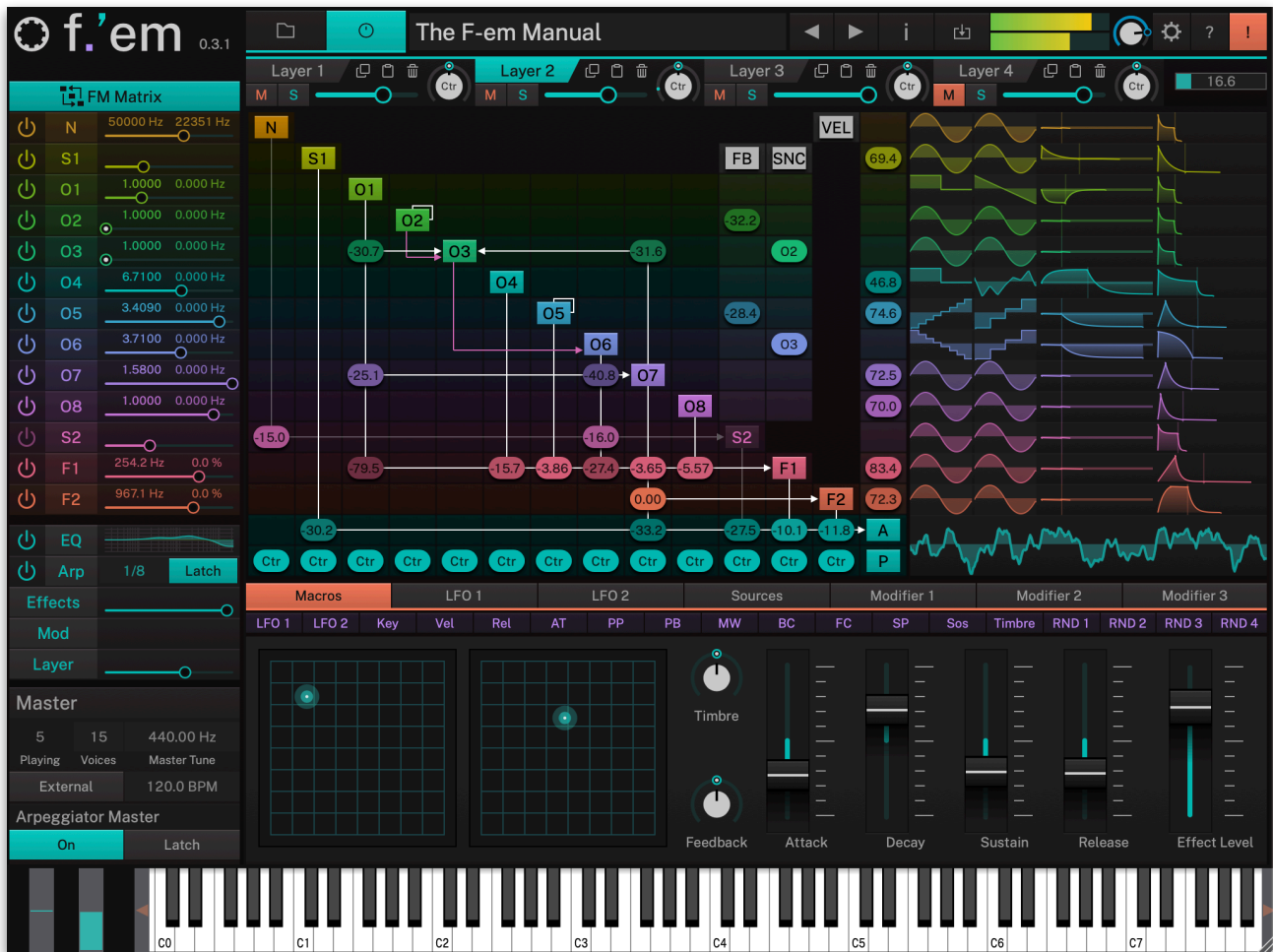# F-em

Frequency Modulation Synthesizer

USER GUIDE

Credits

Programming: Wolfram Franke

Additional Programming: David Rowland

User Interface Design: Sasha Radojevic

Sound Design and Sample Content: Alessandro Cardinale, Ed Ten Eyck, Jay de Miceli, Klaus-Peter Rausch (KPR), Stranjah, Vasil Ivanov (The Editor), Yuli Yolo

Manual: Wolfram Franke, Holger Steinbrink

Version: 0.3.1

Table of Contents

1. Introduction

Thank you for purchasing our F-em Frequency Modulation Synthesizer.

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

F-em is a premium quality instrument that combines an advanced FM synthesis engine with a stunning array of programming options.

The Main Edit section lets you dive much deeper into the FM synthesis engine and offer lots of parameters to control nearly everything.

The deep sound creation possibilities include four sound layers with eight operators each, that offer typical analog waveforms and separate pitch and level LFOs as well as pitch and level envelopes. There are two additional sample oscillators and a white and pink noise generator. Followed by two serial multimode filters and a four-band equalizer with shelf and bell curves.

Each layer can be 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. F-em offers 2 Flow LFOs, modulation modifiers and a wide range of sources.
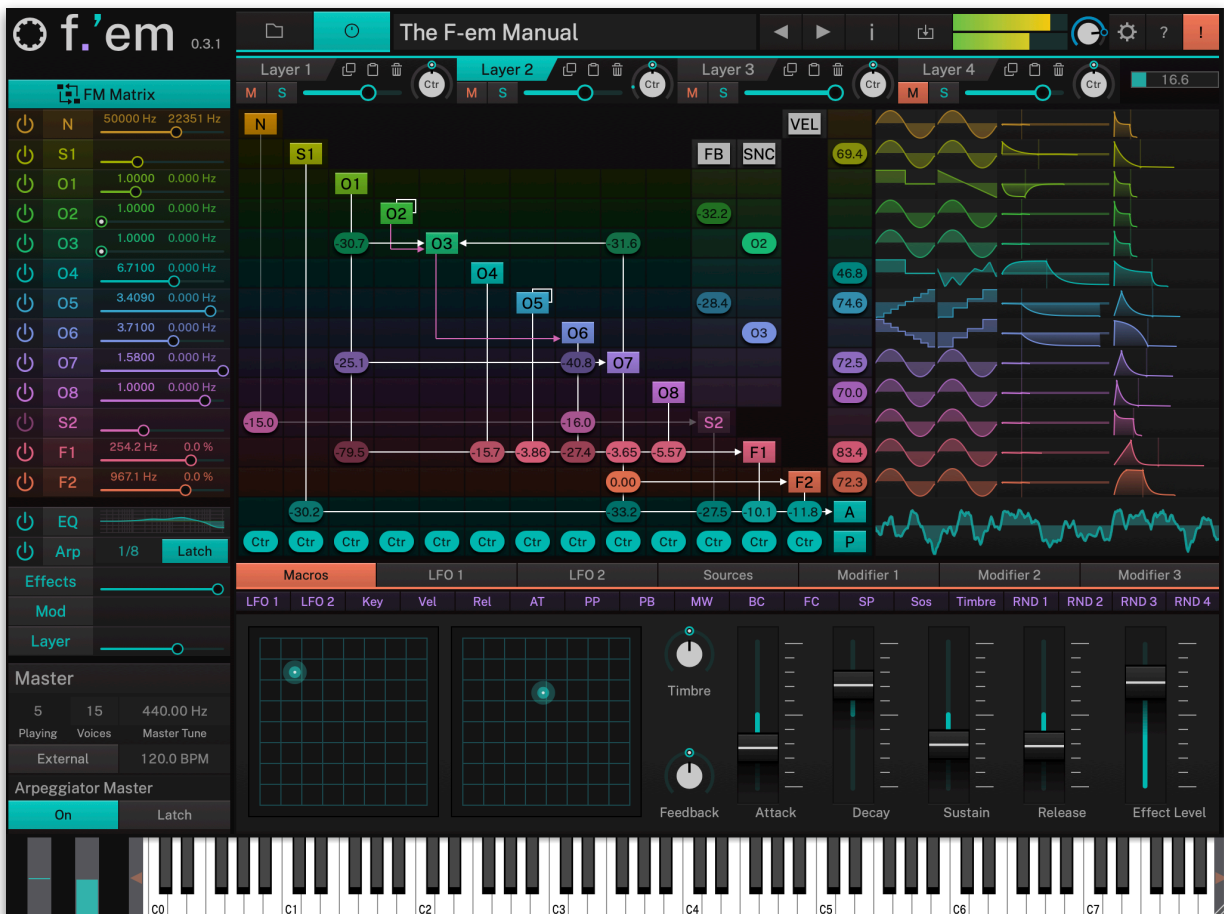
Some Basics

What is an Instrument?

An instrument is a sound program of up to four layers including all sound data, effect settings and global instrument parameters.

What is a Layer?

An instrument can contain up to four sound layers that can be edited separately. Basic options and parameters (e.g. Arpeggiator, Effects) are used by all layers. Each layer offers a full range FM engine.

What is FM?

Frequency Modulation (FM) is a synthesis method that uses the frequency of a waveform to modulate the frequency of another. In FM synthesis, these waveform generators mostly called 'operators' that are comparable to oscillators.

Operators are arranged in different relationships to one another; these arrangements are known as algorithms. Depending on the algorithm, one or more operators is usually routed directly to the audio output. Some FM synthesizers offer predefined algorithms, F-em uses a completely free programmable FM matrix.

The primary operator is known as the carrier, and the other operators are known as modulators. When a modulator does not have a direct link to the audio output it is being used to alter the sound of the carrier.

More than one modulator can be linked to a carrier, which allows the introduction of even more complex harmonics into the output of the carrier. The waveshapes can become very complex.

Also, a feedback loop can be set up. This a way for an operator to modulate itself, or to provide additional modulation for an entire stack of operators.

In F-em, each operator offers synchronisation, a Pitch LFO and a Level LFO as well as a Pitch envelope (with Level Scaling) and a Level envelope. Besides this, 2 sample oscillators and 2 filters are available for FM purposes.
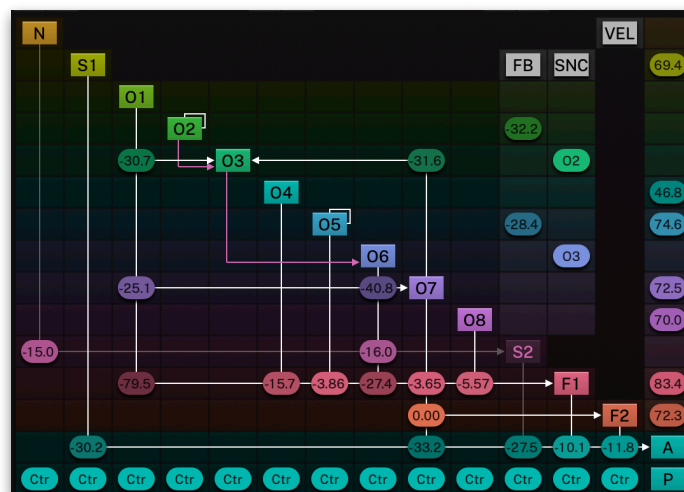
The FM Matrix

2. Installation & Authorisation

The installation including all sample and sound data requires approximately 2 GB space on your hard drive.

F-em 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! F-em runs only in 64 bit host applications.

1. Double click "f-em 1.0.0.exe" to launch the installer for F-em. In this example, 1.0.0 refers to the current version number of the file, which most likely differs from this example.

2. Follow the on-screen instructions.

Installation locations on Windows:

VST 64-bit: C:\Program Files\Steinberg\VstPlugins\

AAX 64-bit: C:\Program Files\Common Files\Avid\Audio\Plug-Ins\

2.2 macOS

✎ Please observe the system requirements of your host application! F-em runs only in 64 bit host applications.

1. Double click "F-em 1.0.0.pkg" file to launch the installer for F-em. In this example, 1.0.0 refers to the current version number of the file, which most likely differs from this example.

2. Follow the on-screen instructions.

Installation location Mac:

AU: /Library/Audio/Plug-Ins/Components/

VST: /Library/Audio/Plug-Ins/VST/

AAX: /Library/Application Support/Avid/Audio/Plug-Ins/

2.3 Linux Ubuntu

1. Copy the F-em 1.0.0. 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 F-em content

1. Create a F-em directory at "~/Tracktion/F-em/" (may already exist)

2. Copy the "Instruments" content to „~/Tracktion/F-em/Instruments"

3. Copy the "Samples" content to „~/Tracktion/F-em/Samples"

4. Restart your host and any open instances of F-em

You should end up with two folders:

"~/Tracktion/F-em/Instruments" containing „.f-em instrument" files

"~/Tracktion/F-em/Samples" containing ".acktionsamples" files

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

2.4 Authorisation

F-em uses a software based copy protection and can only run with an activated license.

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

Click on the Unlock button in the headline of the plug-in window to open the authorisation window.

Please enter your email address and password from Tracktion Marketplace and click on "Register" to unlock F-em. 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 F-em without limitations.

3. Basic Operations

3.1 Loading Instruments on the Browser Page

Click on the folder button in the head section 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" (macOS) or "Ctrl" (Windows) key is used in conjunction with selection – this will select individual tags.

Use the **Search** field in the right area to perform a text search of the Instrument library. With the left and right arrows you can switch through the patches.

To select sound packs or a desired author, click on the corresponding name in the left window section. Same is valid for **Category**, **Style** or **Character** search. You can also combine search attributes. To set back a search, click on the corresponding **X** in the upper right are of an attribute column. The actual search results are always displayed in the right sound result area.

You can tag your favourite sounds by clicking on the star symbol right besides the sound name in the right area. Favourite sounds will be shown with a yellow star symbol. Click on „Show only favourites" to show your „best only" sounds

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

Another way to search and load a sound is to click on the sound name field in the **Header** section. Here, you can also perform a name search.

3.2 Saving Instruments

To save an Instrument, click on the **Save** button in the **Header** section. User Instruments are added to the Instrument library. The file ending contains „.fminst". A newly saved preset is appended to the end of the file browse list. When launching F-em, 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.

3.3 The Macro Parameters in the Browser Page
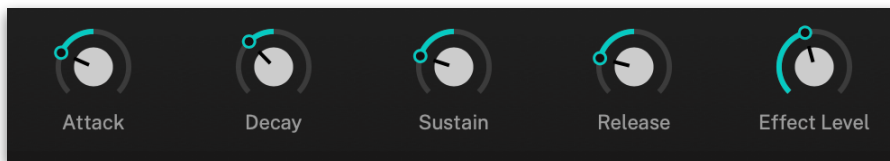
The Macro parameters can be found on two location: At the bottom of the Browser page you will find additional parameters regarding the overall Level envelopes and Effect levels. These macro parameters have influence on the Level envelopes and Effect levels of all Layers.

Why are these parameters located on the Loading page? Because If you load a sound and want to change some important settings directly, you do not have to navigate to the corresponding sound edit pages.

Keep in mind, that the Macro parameters produce an offset to the original settings. For example, if all Level envelopes are have an Attack of 0, a negative setting of the Macro Attack dial won't produce any result.

Another Macro parameter location can be found in the Macros tab. Read more on page 48.

3.4 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.
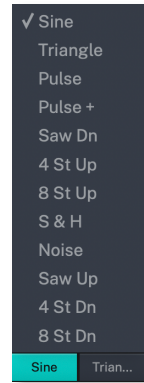
Selection Field Button

A simple click on a selection field calls up the corresponding display page. Selection field buttons light up, when clicked.

Pop-up Menus

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

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.

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.
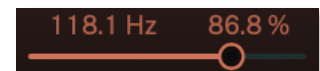
Graphical Elements

Click on the corresponding graphic and move the mouse vertical or horizontal to change settings. This is valid for the envelopes as well as the equalizer curves.

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

F-em 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.

4. The User Interface

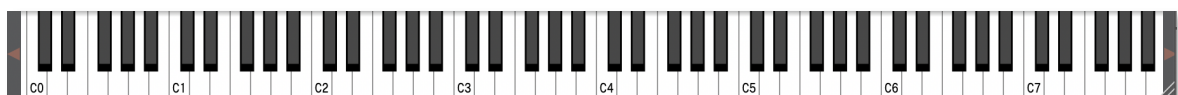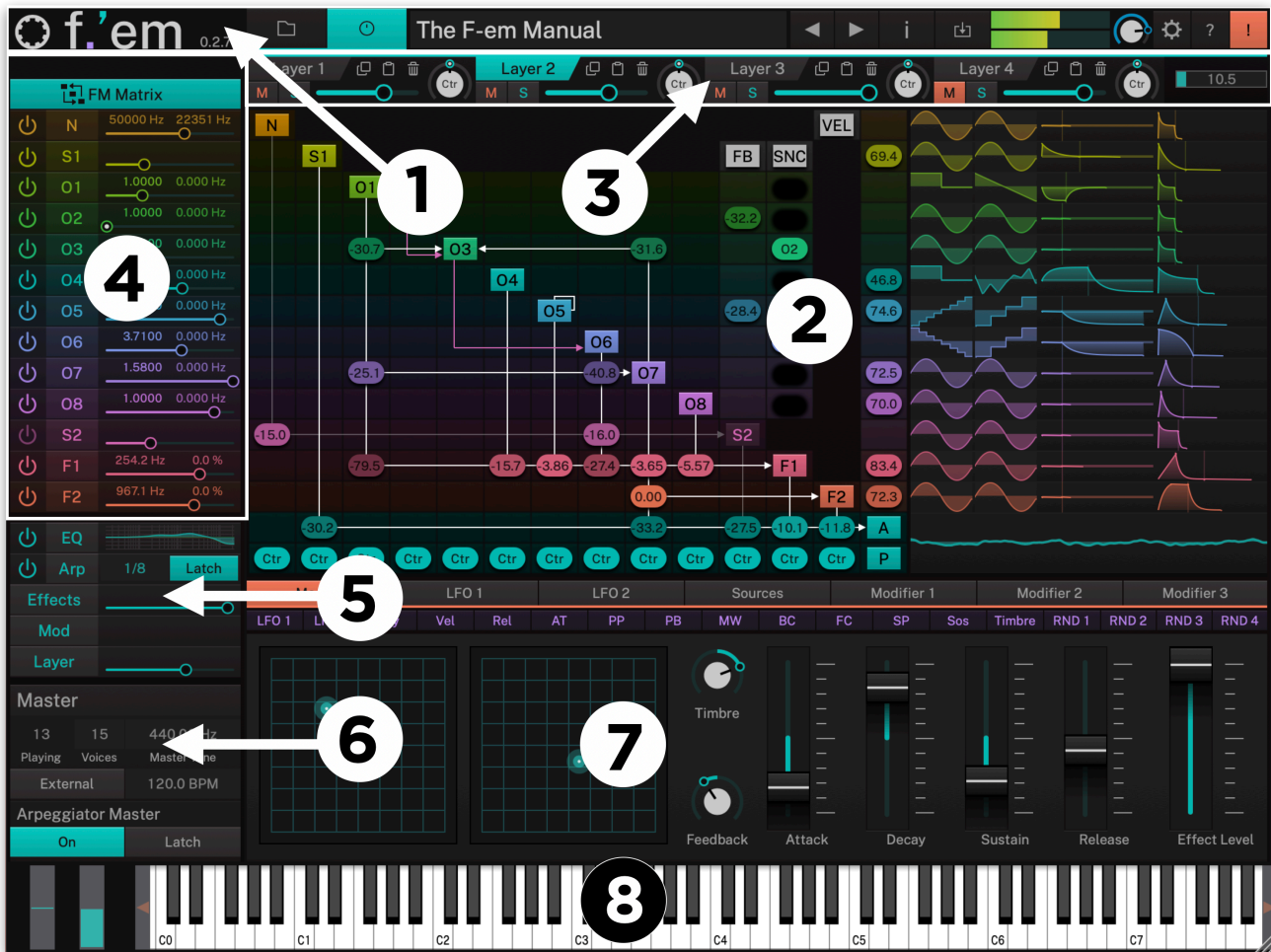F-em is divided into different sections which are described in detail within the next chapters.

The **Header** section **(1)** offers basic instrument functions.

The **Main Edit (2)** section includes lot of controls to edit a Sound Layer in detail.

The **Layer** section **(3)** lets you select and control the 4 Sound Layers.

On the left side, you find certain sub sections **(4)** for controlling FM Operators, Filters, the Arpeggiator, Effects etc.

In this section **(5)** you find: An **EQ** page includes a four-band equalizer, the **Arp** page offers a powerful arpeggiator, the **Effects** page offers four effects for each layer. The **Mod** page contains an extensive modulation matrix and the **Layer** page contains various options and parameters for each of the four layers.

The **Master** and **Arpeggiator Master** section **(6)** offers global parameters.

The **Modulation Sources** section **(7)** offers the Macro page, 2 Flow LFOs, lots of additional mod sources as well as modifier sources.

The **Virtual Keyboard (8)** provides a virtual full-range keyboard with 128 keys

11
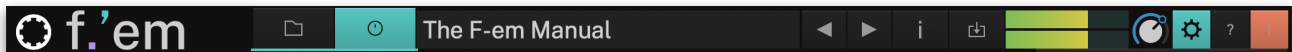
4.1 Header Section

The Header section offers global and basic settings.

- Click on the **f.´em** logo to open a pop-up window with additional information about the plug-in version number as well as the development team. Here, you can also check for updates.

- Click on the **Folder** button to open the **Browser** section. Read more in the chapter "3.1 Loading Instruments on the Browser Page" on page 8 of this manual.

- Click on the **Program Interface** button to open the **Main Edit** page. Read more in the chapter "5. The Main Edit Section" on page 16

- The **Instrument name field** displays the name of the actual Instrument. Click on the name to open the sound list.

- Click on an **Arrow** button to switch to the previous/next sound.

- The **Instrument information field** opens a pop-up window where you can enter several information regarding the actual sound. These information will be saved with the sound.

  - **Name**: Here you can enter a name for the actual instrument.

  - **Category**: Here you can enter an instrument category, e.g. Lead.

  - **Author**: Here you can enter the name of the sound designer.

  - **Style**: Here you can enter a sound or music style of the actual instrument.

  - **Character**: Here you can enter a sound character of the actual instrument.

  - **Description**: here you can enter a further description regarding the instrument.

- The **Save** button lets you save the actual Instrument. You can also change the name of the Instrument during the storing process.

- The **Master Volume** dial allows you to adjust overall volume of F-em. The meter left besides the Volume dial displays the outgoing audio signal.

- Click on the **Global** button (gear symbol) to open a pop-up window with additional parameters (see picture on the right). Click the **Global** button again to close the Global menu.

  - **Author**: Here you can enter the name of the sound designer.

  - **Zoom** - click on the corresponding zoom button to change the size of the plug-in user interface. The regular size is 100%. You can also drag the handle in the lower right GUI corner to change the size of the user interface to your needs.
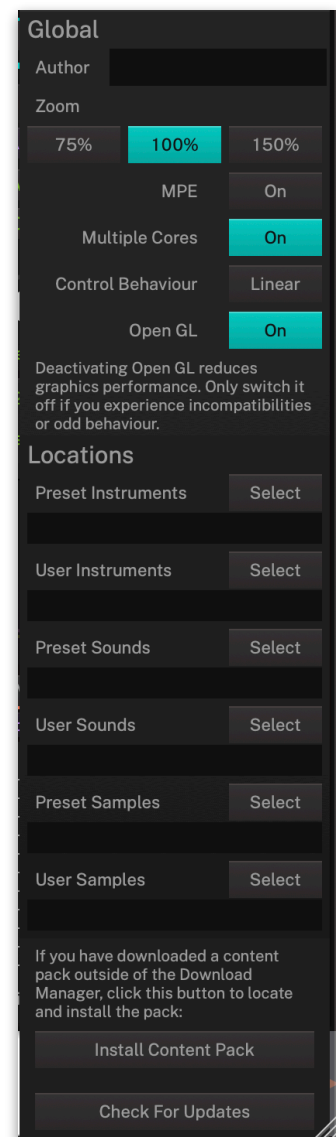
- **MPE** activates/deactivates the global MPE mode (MIDI Polyphonic Expression) for all Layers. Read more about the MPE functionality of F-em on page 46.

- **Multiple Cores** - when running multiple instances of F-em, turning off the multi core processing can improve CPU performance on certain systems – typically older computers. Which setting works better, depends on your computer hardware, your operating system and finally your host application. Therefore, we recommend to try out both settings in different work situations.

- **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.

- **Open GL** - Activates/deactivates the Open GL graphic engine. If deactivated, the graphic performance is reduced. We recommend to switch **Open GL** off only, if you experience incompatibilities or odd behaviour.

- **Locations** - Click on the corresponding button (e.g. **User Sounds**) to open the folder where the sounds or samples are placed. If you want to move the preset/sample content, you need to show F-em the new data path. This can be done within the Locations window.

- **Install Content Pack** - If you have downloaded a content pack outside of the Tracktion Download Manager, click this button to locate and install the pack.

- **Check For Updates** - Click on this button to open the Tracktion Marketplace. If an update for F-em is available, it will be shown here. Keep in mind that you need an active internet connection to check for updates.

- **Question Mark** button - click on this button to open this PDF manual.

- **Panic** button: Click on the exclamation mark button to send and execute 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

F-em 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. The orange arrows to the left and right side of the keyboard can be used to move the keyboard range, that goes from C-2 to G8.
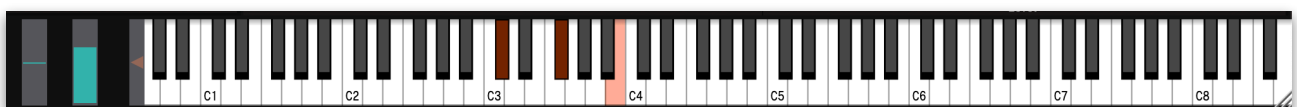
4.3 Master Section

Here, you can make global settings for playback.

- **Playing**: Shows the number of the current played voices. Keep in mind, that every of the 4 Layers plays its own voice, so the total number of played voices is 4 when all four Layers are active and you play a single note (and Unison is set to 1).

- **Voices**: Determines the maximum number of voices that F-em produces (up to 256 voices, default is 40). 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.
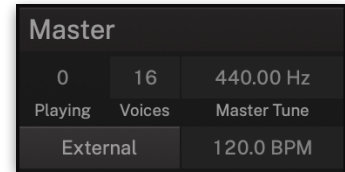
> Please keep in mind that the chosen number of unison voices will respectively add more CPU usage.

- **Master Tune**: 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 (Ab of A 440), baroque for some special church music (Chorton pitch) at 466 Hz (A# of A 440), and classical pitch at 430 Hz.

- **External**: When enabled, the arpeggiator(s) and all clock-based parameters (LFOs, Envelopes, Clocked Delays etc.) are synchronized to the tempo of your host application. If disabled, you can specify a desired tempo in BPM (see Tempo option below).

- **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 double click on **Tempo**, it will be set to default 120 BPM.

4.4 Arpeggiator Master Section

The arpeggiator is used to sequence rhythmic patches. Each of the 4 layers can have its own arpeggiator settings, but this section controls the global arpeggiator settings for all layer arpeggiators simultaneously.
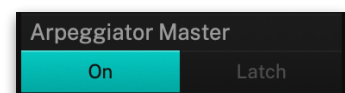
- **On**: Toggles the operation state for all arpeggiators to on or off.

- **Latch**: Enables the arpeggiators to remember the last set of keys played on the keyboard. When enabled, the arpeggiator sequences 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 arpeggiators with Latch enabled, press the "On" button.

4.5 Working with Layers

F-em offers 4 Layers. Each Layer offers a full range FM synthesis engine, an equalizer, an arpeggiator, an effect section and comprehensive modulation options as LFOs, sources and modifiers.

Finally, the audio signal of the 4 Layers is mixed together.
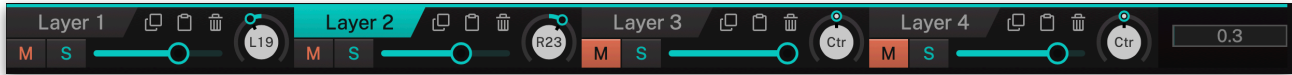
- **Layer** button: Click on the corresponding Layer button to select the desired Layer for editing. The parameter page switches immediately and shows the settings for the current Layer.

- **Mute/Solo**: Click on the **Mute** button (M) to mute the desired Layer. Click on the **Solo** button (S) to solo the desired Layer. This mutes all other Layers. Click the Solo/Mute button again to deactivate the solo/mute function.

- **Layer Volume** slider: Controls the output level of the corresponding Layer.

- **Layer Panner** dial: Controls the panning of the corresponding Layer.

- **Copy/Paste/Init** buttons: Click on the first button (with two overlay rectangles) to copy the settings of the corresponding Layer into the clipboard. Click on the second button (with the clipboard symbol) to paste the settings of the Layer in the clipboard to the corresponding Layer. Double click on the third paper bin symbol to to initialize the corresponding Layer to its basic settings.

- **CPU Usage Display**: The CPU usage display right beside the 4 Layers is used to monitor how much of your computer's CPU is being used by the instrument. If you play too much F-em voices, the global performance of your system and the audio may suffer.

5. The Main Edit Section

The Main Edit section includes dedicated subpages to edit each Sound layer in detail.

> ✎ Hint: Some sound 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 in the corresponding chapter on page 49.

5.1 Basic Sound Editing Operations

For deeper sound editing, it is useful to know, how F-em is structured and how to select subpages.

Here is a short overview, how to edit a sound:

• Create a default Layer: We recommend to start with a single default Layer. Select **Layer 1** and initialize it. Click on the **Solo** button, so that you hear only this layer.

• Build up an FM interconnection: First of all, you need to set up a FM basic structure. This must be done in the **FM Matrix**. Click on the FM Matrix selection field to open the FM Matrix page.

• Here, you find a basic structure of your sound. By default, **Operator 8** (O8) is routed to the amplifier (A). When you press a key on your MIDI master keyboard, you should hear a raw sine tone. In our case, O8 is called a carrier.

• To edit the sound parameters of Operator 8, simply click on the O8 selection field. Here, you can make edits regarding the frequency, the waveform and the corresponding modulation sources for Operator 8, as Pitch LFO, Level LFO, Pitch Envelope, Level Scaling and Level Envelope.

• Click on the selection fields of the other operators (O1 to O7) to open the corresponding operator edit pages.

• In the same way, you can open a Noise generator (N), two Sample Oscillators (S1 and S2) as well as two filters (F1 and F2). Keep in mind, that every module can be deactivated by clicking the On/Off button.

• You can also select a dedicated EQ for each layer, an arpeggiator (Arp), 4 Effects, a modulation matrix (Mod) and a page with basic Layer parameters.
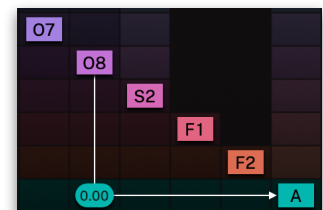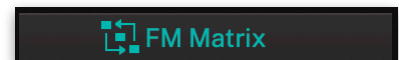
> ✎ Every operator/oscillator edit page, as well as the effects, the modulation matrix and the arpeggiator will be described in detail on the next pages.
> So let's start with the FM Matrix!

5.2 The FM Matrix

To open the FM Matrix page, click on the FM Matrix selection button.
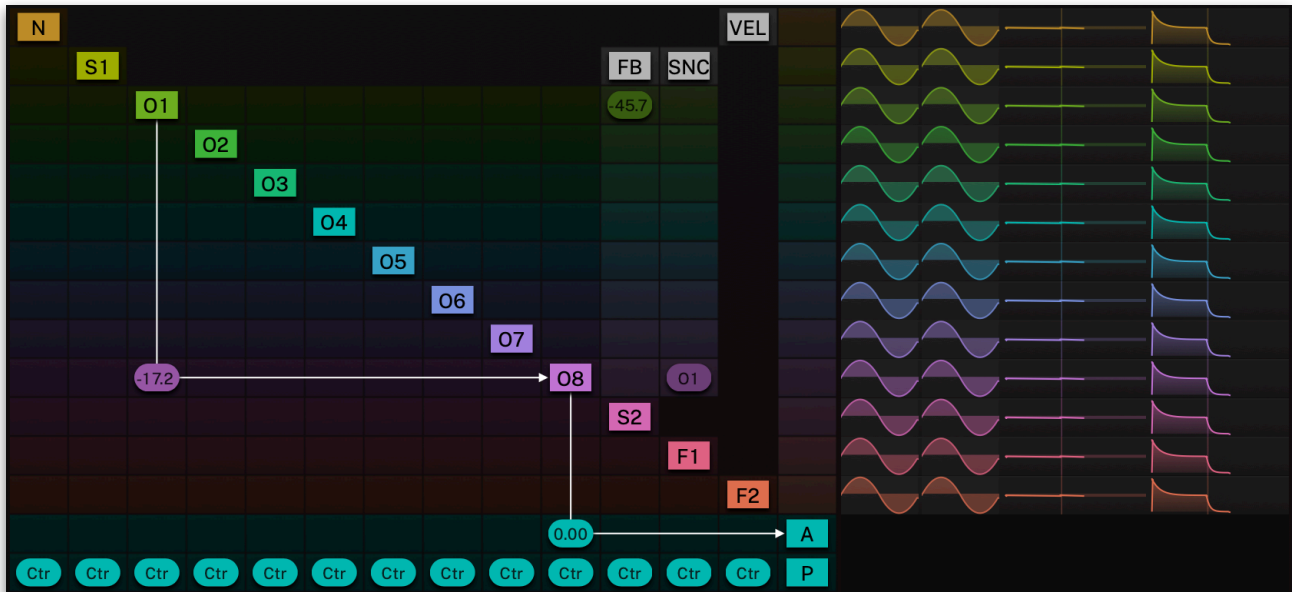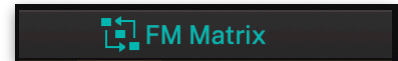
The FM Matrix page with a simple FM connection

Here, you setup you FM connections. As you might know, each operator can be work as a carrier (you hear the corresponding operator audio signal) or as a modulator (you might not hear the audio signal, but it works as modulation source for other operators (or the filters).

By default, Operator 8 (O8) is set up as a carrier (see picture above). This is represented by the thin white connection lines from O8 to A (which stands for Amplifier). By clicking and moving the mouse over the intersection, you can edit the volume. You can also use the slider in the left section on the O8 selection field. You can switch off operators by clicking pin the corresponding button (e.g. O1). The white connection line is greyed out. Click again on the button again, to activate the operator.

In our example (see picture above), we have set up O1 as a carrier for Operator 8. This can be done by clicking on the intersection of O1 and O8 in the matrix. Keep in mind that you need to click and move the mouse up or down to adjust the modulation amount. In our example, the amount is -17.2 which means, that operator 1 modulations operator 8. Here, Operator 1 can't be heard because there is no connection to the amplifier (A),

Changing the ratio and frequency of Operator 8 will change the modulation of Operator 1 and so also the sound result. This can be done by using the two value selections on the selection field for O1. Depending on the values, the sound can change dramatically.
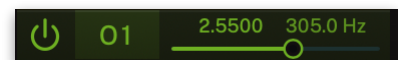
You can also introduce a feedback for a desired operator. Click on the corresponding feedback selection field in the FB column and move the mouse up or down. In our example, we introduced a small amount of feedback for Operator 1.

You can also sync two operators. In our example, we have set up sync for Operator 8 which will be synchronized to Operator O1. Click on the

corresponding Sync selection field in the SNC column and move the mouse up or down.

Each operator can be controlled by velocity. To do so, click on the corresponding velocity selection field in the VEL column and move the mouse up or down. In our example, we introduced maximum velocity amount for Operator 1 and 8.

In the right section of the FM matrix page you will find miniature graphics, that represents the current settings of the operators. The first two waveforms show the Pitch and Level LFO for each operator. The next graphic shows the Pitch envelope and the last one the Level envelope. To navigate directly to the corresponding edit page, click on the desired graphic.
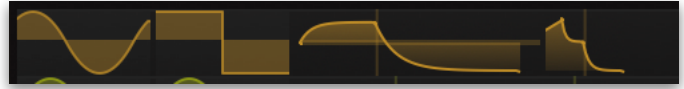
Each operator can be placed in the stereo panorama. This is done in the P(anning) row under the A(mplifier) row at the bottom of the FM Matrix page. Click on the corresponding selection field and move the mouse up or down.

Below the miniature graphics you find a simple oscilloscope which displays a graphical representation of the current audio signal.

5.3 Noise Generator

On this page, you find the noise generator controls. To open this page, click on the N selection field in the left section. Each Layer features its own noise generator.
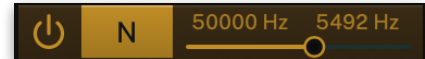
- **Noise Activation**: Click on this button, to activate/ deactivate the Noise generator.

- **Noise Selection**: Click on this pop-up menu to select the noise type. You can choose between White noise (which contains all frequencies with equal intensity) and Pink noise (where each octave interval carries an equal amount of noise energy).

- **Noise Filter**: Click on the pop-up menu to select the desired noise filter type. A description of the available filter types can be found in chapter „5.7 Filter 1 and 2".

- **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. The Cutoff value is shown in Hz ion the left section and also as graphical representation.

- **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.

- **Burst Rate** (the upper parameter in the left section of the Noise generator): Burst noise consists of sudden step-like transitions between two or more current levels. The lower the value, the more popping or crackling sound is produced.

Pitch LFO, Level LFO, Pitch Envelope, Level Scaling and Level Envelope are described in separate sections in chapter 5.6, because they are identical for all Operators/Sample Oscillators/ Noise generator.

5.4 Operator 1 to 8

On these pages, you find the Operator controls. To open an Operator page, click on the corresponding selection field O1 to O8 in the left section. Each Layer features its own set of 8 operators. Because every Operator features the same parameters, it will be described at once.
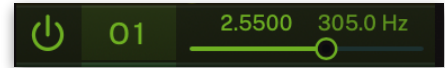
The Copy/Paste buttons lets you copy all settings from this operator and paste it to any other, regardless of its Layer.

- **Operator On/Off**: Click on this button, to activate/ deactivate the corresponding Operator.

- **Waveform**: Click on this button to open a pop-up menu. Here, you can select the desired operator waveform (Sine, TX2 - 8, Triangle, Square, Saw). The TX waveforms are based on a classic FM synthesizer from the 1980s.
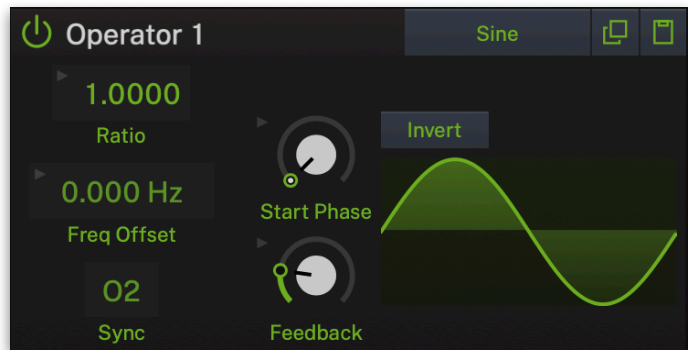
- **Invert**: By clicking this button, the polarity of the selected waveform is inverted. When the button lights up, the phase of the waveform has been inverted.

- **Ratio**: The Frequency Ratio selection causes the frequency of the operator to follow the keyboard. In other words, higher notes produce higher frequencies. 1.00 is the fundamental pitch, so the values 0.25 and 0.50 are considered sub-harmonic. As with the notes on a piano, doubling the coarse tuning value raises the frequency by an octave, and lowering the value by half produces a frequency that is an octave lower. For example, a value of 2.00 is an octave higher than a value of 1.00, and a value of 0.50 is an octave lower than a value of 1.00.

- **Freq Offset**: This option is comparable to the pitch parameter of an oscillator. Based on the fundamental pitch, the operator can be tuned in Hertz to positive and negative values.

- **Sync**: Synchronizes the current operator to the selected one (O1 to =8). Synchronization means, that the operator is reset with each full wave cycle of the selected master operator. It produces a rich sync-sound. Any operator can be synced to any other operator besides itself. Recursive sync is prevented, the second sync connection is disabled in this case.

- **Start Phase**: Controls the phase with which the Operator begins its cycle when a note is played. Typically, you set **Startphase** to "Free" if you want to simulate a typical analog synthesizer waveform or if you want to have some life in the sound. When you set up more than one Operators 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.

- **Feedback**: Controls the feedback level; i.e, the amount of self-modulation an operator will apply to itself. The higher the feedback, the more complex the sounding result. Feedback can also set in the FM Matrix. If you make any changes here, it affects the Feedback dial and vice versa.

5.5 Sample Oscillator 1 and 2

On this pages, you find the controls for the 2 sample oscillators. To open this page, click on the corresponding selection field S1 or S2 in the left section. Each Layer features 2 separate sample oscillators. Because both Sample Oscillators feature the same parameters, it will be described at once.
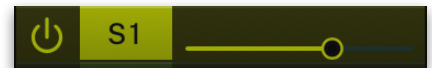
The Copy/Paste buttons lets you copy all settings from the current Sample Oscillator and paste it to any other regardless of its Layer.


- **Sample Oscillator On/Off**: Click on this button, to activate/ deactivate the corresponding Sample Oscillator.

- **Sample Selection**: Click on this button to open a window, where you can select a sample. After loading, the sample name will be displayed next to the Browse button. See also the section „Loading Samples" on the next page.
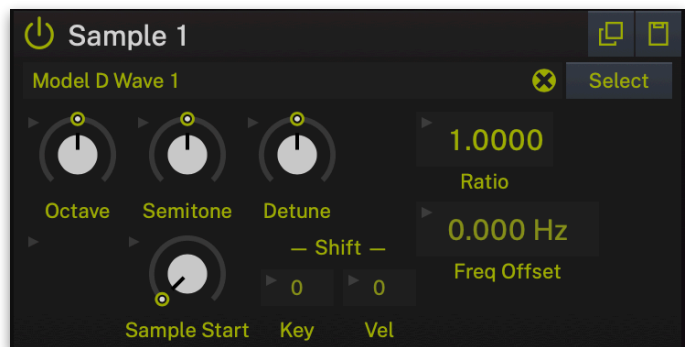
- **Octave**: Sets the basic pitch of the Sample Oscillator in steps of an octave.

- **Semitone**: Sets the pitch of the Sample 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 Sample 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 Sample Oscillator and an equivalent negative setting for another.

- **Ratio**: The Frequency Ratio selection causes the frequency of the Sample Oscillator to follow the keyboard. In other words, higher notes produce higher frequencies. 1.00 is the fundamental pitch, so the values 0.25 and 0.50 are considered sub-harmonic. As with the notes on a piano, doubling the coarse tuning value raises the frequency by an octave, and lowering the value by half produces a frequency that is an octave lower. For example, a value of 2.00 is an octave higher

than a value of 1.00, and a value of 0.50 is an octave lower than a value of 1.00.

- **Free Offset**: This option is comparable to the pitch parameter of an oscillator. Based on the fundamental pitch, the Sample Oscillator can be tuned in Hertz to positive and negative values.

- **Sample Start**: 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.

- **Shift**: The **Key** option 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. The **Vel** option 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.

Loading Samples

To load samples into a Sample Oscillator, you can either click on the **Browse** button or drag & drop a samples from your operating system's file manager onto the sample name field of the respective oscillator. Upon dropping the sample file, it is copied into the user sample library folder.

Clicking on the **Sample Selection** button opens a browser window where you can toggle between **Preset** samples which comes with F-em and **User** samples. The user sample list depends on the content of F-em's user sample folder on your hard drive. 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.
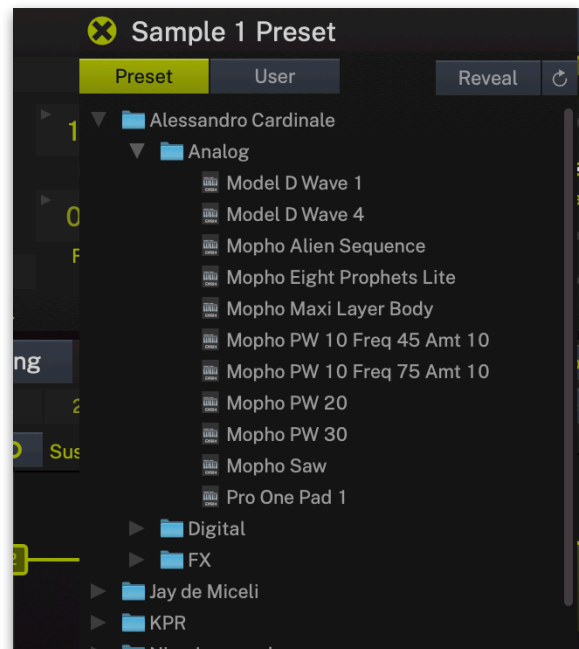
Data path for the user sample folder:

Windows: C:\ProgramData\Tracktion\F-em\Samples

Mac: /Library/Audio/Presets/Tracktion/F-em/Samples

Linux: /var/Tracktion/F-em/Samples

5.6 Common Modulation Sources

Each Operator (O1 to O8), both Sample Oscillators (S1 and S2), the Noise Generator (N) and also both Filters (F1 and F2) offer the same modulation source that will be described in detail on the next pages.

Pitch LFO

This low-frequency oscillator (LFO) is used to modulate the pitch of the corresponding tone generator, respective the filter cutoff for Filter 1 and 2 periodically, but you can freely assign it to other modules as well. 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.

- **Pitch LFO On/Off**: Click on this button, to activate/deactivate the corresponding Pitch LFO.

- **Assign**: Click on the "Assign" button to call up the modulation assignment for the corresponding LFO. Read more about this on page 49.

- **Waveform**: Selects the waveform the Pitch LFO generates: Sine, Triangle, Pulse, Pulse +, Saw down, 4 and 8 steps Up, Sample & Hold, Noise, Saw Up, 4 and 8 steps down. The actual waveform is displayed below the waveform button.

- **Mono**: If deactivated, each voice of a sound has its own Pitch LFO starting at the selected **Start Phase** 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 start phase and each following voice synchronizes to it.

- **Clock**: Activates or deactivates the synchronization of the **Rate** to the internal clock, that is delivered by the host application. Please note that this parameter only synchronizes the rate of the Pitch LFO. If you want to synchronize to the note start, set **Start Phase** accordingly.

- **Rate**: Controls the Pitch 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.

- **Depth**: A setting of 100% means that the Pitch LFO fully affects the modulated signal in the positive direction. Lower settings reduce the strength of the Pitch LFO modulation. Negative settings produce a modulation in the negative direction.

- **Start Phase**: Controls the phase with which the Pitch 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.

- **Delay**: Controls the time before the Pitch LFO waveform is faded in or out. Depending on the setting of **Fade**, it either causes the Pitch 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 in or fade out of the Pitch LFO waveform. When set to center, the Pitch LFO stays at its maximum amplitude forever. When the parameter is set below center, the Pitch 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**.

> ✎ Keep in mind, that this LFO modulates the pitch for the selected Operator, Noise Generator or Sample Oscillator. All LFO parameters are identical.

Level LFO

This low-frequency oscillator (LFO) is used to modulate the level of the corresponding tone generator or filter periodically, but you can freely assign it to other modules as well. 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.
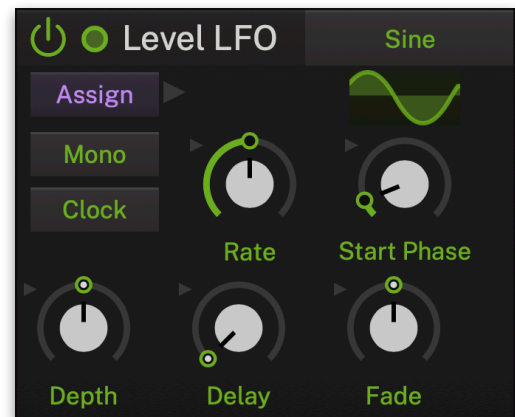
- **Level LFO On/Off**: Click on this button, to activate/deactivate the corresponding Level LFO.

- **Assign**: Click on the "Assign" button to call up the modulation assignment for the corresponding LFO. Read more about this on page 49

- **Waveform**: Selects the waveform the Level LFO generates: Sine, Triangle, Pulse, Pulse +, Saw down, 4 and 8 steps Up, Sample & Hold, Noise, Saw Up, 4 and 8 steps down. The actual waveform is displayed below the waveform button.

- **Mono**: If deactivated, each voice of a sound has its own Level LFO starting at the selected **Start Phase** 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 start phase and each following voice synchronizes to it.

- **Clock**: Activates or deactivates the synchronization of the **Rate** to the internal clock, that is delivered by the host application. Please note that this parameter only synchronizes the rate of the Level LFO. If you want to synchronize to the note start, set **Start Phase** accordingly.

- **Rate**: Controls the Level 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.

- **Depth**: A setting of 100% means that the Level LFO fully affects the modulated signal in the positive direction. Lower settings reduce the strength of the Level LFO modulation. Negative settings produce a modulation in the negative direction.

- **Start Phase**: Controls the phase with which the Level 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.

- **Delay**: Controls the time before the Level LFO waveform is faded in or out. Depending on the setting of **Fade**, it either causes the Level 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 in or fade out of the Level LFO waveform. When set to center, the Level LFO stays at its maximum amplitude forever. When the parameter is set below center, the Level 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**.

Pitch Envelope

This multistage envelopes with up to 32 nodes is used to modulate the pitch of a tone generator (O1 to O8, Noise as well as S1 and S2). For Filter 1 and 2 it is called Cutoff Envelope and modulates, was is says, the cutoff frequency. Envelopes are essential parts of any synthesizer. They are mostly used to change the volume of a sound over time, which is essential for any sound from any acoustic or electronic instrument. In this special case, this envelope controls the pitch but you can freely assign it to other modules as well.

The **Copy/Paste** buttons lets you copy all settings from this envelope and paste it to any other, regardless of its Layer.

Click on the **Magnifying Glass** symbol in the upper right corner to maximize the envelope curve display. Here you can find additional parameters. To close the maximized view, click on the X symbol in the upper left corner of the envelope window.

Basically, this 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 (up to 32), 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.

- If you have setup other Pitch Envelopes for the same Layer, the corresponding curve is shown as overlay within the current envelope. This might be helpful when comparing envelopes.

- **Breakpoint** handle: Controls the time for 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 49.

- **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**: Click on this button to activate the sustain loop (lights up when active). With the two fields above the **Sustain Loop** button 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.

- **Stages Loop**: Click on this button to activate the stage loop (lights up when active). With the two fields above the **Stage Loop** button you can define the stage loop area. This area will be displayed within the envelope graphic. We recommend to

make useful loop settings which means lower settings for the stage loop start as for the stage loop end.

- **Clock**: Activates the synchronization of the Envelope breakpoints to the internal clock, that is delivered by the host application. If activated, a grid is displayed within the envelope representation. Breakpoints will snap to the grid. The density of the grid can be adjusted by the value field above the Clock button.

- **Depth**: A setting of 100% means that the Pitch Envelope fully affects the modulated signal in the positive direction. Lower settings reduce the strength of the Pitch Envelope modulation. Negative settings produce a modulation in the negative direction.

- **Keytrack**: This parameter affects the overall duration of the envelope. When the value is above zero the envelope duration will decrease as higher notes are played. The higher the value of this number, the greater the difference will be between the lower notes and the higher notes.

- **All Rates** (only available in maximized view): This is an offset parameter that affects all envelope rates simultaneously. You can speed up or slower the expiration speed of the envelope.

- **Decay** (only available in maximized view): This is an offset parameter that affects the decay envelope rates simultaneously. You can speed up or slower the expiration speed of the decay stage.

- **Release** (only available in maximized view): This is an offset parameter that affects the release envelope rates simultaneously. You can speed up or slower the expiration speed of the release stage.

> ✏️ Keep in mind, that this envelope modulates the pitch for the selected Operator, Noise Generator or Sample Oscillator. All envelope parameters are identical.

Level Scaling

Level scaling changes the loudness of the sound by using the played key number as a modulation source.

The **Copy/Paste** buttons lets you copy all settings from this scaling option and paste it to any other, regardless of its Layer.
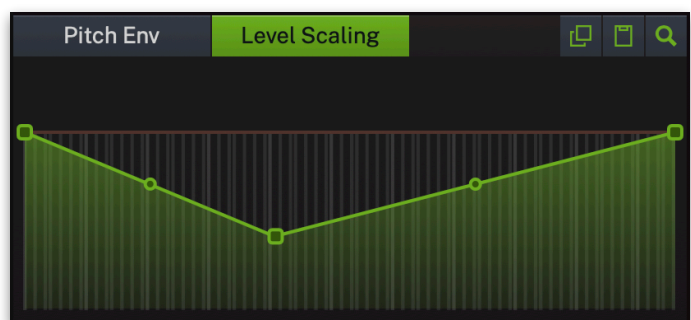
Click on the **Magnifying Glass** symbol in the upper right corner to maximize the scaling graphic display for a better overview. To close the maximized view, click on the X symbol in the upper left corner of the envelope window.

- To add more breakpoints, double click on the line between two points. A new stage number will be added automatically. To delete a stage, hold the ALT key and double click directly onto the corresponding point.

- **Breakpoint** handle: Controls the level of the respective keyboard section. These parameters can be adjusted by clicking on the corresponding breakpoint handle and moving it in the display horizontally or vertically.

- **Slope** handle: Controls the slope of the respective area. These parameters are adjusted by clicking on the rounded rectangular handles in the 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.

Level Envelope

This multistage envelope with up to 32 nodes is used to modulate the level of a tone generator (O1 to O8, Noise as well as S1 and S2) or the filter output. Envelopes are essential parts of any synthesizer. They are mostly used to change the volume or cutoff frequency of a sound over time, which is essential for any sound from any acoustic or electronic instrument. In this special case, this envelope controls the level but you can freely assign it to other modules as well.

The **Copy/Paste** buttons lets you copy all settings from this envelope and paste it to any other, regardless of its Layer.

Click on the **Magnifying Glass** symbol in the upper right corner to maximize the envelope curve display. Here you can find additional parameters. To close the maximized view, click on the X symbol in the upper left corner of the envelope window.

Basically, this 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 (up to 32), 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.

- If you have setup other Level Envelopes for the same Layer, the corresponding curve is shown as overlay within the current envelope. This might be helpful when comparing envelopes.

- **Breakpoint** handle: Controls the time for 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 49.

- **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.

- **Declick**: 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. Remember to check every sound layer when hunting for clicking.

- **Sustain Stages Loop**: Click on this button to activate the sustain loop (lights up when active). With the two fields above the **Sustain Loop** button 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.

- **Clock**: Activates the synchronization of the envelope breakpoints to the internal clock, that is delivered by the host application. If activated, a grid is displayed within the envelope representation. Breakpoints will snap to the grid. The density of

the grid can be adjusted by the value field above the Clock button.

- **Velocity**: Specifies how much volume level will be affected by keyboard velocity. Use this feature to give more expression to the sound. With a setting of 0.00 velocity will have no effect on the volume. For positive settings the volume rises with higher velocities.

- **Keytrack**: This parameter affects the overall duration of the envelope. When the value is above zero the envelope duration will decrease as higher notes are played. The higher the value of this number, the greater the difference will be between the lower notes and the higher notes.

- **All Rates** (only available in maximized view): This is an offset parameter that affects all envelope rates simultaneously. You can speed up or slower the expiration speed of the envelope.

- **Decay** (only available in maximized view): This is an offset parameter that affects the decay envelope rates simultaneously. You can speed up or slower the expiration speed of the decay stage.

- **Release** (only available in maximized view): This is an offset parameter that affects the release envelope rates simultaneously. You can speed up or slower the expiration speed of the release stage.

5.7 Filter 1 and 2

On this page, you find the filter and drive controls. To open this page, click on the F1 or F2 selection field in the left section. Each Layer features 2 filters with drive.

As for the Operators, the Filters features hardwired modulations sources as a Cutoff LFO, a Level LFO, a Cutoff Envelope and Level Scaling and finally a Level Envelope.

> ✎ To hear the effect of a filter and the drive parameter, you need to route an operator/noise generator/sample oscillator signal through the corresponding filter. This needs to be done in the FM Matrix.

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. F-em offers various Filter types abbreviated LP, BP, HP or BS, followed by a number denoting the slope in decibel per octave. F-em offers two filter stages that are arranged in serial wiring.

The **Copy/Paste** buttons lets you copy all settings from this envelope and paste it to any other, regardless of its Layer.
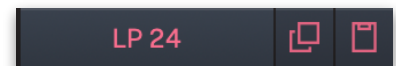
- **Filter On/Off**: Click on this button, to activate/deactivate the corresponding Filter module.

- **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 operator output. Keep in mind that FM is only available for the 12 and 24 dB filter types as well as for all Ladder types.

- **Invert**: Inverts the phase of the corresponding filter.

Drive

The Drive stage offers an overdrive effect, also known as saturation or distortion. It is routed behind the corresponding Filter so it works on the filtered sum of all 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.

> ✎ Cutoff LFO, Level LFO, Cutoff Envelope, Level Scaling and Level Envelope are described in separate sections in chapter 5.6, because they offer identical parameters for all modules.
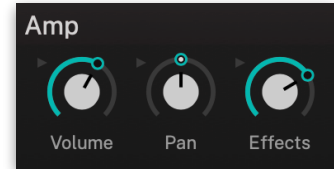
5.8 EQ - Equalizer and Amp

On this page, you find the amplifier controls as well as a 4-band equalizer. To open this page, click on the EQ selection field in the left section. Each Layer features its own equalizer and amp controls.

Amp

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

- **Volume**: The signal generated by the FM tone generator and filter modules of F-em 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 an ADSR **Envelope**, which is hardwired to **Volume** and can be found at the bottom of the Browser page (Attack, Decay, Sustain, Release). Furthermore, a modulation assignment where the note velocity controls volume is already set up for you when you initialize a layer.

- **Pan**: Controls the placement of the Layer in the stereo panorama. This control is identical with the **Panning** dial. A modulation of this parameter can have a range of up to 200 %. That means, if you place the Layer fully to the left and modulate it e.g. with LFO 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 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

Each layer offers a four-band equalizer (Low/Mid1/Mid2/High). You can edit the equalizer by using either the controls or the response graphic.
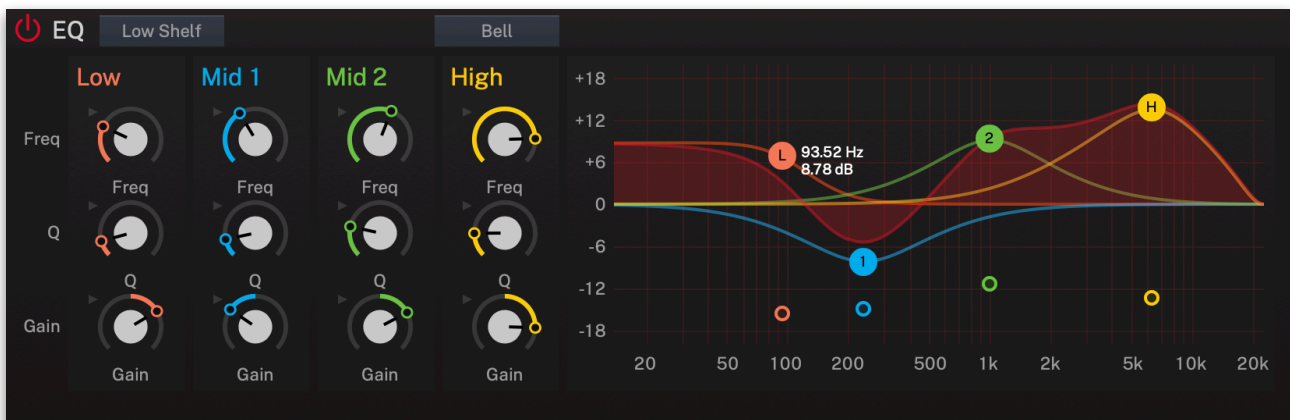
- **Low/High Band Selection**: Selects the filter type of the first and last band. A **Shelf** filter type looks like a normal or an inverted "S" and it sounds and works similar to a low or high pass filter with adjustable stop band attenuation or boost. A **Bell** filter looks like a normal or inverted 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 off 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 representation, drag the desired button (L, 1, 2, or H) and move it to the left or the right to change the frequency. Move it up or down to change the gain. The colored small buttons under the curve are for editing the quality of the desired band.

5.9 Arpeggiator

On this page, you find the arpeggiator controls. To open this page, click on the Arp selection field in the left section. Each Layer features its own arpeggiator. All arpeggiators can be controlled in the Arpeggiator Master section.
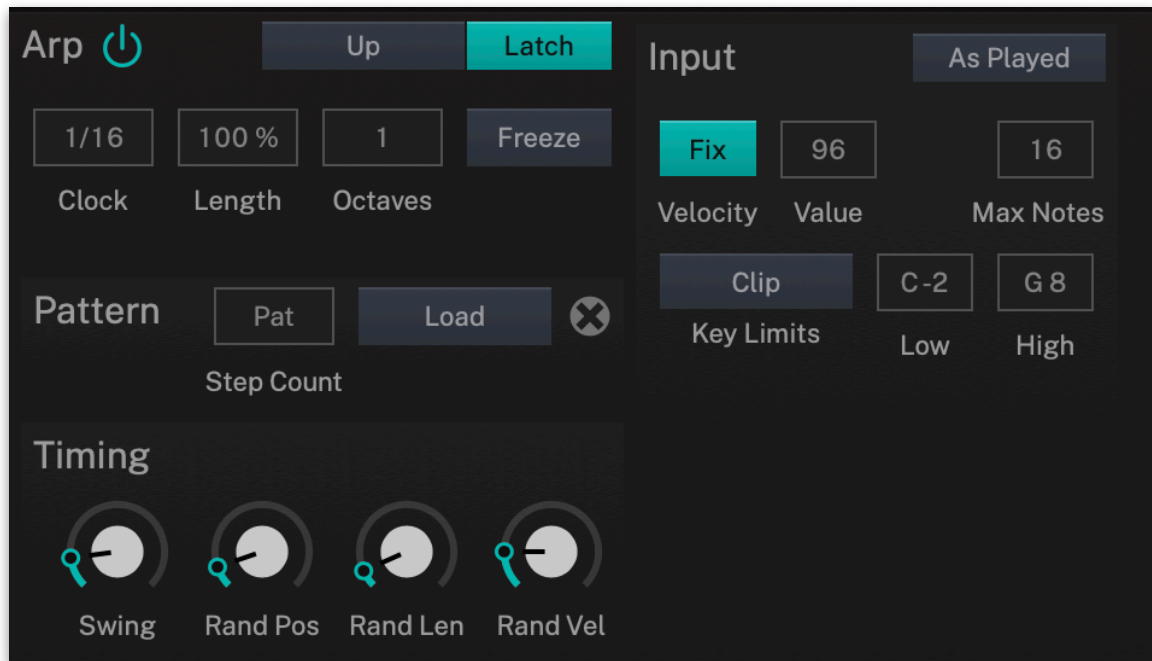
- **Activate**: Toggles the operation of the arpeggiator. The arpeggiator is used to sequence rhythmic patches, disabling it allows the user to access individual hits and elements.

- **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 "Activate" button.

- **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 **Octave**). 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 **Octave**) 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
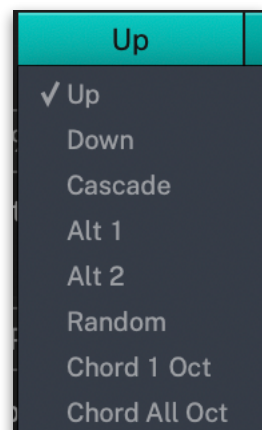
with all notes from the note list. This means that you have to play at least two notes to hear the effect. The note list is not advanced. If "Chord All Oct" is selected, the Arpeggiator plays a chord with all notes from the note list. If **Octave** is set to 2 or higher, all notes of the selected octaves are played simultaneously.

- **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.

- **Octave**: 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 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 F-em. 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 turquoise). 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.

5.10 Effects 1 to 4

To open the Effects page, click on the Effects selection field in the left section. Here, you find all parameters regarding to the four effect busses. Each Layer features its own effect bus. For each effect bus, one of 9 different effect types can be selected. Use the effect send slider to send the audio signal produced by the FM synthesis engine into the corresponding effect section.

The Effect Types in Detail

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

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

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.
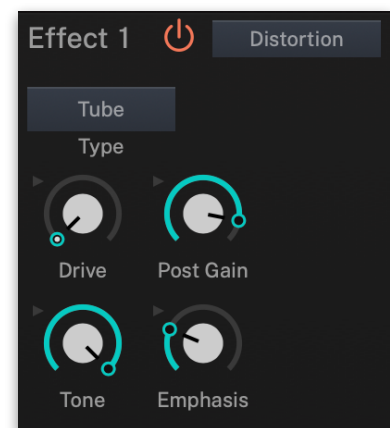
Chorus

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, it

automatically decreases the depth in the chorus so that the detune effect stays the same. If you want to change the detune, adjust **Depth**.
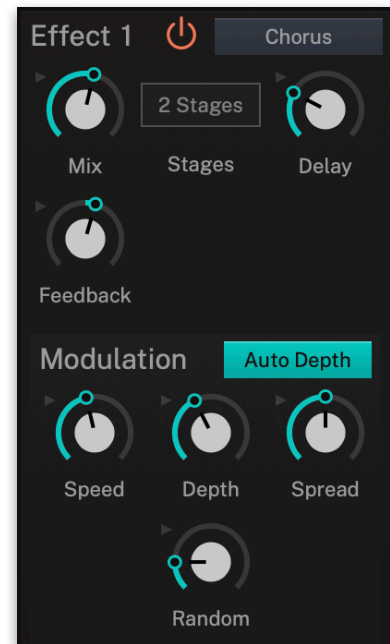
- **Mix**: Adjusts the relation between the dry signal at the full left to the wet effect signal at the full right.

- **Stages**: This effect 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. Click & drag the mouse to switch between the stage types.

- **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. Click & drag the mouse to switch between the stage types.

- **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.
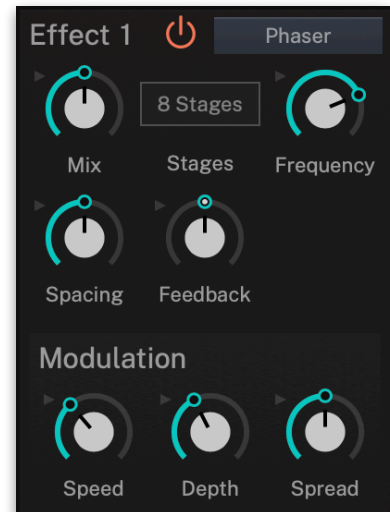
Compressor

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.
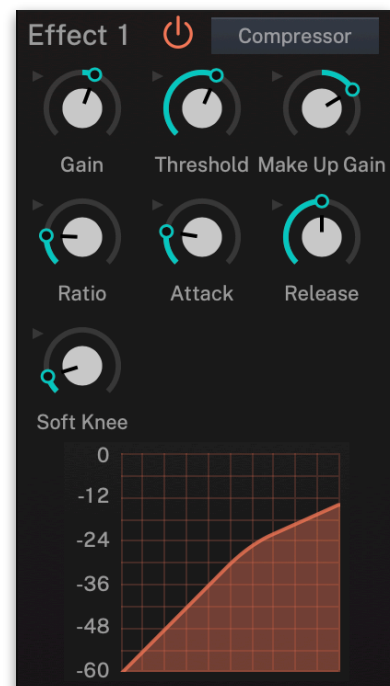
- **Ratio**: Sets the compression ratio. 1:1 means that no compression takes place, while higher values result in an increasingly stronger compression.

- **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 1 and 2" 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 LP6-24, Sallen-Key LP)

- **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

F-em 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.
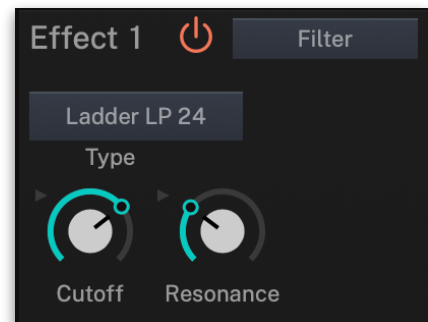

- **Mix**: Adjusts the relation between the dry signal at the full left to the wet effect signal at the full right.

- **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 synchro-nized to the current tempo in the Master section.

- **Feedback**: Controls the level of the signal that is fed back into the delay.
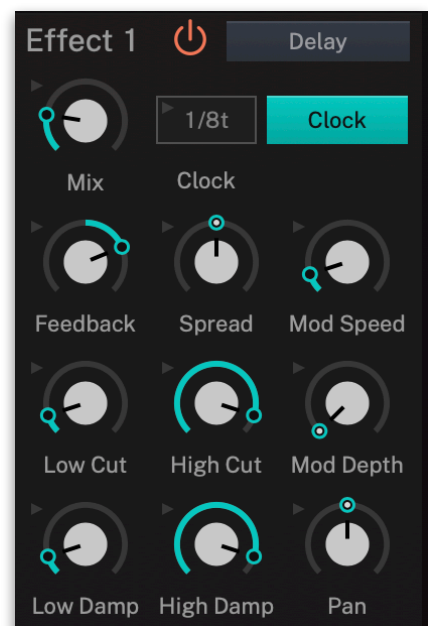
- **Spread**: Adjusts the relative time of the first repetition compared to the delay length. A value of 0 % means that the delay of both stereo sides produce their first repetition at the same time while smaller values result in the left side repeating first and larger values result in the right side 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

F-em 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
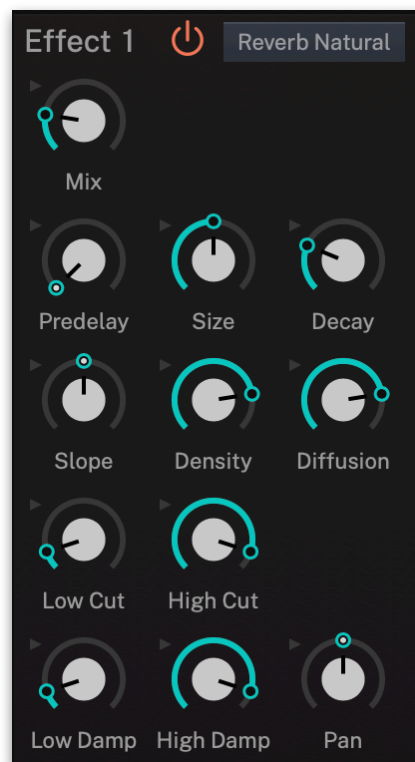
shorter ones. If the reverb starts to sound not dense enough, you might try to increase the Decay of the sound, maybe a snare drum, by a little to smooth it out.

- **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 corresponding reverb effect.

5.11 The Mod Matrix

On this page, you find the modulation matrix controls. To open this page, click on the Mod selection field in the left section. Each Layer features its own modulation matrix with up to 200 modulation slots.

> ✏️ We strongly recommend to use the Mod Matrix only as an overview of your assigned modulations, not for setting up them. Because of the massive number of sources and amounts, this is an extensive amount of work. Therefore, we have implemented the Assign function which makes modulation assignments very easy. Go for it!

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". F-em offers 200 independent modulation assignments (slots) each with individual settings of two sources, destination and amount - and that for everyone of the 4 Layers. A total of 800 modulation options per sound.

To set up a modulation, click on the first Source pop-up menu and select a desired source, e.g. "Flow 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. "Pan Mix Op 8". Finally, set amount to a desired value, for example to 50.0%. Now you have set up a periodic panning modulation that can be controlled by the 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.

Again, you should use the comprehensive modulation assignment of F-em to easy add modulations. Read more about this in the chapter "Modulation Assignment".

5.12 The Layer Page

Here you find the basic Sound controls as Unison Mode, Key Range and Velocity Range as well as a comprehensive Trigger management. To open this page, click on the layer selection field in the left section. Each Layer features its own Layer page.
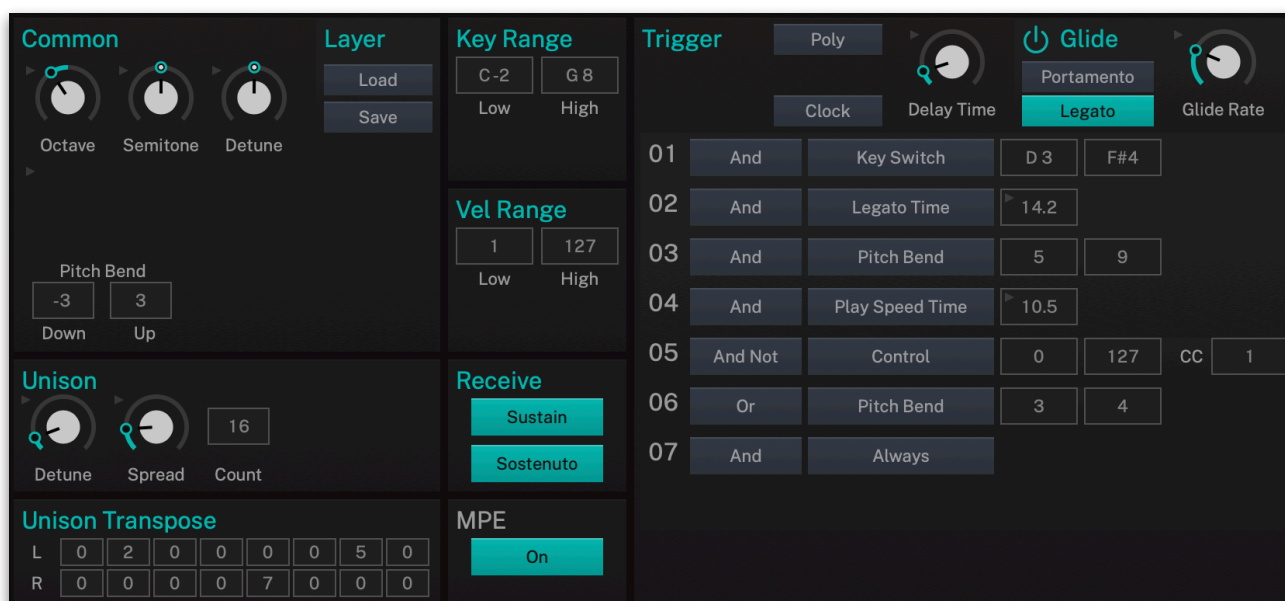
Common

- **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.

Load and Save

Here, you can save all settings of a Layer on your hard drive and load it when desired on the current Layer. The saved Layer contains the file extension .fmsound

Pitch Bend

Here you can determine the bend range for the Pitch bend. The range goes from -60 to 60 semitones.

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. With **Fade**, you can define a fade zone in which the volume of the note raises or falls. If you, for example, set C3 for **Low** and 5 for **Low Fade**, the full volume is reached at F3 (5 semitones higher than C3).

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. With **Fade**, you can define a fade zone in which the velocity of the note rises or falls.

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.

MPE

Click on this button, to activate the MPE option for the current Layer. The global MPE option must also activated.

In this mode, F-em reacts on incoming MPD data that is sent by a MPE Controller as for example the Roli Seaboard.

MIDI Polyphonic Expression (or MPE) is a method of using MIDI to enable expressive electronic musical instruments to control multiple dimensions of sound polyphonically. In MIDI, channel-wide messages (such as pitch bend, CCs, and channel aftertouch) are applied to all notes being played on a single channel; therefore, in MPE each note is assigned its own channel so that those messages can be applied to each note individually.

An MPE instrument typically has three dimensions of expression / control – left-right (X-axis); front-back (Y-axis); and pressure (Z-axis) – each axis can be mapped to various parameters of sound and be applied on a per-note basis.

Each MPE dimension is mapped to the F-em sound engine as follows:

- Left-right / X-axis (sent as MIDI Pitch Bend) – This controls the pitch of the note, with the range set using the Pitch Bend range (see above).

- Front-back / Y-axis (sent as MIDI CC 74) – This controls the timbre modulation assignments.
- Pressure / Z-axis (sent as MIDI Channel Aftertouch) – this controls the Aftertouch modulation assignments.

Trigger

F-em 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 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 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 hi-hat sounds where an open hi-hat sound cuts off the closed hi-hat 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. F-em offers three XOR groups so that you can have an alternating hi-hat, 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.
- **Rotate #**: Here you can define, which variation group is used for the actual Sound layer.
- **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.

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 Operators.

- **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.

5.13 The Macros Tab

Here you find two Pad controllers and Macro dials and slider.

The pad controllers let you drastically change the sound of the selected Instrument. Use the green point to freely move it in X and Y axis. You can assign the X and Y axis sources independently in the Sources tab.

The 5 sliders control the overall Level envelopes and Effect levels of all Layers. These parameters can also be found at the bottom of the Loading page. Keep in mind, that the macro parameters produce an offset to the original settings. For example, if all Level envelopes are have an Attack of 0, a negative setting of the Macro Attack dial won't produce any result.

There are two dials to control the overall Timbre and Feedback.

> ✎ The Macro slider parameters can also be found at the bottom of the Browser page.

6. Modulation Assignment

6.1 About Modulations

F-em features a powerful modulation architecture. Any of the available modulations sources like LFOs, Envelopes, MIDI Note Numbers, Velocity, Pitch Bend Wheel and so on can modulate any of the continuous sound parameters like Operator Frequency Offset, 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 LFOs and the Envelopes are located in the upper left corner of its section (see picture opposite), while the further available sources are collected in a dedicated page section at the bottom of the user interface. For ease of understanding, we call them all just Assign buttons.

LFO 1 Modulation section at the bottom of the Main Edit Page

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:

destination = source x depth

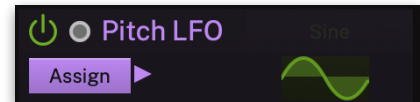However, just assigning one modulation source to a destination is often not enough. Imagine controlling a pitch vibrato by the modulation wheel. For this to work, you need two modulation sources, where one controls the depth of the other, in this case the LFO controlled in depth by the Modulation Wheel. F-em 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:

destination = source 1 x source 2 x 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 Operator 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 800 modulations per sound (if you use all 4 Layers) but F-em 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 F-em sections changes slightly and the selected Assign button lights. 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 LFO 1 to the Feedback of Operator 8 but a negative one to Noise Cutoff. 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 1 (F1) page and set up Filter 1 as desired. We recommend to set Cutoff to a lower setting, so that it sounds noticeably duller.

2) Click on the MW Assign button in modulation assignment section. The user interface will change slightly and is now ready for destination selection and amount setting.

3) 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 colored line.

4) Now the modulation wheel controls the cutoff frequency of Filter 1 with the corresponding amount. A little rectangle beside the Cutoff dial shows, that a modulation assignment has made. Click on this rectangle to view the modulation source(s) and the corresponding amount.

5) 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. Now you can move the Filter 1 cutoff knob to add the amount.

6) If you now play with different velocities, the filter modulation controlled by the Mod wheel 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 Operator Start Phase parameter by an Envelope always results in the same behavior, since the Envelope always starts at 0 while Start Phase, as its name already says, only affects the start phase of the Operator and doesn't do anything afterwards

6.2 Modulation Sources

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

These sources are located at the bottom of the user interface and stays always active, regardless of the selected edit page above.

Flow LFO 1 and LFO 2

Macros	LFO 1	LFO 2	Sources	Modifier 1	Modifier 2	Modifier 3

| LFO 1 | LFO 2 | Key | Vel | Rel | AT | PP | PB | MW | BC | FC | SP | Sos | Timbre | RND 1 | RND 2 | RND 3 | RND 4 |

Flow LFO 1 — Mono — Clock

Flow LFO Slot Matrix — Slots 8

Assign — Level — Delay — Fade

Shape	Sine	Triangle	Pulse +	4 St Up	4 St Dn	Pulse	Saw Dn	Pulse
Slot								
Edit								
Rate	1.00 Hz	1.00 Hz	13.5 Hz	12.9 Hz	318 mHz	1.00 Hz	1.00 Hz	1.00 Hz
Phase	8.6 °	Free °	Free °	Free °	Free °	Free °	Free °	Free °
Symmetry	50.0 %	50.0 %	50.0 %	50.0 %	50.0 %	50.0 %	50.0 %	50.0 %
Level	100 %	100 %	100 %	100 %	64.8 %	100 %	100 %	100 %

F-em offers 2 Flow 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 F-em offers free modulation assignments, you can also modulate other aspects of the sound with the LFO like e.g. one of the **Envelopes Decay**, or the **Band Frequencies** of the Equalizer.

The F-em Flow LFOs contain a unique functionality: Both Flow 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** or **LFO 2** button to call up the corresponding Flow LFO.

- **Assign**: Click on the "Assign" button to call up the modulation assignment for 49
- **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 **Start Phase** and running with its rate. If activated, the sound only uses one LFO for all voices. The first voice of any overlapping notes determines the start phase 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 **Start Phase** accordingly.

- **Level**: Controls the amount of modulation applied by the Flow LFO.

- **Delay**: Controls the time before the Flow LFO waveform is faded in or out. Depending on the setting of **Fade**, it either causes the Flow 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 fade in for the Flow LFO waveform. When set to center, the Flow LFO stays at its maximum amplitude forever. When the parameter is set below center, the Flow 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**.

- **Shape**: 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 Down as well as a user definable Curve)see below). The current selected 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.

- **Start Phase**: Controls the phase in degree 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. If you want to have a Start Phase of 0°, just set it to 360° rather than trying to move the parameter as close as possible to Free.

- **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 Level**: A setting of 100% means that the LFO fully affects the modulated signal. Lower settings reduce the strength of the LFO modulation, negative settings inverts the waveform.
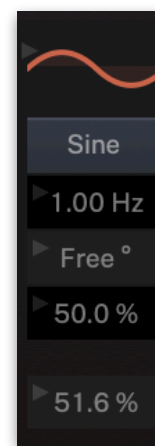
The user-definable LFO Curve

When Curve is selected as LFO Shape, you can easily draw your own LFO waveform. This is how its done:

1) Select Curve as Shape for the desired Flow LFO slot.

2) Click on the Edit button below the curve graphic of the desired LFO.

3) In the curve editor window, you can easily draw your desired LFO curve. Up to 32 stages are available. To edit the number of stages, click on the

Stages value and move the mouse pointer up or down. Stage points will be placed within a grid, that is defined with the Grid parameter.

4) You can also adjust the slope by clicking on the rounded rectangular handles in the 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.

5) To close the LFO Curve editor, click on the X in the upper left corner.

Other Mod Sources

Beside the Flow LFOs and Operator LFOs and Envelopes, F-em offers a lot of other modulation sources. Click on the **Sources** 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. F-em 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 Operator 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 1980s. All parameters that control more or less some kind of pitch, like Cutoff, Redux Rate and certainly Operator Pitch can be played perfectly chromatically when the modulation depth is at maximum.
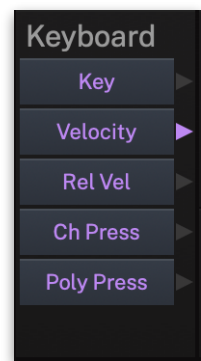
  - **Velocity**: On almost all MIDI keyboards, hitting keys soft or hard results in MIDI notes with different velocities between 1 and 127 (0 is never sent on note on messages as it means „interpret this note as note off" in the MIDI world). This is the same behavior as on a piano where harder played 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 a 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 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. You can also use Velocity to modulate parameter values of other modulation sources, e.g.

modulate a Level Envelope 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 can 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. Poly pressure is an important modulation source when using the MPD (MIDI Polyphonic Expression) option of F-em.

- **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 overall 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 **LFO1** and **MW** in the Assign section, followed by adjusting the **Pitch** modulation depth on the Layer page (in the Common Assign section). You can also use it to control other aspects of a sound, e.g. fade in a **LFO**, 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
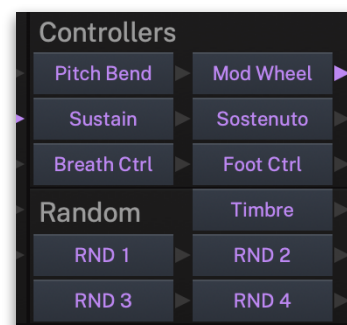
voices of the respective MIDI channel are played. If a value smaller than 64 is received, the "Key Off" envelope phases are played. This means, that a Sustain pedal modulation can be used as switch, e.g. to activate/ deactivate parameter functions.

- **Sostenuto**: This pedal message (MIDI CC #66) can be also used as modulation sources.

- **Breath Ctrl**: A breath controller (MIDI CC #2) 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.

- **MPE Timbre**: The forward/backward movement (Y axis) of a key on a MPE-capable MIDI controller is typically used for continuous control of timbre. Here, you can control every desired parameter to change your sound from subtile to massive.

- **Random** sources (**RND 1, 2, 3, 4**): Upon playing a note, F-em 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. Vintage 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 the carrier **Operator(s) Pitch** and maybe to **Filter Cutoff**, all with some small but different depth setting. You can also use stronger depth settings to create typical sample & hold effects.

- **Pad XY** sources:

- These 2 pad controller (Left XY pad and Right XY Pad) let you drastically change the sound of the selected Instrument. Use the green point to freely move it in X and Y axis. You can assign the X and Y axis sources

independently by clicking on the small A(ssign) button. The source signal an either act as unipolar or bipolar controller, depending on the status of the Bi button. Based on the assigned synthesis parameters, the pads can produce subtle sound changes as well as extrem morphings. For real-time editing, both Pad controllers can be found in the Macros tab.

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			LFO 2			Sources 1			Modifier 1			Modifier 2			Modifier 3	
LFO 1	LFO 2	Key	Vel	Rel	AT	PP	PB	MW	BC	FC	SP	Sos	Tim	RND 1	RND 2	RND 3	RND 4

Modifier 01-10

01	Unison	x	Max	88.3	Assign ▶	06	Off	x	Constant	0.00	Assign ▶
02	Sample 1 ...	x	Constant	100	Assign ▶	07	Off	x	Constant	0.00	Assign ▶
03	Sample 1 ...	x	Constant	100	Assign ▶	08	Off	x	Constant	0.00	Assign ▶
04	Off	x	Constant	0.00	Assign ▶	09	Off	x	Constant	0.00	Assign ▶
05	Off	x	Constant	0.00	Assign ▶	10	Off	x	Constant	0.00	Assign ▶

Click on the **Modifier 1**, **2** or **3** buttons to call up the corresponding modifier section. Up to 30 different modifiers can be used for every Layer (a sum of 120 Modifiers per sound).

- **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.

- **Assign**: Click on the "Assign" button to call up the modulation assignment for the corresponding Modifier slot. Read more about this on page 49.

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

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 ommited. 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 x 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 • 0.5 + 0.5

Switch

Returns 0 if **a** is smaller than **b**, else returns 1. Performs the following operation: y = a • (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 • |a|

Cubic

Returns the cubic value of **a**, resulting in an s-shape. Performs the following operation:
y = a • a • 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 = clip (a • b • 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 •(1 - b) + a • 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[O] = 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 [O] = 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

or write a request via our support form on:

https://tracktion.zendesk.com/hc/en-us/requests/new

Tracktion Software Corporation

11644 NE 80th St

Kirkland, WA 98033

USA

www.tracktion.com

© 2021 Tracktion Software Corporation. All rights reserved.