Tera Pro

Modular Analog/Digital Polysynth

VirSyn Software Synthesizer

Introduction

TERA PRO – THE MODULAR POLYSYNTH

Tera Pro brings together all major sound synthesis methods on one table. No matter if you want to recreate sounds from analogue legends or explore modern digital sounds with FM, Waveshaping or Wavetable synthesis. Add some flavour of physical modelling and modules for additive synthesis. Combine them altogether for new sound experiences far beyond your wildest dreams.

Tame this overwhelming amount of possibilities with Tera Pros intelligent user interface showing you only exactly what you need, paired with realtime oscilloscopes and level indicators helping you seeing what's going on. Connect the modules with a flexible routing method without the visual cluttering of real wire cables needed in conventional modular synthesisers.

But all these marketing blurbs mean nothing until you hear the first sounds...

Tera Pro supports AUv3, Audiobus, Ableton Link for integration with other Music Apps and works standalone too.

FEATURES

Modular Analog/Digital Synthesizer

- Modular polyphonic synthesizer with free wiring between modules.
- Mono/Dual/Split Layer sound architecture.
- Analogue synthesis
- Additive Synthesis
- Spectrum synthesis
- FM / Waveshaping / Physical modelling
- Vintage voice modelling for true analog warmth.
- Unison mode (up to 8x)
- Monophonic or polyphonic with up to 128 voices.
- More than 1800 factory presets, unlimited user presets.
- Complete effect rack with everything you need to finish the sound.

Live

- Play melodies live with the onscreen keyboard. Drag fingers for slides and vibrato.
- Use a CoreMIDI compatible MIDI Keyboard to control Tera Pro.
- MIDI Polyphonic Expression (MPE) supported.
- Select from dozens of scales and play them with scale optimised keyboard layout.

Programmable Arpeggiator

- Uses programmable sequences with up to 32 steps.
- Can trigger single notes and chords.
- Unique randomiser generates Arpeggios with 100% usability.
- 32 Arpeggios included, unlimited user arpeggios possible.
- For each step you can program tie, accent, transposition and note order

Global

- Audiobus compatible (Output).
- AudioUnit AUv3
- Audio recorder for exporting live sessions
- Support for virtual MIDI input connections.
- MIDI Learn function for parameters.
- Exchange user presets with File Sharing.

Basics

The user interface of Tera Pro is quite simple and is divided into three different main screens:

- Synthesis page for sound generating and processing modules
- Arpeggiator page
- Effects page

Top bar

Functions from left to right:

- **VirSyn** Logo opens About box (and can be used as MIDI Panic button)
- **TERA** – Switch to Synthesis page.
- **ARP** – Switch to ARP page for Arpeggiator settings
- **FX** – Switch to Effects page
- **Patch name** – tap to open patch selection dialog
- **Left**/**right** buttons to scroll through presets
- **Layer selection**: **A**, **B**, **A+B**, **A|B** for Layer and Split sounds
- Display for number of active voices and cpu load
- **Dice** – tap to generate a new random preset.
- List of recordings already made
- Recorder – Start/Stop audio recording
- **?** – open help overlay for current page.

INTERFACES

KEYBOARD

Tap on the gear icon to open the keyboard menu:
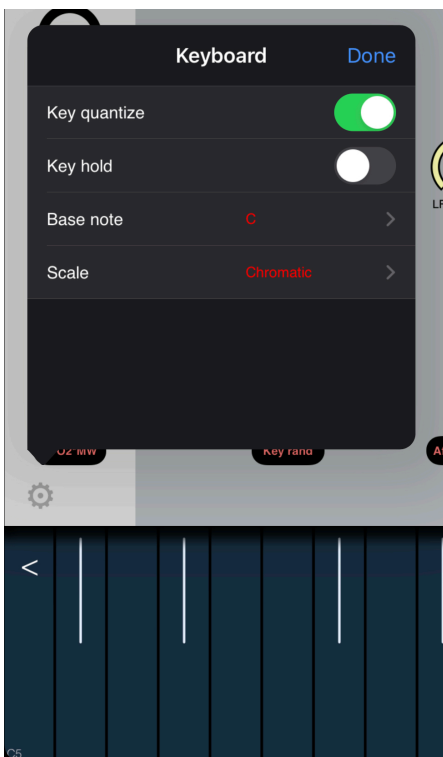
- **Key quantize** – If Key Quantize is set: swiping over the keys is constrained to exact tuning (Glissando), otherwise pitch bends continuously (Glide).

- **Base note / Scale** – select the key and the scale used by the onscreen midi keyboard

- **Keyboard range** – Tap to select keyboard range, double tap to size the keyboard.

- **Key hold** – If set all keys tapped remain on until tapped again. Use this to hold notes while designing sounds.

SCALE

If Scale is set to another value than 'Chromatic', notes played on the live keyboard or entered in the sequencer 'snap' to notes allowed by the scale set. The following scales are available:

Chromatic	Chord Major	Algerian
Diatonic Major	Chord (m)	Arabic
Diatonic minor	Chord (6)	Bali Island Pelog
Pentatonic Major	Chord (7)	Byzanthine
Pentatonic minor	Chord (maj7)	Chinese
Blues Major	Chord (maj7+5)	Egyptian
Blues minor	Chord (m7)	Hawaiian
Whole-tone	Chord (sus4)	Hindu
Ionian	Chord (7sus4)	Japanese
Dorian	Chord (dim)	Mongolian
Phrygian	Chord (dim7)	Oriental
Lydian	3rd Interval	Persian
Mixolydian	4th Interval	Ryukyu
Aeolian	5th Interval	Spanish
Locrian		

The base note transposes all live played/sequenced notes if set to another value than 'C'.

KEYBOARD

Tera Pro can be played by either the on-screen keyboard or a CoreMIDI compatible Hardware controller keyboard.

The on-screen keyboard seems to miss a pitch bender / modulation wheel but this functions are there in a more touch screen compatible way:

- Pitch bending is done with moving the fingers left/right. (if "Q" is not set)
- Modulation wheel is simulated by moving fingers up/down.

SOUND PRESETS

The presets in Tera Pro are organised in a user definable folder structure.

Tap on the preset name in the top bar to open a popup window with a list of all available folders/presets.

Popup window for presets:

Sharing sound presets:

The presets can be shared with other users as files or zipped folders for complete preset banks. Tap on the share button in the preset list to open the preset sharing dialog:
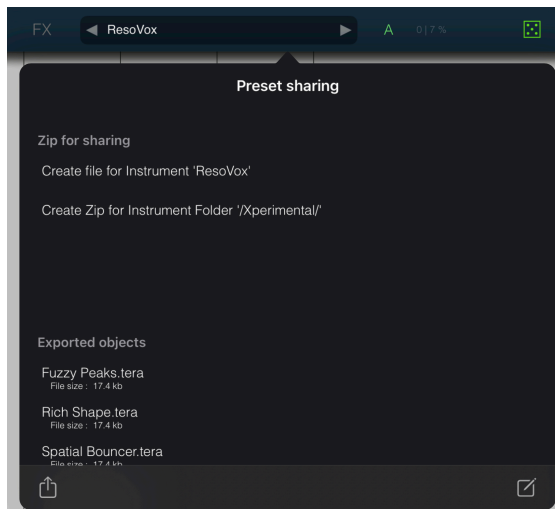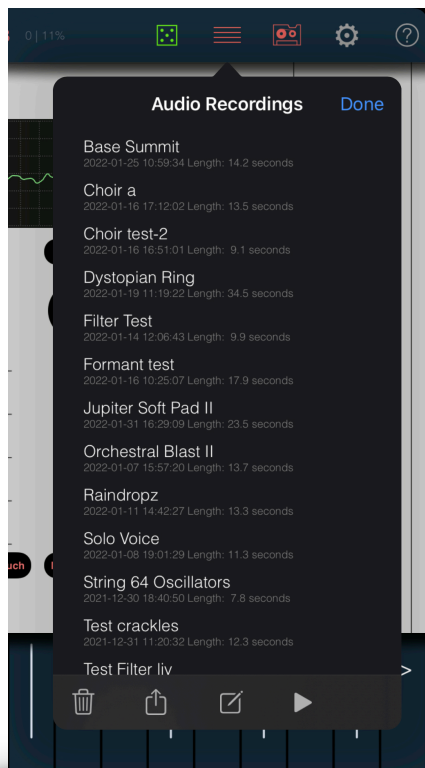
Save preset:

Tap on the Disc Icon and you'll get the following dialog:

Tap on the category you want this sound to be in and name the preset to your liking.

AUDIO RECORDINGS

Tap on the record button (red circle) in the top bar to bring up the Audio Recorder popover which let you enter the recorder settings and you can start recording of audio from here:

Recording starts after a count–in of one bar indicated by the metronome and stops with the next tap on "REC". Start the recording with a tap on the "Record" button.

The recorded audio is saved in a wave file with the preset name set as default file name. You can access the the files in the iTunes File sharing folder for Tera Pro.

In addition the midi note on/off information coming from playing the onscreen or external keyboard is recorded into a file with the same name and the file extension ".mid". If the arpeggiator is active the notes from the arpeggiator are recorded in the MIDI file.

To change the file name just tap on the name field and enter the new file name.

Tap on the right arrow of an existing recording in the list to replay it.

Synthesis

MODULAR SYNTHESIS

The modular synthesizer uses independent sound generating and processing modules. The modules are not hardwired together but can be freely connected together using virtual wires (patch cords).

These modules can be accessed on the "TERA" Page. Please note that only modules really used (i.e. patched to some input of another module) are shown in the user interface.

Tap on a free space in the modules pane and drag left/right to reveal modules not visible. Tap on the module title to expand/collapse the modules user interface.

Common

Master tuning / Portamento.

Oscillator 1/2/3

Main oscillators covering analogue/additive/FM synthesis.

Spectrum Oscillator

Additive/Wavetable Oscillator using an Osc/Filter model.

Wave Shaper

Waveshaping / Wavefolding synthesis.

Ringmodulator

Multiplication of two input signals.

Noise generator

White and Pink noise.

Wave mixer

Mixer 3 + 2 in 1.

2 Multimode filter

Filter with adjustable slope (24/18/12 dB) and characteristic (Lowpass / Highpass / Bandpass / Bandreject).

Formant filter

3 Bandpass or bandreject filter for formant sounds and phasing effects.

Vocal filter
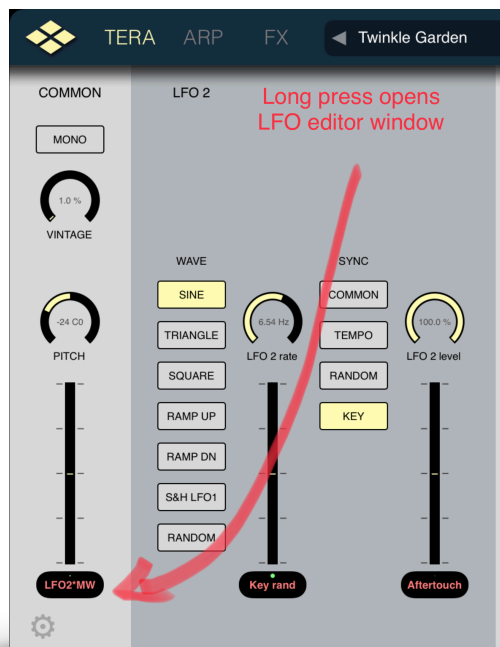
Two morphable 16-pole filter emulating human vocal tract.

Karplus modul

Physical modelling for simulating string and wind instruments.

Amplifier

Final amplifier with overdrive and LoFi effects.

The modulation modules open their UI on demand, this means you have to long press on a modulation source input patched to the modulation module:

4 Multi segment envelope generator

syncable Envelopes with up to 64 breakpoints.

4 Envelope generator

Envelopes with D–ADSR characteristic.

4 LFO

Low frequency oscillator with synchronisation and Sample&Hold function.

ROUTING OF MODULES

To show you the routing of modules we will build a very simple synthesizer: An oscillator with an envelope generator for the volume envelope as the most basic sound modulation.

The routing between the modules can be made on two different levels:

1. Routing of sound output of some module to the sound input of another.

For our example we need one oscillator (we take Oscillator 1, but it can be any) and want the output of this oscillator be routed to the input of the Amplifier module (This module must be used by any patch).

Tap on the input jack of the AMP Module and select "Oscillator 1" from the upcoming list of Sound module outputs:

2. Routing of control signals to modulate sound parameters.

Next step is to route the control signal of an envelope generator – let's take ADSR1 in this example – to modulate the volume of the sound. The Volume parameter can be set with the circular knob right above the label "VOLUME". Luckily the modulation input for the Volume parameter (it's right below the Volume knob) is already prewired with "ADSR 1". If you want another modulation source just press to open a modulation selector. To edit the ADSR 1 envelope press long on the modulation input jack and the ADSR 1 parameter dialog magically opens:
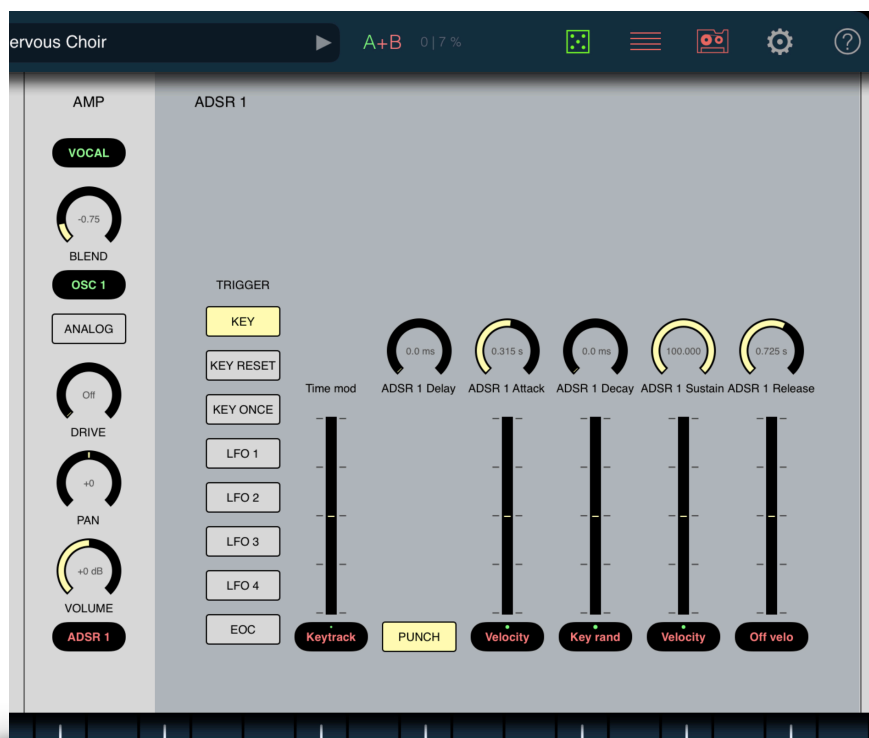
COMMON

The keyboard module provides you with access to parameters influencing the processing of note events coming from the Keyboard or the internal arpeggiator.
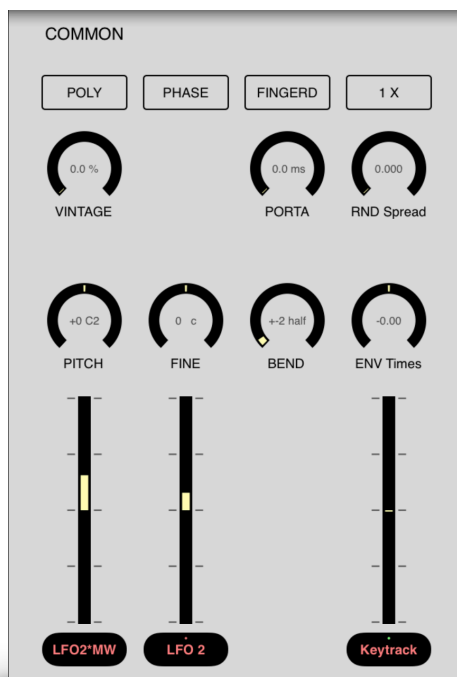
Master Pitch coarse/fine

Sets the absolute pitch for all oscillators and the waveguide module in the range of +/– 24 semitones if tracking is set. Otherwise the absolute pitch of the oscillator is set.

Pitch Mod

Modulate the master pitch with up to two selectable modulation sources. Select a modulation source by tapping into the square field below the slider and set the amount of modulation with the slider.

PORTA

Smooth transition of the pitch between two notes played in succession. The time to reach the final pitch can be adjusted in the range of 0 milliseconds to 70 seconds. This effect is typical for many lead sounds on monophonic synthesizers, but can also be used here in polyphonic mode.

FINGERD

In normal mode all notes played will have the portamento effect. In fingered mode only legato–played notes will have this effect.

MIDI Bend

Adjusts the pitch bend range (up to +/– 48 semitones) applied to the sound if an external MIDI Keyboard with a Pitch bender control is connected to Tera.

Poly/Mono

Set synth to monophonic (only one note at one time) or polyphonic (max 128 notes at the same time)

Phase

With "Free Phase" all oscillators are running freely. This has the effect that the same note value played repeatedly can sound slightly different from each other. This is the preferred mode for most sounds. With "Sync Phase" the oscillators starts synchronously on each note start. Then repeated notes all sound equal. This is the preferred mode for drum sounds. Note that this mode can lead to clicks in the onset of sounds.
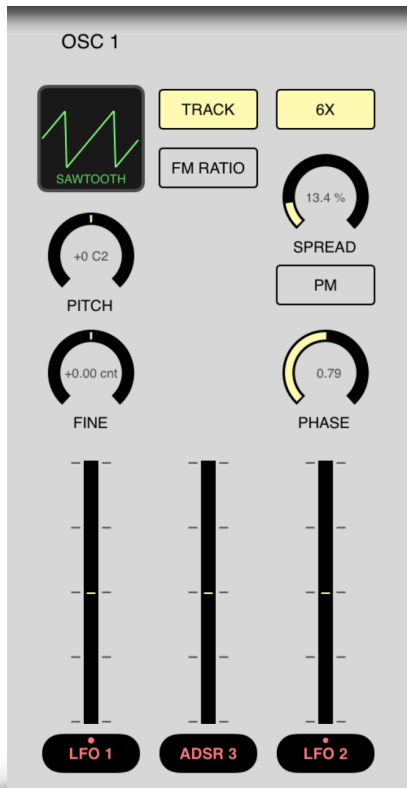
Unison

Select from 1 x to 8 x the number of unison voices (simultaneously played voices per Note) each with slightly different frequency.

RND Spread

Set the amount of random pitch variation on unison voices.

OSCILLATOR 1

Oscillators provides you with the base material for sound synthesis. They generate periodic oscillations of analog and digital waveforms with controllable frequency. The oscillators in Tera use anti-aliasing algorithms to produce waveforms without digital distortion even in the highest register. Nevertheless, Tera's oscillators provide you with an outstandingly fat sound in the lower registers.

Pitch coarse/fine

Sets the relative pitch for this oscillators in the range of +/- 4 octaves if tracking is set. Otherwise the absolute frequency of the oscillator is set.

If FM RATIO button is set, then the pitch can be set in integer multiples of the master pitch. The range is from 1:1 to 1:17. This mode is especially suited for setting the typical carrier/modulator frequency ratios used in FM synthesis.

Waveform

Select the waveform used by the oscillator from a list of 64 waveforms including the classical waveforms sawtooth, square, triangle and sine. If you select "Custom" from the waveform menu a dialog pops up where you can specify the level of up to 256 partials which determine the waveform. To open the waveform list just tap on the module box.

PM - Phase modulation

With phase modulation activated (Button PM) the oscillator's output is added to a phase shifted copy of itself. The phase shift is controlled by the phase modulation knob.

If you select the waveform sawtooth then the result is a pulse waveform with variable pulse width. In contrast to traditional pulse implementations you can use this for all the available waveforms.

Below the Phase knob you find a slider/selector combination to modulate the phase with one of the many possible modulation sources.

Pitch Mod

Modulate the oscillator's pitch with up to two selectable modulation sources. Select a modulation source by tapping into the square field below the slider and set the amount of modulation with the slider.

Track

This button activates keyboard tracking. When switched on the oscillator will track the keyboard at the rate of one semitone for each key. When switched off the oscillator will have a fixed pitch independent of the notes played on the keyboard.

6x - Spread

To make wonderful fat and powerful supersaw like sounds you need several oscillators tuned nearly in unison. Just tap the 6x button and Oscillator 1 will then output six detuned waves concurrently. The amount of detuning between this six oscillators can be adjusted with the "Spread" knob.

OSCILLATOR 2

In addition to the functions of Oscillator 1 this one has a linear FM input for FM synthesis.

Pitch coarse/fine

Sets the relative pitch for this oscillators in the range of +/- 4 octaves if tracking is set. Otherwise the absolute frequency of the oscillator is set.

If FM RATIO button is set, then the pitch can be set in integer multiples of the master pitch. The range is from 1:1 to 1:17. This mode is especially suited for setting the typical carrier/modulator frequency ratios used in FM synthesis.

Waveform

Select the waveform used by the oscillator from a list of 64 waveforms including the classical waveforms sawtooth, square, triangle and sine. If you select "Custom" from the waveform menu a dialog pops up where you can specify the level of up to 256 partials which determine the waveform. To open the waveform list just tap on the module box.

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Below the Phase knob you find a slider/selector combination to modulate the phase with one of the many possible modulation sources.

Pitch Mod

Modulate the oscillator's pitch with up to two selectable modulation sources. Select a modulation source by tapping into the square field below the slider and set the amount of modulation with the slider.
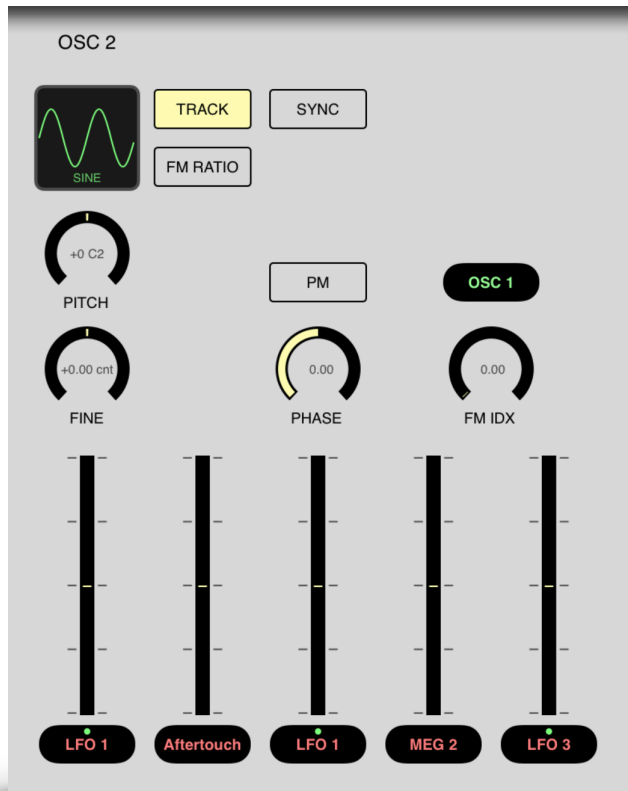
Track

This button activates keyboard tracking. When switched on the oscillator will track the keyboard at the rate of one semitone for each key. When switched off the oscillator will have a fixed pitch independent of the notes played on the keyboard.

FM-Input

With this input you can modulate the oscillator frequency with the output signal of another module, for example a filter or an oscillator. The amount of the modulation, called Modulation index in FM synthesis terminology, can be adjusted in the the range of 1 to 16.

FM-Index

The modulation index can be modulated with up to two selectable modulation sources. Select a modulation source by tapping into the square field below the slider and set the amount of modulation with the slider.

OSCILLATOR 3

Oscillator 3 has the same functions as Oscillator 2, except the phase modulation feature.

Pitch coarse/fine

Sets the relative pitch for this oscillators in the range of +/- 4 octaves if tracking is set. Otherwise the absolute frequency of the oscillator is set.

If FM RATIO button is set, then the pitch can be set in integer multiples of the master pitch. The range is from 1:1 to 1:17. This mode is especially suited for setting the typical carrier/modulator frequency ratios used in FM synthesis.

Waveform

Select the waveform used by the oscillator from a list of 64 waveforms including the classical waveforms sawtooth, square, triangle and sine. If you select "Custom" from the waveform menu a dialog pops up where you can specify the level of up to 256 partials which determine the waveform. To open the waveform list just tap on the module box.

Pitch Mod

Modulate the oscillator's pitch with up to two selectable modulation sources. Select a modulation source by tapping into the square field below the slider and set the amount of modulation with the slider.

Track

This button activates keyboard tracking. When switched on the oscillator will track the keyboard at the rate of one semitone for each key. When switched off the oscillator will have a fixed pitch independent of the notes played on the keyboard.
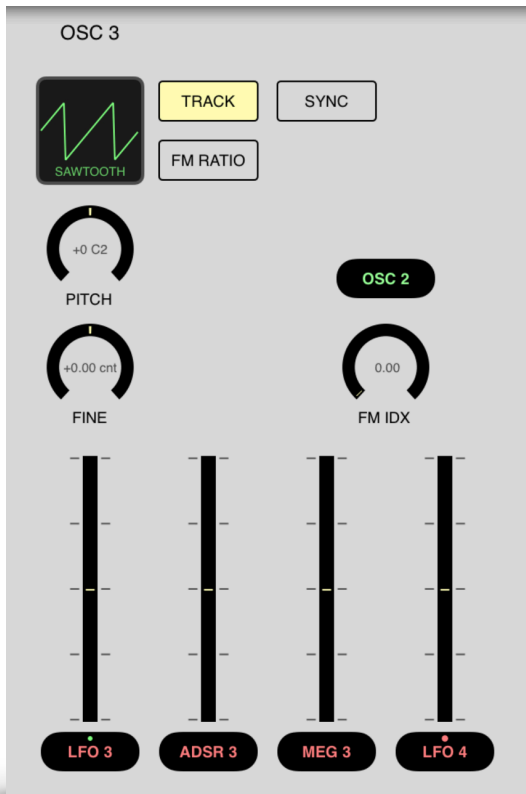
FM-Input

With this input you can modulate the oscillator frequency with the output signal of another module, for example a filter or an oscillator. The amount of the modulation, called Modulation index in FM synthesis terminology, can be adjusted in the the range of 1 to 16.

FM-Index

The modulation index can be modulated with up to two selectable modulation sources. Select a modulation source by tapping into the square field below the slider and set the amount of modulation with the slider.
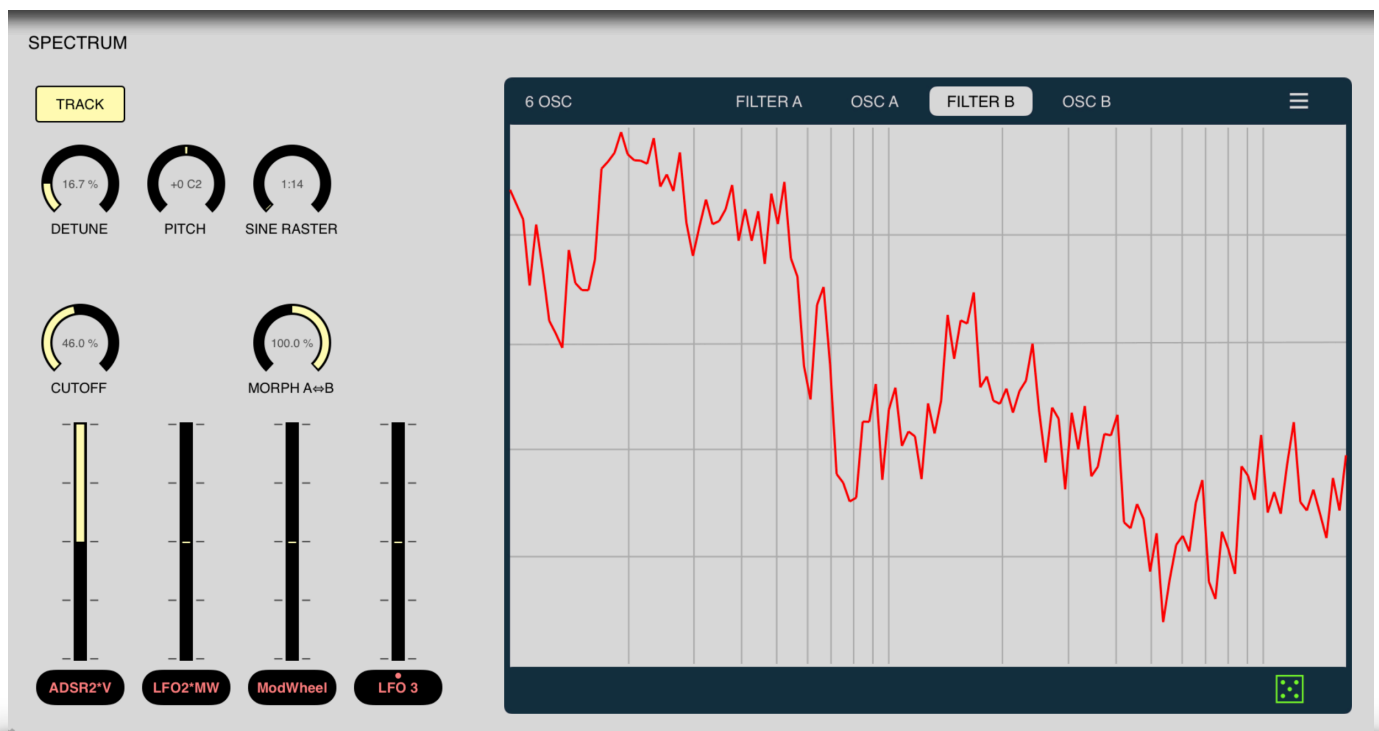
SPECTRUM OSCILLATOR

This module lets you create sounds with the help of spectrum synthesis, which allows the construction of very complex formant structures. Up to six independent oscillators are the basis of this module. It's therefore ideally suited for generating ultra fat or chorus like sounds. By strongly detuning the six oscillators you can generate inharmonic sounds well–suited for simulating (for example) cymbals. The waveform of the oscillators can be defined freely by specifying up to 128 partials. With the Raster parameter the number of harmonics present in the spectrum can be reduced.

Internally this is done through wavetable synthesis and therefore the sound results can be quite similar. The difference is that in this case the wavetables are automatically generated from the user specification of partial tones and filter resonance characteristics which enables a more musical view on this popular synthesis technique.

With its 128 pole dynamic resonance filter the base sound of the oscillators will be filtered in a way that is comparable to the filtering of a resonant body of physical instruments. The Morph function LETS you smoothly blend between two different resonant bodies. The filter frequencies of all formants can be shifted towards lower frequencies with the spectrum parameter.

The spectrum oscillator has two additional outputs for stereophonic effects. These outputs are highly uncorrelated and contain a subset of the six oscillators used by this module.
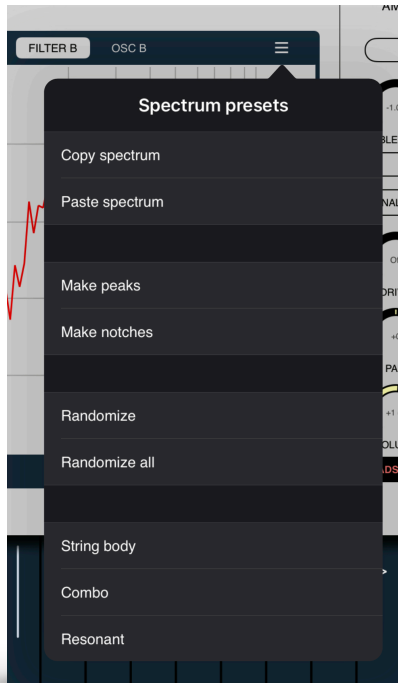
Spectrum window

In the spectrum window you can directly draw the desired filter and oscillator spectrum. Just draw the spectrum by touching inside the spectrum window and moving around. "Filter A" and "Filter "B" selects the two different filter curves for editing.

Touch on the menu symbol at the top right position for more functions:

Copy spectrum – copy spectrum to clipboard

Paste Spectrum – paste spectrum from clipboard

Make Peaks– adds comb filter to spectrum

Make Notches – subtracts comb filter from spectrum

Randomize – Visible Filter/Osc settings are affected

Randomize all – All Filter/Osc settings are affected

Other entries represent filter presets which can be used as a basis for your own experiments.

Oscillator menu

With this menu you can choose among the following oscillator configurations:
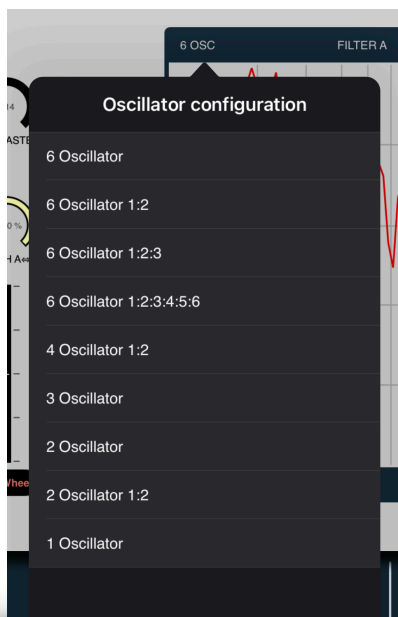
6 Osc – 6 oscs with same pitch

6 Osc 1:2 – 3 oscs with base pitch, 3 oscs in suboctave

6 Osc 1:2:3 – 3 * 2 oscs tuned in ratio 1:2:3

6 Osc 1:2:3:4:5:6 – 6 oscs tuned in ratio 1:2:3:4:5:6 (subharmonic)

4 Osc 1:2 – 2 * 2 oscs tuned in ratio 1:2

3 Osc – 3 oscs with same pitch

2 Osc – 2 oscs with same pitch

2 Osc 1:2 – 2 * 1 osc tuned in ratio 1:2

1 Osc – 1 oscillator

TRACK

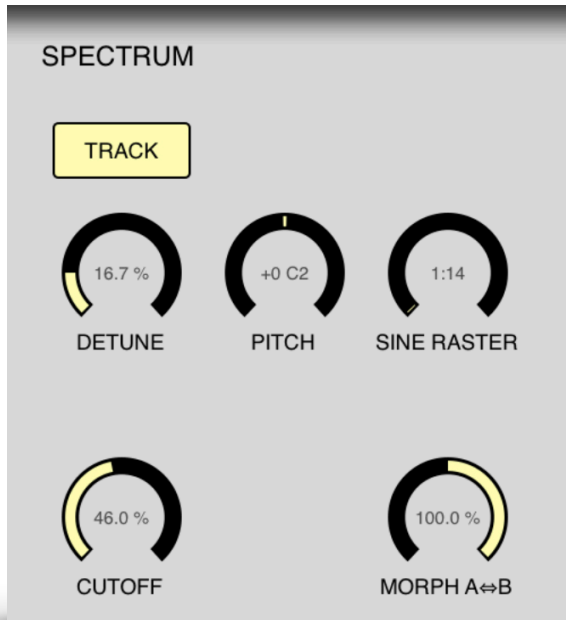This button activates keyboard tracking. When switched on the oscillator will track the keyboard at the rate of one semitone for each key. When switched off the oscillator will have a fixed pitch independent of the notes played on the keyboard.

DETUNE

With this parameter you can set the detuning of the 6 independent oscillators of the multi oscillator in the range from 0.002 to 2.0. Small values lead to complex, chorus like sounds. With increasing values the sound becomes more and more dissonant and bell like.

PITCH

Coarse tuning of the oscillator relative to the master tuning. Sets tuning within a range of +/- 48 semitones.

RASTER

With the raster parameter you can change the number of harmonics present in the resulting spectrum:

1:1 – all harmonics present.

1:2 – only every second harmonic is present (like a square wave)

1:3 – only every third.

1:4 – …

The resulting sounds gets thinner the higher this parameter is set.

CUTOFF

The filter spectrum can be shifted to the lower end using the spectrum parameter. The drawn filter curve is completely active only if the spectrum knob is fully opened. The more this knob is closed, the more the spectrum is shifted to the left and the sound GETS darker.

MORPH

With the morph parameter you can blend between Oscillator/Filter 1 and Oscillator/Filter 2.

Oscillator Synthesis method

On the oscillator pages you can set the method of partial sound generation used to build the wavetables:

– Additive generator (Sine oscillators)

– Wavelet Synthesis

– Walsh function Synthesis

– Operator Synthesis (FM like)

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KARPLUS

This module is the basic building block for physical modelling synthesis. First of all it sums the signals from its two audio inputs (INPUT and FEEDBACK), and is attenuated by the input knob and the feedback knob settings. The sum is then delayed with a pitch synchronous delay time and low-pass filtered with an adjustable cutoff frequency. In the simplest form one can use this module as a delay line with real-time control of the delay time.
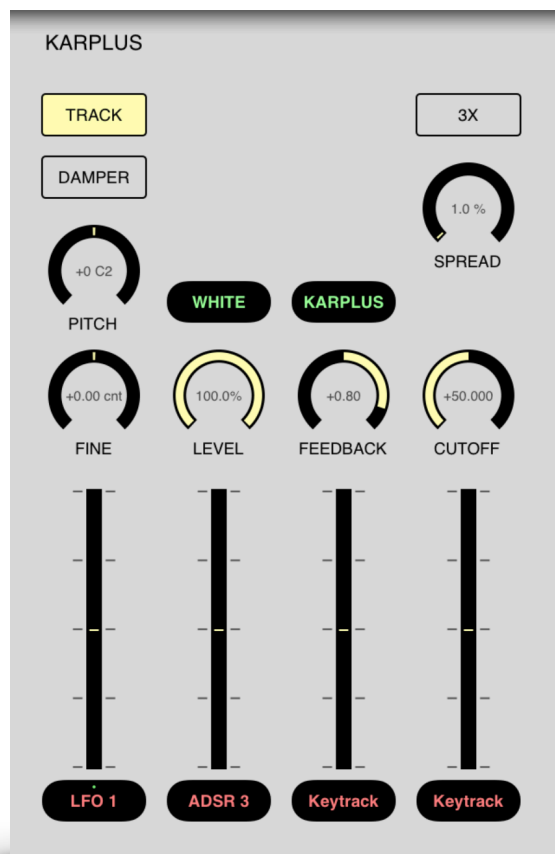
But how can we make a physical model of an instrument with this delay line ?

To simulate the travelling wave of a string or an air tube the output of the delay line has to be patched to the FEEDBACK Input to get a closed feedback loop. This loop needs some initial energy put into INPUT to resonate and thereby produce a sound. To accomplish this, patch some oscillators or the noise generator to INPUT and modulate the input level by a very short attack/decay envelope. The spectrum of this input signal strongly determines the resulting sound in a very interesting manner: the resulting sound is filtered by the spectrum of the input signal !

If, for example, a mix of three sine tones with fixed frequencies are used as excitation sources the resulting output sounds like a formant filter with three bandpass filters having fixed centre frequencies and infinite slopes.

The built-in loop filter (6dB lowpass) corresponds to the damping of high frequencies in the physical model. Due to this loop filter higher harmonics decay much faster than lower harmonics. With the cutoff frequency set to 0 you get the typical sound of plucked strings, like a guitar. For steady state sounds you should set the cutoff frequency to maximum.

In the physical model the source for the FEEDBACK input is usually the WaveGuide output simulating the reflection of the travelling wave on both ends of a string or tube. But you can choose any audio source if you want to use this module for other purposes. The input level is controlled in an unusual manner by the feedback slider. This slider gives you control over the decay time of the sound in the range of 50 ms to 200 seconds. Negative values simulate the travelling wave of a tube with one open end and one closed end resulting in a hollow, square-like sound. In this case the pitch is one octave lower than for positive feedback values.

INPUT

Choose an audio source for the initial energy impulse by clicking into this field and selecting a source from the popup menu.

LEVEL

Adjusts the level for Input 1. Modulate this input with a short attack/decay envelope to simulate the injection of an energy impulse.

FEEDBACK INPUT

Second input for the delay line. The input level can be adjusted with the feedback slider. For physical modelling synthesis the WaveGuide output has to be patched to this input.

TRACK

This button activates keyboard tracking. When switched on the WaveGuide will track the keyboard at the rate of one halftone for each key. When switched off the WaveGuide will have a fixed pitch independent of the notes played on the keyboard.

COARSE

Coarse tuning of the oscillator relative to the master tuning adjusted in the keyboard module within a range of +/- 48 semitones.

FINE

Fine tuning of the pitch in the range of -50 to +50 cent with variable resolution from 0.02 cent to 2.0 cent for one slider increment. (100 cent = 1 halftone step)

CUTOFF

Controls the cutoff frequency of the loop filter (6dB lowpass). Low cutoff values lead to faster decay of the higher harmonics compared to the lower harmonics. In addition the decay time of all harmonics is always proportional to the pitch and influenced by the feedback slider.

FEEDBACK

Adjusts the level for the feedback input. If the WaveGuide output is patched to the feedback input you get a closed loop with the feedback slider controlling the decay time of the sound in the range of 50 ms to 200 seconds

Ringmodulator

RINGMOD

OSC 1

OSC 2
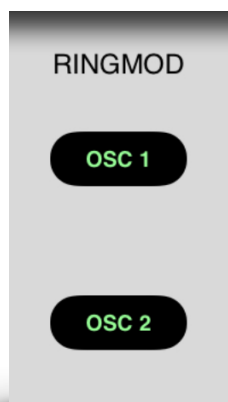
This module combines two input signals by multiplying them. The resulting spectrum contains the sum-of and the differences-between the frequency components of the two input signals. When the frequency of one input signal is below 20 Hz the result is a tremolo effect. But when the frequency is in the audible range the result is a change in timbre. If the pitch of the input signals are in an integer ratio to one another, the resulting spectrum is harmonic, otherwise the spectrum is inharmonic.

Example:

Sine tone with 1000Hz at input 1

Sine tone with 300Hz at input 2

Result on Output: two sine tones with 1300Hz and 700Hz.

INPUT 1

Choose an audio source for input 1 by tapping into this field and selecting a source from the popup menu.

INPUT 2

Choose an audio source for input 2 by tapping into this field and selecting a source from the popup menu.

NOISE GENERATOR

The noise generator provides you with two different types of noise. Use the noise generator to produce sounds simulating wind, ocean breakers or unpitched drums:

White noise

The spectrum of white noise contains all frequencies with equal power. It sounds very bright and sharp.

Pink noise

Lowpass filtered white noise. The lowpass filter has a fixed cut-off frequency of 0 Hz and a slope of 3dB/octave. Pink noise sounds darker and smoother than white noise.

Every voice in Tera has its own noise generator to avoid undesirable correlations when many voices are sounding simultaneously.

There are no user interface elements for the noise generator. The module appears only in the menu of the module inputs as one of the selectable sound sources.

SUBHARMONIC OSCILLATOR

Oscillator 1 has an additional output for subharmonic sounds. The pulse width of this sound is always controlled by the wave modulation slider, even if wave modulation is deactivated. The subharmonic sound is a mix of six pulse oscillators with a frequency ratio of 1:2:3:4:5:6.

This module is inspired by Oskar Sala's Mixtur-Trautonium and the Ondes Martenot, both legendary instruments created in the beginning of the 20th century.

WAVE SHAPER

The Wave Shaper/Folder is a nonlinear processor altering the shape of the input waveform.

Like the index in FM synthesis the input level of the shaper controls the dynamic evolution of the produced spectrum: the spectrum becomes richer as the input level is increased. The shaping function controls the resulting spectrum. This kind of synthesis is also known as Waveshaping.

INPUT

Choose an audio source for input by tapping into this field and selecting a source from the popup menu.

Shaping function

Selects one of the shaping functions. If you select "Custom" an editor pops up on long press which lets you specify the shape for your own. Alternatively you can specify the output spectrum with additive synthesis (HARMONIC) or FM synthesis assuming that the shaper is fed by a pure sine wave.

LEVEL

Adjusts the level of the input signal. Increasing the input level brightens the sound spectrum.

Tip

For typical waveshaping sounds use one of the envelope generators to modulate the input level of the shaper module. This gives you interesting, dynamically evolving timbres.

Use the preset wave shape Buchla 259 to emulate a typical west coast style Wave folder.

MIXER

With this module you can mix down up to 5 input signals to the mix outputs.
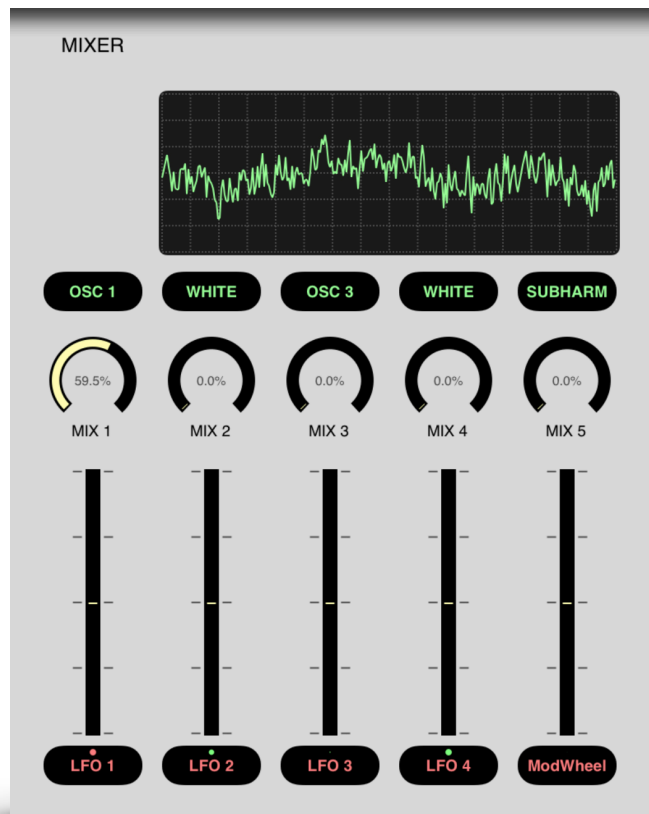
The three outputs of the wave mixer provides you with the following sub mixes:

Mix 1-3

Mix of the signals on input 1 to 3.

Mix 4-5

Mix of the signals on input 4 and 5.

Therefore you can also use this mixer as two independent mixers with, respectively, 2 and 3 inputs.

Mix

Mix of all input signals.

Mix (dark)

Mix of all input signals with a filter to damp high frequencies slightly.

Mix (light)

Mix of all input signals with a filter to boost high frequencies slightly.

INPUT 1-5

Choose the audio source for the inputs 1 to 5 by tapping into the appropriate field and selecting an audio source from the popup menu.

Filter 1 / 2

The two multimode filters are the most important sound design tools in Tera Pro. They simulate the characteristics of the filter types used in traditional analogue synthesisers. With the built-in overdrive you get a soft distortion. Several filter characteristics with 6, 12, 18 and 24 dB/octave slopes are available. The filter resonance can be adjusted up to the level of self-resonance, where the filter behaves like a sine oscillator.
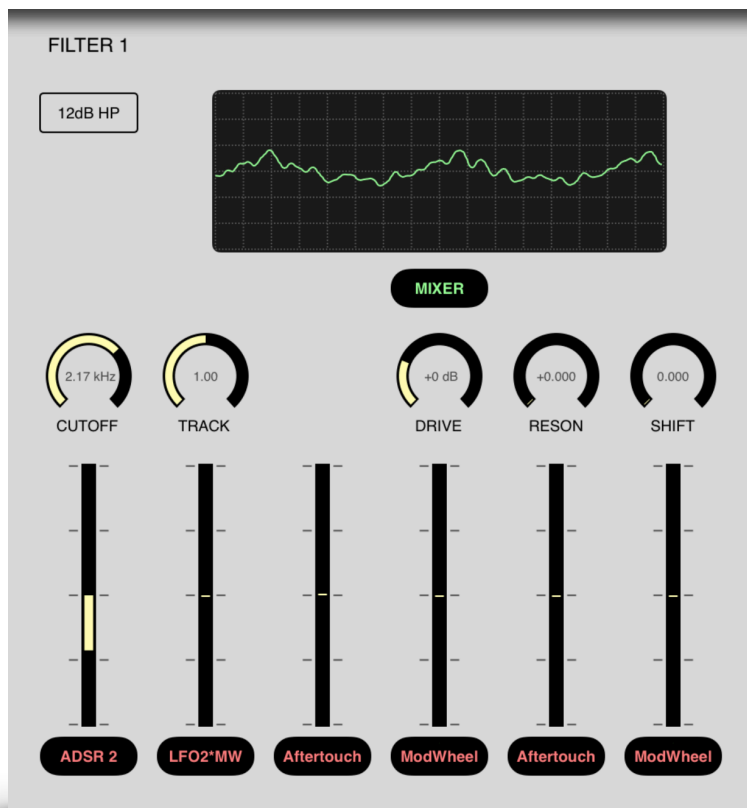
Here are the available filter characteristics in detail:

Lowpass 24dB Typ I

The lowpass filter attenuates frequencies above the cutoff frequency with 24dB/octave giving a warm and fat sound. The design of this filter is also known as "State variable filter".

Highpass 24dB Typ I

The highpass filter attenuates frequencies below the cutoff frequency with 24dB/octave giving a bright and sharp sound. The design of this filter is also known as "State variable filter".

Bandpass 12dB

The bandpass filter attenuates frequencies below and above the cutoff frequency with 12dB/octave giving a nasal and thin sound.

Bandreject 12dB

The bandreject filter attenuates frequencies near the cutoff frequency with 12dB/octave giving a phaser like sound.

Lowpass 12dB

The lowpass filter attenuates frequencies above the cutoff frequency cutoff with 12dB/octave giving a warm and natural sound.

Highpass 12dB

The highpass filter attenuates frequencies below the cutoff frequency with 12dB/octave giving a bright and thin sound.

Lowpass 24dB Typ II

This lowpass filter has a classical cascade design and attenuates frequencies below the cutoff frequency with 24dB/octave giving a warm and dark sound.

Lowpass 18dB Typ II

This lowpass filter has a classical cascade design and attenuates frequencies below the cutoff frequency with 18dB/octave giving a 303 like sound.

The filter parameter in detail:

INPUT

Choose an audio source for input 1 by clicking into this field and selecting a source from the popup menu.

Filter type

Selection of the filter characteristic. tap in the filter box for selection menu.

DRIVE

Adjusts the filter input level in the range of –10dB to +30dB. Actual input levels above 0dB results in a soft distortion of the input signal. Note that with increasing input level the filter resonance fades.

TRACK

Adjusts the keyboard tracking of the filter cutoff frequency. With a value of 1 the keyboard will control the cutoff frequency at a rate of one halftone per key. With a value of 0 keyboard tracking is off.

CUTOFF

Adjusts the filter cutoff frequency. For a lowpass filter higher values result in brighter tones and lower values result in darker tones. The range is from 20Hz to 20kHz. The cutoff frequency which is shown in the info window below the cutoff slider is the value for the key A4 (440Hz) if keyboard tracking is active.

RESONANCE

Adjusts the filter resonance which emphasizes frequencies close to the cutoff frequency. With very high values the filter will start to oscillate itself, producing a sine wave.

SHIFT

Each filter is internally built by two or more sub–filters connected in series. This parameter shifts the cutoff frequency of these subfilters. The effect depends on the filter characteristic:

Lowpass   adjusts the slope of the filter
Highpass   adjusts the slope of the filter
Bandpass   adjusts the bandwidth of the passband
Bandreject   adjusts the bandwidth of the stopband

This parameter has no effect for the filter types Lowpass 12dB and Highpass 12dB.

FORMANT FILTER

The Formant filter contains three independent filters which can be used in two different configurations:
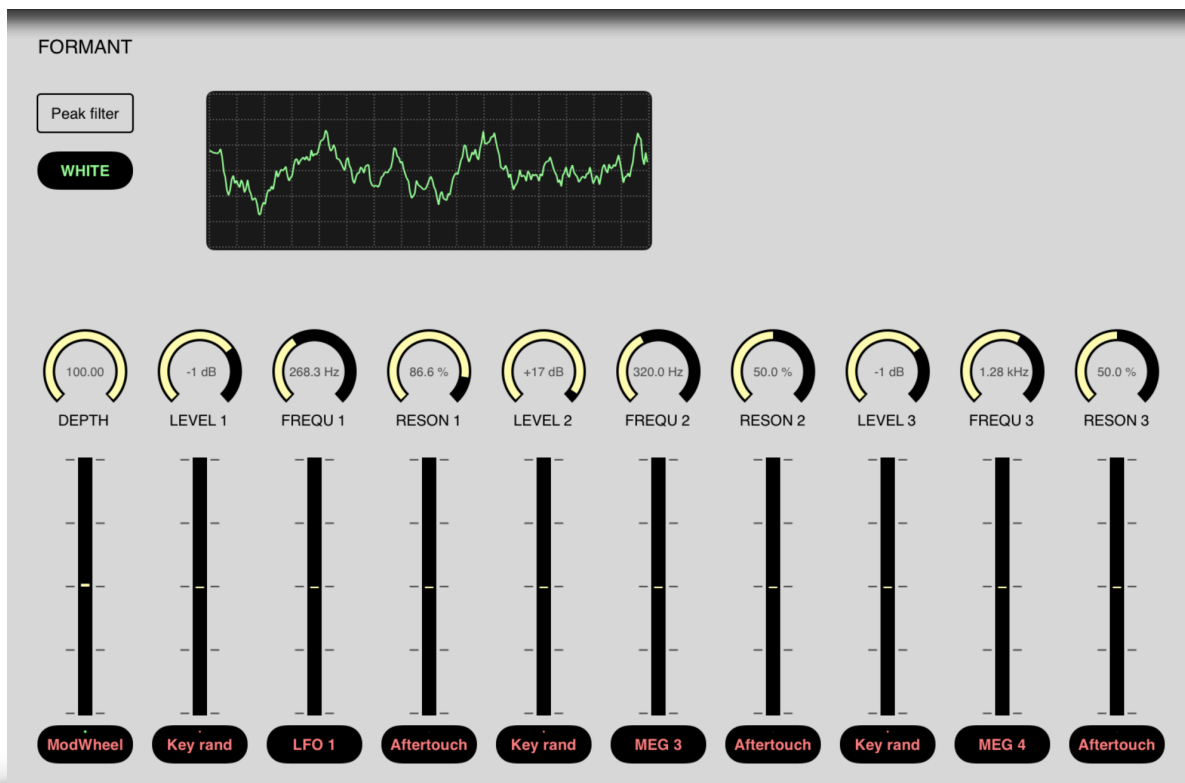
3 band pass filter in parallel

Formant filter for fixed and dynamically controllable formants. This configuration is useful for simulating vocal sounds and for filter morphing.

3 band reject filter in series

This filter configuration results in sounds comparable to comb filters and phaser effects useful for „spacey" sounds.

As opposed to common fixed filter banks, with this one you can dynamically control all filter parameters using different modulation sources !

INPUT

Choose an audio source for input by tapping into this and selecting a source from the popup menu.

Peak filter/Notch filter

Switches the filter configuration between „parallel bandpass" and „serial band reject" filters.

DEPTH

The depth parameter controls the mix of the filtered and the original input signal. The range is from 0.0 to 1.0. With depth set to 1.0 only the filtered signal is sent to the output. With depth set to 0.0 only the original signal is sent to the output.

FREQU 1 .. 3

Adjusts the centre frequency of the peak/notch filters. All frequencies below and above the centre frequency will be attenuated for the bandpass filter. All frequencies around the centre frequency will be attenuated for the bandreject filter. The range is from 20 Hz to 20000 Hz.

Q 1 .. 3

Adjusts the bandwidth of the bandpass filter relative to the centre frequency:

bandwidth [Hz] = center frequency [Hz] / resonance

The range is from 0.5 to 512. For a centre frequency of 1000 Hz this results in a bandwidth ranging from 2000 Hz to 2 Hz.

L 1 .. 3

The output level for the bandpass filter relative to the other two bandpass filters. The range is from –50 dB to +20 dB.

VOCAL FILTER

This module allows for real-time morphing between two 16 pole filters selectable from a set of predefined vocal filters simulating the vocal tract of a human body.
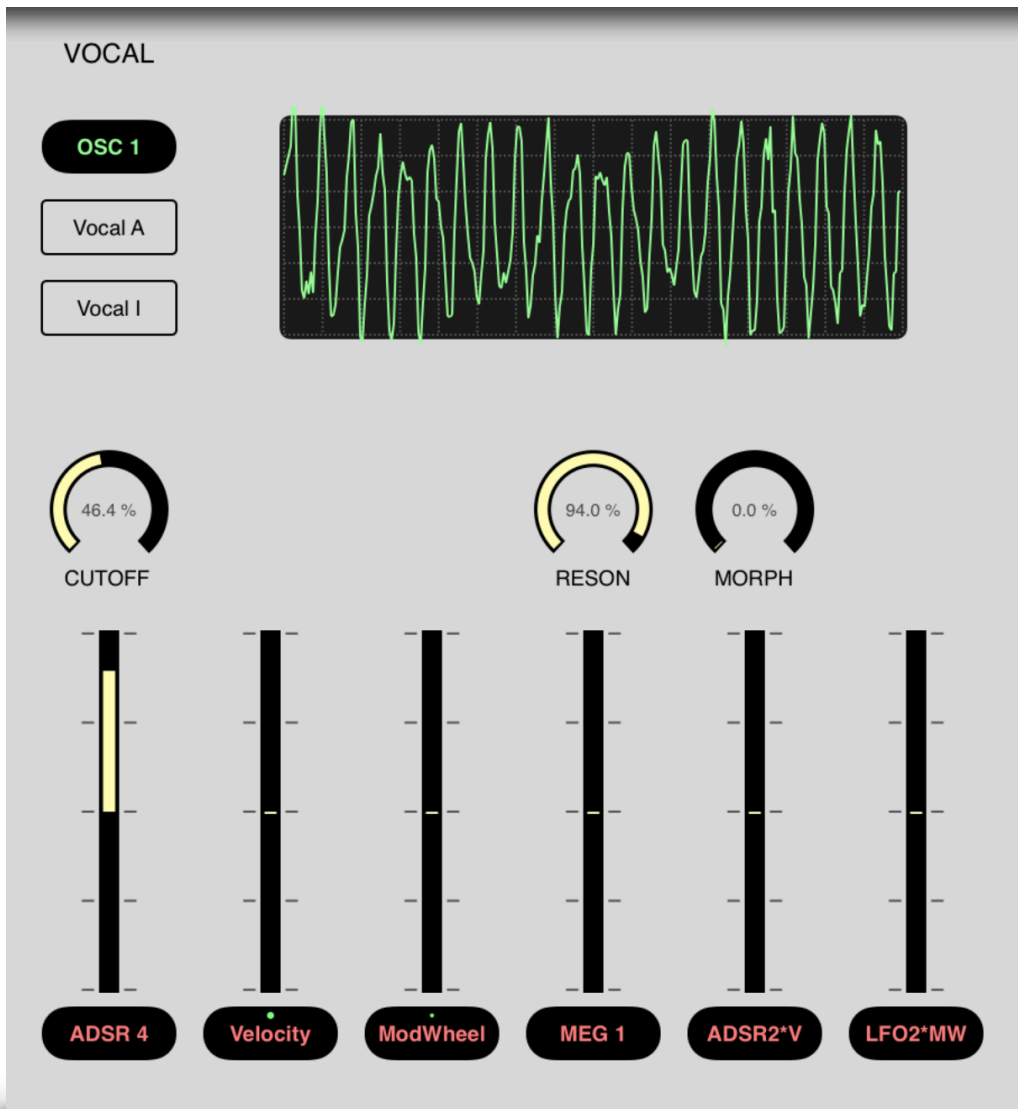
Filter selection 1/2

Selects the filter characteristic from a list of predefined presets.

CUTOFF

Shifts the base frequency of the filter.

RESON

Changes the resonance characteristic of the filter.

MORPH

Blends between the two preselected filter characteristics.

AMPLIFIER

The amplifier is the last module in every patch. This module mixes the selected input sources of all voices down to a stereo output. All parameters act independently on every voice before the final stereo mix is done. The pan position for example, can therefore be different for every voice in a polyphonic part. The stereo output is routed to the effects section for further processing.
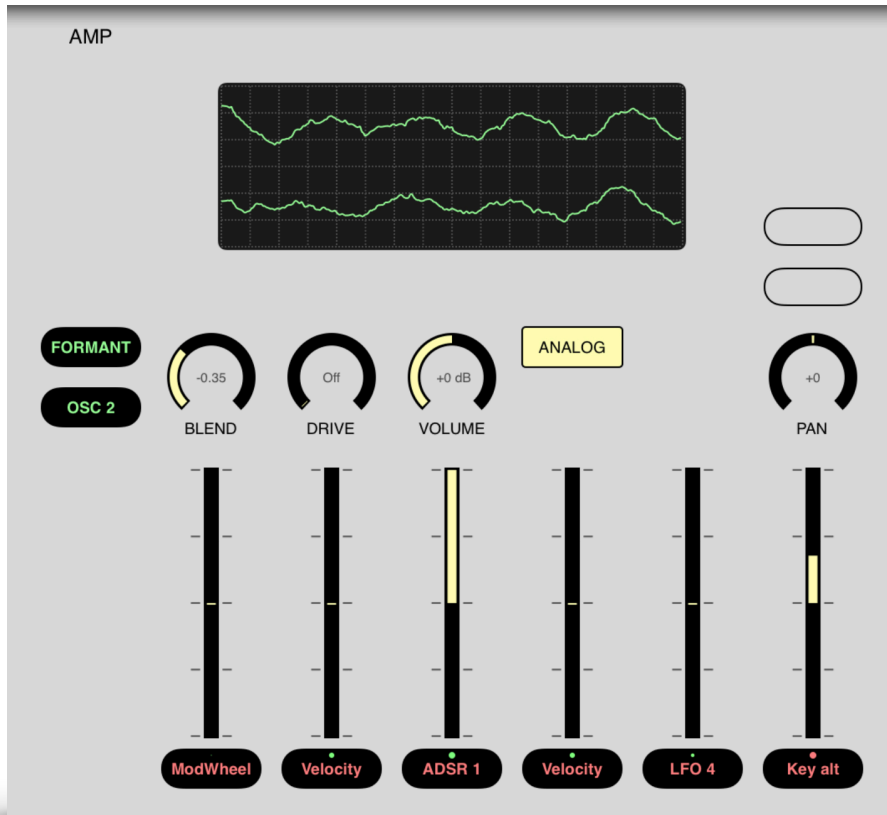
INPUT 1/2

Choose an audio source by tapping into this field and selecting a source from the context menu.

BLEND

Mix between input 1 and 2

OVERDRIVE

Adjust the amount of ranging from „Off" to +64dB (maximal distortion).

A soft distortion unit is inserted between the input of the amplifier and the volume slider. For this reason the degree of distortion is independent of the volume setting. If the volume is modulated by an envelope generator – as in most sound programs – the degree of distortion is also independent of the dynamic envelope. If you modulate the amplitude with an envelope generator before the signal goes into the amplifier unit then the degree of distortion varies with the dynamic envelope just as usual. Every voice has its own distortion unit, so there's no inter–modulation between the notes in polyphonic play. Therefore the sound is quite different in comparison to a standard distortion unit.

VOLUME

Output level ranging from –40dB to +40dB

PAN

Adjusts the position of the sound in the stereo panorama. Opposed to the pan slider in the part mixer this one works independently for each voice ! If for example the pan is modulated by an LFO this means every voice can have an independent position.
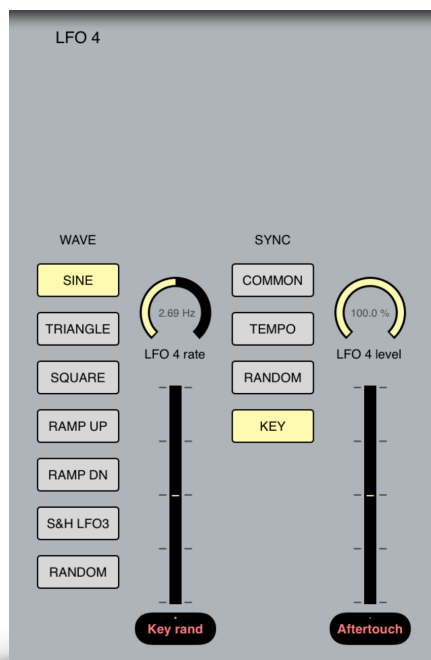
DIRECT LEFT/RIGHT inputs

These two inputs are routed directly to the stereo output of the amplifier. This makes it possible to produce stereophonic patches directly in the synth patch.

Modulation

LOW FREQUENCY OSCILLATORS (LFO)

Tera Pro provides you with four low frequency oscillators which can be freely assigned to modulate one or more sound parameters.

This module provides you with oscillations useful for periodic and random sound parameter modulations. The LFO can produce seven different modulation waveforms. Every LFO can also be used to Sample & Hold the waveform of another LFO. The rate of the LFO can be synced to the tempo of the Arpeggiator.

Wave

Selects one of the following waveforms:

1 Sine

2 Triangle

3 Square

4 Ramp up

5 Ramp down

6 S&H (Sample & Hold)

7 Random – Aperiodic waveform.

Using the waveform S&H produces a Sample & Hold effect. In this mode the module takes samples of the output of the next LFO at the rate of the LFO frequency. For Example LFO 2 takes samples from LFO 1. To create the traditional Sample & Hold effect select a random modulation waveform for the sampled LFO.

Wave

Sets the rate of the modulation oscillator. The rate of the oscillator can be specified in two ways:

If the Sync parameter is set to "Tempo sync'd" then the rate depends on the tempo of the Arpeggiator. Then you specify the cycle length in terms of note length with a range from 1/32 triplet to 4/1 dotted.

Otherwise the rate can be specified in a range from 1 cycle every 100 seconds to 689 cycles / second.

Sync

Sets the LFO phase and rate synchronisation. The following modes are available:

Common	– All voices of a part are modulated in sync.
Tempo Sync'd	– The rate is synced to the tempo of the internal Arpeggiator
Random	– Every voice has its own LFO starting with random phase on key down. The modulation is therefore phase–shifted for every voice of a polyphonic chord.
Key	– Every voice has its own LFO starting with zero phase on key down.

STANDARD ENVELOPES

The envelope generator lets you shape the parameters of a sound over time, beginning from when a key is hit, continuing as the key is held, and adjusting how the sound decays after the key is released. You can also trigger the envelope by one of the four LFOs instead of hitting a key. TERA uses envelopes with ADSR characteristic (like most analogue synthesizers do), extended by a delay phase and the option to thicken short attack/decay envelopes with the punch option.
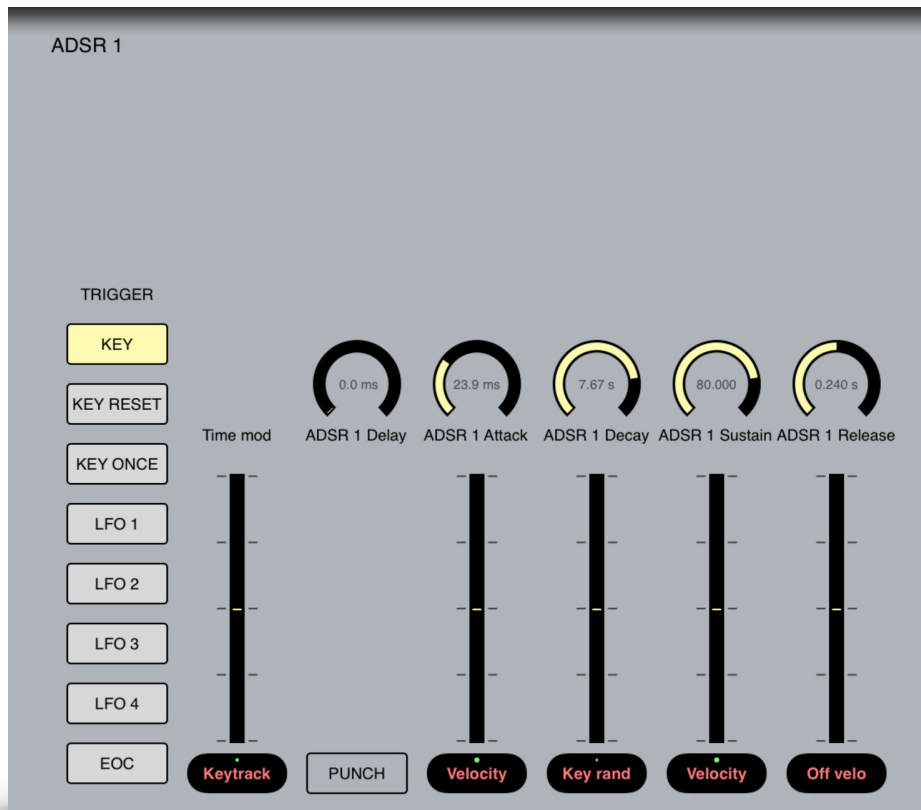
For every single voice TERA provides you with four independent envelope generators.

With LFO1 .. LFO4 selected as trigger source you can directly open the LFO with another press on the corresponding LFO button.

This is how the envelope evaluates after triggering:
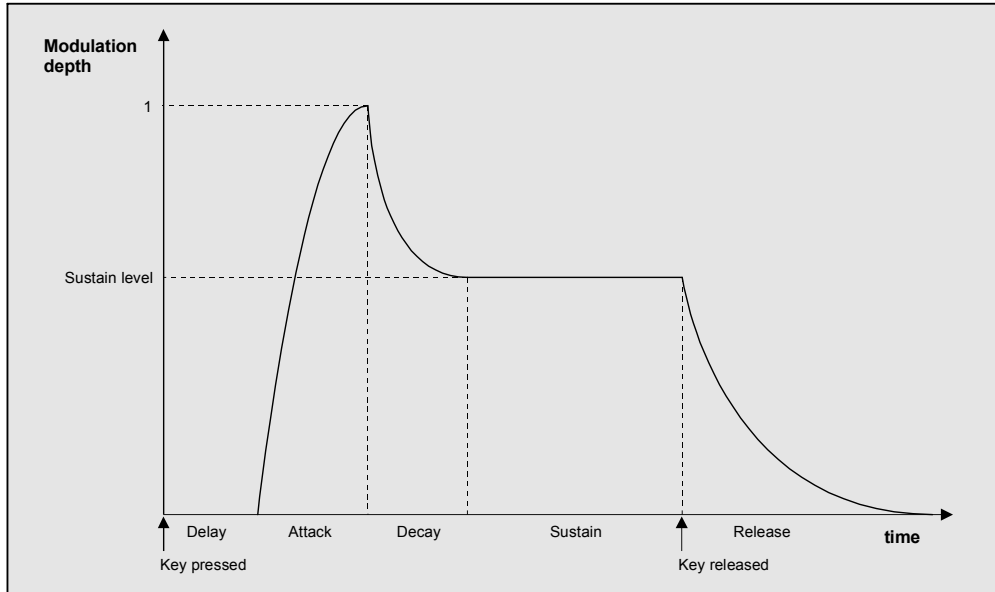
DELAY

Sets the delay time between note triggering and the start of the attack phase. Range is from 0.7 ms to 132 seconds.

ATTACK

Sets the time the envelope needs to reach the maximum level. The curvature is logarithmic, which means that from the start the signal builds up quickly and slowly rises when approaching the maximum level. Range is from 0.7 ms to 132 seconds.

DECAY

Sets the time the envelope needs to reach the sustain level after the attack phase has ended. The curvature is exponential, which means that from the start the signal decays quickly and slowly falls when approaching the sustain level. Range is from 0.7 ms to 132 seconds. The decay time has no effect if the sustain level is set at maximum.

SUSTAIN

Sets the sustain level the envelope will stay until release. Range is from 0 to 1.

RELEASE

Sets the time the envelope needs to reach the level 0 after the key is released. The curvature is exponential, which means that from the start the signal decays quickly and slowly falls when approaching the level 0. If the release time is set to 0 then the release time automatically follows the decay time. This results in an Attack/Delay only envelope.

PUNCH

The punch switch delays the start of the decay phase by a few milliseconds. The result is a „thicker“ attack similar to a compressor effect. This effect is audible only on short attack/decay time envelopes.

Trigger mode

With the trigger mode you decide when and how the envelope is triggered. The following modes are available:

KEY

The envelope starts on key down with the last actual level.

KEY RESET

The envelope starts on key down with the level 0.

KEY ONCE

Like Key trigger, but the envelope won't start with legato played notes.

LFO 1

The envelope starts with every new cycle of LFO 1.

LFO 2

The envelope starts with every new cycle of LFO 2.

LFO 3

The envelope starts with every new cycle of LFO 3.

LFO 4

The envelope starts with every new cycle of LFO 4.

EOC

Every new cycle of the envelope starts whenever the envelope comes to the end (**E**nd **O**f **C**ycle). Therefore the envelope is cycling indefinitely… Check out the patches named "Krell" to get some alien world examples. It starts to get interesting whenever you randomise the envelope stage times with "Key rand" – then you get indefinitely varying sound creatures lurking around…

MULTI ENVELOPES

Tera Pro provides you with four multi segment envelopes which can be freely assigned to modulate one or more sound parameters.

These envelopes have up to 64 time/level segments giving you all from simple Attack/Decay style envelopes up to complex rhythmical patterns which can even be synchronised to MIDI tempo.

The four envelopes are completely independent from each other: They can have different number of segments with free time/level values.

The main envelope editor figured below provides you with a context menu for copying and pasting envelopes through the clipboard and for selecting them from about 64 predefined envelopes.
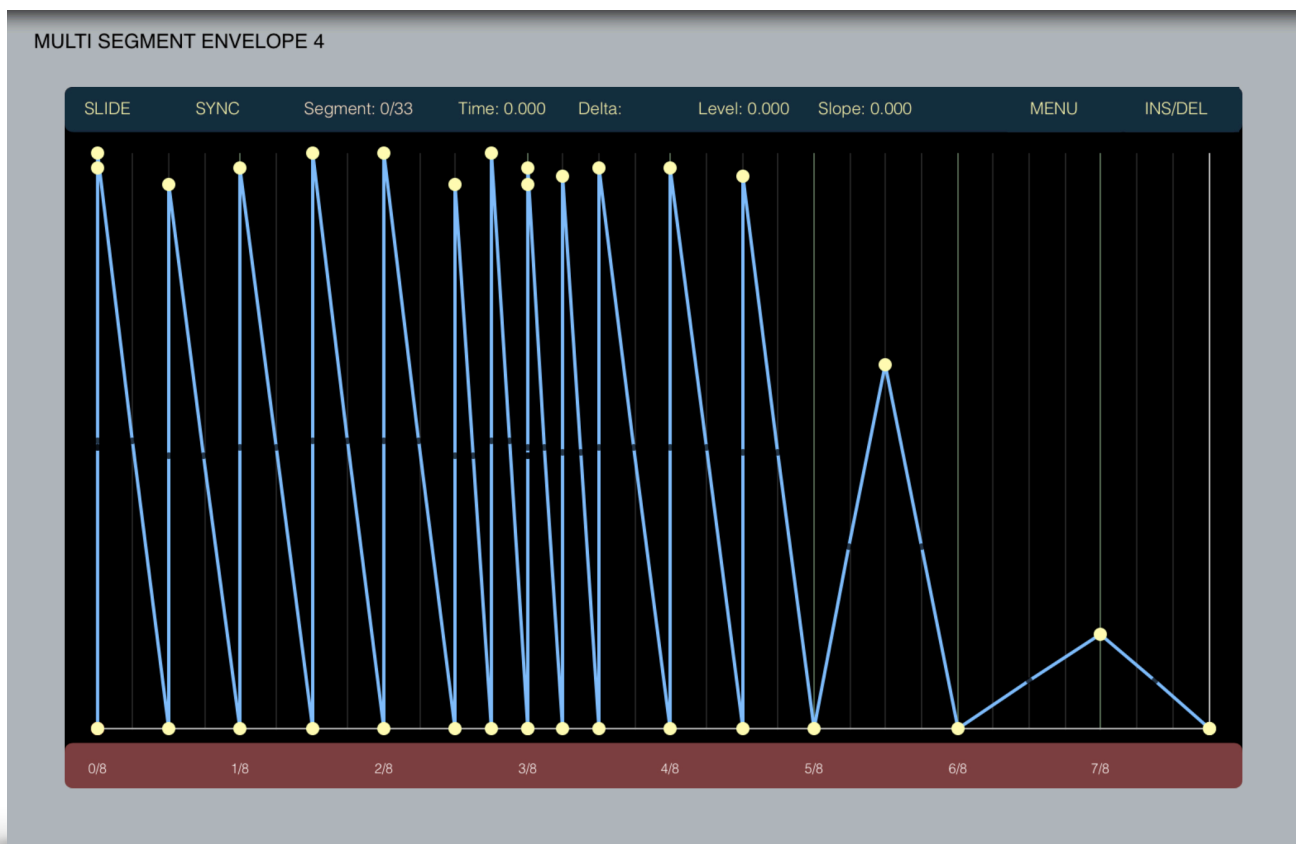
Editing the envelopes is an easy and straightforward process:

You can change the time or level of any segment by dragging the small yellow circle at the end of each segment into the desired position. Clicking the small black dot within the segment changes the slope. All editing take effect immediately!

With this dragging you also have selected this segment and then you can use alternatively the value fields for delta, level and slope for editing these values. Tap into the appropriate value field and drag your finger up or down for changing the values.

To insert a new segment/delete an existing one first tap on the "Insert/Delete" Button to switch mode and then tap the segment you want to insert/delete.

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The two grey vertical lines are the loop marker positions for the envelope. They are managed automatically by Tera Pro. The looping of the envelope between these two lines come automatically in effect as soon as more than one envelope segment is inside the marker positions.

You can scroll and zoom in the envelope window concurrently by tapping on the time bar below the envelope window and drag up/down for zooming and drag left or right for scrolling in the window.

If you want to edit a segment without changing the positions of other segments switch off the slide mode with a tap into the mode field above the envelope.

To create tempo synchronised envelopes tap into the sync field which will show then the value "On" and the time bar will show you then note values instead of absolute time.

Arpeggiator
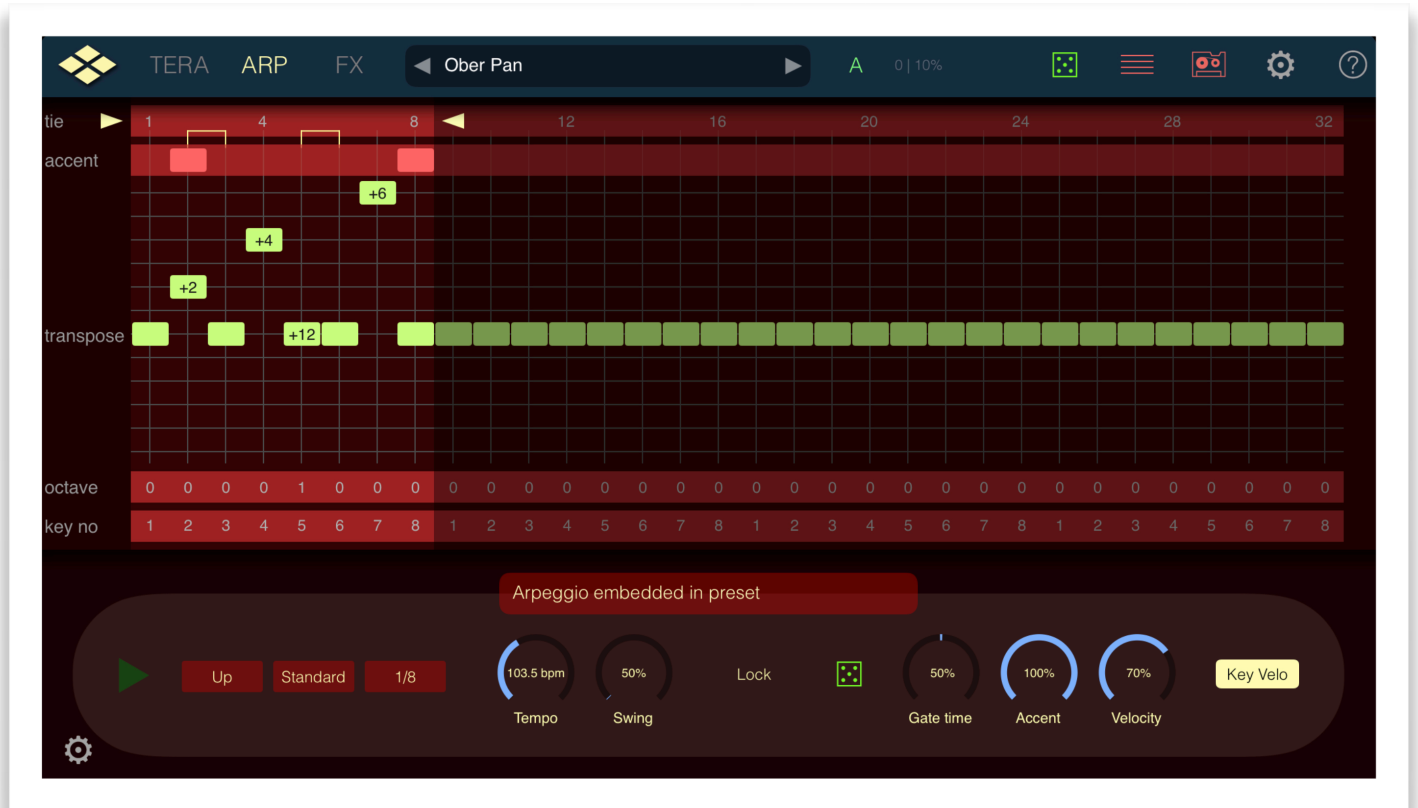
OVERVIEW

Tera Pro features an arpeggiator with a very flexible step matrix editor for programming and a unique "rolling dice" random arpeggio generator. Every sound preset can store it's own arpeggio pattern. You can build arpeggios from scratch or use one of the 32 preset arpeggios as foundation. You can also create an unlimited number of user arpeggio presets.

Just play a chord or a single note and it gets transformed into a grooving monophonic synth line or a rhythmical chord pattern.

GLOBAL SETTINGS
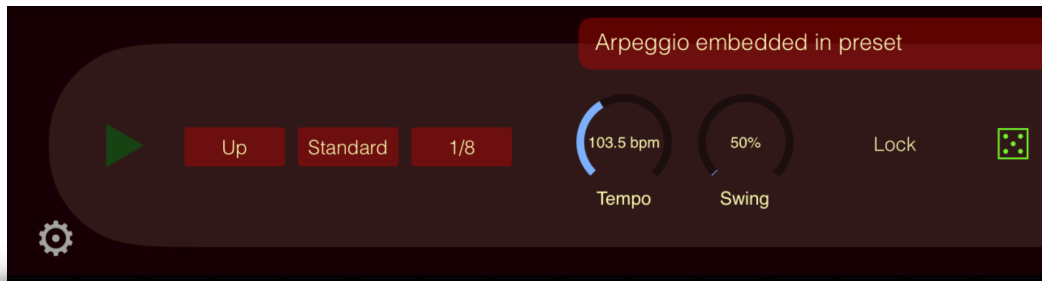
Tap on the arpeggio button to start/stop the arpeggio.

The mode parameter determines the way the pressed keys are sequenced. Available modes are: Up/Down/Alternate/Random/Incl/Excl.

The trigger parameter controls the way pressed notes and chords are handled:

standard – all currently pressed notes are used for the arpeggio.

hold – the last pressed notes are taken and the arpeggio continues even after you lift of all notes.

key sync – like "standard, but the arpeggio restarts at step 1 with every new press of some notes.

The clock parameter sets the basic step length.

The Tempo can be set in bpm (40–300) with the tempo knob. To enter a tempo value by tapping a rhythm tap on the tempo display in the top bar of the main window.

With Shuffle every other step in the arpeggio is moved forward or backward in timing.

The fix parameter is a performance function and if on the global settings of the arpeggio don't change on sound or arpeggio preset selection. This is useful if you want to use different arpeggios for one sound or different sounds for one arpeggio with the same settings.

Note that if fix is on the selection of another preset doesn't load the presets own arpeggio, the currently active arpeggio stays as it is.

Some more parameter are available on the right side of the Arpeggio page:
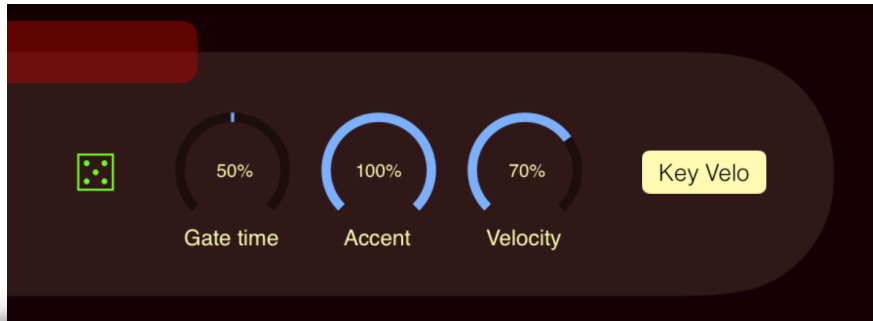
With the "rolling dice" you can get another randomly generated arpeggio with each tap on the dice and be surprised how useful they can be !

The gate parameter controls the length of each note relative to the step time.

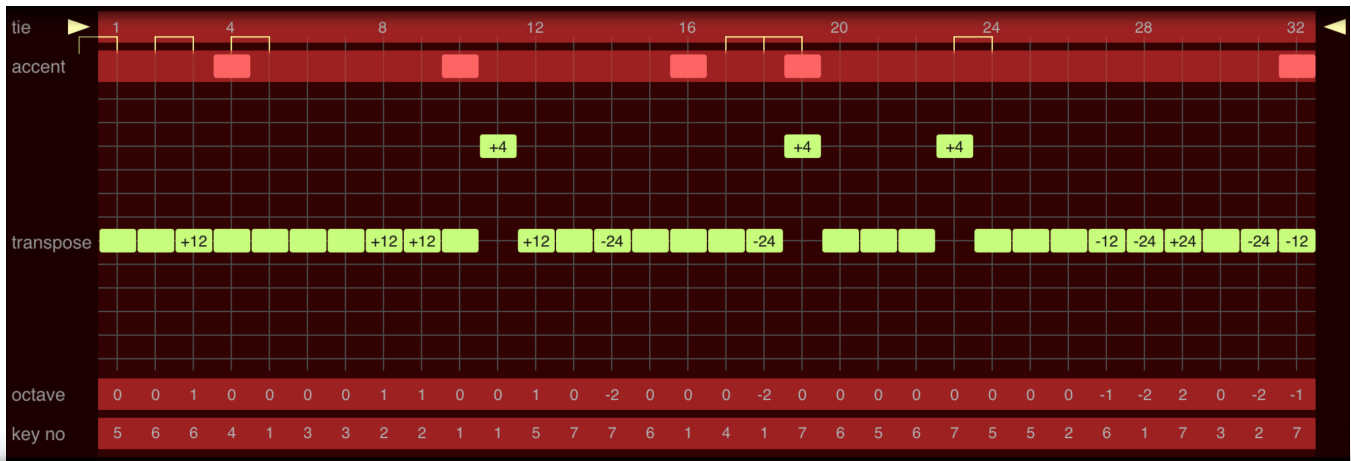With the accent row in the step matrix you can boost the velocity for selective steps. The velocity used for accented steps can be set with the accent knob shown above.

If "key velo" is set the arpeggiator uses a constant velocity set with the "velocity" knob above for all steps which are not accented. Otherwise the arpeggiator uses the velocity with which the notes were pressed.

Step Matrix Editor

The step matrix editor is the programmable "heart" of the Arpeggiator. The matrix rows represents the five parameters which can be set individually for each step in the pattern, the steps being represented by

row																																
octave	0	0	1	0	0	0	0	1	1	0	0	1	0	-2	0	0	0	-2	0	0	0	0	0	0	0	-1	-2	2	0	-2	-1
key no	5	6	6	4	1	3	3	2	2	1	1	5	7	7	6	1	4	1	7	6	5	6	7	5	5	2	6	1	7	3	2	7

the columns of the matrix.

The current position of the arpeggio in the timeline is shown by a running highlight in the position row on top of the matrix. Tap and drag in the position row to change the length of the pattern from 1 to 32 steps.

In the next row you can set or reset the tie control for each step. With tie activated on a step the note before this step will get played legato to the note in the step. If both steps play the same note no note off message is generated which means that the note just gets played longer. With legato and slew set on the keyboard page the tied notes also gets the typical 303 type note glide effect.

In the next row you can set an accent on each step you like to boost the velocity for this step.

In the note and octave row you can set the transposition relative to the played notes. Tapping directly on the yellow transpose marker mutes this step.

Parameter row functions

In each parameter row you can set the following functions from left to right:

Rolling dice: tap to get another random set of step parameter for this row.

Left arrow: rotate the contents for this parameter in the row on step to the left.

Right arrow: rotate the contents for this parameter in the row one step to the right.

Reset sign: reset the parameter for this row to their default value.

Effects

Cube Synth Pro comes with a set of 7 studio quality effects which can be used simultaneously and in any arbitrary order.

Common

- **Dry/Wet -** Balances between original and delayed signal.
- **Level -** global output audio level
- **Dice -** each tap on the dice gives you another random global effect setting.

The signal flow of the effect is from left to right. To change the position of one effect in the fx chain just tap on the name of the effect and drag it to the new position in the chain.

Drive

The Drive effect applies typical analog distortion to the audio input in various degrees. You can select between four different basic characteristics: hard and soft transistor type distortion, a warm tube amplifier simulation and a typical tape (magnetic recording) simulation. To further color the distorted sound a Tone filter is applied to the signal. It basically cuts the spectrum in two halves around the cutoff frequency and the tone parameter balances the high and the low part of the spectrum.

- **Type -** Select the basic distortion characteristic (Hard/Soft/Tape/Tube)

- **Drive -** Input level before distortion unit. Controls the amount of distortion.

- **Level -** Output level - Use this to compensate for excessive gains in the distortion stage.

- **Cutoff -** Tone filter split frequency.

- **Tone -** Balance between High pass and Low pass. Set Tone to 0.5 for no colouring.

Parametric Equalizer

The **Parametric equalizer** offers continuous control over the audio signal's frequency content, which is divided into three bands of frequencies. The parameters can either be set by the knobs in the user interface or in the graphic representation of the filter curve by dragging the filter points.

- **HiGain -** Gain of the High pass shelf filter.

- **HiFreq -** Cutoff frequency of the High pass shelf filter.

- **MidGain -** Gain of the Band pass filter.

- **MidFreq -** Cutoff frequency of the Band pass filter.

- **MidQ -** Q-Factor (Bandwidth) of the Band pass filter.

- **LoGain -** Gain of the Low pass shelf filter.

- **LoFreq -** Cutoff frequency of the Low pass shelf filter.

Delay unit

The delay effect provides you with two different types of delay effects using two delay lines. The delay times are set by note lengths and are relative to the currently set tempo. The delayed signal can be feed back to the input for echo effects. You can select between a stereo delay and a ping-pong delay where the echoes bounces across the stereo field.
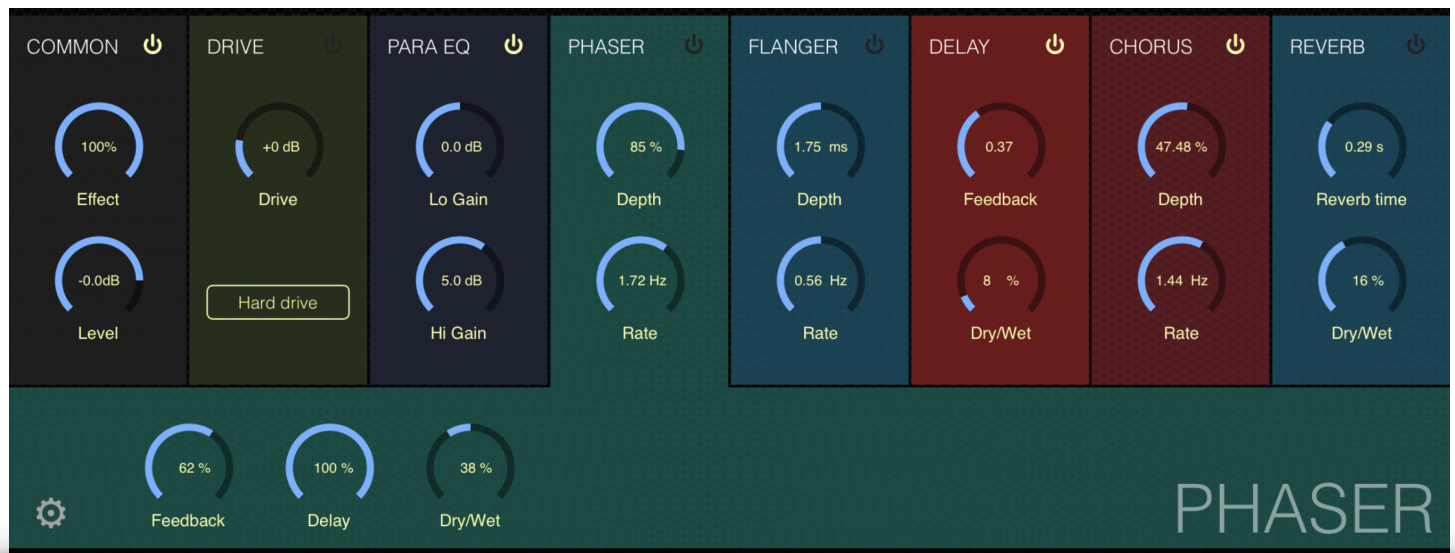

- **Dry/Wet** - Balances between original and delayed signal.

- **Delay L/R**- Sets the the delay time in note lengths for the left/right channel.

- **Link** - Toggle to link left and right delay times.

- **Feedback** - Controls the decay time of the echoes.

- **Cutoff** - A lowpass filter is build into the feedback loop of the delay. This parameter controls the cutoff of this filter resulting in successive echoes sounding darker.

- **Cross -** Ping Pong style delay. The echo jumps between left and right channel.

- **Sync -** Synchronise delay time to the current song tempo.

- **Tempo -** Set the song tempo manually (see below*)

- **Tap Tempo -** tap quarter notes to set the tempo the musical way.

* The tempo can be synced to Ableton Link enabled Apps. Enable Ableton Link in the Settings Dialog of AudioLayer. In this case you can no longer set the tempo manually here.

Phaser modulation effect

The Phaser is built of an 8-pole allpass filter, which gives you a very short and frequency dependent delay time. This delay time is modulated by an LFO.

- **Dry/Wet** - Balances between original and delayed signal.
- **Depth** - This parameter controls the depth of the delay modulation.
- **Feedback** - Sets the feedback amount for the all pass filters of the phaser to further increase the phaser effect.
- **Delay** - The mean phase delay for the phaser.
- **Rate** - The modulation rate of the effect.

Flanger modulation effect

The Flanger is composed of two independent delay lines with feedback for the left and the right channel respectively. The delay time of both delays is modulated by one LFO with adjustable frequency. For typical flanging effects the delay time should be below 5 milliseconds.

- **Dry/Wet** - Balances between original and delayed signal.
- **Depth** - This parameter controls the depth of the delay modulation.
- **Feedback** - Sets the feedback amount for the delay lines of the flanger to further increase the flanger effect.
- **Delay** - The mean delay time of the chorus. Set this to 1-2ms for typical flanger effect.
- **Rate** - The modulation rate of the effect.

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Chorus modulation effect

Use the chorus/ensemble effect to fatten the sound and give it more width in the stereo panorama. The chorus effect uses four feedback delay loops which can be modulated by different modulation rates to allow for a full ensemble effect. Use the spread parameter to set the range of the modulation rates.

- **Dry/Wet** - Balances between original and delayed signal.
- **4x** - Switch this off for a more traditional chorus sound using only 2 delay lines.
- **Depth** - This parameter controls the depth of the delay modulation.
- **Feedback** - Sets the feedback amount for the delay lines of the chorus to further increase the chorus effect.
- **Delay** - The mean delay time of the chorus. Set this to 10-20ms for typical chorus effect.
- **Spread** - Use this to decrease the periodicity of the modulation.
- **Rate** - The modulation rate of the effect.

Reverb effect

The reverberation effect is used to simulate the sound reflections off the walls in different kind of rooms. This reflections disperse the sound, enriching it by overlapping the sound with its reflections.

The reverb unit in AudioLayer is a stripped down version of our hi-end reverb REFLECT which is famous for his natural and smooth sound.

- **Tail delay -** Sets the amount of time that elapses between the direct sound and the reverberation tail. This time correlates with the perceived size of the simulated room. The adjustable range is from 0 ms to 300 ms.

- **Reverb Time** - The time the reverb takes to die away by 60dB after the end of the initial sound. This parameter correlates with the perceived size of the simulated room. The adjustable range is from 50 ms to 300 seconds.

- **Room Size** - The room size parameter sets the average distance between the reflecting walls.

- **Diffusion** - The Diffusion parameter controls the echo density of the resulting reverberant sound. The adjustable range is from 0% to 100%. A higher value corresponds to a higher echo density which is in most cases more natural sounding.

- **Absorption** - The Absorption parameter simulates the surface and air absorption of high frequencies. The adjustable range is from 0% to 100% . Due to absorption the reverberation time is shorter for high frequencies than for low frequencies. With higher absorption high frequencies will decay much faster.

- **Modulation** - Modulates the delay lines slightly to further improve the naturalness of the reverb tail. High values may lead to pitch shifting effects.

- **Tail stereo -** Adjusts the stereo image width of the reverberation tail.

- **Tail cut-off / tone -** Low and High pass filter for the reverb tail. Set crossover frequency and balances the output of the low/high pass.

- **Dry/Wet** - Balance between the dry and reverberant part of the output signal.

Appendix

TROUBLESHOOTING

Fresh installation

Sometimes Apps having problems after first installation or update. Please try the following:

- trash the app
- boot your device. (most important !)
- repurchase again from the AppStore (you don't have to pay again, they know that you already have)
- sometimes another reboot can be helpful.
- This removes usually problems of this kind and has only be done once.

Support e-mail

If you have any problem please send us a support form from our website mobile.virsyn.com
This is the best way to contact us.