64 bits Delay-based Sound Synthesizer
for Windows

U S E R   M A N U A L

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System requirements & Installation

Requirements:

-Windows 7+ x64, at least 4GB Ram, any modern 64 bits multicore CPU and any VST® compatible host are the only requirements.

-Depending on your setup, you *might* need to install *Microsoft Visual C++ x64 Redistributable 2015-2019* (it should be present by default in Windows 10+). This is likely the case if Synlay crashes on loading or cannot be detected by your plugin host. Please make sure the x64 version is installed, otherwise download it from Microsoft.

Installation:

Because of our strict policy about "portability" and transparency, we are not providing any automated installer.

-Please ***delete any previous version of Synlay***, since Synlay 2 has the same name and ID as Synlay 1.x and is 100% compatible with existing DAW projects and presets

-For usage as VST2 simply ***rename*** `Synlay.vst3` as `Synlay.dll`, and copy the whole Synlay distribution folder to your preferred plugins folder

-For usage as VST3: copy the whole Synlay distribution folder to your preferred plugins folder. Unfortunately, you shall also ***move*** the included `VST3 Presets` folder to `C:\Users\[your name]\Documents\`, since the VST3 standard imposes you this fixed location for factory presets (we can't do anything against that, despite several attempts)

Note: Synlay's binary is protected against tampering, ripping and reverse-engineering, for this reason some anti-viruses could mistake it for "malicious" software. In the event of "false positives", we can't do anything other than <u>trashing such complaints</u>. Please remove any antivirus, they are often worse than actual viruses and will just slow down your machine !

Key Features

-Advanced yet relatively compact delay-based (Karplus-Strong) synthesizer capable of producing very diverse, original timbres and of high quality

-Stereo noise source and true stereo processing, for naturally wide stereo sounds

-External audio input can also be used as excitation signal for the waveguides in place of the internal noise source

-Biquad resonant multimode filter with up to 8 stages for the excitation source, (polyphonic and modulable)

-ADSR envelope with Attack and Release velocity scaling and "one-shot" mode for the excitation source

-Three parallel, independent digital waveguides

-16 polyphonic voices or Mono mode

-Adjustable, detuning-free feedback HF damp filters

-Feedback can be either positive or negative and can be automatically scaled with pitch, for all notes to have the same release time

-Optional post-waveguide envelope to shorten the natural feedback decay

-100% devoid of clicks or artifacts

-Fully configurable LFO for vibrato or tremolo

-Master multimode / spectral filter, to furtherly shape the result (not polyphonic, not modulable)

-Extensive Velocity and Aftertouch support and MIDI automation

-Polyphonic Glide support

-Identical sound at any working sample rate

What's new in v2 ?

The whole program and all its components, in particular the digital waveguides, have been re-engineered and re-written from scratch, resulting in highly improved code and much lower CPU load. Despite the abundant work done behind the scenes, Synlay 2 is 100% compatible with existing presets and DAW projects, and efforts have also been put to have it sounding identically
Unfortunately, some little inconsistencies or imperfections could NOT be fixed, like the envelope “exploding” when releasing a key during A or D stages, or the linearity of the feedback knobs, because doing so would have broken compatibility with existing presets.

-An external either mono or stereo signal can now be used as excitation source, rendering the old *Tuned Delays* plugin by Elena Design deprecated

-Negative values are now allowed for filter resonance (more a Q factor), to smooth cut-off transition zone

-The white noise source can now optionally be resetted at every key press, to generate identical sequences. This can be sometimes useful for basses or percussive sounds, to have a predictable yet less realistic attack

-More radical abatement of the DC peak for cleaner sounds and pretty no residual low-frequency noise

-Spectral master filter to further shape the final sound or simply equalize the external input, which can be used in alternative to the standard multimode IIR filter

-The whole panel has been re-arranged and some details re-designed for improved ergonomy and rationality; most readouts for entering values manually have been removed and replaced by popup readouts; all controls are now consistently “ghosted” when disabled by other controls

-The range of the LFO controls was inconsistent and have been fixed and their behaviour improved; LFO width range for pitch modulation can now be extended from +/- one semitone to +/- two octaves by a little switch, when really needed for special effects

-Clicks no longer occur when retriggering the LFO (Sync enabled) in AM mode

-Clicks no longer occur repeating a same note with lower Velocity when feedback velocity modulation is set

-The source filter can now be completely disabled when not needed

-Some buttons could not be pressed twice without moving the mouse pointer between presses, this also has been fixed

-The option to limit Pitch Bending to the highest note has been removed for technical reasons, but it was pretty never used anyway

-Mono retrigger priority is now always assigned to the previously held note – even in this case, other options were just confusing and never used

-A new Dup Notes option has been added, to allow a same key pressed repeatedly to engage new polyphonic voices

-Many new and better factory presets added, many old and not-so-good ones removed, many existing ones revised

-Some factory presets had unwanted MIDI-controls assignments, now removed

1. Overview

Synlay is a **de*lay*-based *syn*thesizer**. Also known as “Karplus-Strong”, this synthesis technique relies on no oscillators and no samples, and at present has only been exploited partially and pretty only for plucked strings emulation, despite its *huge* potential.

A very short delay with feedback (i.e. an actual “echo”), is applied to an excitation signal, consisting generally of white noise or a synthesized pulse. However, the delay time is so short that the “echo” is perceived as a *pitch*, rather than a distinct repetition of the input signal – it is a “tuned delay”.

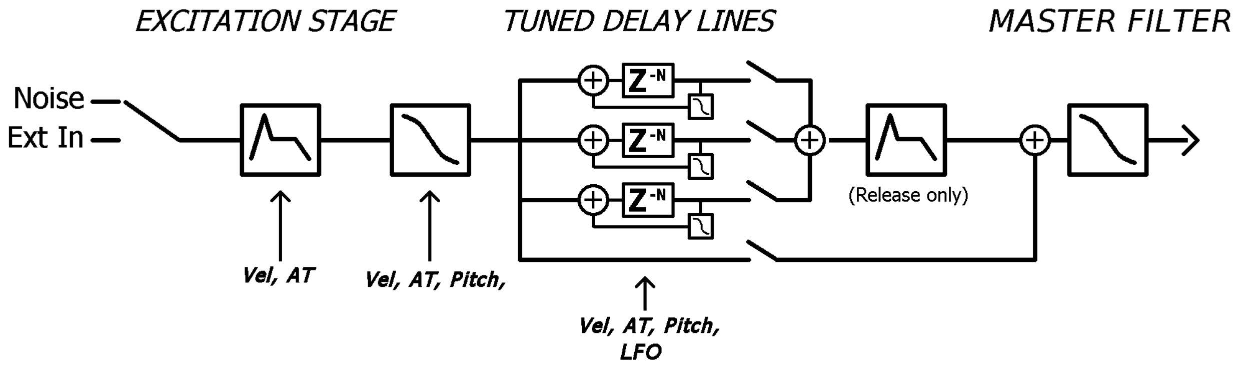
This principle can be appreciated accidentally when moving back and forth before a wall reflecting the background noise of the city, for example: you can perceive a pitch, changing as you move closer or farther from the wall, as an effect of the changing delay time of the reflected noise, mixed back with the direct noise – just the feedback (echo) is missing.

More technically, the very process can be seen as the convolution of the excitation signal with a decaying pulse train of given pitch.

Beyond some plucked strings plugins, we have yet to find any synth where this technique is employed *extensively* enough to be used for any kind of instruments, and to be exploited and *explored* without constrains (if we exclude Dulcet, an early but nice attempt, and perhaps very few other, all with their intrinsic limitations.) This motivated us to create a *professional* delay-based synthesizer, in order to freely and fully explore the true potential of this not popular enough (if compared to additive, subtractive, FM, wave-table based, etc) synthesis method.

Despite being relatively simple conceptually, the actual implementation of high quality delay-based synthesis required special and carefully engineered delay lines, or *digital waveguides*, resulting in a quite complex piece of software, with respect to the initial, non-published proof-of-concept prototype. The waveguides employed by Synlay are very accurate clockworks, capable of 1. precise fractional delay, to resolve every pitch with continuity and without any spectral attenuation; 2. click-free modulation of delay time (pitch); 3. pitch compensation to avoid detuning when using the feedback damping filter; 4. automatic scaling of the feedback amount in function of pitch, for all notes to decay with the same length; 5.automatic DC peak abatement; 6. “ground” signal abatement for a cleaner sound

2. Principle of operation



The above schematic illustrates Synlay's operating principle in a simplified manner.

The excitation source can be chosen from the internal white/pink stereo noise generator (default usage) or from an external audio input. An external input can be used in many creative ways: using Synlay to “harmonize” inharmonic material like drums, using a microphone to “musicalize” noises (ticks, plucks, …) or even blowing in as it were a breath controller for flute sounds, or even cascading a different synthesizer or sound source with Synlay for generation of original and complex timbres.

Every circuit inside Synlay is duplicated, for true stereo processing; this allows, together with the stereo noise generator, for true spatial sounds without resorting to any reverbs or ambience effects. Reverberating sounds can be easily produced by intelligent setting of the ADSR, as illustrated by several factory presets.

The input signal amplitude is controlled by the Input ADSR envelope, triggered by every MIDI key press.

The input signal is then subjected to filtering by a biquad multimode resonant filter with up to eight stages. This filter can be extensively modulated by either key velocity, aftertouch or key position.

The resulting signal finally excites three parallel and independent waveguides (“tuned delays”), each one with adjustable pitch, level, both negative or positive feedback, feedback HF damping and L/R detuning for a chorus effect. Feedback amount can be modulated by key velocity or aftertouch, and every waveguide module can be disabled by the apposite main switch when not used.

A positive feedback convolves the excitation signal with a pulse train, resulting in a “sawtooth-like” timbre with all harmonics. A negative feedback convolves the excitation signal with a train of alternate negative and positive pulses instead, resulting in a “square wave-like” timbre one octave lower and with only odd harmonics. The result is a sound whose spectrum is the product (filtering) of the source spectrum with a comb of varying pitch: every harmonic will have its magnitude constantly modulated by the respective frequency in the source spectrum, leading to a dynanic and realistic sound, not sounding “synthetic” at all if compared to a flat sawtooth or square waveform.

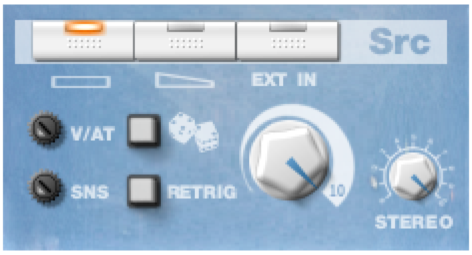
The output of the waveguides is then subjected to a further auxiliary envelope consisting of a Release stage only, whenever the naturally decaying feedback should be shortened or trimmed.

The filtered excitation signal bypassing the waveguides can be forwarded and summed to the output of the auxiliary envelope, whenever hearing it directly were desireable.

A final master filter allows for accurate frequency shaping of the resulting sound, where required. Either a multi-mode, non-resonant single-stage IIR filter or a 128 bands zero-phase spectral filter are available. Since the underlying synthesis process is substantially a convolution, applying a filter upstream or downstream the waveguides leads to an identical result. However, two distinct and different filters are provided, being the source filter substantially modulable for expression purposes and physically replicated for every polyphonic voice, while the master filter is non-modulable and is common to all voices. For this reason, the master filter can also be used to equalize the input signal in case an external source is selected, as in case of a microphone signal to remove low-frequency bumps.

3. Detailed Usage

Excitation section: Source



The three main “radio” buttons on top select the excitation source, from **White** stereo noise, **Pink** stereo noise (richer with low frequencies, ideal for basses) or an **external** stereo signal.

The **V/AT** and **SNS** trimmers adjust the ratio of Velocity vs. Aftertouch which modulates the amplitude of the noise generator, and the modulation sensitivity, respectively. For example, by setting theV/AT to zero (min) and SNS to one (max), key velocity will fully control the volume of the noise source; by setting V/AT to one (max), only the Aftertouch will be effective; by setting V/AT to 0.5 both Velocity and Aftertouch will control volume in a balanced way; by setting SNS to zero, noise volume will be fixed and won't be influenced by key pressure in any way.

The stereo noise generator is a “pseudo-random” noise generator, i.e. with completely deterministic behaviour depending entirely on the initial Left and Right “seeds”. The **Retrigger** switch allows to re-trigger the noise generator with same seeds for generating an identical signal everytime a key is pressed, instead of operating in “free run” mode.
The **Seed** button (dice icon) generates a new pair of Left and Right random “seeds” everytime it is pressed.
Note that, for short noise pulses (see Envelope), seeds tend to affect the stereo positioning of the resulting signal and its spectral content; this is absolutely normal.
Retrigger / seeding feature is normally just needed for plucked or percussive sounds only (short excitation impulses with fast attack), especially basses, to prevent random variations of the attack spectrum from note to note, which may render some notes weaker than others.
Every new seed pair obtained with the dice button will be stored in the current Preset.
In case a timbre with a short excitation pulse should “jump” too much thru the stereo panorama from note to note, you can either 1. set Stereo widh to zero and add spatiality adjusting L/R detuning (see later) or simply using an external effect, or 2. enable Retriggering and press the Seed button repeatedly, until a seed pair giving the wanted stereo positioning is found.

The four controls described above only affect the stereo noise generator.

The large knob adjusts the main **Volume** of the excitation source and the **Stereo** knob adjusts its stereo width. Both controls affect the white noise generator as well as the external signal, if selected.

Using an **external signal** as excitation source has many creative purposes.
An external drum track can be “harmonized” with Synlay, exactly as it was done by the (now deprecated) *Tuned Delays* plugin by Elena Design. Even another synth can be cascaded and harmonized with Synlay, to obtain complex and very rich timbres.
A microphone can be used aswell, to harmonize your voice in a “metallic” way, or to generate musical sounds from inharmonic noises (hits, pops, rubbing or even blowing, using

your mic as a kind of breath control).
For more information about using an external source, please consult the dedicated chapter.

Excitation section: Source Envelope

The four **A**, **D**, **S**, **R** little knobs adjust the envelope Attack time, Decay time, Sustain level and Release time, respectively, as in most plain ADSR envelopes. The two **"VEL"** trimmers under the A and over the R knobs adjust the amount of time scaling by Velocity: when set to a value larger than zero, higher Velocity values will shorten the respective envelope time proportionally. By proper adjustment it is therefore possible, for example, to dynamically control with Velocity the attack and release time of instruments like violins or trumpets, for increased expression and realism.

The "**1-Shot**" button on top causes the envelope generator to ignore any key release during the Attack and Release stages, for them to always come to their natural completion.

The "Post" button enables the **Post-Envelope**. Despite its controls have been placed in the Excitation/Envelope section because that seemed more convenient, this auxiliary envelope actually operates *downstream* the three delay lines, and it is composed of a Release stage only. It can be extremely useful to shorten the natural decay of a sound caused by its resonance in the waveguides, when setting a relatively high Feedback value (see later). In fact, a high Feedback setting is often desired for a richer timbre, yet the resulting long natural decay may not always be wanted.
The Post-Envelope Release time is controlled by the large knob labeled with an "**R**", and it is enabled only when the Post button is switched On.

The Envelope is triggered by every key press, and this also is true when using an external excitation source; in this case, it is often preferred to set Attack and Release times to zero and Sustain level to the maximum, for the Envelope to rather behave like a simple "gate".

Excitation section: Source Filter

The excitation signal subjected to the Envelope is then processed by a polyphonic biquad resonant and multi-stage source filter before reaching the dealy lines. This filter is replicated for every polyphonic voice and responds to several kinds of modulation sources.

The **four main "radio" buttons** on top allow selection of the filter type from Low-Pass, High-Pass, Band-Pass and Band-Reject. These buttons can also be completely disabled by switching the currently active button: in this case the source filter will be disabled, as it may often be desired when using an external source.

The large “**Khz**” knob adjusts the filter cut-off frequency, from 0 to 15 Khz.

The “**Q**” knob adjusts the filter q-factor: a negative value will smooth the transition zone, while a positive value will cause resonance.

The “**STGS**” knob selects the number of filter stages, i.e how many identical filters (up to eight) are stacked, for a more and more abrupt frequency response.

For expression purposes, cut-off frequency can be modulated in several ways by switching the respective little squared “radio” buttons:

***KF** (“Key-follow”)* - the frequency of the played note is directly assigned to the filter cut-off. This allows for sound generation already at filter stage in combination with a suitably high Q factor, without even using the delay lines, or in addition to them (but be aware of the resonance which may result !). When KF is enabled, the Khz knob is disabled by having no purpose

***V/AT** (“Velocity / Aftertouch scaling”)* - cut-off frequency set with the Khz knob is scaled by Velocity / Aftertouch (see Source section): for a sensitivity (“**SNS**”) level set to its maximum, a low value of Velocity or Afteroutch will cause the resulting cut-off frequency to be close to zero, and a maximum value will result in the cut-off value set. By enabling the “**INV**” switch, the inverse behaviour is adopted

***KS** (“Key scaling”)* - the cut-off frequency set is proportionally increased when playing notes higher than central A and decreased when playing lower notes, by an amount dictated by the neighbour **slider**, whose position zero is at the center. By specifying a negative key-scaling value, the opposite occurs

Tuned Delays: Main Bar

This is the section where actual sound generation occurs, and the excitation signal is turned into a musical sound.
The main Tuned Delays bar offers three global controls.

The “**Direct**” switch allows the excitation signal to bypass the three digital waveguides, whenever hearing it directly may be desireable.

The “**Autoscale FB**” switch allows automatic internal scaling of the Feedback value set according to the note played, for all notes to decay with the same length. When this switch is disabled, higher notes will tend to extinguish faster and lower notes more slowly, as it is the natural behaviour of a waveguide.
Note that when a non-zero Damp factor is specified (see later), higher notes will tend to extinguish faster anyway: this is completely normal and unavoidable.

The “**Tuning Drift**” trimmer and its respective readout / edit box allows to specify a signed detuning amount thru the keyboard. A slight tune drift may often contribute to a richer sound, to avoid octave intervals sounding too “flat”.

Tuned Delays: Digital Waveguides

The core of Synlay are the three identical, independent and parallel Digital Waveguide modules, or “tuned delays”.

The **Tuning** slider and its respective readout / edit box adjusts the waveguide tuning, in semitones including fractions.

The large **main button** at top-right enables or disables the waveguide module. When a module is not needed, it is always better to turn it off, in order to save CPU.

The first large knob labeled with a curved arrow controls the **Feedback** amount and sign of the waveguide. A higher value produces a more distinct sound (more prominent harmonic peaks with respect to the excitation signal), and even a longer natural decay. A positive value produces a “sawtooth-like” timbre, with all spectral harmonics, while a negative value produces a “squarewave-like” timbre, with just odd harmonics and one octave lower.

As usual, the “**V/AT**” and “**SNS**” trimmers placed on the left allow scaling of the Feedback factor set with key Velocity and/or Aftertouch.

The second large knob adjusts the **Volume** of the waveguide module.

The “**L/R Det**” knob detunes the Left and Right channels by a variable and opposite amount, to produce a kind of stereo chorus.

Finally, the “**Damp**” knob adjusts the amount of high-frequency damping thru the feedback line of the waveguide. Damping allows to easily and realistically simulate the sound of plucked strings or other percussive instruments, since in real world higher frequencies always tend to dissipate faster. The feedback damp filter is perhaps what contributes the most to the sound realism achievable by Synlay and by Karplus-Strong synthesis in general.

Low Frequency Oscillator

The LFO is polyphonic and every played note gets an independent one.

The two large "radio" buttons in the LFO section bar select the type of LFO **waveform**, which can be either sinusoidal or random. The random wave is a smoothed pseudo-random oscillation (brick-walled white noise), with a rate of change (cut-off frequency) consistent with the selected LFO frequency. Both buttons can be turned Off, to disable the LFO.

The little "**Dest**" box when clicked opens a menu to choose the LFO target, or destination: FM (frequency modulation of "vibrato", will affect the pitch of the waveguides), AM (amplitude modulation or "tremolo", will modulate the amplitude of the waveguides output mixed with the direct signal, if enabled)

The large knob labeled with a "**W**" adjusts the LFO modulation width, whose units depend on the target: semitones for FM, or a percentage of modulated amplitude for AM.
In FM mode, the default knob range is zero to one semitone (peak), but it can be extended to 24 semitones (2 octaves peak) for special effects by enabling the small **EXT** switch.

LFO width can be further scaled with the Modulation Wheel or the Aftertouch up to the value set, by enabling the mutually exclusive buttons on the left, labeled with "**MW**" and "**AT**" respectively.

The second large knob adjusts the LFO **frequency** from 0 to 10 Hz. The frequency can be further modulated with the Aftertouch from zero up to the value set and by a variable amount using by the small "**AT**" trimmer on the right.

When the "**Sync**" option is enabled, LFO phase is resetted at every key press instead of running freely, and the retrigger phase can be specified with the four-positions **phase selector** just above (0, 90, 180 or 270 degrees relative to a sine wave). Phase consistency on re-tigger is warranted even when using the Random waveform.

→ *Hint*: an LFO frequency equal to zero can be specified, in which case the LFO will be "frozen" on a fixed value depending on its phase. In conjunction with the Sync option and retrigger phase suitably set plus Aftertouch scaling, you can modulate the sound pitch with key pressure, when required for expression purposes and as illustrated in some factory presets.

Master Filter

The sound generated by the three waveguides mixed with the direct excitation signal, if enabled, can be furtherly shaped by a master, non-modulable filter.

The spectrum of the *un-filtered* signal can be plotted constantly, for convenience, by the builtin **spectrum analyzer**, when enabled by the little switch labeled with a **monitor** icon.

The five large "radio" buttons allow selection of the master **filter type**, from IIR Low-Pass, High-Pass, Band-Pass, Band-Reject or Spectral.
The Cut-off frequency for the first four types (IIR) can be adjusted with the large "**Khz**" knob underneath from 0 to 15 KHz, which is of course disabled when the Spectral filter type is selected.

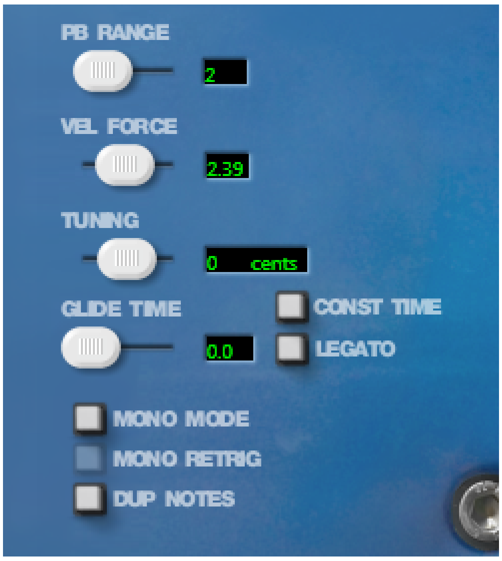
The **spectral filter** used is a zero-phase, 256 points convolution filter, and as such has a latency of 128 samples, a delay which is however not appreciable for most purposes.
Its frequency response curve can be hand-drawn directly inside the Analyzer box.
Every horizontal grid line corresponds to 10 dB, from +10 dB on top down to -∞ (with a linear "roll-off" region from -60 dB to silence). The 0 dB line is marked in green; any curve value above this point will cause amplification.
The little **Reset** button labeled with a flat line resets the curve to a constant 0 dB level.
Frequency is represented on a logarithmic scale, from 0 Hz to 20 Khz; the upper limit is fixed at 20 Khz, for a same curve drawn to be consistent across different working sampling rates (this is the same range covered by the spectrum analyzer). The curve value drawn at 20 KHz will be replicated up to the working Nyquist frequency as a flat response line (not displayed). The frequency corresponding to the current mouse X coordinate is displayed by a momentary box within the filter bar.
The little button labeled with a **camera** icon captures the *spectral envelope* of the external signal, to be used as frequency respose curve, which can be further hand-edited if required.
A **context menu** is also available by right-clicking in the editor box. You can either **Load** or **Save** the current frequency response curve in a propertary .spec binary format. You can also **load a short .wav file** containing an Impulse Response, to be converted to a response curve (since the filter uses a 256 samples kernel, shorter impulse responses will be zero-padded and longer ones will be truncated; the result will always be normalized to 0 dB; wav files longer than 16384 samples will be rejected).
Note that the last curve drawn will remain stored in the current preset memory, even when the spectral filter is disabled, otherwise it could not be retrieved when enabling it again. For this reason, a **Delete Curve** menu is also available for erasing the preset curve memory, thus saving unused space in the current preset or preset file when no spectral filter is used.

The Master Filter can be disabled when not needed, by switching the currently active "radio" button Off.

The small **Clip** led is a clipping indicator which warns you when the output signal reaches 0 dB (+/- 1.0 float), after which distortion will occur. Please lower the Source Volume or the

Sustain level in case of clipping.

Master Controls



On the rigthmost side of the panel you can find the master synth controls.

"**PB Range**" adjusts the pitch-bender range, in semi-tones

"**Vel Force**" adjusts the global velocity curve, using a power law with exponent ranging from 1 (no scaling) to 4 (more force required)

"**Tuning**" adjusts the global synth tuning, from -100 to 100 cents

"**Glide Time**" adjusts the speed of the polyphonic glider, from 0 (gliding disabled) to 8 (very slow gliding). The two little switches on the right enable **Constant Time** gliding mode (vs. Constant Rate) and gliding on **Legato** only, respectively.

"**Mono Mode**" limits polyphony to just one voice, while "**Mono Retrig**" enables re-triggering of further notes played over in mono mode, when otherwise only a pitch change would occur.

"**Dup Notes**" allows a same key pressed over and over to engage new polyphonic voices for the same note, instead of re-triggering the same voice as it happens with a piano.

Synlay logo

Finally, by clicking on the **Synlay logo** sticker, an About panel will open showing program information, version and registration data. You will also be able to access this Manual directly by clicking on the **Manual icon** found inside the About panel.
In the unlicensed version, the same panel will be displayed automatically everytime the plugin interface is opened, where the user can find the button and link to donate via PayPal.

4. Using an external signal

To use an external signal, you shall configure your VST host to warrant Synlay **both MIDI and audio inputs**, identically as you would do with a sampler, or with a MIDI-controlled audio effect, for example. How this is accomplished depends on the host used, since every one adopts a different routing logic.

At preset, *Reaper* appears having a 'bug' for which a synth provided with audio input(s) will have the incoming audio mixed with the audio output. The only workaround is configuring Synlay to accept the audio input on a new virtual input pair (3,4) disabling the default pair (1,2), and routing the audio from an audio track (muting its master send) to the MIDI track hosting Synlay to inputs 3,4. Other hosts seem not suffering similar problems though. To verify that everything is routed properly, *make sure no input audio is audible as long as no MIDI key is pressed*.

Depending on your host audio routing, you may need to set Stereo Width to zero in Synlay Source section, when using a monophonic source as a microphone, unless your host is duplicating the mono signal to both Left and Right channels.

It is advisable to set the Envelope to act like a **gate** when using an external signal, by adjusting the Attack and Release times to zero and Sustain level to the maximum.

In general it is preferrable to disable the Source filter when using an external signal, and to just rely on the Master filter to equalize or shape the input, if needed. Using the Source filter with an external signal is in general pointless, since the spectral features of the excitation signal are dictated by the external signal and there is not any noise source to be shaped; also, the Source filter would waste more CPU power for nothing, being replicated for every note played.

The factory preset “EXT IN” can be used as a starting point to test processing of an external signal.

An external signal can also by used by the Master Filter when operating in Spectral mode, to capture and use its spectral envelope as frequency response curve. Smart usage of this feature allows several tricks. For example, you may “vocalize” a timbre by sampling the spectral envelope of some vowels using your microphone; however, the fidelity of the result may vary depending on many factors, also considering that for some vowels more than others, the formants envelope is not constant across any pitch. You can also sample the spectral envelope of an acoustic instrument, to apply its impulse response to a timbre (e.g a violin or guitar body)

WARNING ! When using a microphone, it is imperative to use headphones and to turn your audio monitors off, otherwise a loud feedback (Larsen) will suddently originate, with risk of damages to your eardrums or audio apparatus !

5. Automations and commodities

-On every control provided with a central default position, a double-click will reset the control to its default central value (usually zero)

-A bubble-help displaying the control purpose will pop up when hovering on a control and standing for a moment

-The exact value of controls not provided with an explicit readout/edit box is displayed in a bubble box when clicking for adjustment. Whenever an exact value should be entered, you may use direct parameter listing (if your VST host allows so, see below)

-Several controls can be assigned to MIDI controllers for convenience. For controls where this feature is provided, a **Learn / UnLearn** menu is also available when right-clicking. After selecting Learn, the control will be assigned to the first MIDI controller message received. The assignment can be removed by selecting UnLearn.

-Most controls are also exposed to the VST host either for track automaion or simply for symbolic listing and manual entering of the respective values (only in those VST hosts allowing so)

6. Faq

-Q: Why no Synlay for Mac any more ?

-A: Synlay 1.7 was available for Mac; however, given the *null* feedback received, we decided to remove it and to drop support.
Depending on pool of users, amount of donations and requests, we *may* commision a Mac porting in future. We are not using any Mac, nor we want one. However, it will be a *commercial* software, if at all, or simply it will be granted to donors of the Windows version only.

-Q: On Windows 7, some graphics look weird and/or with white “speckles”

-A: Sadly this is a bug with the graphics framework used to load images. We can't do antyhing to fix it at preset. Windows 10+ is not affected though.

-Q: I can't see any factory presets ! Where are they ?

-A: Evidently you did not read the install instructions.
When Synlay operates in VST3 mode, you must move the included VST3 Presets folder to C:\USERS\[YOUR NAME]\DOCUMENTS.
This is an imposition of the VST3 standard and we can't do anything !
If installed properly, the included factory presets shall be listed automatically by your VST host. Synlay has no internal preset browser.