VirSyn Software Synthesizer
Owner's Manual

Version 1.0

Virtual integrated realtime synthesis

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VirSyn

**The professional tool for
sound design and
computer music production**

Copyright

Trademarks

Caution !

1. english Edition 06/2000

Contents

1 Introduction

1.1 Prologue

Virtual integrated relatime synthesis

Explore new sound spaces with VirSyn beyond pure emulation of synthesizer legends. Use the power of analog and digital synthesis combined with new physical models. New user interface elements navigate through 8-dimensional sound spaces exceeding the limits of traditional sound morphing. 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 12 modular synthesizer can be used simultaneously, all having the same user interface helping you operating this multi-modular system without loosing track. Distortion, echo/delay and chorus/flanger/phaser effects for every modular synthesizer boosts the sound into outer space. VirSyn gives your production an individual character with its unique fat sound. Precise timing even with an external sequencer let you feel the rhythm. Produce complete productions with unequaled ease using VirSyn as a stand alone tool. Inspire yourself from the new-designed 64 step-sequencer, extending the concept of the classic analog sequencer with variable rhythmic structures and chord sequences. Or use the sequencer for realtime transposition of keys to play arpeggios never heard before. Arrange up to 256 sequences in the song editor to complete the composition. The integrated mixing console together with a roomsimulator and a global chorus effect helps you in the final production step.

1.2   System requirements

For proper operation of VirSyn the system requirements are:

- **PC with Pentium Prozessor 200 MHz or better.**
- **Microsoft Windows 95/98.**
- **at least 32 MB main memory.**
- **Minimal screen resolution 800X600.**
- **16-bit Soundcard.**
- **DirectX Driver version 5.0 or higher.**
- **30 MB free space on your hard disk.**

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

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

1.3   Installation

To install VirSyn on your hard disk, insert the VirSyn CD-ROM into your CD-ROM drive. The installation program starts automatically 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 .

Please read carefully the copy protection remarks on the next page before starting VirSyn the first time !

1.4   Copy protection

VirSyn is copy protected with a license disk. After first starting the program you will be asked to insert the license disk. VirSyn will then be registered for exclusive use on this computer. Future program starts do not require the license disk to be insert. But if you want to install VirSyn on another computer first you have to transfer the license from the program back to the license disk. Insert the license disk into drive A: and use the function Transfer license in the system menu to accomplish this task. After this transfer the license is no longer valid for this computer. Then install VirSyn on the new computer using the original CD-ROM. You will be asked to insert the license disk on the first start of the program on the new computer. In case that you can't transfer the license to disk, because your computer crashed completely you can use the emergency license disk after reinstalling VirSyn on the new computer.

<u>IMPORTANT !</u>

Never unistall or delete VirSyn without transferring the license to a disk before !

2 Concept

2.1   Block diagram

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

Project

		Part 1	
MIDI	M I D I M a p p i n g	64 step sequencer arpeggiator → modular synthesizer up to 32 voices → part effects up to 3 inserts	M I X E R → OUT

☐

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Part 12

64 step sequencer arpeggiator → modular synthesizer up to 12 voices → part effects up to 3 inserts

song pattern sequencer

global effects Reverb / Chorus
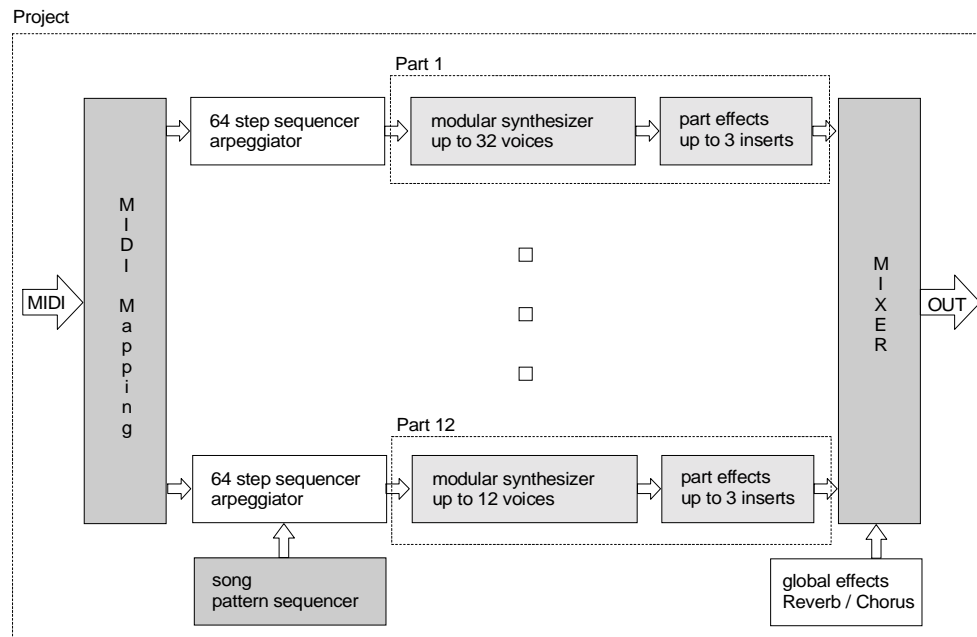
2.2   Part

VirSyn is multitimbral by design. The **part** is the basic unit of this multitimbral structure. Every part contains an independent modular synthesizer and up to three insert effects. Up to 12 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 VirSyn. The synthesizers of every part can have its own structure independent of all other parts. Thus VirSyn is not only multitimbral but even „multimodular" !

The three insert effects can be used to refine the sound output of the synthesizer. VirSyn provides you with an Overdrive effect with adjustable clipping characteristic, an Echo/Delay effect and a modulation effect usable as 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 VirSyn are organized in 128 sound banks with every bank containing up to 128 sound programs. The selection of a sound bank and program can be done manually or by using the MIDI bank and program change commands.

2.3   Project

A project is the top-level structure in VirSyn containing all parameter settings. It can be stored under a user definable name with the menu command **Project** > **Save as** and recalled later with the menu command **Project** > **Load**.. The name of the currently loaded project together with the project titel is shown in the titel bar of VirSyn. The project title can be entered in the project info dialog which can be opened with the menu command **Project** > **Info.**

The following parameters are stored in a project:

- The sound programs for all 12 parts.

- Step sequencer pattern memory (256 pattern * 64 steps/chords).

- Song sequencer with 256 patterns.

- Global effects parameter.

- Mixer settings.

- Info for project description and title.

The first sound bank with the fixed name „Song bank", is a special sound bank containing all 12 sound programs used by a project. By saving a project all currently active sound programs for the 12 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 changes.

2.4   MIDI Mapping

You can associate a distinct MIDI channel to 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 on the block diagram, you will see that every part has ist own step sequencer. If the step sequencer is stopped or not activated for this part, the 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.

Due to this realtime functions of the sequencer, VirSyn is ideally suited for complex live sessions !

CONCEPT

3 User interface

3.1   Overview

The user interface of VirSyn is designed to give you an optimal operating environment with mouse and keyboard. For this reason we avoided imitating traditional hardware synthesizers. Big sliders enable you to modify sound parameters precise and easy by using mouse, keyboard or external MIDI controller. All parameters are divided into ten property sheets. We used a fixed placement of user interface elements to avoid confusing changes of the user interface when you select another part to edit.

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 your MIDI controller box.

VirSyn gives you access to the sound engine in four user interface levels with increasing complexity:

- Select one of the 800 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.

- Construct your own synthesizers by patching the modules at your own and by changing the routing of modulations using the modulation matrix.

3.2   Screen layout

This section describes the distinct screen areas and their underlying functionality.
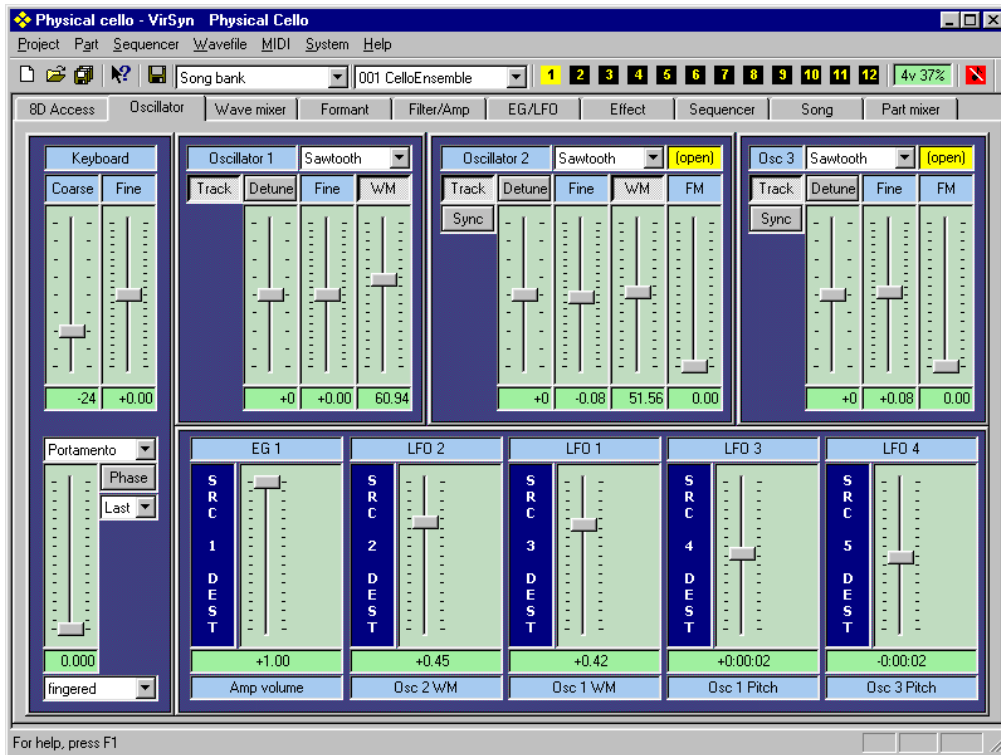
The main screen is divided in the following screen areas – listed from top to bottom:

- **Titlebar**
  Shows you the name and the title of the currently loaded project.

- **Menubar**
  Gives you access to the main functions of VirSyn.

- **Toolbar**
  Gives you fast access to the most important functions from the menu.

- **Property sheets**
  Gives you access to the parameters of the modular synthesizer, the effects, the step- and song sequencer and the part mixer divided in ten property sheets.

- **Statusbar**
  Shows you additional information to the currently selected menu command.

3.3   Menubar

3.3.1   Project

The project menu gives you access to the following commands working on projects:

- **New**

  Create a new project.

  Remark:
  VirSyn opens a special project named „new" to create a new project. The underlying file is named „new.vsp". You find this file in the directory „VirSyn" in contrast to the normal project files in the directory „VirSyn\Project". By overwriting this file with your own project file you can change the inital settings loaded on creating a new project.

- **Open**

  Opens an existing project. Changes made to the previously active project are getting lost !
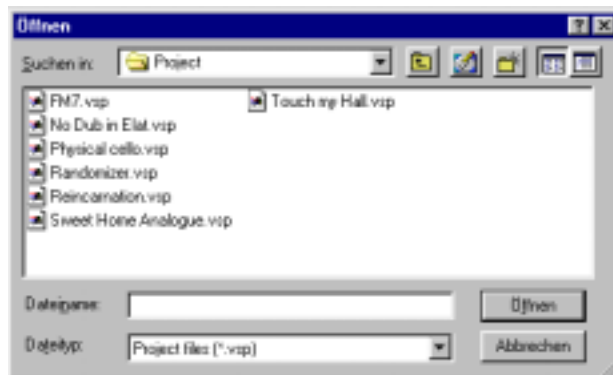
- **Info**
  Shows up a dialog for viewing and editing the project information. This information contains the project title and a short description of your project and a web address (URL). Pressing the button **Go!** brings up your favourite internet browser and loads the specified website.
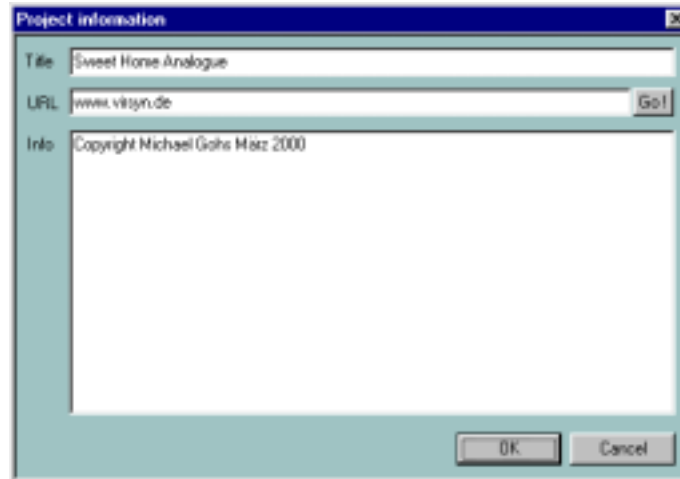
- **Save**
  Saves the current project.

- **Save as**
  Saves the current project in a project file with a new name.
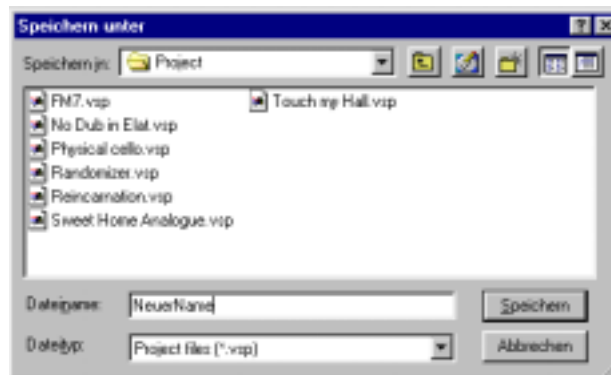
- **1 .. 4**
  Use the numbers and project names listed at the bottom of the Project menu to open the last four projects you used. Choose the number that corresponds with the project you want to open..

- **Exit**
  Closes the program. Before actually closing, VirSyn asks you to save the current project.

3.3.2   Part

The parameters for one part are divided into the property sheets for the modular synthesizer and the part effects. On these sheets you can view and edit the parameter settings for the currently selected part. To view or edit the parameter settings of another part, you first have to select the part using the part selection buttons or the part menu.

- **Save program as...**
  Saves the sound program of the currently selected part. This command pops up a dialog where you can specify the bank and the program place into which you want to save the program. This dialog can also be used to change the name of a bank or program.
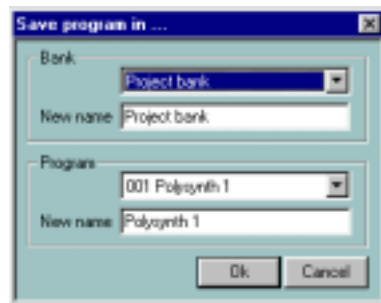
- **1**
  Selects part 1 for viewing/editing.

- **...**

- **12**
  Selects part 12 for viewing/editing.

3.3.3   Sequencer

This menu provides you with commands to manipulate the patterns of the step sequencer.

- **Copy pattern**
  Use this function to copy the currently selected pattern to the clipboard.
  The copied pattern contains all steps of all parts.

- **Copy part**
  Like copy pattern, but only the 64 steps of the currently selected part are copied to the clipboard.

- **Copy pattern measure**
  Like copy pattern, but only the 16 steps of the currently selected measure of all parts are copied to the clipboard.

- **Copy part measure**
  Like copy pattern measure, but only the 16 steps of the currently selected part are copied to the clipboard.

- **Paste**
  This function copies the contents of the clipboard into the currently selected pattern.

- **Clear pattern**
  This function initializes the currently selected pattern with default settings.
  The settings of all 64 steps for all parts are initialized.

- **Clear part**
  Like clear pattern, but only the 64 steps of the currently selected part are initialized.

- **Clear pattern measure**
  Like clear pattern, but only the 16 steps of the currently selected measure of all parts are initialized.

- **Clear part measure**
  Like clear pattern measure, but only the 16 steps of the currently selected part are initialized.

- **Shift pattern left**
  This function shifts all steps of the currently selected pattern one step to the left.

- **Shift pattern right**
  This function shifts all steps of the currently 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.

3.3.4 Wavefile

- **Record to wavefile**
  The audio ouput of VirSyn can be written directly to a disk file. This menu command opens a file dialog where you can specify the name of the file to be created. The recording starts immediatly after clicking Ok in this file dialog. 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) in stereo with 16 bit resolution. The samplerate depends on the actual setting in the menu **System > Samplerate**.

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

  During the recording the menu item **Record to wavefile** is checked to indicate that the recording is active.

- **Stop recording**

  This command stops the current recording. The wave file written during recording can't be processed by other programs before stopping the recording.

3.3.5   MIDI

- **Panic**
  This command stops 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.

The following four menu items control the synchronization of VirSyn to an external sequencer. The currently active setting is indicated by the menu item having a check mark.

- **Internal sync**
  With this setting the synchronization to an external sequencer is disabled. The tempo of the internal sequencer is controlled by the tempo slider residing on the property sheet „Sequencer". If any of the LFOs or the delay time of the part delay effect is synchronized to „MIDI", they are synchronized to this internal tempo.

- **External Sync**
  This setting synchronizes the internal tempo to the incoming MIDI clock events usually coming from an external sequencer. This also synchronizes all LFOs and the delay time of the part delay effect, if they are synchronized to „MIDI".

- **External sync song**
  In addition to the function **External sync** described above, this setting allows remote control of the internal **Song sequencer** through the MIDI commands **Start / Stop** and **Continue**.

- **External sync pattern**
  In addition to the function **External sync** described above, this setting allows remote control of the internal **Step sequencer** through the MIDI commands **Start / Stop** and **Continue**.

- **Controller assignment...**
  This command opens a list showing the actual assignment of MIDI controllers to sound parameters. You can change these assignments by using the MIDI learn function described later in this chapter.
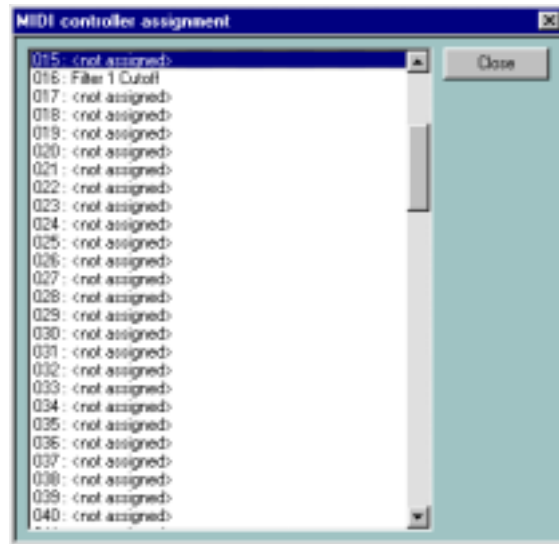
3.3.6　　　System

- **Preferences**
  Opens a dialog to view or edit the settings of the system parameters. This dialog allows you to choose the Audio und MIDI Drivers. The changes you make take effect immediatly, so you can test your settings without leaving the dialog.
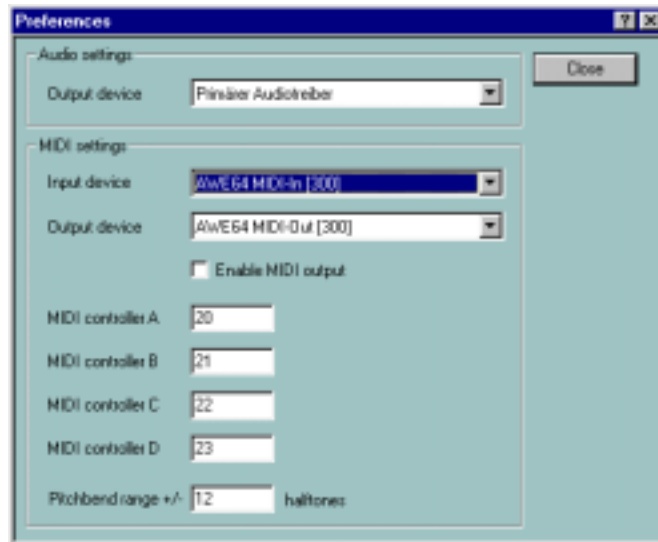
Audio settings

Output device

This combo box allows you to select the audio device used for the sound output of VirSyn. The list you can select from shows only DirectX compatible devices. VirSyn currently does not support MME drivers.

MIDI settings

Input device
This combo box allows you to select the MIDI input device used to receive MIDI messages from an external source from a list of available devices. To control VirSyn from a Software Sequencer running on the same computer you have to use a loopback device like the well known „Hubi's loopback device".

Output device
This combo box allows you to select the MIDI output device used to transmit MIDI controller messages from the list of available devices. This messages are send by user interface elements assigned to a MIDI controller with the MIDI learn function. With this function you can record parameter changes with an external sequencer. To send MIDI commands to an external sequencer program running on the same PC choose a loopback device as MIDI output.

Enable MIDI Output
Used to enable ∕ disable sending of MIDI messages to the MIDI output device.

MIDI Controller A
This edit field allows you to specify the MIDI controller (0 .. 127) used as modulation source **Controller A** for the modulation matrix.

MIDI Controller B
Same as above, but for modulation source **Controller B**.

MIDI Controller C
Same as above, but for modulation source **Controller C**.

MIDI Controller D
Same as above, but for modulation source **Controller D**.

Pitchbend range

The MIDI pitchbend messages modulate the base pitch of all oscillators in VirSyn. The depth of this modulation can be specified here in units of one halftone step. If you enter 0, the build in routing from the pitchbender to the pitch input of the oscillators is disabled.

- **Samplerate**

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

You can select from one of the following sample rates:

48000 Hz	DAT compatible.
44100 Hz	CD compatible.
32000 Hz	nearly as good as CD, but needs lesss computing power.
22050 Hz	LoFi

- **Transfer license**

This function copies your license back to the license disk. After copying, the license is no longer valid for this computer. Use this function only if you want to install VirSyn on another computer.

3.3.7 Help

- **Help topics**
  Calls the online help search dialog.

- **Index**
  Calls the online help with the main index.

- **Context help**
  Use the Context help command to obtain help on some portion of VirSyn.
  When you choose this menu item or the Toolbar's Context help button, the
  mouse pointer will change to an arrow with a question mark. Then click on
  the user interface element or module you need help for. The help topic will
  be shown for the item you have clicked on.

- **VirSyn on the Web**
  Use this command to jump directly to the internet website of VirSyn. You
  must have access to the internet and an internet browser for this function to
  work.

- **About VirSyn**
  Use this command to display the copyright notice and version number of
  your copy of VirSyn.

3.4   Toolbar

The toolbar is displayed across the top of the application window, below the menu bar. The toolbar provides quick mouse access to the most important menu commands used in VirSyn. Every button in the toolbar has a corresponding menu item with the same function. The combo boxes for bank and program allows you to select the sound program for the currently active part.

The toolbar commands from left to right:

	New	Create a new project.
	Open	Open an existing project.
	Save Project	Save current project.
	Context help	Activate context sensitive help.
	Save program	Save sound program for currently active part.

Select bank/program   Select sound program from up to 128 sound banks each containing up to 128 programs.

Select part   Select part and monitors incoming MIDI notes.

Voices   Displays number of voices and processor usage in %.
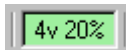
Panic   Switch off all notes currently playing.

3.5 Property sheets

The main screen of VirSyn is divided into 10 property sheets each containing a subset of the parameters. Click on the appropriate label tab to make the desired property sheet visible. The first eight 8 property sheets contain the sound parameters for the currently selected part. With just two mouse clicks you can reach any of the over 2500 sound parameters:

1. Select one of the twelve avaliable parts – each a complete modular synthesizer !

2. Select the property sheet containing the parameter you want to edit.
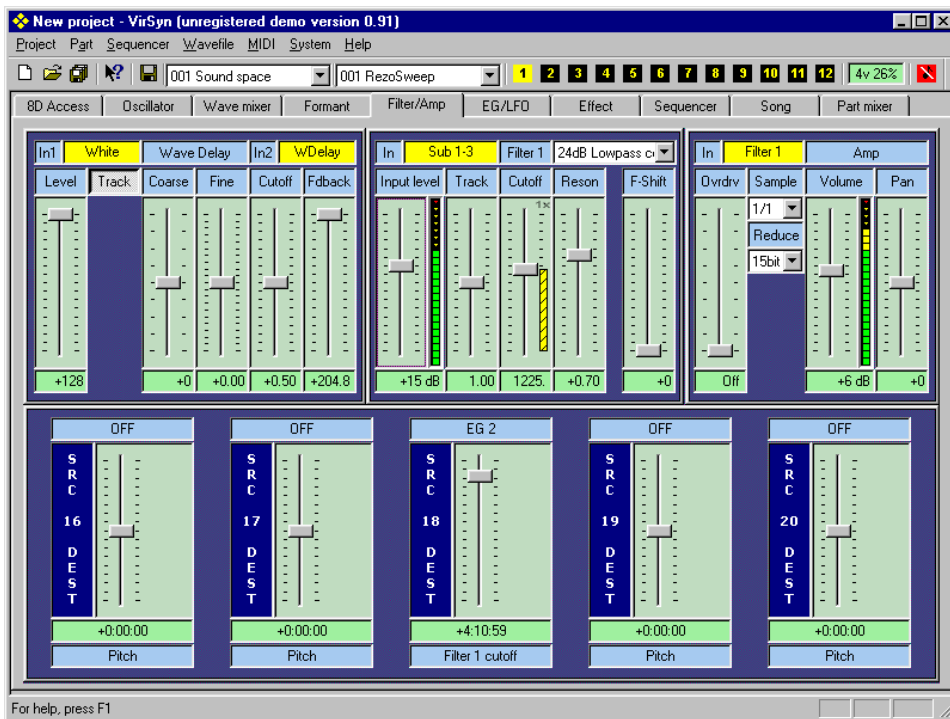
The other two property sheets **Song** and **Part mixer** and the global effects on the sheet **Effects** are common for all parts. This means that this parameters are independent from the currently active part.

3.6   Slider

The slider is the most important user interface element in VirSyn 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 of the slider to move the position while keeping the mouse button pressed.

   Click with the left mouse button above the knob to increase the parameter value about 1/10 of the whole parameter range.

   Click with the left mouse button below the knob to decrease the parameter value about 1/10 of the whole parameter range.
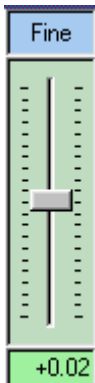
   If you keep the mouse cursor above the slider for a short amount of time, a small popup window with a short parameter description appears.

- ▪ **Keyboard**
   The slider have to be selected before it can react on keyboard commands. You can select the slider either by clicking the knob with the left mouse button or using the **TAB** keys. The selected slider is displayed with a dotted frame.

   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** ↑ und **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 key **0** on the numerical keypad (with **Num Lock** set on) to set the parameter to its default value. This is usually the center position of the slider.

▪ **Context menu**
Click on the slider with the right mouse button and you will get the following context menu:
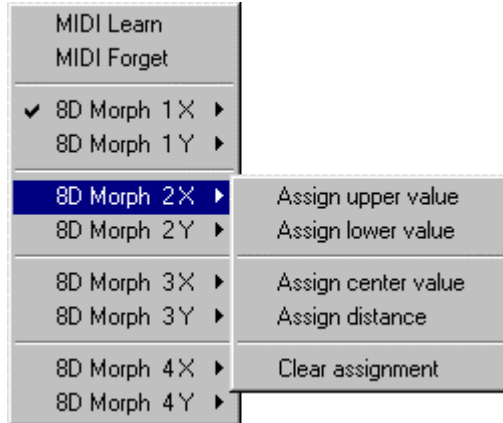
This context menu is only available for sliders residing on the property sheets for the modular synthesizer and the part effects. This menu provides you with functions to control sound parameters by an external MIDI controller or to assign the parameter to one of the sound space dimensions controlled by the property sheet **8D Access**.

• **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. VirSyn 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. This assignment is independent of the receiving MIDI channel and the MIDI controller always controls the currently active part.

- **MIDI Forget**
  This command deletes the assignment of a MIDI controller to the sound parameter controlled by this slider.

- **8D Morph 1X .. 4Y**
  With this submenu you can assign the sound parameter to one of the 8 dimensions of the sound space on the **8D Access** property page. Up to 8 sound parameters can be assigned to (and therefore be controlled by) one dimension. Existing assigments are indicated by a checkmark on the menu item.

  - **Assign upper value**
    The actual value of the sound parameter is assigned as maximum value for the selected dimension 1X .. 4Y.
  - **Assign lower value**
    The actual value of the sound parameter is assigned as minimum value for the selected dimension 1X .. 4Y.
  - **Assign center value**
    The actual value of the sound parameter is assigned as center value for the selected dimension 1X .. 4Y.
  - **Assign distance**
    The distance of the actual parameter value to the **center value** assigned before is measured. The center value +/- the measured distance defines now the new parameter range for the selected dimension 1X .. 4Y.
  - **Clear assignment**
    Clears an existing assigment of the sound parameter for the selected dimension 1X .. 4Y.
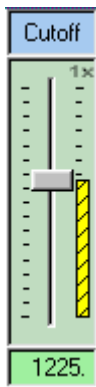
Assignment display

After succesful assignment the adjusted range will be shown as a small bar on the right side of the slider. The hatching direction indicates how the assigned sound dimension controls the sound parameter: in the same or in opposite direction. The assigned dimension 1X .. 4Y is shown in the right upper corner of the slider.

Tip

With the MIDI Learn function normally 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.

3.7   Listbox

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

| Sawtooth ▾ |
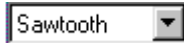
- ▪ **Mouse**
  With a click on the arrow you can open the listbox showing you a list of values. Click on the value you want to select.

  If you keep the mouse cursor above the listbox for a short amount of time, a small popup window with a short parameter description appears.

- ▪ **Keyboard**
  The listbox have to be selected before it can react on keyboard commands. You can select the listbox either by clicking on it with the left mouse button or using the **TAB** keys. The selected listbox is displayed with a dotted frame and a blue colored background.

  Use the cursor keys ↑ and ↓ to select the list box entries one after another.

  You can open the listbox with the keys **Alt** + ↓.

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

- ▪ **Context menu**
  Click on the listbox with the right mouse button and you will get the same context menu as described for the slider having the same functionality.

  This context menu is only available for listboxes residing on the property sheets for the modular synthesizer and the part effects.

3.8   Buttons

Buttons are used to switch certain functions like **Keyboard-tracking** on or off.

- **Mouse**
  Click on the button to switch the function on or off.

  If you keep the mouse cursor above the button for a short amount of time, a small popup window with a short parameter description appears.

- **Keyboard**
  The button have to be selected before it can react on keyboard commands. You can select the button either by clicking on the button with the left mouse button or using the **TAB** keys. The selected button is displayed with a dotted frame.

  Use the space key to toggle the state of the button.

- **Context menu**
  Click on the button with the right mouse button and you will get the same context menu as described for the slider having the same functionality.

  This context menu is only available for buttons residing on the property sheets for the modular synthesizer and the part effects.

3.9   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 cable connected with an output of another module. The name of the connected output is shown in the audio input window.
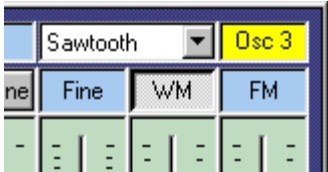
The audio input window has a yellow background, if the modul is active otherwise it hase a blue background. VirSyn automatically deactivates modules whose outputs are never used to preserve processor resources.

Click on the slider with the right mouse button and you a context menu where you can select the output you want to be connected with this audio input.
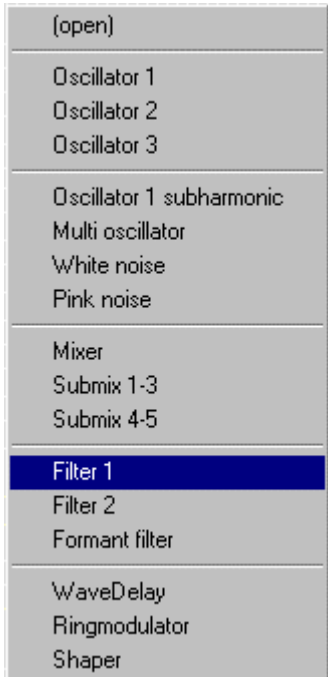
The context menu on the right shows you all available module outputs of VirSyn.

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

4 Modular synthesizer

4.1   Overview

The modular synthesizer is the most important part of VirSyn and is responsible for the sound generation. It provides you with up to 12 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 Stimmen   = 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 the modules.

VirSyn works with a fixed number of modules. This gives you the following advantages:

- The patch connections between the modules are treated as parameters like any other sound parameter. That way connections are stored together with the sound parameters in the sound program. Thus 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 12 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 highly optimize the synthesis algorithms. This has the consequence of reduced processor contention and thereby getting an increased voice yield. The advantage shows up especially on complex setups with many differently constructed instruments in a project.

4.2   8D Sound Access

The property sheet **8D Sound Access** provides you with a unique user interface to navigate through a user definable sound space consisting out of 8 independent sound dimensions.
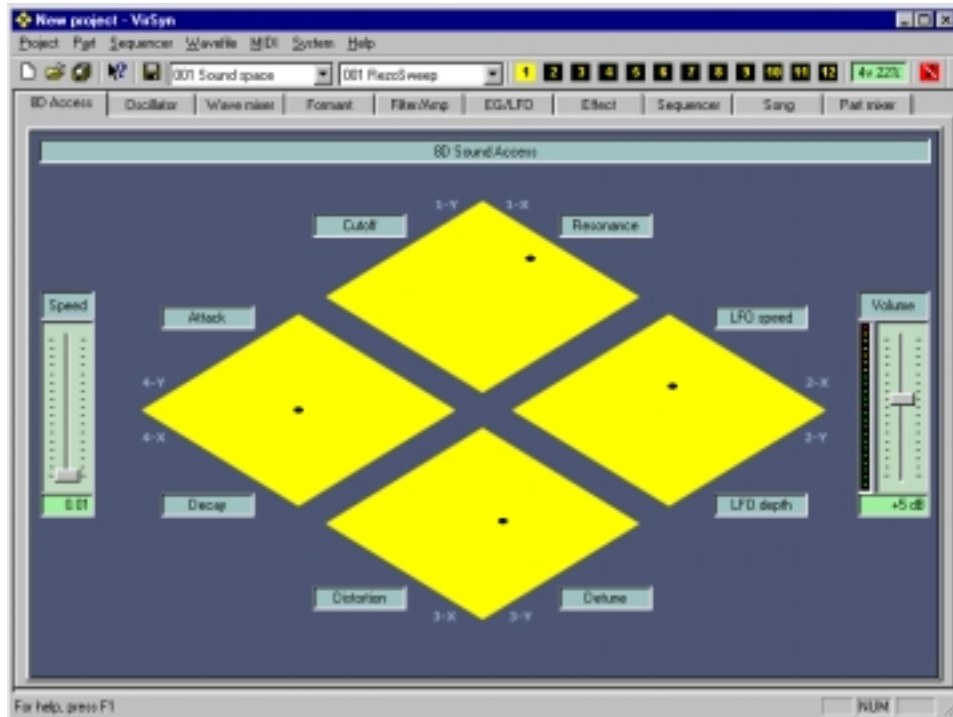
Every dimension can control up to 8 sound parameters simultaneously. This means that you can control up to 64 sound parameters (8 dimensions x 8 parameters) with a single click on the mouse button. Or you can use up to 8 MIDI controllers to travel remotely through this sound space. You can control the 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 the diamond shapes.

For easy navigation through this sound space VirSyn 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 possibilty to remotely control your sound voyage with an external sequencer or with a hardware MIDI controller.

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 have to select the property sheet containing the respective user interface element. Then open the context menu for this element with the right mouse button to assign the sound parameter to the sound space as described in section **3.6 Slider**.

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 continously from their starting position to the end position in this time.

The 8 edit fields grouped around the four diamond shapes represent the 8 sound dimensions and provide you with the following functions:

1   You can overwrite the default text „Morph ..." to name the sound dimension which then is stored together with the sound program.

Click into the edit field with the right mouse button and you will get a context menu with the functions:

2   MIDI Learn: assigns the sound parameter to a MIDI controller.

3   MIDI Forget: clears assignment to the MIDI controller.

4   Show 8D parameter list : Displays a list of all sound parameters assigned to this sound dimension.
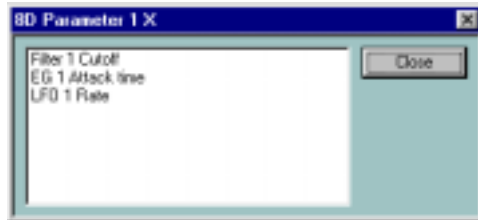
4.3   Modulation Matrix

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 21 sources, to a modulation destination, likewise selectable from 81 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.

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 build in:

1.   pitch bender   >> pitch

2.   keytrack   >> pitch of all oscillators, cutoff frequency for the two
                             multimode filter and the pitch of the WaveDelay module.

The user interface elements:
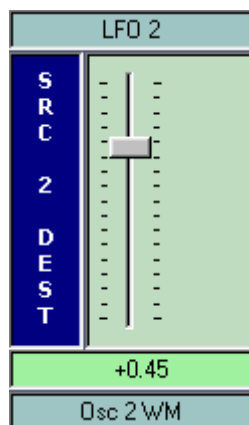
- **Modulation source**
  Source which controls the sound parameter specified in the modulation destination.

- **Modulation amount**
  Slider 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:
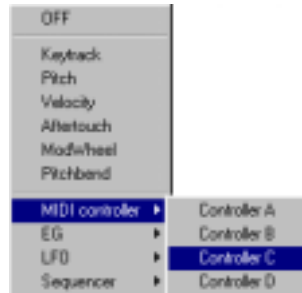
- **Select modulated destination**
  Select a modulation destination by clicking into the destination field with the right mouse button and selecting a source from the context menu:

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

4.4   Modules

This chapter describes every module of VirSyn's synthesizer section in detail. This 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. VirSyn checks for each module whether its output is used by any active module. If not the module is switched off.

Module overview:

- **Keyboard**
  Master tuning / Portamento.

- **Oscillator 1**
  Main oscillator with wave modulation.

- **Oscillator 2**
  Oscillator with wave modulation and FM input.

- **Oscillator 3**
  Oscillator with FM input.

- **Multi Oscillator**
  6 Oscillators with detune and wave modulation.

- **Ringmodulator**
  Multiplication of two input signals.

- **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.

- **WaveDelay**
  Physical modeling for simulating string and wind instruments.

- **Amplifier**
  Final amplifier with overdrive and LoFi effects.

- **4 Envelope generator**
  Envelopes with D-ADSR characteristic.

- **4 LFO**
  Low frequency oscillator with synchronization and Sample&Hold function.

4.4.1　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:

The parameters **Coarse** and **Fine** determine the base pitch used for all oscillators, the WaveDelay and influences the tracking of the filters. This parameters also take effect on the modulation sources **Pitch** and **Keytrack.**

- **Coarse**
  Pitch adjustable in the range of –48 to +48 halftone steps.

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

- **Portamento/Glissando**
  Smooth (Portamento) or stepwise transition (Glissando) 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 (portamento off) to 132 seconds. This effect is typical for many lead sounds on monophonic synthesizers, but can be used here in polyphonic mode also !

- **fingered/normal**
  In normal mode all notes played will have the portamento/glissando effect. In fingered mode only legato played notes will have this effect.

▪ **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. Very important for drum sounds !

Without phase synchronization the oscillators generate their waveforms continously, and every note starting picks up a random phase. The old analog synthesizers are all working this way. This is one of the main reasons for the warmth of their sound. Compare the sound of slightly detuned 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, ...

▪ **Last / Low / High**
The key priority controls which note will be played if you press more than one key in monophonic mode. With key priority **Last** always the last note will be played. With key priority **Low** or **High** the lowest or the highest key will be played.
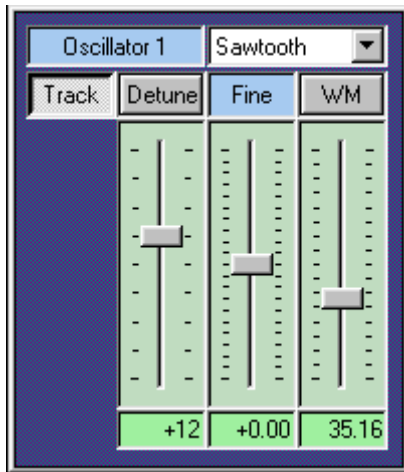
If the currently played key is released with key priority low or high then the lowest/highest note restarts playing if you have pressed more than one key. This is the same behaviour as on most classic monophonic synthesizers.

Tip

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

4.4.2   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 VirSyn are using antialiasing algorithms producing waveforms without digital distortion even in the highest register. Nevertheless VirSyn oscillators provides you with an outstanding fat sound in the lower registers.

- **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 osillator from a list of 64 waveforms including the classical waveforms sawtooth, square, triangle and sine.

- **Detune/Ratio**
  Coarse tuning of the oscillator relativ to the master tuning adjusted in the keyboard module. With the button Detune you can choose between two modes:

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

  Ratio: 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.
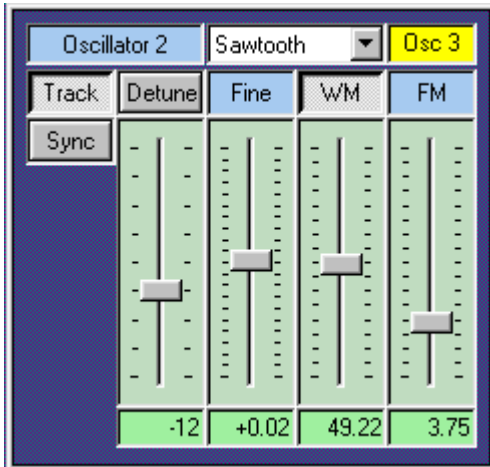
- **Fine**
  Fine tuning of the oscillator. The adjustable range depends on the detune/ratio mode:

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

  Ratio: Sets tuning in fractional multiples of the master tuning.

- **WM Wave modulation**
  With wave modulation activated the oscillator 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 waveforms available !

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.

4.4.3 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 osillator from a list of 64 waveforms including the classical waveforms sawtooth, square, triangle and sine.

- **Detune/Ratio**
  Coarse tuning of the oscillator relativ to the master tuning adjusted in the keyboard module. With the button Detune you can choose between two modes:

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

  Ratio: 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.
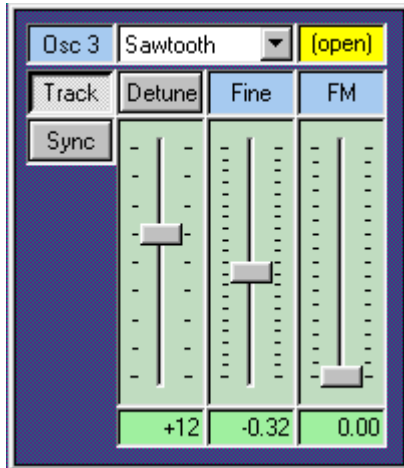
- **Fine**
  Fine tuning of the oscillator. The adjustable range depends on the detune/ratio mode:

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

  Ratio: Sets tuning in fractional multiples of the master tuning.

- **WM Wave modulation**
  With wave modulation activated the oscillator 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 waveforms available !

- **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 with the context menu of the yellow 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.

4.4.4   Oscillator 3

Oscillator 3 has the same functions as oscillator 2, except the wave modulation feature.
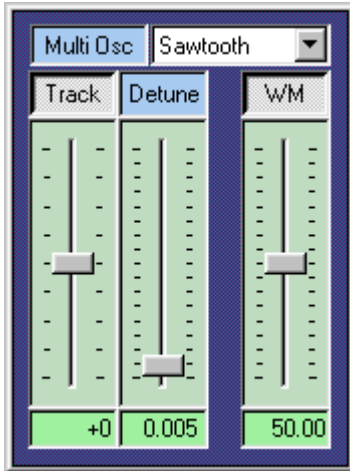
- **Track**
  This button activates keyboard tracking. When switched on the oscillator will track the keyboard at the rate of one semitone for each key. When switched off the oscillator will have a fixed pitch independent of the notes played on the keyboard.

- **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 osillator from a list of 64 waveforms including the classical waveforms sawtooth, square and triangle and sine.

- **Detune/Ratio**
  Coarse tuning of the oscillator relativ to the master tuning adjusted in the keyboard module. With the button Detune you can choose between two modes:

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

  Ratio: 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. The adjustable range depends on the detune/ratio mode:

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

  Ratio: 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 with the context menu of the yellow 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.

4.4.5   **Multi Oscillator**

The Multi Oscillator is composed of six independent oscillators. It's ideally suited for generating ultra fat or choruslike sounds. By strongly detuning the six oscillators you can generate inharmonic sounds well suited for simulating for example cymbals.

- **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 osillator from a list of 64 waveforms including the classical waveforms sawtooth, square and triangle and sine.

- **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, choruslike sounds. With increasing values the sound becomes more and more inharmonic.

- **WM Wave modulation**
  With wave modulation activated the oscillator 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 waveforms available !

Tip

With the settings detune = 0.7, waveform =Sawtooth, WM = 80 you can simulate the base sound of the 808 generator for cymbals.

4.4.6　　　Ringmodulator

This module combines two input signals by multiplication. The resulting spectrum contains the sum and differences of 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.
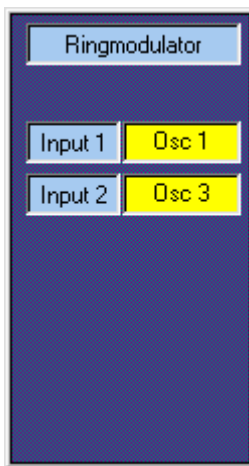
- **Input 1**
  Choose an audio source for input 1 by clicking into this field with the right mouse button and selecting a source from the context menu.

- **Input 2**
  Choose an audio source for input 2 by clicking into this field with the right mouse button and selecting a source from the context menu.

Note

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

4.4.7 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 cutoff frequency of 0 Hz and a slope of 3dB/octave. Pink noise is sounding darker and smoother than white noise.

Every voice in VirSyn 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.

4.4.8   Wave mixer

With this module you can mix down up to 5 input signals to the mix ouputs. The three ouputs 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.

That way you can use this mixer also as two independent mixers with 2 and 3 inputs respectively.
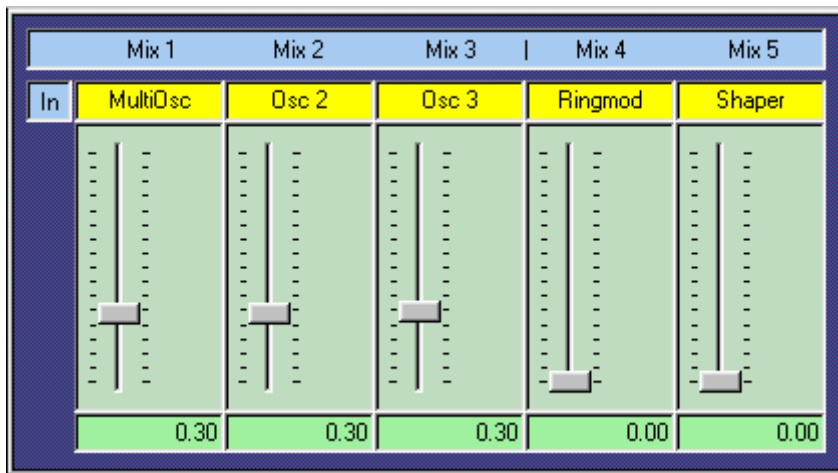
▪ **Input 1-5**
   Choose the audio source for the inputs 1 to 5 by clicking into the appropriate field with the right mouse button and selecting an audio source from the context menu.

4.4.9 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 control 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**.
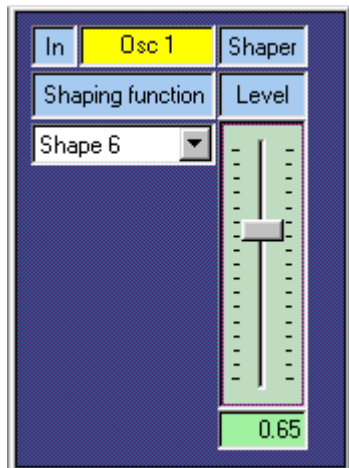
▪ **Input**
Choose an audio source for input 1 by clicking into this field with the right mouse button and selecting a source from the context menu.

▪ **Shaping function**
Selects one of the shaping functions. The entries named **FM...** result in sounds comparable to FM synthesis.

▪ **Level**
Adjusts the level of the input signal. With increasing input level the spectrum of the sound becomes brighter.

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 evaluating timbres.


4.4.10   Multimode filter

The two multimode filter are the most important sound design tools in VirSyn. They simulate the characteristics of the filter types used in traditional analogue synthesizers. With the build 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.

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 **phaserlike sound.**

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

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

- **Lowpass 24dB cc**
  This lowpass filter has a classical cascade design and attenuates frequencies below the cutoff frequency with 24dB/octave giving a **warn 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:
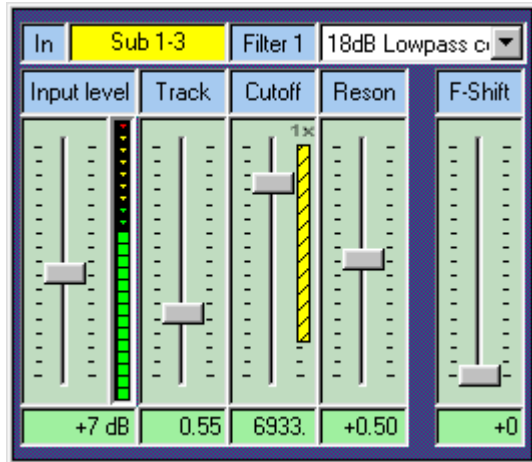
- **Filter input**
  Choose an audio source for input 1 by clicking into this field with the right mouse button and selecting a source from the context menu.

- **Filtertyp**
  Selection of the filter characteristic.

- **Input level**
  Adjusts the filter input level in the range of –10dB to +30dB. 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 is fading.

- **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 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 by itself producing a sine wave (even if no input source is connected !).

- **F-Shift**
  Each filter is internally build by two or more subfilters 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.

4.4.11   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 „spacy" sounds.

As opposed to common fixed filter banks with this one you can dynamically control all filter parameters using the modulation matrix !
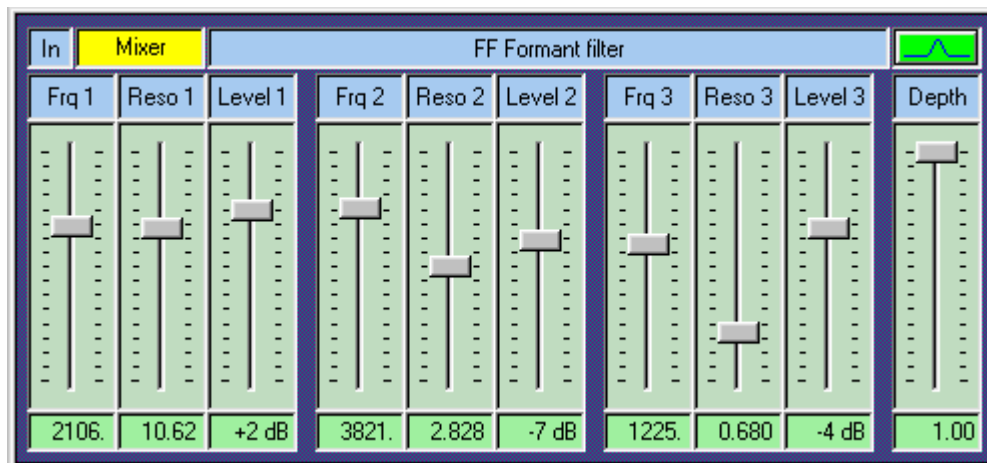
- **Input**
  Choose an audio source for input 1 by clicking into this field with the right mouse button and selecting a source from the context menu.

- **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 send to the output. With depth set to 0.0 only the original signal is send 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 relativ 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 phaserlike sounds.

4.4.12   WaveDelay

This module is the basic building block for physical modeling synthesis. First of all it sums the signals from its two audio inputs In1 and In2, attenuated by the input slider and the feedback slider settings. The sum is then delayed with a pitch synchronous delay time and lowpass filtered with an adjustable cutoff frequency. In the simplest form one can use this module as a delay line with realtime 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 producing 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 source 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 normally is 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 simulates 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:
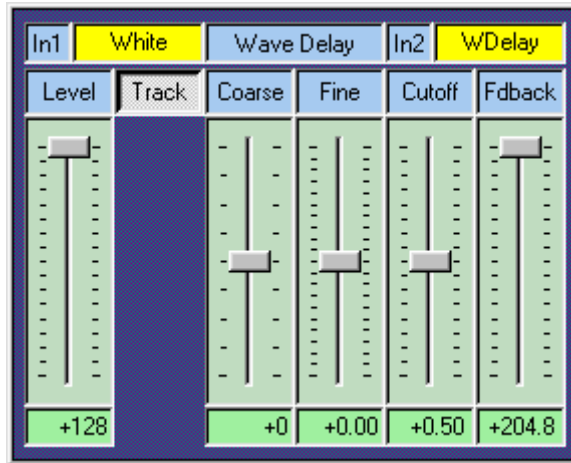
- **Input 1**
  Choose an audio source for the initial energy impulse by clicking into this field with the right mouse button and selecting a source from the context menu.

- **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 modeling 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 relativ 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 bis +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 normally 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 for example an LFO can cause some noise.

4.4.13   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 slider 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 build into the amplifier module:

- **Overdrive**
  A soft distortion unit which 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 intermodulation between notes in polyphonic play. Thus the sound is quite different in comparision to a standard distortion unit.

- **Sample**
  With this parameter you can change the samplerate for this program independent of the samplerate of programs used in other parts. The setting here is always relativ to the physical samplerate used by the soundcard set in the system menu. Use this function to get Lofi sounds comparable to old digital synths. With subsampling activated, the processor contention reduce, and therefore you have more simultaneous voices available. Especially bass sounds often don't need high samplerates.

- **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 LoFi sounds !

The Amplifier parameter:
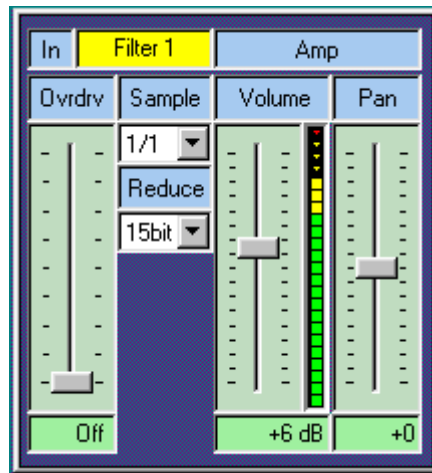
- **Input**
  Choose an audio by clicking into this field with the right mouse button and selecting a source from the context menu.

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

- **Volume**
  Output level ranging from –40dB bis +40dB

- **Pan**
  Adjusts the position of the sound in the stereo panorama. As 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.

4.4.14   Envelope generator

The envelope generator lets 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. VirSyn 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 voiceVirSyn provides you with four independent envelope generators.

Phase transition diagram of VirSyn Envelopes:
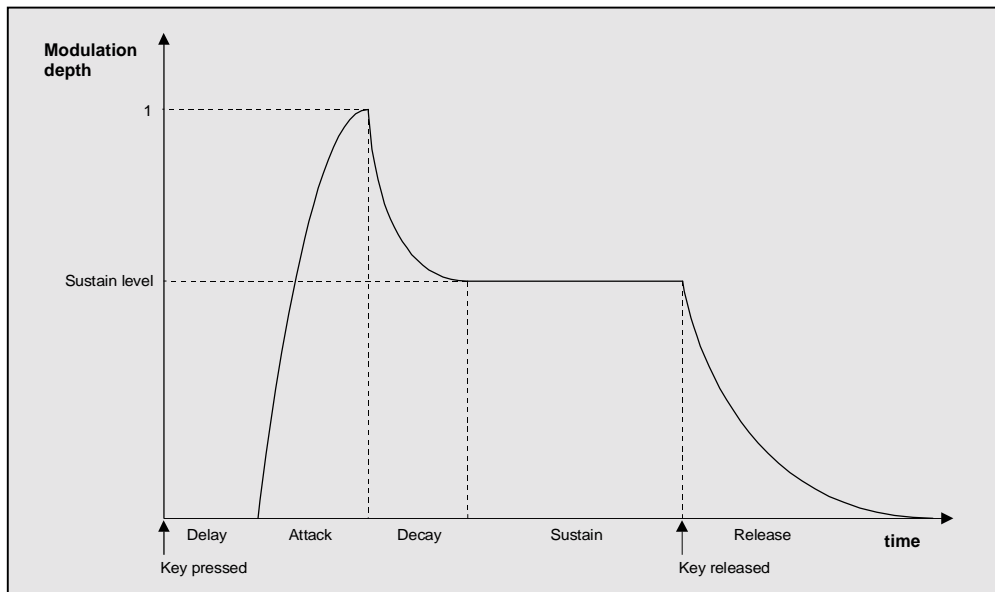
Die Parameter des Envelope generators:
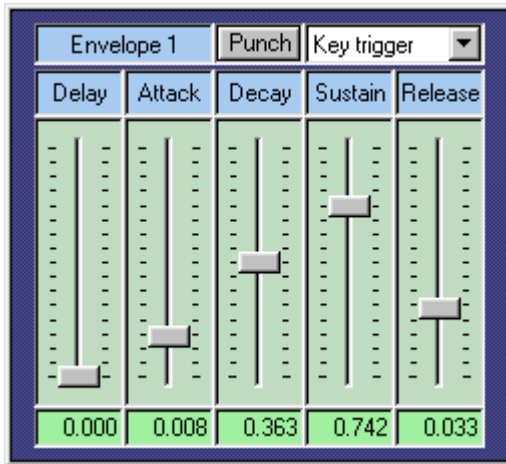
- **Delay**
  Sets the delay time between triggering and start of the attack phase.
  Range is from 0.7 ms to 132 seconds.

- **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. Range is from 0.7 ms to 132 seconds.

- **Punch**
  With the punch switch the start of the decay phase is delayed a few milliseconds. The result is a „thicker" attack similiar to a compressor effect. This effect is audible only on short attack/decay time envelopes.

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

  - **Key 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 VirSyn provide you with extreme short time settings below 1/1000 seconds. Modulating the filter cutoff frequency with extremly 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.

4.4.15   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 sync'd to the external/internal MIDI Clock. The phase of the waveform can be controlled by the wave sync parameter.
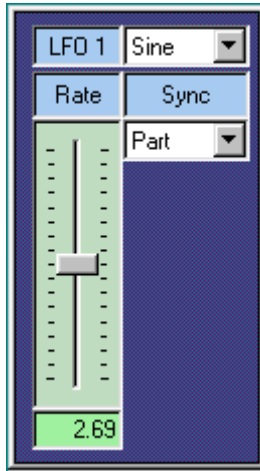
- **Waveform**
  Selects one of the following waveforms:

- **Sine**

    - **Triangle**

    - **Square**

    - **Ramp up**

    - **Ramp down**

    - **S & H <n>**
      Using the waveform S & H <n> produces a Sample & Hold effect. In this mode the module takes samples of the output of another LFO <n> at the rate of the LFO frequency. To create the traditional Sample & Hold effect select a random modulation waveform for LFO <n>.

    - **Random**
      Aperiodic waveform.

- **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 triplett 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 sync'd 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 ist 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 ist own LFO starting with zero phase on key down.

5 Effects

5.1   Overview

▪ **Global Effects**
The global effects Reverb and Chorus are shared by all parts and exists only once. You can use them for all 12 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 has 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.
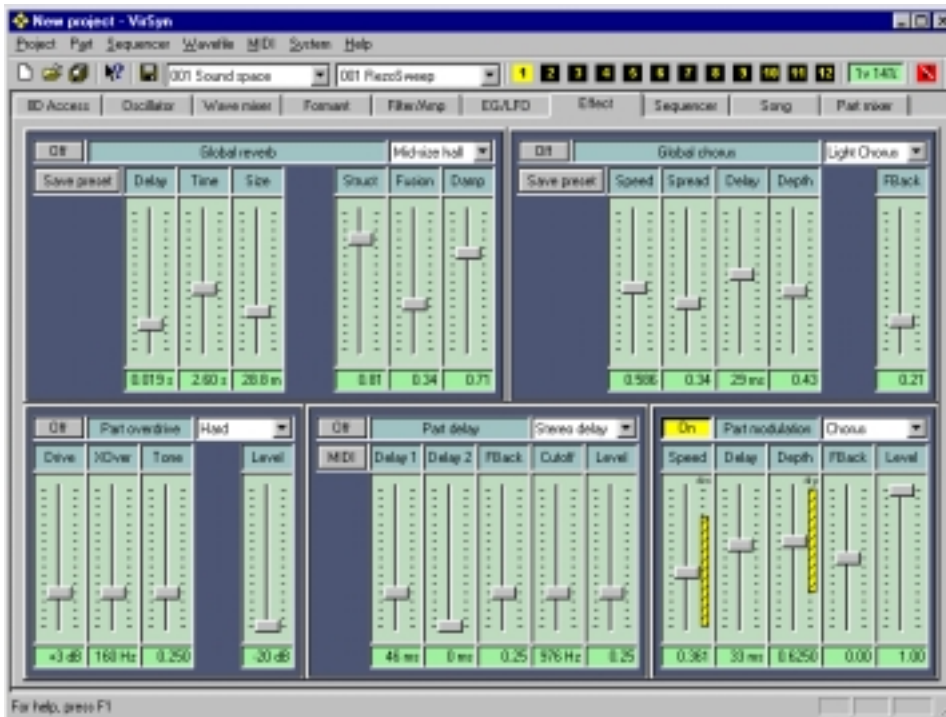
5.2   Global effects

5.2.1   Reverb

The reverberation effect is used to simulate the reflections of sound 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 in the special effect preset „Project". If you want to use the reverb settings in other projects, you can store the settings in a preset too.
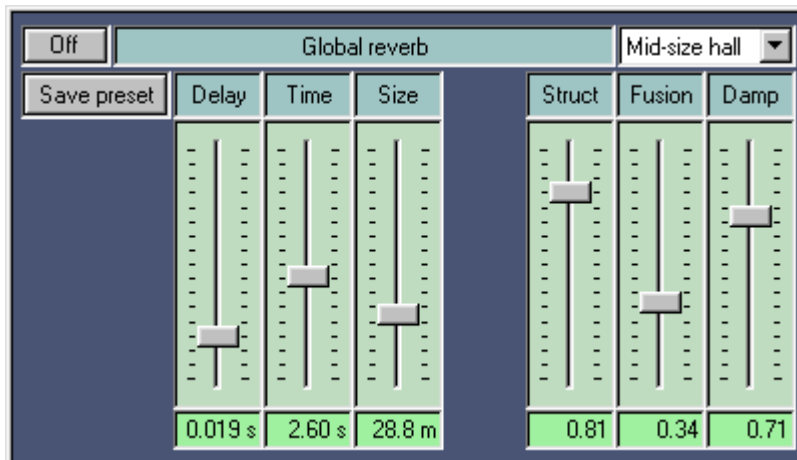
- ▪ **ON/OFF**
  Switches the reverb on and off. If the reverb effect is not used, it should be switched off to lower the processor contention.

- **Select Preset**
  Selects from 32 global presets containing user definable parameter settings. The first entry is not really a preset. It has always the name „Project" and contains the reverb parameter settings stored in the currently loaded Project.

- **Save Preset**
  Saves the current settings of the reverb parameters to one of the 32 global presets. Choose a preset place where you want to store the preset from the combo box and enter the name for this preset. Click on „Ok" to store the preset.
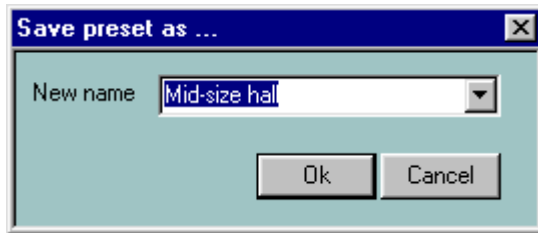
- **Delay**
  Sets the amount of time that elapses between receiving a direct sound and its first reflection. This time correlates with the perceived size of the simulated room. The adjustable range is from 0 ms to 100 ms.

- **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, labelled „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. The higher the damping the faster the high frequencies will decay.

5.2.2　　　Chorus

The chorus effect changes the sound of a single instrument into that of a group of instruments playing in unison. In VirSyn 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 in the special effect preset „Project". If you want to use the chorus settings in other projects, you can store the settings in a preset too.
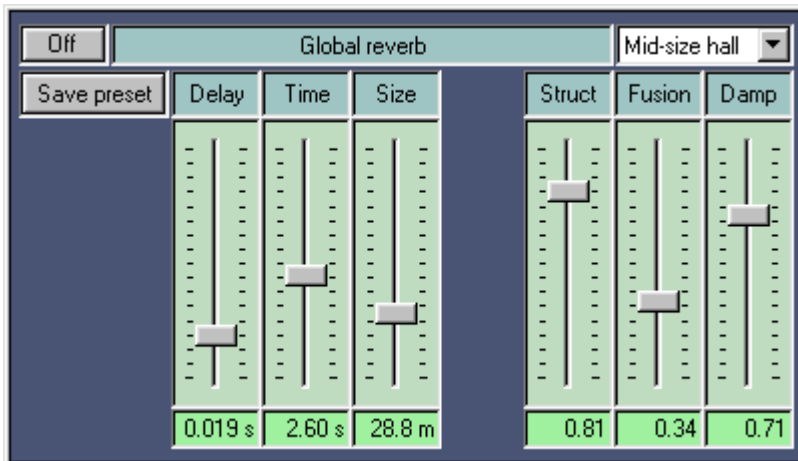
- ▪ **ON/OFF**
  Switches the chorus on and off. If the chorus effect is not used, it should be switched off to lower the processor contention.

- **Select Preset**
  Selects from 32 global presets containing user definable parameter settings. The first entry is not really a preset. It has always the name „Project" and contains the chorus parameter settings stored in the currently loaded Project.

- **Save Preset**
  Saves the current settings of the chorus parameters to one of the 32 global presets. Choose a preset place where you want to store the preset from the combo box and enter the name for this preset. Click on „Ok" to store the preset.
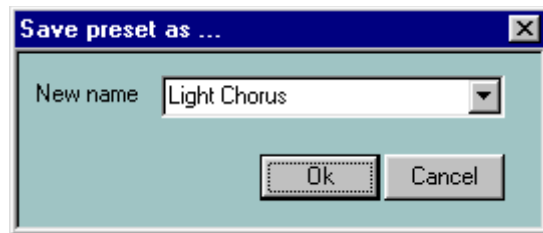
- **Speed**
  Sets the average speed (frequency) of the LFOs modulating the delay time. The adjustable range is from 0.011 Hz to 75 Hz (cycles/second).

- **Spread**
  Sets the relative difference between the speeds of the four LFOs modulating the delay time. The adjustable range is from 0 to 1. With this parameter 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.

5.3   Part effects

5.3.1   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 has its own overdrive effect. The parameters are stored together with the sound program.
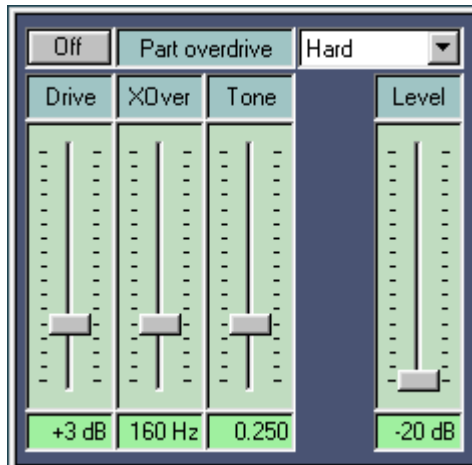
- ▪ **ON/OFF**
  Switches the part overdrive on and off.

- ▪ **Drive**
  Sets the amount of distortion by amplifiying the input signal.
  Range is from -12 dB to +48 dB.

- ▪ **XOver**
  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.

5.3.2   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 echos. VirSyn provides you with three different delay effect types each composed of two delay lines: stereo delay, mono delay and cross delay. Every part has its own echo/delay effect. The parameters are stored together with the sound program.
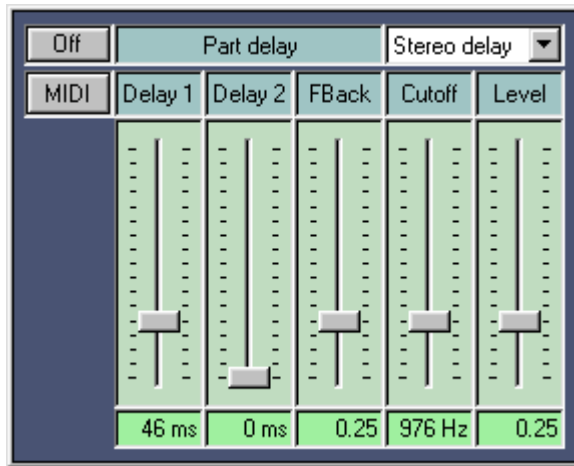
- ▪ **ON/OFF**
  Switches the part echo/delay effect on and off.

- ▪ **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 triplett 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 triplett 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.

5.3.3 Modulation

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

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

- **Flanger**
  The Flanger is composed of two independet 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.
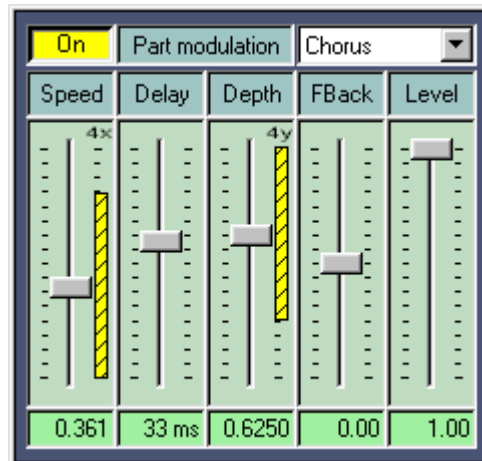
- **ON/OFF**
  Switches the part modulation effect on and off.

- **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.

6 Pattern sequencer

6.1   Overview

The pattern sequencer of VirSyn is based on the concepts of the classic analogue step sequencer. Every part has ist own sequencer with a maximum of 64 steps for each pattern. The pattern sequencer repeats playing one pattern containing up to 64 steps for each part. The pattern played can be selected from 256 patterns, organized in 4 groups with 8 pattern banks each containing 8 patterns. The 64 steps are divided into 4 measures with 16 steps each. This set of 256 patterns is stored together with the project. The sequence in which the pattern are played can be programmed using the **Song Sequencer**.

The structure of a pattern:

Pattern

	Measure 1		Measure 2		Measure 3		Measure 4	
Part 1	Step 1 ▪▪▪ Step 16	Step 17 ▪▪▪ Step 32	Step 33 ▪▪▪ Step 48	Step 49 ▪▪▪ Step 64
Part 2	Step 1 ▪▪▪ Step 16	Step 17 ▪▪▪ Step 32	Step 33 ▪▪▪ Step 48	Step 49 ▪▪▪ Step 64
Part 3	Step 1 ▪▪▪ Step 16	Step 17 ▪▪▪ Step 32	Step 33 ▪▪▪ Step 48	Step 49 ▪▪▪ Step 64
Part 4	Step 1 ▪▪▪ Step 16	Step 17 ▪▪▪ Step 32	Step 33 ▪▪▪ Step 48	Step 49 ▪▪▪ Step 64
Part 5	Step 1 ▪▪▪ Step 16	Step 17 ▪▪▪ Step 32	Step 33 ▪▪▪ Step 48	Step 49 ▪▪▪ Step 64
Part 6	Step 1 ▪▪▪ Step 16	Step 17 ▪▪▪ Step 32	Step 33 ▪▪▪ Step 48	Step 49 ▪▪▪ Step 64
Part 7	Step 1 ▪▪▪ Step 16	Step 17 ▪▪▪ Step 32	Step 33 ▪▪▪ Step 48	Step 49 ▪▪▪ Step 64
Part 8	Step 1 ▪▪▪ Step 16	Step 17 ▪▪▪ Step 32	Step 33 ▪▪▪ Step 48	Step 49 ▪▪▪ Step 64
Part 9	Step 1 ▪▪▪ Step 16	Step 17 ▪▪▪ Step 32	Step 33 ▪▪▪ Step 48	Step 49 ▪▪▪ Step 64
Part 10	Step 1 ▪▪▪ Step 16	Step 17 ▪▪▪ Step 32	Step 33 ▪▪▪ Step 48	Step 49 ▪▪▪ Step 64
Part 11	Step 1 ▪▪▪ Step 16	Step 17 ▪▪▪ Step 32	Step 33 ▪▪▪ Step 48	Step 49 ▪▪▪ Step 64
Part 12	Step 1 ▪▪▪ Step 16	Step 17 ▪▪▪ Step 32	Step 33 ▪▪▪ Step 48	Step 49 ▪▪▪ Step 64

One step can represent one note (for monophonic parts) or a chord (for polyphonic parts) containing a maximum of four notes. In Arpeggio mode the note currently played on the keyboard is transposed by the step note intervall set by the step slider. With the parameter Arpeggio key you can program which of the currently pressed keys is played on a given step position.

To program complex rhythmic structures, the note length and the gate time can be set individually for each step. The control sequence A or B can be used to adjust one or more sound parameters for each step. The values set by the control sequence A and B can be used as a modulation source in the modulation matrix.

The tempo can be set for each part individually, but normally the tempo is synchronized to part 1.

6.2   Common parameter

This section describes the common parameters and functions of the pattern sequencer.
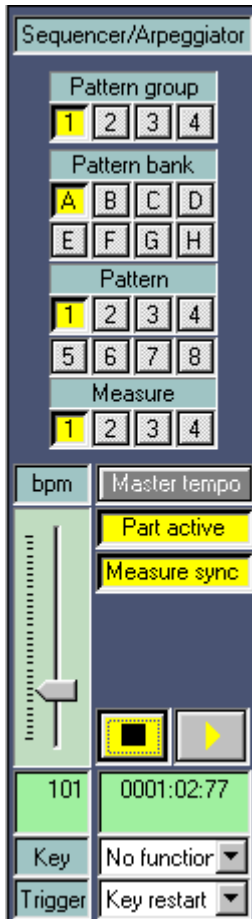
- **Pattern group**
  Select pattern group 1-4.

- **Pattern bank**
  Select pattern bank A-H.

- **Pattern**
  Select pattern 1-8.

  The selection of a new pattern will not take effect until the currently played pattern has ended.

- **Measure**
  Selects the subset of 16 Steps (one measure) visible on the property sheet::

  - **Measure 1**
    Steps 1-16

  - **Measure 2**
    Steps 17-32

  - **Measure 3**
    Steps 33-48

  - **Measure 4**
    Steps 49-64

- **Tempo**
  This slider controls the tempo of the sequence in beats per minute (bpm). The range is from 40 – 295 bpm. Part 1 is the master track which determines the tempo for all parts where **Tempo Sync** is set to „**Master tempo**". For parts where **Tempo Sync** is set to „**free**", the tempo slider controls the tempo relative to the master tempo. In this case the range is from 50% – 150% relative to the master tempo.

  The tempo slider also controls the synchronization of the LFOs and the Delay effect. With external synchronization the tempo slider is controlled by the MIDI clock.

  The tempo can be modulated by a source with the modulation matrix ! With a sync'd LFO as source you'll get the opportunity to alter the groove of a sequence.

- **Tempo sync**
  With this button you can synchronize the tempo of the part to the master part 1:

  - **Master tempo**
    The tempo of the selected part is sync'd to part 1.

  - **Free**
    The tempo of the selected part can be adjusted relativ to the tempo of part 1 in the range of 50-150%.

- **Activate sequence**
  This button activates/deactivates the sequencer for this part. Use this for example to get some parts played by the sequencer and some parts played live with the keyboard.

▪ **Measure sync**
This button activates/deactivates the measure sync function for this part / sequencer track. With measure sync activated the sequence restarts synchronously to the restart of the master track (part 1). Otherwise the sequence restarts when it reaches the end of the pattern. This function is very important to get sequences of different length synchronously restarted on step 1 when the master part 1 is restarting. But its also very interesting to let them run free...

▪ **Stop / Play**
These buttons stop the sequencer or restart the sequencer on step 1.

▪ **Measure info**
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).

▪ **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:

• **No function**
The keyboard has no function.

• **Live 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.

• **Arpeggio**
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 note sliders. The Arpeggio key parameter determines which note of the chord will be played on a given step transposed by the step value.

- **Select pattern**
  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
D	selects pattern 2
E	selects pattern 3
F	selects pattern 4
G	selects pattern 5
A	selects pattern 6
B	selects pattern 7
H	selects pattern 8
C#	selects previous pattern
D#	selects next pattern
F#	selects previous pattern bank
G#	selects next pattern bank

- **Mute part**
  Use the keyboard to mute a part. In every octave on the keyboard the notes have the same function:

C	mute/activate part 1
C#	mute/activate part 2
D	mute/activate part 3
C#	mute/activate part 4
E	mute/activate part 5
F	mute/activate part 6
F#	mute/activate part 7
G	mute/activate part 8
G#	mute/activate part 9
A	mute/activate part 10
B	mute/activate part 11
H	mute/activate part 12

- **Trigger mode**
  The trigger mode controls the way the keyboard starts and stops the sequencer. This setting takes no effect if the **Key function** parameter is set to „No function".

  - **Key Restart**
    Sequencer starts with key down and stops with key up.

  - **Key Hold**
    Sequencer starts with key down and stops only if the sequencer is stopped.

6.3 Step parameter

This section explains the parameter available for each step. The user interface elements are described from top to bottom as they appear on the screen.

▪ **Step buttons A**
The Step buttons control the end of the step sequence and of the **Control sequence A.** Just press the button for the first step which should not be played. The color of the step button is used to indicate the current step and the step length:

- **Red**
  Step is beyond the end of the sequence.

- **Yellow**
  Step is currently played.

▪ **Control A**
The control sequence A can be used to adjust some sound parameters for each step. To control a sound parameter use the modulation matrix with the modulation source „**Sequencer > Control A**" and the sound parameter as modulation destination. With this method it's possible to control more than one sound parameter with Control A..

▪ **Step buttons B**
The Step buttons control the end of the control sequence B. Using a longer or shorter length of control sequence B than the length of sequence A gives you the possibility to change some sound parameters asynchronously to the step sequence.

▪ **Control B**
The control sequence B can be used to adjust some sound parameters for each step. To control a sound parameter use the modulation matrix with the modulation source „**Sequencer > Control B**" and the sound parameter as modulation destination. With this method it's possible to control more than one sound parameter with Control B..

▪ **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:
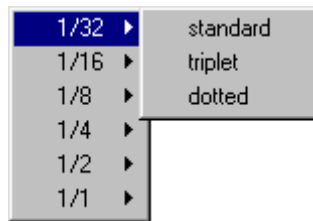
This menu allows you to use note lengths ranging from 1/32 to 1/1 with the variants triplett (2/3 standard length) and dotted (3/2 standard length).
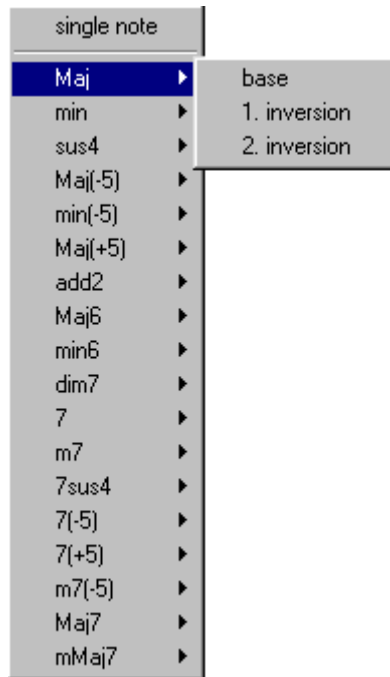
Pressing the Ctrl Key together with the right mouse click changes the step time for all steps.

▪ **Gate time**
The gate time is the ratio between the time the note is on and the time the note is off. To change the gate time click on the button one ore more times to set the desired time:

- **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.

- **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 slider.

  Click into the field with the right mouse button, and you get a context menu to choose the arpeggio key:
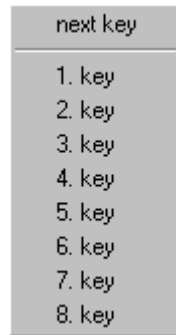
next key
1. key
2. key
3. key
4. key
5. key
6. key
7. key
8. key

  Pressing the Ctrl Key together with the right mouse click changes the step time for all steps.

- **Step note**
  The step note sliders are used to define the pitch of single notes and the base pitch for chords played. The range is from –3 octaves to +3 octaves in steps of one halftone. In Arpeggio mode the slider defines the transposition of the note played on the keyboard.

- **Step info**

  The step info fields below the step note sliders shows you the currently assigned pitch or chord. Click into this field with the right mouse button to get a context menu with functions to switch between single note and chord mode by selecting a chord and alteration to be played:

single note	
Maj ▶	base
min ▶	1. inversion
sus4 ▶	2. inversion
Maj(-5) ▶	
min(-5) ▶	
Maj(+5) ▶	
add2 ▶	
Maj6 ▶	
min6 ▶	
dim7 ▶	
7 ▶	
m7 ▶	
7sus4 ▶	
7(-5) ▶	
7(+5) ▶	
m7(-5) ▶	
Maj7 ▶	
mMaj7 ▶	

  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.

- **Skip step**

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

- **Mute step**

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

7 Song sequencer

7.1   Overview

The song sequencer is used to program and playback a sequence of patterns. For each pattern in this sequence you can define the number of repeats for the pattern and the parts to be muted. Song playback starts with the currently hilighted song position in the list editor. To start playback on an arbitrary position, click on the respective line before pressing the start button. You can even jump to another song position during playback by just clicking on the line you want to jump to.

The sequence of patterns programmed in the songeditor is stored together with the project.
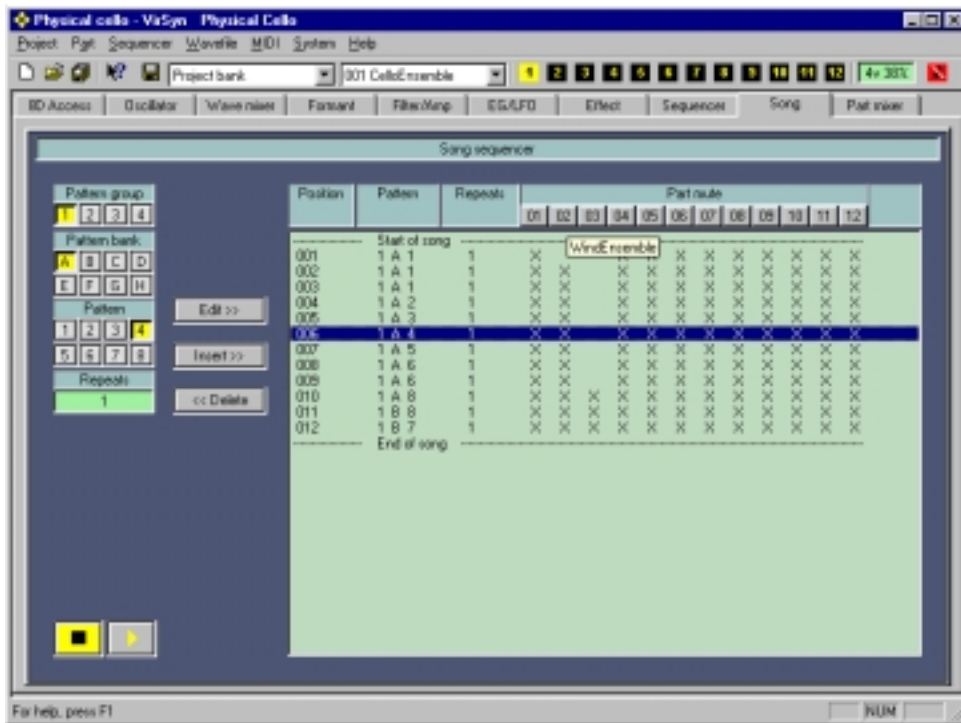
7.2   Operation

This section explains the base operations of the song sequencer and how to insert, edit and delete a pattern in the song list.
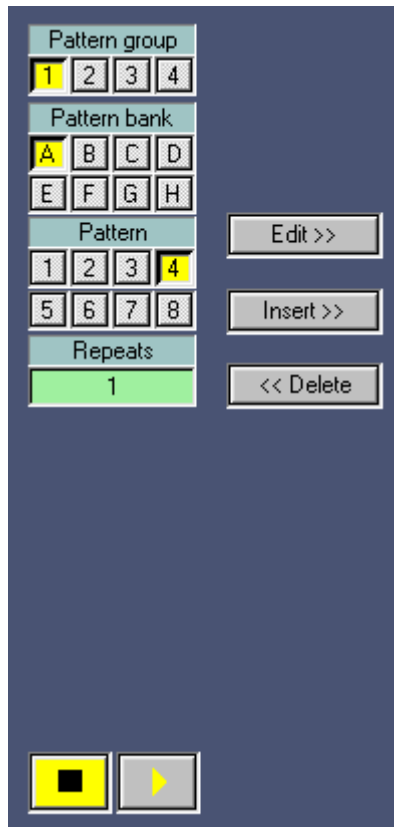
- **Pattern selection**
  Selects the pattern for the next edit or insert operation as described in the chapter **Pattern sequencer.**

- **Repeats**
  Click into this field and enter the number of repeats for the current song position.

- **Play**
  This button starts the song sequencer. The sequencer starts playing on the hilighted song position.

- **Stop**
  This button stops the song sequencer. The sequencer stops automatically if the end of the song is reached.

▪ **Song list**
The song list window shows you the list of patterns played by the song sequencer. Up to 256 patterns can be inserted into the song list. The first line and the last line represent the beginning and the end of the song list and can't be edited or deleted.

From left to right the following parameters are displayed:

- **Song position**
  The song position numbered from 001 to 256.

- **Pattern**
  The patterns for every song position in the format:
  <pattern group> <pattern bank> <pattern>.

- **Repeats**
  The number of repeats of the pattern.

- **Part mute**
  The mute state of the 12 parts:
  X = Part is not muted.

▪ **Edit**
With this button you can edit one line in the song list.

1. Select the line you want to edit in the song list.

2. Select the new pattern with the **Pattern selection** described above.

3. Enter the number of repeats for the pattern.

4. Press the button **Edit >>** to alter the line selected.

- **Insert**
  With this button you can insert a new line into the song list:

  1. Select the line after which you want to insert the new line into the song list.

  2. Select the new pattern with the **Pattern selection** described above.

  3. Enter the number of repeats for the pattern.

  4. Press the button **Insert >>.**

- **Delete**
  With this button you can delete a line from the song list:

  1 Select the line after which you want to delete in the song list.

  2 Press the button **<< Delete**.

- **Part mute buttons**
  In the title of the songlist there are 12 buttons corresponding to part 1-12. This buttons provides you with the following functions:

  - **Mute part**
    The 12 part mute buttons can be used to mute parts for the pattern at the currently hilighted song position. Parts not muted are shown marked with an **X**.

  - **Part info**
    Keeping the mouse pointer above one of the part buttons for a few seconds without clicking pops up a small window showing you the program name for this part.

8 Part mixer

8.1   Overview
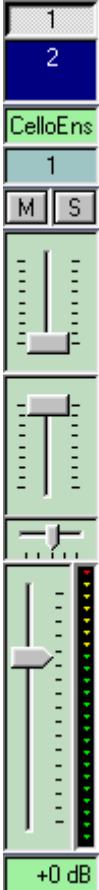
This is the main mixing console where all parts are finally mixed down to the stereo output. In the part mixer you can also adjust the send level for the global reverb and chorus effect. The effect output of this global effects is mixed to the stereo output in the master section. Every part can be individually 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 possibilty 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.
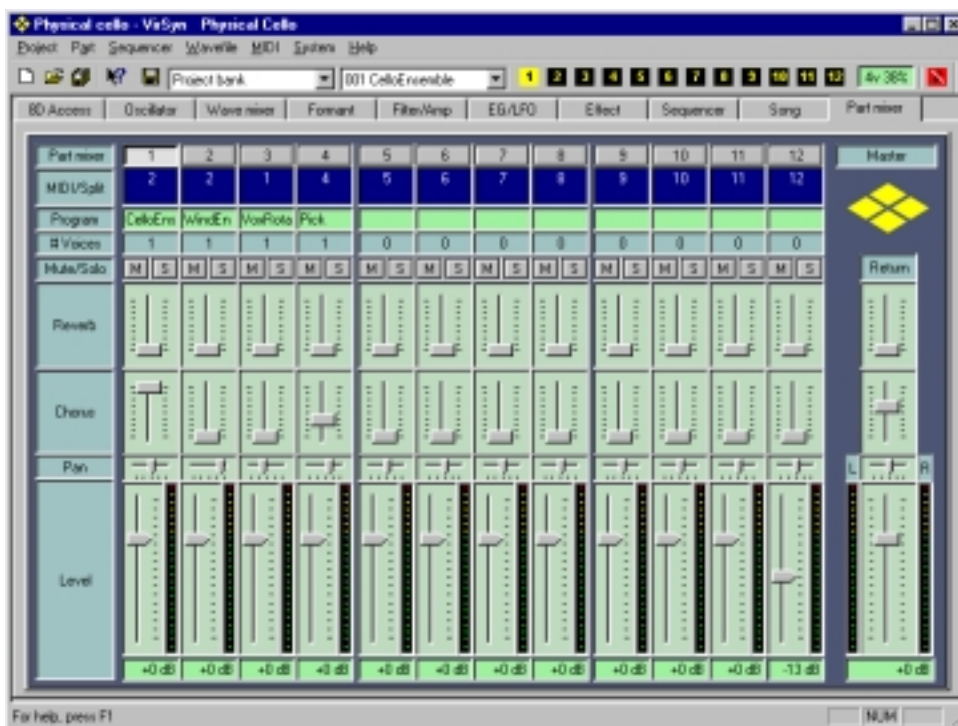
8.2 Operation

8.2.1 Part section

- **Part button**
  This buttons select the part to be viewed / edited. It has the same function as the part buttons in the toolbar and the part menu.

- **MIDI/Split**
  This edit field is used to display and alter the MIDI channel assigned to this part. All MIDI Note On/Off and Controller messages from this channel are directed to this part. It also display the keyboard zone the part is responding to MIDI input messages. If for example **C1-C3** is displayed only notes between C1 and C3 will be played by this part. Clicking into the edit field with the right mouse button for the part to be changed, you'll get a context menu where you can alter those settings as described later in this section.

- **Program info**
  Displays the name of the sound program for every part. Leaving the mouse pointer above one of this info fields for a moment pops up a tooltip window containing the full name.

- **# 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).

- **Mute/Solo**
  The buttons **M** and **S** let you mute or solo the part.

- **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.

- **Part level**
  This slider is used to adjusts 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).

- **MIDI channel and Keyboard split zone**
  With the context menu of the **MIDI/Split**: field you can alter the settings for MIDI channel and for keyboard split:

Channel 1
Channel 2
Channel 3
Channel 4
Channel 5
Channel 6
Channel 7
Channel 8
Channel 9
Channel 10
Channel 11
Channel 12
Channel 13
Channel 14
Channel 15
Channel 16
Clear Splitpoint
Set Splitpoint
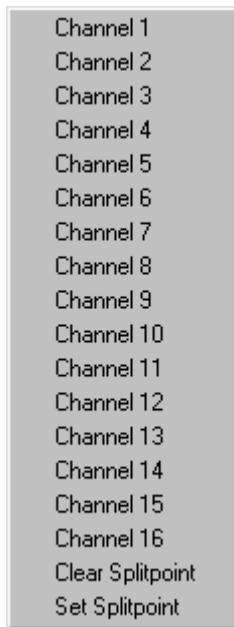
- **Channel 1 – 16**
  Selects the MIDI channel the part is responding to MIDI messages.

- **Clear Splitpoint**
  With this function you can clear a previously set keyboard split zone for this part. Afterwards the part will respond to all notes of the MIDI channel set.

- **Set Splitpoint**
  With this function you can define a keyboard split zone. Afterwards the part will respond only to Note On messages coming from this keyboard zone. By selecting this function you will be guided through two dialogs where you can specify the lowest and the highest key this part should respond to:

Press the lowest key of the keyboard zone you want to define after the following dialog appears:
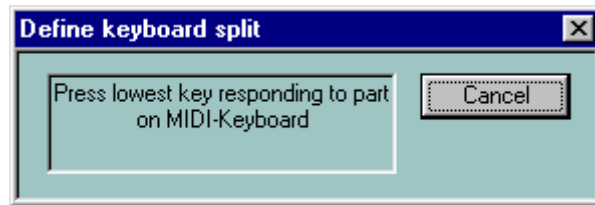
Press the highest key of the keyboard zone you want to define after the following dialog appears:
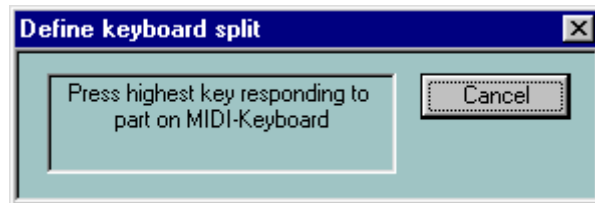
With the **Cancel** button you can abort programming the keyboard zone.

- **Number of voices**
  You can set the number of simultaneous voices for this part with the context menu of the field **# Voices** :

```
0 Voices
1 Voice
2 Voices
3 Voices
4 Voices
5 Voices
6 Voices
7 Voices
8 Voices
9 Voices
10 Voices
11 Voices
12 Voices
16 Voices
20 Voices
24 Voices
28 Voices
32 Voices
```
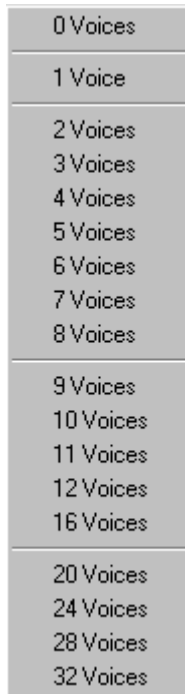
The number of voices used for this part can be set from 0 (part inactive), 1 (monophonic part) to a maximum of 32 (polyphonic part). Please note that in this version (1.0) all parts together can have no more than 64 voices. In addition to this the total amount of voices you can use depends on the processing power of your computer. If the processor usage exceeds the limit of 90% VirSyn automatically reduces the number of voices for some parts.

8.2.2   Master section
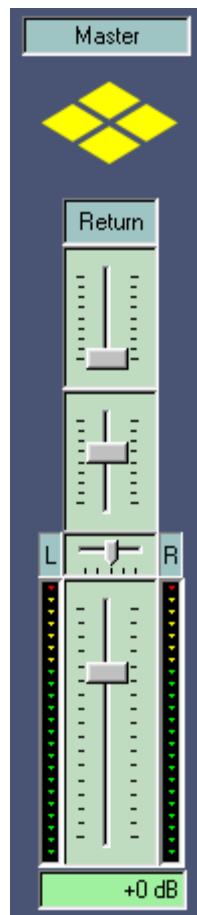
- **Reverb return**
  Adjusts the sound level returned from the global reverb effect.

- **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 respectively. Before the stereo output signal is sent to the soundcard two analogue style distortion units prevent the signal from digital clipping.

Appendix

(A)   Technical data

General

- **Multi modular synthesis**

  - Realtime software synthesizer.

  - 12 independent modular synthesizer (parts).

  - For every part you have a sound program with wireless module patching, settings for the sound parameters and the modulation matrix.

  - Multi layer/key split: up to 12 sound programs can be played simultaneously or can be assigned to different keyboard zones.

- **Intelligent modulation matrix**

  - Route 21 modulation sources to 81 possible destinations.

  - The characteristics of the modulation is automatically adjusted due to the used sourse and destination.

  - Nearly every parameter can be controlled by the modulation matrix.

- **Sound library**

  - The supplied sound library contains over 800 sound programs. (including 808/909/303...)

- **3 user interface levels**

  - Navigate through 8-dimensional sound spaces with controllable speed.

  - Change every parameter of the sound program.

  - Change the module patching to create your own synth.

- **System info**

  - Up to 64 voices assigned freely to the 12 modular synthesizers available.

  - Samplerate: 44100/32000/22050 Hz.

  - Context sensitive online help.

  - Audio output can be written directly to disc (WAV format)

  - Automatical optimization to the minimal possible latency of your sound card.

  - Interfacing to external software sequencer using MIDI loopback devices, for example Hubi´s MIDI Loopback

Modules

- **Keyboard**

  - Master tuning

  - Portamento / glissando function also working polyphonic (!) with optional "fingered" setting, this means activation only on legato played notes.

  - Key priority for monophonic play: Last / Lowest / Highest Note.

- **Oscillators**

  - Antialiasing algorithms for clean "analogue" base sound – even in the highest register.

  - 64 base waveforms including Saw, Square, Triangle, Sine.

  - Wave modulation can be used for pulse width modulation of every (!) waveform.

  - FM inputs allow FM synthesis.

  - Hard sync.

  - Phase switch to allow "running through" oscillators.

  - 3 independent base oscillators.

  - Subharmonic generator containing six oscillators with frequency ratios 1:2:3:4:5:6.

  - Multi oscillator containing six independent oscillators with adjustable detuning for ultra fat sound and choirus effects.

  - White/Pink noise generator with phase synchronization.

- Physical modeling WaveDelay for plucked strings, woodwinds and extreme formant sounds.

▪ **Filter**

- 2 Multimode filter 12dB/18dB/24dB Lowpass/Highpass/Bandpass/Bandstop

- Input with built in overdrive.

- Resonance parameter can be set to self-oscillation even without input signal !

- Formant filter with 3 Bandpass/Bandstop Filters.

▪ **Special modules**

- Wave mixer 5/1 or 3/1 + 2/1.

- Waveshaping synthesis for dynamic evolution of sound spectra.

- Ringmodulator for inharmonc sounds.

- Harmonic Overdrive allowing distortion effect without intermodulation on polyhonic sounds.

- Amplifier allows adjustment of level and modulation of panorama position for every voice independently.

- Subsampling – LoFi sounds (and reduced processor contention) through virtual reduction of sample rate.

- Bit Reduction – digital noise through reduction of sample resolution.

▪ **Modulators**

- 4 very fast ADSR envelope generators with special "Punch" effect.

- 4 LFOs with 6 waveforms with selectable phase and MIDI synchronization. The LFOs can be used as Sample&Hold circuits also.

- 4 external MIDI controllers.

Integrated production environment

- **Effects**

  - 3 Insert effects for every part:

  - Distortion with characteristics: Hard, Soft, Tape, Tube and simple tone control.

  - Stereo delay / cross delay / echo with MIDI synchronization.

  - Modulation effect

  - Chorus with four delay lines

  - 8-pole phaser

  - Flanger

  - 2 global effects:

  - Roomsimulator (from shoe box to Taj Mahal ...)

  - Chorus effect

- **Integrated mixing console**

  - 12/2 channels.

  - Mute / Solo function.

  - 2 Effect sends for every channel.

  - Level meter for every channel and the stereo sum.

- **Step sequencer / Arpeggiator**

  - 12 Track step sequencer with step light programming.

  - MIDI clock synchronization.

  - Independent or synchronized tempo for every track.

  - One pattern can contain up to 64 notes or chords.

  - 2 Parameter can be used as modulation source for every step.

  - Gate time / note duration adjustable for every step.

  - Mute/Skip function.

  - Timing precision down to one sample.

- Realtime pattern selection using the MIDI keyboard.

- Realtime transposition of patterns.

- Programmable arpeggiator.

- Live record function.

▪ **Song sequencer**

- Song sequencer for pattern arrangement.

- Pattern / Song concept.

- 256 pattern can be used for every project.

▪ **MIDI-Implementation**

- Control up to 8 sound parameters simultaneously with one MIDI controller using the 8D sound access page.

- MIDI Learn function

- MIDI Bank/Program Select.

- Timing resolution is below 2 ms.

(B)   MIDI Implementation Chart

Model: VirSyn 1.0   MIDI-Implementation Chart   Version: 1.00

Function		Transmitted	Recognized	Remarks
Basic Channel	Default	X	1-16	
	Changed	X	1-16	
	Default	X	Mode 3	
Mode	Messages	X	X	
	Altered	N/A	X	
Note number		X	0-127	
	True voice		0-127	
Velocity	Note On	X	O	
	Note Off	X	X	
Aftertouch	Key's	X	O	
	Channel	X	O	
Pitch Bend		X	O	
Control Change		X	O	MIDI learn function
Program Change		X	O	MIDI Bank change 1-127
	True Number	N/A	0-127	supported
System exclusive		X	X	
	Song Position	X	X	
System common	Song Select	X	X	
	Tune Request	X	X	
System Realtime	Clock	X	O	Song play, stop, continue
	Commands	X	O	
	Local On/Off	X	X	
Aux Messages	All Notes Off	X	O	
	Active Sensing	X	X	
	System Reset	X	X	

Notes: O = Yes, X = No, N/A = Not Applicable

VirSyn Owner's manual

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www.virsyn.com