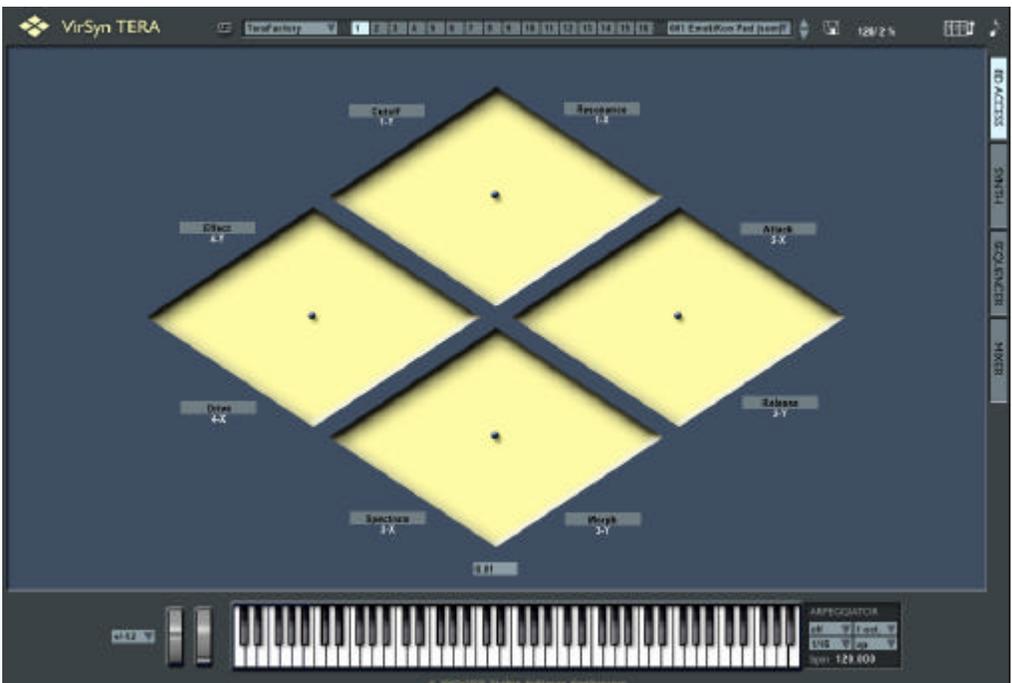


VirSyn TERA

Owner's manual 3.0



Virtual integrated realtime synthesis

Harry Gohs

VirSyn Software Synthesizer

**Professional tools for
sound design and
computer music production**

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1 Introduction

1.1 Prologue

Virtual integrated realtime synthesis

Explore new sound spaces with VirSyn TERA that go beyond the pure emulation of synthesizer legends. Use the power of analog and digital synthesis combined with new physical models and new user interface elements to navigate through 8-dimensional sound spaces exceeding the limits of traditional sound morphing. THE well-designed algorithms and the economical modular structure provides you with a comparatively high voice yield. Wireless patching of modules combined with an intelligent modulation matrix simplifies the development of new synthesizers from scratch. Up to 16 modular Synthesizers can be used simultaneously, all having the same user interface helping you operate this multi-modular system without losing track. Distortion, echo/delay and chorus/flanger/phaser effects for every modular synthesizer help boosts the sound into outer space. VirSyn TERA gives your production an individual character with its unique fat sound. Precise timing even with an external sequencer lets you feel the rhythm. Produce complete productions with VirSyn TERA as a stand-alone tool. Inspire yourself from the newly-designed 64 step-sequencer, expanding the concept of the classic analog sequencer with variable rhythmic structures and chord sequences. Or use the sequencer for real-time transposition of keys to play never-before heard arpeggios. Arrange up to 256 sequences in the song editor to complete the composition. And the integrated mixing console, together with a room simulator and a global chorus effect, helps you in the final production step.

1.2 System requirements

For proper operation of VirSyn TERA the system requirements are:

- ✍ **PC with Pentium II Processor 400 MHz or better.**
- ✍ **Microsoft Windows XP.**
- ✍ **at least 512 MB main memory.**
- ✍ **Minimal screen resolution 1024X768 / min. 16-bit colors.**
- ✍ **16-bit Soundcard.**
- ✍ **ASIO/ DirectX Driver.**
- ✍ **300 MB free space on your hard disk.**

or

- ✍ **Mac with G3/G4 Processor 300 MHz or better.**
- ✍ **Mac OS X 10.3.**
- ✍ **at least 512 MB main memory.**
- ✍ **Minimal screen resolution 1024X768.**
- ✍ **CoreAudio.**
- ✍ **300 MB free space on your hard disk.**

If you want to use VirSyn TERA together with an external keyboard you will need:

- ✍ **MIDI Keyboard**
- ✍ **MIDI Interface**

You can use VirSyn TERA also in the following host environments:

- ✍ **VST (WinXP/OSX)**
- ✍ **AudioUnit (OSX)**
- ✍ **RTAS (WinXP/OSX)**

1.3 Installation

VirSyn TERA is delivered on a hybrid CD containing both the Mac and the PC version of the program. Therefore you can crossgrade without ordering a new CD free of charge. The included sound banks and projects can be used independent of the Computer platform.

To install VirSyn TERA on your hard disk, insert the installation CD into your CD-ROM drive. The installation program starts automatically (on the Mac you have to start the installation manually) after inserting the CD-ROM. If for some reason it does not, you can start the installation program manually by opening the CD-ROM with the explorer and starting „setup.exe“. The installation program leads you through the rest of the installation process.

After starting the program the first time you will be asked for the registration code which is printed on the CD cover. After entering this code successfully, the program is licensed on your computer. Please keep the registration code together with the license number in a safe place! You will need them for every update and reinstallation of VirSyn TERA.

Remarks:

For installation as a VST2 plug-in you have to copy the file TERA3.dll in the installation folder of TERA manually into the vstplugins folder of your favourite VST2 host program.

1.4 What's new in Version 3.0 ?

1.4.1 Multi sampling

Each of the three main oscillators can use now samples instead of the built-in waveforms. The samples are organized in up to 16 key zones and 2 morph zones, making for a total of up to 32 sample files for each oscillator. Due to its high quality re-sampling algorithm, you can use a single sample for the whole tonal range!

1.4.2 Terrain filter

This new module allows for real-time morphing between two 16 pole filters selectable from a set of predefined vocal filters.

1.4.3 Arpeggiator

In addition to the high-level arpeggio functions of the built in sequencer, TERA 3 now offers a simple standard arpeggiator for each part.

1.4.4 Stereo routing

The amplifier module now has two additional stereo inputs allowing for stereo effects inside the synthesis patch. The spectrum oscillator has a new stereo output for stereophonic wavetable synthesis.

1.4.5 Envelopes

The four multi-segment envelopes in TERA can now be used in A new “free” mode where each envelope can be used independently. These envelopes can be saved and loaded from special envelope presets.

The standard envelope generators have also gotten a new Attack/Decay only mode.

1.4.6 Spectrum oscillator

The Spectrum oscillator has many new functions:

- uncorrelated stereo output
- spectrum presets
- separate spectrum parameters for two sets of oscillators

In addition to the additive sine synthesis used in TERA 2 as a basis for the wavetable generation process, the following methods can also be used:

- wavelet synthesis
- walsh function synthesis
- operator synthesis

1.4.7 Voltage controlled amplifier (VCA)

This new module works just like a ring modulator, except here one input is used to modulate the amplitude of the other input.

1.4.8 New module inputs

The outputs of the envelopes and the LFO's can now be used as inputs for the signal processing modules of TERA. previously, these outputs could only be used in the modulation matrix.

1.4.9 New Modulation sources

Two modulation sources have been added:

- **Key alternate**: can be used to get a different sound on every other note (IE: like bow up/down in natural string instruments)
- **Key random**: creates a random control signal on every note.

1.4.10 Modulation matrix

The modulation matrix has been given a copy/paste function for transferring modulations between presets. Modulated parameters are indicated in the user interface through a color change of the parameter knobs.

1.4.11 8D control

The unique 8D morph control of TERA is now also usable for the six special synthesis pages introduced in TERA 2. The controller settings can be copied and pasted between presets and each dimension can be labelled individually.

1.4.12 “Fly over” tool tip help system

You can get more information on a particular parameter by moving the mouse over it, revealing a small window with detailed parameter info.

1.4.13 New sounds

The new TERA sound banks contain 256 presets, including some from the well-known sound designer Howard Scarr.

For the new sample module a set of about 35MB of multi-samples is included. In addition to the traditional bank structure all presets are now available in a hierarchical folder structure for easy sound categorization and searching.

1.4.14 General

- MIDI controller editor for easier assignment of MIDI controllers to parameters.
- Selection fields (eg. Wave selection) can now be assigned to MIDI controllers
- Receive MIDI input from multiple devices simultaneously.
- Scroll wheel can be used for parameter adjustments on Windows XP.
- Smaller user interface makes plugin usage more convenient.
- Parameter fine adjustment control with Shift+Drag function.
- Glissando function (chromatic glide)
- Each Sequencer part can now control an external synthesizer (Windows XP)
- RTAS plugin for windows XP and OS X.

1.5 Mac/PC Platform differences

The Mac and the PC version have the same functionality, but they differ slightly in the use of the mouse and in certain key strokes according to the usual user interface on their respective platform.

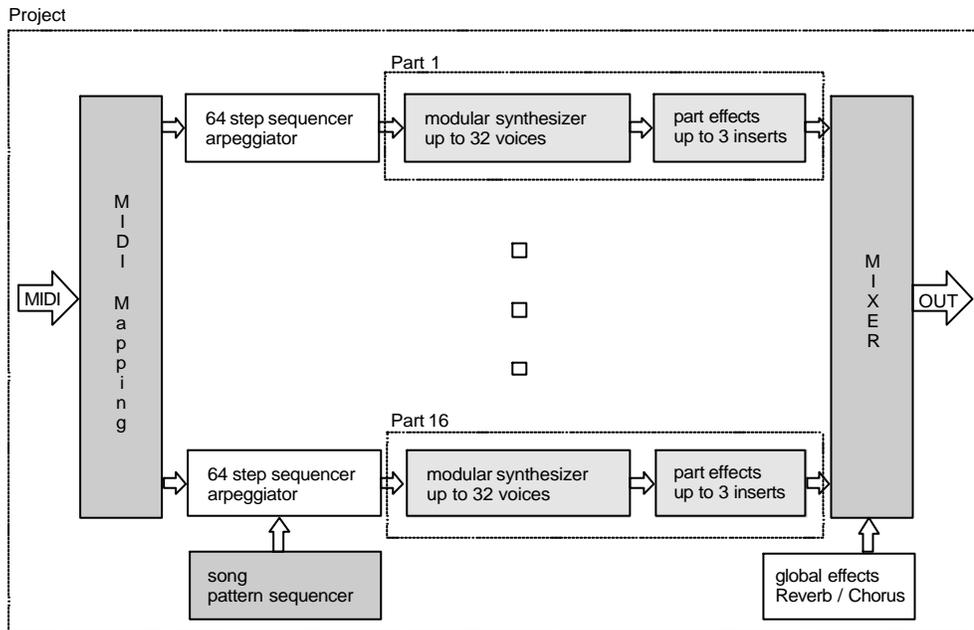
The key strokes and the usage of the mouse as described in this manual are valid for the PC platform only. The following table shows the corresponding key strokes and mouse usage on the Mac:

| PC | Mac | Function |
|--------------------------|------------------------------|--|
| Right MOUSEBUTTON | ctrl+ MOUSEBUTTON | Context menu |
| ctrl+ MOUSEBUTTON | alt+ MOUSEBUTTON | 8D Editor, Spectrum window |
| F1-F5 | Command+1 – Command+5 | Selects the different screen pages (available only in stand-alone version) |
| Enter | Enter | Starts Sequencer |
| Space | Space | Stops Sequencer |
| ? | ? | Increments selected parameter |
| ? | ? | Decrements selected parameter |
| Pg? | pg up | Increments selected parameter coarse |
| Pg? | pg dn | Decrements selected parameter coarse |
| Pos 1 | home | Sets maximum value for selected parameter |
| End | end | Sets minimum value for selected parameter |
| Del | Backspace | Sets default value for selected parameter |
| not available | Command+N | Opens new project |
| not available | Command+O | Opens an existing project |
| not available | Command+S | Saves the current project |
| Alt+F4 | Command+Q | Quits TERA |

2 Concept

2.1 Block diagram

VirSyn TERA is designed to be used as a professional tool for sound design and computer music production. The following block diagram shows you the components of TERA's integrated production environment:



VirSyn TERA can be used as either a standalone program or as a Plug-in instrument in one of the supported host environments; TERA's features are almost identical for all versions. The manual sections which only apply to some versions are specially marked with **VST: AudioUnit:RTAS:**

2.2 Part

VirSyn TERA is multi-timbral by design. The **part** is the basic unit of this multi-timbral structure. Every part contains an independent modular synthesizer and up to three insert effects. Up to 16 parts can be used at the same time. The modular synthesizer contains 25 modules, including oscillators, filters with self-resonance and the modulation matrix. The modules can be freely patched using the unique „wireless“ patching method of TERA. The synthesizers of every part can have its own structure independent of all other parts. Therefore TERA is not only multi-timbral, but IS even „multi-modular“ !

The three insert effects can be used to refine the sound output of the synthesizer. TERA provides you with an Overdrive effect with adjustable clipping characteristics, an Echo/Delay effect and a modulation effect usable as either a Chorus, Flanger or Phaser.

The sound parameter for one part, including the patching of the modules and the effect parameters, can be stored in a sound program. The sound programs of TERA are organized in sound banks with every bank containing up to 128 sound programs. The selection of sound programs can be done manually or by using the MIDI program change commands.

2.3 Project

A project is the top-level structure in TERA containing all parameter settings. It can be stored under a user definable name with a click on the Save button and choosing “Save project...” from the popup menu. To start with an empty project just open the project named “empty”.

The following parameters are stored in a project:

- ✎ The sound programs for all 16 parts.
- ✎ Step sequencer pattern memory (512 pattern * 64 steps/chords).
- ✎ Song sequencer with 512 measures.
- ✎ Global effects parameter.
- ✎ Mixer settings.

The first sound bank with the fixed name „Song bank“, is a special sound bank containing all 16 sound programs used by a project. By saving a project all currently active sound programs for the 16 parts are automatically stored in the Song bank. This ensures that the sound programs used by a project are preserved, even if the original sound programs change.

VST:AudioUnit:RTAS

In Plug-in operation you normally don't need to load and save projects in TERA itself -- the projects are stored and loaded together with the host's load and save operations.



2.4 MIDI Mapping

You can connect a distinct MIDI channel to each and every part in the part mixer. The default setting in the part mixer associates part 1 to MIDI channel 1, part 2 to MIDI channel 2 and so on. The MIDI events coming from any MIDI input channel are redirected to the associated parts.

With the key split function of the part mixer you can restrict the redirection of MIDI note events to a user definable keyboard zone. This function enables you to assign different parts (with different sound programs) to separate key zones of the keyboard.

Having a look at the block diagram, you will see that every part has its own step sequencer. If the step sequencer is stopped or not activated for a particular part, these note events are redirected directly to the modular synthesizer. While the sequencer is running and activated for this part, the incoming note events are manipulated by the step sequencer. This enables you to transpose sequences, select sequencer patterns or to arpeggiate chords controlled by the notes you play on the keyboard.

Thanks to the real-time functions of the sequencer, TERA is perfectly suited for complex live sessions!

2.5 Microtuning

Microtuning gives you the possibility to use other tunings than well-tempered. In TERA you can change the tuning for each part separately. In TERA's installation folder is a new subfolder called "Tuning" created for this purpose. TERA looks there for Scala compatible tuning definition files (.scl, .tun) named "part1.scl", "part2.scl" and so on. If you want to change the tuning for part 1, for example, you simply have to put the corresponding Scala file into the subfolder "Tuning" and rename it to "part1.scl". On the next start -up of TERA it will then use this tuning for part 1.

C O N C E P T

3 User interface

3.1 Overview

The user interface of VirSyn TERA is designed to give you an optimal operating environment with mouse and keyboard. For this reason we avoided imitating traditional hardware synthesizers. Big knobs and graphical editors enable you to modify sound parameters precisely and easily by using the mouse, keyboard or an external MIDI controller. All parameters are divided into four property sheets.

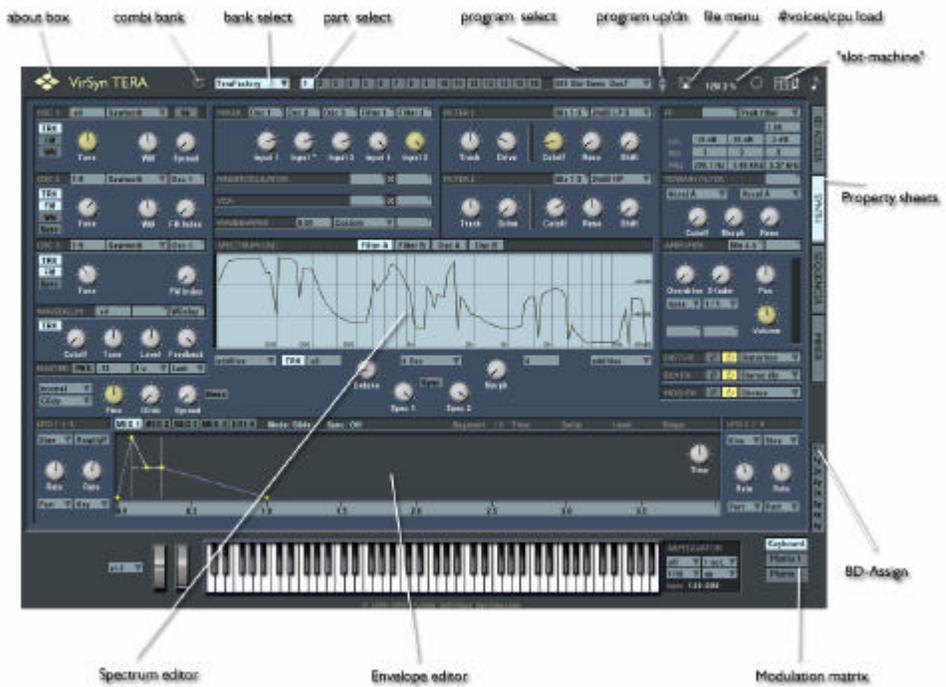
The MIDI Learn function lets you assign a MIDI controller to any sound parameter. Then you can control sound parameters by turning the knobs of your external keyboard or MIDI controller box.

TERA gives you access to the sound engine via five user interface levels with increasing complexity:

- ✍ Select one of the 1600 supplied sound programs using bank/program change commands.
- ✍ Navigate through the 8-dimensional sound space using the 8D access page to control up to 64 sound parameters simultaneously. Define your next sound destination with the mouse or external MIDI controllers and reach the destiny of your voyage with adjustable travelling speed.
- ✍ Modify the basic sound parameters on their respective sound parameter page.
- ✍ Create your own sounds using one of the six prewired Synthesizers **microTERA / specTERA / fmTERA / formanTERA / waveTERA / anaTERA.**
- ✍ Construct your own synthesizers by patching the modules and modulation routings ON your own with **profiTERA.**

3.2 Screen layout

This section describes the distinct screen areas and their underlying functionality.



The main screen is divided in the following screen areas – listed from top to bottom:

- ✍ **Toolbar**
Gives you fast access to the most important functions and information.

- ✍ **Property sheets**
Gives you access to the parameters of the synthesizer, the effects, the step- and song sequencer and the part mixer divided into four property sheets.

- ✍ **Bar on the right side**
Lets you select the property page and choose the 8D-Parameter you want to edit. With the button “Matrix” you can open the modulation matrix editor. These functions are only available for the **profiTERA** synthesizer.

3.3 Toolbar

The functions of the user interface elements are explained from left to right in the following sections. The first on the left is – of course – the logo of our company and the product. If you click on it you'll get the About box, with a right click you can alter the color and background of the user interface.

3.3.1 Combination bank

This switch LETS you choose between the normal program banks and the new Combination banks. A Combination bank gives you the opportunity to create and manage layer sounds more easily than before. If you switch to combination mode the organization of the presets changes: the bank is divided in 32 groups with 4 presets in each group. If one of these linked presets is selected, 4 presets are loaded in four consecutive parts having the same MIDI channel and can be played together as a layered or split sound.

3.3.2 Sound bank

This field shows you the sound bank used for the current selected part. You can select another sound bank by clicking into this field and choosing the new bank from the open file dialog.

3.3.3 Part selection

On the “Synthese” page you will find all parameters of the synthesizer for the selected part. To view or edit the parameter settings of another part, you first have to select the part using the part selection buttons. The led's for these buttons also indicate incoming MIDI events by flashing.

3.3.4 Sound program

This field shows you the sound program used for the currently selected part. You can select another sound program by clicking into this field and choosing the desired program from A pop-up menu. If you want to edit the name of a preset just click into the text part of this field and type the new name.

3.3.5 File menu

The file menu provides you with all operations you need for managing projects, banks and presets.



- ✎ **Load bank / Save bank**
Loads or saves the bank of the currently selected part.
- ✎ **New bank**
Creates an empty bank with all presets filled with default settings for the selected synth.
- ✎ **Load / Save project**
Loads / saves projects (see “Projects” for details).
- ✎ **Load / Save preset**
Loads / saves the currently selected sound program.

- ✎ **Copy preset**
Copies the currently selected sound program onto the clipboard.
- ✎ **Paste preset**
Copies sound program from the clipboard. If you want to convert a sound program from any of the specialized synth types to **profitERA**, just copy this program and paste it into an empty **profitERA** program. This gives you the opportunity to get free access to all the parameters of a patch which originated from one of the pre-wired synth types.
- ✎ **Paste 8D settings**
Copies only the 8D assignments from the sound program currently in the clipboard.
- ✎ **Paste Modmatrix**
Copies only the modulation matrix from the sound program currently in the clipboard.

✍ **New presets**

The current sound program will be replaced by a default preset with the selected type.

✍ **Convert to profiTERA**

This converts the current program into a profiTERA compatible Patch. This gives you the opportunity to modify presets built with one of the six specialized synth models at a deeper level. But remember that once converted, there's no way to go back...

✍ **Randomize**

This works like a "Slot machine" : Every try is a completely usable (randomly-generated) Sound !

✍ **Waverecorder**

This pops-up the wave recorder described below.

3.3.6 Wavefile recorder



The audio output of TERA can be recorded directly to a disk file. Click into the field to the right of the transport buttons to open a file dialog where you can specify the name of the file to be created. The maximum recording time is only limited by the amount of free disk space. You will need approximately 10.5 Mb free space to record one minute (assuming a samplerate of 44100 Hz). The file is written in Windows PCM format (*.WAV) for PC or Mac AIFF Format in stereo with 16 bit resolution. The sample rate depends on the actual setting in TERA.

If your hard disk is too slow or heavily fragmented you might hear drop-outs in the audio output, but these interruptions are not recorded to the file.

During the recording the record button is red and instead of the filename the current recording time is shown to indicate that the recording is active.

The stop and the pause buttons stops the current recording. The wave file written during recording can't be processed by other programs until the recording itself has been stopped and saved.

After pressing the pause button you can continue recording by re-pressing the record button.

3.3.7 MIDI Control editor

In addition to the possibility of assigning MIDI controllers to parameters with the MIDI learn function you can also edit the MIDI assignments directly with the MIDI control editor. Click on the MIDI symbol and the user interfaces shows a yellow spot on each parameter with a controller number inside. Just click into the spot and enter a new controller number to change the assignment:



3.3.8 MIDI Panic

The right-most top button (which looks like a music note) stops and silences all pending notes for all parts. Use this to cancel „hanging“ notes due to missing note off commands, which can occur if the connection to your external keyboard is broken.

3.3.9 System

You will find the System settings on the page titled “Mixer”. These settings are only available in the stand-alone WindowsXP version. The plug-in version automatically uses the settings of the host software. For the stand alone Mac OS X version this settings are done with the OS X system program “Audio/MIDI Configuration”.



This module allows you to choose the Audio and MIDI Drivers. The changes you make are immediately effective, so you can test your settings easily.

Audio output

This field allows you to select the audio device used for the sound output. TERA currently supports DirectX compatible drivers and ASIO 2.0 drivers for the Windows version.

If you use A DirectX driver, the field „Latency“ shows the actual latency of the system. Usually this is an optimal value for your system and needs no change. But if you encounter problems in the output (such as dropouts) you can set the latency to a higher value. Just click into the latency field and move the mouse up or down while pressing the mouse button to change the value.

If you use ASIO drivers, the button „ASIO“ allows you to call up the settings dialog of your soundcard system.

MIDI Input

This combo box allows you to select from a list of available devices the MIDI input device used to receive MIDI messages from an external source. You can select more than one input driver; the incoming messages from all drivers are then merged together. To control TERA from a Software Sequencer running on the same computer you have to use a loopback device like the well-known „Hubi’s loopback device“ or MIDI Yoke.

MIDI Output

This combo box allows you to select from a list of available devices the MIDI output device used to transmit MIDI controller messages. These messages are sent by user interface elements assigned to a MIDI controller with the MIDI learn function. With this function you can record parameter changes to an external sequencer. To send MIDI commands to an external sequencer program running on the same PC choose a loopback device as MIDI output. The internal sequencer sends notes from all sequencer parts set to a channel other than “TERA” to this output device. These notes are delayed by the time set in the MIDI Output delay field to compensate for delays.

MIDI Output delay

This spin field let you adjust the output delay of the sequencer notes in seconds.

Samplerate

The Sample rate menu sets the sample rate for the audio output. The sample rate selected must be supported by your soundcard, otherwise the quality of audio output could be deteriorated.

UI Design

Click into this field and be surprised !

3.4 Property pages

The main screen of VirSyn TERA is divided into 4 property pages each containing a different part of TERA. Click on the appropriate label tab on the right side to make the desired property page visible. The first 2 property pages, **8D-Access** and **Synth**, contain the sound parameters for the currently selected part. The other two property sheets, **Sequencer** and **Mixer**, are common for all parts. This means that these parameters are independent from the currently active part.



3.5 Knob

The Knob is the most important user interface element in TERA and provides you with easy and precise access to the associated sound parameter. Due to the smooth interpolation of slider movements you will not be able to hear any undesirable parameter stepping effects!



☞ Mouse

Click with the left mouse button on the knob and move it up or down while keeping the mouse button pressed. Pressing the shift key while dragging allows for fine adjustment of the

parameter. If one of the 8D assign buttons “1x” .. “4y” is active then the modulation range regarding the selected 8d-parameter is changed. This range shows up as green (positive modulation) or red (negative modulation) circle segment. Windows XP users can adjust the parameter also with the scroll wheel.

☞ Keyboard

The slider HAS to be selected before it can react to keyboard commands. You can select the slider either by clicking the knob with the left mouse button or even just by moving the mouse over the knob. When the knob is selected the numerical value of the parameter is displayed instead of the label.

Use the cursor keys **?** and **?** to increase or decrease the parameter value by one increment, which is usually 1/128 of the whole parameter range.

Use the keys **Pg ?** and **Pg ?** to increase or decrease the parameter value by 1/10 of the whole parameter range.

Use the keys **Pos 1** and **End** to set the parameter to its maximum or minimum value.

Use the **del** key to set the parameter to its default value. This is usually the zero position of the knob.

☞ Context menu

Click on the slider with the right mouse button and you will get the following context menu:



This context menu is only available for knobs on the Synthese page. This menu provides you with functions to control sound parameters by an external MIDI controller or to assign the parameter to the modulation matrix.

? **Modulation**

Opens a dialog where up to two modulation sources can be connected to this parameter. To get an overview over all assignments in the modulation matrix you can call up a dialog using the key F5.

? **MIDI Learn**

After clicking on the MIDI learn menu item you should turn the knob (or move the slider) of your external MIDI controller hardware. TERA analyzes the incoming stream of MIDI controller messages and assigns the sound parameter to the recognized MIDI controller. After this assignment the sound parameter follows exactly the external controller changes.

? **MIDI Forget**

This command deletes the assignment of a MIDI controller to the sound parameter controlled by this knob.

Tip

With the MIDI Learn function usually only one sound parameter can be controlled by a single MIDI controller. The controlled range thereby is always the full range of the sound parameter. But using the commands explained above you can control more than one parameter each with an individual range:

1. assign all sound parameters you want to control simultaneously to one sound dimension.
2. assign a MIDI controller to this sound dimension using the MIDI learn function in the 8D Access property sheet.

3.6 Listbox

The listbox is used for parameters where you have to select from a list of options, waveforms and so on.



Mouse

With a click into this field you can open the listbox to show a menu where you can select the desired value. Windows XP users can also adjust the parameter with the scroll wheel.

Keyboard

The listbox has to be selected before it can react to keyboard commands. You can select the listbox either by clicking on it with the left mouse or just moving the mouse above it. The selected listbox is displayed with a yellow colored background.

Use the cursor keys **?** and **?** to select the list box entries one after another.

Use the keys **Pos 1** and **End** to select the first or last entry in the listbox.

3.7 Spin fields

Spin fields are a space-saving alternative to sliders and knobs for parameter input.



✎ Mouse

Click with the left mouse button on the spin field and move it up or down while keeping the mouse button pressed. If one of the 8D assign buttons “1x” .. “4y” is active then the modulation range regarding the selected 8d-parameter is changed. This range shows up as green (positive modulation) or red (negative modulation) bar. Windows XP users can also adjust the parameter with the scroll wheel.

✎ Keyboard

The spin field has to be selected before it can react to keyboard commands. You can select the spin field either by clicking on it with the left mouse button or just by moving the mouse over the field. When the spin field is selected the numerical value of the parameter is displayed instead of the label.

Use the cursor keys **←** and **→** to increase or decrease the parameter value by one increment, which is usually 1/128 of the whole parameter range.

Use the keys **Pg ↑** and **Pg ↓** to increase or decrease the parameter value by 1/10 of the whole parameter range.

Use the keys **Pos 1** and **End** to set the parameter to its maximum or minimum value.

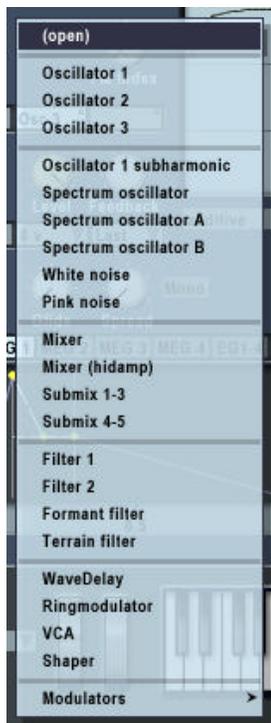
Use the **del** key to set the parameter to its default value. This is usually the zero position of the knob.

3.8 Patching of modules



If you create a new sound program starting from scratch, you first have to connect the modules in the desired way. The audio input windows of the modules represent the patch cables. They are the input jackets where you can plug in a virtual cable connected to the output of another module. The

name of the connected output is shown in the audio input window. TERA automatically deactivates modules whose outputs are not used in order to preserve processor resources.



Click on one of the audio input listboxes and you'll get a menu where you can select the output you want to be connected to this audio input.

The menu on the left shows you all the available module outputs of TERA.

(open) is not really an output; use it if you don't want this input be connected to any output.

4 Synthesizer models

4.1 Introduction

In addition to TERA's main modular synthesizer **profitERA**, there are also six pre-wired synthesizer models which greatly simplify the creation of your own sound presets. These models lead you through a broad range of synthesis techniques, both traditional and innovative. The wiring of the modules and the routings for the modulation matrix are predefined. But sound presets made with one of this models can easily be transferred to the more powerful and complex modular synth **profitERA** through the copy and paste preset functionality provided by the main file menu. There you can process your preset further with the full functionality of TERA's modular system.

4.2 MicroTERA



The waveshaper is the core of microTERA's synthesis engine. It works as a distortion unit that takes (for example) a sine wave as input and delivers an arbitrary spectrum as output. In contrast to other synths using also the waveshaping synthesis, microTERA uses a new technique to define the output spectrum of the waveshaper.

There are three methods used in TERA to specify the shape:

1. The simplest method is just drawing the waveshaping curve itself. This method mostly just gives "distorted" results and is not very intuitive. It is very hard to control the resulting spectrum.
2. The second method comes from the fact that FM synthesis can be simulated by a waveshaper. Here the user controls the harmonics of the FM modulator and the synth calculates the needed waveshape from this specification. This gives much better control of the resulting spectrum, but always sounds "FMish".
3. microTERA uses a new technique to specify the resulting spectrum directly: The user specifies directly the harmonic amplitudes for each partial the resulting spectrum should have if the waveshaper is fed by a full signal sinus tone. This gives you total control and you can precisely define the spectrum. The spectrum variations arising from the modulation of the input level of the waveshaper (Parameter "Spectrum") are also distinctly different from those of the FM synthesis. Therefore the resulting range of sounds is very unique and can't be achieved by other synthesis methods.

The basic structure of microTERA is very simple: 3 sine oscillators are feeding a waveshaper and the output of this is fed to the amp. It's uniqueness lies completely in the waveshape algorithm. As soon as the level is getting lower the spectrum is getting darker. Therefore the Spectrum parameter can be seen as A kind of complex "Cutoff". Things get even more interesting when more than one sine tone with A slightly different frequency is fed into the waveshaper: due to the non-linear behaviour of the shape, the new harmonic partials (or inharmonic partials if the feeding sine tones have an irrational frequency ratio) are generated by the shape itself. If the pitch difference between the sine waves is very small, this results in subtle timbre changes in the output. Another speciality is the Noise parameter. It's not some commonly used pink or white noise generator, but instead it's a feedback loop in the waveshape itself that results in a sonically chaotic behaviour which is perceived as different kinds of noise.

These simple methods create an astonishing variety of sounds which are very unique and can't be reproduced by any other contemporary synth!

4.3 SpecTERA



This synth focuses on the spectrum module of TERA. The spectrum module can be thought of as a set of 6 Oscillators. Each oscillator has two sources that can be morphed in real-time. Each source has a partial editor to define the harmonics of the source and an arbitrary formant filter with a real-time "Spectrum" parameter which shifts the whole filter in the frequency range and sounds like the "Cutoff" parameter of a conventional synth filter. Technically this module combines the power of Wavetable synthesis with an easy to use graphical user interface.

These six oscillators can be detuned in a wide range and the number and base tunes can be selected from a set of presets. The output of this complex module - which is in fact a synth in itself - can then be further modified by a conventional filter module with selectable type (Lowpass/Highpass/Bandpass/Bandstop) and slope (24/18/12 dB/Octave).

4.4 FmTERA



This synth looks at a first glance like a simple 3 operator FM synth, but if you look closer it goes far beyond classical FM synthesis. The carrier signal is first fed into one of the 2 multimode filters of TERA before they modulate the other two operators! This gives you a unique new type of control over this kind of synthesis making it much easier to create interesting sounds which are not only in the typical FM range.

4.5 FormanTERA



This synth is focused on the Formant filter module of TERA. The 3 main oscillators of TERA are feeding the formant filter and a standard filter in parallel. The output of both filter modules can then be morphed for the resulting output. The formant filter itself consists of three bandpass/bandstop filters with the controllable parameters "Frequency", "Resonance" and "Level". This kind of filter is ideally suited for voice-like sounds. Using the new multi-segment envelopes it's even possible to teach TERA how to speak!

4.6 WaveTERA



This synth is focused on the Wavedelay module of TERA, giving you access to a wide range of physical modelling instruments. Strings, Guitars and Woodwind instruments are especially conducive to this synthesis technique, but the open structure of this synth can be used for creating much more weird things too...

The signal flow is very easy: an energy signal with a detailed user definable spectrum represents the "Excitation energy" for the modelled instrument. This excitation signal is shaped by an envelope generator and fed into the wave delay loop. This delay loop has a delay length normally proportional to the tune of the resulting sound representing, for example, the length of the modelled guitar string. The cutoff frequency and feedback parameter of this loop determines how the spectrum evolves over time. The resulting signal is then further modified by a standard multimode filter to give the user more control over the timbral evolution of the sound.

4.7 AnaTERA



This synth is an enhanced classically modelled analogue synth. The 3 main oscillators of TERA are feeding the 2 standard multimode filters in parallel. The output of both filter modules can then be morphed for the resulting output. But keep in mind that despite this simple base structure you can do much more than with any previously existing analogue synths due to the multi-segment envelopes and the possibility to specify your own waveforms for each oscillator.

Taking into account that Osc 1 can be put in a special mode where it's split into six separate oscillators you can use up to 8 Oscillators for a single patch !

5 Modular synthesizer

5.1 Overview

The modular synthesizer called **profiTERA** is the most flexible synth of TERA. It provides you with up to 16 instruments (parts) you can use for your music production. On the property sheet **Part mixer** you can set the number of voices simultaneously available for each part separately:

- 1 voice = monophonic synthesizer
- 2.. 32 Voices = polyphonic synthesizer

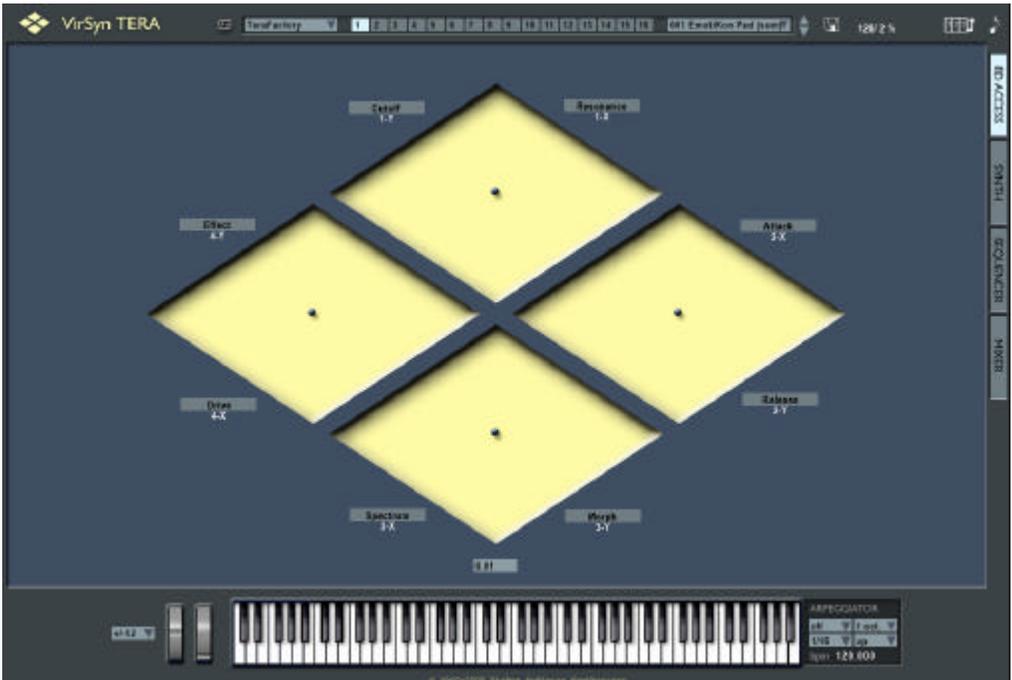
As the name „Modular synthesizer“ implies, every synthesizer is constructed out of independent modules and the synthesizer design is determined through the connections between these modules.

AS TERA works with a fixed number of modules, this gives you the following advantages:

- The patch connections between the modules are treated like any other sound parameter. In that way, connections are stored together with the sound parameters in the sound program. Therefore every sound program contains the construction plan describing the synthesizer structure used !
- The user interface of differently constructed synthesizers is always the same, independent of the number of modules used and their connections. This allows you to switch between the 16 instruments with the part buttons without changing the user interface. The assignment of MIDI controllers to sound parameters is also independent of the synthesizer construction.
- The fixed module structure allowed us to optimize the synthesis algorithms. This has the consequence of reducing processor contention and increasing THE voice yield. The advantages are especially apparent in complex set-ups with many differently constructed instruments in a project.

5.2 8D Sound Access

The property sheet **8D Sound Access** provides you with a unique user interface to navigate through the user definable sound space consisting of 8 independent sound dimensions.



Every dimension can control up to 8 sound parameters simultaneously. This means that you can control up to 64 sound parameters (8 dimensions x 8 parameters) with a single click of the mouse button. Or you can use up to 8 MIDI controllers to travel remotely through this sound space. You can control the travelling speed with the parameter **Speed**, giving you some kind of „8-dimensional portamento“.

The four diamond shaped objects represent the user interface to this 8 dimensional sound space. Every diamond represents 2 dimensions (X and Y). The actual sound position is indicated by the four small black spots inside each of the diamond shapes.

For easy navigation through this sound space TERA provides you with a range of possible navigation methods:

1. Click into one of the four diamond shapes to alter the sound position for 2 dimensions simultaneously.
2. Click into one of the four diamond shapes and hold the mouse button while moving through the sound space.
3. Use method 1 or 2 while holding the **Ctrl**-Key to alter the sound position for all 8 sound dimensions simultaneously.
4. Use the **MIDI learn** function to assign a MIDI controller to one or more of the 8 sound dimensions. This provides you with the possibility to remotely control your sound voyage with an external sequencer or with a hardware MIDI controller. To assign a dimension to a MIDI controller just right click near the dimension you want to be controlled.

To program the sound space you have to assign different sound parameters to some or all of the 8 dimensions. To assign a sound parameter to one of the 8 dimensions you first have to select the property page "Synthese". Then choose the dimension by clicking on one of the buttons shown on the right. Now you can adjust the range of parameter change as described in the section Spinfield/Knob.

With the **Speed** slider you can adjust the time needed to smoothly reach a new position in the sound space. With this feature you can glide from one sound position to the next. This parameter is adjustable in a range from 10 ms to 100 seconds. All sound parameters change continuously from their starting position to the end position in this time.



5.3 Modulation Matrix

The modulation matrix is used to control any sound parameter by a selectable modulation source. The modulation matrix routes a modulation source, selectable out of 32 sources, to a modulation destination, also selectable from 87 possible destinations. The modulation amount/depth is adjustable. The usable range for the modulation amount is optimized for every combination of modulation source and destination. Up to 20 routings can be used simultaneously. Please note that this matrix can only be accessed from **profiTERA**-type synths; all others have predefined routings. The settings of the modulation matrix can be copied to another preset with the previously described file menu.

The modulation amount of the routings 11-20 can be used as destination in the routings 1-10. This gives you a second level of modulation.

In addition to the modulation matrix VirSyn has two fixed routings BUILT-in:

1. pitch bender >> pitch
2. keytrack >> pitch of all oscillators, cut-off frequency for the two multimode FILTERS and the pitch of the WaveDelay module.

Programming the modulation matrix can be done in two ways:

1. Click on “Matrix 1” or “Matrix 2” to open all Modulations together:



2. Right clicking on the parameter you want to modulate and opening a dialog to directly edit two modulations:



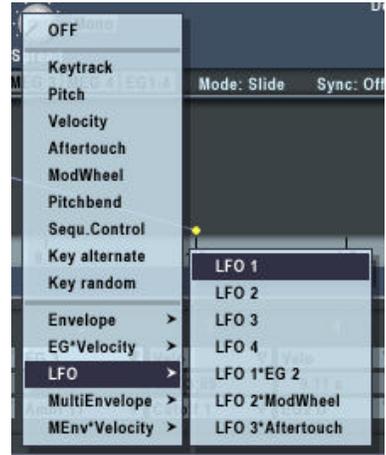
The user interface elements:

- ✍ **Modulation source**
Source which controls the sound parameter specified in the modulation destination.
- ✍ **Modulation amount**
Knob to adjust the modulation amount. The amount can be positive or negative, controlling the depth and direction of the modulation. The amount is displayed in physical units, depending on the selected destination.
- ✍ **Modulation destination**
Sound parameter controlled by the modulation source.

The following steps create a new routing:

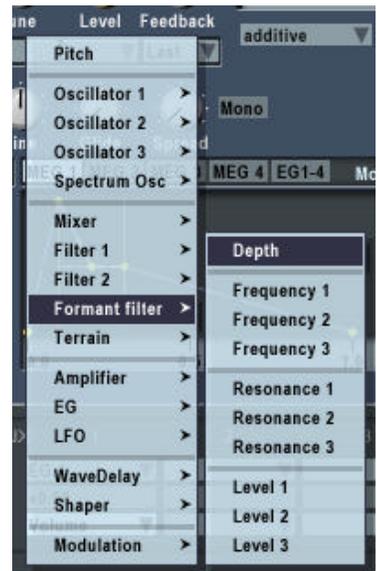
✎ **Select modulating source**

Select a modulation source by clicking into the source field with the right mouse button and selecting a source from the context menu.



✎ **Select modulated destination**

Select a modulation destination by clicking into the destination field with the right mouse button and selecting a source from the context menu.



✎ **Adjust modulation amount**

Adjust the modulation amount to get the desired depth and direction.

5.4 Modules

This chapter describes every module of TERA's synthesizer section in detail. These modules are the basic building blocks you can use to construct your own synthesizer. Modules which are not used are automatically switched off to reduce processor contention. TERA checks each module for whether its output is used by any active module. If not the module is switched off.

Module overview:

- ✍ **Keyboard**
Master tuning / Portamento.
- ✍ **Arpeggiator**
Standard arpeggio function.
- ✍ **Sample Oscillator**
Multisample MODULE "hidden" in Oscillator 1..3.
- ✍ **Oscillator 1**
Main oscillator with wave modulation.
- ✍ **Oscillator 2**
Oscillator with wave modulation and FM input.
- ✍ **Oscillator 3**
Oscillator with FM input.
- ✍ **Spectrum Oscillator**
6 Oscillators with detune and spectrum filter.
- ✍ **Ringmodulator**
Multiplication of two input signals.
- ✍ **VCA**
Voltage controlled amplifier.
- ✍ **Noisegenerator**
White and Red noise.
- ✍ **Wavemixer**
Mixer 3 + 2 in 1.

- ✍ **Shaper**
Waveshaping module (dynamic distortion).
- ✍ **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.
- ✍ **Terrain filter**
Two morphable 16 pol filter with predefined characteristic.
- ✍ **WaveDelay**
Physical MODELLING for simulating string and wind instruments.
- ✍ **Amplifier**
Final amplifier with overdrive and LoFi effects.
- ✍ **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 synchronization and Sample&Hold function.

5.4.1 Keyboard

The keyboard module provides you with access to parameters influencing the processing of note events coming from an external Keyboard or the internal sequencer:



✍ **Semitone / fine**
Master tuning

✍ **Glide/Glissando**
Smooth or stepwise chromatic 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 seconds to 132 seconds. This effect is typical for many lead sounds on monophonic synthesizers, but can also be used here in polyphonic mode.

✍ **fingered**
In normal mode all notes played will have the glide effect. In fingered mode only legato-played notes will have this effect.

✍ **Priority**
The key priority function controls which note will be played if you press more than one key in monophonic mode. With key priority **Last** the last note will always be played. With key priority **Low** or **High** the lowest or the highest key will be played.

If the currently played key is released with key priority low or high then the lowest/highest note restarts playing if you have pressed more than one key. This is the same behaviour as on most classic monophonic synthesizers.

✍ **Bend**
Adjusts the bend range of the pitch bender.

✍ **Mono**

With this function the sound program can only be played in monophonic mode. If you assign more than one voice to this part, then all voices are played with the same base tune which can be detuned slightly by the spread parameter.

✍ **Spread**

Controls the detuning of the voices in Mono mode.

Tip

For the typical 303 Slide effect use fingered Portamento and switch the filter envelope to triggermode „single“ !

5.4.2 Arpeggiator

In addition to the high level arpeggio functions of the built in sequencer, TERA 3 now offers a simple standard arpeggiator for each part.



✍ **Arpeggio type**

For each part you can switch on an independent arpeggiator. With the hold mode the arpeggio doesn't stop when all keys are released.

✍ **Clock**

Here you can select the base note duration and predefined rhythms.

✍ **Range**

The range for the arpeggiated notes from 1 to 8 octaves

✍ **Mode**

Controls how the held chords are broken into single note sequences. Possible values are "up", "down", "alternate" and "random".

5.4.3 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.

The parameters **Semitone** and **Fine** of Oscillator 1 determines the base pitch used for all oscillators AND the WaveDelay and influences the tracking of the filters. These parameters also take effect on the modulation sources **Pitch** and **Keytrack**.



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.

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. Select "Sample" to open the multi-sample dialog described later.

Semitone

Coarse tuning of the oscillator relative to the master tuning.

FM off: Sets tuning within a range of +/- 48 semitones.

FM on: Sets tuning in integer multiples of the master tuning. 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.

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)

✍ **WM Wave modulation**

With wave modulation activated the oscillator's output is added to a phase shifted copy of itself. The phase shift is controlled by the wave modulation slider.

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.

✍ **Phase**

When phase synchronization is activated all oscillators will restart its waveform cycle with every key pressed on the keyboard. This means that if you repeat playing the same note, every note sounds exactly like the one before. This is very important for drum sounds !

Without phase synchronization the oscillators generate their waveforms continuously, and every starting note picks up AT a random phase. This is the way old analog synthesizers worked, and one of the main reasons for the warmth of their sound. Compare A slightly detuned sound with and without phase synchronization and you'll hear the difference immediately.

Phase synchronization also works for the noise generator. Use this for snare drums, hihats, ...

Extra feature

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.

5.4.4 Oscillator 2

Oscillator 2 HAS a linear FM input for FM synthesis in addition to the functions of oscillator 1.

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.

Sync

With synchronization activated the waveform cycle will be restarted with every cycle of oscillator 1. The pitch of the output signal will then be controlled by the master oscillator. The pitch of the synchronized oscillator will now control the timbre of the output.

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 which determine the waveform. Select “Sample” to open the multisample dialog described later.

Semitone

Coarse tuning of the oscillator relative to the master tuning.

FM off: Sets tuning within a range of +/- 48 semitones.

FM on: Sets tuning in integer multiples of the master tuning. 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.

✍ **Fine**

Fine tuning of the oscillator.

FM off: 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)

FM on: Sets tuning in fractional multiples of the master tuning.

✍ **WM Wave modulation**

With wave modulation activated the oscillator's output is added to a phase shifted copy of itself. The phase shift is controlled by the wave modulation slider.

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

✍ **FM-index**

With this input you can modulate the oscillator frequency with the output signal of another module, for example a filter or an oscillator. The modulation source can be selected by clicking into the field displaying the currently connected source. The amount of the modulation, called **Modulation index** in FM synthesis terminology, can be adjusted in the the range of 1 to 16.

Tip

For typical synthesizer sync sounds, modulate the pitch of the synchronized oscillator by an envelope or a LFO, keeping the pitch higher than the pitch of the master oscillator 1.

5.4.5 Oscillator 3

Oscillator 3 HAS the same functions as oscillator 2, except the wave modulation feature.

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.



Sync

With synchronization activated the waveform cycle will be restarted with every cycle of oscillator 1. The pitch of the output signal will then be controlled by the master oscillator. The pitch of the synchronized oscillator will now control the timbre of the output.

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. Select “Sample” to open the multi-sample dialog described later.

Semitone

Coarse tuning of the oscillator relative to the master tuning.

FM off: Sets tuning within a range of +/- 48 semitones.

FM on: Sets tuning in integer multiples of the master tuning. 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.

✍ **Fine**

Fine tuning of the oscillator.

FM off: 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)

FM on: Sets tuning in fractional multiples of the master tuning.

✍ **FM**

With this input you can modulate the oscillator frequency with the output signal of another module, for example a filter or an oscillator. The modulation source can be selected by clicking into the field displaying the currently connected source. The amount of the modulation, called **Modulation index** in FM synthesis terminology, can be adjusted in the the range of 1 to 16.

5.4.6 Sample MODULE

Each of the three main oscillators can use now samples instead of the built in waveforms. The samples are organized in up to 16 key zones and 2 morph zones making up for a total of 32 sample files for each oscillator. Due to it's high quality re-sampling algorithm you can use a single sample for the whole tonal range ! After selecting "Sample" as the waveform in one of the three main oscillators the following multi sample dialog opens:



In the upper section you'll find parameters valid for all samples:

- ✎ **New multi sample**
Creates a new multisample.
- ✎ **Save multi sample**
Saves the data for the multi sample in a file structure independent from the presets. When saving the preset, all changes done in the multi-sample dialog are saved automatically.

✍ **Open multi sample**

Opens an existing multi sample from a file. Every multi sample consists of a description file with the extension “vsi” and a folder containing the used samples.

✍ **Morph source selection**

The selected morph source controls the smooth blending between Samples in Zone A and Zone B.

✍ **Modulation curve**

This parameter influences the way in which the blending is done: A value of zero means linear blending.

✍ **Play mode**

One shot – samples are played once from start to end.

Loop – samples are played endlessly according to the loop points set.

✍ **Loop fade**

Controls the blending of the loop start and endpoint. A Loop fade of 0 means no blending at all – the hard transition between the loop end and start point can lead to click noise. With higher values the transition goes more and more smoothly.

In the lower section you can load and edit the samples themselves:

✍ **Samples A / Samples B**

Click into this field to load a sample from a WAV- or AIFF file. With a right click and selecting „Remove“ you can unload the sample.

✍ **Orig key**

Use to set the original key of the Sample.

✍ **Hi key**

Use to set the highest key this sample should be used. The lowest key is determined automatically from the settings of the other samples.

✍ **Loop start**

Sets the loop start point relative to the length of the sample in percent.

✍ **Loop end**

Sets the loop end point relative to the length of the sample in percent.

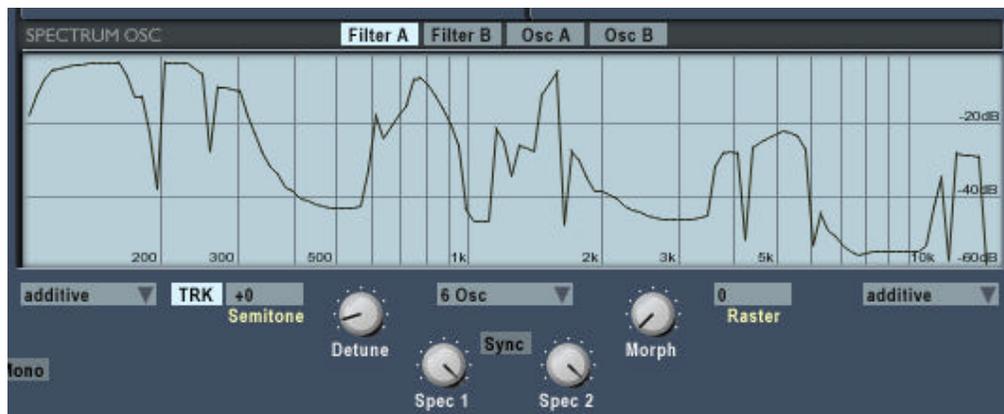
Remark: if one of the oscillators uses a multi-sample, some of the standard functionality of the oscillators like FM or wave modulation are no longer functional!

5.4.7 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.

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.



Description of user interface elements:

 **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 |
| 6 Osc Major | - 6 oscs major chord tuning |
| 6 Osc Minor | - 6 oscs minor chord tuning |

☞ **Spectrum window**

In the spectrum window you can directly draw the desired filter and oscillator spectrum. Just click with the mouse into the window and draw the spectrum by dragging the mouse. When the mouse button is released the changes are in effect immediately. “Filter A” and “Filter “B” selects the two different filter curves for editing. If you hold the shift or the ctrl key while drawing, comb like structures are drawn.



With a right button mouse click into the spectrum window you’ll get a menu with the following 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

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

☞ **Sync**

With Sync activated the modulation of the spectrum parameter affects both Sources A and B. Otherwise Source A and B can be modulated independently.

✎ 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 inharmonic.

✎ Semitone

Coarse tuning of the oscillator relative to the master tuning of oscillator 1.

Sets tuning within a range of +/- 48 semitones.

✎ Spectrum modulation 1 / 2

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. Spec 1 and Spec 2 independently influence the spectrum shift for two different subsets of oscillators.

✎ Synthesis method

In addition to the additive sine synthesis used in TERA 2 as a basis for the wavetable generation process the following methods can be used:

- Wavelet Synthesis
- Walsh function Synthesis
- Operator Synthesis

This parameter can be set for two different subset of oscillators independently.

✍ **Morphing**

With the morph parameter you can blend between the two filter curves.

✍ **Raster**

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

0 – all harmonics present.

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

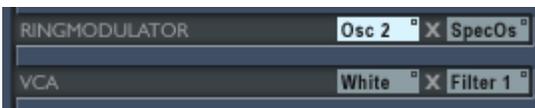
2 – only every third.

3 - ...

5.4.8 Ringmodulator

This module combines two input signals by adding and subtracting them together. 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: two sine tones with 1300Hz and 700Hz.



✎ Input 1

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

✎ Input 2

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

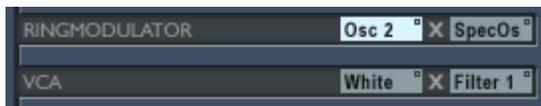
Note

To get an audible output both inputs have to be assigned to audio sources !

5.4.9 VCA

This module works like a voltage controlled amplifier. The signal of input 1 is amplitude modulated by the signal on input 2. The resulting spectrum contains the sum and differences of the frequency components of the two input signals like the ringmodulator, but in addition it also contains the original frequency. 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: two sine tones with 1300Hz and 700Hz, 1000 Hz and 300 Hz.



- 🔗 **Input 1**
Choose an audio source for input 1 by clicking into this field and selecting a source from the popup menu.
- 🔗 **Input 2**
Choose an audio source for input 2 by clicking into this field and selecting a source from the popup menu.

Note

To get an audible output both inputs have to be assigned to audio sources !

5.4.10 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 context menu of the module inputs as one of the selectable sound sources.

5.4.11 Wave 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.
- ✍ **Mix**
Mix of all input signals.
- ✍ **Mix (hidamp)**
Mix of all input signals with a filter to damp high frequencies.

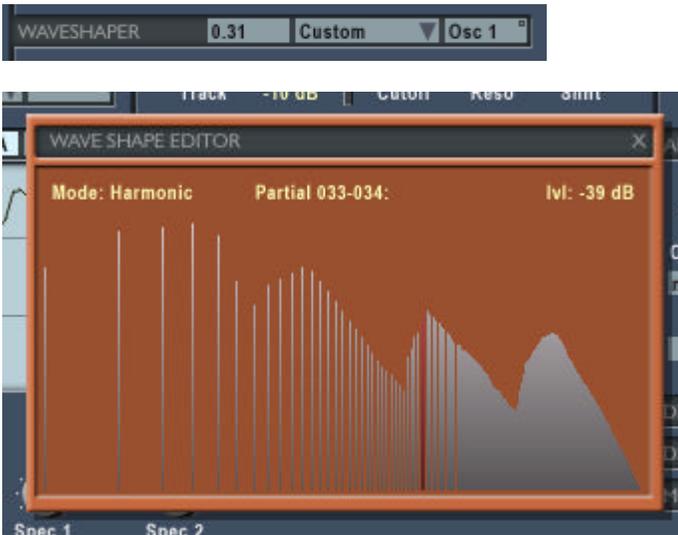
Therefore you can also use this mixer as two independent mixers with, respectively, 2 and 3 inputs.



- ✍ **Input 1-5**
Choose the audio source for the inputs 1 to 5 by clicking into the appropriate field and selecting an audio source from the popup menu.

5.4.12 Shaper

The waveshaper 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 1 by clicking 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 which lets you specify the shape for your own (for details see “microTERA”)

✎ 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.

5.4.13 Multimode filter

The two multimode filters are the most important sound design tools in TERA. They simulate the characteristics of the filter types used in traditional analogue synthesizers. With the built-in overdrive you get a soft distortion. The input level meter shows you the degree of distortion. Several filter characteristics with 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**

The lowpass filter attenuates frequencies above the cutoff frequency with 24dB/octave giving a **warm and fat sound**.

🔊 **Highpass 24dB**

The highpass filter attenuates frequencies below the cutoff frequency with 24dB/octave giving a **bright and sharp sound**.

🔊 **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 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 cc**

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 cc**

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:

✎ **Filter 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.

✎ **Drive**

Adjusts the filter input level in the range of -10dB to $+30\text{dB}$. The actual peak level of the input is indicated by the peak level meter on the right side of the slider. 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.

✍ **Reson**

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 (even if no input source is connected!).

✍ **F-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.

5.4.14 Terrain filter

This module allows for real-time morphing between two 16 pole filters selectable from a set of predefined vocal filters.



The Terrain filter supplies the following parameters:

- ✍ **Input**
Select the audio source for THE filter.
- ✍ **Filter selection 1 / 2**
Selects the filter characteristic from a list of predefined presets.
- ✍ **Cutoff**
Shifts the base frequency of the filter.
- ✍ **Resonance**
Changes the resonance characteristic of the filter.
- ✍ **Morph**
Morphs between the two preselected filter characteristics.

5.4.15 Formant filter

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

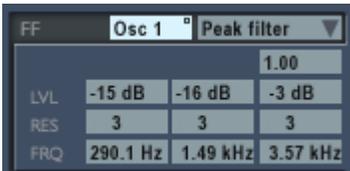
✎ **3 band pass filter in parallel**

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 the modulation matrix !



✎ **Input**

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

✎ **Bandpass/Bandstop**

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.

✎ **Frequency 1 .. 3**

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

✎ **Resonance 1 .. 3**

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

$$\text{bandwidth [Hz]} = \text{center frequency [Hz]} / \text{resonance}$$

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

✎ **Level 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.

Tip

With the characteristic set to „bandreject“ and modulating the center frequency slightly with different LFOs you get „natural“ sounding pads. Using extreme values for the modulation amount you'll get phaser-like sounds.

5.4.16 WaveDelay

This module is the basic building block for physical modelling synthesis. First of all it sums the signals from its two audio inputs (In1 and In2), 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 cut-off 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 **Input 2** to get a closed feedback loop. This loop needs some initial energy put into **Input 1** to resonate and thereby produce a sound. To accomplish this, patch some oscillators or the noise generator to Input 1 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 center 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 input 2 IS usually the Wave Delay 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.

The WaveDelay parameter:



✍ **Input 1**
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.
- ✍ **Input 2**
Second input for the delay line. The input level can be adjusted with the feedback slider. For physical modelling synthesis the WaveDelay output has to be patched to this input.
- ✍ **Track**
This button activates keyboard tracking. When switched on the WaveDelay will track the keyboard at the rate of one halftone for each key. When switched off the WaveDelay 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 input 2. If the WaveDelay output is patched to input 2 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

Tips

The envelope used to modulate the level of input 1 controls the roughness of the sound attack. A very short impulse achieved by short attack/decay time settings gives you a clean attack to the sound. Scaling the attack/decay times with keytracking simulates the natural behaviour of instruments having rougher attacks on lower pitches. Modulate the envelope times using **Keytrack** as a modulation source to achieve this effect.

To have better control over the decay time for different pitches modulate the **Feedback** parameter with the modulation source **Keytrack**.

Due to the construction of the Wave Delay, modulating the pitch with an LFO, for example, can cause some noise.

5.4.17 Amplifier

The amplifier is the last module in every program. This module mixes the selected input source 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. In contrast the pan knob in the part mixer acts on all voices of a part. The stereo output is routed to the global and insert effects and to the part mixer.

In addition to this base functionality the following special effects are built into the amplifier module:

☞ **Overdrive**

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.

☞ **Sample**

With this parameter you can change the sample-rate for this program independent of the sample-rate of programs used in other parts. The setting here is always relative to the physical sample-rate used by the soundcard set in the system menu. Use this function to get the kind of Lo-fi sounds comparable to old digital synths. With sub-sampling activated, the processor contention is reduced, and you have more simultaneous voices available. Bass sounds especially often don't need these high sample-rates.

☞ **Bit reducer**

The amplitude resolution of the amplifier output can be reduced down to 1 bit by this parameter. This adds some kind of „digital“ noise to the output comparable to the noise produced by the first generation of digital synthesizers. Good for Lo-Fi sounds !

The Amplifier parameter:



- ✎ **Input A/B**
Choose an audio source by clicking into this field and selecting a source from the context menu.
- ✎ **X-Fader**
Mix between input A and B

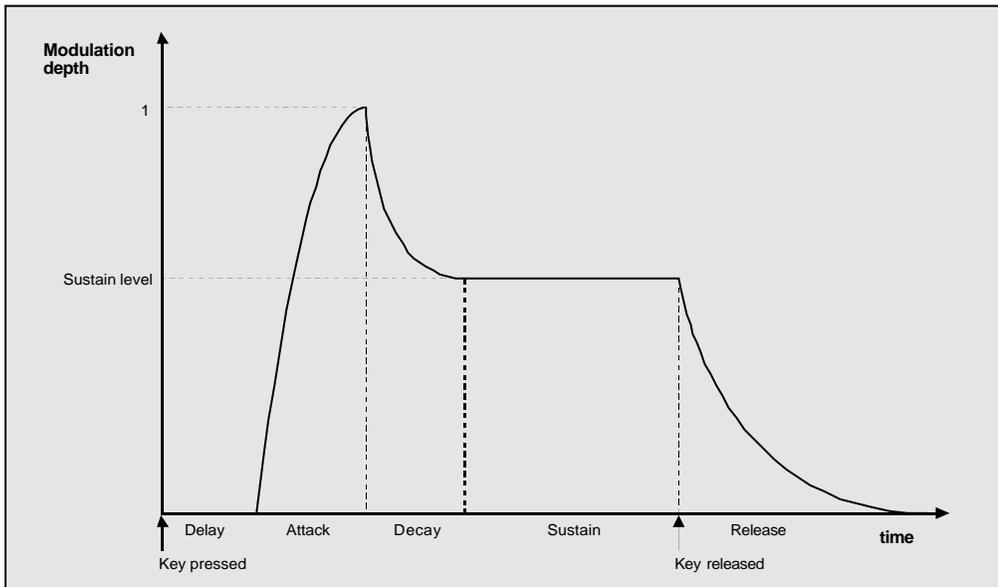
- ✎ **Overdrive**
Adjust the amount of ranging from „Off“ to +64dB (maximal distortion).
- ✎ **Volume**
Output level ranging from -40dB to +40dB
- ✎ **Pan**
Adjusts the position of the sound in the stereo panorama. Opposed to the pan slider in the part mixer this one works independently for each voice ! If for example the pan is modulated by an LFO this means every voice can have an independent position.
- ✎ **Sample**
Virtual reduction of the samplerate for this part only:
 - 1/1 no reduction
 - 1/2 only every second sample is used (half samplerate)
 - 1/4 only every fourth sample is used (quarter samplerate)
- ✎ **Reduce**
Reduction of the output resolution. You can select between maximum resolution (32-bit) and reduction to 15 .. 1 Bit.
- ✎ **Auxilliary 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.

5.4.18 Standard envelope generator

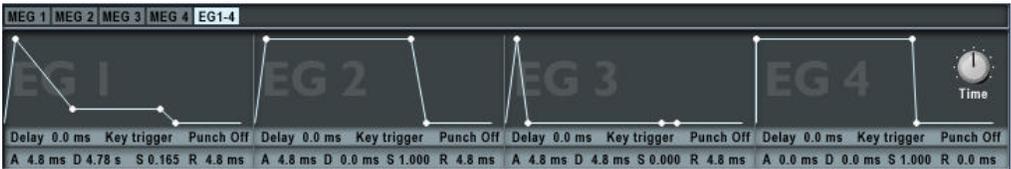
The envelope generator let you shape sound parameters over the time from when a key is hit, then when the key is hold, and how the sound decays after the key is released. You can also trigger the envelope by one of the four LFOs instead of hitting a key. TERA uses envelopes with ADSR characteristic, like most analogue synthesizers do, extended by a delay phase and the possibility to thicken short attack/decay envelopes with the **punch** option.

For every voice TERA provides you with four independent envelope generators.

Phase transition diagram of TERA Envelopes:



The Parameter of the standard envelope generators:



Delay

Sets the delay time between triggering and 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, that means from start the signal builds up quickly and rising slower 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, that means from start the signal decays quickly and falling slower 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 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, that means from start the signal decays quickly and falling slower 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 a AD only envelope.

Punch

With the punch switch the start of the decay phase is delayed 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.

The parameters attack / decay / sustain and release can also be edited directly in the graphic screen.

✍ **Trigger mode**

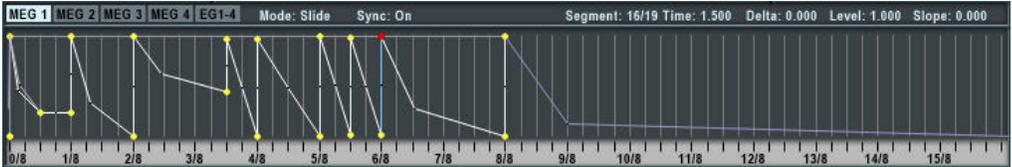
With the trigger mode you decide when and how the envelope is triggered. The following modes are available:

- ? **Key trigger**
The envelope starts on key down with the last actual level.
- ? **Key reset**
The envelope starts on key down with the level 0.
- ? **Key single**
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.

Note

The envelope generators in TERA provide you with extreme short time settings below 1/1000 seconds. Modulating the filter cutoff frequency with extremely fast envelopes can lead to oscillations with high sound levels. Reduce the output level of the filter if you want to avoid the resulting distortion.

5.4.19 Multisegment Envelopes



TERA provides you with four multi segment envelopes which can be used as a modulation source for many purposes. The envelopes can have up to 64 time/level segments giving you all from simple Attack/Decay style envelopes up to complex rhythmic patterns which can even be synchronized to MIDI tempo.

The four envelopes are normally not independent from each other: They have the same number of segments and the time when a specific segment starts is always the same for all four.

If you want one or more of these envelopes completely independent of the others you just have to set the option “Sync to MEGx” to “Off”. Then this envelope works independently from all others.

You can change the time or level of any segment by clicking on the small circle representing the segment and dragging it into the desired position. All editings are in effect immediately !

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

To insert a new segment click into the desired position in the envelope window while holding the shift-key down.

To delete a segment click into the small circle representing the segment you want to delete while holding the shift-key down.

The two black horizontal lines are the loop marker positions for the envelope. They are managed automatically by TERA. 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 simultaneously the following way: click on the time bar below the envelope window and drag up/down for zooming and left / 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 click into the mode field above the envelope.

To create tempo synchronized envelopes click 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.

The main envelope editor figured below provides you with a context menu for copying and pasting envelopes through the clipboard and for loading and saving your own envelope presets. In addition to this you can simply select there from about 64 predefined envelopes.

5.4.20 LFO

This module provides you with a Low Frequency Oscillator useful for periodic and random sound parameter modulations. The LFO can produce five 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 external/internal MIDI Clock. The phase of the waveform can be controlled by the wave sync parameter.



Waveform
Selects one of the following waveforms:

- ? **Sine**
- ? **Triangle**
- ? **Square**
- ? **Ramp up**
- ? **Ramp down**
- ? **S & H**
- ? **Random - Aperiodic waveform.**

Using the waveform S & H <n> 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.

✍ **Rate**

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

If the Wave sync parameter is set to MIDI then the rate depends on the tempo of the internal sequencer or the MIDI clock. 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 synchronization. The following modes are available:

? **Part**

All voices of a part are modulated in sync.

? **MIDI**

The rate is synced to the external MIDI Clock or to the internal sequencer tempo (depending on the settings in the system menu **external sync**).

? **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.

6 Effects

6.1 Overview

Global Effects

The global effects Reverb and Chorus are shared by all parts and exists only once. You can use them for all 16 Parts. For every part you can adjust the effect send level for the Reverb and the Chorus effect in the part mixer. The global effect output is added to the main output. The effect return levels can also be adjusted in the part mixer. The global effect settings are stored together with the project.

Part Effects

Every part have its own set of three part effects (Overdrive/Delay/Modulation). They are designed as insert effects with the input coming from the output of the amplifier module of the synthesizer and the output of the last effect (Modulation) is patched to the channel input of the part mixer. . The part effect settings are stored together with the sound program.

6.2 Global effects

6.2.1 Reverb

The reverberation effect is used to simulate the sound reflections off walls in different kind of rooms. This reflections disperse the sound, enriching it by overlapping the sound with its reflection. The reverb effect is global, this means only one reverberation type can be used at one time, but every part has an individual amount of reverberation set in the Mix page. The Reverb output is mixed to the main output in the Part mixer.

The effect parameters can be set individually, or they can be selected from a list of 32 global presets. The parameters are stored together with the project.



☞ **ON/OFF**

Switches the reverb on and off. If the reverb effect is not used, it should be switched off to lower the processor contention.

☞ **Select Preset**

Selects from 32 global presets.

☞ **Delay**

Sets the amount of time that elapses between receiving a direct sound and its first reflection. This time correlates with the perceived size of the simulated room. The adjustable range is from 0 ms to 100 ms.

✍ **Time**

The reverberation time indicates the time required for a sound to die away to -60 dB of its amplitude after its source shuts off. This time correlates with the perceived size of the simulated room. The adjustable range is from 50 ms to 330 seconds. In the uppermost position, labeled „endless“ the sound never dies out.

✍ **Size**

The room size parameter sets the average distance between the reflecting walls. The adjustable range is from 1 meter to 100 meters.

✍ **Structure**

The room structure parameter sets the relative distance between walls facing each other. The adjustable range is from 0 to 1. A high value correspond to equal distances between walls facing each other. For natural sounding rooms take a value around 0.7 .

✍ **Fusion**

The fusion parameter controls the echo density of the resulting reverberant sound. The adjustable range is from 0 to 1. A high value correspond to high echo density.

✍ **Damping**

The damping parameter simulates the different absorption of high frequencies by the wall material. The adjustable range is from 0 to 1. The reverberation time is shorter for high frequencies than for low frequencies. As higher the damping as faster the high frequencies will decay.

6.2.2 Chorus

The chorus effect changes the sound of a single instrument into that of a group of instruments playing in unison. In TERA four modulated delay lines are used to achieve this effect. The amount of delay of each line is modulated by a separate LFO. The outputs of the four delay lines are spread around the stereo panorama. The chorus effect is global, this means only one chorus type can be used at one time, but every part has an individual amount of chorus set in the Mix page. The output of the chorus effect is mixed to the main output in the Part mixer.

The effect parameters can be set individually, or they can be selected from a list of 32 global presets. The parameters are stored together with the project.



ON/OFF

Switches the chorus on and off. If the chorus effect is not used, it should be switched off to lower the processor contention.

Select Preset

Selects from 32 global presets containing parameter settings.

Speed

Sets the average speed (frequency) of the LFOs modulating the delay time. The adjustable range is from 0.011 Hz to 75 Hz (cycles/second).

Spread

Sets the relative difference between the speeds of the four LFOs modulating the delay time. The adjustable range is from 0 to 1. If this parameter is set to 0, all LFOs oscillate with the same frequency.

✍ **Delay**

This parameter sets the average delay time of the four delay lines. The adjustable range is from 0.1 ms to 93 ms. The value for a typical chorus effect should be between 10 and 20 ms. But try other values too, you will enter a world of new effects!

✍ **Depth**

This parameter controls the depth of the delay modulation. The adjustable range is from 0 to 1. The prominence of the chorus effect is also affected by the settings of the part and master chorus amount in the Mix page.

✍ **Feedback**

The feedback parameter controls the amount of feedback for all four delay lines. The adjustable range is from 0 to 1.

6.3 Part effects

6.3.1 Overdrive

The part overdrive effect provides you with different types of distortion from hard clipping to soft distortion simulating old tube amplifiers. The sound balance can be adjusted using a combined High/Lowpass filter. Every part have its own overdrive effect. The parameters are stored together with the sound program.



ON/OFF

Switches the part overdrive on and off.

Drive

Sets the amount of distortion by amplifying the input signal. Range is from -12 dB to +48 dB.

Cutoff

This parameter sets the crossover frequency of the distortion filter. The distortion filter consists of a lowpass filter and a highpass filter with a cutoff frequency equal to the crossover frequency. The relative amount of lowpass and highpass filtered signal power is controlled by the Tone parameter.

✍ **Tone**

This parameter controls the relative amount of lowpass and highpass filtered signal power. The cutoff frequency of both the lowpass and highpass is controlled by the XOver parameter.

✍ **Select Overdrive type**

Selects between four distortion characteristics:

? **Hard**

Hard clipping distortion characteristic.

? **Soft**

Soft clipping distortion characteristic.

? **Tape**

Distortion simulating the characteristic of magnetic tape recordings.

? **Tube**

Soft distortion simulating old tube amplifiers.

6.3.2 Delay / Echo

The part echo/delay effect provides you with three different types of delay effects using two delay lines. The delay times can be synchronized to an external MIDI clock. The delayed signal can be feed back to the input for echo effects. The delay time can be synchronized to the MIDI clock or to the internal sequencer tempo to realize rhythmic echoes. TERA provides you with three different delay effect types each composed of two delay lines: stereo delay, mono delay and cross delay. Every part have its own echo/delay effect. The parameters are stored together with the sound program.



ON/OFF

Switches the part echo/delay effect on and off.

MIDI

Switches the synchronization to the MIDI clock on and off. The synchronization source can be the external MIDI clock or the tempo of the internal sequencer, depending on the sync settings internal/external in the menu **System > MIDI**. With sync switched on the delay times are shown in note length units instead of milliseconds.

✎ **Delay 1**

Sets the delay time of delay line 1 ranging from 0 ms to 728 ms. If MIDI sync is activated the range is from 1/32 triplet to 1/4.

✎ **Delay 2**

Sets the delay time of delay line 2 ranging from 0 ms to 728 ms. If MIDI sync is activated the range is from 1/32 triplet to 1/4.

✎ **Feedback**

This parameter controls the feedback of the delay line. This controls the decay of the amplitude of the echoed sound.

✎ **Cutoff**

A lowpass filter is build into the feedback loop of the delay. This parameter controls the cutoff frequency of this feedback filter. This results in successive echoes sounding darker.

✎ **Level**

The level parameter controls the output level of the effect. The adjustable range is from 0 to 1.

✎ **Select effect type**

With this listbox you can select between three different delay line configurations:

? **Stereo delay**

Both delay lines are used independently for the left and right input.

? **Mono delay**

The delay lines are connected in series for monophonic dual tap delay effects.

? **Cross delay**

The feedback loops of the two delay lines are cross coupled. Thus the delayed sound bounces between the stereo channels.

6.3.3 Modulation

The part modulation effect provides you with three different types of modulation effects:

✎ Chorus

The Chorus effect have the same structure as the global Chorus effect: four Delays modulated by four independent LFOs.

✎ Flanger

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.

✎ Phaser

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.



✎ ON/OFF

Switches the part modulation effect on and off.

✍ **Select modulation effect type**

With this listbox you can select between the three modulation effects: Chorus, Flanger and Phaser.

✍ **Speed**

Sets the average speed (frequency) of the LFOs modulating the delay time. The adjustable range is from 0.011 Hz to 75 Hz (cycles/second).

✍ **Delay**

This parameter sets the average delay time of the four delay lines. The adjustable range is from 0.1 ms to 93 ms for the chorus and the flanger effect. For the phaser effect the delay time is considerably smaller and depends on the frequency spectrum of the input signal. The value for a typical chorus effect should be between 10 ms and 20 ms. For a typical flanging effect the value should be below 5 ms.

✍ **Depth**

This parameter controls the depth of the delay time modulation. The adjustable range is from 0 to 1.

✍ **Feedback**

The feedback parameter controls the amount of positive or negative feedback for all four delay lines. The adjustable range is from -1 to 1.

✍ **Level**

Sets the output level of the part modulation effect. The adjustable range is from 0 to 1. With a value of 1 the output is a 1:1 mix of the original and the effect sound.

7 Pattern sequencer

7.1 Overview

The pattern sequencer of TERA is based on the concepts of the classic analogue step sequencer.



The sequencer in TERA is divided into two main parts:

In the upper screen area you will find the song sequencer where you can compose a song containing up to 16 tracks. The patterns are selected in the pattern pool on the left side of the sequencer and then put into the song with a drag&drop operation.

In the lower screen area you will find the step sequencer where the current selected pattern can be edited. Patterns are automatically selected if you click on a pattern in the song sequencer or pattern pool.

The pattern played can be selected from 512 patterns, organized in 4 banks with 8 pattern groups each containing 16 patterns. This set of 512 patterns is stored together with the project. They can be used in a song with a maximum of 16 tracks each and a maximum length of 512 bars. All the patterns are prefilled with examples which can be used as startup and inspiration source for your own projects.

Each pattern consists of up to 64 steps and one step can represent a note (for monophonic parts) or a chord (for polyphonic parts) containing a maximum of four notes. To program complex rhythmic structures, the note length and the gate time can be set individually for each step. Two control sequences can be used to adjust one or more sound parameters for each step. The values set by the control sequences can be used as a modulation source in the modulation matrix.

Each track can have an individual key function assigned. This opens up many possibilities to influence sequences in realtime for live sessions.

7.2 Common parameter

This section describes the common parameters and functions of the pattern sequencer.

7.2.1 Insert/Delete patterns

✍ **pattern bank**

select pattern bank 1-4.

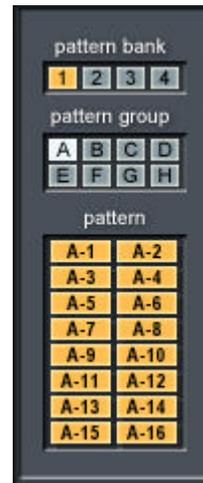
✍ **pattern group**

select pattern group A-H.

✍ **pattern**

select pattern 1-16.

The current selected pattern is shown automatically in the step sequencer and can be edited there immediately.



One pattern can be inserted into the song by drag&drop operation from the pattern pool to the desired position in the song sequencer grid. Just click on the pattern in the pattern pool and hold the mouse button pressed while moving the pattern to the desired position in the song and then release the mouse button.

The last pattern selected can be copied more than once by simply clicking into the desired song position while holding down the shift-key.

To delete a pattern in the song editor you can drag&drop the pattern outside the song editor or just click on the pattern while holding down the shift- and ctrl-key.

7.2.2 Transport area

cycle

With cycle switched on the song will be repeated between the start and stop marks. This song marks can be adjusted through dragging with the mouse.



bpm - Tempo

This slider controls the tempo of the sequence in beats per minute (bpm). The range is from 40 – 300 bpm. Click into the right half of the field and dragging up/down changes the tempo coarse. Doing the same in the left half of the field you can fine adjust the tempo. The tempo also controls the synchronization of the LFOs and the Delay effect. If used as a VST Instrument the host determines the tempo alone and this field can't be edited!

shuffle

Parameter to adjust the shuffle groove.

position

Shows you the actual position of the sequencer in the following format:

<measure.quarter.ppq>

One quarter note is thereby divided into 96 ppq (parts per quarter).

stop

Stops the sequencer on current position. With a second click the song position is reset to the beginning of the song. The sequencer can also be stopped pressing the space bar

play

Starts the sequencer. The sequencer can also be started pressing the enter key.

7.2.3 Song editor

| | key function | preset | output | m | s | 1 | 2 | 3 | 4 | 5 |
|---|--------------|------------------------|--------|---|---|-----|-----|-----|-----|---|
| 1 | none | 001 EmotiKon Pad (sum) | TERA | | | A-1 | A-2 | A-3 | A-4 | |
| 2 | none | 002 Encouter Pad (sum) | TERA | | | | | | | |
| 3 | none | 003 Overestimated ;-) | Do | | | | | | | |
| 4 | ... | ... | ... | | | | | | | |

✎ Key function

The key function parameter give you realtime control of the sequencer through the notes played on the keyboard. The following functions are available:

? **none**

The keyboard has no function.

? **arpeggio and arpeggio/hold**

The chords played on the keyboard are broken into single notes played in the sequence defined by the arpeggio key parameter. The pitch of the notes are transposed by the step notes. The Arpeggio key parameter determines which note of the chord will be played on a given step transposed by the step value.

? **pattern sel**

Use the keyboard to select the next pattern to be played. In every octave on the keyboard the notes have the same function:

C selects pattern 1

C# selects pattern 2

D selects pattern 3

D# selects pattern 4

...

? **record**

The notes played on the keyboard are used to program the sequencer steps in realtime. The current step note of the pattern will be replaced by the played note.

✎ **Sound program**

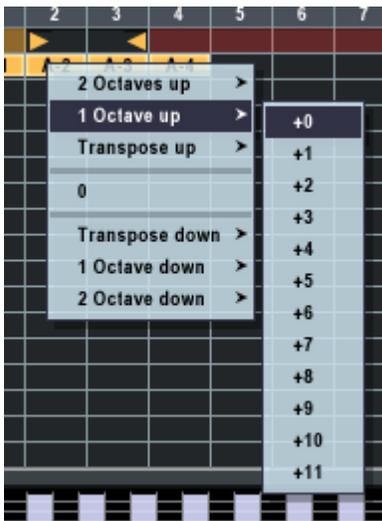
Click in this field to get a menu selecting a sound program for the track/part.

✎ **Mute / Solo**

The corresponding part will be muted or played solo.

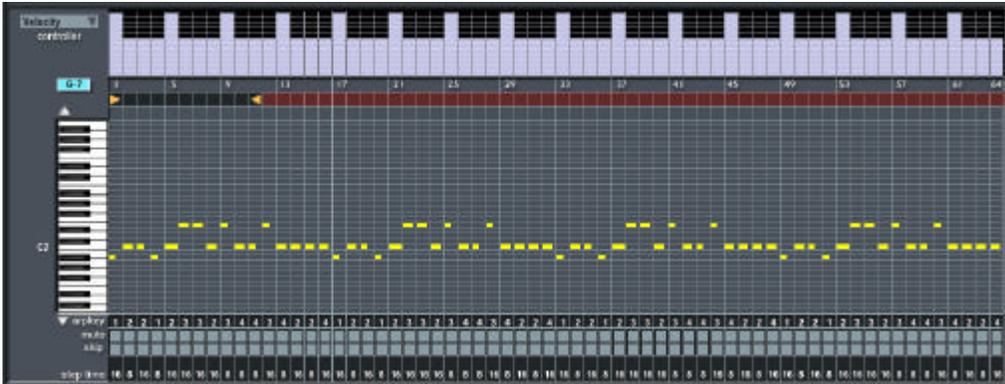
✎ **Pattern transposition**

Right click into a pattern in the song editor opens a menu for pattern transposition:



7.3 Step parameter

This section explains the parameter available for each step.



✎ Velocity / control

With the two buttons velocity and control you can select a control sequence for editing. For every step you can draw the control value directly into the control sequence window. The control sequences can be used as modulation source in the modulation matrix to control arbitrary sound parameters.

✎ sequence start/stop

The two red arrow buttons represent the start and stop position of the cyclic played pattern. This positions can be changed by dragging the arrows to the desired positions – even while the sequencer is already running. If the starting point of the sequence is after the stop position the sequence is simply played reversed!

✎ octave up/down

Click one of the blue arrow buttons on the left side of the keyboard to shift the note range shown in the step editor one octave up or down.

 **Edit Step note**

Click into the sequencer grid to change the note values for a step. If you hold the mouse button pressed you can draw more than one note into the grid editor at once.

 **Gate time**

The gate time is the ratio between note on time and note off time. To change the gate time click into the step while holding the Ctrl-key pressed.

This are the values which can be set:

? **25 %**

The note is on for 25% of the step time, for 75% of the time the note is off.

? **50 %**

The note is on for 50% of the step time, for 50% of the time the note is off.

? **75 %**

The note is on for 75% of the step time, for 25% of the time the note is off.

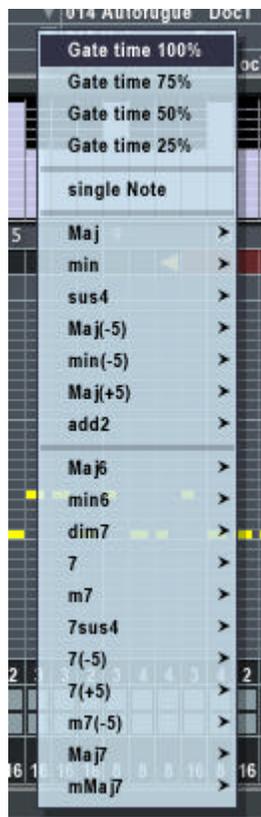
? **100 %**

The note is on for 100% of the step time. Envelopes using the triggermode „Key single“ , won't be triggered with the next step. This is important for 303 like pitch bending.

You can also use the chord menu described below to change the gate time.

✎ **Chord menu**

Clicking with the right mouse button into the step editor will open the chord menu:



If you use the chord function up to 4 voices are played simultaneously, but note that if less than 4 voices are assigned to this part not all voices are played.

Arpeggio key

If the Arpeggio **Key function** of the sequencer is enabled, every chord played on the keyboard is broken into a sequence of single notes. The sequence in which the notes of the chord will sound can be controlled by assigning each key of the chord to one or more steps. The pitch of the note played is the pitch of the associated chord note transposed by the setting of the step note.

Click into the field with the right mouse button, and you get a context menu to choose the arpeggio key:



Mute step

This Button gives you the opportunity to mute the current step. The sequencer waits quietly the specified step time and then proceeds with the next step.

Skip step

This Button gives you the opportunity to skip the current step. The sequencer immediately proceeds with the next step.

 **Step time**

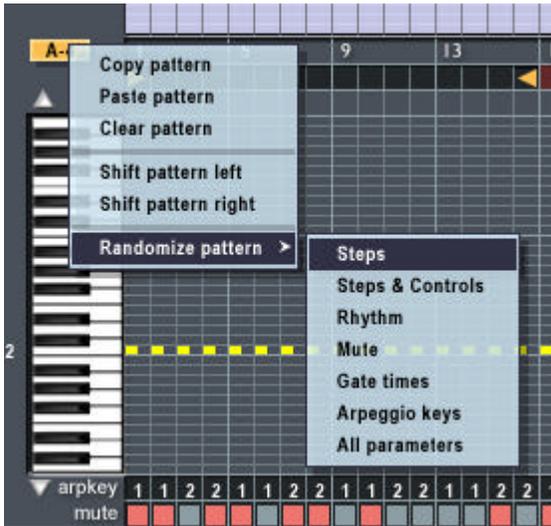
The step time determines the time to proceed to the next step. Click into this field with the right mouse button, and you get a context menu to choose the step time:



This menu allows you to use note lengths ranging from 1/32 to 1/1 with the variants triplet (2/3 standard length) and dotted (3/2 standard length).

7.4 Pattern manipulation

This menu provides you with commands to manipulate the patterns of the step sequencer. This menu can be accessed by a right mouse click into the step info field in the step sequencer.:



- ? **Copy pattern**
Use this function to copy the current selected pattern to the clipboard.
- ? **Paste pattern**
This function copies the contents of the clipboard into the current selected pattern.
- ? **Clear pattern**
This function initializes the current selected pattern with default settings.

- ? **Shift pattern left**
This function shifts all steps of the current selected pattern one step to the left.
- ? **Shift pattern right**
This function shifts all steps of the current selected pattern one step to the right.
- ? **Random pattern**
With the functions in this menu you can create random patterns. Each of these functions set a different subset of parameters with random values.
 - ? **Steps**
The note values are set randomly.
 - ? **Steps & Controls**
The note values and the settings for control A and B are set randomly.
 - ? **Rhythm**
The steptime between notes is set randomly.
 - ? **Mute**
The state of the mute switches are set randomly.
 - ? **Gate times**
The gate times (ratio between note on and note off) are set randomly.
 - ? **Arpeggio keys**
The arpeggio key assignments are set randomly.
 - ? **All parameters**
All step parameters are set randomly.

8 Part mixer

8.1 Overview

This is the main mixing console where all parts are finally mixed down to the stereo output. In the part mixer you can also adjust the send level for the global reverb and chorus effect. The effect output of this global effects is mixed to the stereo output in the master section. Every part can be individually muted or soloed, the panorama position can be adjusted and last but not least the level for the part can be adjusted. The actual sound level can be monitored using the peak level meter.

The part mixer also provides you with the possibility to assign each part a MIDI channel and a keyboard zone the part should respond to MIDI Input (Key split). The number of voices you can play simultaneously for one part is also specified here.



8.2 Operation

8.2.1 Part section



✂ Program selection

Displays the name of the sound program. Click into this field to change the sound program for this part.

✂ MIDI-channel

This edit field is used to display and alter the MIDI channel assigned to this part. Click into this field to open a menu for selection of a MIDI-channel.

✂ # Voices

Defines the number of voices which can be simultaneously played by this part. You can alter the number of voices from the context menu displayed after you clicked into this field with the right mouse button. (see below).

✂ Key low / Key high

This two fields allow you to define a keyboard split zone. Choose the upper and lower key of the desired split zone by clicking into this fields and choosing the keys from the popup menu.

✂ Reverb send

Adjusts the sound level send to the global reverb effect.

✂ Chorus send

Adjusts the sound level send to the global chorus effect.

✂ Panorama

Adjusts the panorama position of the part in the master stereo output.

✂ Mute/Solo

The buttons **M** and **S** let you mute or solo the part.

✍ **Output**

When used as plugin the part outputs can be routed to up to 8 stereo channels which can be mixed down separately by the host program. The routing works in two ways:

if “Master” is selected as output for a part on the Mix page, this part is routed to the master section of TERA. The master signal – together with the effect returns – is then output to the stereo channel 1.

if “Out2”- “Out8” is selected as output for a part on the Mix page, this part is directly routed to the stereo channel 2 – 8.

✍ **Part level**

This slider is used to adjust the sound level for this part. The peak level meter next to the slider displays the current peak level for this part. The field below the slider displays the actual settings of the slider in dB. The level can be adjusted from –57 dB to +12 dB. Moving the slider to the bottom will mute the part (OFF).

✍ **Number of voices**

You can set the number of simultaneous voices for this part with the context menu of the field **# Voices**. The number of voices used for this part can be set from 0 (part inactive), 1 (monophonic part) to a maximum of 32 (polyphonic part). Please note that in this version (1.0) all parts together can have no more than 64 voices. In addition to this the total amount of voices you can use depends on the processing power of your computer. If the processor usage is too high audio dropouts may occur. If this happens please reduce the number of voices for some parts.

8.2.2 Master section



- ✎ **Reverb return**
Adjusts the sound level returned from the global reverb effect.
- ✎ **Chorus return**
Adjusts the sound level returned from the global chorus effect.
- ✎ **Panorama**
Adjusts the balance between the left and right stereo outputs.
- ✎ **Master volume**
Adjusts the sound level of the master stereo output ranging from -57 dB to $+12$

dB. Next to the slider are two peak level meters displaying the peak level of the left and right output. Before the stereo signal is sent to the soundcard two analogue distortion units prevent the signal from digital clipping.



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