below it to adjust the output level of the generator.

Tuning stereo signals

When tuning a stereo signal, the Tuner plug-in analyzes the sum of the two channels. If the channels are not phase coherent, the Tuner plug-in may not be able to measure the frequency of the signal.

Part 2

Instruments

CHAPTER 2 Instrument Plug-ins

OVERVIEW

Performer Lite includes six instrument plug-ins: BassLine, PolySynth, Nanosampler, Modulo, Model 12 and Proton. The following sections discuss the settings for each individual instrument.

For general information about using virtual instruments, see chapter 12, “Instrument Tracks” (page 75) in the *DP User Guide*.

BassLine 53
PolySynth 54
Nanosampler 55
Modulo 62
Model 12 74
Proton 78
MIDI Control of instrument settings 81

BASSLINE

BassLine (Figure 2-1) is a monophonic, analog-modeled bass synth instrument plug-in.

Figure 2-1: BassLine.

Oscillator

BassLine provides a single oscillator that can produce a sawtooth wave, square wave or a hybrid combination of the two by turning the *Waveform* knob (Figure 2-1) to a position somewhere in between the highest and lowest setting.

Use the *Range* knob to specify the octave. As was often the case on vintage analog synths, the three settings (8, 16 and 32) refer to the length (in feet) of organ pipes. A pipe twice as long produces notes an octave lower. *Detune* produces an increasingly fatter sound and widening stereo image.

Pitch modulation

Bend range (specified in half-steps from 1 to 12, where 12 is an octave) determines the response to pitch bend. *Glide* provides classic portamento from one note to the next.

Filter

This is a classic low-pass filter with *cutoff* frequency and filter *resonance* controls.

Filter modulation

The Filter Modulation section lets you apply a simple one-stage decay envelope to the filter frequency. *Decay* determines the speed of the envelope and *Envelope Amount* (*Env Amt*) determines the strength of the envelope applied to the filter. *Velocity* lets you control envelope amount via note-on velocity. A higher settings produces greater on-velocity response.

Amplifier

Control the output level of the instrument with the *Volume* knob (which is mapped to controller #7). *Overdrive* produces classic analog-style signal overload effect.

Amplifier modulation

Control the length of each note with *Decay*. Apply note-on velocity response with the *Velocity* knob.

Legato

BassLine is a monophonic instrument and can therefore only play one note at a time. When *Legato* mode is disabled, holding one note and then playing another note ends the first note and retriggers the amplitude envelope. When Legato is enabled, the transition to the second note is made with pitch glide from one to the other. In addition, the envelope is not retriggered, so the decay continues until all notes are released.

POLYSYNTH

PolySynth (Figure 2-2) is a polyphonic synthesizer inspired by the Roland Juno 106 and other one-oscillator analog synths from the 1980's.

DCO

The heart of PolySynth is the *DCO* (*Digitally Controlled Oscillator*) section. Here, you will find oscillator level controls for triangle, sawtooth, rectangle, sub-frequency 1 and sub-frequency 2 oscillators. You can also add *Noise* and adjust stereo *Detune*. Use the *Range* buttons to specify the octave. *Bend* range (specified in half-steps from 1 to 12, where 12 is an octave) determines the response to pitch bend.

LFO

The *LFO* (*Low Frequency Oscillator*) section allows you to apply periodic modulation effects. *Speed* controls LFO rate. *PWM* (*Pulse Width Modulation*) uses the LFO to change the shape of the rectangle and sub-frequency oscillators, resulting in cyclic timbral changes to the sound. The *Vibrato/Wah* slider uses the LFO to either modulate pitch for a vibrato effect or modulate the resonant low-pass filter for the classic "wah" sound.

Filter

This is a classic low-pass filter with *cutoff frequency* (*FREQ*) and filter *resonance* (*RES*) controls. Apply note-on velocity response with the *velocity* (*VEL*) control. Apply the ADSR envelope (or its inverse) to the filter with the *envelope* (*ENV*) control. *Key Tracking* makes the filter cutoff frequency change relative to the root frequency of the note being played. Full (3/3) tracking produces a full range key tracking over the entire keyboard, where the 2/3 and 1/3 settings reduce the tracking effect by scaling the change proportionally as you move up the keyboard.

Envelope (ENV)

The *ADSR* envelope (*Attack, Decay, Sustain, and Release*) controls the loudness of each note, and it can also modulate the filter. The Attack, Decay, and Release parameters are rate or time controls. Sustain controls amplitude (level). When a note is first triggered, the envelope generator will begin to rise to its full level at the rate set by the Attack. When it reaches the peak level it then falls at the rate set by the Decay parameter to the level set by the Sustain control. The envelope will remain at

Figure 2-2: PolySynth.

the sustain level as long as the key is held down. When a key is released, it will return to zero at the rate set by the Release parameter.

Amplifier (AMP)

Control the output level of the instrument with the *Volume* slider (which is mapped to controller #7). *Velocity (VEL)* controls note-on velocity sensitivity. For example, if you set Velocity to zero, all notes will play at the same volume. Higher values cause notes played softly to sound at a low volume and notes played hard to sound at a higher volume.

Effects

Add *Chorus* and *Distortion* (*DIST*) to the sound as desired.

NANOSAMPLER

Nanosampler (Figure 2-3) is a sample player instrument. You can load a mono or stereo audio file of any length into one instance of Nanosampler and then play the sample, or individual slices of it, from a MIDI track or a MIDI controller. For longer samples, Nanosampler plays the first 20 seconds. Only one sample can be loaded at a time, but you can create and operate as many instances of Nanosampler as you need (and your computer processing resources allow).

Sample menu

The *Sample menu* (Figure 2-3) displays the currently loaded sample. Click to access the menu items explained below. These features work the same way as described for Model 12 in “Model 12 sample management” on page 77. You can assemble your own Nanosampler library by building it from your computer desktop, as explained in “Building your user library” on page 77.

Factory: This sub-menu provides access to Performer’s included (factory) samples, which offer a broad, basic palette of instrument sounds for you to work with.

Shared: Use the *Copy To* menu command (see below) to copy a sample to this Shared sub-menu. *Shared* samples are stored in the system Library Application Support folder and can therefore be made available to any users that you wish to share them with. If this menu item is grayed out, there are no Shared samples yet.

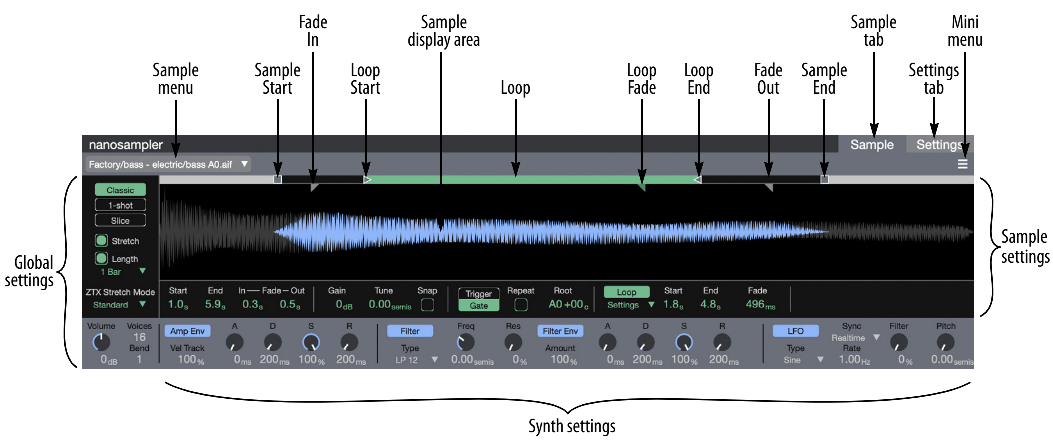


Figure 2-3: Nanosampler.

User: Use the *Copy To* command (see below) to copy a sample to this User sub-menu. *User* samples are stored in your user directory and are therefore available only when you are currently logged in to your computer with your user account. You can use the user directory to protect your own sample content. If this menu item is grayed out, there are no User samples yet.

Project: *Project* samples are stored with the Performer project itself. By keeping them with the rest of the files associated with the project, you don't have to worry about losing them when exchanging the project with a colleague, archiving the project, transferring it, etc.

Open: Lets you load a sample from your hard drive.

Reveal In Finder/Explorer: Displays the audio file on the computer desktop.

Copy To: Lets you copy the current sample to the Shared, User or Project sub-menus.

Loading a sample using drag and drop

To load a sample into Nanosampler via drag and drop, drag a mono or stereo sample from your computer desktop, the Soundbites list, a clippings window or other draggable source into the Nanosampler sample display area (Figure 2-3).

Global settings

Once a sample is loaded, the global settings (Figure 2-3) determine the overall operation of Nanosampler.

Classic mode

In *Classic* mode (Figure 2-3), Nanosampler acts like a traditional sampler, where the sample plays back at different pitches depending on the key pressed, with either Gate or Trigger playback (see "Gate/Trigger (Classic and Slice modes only)" on page 59).

1-shot

1-shot mode (Figure 2-4) is similar to Classic, except that the sample plays in Trigger mode only, and polyphony is locked at one voice. All Trigger, Gate and Loop options are disabled. The result is that the sample plays in its entirety as soon as it is triggered, unless it gets retriggered, which causes it to play from the beginning again.

Slice

In *Slice* mode (Figure 2-5), the sample is divided into individual slices triggered by a their own MIDI note. Unlike with Classic and 1-shot modes, the MIDI note does not affect the pitch of the slice but instead triggers the audio at its original pitch.

For more information about Slice mode, see "Editing slices in the waveform display" on page 58 and "Slice mode settings" on page 59.

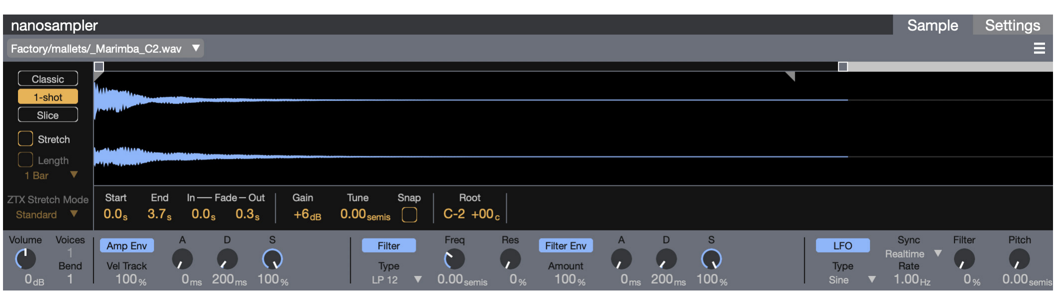


Figure 2-4: 1-shot mode.

Stretch

When *Stretch* (Figure 2-3 on page 55) is enabled, the sample will maintain its original duration at any pitch that you play.

When Stretch is disabled, the sample length becomes proportionally shorter or longer, respectively, as you go up or down from the root pitch.

Length

When Stretch (above) is enabled, the *Length* option becomes available (Figure 2-3 on page 55) and lets you choose from the menu a specific metric length for the sample (at all pitches). For example, if you choose 1 bar, the sample will become one measure long at whatever the sequence tempo is when the sample is triggered. This option is commonly used for drum loops and other rhythmic material.

ZTX Stretch mode

When using the Stretch option (above), use the *ZTX Stretch Mode* menu (Figure 2-3 on page 55) to choose the desired time-stretching algorithm. *Standard* produces more of the conventional “sampler” effect, while *Formant-corrected* preserves the tonal characteristics at different pitches. For details, see “ZTX Formant-Corrected” on page 943 in the *DP4 User Guide* and “ZTX Standard” on page 944 in the *DP4 User Guide*.

Volume

Control the output level of Nanosampler with the *Volume* knob (Figure 2-3 on page 55), which is mapped to controller #7.

Voices

The *Voices* setting (Figure 2-3 on page 55) lets you choose how many notes you can play at one time. For example, if you are playing a bass sample, you can probably set the polyphony to 2, as you’ll likely never play more than two bass notes at the same time (perhaps only when they overlap a little bit from one to the next). On the other hand, if you

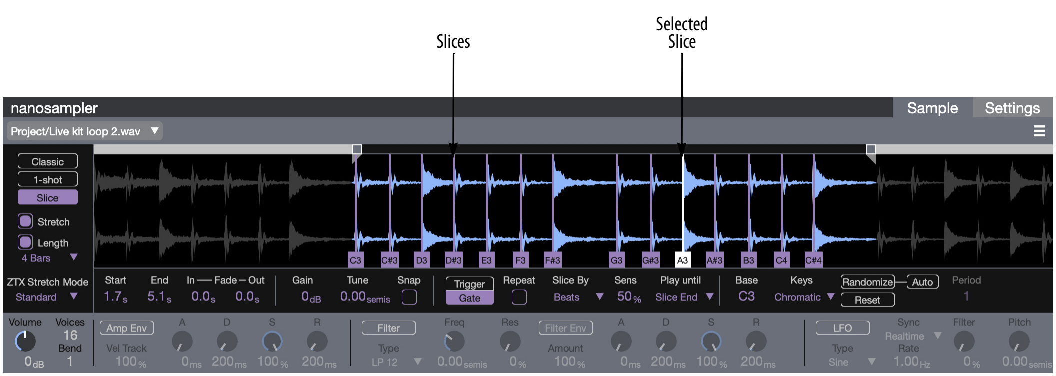


Figure 2-5: Slice mode.

are playing a keyboard pad sample, you might want as much as 10 or 12, depending on the length of the decay of the notes, the music you will be playing, and any other factor that might impact how many notes are sounding at one time.

Bend

Bend range (Figure 2-3 on page 55) is specified in half-steps from 1 to 12, where 12 is an octave, and determines Nanosampler's response to pitch bend.

Sample tab

The Sample tab (Figure 2-3 on page 55) displays the waveform and sample settings.

Waveform display

The waveform display (Figure 2-3 on page 55) shows the currently loaded sample. Right-click (or Ctrl/Win-click) to quickly access the mini-menu commands (see "Mini-menu" below).

Editing slices in the waveform display

In Slice mode (Figure 2-5 on page 57), use the following techniques to edit slices:

To do this	Do this
To select a single slice	Click its stem.
To select an adjacent slice	Press the left/right arrow key.
To select multiple adjacent slices	Shift-click them.
To select non-adjacent slices	Command/Ctrl-click them.
To remove one or more slices	Select them and press delete.
To add a slice	Double-click on the waveform.
To move a slice	Drag it.
To audition a slice	Click its note name. The slice plays according to the sample settings (such as trigger/gate).

Also see "Slice" on page 56 and "Slice mode settings" on page 59.

Mini-menu

The Nanosampler mini-menu (Figure 2-3 on page 55) has the following commands:

Reverse Sample: Plays the sample backwards.

Normalize Sample: Boosts the level of the entire sample by whatever amount is needed to make the loudest peak reach digital full scale (zero dB).

Show Separate Channels/Mono Sum: If the sample material consists of stereo audio, these two commands let you view it as separate side-by-side channels or as a single waveform that is a mono sum of the two channels.

Sample settings

The sample settings (Figure 2-3 on page 55) are displayed below the waveform, where they can be edited numerically. There are graphic handles above the waveform (also shown in Figure 2-3) for basic settings such as sample start/end, fade in/out, loop start/end, etc.

Start/End

If you would like to specify sample start/end times that differ from the very beginning and end of the sample, you can set them graphically in the LCD display or by numerically adjusting the sample *start* and *end* settings (Figure 2-3).

Fade In/Out

The *Fade In* and *Fade Out* points (Figure 2-3) are set relative to the Sample Start and End times and allow for a smooth fade of the desired length at the start or end of the sample.

Gain

Use *Gain* (Figure 2-3) to adjust the overall level of the sample in dB. Positive values raise the level; negative values lower it.

Tune

The *Tune* setting (Figure 2-3) lets you adjust the pitch of the note up or down. The adjustment is specified in semitones and cents (a hundredth of a semitone), so it can be very accurate. To adjust down, enter a negative number.

Snap
When *Snap* is enabled (Figure 2-3), the sample start/end, loop start/end and slice start points will snap to the best value in the local range to prevent clicks.

Root (Classic and 1-shot modes only)
The *Root* setting (Figure 2-3) determines which MIDI note (and therefore which key on your MIDI keyboard) triggers the original sample. Adjusting it lets you shift the root pitch of the sample up or down on the keyboard.

Gate/Trigger (Classic and Slice modes only)
In Classic mode, you can choose between *Gate* or *Trigger* playback (Figure 2-3). Gate stops playback as soon as a MIDI key is released (after the amplitude envelope release finishes). Trigger ignores note-off messages and always plays until the end of the sample, or until the sample is retriggered.

Repeat (Gate mode only)
With Gate enabled, the *Repeat* option (Figure 2-3) causes sample playback to automatically play again after completing a full cycle, repeating the full cycle over and over for as long as the note is held. In Slice mode, each slice will repeat individually for as long as its note is held.

Loop (Gate mode only)
If you would like the sample to sustain when you hold down keys, enable the *Loop* button (Figure 2-3). You can then set the loop Start and End points, either numerically or graphically using the handles above the waveform display, as shown in (Figure 2-3).

Use the *Fade* control (Figure 2-3) to create a smooth transition from the loop end to the loop start. You can save the loop in the audio file itself using the *Save loop in sample* command in the *Settings* menu. You can load a previously saved loop from the audio file using the *Load loop from sample* command in the same menu.

Slice mode settings

The settings in this section are available in Slice mode only (Figure 2-5 on page 57). Also see “Slice” on page 56 and “Editing slices in the waveform display” on page 58.

Slice By
Use the *Slice By* menu (Figure 2-5) to choose how slice locations are determined, as follows:

Slice By	Setting	Explanation
Beats	Sensitivity	Automatic beat detection finds slices by searching for transients in the audio. Adjust the *Sensitivity* setting to find more or fewer slices.
Mensural	Length	Slices are placed in even, metric increments determined by the *Length* setting (16th note, 8th note, etc.)
Division	Number	Slices are placed in equal increments, according to the *Number* specified.
Manual	(None)	Slices are placed manually. See “Editing slices in the waveform display” on page 58.

If you edit slices while in Beats, Mensural or Division mode, the Slice By mode switches to Manual.

Play Until

When a slice is playing, the point it plays until depends on the *Play Until* setting (Figure 2-5), which has three options:

Play Until	Explanation
Slice End	The slice plays.
Sample End	Playback begins at the start of the slice and proceeds all the way to the end of the sample.
Final Note	The slice plays, and then each subsequent slice will play sequentially. (See below for more info.)

Final Note

The *Final Note* setting plays slices sequentially in order of their assigned MIDI note. This produces the same result as the play until *Sample End* mode when the slices have the default ascending order. However, a completely different result can be obtained when the slice order is randomized (see below).

Base (Base Note)

The *Base* (Base Note) option (Figure 2-5) lets you specify which MIDI key triggers the first slice. Subsequent slices are assigned to consecutive notes, according to the setting in the *Keys* menu: *Chromatic*, *White* or *Black* keys.

Randomize

The *Randomize* button (Figure 2-5) randomizes the order of the MIDI notes assigned to the existing slices. No slices are added or removed; only the existing slices are shuffled. Clicking the Randomize button is a 'one-shot' operation: the result remains as is, unless you click the button again. (However, see "Auto Randomize" below.)

☛ If you would like to limit the randomization to only certain slices, select them before clicking the Randomize button. For details about selecting slices, see "Editing slices in the waveform display" on page 58.

To get back to the default ascending MIDI note ordering (starting from the base note), click the *Reset* button.

Auto Randomize

The *Auto* button connected to the Randomize button (Figure 2-5) automatically randomizes the slices after playback cycles through the number of periods specified by the *Period* setting. A new randomization occurs after each period reset.

For example, consider the following setup:

Sample setting	Set to
Repeat	Enabled
Gate/Trigger	Gate
Play Until	Slice End
Randomize	Auto
Period	2

With the above settings, if you hold down a MIDI note, a slice will repeat two times before the randomization resets and then another random slice will repeat twice, and so on, until you release the key.

The *Play Until Final Note* setting is a lot of fun with Auto randomization because the sequence of slices will shuffle each time you complete a full note/slice sequence.

The Settings tab

The *Settings* tab (Figure 2-6 on page 61) provides graphic control of the Envelope, Filter and LFO settings. The settings across the bottom of the Nanosampler window can be accessed even when the Settings tab is hidden.

Amplitude Envelope

To apply the *Amplitude Envelope* (Figure 2-6), engage its button. Doing so applies a standard ADSR envelope to the sample. The *ADSR* envelope (*Attack, Decay, Sustain, and Release*) controls the loudness of each note over time. The Attack, Decay, and Release parameters are rate or time controls. Sustain controls amplitude (level). When a sample is first triggered, the envelope generator will begin to rise to its full level at the rate set by the Attack. When it reaches the peak level it then falls at the rate set by the Decay parameter to the level set by the Sustain control. The envelope will remain at the sustain level as long as the key is held down. When the key is released, volume will return to zero at the rate set by the Release parameter.

Velocity Tracking (Vel Track) allows you to apply the ADSR envelope to varying degrees, depending on MIDI note-on velocity (how hard you play notes). For example, if you set Velocity to zero, all notes will simply play at the volume set by the volume knob. Higher velocity values cause notes to be played louder, the harder you play.

Filter

To apply the *Filter* (Figure 2-6), click the Filter button to engage it. Choose the desired filter type from the menu and then adjust the *cutoff frequency* (*Freq*) and filter *resonance* (*Res*). Apply note-on velocity response with the *velocity tracking* (*Vel Track*) control. *Key Tracking* (*Key Track*) makes the filter cutoff frequency change relative to the root frequency of the note being played. Full tracking (100%) produces a full range key tracking over the entire keyboard, where lower settings reduce the tracking effect by scaling the change proportionally as you move away from the filter cutoff frequency on the keyboard.

Filter envelope

The *Filter Envelope* (*Filter Env* in Figure 2-6) controls how the filter cutoff frequency evolves as a note sustains. When the envelope is disabled, the filter frequency remains constant, at its current setting, for as long as the note plays. If the Filter Envelope is enabled, the attack setting determines how long the cutoff frequency takes to reach its maximum value, as determined by the Filter Envelope *Amount* setting. The subsequent *Decay*

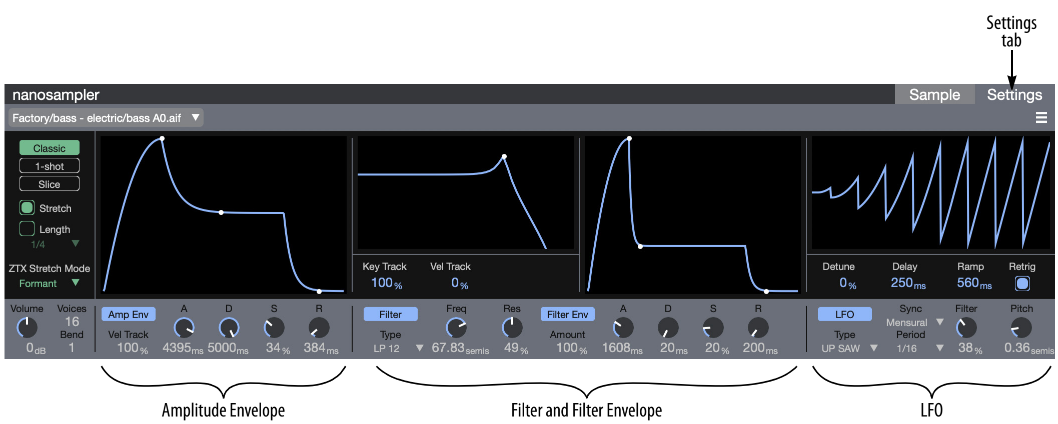


Figure 2-6: The Settings tab.

(*D*), *Sustain* (*S*) and *Release* (*R*) settings control how the cutoff frequency returns to its original, specified value.

LFO

The Nanosampler LFO has the following settings:

Setting	Unit	Explanation
Type	Menu	Lets you choose the wave shape of the LFO: Sine, Up/Down Sawtooth, Square, Sample & Hold, Sample & Ramp and Random Walk.
Sync	Menu: Real time Mensural	Determines whether the LFO is locked to real time or tempo. Mensural makes the LFO synchronize to the tempo of the project's time line.
Mensural	Beats	LFO rate is adjusted in beat values (whole note, half note, quarter note, etc.) A quarter note is equal to one beat.
Rate	Hertz (Hz)	The number of cycles per second at which the LFO oscillates.
Filter	percent	Modulates the filter cutoff frequency with the LFO.
Pitch	percent	Modulates the pitch of the sample with the LFO.
Detune	Percent	Changes the rate of the LFO based on the note pitch. (Has no effect when LFO is synced to tempo.)
Delay	ms	The amount of time before the LFO begins oscillating after being triggered.
Ramp	Milliseconds (ms)	Applies an amplitude ramp (from zero to 100%) to the LFO.
Retrig (Retrigger)	On/off	Enable retrigger to restart the LFO each time a note is played (polyphonic LFO); disable it to keep the LFO running from the first note played (monophonic LFO).

MODULO

Modulo (Figure 2-7) is a two-oscillator virtual synthesizer instrument, which features digital waveforms with a unique phase modulation system.

Patches

A *patch* is a snapshot of all the settings in the Modulo window. Modulo ships with dozens of useful presets organized by category in several dozen banks, such as *Synths*, *Leads*, *Strings*, etc. The Presets section at the bottom of the Modulo window (Figure 2-7) lets you choose, modify, compare, revert and save presets. Modulo can support an unlimited number of presets.

Bank

A bank can hold up to 128 presets. When you choose a bank, its patches (presets) are displayed in the Patch menu. Use the +/- buttons next to the menu to choose the next or previous bank in the list. To create, rename or delete banks, see “Managing patches and banks” on page 63.

Patch

The Patch menu displays all of the patches in the bank currently chosen in the Bank menu. When you choose a patch from the menu, its settings are loaded into the Modulo window. Use the +/- buttons next to the menu to choose the next or previous preset in the list. Hold down the Option key while clicking the +/- buttons to stay within the current bank when browsing presets. To create, rename, move, duplicate or delete patches, see “Managing patches and banks” on page 63.

Save

When you first choose a patch, the *Save* button is not available. As soon as you change any parameter in the Modulo window, Save becomes active, and the patch name becomes italic in the Patch menu to indicate that the patch has been modified. Click Save to store the changes you made to the patch (replacing the original version).

If you wish to “save as” in order to preserve the original patch, see “Managing patches and banks” on page 63.

Revert

When you first choose a patch, the *Revert* button is not available. As soon as you change any parameter in the Modulo window, Revert becomes active, and the patch name becomes italic in the Patch menu to indicate that the patch has been modified. Click the Revert button to permanently discard any changes you’ve made to the patch.

A/B (compare)

When you first choose a patch, the *A/B* button is not available. As soon as you change any parameter in the Modulo window, Compare becomes active, and the patch name becomes italic in the Patch menu to indicate that the patch has been modified. Click the A/B button repeatedly to toggle between the original patch and the modified version. The original patch is indicated by non-italic text; the modified patch name is italic. Note that the modified patch gives you access to the *Save* and *Revert* buttons, which do not apply to the original patch (since it is already saved).

Managing patches and banks

Click the *File* button to display patch and bank management controls in the Modulo window, as shown in Figure 2-8. These controls let you manage Modulo banks and patches. Click the *Edit* button to return to the main Modulo window.

Source

The *Source* section (Figure 2-8) lets you choose the patch you wish to work with. Click *Delete* to permanently remove the current source patch. Click *Save* to save any changes that have been made to the current source patch (such as any edits to the name, author or other patch text).

Figure 2-7: Modulo.

Destination
The *Destination* controls (Figure 2-8) let you move or "save as" a patch to a new location in any bank. The *Move to* button places the patch at the chosen destination location and also empties the patch's current location. The *Save to* button preserves the source patch and makes a copy of it, along with any changes that have been made, to the chosen destination location. *Save to* is similar in concept to the standard *Save As* command.

Patch name, author, etc.
Edit the text in these text fields as desired. To save your changes, click the *Save* or *Save to* button (in the Source or Destination sections).

New/rename/delete/export/import bank
Use these three buttons to manage banks as follows:

Goal	Action
To create a new bank	Type in a name in the text box on the right and click *New Bank*.
To rename a bank	Choose it from the menu, type in a new name in the text box on the right and click *Rename Bank*.
To delete a bank	Choose it from the menu and click *Delete Bank*.
To export a bank	Choose it from the menu and click *Export Bank*.
To import a bank	Click *Import Bank* to locate the bank you wish to import on your hard drive.

Revert
Click *Revert* to discard any unsaved changes you've made to the current preset.

Edit
Click *Edit* to return to the main Modulo window.

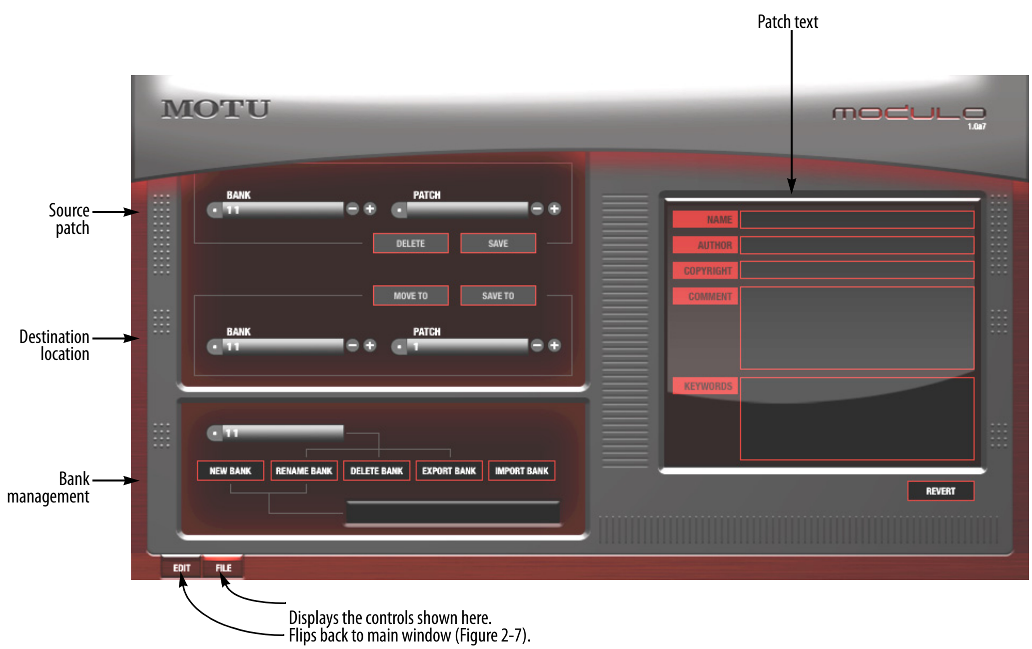


Figure 2-8: Click the File button to display the patch and bank management controls.

Master controls

The Master controls (Figure 2-9) provide global controls that affect the overall performance of Modulo. They are presented as a row of knobs below the LCD display as shown in Figure 2-9.

Figure 2-9: The Master controls.

Glide

Glide is a continuous, smooth glide in pitch from one note to another. You can adjust it from zero (no portamento) to 100% (full portamento).

Detune

Detune splits each oscillator into two separate oscillators panned left and right and detuned from one another by a percentage from zero to 100%, where 100% is a full semitone. Detune can be used with all three modes (mono, legato or poly). Detune is good for thickening sounds.

Mix

Mix controls the relative amount of each oscillator present in the signal. When the mix knob is in the center position, you have equal amounts (50%/50%) of each oscillator. To hear oscillator 1 only, turn the knob all the way left (100%/0%). To hear oscillator 2 only, turn the knob all the way right (0%/100%).

Noise

Use the *noise* control to add white noise to the sound.

Volume

The *Volume* slider (which is mapped to controller #7) controls Modulo's overall volume.

MIDI settings (mono, poly, legato, bend)

Just like classic analog synths, Modulo can operate in mono, legato or poly mode. Click the button shown in Figure 2-9 to enable the desired mode.

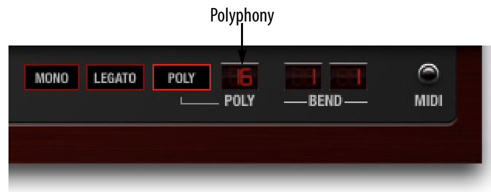


Figure 2-10: MIDI settings.

In *mono* mode, only one note can be played at a time. Each new note will cut off the currently sustained note, if any.

Legato mode is an alternative form of mono mode where the envelope is not retriggered; only the pitch changes.

In *poly* mode, Modulo can play two or more notes simultaneously, up to the limit you select next to the poly button. For example, if polyphony is set to 16, Modulo can play and sustain up to 16 notes at a time.

Polyphony

The polyphony setting determines how many notes can play at a time. The maximum allowed polyphony for one instance of Modulo is 16. Beware, however, that higher polyphony settings place higher demand on your host computer's processing resources. Therefore, the ideal polyphony setting is that which matches the highest number of notes you will actually need (the highest number of notes you will play simultaneously, taking into account their releases and any resulting overlapping).

Bend

The bend parameters control how Modulo responds to pitch bend. The pitch bend range is split at zero into two pitch bend ranges: *upper* and *lower*. The upper range determines how much pitch bend occurs between the zero position on your pitch bend wheel (or other controller) and its highest position. The lower setting determines the range from zero to the pitch bend wheel's lowest position. By setting them to different values, you can more easily bend up and down by different amounts.

The upper and lower pitch bend ranges offer a maximum range of one octave (12 semitones) each, for a combined maximum of two octaves.

MIDI activity light
The MIDI activity light illuminates when Modulo receives MIDI data. This can be a useful trouble-shooting tool. If Modulo is not making any sound when you play it, but the MIDI light does blink, then you can focus your troubleshooting efforts on the audio signal path.

Status
The status LCD (Figure 2-11) displays information about the parameter you are currently modifying or targeting with the cursor. If you modify an oscillator setting, the LCD also displays the oscillator waveform.

Figure 2-11: The Status LCD.

Oscillators
Modulo provides two oscillators (Figure 2-12).

Signal flow
Modulo's overall signal flow bears a striking resemblance to a Prophet 5. But it is best not to think of each oscillator as a separate synthesizer. Instead, think of the oscillators as going into a mixer before being processed.

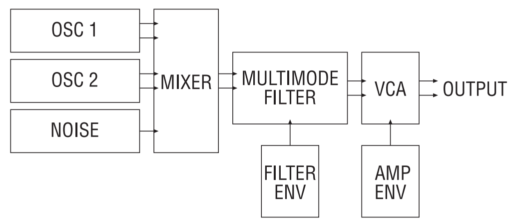


Figure 2-13: Modulo signal flow.

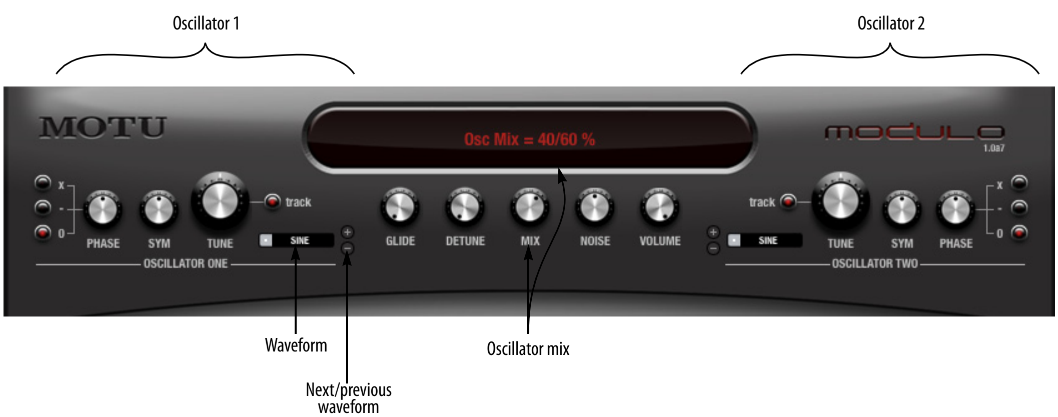


Figure 2-12: The Oscillators.

Oscillator Mix

Turn the Oscillator Mix knob to control how much of each oscillator is included in the signal (Figure 2-12). To completely turn off oscillator 1 or 2, turn the Mix knob all the way right or left, respectively.

Waveforms

Choose the desired waveform for the oscillator from the menu, or use the +/- buttons to cycle through the list of waveforms. Each oscillator provides a variety of standard subtractive synthesis waveforms. An extensive library of waveshapes are also provided.

Sine

This is a standard sinusoidal waveform.

Sawtooth

This is a standard sawtooth wave. Use the symmetry parameter to morph the sawtooth waveform between a downward triangle and an upward triangle waveform:

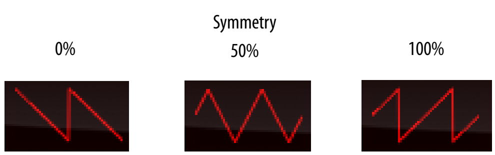


Figure 2-14: Use symmetry to morph between these three basic waveforms.

A sawtooth wave (50% symmetry) has energy at all harmonics, and the strength of higher partials falls off linearly. A triangle wave has less energy at high partials, and strength falls off exponentially as the square of the partial number.

Rectangle

This is a standard square wave. The pulse width can be modified into a rectangle waveform using the symmetry control (“Symmetry” on page 68).

Figure 2-15: Adjust the symmetry control to create a rectangular wave form.

Waveshapes

Waveshapes are sets of complex waveforms that provide a rich assortment of harmonic content and tone color for a sound. A wide variety of waveshapes are provided. Modulo waveshapes have adjustable symmetry, just like the pure waveforms (see “Symmetry” on page 68).

Waveshapes can produce spectra that differs greatly from standard subtractive synthesis waveforms. Changing waveshapes is an easy and rewarding way to get into patch programming. Simply find a patch you like, and audition different waveshapes.

Modulating the oscillator phase shift is an easy way to add movement to your patches. If you modulate the phase of oscillator 1, and modulate the phase of oscillator 2 at the same time, the combined effect is that the sound will never be quite the same at any given moment.

Because waveshapes can sound like filtered subtractive waveforms, sometimes you can use waveshapes without any filtering at all, which saves CPU overhead.

Tune

Use the *Tune* knob to offset the pitch of the oscillator from key tracking. When the *Track* button (explained in the next section) is disabled, the range is expressed in absolute frequency (Hz or kHz); the range is from 8.2 Hz to 22.1 kHz.

When the Track button is enabled, the pitch is expressed in semitones relative to the root pitch of the note being played. The range is from -60.00 semitones to +84.00 semitones.

Tracking

When the *Track* button is enabled, the oscillator adjusts its frequency relative to the note being played. Accordingly, when tracking is enabled, the *pitch* of the oscillator is expressed in the number of semitones relative to the root pitch of the note being played.

Symmetry

The *symmetry* knob adjusts pulse width on rectangle waves, but symmetry can also be applied to sine waves, sawtooths and even waveshapes. Doing so produces interesting changes in harmonic content. Further interesting effects can be achieved by modulating the symmetry (see “Modulation” on page 71).

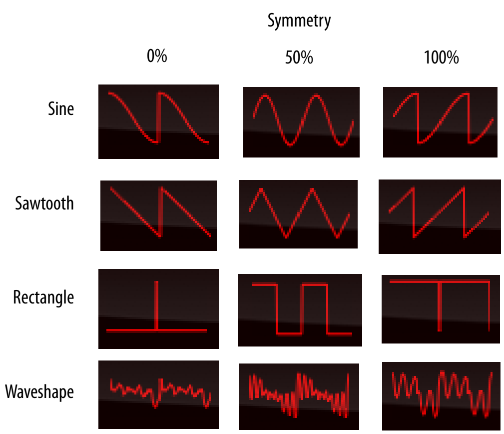


Figure 2-16: Symmetry.

Phase

Three *phase* shift modes are provided: *off* (0), *subtract* (-) and *multiply* (x). When engaged, the oscillator is split into two separate oscillators that can be offset from one another by the phase knob from 0 to 180 degrees, and then either *subtracted* from each other or *multiplied* by each other. The results can produce significant timbral differences, depending on the chosen waveform. As you turn the Phase knob, you’ll often discover “sweet spots” where significant timbral change occurs.

Modulating oscillator settings

Oscillator pitch, phase shift and symmetry can be modulated. For further details, see “Modulation” on page 71.

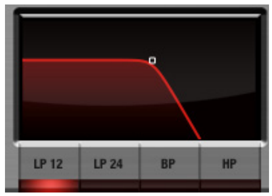
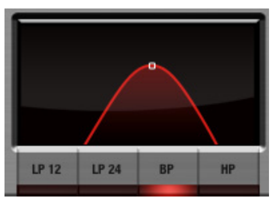
Filter

Modulo provides a multimode filter.

Figure 2-17: The filter.

Filter type

The filter can be assigned to one of four different filter types (shown below) using the illuminated buttons along the bottom of the filter graph:

Symbol	Filter type	Sample
LP12	Lowpass 12dB slope	
LP24	Lowpass 24dB slope	
BP	Bandpass	
HP	Highpass	

Enabling/disabling the filter

In accordance with classic analog synthesizer design, the filter is always present in the signal path. However, you can “disable” it by choosing one of the low pass filters and turning the cutoff frequency all the way up (to 22.1kHz). Alternatively, you could choose the high pass filter and turn the cutoff frequency all the way down (to 8 Hz).

Cutoff frequency

The *Cutoff* knob determines the filter’s *cutoff* or *center frequency*, which can be set as a number of cycles per second (Hz or kHz) when key *tracking* is disabled. When key tracking is enabled, Hz and kHz are not meaningful because the filter’s center frequency is not fixed at an absolute frequency; instead, it is relative to the note being played. Therefore, Modulo expresses filter cutoff/center frequency in semitones relative to the pitch being played.

Resonance (RES)

Resonance emphasizes the cutoff/center frequency of the filter.

Tracking

The filter is equipped with key tracking (the *track* button), which causes the cutoff frequency to “track” (change relative to) the frequency of the note being played. Key tracking helps avoid undesirable artifacts. For example, a lowpass filter will cause notes to get more dull as you play higher pitches (which have higher frequencies).

Using the filter graph

Use the control point on the filter graph to adjust the frequency and resonance graphically.

Envelope Amount (ENV AMT)

The filter has a dedicated envelope for sweeping the cutoff frequency (see “Filter envelope” on page 70). The *envelope amount* knob controls the range of frequencies, specified in semitones, over which the cutoff frequency is modulated by the envelope.

Modulating filter settings

Filter cutoff frequency and resonance can be further modulated. For details, see “Modulation” on page 71.

Envelopes

Modulo provides three dedicated four-stage ADSR (*Attack, Decay, Sustain, and Release*) envelopes for amplitude, filter and general purpose parameter modulation. The three envelopes share the same graph and controls, so to choose one for viewing and programming, click the desired selector button just below the graph (Figure 2-18).

Figure 2-18: Envelopes.

The four envelope stages

The envelope graph provides a visual indication of the four stages for each envelope:

Symbol	Stage	Explanation
A	Attack	The initial stage of the envelope, specified in the amount of time for it to fully open up.
D	Decay	The amount of time between the end of the attack and the beginning of the sustain.
S	Sustain level	The level at which the envelope remains open, where zero is completely closed and 1.00 is fully open.
R	Release	The final stage of the envelope, where it closes down to zero, specified in the length of time from the end of the hold to the moment when it reaches zero.

Amplitude envelope

Modulo's amplifier is not presented graphically in the Modulo window. Instead, there is a dedicated overall volume control (Figure 2-9 on page 65), a dedicated velocity > volume slider (see "MIDI note-on velocity (VEL)" on page 72) and the amplitude envelope, which is dedicated to overall amplitude modulation.

The amplitude envelope triggers polyphonically. That is, each separate note that is played is given its own unique envelope cycle. The only exception is when Legato mode ("MIDI settings (mono, poly, legato, bend)" on page 65) is engaged. In this case, the envelope is applied only to the first held note and it is not retriggered until the held note is released and a new note is played.

Filter envelope

The *Filter envelope* can be applied to the filter cutoff frequency. Set the filter envelope shape as desired and then use the *Envelope Amount* (*ENV AMT*) knob in the filter section (Figure 2-17) to control the range over which the cutoff frequency is modulated by the envelope.

LFOs

Modulo provides two LFOs (Low Frequency Oscillators), which can be used to modulate filter cutoff frequency and all continuously variable oscillator parameters. The two LFOs share the same graph and controls, so to choose one for viewing and programming, click the desired selector button just below the graph.

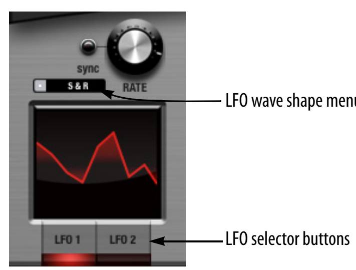
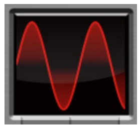
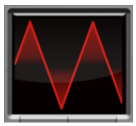
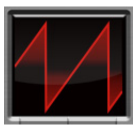
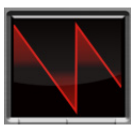
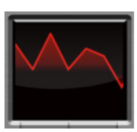
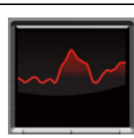


Figure 2-19: The LFOs.

LFO wave shape

Choose the desired LFO wave shape from the wave shape menu (Figure 2-19):

LFO wave shape	Example
Sine	
Triangle	
Up Saw	
Down Saw	
Square	
Sample and hold	
Sample and ramp	
Random walk	

LFO rate

The LFO *rate* knob controls the number of cycles per second at which the LFO oscillates. The range is from 0.0001 Hz to 25 Hz.

Polyphonic LFO triggering

LFO 1 is polyphonic. This means that each note is given its own unique LFO onset when it is played.

LFO 2 is monophonic. This means that an initial held note determines the LFO onset, but it is not retriggered by subsequent notes until the original held note is released.

LFO sync

The sync button (Figure 2-19), when enabled, makes the LFO synchronize to the tempo of the sequence time line. Accordingly, when sync is enabled, the LFO rate parameter is expressed in beat values (whole note, half note, quarter note, etc.)

Modulation

Modulo has a powerful modulation architecture. The modulation section (Figure 2-20) is most easily understood if you think of it as a modular synthesizer that uses control voltages to manipulate the sound.

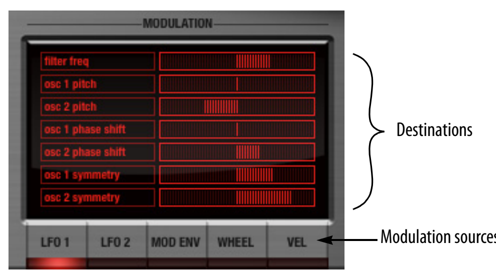


Figure 2-20: The modulation section.

Modulation sources

There are five modulation sources (buttons) in the modulation section (Figure 2-20). Click each source button to view its possible destinations above (the parameters that it can modulate).

Assigning a source to a destination

To assign the modulation source to a destination, click the source button, and then click and drag on the touch-sensitive LED graph to the right of the destination you wish to modulate.

Assigning a source to multiple destinations

In Modulo, sources can modulate multiple destinations simultaneously. LFO 1 could modulate all seven of its destinations at once.

Source highlighting

Source buttons have three illumination states:

Illumination state	Meaning
Dark	The source is not being used in the current preset.
Red	The source is currently being viewed.
Orange	The source is not currently being viewed, but it has destination assignments.

LFO 1 and 2

LFO 1 can modulate filter cutoff frequency, individual oscillator pitch, individual oscillator phase shift, and oscillator frequency.

LFO 2 can modulate oscillator mix (“Oscillator Mix” on page 67), oscillator 1 & 2 pitch (both at the same time), oscillator phase shift and oscillator symmetry. Note that phase shift and symmetry can be modulated by both LFO1 and 2 at the same time.

The modulation envelope (MOD ENV)

The envelope can modulate oscillator mix, pitch, phase shift and symmetry.

Mod wheel (WHEEL)

Click the *wheel* button (Figure 2-21) to assign the mod wheel of your MIDI controller as a modulation source to filter cutoff frequency, filter resonance, oscillator phase or oscillator symmetry. You can also assign it to control the amplitude of LFO 1 as a modulation source for the filter cutoff frequency. In effect, you are manually controlling the strength of the effect of the LFO on the cutoff frequency. Similarly, mod wheel can control LFO 1 as a modulation source for oscillator pitch (Figure 2-21). These modulations can be used together with the modulations from LFO 1.

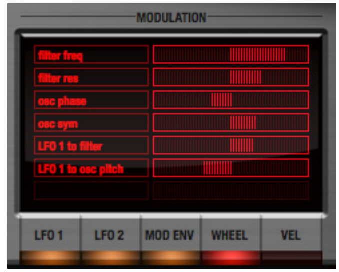


Figure 2-21: The mod wheel modulation destinations.

MIDI note-on velocity (VEL)

Click the VEL button to assign MIDI note-on velocity as a modulation source for any combination of the following destination parameters: overall volume, oscillator mix, oscillator phase, oscillator symmetry and/or filter cutoff frequency.

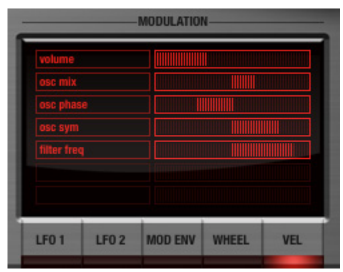


Figure 2-22: Note-on velocity modulation destinations.

Host application plug-in automation

Modulo fully supports the plug-in automation features of Performer Lite. Each parameter in Modulo has its own dedicated automation input. This automation operates independently from Modulo’s modulation section, although it can be combined with any modulation sources.

Contextual menus

Hold down the Control key (or the right-hand button your dual-button mouse) and click on am item in the Modulo window to view a contextual menu of additional options for that item. What appears in the contextual menu depends on what you Control-click. Here is a brief summary of the various commands you will see in the contextual menus.

Copy this item to all others
Applies all settings from the source oscillator, envelope or LFO to all of the others. For example, if you Control-click oscillator 1 and choose *Copy this oscillator to all others*, then oscillator 1's settings are applied to oscillators 2.

Learn controller mapping
Lets you assign any external MIDI controller to the parameter. To do so, choose this command and send the controller message you wish to use (move the knob or slider).

Forget controller mapping
If the parameter is currently assigned to a MIDI controller for external control, you'll see this menu item, which clears the MIDI controller (and disconnects the external control). You can then reassign it, if you wish.

Copy this setting to all others
Applies the setting from the source parameter to all other similar parameters. For example, if you Control-click the *Symmetry* parameter for oscillator 1 and choose *Copy this setting to all others*, then oscillator 1's Symmetry value is applied to the Symmetry parameter for oscillators 2.

Figure 2-23: Model 12.

MODEL 12

Model 12 (Figure 2-23) is a twelve-part virtual drum module. Hundreds of drum sounds and dozens of preset drum kits are included.

Drum kits

To load a preset drum kit, choose it from the plug-in window Preset menu. You can also use the plug-in preset management features in the Preset menu to create your own drum kit presets and otherwise manage them.

Master volume, tune and stretch

Use the master volume, tune and stretch knobs (Figure 2-23) to adjust the overall volume, pitch and sample length of all the instrument sounds in the drum kit. As you adjust each parameter, the LCD display shows its current settings, as shown in Figure 2-23. Master volume is mapped to controller #7.

Parts

There are twelve parts in Model 12 (Figure 2-23). Each part (Figure 2-24) can load a single drum sound (sample), such as kick, snare, hi-hat, etc. which you then trigger by playing the part's associated MIDI pitch (such A1, B1, etc.) Each part has the following controls:

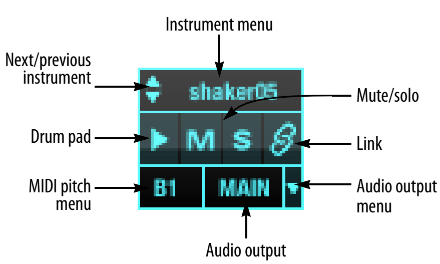


Figure 2-24: A Model 12 part.

Instrument menu

Choose the desired instrument from the *Instrument* menu (Figure 2-24). Or click the *Next/ previous instrument* buttons to browse. The Instrument menu provides the options explained below. For further details, see “Model 12 sample management” on page 77.

Menu item	Explanation
Factory	Lets you choose factory sounds from sub-menus organized by instrument category.
Shared	This menu item appears after you first use the Copy to Shared command (explained below), which copies samples to a shared directory. For example, you might have several networked computers at your studio, and you would like to provide shared Model 12 samples to all of the computers.
User	Lets you access sounds located in your own user sounds, as you've organized them. See “Model 12 sample management” on page 77.
Project	Lets you choose any sound that has been saved with the project.
Open	Loads any audio file from your hard drive.
Reveal in Finder/ Reveal in Explorer	Displays the audio file on the computer desktop.
Copy to User	Copies the currently loaded sample to your user directory, displayed in the User sub-menu (explained below). This menu item is grayed out when a factory sample is loaded, since factory samples are always available in the Factory sub-menu.
Copy to Project	Copies the currently loaded instrument sound to the project folder to consolidate the audio for archiving purposes, transfer, exchanging the project with other users, etc. This menu item is grayed out when a factory sample is loaded, since factory samples are always available in the Factory sub-menu.
Copy to Shared	Copies the currently loaded instrument sound to a shared (public) sample directory which can be made accessible to other users. This menu item is grayed out when a factory sample is loaded, since factory samples are always available in the Factory sub-menu.
Clear Sample	Removes the current sample from the part.
Audition on Load	Plays the sample when you load or browse it.

Drum pad

Click the *drum pad* (Figure 2-24) to trigger the sample. The drum pad also illuminates whenever it is triggered via MIDI.

MIDI pitch menu

Choose the *MIDI pitch* (Figure 2-24) you wish to assign to the instrument. This is the MIDI note you will play to trigger the sample.

Audio output

Each part can be assigned to one of several different audio output destinations. By default, each parts is assigned to *Main*, which is the audio output of the Performer Lite instrument track on which Model 12 is instantiated. Alternatively, you can use the audio output menu (Figure 2-24) to assign each part to one of six auxiliary (aux) stereo output bundles. See “Send and Aux output mapping” on page 77.

Link

Click the *link* button (Figure 2-24) on two or more parts to link them together. The classic use for this feature is open and closed hi-hat sounds. The closed hat sound will cut off the open hat sound, and vice versa. In other words, linked instruments can’t play at the same time.

Mute/solo

The *Mute* and *Solo* buttons (Figure 2-24) mute and solo the individual part.

Part settings

Click a *Part Select* button (Figure 2-23) to view settings for the part in the main portion of the window as shown in Figure 2-25.

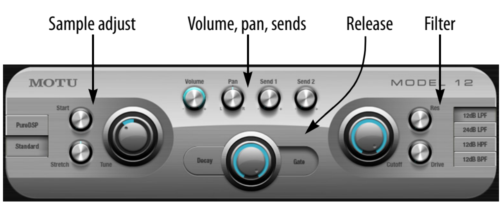


Figure 2-25: Part settings.

Volume, pan and sends 1/2

The *volume* and *pan* knobs (Figure 2-25) controls the level and panning of the individual part, relative to all other parts in Model 12.

The two *send level* knobs (Figure 2-25) route audio signal from the part to Model 12’s two send busses. These busses appear in the *Instruments* tab of Performer Lite’s Bundles window. You can then create mono or stereo instrument output bundles that can be used as inputs for aux tracks for further routing and processing. See “Send and Aux output mapping” on page 77.

Release

The *Decay/Gate* controls (Figure 2-25) let you specify the length and release of the sample.

Click the *Decay* button to make the sample continue playing for as long as you hold down the note (such as a drum loop, for example), and then fade out when you release the note. The fade-out time can be adjusted using the knob, and it can range from 1 ms to 5.00 seconds.

The *Gate* button creates the classic gated drum effect, where the sample is triggered and then immediately cut off after the specified gate amount, which can be set as a percentage of the sample length, where 100% is the length of the entire sample.

Sample adjust section

The *sample adjust* controls (Figure 2-25) let you modify the length, pitch and start time of the sample. Velocity control and randomization can be added to produce a more dynamic and “human” feel.

Sample start

The *start* knob (Figure 2-25) in the sample adjust section controls where the sample begins playing. A setting of 0.0% begins playback at the very beginning of the sample. A setting of 100% begins playback at the very end of the sample. Click the Start knob to view additional settings shown below:

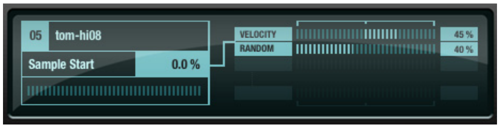


Figure 2-26: Sample start.

Click anywhere in the touch-sensitive LED strips to make a setting. Double-click to return the setting to zero (or its default value).

Velocity (Figure 2-26) allows you to control the sample start time based on note-on velocity. With positive values, the harder you play, the later the sample start time. With negative velocity values, the harder you play, the earlier the start time, relative to the current sample start setting. For example, you could set the sample start to 50% and velocity to -100% (negative 100%). The result is that the harder you play, the closer the sample trigger gets to its beginning. This is very effective for adding a great deal of dynamic control over samples with percussive attacks.

Random (Figure 2-26) lets you specify a range, starting from the beginning of the sample, over which the start time will be randomly played each time the sample is triggered.

Sample tune

The *Tune* knob (Figure 2-25) in the sample adjust section controls the pitch of the sample in hundredths of a semi-tone, where the range is from -12.00 to +12.00 semitones. Click the Tune knob to view additional settings shown below:

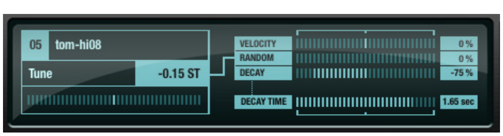


Figure 2-27: Sample tune.

Velocity (Figure 2-27) modifies the tuning of the sample based on MIDI note-on velocity, where positive values make the pitch go higher as you play harder, and negative values make the pitch go lower as you play harder.

Random (Figure 2-27) lets you specify a range, extending from the root pitch of the sample, over which the tuning will be randomly played each time the sample is triggered

Decay (Figure 2-27) causes the pitch modulation to decay over the time period specified by *Decay Time*. Positive decay values start high at the initial attack and then "bend" down to the root pitch; negative decay values start low and bend up to the root. Use decay time to control the length of the bend.

Sample stretch

The *Stretch* knob (Figure 2-25) lets you lengthen the sample up to 100% of its original length or shorten it to 50% of its original length. Choose either *Standard* or *PureDSP* time-stretching. Standard time-stretching causes the pitch to go up when you shorten the sample and down when you lengthen it. PureDSP maintains the original pitch.

Filter

The *Filter* section for each part (Figure 2-25) provides four filter types: a lowpass (*LPF*) with 12dB slope, lowpass with 24dB slope, high pass (HPF) with 12dB slope and band pass (BPF) with 12dB slope. Click the button for the desired filter type.

Filter cutoff (center) frequency (*Cutoff*), resonance (*Res*) and *Drive* all provide velocity and random controls, similar to those explained earlier for Sample Tune. In addition, cutoff frequency provides *Decay* and *Decay* time similar to the Sample Tune decay features explained earlier.

Send and Aux output mapping

The send and aux output mapping for Model 12 is as follows:

Send/aux	Model 12 mapping
Send 1	Model12 3-4
Send 2	Model12 5-6
Aux 1	Model12 7-8
Aux 2	Model12 9-10
Aux 3	Model12 11-12
Aux 4	Model12 13-14
Aux 5	Model12 15-16
Aux 6	Model12 17-18

Model 12 sample management

Model 12 organizes samples into four categories:

Sample Category	Where the samples are stored
Factory	In the MOTU application support folder: /Library/Application Support/MOTU/Model 12/ Model 12 Data.bundle
User	In your user directory application support folder: User/Library/Application Support/MOTU/Model 12/ Samples/
Shared	In the MOTU application support folder: /Library/Application Support/MOTU/Model 12/ Shared/Samples/
Project	In the Performer Lite project folder: Project/Plug-in Data/Model 12/Samples/

The *Factory* samples are always available. *User* samples are stored in your user directory and are therefore available only when you are the currently logged in user. You can use the user directory to protect your own sample content. *Shared* samples are stored in the system library application support folder and can therefore be made available to any users that you wish to share them with. *Project* samples are stored with the Performer Lite project itself. By keeping them with the rest of the files associated with the project, you don't have to worry about losing them when exchanging the project with a colleague, archiving the project, transferring it, etc.

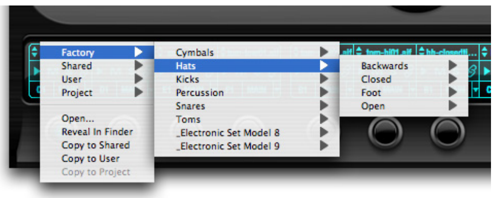


Figure 2-28: The Model 12 Instrument menu.

Building your user library

To add a single sample to your user library, load it and then choose *Copy to User* from the instrument menu (Figure 2-28). To view the folder on the computer desktop, load a user sample (if one is not currently loaded) into a part and then choose *Reveal in Finder/Explorer* from the instrument menu (Figure 2-28). To add many samples at once, you can simply place them in the User folder in the Finder.

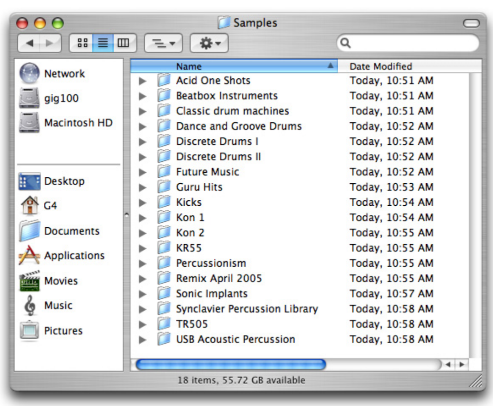


Figure 2-29: Organizing the Model 12 sample library on the computer desktop.

You can organize them into folders and sub-folders as desired (Figure 2-29), which then appear as sub-menus in the instrument menu (Figure 2-30):

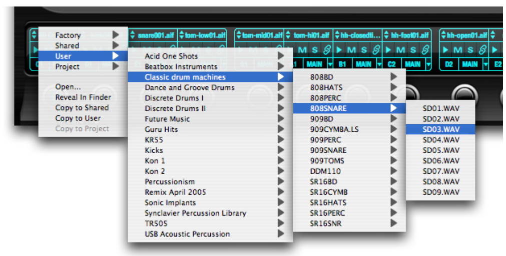


Figure 2-30: Organizing user samples into sub-folders causes them to appear as sub-menus in the Model 12 instrument menu.

Using aliases or shortcuts

If you wish to include audio files that you prefer to store elsewhere on your hard drive(s), make Mac aliases or Windows shortcuts of them and place them in the Model 12 sample library folder(s).

PROTON

Proton (Figure 2-31) is a two-operator FM (frequency modulation) synth. The frequency of the *Carrier* signal (sine wave) is modulated by the *Modulator* oscillator. Additional controls allow you to program a wide variety of classic FM sounds, including gongs, bells, Rhodes pianos, plucked strings and many others.

About FM synthesis

If you modulate an oscillator's frequency with the output of another oscillator at sub-audible rates (below approximately 80 Hz), the resulting effect on the signal is vibrato. As you raise the modulator's frequency into the audible range (above approximately 80 Hz), the vibrato's character transforms into a change of timbre. At

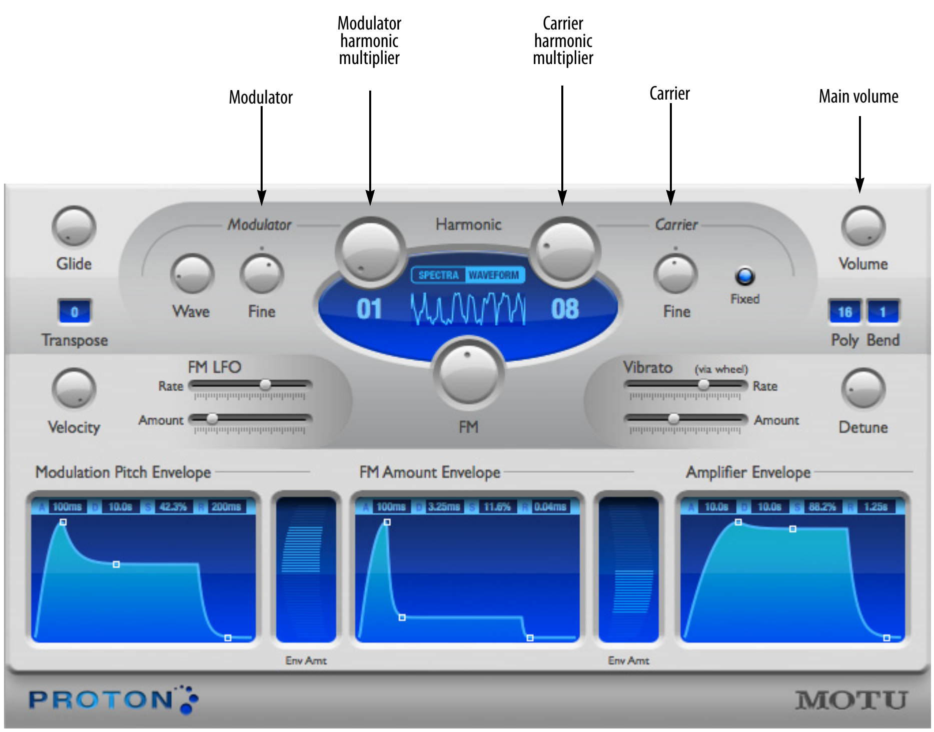


Figure 2-31: Proton.

that point, it becomes frequency modulation instead of vibrato. Different frequency ratios of modulator and carrier produce a variety of interesting timbres. Proton provides controls for easily applying frequency multipliers (harmonics) to both the carrier and the modulator signal. The graphic display in the center of the window can show the resulting spectra or periodic waveform in real time.

The Carrier and Modulator

The *Carrier* signal (Figure 2-31) is a sine wave equal to the frequency of the note being played. (For example, the A above middle C has a frequency of 440 Hz.) The *Modulator* modulates the *Carrier* signal. In its most basic form, the Modulator is also a sine wave that is based on the frequency of the note being played.

Harmonic multipliers

Harmonic frequencies are the frequencies you get when you multiply a base frequency by integers. The *Harmonic* knobs allow you to multiply the modulator and the carrier frequencies by whole integer multiples. For example, a harmonic multiplier of 1 (01) simply produces the base frequency of the note being played. If you set the modulator harmonic to 1 and the carrier harmonic to zero (with all other controls disabled), the result is a sine wave at the base frequency. (Note that Proton does not allow you to set the modulator harmonic to zero, as this would zero out the modulator's effect and produce no frequency modulation at all — with the result being just a simple sine wave. If both the modulator and carrier harmonic multipliers were set to zero, the result would be silence: a "direct current" flat-line signal with no modulation at all.)

If you set both the modulator and carrier harmonic multipliers to 1, you get a signal being modulated at its own frequency, and things start to get interesting. As you adjust the modulator and carrier harmonic multipliers to different ratios (such as 1:3, 1:4, 2:5, 4:3, etc.), the resulting harmonic content becomes increasingly more complex and interesting.

Modulator and Carrier Fine controls

Both the modulator and carrier supply a *Fine* knob (Figure 2-31), which lets you fine-tune their frequency over a range from half a harmonic below (-0.5) to half a harmonic above (+0.5) their base frequency.

Wave

The modulator *Wave* knob (Figure 2-31) provides an interesting twist to conventional FM synthesis, which customarily uses sine waves as the basic waveform for both the modulator and the carrier. Proton's Wave knob, however, changes the waveshape of the modulator to a special-purpose wavetable, which you can index as you turn the knob. This specialized modulator waveform provides a rich variety of timbres to work with, all within Proton's simple and intuitive two-operator design.

FM

Use the *FM* knob (Figure 2-31) to control the amount of frequency modulation being applied — essentially it controls the strength of the modulator's affect on the carrier. The amount of frequency modulation can also by controlled with MIDI note-on velocity. See "Velocity" below.

Fixed frequency carrier

Enabled the *Fixed* frequency button (Figure 2-31) to disable key tracking of the carrier up and down the keyboard. When disabled, the carrier frequency remains proportional to the pitch of the note being played.

The Spectra/Waveform display

In the Spectra/Waveform display (Figure 2-32), click the *Spectra* button to view a spectral graph of the frequencies being produced by the current settings in the plug-in. Blue lines represent components that are harmonic and pink lines represent components that are enharmonic. Click *Waveform* to see one cycle of the periodic waveform being produced by the current settings in the plug-in.

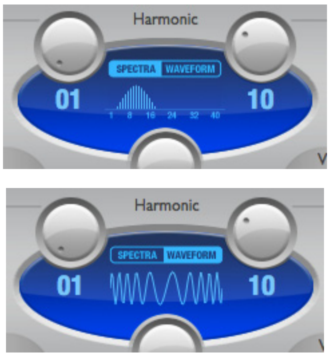


Figure 2-32: The Spectra/Waveform display.

Glide

Glide (Figure 2-31) is a continuous, smooth glide in pitch from one note to another. When the polyphony setting is set to 1, you can adjust Glide from zero (no portamento) to 100% (full portamento).

Transpose

Transpose (Figure 2-31) lets you transpose the MIDI input of Proton by a number of half-steps. The range is two octaves (-24 to +24).

Volume

Volume (Figure 2-31) controls the overall level of Proton's output. It is mapped to controller #7.

Polyphony

The *Polyphony* (Figure 2-31) setting determines how many notes can play at a time. The maximum allowed polyphony for one instance of Proton is 16. Beware, however, that higher polyphony settings place higher demand on your host computer's processing resources. Therefore, the ideal polyphony setting is that which matches the highest number of notes you will actually need (the highest number of notes you will play simultaneously, taking into account their releases and any resulting overlapping).

Bend

Bend range (Figure 2-31) is specified in half-steps from 1 to 12, where 12 is an octave, and it determines Proton's response to pitch bend.

Velocity

Velocity (Figure 2-31) lets you control the overall loudness of each note, as well as the amount of FM applied, via note-on velocity. A higher settings produces greater on-velocity response.

FM LFO

The *FM LFO* (Figure 2-31) lets you control the amount of FM applied with an adjustable LFO. *Rate* controls the speed of the LFO and *Amount* controls how much FM is applied.

Vibrato

The *Vibrato* controls (Figure 2-31) allow you to apply vibrato. *Rate* controls the speed of the vibrato and *Amount* controls the depth.

Detune

Detune (Figure 2-31) splits the carrier into two separate oscillators panned left and right and detuned from one another slightly. Detune is good for thickening sounds.

Modulation Pitch Envelope

The *Modulation Pitch Envelope* (Figure 2-33) applies a standard ADSR (attack, decay, sustain, release) envelope to modulator frequency. Attack, decay and release are expressed as amounts of time, whereas sustain is expressed as a percent of the envelope amount. The envelope range is +/- one octave. You can either drag the control points to adjust all ADSR values, or you can click each ADSR text field to edit the value directly. Use

Envelope Amount to control the strength of the envelope. Drag up to bend the pitch up; drag down to bend it down.

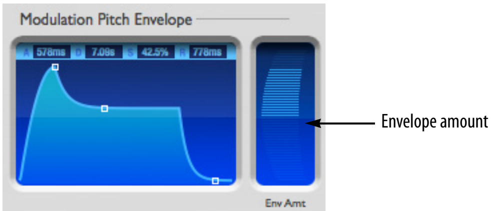


Figure 2-33: The Modulation Pitch Envelope.

FM Amount Envelope

The *FM Amount Envelope* (Figure 2-34) applies a standard ADSR (attack, decay, sustain, release) envelope to modulator amplitude. Attack, decay and release are expressed as amounts of time, whereas sustain is expressed in percent. You can either drag the control points to adjust all ADSR values, or you can click each ADSR text field to edit the value directly. Use *Envelope Amount* to control the strength of the envelope. Drag up to apply the envelope; drag down to invert the envelope.

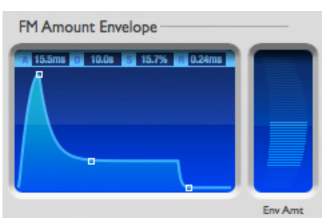


Figure 2-34: The Modulation Pitch Envelope.

Amplifier envelope

Use the *Amplifier Envelope* controls (Figure 2-35) to apply a standard ADSR amplitude envelope to each note. The *ADSR* envelope (*Attack, Decay, Sustain, and Release*) controls the loudness of each note over time. The Attack, Decay, and Release parameters are rate or time controls. Sustain controls amplitude (level).

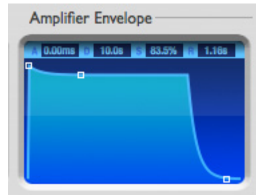


Figure 2-35: The Amplifier Envelope.

MIDI CONTROL OF INSTRUMENT SETTINGS

The settings in Model 12, Nanosampler and Performer Lite's other instruments can be controlled from your MIDI controller.

To assign a knob, fader, or other control device on your MIDI controller to a knob in Modulo, Control-click the knob to open the contextual menu and choose *Learn Controller Mapping* from the contextual menu. Then send MIDI data from your controller to complete the assignment.

Performer Lite's other five instruments can be controlled by sending the MIDI NRPN's listed below from your controller. Modulo can also be controlled via the NRPN's listed here.

BassLine

NRPN 0	Glide
NRPN 1	Legato Mode
NRPN 2	Lower Pitch Bend Range
NRPN 3	Volume
NRPN 4	Velocity->Volume
NRPN 5	Osc Mix
NRPN 6	Octave Transpose
NRPN 7	Stereo Detuning
NRPN 8	Filter Freq
NRPN 9	Filter Resonance

NRPN 10	Filter Env Decay
NRPN 11	Filter Modulation
NRPN 12	Filter Velocity Modulation
NRPN 13	Overdrive
NRPN 14	Amp Env Decay

PolySynth

NRPN 0	Pitch Bend Range
NRPN 1	Volume
NRPN 2	Velocity->Volume
NRPN 3	Triangle Oscillator Level
NRPN 4	Sawtooth Oscillator Level
NRPN 5	Rectangle Oscillator Level
NRPN 6	Sub 1 Oscillator Level
NRPN 7	Sub 2 Oscillator Level
NRPN 8	Noise Level
NRPN 9	Octave Transpose
NRPN 10	Stereo Detuning
NRPN 11	LFO Rate
NRPN 12	PWM Modulation
NRPN 13	Vibrato/Wah-Wah
NRPN 14	Filter Keyfollow
NRPN 15	Filter Freq
NRPN 16	Filter Resonance
NRPN 17	Filter Envelope Modulation
NRPN 18	Filter Velocity Modulation
NRPN 19	Attack
NRPN 20	Decay
NRPN 21	Sustain Level
NRPN 22	Release
NRPN 23	Distortion
NRPN 24	Chorus

Modulo

NRPN 0	Polyphony
NRPN 1	Lower Pitch Bend Range
NRPN 2	Upper Pitch Bend Range
NRPN 3	Polyphonic Mode
NRPN 4	Legato Mode
NRPN 5	Portamento Time
NRPN 100	Osc Mix
NRPN 101	Noise Level
NRPN 200	Osc 1 Waveform
NRPN 201	Osc 1 Key-Follow
NRPN 202	Osc 1 Phase Mode
NRPN 203	Osc 1 Pitch Offset
NRPN 204	Osc 1 Phase Shift
NRPN 205	Osc 1 Symmetry
NRPN 220	Osc 2 Waveform
NRPN 221	Osc 2 Key-Follow
NRPN 222	Osc 2 Phase Mode
NRPN 223	Osc 2 Pitch Offset
NRPN 224	Osc 2 Phase Shift
NRPN 225	Osc 2 Symmetry
NRPN 300	Filter Mode
NRPN 301	Filter Freq
NRPN 302	Filter Resonance
NRPN 303	Filter Key-Follow
NRPN 400	Amp Env Attack
NRPN 401	Amp Env Decay
NRPN 402	Amp Env Sustain Level
NRPN 403	Amp Env Release
NRPN 410	Filter Env Attack
NRPN 411	Filter Env Decay

NRPN 412	Filter Env Sustain Level
NRPN 413	Filter Env Release
NRPN 420	Mod Env Attack
NRPN 421	Mod Env Decay
NRPN 422	Mod Env Sustain Level
NRPN 423	Mod Env Release
NRPN 500	LFO 1 Waveform
NRPN 501	LFO 1 Sync Mode
NRPN 502	LFO 1 Rate
NRPN 503	LFO 1 Sync Period
NRPN 520	LFO 2 Waveform
NRPN 521	LFO 2 Sync Mode
NRPN 522	LFO 2 Rate
NRPN 523	LFO 2 Sync Period
NRPN 600	Filter Env->Filter Freq
NRPN 700	Mod Env->Osc Mix
NRPN 701	Mod Env->Osc 1 Freq
NRPN 702	Mod Env->Osc 2 Freq
NRPN 703	Mod Env->Osc 1 Phase Shift
NRPN 704	Mod Env->Osc 2 Phase Shift
NRPN 705	Mod Env->Osc 1 Sym
NRPN 706	Mod Env->Osc 2 Sym
NRPN 800	LFO 1->Filter Freq
NRPN 801	LFO 1->Osc 1 Pitch
NRPN 802	LFO 1->Osc 2 Pitch
NRPN 803	LFO 1->Osc 1 Phase Shift
NRPN 804	LFO 1->Osc 2 Phase Shift
NRPN 805	LFO 1->Osc 1 Sym
NRPN 806	LFO 1->Osc 2 Sym
NRPN 900	LFO 2->Filter Freq
NRPN 901	LFO 2->Osc Mix
NRPN 902	LFO 2->Osc 1 & 2 Pitch
NRPN 903	LFO 2->Osc 1 Phase Shift
NRPN 904	LFO 2->Osc 2 Phase Shift
NRPN 905	LFO 2->Osc 1 Sym
NRPN 906	LFO 2->Osc 2 Sym
NRPN 1000	Velocity->Volume
NRPN 1001	Velocity->Osc Mix
NRPN 1002	Velocity->Osc 1 & 2 Phase Shift
NRPN 1003	Velocity->Osc 1 & 2 Sym
NRPN 1004	Velocity->Filter Freq
NRPN 1100	Wheel->Filter Freq
NRPN 1101	Wheel->Filter Res
NRPN 1102	Wheel->Osc 1 & 2 Phase Shift
NRPN 1103	Wheel->Osc 1 & 2 Sym
NRPN 1104	Wheel->LFO 1->Filter Freq
NRPN 1105	Wheel->LFO 1->Osc Pitch
NRPN 1200	Stereo Detuning
NRPN 1201	Volume

Nanosampler

NRPN 0	Polyphony
NRPN 1	Pitch Bend Range
NRPN 2	Volume
NRPN 3	Velocity Sensitivity
NRPN 4	Tuning
NRPN 5	Sample Start
NRPN 6	Sample End
NRPN 7	Loop Enable
NRPN 8	Loop Start
NRPN 9	Loop End
NRPN 10	Loop Crossfade
NRPN 11	LFO Trigger Mode

NRPN 12	LFO Sync Mode
NRPN 13	LFO Waveform
NRPN 14	LFO Rate
NRPN 15	LFO Synced Rate
NRPN 16	LFO Delay
NRPN 17	LFO Fade
NRPN 18	LFO Detuning
NRPN 19	LFO Pitch Modulation
NRPN 20	Amp Env Attack
NRPN 21	Amp Env Decay
NRPN 22	Amp Env Sustain Level
NRPN 23	Amp Env Release
NRPN 24	Filter Env Attack
NRPN 25	Filter Env Decay
NRPN 26	Filter Env Sustain Level
NRPN 27	Filter Env Release
NRPN 28	Filter Enable
NRPN 29	Filter Mode
NRPN 30	Filter Frequency
NRPN 31	Filter Resonance
NRPN 32	Filter Key-Follow
NRPN 33	LFO Filter Modulation
NRPN 34	Filter Velocity Tracking
NRPN 35	Envelope Filter Modulation

Proton

NRPN 0	Polyphony
NRPN 1	Pitch Bend Range
NRPN 2	Glide
NRPN 3	Transpose
NRPN 4	Velocity->Volume
NRPN 5	Volume
NRPN 6	Stereo Detuning
NRPN 7	Carrier Harmonic
NRPN 8	Carrier Fine Tuning
NRPN 9	Carrier Key Follow
NRPN 10	Modulator Harmonic
NRPN 11	Modulator Fine Tuning
NRPN 12	Modulator Waveform
NRPN 13	FM Amount
NRPN 14	Modulator Pitch Envelope Modulation
NRPN 15	FM Envelope Modulation
NRPN 16	FM LFO Rate
NRPN 17	FM LFO Modulation
NRPN 18	Vibrato LFO Rate
NRPN 19	Vibrato Amount
NRPN 20	Modulator Freq Attack
NRPN 21	Modulator Freq Decay
NRPN 22	Modulator Freq Sustain Level
NRPN 23	Modulator Freq Release
NRPN 24	FM Attack
NRPN 25	FM Decay
NRPN 26	FM Sustain Level
NRPN 27	FM Release
NRPN 28	Amp Attack
NRPN 29	Amp Decay
NRPN 30	Amp Sustain Level
NRPN 31	Amp Release

Model 12

NRPN 0	Master Volume
NRPN 1	Master Tune
NRPN 2	Master Stretch
NRPN 3	Voice Selection

NRPN 101	Part 1 Mute
NRPN 102	Part 1 Solo
NRPN 103	Part 1 Link
NRPN 104	Part 1 Output Assignment
NRPN 105	Part 1 Output Assignment
NRPN 106	Part 1 Sample Start
NRPN 107	Part 1 Sample Start Velocity Mod
NRPN 108	Part 1 Sample Start Random Mod
NRPN 109	Part 1 Standard Tune
NRPN 110	Part 1 Tune Velocity Mod
NRPN 111	Part 1 Tune Random Mod
NRPN 112	Part 1 Tune Decay Amount
NRPN 113	Part 1 Tune Decay Time
NRPN 114	Part 1 Tune Mode
NRPN 115	Part 1 Stretch
NRPN 116	Part 1 PureDSP Tune
NRPN 117	Part 1 Filter Frequency
NRPN 118	Part 1 Filter Frequency Velocity Mod
NRPN 119	Part 1 Filter Frequency Random Mod
NRPN 120	Part 1 Filter Frequency Decay Amount
NRPN 121	Part 1 Filter Frequency Decay Time
NRPN 122	Part 1 Filter Mode
NRPN 123	Part 1 Resonance
NRPN 124	Part 1 Resonance Velocity Mod
NRPN 125	Part 1 Resonance Random Mod
NRPN 126	Part 1 Drive
NRPN 127	Part 1 Drive Velocity Mod
NRPN 128	Part 1 Drive Random Mod
NRPN 129	Part 1 Sample End
NRPN 130	Part 1 Decay Time
NRPN 131	Part 1 Decay Mode
NRPN 132	Part 1 Volume
NRPN 133	Part 1 Volume Velocity Mod
NRPN 134	Part 1 Pan
NRPN 135	Part 1 Pan Velocity Mod
NRPN 136	Part 1 Pan Random Mod
NRPN 137	Part 1 Send 1 Level
NRPN 138	Part 1 Send 2 Level
NRPN 201	Part 2 Mute
NRPN 202	Part 2 Solo
NRPN 203	Part 2 Link
NRPN 204	Part 2 Output Assignment
NRPN 205	Part 2 Output Assignment
NRPN 206	Part 2 Sample Start
NRPN 207	Part 2 Sample Start Velocity Mod
NRPN 208	Part 2 Sample Start Random Mod
NRPN 209	Part 2 Standard Tune
NRPN 210	Part 2 Tune Velocity Mod
NRPN 211	Part 2 Tune Random Mod
NRPN 212	Part 2 Tune Decay Amount
NRPN 213	Part 2 Tune Decay Time
NRPN 214	Part 2 Tune Mode
NRPN 215	Part 2 Stretch
NRPN 216	Part 2 PureDSP Tune
NRPN 217	Part 2 Filter Frequency
NRPN 218	Part 2 Filter Frequency Velocity Mod
NRPN 219	Part 2 Filter Frequency Random Mod
NRPN 220	Part 2 Filter Frequency Decay Amount
NRPN 221	Part 2 Filter Frequency Decay Time
NRPN 222	Part 2 Filter Mode
NRPN 223	Part 2 Resonance
NRPN 224	Part 2 Resonance Velocity Mod

NRPN 225	Part 2 Resonance Random Mod
NRPN 226	Part 2 Drive
NRPN 227	Part 2 Drive Velocity Mod
NRPN 228	Part 2 Drive Random Mod
NRPN 229	Part 2 Sample End
NRPN 230	Part 2 Decay Time
NRPN 231	Part 2 Decay Mode
NRPN 232	Part 2 Volume
NRPN 233	Part 2 Volume Velocity Mod
NRPN 234	Part 2 Pan
NRPN 235	Part 2 Pan Velocity Mod
NRPN 236	Part 2 Pan Random Mod
NRPN 237	Part 2 Send 1 Level
NRPN 238	Part 2 Send 2 Level
NRPN 301	Part 3 Mute
NRPN 302	Part 3 Solo
NRPN 303	Part 3 Link
NRPN 304	Part 3 Output Assignment
NRPN 305	Part 3 Output Assignment
NRPN 306	Part 3 Sample Start
NRPN 307	Part 3 Sample Start Velocity Mod
NRPN 308	Part 3 Sample Start Random Mod
NRPN 309	Part 3 Standard Tune
NRPN 310	Part 3 Tune Velocity Mod
NRPN 311	Part 3 Tune Random Mod
NRPN 312	Part 3 Tune Decay Amount
NRPN 313	Part 3 Tune Decay Time
NRPN 314	Part 3 Tune Mode
NRPN 315	Part 3 Stretch
NRPN 316	Part 3 PureDSP Tune
NRPN 317	Part 3 Filter Frequency

NRPN 318	Part 3 Filter Frequency Velocity Mod
NRPN 319	Part 3 Filter Frequency Random Mod
NRPN 320	Part 3 Filter Frequency Decay Amount
NRPN 321	Part 3 Filter Frequency Decay Time
NRPN 322	Part 3 Filter Mode
NRPN 323	Part 3 Resonance
NRPN 324	Part 3 Resonance Velocity Mod
NRPN 325	Part 3 Resonance Random Mod
NRPN 326	Part 3 Drive
NRPN 327	Part 3 Drive Velocity Mod
NRPN 328	Part 3 Drive Random Mod
NRPN 329	Part 3 Sample End
NRPN 330	Part 3 Decay Time
NRPN 331	Part 3 Decay Mode
NRPN 332	Part 3 Volume
NRPN 333	Part 3 Volume Velocity Mod
NRPN 334	Part 3 Pan
NRPN 335	Part 3 Pan Velocity Mod
NRPN 336	Part 3 Pan Random Mod
NRPN 337	Part 3 Send 1 Level
NRPN 338	Part 3 Send 2 Level
NRPN 401	Part 4 Mute
NRPN 402	Part 4 Solo
NRPN 403	Part 4 Link
NRPN 404	Part 4 Output Assignment
NRPN 405	Part 4 Output Assignment
NRPN 406	Part 4 Sample Start
NRPN 407	Part 4 Sample Start Velocity Mod
NRPN 408	Part 4 Sample Start Random Mod
NRPN 409	Part 4 Standard Tune
NRPN 410	Part 4 Tune Velocity Mod

NRPN 411	Part 4 Tune Random Mod
NRPN 412	Part 4 Tune Decay Amount
NRPN 413	Part 4 Tune Decay Time
NRPN 414	Part 4 Tune Mode
NRPN 415	Part 4 Stretch
NRPN 416	Part 4 PureDSP Tune
NRPN 417	Part 4 Filter Frequency
NRPN 418	Part 4 Filter Frequency Velocity Mod
NRPN 419	Part 4 Filter Frequency Random Mod
NRPN 420	Part 4 Filter Frequency Decay Amount
NRPN 421	Part 4 Filter Frequency Decay Time
NRPN 422	Part 4 Filter Mode
NRPN 423	Part 4 Resonance
NRPN 424	Part 4 Resonance Velocity Mod
NRPN 425	Part 4 Resonance Random Mod
NRPN 426	Part 4 Drive
NRPN 427	Part 4 Drive Velocity Mod
NRPN 428	Part 4 Drive Random Mod
NRPN 429	Part 4 Sample End
NRPN 430	Part 4 Decay Time
NRPN 431	Part 4 Decay Mode
NRPN 432	Part 4 Volume
NRPN 433	Part 4 Volume Velocity Mod
NRPN 434	Part 4 Pan
NRPN 435	Part 4 Pan Velocity Mod
NRPN 436	Part 4 Pan Random Mod
NRPN 437	Part 4 Send 1 Level
NRPN 438	Part 4 Send 2 Level
NRPN 501	Part 5 Mute
NRPN 502	Part 5 Solo
NRPN 503	Part 5 Link
NRPN 504	Part 5 Output Assignment
NRPN 505	Part 5 Output Assignment
NRPN 506	Part 5 Sample Start
NRPN 507	Part 5 Sample Start Velocity Mod
NRPN 508	Part 5 Sample Start Random Mod
NRPN 509	Part 5 Standard Tune
NRPN 510	Part 5 Tune Velocity Mod
NRPN 511	Part 5 Tune Random Mod
NRPN 512	Part 5 Tune Decay Amount
NRPN 513	Part 5 Tune Decay Time
NRPN 514	Part 5 Tune Mode
NRPN 515	Part 5 Stretch
NRPN 516	Part 5 PureDSP Tune
NRPN 517	Part 5 Filter Frequency
NRPN 518	Part 5 Filter Frequency Velocity Mod
NRPN 519	Part 5 Filter Frequency Random Mod
NRPN 520	Part 5 Filter Frequency Decay Amount
NRPN 521	Part 5 Filter Frequency Decay Time
NRPN 522	Part 5 Filter Mode
NRPN 523	Part 5 Resonance
NRPN 524	Part 5 Resonance Velocity Mod
NRPN 525	Part 5 Resonance Random Mod
NRPN 526	Part 5 Drive
NRPN 527	Part 5 Drive Velocity Mod
NRPN 528	Part 5 Drive Random Mod
NRPN 529	Part 5 Sample End
NRPN 530	Part 5 Decay Time
NRPN 531	Part 5 Decay Mode
NRPN 532	Part 5 Volume
NRPN 533	Part 5 Volume Velocity Mod
NRPN 534	Part 5 Pan

NRPN 535	Part 5 Pan Velocity Mod
NRPN 536	Part 5 Pan Random Mod
NRPN 537	Part 5 Send 1 Level
NRPN 538	Part 5 Send 2 Level
NRPN 601	Part 6 Mute
NRPN 602	Part 6 Solo
NRPN 603	Part 6 Link
NRPN 604	Part 6 Output Assignment
NRPN 605	Part 6 Output Assignment
NRPN 606	Part 6 Sample Start
NRPN 607	Part 6 Sample Start Velocity Mod
NRPN 608	Part 6 Sample Start Random Mod
NRPN 609	Part 6 Standard Tune
NRPN 610	Part 6 Tune Velocity Mod
NRPN 611	Part 6 Tune Random Mod
NRPN 612	Part 6 Tune Decay Amount
NRPN 613	Part 6 Tune Decay Time
NRPN 614	Part 6 Tune Mode
NRPN 615	Part 6 Stretch
NRPN 616	Part 6 PureDSP Tune
NRPN 617	Part 6 Filter Frequency
NRPN 618	Part 6 Filter Frequency Velocity Mod
NRPN 619	Part 6 Filter Frequency Random Mod
NRPN 620	Part 6 Filter Frequency Decay Amount
NRPN 621	Part 6 Filter Frequency Decay Time
NRPN 622	Part 6 Filter Mode
NRPN 623	Part 6 Resonance
NRPN 624	Part 6 Resonance Velocity Mod
NRPN 625	Part 6 Resonance Random Mod
NRPN 626	Part 6 Drive
NRPN 627	Part 6 Drive Velocity Mod
NRPN 628	Part 6 Drive Random Mod
NRPN 629	Part 6 Sample End
NRPN 630	Part 6 Decay Time
NRPN 631	Part 6 Decay Mode
NRPN 632	Part 6 Volume
NRPN 633	Part 6 Volume Velocity Mod
NRPN 634	Part 6 Pan
NRPN 635	Part 6 Pan Velocity Mod
NRPN 636	Part 6 Pan Random Mod
NRPN 637	Part 6 Send 1 Level
NRPN 638	Part 6 Send 2 Level
NRPN 701	Part 7 Mute
NRPN 702	Part 7 Solo
NRPN 703	Part 7 Link
NRPN 704	Part 7 Output Assignment
NRPN 705	Part 7 Output Assignment
NRPN 706	Part 7 Sample Start
NRPN 707	Part 7 Sample Start Velocity Mod
NRPN 708	Part 7 Sample Start Random Mod
NRPN 709	Part 7 Standard Tune
NRPN 710	Part 7 Tune Velocity Mod
NRPN 711	Part 7 Tune Random Mod
NRPN 712	Part 7 Tune Decay Amount
NRPN 713	Part 7 Tune Decay Time
NRPN 714	Part 7 Tune Mode
NRPN 715	Part 7 Stretch
NRPN 716	Part 7 PureDSP Tune
NRPN 717	Part 7 Filter Frequency
NRPN 718	Part 7 Filter Frequency Velocity Mod
NRPN 719	Part 7 Filter Frequency Random Mod
NRPN 720	Part 7 Filter Frequency Decay Amount

NRPN 721	Part 7 Filter Frequency Decay Time
NRPN 722	Part 7 Filter Mode
NRPN 723	Part 7 Resonance
NRPN 724	Part 7 Resonance Velocity Mod
NRPN 725	Part 7 Resonance Random Mod
NRPN 726	Part 7 Drive
NRPN 727	Part 7 Drive Velocity Mod
NRPN 728	Part 7 Drive Random Mod
NRPN 729	Part 7 Sample End
NRPN 730	Part 7 Decay Time
NRPN 731	Part 7 Decay Mode
NRPN 732	Part 7 Volume
NRPN 733	Part 7 Volume Velocity Mod
NRPN 734	Part 7 Pan
NRPN 735	Part 7 Pan Velocity Mod
NRPN 736	Part 7 Pan Random Mod
NRPN 737	Part 7 Send 1 Level
NRPN 738	Part 7 Send 2 Level
NRPN 801	Part 8 Mute
NRPN 802	Part 8 Solo
NRPN 803	Part 8 Link
NRPN 804	Part 8 Output Assignment
NRPN 805	Part 8 Output Assignment
NRPN 806	Part 8 Sample Start
NRPN 807	Part 8 Sample Start Velocity Mod
NRPN 808	Part 8 Sample Start Random Mod
NRPN 809	Part 8 Standard Tune
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NRPN 811	Part 8 Tune Random Mod
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NRPN 813	Part 8 Tune Decay Time
NRPN 814	Part 8 Tune Mode
NRPN 815	Part 8 Stretch
NRPN 816	Part 8 PureDSP Tune
NRPN 817	Part 8 Filter Frequency
NRPN 818	Part 8 Filter Frequency Velocity Mod
NRPN 819	Part 8 Filter Frequency Random Mod
NRPN 820	Part 8 Filter Frequency Decay Amount
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NRPN 822	Part 8 Filter Mode
NRPN 823	Part 8 Resonance
NRPN 824	Part 8 Resonance Velocity Mod
NRPN 825	Part 8 Resonance Random Mod
NRPN 826	Part 8 Drive
NRPN 827	Part 8 Drive Velocity Mod
NRPN 828	Part 8 Drive Random Mod
NRPN 829	Part 8 Sample End
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NRPN 833	Part 8 Volume Velocity Mod
NRPN 834	Part 8 Pan
NRPN 835	Part 8 Pan Velocity Mod
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NRPN 837	Part 8 Send 1 Level
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NRPN 902	Part 9 Solo
NRPN 903	Part 9 Link
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NRPN 905	Part 9 Output Assignment
NRPN 906	Part 9 Sample Start

NRPN 907	Part 9 Sample Start Velocity Mod
NRPN 908	Part 9 Sample Start Random Mod
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NRPN 912	Part 9 Tune Decay Amount
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NRPN 916	Part 9 PureDSP Tune
NRPN 917	Part 9 Filter Frequency
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NRPN 922	Part 9 Filter Mode
NRPN 923	Part 9 Resonance
NRPN 924	Part 9 Resonance Velocity Mod
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NRPN 926	Part 9 Drive
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NRPN 932	Part 9 Volume
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NRPN 936	Part 9 Pan Random Mod
NRPN 937	Part 9 Send 1 Level
NRPN 938	Part 9 Send 2 Level
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NRPN 1002	Part 10 Solo
NRPN 1003	Part 10 Link
NRPN 1004	Part 10 Output Assignment
NRPN 1005	Part 10 Output Assignment
NRPN 1006	Part 10 Sample Start
NRPN 1007	Part 10 Sample Start Velocity Mod
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NRPN 1014	Part 10 Tune Mode
NRPN 1015	Part 10 Stretch
NRPN 1016	Part 10 PureDSP Tune
NRPN 1017	Part 10 Filter Frequency
NRPN 1018	Part 10 Filter Frequency Velocity Mod
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NRPN 1020	Part 10 Filter Frequency Decay Amount
NRPN 1021	Part 10 Filter Frequency Decay Time
NRPN 1022	Part 10 Filter Mode
NRPN 1023	Part 10 Resonance
NRPN 1024	Part 10 Resonance Velocity Mod
NRPN 1025	Part 10 Resonance Random Mod
NRPN 1026	Part 10 Drive
NRPN 1027	Part 10 Drive Velocity Mod
NRPN 1028	Part 10 Drive Random Mod
NRPN 1029	Part 10 Sample End
NRPN 1030	Part 10 Decay Time

NRPN 1031	Part 10 Decay Mode
NRPN 1032	Part 10 Volume
NRPN 1033	Part 10 Volume Velocity Mod
NRPN 1034	Part 10 Pan
NRPN 1035	Part 10 Pan Velocity Mod
NRPN 1036	Part 10 Pan Random Mod
NRPN 1037	Part 10 Send 1 Level
NRPN 1038	Part 10 Send 2 Level
NRPN 1101	Part 11 Mute
NRPN 1102	Part 11 Solo
NRPN 1103	Part 11 Link
NRPN 1104	Part 11 Output Assignment
NRPN 1105	Part 11 Output Assignment
NRPN 1106	Part 11 Sample Start
NRPN 1107	Part 11 Sample Start Velocity Mod
NRPN 1108	Part 11 Sample Start Random Mod
NRPN 1109	Part 11 Standard Tune
NRPN 1110	Part 11 Tune Velocity Mod
NRPN 1111	Part 11 Tune Random Mod
NRPN 1112	Part 11 Tune Decay Amount
NRPN 1113	Part 11 Tune Decay Time
NRPN 1114	Part 11 Tune Mode
NRPN 1115	Part 11 Stretch
NRPN 1116	Part 11 PureDSP Tune
NRPN 1117	Part 11 Filter Frequency
NRPN 1118	Part 11 Filter Frequency Velocity Mod
NRPN 1119	Part 11 Filter Frequency Random Mod
NRPN 1120	Part 11 Filter Frequency Decay Amount
NRPN 1121	Part 11 Filter Frequency Decay Time
NRPN 1122	Part 11 Filter Mode
NRPN 1123	Part 11 Resonance
NRPN 1124	Part 11 Resonance Velocity Mod
NRPN 1125	Part 11 Resonance Random Mod
NRPN 1126	Part 11 Drive
NRPN 1127	Part 11 Drive Velocity Mod
NRPN 1128	Part 11 Drive Random Mod
NRPN 1129	Part 11 Sample End
NRPN 1130	Part 11 Decay Time
NRPN 1131	Part 11 Decay Mode
NRPN 1132	Part 11 Volume
NRPN 1133	Part 11 Volume Velocity Mod
NRPN 1134	Part 11 Pan
NRPN 1135	Part 11 Pan Velocity Mod
NRPN 1136	Part 11 Pan Random Mod
NRPN 1137	Part 11 Send 1 Level
NRPN 1138	Part 11 Send 2 Level
NRPN 1201	Part 12 Mute
NRPN 1202	Part 12 Solo
NRPN 1203	Part 12 Link
NRPN 1204	Part 12 Output Assignment
NRPN 1205	Part 12 Output Assignment
NRPN 1206	Part 12 Sample Start
NRPN 1207	Part 12 Sample Start Velocity Mod
NRPN 1208	Part 12 Sample Start Random Mod
NRPN 1209	Part 12 Standard Tune
NRPN 1210	Part 12 Tune Velocity Mod
NRPN 1211	Part 12 Tune Random Mod
NRPN 1212	Part 12 Tune Decay Amount
NRPN 1213	Part 12 Tune Decay Time
NRPN 1214	Part 12 Tune Mode
NRPN 1215	Part 12 Stretch
NRPN 1216	Part 12 PureDSP Tune

NRPN 1217	Part 12 Filter Frequency
NRPN 1218	Part 12 Filter Frequency Velocity Mod
NRPN 1219	Part 12 Filter Frequency Random Mod
NRPN 1220	Part 12 Filter Frequency Decay Amount
NRPN 1221	Part 12 Filter Frequency Decay Time
NRPN 1222	Part 12 Filter Mode
NRPN 1223	Part 12 Resonance
NRPN 1224	Part 12 Resonance Velocity Mod
NRPN 1225	Part 12 Resonance Random Mod
NRPN 1226	Part 12 Drive
NRPN 1227	Part 12 Drive Velocity Mod
NRPN 1228	Part 12 Drive Random Mod
NRPN 1229	Part 12 Sample End
NRPN 1230	Part 12 Decay Time
NRPN 1231	Part 12 Decay Mode
NRPN 1232	Part 12 Volume
NRPN 1233	Part 12 Volume Velocity Mod
NRPN 1234	Part 12 Pan
NRPN 1235	Part 12 Pan Velocity Mod
NRPN 1236	Part 12 Pan Random Mod
NRPN 1237	Part 12 Send 1 Level
NRPN 1238	Part 12 Send 2 Level

CHAPTER 3 MOTU Instruments Lite Soundbank

OVERVIEW

The MOTU Instruments Lite soundbank provides approximately 1.2 GB of multi-sample instruments. You'll find acoustic drum kits, electronic drum kits, acoustic pianos, electric pianos, a church organ, electric organs, acoustic guitars, electric guitars, strings, brass, woodwinds, ethnic instruments, voices, percussion and more.

This versatile collection covers a variety of musical genres and styles.

How it works **93**
Accessing sounds **93**
A quick tour of UVIWorkstation **93**

HOW IT WORKS

To use the MOTU Instruments soundbank, you must load it into the free UVIWorkstation application, which runs as an instrument plug-in within Performer Lite.

ACCESSING SOUNDS

1 Create a UVIWorkstation instrument track in Performer Lite, as usual: choose *Project menu > Add Track > Instrument Track > UVI > UVIWorkstation (stereo)*.

2 Double-click the browser bar at the top of the window to open the browser.

3 Double-click the UFS file to open it, and then navigate through the preset menus.

4 Double-click the preset you wish to load.

The Soundbanks section of the browser

For your convenience, the Soundbanks section of the UVIWorkstation browser will display any UFS soundbank files placed in the following location:

macOS: /library/Application Support/UVISoundbanks

Windows: /Program Files/ UVISoundbanks

Simply place the MOTU Instruments.UFS file in this location and restart UVIWorkstation. The soundbank will automatically appear in the Soundbanks section of the browser.

You can also customize this location. For details, see the UVIWorkstation PDF manual.

A QUICK TOUR OF UVIWORKSTATION

For complete information about UVIWorkstation, please refer to the UIVWorkstation PDF manual. This section provides a quick tour to get you started.

The UVIWorkstation window

When you load a MOTU Instruments soundbank preset in UVIWorkstation, you'll see something similar to Figure 3-1, which shows the Edit tab for the *Twelve String* guitar preset.

Browsing presets

Click the Browser button or double-click the preset to browse other presets in the soundbank. For details about using the browser see the UVIWorkstation manual.

The Edit tab and macro controls

The Edit tab displays Macro controls (knobs and buttons) that let you modify the sound of the preset. The Macro Help section provides tips for using the macro controls for each preset.

The Effects tab

Click the Effects tab (Figure 3-1) to gain access to the effects rack for the preset (Figure 3-2 on page 94). This gives you more detailed control over the effects being used for the preset. Here, you can add, remove and modify effects from UVIWorkstation's considerable arsenal of effects processors.

Figure 3-2: The Effects tab.

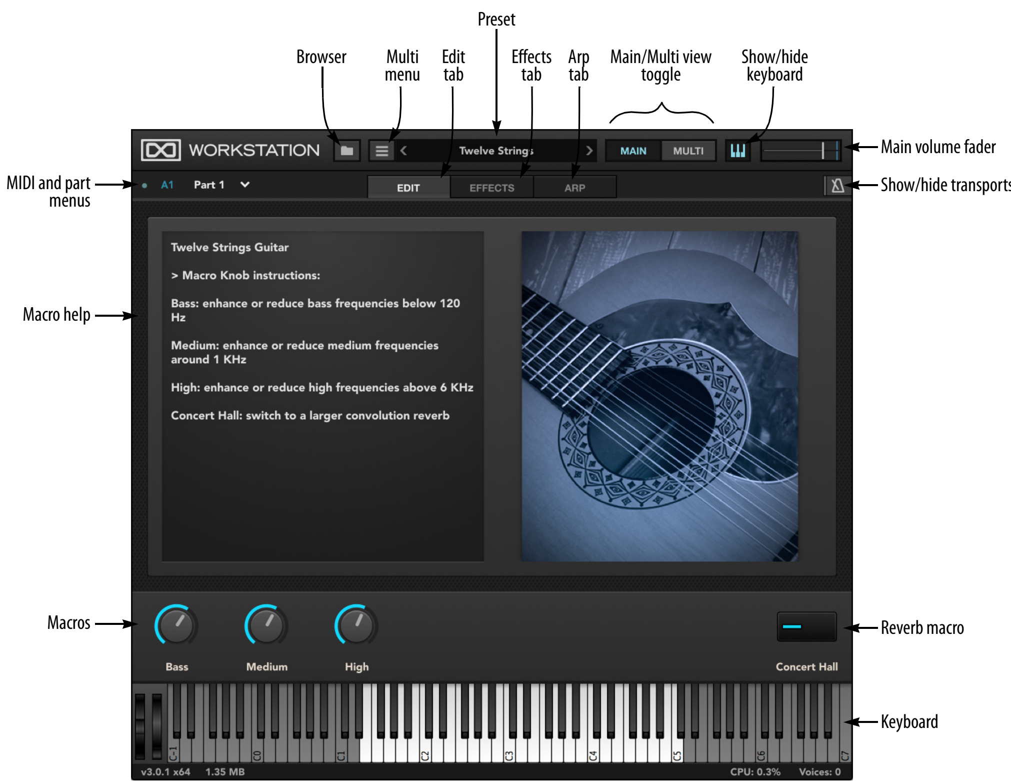

Figure 3-1: A twelve-string guitar preset from the MOTU Instruments soundbank.

Multi view

Multi view (Figure 3-1) allows you to load up to four different presets simultaneously and then assign them to either different MIDI channels (for multi-timbral operation), or the same MIDI channel, where you can then map them to different regions on your controller keyboard to mix and blend them. For further details, please refer to the UVIWorkstation PDF manual.

On-screen keyboard

Use the on-screen keyboard (Figure 3-1) to be able to play the preset using your on-screen cursor.

Other features

To learn about other features in UVIWorkstation, including the Preset Browser, Transport Bar and Arp (Arpeggiator) Tab, please refer to the UVI Workstation manual.

Part 3

Appendices

APPENDIX A Glossary

32-bit: Refers to the number of bits used to describe an individual sample. In Performer Lite instruments, audio is generated internally at 32-bit resolution (32 bits to describe each sample).

Amplitude envelope: Also see *Envelope*. Modulates the volume of a sound over time according to the settings of each envelope stage.

Audio Unit (AU): A standard Mac OS X plug-in format. Programs like Performer Lite can host AU plug-ins.

Automation: The process of changing a plug-in parameter over time. For example, a filter cutoff frequency can be automated by sending a stream of automation data values that change over time.

Bend: The process of changing the pitch of a note smoothly, or the range over which such a change can occur.

Buffer: A small portion of computer memory that is used to temporarily store audio data as it is being moved or processed. Larger buffer sizes can increase system latency. See "Buffer Size" on page 23 in the *DP Getting Started* Guide.

Bus: A connection from one point in a mixing environment to another.

Cents: a unit of measurement for pitch transposition. There are 100 cents in a semitone and twelve semitones in a octave. 50 cents is a quarter of a tone.

CoreAudio: The term used to refer collectively to the built-in audio services provided by Mac OS X.

CoreMIDI: The term used to refer collectively to the built-in MIDI services provided by Mac OS X.

CPU: Central Processing Unit. This is the "brain" of a computer, where the majority of the computing is done.

Cutoff frequency: The frequency above or below which a digital signal processing filter is applied.

Effects: Signal processing applied to an audio signal.

Envelope: A modulation profile that changes over time, applied to an audio signal. For example, in samplers, amplitude envelopes are applied to samples to produce more dynamic-sounding notes, with distinct attack, sustain and release characteristics.

Filter: An audio signal processor that modifies an incoming signal in some way.

Filter modulation: The process of changing the filter cutoff frequency over time. For details, see "Modulation" on page 172.

Fine-tune: A pitch control setting that allows you to change pitch in cents (a hundredth of a semitone). Also see *cents*.

Frequency: The rate at which an audio signal oscillates. Also see *cutoff frequency*.

Gain: Volume, amplitude.

General MIDI: A standardized, basic sound set with a standard organization adopted by the electronic music instrument industry to provide users with a familiar sound set, regardless of the instrument being used to produce the sounds.

Hardware buffer size: The size of a small amount of computer memory used to transfer digital audio data to and from external audio hardware. See “Buffer Size” on page 23 in the *DP Getting Started* Guide.

High Pass Filter: A signal processor that allows frequencies above its threshold to go through and mutes frequencies below its threshold.

HPF: See *High Pass Filter*.

Instance: An instantiated plug-in (see *Instantiate* below).

Instantiate: The process of opening a plug-in within Performer Lite as the host software.

Insert: A point in a signal chain where an additional signal path loop (out and then back in again) can be added.

I/O buffer size: See *Buffer*.

Key follow: The amount of keyboard tracking that can be applied to an oscillator.

Layer: Multiple presets (patches) that are played simultaneously by the same note.

Latency: a very short delay that can occur between when a MIDI note is played and the resulting sound is triggered. See “Buffer Size” on page 23 in the *DP Getting Started* Guide for further information.

LFO: See *Low Frequency Oscillator*.

Low Frequency Oscillator: A low frequency signal that is used as a control signal for a signal processor (such as a filter).

Low Pass Filter: A signal processor that allows frequencies below its threshold to go through and mutes frequencies above its threshold.

LPF: See *Low Pass Filter*.

MAS: A plug-in format for Performer Lite.

MIDI: Musical Instrument Digital Interface. A command and control protocol for electronic musical instruments and software.

Modulation: The process of modifying a signal over time.

Mono: One channel.

Multi-timbral: Producing (or the ability to produce) more than one type of instrument or sound at a time.

Normalize: To boost the amplitude of an audio signal by whatever constant amount is needed to make the loudest peak reach digital full scale (zero dB).

Note-on/off velocity: A parameter of MIDI note data event that specifies the strength of the attack and release of the note.

Octave: A frequency that is higher or lower by a factor of 2. For example, the A above middle C is 440 Hz. An octave higher is 880 Hz. Two octaves higher is 1760 Hz.

Off-velocity: See *Note-on/off velocity*.

On-velocity: See *Note-on/off velocity*.

Parallel filters/effects: Two or more signal processors that can be applied independently to the same input signal.

Plug-in: A piece of software that operates within a host application.

Polyphony: The characteristic of sounding two or more notes at the same time. More specifically, this setting determines the maximum number of notes a part can play simultaneously.

Preset: A reference to a patch, which is a "snapshot" of all of the settings in plug-in.

RAM: Random Access Memory. This is the portion of a computer where data is temporarily stored during the computer's operation. When you restart or shut off the computer, RAM is flushed.

Release velocity: See *note-off velocity*.

Resonance: A boost in amplitude around the cutoff frequency. Also see *cutoff frequency*.

Serial effects: Two or more signal processors that can be applied one after the other to an audio signal.

Stack: Multiple presets (patches) that are played simultaneously by the same note.

Threshold: A specific frequency, amplitude or other audio signal characteristic that is used as a trigger.

Trigger: Something that initiates something else. For example, a MIDI note-on event can *trigger* a sound.

Velocity: See *note-on/off velocity*.

Virtual instrument: Software that produces sounds in a similar fashion to real acoustic or electronic instruments.

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