ADAM SZABO

VIPER

USER MANUAL 1.2.0

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1. Introduction

1.1 Welcome

Thank you very much for using Viper! You now have a very powerful synthesizer in your arsenal, which aims to deliver the sound characteristics of legendary hardware synthesizers in a virtual format. We can call Viper a **Vir**tual **Per**formance synth, with a vast array of features like: multiple oscillator types with user waveforms, Sync, Frequency Modulation, Ring Modulation, Unison up to 8 voices, filter saturation, filter routings, mod matrix, high quality effects, a powerful Arpeggiator, and much more!

A lot of time and research has been invested to make the sound engine of Viper capable of delivering a wide array of sounds, everything from smooth, lush pads, huge basses, and powerful leads, typically associated with traditional analog synthesizers. This was achieved with careful coding, experimenting, analysing hardware synths, and using the ear to fine tune the controls. A constant randomization is also active in the synth to slightly alter its sound to mimic electronic components.

The engine is written in optimised code to use as little CPU consumption as possible, while maintaining a high quality sound engine, finding a good balance between the two. The synth automatically turns off any parts that are not used to further reduce CPU usage. Viper also has a smoothing algorithm for its controls to remove the 'steppyness' when it goes between values to give that extra polished sound.

1.2 System Requirements

Windows Vista or higher with SSE2 support (or higher). A 32 or 64-bit compatible DAW or Host.

1.3 Installation

Viper is available in 3 different plugin versions:

VST2.4 32-bit: Viper.dll

VST2.4 64-bit: Viper (64).dll

**VST3 64-bit:** Viper (64).vst3

Installing Viper is very easy, simply extract the entire 'Viper' folder into your correct VST folder location, then scan for the new plugin.

Consult your DAW or Host where the VST folders are located, but usually it is best to copy the **VST2.4 32-bit** Viper to: **C:\Program Files (x86)\VstPlugins\Viper**

The **VST2.4 64-bit** to: **C:\Program Files\VstPlugins\Viper**

The **VST3 64-bit** usually goes to: **C:\Program Files\Common Files\VST3\Viper**

Make sure the rest of the folders that contain data are located where the VST plugin file is, as that is where it will look for them. Do not copy them to a different folder from the VST file, otherwise it will not find the data. This is how it should look like (using the 64-bit .dll version for example):
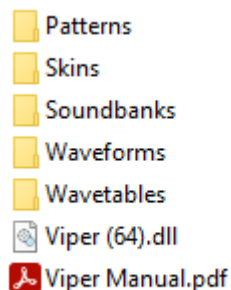
1.4 Registration

Once you start your host, insert Viper and follow the on-screen instructions. Enter your name and serial number into the text fields (you can use Ctrl + C to copy and Ctrl + V to paste) and click 'Register' to activate your license. Remove the current instance of Viper from the host, and insert it

again. If everything went well, the activation was successful and Viper has been unlocked. If the registration window appears again, the data was incorrect and must be entered once more.

1.5 Troubleshooting

If your DAW or Host does not find Viper, cannot load it, or otherwise shows an error of some kind, you might be missing the **Visual C++ Redistributable Package 2013**. Please download and install **vcredist_x64.exe** for the **64-bit**, and **vcredist_x86.exe** for the **32-bit** version:

https://www.microsoft.com/en-us/download/details.aspx?id=40784

If that still did not help, there might be other packages missing on your system. You can try installing an all in one package with all the different Redistributable Packages available:

https://www.techpowerup.com/download/visual-c-redistributable-runtime-package-all-in-one

2. Basic Operation

2.1 Interface and Controls

Viper is a multi-page synth, and has a very clean and logical layout of the interface. Unlike other virtual instruments, Viper uses dynamic tabs to separate the different sections of the synth. The whole synth is divided into seven parts:

- **Oscillator section**
- **Filter section**
- **LFO section**
- **Effects**
- **Modulation Matrix**
- **Arpeggiator**
- **Browser**

The dynamic tabs (along the left-hand side of the synth) show the most important controls and values from each section, making it possible to have a fast overview of the patch and how it is made, without having to go into each section separately. Clicking on one of these tabs, opens that part of the synth and allows for editing of its parameters.

The main display of Viper shows lots of important settings, and displays extra information regarding the current patch.
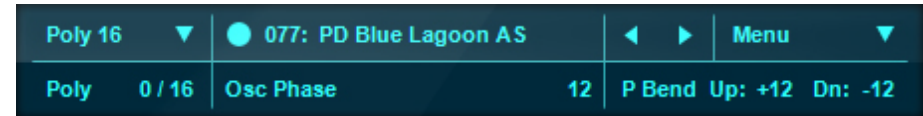
On the left is the **voice selector** and **polyphony display**. The middle section shows the patch number and name, and below that is the **parameter value display**. Whenever you hover over or change a value of a parameter, it will show the name and value of that parameter in the display. Clicking on the circle beside the patch name allows renaming of the preset. To the right of the current preset name are the **program select** arrows for switching the next or previous preset. On the right side of the display is the **main menu**, and the **Pitch Bend** semitone selectors. These selectors set the range of the Pitch Bend knob.

The controls of Viper consists of buttons and selectors that can be clicked, and knobs which can be controlled by dragging up and down in a vertical motion, or by using the **mouse wheel** to move it one step at a time. Knobs can either be unipolar, its values usually ranging from 0-127 or bipolar, ranging from -64 to +63, where the center position is +0. Holding the **Shift** key while dragging enables you to have higher accuracy and fine tune the control.

Double clicking, or clicking on the knob while holding the **Ctrl** key, resets it to its default position.

2.2 Pitch Bend, Mod and Master

The Pitch Bend (**P Bend**) control located beside the main

display is used to temporarily shift the pitch up or down. When released, it automatically snaps back to the center position. It can also be controlled by MIDI as well. Using the Pitch Bend semitone selectors (**P Bend Up** and **P Bend Dn**) in the main display, you can specify the bend range in semitones when Pitch Bend control is turned up or down.

The Modulation control (**Mod**) beside the Pitch Bend, is used as a modulation source for any type of modulation. It can be assigned in the modulation matrix to control all kinds of parameters in the patch. The Mod control can also be controlled by MIDI and is assigned to the MIDI CC#1 by default.

On the top right corner of the interface is the Master gain control. It sets the volume of Viper's audio output after all the effects, and is the last control in the signal chain. It can be used to increase the main volume by +12dB or reduce

it to complete silence (-INF). In the center position there is no gain applied (+0.0dB). The VU meter beside the Master control shows the approximate signal level of the

synths output. If the signal is too loud, the red clipping meters will show up, and an internal **limiter** will stop the signal from clipping and being distorted. If the signal stays below the red meters, the limiter is automatically bypassed to save CPU usage and will not be active.

2.3 MIDI Learn

Most knobs on Viper can be controlled with MIDI Control Changes (CC). You can easily assign a MIDI CC to a knob by right clicking on the parameter and selecting **'Midi Lean'**. The selected knob will wait for a MIDI input. If you want to cancel the assignment, right click on the knob again and select **'Stop Midi Lean'**. Once the knob is waiting for a MIDI input, and you turn a parameter on your MIDI controller, the knob will be assigned to that CC number. The assigned knob will be linked to that MIDI CC indefinitely in all Viper instances, until you remove the link by right clicking on the knob again and selecting **'Remove Midi CC #7'** for example. To clear and remove all learned MIDI links, you can select **'Reset Midi CC'** from the main menu.

2.4 Loading and Saving Presets/Banks

In Viper, the presets are saved in standard Cubase .fxp, and banks (containing 128 presets) in .fxb format. You can load or save banks from Viper's main menu: **Load Bank (.fxb)** or **Save Bank (.fxb)**. To load or save individual presets, select **Load Preset (.fxp)** or **Save Preset (.fxp)**. You can also copy and paste presets using the menu. When you load or paste a preset, it will be loaded to the currently selected program number. To rename a preset, click the circle beside the preset name on the main display. To cancel editing, press the **Esc** key, or click anywhere outside the edit box. To accept the change press **Enter**. If you would like to start a preset from scratch and load all the parameters default setting, select the **Init Preset** from the menu.

2.5 Section Locking

Sometimes when you change presets, you might like a certain effect's or filter's parameter setting, but they get lost upon preset change. What if you want to keep these settings while you browse the sounds? In Viper it is possible to lock parameters so they won't change their values when you load a new preset. Most pages in Viper are divided into different sections, and these are used to group various controls together. They also have their own names like: OSCILLATOR 2 FM, MIXER, AMPLIFIER and so on. Double

clicking on the title area of the section will change its color, and now that section will be locked. You can lock as many sections as you like. To unlock, double click the area again. Once the section is locked, you can still edit its parameters, but changing a preset will not alter their values.

2.6 Voicing and Polyphony

The **Voice Selector** in the main display sets how many voices can play at once. **Retriggered** means all envelopes will retrigger when a new key is pressed even while holding a previous one. **Full Portamento** is the sliding of notes at every key press, while **Legato** only slides new notes when a previous key is being held down.

- **Poly:** 8, 16, 32, 64 or 128 voices.
- **Mono 1:** Monophonic, Retriggered, Full Portamento.
- **Mono 2:** Monophonic, Retriggered, Legato.
- **Mono 3:** Monophonic, Full Portamento.
- **Mono 4:** Monophonic, Legato.
- **Midi Out:** Sends out midi data from the Arpeggiator.
- **Hold:** Holds the last pressed note indefinitely (to stop the held note, select a different voicing option).

2.7 Main Menu

The Main Menu in Viper holds a few different important functions like managing sounds, and changing the GUI size.

- **Soundbanks:** If you have several .fxb banks in the 'Soundbanks' folder where the Viper VST file is located, one can can easily and conveniently load these soundbanks within Viper.

- **Load Bank (.fxb):** Loads a soundbank in .fxb format.

- **Save Bank (.fxb):** Saves a soundbank in .fxb format.

- **Load Preset (.fxp):** Loads a preset in .fxp format.

- **Save Preset (.fxp):** Saves a preset in .fxp format.

- **Import .mid Bank:** Imports .mid Virus soundbanks.

- **Init Preset:** Initializes all the parameters to their default values.

- **Restore Preset:** Restores the currently loaded preset's parameters to their original values.

- **Copy Preset:** Copes the preset to a temporary folder.

- **Paste Preset:** Loads the copied preset from the temporary folder.

- **Reset Midi CC:** Removes all Midi CC assignment.

- **Enable Tooltip:** Enables a tooltip to appear at the mouse cursor when hovering over a parameter. The tooltip contains useful information about the parameter's function.

- **View Schematic:** Displays the signal flow of Viper.

- **GUI Size:** Changes the size of Viper from a selection of pre-defined values. It is also possible to change the size by dragging the bottom right corner. Right clicking the corner resets to the original size.

- **Skins:** Changes the skin of Viper from the available skin files in the 'Skins' folder.

- **User Manual:** Opens this manual.

- **adamszabo.com:** Opens the Adam Szabo website in the internet browser.

- **About…:** Displays the 'About' window.

2.8 Signal Flow

The schematic below shows the internal structure of the audio path in Viper. Each line represents an audio signal. Oscillator 1 drives the Sync and FM (Frequency Modulation) of Oscillator 2. The output of Oscillator 1, 2, and the Sub Oscillator create the RM (Ring Modulation), which has its own output. The signals can be then routed to the filters in various ways. The sum of Oscillator 1, 2, 3, Sub Oscillator, Noise and RM, can either go to Serial 4-6 or Parallel filter modes. In Split mode, Oscillator 1 and the Sub Oscillator go to Filter 1, and Oscillator 2, 3, Noise and RM go to Filter 2. The signal from the filters then goes to the Amp section, where it enters the Effects in this order: Filter Effects, Distortion, Character, Phaser, Chorus, Delay, Reverb, Equalizer, and finally the Limiter.
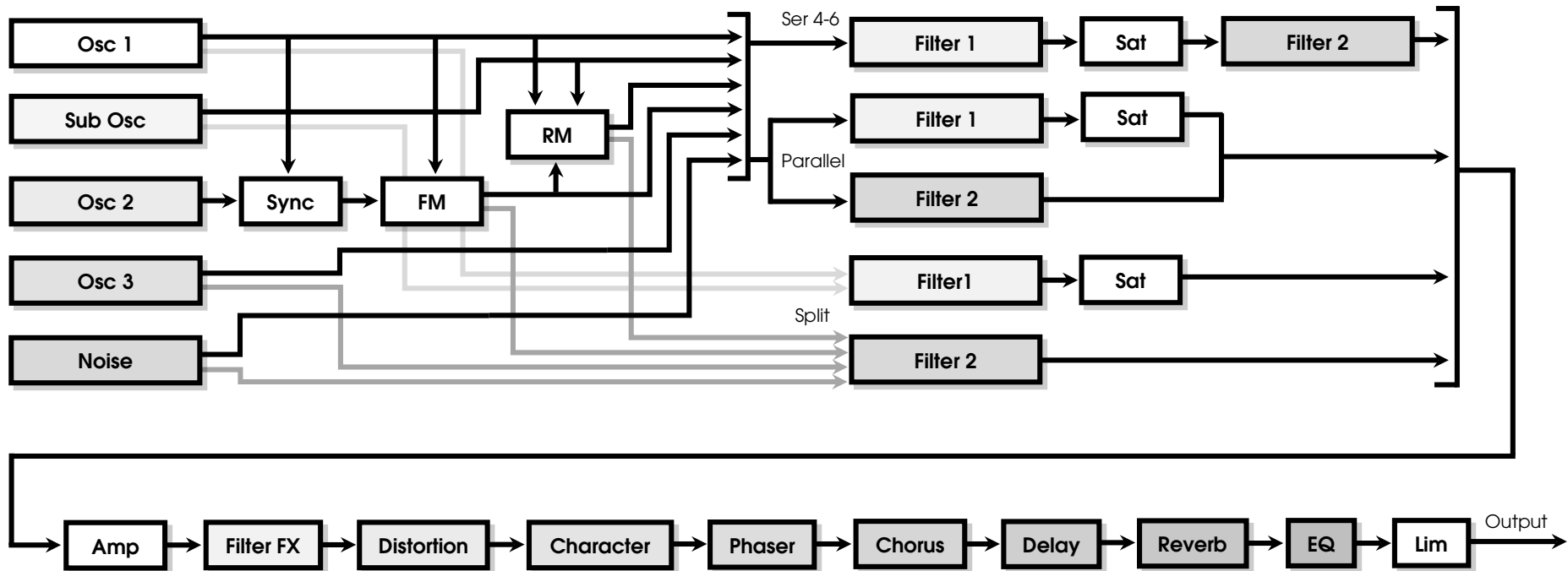
2.9 Optimising Your Patches

Viper is designed to use as little CPU usage as possible, however one can still create a patch that will increase CPU and adversely impact the computer. This chapter will give you some tips on optimising your patches to use less CPU. It is best to come back here after reading through the manual and familiarising yourself with all the functions of Viper.

Polyphony and Unison
High Polyphony and Unison count are one of the biggest contributors to high CPU usage. When designing a patch, one should ask the question: how many voices will I need?
If you know the sound will be a bass, a lead, or it will always play one note at a time, it is best to set the voicing to Mono. It is great to use Unison to thicken the sound, but maybe one can get a very similar thickness using only 3x Unison and Chorus, instead of using 8x Unison? Always try to keep the Polyphony and Unison to a minimum where you can!

Oscillators
As you use more oscillators, the CPU required to generate sound will increase. The Sub Osc, Oscillator 3 and Noise are bypassed when not in use, so always plan ahead and decide the number of oscillators you will need. If you want a big saw sound, try using the Multisaw instead of Unison.

Envelopes
The Release stage in the envelopes also affect the CPU usage. If the Release is high, it will take longer for the voices to end, and the Polyphony can add up very quickly! The Amp and Filter envelopes are always active, but Envelope 3 and 4 are off when not in use. Instead of assigning them to a parameter, try reusing the Amp or Filter envelopes.

Filters
The Filters are also turned off when not in use, and are bypassed by the Filter Balance control. If it is all the way to Filter 1, Filter 2 is turned off, and vice versa. It is also better to use the Serial 6 on Filter 1 to create a simple 24dB filter, instead of enabling both 12dB filters in Serial 4.

LFO and the Matrix
If you want to control the pitch of the Oscillator, or control the Filter, use a dedicated LFO modulation control instead! If you assign it in the Modulation Matrix, the Matrix will not by bypassed anymore and will use additional CPU!

Effects
Turn off any effect which you are not using even if their Mix is 0%. If you run several Vipers with the same kind of effects, it might be better to use external effects and sending them to all the Viper instances.

3. Oscillator Section

3.1 Oscillators

The Oscillators are the heart of any synthesizer, this is where the sound waves are generated. Viper has five sound generation sources: two main Oscillators, a third Oscillator, a Sub Oscillator, and a Noise generator. These can be combined in many different ways to create a vast array of sounds.

Oscillator 1 and 2
The two main Oscillators of Viper contain multiple Oscillator types, and they have dedicated controls to change their behaviour:

- **Classic:** In Classic mode, it is possible to blend between a waveform, a Sawtooth, and a Pulse. The **Shape** knob mixes between the different waveforms. When the knob is at 0, one can select a waveform with either the arrows, or by clicking on the name from the list. Sine and Triangle are generated inside the synthesizer, but the rest are wave files. You can also load your own single cycle waves, if you copy them to **Viper/Waveforms/** in **16-bit, Mono .wav** format. When the Shape knob is at the center, the shape of the wave is a Sawtooth waveform. When it is at maximum, is it a Pulse, and then you can control its width with the **PW** (Pulse Width) control.

- **Multisaw:** The Multisaw is used to generate several Sawtooth waveforms in a single oscillator. The **Voices** control the amount of Saw waves (from 1 to 9), while the **Detune** controls their frequency offset to make the sound thicker. In Multisaw mode, the oscillator also generates an equal amount of sub oscillators an octave below the main oscillator.

- **Wavetable:** The Wavetable oscillator can crossfade between multiple waves from a wavetable .wav file. These are located in **Viper/Wavetables/** in **16-bit, Mono .wav**. The **Morph** control sweeps through the available waves, while the **Inter** knob sets the interpolation amount for the blending of the waves.

- **Grain Osc:** The Grain oscillator adds 'grains' to the selected wavetable, creating a sync like sound. The **Grain** knobs changes the grain size. The grains are also key tracked and follow the notes being played.

- **Formant Osc:** The Formant oscillator is the same as the Grain Osc, except the generated grains are not key tracked and this creates a different texture to the sound, similar to changing the 'formant' of the sound.

Oscillator 3

The third Oscillator is a simple wave generator to complement the main Oscillators. It can be turned off to save CPU. One can select the Factory or User waveforms, including Saw, Pulse, Sine and Triangle. There is also a 'Slave' mode, where Oscillator takes its characteristics from Oscillator 2. The wave type, and semitone will be the same, but the detune value will be opposite that of Oscillator 2.

Sub Oscillator

The Sub Oscillator is used to thicken the sound by adding a basic wave (Pulse, Triangle, Saw, Sine) an octave below the pitch of Oscillator 1. The **PW** control can change the Pulse Width of the wave when the 'Pulse' is selected. If Oscillator 1 or 2 is in 'Multisaw' mode, this oscillator is disabled and replaced by the Multi Sub oscillators instead.

Noise

The noise generator adds 'whitenoise' to the sound. The **Color** knob controls the tone of the noise generated. When set to the middle (+0), it is neutral, it is a white noise. When negative (-64), it is dark, low passed with less top and more bass. Positive (+63), is a high passed noise, thin and bright.

Punch

The Punch generates a little click at the beginning of each key press, to add more transient and bite to a sound. The **Intensity** control determines the volume or intensity of the

punch. This can only be heard if the Amp Attack is set to a low value, otherwise a high attack will take away the punch.

Other controls for the Oscillators include:

Semi

Adjusts the tuning of the Oscillator in semitones ranging from -48 to +48.

Key Flw

Controls how much the pitch of the oscillator follows the keyboard notes. At value +32 the keys are in 'normal' mode, working as one would expect. Higher values push the pitch of the keys further. At negative ranges the pitch is inverted, meaning high keys will create low notes, and low keys create high notes.

Vel > Shp

This determines how much the note velocities control the oscillator shape parameter.

Detune 2/3

The Detune fine-tunes the pitch of Oscillator 2 upward, or Oscillator 3 downward, in relation to Oscillator 1.

3.2 Oscillator Common

The Oscillator Common section contains controls which affect multiple oscillators and their behaviour in relation to each other.

Sync

This enables oscillator synchronisation between Oscillator 2 and Oscillator 1 where Oscillator 1 (Master) will reset the wave cycle of Oscillator 2 (Slave). Play with the Semitone controls to hear the effect.
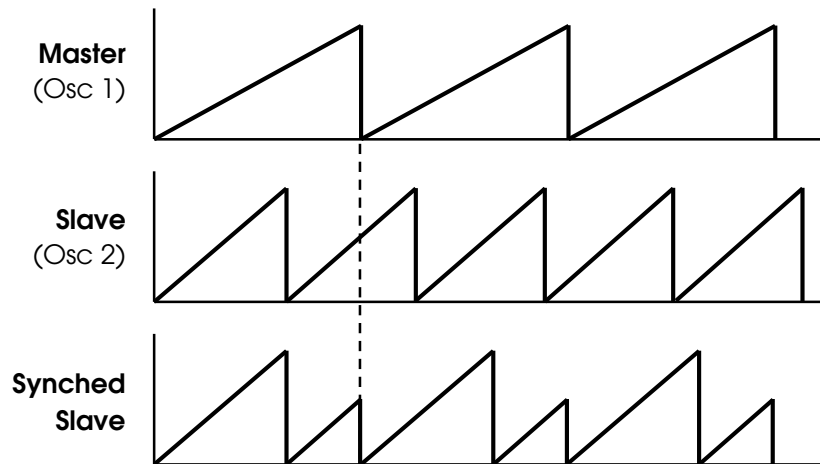
Phase

When the Phase knob is at 0 (Free), the oscillators are 'free-running', meaning their phase is remembered and continue where it left off at each key press. This simulates old analog synthesizer behaviour. From values 1-127 the phase of Oscillator 1 is fixed at 0°, Oscillator 2 phase is shifted forward, and Oscillator 3 phase is shifted backward by the same amount.

Porta

The Portamento knob determines the rate at which the pitch of notes glides from one to the next. At value 0 the portamento is off, and no pitch gliding occurs. The higher the portamento value, the longer the pitch glide will be.

Transp

The Transpose knob offsets the global semitone for all oscillators. Use this to transpose the entire patch down 64 semitones, or 63 semitones up.

Env > Osc 2

This determines how much the filter envelope modulates the pitch of Oscillator 2.

Vel > PW

This parameter sets how much the velocity affects the Pulse Width of the Classic oscillators when the waveforms is a Pulse. If the oscillators are in Multisaw mode, this affects the detune parameter. When the oscillators are either in Grain or Formant mode, this changes the Grain and Formant parameter respectively.

3.3 Oscillator 2 FM

In Viper, Frequency Modulation is the process where the frequency of Oscillator 2 (Carrier) is being modulated by Oscillator 1 (Modulator). One can create complex and interesting timbers, everything from subtle to aggressive sounds using this form of modulation.

Modulator (Osc 1)

Carrier (Osc 2)

FM wave
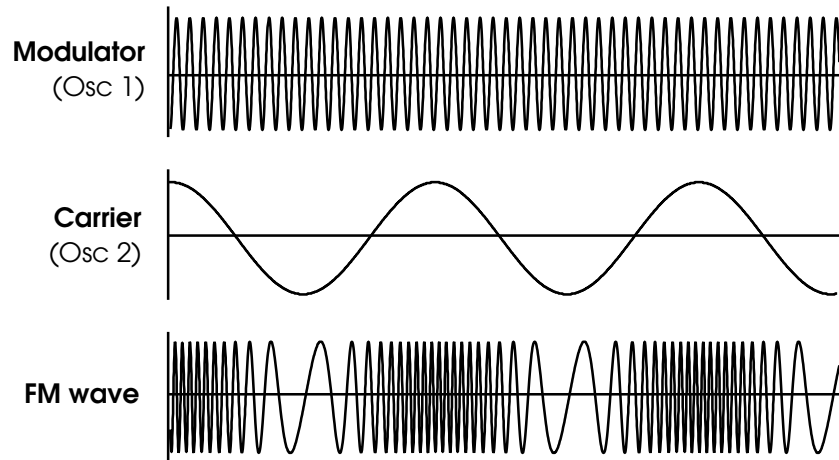
When Oscillator 2 is set as the Classic oscillator, these are the options available for the different FM modes:

FM Mode
Selects the waveform type for the FM Modulation:

- **Pos Triangle (Uni):** A unipolar positive triangle wave generated by Oscillator 1.

- **Triangle (Bi):** A bipolar triangle wave generated by Oscillator 1.

- **Wave (Osc 1):** The waveform coming out from Oscillator 1. (Select a Sine waveform to create typical DX7-FM style sounds.)

- **Noise:** The output of the Noise signal. Note that the Color setting for the Nosie affects the FM as well!

When Oscillator 2 is set as the Multisaw oscillator, the FM mode is disabled in this instance. If Oscillator 2 is either a Wavetable, a Grain Osc, or a Formant Osc, The FM Mode has these options to choose from:

- **Freq Mod:** The output of Oscillator 1 modulates the frequency of Oscillator 2.

- **Phase Mod:** The output of Oscillator 1 modulates the phase of Oscillator 2. Use this to create a different type of timbre from classic frequency modulation.

- **Phase Mod Alt:** Same as the 'Phase Mod', except the range of the modulation is higher.

FM Amt

Sets the amount of Frequency Modulation applied to Oscillator 2.

Env > FM

This determines how much the FM Amount is modulated by the filter envelope.

Vel > FM

Affects how much the velocity controls the FM Amount.

3.4 Mixer

The Mixer is the area where you balance the different sound sources within the oscillator section.

Osc 1/2 Bal

Adjusts the balance between Oscillator 1 and Oscillator 2. At value -64 you only hear Oscillator 1 and at +63 is Oscillator 2 (including FM) only. At value +0, both oscillators are equally balanced.

Osc 3

This adjusts the volume of Oscillator 3. Set the value to 64 to make Oscillator 3 equally balanced to Oscillator 1 and 2.

Sub Osc

Adjusts the volume of the Sub Oscillator. When an oscillator is in Multisaw mode, this crossfades between the Multi Sub oscillators that are one octave lower.

Osc Vol

Determines the overall volume of Oscillator 1, 2, 3 and Sub Oscillator in relation to the Ring Modulator and Noise, just before entering the filters. At value -64 only the Ring Modulator and Noise are audible, at +0 all the oscillators are at maximum volume, and from +0 onward, only the filter saturation amount is affected.

Noise

Sets the volume of the Noise. When the parameter is at 0, the Noise generator is bypassed to save CPU.

Ring Mod

Sets the output level of the Ring Modulator, which multiplies the output of Oscillator 1 (and Sub Osc), and Oscillator 2 to create additional harmonic overtones. This can be seen as one oscillator modulating the other in its amplitude, to create interesting bell-like tones. Try setting both oscillators to a Sine wave, and play with their pitch settings.

Oscillator 1

Oscillator 2

Ring Modulation
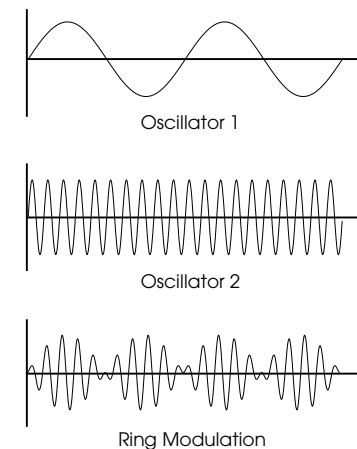
3.5 Amplifier

The Amplifier section is used to manipulate the volume of the patch before the master, and is controlled with the **Volume** knob. One can easily control this parameter with the velocity using the **Vel > Vol** parameter. Positive values make the sound louder, negatives ones make it quieter when increasing the note velocity. The **Pan** control adjusts the sound in the stereo field, and can also be controlled with the velocity with the **Vel > Pan** knob.

3.6 Unison

When Unison is enabled, Viper will play several instances of the same note at the same time. The voices are then stacked up and can be detuned against each other and spread across the stereo field for a much fuller and richer sound.

Note that enabling unison reduces the polyphony by the same factor equal to the unison voices. So if your polyphony is set to 8, and unison is 2, you will have a maximum polyphony of 4 voices.

Voices
Sets the number of voices to be played in unison.

Detune
How much the stacked voices are detuned against each other.

Pan Sprd
The Pan Spread controls the stereo separation between the voices. At value 0, all voices are panned center, at 127 all voices are spread evenly across the stereo field. This parameter is also enabled when the filter routing is set to 'Split' mode, and then affects the stereo separation of the signals from filter 1 and 2.

LFO Phs
This determines the phase offset between the LFOs in each voice. At +0 all LFOs are the same phase for each voice, at positive values the phases are pushed forward, and at negative they are pushed backward.
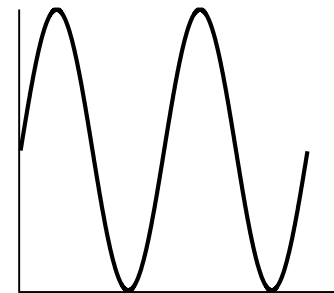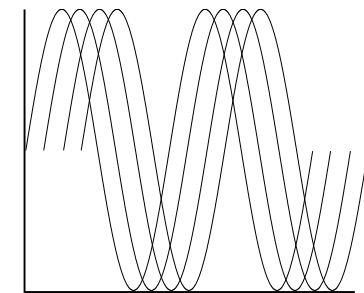
No separation   Positive separation

4. Filter Section

4.1 Filters

One of the most important parts of any synthesizer is the filter. Filters work by removing certain frequencies of the signal, leaving only those that are needed, thus enabling one to drastically alter the tonal quality of the oscillators. This is where the sound becomes organic, and comes alive!

Viper is equipped with two main filters which include several filter types, multiple filter routings and lots of different saturation modes. The frequencies are filtered at the **Cutoff** point, where **Resonance** can be applied to attenuate those frequencies. There are four filter types to choose from, and each behaves differently:

- **Low Pass:** Removes the high frequencies and allows the lower frequencies to pass through.

Magnitude

Frequency

Cutoff point
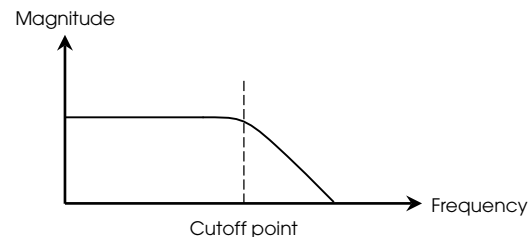
- **High Pass:** The opposite of a low pass, it removes the low frequencies and allows the high ones to pass.

Magnitude

Frequency

Cutoff point
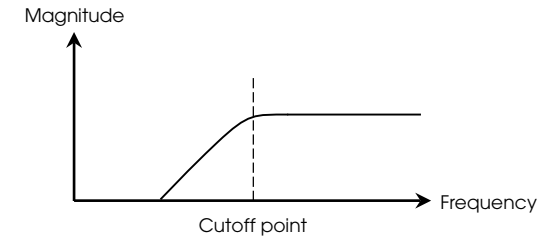
- **Band Pass:** Allows only a narrow band of frequencies to pass through. The resonance determines the width of the band in this case.

Magnitude

Frequency

Cutoff point
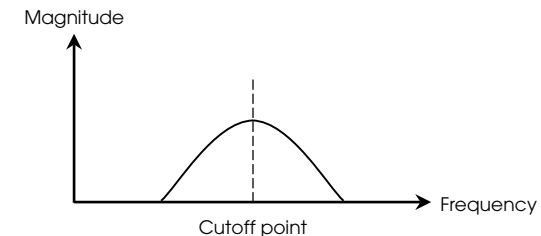
- **Band Stop:** Also called a 'notch' filter, it is the inverse of band pass. It cuts through a band of frequencies.

Magnitude

Frequency

Cutoff point
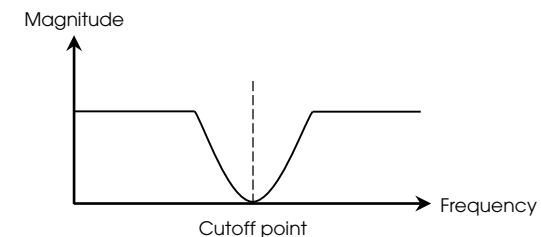
4.2 Filter 1 and 2

The Filters in Viper are designed to be very flexible and provide you with a lot of options to mould and sculpt your sound. Filter 1 and 2 are the same design and use many of the same controls.

Cutoff
Maybe the most important filter parameter is the Cutoff. This sets the frequency where the filter operates, and modulating this parameter with an envelope or an LFO is the key to get a sound to come alive.
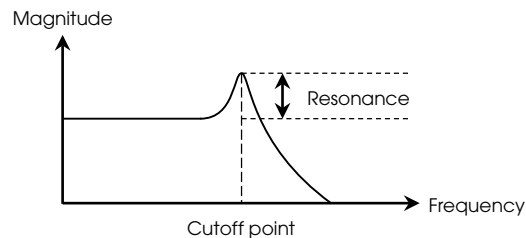
Reso
If the output of the filter is fed back into its input, feedback occurs, and creates a peak at the cutoff point. This is called Resonance. The Resonance knob controls the depth of this effect.

Filter 1/2 Type
Selects the type of filter to be used.

Env Amt
The Envelope Amount parameter determines how much the Filter Envelope affects the cutoff frequency.

Env Pol
The Envelope Polarity button determines which way the Filter Envelope will sweep the cutoff frequency of the filter. Normal polarity means that attack sweeps upwards and decay sweeps downwards. Negative polarity inverts the filter envelope upside-down so that the attack sweeps downwards and decay sweeps upwards.

Key Flw
The Key Follow parameter determines how much the MIDI note pitch affects the cutoff frequency. Set to +0, all notes have the same cutoff frequency as specified by the Cutoff parameter. Positive values will cause the cutoff frequency to increase as you play up the keyboard, negative values will cause it to increase as you play down the keyboard.

Reso > Vel
Controls how much the resonance of the filter is affected by the note velocities.

Env > Vel
Determines to what extent the Env Amt control is affected by the note velocities.

4.3 Filter Common

The Filter Common area contains parameters that allow us to further change the behaviour of the filters and make advanced routings or add saturation to the sound.

Filter Saturation
In the signal chain of Viper, Filter 1 is followed by a saturation stage. This is used to emulate old analog gear where it was possible to push the inputs of the filter beyond its limits causing distortion. Viper has many different saturation types, some just add overtones, whilst others can completely warp and destroy the sound. These distortions are 'polyphonic' which means the distortions are per voice.

- **Off:** No saturation is applied. When turned off, the entire saturation code is disabled to save CPU.

- **Light, Soft, Medium, Hard:** Four different distortion curves each with varying degree of saturation.

- **Digital:** A digital saturation type or 'clipper', which clips the signal peaks after a certain threshold.

- **Wave Shaper:** Sinusoidal waveshaping. Can cause drastic changes to a waveform comparable to Frequency Modulation. This Wave Shaper already starts to take effect at low oscillator volume.

- **Rectifier:** Full wave rectification. Inverts the lower half of the signal, therefore causing distorting.

- **Bit Reducer:** Reduces the bit depth of the signal, which generates digital quantization noise.

- **Rate Reducer:** Reduces the sample rate of the signal, which causes digital aliasing.

- **Rate + Follow:** Same as the Rate Reducer but with Key Follow; the sampling rate follows the played pitch.

- **Low Pass:** A very simple 1 pole low pass filter.

- **Low + Follow:** Same as Low Pass, but with Key Follow; the cutoff frequency follows the played pitch.

- **High Pass:** A very simple 1 pole high pass filter.

- **High + Follow:** Same as High Pass, but with Key Follow; the cutoff frequency follows the played pitch.

- **Analog:** This saturation emulates the distortion of the filters in old analog synthesizers.

Filter Routing

This Filter Routing selector offers several filter routing options which allow you to operate the filters in series, that is patch one after the other in the signal chain, or in parallel, which means side by side in the signal chain.

- **Serial 4:** The filters are routed in series. Both filters are 2 pole, with a 12dB/Oct slope. This results in a 4 pole filter with a 24 dB slope.

- **Serial 6:** The filters are routed in series. Filter 1 is 4 pole, 24 dB/Oct, and Filter 2 is 2 pole, 12 dB/Oct. This results in a 6 pole filter with a 36 dB slope.

- **Parallel:** The filters are routed in parallel. Both filters are 2 pole with a 12 dB/Oct slope. In this configuration, the saturation applied to Filter 1 is not affected by Filter 2.

- **Split:** The filters here are in parallel too, with 2 poles each, 12 dB/Oct, but the difference is that the filters each receive different signals. Oscillator 1 and the Sub Oscillator are routed through Filter 1, while Oscillators 2, 3, Noise and the output of the Ring Modulator are routed through Filter 2. Filter 1 is panned hard left, and Filter 2 is panned hard right. The depth of panning can be controlled via Unison Pan Spread (Oscillator Page).

Serial 4 Mode

Serial 6 Mode

Parallel Mode

Split Mode
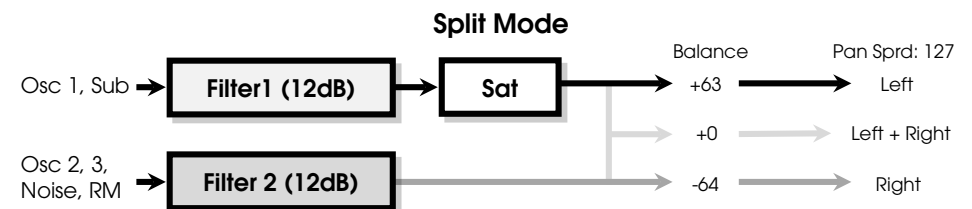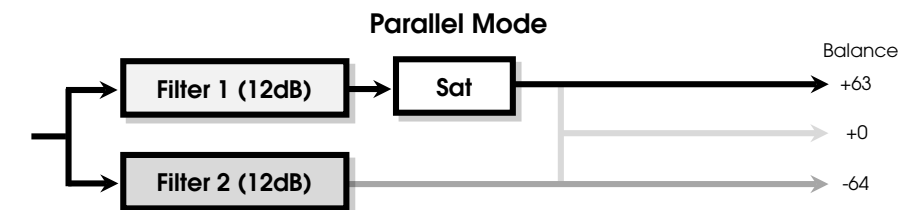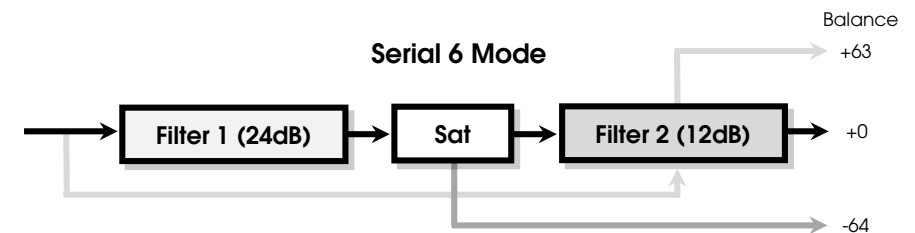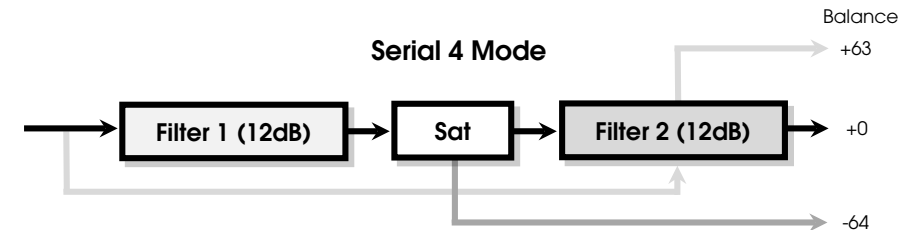
Cutoff Link

The Cutoff Link button synchronises the cutoff frequencies between both filters. When the control is **Off**, the Cutoff parameters are totally independent from each other and control the respective frequencies for Filter 1 and 2. When switched **On**, the upper Cutoff knob also controls Filter 2. The lower Cutoff knob becomes a bipolar (-64 to +63) offset to Filter 1 cutoff, instead of an independent value. At the center position of Cutoff 2, the frequencies of the two filters are identical.

Filter Link

When enabled, Env Amt, Resonance and Key Follow become linked together with their counterparts in Filter 2, so that both filters can be adjusted simultaneously.

Osc Vol

This control is the same from the Oscillator Page in the Mixer. It is shown here to be easily accessible when controlling the filter saturation. Move the knob to positive values (+0 to +63) to increase the filter saturation amount. Saturation is also referred to as 'Drive' in other synths.

Filter Balance

This balances the output levels of the two filters. In Serial 4 and Serial 6 modes, the output of Filter 1 is routed through Filter 2, and then to the amplifier section. If Filter Balance is set to the far left position (-64), you will hear Filter 1 plus Saturation only, whereas Filter 2 is blended into the mix as you rotate the knob towards the center position (+0). When the knob is set to the far right (+63), you will hear Filter 2 exclusively, whereas Filter 1 plus Saturation is blended into the mix as you rotate the knob towards the center. You must leave the Filter Balance in the center position if you want both filters in series (in equal amounts) in the signal path.

In Parallel and Split modes, Filter Balance simply balances the outputs of Filter 1 and Filter 2.

Key Base

This determines the base note for the filter Key Follow, that is, notes below and above this point will be affected by the Key Follow. Key Base affects both filters simultaneously.
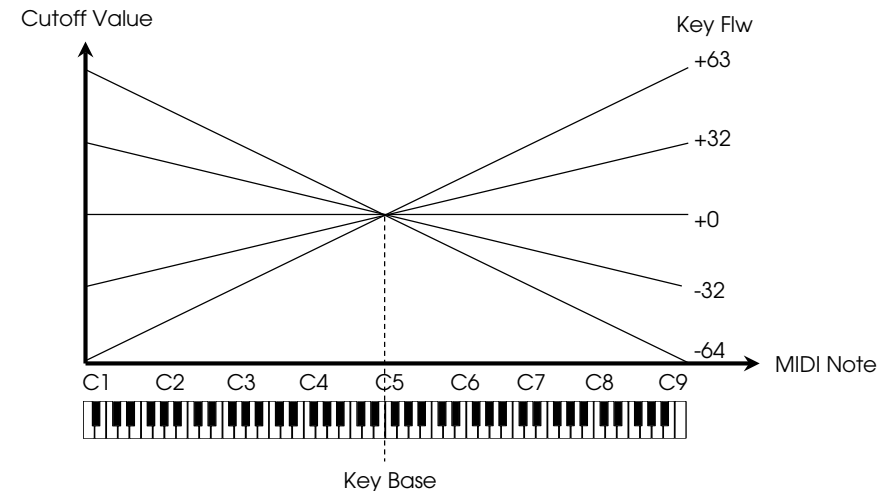
4.4 Envelopes

An envelope is used to control a certain parameter (or parameters) during, and immediately after each note you press. This can be used to make a sound fade in or out, or change its timbre over time.

Viper has four independent envelopes: an amplitude, a filter, and two modulation envelopes. The Amp Envelope controls the volume of a sound, whereas the Filter Envelope controls the filter cutoff over time. Modulation Envelope 3 and 4 are freely assignable to control a parameter in the mod matrix. The Amp and Filter envelopes can also modulate parameters in the matrix. Use the selectors beside the envelope graphs to select the envelope to edit: **F** (Filter), **A** (Amp), **M3** (Modulation 3), **M4** (Modulation 4). You can also click and drag the square handles on the graph to change the parameter values.

All envelopes in Viper can be described by four main stages called **Attack**, **Decay**, **Sustain** and **Release**. The Sustain also has a **Slope** control.

Attack
Determines the amount of time it takes for the envelope to rise to its maximum volume after the start of a note. The higher the Attack value, the longer this rise takes.

Decay
This determines how long it takes for the envelope to decay from its maximum level to the sustain level.

Sustain
This parameter specifies the sustain level that is reached after the decay stage ends. The sustain stage lasts as long as a key is pressed.

Slope
This determines how long the envelope will remain at the sustain level. At +0, the Sustain level remains constant through to the end of the note. At positive values, the envelope will rise back up to maximum. At negative values, the envelope will sweep down to its minimum level.

Release
This determines how long the envelope will take to fall to its minimum level after the key is released.
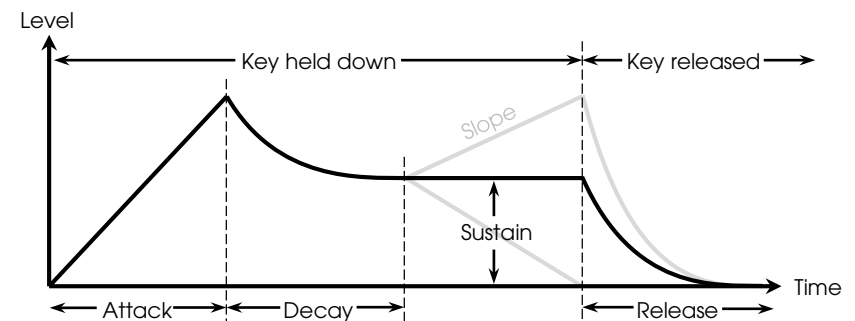
5. LFO Section

5.1 LFOs

LFO is an abbreviation of Low Frequency Oscillator, and is often used to add some cyclic movement to otherwise static sounds. An LFO is similar to a normal oscillator, but it oscillates significantly slower so that its output signal is too low for human hearing and only useful as a modulation source for other parameters. LFOs are used in much the same manner as envelopes, with the major difference that they are repeated indefinitely, so it can modulate the pitch of an oscillator to create a vibrato effect, or modulate the amplitude to create tremolo.

There are three LFOs in Viper, and each one has a different assortment of parameter destinations it can modulate directly. Two of the LFOs are the main ones and they can also be turned into a simple envelope! The third LFO is a simpler one that can be used as an additional modulation source.

LFO Wave
Selects the waveshape of the LFO. Choose between Sine, Triangle, Saw, Pulse, Sample and Hold (stepped random) and Sample and Glide (smooth random).

Shape 1/2
With the shape control for LFO 1 and 2, you can bend or morph the selected LFO waveform.

Shape	-64	+0	+63
Sine			
Triangle			
Saw			
Pulse			
S&H			
S&G			
Wave	Zoom in wave		

LFO Sync
The Sync selector switches the rate of the LFO to be either free from the host clock, or synchronise with the tempo of the host, expressed as a fraction of one bar in 4/4 time.

Rate

Sets the speed at which the LFO will oscillate in Hertz. If LFO Sync is enabled, the rate will set the values in synced rates.

Key Flw

Controls how much the LFO rate is affected by MIDI note number. Key Follow does not apply when LFOs are in Envelope Mode or are synchronized to the clock.

Phase 1/2

Sets the starting point of the wave cycle, and forces it to restart upon pressing a new note. When the phase is set to 0 (Rand), the phase of the LFOs are random.

Fade In (LFO 3)

This determines how long it takes for the LFO to reach full amount. If set to 0, the LFO is instantaneous, if set higher, the LFO will slowly fade up in its amplitude.

← Fade In →
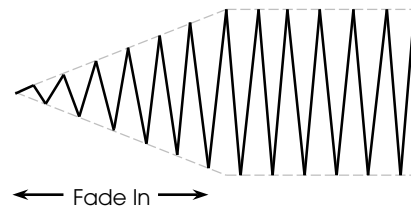
Mono

In polyphonic mode, each of the voices are assigned a dedicated LFO, and oscillate in different phase positions from each other. When Mono is 'On', all voices share a common LFO. This only applies if the parameters Env and Phase are both set to 'Off'.

Env 1/2

When enabled, the LFO will cycle through the waveform only once, then stop. This turns the LFO into an additional envelope which is shaped according to the currently selected waveform. The start of the waveform can be specified with the Phase control.

5.2 LFO Modulations

The LFOs have dedicated controls for modulating various parameters. This allows us to quickly and easily assign different modulations without having to go to the matrix. The controls are bipolar, meaning the LFOs can be inverted when setting the controls in the negative range. There are also buttons for some controls to link them together.

LFO Modulation 1

These are the destinations controlled by LFO 1.

- **Osc1 Pitch:** How much LFO 1 modulates Oscillator 1 pitch.

- **Osc 2 Pitch:** How much LFO 2 modulates Oscillator 2 pitch. Oscillator 3 receives this modulation as well.

- **Pulse Width 1+2:** How much LFO 1 modulates the Pulse Width of both main oscillators at the same time.

- **Resonance:** How much LFO 1 modulates the resonance of both filters at the same time.

- **Filter Gain:** How much LFO 1 modulates the total level of all signals (and thus the subsequent saturation level) before entering the filters.

- **Target:** A freely selectable sound parameter for LFO modulation. For example: select 'Noise Volume' to control the volume of the Noise with this LFO.

- **Amount:** Determines the amount of modulation applied to the parameter selected in the Target selector.

LFO Modulation 2

These are the destinations controlled by LFO 2.

- **Filter 1:** How much LFO 2 modulates the cutoff frequency of Filter 1.

- **Filter 2:** How much LFO 2 modulates the cutoff frequency of Filter 2.

- **Shape 1+2:** How much LFO2 modulates the shapes of oscillators 1 and 2 simultaneously.

- **Pan:** How much LFO 2 modulates the stereo pan position.

- **FM:** How much LFO2 modulates the amount of Frequency Modulation.

- **Target:** A freely selectable sound parameter for LFO modulation. For example: select 'Noise Volume' to control the volume of the Noise with this LFO.

- **Amount:** Determines the amount of modulation applied to the parameter selected in the Target selector.

LFO Modulation 3

LFO 3 does not have dedicated controls for parameter modulation, however it also has a selector like the other LFOs, but instead of showing all the available targets for modulation, you can select: **Osc 1 Pitch**, **Osc 1+2 Pitch**, **Osc 2 Pitch**, **Osc 1 Pulse Width**, **Osc 1+2 Pulse Width** and **Osc 2 Pulse Width**. This does not mean you cannot assign LFO to any other parameter in the Modulation Matrix.

6. Matrix Section

6.1 The Matrix

In order to create more complex sounds we can use a modulation matrix, where you can choose from a set of

sources and link them to almost any parameter. In Viper, 6 sources can be routed to 3 destinations each, allowing for some very complex movement within a sound.

The Sources and Targets are divided into Slots, and every Slot contains 1 Source and 3 Target selectors and 3 Amount controls.

Source
Specifies the modulation source (Envelopes, LFOs etc.) to control a parameter specified by the Target selector. This source can be simultaneously routed to a maximum of three destinations per slot.

Target
Specifies a destination parameter that should be modulated by the selected source.

Amount
Modulation amount for the specified target. The range is bipolar so that modulation can be inverted, and amounts are internally doubled so that unipolar destinations can be modulated over their entire range. The Amount parameter itself can be controlled by another modulation source. The target is for example 'Slot2 Amount1'.

6.2 Matrix Sources

There are several modulation sources to choose from and a source can modulate up to three target parameters in a multitude of ways:

- **Pitch Bend:** Assign the Pitch Bend to modulate other parameters as you move it up and down.

- **Modulation:** Refers to the Modulation control (Mod).

- **Envelopes:** Use any of the Amplitude, Filter, Mod 3 and Mod 4 Envelopes for extra modulation.

- **Bipolar and Unipolar LFOs:** The output of the LFOs can be used in the matrix for controlling parameters other than their dedicated modulation controls. Their output can either be Bipolar or Unipolar. Bipolar values oscillate between -0.5 and 0.5 while Unipolar ones are only in the positive range from 0 to 1.

Bipolar

Unipolar
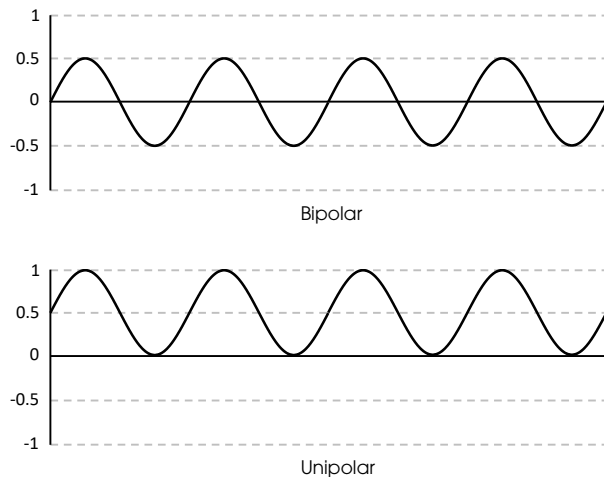
- **Velocity:** Sends values from the MIDI Note velocities. The harder you hit a key, the higher the value it will send.

- **Key Follow:** Follows the values of the Pitch of the MIDI Notes. The higher on the keyboard you play, the higher value it will send.

- **Random:** Sends a random value between 0 and 1, or -1 and 0 if the Target Amount is set to negative.

- **Arp Input:** Sends the step height values from the Arpeggiator in bipolar range (-0.5 to 0.5).

- **Aftertouch:** Sends values based on the pressure applied to MIDI keys.

- **Sustain Pedal:** Sends values from the pressure of the Sustain Pedal.

- **Constant Values:** The controls in Viper use a resolution of 128 steps according to the MIDI specification. Sometimes you may want to tune a parameter so precisely that the value you need is somewhere in between two of these 128 positions. The **1% Constant**, and **10% Constant** provide a way for you to apply a constant offset and fine-tune the target parameter.

- **Arp Sequencer:** Sends the step height values from the Sequencer Steps (0 to 1).

- **Alternating:** Sends alternating values (-0.5 and 0.5) on each key press.

- **Unison Voice:** Sends the Unison voice numbers as values increasing to 1, so each voice can modulate a parameter by different amounts. For example a 3 voice Unison will send 0, 0.5, 1. A 4 voice Unison will be 0, 0.333, 0.666, 1.

7. Arpeggiator Section

7.1 Arp Settings

The Arpeggiator (Arp) module generates a melodic or rhythmic sequence from incoming MIDI notes. It takes a chord and outputs individual notes, and plays them in a cyclic pattern, one after the other. There is also a Sequencer above the Arp that plays in sync. This can also send values to modulate parameters. To enable the Arpeggiator you need to select an order in which way it should play the notes, and this order is defined by the **Mode** Setting.

- **Off:** The Arpeggiator is turned off and all other controls are inactive.

- **UP:** Held notes are arpeggiated in ascending order, starting with the lowest note.

- **Down:** Held notes are arpeggiated in descending order, starting with the highest note.

- **Up Down:** Held notes are arpeggiated in ascending then descending order, starting with the lowest note.

- **As Played:** Held notes are arpeggiated in the sequence that they are played.

- **Random:** Held notes are arpeggiated in a random sequence.

- **Chord:** Plays all held notes as chords in a sequence.

- **Arp Note:** In this mode the velocities of the Arpeggiator transposes the pitch of the oscillators. This way you can create melodic sequences! The velocity of the Arpeggiator steps will be bipolar and control the pitch in semitones. When the value is +0 no transposing occurs, when it is positive it moves it up, when it is negative it moves it down. In this mode, it is a good idea to set the voicing to Mono, to allow the Arpeggiator to play the sequence in a proper order.

- **Arp Out:** The arpeggiator will no longer send any notes, but if you select 'Arp Input' as a modulation source in the Mod Matrix, the velocity values of each step can be used to modulate the selected destination parameter. In this mode the Arp Step length has no effect, all steps reach to the next regardless of their length.

Pattern

Selects one of the rhythmic patterns for the Arpeggiator.

Octave

Determines the transposition range of the arpeggiator in octaves. With Octave set to 1, the arpeggiator only cycles through the currently held notes.

Note Length

Scales the lengths of all notes. The center position leaves the notes at their original length. Positive values lengthen the notes while negative values shorten them relative to their original lengths.

Swing

This control shuffles the position of every other 16th note, backwards by the specified amount. This lends the rhythm a swing feel.

Sync

When enabled, the Arpeggiator rate is synced to the host tempo, otherwise it runs freely.

Sync Rate: The Arpeggiator's rate expressed as a fraction of a bar in 4/4 time.

Rate: When sync is disabled, the Rate knob controls the internal rate of the Arpeggiator.

7.2 Arp Pattern

The Pattern editor is a very powerful feature of the Arpeggiator, allowing you to create your own rhythmic sequences or melodies up to 32 steps long. You can start editing a pattern when you select **'User'** from the Pattern list, or as soon as you edit a sequence, it will switch to being a User pattern.

The Pattern editor contains two rows of numbers, one at the top, and one at the bottom. The top ones are for enabling or disabling a step, making the previous one longer or shorter. The bottom ones are for setting a loop point where we tell the Arpeggiator how many bars it should play, before starting from the beginning.

You can edit a steps length and height by dragging its sides, or the empty area beside or above it.

Drag the loop numbers or press the **<<** and **>>** buttons to shift the steps within the loop point. The **Reset** button will load the default User pattern, while the **Fill**, will copy the looped area to the rest of the pattern. You can also randomize the **Step** or the **Velocity** and the **Length** of the steps of the looped section.

Use the Save icon to save a pattern to **'Viper/Patterns'** for it to appear in the Pattern selector list.

8. Effects Section

Viper offers several effects to further enhance the sound. The order in which they are processed can be seen on the signal flow graph on page 8. Each effect type has an On/Off button to enable or disable the effect.

8.1 Character
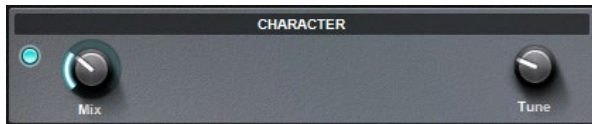
The Character is a bass booster to emulate the bottom end warmth associated with certain analog synthesizers. The **Mix** controls the effect intensity and the **Tune** determines the frequency where the effect occurs.

8.2 Distortion

The Distortion is very similar to the Saturation in the filter section with one major difference: the Saturation affects each voice separately, while the Distortion processes all voices collectively. This makes a huge difference in tone, and this means that the intensity of the distortion is enhanced as you play more notes simultaneously. The **Mix** controls the level of distortion, **Drive** controls the distortion applied to the signal, and the **Hi Cut**, can be used to remove some harshness of the Distortion. The Bit Reducer and Rate Reducer has a **Quality** parameter instead of a Hi Cut to remove some of the high frequencies for a warmer sound.

8.3 Filter Effects

The Filter Effects are a collection of effects based on filters like the All Pass filter. The **Mix** parameter controls the intensity of the effect in all cases.

Ring Modulator
The Ring Modulator multiplies the incoming audio with a sine wave creating combination of the 'sum' and 'difference' frequencies of the two signals. The **Freq** parameters controls the frequency of the sine wave, and the **Phase** controls the phase relationship of the sine wave for the left and right channels, thereby widening the stereo image.

Frequency Shifter

The Frequency Shifter is similar to the Ring Modulator, it uses a sine wave to modulate the output of the synth, but in such a way as to produce only the 'sum' or 'difference' frequencies of the two signals. In contrast to a Pitch Shifter, which preserves the harmonics of the signal, the Frequency Shifter compresses and expands the harmonics. The **Freq** controls is used the tune the frequency of the modulator sine wave. Use the **Shape L/R** parameters to mix the sum and difference frequencies for each channel. If these parameters are in the middle (+0) they will produce the same result as the Ring Modulator.

Vowel Filter

This effect is an easy way to make any sound seem as though it is being vocalized, that is through a vocoder or a talk box. Use the **Freq** controls the sweep through the vowels and make the sound seem as if it talks. The **Reso** is used to reduce or intensify the peaks of the vowel filter. Use the **Phase** knob to offset the position of the vowels between the left and right channels, thereby creating a wide stereo image.

Comb Filter

A Comb Filter is a very short delay that is added back to the signal causing certain frequencies to cancel each other out causing a 'Comb' effect in the spectrum. Use this to create pitched overtones or warp the sound in different ways. Use the **Freq** control to determine the pitch at which the comb filter will resonate. Adjust the **Reso** to reduce or intensify the resonant effect (feedback amount) of the comb filter. Use the **Phase** to offset the frequency of the comb filter between the left and right channels, thereby creating a wide stereo image.
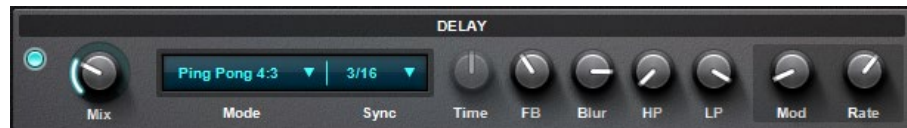
8.4 Phaser

The Phaser creates notches in the frequency spectrum and shifts them up and down with an internal LFO. This result in a sweeping effect typically used in the 60s and 70s on psychedelic music. The **Mix** control crossfades between the dry and wet signal, however, leaving it at the 50% value produces the most noticeable phasing effect. You can select the amount of notches using the **Stage** selector, and vary the size of their bandwidth with the **Spread** parameter. The **FB** (feedback) controls how much the wet signal is fed back into the phaser creating high peaks in the spectrum. The **Freq** knob determines the center frequency about which the phaser will sweep. The **Rate** controls the speed of the internal LFO and the **Mod** controls the amount or range of the sweeping effect.

8.5 Chorus
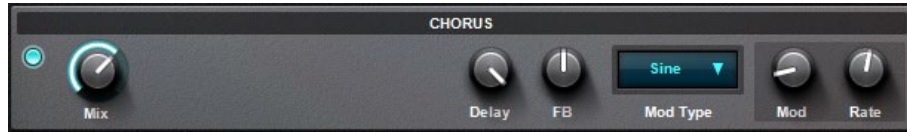
The Chorus works by duplicating the dry signal and modulating its delay time with an LFO to create a detuned effect. This can be used to thicken and add depth to a sound. The **Mix** is used to balance the dry and wet signals, and if set to 100%, one can create a vibrato effect. **Delay** determines the delay time between the original signal and the output of the chorus. Longer times result in a warmer sound, whilst very short times result in a flanging effect. **FB** (feedback) feeds the signal back into the chorus to create highly resonant sounds. **Mod Type** determines the wave shape, while **Rate** adjusts the rate of the internal LFO. **Mod** controls the depth of modulation applied to the chorus delay time.

8.6 Delay

The Delay effect is used to create a series of echoes. You can also modulate the delay time to create phase shifting in the stereo panorama, or create Tape Delay type effects where the delay speeds up or slows down. Delay time can also be locked into sync with the host clock to be matched up to the given song tempo. The Delay also offers several pattern algorithms where the left and right channels are delayed so that the echoes are repeated in different beats.

Use the **Mix** to increase the intensity of the delay. The **Mode** selector switches between different delay algorithms, with varying time between the left and right channels. The Pattern algorithms are always synced to the host. Use the **Sync** selector to choose from synchronized delay times expressed as 4/4 bar divisions. When Sync is 'Off' the **Time** parameter becomes available to set the delay time manually in milliseconds. **FB** (feedback) controls how much of the output signal is fed back into the delay input, affecting the number of repetitions you will hear. **Blur** smears the echoes to create a smoother delay sound (sort of like a mini reverb). **HP** (High Pass) and **LP** (Low Pass) are for filtering and coloring the delays. Use the **Rate** and **Mod** to change the amount and speed of modulation applied to the Delay Time by the delay's own LFO. This can be used to create a much livelier and organic delay, and can add depth and warmth to a sound.

8.7 Reverb

The Reverb effect simulates sound reflections from surrounding walls or objects to create the illusion of a sound being played back in a spatial environment such as a living room, hall or cathedral.

The **Mix** controls the amount of Reverb applied to the signal. The **Pre Delay** is used to delay the room reflections to create an illusion of a wider space, and you can use the **Sync** switch to sync it with the host tempo. The **Shimmer** control creates overtones and sets how much of this is heard. Use this to create otherworldly shimmering or glistening sounds. The **Semi 1** and **Semi 2** knobs set the Shimmer's pitch in semitones. The **Size** parameter changes the size of the reverberation room, it can be anything from a tiny room to a huge hall! The **Warm** control is a filter that cuts off the highs for a less digital sounding reverb, while **HP** and **LP** filters can be used to gradually color the reverb reflections. The **Mod** and **Rate** parameters add modulation to the reverb to smear the sound, and add a chorus effect for extra warmness.

8.8 Equalizer

The Equalizer (EQ) allows you to adjust the sound characteristics of the audio signal. It works by controlling the different frequencies of sound, such as bass, midrange, and treble, to create a desired audio balance. By adjusting the equalizer settings, you can enhance or reduce specific frequency ranges to suit your preferences. For example, increasing the bass frequency can make the sound deeper and more powerful, while reducing the treble can make it less sharp. The EQ in Viper consist of a Low Shelf, a High Shelf and a Peak filter. It is the last effect in the effect chain. The shelf filters (High and Low) are for boosting or cutting bass and/or treble, while the peak filter (Mid) which has a variable bandwidth control, can be used for a wide, or precise midrange equalization.

The **Gain** parameters are bipolar, and used to either boost or cut the selected frequencies. **Freq** controls the frequency where the gain is applied for each filter. **Q** adjusts the width of the peaking filter. Low Q values produce a wide peak, while high values create a narrow peak.

9. Browser Section

9.1 The Browser

Viper has a browser which allows for an easy and fast overview of all the 128 presets that are contained within one soundbank. You can select a preset by simply clicking on the name of the preset, and it will be loaded and highlighted. While the Browser tab is open, the computer **keyboard arrow keys** can also be used to switch between presets. Right clicking on a preset name brings up a list with a few options: **Copy Preset** will copy the selected preset, **Paste Preset**, will paste the copied preset into the selected slot, while **Init Preset** will initialize the preset.

Collecting and Saving your favourite sounds

If one has lots of sounds for Viper, sometimes one wishes to create a new soundbank containing all the different favourite sounds. There is an easy way to do this using the browser: open two Viper instances side by side so that both are visible, and open the Browser tab in each. Load the **Viper Empty Soundset.fxb** from the **Soundbanks/Factory** folder in one of them, and open another soundbank in the

other Viper instance. Right click on a patch that you like, and select **Copy Preset**, then in the Viper with the empty soundbank, right click on an Init preset, and click **Paste Preset.** When you are happy with the collected sounds, simply go to the main menu and select **Save Bank (.fxb)**.