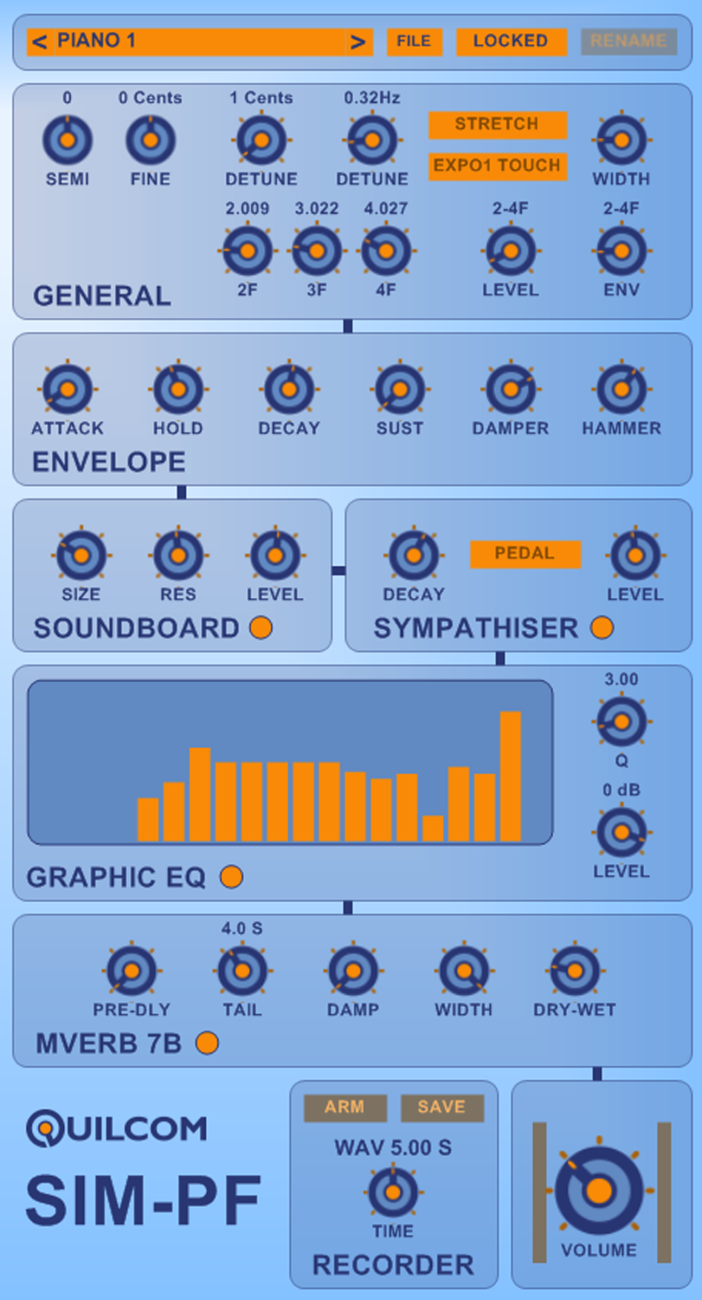
**Quilcom SIM-PF**



**Design**

The Quilcom SIM-PF is a plugin designed specifically for simulating a Pianoforte.

The brief I set myself was to use conventional synthesiser techniques and to use NO sample clips or complex Physical Modelling like Waveguides etc.

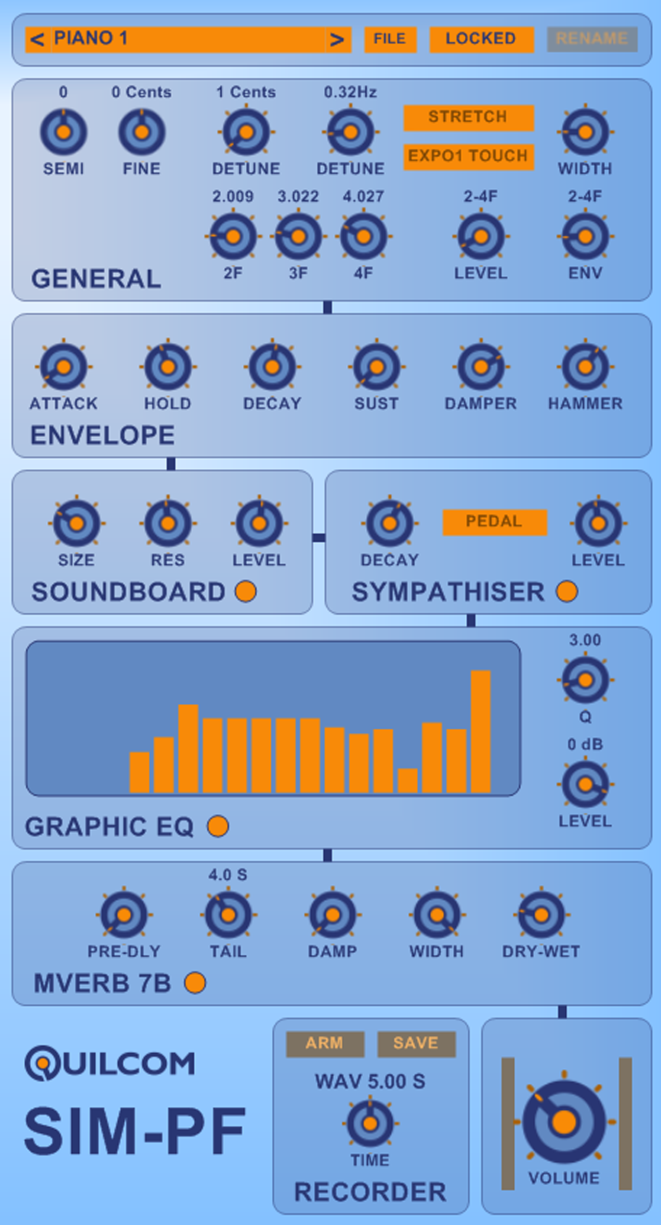
Generally speaking, using sets of samples will give more authentic results but, for truly convincing sounds, you would need many gigabytes of disk space and sophisticated direct-from-disk streaming for the sustaining parts. You would also need multiple Round Robin or random sample selection methods, velocity-based selection of samples and detailed key-zoning/mapping. Finally, at least the attack phases would need to be loaded into RAM for latency-free playing, thus needing a high specification and more expensive PC (for the better samplers) and long load times.

For reducing the hard drive space, developers often use reduced length sample set and have to make loop points for the “sustaining” phase (actually slow decay). If I listen to a sample-based piano note I can often hear the unnatural looping because true piano sounds change their timbre smoothly as they decay. This can’t happen with reduced sample lengths which have to loop while the sound fades.

Another advantage of synthesising the sound is that of parameter control. Even when all the conditions are right for the sampling session, the sound is “baked in” and only superficial variations can be made after the event.

The SIM-PF should not be expected to sound as authentic as an expensive sample-based instrument, but it does offer you the advantages outlined above.

Rather than go into the various techniques used at this point, the operating instructions below will touch on the theory relevant to the various panels’ functions in the instrument.



**Overview**

At the top is the preset manager. The main preset is PIANO 1 which I “tuned” carefully to give a general purpose sound. This makes a useful starting point to adapt the sound and touch to your taste and needs. In practice I found it necessary to tweak settings depending on the music being played.

The other presets give an idea of the range of the instrument. Then we go down to the GENERAL panel which is where you can affect the base sound of the instrument.

Next is the ENVELOPE generator which controls the volume contour and can modulate the inharmonics controlled on the GENERAL panel.

The signal then feeds a SOUNDBOARD simulator and then the SYMPATHISER for sympathetic string resonance.

The harmonic balance of the signal can then be tailored with the Graphic eq and finally reverb can be added to give space to the sound, which can be recorded to be used in any sampler of choice.

So next are descriptions of the various panels with more details and theory…



This is the preset (patch) manager. Click on the preset name to open a drop-list selector, or page through them with the arrow keys. The FILE button is for saving, loading, copying and pasting of presets. You can also save and load a whole bank. The FlowStone presets are saved as text files, which can be read and edited in Notepad for example.

When LOCKED is selected, any changes you make to the controls will NOT be remembered by the DAW. This enables you to experiment and easily get back to the original sound by just re-selecting the preset.

When the LOCKED button is clicked, and then shows UNLOCKED, the current settings will be saved with the DAW song file. The greyed out RENAME button will become available to call the preset whatever you want.



The VOLUME knob is for the plugin’s audio output and features two bar graphs which indicate average peak values. Maximum is clipping point. If clipping happens, even for a very short time, the central ring of the knob turns red and holds for 1 second, so you don’t miss short clipping times.



The recorder allows you to capture any sound from the synth for up to 10 seconds, the length of the created WAV file being set with the TIME knob. This allows you to capture a sound to use elsewhere.

Set up the sound you want and the recording time. Click ARM and then the recording will start when you press any MIDI note. A bar shows the progress.

If you want to save it, click on SAVE and use the standard Windows dialogue box to choose a location and name. If you’re not happy with it, just repeat the above without saving. The internal buffer is over-written.



On the GENERAL panel the SEMI and FINE knobs allow you to tune the whole synth to other instruments. The SEMI knob allows a full plus or minus octave in semitone steps.

The EXPO1 TOUCH is a selector for choosing a touch style to match your playing technique and keyboard. It controls the way the raw incoming velocity value is mapped to the internal velocity sensitive components. If you have the option to alter this touch response curve on your keyboard, you should choose VEL DIRECT on the selector and then no change is made to the values by the synth.

The synth’s sound is panned left-right on a per-octave basis. The WIDTH knob controls the stereo width of the oscillators, going from mono through normal when centred (double click) to extra-wide when fully clockwise.

**Oscillator bank**

A piano has 3 strings for most notes and these 3 strings are slightly detuned per-note to give a much more interesting sound. Accordingly, there are 3 oscillators sounding which can be detuned in terms of Hz for controlling the beat frequency across the span, and as a proportional cent (1/100 of a semitone) across the span. Obviously you can have both and this gives a widely adjustable range to the timbre.

The waveform tables for the 3 string oscillators make use of what I call Phase Scrambling. Each is a single cycle extracted using my Quilcom Wavemaker 4. This can take a waveform and perform a Fourier Transform, keeping the partial *amplitudes*, but randomly changing their *phases*. This results in very similar harmonic spectra but very different waveform *shapes*. The purpose of this is to reduce unnatural “flanging” sound when detuned.

The DETUNE knobs set these static tuning offsets in cents or Hz.

A steel string set with high tension will produce *inharmonic* tones, due to the displacement of the string as it vibrates. Higher harmonics will have a higher proportional upward tuning; they are always *sharp* with respect to the fundamental of each string and so provide more beating effects which contribute to part of the piano timbre. They not only beat with the fundamental but with each other. In the SIM-PF you have 3 further oscillators which can be set to inharmonic ratios to the fundamental. Fine tuning for 2F, 3F and 4F are provided. The *amplitude* ratios are internally fixed so as to reduce with harmonic number, based on real spectra I have looked at. Experimentally I found no audible benefit from going above the 4th harmonic.

There are 2 knobs for setting the levels of the inharmonic oscillators. The LEVEL knob is a static setting for the amount of inharmonics and the ENV knob sets the amount of influence from the Envelope generator. The settings of both knobs are summed. By using the envelope to control the level, the amount of inharmonics present decays with the amplitude of the sound, as it would in a real piano. There’s also a small pitch modulation from the envelope generator.

The STRETCH button turns on stretch tuning which simulates the famous Railsback curve. This curve was derived from measuring the fundamental tuning deviation set by professional piano tuners. The reason for this tuning method is due to those inharmonic overtones mentioned above. They need to sound in tune with other notes higher or lower on the keyboard. So, for example, if the second harmonic is sharp the whole note must sound in tune with the whole sound 1 octave higher. Middle C is normally set to be “correct”, lower notes are slightly flattened and higher ones sharpened. It’s quite a subtle effect to my ears!



The ENVELOPE generator controls the volume contour and can also be used to control the inharmonic oscillators’ level. I have set the operating ranges purely for simulating a piano envelope, but there is sufficient flexibility for making different sounds, as per some of the presets.

The ATTACK knob has 2 areas of operation. Up to 9 o’clock the attack time is not influenced by velocity. Beyond this, higher velocity will start to *reduce* the attack time. That allows for new sounds like my BOWED PIANO preset and allows for more expressive control when a softer attack is set.

HOLD time is provided because a real felt hammer hitting a string will have a contact time of 3-5mS.

DECAY allows you to go beyond simulating the longer decay times of a Grand piano and the shorter decay times of an Upright piano. If you play a piano MIDI file, you can reduce the sound to staccato by setting shorter decay times.

The decay time is key-tracked internally to provide much shorter notes at the high end of the keyboard. This effect is spread non-linearly across the key span, as in a real piano.

The SUST sustain level knob would normally be set at zero for a piano, but advancing it can give a very different sound.

I’ve named the release time knob DAMPER since that is its function in this simulation. Note that a Grand piano will take longer to dampen than an upright, because there is more energy stored in the longer strings and larger soundboard.

The HAMMER knob controls level of the initial sound of the hammer striking the string. It’s a short decay envelope derived from the main envelope signal. With a fast attack set and high velocity, the strike sound is more prominent. It controls the amount of filtered noise which ring-modulates the audio from the oscillators and thus simulates the interaction of the felt with the string for the first 5mS or so.

A real piano envelope decay is quite complex, but typically resolves into 2 decay phases, depending on the energy supplied when striking a key. This is simulated in the envelope generator system to add a short initial peak and decay to the signal. It has the effect of providing a higher dynamic range and a more realistic and variable strike sound. Not so easy with fixed sample sets!

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As with an electric guitar, the sound produced by the string *on its own* is really quiet. In a real piano the string vibration is coupled to a soundboard which amplifies and projects the sound, in principle similar to an acoustic guitar. When I examined the frequency response of typical soundboards I noticed they took a form similar to a comb filter; lots of regular peaks and troughs getting closer together at higher frequencies. This is the method used in the SIM-PF.

The SIZE knob sets the delay time in the comb filter. At a low setting this simulates a smaller soundboard with a shorter delay, and increasing the SIZE sets a longer delay. This is not *strictly* analogous to piano size though, but it allows you to set a colouration according to taste.

The RES resonance knob sets the height of the peaks of the comb filter. At minimum the effect is subtle and at maximum the sound is very “ringy”. If you chose to automate this knob in your DAW it can add a mild to extreme flanginess to the sound.

The LEVEL knob sets the amount of filtered signal added to the dry signal.

The orange LED switch turns the SOUNDBARD on/off, and when off reduces CPU usage.

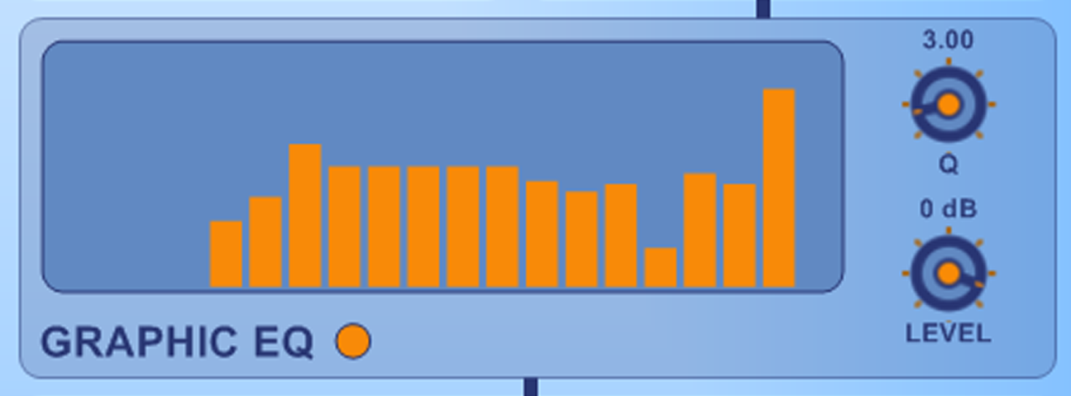
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One of the elements of a real piano sound is sympathetic resonance. This is when a string is slightly energised by the vibration of another string, typically an octave or two apart. This resonance is only noticeable when the sustain pedal is pressed, so all the string dampers are off.

The SYMPATHISER uses the core engine of my Quilcom Harpverb which is actually a chromatic reverb (since there are no actual strings to resonate!). The DECAY sets the time of the tail and the LEVEL sets the amount of pseudo-resonance added to the dry signal.

When PEDAL is selected, as shown, the simulated resonance is added to the signal from the SOUNDBOARD only when the sustain pedal is pressed (you get a border showing around the switch when the pedal is down). It’s then mixed with the dry signal from the oscillators. If you don’t have a sustain pedal, or you want to hear the chromatic reverb all the time, change the switch to REVERB.

The orange LED switch turns the SYMPATHISER on/off. When turned off it doesn’t use CPU. It’s quite CPU hungry due to all the processes running to achieve the effect, so only use it if you need it!



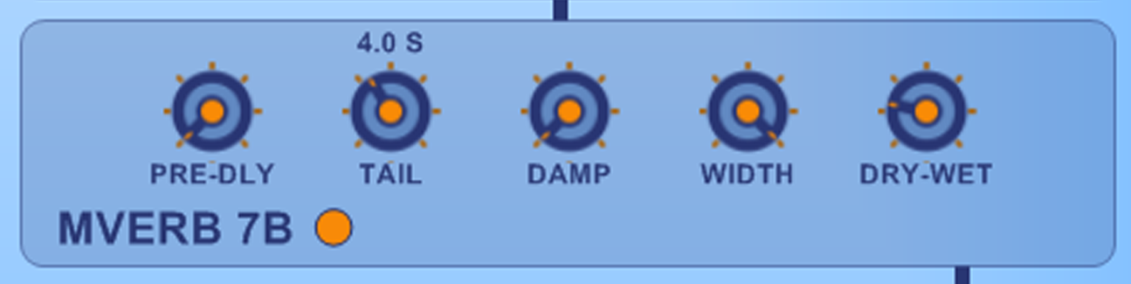
The summed signal is then passed to a graphic equaliser for further shaping the timbre and frequency response across the key span (and beyond!). If you don’t need it, turn it off with the orange LED switch.

When you pass the mouse over the adjustment bars you’ll see a readout of the centre frequency of each bar. These frequencies are chosen to match the key span in half-octaves. So, for example, one of them is set to 262Hz which is roughly Middle C.

When you click and drag on a bar you’ll see a readout of the dB offset in the range of +/- 18dB. You can also “draw” the frequency response by dragging the mouse over the bars in one sweep. A right-double-click will reset all the bars to 0dB. If you wish, you can automate individual bars in the DAW.

The Q knob sets the bandwidth of *all* the filters. When set to 3, the response is pretty much flat when all the bars are at 0dB. Turning it higher increases the colouration of the signal.

If you set a “peaky” response curve you can offset the overall gain of the eq using the LEVEL knob.



The MVERB 7B uses a wonderful reverb engine created by Martin Vicanek and I think is of very high quality. If you prefer to use your own favourite, just turn it off with the orange LED switch and it won’t use CPU.

The knobs do what you would expect of a digital reverb:

PRE-DLY is a delay time before the reverb is introduced.

TAIL is the length of the reverb tail; the time it takes to drop to -60dB.

DAMP reduces the high frequency content during the decay of the tail. It simulates a space which behaves this way.

WIDTH controls the width of the *reverb* in the stereo field. Not to be confused with the WIDTH control for the oscillators (in the GLOBAL panel).

DRY-WET sets the balance between the incoming signal and the reverberation signal.