

bi:tek

organic synthesizer

User Guide v2.0



Credits

Project Management: Taiho Yamada

Programming: Wolfram Franke

Additional Programming: David Rowland

User Interface Design: Kristina Childs

Sound Design: Taiho Yamada, Mike Wall, Ed Ten Eyk, Christian Halten

Sample Content: Mike Wall, Taiho Yamada, James Woodburn

Manual: Wolfram Franke, Holger Steinbrink

Table of Contents

Credits	2
Table of Contents	3
1. Introduction	4
Conventions	5
2. Installation & Authorisation	6
2.1 Windows PC	6
2.2 OS X	6
2.3 Linux Ubuntu	7
2.4 Authorisation	7
3. Basic Operation	8
3.1 Loading and Saving Instruments	8
3.2 Control Elements	9
4. The Basic User Interface	11
4.1 Header Section	12
4.2 Keyboard Section	13
4.3 Arpeggiator Section	13
5. The Wild User Interface	14
5.1 Biotek 2 Ring	14
5.2 Master Controls	15
5.3 Macro Controls	15
5.4 Sound Layer Section	15
6. The Edit User Interface	17
6.1 Sound Layer Section	17
6.2 Instrument Page	18
6.3 Sound Page	19
6.4 Oscillator Page (Oscs)	22
6.5 Filter / Amp Page	33
6.6 Effects Page	36
6.7 Arpeggiator (Arp) Page	42
6.8 Matrix Page	45
6.9 Modulation Sources	45
6.10 Modulation Assignment	52
7. Appendix	55
7.1 Modifiers and their Operations	55
8. Product Support	59

1. Introduction

Thank you for purchasing our Biotek 2 Organic Synthesizer.

We hope you enjoy it as much as we do, have a lot of fun with it and make great music with it.

Biotek 2 is a premium quality instrument that combines an advanced synthesis engine with a stunning array of samples sounds from natural, urban and mechanical environments. Built on the entirely new Acktion platform, organic “synthesizer” sounds like nothing else.

Biotek 2's Wild user interface is designed to place a wide range of sound sculpting tools right at your fingertips. Rather than a confusing mix of menus and layers, the 20+ controls on the Wild interface give you the ideal set of variable parameters for each sounds.

The Edit pages let you dive much deeper into the synthesis engine and offer lots of parameters to control nearly everything.

The deep sound creation possibilities include four sound layers with four oscillators that offers the typical analog waveforms and granular sample playback, extensive frequency modulation and a separate white and pink noise generator. Followed by two serial multimode filters and a drive module with several types and a four-band equalizer with shelf and bell curves.

Each sound is mixed into four effect busses, each featuring a sum overdrive, a compressor, a modulation effect like chorus, flanger, phaser or sum filter, and a space effect like delay or clocked delay, or natural, plate or non-linear reverb.

Freely assignable modulations with an easy-to-understand user interface offer maximum flexibility easily surpassing even expensive hardware synthesizers. Biotek 2 offers 8 LFOs, 4 envelopes and a wide range of sources.



Conventions

Instrument

An Instrument is a collection of one or more Sounds including all sound data and global instrument parameters.

Sound

An Instrument can contain one or more Sound layers that can be edited separately.



Read more how to select and edit Sound layers in the chapter “6. The Edit User Interface” on page 17.

2. Installation & Authorisation

The installation including all sample and sound data requires approximately 3 GB space on your hard drive.

Biotek 2 can be installed and unlocked on up to 4 different computers with a single license.

2.1 Windows PC



Please observe the system requirements of your host application! Biotek 2 runs within 32 bit as well as 64 bit host applications.

1. Refer to the folder where the downloaded Biotek 2 zip archive is located.
2. Double click on the archive file to extract it.
3. Double click on the “install_biotek_2.exe” file. This launches a special installation program.
4. Follow the on-screen instructions.

Installation location Windows:

VST 32-bit: VST 64-bit: AAX 32-bit: AAX 64-bit:

C:\Program Files (x86)\Steinberg\VstPlugins\Biotek2.dll

C:\Program Files\Steinberg\VstPlugins\Biotek2 (64-bit).dll

C:\Program Files (x86)\Common Files\Avid\Audio\Plug-Ins\Biotek2.aaxplugin

C:\Program Files\Common Files\Avid\Audio\Plug-Ins\Biotek2 (64-bit).aaxplugin

2.2 OS X

1. Refer to the folder where the downloaded Biotek 2 zip archive is located.
2. Double click on the archive file to extract it.
3. Double click on the “Biotek 2.pkg” file. This launches a special installation program.
4. Follow the on-screen instructions.

Installation location Mac:

AU: /Library/Audio/Plug-Ins/Components/Biotek 2.component

VST: /Library/Audio/Plug-Ins/VST/Biotek 2.vst

AAX: /Library/Application Support/Avid/Audio/Plug-Ins/Biotek 2.aaxplugin

2.3 Linux Ubuntu

1. Copy the Biotek 2.so file into your VST plugins directory. Common locations are shown below.
2. Ensure your host is set to scan the location
3. Perform a plugin scan in your host

Common Linux VST directories:

/usr/lib/vst /

usr/local/lib/vst

~/.vst

Please follow these instructions to install the Biotek 2 content

1. Create a Biotek 2 directory at "~/Tracktion/Biotek 2/" (may already exist)
2. Copy the "Instruments" content to "~/Tracktion/Biotek 2/Instruments"
3. Copy the "Samples" content to "~/Tracktion/Biotek 2/Samples"
4. Restart your host and any open instances of Biotek 2

You should end up with two folders:

"~/Tracktion/Biotek 2/Instruments" containing ".biotek 2instrument" files

"~/Tracktion/Biotek 2/Samples" containing ".acktionsamples" files

To authorize the plugin on Linux, you must use the offline authorization method.

2.4 Authorisation

Biotek 2 uses a software based copy protection and can only run with an activated license.

After installation, please load Biotek 2 as instrument within your DAW. the plug-in is currently locked and running in demo mode (a noise occurs periodically).

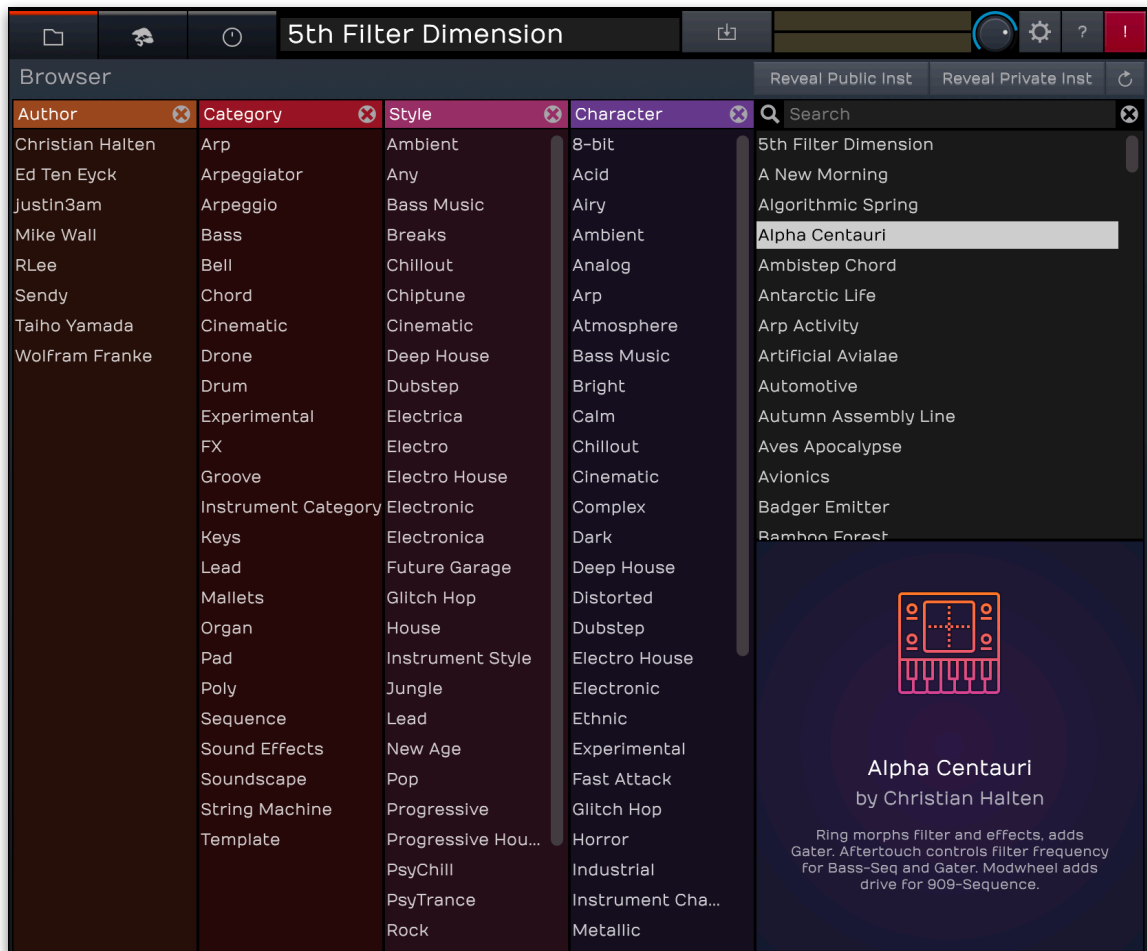
Please enter your email address and password from Tracktion Marketplace and click on "Register" to unlock Biotek 2. You need an active internet connection for this procedure.

You can alternatively click on the "Offline Unlock" button, if you computer has no internet connection. Please follow the onscreen-instructions.

After a successful authorisation, you can use Biotek 2 without limitations.

3. Basic Operation

3.1 Loading and Saving Instruments



Click on the folder button in the main area to open the **Browser** section. Here you can quickly locate e.g. categories or styles of Instruments by selecting one or more tag. Standard selection is cumulative unless the “command” (Mac) or “Ctrl” (Windows) key is used in conjunction with selection – this will select individual tags.

Use the **Search** panel in the upper right corner to perform a text search of the Instrument library.

To load a selected Instrument in the instrument result list, simply double click on its name.

To save an Instrument, click on the **Save** button in the Head section. User Instruments are added to the Instrument library. A newly saved preset is appended to the end of the file browse list. When launching Biotek 2, the files are scanned starting with your user’s instruments folder, followed by the public instruments folder (the one with the factory instruments). This results in the order that your creations come first followed by the ones from the factory sound bank.



3.2 Control Elements

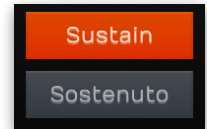
Dials

To set a value, click on the dial, hold down the mouse button and move the mouse up and down. A double click resets the parameter value to its default.



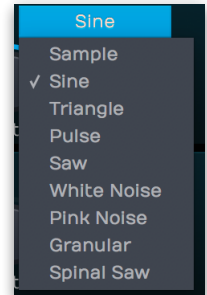
Buttons

A simple click on a button activates or deactivates its function. Buttons light up when they have been used.



Pop-up Menus

Click on the corresponding slot to open a pop up menu where you can choose the desired parameter.



Fader

Click on the corresponding fader and hold it. Move your mouse to the left or the right to change the value. Some faders need to move vertically to change values.



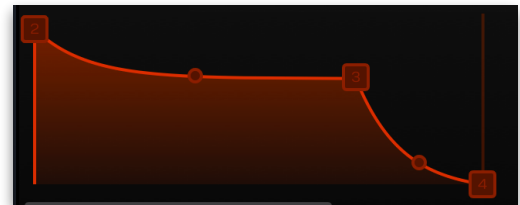
Value Selection Fields

Click in a Value Selection Fields and move the mouse up or down to easily change the parameter value. Double click in a Value Selection Field to set the parameter value to its default value.



Graphical Elements

Click on the corresponding graphic and move the mouse vertical or horizontal to change settings. This is valid for the BFM algorithms of the oscillators, the envelopes as well as the LFO slot slider.



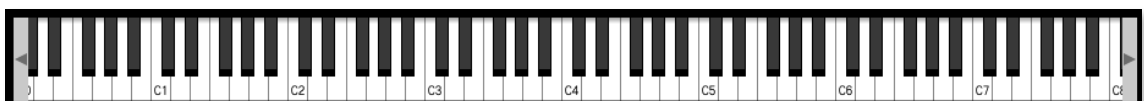
Wheels

To change the value, click on the wheel and drag it up or down. The Pitch wheel snaps automatically back into its center position as soon as you release the mouse button.



The Virtual Keyboard

Biotek 2 provides a virtual keyboard with 128 keys. Click on a key to play the corresponding note. The vertical position of the initial click of the key determines the velocity.



Biotek 2 Ring

Click on the yellow button within the X-Y pad and move the mouse. A single click on any position within the pad will set the button to this position.



4. The Basic User Interface

Biotek 2 is divided into different sections which are described in detail within the next chapters.



Hint: Some parameters may display a little right-pointing triangle symbol. If it is highlighted, this indicates, that this parameter is modulated by on or more modulation sources. Upon clicking such a highlighted Modulation indicator, it opens a popup menu showing the modulation sources and amounts. Clicking one of these entries activates the respective Modulation Assignment View. Read more about modulations on page 52.

The **Header Section (1)** includes basic functions as well as the Preset Loading page.

The **Main Section (2)** in Wild mode includes all macro controls to edit an Instrument quickly. It contains certain sub sections for controlling Oscillators, Filter, Envelopes, LFOs, Effects as well as the Flow LFOs. In Edit mode, this section includes lot of controls to edit an Instrument as well as a Sound in detail.



In the **Sound Layer (3)** section on the left you can make basic settings regarding the Instrument's Sound layers.

The **Virtual Keyboard (4)** provides a virtual keyboard with 128 keys

The **Arpeggiator Section (5)** on the left offers some basic arpeggiator parameters.

4.1 Header Section

The Header section offers global and basic settings.



- Click on the **Biotek** logo to open a pop-up window with additional information about the plug-in version number as well as the development team.
- Click on the **Folder** button to open the **Browser** section. Read more in the chapter “3.1 Loading and Saving Instruments” on page 8 of this manual.
- Click on the **Wild/Edit** buttons to change the view of the Main section. In Wild mode, only macro parameters and the Biotek 2 pad are visible. In Edit mode, all parameters are visible for detailed editing of a Instrument or Sound. Read more about Wild and Edit parameters on the corresponding chapters.
- The **Instrument name field** displays the name of the actual Instrument.
- The **Save** button lets you save the actual Instrument. You can also change the name of the stored Instrument.
- The **Master Volume** dial allows you to adjust overall volume of Collective. The meter displays the outgoing audio signal.
- Click on the **Global** button (Gear symbol) to open a pop-up window with additional parameters. Click the **Global** button again to close the Global menu.
 - **Zoom** - click on the corresponding zoom button to change the size of the Biotek 2 plug-in user interface. The regular size is 100%. You can also drag the handle in the lower right corner of Biotek 2 to change the size of the user interface to your needs.
 - **Multiple Cores** - when running multiple instances of Biotek 2, turning off the multi core processing can improve CPU performance on certain systems – typically older computers. We strongly recommend to activate this option.
 - **Control Behaviour** - when using a mouse to adjust the rotary controls, the input movement can be selected to be linear or circular. If set to “Circular”, you change a value of a rotary control by clicking on it and moving your mouse in a circular motion around the control. You don't need to stay inside the control bounds, though, you have a wide range to fine-tune the value. If set to “Linear”, rotary controls behave like value display controls. Moving the mouse up or right increases a value, moving it left or down decreases it.
 - **Ring Display** - to reduce CPU load, the Biotek 2 pad graphic can be simplified. “Eco” removes animation while “Low” reduces animation and graphic transitions.

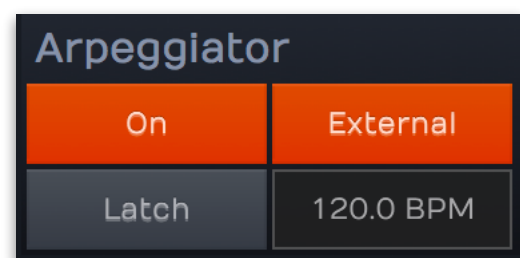
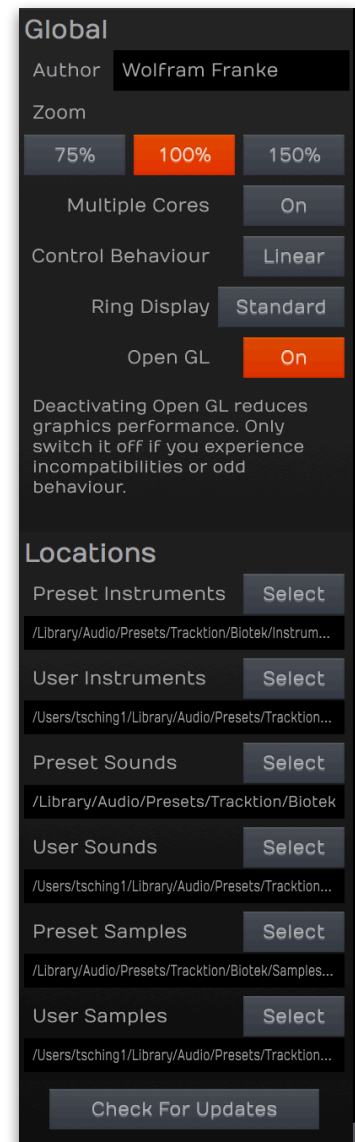
- **Open GL** - if deactivated, the graphic performance of Biotek 2 is reduced. We recommend to deactivate this parameter only if you experience incompatibilities or odd behavior.
- **Locations** - Click on the corresponding button to locate the folder on your hard disk where your Instruments, Sounds or Samples are placed. If you want to move the sample content you need to show Biotek 2 the corresponding data path.
- **Check For Updates** - opens a window that leads to the Tracktion Marketplace (if an internet connection is available). Here, available product updates will be shown for downloads.
- **Question Mark** button - click on this button to open the PDF manual of Biotek 2.
- **Panic** button: Sends and executes an “All Notes Off” command to terminate stuck notes. **Panic** immediately sets all envelopes to their release phases. Panic also stops the arpeggiator playing when running in Latch mode.

4.2 Keyboard Section

Biotek 2 provides a virtual keyboard with 128 keys and two wheels. Click on a key to play the corresponding note. The vertical position of the initial click of the key determines the velocity.

4.3 Arpeggiator Section

- **On**: Toggles the operation of the arpeggiator. The arpeggiator is used to sequence rhythmic patches, disabling it allows the user to access individual hits and elements.
- **External**: When enabled, it synchronizes the tempo to your host application. If disabled, you can specify a tempo in BPM.
- **Latch**: Enables the arpeggiator to remember the last set of keys played on the keyboard. When enabled, the arpeggiator sequence will continue to play even after the keys have been released. Once all keys are released, the next chord to be played will clear the latch memory and only the newly played chord will be arpeggiated. To stop the arpeggiator with the latch enabled, press the “On” button.
- **Tempo**: Adjusts the tempo when External is disabled. This can be useful if you prefer a rhythmic pattern to be half or double time. By pressing the “Option” (Mac) or “Alt” (Windows) key and click on **Tempo**, it will be set to default 120 BPM.



5. The Wild User Interface

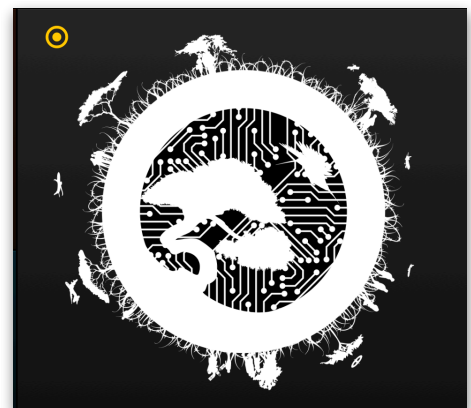
This user interface includes all macro controls to edit an Instrument quickly. All controls (except for the master section) are patch specific macro controls pre-assigned by the sound design team to numerous features.

Click on the Wild button in the Header section to select the Wild user interface



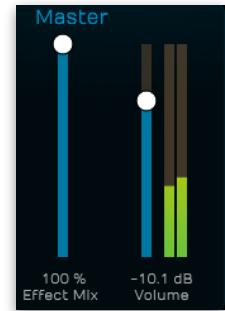
5.1 Biotek 2 Ring

The Biotek 2 Ring is a pad controller which lets you drastically change the selected Instrument. Use the orange pad controller to freely move it in X and Y axis. Based on the assigned synthesis parameters, the Biotek 2 Ring can produce subtle sound changes as well as extrem morphings. Read more about modulation controller assignment on page 52 of this manual.



5.2 Master Controls

The Master controls are universal to all patches and allows you to adjust overall Effect Mix and Master Volume of Biotek 2.



5.3 Macro Controls

Controls can be positive or negative in value – for example the oscillator dials on the right image offers settings for the pitch of each oscillator.

Holding the “Option” (Mac) or “Alt” (Windows) key and clicking on any parameter sets the control to its default location.

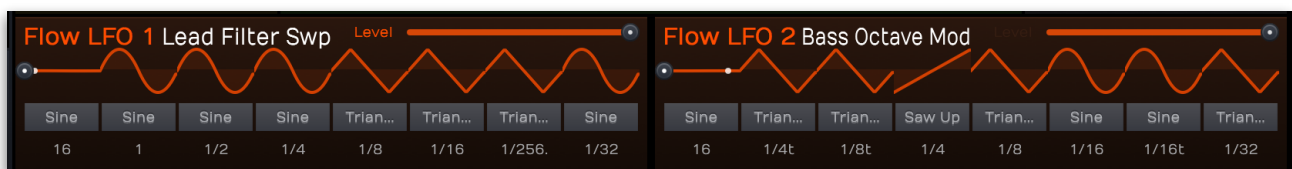
Based on the assigned synthesis parameters, the Macro controls can produce subtle sound changes as well as extrem morphings. Read more about Macro controller assignment on page 51 of this manual.



Macro Control Sections

Biotek 2 offers six Macro Control sections that can be named to best suit the instrument. Mostly, they will be used to control common synthesizer parameters as Oscillators, Filter, Envelopes and Effects. But you can create your own Macro sections without any limitations.

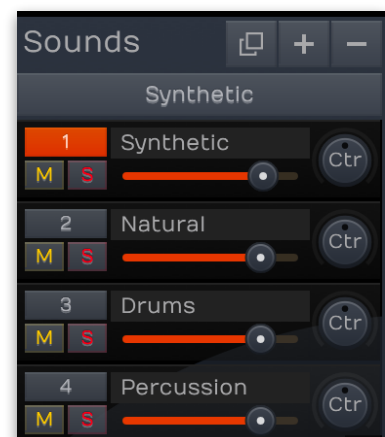
The Flow LFOs offer detailed parameters to control LFO 1 and 2. Read more about the LFOs in the chapter “Modulation Sources” on page 45.



5.4 Sound Layer Section

In the left area of the user interface you will find the controls to select and edit the Sound layers. Keep in mind that some Instruments might use only one Sound layer.

- **Select Sound Layer:** Click on the pop-up menu to select the desired Sound layer. The layer number lights up and the name of the Sound layer is displayed within the pop-up menu button. Alternatively you can directly click on the layer number to select it.
- **Sound Layer Name:** To rename a layer just click on its name and enter a new one.
- **Mute/Solo Sound Layer:** Click on the Mute button (M) to mute the desired Sound layer. Click on the Solo button (S) to solo the desired Sound layer. This mutes all other Sound layers. Click the Solo/Mute button again to deactivate the solo/mute function.



- **Sound Layer Volume:** Controls the output level of the corresponding Sound layer.
- **Sound Layer Panner:** Controls the panning of the corresponding Sound layer.

There are also some basic controls for adding/removing Sound layers.

- **Duplicate Layer:** Click on the **Duplicate** button to duplicate the actual Sound layer including all parameter settings.
- **Add Layer (+):** Click “+” to add a Sound layer with initial settings.
- **Remove Layer (-):** Click “-” to remove the actual Sound layer.

6. The Edit User Interface

This user interface includes dedicated subpages to edit an Instrument or Sound layer in detail.

Click on the Edit button in the Header section to select the Edit user interface.



Hint: Some parameters may display a little right-pointing triangle symbol. If it is highlighted, this indicates, that this parameter is modulated by on or more modulation sources. Upon clicking such a highlighted Modulation indicator, it opens a popup menu showing the modulation sources and amounts. Clicking one of these entries activates the respective Modulation Assignment View. Read more about modulations on page 52.

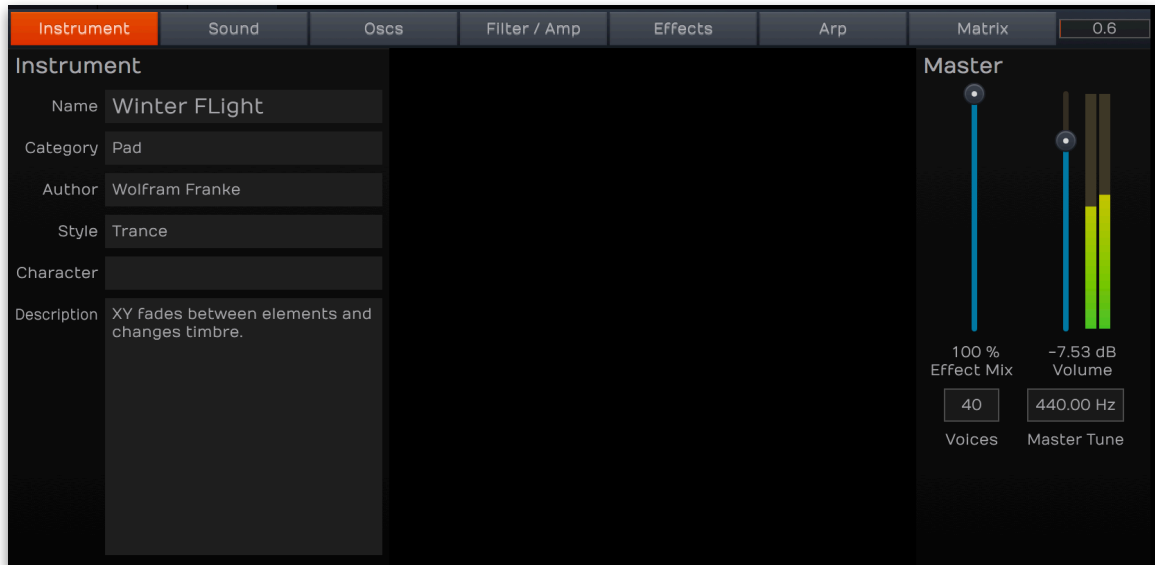


6.1 Sound Layer Section

For more information regarding the Sound Layer section please refer to chapter 5.4 on page 15.

6.2 Instrument Page

Click on the Instrument button to select the Instrument page. Here you can make some entries regarding the Instrument description. All entries will be automatically used as Tags to search and find an Instrument. You can also make some basic settings regarding voices and tuning.



Instrument Tags

- **Name:** Here you can enter a name for the Instrument. This name is automatically used when saving the Instrument. It will also be displayed in the Head section.
- **Category:** Here you can enter a sound category (e.g. Pad, Bass or Lead).
- **Author:** Here you can enter the name of the sound designer.
- **Style:** Here you can enter a sound or music style (e.g. Ambient, Electro)
- **Character:** Here you enter desired sound character attributes.
- **Description:** Here you can enter a short description of the Instrument (e.g. Wide ambient fluffy space pad).

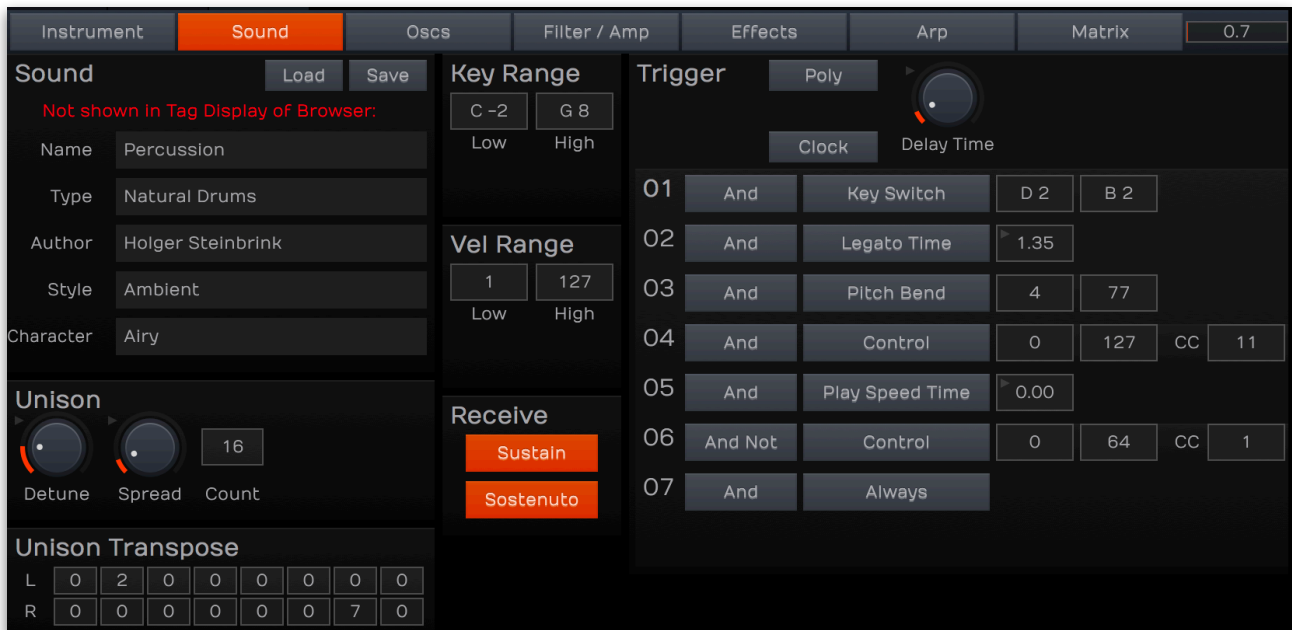
Master Section

The Master section controls are universal to all sound layers and allow you to adjust the Instrument's overall Effect Mix and Master Volume. There are two additional parameters:

- **Voice Count:** Voice count determines the maximum number of voices that the current Instrument produces. Decreasing the number of voices improves efficiency and therefore reduces load on CPU. If voice count is set too low, then "voice stealing" can occur with the possibility of audible artifacts.
- **Master Tuning:** 440.00 Hz is widely used as concert pitch in the United Kingdom and the United States. In continental Europe, the frequency of the note A commonly varies between 440 Hz and 444 Hz. In the period instrument movement, a consensus has arisen around a modern baroque pitch of 415 Hz (A_b of A 440), baroque for some special church music (Chorton pitch) at 466 Hz (A[#] of A 440), and classical pitch at 430 Hz.

6.3 Sound Page

Here you find the basic Sound controls as Unison Mode, Key Range and Velocity Range as well as a comprehensive Trigger management.



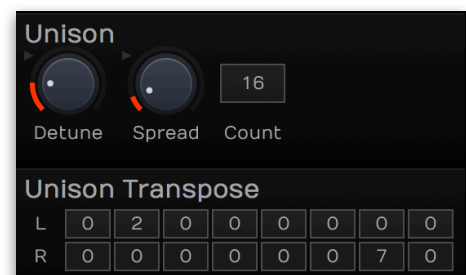
Sound Section

Here you can make some entries regarding the Sound description. Keep in mind that these entries will not be used as Tags.

- **Name:** Here you can enter a name for the actual Sound layer.
- **Type:** Here you can enter a type, e.g. Rhythm Layer.
- **Author:** Here you can enter the name of the sound designer.
- **Style:** Here you can enter a sound or music style of the actual Sound layer.
- **Timbre:** Here you can enter at the timbre of the actual Sound layer.

Unison

- **Detune:** Controls the detuning of the unison voices. Each voice is detuned differently.
- **Spread:** Spreads all unison voices automatically within the stereo panorama. Higher values will alternate the played voices between left and right panorama position.
- **Count:** Determines the number of played oscillators (up to 16). 1 means, that a note triggers only one voice.
- **Unison Transpose:** Here you can transpose every unison voice in semitones for the left and right panorama position (L + R) independently. The number of available transpose slots is defined with the **Count** parameter. Make sure to use corresponding **Spread** settings to clearly hear the stereo effect.

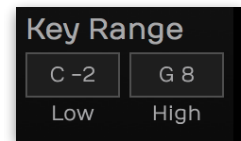




Please keep in mind that the chosen number of unison voices will respectively add more CPU usage.

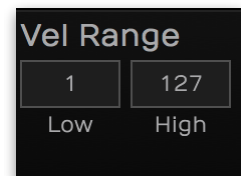
Key Range:

The key range can be restricted for the Sound layer's tone generation. Set the **Low** parameter to C- 2 and the **High** parameter to G 8 if you want to use the full keyboard range. Only notes with a key number higher/lower or equal to the selected values are passed through.



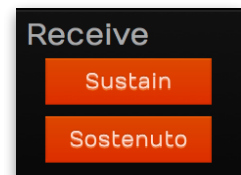
Vel(ocity) Range:

Low allows you to limit the velocity range in which the instrument is played. Only notes with a velocity higher or equal to the selected value are passed through. Set this parameter to 0, if you want to turn velocity switching off. **High** is the counterpart to the **Low** parameter. Only notes with a velocity lower or equal to the selected value are passed through.



Receive

- **Sustain:** Click on the Sustain button to determine, if the Sound reacts on incoming sustain controller data (MIDI CC# 64).
- **Sostenuto:** Click on the Sostenuto button to determine, if the Sound reacts on incoming sostenuto controller data (MIDI CC# 66). Similar to sustain, it holds only notes that were played when the pedal was pressed. Use it to hold chords and play melodies over the held chord.



Trigger


Biotek 2 offers a very comprehensive trigger system that allows you to control the triggering behavior for the actual Sound layer. You can use this section to create articulations or to add release sounds. The trigger table offers an overview for up to 8 logical operations that can be defined.

- **Poly** modes: **Poly** means that each voice is played normally like on a piano where you can play single melodies as well as chords. **Mono** results in only one sounding voice, no matter how many notes you play. When you play a number of notes without releasing older notes and then release the last played note, the previous older note can be heard. If you release that note, again the previous note can be heard and so on until you've released all notes. **XOR 1 ... 3** are allocation modes especially made for percussion sounds and this mode works over all sounds in all parts of the current Instrument. When a note is played, any other voice on any sound with the same XOR allocation setting is stopped and the voice of the new note sounds. This setting is used e.g. on hihat sounds where an open hihat sound cuts off the closed hihat sound in a different part and vice versa. Other typical percussion instruments might include an open and closed triangle, a slowly and quickly stroked guiro, a brushed and stroked snare or an open and muted crash cymbal. Biotek 2 offers three XOR groups so that

you can have an alternating hihat, a triangle and a guiro (or other sounds or combinations) at once in an Instrument.

- **Delay Time:** Determines the time, the trigger condition is delayed.
- **Clock:** If activated, the **Delay Time** can be set in musical values.
- **Trigger Table:** Here you can combine up to 8 conditions using logical operations. If you set up the first condition (every trigger type unequal to “Always”, the second condition is shown automatically. For each of the conditions, you can make the following settings in the first column: “And” combines two conditions logically. The expression is true if both conditions become true. “Or” combines two conditions logically. The expression is true if one of the conditions becomes true. “And Not” inverts the condition. A condition that was previously true becomes false, and vice versa. In the second column, the trigger type can be selected. The following types are available: Always, Key Switch, Play Speed Time, Play Speed Clock, Legato Time, Legato Clock, Pitch Bend, Control. Based on the type settings, the third column offers specified parameters to define the trigger values of the selected type.

Trigger

Poly 

Clock **Delay Time**

01	And	Key Switch	D 2	B 2	
02	And	Legato Time	1.35		
03	And	Pitch Bend	4	77	
04	And	Control	0	127	CC 11
05	And	Play Speed Time	0.00		
06	And Not	Control	0	64	CC 1
07	And	Always			

6.4 Oscillator Page (Oscs)

An Oscillator is the most important module in any synthesizer. It is used to generate a waveform to produce any sound at all. This waveform, be it either repeating after each cycle (oscillating) or playing a more or less continuous stream of recorded data (sample playback) is played with a certain pitch or speed and sent to other synthesizer modules like a filter and/or amplifier performing further processing on this signal.

Biotek 2 features four Oscillators with typical analog waveforms and the ability to play back samples - either in a normal way or as granular stream. The Oscillators offer nearly the same parameters with the only exception that if a waveform or a (granular) sample is selected, the parameters differ.



Hint: Some parameters may display a little right-pointing triangle symbol. If it is highlighted, this indicates, that this parameter is modulated by on or more modulation sources. Upon clicking such a highlighted Modulation indicator, it opens a popup menu showing the modulation sources and amounts. Clicking one of these entries activates the respective Modulation Assignment View. Read more about modulation assignment on page 52.

Common

- **Ring Mod:** Controls the depth of ring modulation between all Oscillators 1 and Oscillator 2. Although it is located in the Common section, both Oscillators generate this modulation with equal parts.





Ring Modulation is similar in sound to Frequency Modulation, albeit with less complex harmonic content. A Ring Modulator combines two input signals in a way that it outputs all sum and all difference frequencies of all input frequencies. That means, if you have a sine wave oscillator with a frequency of 200 Hz and another one with a 300 Hz frequency, the ring modulator produces a signal consisting of a 100 Hz sine wave ($300 \text{ Hz} - 200 \text{ Hz} = 100 \text{ Hz}$) and a 500 Hz sine wave ($300 \text{ Hz} + 200 \text{ Hz} = 500 \text{ Hz}$). You can easily imagine that more complex waveforms produce far more complex output signals. Note that you shouldn't set the pitches of the two oscillators to similar or identical frequencies, since this might result in signals somewhere in the subsonic range or even 0 Hz (DC), although you can get interesting pumping effects with it.

- **FM Algorithm:** Click on the algorithm diagram and move the mouse up or down to select one of the 10 FM algorithms.

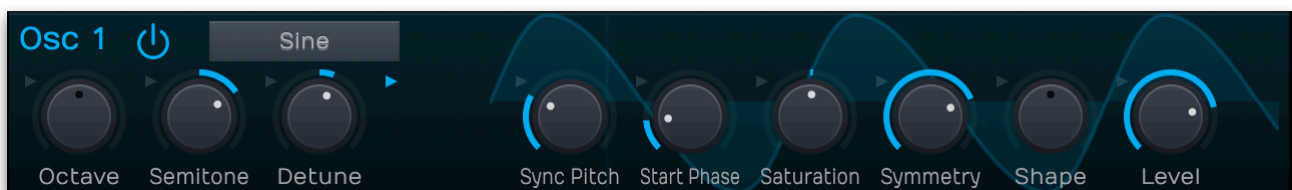


Frequency Modulation is a technique invented by John Chowning in the late 1960's. It is basically a very fast change in frequency of one signal (typically called Carrier) performed by the amplitude of a second signal (called Modulator). Typical FM synthesizers work with two sine oscillators. Changing the amplitude of the Modulator changes the harmonic content of the Carrier. Changing the frequency ratio of the Modulator changes the structure of the harmonic content, i.e. all harmonics, all odd harmonics or any other structure including harmonics that sound „out of tune“, since they might not relate directly to the pitch of the Carrier, which is typically used on bell sounds. Biotek 2 goes a little further. It not only offers periodic FM, it also offers 10 FM algorithms as known from the Yamaha DX synthesizers.

- **Octave:** Sets the basic pitch of all oscillators in steps of an octave.
- **Semitone:** Sets the pitch of all oscillators in semitone steps. The standard setting for this parameter is 0.
- **Detune:** Fine-tunes all oscillators in steps of 1/128th of a semitone.

Oscillators

- **Waveform:** Selects the waveform (Sample, Sine, Triangle, Pulse, Saw, Sample, White Noise, Pink Noise, Granular, Spinal Saw) for the Oscillator. Except for the types "Sample" and "Granular", the waveforms are periodic, i.e. oscillating. The shape of the selected waveform is displayed for the corresponding oscillator. **Sync Pitch, Start Phase, Saturation, Symmetry** and **Shape** have a direct influence on the graphical view of the waveform shape.



Sine oscillator type



Biotek 2's **Sine** oscillator type is actually more than that: it is able to play a sine wave but it can generate a good approximation of all other typically analog waveforms as well and any waveform in-between. But let's start with the sine wave: it is the most basic waveform consisting of only the fundamental, i.e. basic pitch, with no additional overtones. This results in the most pure sound. In analog circuitry, it is quite difficult to generate a perfect sine wave; to save cost it was often created with a simple, self-oscillating filter circuitry, but this wasn't perfect and introduced some form of distortion resulting in additional overtones. To simulate this behavior, Biotek 2 offers a number of parameters that can change the waveform from sine to triangle, sawtooth or square or a waveform in-between by either modifying the phase of the oscillator or clip its signal output.



The **Triangle** waveform contains only odd harmonics and can be used for drum or percussion sounds, or to create hollow pad sounds.



The **Pulse** waveform depending on the setting of Pulse Width, which is controlled by the **Symmetry** parameter. A value of 100 % results in a pure square wave only containing odd harmonics. The more you reduce the value, the more even harmonics are added to the signal, caused by the asymmetric duty cycle. A value around 1 % results in a very sharp sounding signal resembling a sawtooth wave with emphasized high frequencies. A value of 0% results in no audible signal.



The **Sawtooth** waveform contains all harmonics in falling magnitude and can be used for all kinds of synthesizer sounds.



The **Spinal Saw** oscillator type generates multiple detuned sawtooth waves in just one oscillator. It allows to create thick sounds without using a lot of processor power, so it is perfectly suited for pads, lead sounds or basses. A similar oscillator type was first introduced in Roland's JP-8000© virtual analog synthesizer and later got adopted by several hardware and software synthesizer manufacturers. Due to the limited processing power back then it had a reduced sound quality compared to simpler oscillator types. This, however, is not the case with the Spinal Saw. It uses the same high-quality algorithm the normal Saw type uses. And it goes to eleven.



The **Noise** generator offers white and pink noise. If any noise is selected, only the **Level** parameter is available.

Octave: Sets the basic pitch of the oscillator in steps of an octave.

- **Semitone:** Sets the pitch of the oscillator in semitone steps. The standard setting for this parameter is 0, but there are cases where different values are interesting as well.
- **Detune:** Fine-tunes the oscillator in steps of 1/128th of a semitone. The audible result of detuned oscillators is a Chorus or Flanger effect. Use a positive setting for one oscillator and an equivalent negative setting for another.
- **Level:** Controls the output level of the Oscillator.
- **Browse** (only available for Sample & Granular): Click on the Browse button to open a window, where you can select a sample. After loading, the sample name will be displayed next to the Browse button.




Read more about loading user samples on the page 31 of this manual.

- **Sample Start** (only available for Sample & Granular): When the Oscillator is set to play a sample or a grain stream, here you can set the sample start between 0 % and 100 %. Please note that the range covers the whole sample. A lot of samples somehow fade out to the end, especially drum or percussion samples, so a high value of **Sample Start** might only produce a very low output signal or no signal at all. Decrease **Sample Start** if this happens.
- **Key Shift** (only available for Sample & Granular): Transposes the multisample split points, but not the pitch of a sample map. Most samples are single zoned, but the drum kits and some bowls use multisample mappings.
- **Velocity Shift** (only available for Sample & Granular): Transposes the multisample velocity split points, but not the velocity of a sample map. Most samples are single zoned, but the drum kits and some bowls use multisample mappings.
- **Remove Sample** (only available for Sample & Granular): Click on the “x” button to remove the actual loaded sample.
- **Sync Pitch** (not available for Noise and Samples & Granular): Adds an synchronized slave oscillator to the selected one (Sine, Triangle, Saw or Pulse). Oscillator synchronization means, that the slave oscillator is reset with each full wave cycle of the master oscillator. It produces a rich sync-sound without using other oscillators as slave or master. The **Sync Pitch** parameter adjusts the pitch of the slave oscillator, producing the typical sync sound.
- **Start Phase** (not available for Noise, Samples & Granular): Controls the phase with which the Oscillator begins its cycle when a note is played. This parameter is shown in the waveform display as a vertical line with a rounded rectangular handle at its bottom end. A dotted line means that the start phase is random while a solid line shows the location in the waveform where the oscillator phase starts. Oscillators in classic analog synthesizers ran freely, which means they weren't stopped after a note had ended but oscillated inaudibly until a new note using a particular oscillator was played. Whether they were audible or not was controlled by the amplifier section, which typically attenuated the oscillator signal to silence when a key was released. Typically, you set **Startphase** to “Free” if you want to simulate a typical analog synthesizer or if you want to have some life in the sound. When you set up all oscillators to play the same waveform and roughly the same pitch, you should also set **Startphase** to “Free” to prevent flanging upon note start, an effect that plagues most sample-based synthesizers even today. You should set **Startphase** to the maximum magnitude of the waveform if you want a sharp click on note start. This is an easy and fast way to get nice and punchy bass drum or percussion sounds. You can reduce the click volume by moving **Startphase** away from this maximum magnitude until you are happy with the result. If you don't want a click upon note start, set **Startphase** to a point where the waveform crosses the center. On the **Sine** and **Triangle** waveform, you can simply move **Startphase** all the way to the right resulting in a start phase of 360° which is equal to 0°.

- **Saturation** (only available for Sine): Controls how much the waveform is driven into clipping. The clipping cuts the top and bottom of the waveform introducing sharp edges producing lots of overtones.
- **Symmetry** (only available for Sine and Pulse): Controls the asymmetry of the selected waveform. Move it to full right to get a perfectly symmetric waveform (100 %). Move it to the left to continuously introduce an asymmetry to the waveform. If the Pulse waveform is selected, **Symmetry** is used to change the pulsewidth.
- **Shape** (only available for Sine): Controls a symmetric phase modification of the waveform.
- **Cluster** (only available for Spinal Saw): Controls the number and level of the individual oscillators between 1 and 11 oscillators. At its leftmost position, it only plays one oscillator. Increasing the value gradually blends in more and more oscillators. Notice that in the lower range, only one additional oscillator is blend in while in higher value ranges, the oscillators are blend in quicker. This is because your ear easily differentiates between one and two oscillators but not so much between ten and eleven oscillators.
- **Detune** (only available for Spinal Saw): Controls the detuning between the oscillators. The first oscillator is not detuned, and the second to eleventh oscillators are detuned with varying strengths. So you must turn up Cluster at least a little bit to hear an effect. Low Detune values result in no or only a small detuning while higher values detune to roughly one semitone.

The Granular Oscillator



 Granular synthesis is a method of playing short sound snippets in quick succession. The resulting timbre might range from repetitive sounds over pitch shift and time stretch effects up to clusters of sounds with a random character. Such a sound snippet is called a "grain" and it is typically selected from a sample where its length is somewhere between 20 ms and 100 ms. Hence the name "granular synthesis". Successive grains might overlap or have gaps between them and each has its own set of playback parameters like pitch, length and a volume envelope (aka the **Window** parameter).

Some basic knowledge how to work with a granular oscillator:

1. **Sample:** At first, you need a sample to work with. In theory, single-cycle waveforms such as sine, sawtooth or pulse could also be used, but since they are static, it doesn't make too much sense to repeatedly start and stop them. A good sample for granular synthesis should have a uniform level with more louder parts than lower parts so that if the grains jump back and forth in it they always

have something audible to play. Acoustic instruments are suited quite well but you can also use drum or percussion loops or anything you like. The only way to find out if a sample is suited is to experiment with it.

2. **Number of Grains:** Granular synthesis works with a number of grains that might sound at once. The respective parameter is called **Density**, in other implementations it might be just called "Number", "Count" or "Inter On-Set Time". All of that only means how quickly the next grain is started. This is done in relation to the grain length so that the next grain is started when the former grain has reached e.g. half of its lifecycle, which would be a **Density** of 2. It might be interesting to start new grains really quickly to create sound clusters or very slowly to create repeating sound patterns.
3. **Window:** The window is the most important aspect of a grain. If a grain is just started and stopped after a couple of milliseconds, the sound is strongly colored spectrally by the hard on- and off-set of the grain. Sometimes you might want that, but most of the time you don't. Therefore, a small envelope must be applied to the grain that gradually blends it in and out. Unfortunately, this comes with a drawback: depending on the character of the sample and the sound you want to achieve, this window might smooth out the grains too much so that sharp transients e.g. at the start of a piano sample get lost. So, there is no "one size fits all" window. Smart people came up with different window functions that all have their strengths and weaknesses, where the most well-known ones are Paraboloid (aka cosine), Blackman, Von-Hann (aka raised cosine, sometimes only called Hann or Hanning) and Hamming. This granular oscillator type offers a good selection of them and it has some additional ones for special purpose. Furthermore, it is desirable to control the speed of the fade-in and -out and the stationary level in-between. Sometimes a very smooth fade-in and -out with no stationary level is wanted, sometimes a sharp attack followed by a smooth release is what you are after, and sometimes you might want a hard on and off switch.
4. **Pitch:** If all grains would always sound with the same pitch, the result would be quite boring. This might be needed for pitch shift or time stretch effects where you want to come as close to the original sound as possible, only with a different pitch or speed, but often you want to have some randomness or detuning happening to create rich and unpredictable sound clusters.
5. **Length:** The same as for the grain pitch applies for its length. Sometimes you want a determined length per grain but adding some randomness or key tracking to the grain length to spice a sound up. As already said, a grain is typically between 20 ms and 100 ms in length. That is the classic approach to granular synthesis. With this, you create typical time stretch effects but you have to keep in mind that the sound might be strongly colored. It more or less sounds like applying a comb filter to the signal. Longer lengths are also fun. With them, you can create rhythmic, repeating sound clusters, or lush, noisy pads.
6. **Randomness:** The key to interesting granular sounds is randomness. Sometimes you want each grain to be played slightly detuned for a rich sound or with varying length so that it doesn't sound like machine gun fire, or want to detune it heavily for noisy sound clusters. Since this is so common, most of the parameters also have an associated randomize parameter so that you don't need to set up modulation assignments for them.

The Granular Oscillator Parameters

- **Zoom View** (Magnifying glass symbol): Since the granular oscillator type has so many parameters, we have added a zoom view to each oscillator covering the top two thirds of the sound editor. To show the zoom view, click on the magnifying glass button of the respective oscillator. To close the zoom view, click on the close button marked with an **x** in the upper left corner.
- **Sample Selection (Browse)**: This is the same as in the Sample oscillator type.
- **Key Shift**: This parameter is the same as the one in the Sample oscillator type. This parameter only has an effect when using a multisample with various key zones. It shifts the sample mapping up- or downwards without changing the playback pitch of the sample. This sounds highly theoretical, so here is a practical example: A piano sample typically consists of individual samples for a certain key interval, because low piano notes have far more overtones than high notes and due to the body of the piano, some harmonics are more emphasized than others. Typically, this interval is three to five notes for each sample. Really high-quality piano multi-samples have samples for every note and those that conserve memory sometimes only have a sample per octave. Now consider playing the same note, e.g. C3. When Key Shift is 0, it plays the sample that has been originally assigned to that note. When lowering Key Shift to e.g. -12, the same note plays the sample from one octave lower, resulting in a much sharper sound character. Vice versa, when you set Key Shift to +12, the note becomes more muffled. You can use Key Shift for so-called "processed" sounds by increasing or lowering Key Shift. You can also modulate Key Shift, e.g. by one of the RND sources to randomly select a different sample for each new note. This can be fun with drum or percussion kits.
- **Vel Shift**: This parameter is the same as the one in the Sample oscillator type. This parameter is similar to **Key Shift**, but it shifts the velocity mapping. Therefore, it only has an effect when the multisample contains more than one velocity zone. Let's use the aforementioned piano sample as an example again: Piano notes sound brighter the harder you play a note on the keyboard. Therefore, piano multi-samples contain individual samples for soft and hard notes, typically between four and ten different samples. When you increase Vel Shift, the same note plays the sample at a higher velocity, resulting in a brighter sound. Decreasing Vel Shift makes the note softer. You can even use extreme values like +127, which means that the sample for the highest velocity is always played, no matter how soft you play.
- **Density**: Controls the number of grains that are sounding at once. The center position means that exactly one grain is sounding at a time. Higher values result in overlapping grains, which means that the next grain is started while the previous one is still sounding. This is one of the typical features of a granular oscillator. At its maximum position, the granular oscillator plays 20 grains at once. Internally, the volume of the grains is then reduced so that the perceived output volume stays roughly the same. Now, what about values below the

center position? They result in an increasingly longer time to start the next grain, allowing you to create rhythmic effects.

- **Length:** Controls the length of the grain that is about to be started. Very short values might color the resulting sound strongly but can be used for time-stretch or pitch-shift effects. In practice, values between 100 ms and 500 ms are a good start.
- **Ln Key** (Length Keytrack): Especially for short Length settings where the grain repetition is already in the audible frequency range, a Length Keytrack of +100% can be applied so that the resulting signal sounds more like a looped sample, i.e. the length follows the pitch. Other values can be used for special effects.
- **Ln Rnd** (Random Length): Controls the randomization amount of the grain lengths. Randomizing the grain length adds some life to the resulting sound. Sometimes, even a little randomness is enough. Other times, you might want to simulate the sound of water drops with all their unpredictability, so you would set Ln Rnd to quite a high value.
- **Pitch:** Controls the pitch of the grain that is about to be started. Why another Pitch parameter when there are already the usual Octave, Semitone, Detune and Pitch (Modulation Amount) controls of the oscillator? The usual pitch controls change or modulate the oscillator pitch continuously, while the dedicated Granular Oscillator Pitch only affects the next grain at its start. That pitch is then used as the base for all further normal pitch modulations. With this feature, you can create rich detuned sound clusters, especially when modulating it by any of the modulation sources. If you only need a random pitch for each grain, see the next parameter.
- **Pt Rnd** (Random Pitch): Controls the random pitch detuning amount of each grain that is about to be started. At its leftmost position, grains are not detuned, higher values create continuously stronger detuning. If you want to create rich sound clusters, set it a little higher than the leftmost position so that you can hear a nice beating in the sound.
- **Pos** (Position): Controls the start position offset of the grain that is about to be started. The granular oscillator has a global timeline. That means that there is a pointer which is advanced internally over the duration of a held note, as you know from a normal sample playback oscillator. The initial position offset of that pointer upon playing a note is controlled by the Position parameter, its speed is controlled by the Speed parameter (see below). When Position is set to its neutral position and Speed is set to 1, it means that the pointer moves from the start of the sample to the end with the same speed as a normal sample oscillator would do (and also loops if the sample contains a loop), triggering grains along its way. Each grain is then played back with its set pitch (see the Pitch parameter). Position is a bipolar parameter because it is an offset to the internal pointer rather than an absolute position. This allows you to modulate this parameter after note on to jump around in the sample timeline to your likeness. For example, if you want to assign an LFO to modulate Position over its whole range, you would set Speed to 0, Position to +50% and assign the LFO with an amount of +50% as well. Upon pressing a note, you should see

and hear that grains are started back and forth along the timeline.

- **Pos Rnd** (Random Position): Controls the maximum amount the position can be changed for each grain that is about to be started. The randomization is bipolar, meaning that the resulting grain could be started after or before the actual position. Using the previous example from the Position parameter, instead of setting up an LFO modulation you could also set Pos Rnd to +50% so that grains would be triggered at arbitrary points in the sample.
- **Speed**: Controls the speed of the internal position pointer, which is advanced over the duration of a held note. It does not control the playback speed of a grain, this is done by Pitch. As written above, when Speed is 1, the position pointer is advanced as a normal sample oscillator would do. When Speed is set to 0, the playback is "frozen in time", meaning that all following grains are started at the current position. Speed can also be negative to move the position pointer backwards, but again, the grains still play forward with their own pitch setting. Speed can go up to 10-fold, which results in a time compression of 1000%. Also, slow speeds like 0.1 or even closer to 0 might be interesting, which means that the sample needs 10 times the time to be played completely.
- **Spd Rnd** (Random Speed): Controls the randomization amount of the Speed parameter. The speed is randomized upon each newly started grain, so the result might be similar as if randomizing the Position parameter, but with more gradual changes.
- **Ramp**: Together with **Sym** and **Window**, this parameter sets up the grain window or volume envelope. A value of 0% means that the grain is played with full volume over its lifecycle, higher values result in longer on- and off-ramp times with shorter stationary times in-between. Consider a grain length of 2 seconds, **Sym** at neutral and a trapezoid window (i.e. linear fade-in and -out). With **Ramp** 0%, the grain plays with full volume for 2 seconds. With **Ramp** 50%, the grain fades in for 0.5 seconds, stays at full level for 1 second and fades out for another 0.5 seconds, totaling 2 seconds. With **Ramp** 100%, the grain fades in for 1 second and immediately fades out for 1 second, with no stationary time in-between.
- **Sym** (Ramp Symmetry): Controls the relation between fade-in and fade-out. In its neutral position, both fades are of equal length. Lowering the value shortens the fade-in time and prolongs the fade-out time accordingly. Turning it fully to the left results in an instant on-time followed by a long fade-out time. This parameter is important when you work with medium to long grain lengths where you want to preserve the percussive nature of the underlying sample. E.g., when set to neutral, a piano sample would gradually blend in removing the hammer sound and a lot of its character. There is even a special window function helping you with sounds like these: the Quadratic window together with Ramp set to 100% and Sym set to -100% results in an envelope with instant attack and a nice release similar to an exponential release.
- **Window** pop-up menu: Selects the grain window function. As written above, grains can create a lot of spectral coloring when starting and stopping them with full volume. Since there is no

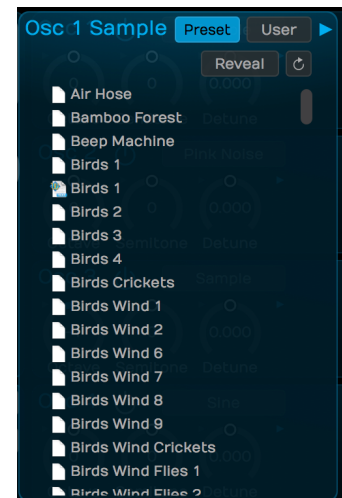
universal window function that works perfectly in all situations, you can choose the one that fits you most:

- **Paraboloid** — quite smooth cosine window keeping sidebands at a low level but still preserving a good portion of the attack and decay part of the grain.
- **Von Hann** — very smooth raised cosine window with much lower sideband levels but also suppressing a lot of the attack and decay.
- **Hamming** — the smoothest raised cosine window with very low sideband levels, best for very short grain lengths.
- **Trapez** — linear fade-in and -out, good for longer grain lengths.
- **Quadratic** — quadratic fade-in and -out approximating an envelope with exponential attack and decay, good with negative Symmetry settings.
- **Square** — just full volume, Ramp and Symmetry have no effect.
- **Start**: This parameter is the same as the one in the Sample oscillator type. It controls the start position of the sample. Note that the granular oscillator never plays grains before this start position so you can use it to limit the sample start to a certain position.

Loading User Samples

To load your own user samples into Biotek 2, you can either click on the **Load** button or drag & drop a samples from your operating system's file manager onto the sample name field of the respective oscillator (note that the oscillator type must be set to **Sample**). Upon dropping the sample file, it is copied into the user sample library folder.

Clicking on the **Load** button opens a browser window where you can toggle between **Preset** samples which comes with your Biotek 2 synthesizer and **User** samples. The user sample list depends on the content of Biotek 2's user sample folder on your hard disk. To locate and open this folder, click on the **Reveal** button. You can easily copy your desired samples into this folder. Make sure to click on the **Reload** button, if your user samples aren't displayed properly in the list.



Data path for the user sample folder:

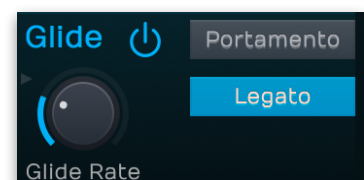
Windows: C:\ProgramData\Tracktion\Biotek 2\Samples

Mac: /Library/Audio/Presets/Tracktion/Biotek 2/Samples

Linux: /var/Tracktion/Biotek 2/Samples

Glide

Enables or disables the Glide effect. "Glide" or "Portamento" describes the continuous gliding from one note to another. This effect can be created on fretless stringed instruments or



some brass instruments (e.g. trombone). It is very common on synthesizers and used throughout all music styles. Please note that Glide affects the pitch of all oscillators.

- **Glide Rate:** Determines the glide time. Low values will give a short glide time in a range of milliseconds that gives a special character to the sound. High values will result in a long glide time of up to several seconds which can be useful for solo and effect sounds.
- **Portamento/Glissando:** If Portamento is activated, a continuous glide is performed on all new notes. If Glissando is activated, the Glide effect reacts in the same manner as by changing the pitch in semitones.
- **Legato:** If activated, a continuous glide is performed only when notes are played legato. Staccato played notes start on the exact pitch of their note.

6.5 Filter / Amp Page



Filter 1 / Filter 2

A filter is used in synthesizers to attenuate or emphasize certain frequency ranges of a source signal. Most filter types consist of a pass and a stop band, some have two pass or two stop bands. The pass band is the frequency range where the input signal can pass (almost) unaffected, the stop band is the frequency range that is attenuated, sometimes fully. Biotek 2 offers various Filter types abbreviated LP, BP, HP or BS, followed by a number denoting the slope in decibel per octave. Biotek 2 offers two filter stages that are arranged in serial wiring.

- Filter Type:** Click on the pop-up menu to select the desired filter type. Note that there are no Band Pass or Band Stop types for 6 dB. A 6 dB filter consists of one so-called pole, which can't be split in half. LP = A low pass filter lets low frequencies pass and attenuates high frequencies. It is the most common filter type found in all synthesizers with subtractive synthesis. BP = A band pass filter lets frequencies around its center frequency pass and attenuates frequencies further away from it. HP = A high pass filter lets high frequencies pass and attenuates low frequencies. BS = A band stop filter, also known as a band reject or notch filter, attenuates frequencies around its center frequency and lets frequencies pass that are further away from it. Comb = Differ from the other filter types greatly, because they don't actually damp any part of the signal, but instead add a delayed version of the input signal to the output. Redux = Is a decimator stage. It reduces the sample rate and bit depth of the incoming signal. Ladder types simulate a classic transistor ladder filter typically found in Moog® or Arp® synthesizers, among others. Since this filter type is typically composed of up



to four one-pole filters, the Ladder Filter is available with 6, 12, 18 and 24 dB slope, all with resonance up to self-oscillation and FM. Sallen-Key LP simulates a Sallen-Key filter found in the Korg® MS-10 and MS-20 synthesizer. This is a 12 dB low-pass filter with a very characteristic sound. Some love it, others hate it, we won't judge.

- **Cutoff:** Controls the cutoff (on low pass or high pass types) or center (on band pass or band stop types) frequency of the filter. If Comb is selected, the frequencies near the cutoff frequency are emphasized. If Redux is selected, the outgoing sample rate of the signal is reduced.
- **Resonance:** Controls the resonance of the filter if set to any of the 12 dB or 24 dB filter types. On 6 dB filter types, this control does nothing. On band stop types, it narrows the stop band, which is almost inaudible. If Redux is selected, the outgoing bit rate of the signal is reduced.
- **FM** = Sets the amount of frequency modulation that is applied to the filter by the oscillator bank output. Keep in mind that FM is only available for the 12 and 24 dB filter types.

Drive

The Drive stage offers an overdrive effect, also known as saturation or distortion. It is routed behind Filter 1 so it works on the filtered sum of all oscillator signals. This also means that the sound of the filter resonance is affected greatly by the Drive stage.



- **Drive Type:** Selects the **Drive Type**. Light, Medium and Hard are typical diode clipping circuitries in different strengths and symmetries. Clip is a hard clipping similar to that of a transistor distortion, Tube simulates a tube or valve overdrive and Fuzz results in a very strong distortion.
- **Drive:** Controls the amount of overdrive.

Amp

The Amplifier section controls the volume, stereo panning and Effects routing.



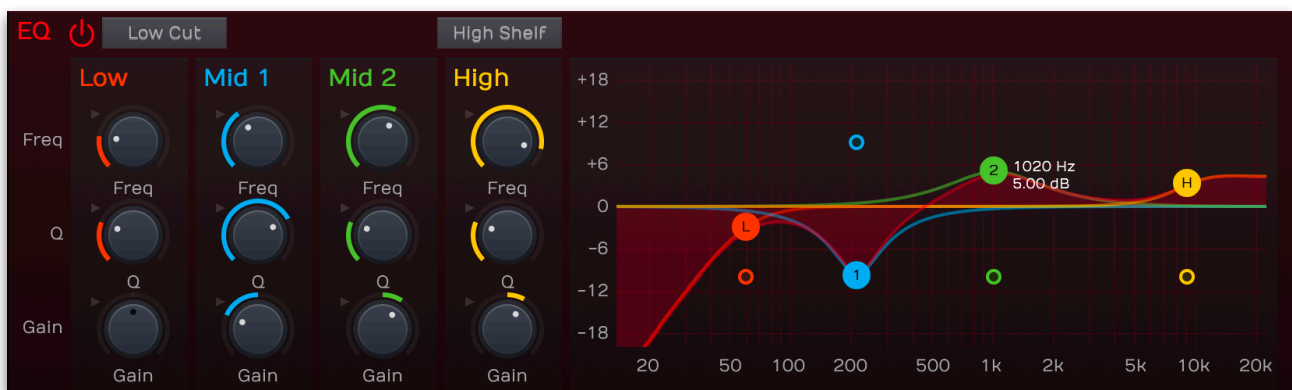
- **Volume:** The signal generated by the various tone generator and filter modules of Biotek 2 must ultimately be formed in volume to be heard. This is the job of the Amplifier. To do so, it needs some form of modulation input to control how a sound rises and decays in volume. This is achieved by **Envelope 4**, which is hardwired to **Volume**. Furthermore, a modulation assignment where the note velocity controls volume is already set up for you when you clear the sound.
- **Pan:** Controls the placement of the Sound layer in the stereo panorama. A modulation of this parameter can have a range of up to 200 %. That means, if you place the Sound layer fully to the left and modulate it e.g. with Envelope 1 with +100 %, the Sound wanders from 100 % left to 100 % right and vice versa. Keep this in mind when you assign a bipolar modulation source like an LFO or Pitch Bend. A depth of -100 % or +100 % would result in a signal sometimes stuck to the left or right. Reduce the depth slightly until you hear a nice continuous movement

of the LFO or a change in panorama when you move the Pitch Bend from its fully down or up position slightly to its center position.

- **Effects:** Controls the mixture between the dry signal output of the Sound layer and the wet effect signal output processed by the effect busses. A setting of 0% results in a completely dry signal, 50 % means that the signal is routed with equal volume into the effect unit and forwarded dry and 100 % means that no signal is sent out dry but it is routed into the Effect bus with full level. A setting of 100 % is recommended when you plan to use a dynamic effect such as a Compressor.

Equalizer

Biotek 2 offers a four-band equalizer per voice. Please read again: per voice. Contrary to other electronic instruments, the equalizer in Biotek 2 is not placed behind the sum, it is part of the Sound layer. While this doesn't make any difference when you just set it up statically, it certainly does so when you modulate its bands. You can edit the equalizer by using either the controls or the response graphic.



- **Low/High Band Selection:** Selects the filter type of the first and last band. A **Shelf** filter type looks like a normal or an inverted “S” and it sounds and works similar to a low or high pass filter with adjustable stop band attenuation or boost. A **Bell** filter looks like a normal or inverted bell and sounds and works similar to a band pass or notch filter, again with variable boost or attenuation and also with variable width. A **Cut** filter is nearly similar to the Shelf filter but with a sharper slew rate.
- **Frequency (Freq):** Controls the cutoff (shelf) or center (bell) frequency of the band. Moving the dial to the left results in a lower frequency of the band's cutoff or center frequency, to the right increases it.
- **Quality (Q):** Controls the quality or width of the band. When using a bell type, moving the dial upwards results in a narrower band, moving it downwards widens it. With the shelf type, it emphasizes or attenuates the frequency band around the cutoff frequency.
- **Gain:** Controls the gain boost or cut of the band. Clicking and moving the dial upwards increases the gain of the band, moving it downwards decreases it.

To edit the equalizer settings within the graphic, drag the desired button and move it to the left or the right to change the frequency. Move it up or down to change the gain. The small second button is for editing the quality of the desired band.

6.6 Effects Page

Here you find all parameters regarding the four effect busses. For each effect bus, one of 9 different effect types can be selected.



Click on one of the corresponding buttons (**Effect 1** to **Effect 4**) to activate it. The corresponding button will light orange.

Use the pop-up menu to select the desired effect. Based on the chosen effect type, different editing parameters are available.



Hint: Some parameters on the Edit pages may display a little right-pointing triangle symbol. If it is highlighted, this indicates, that this parameter is modulated by on or more modulation sources. Upon clicking such a highlighted Modulation indicator, it opens a popup menu showing the modulation sources and amounts. Clicking one of these entries activates the respective Modulation Assignment view.

Distortion

This effect offers an overdrive, also known as saturation or distortion. It affects the sum of all voices routed into this effect bus, which creates a different distortion depending on the number of voices you play due to intermodulation.

- **Type:** Selects the **Drive Type**. Light, Medium and Hard are typical diode clipping circuitries in different strengths and symmetries. Clip is a hard clipping similar to that of a transistor distortion, Tube simulates a tube or valve overdrive and Fuzz results in a very strong distortion.
- **Drive:** Controls the amount of overdrive.
- **Post Gain:** Controls the output level of the distorted signal.

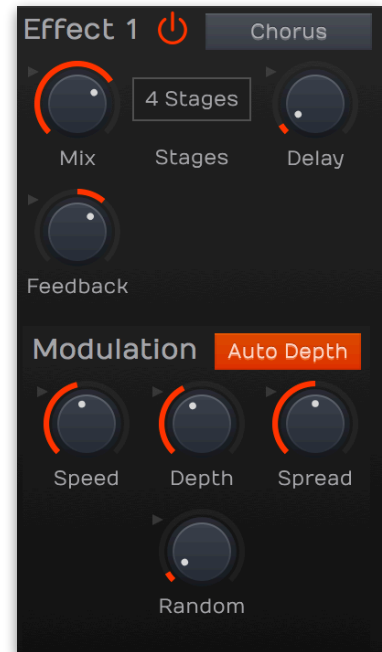


- **Tone:** Controls high-frequency attenuation of the drive signal.
- **Emphasis:** Controls the feedback of the **Tone** parameter.

Chorus

The Chorus multiplies the signal by generating delayed copies of it and slightly modulating the speed with which these delayed signals are played back resulting in a detune effect. The signal then sounds like an ensemble or, as the name says, a chorus. **Speed** and **Depth** work hand in hand in: when you increase Speed, Biotek 2 automatically decreases the depth in the chorus so that the detune effect stays the same. If you want to change the detune, adjust **Depth**.

- **Mix:** Adjusts the relation between the dry signal at the full left to the wet effect signal at the full right.
- **Stages:** Biotek 2 offers three chorus types with either 2, 4 or 6 independent delay lines, where half of them is routed to the left input and output and the other to the right input and output.
- **Delay:** Sets the minimum delay length. For a chorus effect, you typically increase it to around a tenth of its value range, for a flanger effect, you would set it to 0. The rest of the value range is just for your pleasure.
- **Feedback:** Controls the level of the signal that is fed back into the chorus. With a small setting of **Delay** and **Depth** and a high positive or negative setting of **Feedback** you can create a typical flanger effect, especially with 2 Stages.
- **Auto Depth:** Adjusts the strength of the modulation automatically to minimize artifacts. **Mod Depth** will be reduced all the more, the higher the settings for **Speed**.
- **Speed:** Controls the basic modulation speed. When using 4 Stage or 6 Stage, the 3rd and all following delays are modulated with a slightly higher or lower speed, depending on the setting of **Spread**.
- **Depth:** Adjusts the strength of the modulation, resulting in the detune effect. When using 4 Stage or 6 Stage, the depths of the 3rd and all following delays are slightly longer or shorter.
- **Spread:** Controls the deviation of the various delays. When **Spread** is set to full left, all modulations happen in sync albeit the 3rd and following stages are offset in phase to let the chorus still sound rich when using 4 Stage or 6 Stage. When you increase Spread, the delay deviations are increasingly stronger.
- **Random:** Adds a random movement to the **Spread** parameter. It produces linear ramps that adapt to **Speed**. This makes sure that the Chorus effect sounds not that static.



Phaser

The Phaser creates 2, 4, 6, 8, 10 or 12 phase-shifted signal copies creating equally spaced peaks and dips in the frequency spectrum. These phase-shifters are modulated in a way that a sweeping effect is created.

- **Mix:** Adjusts the relation between the dry signal at the full left to the wet effect signal at the full right.
- **Stages:** Sets the number of stages that are used for creating the phase-shifted signal copies.
- **Frequency:** Sets the center frequency of the phaser. Note that **Depth** and **Spacing** both influence the resulting frequencies of the phase shifters.
- **Spacing:** Controls the relative frequencies of the phase shifters. This spreads the peaks and dips in the frequency spectrum
- **Feedback:** Controls the level of the signal that is fed back into the phaser.
- **Modulation Speed:** Controls the modulation speed
- **Modulation Depth:** Sets the depth of the modulation.
- **Modulation Spread:** Controls the deviation between the left and right stereo side.



Compressor

The Compressor reduces signals which exceed a certain **Threshold** level in volume by an adjustable **Ratio**. The speed of this level reduction is controlled by **Attack**, when the signal starts to exceed the threshold, and **Release**, when it drops below the threshold again. When you set its parameters to extreme values, the signal might start to „pump“, so use your ears to set the values properly. This pumping effect might even be wanted, so use it to your liking. However, extremely short **Attack** or **Release** settings might create audible noise in the signal, which happens because the compressor changes its settings far in the audio range rather than creating a smooth envelope.

- **Gain:** Sets an input gain offset. Sometimes, the input signal is a little bit too low or loud, which you can change with **Gain**.
- **Threshold:** Sets the threshold above which the Compressor starts to reduce the signal level.
- **Make Up Gain:** Sets the output gain offset. A compressor reduces loud signals and **Make-up Gain** allows you to boost up this reduction back to a nice level.



- **Ratio:** Sets the compression ratio. 1:1 means that no compression takes place, while higher values result in an increasingly stronger compression.
- **Attack:** Sets the speed of the level reduction when the signal exceeds the **Threshold** level.
- **Release:** Sets the speed of the return to normal level when the signal drops below the **Threshold** level again.
- **Soft Knee:** With the setting of 0, the signals above the threshold is compressed instantly according to the set ratio (hard knee). Higher values let the onset of compression sound more gradual while producing a less drastic result.
- **Characteristic Diagram:** Displays the characteristic curve of the compression based on **Threshold** and **Ratio**.

Filter

The Filter effect filters the sum of all voices routed into this effect bus. Please refer to the chapter “Filter/Amp” for a thorough description of the various filter types.

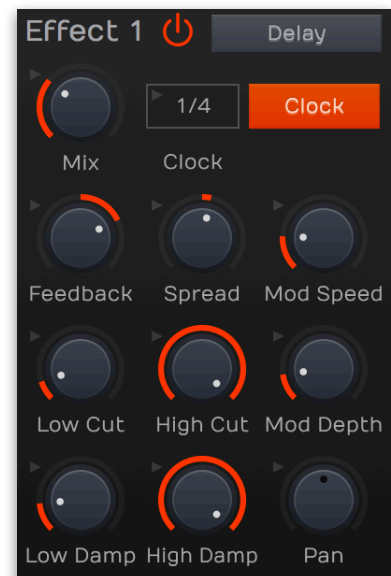
- **Type:** Selects the filter type (LP6/12/24, HP 6/12/24, BP12/24, BS 12/24, Comb, Redux, Ladder, Sallen-Key).
- **Cutoff:** Controls the cutoff (on low pass or high pass types) or center (on band pass or band stop types) frequency of the filter.
- **Resonance:** Controls the resonance of the filter if set to any of the 12 dB or 24 dB filter types. On 6 dB filter types, this control does nothing. On band stop types, it narrows the stop band, which is almost inaudible.



Delay

Biotek 2 offers a time-based and a clock-based stereo modulation delay. Stereo means that the left and the right signals are processed independently, while modulation means that the Delay can slightly or heavily detune its output periodically making the signal richer or creating special effects.

- **Mix:** Adjusts the relation between the dry signal at the full left to the wet effect signal at the full right.
- **Time:** Sets the delay time in milliseconds. This parameter is only editable when **Clock** is deactivated.
- **Clock:** Activate it to set the delay time in clock divisions synchronized to the current tempo.
- **Feedback:** Controls the level of the signal that is fed back into the delay.
- **Spread:** Adjusts the relative time of the first repetition compared to the delay length. A value of 0 % means that the delay of both stereo sides produce their first repetition at the



same time while smaller values result in the left side repeating first and larger values result in the right side repeating first. This allows you to create ping-pong delays. E.g., when setting **Clock** to 1/4 and **Spread** to -50 % you get an 1/8 ping pong delay. Other interesting **Spread** percentages are 33 % for triplets or 350 % for dotted notes, provided that **Clock** is set to a straight value. If **Clock** is set to a dotted value, you would, for example, set **Spread** to 333 % to have one of the repetitions straight. Also interesting are only slight deviations from 0 %, creating deeper delays. Sounds complicated, but offers a lot of fun.

- **Mod Speed:** Controls the modulation speed.
- **Low Cut:** Removes low frequencies before the signal enters the delay stage so that the first repetition is already filtered.
- **High Cut:** Removes high frequencies before the signal enters the delay stage so that the first repetition is already filtered.
- **Mod Depth:** Adjusts the strength of the modulation, resulting in a detune effect.
- **Low Damp:** Removes low frequencies in the feedback stage.
- **High Damp:** Removes high frequencies in the feedback stage of the delay meaning that each repetition has less and less high frequencies.
- **Pan:** Determines the stereo panning of the delay effect.

Reverb Natural / Reverb Plate / Reverb Non-Linear

Biotek 2 features high-quality reverb algorithms in different flavors. They all share common parameters.

The reverb natural algorithm simulates the reverberation that occurs in natural rooms. The names suggest the basic size of the simulated room although you can create artificially sounding rooms by changing the reverb time to unnatural values.

The Reverb Plate algorithm simulates a reverb plate. In the late 1950's, the German company EMT (Elektro-Mess-Technik) introduced huge steel plates fitted with a loudspeaker and a microphone. A signal would be sent to the loudspeaker and recorded by the microphone including the vibrations the steel plate generated resulting in a smooth but slightly artificial sounding reverb. The Plate Reverb and its digital simulation is suited well for drum and percussion sounds due to their dense initial reflections.

The non-linear reverb algorithm generates typical gated reverb effects (although it works completely differently) or inverse reverbs.

Since it doesn't have an internal feedback it must produce the whole reverb in one turn to create the gate effect. This means that longer reverb times sound less dense than shorter ones. If the reverb starts to sound not dense enough, you might try to increase the Decay of the sound, maybe a snare drum, by a little to smooth it out.

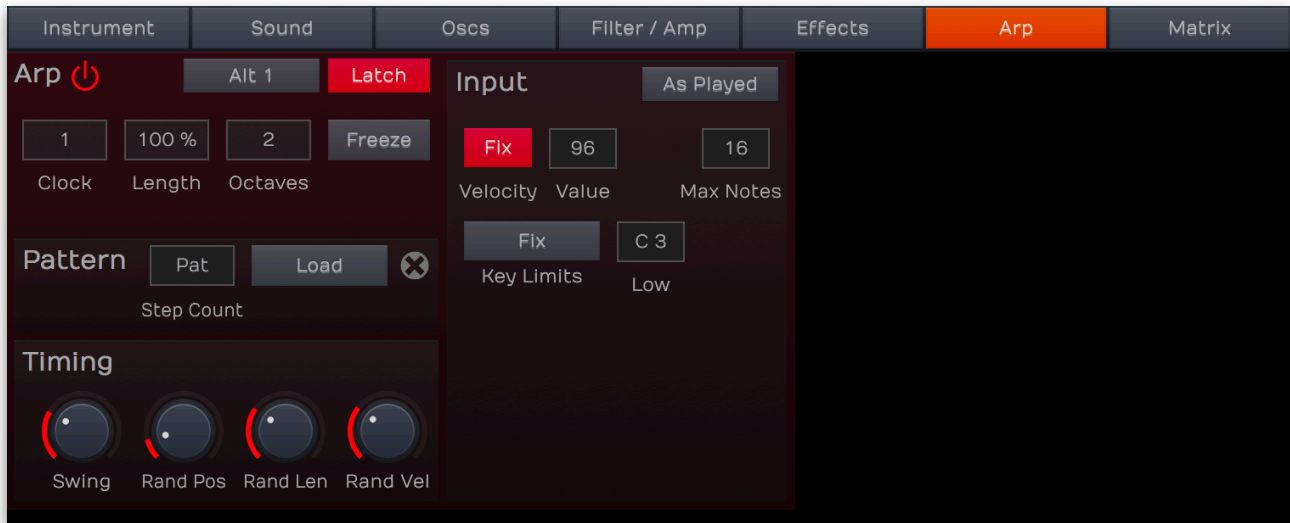


- **Mix:** Adjusts the relation between the dry signal at the full left to the wet effect signal at the full right.

- **Predelay:** Sets the delay before the signal enters the reverb stage. Typically you set it to around a hundredth to a tenth of the reverb **Decay**. **Predelay** strongly affects the perceived size of the reverb effect since it controls the time until the first reflection is generated.
- **Size:** Sets the length of the room size. Lower settings simulate a small room while higher settings simulate a big hall or church.
- **Decay:** Sets the time the reverb needs to decay.
- **Slope:** Controls the slope of the reverb. Setting it to center results in a typical gated reverb where the reverb sound stays equally loud before it is cut off abruptly. Lower values let the reverb decay slightly while higher values result in an inverse reverb which gets louder over its time.
- **Density:** Controls the initial density of the reverb. Lower values result in audible early reflections while higher values produce a dense reverb start.
- **Diffusion:** Controls the density or diffusion of the reverb the longer it sounds. Lower values result in more audible individual reflections while higher values result in a dense reverb tail. Note that even a setting of 0 will still create a reverb becoming more and more diffuse.
- **Low Cut:** Removes low frequencies before the signal enters the reverb stage so that the reverb sound is filtered.
- **High Cut:** Removes high frequencies before the signal enters the reverb stage so that the reverb sound is filtered.
- **Low Damp:** Removes low frequencies in the feedback stage of the reverb meaning that the low frequencies are dampened over time.
- **High Damp:** Removes high frequencies in the feedback stage of the reverb meaning that the high frequencies are dampened over time.
- **Pan:** Determines the stereo panning of the reverb effect.

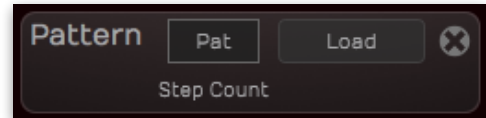
6.7 Arpeggiator (Arp) Page

Biotek 2 offers an extensive arpeggiator. To use it, click on the activation button right to the “Arp” characters.

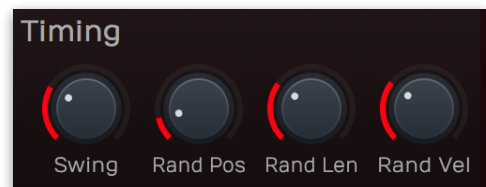


- **Direction** pop-up menu: Sets the direction that is used to play back the arpeggio. If “Up” is selected, the note list is played forward and the octaves are transposed upward. The arpeggio starts in the original octave and goes up to the highest octave (determined with **Octaves**). Then the arpeggio is repeated. If “Down” is selected, the note list is played backward and the octaves are transposed downward. The arpeggio starts in the highest octave (determined with **Octaves**) and goes down to the original octave. Then the arpeggio is repeated. If “Alt 1” is selected, the note list is first played forward and the octaves are transposed upward. After reaching the last note of the note list in the highest octave to play, the note list is played backward and the octaves are transposed downward down to the first note of the note list in the original octave. Then the arpeggio is repeated. “Alt 2” reacts nearly in the same way as Alt 1 with a slight variation. If “Random” is selected, the note list will be played randomly. If “Chord 1 Oct” is selected, the Arpeggiator plays a chord with all notes from the note list. This means that you have to play at least two notes to hear the effect. The note list is not advanced. If “Chord All Oct” is selected, the Arpeggiator plays a chord with all notes from the note list. If **Octaves** is set to 2 or higher, all notes of the selected octaves are played simultaneously.
- **Latch**: Enables the arpeggiator to remember the last set of keys played on the keyboard. When enabled, the arpeggiator sequence will continue to play even after the keys have been released. Once all keys are released, the next chord to be played will clear the latch memory and only the newly played chord will be arpeggiated.
- **Clock**: Sets the note value for the steps of the rhythm pattern in a range from whole notes to thirty-second triplet notes. Triplets (e.g. 1/8T) and dotted notes (e.g. 1/16.) are available for every note value. Clock offers a range of useful musical values.
- **Length**: Sets the length of the generated arpeggio notes in percent based on the note values made with the **Clock** parameter.

- **Octaves:** Determines the range of the single notes in octaves. When it is set to 1, the note list will be played back in the same octave as originally entered. Higher values mean that the note list is repeated in higher or lower octaves. The octave in which the arpeggio starts is determined by the **Direction** parameter.
- **Freeze:** If activated, the arpeggiator splits up all played notes and generates a continuous arpeggio even when the chord is released.
- **Pattern:** Allows you to import your own pattern sequences as a standard MIDI file (.mid). Click on the **Load** button to load a corresponding MIDI file. All tracks of this MIDI file are summed into one sequence before it is read into the arpeggiator pattern. All arpeggiator settings like Direction, Clock Division, Tempo etc. are performed in Biotek 2. The MIDI file import only sets the steps that make up the arpeggiator pattern. The pattern starts at 0 (i.e. song start). The “neutral” time base is 16th but the sequence can contain arbitrary notes like 32th, 64th, quarter or half notes or anything else. The pattern notes are read in with the position they have in the MIDI file. So, if you would place the first note at bar 4 of the song, it would need these 4 bars upon note press until the first arpeggio note is generated. If you want to have a simple, neutral arpeggio pattern, set all notes to C3 in the MIDI file. The result is that the arpeggio generates its output notes based entirely on the played input notes, e.g. when playing a C major chord the arpeggiator generates C, E, and G notes in sequence

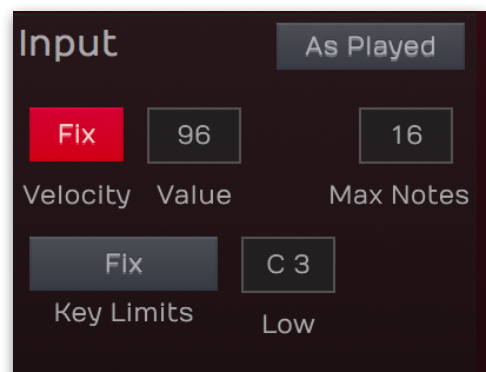


- **Timing Swing:** Determines how much the timing affects an arpeggio step. If set to 0%, the arpeggio is played back without any shuffled timing. Higher settings increase the shuffling of the notes and creates a typical swing.



- **Timing Random Position:** Creates random timing positions for the played notes. The higher the value, the more the timing differs from the original timing.
- **Timing Random Length:** Creates random note lengths. The higher the value, the more the note lengths differ from the original note length.
- **Timing Random Velocity:** Creates random velocities. The higher the value, the more the velocities differ from the original velocities.

- **Input Order:** Sets the order in which the note list is arranged. With this parameter, you can determine how the notes you originally play are split up for the arpeggio. If “As played” is selected, the notes are sorted in the exact order you played them. If you e.g. press E1, G1 and C1, the note list looks exactly like that. If “Ascending” is selected, the notes are sorted



from the lowest note to the highest note. If you e.g. press E1, G1 and C1, the note list is sorted to C1, E1 and G1. “Descending” is the opposite of Ascending. The example would be sorted as G1, E1 and C1.

- **Input Velocity:** Click on the **Fix** button to activate it (it lights red). From now on, all incoming velocities will be ignored and the arpeggio notes are played with a static velocity which is determined with **Value**.
- **Input Maximum Notes:** Sets the maximum notes that are played within an arpeggio. We recommend to set this value to 3 or higher if you want to play regular chords.
- **Key Limits:** **Off** means, that the arpeggiator only plays notes that are played by keyboard. **Fix** means that the arpeggiator only uses the note that is defined for “Low”. **Clip** means, that the arpeggiator only uses notes that lay between “Low” and “High”. **Fold Octave** means that notes below “Low” and notes higher than “High” will be transposed by one octave. **Fold Range** transposes all notes by the whole range.

6.8 Matrix Page

A modulation can be described as a generating unit's influence upon a sound parameter. The terms used in this context are "Source" and "Destination". Biotek 2 offers 200 independent modulation assignments (slots) each with individual settings of two sources, destination and amount.



To set up a modulation, click on the first Source pop-up menu and select a desired source, e.g. "LFO 1". If desired, select another source in the Source 2 pop-up menu that influences Source 1, e.g. "Mod Wheel". This is not necessary for a regular modulation. Now select a destination, e.g. "Pitch". Finally, set amount to a desired value, for example to 0.28. Now you have set up a vibrato that can be controlled by Mod Wheel.

On the first page of the Matrix page you will find the first 15 modulation slots. Click on the "016" button to select the next 15 ones and so on.

You can also use the comprehensive modulation assignment of Biotek 2 to easily add modulations. Read more about this in the chapter "Modulation Assignment".

6.9 Modulation Sources

In the next sub chapters all modulation sources as for example LFOs or envelopes are described in detail.

The LFOs

Biotek 2 offers 8 LFOs. These low-frequency oscillator (LFO) are used to modulate parameters periodically. The LFO usually offers the same or similar waveforms as an audio oscillator, but it doesn't produce any sound by itself, it usually oscillates in subsonic frequencies around 1 Hz, sometimes slower, sometimes faster. It is typically used to modulate **Pitch** to create a vibrato, **Volume** to create a tremolo or **Filter Cutoff** to create an auto-wah effect. Since Biotek 2 offers free modulation assignments, you can also modulate other aspects of the sound with the LFO like e.g. one of the **FM Depths**, or the **Band Frequencies** of the Equalizer.

The Biotek 2 LFOs contain an unique functionality: Every LFO provides up to 8 slots with independent parameter settings. This means, that you can set up a complex LFO arrangement based on 8 completely different LFOs which can be switched through by another modulation source, e.g. the Mod Wheel.

Click on the LFO 1-2, LFO 3-4, LFO 5-6 or LFO 7-8 buttons to call up the corresponding LFOs.



- **Assign:** Click on the “Assign” button to call up the modulation assignment for the corresponding LFO. Read more about this on page 52.
- **Slots:** Here you can determine the number of LFOs (from 1 to 8) that act within the corresponding LFO.
- **Slot Selector:** The white button within the LFO graphics shows, which of the LFO is active. Move the button horizontally to select the desired LFO. The setting of this button is important when assigning a modulation to it.
- **Mono:** If deactivated, each voice of a sound has its own LFO starting at the selected **Startphase** and running with its rate. If activated, the sound appears to only have one LFO for all voices. The first voice of any overlapping notes determines the startphase and each following voice synchronizes to it.
- **Clock:** Activates or deactivates the synchronization of the **LFO Rate** to the internal clock. Please note that this parameter only synchronizes the rate of the LFO, to also synchronize it to the note start, set **Startphase** accordingly.
- **Delay:** Controls the time before the LFO waveform is faded in or out. Depending on the setting of **Fade**, it either causes the LFO to start with maximum amplitude for the time set by **Delay** before it is faded out, or to start with zero amplitude before it is faded in.
- **Fade:** Controls the fade out or in of the LFO waveform. When set to center, the LFO stays at its maximum amplitude forever. When the parameter is set below center, the LFO waveform is

faded out after the time set by **Delay**, when it is set above center, it is faded in after the time set by **Delay**.

- **Wild Page Sound:** On the Wild page, only LFO 1 and 2 can be displayed (Flow LFO 1 and 2). Here you can determine, which Sound layer LFO is shown on the Wild page.
- **Waveform:** Selects the waveform the LFO generates: Sine, Triangle, Pulse, Pulse +, Saw down, 4 and 8 Steps Up, Sample & Hold, Noise, Saw Up, 4 and 8 Steps up. The actual waveform is displayed graphically.
- **Rate:** Controls the LFO rate or speed in Hz when **Clock** is off. Changes are continuous and **Rate** can be modulated. If **Clock** is activated, **Rate** can be set in musical values, e.g. 1/16.
- **Startphase:** Controls the phase with which the LFO begins its cycle when a note is played. The start position is shown in the LFO waveform display. A setting of "Free" means, that the start phase is random.
- **Symmetry:** Controls the asymmetry of the selected LFO waveform. Change it to the lower or higher settings than 50% to continuously introduce an asymmetry to the LFO waveform.
- **Output Amount:** A setting of 100% means that the LFO fully affects the modulated signal. Lower settings reduce the strength of the LFO modulation.

The Envelopes

Biotek 2 offers 4 multistage envelopes with up to 32 nodes. Envelopes are essential parts of any synthesizer. They are used to change the volume of a sound over time, which is essential for any sound from any acoustic or electronic instrument. They can also be used to control other aspects of the timbre over time, i.e. control **Filter Cutoff**, **Oscillator Pitch** or anything that can be modulated. Biotek 2 offers up to four envelopes per Sound layer. **Envelope 4** is always assigned to **Volume** but you can freely assign it to other modules as well. Each of the four envelopes offer the same parameters (except **Env 4**, which contains no Release Loop).

Click on the **Env 1-2** or the **Env 3-4** buttons to call up the corresponding envelopes.



Basically, an envelope offers three time stages (nodes 1, 2 and 3) which controls Attack, Decay and Release. Sustain is a time parameter and is defined by the length of the played note that triggers the envelope.

- To add more stages, double click on the line between two numbers. A new stage number will be added automatically. To delete a stage, double click directly onto the corresponding number. Add more stages, if you need a more precise control for modulations. Keep in mind that stages can overlap, so that you cannot see them. Click and move a stage breakpoint for a better overview.
- **Breakpoint** handle: Controls the time of the respective envelope stage. These parameters can be adjusted by clicking on the corresponding breakpoint handle and moving it in the envelope display horizontally or vertically. Please note that a time of 0 s really means 0 s, which can result in a click.
- **Slope** handle: Controls the slope of the respective envelope stage. These parameters are adjusted by clicking on the rounded rectangular handles in the envelope display and move it vertically. The slope of the stage can be changed gradually between fully exponential when the handle is moved fully down, to linear when the handle is in the center between the previous and next stage up to an inverse exponential shape when the handle is moved fully up. Note that the handle isn't displayed for very short envelope stages. If you want to change the slope of a very short envelope, just increase the respective time temporarily, adjust the slope and decrease it back to the previous value.
- **Assign**: Click on the "Assign" button to call up the modulation assignment for the corresponding envelope. Read more about this on page 45.
- **Envelope Triggering**: Normally, an envelope is triggered by an incoming MIDI note. If you play more than one note, every voice contains its own envelope. This is called "Poly". You can also set an envelope to a monophonic behavior, so that the first played note ("First") or the last played note ("Last") triggers on envelope which acts for all played voices. It means that the envelopes of all voices of a Sound layer behave like a single envelope. This common envelope starts as soon as the first/last note is played. The sustain level remains until the last note is released. Afterwards the release phase is active.
- **Sustain Loop Start**: The Attack phase of an envelope can be looped by desire. Here you can determine the number of loops for this phase.
- **S(ustain) Loop / Sustain**: Click on the **S Loop** button to activate the a sustain loop (it lights orange). With the two fields above the **S Loop** button and **Sustain** you can define the sustain loop area. This area will be displayed within the envelope graphic. We recommend to make useful loop settings which means lower settings for the sustain loop start as for the sustain loop end.
- **R(elease) Loop / Release**: Click on the **R Loop** button to activate the a release loop (it lights orange). With the two fields above the **R Loop** button and **Release** you can define the release loop area. This area will be displayed within the envelope graphic. We recommend to make useful loop settings which means lower settings for the release loop start as for the

release loop end. keep in mind that **ENV 4** does not offer a release loop.

- **Declick** (only for Envelope 4): This button appears, when the Attack rate is smaller than 0.5 milliseconds. Click this button to set it to 0.5 milliseconds automatically to prevent sound clicks, when using sine oscillators or other hollow oscillator or filter settings. Remember to check every sound layer when hunting for clicking.

Other Mod Sources

Beside LFOs and Envelopes, Biotek 2 offers a lot of other modulation sources. Click on the **Source 1** or **Source 2** button to call up the corresponding sources.



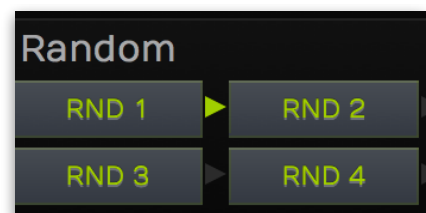
- **Keyboard** sources:
 - Each **Key** on a MIDI keyboard has a certain note number. Low notes have low numbers while high notes have high numbers. Biotek 2 converts this note number into a bipolar modulation source with MIDI note C3 (MIDI note #60) as center, i.e. where no modulation occurs. Lower notes result in a negative modulation while higher notes result in a positive one. Keytrack is applied to Oscillator Pitch on melodic sounds by default to allow sounds to be played melodically (but you can certainly change this setting). Use Keytrack on other parameters as well, for example Filter Cutoff to make higher notes sound brighter or envelope Decay or Release to shorten the sound slightly on higher notes. Another interesting modulation assignment is Redux Rate, especially with samples, to create low-fi sounds reminding of early samplers of the 1980's. All parameters that control more or less some kind of pitch, like Cutoff, Transient Generator Speed, Redux Rate and certainly Oscillator Pitch can be played perfectly chromatically when the modulation depth is at maximum.
 - **Velocity**: On almost all MIDI keyboards, hitting keys soft or hard results in MIDI notes with different velocities between 1 and 127 (0 is never sent on note on messages as it means „interpret this note as note off“ in the MIDI world). This is the same behavior as on a piano where harder played keys sound louder and brighter than softer played keys. You can use Velocity to change any parameter directly, but typically, as found on most hardware synthesizers, you might use it to control the depth of the LFO or one of the envelopes modulating a

certain parameter. For example, modulating Filter Cutoff is usually not performed directly by Velocity, but an envelope (say Envelope 1) is applied to it controlled by Velocity, maybe accompanied by some additional modulation without Velocity. Just think again of a piano: the harder you strike a key, the more brilliant is the attack phase, but the final decay of the piano is almost as dull as when you stroke the key softly. There is one exception to the above: When you apply Velocity to Amplifier Volume, you don't need to activate the Envelope 4 Assign button, this is done internally anyway. Otherwise, the voice would sound infinitely since the Velocity modulation would have been added to the output volume. You can also use Velocity to modulate parameter values of other modulation sources, e.g. modulate Envelope 4 Attack with a negative value to create a sharper attack phase the harder you play a key or reduce the LFO Fade time to let a vibrato come in faster the harder you play.

- **Release Velocity (Rel Vel):** After releasing a key on your MIDI keyboard, the release velocity can also be used as modulation source.
- **Channel Pressure (Ch Press):** Also known as Aftertouch be also be used as a modulation signal. The majority of contemporary keyboards are capable of generating this messages. When you press harder on a key you are already holding down, a MIDI Channel Pressure message is generated. This feature makes sounds even more expressive (e.g. through vibrato). Channel Pressure is unipolar.
- **Poly Pressure (Poly Press):** The minority of contemporary keyboards are capable of generating this messages. However, most sequencer software is able to produce such messages. For every key you are pressing harder after already holding down, a single MIDI Poly Pressure message is generated. This feature makes sounds even more expressive (e.g. through vibrato). Poly Pressure is unipolar.
- **Controllers** sources:
 - The **Pitch Bend** wheel or some similar control found on most MIDI keyboards. Most of the time it has a spring attached to it to reset the wheel to its center position automatically when no force is applied to it. Pitch Bend is a bipolar modulation source. Use Pitch Bend typically to change the Oscillator Pitch slightly for bent notes or manually generated vibrato, or modulate Filter Cutoff to brighten or dampen a sound.
 - The **Modulation Wheel (Mod Wheel)** is another control found on most MIDI keyboards right beside the Pitch Bend wheel. Sometimes its integrated into the Pitch Bend wheel, e.g. as a pressure sensor or part of the joystick. It's a unipolar modulation source and, depending on the keyboard manufacturer, the control stays at the position where it has been set to or moves back to a zero position when no force is applied to it. Use the modulation wheel to blend in modulations, as the name implies, like a pitch vibrato, by pressing both **L1** and **MW** in the Assign

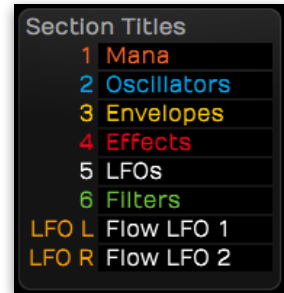
section, followed by adjusting the **Oscillator Pitch** modulation depth. You can also use it to control other aspects of a sound, e.g. fade in the **Ring Modulator**, increase **Drive** or change the **Effect Mix**.

- **Sustain**: This pedal message (MIDI CC #64) can be used as modulation sources. If a value greater or equal than 64 is received, the “Key Pressed” envelope phases of all voices of the respective MIDI channel are played. If a value smaller than 64 is received, the “Key Off” envelope phases are played. This means, that a Sustain pedal modulation can be used as switch, e.g. to activate/deactivate parameter functions.
- **Sostenuto**: This pedal message (MIDI CC #66) can be also used as modulation sources.
- **Breath Ctrl**: A breath controller can be used to add more expression to a sound, because it is controlled by human mouth and allows a very slight control.
- **Foot Ctrl**: Foot switches send value 0 (off) and 127 (on) on their controller numbers. There is no dedicated “MIDI Foot Switch Message” so the reception of the MIDI Messages they originally generated depends on the settings of the parameters of the current Sound.
- **Random sources (RND 1, 2, 3, 4)**: Upon playing a note, Biotek 2 generates 4 random values for this voice that you can use to create slight deviations resulting e.g. in a livelier sound. These values are held as long as the voice plays, so these Random Values are 4 simple sample&hold modules with random input. Old analog synthesizers, for example, always sounded a little out of tune or had slightly differently calibrated filter circuits so that the same note played in repetition always sounded a little different. With these modulation sources you can mimic this behavior. Just assign one of these Random Values to **Oscillator 1 Pitch**, another to **Oscillator 2 Pitch** and maybe a third one to **Filter Cutoff**, all with some small but different depth setting. You can also use stronger depth settings to create typical sample&hold effects.
- **Ring**: Sets the modulation assignment for the Biotek 2 Ring. Click on the corresponding **A** button (Assign) to set up the desired modulation. use the X and Y knob to control the modulation range.
- **Wild Page Section 1** sources: Here you find four nameable sections which let you define two modulation assignment for each section. The parameter and its name will be shown on the Wild page. Click on the corresponding **A** button (Assign) to set up the desired modulation. Each parameter can be set as an unipolar or a bipolar control (**Bi**). Keep in mind that you can rename the section itself as well as also the modulation parameter separately.
- **Wild Page Section 2** sources: Here you find 12 more modulation sources within three sections. The parameter and its name will be shown on the Wild page. Click on the



corresponding **A** button (Assign) to set up the desired modulation. Each parameter can be set as an unipolar or a bipolar control (**Bi**). Keep in mind that you can rename the section itself as well as also the modulation parameter separately.

- **Section Titles:** On the left, you find a list of all sections that are displayed on the Wild page. Here you can also change the name for the corresponding section as you desire.



The Modifiers

Modifiers allow you to apply mathematical functions on modulation signals. Depending on the function type selected, calculation will affect two source signals or a source signal and a constant parameter. You can use up to four independent modifier units. The result of each operation is not directly audible, but is used as input source for the modulation assignment. Moreover, you can use it as source for yet another modifying process.

LFO 1-2				LFO 3-4				LFO 5-6				LFO 7-8				Env 1-2				Env 3-4				Sources 1				Sources 2				Modifier 1				Modifier 2				Modifier 3							
L1	L2	L3	L4	L5	L6	L7	L8	E1	E2	E3	E4	Key	Vel	Rel	AT	PP	PB	MW	BC	FC	SP	Sos	R1	R2	R3	R4																					
Modifier 01-10																																															
01		Mod Wheel				x				LFO 4				15.9				Assign				▶		06		Off				x				Constant				0.00				Assign				▶	
02		Pitch Bend				x				Constant				18.9				Assign				▶		07		Off				x				Constant				0.00				Assign				▶	
03		Ch Press				x				LFO 2				9.79				Assign				▶		08		Off				x				Constant				0.00				Assign				▶	
04		Sustain				x				Constant				9.99				Assign				▶		09		Off				x				Constant				0.00				Assign				▶	
05		Random 1				x				Unison				10.7				Assign				▶		10		Off				x				Constant				0.00				Assign				▶	



See the **Operation** parameter described in the Appendix for further details.

Click on the **Modifier 1**, **2** or **3** buttons to call up the corresponding modifier section. Up to 30 different modifiers can be used.

- **Source** slots: Select the source signals that are used for the calculation.
- **Operation:** Determines which kind of operation will be performed on the selected input sources. The result of a modifier operation always lies within the range -max...0...+max. When it is assigned to a parameter in the Modulation Matrix, it is scaled to the range of the selected parameter.
- **Constant:** Defines a value for modifier functions that require a constant parameter.

6.10 Modulation Assignment

Biotek 2 features a powerful modulation architecture. Any of the available modulations sources like LFO, Envelopes, MIDI Note Number, Velocity, Pitch Bend Wheel and so on can modulate any of the continuous sound parameters like Oscillator Level, Filter Cutoff, Equalizer Frequency etc. To make assigning these modulation sources to destinations as easy as possible, each modulation source sports a so-called Assign button. The **Assign** buttons for the internal modulation

sources like the LFO and the Envelopes is located in the upper left corner of its section, while the further available sources are collected in a dedicated section called Assign. For ease of understanding, we call them all just Assign buttons.

You can think of this modulation assignment as connecting a patch cable from the output of the modulation source to the input of the destination parameter, with the added functionality of being able to control its modulation depth. The formula for this is like this:

$$\text{destination} = \text{source} \times \text{depth}$$

However, just assigning one modulation source to a destination is often not enough. Imagine controlling a pitch vibrato by the modulation wheel. For this to work, you need two modulation sources, where one controls the depth of the other, in this case the LFO controlled in depth by the Modulation Wheel. Biotek 2 certainly offers this as well. Internally, this results in a multiplication between the two sources, so it doesn't matter which Assign button is pressed first. Mathematically it looks like this:

$$\text{destination} = \text{source 1} \times \text{source 2} \times \text{depth}$$

You are not restricted to modulate a parameter by only one modulation source (or the aforementioned two multiplied sources). You can assign as many modulation sources as you like to any parameter. A typical example is modulating Filter Cutoff by an envelope plus letting it follow the played note, so that higher notes have a brighter sound than lower notes. Or, modulating the Oscillator Pitch of a melodic sound by both the played note (since you want to play a sound melodically) and adding some vibrato to it. The only limitation is that you cannot assign more than 200 modulations per sound but Biotek 2 reminds you politely if this unlikely situation is about to ever occur.

You can even modulate parameter values of other modulation sources, like changing the decay phase of the envelopes by Keytrack resulting in shorter decays when playing higher notes. Just let your imagination go crazy.

As soon as you press one of the Assign buttons, the view of the Biotek 2 sections changes slightly and the selected Assign button lights green. All continuous parameter controls are now bipolar, which means that you can modulate each parameter in either direction. You might use this to e.g. apply a positive modulation of Envelope 1 to Filter Cutoff but a negative one to Oscillator 1 Pitch. Non-continuous parameters like switches or selection buttons are disabled.



Please keep in mind that the assignment mode is active as long as one or two assignment buttons are pressed.

Here is a short example how to proceed when setting up an envelope modulation for cutoff frequency:

- 1) Go to the Filter / Amp section and set up Filter 1 as desired.
- 2) Click on the Env 1 - 2 button to call up the Envelope graphic and set up Envelope 1 as desired.
- 3) Click on the Assign button in the Envelope 1 section. The user interface will change slightly and is now ready for destination selection and amount setting.

- 4) Move the Filter 1 cutoff knob to set up the modulation amount. As an orientation, the original position of the cutoff knob is also shown. The amount range will be shown as green line.
- 5) Now Envelope 1 controls the cutoff frequency of Filter 1 with the corresponding amount.
- 6) If you want to add more control to this modulation, you can also set up Velocity in addition to the regular envelope modulation. Click on the Vel button above the Envelope section (it will light green). Now you can move the Filter 1 cutoff knob to add the amount.
- 7) If you now play with different velocities, the filter modulation controlled by envelope is also influenced by the actual played velocity.



Please note that some combinations of modulation sources and destination parameters are less audible than others or even inaudible. E.g., modulating the Oscillator Startphase parameter by an Envelope always results in the same behavior, since the Envelope always starts at 0 while Startphase, as its name already says, only affects the start phase of the Oscillator and doesn't do anything afterwards

7. Appendix

7.1 Modifiers and their Operations

Modifiers are used to generate a modulation signal by using one or two input modulation signals processed by a certain algorithm!

Most modifiers use two input modulation signals, some use only one due to their nature. For those modifiers that use two input modulation signals, you might choose “Constant” as the second input signal and set a value in the adjacent box. This can also be helpful to understand how some modifiers work, so you are invited to experiment.

Some modifiers use conditional comparisons to decide what to return. Those conditions are noted as follows:

$y = a > 0 ? a : b$

This means: If the comparison before the question mark evaluates to true, the expression behind the question mark is assigned to y, else if it evaluates to false, the expression behind the colon is assigned to y.

There is only one basic arithmetic modifier, the multiplication. Other basic arithmetics like addition, subtraction and division are omitted. This is because adding or subtracting a second modulation to a parameter means to just assign it to it with a positive or negative depth. Division, however, would rarely result in any usable modulation output.

x

Returns **a** multiplied by **b**. This modifier is equal to what happens in the modulation matrix when you activate two assign sources at once, but it can be used to multiply up to four sources for one modulation.

Performs the following operation: $y = a \times b$

Min

Returns the smaller value of the values **a** and **b**. Performs the following operation: $y = a < b ? a : b$

Max

Returns the larger value of the values **a** and **b**. Performs the following operation: $y = a > b ? a : b$

Average

Returns the average between **a** and **b**. Performs the following operation: $y = (a+b) : 2$

Abs

Returns the absolute value of **a**. Performs the following operation: $y = |a|$

Abs Offset

Returns **a** unaffected when it is bigger or equal than **b**. Mirrors **a** at the axis of **b** if **a** is smaller than **b**. Performs the following operation: $y = b + |b-a|$

Unipolarize

Converts a modulation that happens between -1 and +1 to between 0 and 1. Note that the result is not always the same compared to e.g. an LFO with unipolar waveform. If a bipolar LFO is set up to fade out to 0, this 0 becomes 0.5 with the Unipolarize modifier, but it would be 0 if an unipolar LFO would be used directly. Performs the following operation:
 $y = a \cdot 0.5 + 0.5$

Switch

Returns 0 if **a** is smaller than **b**, else returns 1. Performs the following operation: $y = a \cdot (1 - b) + b$

LinExp

Approximation returning typical volume levels with roughly 3.5 dB change per 0.1. Very low values deviate to 0. Examples:

a	y
0.0	0.0
0.1	0.0055
0.2	0.0152
0.5	0.1
0.8	0.4208
0.9	0.6522
1.0	1.0

LinLog

Approximation of the inverse LinExp function.

Quadratic

Returns the quadratic value of **a**, taking care of the sign. Results in an s-shape. Performs the following operation:
 $y = a \cdot |a|$

Cubic

Returns the cubic value of **a**, resulting in an s-shape. Performs the following operation:
 $y = a \cdot a \cdot a$

Step

Transforms **a** into a steppy signal with the number of steps set between 1 and 100 with operand **b**. **b** = 0 returns **a** unaffected.

Performs the following operation:

```

b = floor(|b| * 100);
clip b to between 1 and 100
y = round(a * b) / b;
```

Reverse

Subtracts **a** from 1 or from -1 if **a** is smaller than zero. Values close to 0 become close to 1 and vice versa. Performs the following operation:
 $y = a < 0 ? (-1 - a) : (1 - a)$

Threshold

Only returns **a** if **a** is greater or equal than **b**, else returns 0. Similar to Switch, but returns **a** rather than 1. Performs the following operation:
 $y = a < b ? 0 : a$

Saturate

Drives **a * b** up very high and clips the result to the range of -1 .. +1. To reduce clipping, just lower **a** or **b**. Performs the following operation:
 $y = \text{clip} (a \cdot b \cdot 100)$

Filter

Filter applies a first-order low pass filter to the value changes of **a** with **b** as cutoff. Performs the following operation:
 $y = y1 \cdot (1 - b) + a \cdot b; y1 = y$

Lag

Returns a linear ramp reaching value **a** with a rate of **b**. The ramp goes up- or downwards depending on the value of **a**. If **y** reaches the value of **a**, it stays there. Performs the following operation:
 $y = a > y1 ? y1 + |b| : a < y1 ? y1 - |b| : a; y1 = 1$

Ramp

Returns a linear ramp from 0 to maximum with a rate of **b** if **a** crosses 0 positively. If the ramp is at its maximum value, it stays there. If **a** crosses 0 negatively, the ramp is reset back to 0. Performs the following operation:

```
y = a > 0 ? y1 + b : 0;
y = y > 1 ? 1 : y;
y1 = y;
```

S & H (Sample & Hold)

Performs the following operation:

```
b = b * b * n;
if (t > b)
{
    t = 0;
    y1 = a; }
y = y1;
t++;
```

b is multiplied by itself to allow finer adjustments in the lower value range. **n** is the maximum hold time of 1 second. **t** is an incremental timer. Upon note start, **t** is set to **n + 1** to ensure that **y** contains a valid value.

S & H Clocked (Clocked Sample & Hold)

Performs the following operation:

```
b = clockdiv(b);
```

(The rest as Sample & Hold)

Delay

Performs the following operation:

```
b = b * b * n;
```

```
x[0] = a;
```

```
y = x [-b];
```

b is multiplied by itself to allow finer adjustments in the lower value range. **n** is the maximum buffer size of 1 second.

Delay Clocked

Performs the following operation:

```
b = clockdiv(b);
```

```
x [0] = a;
```

```
y = x [-b];
```

b is transformed to a clock division.

8. Product Support

If you have any questions, please contact us by email:

support@tracktion.com

Tracktion Software Corporation

23515 NE Novelty Hill Rd

Suite B211-141

Redmond, WA 98053

USA

www.tracktion.com

© 2018 Tracktion Software Corporation. All rights reserved.