SURGE

User Manual

Revision 5 (v 1.5.0)

Table of Contents

Getting Started

Thank you for choosing SURGE!

This chapter is intended to give you a brief overview to some concepts that are specific to SURGE and a general introduction of the synthesizer.

Installing SURGE

On Mac OS X

On Mac, SURGE is delivered as a Plug-in Instrument for both the Audio Unit (AU) and VST Plug-in interfaces. To use it, a host application compatible with one of the plug-in interfaces is required.

System Requirements:

- Mac OS X 10.3.9 or newer
- An Intel CPU or PowerPC G4/G5 CPU (1 GHz or faster)
- AU or VST-compatible host application

To install, drag the file “SURGE.component” to the “Components” link, “SURGE.vst” to the “VST” link and the directory “Vember Audio” to the “Application Support” link as instructed in the DMG file. This will install SURGE for all users on your computer.

If you wish to install SURGE only for a single user, drag the contents to the following paths instead (~ is your home directory):
SURGE.component -> ~/Library/Audio/Plug-Ins/Components/
SURGE.vst -> ~/Library/Audio/Plug-Ins/VST/
Vember Audio -> ~/Library/Application Support/

Audio Units, AU is a trademark of Apple Computer, Inc
VST is a trademark of Steinberg Media Technologies GmbH

On Windows

On the windows platform, SURGE is delivered as a VST plug-in instrument and need a compatible host application to work.

System Requirements:

- Windows 2000/XP or newer
- A reasonably fast (1 GHz or faster) CPU with SSE support (Pentium 3, Athlon XP or better)
- VST-compatible host application

Make sure you install it in a directory in which your host application will search for VST plug-ins. There is usually a directory named vstplugins created by the host application for this purpose. (see your host application's documentation for more information)

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64-bit version of SURGE (Windows x64)

As of version v1.1.0, SURGE ships as both a 32-bit (x86) and 64-bit (x64) plug-in for the VST/windows platform.

To use the 64-bit version you need the following:

- A CPU supporting the x64 (AMD64/EM64T) instruction set
- A 64-bit OS (like Windows XP x64 edition, Windows 2003 x64 or Vista x64)
- An application capable of hosting 64-bit VST plug-ins

Applications known to support 64-bit VST plug-ins at this time include Plogue Bidule and Cakewalk SONAR.

Upgrading from versions earlier than v1.2.0 (Windows)

Prior to v1.2.0 all patches were stored under a single directory tree named patches (located in vstplugins\surgedata). This directory is now split up into 3 depending on who created the patch.

patches_factory – Patches created in-house by Vember Audio

patches_3rdparty – Patch packs created by users and 3rd parties.

patches_user – This is where patches you store in SURGE end up.

If you upgrade from an earlier version than v1.2.0, SURGE will no longer look in the old “patches” directory. You should move any patches you've created yourself from “patches” to “patches_user”. You should only make this with sounds you've created yourself as an updated version of the factory set and 3rd party patches are installed properly by the installer.

Introduction to the User Interface

The user-interface of SURGE is divided into three main sections: Patch/Global, Scene and FX to reflect what part of the synth they control. Keeping this structure in mind will make it easier to understand the layout.

Illustration 1: The three sections the user-interface of SURGE is divided into.

There are two setups of all controls within the Scene section of the user interface. The state of the Scene-select buttons determine which one of the two Scenes you are currently editing.

About Scenes

Every patch in SURGE contains two scenes (A & B) and an effect-section. A scene is similar to a traditional synthesizer patch as it stores all the information used to synthesize a voice. Since there's two scenes in each patch it's possible to have layered or split sounds stored within a single patch.

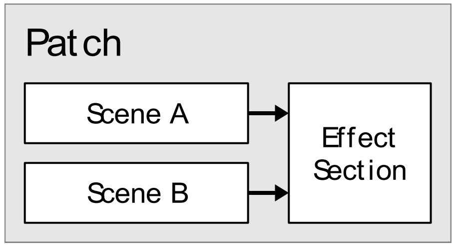


Illustration 2: Both scenes and all effect settings are stored in every patch

Browsing Patches

Browsing patches in SURGE is easy, just press the -/+ buttons until you find something you like. If you click the patch-name field (anywhere in the white area) a menu will list all available patches. A right-click will bring up a menu with just the patches of the current category.

Illustration 3: The patch browser

The patches are grouped into categories, which themselves are grouped into three sections depending on who created them.

- Factory Patches - Patches created in-house by Vember Audio. Categorized by sound type.
- 3rd party Patches - Patch packs created by users and 3rd parties. Categorized by creator.
- User Patches – Your own patches will be stored here. How you categorize them is entirely up to you.

In the drop-down menu there's a division line between the categories of the different sections to indicate the split.

Modulation routing

Modulation routing in SURGE is a bit different than most synthesizers, but it's dead easy. Just select the modulation source you want to use, activate the modulation mode with a second click and drag the slider to the position you want the parameter to be at when fully modulated.

When the modulation mode is active the modulation source flashes (green/blue) and all sliders that can be modulated by the modulation source turn blue. A transparent 'ghost slider' will show the non-modulated position while in modulation mode.

A third click on the modulation source button disengages the modulation mode.

Illustration 4: modulation routing step-by-step.
1-2) Select modulation source by clicking it.
4) Activate modulation mode by clicking it a second time.
5) Modulate-able sliders now appear blue.
6) Drag the slider to the desired position when fully modulated . A “ghost slider” will display the original position.

Those screen-shots are from an older version of SURGE. The newer releases look slightly different.

As entering/leaving the modulation mode is something you will do often there's several ways to activate/deactivate the modulation mode:

- Clicking an already selected modulation source again
- Holding down the alt-key
- Pressing the TAB key
- Pressing the middle, 4^{th} or 5^{th} mouse button. (cursor can be anywhere in the window)

The last three of the alternatives depend on the host application to forward the correct mouse/keyboard-messages to the plug-in. They may not work in all hosts because of this. Whether the middle, 4^{th} and 5^{th} mouse buttons will work is also dependent on how the mouse driver of the operating system is configured.

Keep in mind that although it might seem like the modulations are set to an absolute position they are in fact relative. If you move the slider's non-modulated position the modulated position will move as well.

User Interface Reference

Common UI elements

Sliders

The most common user-interface control in SURGE is the slider. They come in both horizontal and vertical orientations but their functionality is otherwise identical.

Sliders are always dragged, there is no jump if you click on the slider tray instead of the slider head, it enters dragging mode nonetheless.

Slider interaction:

LMB	*Drag slider*
LMB+RMB	*Drag slider (fine)*
Shift+LMB	*Drag slider (fine)*
Shift+LMB+RMB	*Drag slider (ultra-fine)*
LMB double click	*Reset parameter to default value*
RMB	*Context menu*

Right-clicking sliders bring up a context-menu that allows you to clear modulation routings and assign a MIDI controller to the slider.

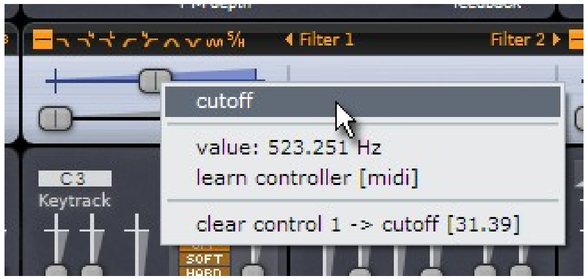


Illustration 5: Slider context menu

The 'learn controller [midi]' command will engage the learning mode. The slider will be assigned to the next controller message received by SURGE. The MIDI-messages recognized are ordinary Channel Controllers as well as RPN/NRPN messages. (Registered Parameter Number/Non Registered Parameter Number)

Some parameters can be have their range extended and/or be synchronized to the host tempo. The options 'extend range' or 'temposync' will show up on the context-menu if they do.

The slider heads give a visual indication whether they can be modulated by the current modulation source when entering the modulation mode (Modulation routing).

Illustration 6: Modulation mode
left) Off, Slider is editing parameter directly.
right) On, Slider is editing the modulation depth from the currently selected modulation source.

The slider tray will have a blue tint if it is modulated by the current modulation source. A half-tint indicates that it is modulated, but not by the currently selected source.

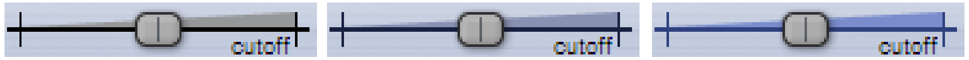


Illustration 7: The amount of blue-tine of slider tray indicates whether the parameter is modulated.

1) Parameter is not modulated

2) Parameter is modulated (half-tint)

3) Parameter by the currently selected modulation source (full tint)

Modulation source buttons

The modulation source buttons have a few additional feats not shown in the introductory modulation chapter. (Modulation routing)

They do change their appearance depending if they're used in the current patch (scene dependent) and will highlight when the mouse is hovering over a destination slider that is modulated by that particular source.

Illustration 8: Modulation sources look different when used

1) Unused modsource

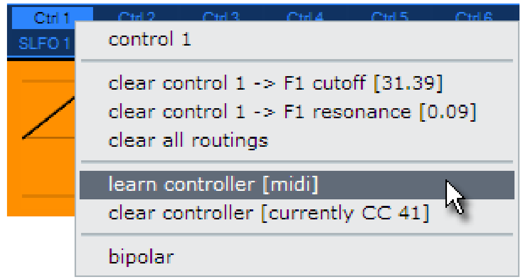
2) Used modsource

3) Modsource that is used by the control the mouse is currently hovering over.

4) Selected modsource

Right-clicking a modulation source button brings up a context-menu that allows you to:

- copy/paste LFO settings (LFO only)
- clear routings to either all destinations or a single destination
- Assign/clear a MIDI controller (CTRL 1-8 only)
- Toggle between bipolar/unipolar (CTRL 1-8 only)
- Rename them (CTRL 1-8 only)



Controller 1-8

What separates these controllers from the rest is that they are assignable by the user to either MIDI CC, RPN or NRPN controllers and their value can be edited on-screen. Choose 'Learn Controller [MIDI]' from the context-menu and it will be assigned to the next MIDI controller received by the synth.

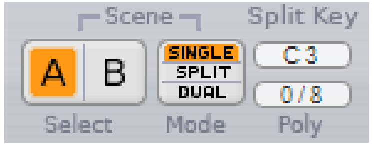
> ***CC*** = *Channel Controller (7-bit)*
>
> ***RPN*** = *Registered Parameter Number (14-bit)*
>
> ***NRPN*** = *Non Registered Parameter Number (14-bit)*
>
> *These are different ways to send controller messages via MIDI. But as Surge will recognize them automatically you just have rotate the knob and Surge will learn it.*

These controllers are stored globally. You can also rename them and choose if their modulation is bipolar (both positive and negative with 0 in the middle) or unipolar (just positive).

Patch/Global section

Scene select/mode

Whether a scene will generate a voice when a key is pressed is determined by the **scene mode** setting:



- **Single** – Notes will be played only by the selected scene.
- **Split** – Notes below the **split-key** will be played by scene A, notes above and including the **split-key** will be played by scene B.
- **Dual** – Notes will be played by both scenes.

Scene select determine which scene is selected for editing and playing (when **scene mode** is set to Single).

> *Right-clicking the **scene select** buttons brings up a context-menu that allows you to copy/paste scenes.*

Poly shows the number of voices currently playing and allows you to set an upper limit to the number of voices allowed to play at the same time. The voice-limiter will kill off excess voices gently to avoid audible artifacts, thus it's not uncommon for the voice count to exceed the limit.

The state of the polyphony limit setting is **not** stored in patches.

Patch browser

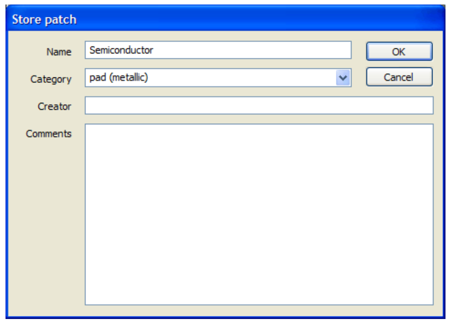
Finding sounds in SURGE is easy, just press the **-/+** buttons until you find something you like. If you left-click the patch-name field (anywhere in the white area) a menu will list all available patches arranged into categories. The categories are further organized into three sections: Factory patches, 3rd party patches and User patches.

A right-click will bring up a menu with just the patches of the current category.

Illustration 9: The patch browser

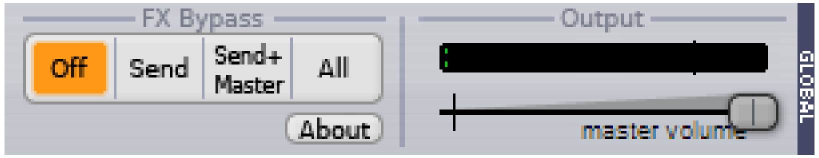
The store dialog

Clicking the store button of the patch browser opens the store dialog. It is where you give patches their name and choose which category they should belong in. You can type new category manually here as well. The patches you store here will end up in the user section on the bottom of the patch menu.



There's also text fields for the name of the patch creator and comments. The comment is not currently shown in the main GUI. (v1.2)

FX-Bypass and Master Volume



FX Bypass lets you quickly hear what a patch sounds like without the effect-units.

- **Off** – All effects are active.
- **Send** – The send effects are disabled.
- **Send + Master** - The send and master-effects are disabled.
- **All** – All effects are disabled.

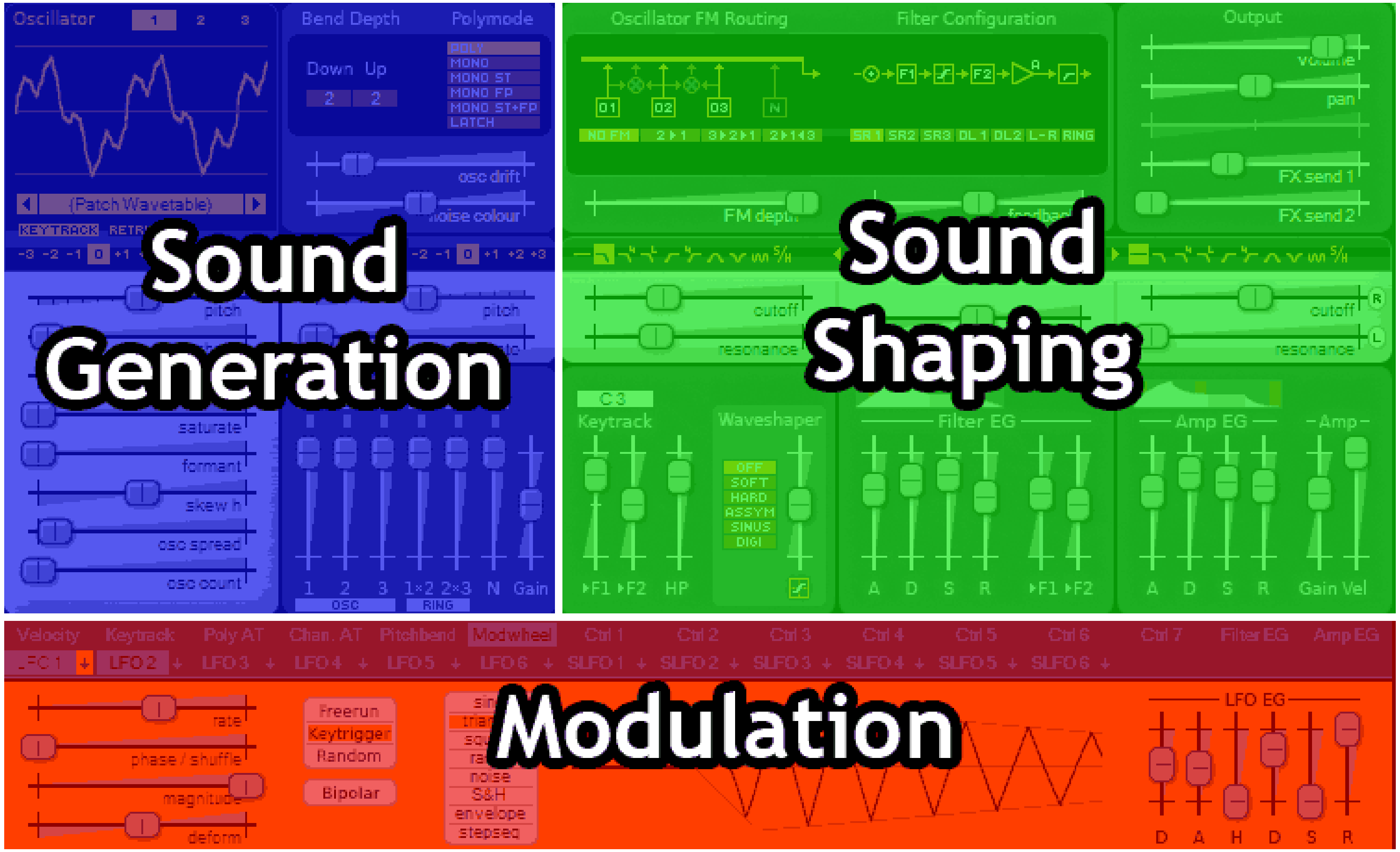
Master volume controls the last gain stage before the output. The VU-meter above it shows the output-level and will become red if it goes above 0 dBFS.

The state of these two settings are **not** stored with patches. They are however stored by the host application in your project files.

Scene section

The UI of the scene section is roughly divided into three parts:

- Sound generation
- Sound shaping
- Modulation



Sound generation

This is where a sound is born. The oscillators generate waveforms according to the notes played, are mixed in the oscillator mixer and the audio is then passed on to the sound-shaping section.

Oscillators

1/2/3-buttons – Chooses the active oscillator for editing.

Display – Shows the active waveform. When the wavetable oscillator is used, it will also work as wavetable picker.

Type – Oscillator type. Chooses which algorithm is used for the oscillator. Available options are classic, sin, wavetable, S&H noise and audio input.

Pitch & octave – Controls the pitch for this particular oscillator. The range of the slider can be extended from its context menu.

Keytrack – When disabled, the oscillator will play the same pitch regardless of the key pressed.

Retrigger – If active, the oscillator will always start immediately at zero phase. This is useful for snappy sounds where you want the attach to sound exactly the same each note.

The rest of the sliders controlling the oscillator is specific to each oscillator type.

Oscillator Mixer

The oscillator mixer has 6 inputs. Each channel has 4 controls.

M – Mute

S – Solo (only play oscillators that have solo active)

Routing (the green box) – Chooses which filter the oscillator is routed to. The middle position (default) will route the output to filter 1 if a serial filterblock configuration is used or both filters for any other configuration.

Slider – Gain control

There is finally an output gain control which affect the level of all the mixer inputs.

Other

Pitch & octave – Controls the pitch for the entire scene. Affects the filter key-tracking and the keytrack modulation source as well. The range of the slider can be extended using the context menu.

Portamento – Portamento is an effect where a new note will slide in pitch from the pitch of the last played note. This setting determine how long the slide will be. A setting of 0 disables Portamento. Can be tempo-synced.

Osc drift – Applies a small amount of instability to the pitch of all oscillators, making them subtly detuned. Although the parameter is shared, the randomness of the instability effect is independent for all oscillators (and eventual unison sub-oscillators).

Noise colour – Affects the frequency spectrum of the noise generator. The middle position results in white noise. Moving the slider to the left emphasizes LF while moving it to the right emphasizes HF.

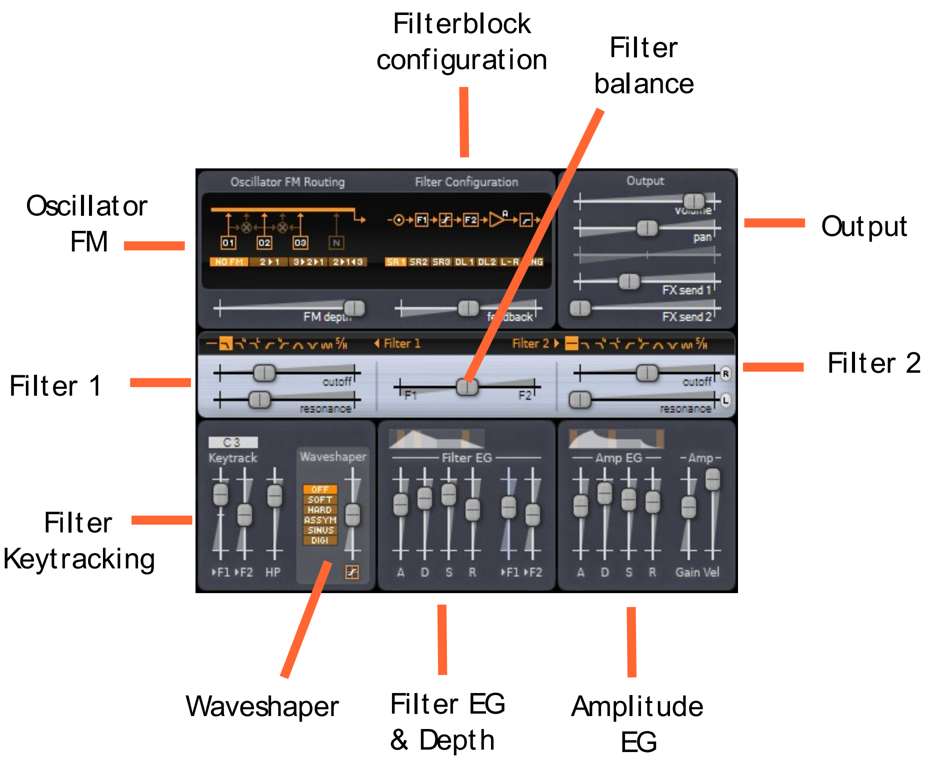
Pitch Bend Down/Up – Pitch Bend Depth. Controls how much the pitch is affected by the pitch bend wheel. (in semitones)

Playmode – Chooses how multiple notes are handled. Poly will allow multiple notes to be played, while Mono will only let the last note play.

Mono has two possible modifiers:

- Single Trigger EG (ST) means that the two envelope generators are not restarted when sliding between two notes (two notes that overlap in time)
- Fingered portamento (FP) means that portamento is only applied when sliding between notes and not when there is time between the played notes.

Sound shaping



Filterblock configuration – Chooses how the filters, waveshaper and the gain stage are connected together.

Feedback – Controls the amount (and polarity) of output that's fed back into the input of the filterblock. It has no effect when using the Serial 1 filterblock configuration (which because of this has a lower CPU load).

Filter balance – Controls how the two filters are mixed. The behavior depend on the filterblock configuration.

Be careful with your monitoring volume when using feedback. It's easy to make really loud high-pitched noises by mistake if you're not familiar with how the synth reacts to feedback.

Don't let this scare you though. There's a lot to be gained from proper and creative use of feedback. Changing the character of filters, making filters interact together, making basic physical models, making sounds that are just about to break apart. It is one of those things that make SURGE special.

Filter controls

Type – Selects the type of the filter. There are 10 choices. Off, 2-pole low-pass, 4-pole low-pass, 4-pole low-pass ladder filter, 2-pole high-pass, 4-pole high-pass, band-pass, notch, comb-filters with both positive and negative polarity and a sample&hold module.

Subtype – Selects variations of each filter type. The difference can vary from subtle to radical depending on how the filter is used. See Filter algorithms in the Technical Reference for information regarding subtypes of each filter type. It is displayed as a number next to the filter type (when available).

Cutoff – Controls the cutoff frequency of the filter.

Cutoff relative switch (small button, filter 2 only) – when active, the cutoff frequency of filter 2 will be set relative to filter 1. This includes any modulations (including the hardwired FEG depth & keytracking).

Resonance – Controls the amount of resonance of the filter.

Resonance link (small button, filter 2 only) – Makes the slider follow filter 1's resonance slider setting.

Keytrack > F1/F2 – Controls how much the pitch of a note affects the cutoff frequency of the filter. A setting of 100% means the filter frequency will follow the pitch harmonically.

Envelope Generators

There are two envelope generators connected to the filterblock. One of them, the Amplitude Envelope Generator (AEG), is hardwired to the gain stage of the filterblock. The other one is hardwired to the two filters, whose depth is set by the **>F1** and **>F2** sliders.

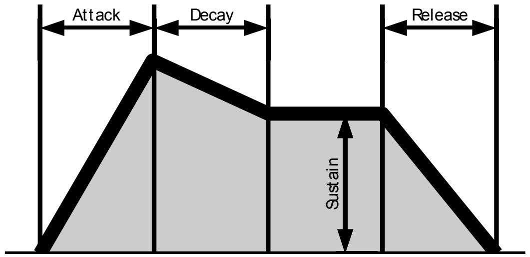


Illustration 10: ADSR envelope structure

The envelope generators are of the 4-stage ADSR type. This is the most common form of EG used in synthesizers and it is named after its four stages **Attack**, **Decay**, **Sustain** and **Release**. If you're new to synthesizer programming the illustration should give you a good idea how they work. The thing you need to remember is that after going through the attack & decay stages the envelope will stick in the sustain stage until the key is released.

Above the envelope stage controls is a graphic representation of the ADSR structure. The orange fields allows you to choose the curvature of the different stages of the envelope.

Other

Keytrack root – Sets the root key of the filter keytracking and the keytrack modulation source. At the root key, the keytrack modulation source will have the value zero. Above/below it it will have positive/negative modulation depending on the distance to the root key in octaves. This parameter does not affect the oscillator pitch.

HP/low-cut – Controls the scene low cut filter. (scene parameter)

FM configuration – Chooses how oscillator FM (frequency modulation) is routed.

FM depth – Sets the depth of the oscillator FM.

Waveshaper type – Chooses type of the non-linear wave-shaping element.

Waveshaper drive – Set the drive amount of the waveshaper.

Amp Gain – Controls the gain element inside the filterblock.

Amp Vel. - Controls how the **Amp Gain** scales with velocity. This is neutral at the maximum position. Other settings provide attenuation at lower velocities, thus this setting will never increase the **Amp Gain** parameter by velocity.

Output stage

The output stage is located after the filterblock in the audio-path. As it's outside the filterblock-structure changing the gain here doesn't have any affect on the timbre of the voice (unlike the previous gain-control which may affect how the feedback and wave-shaping acts). It can still change the timbre of the effect section if non-linear effects (like distortion) are used.

Volume – volume control

Pan – Pan/balance control

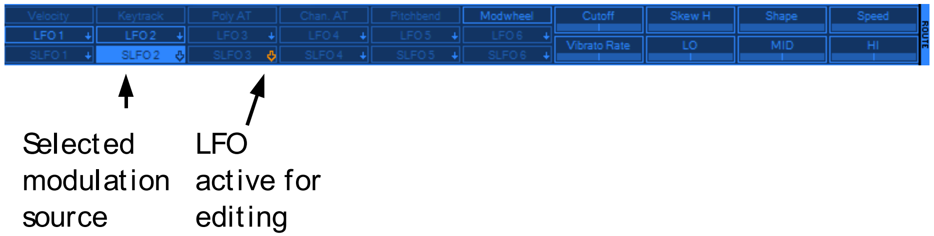
Send 1/2 – Send level to Send effect 1/2. (scene parameter)

Modulation

The modulation section of the scene is different from the sound generation and shaping sections as no audio data is passed through it. Instead it allows you to control the parameters in the other sections from various sources. (see Modulation routing)

Modulation source selection bar

The modulation source selection bar lets you choose which modulation source is selected for modulation routing. It also lets you choose which LFO that are active for editing by using the mini-buttons. When you click the main button of one of the LFOs both the modulation source state and the LFO editor state will be changed.



By using the mini-button next to the main one you can select a different LFO for editing than the modulation source. This lets you modulate the parameters of one LFO with another.

The sub-chapter Modulation source buttons contain more information about how the buttons work.

LFO Overview

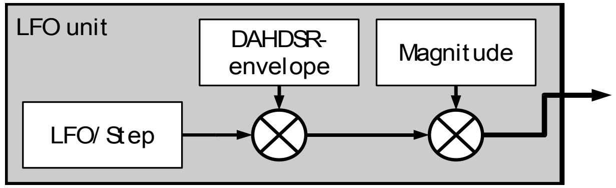


Illustration 11: LFO-unit structure

The LFOs (Low Frequency Oscillator) in surge are very flexible and come with a built in DAHDSR-envelope which lets the LFO work as a dedicated envelope generator or shape the magnitude of the LFO over time.

Parameters

Waveform – Selects the shape of the LFO.

Waveform	*Description*	*Deform action*
Sine	Sine wave	Vertical bend
Triangle	Triangle wave	Vertical bend
Square	Pulse wave	Pulse width
Ramp	Ramp wave (sawtooth)	Vertical bend
Noise	Smooth noise	Correlation
S&H	Step noise	Correlation
Envelope	The LFO waveform output is one, making the LFO-unit as a whole work as an envelope generator.	Envelope shape
Stepseq	The 'stepseq' waveform is a special case that has an additional editor. It can be used to draw waveforms or be used like a step-sequencer.	Smooth/Spikyness

Rate – Controls the rate of the LFO oscillation. When waveform is 'stepseq' 1 step equals the whole cycle. Can be tempo-synced.

Phase/Shuffle - Controls the starting phase of the LFO waveform.

Magnitude – Controls the magnitude of the LFO. This is the parameter you should use if you want to control the depth of an LFO with a controller. (like controlling vibrato depth with the modulation wheel)

Deform – Deform the LFO shape in various ways. The effect varies with the LFO waveform.

Trigger mode (Freerun/Keytrigger/Random) – Chooses how the LFO is triggered when a new note is played.

Unipolar - If active, the LFO-output will be in the [0 .. 1] range. If not [-1 .. 1]

LFO EG

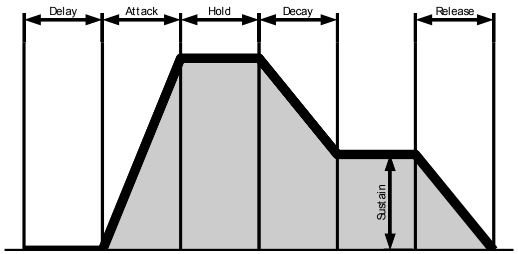


Illustration 12: 6-stage DAHDSR envelope

The LFO Envelope Generators are of the 6-stage DAHDSR type that are multiplied with the waveform generator.

Stepseq

The 'stepseq' waveform is a special case. Instead of the graphical preview there is an editor that allow you to draw the output waveform with up to 16-steps. The two green markers define loop-points that the LFO will repeat once it gets into the loop. The left mouse button is used for drawing while the right one can be used to clear the values to 0. Holding down shift while drawing will quantitize the values to 1/12th steps, hence if the LFO is used to modulate pitch by an octave, each step will represent a semitone.

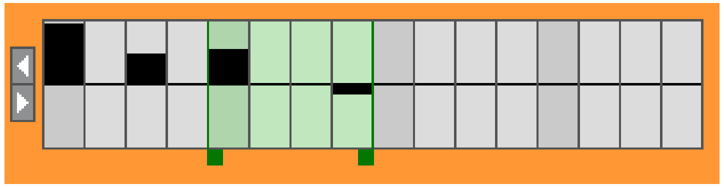


Illustration 13: Stepseq editor

The step-sequencer of Voice LFO 1 has an extra pane at the top of the step-editor that will retrigger the two regular envelopes of the voice (AEG and FEG) at each step if it is checked (black) at that particular step.

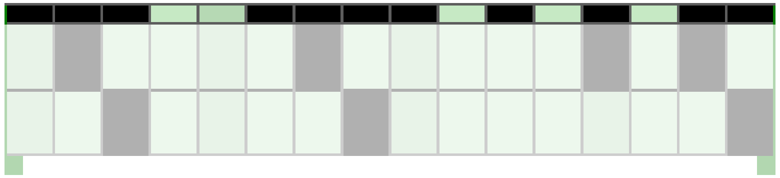


Illustration 14: Envelope retrigger pane of Voice LFO 1

The deform parameter give this waveform a lot of flexibility. A value of 0% will output the steps just as they look on the editor. Negative values will give an increasingly spiky waveform while positive values will make the output smoother.

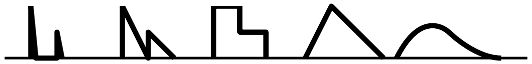
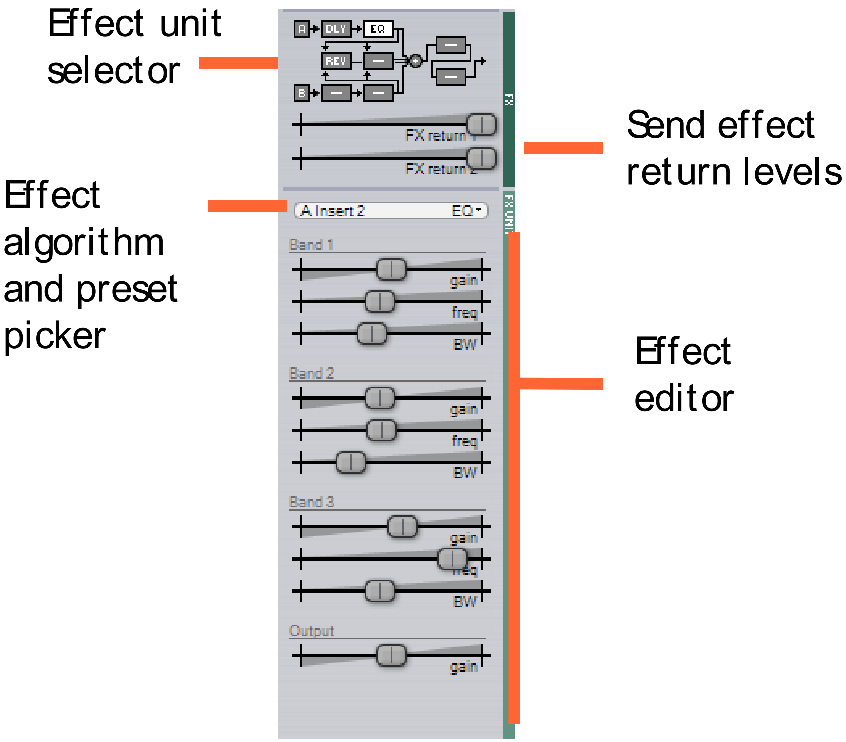


Illustration 15: Effect of the deform parameter on the stepseq waveform

FX section

The FX section lets you control the 8 effect units of the effect block stored in every patch.



The effect unit selector chooses which effect unit is active in the effect editor. A right-click disables/enables that particular unit (this setting is stored within patches unlike the global FX bypass setting).

The effect algorithm/preset-picker lets you assign an effect to the unit selected in the effect unit selector. The effect is assigned by selecting one of the preset settings for that effect from the menu. You can also save your own effect presets which will be stored globally with the synth.

Technical Reference

SURGE Hierarchy

Overview

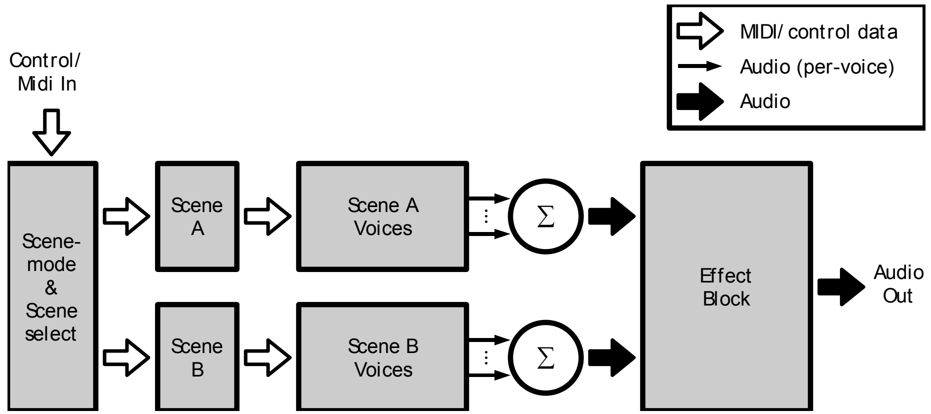


Illustration 16: Block diagram of the synthesizer engine.

Illustration 16 shows an overview of the synthesizer engine of SURGE.

Voices

Illustration 17 shows most audio and control-paths of a single voice. Not all processing elements of the voice are shown in the diagram.

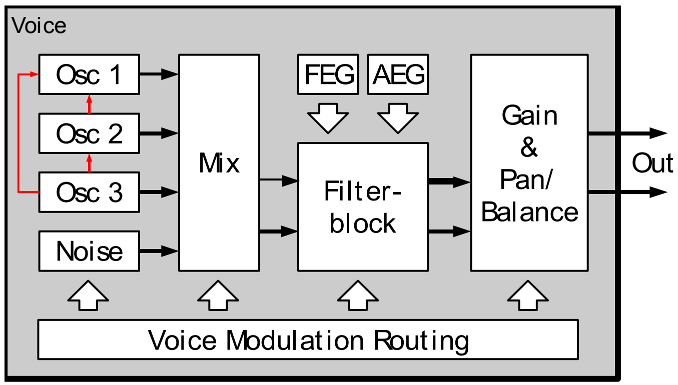
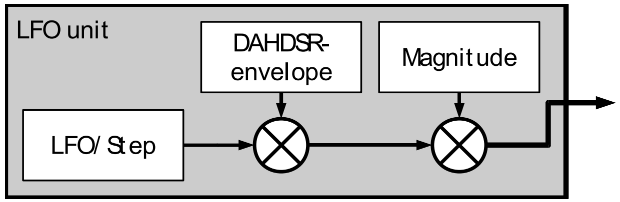


Illustration 17: Block diagram of a synthesizer voice

LFOs

Each voice has 6 modulation source called LFOs (Low Frequency Oscillator) that you can use for modulation purposes. Each scene has an additional 6 LFOs making each voice capable of receiving modulation from a total of 12 LFOs.

Calling them LFOs is a great understatement as they have an integrated envelope generator and can function as a 16-step waveform-generator as well.



More information about the LFOs in the UI reference. ([LFO Overview](#))

The effect block

SURGE has 8 effect units, arranged into an 'effect block'.

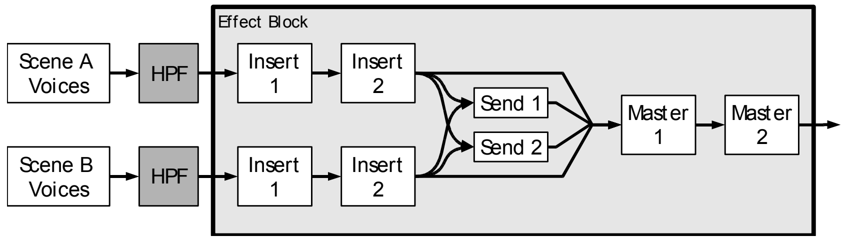


Illustration 18: The effect block

See the chapter [FX section](#) for more information.

Modulation routing in-depth

How the modulation routing works internally isn't something you normally have to think about when using SURGE. Just activate the modulation mode with the desired source and see which of the sliders that become blue. Nonetheless, it is useful to know which limitations are present and why.

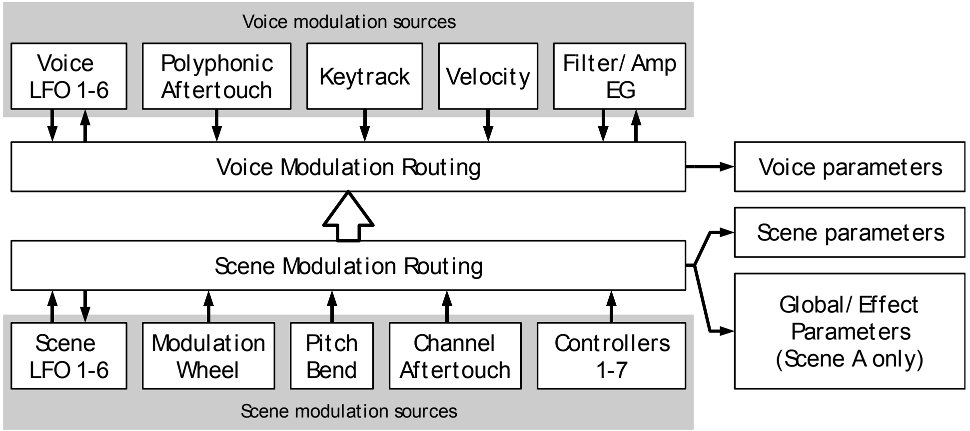


Illustration 19: Modulation routing behind the scenes

The thing to remember is that the voice modulation sources can't modulate the scene parameters, global/effect parameters or the scene LFOs. Other that that it should be pretty straightforward.

Oscillator algorithms

SURGE provide 5 different oscillator algorithms. Each capable of generating sound in different ways with a different set of controls. They're not just different waveforms.

Classic

The classic oscillator algorithm consists of a main oscillator that can generate a pulse wave, a sawtooth wave, a dual-saw wave or anything in between.

A sub-oscillator provide a pulse-wave one octave below the main oscillator. Changing the pulse-width of the sub-oscillator does affect the main oscillator as well, as they will both change levels at the same time except that the main oscillator does it twice as often.

The classic algorithm is also capable of oscillator self-sync. Note that the sub-oscillator will be used as the base-pitch for the sync.

The algorithm provides unison at the oscillator-level with up to 16 instances. Unlike the wavetable-oscillator the cost of unison in terms of CPU usage for the classic oscillator is quite modest. The unison oscillator-instances are affected by the scene-level Osc-Drift parameter independently.

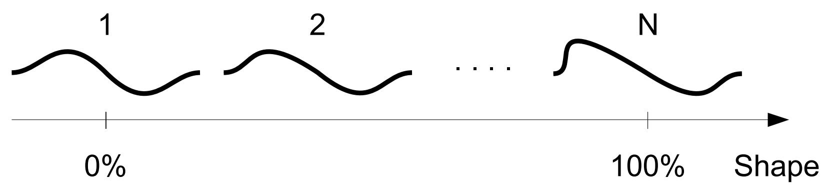
Name	*Description*	*Range*
Shape	Waveform shape -100% = pulse, 0% = saw, 100% = dual saw	-100 .. 100 %
Width	Pulse-width (pulse) or relative phase (dual saw)	0 .. 100 %
Sub-width	Pulse-width of sub-oscillator.	0 .. 100 %
Sub-level	Sub-oscillator mix. 0% = only main, 100% = only sub	0 .. 100 %
Sync	Oscillator self-sync	0..60 semitones
Osc-spread	Detuning of unison oscillators. 100% = 1 semitone in both directions Can be switched between relative (default) and absolute using the context-menu of the slider.	0 .. 100% 0..16Hz
Osc-count	Number of oscillators used for unison. 1 = disabled	1 .. 16

Sinus

The sinus oscillator algorithm generates a simple sine-wave. It has no non-standard parameters.

Wavetable

A wavetable in SURGE consists of up to 1024 single-cycle waveforms. Using the Shape parameter it is possible to sweep across the waveforms in the wavetable.



The individual waves are equidistant in the table. When the shape setting is between two individual waves they will be mixed to ensure smooth travel. You can't edit the wavetable contents directly within SURGE but it is possible to generate custom wavetables with external software.

By modulating the shape parameter it is possible to create motion, dynamic response to playing or just sonic variation. What real-life property, if any, the shape parameter is supposed to mirror depend on each wavetable. Common cases are:

- Analyzed from sounds that evolve over time. The behavior can be recreated by letting shape increase over time by modulation. It's the most common among the analyzed wavetables.
- Analyzed from static sounds over different pitches to capture the formant shift of a sound. The behavior can be recreated by modulating shape by the keytrack modsource.

- A parameter of a mathematical equation.

In the end it's just a set of data and SURGE doesn't care how it was generated, all that matters is how it sounds.

The wave-table oscillator has some interesting sonic characteristics. It outputs the waveform in a stair-stepped fashion, making no attempts to 'smooth the steps' in the process, but does so in a manner that is completely band-limited. This makes it similar in sound to 1980s era wave-table synths and samplers which didn't use resampling but had dedicated D/A-converters for each voice instead and changed the pitch by varying the sample rate of the individual D/As.

The fact that the steps aren't smoothed causes an artifact known as harmonic aliasing. This is not to be confused with inharmonic aliasing which sounds somewhat similar to an AM-radio being tuned and is generally nasty. Instead, this artifact will cause the harmonics of the waveform to repeat themselves and fill up the entire audible spectra even at low pitches, just like a square-wave would, preventing the waveform from sounding dull. As this artifact is completely harmonic it is also musically pleasing. Nonetheless, it may sound a bit out of place on very smooth waveforms but the effect can be filtered out by a lowpass-filter in the filterblock if desired. Some of the wave-tables, such as the regular triangle wave, are large enough for this artifact to never appear in the normally used range for this specific reason.

The important thing is that it, just like the other oscillators in SURGE, doesn't output any inharmonic aliasing whatsoever or any audible levels of interpolation-noise, two artifacts which has played a big part in giving digital synthesizers a bad name.

For developers & advanced users:

There is a reference for the .wt file-format used by the wavetables. It is located in:

surgedata/wavetables/wt fileformat.txt

Name	***Description***	***Range***
Shape	Waveform shape. 0% = first, 100% = last	0 .. 100 %
Skew V	Vertical skew of the waveform	-100 .. 100 %
Saturate	Soft saturation of the waveform	0 .. 100 %
Formant	Compresses the waveform in time but keeps the cycle-time intact	0..60 semitones
Skew H	Horizontal skew of the waveform	-100 .. 100 %
Osc-spread	Detuning of unison oscillators. 100% = 1 semitone in both directions Can be switched between relative (default) and absolute using the context-menu of the slider.	0 .. 100% 0..16Hz
Osc-count	Number of oscillators used for unison. 1 = disabled	1 .. 7

Window

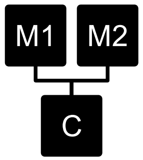
The window oscillator (added in v1.5) is another shot at wavetable synthesis that is quite different from the previous wavetable algorithm.

The wave, which can be any waveform included with Surge, is multiplied by a second waveform, the window, which can be one of 9 waveform types that are specifically made for the window oscillator. The formant parameter controls the pitch of the wave independently of the window, but as the wave is always restarted when the window is the pitch will stay the same. Instead, the timbre of the sound will change dramatically, much depending on which window is selected.

Unlike the wavetable algorithm, the window oscillator uses a more traditional resampling approach which doesn't result in harmonic aliasing. Obviously, being part of a Vember Audio product, the sound quality is still top-notch.

Name	Description	Range
Shape	Waveform shape. 0% = first, 100% = last (doesn't interpolate)	0 .. 100 %
Formant	Pitch of the wave independently of the window	-60 .. 60 semitones
Window	Chooses the window waveform.	-
Osc-spread	Detuning of unison oscillators. 100% = 1 semitone in both directions Can be switched between relative (default) and absolute using the context-menu of the slider.	0 .. 100% 0..16Hz
Osc-count	Number of oscillators used for unison. 1 = disabled	1 .. 7

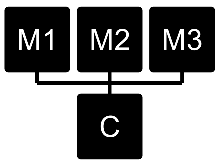
FM2



FM2 provides a miniature FM-synthesizer voice in an oscillator that is specifically tailored towards making nice and musical FM sounds. A single sine carrier is modulated by two sine modulators, whose ratios to the carrier are always integer thus the resulting waveform is always cyclic. However, “Mx Shift” lets you offset the modulators slightly in an absolute fashion, creating an evolving and pleasing detune effect.

Name	Description	Range
M1 Amount	Modulation amount of the first modulator	0 .. 100 %
M1 Ratio	Ratio of the first modulator to the carrier	1 .. 32
M2 Amount	Modulation amount of the second modulator	0 .. 100 %
M2 Ratio	Ratio of the second modulator to the carrier	1 .. 32
Mx Shift	Absolute detuning of the modulators	-10 .. 10 Hz
Mx Start Phase	Changes the initial phase of the modulators to give you different variations of the waveform.	0 .. 100 %
Feedback	Modulation amount of the carrier to itself	0 .. 100 %

FM3



As a contrast to FM2, FM3 is the algorithm of choice for scraping paint of walls. The modulators have a larger range, the ratios can be non-integer and there's a third modulator which has its rate set as an absolute frequency.

Name	Description	Range
M1 Amount	Modulation amount of the first modulator	0 .. 100 %
M1 Ratio	Ratio of the first modulator to the carrier	0.0 .. 32.0
M2 Amount	Modulation amount of the second modulator	0 .. 100 %
M2 Ratio	Ratio of the second modulator to the carrier	0.0 .. 32.0
M3 Amount	Modulation amount of the third modulator	0 .. 100 %
M3 Ratio	Frequency of the third modulator	14Hz .. 25kHz
Feedback	Modulation amount of the carrier to itself	0 .. 100 %

S&H-Noise

S&H is an abbreviation for 'Sample and Hold'.

The S&H-Noise oscillator algorithm works like a pulse oscillator, but instead of always switching between +1 and -1 the levels used are determined stochastically.

The correlation parameter determine how new levels are calculated. A setting of 0% will have no memory and each new level will effectively be a random number (white noise). A lower setting will favor new values that is closer to the previous level and will provide a noise with a darker spectra. Higher values will favor values as far away from the previous one as possible, with 100% resulting in a harmonic pulse-wave.

Name	Description	Range
Correlation	Noise correlation. 0% = white noise, 100% = pulse	-100 .. 100 %
Width	Pulse-width (pulse)	0 .. 100 %
Sync	Oscillator self-sync	0..60 semitones
Osc-spread	Detuning of unison oscillators. 100% = 1 semitone in both directions Can be switched between relative (default) and absolute using the context-menu of the slider.	0 .. 100% 0..16Hz
Osc-count	Number of oscillators used for unison. 1 = disabled	1 .. 16

Audio Input

Audio Input lets you route external audio into the voice-architecture of SURGE.

Name	*Description*	*Range*
Input	Chooses which input is used. -100% = left, 0% = both, 100% = right	-100 .. 100 %
Gain	Input gain in dB.	-48 .. +48 dB

Some problematic VST host applications will refuse to feed instrument plug-ins with audio input unless they are configured as a regular effect. Making a copy of the file “surge.dll” named “surge_fx.dll” in the same directory will cause that copy of SURGE to identify itself as an effect instead of an instrument which will make it work in such hosts.

Filter algorithms

There are 9 filter algorithms available (+ off) for each of the 2 filter units in the filterblock. Each of the algorithms have different subtypes, which alter their sound.

Most of the filter-(sub)types have some non-linear elements in them to allow them to self-oscillate in a stable and predictable manner. This means they will sound different depending on how hard they're driven, which can be conveniently controlled with the Pre-Filter Gain setting. For example, if the resonance peaks of a filter is too loud, increase the Pre-Filter Gain to make the rest of the signal more dominant (and if needed decrease the gain at the output stage of the voice to compensate).

Subtypes for LP12/LP24/HP12/HP24/BP

Depending on the setting of the subtype switch, the characteristics and behavior of these filters will be altered, although their main purpose remains the same.

Subtype	*Description*
1	Clean with a strong resonance, capable of self-oscillation. Handles transient behavior extremely well. (default)
2	Chesty, somewhat distorted sound with a more held-back resonance. Capable of self-oscillation. (default in v1.2.2)
3	The smoothest subtype, capable of lower resonance than the others, which is suitable when you do not want the sound of the filter to be noticed but only to roll-off a part of the spectrum.

LP12

2-Pole Low-Pass filter.

LP24

4-Pole Low-Pass filter.

HP12

2-Pole High-Pass filter.

HP24

4-Pole High-Pass filter.

BP

2-Pole Band-Pass filter.

LP24L

4-Pole Low-Pass ladder filter. You can select at which stage (1-4) the signal is output using the sub-type control. Has stable self-oscillation.

Notch

2-Pole Band-Reject filter.

Subtype	*Description*
1	Default subtype
2	Included for compatibility with v1.2.0 (smaller resonance range)

Comb

Delay-Based Comb filter.

Subtype	*Description*
1	Positive feedback, 50% dry/wet mix
2	Positive feedback, 100% wet mix
3	Negative feedback, 50% dry/wet mix
4	Negative feedback, 100% wet mix

When the sub-type is set to 2 (or 4) and resonance is 0% the comb-filter will work purely as a delay-unit (with sub-sample precision). This can be used together with the other filter-unit along with filterblock feedback to provide interesting options. The “wind/clarinet” and “pluck (fast)/simple waveguide" presets showcase how this ability can be used for simple physical modeling. They only use the oscillator section to ignite the sound, the rest is in the filterblock.

Sample & Hold

Sample & Hold module. Will sample the audio at the rate set by the cutoff-frequency. Resonance will emphasize oscillations around the cutoff frequency, not unlike the resonance peak of a lowpass-filter.

Effect algorithms

SURGE has 8 effect units which each can run one of the 9 provided algorithms.

Delay

The delay algorithm in SURGE is very versatile and can work well both as an echo/delay- and chorus-effect.

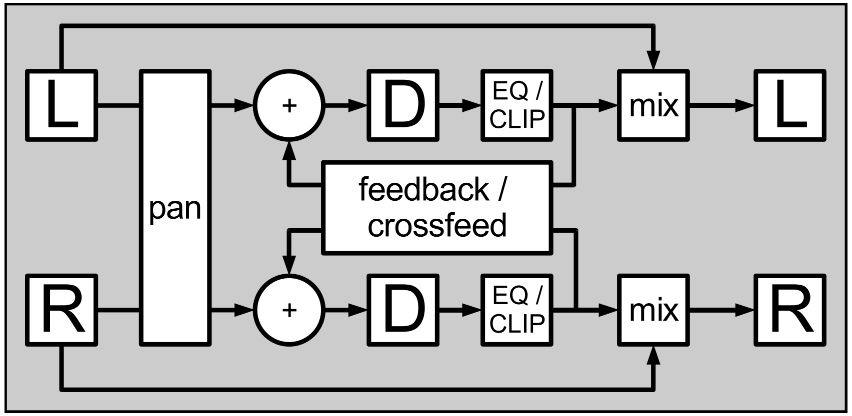


Illustration 20: Delay algorithm block diagram

There is an LFO connected to the delay-lines (not shown in diagram) which can provide stereo-widening/detuning of the delay-line.

Name	*Description*	*Range*
Pan	Routes the two channels to the delay-units by panning. The gain of the input-channels remain unaffected, it's only their stereo location that changes. (a sound only heard in the left channel will still be heard when pan is set to 100% here, but only in the right channel.)	-100 .. 100 %
Delay time L/R	Delay time for the two channels. Can be tempo-synced.	0.004 .. 32 s 1/512 .. 16 bars
Feedback	Amount fed from the channel to its own input	-inf .. 0 dB
Crossfeed	Amount fed from the channel to the input of the opposing channel	-inf .. 0 dB
Low/High-cut	EQ controls of the delayed signal	14Hz .. 25kHz
Modulation rate	Rate of the modulation LFO (triangle). This parameter is inexact due to implementation.	0.008..1024 Hz
Modulation depth	Indirect control of the modulation LFO depth. The effect adjust the depth to match the detuning in cents set here.	0 .. 200 cents
Mix	Blend control between the dry and the wet signal. 0% = 100% dry, 0% wet 100% = 0% dry, 100% wet	0 .. 100 %
Width	Gain scaling of the Side-component of the wet signal	-24 .. 24 dB

Reverb

The reverberation algorithm simulates room acoustics and is suitable both at adding ambience to sounds and creating special effects.

Name	*Description*	*Range*
Pre-delay	The amount of delay applied to the signal before it is fed to the reverberation unit. Can be tempo-synced.	0.004 .. 32 s 1/512 .. 16 bars
Room-shape	Selects between 4 room shapes that has different sounds. (changing this parameter will interrupt the signal)	0 .. 3
Size	Changes the apparent size of the simulated room. (changing this parameter will interrupt the signal)	0 .. 100 %
Decay time	The time it takes for the reverberation to ring-out. (-60 dB)	0.063 .. 64 s
HF-damp	Amount of HF damping applied to the signal inside the reverberator.	0 .. 100 %
Low cut, Band1 freq/gain, High cut	Post-reverb equalizer controls.	
Mix	Blend control between the dry and the wet signal.	0 .. 100 %
Width	Gain scaling of the Side-component of the wet signal	-24 .. 24 dB

Chorus

4-stage chorus algorithm.

Name	Description	Range
Time	Delay time used as chorus mid-point.	0 .. 1/8 s
Mod rate	Rate of modulation LFO. Can be tempo-synced.	0.008..1024 Hz 64..1/2048 bar
Mod depth	Depth of modulation LFO	0 .. 100 %
Feedback	Amount fed from the output back into the input	-inf .. 0 dB
Low/High-cut	EQ controls of the chorused signal	14Hz .. 25kHz
Mix	Blend control between the dry and the wet signal.	0 .. 100 %
Width	Gain scaling of the Side-component of the wet signal	-24 .. 24 dB

Phaser

4-stage phaser algorithm.

Name	Description	Range
Base freq	Base frequency for all the stages	-100 .. 100 %
Feedback	Feedback of the phaser	-100 .. 100 %
Q	Q setting for the stages	-100 .. 100 %
Rate	Rate of modulation LFO. Can be tempo-synced.	0.008..1024 Hz 64..1/2048 bar
Depth	Depth of modulation LFO	0 .. 100 %
Stereo	LFO Phase relation between stereo channels 0% = 0 degrees, 100% = 180 degrees	0 .. 100 %
Mix	Blend control between the dry and the wet signal.	0 .. 100 %

Rotary Speaker

Rotary speaker simulator algorithm.

Name	Description	Range
Horn rate	Rate of HF horn rotation. The LF horn is a lower multiple of this rate. Can be tempo-synced.	0.008..1024 Hz 64..1/2048 bar
Doppler depth	The amount of Doppler shift used in the simulation. (vibrato)	0 .. 100 %
Ampmod depth	The amount of amplitude modulation used in the simulation. (tremolo)	0 .. 100 %

Distortion

Distortion algorithm. Provides plenty of EQ options as well as a feedback loop to alter the tonality of the clipping stage.

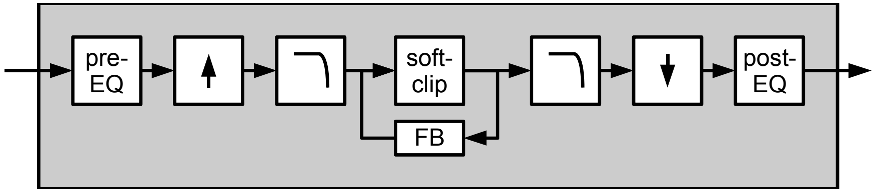


Illustration 21: Distortion algorithm block diagram

Name	Description	Range
Pre-EQ gain/freq/BW	Parametric EQ band prior to the clipping stage	
Pre-EQ High cut	High cut element prior to the clipping stage	14Hz .. 25kHz
Drive	Drive of the clipping stage	-24 .. +24 dB
Feedback	Feedback loop around the clipping stage	-100 .. 100 %
Post-EQ gain/freq/BW	Parametric EQ band after the clipping stage	
Post-EQ High cut	High cut element prior to the clipping stage	14Hz .. 25kHz
Output gain	Output gain	-24 .. +24 dB

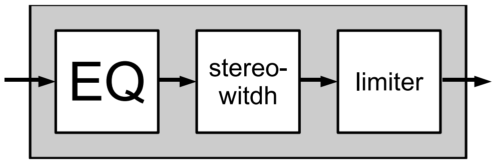
EQ

The EQ unit provide 3-bands of fully parametric equalizing. This high-quality algorithm has a much better response at high frequencies than digital equalizers usually have.

Name	*Description*	*Range*
Band 1/2/3 Gain	Band gain	-48 .. +48 dB
Band 1/2/3 Freq	Band frequency	14Hz .. 25kHz
Band 1/2/3 Bandwidth	Band bandwidth	0 .. 5 octaves
Output gain	Gain control	-48 .. +48 dB

Conditioner

The conditioner is a simple EQ, stereo image control and a limiter built into one unit. The limiter applies make-up gain automatically.



Name	*Description*	*Range*
Bass	LF boost/cut	-12 .. +12 dB
Treble	HF boost/cut	-12 .. +12 dB
Width	Stereo width. 0% = mono, 100% = stereo, -100% = reverse stereo	-100 .. 100 %
Balance	Stereo balance	-100 .. 100 %
Threshold	Limiter threshold level.	-48 .. 0 dB
Attack	Limiter attack rate	-100 .. 100 %
Release	Limiter release rate	-100 .. 100 %

Name	*Description*	*Range*
Output	Limiter output attenuation	-48 .. 0 dB

Frequency Shifter

Frequency shifter algorithm. Provides a delay unit and a feedback loop to give consecutively shifted repeating delays.

Name	*Description*	*Range*
Shift Left	Amount of frequency shift (in hertz) for the left channel. The range can be extended from the sliders context menu.	-10 .. 10 Hz / -1 .. 1 kHz
Shift Right	Amount of frequency shift (relative to the left channel) for the right channel.	-100 .. 100 %
Delay	Delay time for the frequency-shifted signal. Can be tempo-synced.	0.004 .. 32 s 1/512 .. 16 bars
Feedback	Feedback around the frequency shifter and delay-unit.	
Mix	Blend control between the dry and the wet signal.	0 .. 100 %

Questions?

Feel free to visit the on-line forum at the Vember Audio website if there is anything you want to ask about.

http://www.vemberaudio.se