The Quilcom ASS: **A**nalogue **S**ounding **S**ynthesiser

Design

I became fascinated by the never-ending discussions about whether analogue synthesisers sounded “better” than their digital counterparts. So I decided to design a plugin synth using the knowledge and experience I got back in the seventies, when I made various synths and effects using analogue electronics. Added to this was further insight gained from extensive reading of related subject matter, regarding vintage analogue gear and its emulation.

I decided it would be fun and relevant to include a switch whereby one could select between Digital and Analogue settings, both to have the increased choice and also for A/B comparisons. I’ve provided a trimmer panel where many of the Analogue related settings can be adjusted and selected.

Then I got to thinking about what counts as “Digital” for comparison purposes. Initially I was intending to use good quality anti-aliasing oscillators, filter and so on and just add the quirks and limitations from the analogue days. But of course Digital technology underwent many improvements over the years and so the comparison would not be complete. To this end I’ve provided optional digital quirks and limitations such as bit depth, aliasing and DAC errors etc. The trimmer panel includes these options, the default being “clean”.

If you set the digital side to include all or many of the artefacts of the early digital era, and then compare with the full Analogue emulation, you can get a better feel for the initial opinions of early digital adopters who often described the digital sound as “harsh” or “cold”. Much to my surprise I find the Analogue side “warmer” and more rounded somehow. Judge for yourself.

Then I got to thinking about effects. Initially I’d planned to have none, but soon realised that effects were also subject to the analogue to digital transition, so it seemed appropriate to add some classic effects, most of which can be switched between Digital and Analogue emulations individually.

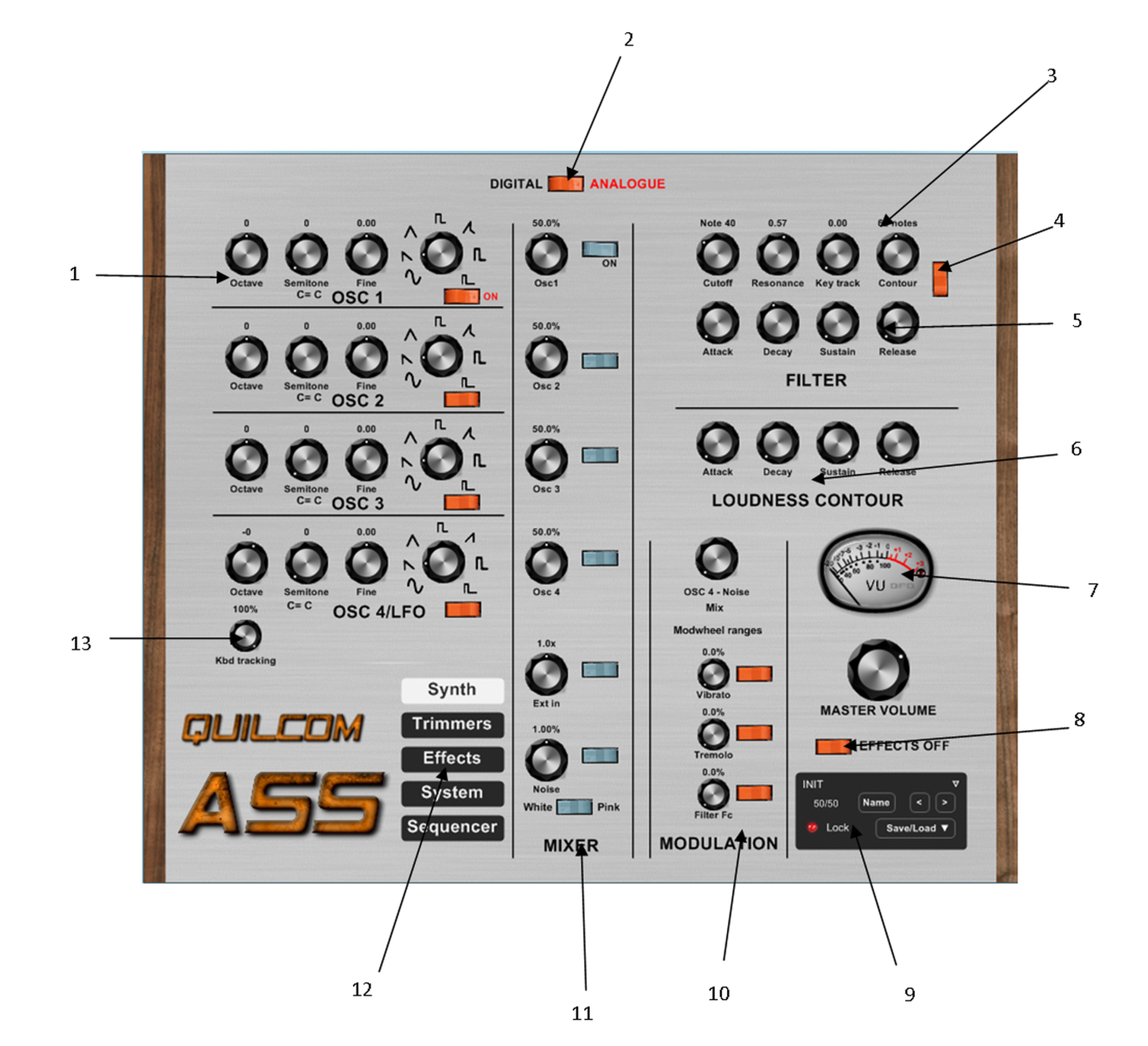
For the topology I chose to lean heavily on the design of the most famous commercial analogue synth for musicians; the Minimoog. It was very tempting to add all sorts of more modern improvements and options, but for the most part I resisted. For the few who may not know too much about this remarkable early instrument I’ve included the operator manual for the Model D and its more recent re-issue. Robert Moog made some very wise choices about what was important for performance and range of possibilities, and what to leave out. As Rick Wakeman said, every knob and switch does something useful.

What follows is a breakdown of the various panels’ control functions and a bit more background to some of them.

You can watch the demo video on YouTube:

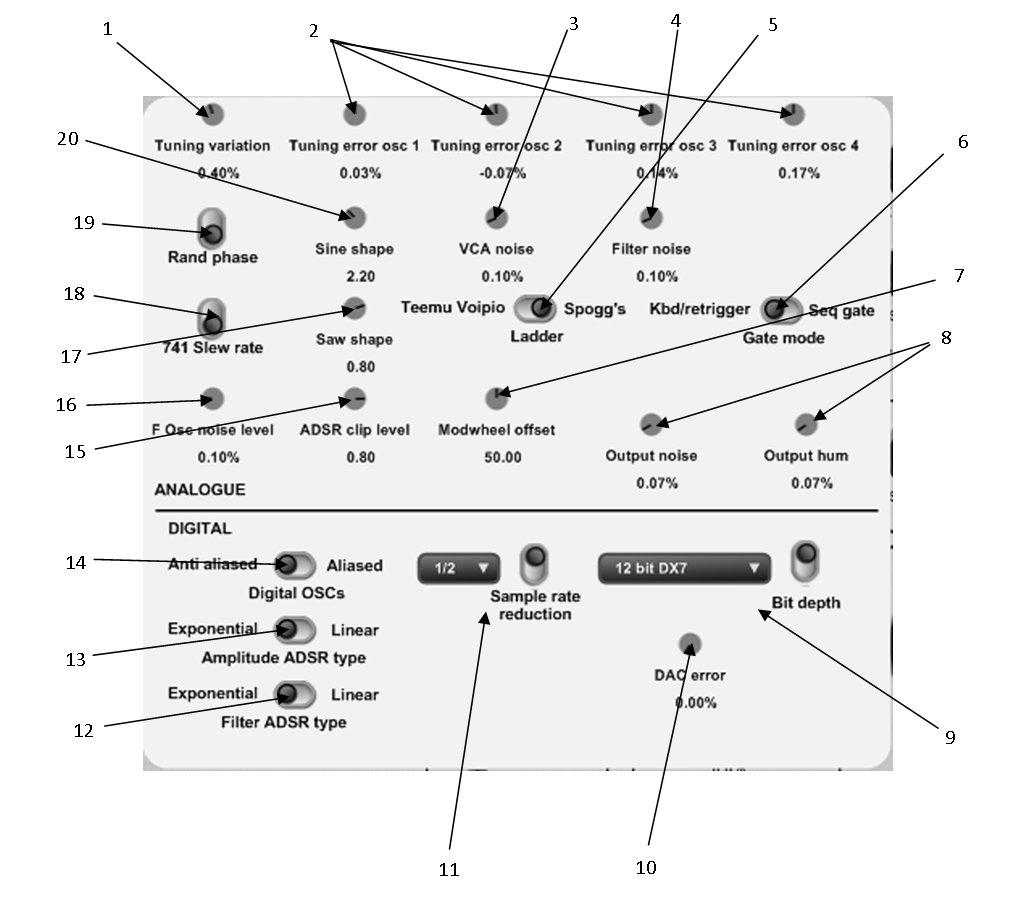
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Main front panel “Synth”



1. There are 4 oscillators provided. 1,2 & 3 are identical. Oscillator 4 can be used for modulation duties, as an LFO or audio rate source. This is the same type of function as Osc 3 on the Minimoog. On Osc 1-3 you can set the pitch range individually. The Semitone knob has an indication of what C3 (middle C) becomes when transposed, which I find very helpful for setting the interval I want. All Oscillators have an orange rocker switch to turn them on, that is to say, to start using CPU cycles. Leave any unwanted ones turned off. The waveforms available are identical to those of the Minimoog and they all sound quite different. In addition there is a sinewave. Osc 4 has an alternative Ramp wave instead of the Tri-Saw (“Shark’s tooth”) since you may want to ramp up or down for modulation effects. Additionally Osc 4 has a keyboard tracking knob (13) (just an on/off switch on the Minimoog). This allows it to operate as a fixed frequency LFO or audio pitch, or can be made to increase the rate for higher keys to the degree you choose. When the Kbd tracking knob is at zero the pitch/rate set is relative to MIDI key 0.
2. This orange rocker switches between the Digital and Analogue emulations. Note that operating this switch creates a reset, so any note or effect sounding will be turned off. This is because some parts of the system need to be in a defined state when started in either mode.
3. This section is the audio filter. It can track the keyboard quite accurately. The Key track knob gives the same options as those provided on the Minimoog by the 2 rockers on the original. The Contour knob controls the amount of effect of the ADSR below, which is dedicated to the filter.
4. This orange rocker can turn the filter off, so it uses no CPU, and can also help with auditioning the effect of the filter.
5. This is an ADSR type envelope generator. Note that I have matched the calibration between Digital and Analogue but there may be some differences for some settings. The Analogue part uses a custom-made envelope generator which emulates the R-C system of the Minimoog. The threshold for Attack to Decay settings is 9.3/5 times the full scale. This gives a somewhat convex curve on the attack phase identical to the Minimoog. In addition the Minimoog is known to have a short flat top at the end of Attack and this is reproduced and can be trimmed (see Trimmers later). This flat top accounts for some of the so-called “punch” on percussive sounds and is akin to a short Hold time on AHDSR generators. On the Digital side you can choose, on the Trimmer panel, between an exponential ADSR and a linear segment one.
6. This is the Loudness Contour (I love those old-fashioned terms!) which controls the amplitude of the sound, and the ADSR method is the same as the filter ADSR above (5).
7. The Volume Unit meter is at the output of the synth. The response is a fast-rise rms with a slow fall. However, peak clipping can get missed so there is a red clip LED which holds the clip indication for 1 second.
8. The whole Effects section can be turned off here if not required, or used for auditioning to allow tweaking of the Synth sound with/without any effects. I found it best to get the basic sound right first, and then bring in whatever effects are wanted.
9. This is the preset manager module. The Lock button allows you to tweak sounds without affecting the original preset, so removing the need to re-load a preset if you get lost or don’t like your adjustments. If it’s off, the DAW will retain your settings in the song. The Synth is reset when changing a preset to avoid unintended noises.
10. The Modulation section can modulate the Synth’s pitch (Vibrato), Amplitude (Tremolo) or the Filter cutoff frequency. Each can have the amount set and be turned on or off by the orange rockers, to gauge the result. The source of the Modulation is always OSC 4 and this can have noise mixed in. As in the Minimoog the MIDI modulation wheel sets the amount of overall transmission of the mixed modulation signal. This can be set with an offset (Trimmer panel) which allows for a minimum modulation amount or to ensure the preset sounds right with no Modwheel present. Note that on the System panel the Modwheel can be disabled for bringing in vibrato from the system panel, should you wish to isolate the effect.
11. The Mixer section combines the sound sources from OSCs 1-4, noise (white or pink) and external input. This mix is then fed into the filter. If you have no external input to process then turn it off with the pale blue rocker, to save CPU. For the Oscillator sources the pale blue rockers have a soft on/off so can be used live or in automation mode without clicking. If turned off in the Mixer only, the CPU for the Oscillators is still used unless the Oscillator *itself* is turned off.
12. These buttons select the panel to be viewed as per their text. Clicking on Synth will remove any other currently viewed panel.
13. This sets the amount of keyboard tracking just for OSC 4. See also (1) above.

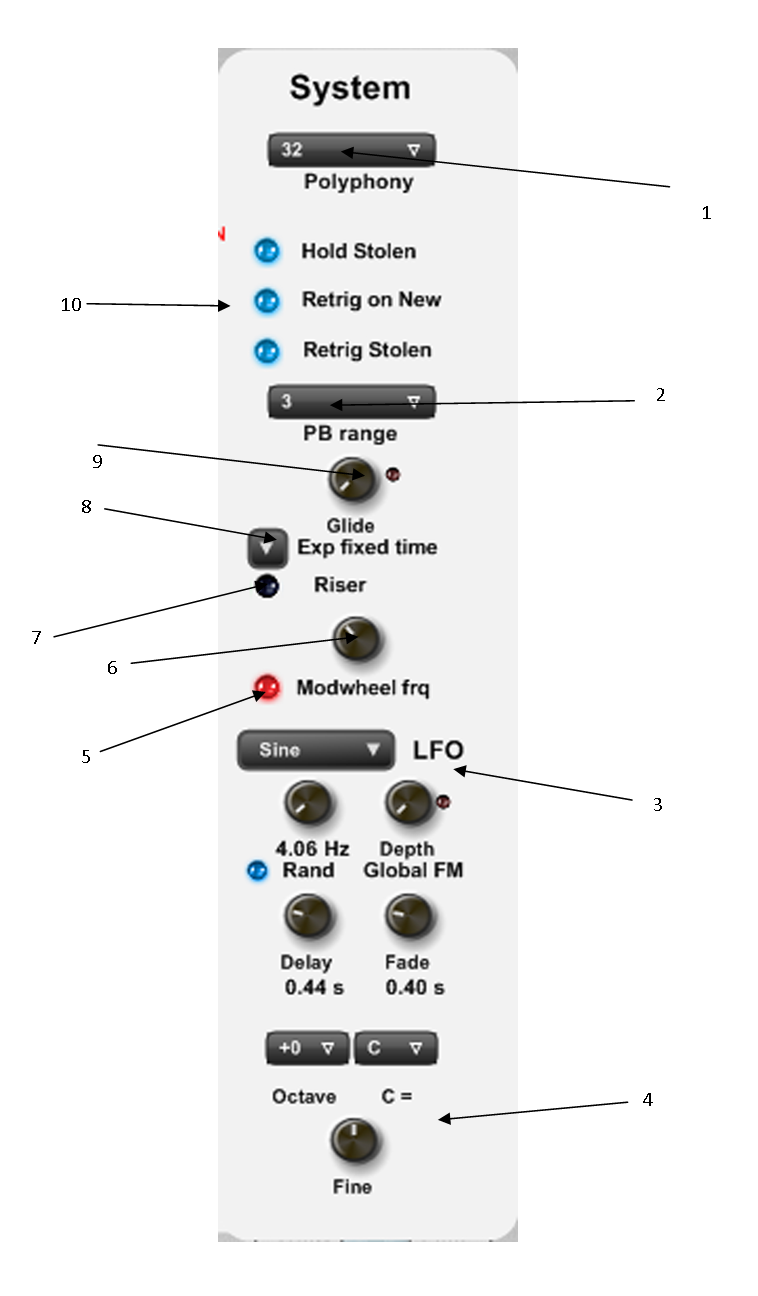
Trimmer Panel



The Trimmer panel allows for detailed setting of Analogue options (upper section) and Digital options (lower sections).

1. A true analogue synth’s oscillator is free running, so when a key is pressed it can happen at any point of a single cycle. Since all analogue oscillators free run, and are not normally in sync, every time you play the same chord the sound will be slightly different dependant on where you catch the waveforms. This is emulated by setting a unique random phase for every key press for all oscillators (enabled by 19). In addition you can set a random tuning variation relating to this random phase with (1). This should normally be a very subtle variation.
2. These trimmers set a tuning error for each OSC 1-4. As is well known, keeping analogue oscillators in tune can be a nightmare, mainly due to the thermal sensitivity of the transistor-based exponential converters. For absolute Fine tuning offset, for beating effects, this is set on the main Synth panel for each OSC. The trimmers here set a parameter that affects the *scaling*, or upper and lower calibration points often only accessible through a hole on the rear panel, if at all. For these trimmers a positive value gives a “stretch” tuning (a term from piano tuning). In this case the keyboard’s lower end will become more flattened and the higher sharper. The centre point is C3 (middle C) where there is correct tuning always. Conversely, if the amount set is negative the high end is flatter and the low end sharper. In this way you can emulate typical VCOs which are never perfect across the whole key range. This can be down to component ageing, poor calibration or imperfect matching (resistors are usually no better than +/- 1% in any case).
3. All real electronic circuits produce noise for various reasons. The most prominent is called 1/f noise, otherwise known as Pink Noise. This trimmer controls how much is injected into the audio stream into the VCA which the Loudness Contour controls. It’s not affected by the filter.
4. See (3) above but this is injected into the filter input, so it’s filtered according to the filter settings. Note that the Ladder Filters will self-oscillate only if there’s at least a little noise present, maybe even less than is easily audible. In an electronic filter this isn’t an issue because there is always some noise present, and this is why the Loudness Contour is wired *after* the filter in the signal path, to improve the S/N ratio.
5. If the Oscillator is the heart of a synth then the Filter is the soul. The Minimoog and Moog modular used what’s called a transistor Ladder Filter and this gave a much sought after sound. There’s lots of information on the internet about how this works and variations on a theme. I’ve provided a switch to choose between 2 types. Both give the appropriate behaviour, namely -24dB/Oct Low pass, decreasing level values below Fc as resonance increases and the ability to self-oscillate. In addition a small amount of non-linearity is provided to emulate the transfer characteristics of a transistor element. For some reason that eludes me I find mine sounds a bit better.
6. This switch only affects the Moog type ADSR (it’s in the Analogue section of the Trimmers). When Kbd/retrigger is selected the ADSR is held on while the key is pressed and will re-trigger from the sequencer. When Seq gate is selected the whole ADSR cycle is controlled by the sequencer gate alone. Which is more suitable will depend on the ADSR settings, the sequencer timing and speed.
7. If you are using the Modulation section the mixed modulation signals are controlled overall by the keyboard’s Modwheel. With this trimmer you can add an offset to simulate the Modwheel being partially opened. This can be useful if you don’t have a Modwheel or if the preset doesn’t sound good with a Modwheel set at zero.
8. In the old days when you turned up the volume to 11, it was normal to hear a bit of hiss and hum from your system. These 2 trimmers allow you to re-create this and the feed is directly to the output. I must say that I felt a little burst of nostalgia for this sound when I first added it.
9. (9) to (14) are trimmer controls that only affect the sound when Digital is selected. (9) allows you to set an emulation of the output DAC’s bit depth which quantises the sound according to the number of discrete levels a DAC could produce. For example, 8 bits can describe only 256 individual levels, so sounds from very old digital systems can be emulated.
10. Very early DACs were not perfect and could suffer from relatively poor monotonicity (every step in one direction should move in that direction by an equal amount). This trimmer allows you to emulate the effect of small errors in the MSB, 2nd MSB and 3rd MSB. This type of imperfection creates high frequency aliased spectra so is more audible on purer sounds, like sine waves, and is most apparent when the volume is changing as a sound decays slowly. I used to hear this on my old DX7 if I auditioned a “pure” low frequency sine wave on headphones, in preparation for making a Bass preset.
11. This drop-list menu allows you to audition the effect of lower sample rates and provides temporal quantisation. Any digital system must use a reconstruction filter to reduce frequencies above half the sample rate and this is provided. The effect is a much reduced brightness with some distortion present.
12. You can choose between a linear or exponential type of ADSR for the filter.
13. You can choose between a linear or exponential type of ADSR for the Loudness Contour.
14. Early digital oscillators (DCOs) used a simple technique based on a counting rate relating to the frequency called for. Each cycle would be at slightly different amplitudes due to the enharmonic relationship between pitch and count increments. This manifests as additional spectra, especially noticeable at higher frequencies, where the cycle length is shorter. (14) allows you to choose between Martin Vicanek’s high quality anti-aliased oscillator set and a set based on the “naïve” technique.
15. The Moog envelope generator used a resistor-capacitor (RC) system to create the “Contour” or envelope. Since the switching from Attack to Decay involved a finite time this gave a small flat top to the envelope shape which only had a small audible effect on fast percussive sounds. This is emulated by providing clipping to the maximum excursion. This trimmer sets the level at which this clipping occurs and leads to a short hold time between Attack and Decay. The effect of this is often described as “punch” since this short dwelling on the maximum amplitude maintains a transient at maximum level for a short time. For slower settings the effect is negligible.
16. As mentioned before, all electronic circuits produce some level of noise. This trimmer injects a small amount of noise into the synth’s frequency control to emulate what an exponential converter and keyboard circuit might produce. It should ideally be set to a point just below where pitch jitter can easily be heard, for the best results.
17. Older electronic oscillators relied on generating a sawtooth by charging a capacitor then rapidly discharging it. This gives rise to a part-exponential rise time shape and this preset can adjust the rise-time shape. The effect is subtle due to the high frequency spectra masking the effect of the ramp shape.
18. A Digital system can change full scale in as little as 1 sample but older electronics, especially those circuits using 741 operational amplifiers, couldn’t achieve such a fast transition. The op-amp specs referred to the maximum rate of change as slew rate and for the 741 it was 0.5 V/uS. So for a 10Vp-p swing this delay becomes significant and gives rise to a limitation of high frequencies, most noticeable with pulse and sawtooth waves which have high and fast transitions. This is emulated by an averaging system and can be turned off and on to audition the effect, which is best heard on the sawtooth waves. It subjectively creates a “warming” effect when enabled.
19. Mentioned in (1) above, this allows selection of a random phase for every note-on event to emulate free-running analogue oscillators. Many thanks to Adam Szabo for providing the technical means to achieve this. Note that if you would like OSC 4 to modulate in exactly the same way for every note (when used as an LFO), this should be turned off.
20. For early analogue oscillators, generating an *accurate* sine wave for *all* audio frequencies was a huge challenge, so the popular technique was to shape a triangle wave using a non-linear transfer, normally reliant on a transistor’s base voltage to collector current relationship or the voltage to current law of a silicon diode junction. To provide similar positive and negative going half cycles, it was necessary to use a trimmer to balance the shape. This is emulated and the trimmer can be used to set this balance. The old technique was never perfect so you could always hear some low level harmonics. The trimmer is set by ear; you adjust it with a sine wave selected and set for the purest sound, or whatever you like!

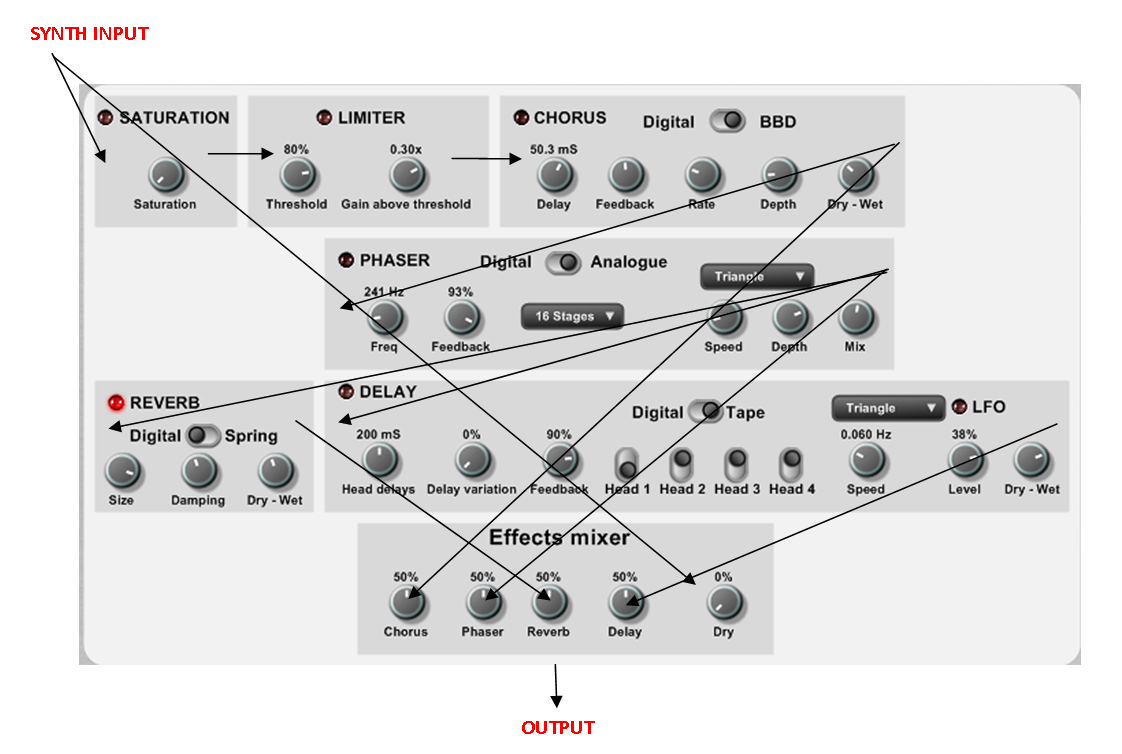
System View



The System view allows settings that generally affect the whole instrument.

1. With this drop-list menu you can set the maximum amount of voices that will sound for polyphonic playing or Monophonic mode for 1 note at a time.
2. This sets the range in semitones for the pitch bend wheel.
3. This is a global LFO. In the Quilcom ASS this is only used for instrument pitch modulation. The red LED by the depth knob comes on when there is a level above zero set. Since the LFO is polyphonic, set the knob to zero if not required, to save CPU. The Blue Rand LED switch provides a small variation of LFO rate for every note pressed, to give a more varied effect for played chords where all the notes are pressed simultaneously. The delay knob holds off the LFO output after a note is sounded and the Fade knob fades in the LFO level. This can be useful for creating delayed vibrato.
4. This is the master tuning for the whole instrument. The C= drop-list menu makes it easier to choose the correct semitone interval (easier for me anyway!).
5. This red LED switch disconnects the Modwheel from the dedicated Modwheel LFO. Please remember that if you wish to use OSC 4 for modulation you may need to disconnect the Modwheel here to get the expected affect. Having the Modwheel Vibrato LFO available means you can use OSC 4 for other purposes than Modwheel-based vibrato.
6. This is the speed of the Modwheel vibrato LFO. This LFO is hard-wired into the whole synth’s pitch system and is always a sinewave.
7. The Riser LED switch is used in conjunction with the glide system above it. This switch turns off the memory of the previously pressed note and ALL notes will glide up from MIDI note 0. This is a fully polyphonic effect and, when combined with a long release time, can sound rather impressive.
8. There are 3 glide (Portamento) types available. Interesting for me is that the Flowstone stock glide type is the same as the vintage type found on the Minimoog. This is exponential fixed time, so a 5 octave glide will take the same time as a 2 note glide and both ranges will slow down as the pitch approaches its destination note. This emulates the R-C delay system found in analogue synths. You also have the options of linear fixed rate or linear fixed time. The fixed rate one will take far longer to glide 5 octaves than 1 octave, and the fixed time one will take the same time whatever the range, but both linear ones will arrive at the final pitch without slowing down as it approaches.
9. This knob sets the glide time or rate. Since the glide is in the polyphonic part remember to turn it to minimum if not required whereupon the red LED will go off and CPU will be saved.
10. These 3 blue LED switches will determine what to do with gating sounds when a note is stolen. Their usefulness really comes into play when Monophonic is set, and even more so when using glide.

Effects panel



The image above shows the signal routing for the Effects section.

The Saturation, Limiter, Chorus and Phaser are connected in series. Any module is bypassed when its red LED switch is OFF.

The Phaser output then feeds the Reverb and Delay in parallel. The main module outputs all connect to the Effects mixer and the dry input can also be mixed in.

What follows is a description of each effect module.

Saturation

The knob controls the amount of drive into a non-linear smooth transfer function and provides distortion based on amplitude. The output level is compensated to give approximately zero overall gain at all settings.

Limiter

This is a 2-stage amplitude limiter. The threshold is the level at which the “Gain above threshold” cuts in. This slope is adjustable from zero for hard clipping to 1x which gives no limiting. This module can therefore provide anything from subtle distortion to clipping distortion for a fuzz box sound. There is also output level compensation to cover most of the range of settings.

Chorus

This is a conventional mono chorus effect which has a wide range of capabilities to include delay and flanging.

When set to BBD mode it emulates the sound of a vintage Bucket Brigade Device system.

Phaser

I used to think that phasing was the same as flanging but it’s quite different. A phaser uses all-pass filters to phase-shift different frequencies by different amounts. When combined with the dry signal this makes a comb filter with entirely different sounds and characteristics to flanging. Flanging delays the signal *equally* across the spectrum and mixes the dry back in.

The quality and depth of the phasing effect is heavily influenced by the number of all-pass filters in the chain. You get a very subtle effect with just one stage and a more pronounced effect with more stages. Here you can choose the number of stages up to 16 to hear the difference. In addition you can adjust the centre frequency of the stages and the amount of feedback to exaggerate the sound. Feedback causes the notches to have a greater depth and to create peaks in the spectrum.

On the right of the module is a simple LFO to modulate the centre frequency for the classic sweeping sound. The mix control mixes the dry with the phase-shifted audio to produce the notches to the degree you desire.

Reverb and Delay

The Reverb and Delay modules are connected in parallel since this is the topology used in the very popular and successful vintage RE-210 Roland Space Echo.

Reverb

When switched to Digital mode the reverb is based on the classic Freeverb design optimised by Martin Vicanek. This is still in the signal path when in Spring mode, to provide controllable diffusion. In Spring mode the behaviour of 2 different springs is emulated using two separate 200 hundred-stage all-pass filters with feedback and filtering etc. The spring system is then fed into the Freeverb. Thanks again to Martin, this was possible due to his optimised filter blocks which keep the CPU down to respectable amounts.

The signature spring “chirp” can be best heard with the “Size” set to minimum, the Damping to maximum and a short transient burst of noise is set on the synth.

Delay

The delay was inspired by the RE-210 as mentioned above, but it’s not an exact emulation.

There are 4 tape “Heads” available which can be switched on individually and are wired in parallel. The Head delays knob sets the base delay time for all the heads which operate as though they were inline on a tape-based system. So, for example, with a delay set to 250mS and only Head 4 turned on, the delay will be 1 second. With all 4 heads on you’ll get a delay output every quarter second.

The Delay variation knob effectively offsets the base delay setting, so effectively the heads are no longer equally spaced on the tape. This can create some interesting repeat patterns which, with suitable feedback, start to approximate a reverberation with embedded repeats.

There is an LFO to the right side, which can modulate the delay time, to really mess things up and provide weird pitch shifts on the delay sounds.

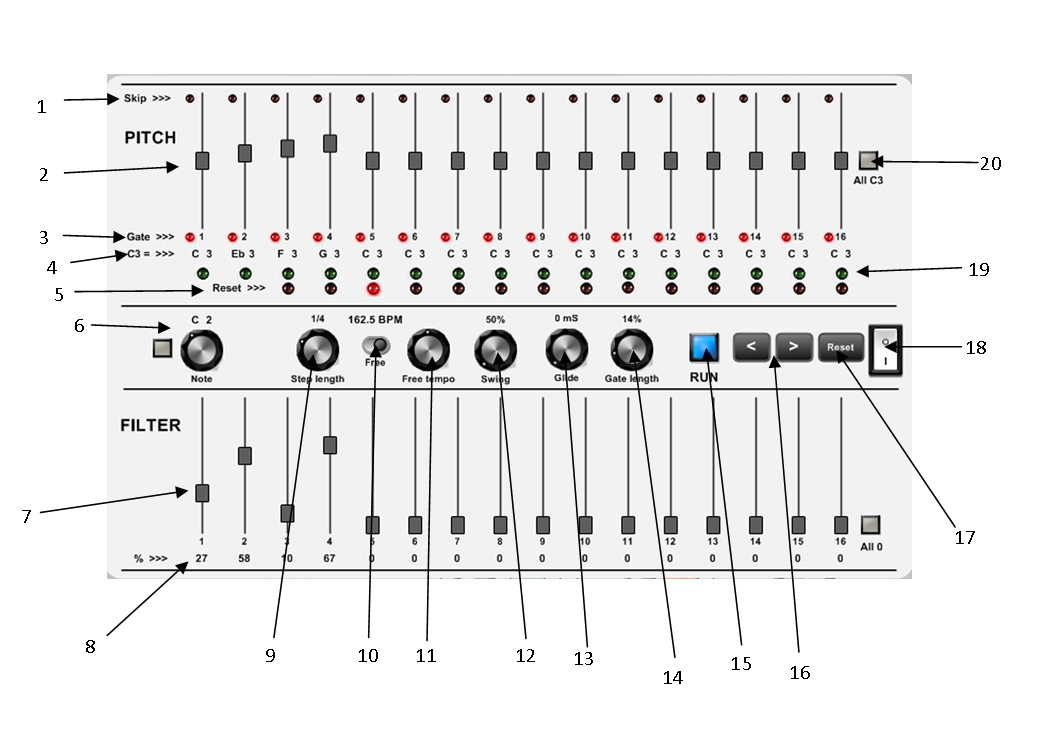
When switched to Tape mode, characteristics of a tape-based system are emulated, including noise, wow and flutter, loss of higher frequencies on repeats and tape saturation. If you turn up the feedback to maximum you’ll hear the classic sound of a tape echo oscillating.

Effects mixer

You can effectively adjust the individual levels of each of the main effects modules here. Be aware that if a module is set to OFF it is bypassed, so its mixer level knob will only control what appears at its input.

Probably the most useful knob is the Dry level knob, so you can mix in the unprocessed sound with the overall effects applied, for example for the early distortion effects to bring in some of the unprocessed sound.

Sequencer View



The Quilcom ASS is provided with a basic sequencer for playing up to 16 steps polyphonically. When turned on, it connects into the system pitch control and filter cutoff frequency control.

The upper sliders (2) set the pitch offset for each step and the lower sliders (7) control the filter cutoff.

When you view the sequencer, if it’s greyed out you need to turn it on with the on/off switch (18).

For the sequencer to run you need the blue RUN switch (15) to be ON and also at least one note pressed. To operate the sequencer you can also turn on and off a note from (6). This note plays the sequencer until it’s turned off again. Set the desired note with the knob (6).

If you want to manually step through the stages, turn RUN (15) off, play or set a note ON and then use the left/right keys (16) or reset to stage 1 using the Reset button (17). The row of LEDs (19) will indicate the current stage or step.

The Pitch sliders have a range of +/- 2 octaves and row (4) shows the C3 (middle C) equivalent for each slider. This helps to set the semitone interval. Because there are 49 different settings for the pitch sliders it may help to hold the Shift key while dragging up/down. A Cntrl+left click will reset a slider to the C3 default. All sliders can be reset to default using the All C3 button (20).

Below each pitch slider is a red LED switch (3) which determines if that stage generates a gate signal. You can use these to create gaps in the sequence since the ADSR won’t be triggered.

The reset LED switches (5) determine when the sequence goes back to step/stage 1. For example, if you want a 4 step sequence then you turn on the Reset LED below step 5. Then when this is reached the sequencer goes back instantly to the start and no gate is created. If more than 1 reset LED is lit it will be the left-most one that resets. By switching between 2 reset positions you can control the length in real time.

Each stage has a Skip option (1) which allows that stage to be bypassed and no gate is generated for that step. You can “play” the sequencer in real time by switching the Skipped steps on and off during the stepping.

The timing or speed of the sequence is set by 3 controls (9, 10, 11). Switch (10) allows you to set a free speed on the Free tempo knob (11) or use the BPM from the host DAW for synchronised playback. The Step length knob (9) sets the length of note per beat, as a ratio. When set to 1/1 at 120 BPM each note is 1 bar (in 4/4 time signature) and the clock rate is 0.5Hz.

You can set the desired amount of swing timing for a more jazz or blues feel with the Swing knob (12). I adopted the Roger Linn system whereby 50% is straight timing and 100% doubles the first beat and halves the second and so on. This gives a very wide range but I reckon around 62% sounds good.

If you want to slide one pitch into the next use the Glide knob (13). This is independent of the System Glide function. The effect will vary with the run speed and if set too long you’ll lose the discreet individual pitches and the sound will be more like LFO modulation of pitch. The glide works on a fixed time exponential algorithm.

The Gate length knob allows you to reduce the gate period. On the Moog 960 module this was fixed at 90% of the step length but here you can set from zero to 100%. This allows for shorter complete ADSR cycles to be repeatedly triggered, assuming the ADSR is set accordingly. For some ADSR settings the gate length will have little effect. Keep in mind that the step time, gate length and ADSR settings all interact in effect.

The Filter sliders (7) mentioned above, simply *add* to the filter cutoff settings on the filter. This means that you need to keep in mind that all filter cutoff sources add up, so you can quickly get to the point where the filter is fully open most of the time. If you have a high manual Cutoff setting and a high sustain on the filter ADSR the sequencer sliders will have little effect.

If you want to create a good range of filter values for the sequencer, start with the manual front panel Cutoff settings quite low or even zero, and then use the filter sliders to adjust the sound.

The bottom % row (8) indicates the percentage of full excursion of the sliders and is not absolutely related to the Cutoff. However this indication may be helpful to ensure sliders can be set to identical values, should this be desired.