- Tactive Instrument Modeller

USER'S MANUAL

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fidelity at work.

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Main Table Of Contents

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GENERAL PART

Introduction 12
Preface 12
About the manual 13
Important safety information 14

Features 16

Noah Hardware 17
Front Panel 17
Back Panel 19
Connections 20

Overview of the Noah Architecture 21
Noah's Slots 21
Instrument Slots 21
Mixer and Effects 21
Slot allocations 22
Number of voices in Noah instruments 23
Triggering the Instruments 23
Input and Output 23
Architecture Block Diagram 24

Operating Noah 25
Starting up Noah 25
The Start Menu 26
Navigating the menus 27
General Menu Structure 27
Opening sub-menus 28
Changing parameters 28
Combined sub-menus and parameters 29
Menus with more than 4 sub-menus or parameters 29
Leaving a menu 29
Examples 30
Single Mode and Multi Mode 32
Presets 32
Preset structure 32
Selecting a preset file – internal and external presets on the Compact Flash Card 34
Selecting a bank 34
Loading presets 34
Presets for a complete Single configuration 35
Presets for a complete Multi configuration 35
Presets for individual instruments of the Multi configuration 35
Presets for Aux Effects 35
Presets for insert effects 35
Presets for the Arpeggiator or the Step Sequencer 35
Previewing presets without keyboard 36
Storing presets 36
Selecting a preset file 36
Naming presets 37
Managing presets 38
Changing presets via MIDI 39
Performance Controller 39
Switching to Control Mode 39
Changing the current allocation 40

The Noah Remote Software 41
Installing the Noah Remote Software 41

Reference Part

Edit Mode: MIDI Menu 44

MIDI Menu 44
Instrument / Devices Menu 44
Slot 1-4 sub-menus 44
Trigger 1/2/3/4 sub-menu 44
Port/Ch1/2/3/4 sub-menus (Port/Channel) 44
Zones1/2/3/4 sub-menu 45
Mixer Menu (Mixer/FX) 45
Clock Menu (MIDI Clock) 45
MultiView Menu 46

Edit Mode: Mixer Menu 47

Mixer Menu 47
Instrument / Slots Menu 47
Slot 1 - Slot 4 sub-menus 47
Anlg/USB Menu 48
Analog sub-menu 48
USB sub-menu 48
Master Menu 49
Aux sub-menu (Aux Master Sends/Returns) 49
Chorus sub-menu 49
Delay sub-menu 49
Reverb sub-menu 50
Outputs sub-menu 50
ADAT sub-menu 50
USB sub-menu 50
MultiView Menu 51

Edit mode: The FX menu 52

Using effects 52
The Aux FX sub-menu - Chorus, Delay, Reverb 52
Adding the effect component of an Aux effect to the signal of another effect 53
Chorus sub-menu (Chorus / Flanger) 53
Delay sub-menu 54
Send/Rtn sub-menu (Send / Return) 54
The different types of Delay 54
Dual Delay 54
LCR Delay 55
Multitap Delay 55
Stereo Delay 56
Synth Delay 56
Reverb sub-menu 57
The Insert 1/2 sub-menus - insert effects 58
Insert effects list 58
Stereo EQ / Parametric EQ 59
Graphic EQ 59
Chorus Effects 60
Ensemble 60
Master Chorus 60
Harmonic Chorus 61
Hexa Chorus / Triple Chorus 62
Flanger Effects 63
Master Flanger 63
Harmonic Flanger 64
Random Flanger 65
Space Flanger 66
Step Flanger 67
Master Phaser 68
SSB Phaser 69
2 Voice Pitch Shifter 70
Stereo Pitch Shifter 70
Feedback Pitch Shifter 71
Autopan 71
Tremolo 72
Auto Wah 72
Amplifier 73
Decimator 74
Distortion 75
Overdrive 75
Resonator 76
Ringmodulator 76
Softclip 77
Stereo Expander 77
Tube Processor 78

Compressor 78
Expander 79
Limiter 79
Gate 80
Dynamics 80
Sources Menu (Preset Sources) 81
The FX Bypass Menu – Muting Effects 81

The System Menu 82
Parameter 82
MIDI sub-menu 82
Controls sub-menu 82
Device sub-menu (System Device) 83
Memory sub-menu 83
Internal Flash / Compact Flash 83

The Utility Menu 84
Presets sub-menu 84
Add Bank Menu 84
Del Bank Menu (Delete Bank) 84
Ren Bank Menu (Rename Bank) 85
Del Pres Menu (Delete Preset) 85

Arpeggiator 86
Introduction 86
Overview 86
Basics 86
MIDI Message Handling 86
Control via the MIDI Keyboard 87
Timing and Synchronization 87
Activating the Arpeggiator 87
Parameter 87
Control Menu 87
Scan Menu 88
Capture Menu 90
Note Menu 90
Output Menu 91
LFO Menu 93
Clock sub-menu (Beat/Clk) 93
Freq sub-menu (Freq Hz) 93
Resync sub-menu 94
KeybCtrl Menu 95
OutTiming Menu 96

The Step Sequencer 97
Parameters 97
Global Menu 97
Steps Menu 98
Example Sequence Configuration 98

Lightwave 99
Introduction 99
Parameters 100
Osc Menu (Oscillators) 100
Common sub-menu 100
Osc 1 / 2 sub-menu 100
Mix Menu 101
Osc 1 / 2 sub-menu 101
Vcf Menu 102
Common sub-menu 102
Vcf 1 / 2 sub-menu 102
Envelope sub-menu (VCF Envelope) 103
Amp Menu (Amplifier) 104
Pan 1 / 2 sub-menu 104
Envelope sub-menu 104
Mod Menu (Modulation) 105
Lfo 1 / 2 sub-menu 105
Free Env sub-menu 106

Six-String (optional) **107**
Introduction 107
What does the Six-String give you? 107
Synthesis Structure 107
The Acoustic model 107
The Electric model 108
Presets 108
Parameter 109
Type/Vol Menu 109
Strings Menu 109
Excit Menu 110
Pluck Menu 110
Noise sub-menu 110
Envelope sub-menu 111
Level sub-menu 111
Damping Menu 111
Microph Menu (Microphone) 112
Pickup Menu 112
Edit sub-menu 112
Body Menu 113
Edit sub-menu (Settings) 113
Slap Menu 114
Pitch Menu 114
Envelope sub-menu 114
LFO sub-menu 115
Global sub-menu 115
Control Menu 116

B-2003 .... **117**
Introduction 117
Drawbars 117
Parameter 117
Pedal Menu 118
Upper Menu 118
Lower Menu 118
Perc&KC Menu 119
Perc sub-menu 119
Modeling Menu 119
ToneWheel sub-menu (Tone Wheels) 119
Drawbars sub-menu 119
Envelope sub-menu 120
Effects Menu 120
Vibrato sub-menu 120
Drive sub-menu 120
Tone sub-menu 121
Rotor sub-menu 121
Horn sub-menu 121
Bass sub-menu 121
Micro sub-menu 122
Control sub-menu 122
External Menu 122
Midi Menu 123

Minimax **124**
Introduction 124
What's so Max about the Mini? 124
Structure and Overview 124
Parameters 125
Osc Menu (Oscillator Bank) 125
Osc 1 -3 sub-menus (Oscillator-1 - 3) 125
Pitch sub-menu 125
Mixer Menu 126
Osc 1-3 sub-menu 126
Noise sub-menu 126
External sub-menu 126
Filter Menu 127
Vcf sub-menu 127
Envelope sub-menu (Filter Envelope) 127
Mod sub-menu (Modulation) 128
Loudness Menu 128
Amp sub-menu (Amplifier) 128
Envelope sub-menu (Loudness Envelope) 128
Control Menu (Controllers) 129
ModWheel sub-menu 129
Glide sub-menu 129
TrigMode sub-menu (Triggers&Modes) 129

Vectron Player 130
Overview 130
Vector Synthesis 130
Parameter 130
Mod sub-menu 130
The Joystick 130

Sequential CircuitsTM Pro-One (optional) 131
Introduction 131
Structure and Overview 131
Parameter 132
Osc Menu 132
Osc A sub-menu 132
Osc B sub-menu 133
Mixer Menu 133
Filter Menu 134
VCF sub-menu 134
Envelope sub-menu 134
Amp Menu 135
Envelope sub-menu 135
Lfo Menu 135
Mod Menu (Modulation) 136
From sub-menu 136
To sub-menu 136
Wheel sub-menu (Wheel Mod) 137
Aftertch sub-menu (Aftertouch) 137
Glide Menu 138
Mode Menu 138
Global Menu 139
Env Fol Menu (Env Follower) 139

Vocodizer 140
Introduction 140
How does a Vocoder actually work? 140
Presets 141
Parameter 141
Analysis Menu 141
LPF EnvF sub-menu 141
BPF EnvF sub-menu 142
HPF EnvF sub-menu 142
Gains sub-menu 142
FltrSet Menu (Filter Settings) 142
Edit sub-menu 142
Freq sub-menu (Frequencies) 143
Matrix Menu 143
Edit sub-menu 143
Synthesis Menu 144
V/U Det sub-menu (V/U Detection) 144
UnvSource sub-menu 145
Level sub-menu 145
Pan sub-menu 145
Sources Menu (Input Sources) 146
InGainAn Menu (Analysis) 146
InGainSy Menu (Synthesis) 147
Output Menu 147

Interpole 148
Introduction 148
Structure and Overview 148
Parameter 149
Channel 1 / Channel 2 Menu 149
Env sub-menu (Envelope) 149
LFO sub-menu 151
Filter sub-menu 151
Link Menu 152
Sources Menu (Source Select) 152

APPENDIX

Noah - Technical Specifications **154**

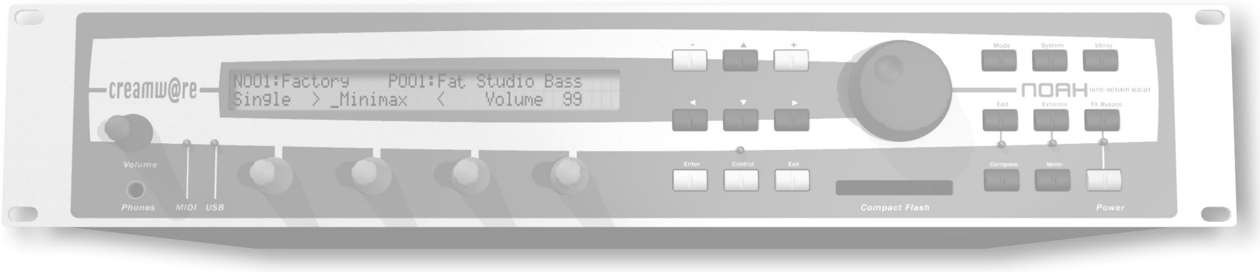
The Hotline **155**

Software Problems 155

Warranty and Disclaimer 157

Index **158**

Noah is state of the Ark



User's Manual

General Part

Introduction

Preface

Dear Noah User,

On behalf of our team at CreamWare, I would like to thank you for choosing Noah as your new sound synthesis engine. We have put many years of effort and our heart's blood into a system designed to be much more than just a commonplace synthesizer. We hope Noah will not only provide you with new sounds and inspiration, but will substantially improve the way you approach music making, thanks to its unparalleled flexibility.

People frequently ask how the product name "Noah" came about. Indeed, the name "Noah" is a metaphor for the fundamental concept of the Noah product. Following the idea of the ark, Noah provides a platform and safe harbor to allow all sorts of classic, current, and future synth technologies to coexist inside a single rugged synth box. We believe Noah's ability to emulate entirely different synth architectures with its "plug-in runner" architecture will set the trend for many synthesizer instruments to come.

It was time for the Noah concept to become a reality. Our friends at Wine Country Sequential have been servicing and repairing classic Sequential Circuits synthesizers for many years.

However, as the availablility of hardware components declines, Wine Country Sequential is finding it increasingly difficult to keep these original products from the 80s alive. Consequently, Wine Country Sequential has initiated the "Software Survival Kit" and has become the first 3rd party to provide faithful recreations of their original classics with the release of the Noah platform.

We at CreamWare are committed to the development of more and more software for the Noah platform, just as we've done for our PCI-card product line. We want you to feel that Noah is one of the best investments you've ever made in music gear. If you like Noah, please help us make it a success by telling all your friends about it.

I am very interested in your thoughts and experiences with Noah. Please tell me personally – my email address is **fh@creamware.de**. Your comments are always appreciated.

We hope your new Noah synthesizer gives you new ideas, much success, and a lot of fun!

Frank Hund

President

CreamWare GmbH

The Creamware hardware is hereby certified to conform to the requirements set forth in the guidelines for electromagnetic acceptability (89/336/EWG).

CreamWare Datentechnik GmbH,
April 2003
Dr. Hans-Ulrich Hund

About the manual

Structure of the manual

In order to enable you to most quickly and easily find what you're looking for, this manual is divided into the following chapters:

- The ***Introduction***, which you're reading right now, describes the setup, care and maintenance of the device.

- The ***General*** section describes the hardware, including all operating controls, tips on cabling, an overview of the internal architecture and the basics of using Noah,

- The ***Reference*** section completely describes the menus of all fixed components (mixer, MIDI, system components, etc.) and of all optionally-loadable instruments and effects, as well as their operation and all of their parameters,

- The ***Appendix*** of the manual in your language contains technical specifications, MIDI implementation tables, terms of the warranty, information regarding our hotline, etc.

- Technically experienced readers may want to refer to the section ***'Technical Reference'***, which contains the MIDI Controller tables and the parameter structures of all modules.

Additionally, once you've installed the Noah software, you can make use of the ***Online Manual*** (installed with the software), which includes additional chapters regarding the operation of Noah via computer.

Be sure to read the **'ReadMe' file**, which may contain information and last-minute changes which were not yet available at the time that the manual was being prepared.

Conventions used in this manual

In addition to normal text, the following text formats are used:

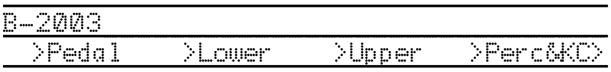
This format is used for a tip providing additional information.

This format indicates especially important information to which you should pay close attention!

Whenever a button is mentioned in the text, you will find a number in parentheses following its name. This is the same number used to identify the corresponding button in the illustration in the *Noah Hardware* chapter.

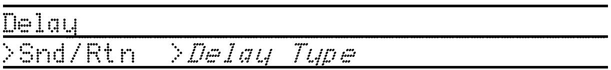
Display illustrations

In some places, you will see pictures of the display (see above). Normally, however - particularly in the Reference section - display illustrations have the following form (see below):



In these illustrations, for technical reasons, the characters ▶, ▷, ✦, and ✚ are replaced by „>", while ✚ is replaced by „<".

In these illustrations, text in italics is used to indicate that the displayed name may vary depending upon which module is loaded.



The full-length name (Long Name) and associated value which appear in the upper line of the display while a parameter is being edited are omitted.

Parameter listings

In the Reference section, each parameter is presented beginning with a heading of the following form:

HiDamp (High Damp) [0, ..., 24000 Hz]

Or in more general form:

Short (Long Name) [minimum value, ..., maximum value]

At far left is the name displayed in the lower line of the display (Short Name). Next, in parentheses - if it differs from the Short Name - is the Long Name which appears in the upper line of the display during editing. Following this, enclosed in square brackets, is the parameter's value range.

MIDI controller assignments and default values for each parameter can be found in the accompanying ***Technical Reference*** booklet.

Important safety information

Please read these instructions through thoroughly.

Please observe the following instructions for operation of the Creamware hardware. Correct and trouble-free operation of the device is guaranteed only under the conditions described here. Furthermore, this information is relevant to product liability.Therefore, it is absolutely essential that you carefully read and follow all instructions presented here!

Transport

Transport the device carefully. Never drop it or allow it to fall or tip over. Always set it down carefully on a stable surface. Owing to the device's own weight, failure to observe these precautions may result in damage to the device which is not covered under the warranty.

Always make certain that the device is in a stable position or is securely installed, both during transport and while it is being used, and that it cannot fall, fall over or slide. Otherwise, owing to the weight of the device, it is possible that personal injury could occur.

Installation

Use of this device in the following types of environment may lead to operational problems:

- in locations where it is exposed to direct sunlight

- in locations which are subject to extreme temperature fluctuations or a high degree of humidity

- in extremely dusty or unclean locations

- in locations where it is exposed to strong vibration

Do not expose the unit to temperatures above +50 degrees Celsius or below -10 degrees Celsius. Do not apply power to the unit unless it has a temperature of at least +10 degrees Celsius.

Power

Before connecting the unit to the line power supply, verify that the line voltage selection switch on the rear panel of the unit is set properly for your line voltage. Never connect the unit to line power of a different voltage than selected by this switch.

STOPP

Warning !! Do not open the unit (this will void the warranty!). Service should be performed only by properly trained, Creamware-authorized technical personnel.

The unit must never be opened unless it has first been disconnected from the line voltage supply (removal of power cord).

Handling

Handle the operating controls with appropriate care, in order to avoid unnecessary damage.

Do not place heavy objects on top of the device.

The device is suitable for use only for the purposes described in this manual. For safety reasons, the device should not be used for other purposes.

In using this device, the applicable VDE regulations are to be observed. The following regulations are especially important: DIN VDE 0100 (Part 300/11.85, Part 410/11.83, Part 481/10.87), DIN VDE 0532 (Part 1/03.82), DIN VDE 0550 (Part 1/12.69), DIN VDE 0551 (05.72), DIN VDE 0551e (06.75), DIN VDE 0700 (Part 1/02.81, Part 207/10.82), DIN VDE 0711 (Part 500/10.89), DIN VDE 0860 (05.89), DIN VDE 0869 (01.85). The VDE publications are obtainable from: VDE-Verlag Gmbh, Bismarckstr. 22, Berlin.

Care, maintenance and cleaning

Dirt and stains can be wiped away with a clean, dry cloth. Never use liquid cleaning agents, solutions or other flammable substances.

Liquids and foreign objects

Never place liquid-containing vessels, glasses, cups etc. near this device. Contact with liquids may result in short circuits, fire or total failure of the device. If a liquid substance has gotten inside the unit or the power supply, it should no longer be used and must be inspected and possibly repaired by a technical specialist. If this occurs while the unit is in use, line power should be disconnected at once.

Take all necessary precautions to prevent metal objects from getting inside the device housing. Should this nevertheless occur, disconnect the line cord immediately. Contact the dealer from whom you purchased the device for further assistance.

Interference with other devices

This device contains a microcomputer which can cause disturbances in television or radio reception. Never operate the unit in close proximity to a radio or television set.

Retain the printed instruction manual !!

Once you have read this instruction manual, please store it in a safe place to ensure that you have it ready at hand whenever it is needed.

A word regarding data

As a result of unforeseeable operational disturbances, it can occur that memory contents arepartially or completely lost. We recommend that you archive your device settings regularly via compact flash card or computer. Creamware accepts no liability for losses or consequential damages arising from loss of data.

Features

* Noah is the first hardware synthesizer with software plug-in technology
* Features truly accurate 1:1 modeling of existing instruments
* Analog, Wavetable, Vector Synthesis, Physical Modelling, Drawbar Organ…
* Comes with six great instruments delivering over 1000 sounds
* Arpeggiator and Step Sequencer
* 2 Up to 4 (Noah/Noah EX) instruments simultaneous

* Effects section with high-end reverb, vocoder and 40 more effects
* Chorus/Flanger, Delay and Reverb as aux effects, 2 freely-assignable insert effects
* Can serve as an effects processor for external signals

* Noah features extensive computer integration:
- USB Interface for audio and MIDI
- Powerful graphical editing software
- New plug-ins downloadable from the Internet
- Can be used as a computer audio interface

* Data storage and exchange using Compact Flash Card (read and write)
* Integrated six-channel mixer
* "Live" control via four freely-assignable infinite rotary controllers
* Illuminated 2 x 40 character LCD display
* Sample rate 44.1 kHz
* 32-bit internal audio data paths

Inputs and Outputs:
* MIDI In, Out, Through
* USB MIDI In and Out
* 2 Analog In, 6,35 mm Phone Jack mono, Input sensitivy switch
* 2 Analog Out, 6,35 mm Phone Jack mono
* ADAT Out optical, 8 channels
* USB Audio In, 2 channels
* USB Audio Out, 6 channels
* WordClock In, BNC
* Headphone, 6,35 mm Phone Jack stereo, level adjustable
* Foot switch

Noah Hardware

Front Panel

1 - Volume / Phones

This rotary control changes the volume of the headphones connected via the associated 6.35 mm jack. The other outputs remain active when headphones are connected.

You can also use this control to preview sounds without a connected MIDI-keyboard.

For more information see: *Operating Noah*: *Previewing of Presets without Keyboard*

2 - MIDI-LED

This LED flashes when the unit is receiving MIDI data.

3 - USB-LED

This LED is lit when Noah is connected to a computer using the USB port.

4 - Display

Displays the menu options and various system settings.

For more information see: *Operating Noah*: *The Start Menu* and *Navigating the Menus*

5 - Continuous Controllers

Use the four Continuous Controllers below the display to access the parameters shown in the lower display line. Usually, each of the four Continuous Controllers is assigned to one of the displayed parameters.

Pressing/Pushing any of the Continuous Controller (Push Function) will open the corresponding menus or reset the assigned parameters to a default value. Press the *Control* Button (9) to switch to *Control* Mode. Now you can use the Continuous Controllers as Performance Controllers to manipulate selected parameters in realtime.

For more information see: *Operating Noah*: *Navigating the Menus* and *Performance Controllers*

6 - Plus / Minus Buttons

Pressing these two buttons lets you load the next or the previous preset or adjust the value of a selected parameter up or down one step at a time.

For more information see: *Operating Noah*: *Navigating the Menus* and *Presets*

7 - Up / Down / Right / Left Buttons

Use these buttons to navigate the menus, in other words: to browse menus or select parameters. If there are more than two menu lines, use the *Up*/*Down* buttons to change to the next line in the display or to move back to the previous line.

For more information see: *Operating Noah*: *Navigating the Menus*

8 - Enter Button

In various menus – e.g. when writing presets –the *Enter* button is used to confirm the entry of a value or a request. You can also use it to open sub-menus.

9 - Control Button

Press the *Control* Button (9) to switch to *Control* Mode. Now you can use the Continuous Controllers as Performance Controllers to manipulate selected parameters "live". When *Control* Mode is active, the LED above the button is lit (red).

For more information see: *Operating Noah*: *Performance Controllers*

10 - Exit Button

Use the Exit button to abort an action, or to move back up through the menu system.

> For more information see: *Operating Noah*: *Navigating the Menus*

11 - Control Wheel

This control changes the value of a selected parameter and/or preset.

The Control Wheel can also be used to select alphanumerical characters when naming files, e.g. presets.

> For more information see: *Operating Noah*: *Navigating the Menus* and *Presets*

12 - Compact Flash Slot

Accepts a standard memory card in the Compact Flash format on which you can store presets, devices, or other data.

> For more information see: *Operating Noah*: *Navigating the Menus* and *Presets*

13 - Mode Button

Switches between operating modes *Single* and *Multi*.

> For more information see: *Overview of the Noah Architecture* and *Operating Noah: Single Mode and Multi Mode*

14 - System Button

Provides access to the *System* menu.

In this menu you will find parameters for the global configuration of Noah as well as for MIDI Echo, MIDI Clock and managing the MIDI Controllers.

> For more information see: *The System Menu*

15 - Utility

Use this button to open a menu with various functions e.g. for managing presets.

> For more information see: *Operating Noah*: *Presets*, *The Utility Menu*

16 - Edit Button

Puts the unit into Edit mode. When *Edit* Mode is active, the LED below the button is lit.

> For more information see: *Edit Mode*

17 - External Button

In several menus, you can use this button to route Read/ Write functions to the Compact Flash Card. When this Routing is active, the LED below the button is lit (red).

18 - FX Bypass Button

Pressing the *FX Bypass* Button (18) temporarily mutes effects. Keeping the button pressed will open a menu where you can specify in more detail which effects will be muted or not when the *FX Bypass* Button (18) is pressed.

When Bypass is active, the LED below the button is lit (red).

> For more information see*: Edit Mode: The FX Bypass Menu – Muting Effects*

19 - Compare Button

Lets you toggle between the current and the previously stored settings of the selected preset.

> For more information see: *Operating Noah: Presets*

20 - Write Button

Opens the menu pages for storing presets.

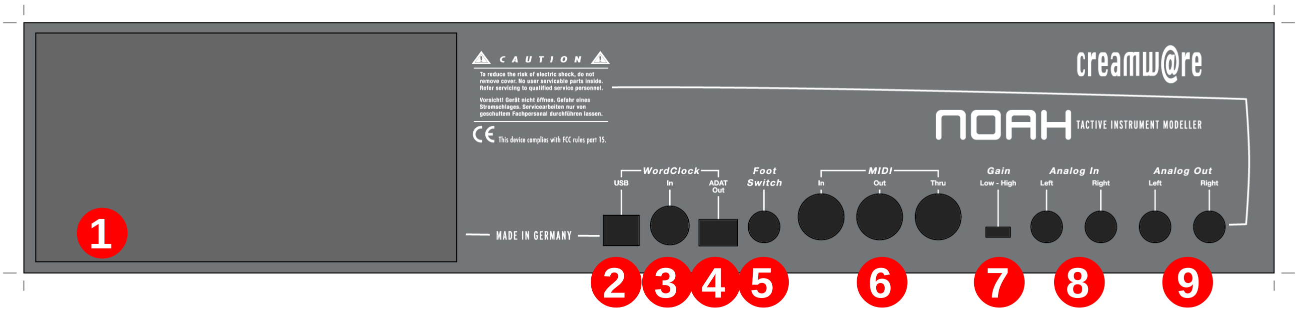
> For more information see: *Operating Noah: Presets*

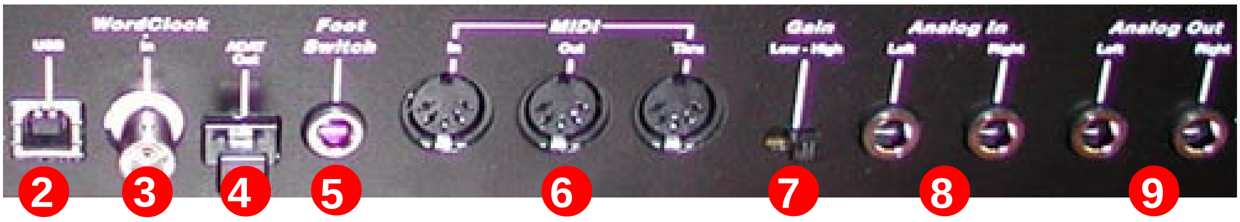
21 - Power Button

Press the button to switch the Noah unit on or off. To start up Noah, press the button once briefly. Keeping the button pressed for a few seconds will switch off the unit.

> Even when switched off, Noah consumes a certain, low amount of power (standby mode). For transport purposes or if you want to switch off the unit for a longer time, you must use the Main Power Switch (1) on the back.

Back Panel





1 - Main Power Switch

This is the main power switch. Use it to switch the Noah unit off if you are not using Noah for a longer time or for transport purposes.

2 - USB Port

Use the USB port to connect a computer from which you can configure Noah easily and conveniently. The USB port can also transmit a USB audio signal between the computer and Noah.

3 - Word Clock Input

Connect a word clock signal from an external device to this BNC terminal.

4 - ADAT Output

Use this optical output to transmit 8 independent audio channels digitally to an external device with a compatible ADAT input.

5 - Footswitch

Currently, the foot switch is not supported.

6 MIDI In / Out / Thru

These jacks connect Noah to a MIDI keyboard, sequencer, or one other MIDI device. Noah receives MIDI data at the MIDI In jack and transmits data through MIDI Out. The incoming MIDI data is passed through to the Thru jack for transmission to other MIDI devices.

7 - Gain Switch

Switches the analog input sensitivity between -10 dB (Low) and -24 dB (High).

8 - Analog In Left / Right

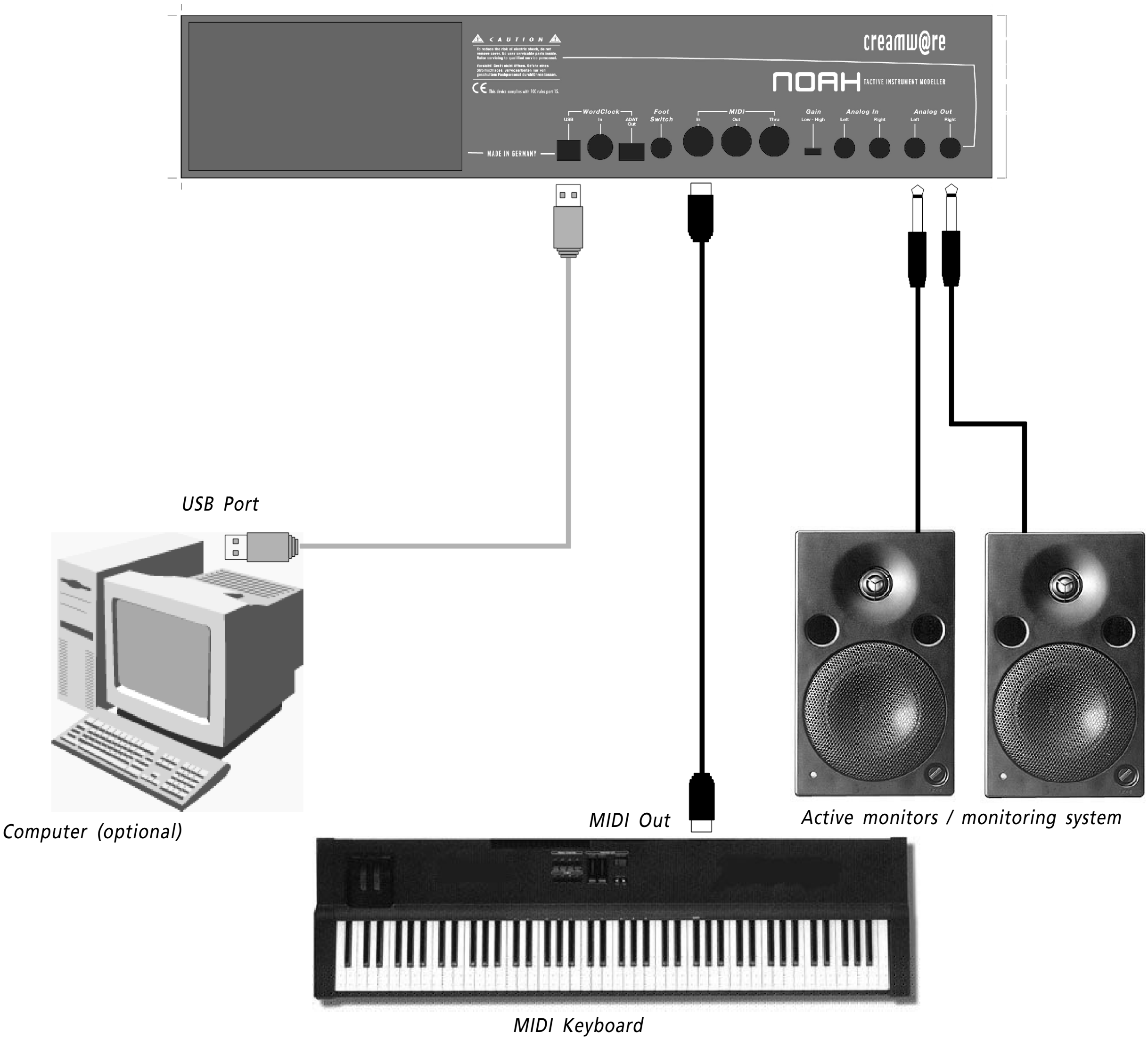
Use these 6.35mm jacks to connect external audio eignals for processing by Noah.

9 - Analog Out Left / Right

The analog outs connect Noah to your audio amplifier or monitoring system.

Connections

The accompanying diagram shows the most important connections to be made to other devices before starting to use Noah.



Overview of the Noah Architecture

Noah processes all digital audio data with 6 special DSPs (Digital Signal Processors) while Noah EX has 10 DSPs. In each system an additional DSP is dedicated to the operating system. The DSPs are reserved for different tasks. Thus, some restrictions arise from the Slot model. Here the term Slot is equivalent to a block of DSPs reserved for a task, such as implementing one of the Noah instruments. Noah's great flexibility lies in the fact that you can load these slots with arbitrary Noah instruments or effects.

All audio data are computed at a single sampling frequency of 44.1 kHz. Other sampling frequencies such as 48 kHz or 96 kHz are not possible.

Noah's Slots

Instrument Slots

The basic version of Noah provides two slots (Slot 1 & 2) for Noah insturments. Each slot provides an equal amount of DSP power. You can either load individual instruments into each slot (Multi operating mode) or one instrument into both slots to extend the number of voices available to the instrument (Single operating mode).

Noah EX provides 4 slots (Slot 1-4), which likewise can all be used for a single instrument (in *Single* mode), with correspondingly higher polyphony, or for different instruments (in *Multi* mode). In Multi mode, slots can be combined freely as desired – each instrument can be loaded using one, two, three or four slots, with corresponding increases in voice count. An instrument loaded into a particular slot occupies all subsequent slots up until the next slot into which an instrument has been loaded. The accompanying diagram shows some of the combinations which are possible.

Each of the instrument slots can be addressed individually through one of the two physical MIDI inputs (MIDI In, USB MIDI In) or by the onboard step sequencer or arpeggiator.

Mixer and Effects

A portion of the DSP resources is reserved for the internal mixer and effects. In principle the Chorus, Delay and Reverb Aux effects are available at all times in the mixer's AUX paths to process signals from either the instrument slots or external sources. With the delay you can select from several versions. The selected version will be loaded into a special Delay slot within the Aux Effects group.

In addition, two Insert effects can be loaded either into a particular mixer channel to process the signal from a specific instrument slot, or into the master channel to process the overall mix.

Because you can load insert effects into the channels reserved for analog input and USB audio, you can also use Noah as an effects processor for external signals.

Slot allocations

The gray box indicates the number of slots occupied by a single instrument.

Noah

Single Mode

[Slot 1 Slot 2]

(1 instrument with double the DSP performance)

Multi Mode

[Slot 1] [Slot 2]

(2 instruments with basic DSP performance)

Noah EX

Single Mode

[Slot 1 Slot 2 Slot 3 Slot 4]

(1 instrument with quadruple DSP performance)

Multi Mode

Different instruments can be loaded into one or more slots each. A few examples:

[Slot 1] [Slot 2] [Slot 3] [Slot 4]

(4 instruments with basic DSP performance)

[Slot 1 Slot 2] [Slot 3] [Slot 4]

(3 instruments, one with double DSP performance, and two with basic DSP performance)

[Slot 1 Slot 2] [Slot 3 Slot 4]

(2 insturments, each with double DSP performance)

[Slot 1] [Slot 2 Slot 3 Slot 4]

(2 instruments with single and triple DSP performance)

Number of voices in Noah instruments

As a result of Noah's flexible architecture, the voice count (polyphony) of a particular Noah instrument depends solely upon the complexity of the algorithms used to realize it. In contrast to some other manufacturers, our top priority is sound quality, and we allow no compromises to be made in this area. As a consequence of the relatively complex and computation-intensive algorithms which this necessitates, the polyphony of Noah instruments is sometimes lower than that offered by competing products of inferior sound quality.

The following overview lists the voice count for all Noah instruments.

Device	1 Slot	2 Slot	3 Slot***	4 Slot***	
MINIMAX	3	6	10	13	
LightWave	6	12	16	16	respectively 2 x 12 = 24 *
Pro One	2	5	8	11	
SixString	2	6	9	12	
Vectron P.	3	7	10	14	
B-2003	full	full	full	full	**
Interpole	-	-	-	-	
Vocodizer	-	-	-	-	

** for maximum polyphony - The limitation to 16 voices in the 4 slot configuration is due to the MidiVoiceControl, which can only manage amaximum of 16 voices.*

*** full polyphony for each slot, meaning up to 4 B-2003s can be played with full poyphony.*

****Noah EX only*

Triggering the Instruments

Noah's instruments can be controlled by external MIDI signals arriving at the MIDI or USB ports (from a master keyboard or sequencer, for example), or by Noah's internal Arpeggiators or Step Sequencers. Altogether Noah provides four independent arpeggiators and step sequencers you can use to control up to four (in the EX version) loaded instruments. You can also use the arpeggiators and step sequencers to control external MIDI devices.

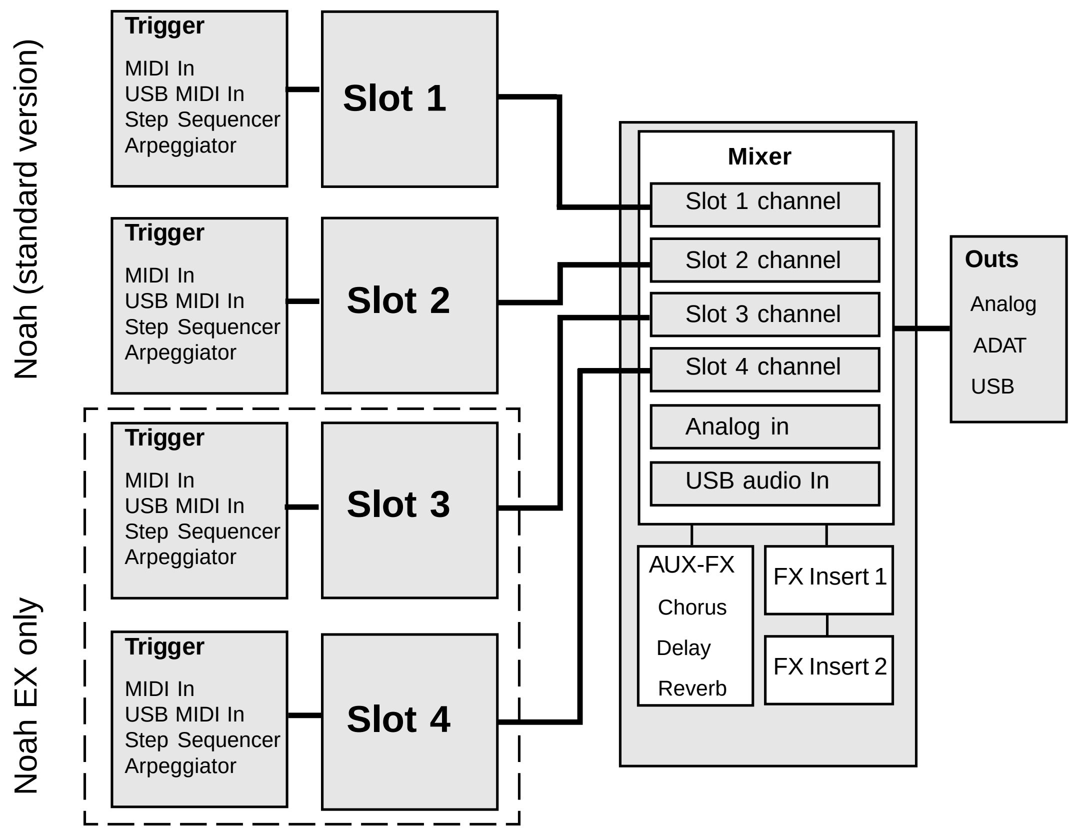
Input and Output

Noah exchanges audio and MIDI data with your other studio gear over several inputs and outputs, and with your computer over the USB interface. Audio can be received over the stereo analog inputs, or the two USB audio channels. You can output 6 audio channels over the USB connection to the computer, or you can use the stereo analog outputs or the ADAT port (8 channels over an optical cable).

MIDI can be transmitted or received over both the MIDI In/Out connections or the USB port. Also, MIDI can be passed directly from the MIDI input to the MIDI Through connection or the USB MIDI output.

For word clock sync to your other studio gear Noah provides an external BNC word clock input connector.

Architecture Block Diagram



Block diagram of the Noah architecture

Operating Noah

In this chapter you will learn about the concept behind Noah's Front panel, how to operate Noah and how to navigate the different menus. It also explains the most important features of the Main Menu, such as selecting an instrument or a preset, which can be accessed without having to change into *Edit* Mode, which will be described separately in a different section.

Starting up Noah

To start up Noah, press the Power button (21) once briefly.
In the display you will see

```
Starting up
Loading operating system
```

Noah will now load its start-up configuration, and on the display you will see:

```
Starting Noah
Loading Performance
```

To switch off Noah, press the Power button (21) and keep it pressed for a few seconds.

The Start Menu

After the operating system and the start-up configuration have been loaded, you will see the following start menu:

```
Starting up
Loading operating system
```

By default, Noah will start up in *Single* Mode (See: Overview of the Noah Architecture) and load the *Minimax* instrument.

At this point, the MIDI channel for the loaded instrument is allocated to Channel 1. If you now set your Master Keyboard to send MIDI messages on this channel and start playing, you should already hear the sound via Noah's analog output or the headphone output.

Preset Bank
This shows the current preset bank. The display will show the bank's number followed by the bank's name. A preceding letter "I" indicates that a preset file from Noah's internal memory has been loaded.

For more detail see the corresponding sections: *Presets: Preset Structure, Selecting Presets* and *Selecting a Bank* in this chapter.

Preset
This shows the currently loaded preset. Unless the cursor is positioned on the bank, use the Control Wheel (11) or the +/- Buttons (6) to navigate the list of presets. Presets are immediately changed by turning the Control Wheel or pressing the +/- Buttons, so there is no need to confirm your selection.

For more detail see the corresponding sections: *Presets: Loading Presets* in this chapter.

Asterisk
When you change parameters, the asterisk indicates that the changes of a loaded preset have not yet been saved. Once you have stored the preset, the asterisk will disappear.

Operation Mode
Single: This is the current operation mode (*Single* or *Multi*). To change the mode, press the associated Continuous Controller (5) or the *Mode* Button (13).

For more information see*: Single Mode and Multi Mode.*

Instrument
When in *Single* Mode, this will show the currently loaded instrument. Turn the associated Continuous Controller (5) to browse the list of available instruments. Pressing the Continuous Controller (5) will load the displayed instrument and replace the current instrument.

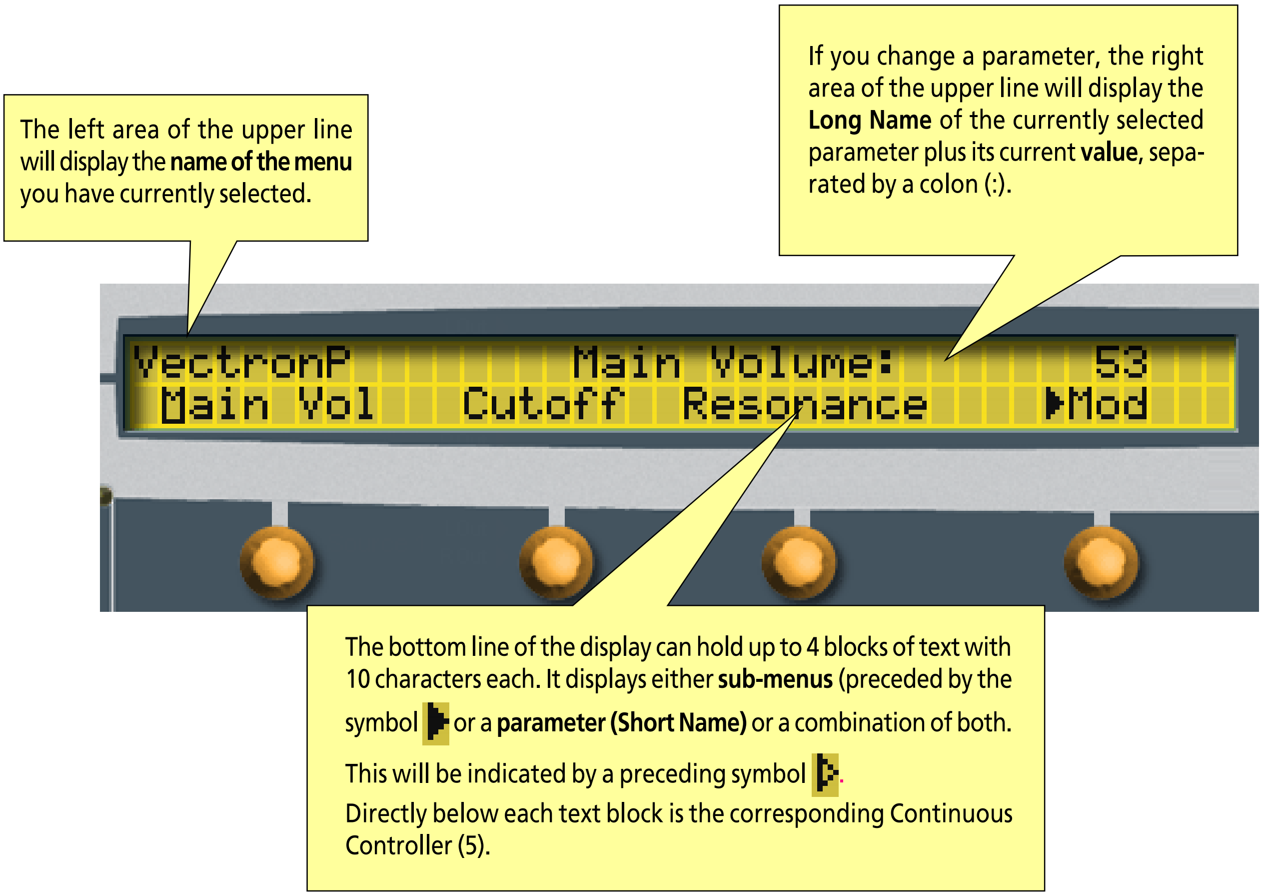
For more information see*: Loading Instruments*

Volume
This shows the current volume level. To change the volume, turn the corresponding Continuous Controller (5).
This parameter corresponds to the Master Volume of the Internal Mixer.

Navigating the menus

General Menu Structure

The display consists of two lines with 40 characters each. Most menus have the same structure, especially in *Edit* Mode, which can be activated or deactivated by pressing the *Edit* Button (16):



Opening sub-menus

To open a sub-menu, press the Continuous Controller (5) below the corresponding sub-menu once briefly.

Alternatively, you can select sub-menus with the Arrow **Right/Left** Buttons (7) and then open the menus using the **Enter** Button.

Changing parameters

To change a parameter, turn the Continuous Controller (5) below the corresponding parameter. While you are turning the dial, the upper display line will show the "Long Name" of the selected parameter. This will often be different from the much shorter "Short Name" (ten characters max.) in the bottom line, as it can contain longer strings. Behind the "Long Name" you can see the current value of the selected parameter. While you are changing parameters, the display will also show the current value in the bottom line instead of the parameter name (Short Name).

Alternatively, you can select individual parameters with the Arrow **Right/Left** Buttons (7) and then change the parameters using the **+/-** Buttons (6) or the Control Wheel (11).

Pressing a Continuous Controller (5) will reset the corresponding parameter to a default value. If there are only two values to choose from, turning the control will toggle between the two values.

You can find a list of all parameters including their default values in the supplementary *Technical References* manual.

Combined sub-menus and parameters

In some menus you will see the symbol preceding a text field in the bottom display line. This indicates that the text field represents both a parameter and a sub-menu and that you can either change the current value by turning the Continuous Controller (5), or open a sub-menu by pressing the Continuous Controller (5).

Example: Turning the Continuous Controller (5) in the Trigger Menu for instruments (Edit/MIDI/Devices/Slot1/Trigger1/) will let you determine the Trigger Source for the instrument, e.g. an arpeggiator or a step sequencer. Once you have selected an arpeggiator, press the Continuous Controller to open the corresponding arpeggiator sub-menu.

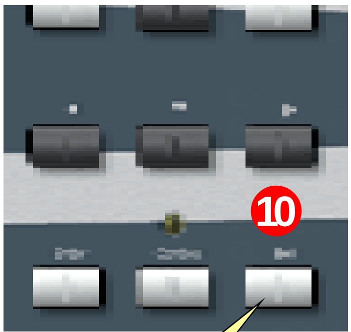
: In this example, **turn** the Continuous Controller (5) to select an Effect. **Press** the Continuous Controller to open the parameter Menu of the selected Effect.

Menus with more than 4 sub-menus or parameters

If a menu contains more than the 4 sub-menus or parameters, which is the maximum number that can be displayed in one line, the symbol will tell you so. Use the Arrow Down button (7) to move to the line "below", where you will find more sub-menus and/or parameters. You will now see the symbol at the end of the line, to remind you of the existence of the sub-menus and/or parameters in the line "above". Occasionally the symbol will be displayed. This means there are even more sub-menus and/or parameters in yet another line "below".

Leaving a menu

Use the *Exit* button to leave the current menu. This will take you back to the level above the current one.



Press the *Exit* Button (10) to leave the menu and move up one level..

Examples

The following systematic step-by-step examples should help you to get better acquainted with the most basic menu functions.

Example 1: We want to load the Organ B-2003 with the preset *P005** and set it to MIDI channel 4. We then want to transpose it up three semitones.

**** The precise name of this preset was not yet available when this manual went to print.***

Selecting and loading instruments

In the Start Menu, turn the Continuous Controller (5) below the instrument display (Default: > Minimax <) until *B-2003* appears on the display.

To actually load a new instrument and replace the current instrument, you must first confirm your selection by pressing the Continuous Controller below the instrument name.

After a few seconds, the display will show the name of the current instrument (in our example: B-2003).

Selecting sounds

Use the -/+ Buttons (6) or the Control wheel to browse the list of presets and to make your selection. The display always shows the name of the currently active preset. This means that you need not confirm your selection.

In our example, we wanted sound P005. The fastest way to select it would be by turning the Continuous Controller until P005 appears on the display. That is all you have to do.

Changing MIDI channel / Edit Mode

To change the MIDI channel of the B-2003, we must first switch to *Edit* Mode, which will make a wide range of parameters available to you.

First press the *Edit* button (16). The LED below will light up to indicate that *Edit* Mode has been activated. At the same time, the display will change and you will now see:

```
Edit
>B-2003     >MIDI      >Mixer     >FX
```

A small arrow in front of the items in the bottom menu line indicates sub-menus (as opposed to editable parameters).

To open a sub-menu, press the Continuous Controller (5) below the desired menu item.

The MIDI channel is set in the MIDI Menu. Press the Continuous Controller (5) below ">MIDI". You are now in the MIDI sub-menu. The display should now show the following:

```
MIDI
>B-2003     >Mixer     >Clock
```

The MIDI channel of the B-2003 is not a global parameter of Noah but is associated with a specific instrument (Device). It is therefore part of the B-2003 sub-menu. Open this sub-menu by pressing the Continuous Controller (5) once more. The display now reads:

```
B-2003
 >Trigger   >Port/Ch    >Zones
```

Now open the sub-menu *Port/Ch.*

```
Port/Channel
  InPort    Channel    OutPort
```

As you can see, this menu contains editable parameters only. The items in the bottom line are not preceded by a small arrow because that would indicate sub-menus.

In general, parameters are displayed in the bottom line in groups of 4 items at the most. Each item can consist of up to 8 characters (Short Name). Below each parameter you can find the corresponding Continuous Controller (5). When you change a parameter by turning the Continuous Controller, you can see the Long Name of the parameter in the right half of the upper line of the display. The parameter name is followed by the current parameter value, which also appears in the bottom line instead of the parameter's Short Name.

The MIDI channel is represented by Parameter *Channel*. Turn the second Continuous Controller (5). The upper line will now display the name *Channel*. For this example, set the value to 4.

Transposing / Leaving a menu

To leave a menu after having changed a parameter, press the *Edit* Button (16) again. This will take you right back to the Start Menu. Use the *Exit* button (10) if you prefer to go back one level.

In our example, the parameter required for transposing is "close at hand" and you can use the *Exit* button (10) to leave the *PortCh (Port/Channel)* Menu and return to the B-2003 Menu.

Now open the sub-menu *Trigger*. You should see the following display:

```
Trigger
 >Direct    TrigOut     Transp
```

Set the parameter Transp1 to "3" by turning the corresponding Continuous Controller (5) clockwise. This will transpose the B-2003 3 semitones up.

Now leave the *Edit* Mode with the *Edit* Button (16). You will be asked if you want to permanently store the changes in the preset. Select No by pressing the Continuous Controller (5) under (No). The LED below is now inactive and you are back in the Start Menu.

Example 2: We want to switch to *Multi* Mode in order to load two instruments simultaneously. We also want to set the volume level for both instruments.

Switch to *Multi* Mode by pressing the *Mode* Button (13). The display will look like this:

```
Bank               Preset:
Multi
```

Selecting instruments

To select instruments for the two (Noah) respectively four instrument slots (Noah EX), press the *Edit* Button (16) to switch to *Edit* Mode.

```
Edit
  >Slots    >MIDI     >Mixer     >Fx
```

The instruments are selected in the *Slots* sub-menu. You must therefore press the first Continuous Controller (5) on the very left.

```
Slots
>Instr.1  >Instr.2 < >Instr.3  >Instr.4>
```

Turning the first Continuous Controller will select the instrument for slot 1. Turning the second Continuous Controller will select the instrument for slot 2. For each slot, select an instrument of your choice, using the corresponding controls.

If you want to select a preset for an instrument, you must first position the cursor on the respective instrument in the bottom display line by using the Arrow Up/Down/Left/Right Buttons (7). Next, use the Control Wheel (11) to select the preset. The preset name is shown in the upper display line.

To edit the instrument itself, press the corresponding Continuous Controller (5). A symbol ▷ will remind you that this feature is available.

By default, each instrument in *Multi* Mode is routed to the MIDI channel corresponding to the instrument's slot number. That is, the instrument in slot 1 is routed to MIDI channel 1; the instrument in slot 2 is routed to MIDI channel 2. Set your Master keyboard to transmit on MIDI channel 1 to play the first instrument; switch to channel 2 to play the second instrument.

Adjusting volume

Leave the menu and move up one level by pressing the *Exit* Button (10).

Then open the *Mixer* sub-menu. You will see the following display:

```
Mixer
  >Slots  >Anlg/USB   Master  >MultiView
```

Open the *MultiView* sub-menu. This menu allows you to change several instruments simultaneously.

```
    Slot1     Slot2     Slot3     Slot4
Vol:127   Vol:127   Vol:127   Vol:127   >
```

For each instrument, the display shows four columns of important parameters from Noah's internal Mixer. The parameter at the top of the list is Volume (Vol). Use the **Arrow Down** Button (7) to access the other parameters.

Use the first two Continuous Controllers (5) to adjust the volume of the two instruments

Leave the *Edit* Mode by pressing the **Edit** Button (16). You will be asked if you want to save the changes.

Select No by pressing the associated Continuous Controller (5) under (*No*).

The LED beneath the **Edit** Button will go out and you will be returned to the Start Menu.

Single Mode and Multi Mode

To switch between *Single* and *Multi* Mode, press the *Continuous Controller* (5) under the menu items *Single* or *Multi* in the Start Menu. Alternatively, press the *Mode* Button (13).

(For more information see: Overview of the Noah Architecture.)

The current operating mode is displayed at the very beginning of the bottom line of the Main Menu.

Single Mode is particularly suitable for live performances, where you only need one sound at a time but want to rely on the maximum number of voices available.

Multi Mode allows you to load 2 (Noah) or up to 4 instruments (Noah EX) at the same time. Choose this mode for live performances, where you need more than one sound simultaneously, or when you use Noah as a sound module together with a sequencer.

Loading instruments

In *Sinle* Mode, you can only load one instrument at a time. In the Start Menu, browse the list of available instruments by turning the second Continuous Controller (5) and press it to confirm your selection. This loads the instrument.

In *Multi* Mode, you must first press the *Edit* Button and switch to *Edit* Mode. Now open the Slots Menu. With Noah EX, each of the four Continuous Controllers (5) is assigned to one instrument slot. With Noah, use the first two Controllers.

The number of slots and the number of voices assigned to an instrument depends on the slot into which the next instrument is loaded. This should be particularly taken into account with Noah EX, where each instrument can occupy a different slot.

For example, if you load the first instrument into slot 1 and a second instrument into slot 3, both instruments will occupy two slots. If you now load a third instrument into slot 4, the first instrument will still occupy two slots (slots 1 and 2) whereas the second and third instrument will occupy only one slot each. If, on the other hand, one instrument is loaded into slot 1 and another into slot 2, the first instrument will occupy one slot (slot 1), the other instrument three. (Slots 2 – 4).

(For more information see: *Overview of the Noah Architecture.*)

If you switch between *Single* and *Multi* Mode, the currently loaded instrument in slot 1 is preserved.

Presets

Noah comes with a wide range of ready-to-use presets for instruments, effects and other components, but of course it also allows you to create and store your own presets.

Preset structure

Presets can comprise different layers within the Noah architecture.

Presets are physically stored in preset files. In Noah's internal memory, separate preset files exist for each of the following structures:

a) The *Multi* configuration

The preset file for the Multi configuration contains all settings for a Multi Mode configuration, including the settings for all its components. This allows you to conveniently switch between two completely different configurations.

b) The *Single* configuration (Instruments)

For each instrument (as well as the Interpole and Vocodizer Effects), there is separate preset file. This contains all settings of a Single Mode configuration – in other words, all the settings for the instrument itself, the Mixer, the Aux and Insert Effects, the Arpeggiator (if used) and the Step Sequencer.

c) The Aux Effects

The Aux Effects Chorus und Reverb as well as the separate Delay types also come with a preset file of their own.

d) The Insert Effects

There is one separate preset file for each Insert Effect. This allows you to load ready-made settings for these Effects, which you can then store as part of a Single or Multi configuration.

e) The Arpeggiator and the Step Sequencer

Both these modules have their own preset file. The four possible instances of these modules, however, are stored within a shared file.

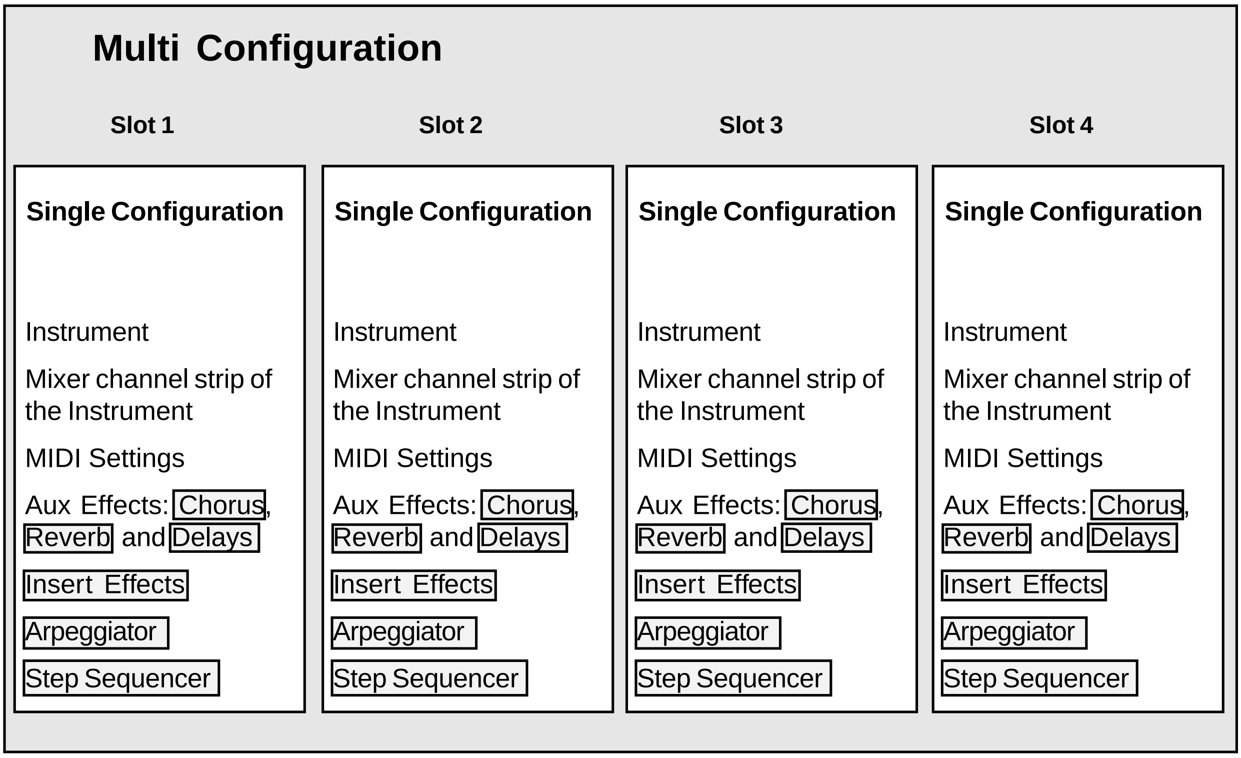
f) Partial presets of the instruments

A few instruments also have partial presets for certain substructures, such as for the string sets of the Six String or for the filter matrix of the Vocodizer. These partial presets can only be loaded but you cannot save any changes.

Storing a preset for a Single or Multi Mode configuration does not change the presets for its individual components. Changing the preset for an instrument or effect does not change the preset for the Single or Multi Mode configuration, either. In other words, a preset for a Single or Multi Mode configuration is an independent file, and does not reference the presets of its components.

In *Multi* Mode, you can load several *Single* presets. The settings of these Single presets are used for the respective instrument, the corresponding Mixer channel strip, and the slot of the MIDI editor. The settings for the Mixer channel strips *Analog In* and *USB In* will not be used.

Usually you will want to load all the presets for a complete *Single* or *Multi* configuration, which already contain the settings for the effects, the Arpeggiator or the Step Sequencer. However, as all these modules come with their own presets, you can easily load the various ready-made settings of these modules and then save them as part of a *Multi* or *Single* configuration.



Noah contains preset files for the various structures - from the comprehensive Multi configuration (which may contain several Single configurations) down to individual effects. Every box in the illustration represents a preset file.

Selecting a preset file – internal and external presets on the Compact Flash Card

Noah's internal memory contains one preset file for each of the structures described above. In addition to these internal presets, you can load additional preset files for any device from a Compact Flash Card. You can also store presets on the Compact Flash Card.

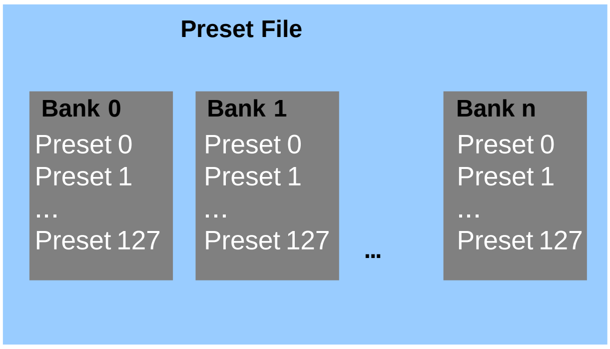
To access preset files stored on the Compact Flash Card, press the External Button (17). The Compact Flash Card must be inserted. Noah will now search the Compact Flash Card for a matching preset file. If, for example, you have loaded the Minimax while in Single Mode, Noah will look for a preset file for the Minimax while you are in the start menu.

When Noah has found a matching preset on the Compact Flash Card, the ”I” (Internal) in the upper left corner of the Start Menu will change to ”E” (External).

The option of choosing between presets from the internal memory or the Compact Flash Card is available in all menus that give you access to substructures.

Selecting a bank

Presets are organized into banks. A bank is like a ”sub-folder” of a preset file that contains the individual settings. A preset file can contain up to 128 banks.



The number of the current bank (000-127) is displayed In the upper left corner of the Start Menu following the letters ”I” (Internal) or ”E” (External).

To switch to another bank, use the Arrow Buttons **Up/Down/Right/Left** to position the cursor on the number of the bank. If the preset file contains more than one bank, you can now select a new bank by using the Control Wheel (11) or the +/- Buttons.

The preset file for Effects, Arpeggiators or Step Sequencers can contain only one bank.

Loading presets

You can select presets from the following menus.

The upper display line generally shows a number and the name of the current preset. Use the Control Wheel (11) or the +/- Buttons to browse the list of presets – that is of course, unless the cursor is positioned on the bank. You can instantly toggle between presets without having to confirm your selection.

Presets for a complete Single configuration

In *Single* Mode, the upper right half of the top line of the Start Menu will show the current preset for a Single configuration. Select the preset as described above.

```
I000:Bank Name  P000:Preset Name
Single  >  Minimax   <     Volume 100
```

Presets for a complete Multi configuration

In *Multi* Mode, the upper right half of the top display line of the Start Menu will show the current preset for a *Multi* configuration. Select the preset as described above.

```
I000:Bank Name  P000:Preset Name
Multi                      Volume 100
```

Presets for individual instruments of the *Multi* configuration

You select presets for instruments in *Edit* Mode. Press the *Edit* Button (16) and open the Slots Menu, which is shown at the beginning of the bottom display line.

Position the cursor on the desired instrument in the bottom display line, using the Arrow Buttons **Up/Down/Right/Left** (7). The upper display line will now show the current bank and the current preset for the instrument. Change the preset as described above.

```
Slots          IOOO P000:Preset Name
 >Minimax >Vocodizer
```

Presets for Aux Effects

Press the *Edit* Button (16) to switch to *Edit* Mode. Now open the FX Menu and from here go to the Aux FX sub-menu. Position the cursor on the desired Aux effect (Chorus, Delay, Reverb) using the Arrow Buttons.

The upper display line will now show the current preset of this effect. Change the preset as described above.

```
Aux FX           POOO:Preset Name
>Chorus   >Delay    >Reverb
```

Presets for insert effects

To load a preset for a certain effect, switch to *Edit* Mode using the *Edit* Button (16). First open the FX Menu and then the sub-menu of the respective Insert Slot (Insert 1, Insert 2). The upper display line will now show the current preset for the currently loaded effect. The effect itself is displayed in the bottom display line at second position. Change the preset as described above.

```
Insert 1         POOO:Preset Name
>Slot1    >Insert-Effect
```

Presets for the Arpeggiator or the Step Sequencer

The presets for the Arpeggiator or the Step Sequencer are displayed in the upper display line of the Parameter Menu for these modules. In *Single* Mode, you can access these parameters by opening this menu from Edit / MIDI / *Instrument* / Trigger / Arpeg4 respectively StepSEQ. In *Multi* Mode, open the menu from Edit / MIDI / Devices / Slot1-4 /Trigger /Arpeg respectively *StepSEQ*.

(For more information, see the descriptions of the Arpeggiator or the Step Sequencer.)

```
Arpeggiator       P000:Preset        Name
 >Control   >Scan    >Capture   >Note  >
```

Previewing presets without keyboard

Even without a connected MIDI keyboard, you can listen to sounds or presets by using the volume control of the headphone output. Depending on the sound category of the respective preset, you will hear a note, an accord or a brief sequence, which will give you a first impression of the preset and its possibilities. (These sound categories are only shown in the preset dialog box of the Noah Remote Software.)

In *Single* Mode, the currently selected preset for an instrument will be played. In *Multi* Mode, you can select the respective instrument whose preset you want to preview. Choose *Edit* Mode, and select the instrument from the Slots Menu, using the Arrow Buttons to position the cursor on the desired instrument.

If you have already made any changes to the preset settings, these changes will be taken into account, i.e. the sound will always be played with the current settings.

Storing presets

An asterisk in the upper right corner of the Start Menu will indicate that parameters in a preset have been changed, but the preset has not yet been saved.

If you have changed the settings for an instrument, an effect or any other component in the *Edit* Mode, you will be asked if you want to store the new settings or not before leaving the *Edit* Mode (by pressing the *Edit* Button (16) once more).

Selecting *Yes* will open the *Write Preset of.*. Menu. You may switch to this menu anytime by pressing the Write Button (20). The following procedure will be the same, though.

Selecting a preset file

In the bottom display line of the menu, you can specify the preset file to which you want to save the current settings by pressing the corresponding *Continuous Controller* (5).

Display in *Single* Mode:

```
Write Preset of:
 >Instr.   >Inserts   >Aux   >Arp/Seq
```

Display in *Multi* Mode:

```
Write Preset of:
  >Multi   >Slots   >Effects  >Arp/Seq
```

This means that you first specify which parts of the total current parameter settings you want to save.

The separate preset files are physically independent files and do not refer to each other. (For more information see: Preset Structure.) Therefore, your choice will determine if the current settings of the Single or Multi configuration will be restored when loading settings again.

Usually you will save the complete *Single* respectively *Multi* configuration, which already contains all effect settings and the parameters for the Arpeggiator and Step Sequencer. In this case, all the current settings will be restored when next loading the *Single* respectively *Multi* configuration.

Example: You have changed an *Insert* effect in *Multi* Mode. You could now save the changes with the preset for the Insert effect, with the preset of the *Single* configuration that contains the instrument or with the preset for the *Multi configuration.*

Save the changed settings with the preset for the *Multi* configuration if you want them to be available as part of that particular *Multi* configuration only. Save the changed settings as part of the *Single* configuration if you want the current settings plus the changed parameters of the effect to be available in *Single* Mode as well. Save them as an effect preset if you want to use them for another instrument as well.

Of course, you can save *all* these different presets one by one. Alternatively, you may decide to save the changes only as part of a *Single* or *Multi* configuration and then later save the effect settings as a separate effect preset.

In *Single* Mode, open the desired sub-menu (*Instrument*, Inserts, Aux, Arp/Seq). In *Multi* Mode, open the sub-menu Multi, Slots, Effects, Arp/Seq. Then select the desired sub-menu/instrument or the desired effect or either the Arpeggiator or Step Sequencer module.

The upper display line will now read ”Write preset to:” In the bottom line, specify to which preset the changes shall be saved.

Write preset to:
I000:*Bankname* P000: *Presetname*

As explained above, you can use the *External* Button (17) to choose a suitable preset file on the Compact Flash Card, which must already exist, though.

Select the bank (not possible with effects, Arpeggiator or Step Sequencer) and the number of the preset.

Confirm by pressing the **Enter** Button (8).

Naming presets

In the pop-up menu *Enter preset name* you can now enter a name for the preset.

You can change this name using the Control wheel (11), which allows you to move back and forth through a set of characters. The cursor – a blinking box – will at this point be positioned on the first character of the name. Choose the desired character and use the Arrow Right Button to move the cursor to the next position. You can use the Arrow Right/Left Buttons to move the cursor back and forth to any desired position within the name and make any necessary changes. A name can contain up to 18 characters.

Sequence of characters:

A, B, C, D, E, F, G, H, I, J, K, L, M, N, O, P, Q, R, S, T, U, V, W, X, Y, Z, [, Y#,], ^, _, \, a, b, c, d, e, f, g, h, i, j, k, l, m, n, o, p, q, r, s, t, u, v, w, x, y, z,{, |, }, →, (blank), !, ", #, $, %, &, ‘, (,), *, +, ,, -, ., /, 0, 1, 2, 3, 4, 5, 6, 7, 8, 9, :, ;, <, =, >, ?, @

The characters are arranged cyclically, i.e. once you have reached the end of the list, you are automatically taken back to its beginning.

Press the Enter Button (8) one more time to store the preset and leave the **Write** dialog.

Depending on the number of stored presets and the number of banks, you might see a warning at this point that there is not sufficient space for storing further presets. In this case, try to optimize the memory settings in the *System/Memory* Menu. If that does not help, you can either delete some presets or store presets on the Compact Flash Card.

Compare Button

If you have changed any settings since you last loaded a preset from the *Single* or *Multi configuration,* you can use the Compare Button to switch between the current and the previous settings.

For example, you can load a preset as a starting point, change its settings and compare them with the original preset to decide if you want to keep the changes or not.

Managing presets

You can find the necessary commands for adding banks or deleting complete banks or individual presets in the *Utility* sub-menu, which you can access with the Utility Button (15). From here, open the *Presets* sub-menu.

If you want to access files on the Compact Flash Card, you must first change to the Compact Flash Card by using the **External Button** (17) before opening the *Utility* Menu.

You can only edit preset files of currently loaded instruments/modules or presets that already exist on the Compact Flash Card.

```
Edit Presetlist of >                 <
>Add Bank >Del Bank >Ren Bank >Del Pres
```

Selecting a preset file

In the *Preset* Menu, use the Control Wheel (11) to select a preset file in the upper display line. The list will contain all preset files that can currently be used and will therefore change according to which operation mode you are using (*Single* or *Multi*), and which instruments, effects or other modules such as the Arpeggiator or the Step Sequencer are currently loaded.

Creating a bank – the Add Bank sub-menu

```
Add Bank                 <              >
 (Write)    (Back)    (Exit)
```

In this menu, you can add a bank to the currently selected preset file.

Naming a bank

The name of the bank is displayed in the right half of the upper display line. You can now enter a name - in the same ways as you name a preset - using the Control Wheel (11).

To create a bank with the name you just entered, press the Continuous Controller (5) below (Write). To leave the current menu and move back one level, press the Continuous Controller (5) below (Back). You can also leave the complete Utility Menu by pressing the Continuous Controller below (Exit).

Deleting a bank – The sub-menu Del (Del Bank)

```
Del Bank                 <              >
 (Write)    (Back)    (Exit)
```

Select the bank you want to delete using the Control wheel. The name of the currently selected bank will be displayed in the upper right corner. Use the Control wheel to cyclically browse through the list of all the banks within the preset file, which you have selected in the menu one level above.

To delete the selected bank, press the Continuous Controller (5) below (Delete). To leave the current menu and move up one level, press the Continuous Controller (5) below (Back). You can also leave the complete Utility Menu by pressing the Continuous Controller below (Exit).

Renaming a bank – The submenu Ren Bank (Rename Bank)

```
Rename Bank              <              >
 (Write)    (Back)    (Exit)
```

The name of the currently selected bank will be displayed in the upper right corner. This is the bank that contains the currently loaded preset.

You can now edit the name of the bank - in the same way as you name a preset - using the Control Wheel (11).

Press the Continuous Controller (5) below (Write) to save the new name. To leave the current menu and move back one level, press the Continuous Controller (5) below (Back). You can also leave the complete Utility Menu by pressing the Continuous Controller below (Exit).

Deleting a preset – The sub-menu Del Pres (Delete Preset)

Dummy	Menu	
Single	>Device	Volume

To delete a preset within the current bank in the preset file (which you have selected in the menu one level above), select a preset using the Control wheel (11). The name of the currently selected preset will be displayed in the upper right corner. Use the Control wheel to cyclically browse through the list of all the banks within the preset file.

Press the Continuous Controller (5) below (Delete) to delete the selected preset. To leave the current menu and move back one level, press the Continuous Controller (5) below (Back). You can also leave the complete Utility Menu by pressing the Continuous Controller below (Exit).

Changing presets via MIDI

You can also change presets via MIDI by sending MIDI Program Change messages. When you send the appropriate program change message from a keyboard or sequencer, the corresponding preset is load.

The MIDI Program Change number corresponds to the preset number (0 – 127), which precedes the preset name in the display.

You can change the bank via MIDI in the same way.

Performance Controller

You can use the 4 Continuous Controllers (5) not only to change parameters in *Edit* Mode, but also as Performance Controllers to change certain parameters while you are playing live. Up to 16 parameters of your choice can be controlled that way.

The assignment of the Performance Controllers is stored as part of a *Single* or *Multi* configuration preset.

Switching to Control Mode

Before you can use the Continuous Controllers (5) as Performance Controllers, you must first switch to Control Mode by pressing the Control Button (9). In control mode, the LED above the button will be lit.

The bottom display line will show the names of the parameters which can be controlled with the corresponding Continuous Controllers (5).

Please note that only 4 of the 16 parameters to which you have assigned a Performance Controller can be controlled simultaneously, as physically there are only 4 Continuous Controllers (5). You can access the next set of 4 controllable parameters by using the Arrow Down (7) button to move to next display line. The Arrow Up button (7) will take you back to the line above.

If the option for sending MIDI Controller messages in the System Menu is enabled, the corresponding MIDI Controller messages are sent when changing parameters. This allows you to transmit the movements of the Performance Controllers as MIDI events and record them e.g. in your sequencer.

Changing the current allocation

The allocation of parameters of a module or a *Single* or *Multi* configuration to a Performance Controller is part of the currently loaded preset. You are free to edit these allocations, meaning that any parameter can be assigned to any Performance Controller.

With the Control Menu opened, change into *Edit* Mode by pressing the *Edit* Button (16). Alternatively, first select *Edit* Mode and then open the Control Menu. Turn the Continuous Controller (5) to which you want to assign a new value. The display will now show all accessible (i.e. loaded) instruments and modules (MIDI, Mixer, FX). Select the desired device by pressing the Continuous Controller (5) as usual. In the same way, you can now either select parameters or browse existing sub-Menus until you reach the desired parameter set. You can browse the parameters by turning the Continuous Controller (5). Leave the Controller set to the desired parameter and then leave Edit Mode. You will be asked if you want to save the current preset.

The Noah Remote Software

You can configure and control all of Noah's functions from your computer. All you need is the NOAH Remote Software, which provides graphical user interfaces for all instruments and modules. You can also use this software to conveniently transfer all data from Noah's internal Flash ROM or the Compact Flash Card to your computer's hard disk and archive them there.

Because of all these benefits, we strongly recommend you to use the Remote Software to control Noah – that is, if you own a computer, of course. Editing directly from the hardware should really be kept for special occasions, such as Live performances.

When you install the software, a separate manual will be copied to your hard disk in Adobe Acrobat format (PDF). Here you can find out about the many features which the software offers and how to operate the instruments and modules.

You will need the free Acrobat Reader software to read this manual. If you have not already installed it on your computer, you can find it on the Noah Software CD.

Installing the Noah Remote Software

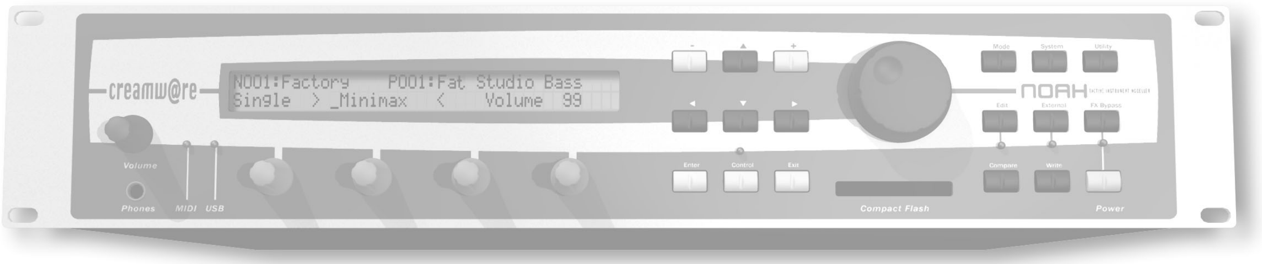
You can find further and much more detailed information about the software and any last minute changes in the documentation provided with your Noah unit. Here we will only describe the general procedure.

1. USB driver installation (PC)

Connect Noah to your computer via a USB cable. Windows should now recognize Noah as a new hardware device and ask for a driver. You can find the driver on the CD provided with your Noah unit.

2. Installing the software

Start the Setup Program (setup.exe) from the CD provided with your Noah unit and follow the instructions.



USER'S MANUAL

REFERENCE PART

Edit Mode: MIDI Menu

MIDI Menu

In addition to the Global MIDI parameters in the System menu (see *System* menu), all MIDI parameters are addressable from this menu system. The various MIDI parameters are distributed throughout three menus (Devices, Mixer, Clock) and their sub-menus. In *Multi Mode,* there is also the **MultiView** menu.

```
MIDI
 >Devices   >Mixer    >Clock
```

Instrument / Devices Menu

Here you'll find all the MIDI settings for the Instruments (Devices). In *Single* Mode, the display will show the name of the Instrument, which you can use to directly access the corresponding slot menu. In *Multi* Mode, you will first see the *Device* menu with sub-menus for the separate slots.

If no instrument is loaded into a slot you can use the MIDI settings to control external MIDI devices connected to Noah's MIDI out port. For example, this allows you to control an external synth with the arpeggiator or step sequencer.

```
Devices
  >Slot 1    >Slot 2    >Slot 3    >Slot 4
```

Slot 1-4 sub-menus

```
Slot1
>Trigger1 >Port/Ch1  >Zones1
```

Trigger 1/2/3/4 sub-menu

```
Trigger1
 >Direct    TrigOut1  Transp1
```

Trigger Source **(Trigger Source) [Direct,Arpeg, StepSEQ]**

Select the MIDI source (MIDI input, Arpeggiator instance of the slot, Step Sequencer instance of the slot) to address the instrument loaded into this slot.

When an Arpeggiator or Step Sequencer is selected as the source, pushing the rotary (5) control directly below the selection opens its control menu.

TrigOut1/2/3/4 (Trigger > Out 1/2/3/4) [On, Off]

If this option is enabled (On) then the MIDI notes generated by the arpeggiator or step sequencer are also sent to the MIDI Out Port (Port/Ch sub-menu).

Transp1/2/3/4 (Transpose 1/2/3/4) [- 127, ... , 127]

Here you can set a transpose value in semi-tones for the instrument loaded into the respective slot.

Port/Ch1/2/3/4 sub-menus (Port/Channel)

```
Port/Channel 1
 InPort1    Channel1  OutPort1
```

InPort1/2/3/4 (In Port 1/2/3/4) [MIDI, USB]

Sets the MIDI port through which the respective slot will receive MIDI data.

Channel1/2/3/4 (Channel 1/2/3/4) [0, ... , 16, omni]

Selects the MIDI channel on which the repective slot transmits and receives MIDI data.

OutPort1/2/3/4 (Out Port 1/2/3/4) [MIDI, USB]

Sets the MIDI Out port through which the respective slot transmits MIDI.

Zones1/2/3/4 sub-menu

```
Zones1
 LowKey1   HighKey1  LowVel1   HighVel1
```

LowKey1/2/3/4 (Low Key 1/2/3/4) [0, ... , 127]

Sets the lowest note value (MIDI note number) of a keyboard zone defined to address a particular instrument. The instrument will not respond to notes below this value.

HighKey1/2/3/4 (High Key 1/2/3/4) [0, ... , 127]

Sets the highest note value (MIDI note number) of the keyboard zone defined to address a particular instrument. The instrument will not respond to notes above this value.

LowVel1/2/3/4 (Low Velocity 1/2/3/4) [1, ... , 127]

Sets the lower limit of a velocity range that the instrument will respond to. The instrument will not respond to values outside this range.

HighVel1/2/3/4 (High Velocity 1/2/3/4) [1, ... , 127]

Sets the upper limit of a velocity range that the instrument will respond to. The instrument will not respond to values outside this range.

Mixer Menu (Mixer/FX)

The mixer and effects can be controlled by MIDI controllers on an assigned MIDI channel.

In the Single operating mode, this channel is the same as assigned to slot #1. In Multi mode, the mixer and effects can be addressed on their own MIDI channel.

To recall a preset of a Multi configuration via MIDI Program Change, the Program Change must likewise be sent on the mixer's MIDI channel. To recall a preset of an individual device *within* the Multi configuration, the Program Change must be sent on the channel assigned to that device.

```
Mixer/FX
  InPort   Channel   OutPort
```

InPort (Mixer/FX In Port) [MIDI, USB]

Selects the MIDI In port for the mixer and effects to receive MIDI data.

Channel (Mixer/FX Channel 1/2/3/4) [1, ... , 16]

Selects the MIDI channel for mixer and effects MIDI transmission and reception.

OutPort (Mixer/FX Out Port) [MIDI, USB]

Selects the MIDI Out port for mixer and effects MIDI transmission.

Clock Menu (MIDI Clock)

This menu contains the parameters to configure the internal MIDI Clock (when using it to synchronize the arpeggiator or step sequencer, for example).

Use the System/MIDI menu to specify the source of the MIDI Clock (Internal, MIDI Input) and whether it should be sent to the MIDI output.

```
MIDI Clock
  Tempo:      BPM      BPM/100
```

Tempo: (display only)

In this menu, you can set the MIDI Clock tempo.

BPM (Beats per minute) [0, ..., 250]

If you have configured the MIDI clock source as *Internal* (System/MIDI menu > ClockSrc), you set the tempo in BPM here. If one of the external sources is configured (MIDI In, USB In), the current tempo is displayed here.

BPM/100 [0, ..., 99]

Here you can fine-tune the tempo in graduations of 1/100ths of a beat per minute (BPM).

MultiView Menu

To provide quick access to the most important MIDI parameters of each loaded instrument in *Multi* mode, the **MultiView** menu shows the menu parameters for **Device 1 - 4** in the form of a table containing parallel entries for each instrument slot.

This enables you to more quickly make changes to multiple instruments (e.g., exchanging MIDI channels) with less navigating around in the submenus of individual instruments.

For each instrument which is loaded (Device 1 - 4), you have access to the following parameters:

Instr1	Instr1	Instr1	Instr1
InP:MIDI	InP:MIDI	InP:MIDI	InP:MIDI>
Ch: 1	Ch: 2	Ch: 3	Ch: 4 >
TS:>Direc	TS:>Direc	TS:>Direc	TS:>Direc>
Trp: 0	Trp: 0	Trp: 0	Trp: 0 >
LKey:C-2	LKey:C-2	LKey:C-2	LKey:C-2 >
HKey:G8	HKey:G8	HKey:G8	HKey:G8 >
LVel: 1	LVel: 1	LVel: 1	LVel: 1 >
HVel:127	HVel:127	HVel:127	HVel:127>
OutP:MIDI	OutP:MIDI	OutP:MIDI	OutP:MIDI>
TOut:Off	TOut:Off	TOut:Off	TOut:Off<

InP (In Port 1/2/3/4) [MIDI, USB]

Sets the MIDI port through which the respective slot will receive MIDI data.

Ch (Channel 1/2/3/4) [0, ... , 16]

Selects the MIDI channel on which the repective slot transmits and receives MIDI data (1-16 / 0=OMNI)

TS (Trigger Source) [Direct,Arpeg1,Arpeg2, Arpeg3,Arpeg4,StepSeq1,StepSeq2,StepSeq3,StepSeq4]

Select the MIDI source (MIDI input, Arpeggiator, Step Sequencer) to address the instrument loaded into this slot.(Direct / Arpeg1-4 /SSEQ1-4)

Trp (Transpose 1/2/3/4) [- 127, ... , 127]

Here you can set a transpose value in semi-tones for the instrument loaded into the respective slot.

Lkey (Low Key 1/2/3/4) [0, ... , 127]

Sets the lower limit of the MIDI note number range within which the instrument in this slot responds. Notes below this value will be ignored by the instrument.

Hkey (High Key 1/2/3/4) [0, ... , 127]

Sets the upper limit of the MIDI note number range within which the instrument in this slot responds. Notes above this value will be ignored by the instrument.

LVel (Low Velocity 1/2/3/4) [1, ... , 127]

Sets the lower limit of the MIDI note-on velocity range within which the instrument in this slot responds. Notes with a velocity below this value will be ignored by the instrument.

HVel (High Velocity 1/2/3/4) [1, ... , 127]

Sets the upper limit of the MIDI note-on velocity range within which the instrument in this slot responds. Notes with a velocity above this value will be ignored by the instrument.

OutP (Out Port 1/2/3/4) [MIDI, USB]

Sets the MIDI Out port through which the respective slot transmits MIDI.

TOut (Trigger > Out 1/2/3/4) [On, Off]

With this option enabled (ON), the notes of the Step Sequencer or the Arpeggiator are also sent to the MIDI output selected under Out Port in the *PortCh* sub-menu.

Edit Mode: Mixer Menu

Mixer Menu

The mixer not only controls the levels of the various audio signals, it's also used to load the effects.

The Mixer parameters are distribuded among 4 menus (***Instrument* / Slots*, Anlg/USB, Master, MultiView***) and their sub-menus:

*depending on *Single/Multi* mode

Mixer			
>Slots	>Anlg/USB	Master	>MultiView

Instrument / Slots Menu

When working in *Single* Mode, this menu lets you open the channel strip of the currently loaded Instrument. When working in *Multi* Mode, this menu is used to access the parameters for the channel strips of the slots 1 – 4 of the Mixer.

Slots			
>Slot 1	>Slot 2	>Slot 3	>Slot 4

Slot 1 - Slot 4 sub-menus

Slot 1				
Volume	Balance	Mute	Mix	>
Reverb	Delay	Chorus		<

Volume [0 - 127]

Controls the channel's output volume.

Balance [left 64, ..., center, ..., right 63]

Controls the perceived signal position in the stereo field.

Mute [On, Off]

Silences the channel.

Mix [On, Off]

On: With this option enabled, the signal from a channel strip is routed to the Master Bus (Mix). By default, this option is enabled.

Off: With this option disabled, the channel strip will be removed from the Master Bus. If the channel has been assigned to a Direct Output using the *Master/Outputs* menu, the signal will still be available at this output.

You can e.g. use this option to remove the slots for the Analysis and Synthesis inputs of the Vocodizer from the Master Bus, thus preventing these signals to appear on the Mixer output in addition to the Vocodizer's own output signal.

Reverb [0 - 127]

Controls the level of the portion of the signal sent to the reverb.

Delay [0 - 127]

Controls the level of the portion of the signal sent to the delay.

Chorus [0 - 127]

Controls the level of the portion of the signal sent to the chorus.

Anlg/USB Menu

Here you'll find the parameters for the mixer channel strips reserved for analog and USB input.

Anlg/USB	
>Analog	>USB

Analog sub-menu

Analog			
Volume	Balance	Mute	Mix >
Reverb	Delay	Chorus	Gain <

Volume [0 - 127]

Controls the channel's output volume.

Balance [left 64, ..., center, ..., right 63]

Controls the perceived signal position in the stereo field.

Mute [On, Off]

Silences the channel.

Mix [On, Off]

On: With this option enabled, the signal from a channel strip is routed to the Master Bus (Mix). By default, this option is enabled. Off: With this option disabled, the channel strip will be removed from the Master Bus. If the channel has been assigned to a Direct Output using the *Master/Outputs* menu, the signal will still be available at this output.

> You can e.g. use this option to remove the slots for the Analysis and Synthesis inputs of the Vocodizer from the Master Bus, thus preventing these signals to appear on the Mixer output in addition to the Vocodizer's own output signal.

Reverb [0 - 127]

Controls the level of the portion of the signal sent to the reverb.

Delay [0 - 127]

Controls the level of the portion of the signal sent to the delay.

Chorus [0 - 127]

Controls the level of the portion of the signal sent to the chorus.

Gain [-162 dB, ..., 24 dB]

The Gain control adjusts the input sensitivity so you can optimize the levels of the input sources before subsequent processing in the mixer.

USB sub-menu

USB			
Volume	Balance	Mute	Mix >
Reverb	Delay	Chorus	Gain <

Volume [0 - 127]

Controls the channel's output volume.

Balance [left 64, ..., center, ..., right 63]

Controls the perceived signal position in the stereo field.

Mute [On, Off]

Silences the channel.

Mix [On, Off]

On: With this option enabled, the signal from a channel strip is routed to the Master Bus (Mix). By default, this option is enabled.

Off: With this option disabled, the channel strip will be removed from the Master Bus. If the channel has been assigned to a Direct Output using the *Master/Outputs* menu, the signal will still be available at this output.

> You can e.g. use this option to remove the slots for the Analysis and Synthesis inputs of the Vocodizer from the Master Bus, thus preventing these signals to appear on the Mixer output in addition to the Vocodizer's own output signal.

Reverb [0 - 127]

Controls the level of the portion of the signal sent to the reverb.

Delay [0 - 127]

Controls the level of the portion of the signal sent to the delay.

Chorus [0 - 127]

Controls the level of the portion of the signal sent to the chorus.

Gain [-162 dB, ..., 24 dB]

The Gain control adjusts the input sensitivity so you can optimize the levels of the input sources before subsequent processing in the mixer.

Master Menu

Here you will find global parameters which typically appear in the master section of a mixer, along with global controls for effects and signal routing.

Master			
>Aux	>Outputs	Headroom	Volume

Headroom [- 186, ... , 0 dB]

Use this controller to turn down the volume of the Master bus. This will give you more headroom.

Internally, the mixer allows for 12 dB of headroom. Overloading should never occur, even when all channels are in use.

The Master Section inserts come before the master fader and receive the reduced-by-12bB signal, thus affording these inserts a similar amount of headroom.

The Aux busses are designed with zero headroom in order to guarantee a maximally high effect signal level. If overloading occurs here, it can be eliminated by lowering the send levels.

Volume [0 - 127] (Master Volume)

This is the global output level control (Master Volume) for Noah.

Aux sub-menu (Aux Master Sends/Returns)

Global sends and returns for the aux effects chorus, delay and reverb are located here.

Use the *Send* controllers in the *Slots* Menu to set the effect amount for the different signals.

The aux effects cannot be used if the signal of the channel strip (Slot 1- 4, Analog, USB) is being sent directly to an output.

Aux		
>Chorus	>Delay	>Reverb

Chorus sub-menu

Chorus		
Active	Send	Return

Active [No, Yes]

The chorus can be switched on or off here.

Send (Chorus Send) [0 - 127]

Adjusts the level of the effect send signal to the chorus.

The effect amounts of the individual signals are adjusted via the Send controls in the Slots menu.

Return (Chorus Return) [0 -127]

Sets the level of the effect return signal from the chorus.

Delay sub-menu

Delay		
Active	Send	Return

Active [No, Yes]

The delay can be switched on or off here.

Send (Delay Send) [0 - 127]

Adjusts the level of the effect send signal to the delay.

The effect amounts of the individual signals are adjusted via the Send controls in the Slots menu.

Return (Chorus Return) [0 -127]

Sets the level of the effect return signal from the delay.

Reverb sub-menu

Reverb		
Active	Send	Return

Active (Reverb Active) [No, Yes]

Switches the reverb on or off.

Send (Reverb Send) [0 - 127]

Controls the level of the reverb send signal.

The Send controls in the Slots menu adjust the individual signal levels.

Return (Reverb Return) [0 -127]

Controls the level of the reverb return signal.

Outputs sub-menu

Configures the routing of signal sources to the physical outputs.

Outputs		
>ADAT	>USB	Analog

Analog [None, Slot1, Slot2, Slot3, Slot4, Analog, USB, Mix]

Selects the signal to be routed to the analog outputs.

ADAT sub-menu

ADAT			
ADAT1/2	ADAT3/4	ADAT5/6	ADAT7/8

ADAT1/2 [None, Slot1, Slot2, Slot3, Slot4, Analog, USB, Mix]

Selects the signal to be routed to ADAT outputs 1/2.

ADAT3/4 [None, Slot1, Slot2, Slot3, Slot4, Analog, USB, Mix]

Selects the signal to be routed to ADAT outputs 3/4.

ADAT5/6 [None, Slot1, Slot2, Slot3, Slot4, Analog, USB, Mix]

Selects the signal to be routed to ADAT outputs 5/6.

ADAT 7/8 [None, Slot1, Slot2, Slot3, Slot4, Analog, USB, Mix]

Selects the signal to be routed to ADAT outputs 7/8.

USB sub-menu

USB		
USB1/2	USB3/4	USB5/6

USB1/2 [None, Slot1, Slot2, Slot3, Slot4, Analog, USB, Mix]

Here you can select which signal gets routed to the USB 1/2 output.

USB3/4 [None, Slot1, Slot2, Slot3, Slot4, Analog, USB, Mix]

Here you can select which signal gets routed to the USB 3/4 output.

USB5/6 [None, Slot1, Slot2, Slot3, Slot4, Analog, USB, Mix]

Here you can select which signal gets routed to the USB 5/6 output.

MultiView Menu

(in *Multi* mode only)

To provide quick access to the most important parameters of each loaded instrument in *Multi* mode, the **MultiView** menu shows the menu parameters for **Slots 1 - 4** in the form of a table containing parallel entries for each instrument. This enables you to more quickly make changes to multiple instruments (e.g., balancing the relative volume levels) with less navigating around in the submenus of individual instruments.

For each instrument which is loaded (Slot 1 - 4), you have access to the following parameters:

Slot1	Slot2	Slot3	Slot4	
Vol:127	Vol:127	Vol:127	Vol:127	>
Bal:0	Bal:0	Bal:0	Bal:0	>
Mute:Off	Mute:Off	Mute:Off	Mute:Off	>
Mix: On	Mix: On	Mix: On	Mix: On	>
Rev:127	Rev:127	Rev:127	Rev:127	>
Del:127	Del:127	Del:127	Del:127	>
Cho:127	Cho:127	Cho:127	Cho:127	>
L R	L R	L R	L R	<

Vol [0 - 127]

Volume of the associated slot.

Bal [- 64, ..., 0, ..., 63]

Pan (balance) of the associated slot.

Mute [Off, On]

Mutes / unmutes the slot.

Mix [Off, On]

Routes the channel strip to the Master Bus (Mix).

Rev [0 - 127]

Reverb send amount of the associated slot.

Del [0 - 127]

Delay send amount of the associated slot.

Cho [0 - 127]

Chorus send amount of the associated slot.

L / R (display only)

Displays the level for the left and right channel of the slot.

Edit mode: The FX menu

Using effects

With Noah you can add two kinds of effects, Aux and Insert:

A) The Chorus/Flanger, Delay and Reverb effects can be routed over the internal mixer's Aux path to be used as Aux effects (also referred to as Send effects).

B) The internal Mixer's channel strips are equipped with Insert slots for two effects you can select from the extensive palette of supplied Insert effects. Because each of the channel strips is supplied with insert slots and led to the master channel, insert effects (maximum two) can also be applied to external signals.

The FX menu thus includes corresponding submenus for the Aux effects (Chorus, Delay, Reverb) as well as for the Insert effects 1 and 2, plus the *Sources* sub-menu, which allows you to specify the source for the effect settings.

```
FX
 >Aux FX  >Insert 1 >Insert 2  >Sources
```

The Aux FX sub-menu - Chorus, Delay, Reverb

Chorus, Delay, Reverb are implemented as Auxilliary Effects; that is, they are available to all mixer channels (Instrument slots A – D, Analog In, USB In) through each channel's Aux Send lines. The settings of the effect parameters are common to all signal sources, however, as only one instance of the respective effect is loaded for all channels.

Note: The output of the Aux Effects is sent only to the mix. If the signal from a channel strip of the internal Mixer is routed to an ADAT or USB output, the effects will not be audible.

Apart from the specific parameters of individual effects, which you adjust in the menu of the effect itself, the following mixer parameters are also relevant for the Aux effects:

*** The parameters of the Mixer / Master / Aux menus (for ease of operation, however, the three send and return controls of the mixer can also be adjusted via the effects menus).**

*** The chorus, delay and reverb of the individual channel strips for Slot 1- 4, Analog In and USB In (menu / *Instrument* (Single mode) or menu Mixer / Slots / Slot 1 - 4 (Multi mode) and Mixer / Anlg / USB / Analog or USB).**

The Aux Effects are always loaded into their own address range on the DSPs and are therefore available independent of other loaded instruments.

The parameters of the three Aux effects – chorus, delay and reverb – are located in menus with the same names:

```
Aux FX
 >Chorus    >Delay   >Reverb
```

Adding the effect component of an Aux effect to the signal of another effect

Often it is desirable for the output of one aux effect to be processed with another aux effect. An example in a typical studio environment would be to add reverb to a delayed signal. If you include the delay and reverb in the mix using two different aux paths, the original signal will have reverb, but the delayed signal will not, and the result will sound unnatural. The experienced recording engineer will probably not route the delay signal back to the mix through a return, but rather through another channel where reverb can be added to it before sending it back along the effects path. Noah, too, allows you to establish such signal routings among the three effects.

Specifically, in the Aux Editor you can send the output of the Chorus to the inputs of the delay and reverb with an independent level for each. Then you can send the output of the delay to the reverb at a level independent of its level in the overall mix.

Chorus sub-menu (Chorus / Flanger)

This chorus effect can also be used as a flanger.

The name "Chorus" hints at the sound produced by this effect. It spreads and thickens the sound passed through it, simulating the sound of multiple instruments of the same type playing together – in other words, a chorus. This effect is achieved by means of a short delay line whose delay time is modulated periodically. Mixing this delayed signal with the original produces the chorus effect.

Like the chorus, a flanger is based on a delay line with a periodically modulated delay time. However, the delay times in a flanger are substantially shorter than those of a chorus. In addition, the flanger feeds back the delayed signal to the delay line input. Therefore, it not only thickens the sound but can add noticeable coloration owing to the comb-filter effect which results from the feedback.

```
Chorus/Flanger
   Type:    Chorus                        >
   Rate     Depth    Feedback   Phase    >
   Send     Return   Dly Send   Rev Send<
```

Type [Chorus, Flanger]

Selects either Chorus or Flanger mode.

Rate [0, ... , 40 Hz]

Adjusts the frequency of the delay time modulation in the chorus/flanger effect.

Depth [0, ... , 127]

Adjusts the amount of delay time modulation in the chorus/flanger effect.

Feedback [- 64, ... , 63]

Controls the level of the feedback signal which produces the comb filter effect, similar to flanging. Negative feedback values invert the phase of the feedback signal, changing the sound of the comb filter effect accordingly.

Phase [0, ... , 180]

Adjusts the phase difference between the modulation signals applied to the left and right channels. This influences the apparent "breadth" of the stereo image.

Send [0, ... , 127]

Adjusts the level of the chorus effect-send signal.

This parameter is a reference to the corresponding control in the mixer – or, in other words, the two controls are coupled to one another. Thus, you do not need to navigate to the mixer menu in order to adjust this setting.

Return [0, ... , 127]

Sets the level at which the chorus is added to the overall mix.

This parameter is a reference to the corresponding control in the mixer – or, in other words, the two controls are coupled to one another. Thus, you do not need to navigate to the mixer menu in order to adjust this setting.

Dly Send (Delay Send) [0, ... , 127]

Controls the level of the chorus effect output added to the delay input signal independent of the delay level in the overall mix. (*see also box: Adding the effect component of an Aux effect to the signal of another effect*).

Rev Send (Reverb Send) [0, ... , 127]

Controls the level of the chorus effect output added to the reverb input signal independent of the chorus level in the overall mix. (*see also box: Adding the effect component of an Aux effect to the signal of another effect*).

Delay sub-menu

Delay		
>Snd/Rtn	>*Delay Type*	*Delay Mode*

Delay Mode **[ms/Bpm]**

This is where you specify if the delay time should be set in millisecond or in beats per minute (BPM). In the latter case, the tempo you set in the *Edit/MIDI/ Clock* menu will be used to calculate the respective delay time.

Depending on the setting of this parameter, names and values of the delay parameters in the menus for the different delay types change accordingly.

Send/Rtn sub-menu (Send / Return)

Snd/Rtn		
Send	Return	Rev Send

Send [0, ... , 127]

Controls the level of the delay effect output

This parameter is a reference to the corresponding control in the mixer – or, in other words, the two controls are coupled to one another. Thus, you do not need to navigate to the mixer menu in order to adjust this setting.

Return [0, ... , 127]

Specifies the level of the delay effect added to the overall mix.

This parameter is a reference to the corresponding control in the mixer – or, in other words, the two controls are coupled to one another. Thus, you do not need to navigate to the mixer menu in order to adjust this setting.

Rev Send [0, ... , 127]

Controls the level of the delay effect output added to the reverb input signal independent of the delay level in the overall mix. (*see also box: Adding the effect component of an Aux effect to the signal of another effect*).

The different types of Delay

In the *Delay Type* field, you can choose among different types of delay with the second of the Continuous Controllers (5). The displayed parameters change according to the type you select.

The available delay types are:

Dual Delay, LCR Delay, Multitap Delay, Stereo Delay, Synth Delay.

Dual Delay

This delay provides the left and right channels with their own independent feedback loops. Filters are included in the feedback loop to dampen the high or low frequencies of subsequent echos.

Dual Delay			
Delay L	HiDamp L	LoDamp L	Feedb L >
Delay R	HiDamp R	LoDamp R	Feedb R <

Delay L/R [0, ... , 4000 ms] (ms-Mode)

Sets delay time directly in milliseconds.

NoteL L/R (Note Length L/R) [1/1, 1/2 dot, 1/2, 1/2 trpl, 1/4 dot, 1/4, 1/4 trpl, 1/8 dot, 1/8, 1/8 trpl, 1/16 dot, 1/16, 1/16 trpl, 1/32, 1/32 trpl] (BPM-Mode)

Set delay time by note values. *Dot* and *trpl* stand for ”dotted” and ”triplet”.

The maximum delay time is four seconds. Depending on tempo and note value, the calculated delay time might exceed this maximum value. However, the delay will never be longer than 4 seconds.

Hi Damp L/R (High Damp L/R) [10, ... , 24000 Hz]

Adjusts the amount of high frequency damping in the feedback loop.

Lo Damp L/R (Low Damp L/R) [10, ... , 24000 Hz]

Adjusts the amount of low frequency damping in the feedback loop.

Feedb L/R (Feedback L/R) [0, ... , 127]

Sets the strength of the feedback - the portion of the delayed signal that is routed back to the input to create multiple receding echos. The degree of feedback determines the fade out time of the repeated echoes.

LCR Delay

The LCR Delay sends the delayed input signal to the left, center or right channel. The delay is adjustable per channel (L/C/R) and a feedback loop produces repeated echos if desired. Filters in the feedback loop allow for high or low frequency attenuation of the echos.

LCR Delay			
Delay L	Delay C	Delay R	>
Level L	Level C	Level R	>
LoDamp	HiDamp	Feedback	<

Delay L/C/R (Delay Left /Center / Right) [0, ... , 4000 ms] (ms-Mode)

Sets delay time /Left, Center, Right) directly in milliseconds.

NoteL L/C/ R (Note Length L/C/R) [1/1, 1/2 dot, 1/2, 1/2 trpl, 1/4 dot, 1/4, 1/4 trpl, 1/8 dot, 1/8, 1/8 trpl, 1/16 dot, 1/16, 1/16 trpl, 1/32, 1/32 trpl] (BPM-Mode)

Set delay time by note values. Dot and trpl stand for ”dotted” and ”triplet”.

The maximum delay time is four seconds. Depending on tempo and note value, the calculated delay time might exceed this maximum value. However, the delay will never be longer than 4 seconds.

Level L/C/R (Level Left /Center / Right) [0, ... , 127]

Adjusts the volume level of each individual delay. Set this to 0 to omit the delay.

Lo Damp (Low Damp) [10, ... , 24000 Hz]

Adjusts the amount of low frequency damping in the feedback loop.

Hi Damp (High Damp) [0, ... , 24000 Hz]

Adjusts the amount of high frequency damping in the feedback loop.

Feedback [0, ... , 127]

Sets the strength of the feedback - the portion of the delayed signal that is routed back to the input to create multiple receding echos. The degree of feedback determines the fade out time of the repeated echoes.

Multitap Delay

The Multitap Delay provides four delays with independently adjustable volume and pan position. The delay time is adjustable for each tap, and a feedback loop is inserted after delay 1 to produce recurring echos. Filters are included in the feedback loop to attenuate the high or low frequencies of subsequent echos. The maximum delay time for each tap is 4.00 s.

Multitap Delay			
Tap 1	Level 1	Pan 1	>
Tap 2	Level 2	Pan 2	>
Tap 3	Level 3	Pan 3	>
Tap 4	Level 4	Pan 4	>
LoDamp	HiDamp	Feedback	<

Tap 1-4 [0, ... , 4000 ms] (ms-Mode)

Sets delay time directly in milliseconds for each of the four taps.

NoteL 1/2/3/4 (Note Length 1/2/3/4) [1/1, 1/2 dot, 1/2, 1/2 trpl, 1/4 dot, 1/4, 1/4 trpl, 1/8 dot, 1/8, 1/8 trpl, 1/16 dot, 1/16, 1/16 trpl, 1/32, 1/32 trpl] (BPM-Mode)

Set delay time by note values. Dot and trpl stand for ”dotted” and ”triplet”.

The maximum delay time is four seconds. Depending on tempo and note value, the calculated delay time might exceed this maximum value. However, the delay will never be longer than 4 seconds.

Level 1-4 [0, ... , 127]

Adjusts the volume level of each individual tap. Set this to 0 to omit the tap.

Pan 1-4 [left 64, ..., center, ..., right 63]

Sets the pan position in the stereo field for each of the four taps.

Lo Damp (Low Damp) [10, ... , 24000 Hz]

Adjusts the amount of low-frequency damping in the feedback loop.

Hi Damp (High Damp) [0, ... , 24000 Hz]

Adjusts the amount of high-frequency damping in the feedback loop.

Feedback [0, ... , 127]

Controls the amount of the delayed signal to be fed back to the input to be delayed again. By skillfully setting indiviual delay times for each tap, interesting rhythmic effects can be produced when feedback is introduced.

Stereo Delay

The signal sent to the delay is delayed for a specified time. The delay time is adjustable independently for each stereo channel, and can be fed back to produce recurring echos. An integrated filter controls the attenuation of the echos depending on frequency and intensity.

Stereo Delay				
Delay L	Delay R	LoDamp	HiDamp	>
Feedback	Cross FB			<

Delay L/R [0, ... , 4000 ms] (ms-Mode)

Sets delay time directly in milliseconds.

NoteL L/R (Note Length L/R) [1/1, 1/2 dot, 1/2, 1/2 trpl, 1/4 dot, 1/4, 1/4 trpl, 1/8 dot, 1/8, 1/8 trpl, 1/16 dot, 1/16, 1/16 trpl, 1/32, 1/32 trpl] (BPM-Mode)

Set delay time by note values. Dot and trpl stand for ”dotted” and ”triplet”.

The maximum delay time is four seconds. Depending on tempo and note value, the calculated delay time might exceed this maximum value. However, the delay will never be longer than 4 seconds.

Lo Damp (Low Damp) [10, ... , 24000 Hz]

Use this control to adjust the amount of low frequency damping in the feedback loop for the respective channel.

Hi Damp (High Damp) [0, ... , 24000 Hz]

Use this control to adjust the amount of high frequency damping in the feedback loop for the respective channel.

Feedback [0, ... , 127]

Sets the strength of the feedback - the portion of the delayed signal that is routed back to the input to create multiple receding echos. The degree of feedback determines the fade out time of the repeated echoes.

Cross FB (Cross Feedback) [Off, On]

Cross feedback, when enabled, feeds the signal from the left channel to the right channel to be delayed, and the signal from the right channel back to the left. The signal path thus forms a figure eight.

Synth Delay

This delay provides the left and right channels with their own independent feedback loops. Filters are included in the feedback loop to dampen the high frequencies of subsequent echos.

Synth Delay				
Delay L	Feedb L	HiDamp L	Level L	>
Delay R	Feedb R	HiDamp R	Level R	>
Cross FB				<

Delay L/R [0, ... , 4000 ms] (ms-Mode)

Sets delay time directly in milliseconds.

NoteL L/R (Note Length L/R) [1/1, 1/2 dot, 1/2, 1/2 trpl, 1/4 dot, 1/4, 1/4 trpl, 1/8 dot, 1/8, 1/8 trpl, 1/16 dot, 1/16, 1/16 trpl, 1/32, 1/32 trpl] (BPM-Mode)

Set delay time by note values. Dot and trpl stand for ”dotted” and ”triplet”.

The maximum delay time is four seconds. Depending on tempo and note value, the calculated delay time might exceed this maximum value. However, the delay will never be longer than 4 seconds.

Feedb L/R (Feedback L/R) [0, ... , 127]

Sets the strength of the feedback - the portion of the delayed signal that is routed back to the input to create multiple receding echos. The degree of feedback determines the fade out time of the repeated echoes.

Hi Damp L/R (High Damp L/R) [0, ... , 24000 Hz]

Adjusts the amount of high frequency damping in the feedback loop.

Level L /R [0, ... , 127]

Sets the level of the delay effect for each channel.

Cross FB (Cross Feedback) [Off, On]

Cross feedback, when enabled, feeds the signal from the left channel to the right channel to be delayed, and the signal from the right channel back to the left. The signal path thus forms a figure eight.

Reverb sub-menu

```
Reverb
Pre Delay  Low Cut   High Cut         >
Room Size  Rev Time   Shape   High Damp>
   Send      Return                   <
```

Pre Delay [0, ..., 200 ms]

Adjusts the delay of the response in milliseconds. The PreDelay is used to separate the hall response from the direct signal.

Low Cut [0, ... , 24000 Hz]

A lowpass filter with a 12dB/Octave slope is located at the reverb input.

> **Rooms or halls we classify as "warm" absorb most of the high frequencies–those over 8 kHz (or less). Use the low pass filter to create a warmer response.**

High Cut [0, ... , 24000 Hz]

A high pass filter, also with a slope of 12db/ocatve, follows the low pass filter in the signal path.

> **The response of many halls lies mainly within the middle frequencies. Together, the low pass and high pass filters form a band pass filter, attenuating both high and low frequencies. Appropriate adjustment of the two filters reproduces the sound of a hall in which the midrange is favored.**

Size (Room Size) [0, ... , 127]

Adjusts the perceived room size.

> **In order to avoid interference, the response is muted for brief periods of time when adjusting the room size.**

Rev. Time (Reverb Time) [0, ... , 127]

Controls the reverb time (duration). The reverb time is unlimited, and can even be adjusted to infinity if desired.

> **As in a real-world acoustical environment, long reverb times correspond to large rooms. In small rooms, use shorter times to produce a natural sounding response.**

Shape [0, ..., 127]

The Shape control lets you change the envelope of the response signal. Lower values correspond to relatively fast rise and fall times, while larger values result in slower rise and fall times. The effect is analogous to moving a wall, or raising or lowering the ceiling in a concert hall. The apparent size of the space changes correspondingly.

> **For percussive sounds like drums or other percussion instruments, smaller values for the Shape parameter are best.**

HighDamp [0, ... , 24000 Hz]

This 6 db/octave filter operates on the reverb response by reducing the high frequencies in the response signal depending on how it is adjusted. There are three ways to adjust the cutoff frequency: a) use the rotary control, b) enter the value directly in the text field, or c) adjust the value directly in the graphic display area.

> **Rooms or halls absorb high frequencies quite strongly. Therefore, settings between 3 kHz and 6 kHz are typical.**

Send [0, ... , 127]

Controls the level of the delay effect output

> This parameter is a reference to the corresponding control in the mixer – or, in other words, the two controls are coupled to one another. Thus, you do not need to navigate to the mixer menu in order to adjust this setting.

Return [0, ... , 127]

Specifies the level of the delay effect added to the overall mix.

> This parameter is a reference to the corresponding control in the mixer – or, in other words, the two controls are coupled to one another. Thus, you do not need to navigate to the mixer menu in order to adjust this setting.

The Insert 1/2 sub-menus - insert effects

FX
>Aux FX >Insert 1 >Insert 2 >Sources

Insert 1
Routing Type

Noah allows for two insert effects to be loaded dynamically. Intern, each channel strip and the Master channel are equipped with two insert fields (Insert1 and Insert2) which are assigned to two hardware insert slots. In other words, if for example an effect is loaded into Insert1 of one of the channel strips then no other effects can be loaded into Insert1 of any of the other channels.

In the Single operating mode you can load an effect into each of the Insert slots of a channel strip, or of the Master channel. Effects loaded into the Master channel process all individual channel signals as well as the analog and USB input signals.

In the Multi operating mode you can use the effects loaded into a physical insert slot of an instrument by assigning them in the Insert fields of the respective channels. Or you can load the effects into the Insert fields of the Master channel whereby all mixer channels and therefore all instruments will be affected.

Routing **[Not Active, Slot1, Slot2, Slot3, Slot4, Analog, USB, Mix]**

Here you can individually deactivate insert slots (Not Active) or select the mixer channel strip (Slot 1 - 4, Analog, USB) to which the effect should be applied. . If *Mix* is selected, the effect is applied to the Master channel and thereby to all channel strips.

Type **[*Insert effect list*]**

Select the desired insert effect for each insert slot.

Insert effects list

No.	*Type*	*Name*
1	*Filter*	*Stereo EQ*
2	*Filter*	*Parametric EQ*
3	*Filter*	*Graphic EQ*
4	*Modulation*	*Ensemble*
5	*Modulation*	*Master Chorus*
6	*Modulation*	*Harmonic Chorus*
7	*Modulation*	*Hexa Chorus*
8	*Modulation*	*Triple Chorus*
9	*Modulation*	*Master Flanger*
10	*Modulation*	*Harmonic Flanger*
11	*Modulation*	*Random Flanger*
12	*Modulation*	*Space Flanger*
13	*Modulation*	*Step Flanger*
14	*Modulation*	*Master Phaser*
15	*Modulation*	*SSB Phaser*
16	*Other*	*2 Voice Pitch Shifter*
17	*Other*	*Stereo Pitch Shifter*
18	*Other*	*Feedback Pitch Shifter*
19	*Modulation*	*Auto Pan*
20	*Modulation*	*Tremolo*
21	*Filter*	*Auto Wah*
22	*Distortion*	*Amplifier*
23	*Distortion*	*Decimator*
24	*Distortion*	*Distortion*
25	*Distortion*	*Overdrive*
26	*Filter*	*Resonator*
27	*Filter*	*Ringmodulator*
28	*Other*	*Soft Clip*
29	*Other*	*Stereo Expander*
30	*Other*	*Tube Processor*
31	*Dynamic*	*Compressor*
32	*Dynamic*	*Expander*
33	*Dynamic*	*Limiter*
34	*Dynamic*	*Gate*
35	*Dynamic*	*Dynamics*

Stereo EQ / Parametric EQ

The EQ provides four bands of equalization. Each band has an adjustable cut/boost, frequency, and Q factor. In the Stereo EQ bands 1 and 4 are implemented as high and low shelving filters and therefore do not have an adjustable Q.

Parametric EQ			
Freq 1	Gain 1	Q1	>
Freq 2	Gain 2	Q2	>
Freq 3	Gain 3	Q3	>
Freq 4	Gain 4	Q4	Bypass <

Freq 1 -4 [20, ... , 20.000 Hz]

Adjusts the frequencies of the respective filter bands.

Gain 1-4 [- 12 , ... , 12 dB]

Adjusts the filter band's cut or boost level in dB.

Q 1-4 [0.7 , ... , 20]

Sets the quality, or Q factor, for a filter band. This value determines the range of frequencies around a center frequency the filter will act on.

In the Stereo EQ, Q1 and Q4 are missing as the upper and lower bands are implemented as shelving filters.

Bypass [Off, On]

This mutes and unmutes the effect.

Graphic EQ

This equalizer is equipped with 8 bands of fixed frequency.

Graph EQ			
63 Hz	125 Hz	250 Hz	500 Hz >
1000 Hz	2000 Hz	4000 Hz	8000 Hz <
Gain			<

63/125/250/500/1000/2000/4000/8000 Hz [- 12 , ... , 12 dB]

Each control regulates a specific frequency band.

Gain [0, ... , 127]

Via this input gain control, you can reduce the incoming signal level as necessary so that overloading does not occur even when specific frequency ranges are strongly boosted.

Bypass [Off, On]

This mutes and unmutes the effect.

Chorus Effects

In addition to the chorus available as an Aux effect, you can also load the following choruses as Insert effects.

The name "chorus" hints at the sound produced by this effect. It spreads and thickens the signal passed through it, simulating the sound of multiple instruments of the same type playing together – in other words, a chorus. This effect is achieved by means of a short delay line with a delay time periodically modulated delay time. Mixing this delayed signal with the original produces the chorus effect. This effect is also useful for creating a stereo impression from a monaural signal.

Ensemble

The Ensemble Chorus is a simple chorus with only a few parameters.

Ensemble			
Rate	Depth	Dry	Wet >
			Bypass <

Rate [0.01, ... , 40 Hz]

Adjusts the frequency of delay time modulation in the chorus effect.

Depth [0, ... , 127]

Adjusts the amount of delay time modulation in the chorus effect, and thus its intensity.

Dry (Dry Level) [0, ... , 127]

Adjusts the level of the original signal.

Wet (Wet Level) [0, ... , 127]

Adjusts the level of the signal after processing by the chorus effect.

Make sure there is always some of the wet component added to the dry signal. Otherwise the effect will not be apparent.

Bypass [Off, On]

This mutes and unmutes the effect.

Master Chorus

This sophisticated device offers options for modifying modulation and chorus tone color in addition to the classic chorus parameters. It covers a wide range of sounds, from especially subtle chorusing to intense chorusing with feedback.

Master Chorus			
Predel L	Predel R	Waveform	Shape >
Rate	Depth	Feedback	Phase >
LoDamp	HiDamp	Dry	Wet >
			Bypass <

Predel L/R (Predelay L/R) [0, ... , 100 ms]

Adjusts the delay time over the range of 0 to 100 ms for both channels of the integrated stereo delay line which is connected in-line ahead of the chorus effect.

Waveform [Sine, Triangle]

Selects either a sine or triangle waveform for modulation of the chorus effect.

Shape [0, ... , 127]

"Warps" the modulation waveform, widening the dips and narrowing the peaks to an adjustable degree, thereby altering the "motion" produced by the modulation.

Rate [0.01, ... , 40 Hz]

Adjusts the frequency of delay time modulation in the chorus effect.

Depth [0, ... , 127]

Adjusts the amount of delay time modulation in the chorus effect, and thus its intensity.

Feedback [- 64, ... , 63]

Controls the level of the feedback signal that produces the comb filter effect, similar to flanging. Negative feedback values invert the phase of the feedback signal, changing the sound of the comb filter effect accordingly.

Phase [0, ... , 180]

Adjusts the phase difference between the modulation signals applied to the left and right channels. This influences the apparent "breadth" of the stereo image.

LoDamp (Low Damp) [10, ... , 24000 Hz]

Adjusts the amount of low-frequency damping in the feedback loop.

HiDamp (High Damp) [0, ... , 24000 Hz]

Adjusts the amount of high-frequency damping in the feedback loop.

> **Through the simultaneous use of both filters, the comb filter effect produced via feedback can be restricted to specific frequencies.**

Dry (Dry Level) [0, ... , 127]

Adjusts the level of the original signal.

Wet (Wet Level) [0, ... , 127]

Adjusts the level of the signal after processing by the chorus effect.

> **Make sure there is always some of the wet component added to the dry signal. Otherwise the effect will not be apparent.**

Bypass [Off, On]

This mutes and unmutes the effect.

Harmonic Chorus

The **Harmonic Chorus** splits the signal into two frequency ranges and applies the effect only to frequencies above the split frequency.

Harmonic Chorus

Split F	Lo Level	Hi Level		>
Rate	Depth	Feedback	Phase	>
LoDamp	HiDamp	Dry	Wet	>
			Bypass	<

Split F (Split Freq) [10, ... , 4000 Hz]

Sets the frequency at which the original signal is split into two frequency ranges. Only the frequencies above the split frequency will be processed for the chorus effect.

Lo Level (Low Level) [0, ... , 127]

Adjusts the volume level of the portion of the signal lying below the split frequency. This part of the signal is not processed via the chorus effect.

Hi Level (High Level) [0, ... , 127]

Adjusts the volume level of the portion of the signal lying above the split frequency. This part of the signal is processed with the chorus effect.

Rate [0.01, ... , 40 Hz]

Adjusts the frequency of delay time modulation in the chorus effect.

Depth [0, ... , 127]

Adjusts the amount of delay time modulation in the chorus effect, and thus its intensity.

Feedback [- 64, ... , 63]

Controls the level of the feedback signal that produces the comb filter effect, similar to flanging. Negative feedback values invert the phase of the feedback signal, changing the sound of the comb filter effect accordingly.

Phase [0, ... , 180]

Adjusts the phase difference between the modulation signals applied to the left and right channels. This influences the apparent "width" of the stereo image.

LoDamp (Low Damp) [10, ... , 24000 Hz]

Adjusts the amount of low-frequency damping in the feedback loop.

HiDamp (High Damp) [0, ... , 24000 Hz]

Adjusts the amount of high-frequency damping in the feedback loop.

Through the simultaneous use of both filters, the comb filter effect produced via feedback can be restricted to a specific band of frequencies.

Dry (Dry Level) [0, ... , 127]

Adjusts the level of the original signal.

Wet (Wet Level) [0, ... , 127]

Adjusts the level of the signal after processing by the chorus effect.

Make sure there is always some of the wet component added to the dry signal. Otherwise the effect will not be apparent.

Bypass [Off, On]

This mutes and unmutes the effect.

Hexa Chorus / Triple Chorus

With the Triple or Hexa Choruses the signal is delayed not with a single delay line, but with 3 or 6 delay lines respectively. The sound of these choruses is particularly full, rich and nuanced.

```
Triple Chorus
   Rate      Depth       Dry        Wet    >
                                 Bypass <
```

Rate [0.01, ... , 40 Hz]

Adjusts the frequency of delay time modulation in the chorus effect.

Depth [0, ... , 127]

Adjusts the amount of delay time modulation in the chorus effect, and thus its intensity.

Spread (Pan Spread) [0, ... , 127] (nur Hexa Chorus)

Controls the distribution of the effect signals between the two stereo channels. At the minimum value, the signals are mixed and sent equally to each channel. At the maximum value, the signals are sent individually to each channel, creating a quasi stereo effect.

Dry (Dry Level) [0, ... , 127]

Adjusts the level of the original signal.

Wet (Wet Level) [0, ... , 127]

Adjusts the level of the signal after processing by the effect.

Make sure there is always some of the wet component added to the dry signal. Otherwise the effect will not be apparent.

Bypass [Off, On]

This mutes and unmutes the effect.

Flanger Effects

Although the chorus supplied as an Aux effect can be switched to Flanger mode, you can also use the following flangers provided as Insert effects.

This effect is similar to the chorus. Like the chorus, a flanger is based on a delay line whose delay time is periodically modulated. However, the delay times in a flanger are substantially shorter than those of a chorus. In addition, the flanger utilizes feedback of the delayed signal back to the delay line input. Therefore, it not only thickens the sound but can add noticeable coloration owing to the comb-filter effect which results from the feedback. The intensity of the effect depends upon the modulation rate, depth and phase settings as well as the dry/wet (original/delayed) mix. This effect is also useful for creating a stereo field from a monaural signal.

Master Flanger

This sophisticated device offers options for modifying modulation and flanger tone color in addition to the classic flanger parameters. It covers a wide range of sounds, from subtle flanging to intense flanging with feedback.

Master Flanger			
Predel L	Predel R	Waveform	Shape >
Rate	Depth	Feedback	Phase >
LoDamp	HiDamp	Dry	Wet >
			Bypass <

Predel L/R (Predelay L/R) [0, ... , 100 ms]

Adjusts the delay time over the range of 0 to 100 msec for both channels of the integrated stereo delay line which is connected in-line ahead of the flanger effect.

Waveform [Sine, Triangle]

Selects either a sine or triangle waveform for modulation of the flanger effect.

Shape [0, ... , 127]

"Warps" the modulation waveform, widening the dips and narrowing the peaks to an adjustable degree, thereby altering the "motion" produced by the modulation.

Rate [0.01, ... , 40 Hz]

Adjusts the frequency of delay time modulation in the flanger effect.

Depth [0, ... , 127]

Adjusts the amount of delay time modulation in the flanger effect, and thus its intensity.

Feedback [- 64, ... , 63]

Controls the level of the feedback signal that produces the typical flanger comb filter effect. Negative feedback values invert the phase of the feedback signal, changing the sound of the comb filter effect accordingly.

Phase [0, ... , 180]

Adjusts the phase difference between the modulation signals applied to the left and right channels. This influences the apparent "breadth" of the stereo image.

LoDamp (Low Damp) [10, ... , 24000 Hz]

Adjusts the amount of low-frequency damping in the feedback loop.

HiDamp (High Damp) [0, ... , 24000 Hz]

Adjusts the amount of high-frequency damping in the feedback loop.

Through the simultaneous use of both filters, the comb filter effect produced via feedback can be restricted to specific frequencies.

Dry (Dry Level) [0, ... , 127]

Adjusts the level of the original signal.

Wet (Wet Level) [0, ... , 127]

Adjusts the level of the signal after processing by the chorus effect.

Make sure there is always some of the wet component added to the dry signal. Otherwise the effect will not be apparent.

Bypass [Off, On]

This mutes and unmutes the effect.

Harmonic Flanger

The Harmonic Flanger splits the signal into two frequency ranges, and applies the effect only to the frequencies above the split frequency.

Harmonic Flanger				
Split F	Lo Level	Hi Level		>
Rate	Depth	Feedback	Phase	>
LoDamp	HiDamp	Dry	Wet	>
			Bypass	<

Split F (Split Freq) [10, ... , 4000 Hz]

Sets the frequency at which the original signal is split. Only the frequencies above the split frequency are processed via the flanger effect.

Lo Level (Low Level) [0, ... , 127]

Adjusts the volume level of the portion of the signal lying below the split frequency. This part of the signal is not processed via the flanger effect.

Hi Level (High Level) [0, ... , 127]

Adjusts the volume level of the portion of the signal lying above the split frequency. This part of the signal is processed via the flanger effect.

Rate [0.01, ... , 40 Hz]

Adjusts the frequency of delay time modulation in the flanger effect.

Depth [0, ... , 127]

Adjusts the amount of delay time modulation in the flanger effect, and thus its intensity.

Feedback [- 64, ... , 63]

Controls the level of the feedback signal that produces the comb filter effect. Negative feedback values invert the phase of the feedback signal, changing the sound of the comb filter effect accordingly.

Phase [0, ... , 180]

Adjusts the phase difference between the modulation signals applied to the left and right channels. This influences the apparent "width" of the stereo image.

LoDamp (Low Damp) [10, ... , 24000 Hz]

Adjusts the amount of low-frequency damping in the feedback loop.

HiDamp (High Damp) [0, ... , 24000 Hz]

Adjusts the amount of high-frequency damping in the feedback loop.

Through the simultaneous use of both filters, the comb filter effect produced via feedback can be restricted to specific frequencies.

Dry (Dry Level) [0, ... , 127]

Adjusts the level of the original signal.

Wet (Wet Level) [0, ... , 127]

Adjusts the level of the processed signal.

Make sure there is always some of the wet component added to the dry signal. Otherwise the effect will not be apparent.

Bypass [Off, On]

This mutes and unmutes the effect.

Random Flanger

The sound of this effect is similar to that of the Master Flanger. However, the modulation waveform is a random signal, resulting in an effect that varies continuously in intensity and character.

Random Flanger			
Predel L	Predel R	Waveform	>
Rate	Depth	Feedback	PhaseInv>
LoDamp	HiDamp	Dry	Wet >
			Bypass <

Predel L/R (Predelay L/R) [0, ... , 100 ms]

Adjusts the delay time over a range of 0 to 100 msec for both channels of the integrated stereo delay line which is located just ahead of the flanger effect.

Waveform [Step, Triangle, Sine]

Selects a sine, triangle or stepped waveform for modulation of the flanger effect. The amplitude of the selected waveform varies randomly.

Rate [0.01, ... , 40 Hz]

Adjusts the frequency of the delay time modulation in the flanger effect.

Depth [0, ... , 127]

Adjusts the degree of delay time modulation in the flanger effect, and thus its intensity.

Feedback [0, ... , 127]

Controls the level of the feedback signal that produces the typical flanger comb filter effect. Negative feedback values invert the phase of the feedback signal, changing the sound of the comb filter effect accordingly.

PhaseInv (Phase Invert) [On, Off]

Selects between in-phase (Off) and out-of-phase (On) modulation signals for the left and right channels of the flanger effect. This influences the perceived "width" of the stereo image.

LoDamp (Low Damp) [10, ... , 24000 Hz]

Adjusts the amount of low-frequency damping in the feedback loop.

HiDamp (High Damp) [0, ... , 24000 Hz]

Adjusts the amount of high-frequency damping in the feedback loop.

Through the simultaneous use of both filters, the comb filter effect produced via feedback can be restricted to specific frequencies.

Dry (Dry Level) [0, ... , 127]

Adjusts the level of the original signal.

Wet (Wet Level) [0, ... , 127]

Adjusts the level of the signal after processing by the chorus effect.

Make sure there is always some of the wet component added to the dry signal. Otherwise the effect will not be apparent.

Bypass [Off, On]

This mutes and unmutes the effect.

Space Flanger

In this flanger not only is the delay position varied, but also the delay length. This lends the Space Flanger a very distinctive character.

Space Flanger			
Predel L	Predel R	Rate	Depth >
Feedback	Phase	LoDamp	HiDamp >
Dry	Wet		Bypass <

Predel L/R (Predelay L/R) [0, ... , 100 ms]

Adjusts the delay time over the range of 0 to 100 msec for both channels of the integrated stereo delay line which is connected in-line ahead of the flanger effect.

Rate [0.01, ... , 40 Hz]

Adjusts the frequency of delay time modulation in the flanger effect.

Depth [0, ... , 127]

Adjusts the amount of delay time modulation in the flanger effect, and thus its intensity.

Feedback [- 64, ... , 63]

Controls the level of the feedback signal which produces the typical flanger comb filter effect. Negative feedback values invert the phase of the feedback signal, changing the sound of the comb filter effect accordingly.

Phase [0, ... , 180]

Adjusts the phase difference between the modulation signals applied to the left and right channels. This influences the apparent "breadth" of the stereo image.

LoDamp (Low Damp) [10, ... , 24000 Hz]

Adjusts the amount of low-frequency damping in the feedback loop.

HiDamp (High Damp) [0, ... , 24000 Hz]

Adjusts the amount of high-frequency damping in the feedback loop.

Through the simultaneous use of both filters, the comb filter effect produced via feedback can be restricted to specific frequencies.

Dry (Dry Level) [0, ... , 127]

Adjusts the level of the original signal.

Wet (Wet Level) [0, ... , 127]

Adjusts the level of the signal after processing by the chorus effect.

Make sure there is always some of the wet component added to the dry signal. Otherwise the effect will not be apparent.

Bypass [Off, On]

This mutes and unmutes the effect.

Step Flanger

The Step Flanger applies a sample-and-hold circuit to the modulation waveform. The resulting flanger effects range from a step-wise evolution of the comb filter through its spectrum to abrupt large jumps.

Step Flanger			
Waveform	Shape	Rate	Depth >
Feedback	Phase	StepRate	Step Lag>
LoDamp	HiDamp	Dry	Wet >
			Bypass <

Waveform [Sine, Triangle]

Selects either a sine or triangle waveform for modulation of the flanger effect.

Shape [0, ... , 127]

"Warps" the modulation waveform, widening the dips and narrowing the peaks to an adjustable degree, thereby altering the "motion" produced by the modulation.

Rate [0.01, ... , 40 Hz]

Adjusts the frequency of delay time modulation in the flanger effect.

Depth [0, ... , 127]

Adjusts the amount of delay time modulation in the flanger effect, and thus its intensity.

Feedback [- 64, ... , 63]

Controls the level of the feedback signal which produces the typical flanger comb filter effect. Negative feedback values invert the phase of the feedback signal, changing the sound of the comb filter effect accordingly.

Phase [0, ... , 180]

Adjusts the phase difference between the modulation signals applied to the left and right channels. This influences the apparent "breadth" of the stereo image.

StepRate (Step Rate) [0.25, ... , 38]

Adjusts the rate at which the modulation signal is "sliced" into steps by the sample-and-hold circuit, specified as a multiple of the modulation frequency. For typical staircase effects, the step rate should be at least twice the modulation rate.

Step Lag [0, ... , 127]

Adjusts the rate at which the modulation signal slews, or makes its transition, from one step value to the next. Larger settings produce more gradual transitions, while a setting of zero yields very hard transitions.

LoDamp (Low Damp) [10, ... , 24000 Hz]

Adjusts the amount of low-frequency damping in the feedback loop.

HiDamp (High Damp) [0, ... , 24000 Hz]

Adjusts the amount of high-frequency damping in the feedback loop.

Through the simultaneous use of both filters, the comb filter effect produced via feedback can be restricted to specific frequencies.

Dry (Dry Level) [0, ... , 127]

Adjusts the level of the original signal.

Wet (Wet Level) [0, ... , 127]

Adjusts the level of the signal after processing by the chorus effect.

Make sure there is always some of the wet component added to the dry signal. Otherwise the effect will not be apparent.

Bypass [Off, On]

This mutes and unmutes the effect.

Master Phaser

The phaser or phase-shifter is similar to the flanger and chorus, but instead of using a delay line, it functions by introducing frequency-dependent phase shifts into the input signal. The key phase-shift frequencies, and therefore the amount of phase shift, are periodically modulated. The phase-shifter effect results when this phase-manipulated signal is mixed with the original signal. The effect is thus similar to the flanger and chorus, but with a sonic character of its own. The intensity of the effect depends upon the modulation rate, depth and phase settings as well as the dry/wet (original/phase-shifted) mix. The effect is also useful for creating a stereo field from a monaural signal.

Master Phaser			
Type	Manual	Res	Waveform>
Rate	Depth	Phase	Shape >
Dry	Wet		Bypass <

Type [6 Stage, 12 Stage]

Choose emulation of either a 6-stage or a 12-stage phaser. The number of stages affects the number of frequency ranges which get attenuated away. A larger number of stages results in a more nuanced sound.

Manual [10, ..., 10000 Hz]

Sets the base operating point of the phaser – thus, the initial positions of the attenuated frequency ranges, before modulation causes them to be shifted up and down.

Res (Resonance) [0, ... , 127]

Controls the amount of feedback, and thus the intensity of the resonances and comb-filter effects which are produced by the phasing action.

Waveform [Sine, Triangle]

Selects the waveform used for modulation.

Rate [0.01, ... , 40 Hz]

Controls the frequency of the phase modulation.

Depth [0, ... , 127]

Adjusts the modulation depth - the strength of the phase modulation.

Phase [0, ... , 180]

Adjusts the phase difference between the modulation signals applied to the left and right channels. This influences the apparent "breadth" of the stereo image.

Shape [0, ... , 127]

Alters the shape of the modulation signal, accelerating rising signals and slowing down falling signals.

Dry (Dry Level) [0, ... , 127]

Adjusts the level of the original signal.

Wet (Wet Level) [0, ... , 127]

Adjusts the level of the processed signal.

Make sure there is always some of the wet component added to the dry signal. Otherwise the effect will not be audible.

Bypass [Off, On]

This mutes and unmutes the effect.

SSB Phaser

What is SSB?

SSB is an abbreviation for Single Sideband and refers to a modulation process by means of which a signal's component frequencies can be shifted. The shift is the same for all frequencies – for example, if a signal contains components at 440 Hz, 880 Hz, 1760 Hz and 3520 Hz, an SSB modulator can shift each of these frequencies by +10 Hz to 450 Hz, 890 Hz, 1770 Hz and 3530 Hz. Therefore, SSB is also referred to as frequency shifting or spectrum shifting. Until now, SSB has been available only in expensive modular systems such as the Moog Modular.

Frequency shifting is not to be confused with pitch shifting. In pitch shifting, all frequencies are multiplied by the same factor (or, in other words, transposed) and harmonic relationships are preserved.

With SSB, by contrast, harmonic relationships are as a rule transformed into inharmonic ones, as the above example illustrates: whereas 880 Hz is an octave above 440 Hz, 890 Hz (880 + 10) is not an octave above 450 Hz (440 + 10) – the shifted frequencies are no longer harmonically related.

If SSB is used to shift a signal's frequency spectrum only slightly (by less than 1 Hz). When the shifted signal is mixed with the original, an effect not unlike a pahser is produced, but it differs significantly from a phaser in the following respect: whereas a phaser creates phase-cancellations that move up and down through the frequency spectrum, those produced by an SSB phaser move only in one direction, according to the sign of the spectrum shift (plus or minus). Larger shifts produce spectra and sounds similar to those produced by ring modulation.

SSB Phaser			
Shift L	Freq L	Shift R	Freq R >
Range	Feedback	Link	>
Dry	Wet		Bypass <

Shift L /R (Freq Shift L/R) [- 64, ... , 63]

Amount by which all frequencies in the signal spectrum are shifted. Both positive and negative values are allowed.

Freq L /R (Frequency) [- 24000, ... , 24000 Hz] (display only)

Display of the frequency shift resulting from the Range and Shift L/R settings.

Range (Shift Range) [0, ... , 100 %]

Multiplier for the Frequency Shift control – sets its effective range. 100 % permits a shift of -/+24000 Hz, while 0.1 % produces a range of -/+ 24 Hz. By setting tyhe range appropriately, frequency shifting can be tuned very finely.

Feedback [0, ... , 127]

Feedback causes the processed signal to be processed over and over again. An already-shifted frequency is shifted again and again by the same amount. With small frequency shifts, this intensifies the phasing effect, while with large shifts, generation of more inharmonic components is the result.

Link [Off, On]

When this function is activated, the frequency shift can be set simultaneously for both left and right signals.

Dry (Dry Level) [0, ... , 127]

Adjusts the level of the original signal.

Wet (Wet Level) [0, ... , 127]

Adjusts the level of the SSB effect signal.

Make sure there is always some of the wet component added to the dry signal. Otherwise the effect will not be apparent.

Bypass [Off, On]

This mutes and unmutes the effect.

2 Voice Pitch Shifter

In contrast to the Stereo Pitch Shifter, the 2 Voice Pitch Shifter does not pitch-shift the two parts of a stereo signal differently, but instead delivers two different shifts of the same signal.

2-V-Pitch-Shifter			
Coarse A	Fine A	Level A	>
Coarse B	Fine B	Level B	>
Speed	Dry	Wet	Bypass <

Coarse A/B [- 24, ... , 12]

Adjusts the detune value in half-tones. Note that high values here require you to use a correspondingly high *Speed* setting for a clean-sounding result.

Fine A/B [- 100, ... , 100]

Adjusts the detune value in fine increments. The control range is +/- 100 cents, where 100 cents corresponds to one semitone.

Level A/B [0, ... , 127]

Controls the volume level of the effect signal.

Speed [0, ... , 127]

This control influences the quality of the effect. The optimal value depends on the nature of the raw material, so you should experiment with it somewhat. In general, the larger the value, the more exact the calculation, although it may require more processing time.

Dry (Dry Level) [0, ... , 127]

Adjusts the level of the original signal.

Wet (Wet Level) [0, ... , 127]

Adjusts the level of the signal after processing by the effect.

> **Make sure there is always some of the wet component added to the dry signal. Otherwise the effect will not be apparent.**

Bypass [Off, On]

This mutes and unmutes the effect.

Stereo Pitch Shifter

Use the Stereo Pitch Shifter to change the pitch of a signal without altering its duration. The pitch shifter can be used to produce a second voice at an adjustable, fixed interval from the original signal. The pitch for each channel can be adjusted independently, as well as the volume and pan position.

Stereo Pitch Shifter			
Coarse L	Fine L	Level L	>
Coarse R	Fine R	Level R	>
Speed	Dry	Wet	Bypass <

Coarse L/R [-24, ... , 12]

Adjusts the detune value in half-tones. Note that high values here require you to use a correspondingly high *Speed* setting for a clean-sounding result.

Fine L/R [- 100, ... ,100]

Adjusts the detune value in fine increments. The range is +/- 100 cents, where a cent is 1/100th of a half-tone (1/2 tone = 100 cents).

Level L/R [0, ... , 127]

Controls the volume level of the effect signal.

Speed [0, ... , 127]

This control influences the quality of the effect. The optimal value depends on the nature of the raw material, so you should experiment with it somewhat. In general, the larger the value, the more exact the calculation, although it may require more processing time.

Dry (Dry Level) [0, ... , 127]

Adjusts the level of the original signal.

Wet (Wet Level) [0, ... , 127]

Adjusts the level of the signal after processing by the effect.

> **Make sure there is always some of the wet component added to the dry signal. Otherwise the effect will not be apparent.**

Bypass [Off, On]

This mutes and unmutes the effect.

Feedback Pitch Shifter

In the Feedback Pitch Shifter, an adjustable amount of the post-pitch-shift signal is fed back into the shifter and shifted again, resulting in a continually varying frequency content.

FB Pitch Shifter			
Coarse	Fine	Feedback	Speed Γ>
Dry	Wet		Bypass <

Coarse [- 24, ... , 12]

Adjusts the detune value in half-tones. Note that high values here require you to use a correspondingly high *Speed* setting for a clean-sounding result.

Fine [- 100, ... , 100]

Adjusts the detune value in fine increments. The control range is +/- 100 cents, where 100 cents corresponds to one semitone.

Feedback [- 64, ... , 63]

Sets the amount of feedback – thus, the amount of processed output signal which is fed back to the input and reprocessed.

Speed [0, ... , 127]

This control influences the quality of the effect. The optimal value depends on the nature of the raw material, so you should experiment with it somewhat. In general, the larger the value, the more exact the calculation, although it may require more processing time.

Dry (Dry Level) [0, ... , 127]

Adjusts the level of the original signal.

Wet (Wet Level) [0, ... , 127]

Adjusts the level of the signal after processing by the effect.

Make sure there is always some of the wet component added to the dry signal. Otherwise the effect will not be apparent.

Bypass [Off, On]

This mutes and unmutes the effect.

Autopan

When used with stereo signals, the Autopan effect exchanges the left and right channels periodically. With a mono signal, the effect operates as a classic panning effect - the signal sweeps periodically from the left channel to the right and back. The modulation is controlled by an LFO with settings for *Waveform*, *Depth*, and *Rate*.

Autopan			
Rate	Waveform	Depth	Phase >
Shape	Output		Bypass <

Rate [0.01, ... , 40 Hz]

Controls the rate, or speed, of the pan modulation.

Waveform [Sine, Triangle]

Selects the waveform the LFO will use for pan modulation. The following waveforms are available: Sine, Square, Saw Up, Saw Down, and Random.

Depth [0, ... , 127]

Controls the depth, or strength of the pan modulation.

Phase [0, ... , 180]

Controls the difference between the phases of the modulation applied to the two signals of a stereo channel.

Shape [0, ... , 127]

Permits alteration of the waveshape of the modulation signal.

Output [0, ..., 127]

Controls the level of the effect output.

Bypass [Off, On]

This mutes and unmutes the effect.

Tremolo

The Tremolo effect modulates the amplitude of an input signal periodically according to the waveshape selected in the *Waveform* field. The *Depth* parameter controls the degree of modulation.

Tremolo			
Rate	Waveform	Depth	Phase >
Shape			Bypass <

Rate [0.01, ... , 40 Hz]

Controls the rate, or speed, of the amplitude modulation.

Waveform [Sine, Triangle]

Selects the waveform the LFO will use to modulate the amplitude.

Depth [0, ... , 127]

Controls the depth, or strength of the modulation of the input signal's amplitude.

Phase [0, ... , 180]

Shifts the phase of the right channel of the modulation signal relative to the left. The modulation of the right channel follows the left.

Shape [0, ... , 127]

Permits alteration of the waveshape of the modulation signal.

Bypass [Off, On]

This mutes and unmutes the effect.

Auto Wah

This effect uses a multimode filter with a cutoff frequency controlled by an envelope follower. The envelope follower tracks the level of the original (pre-distortion) signal, since the distortion effect "flattens out" the signal dynamics. When the filter resonance is increased, the sounds produced resemble the words "wah-wah", hence the name of the effect. A mono and a stereo version of the effect are provided.

Auto Wah			
Cutoff	Res	Type	>
Attack	Decay	Depth	Bypass <

Cutoff (Cutoff Frequency) [0, ... , 127]

Sets filter cutoff or center frequency, with the current value displayed in Hz.

Res (Resonance) [0, ... , 127]

Sets the amount of filter resonance adjustable from 0 to 127.

Type (Filter Type) [LPF, BPF]

High-pass and low-pass modes can be selected. The filter slope is 12 dB/octave in all modes.

Attack (Envelope Attack) [0, ... , 127]

Controls the response time of the envelope follower to rising levels of the audio signal.

Decay (Envelope Decay) [0, ... , 127]

Controls the response time of the envelope follower to falling levels of the audio signal.

Depth (Envelope Depth) [0, ... , 127]

Some signals are too low to effectively control the filter, while others are too high. Use this control to adjust the influence of the signal over the filter.

Bypass [Off, On]

This mutes and unmutes the effect.

Amplifier

This amplifier is in no way inferior to a genuine amplifier. A sophisticated tube simulation allows for extremely realistic amplifier characteristics from a smooth, round tone to a heavily distorted overdrive. Onboard EQs extend the tonal range for the greatest possible flexibility for sound design.

Amplifier			
In Gain	Boost	Drive	>
Bass	Mid	Treble	
Bass	Mid	Presence	Treble >
Post-EQ	Output	Dist	Speaker >
			Bypass <

In Gain (Input Gain) [0, ... , 127]

Sets the input volume of the amplifier. It is important to supply the amp with a sufficient level without overdriving it. Otherwise reduce the input level to avoid undesirable digital overs.

Boost [No, Yes]

The amplifier has two virtual vacuum tubes. With Boost on, the second tube operates with a higher load in such a way as to provide more distortion.

Drive [0, ... , 127]

Controls the intensity of the distortion. Higher Drive values produce higher volumes, so it is usually necessary to adjust the output volume (OutLev) to compensate.

Bass (Pre-EQ Bass) [- 64, ..., 63]

Controls the portion of the bass signal fed to the tube simulator.

> **Pre EQ is enabled only if the distortion switch is switched on.**

Mid (Pre-EQ Mid) [- 64, ..., 63]

Controls the portion of the mid frequency signal fed to the tube simulator.

> **Pre EQ is enabled only if the distortion switch is switched on.**

Treble (Pre-EQ Treble) [- 64, ..., 63]

Controls the portion of the high frequency signal fed to the tube simulator.

> **Pre EQ is enabled only if the distortion switch is switched on.**

Bass (Post-EQ Bass) [- 64, ..., 63]

Another EQ lies downstream from the Pre EQ to further process the sound. **Bass** permits boost / cut of bass frequencies below approximately 145 Hz.

> **Post EQ remains in effect even when the Distortion is disabled. You can therefore polish up the sound of a clean (undistorted) signal with Post EQ.**

Mid (Post-EQ Mid) [- 64, ..., 63]

Another EQ lies downstream from the Pre EQ to further process the sound. **Mid** permits boost / cut of lower midrange frequencies around approximately 555 Hz.

> **Post EQ remains in effect even when the Distortion is disabled. You can therefore polish up the sound of a clean (undistorted) signal with Post EQ.**

Presence (Post-EQ Presence) [- 64, ..., 63]

Another EQ lies downstream from the Pre EQ to further process the sound. **Presence** permits boost / cut of upper midrange frequencies around approximately 1200 Hz.

> **Post EQ remains in effect even when the Distortion is disabled. You can therefore polish up the sound of a clean (undistorted) signal with Post EQ.**

Treble [- 64, ..., 63]

Another EQ lies downstream from the Pre EQ to further process the sound. **Treble** permits boost / cut of treble frequencies above approximately 1550 Hz.

> **Post EQ remains in effect even when the Distortion is disabled. You can therefore polish up the sound of a clean (undistorted) signal with Post EQ.**

Post-EQ [Off,On]

Permits EQ to be switched on and off for quick with / without comparison.

> **Keep in mind that strong boosting of the signal in all ranges simltaneously can result in internal distortion which cannot be eliminated via the output level control. However, this applies mainly for extreme combinations which are in any case not particularly useful from the sound point of view.**

Output (OutputLev) [0, ... , 127]

Sets the Amplifier's output level. If digital-sounding distortion occurs, you should reduce the level somewhat.

Dist (Distortion) [Off, On]

Enables the tube stage.

Speaker [Off, On]

The guitar amp features a speaker simulator further down the chain. This produces an unsurpassed realism for the amplified sound. You can switch this option off, or use it to produce somewhat more atypical, harsh sounds (effects not necessarily imitative of classic speakers) which can be useful nonetheless.

Bypass [No, Yes]

Switches the Amplifier on or off.

Decimator

The Decimator lets you play a signal at a different bit resolution and/or sample rate than that of the rest of the system. In effect, the signal is internally resampled. Aliasing and quantization noise can be deliberately produced, depending upon the settings used. The sample rate can be modulated by an LFO.

```
Decimator
SampleRate Decimate Bit Depth    Chop  >
 LFO Rate LFO Depth High Damp   Bypass <
```

SampleRate (Sample Rate) [1, ... , 48.000 Hz]

Controls the sample rate which is used for the conversion.

Decimate (Decimate On/Off) [On, Off]

Switches Sample Rate on or off.

Bit Depth [1, ... , 16]

Controls the number of bits which is used for the Bit quantization.

Chop (Chop On/Off) [On, Off]

Switches Bit quantization on or off.

LFO Rate [1, ... , 40 Hz]

Sets the rate at which the sample rate is modulated by the LFO.

LFO Depth [0, ... , 127]

Sets the intensity of the modulation of the sample rate by the LFO.

High Damp (6 dB High Damp) [0, ... , 24.000 Hz]

This filter on the Decimator's output can be used to tone down the high-frequency content.

Bypass [Off, On]

This mutes and unmutes the effect.

Distortion

This effect distorts an input signal. The signal can be processed upstream with a highpass filter and downstream with a lowpass filter. The distortion is produced by adjusting the *Drive* control and the resulting signal can be attenuated as appropriate for output with an output gain control.

Distortion			
Highpass	Drive	Lowpass	Output >
			Bypass <

Highpass (Pre EQ Highpass) [10, ... , 1.000 Hz]

Lets you remove low frequencies from the signal before it is distorted.

Drive [0, ... , 127]

Controls the amount of distortion. The level of the overall signal is also increased along with the distortion.

Lowpass (Post EQ Lowpass) [0, ... , 24.000 Hz]

Lets you remove high frequencies from the signal after it has been processed for distortion.

Output [0, ... , 127]

Because distortion processing increases the level of the signal, you can attenuate it with the output control as necessary.

Bypass [Off, On]

This mutes and unmutes the effect.

Overdrive

This classic effect distorts the signal with the characteristics of an overdriven tube amplifier. A highpass filter limits the effect to the higher frequencies if desired. Because the *drive* control increases the overall signal level, an *output* control is provided at the output to attenuate the signal as required. A parametric equalizer downstream from the distortion stage allows for detailed fine-tuning of the signal.

Overdrive				
Highpass	Drive	Color		Output >
Freq 1	Gain 1	Q1		>
Freq 2	Gain 2	Q2		>
Freq 3	Gain 3	Q3		>
Freq 4	Gain 4	Q4	EQ	Byp >
				Bypass <

Highpass (Pre EQ Highpass) [10, ... , 1.000 Hz]

Use this filter to limit the distortion to the upper frequencies.

Drive [0, ... , 127]

Controls the amount of distortion. The level of the overall signal is also increased along with the distortion.

Color [0, ... , 127]

Adjusts the tone quality of the tube emulator from soft to hard.

Output [0, ... , 127]

Because distortion processing increases the level of the signal, you can attenuate it with the output gain control as necessary.

Freq 1 -4 [20, ... , 20.000 Hz]

Adjusts the frequencies of the respective filters.

Gain 1-4 [- 12 , ... , 12 dB]

Adjusts the filter band's cut or boost level in dB.

Q 1-4 [0.7 , ... , 20]

Sets the quality, or Q factor, for a filter band. This value determines the range of frequencies around a center frequency the filter will act upon.

EQ Bypass [On, Off]

Switches the downstream EQ on or off.

Bypass [Off, On]

This mutes and unmutes the effect.

Resonator

This effect implements a comb filter with optional onboard LFO modulation.

Resonator			
Freq	Res	Damp	Waveform>
Rate	Depth	Phase	Shape >
Dry	Wet		Bypass <

Freq (Frequency) [45, ... , 10000 Hz]

Sets the basic frequency for the filter. This frequency will be modulated by the LFO.

Res (Resonance) [0, ... , 127]

Determines the strength of the comb filter effect. The greater the resonance, the more pronounced the effect.

Damp (Damping) [0, ... , 127]

Controls the resulting overall tone color of the filter. Higher values produce darker, or softer results.

Waveform [Sine, Triangle]

Selects the LFO waveform used to modulate the filter frequency.

Rate [0.01, ... , 40 Hz]

Controls the frequency of the LFO (modulation rate).

Depth [0, ... , 127]

Sets the strength of the modulation of the adjusted filter frequency by the LFO.

Shape [0, ... , 127]

Permits alteration of the waveshape of the modulation signal.

Phase [0, ... , 180]

Controls the difference between the phases of the modulation applied to the two signals of a stereo channel.

Dry (Dry Level) [0, ... , 127]

Adjusts the level of the original signal.

Wet (Wet Level) [0, ... , 127]

Adjusts the level of the signal after processing by the resonator effect.

Bypass [Off, On]

This mutes and unmutes the effect.

Ringmodulator

The Ring Modulator multiplies an audio input signal by an internal sine wave and outputs the result. If you modulate the sine wave using an LFO, all kinds of sonic effects evolve, from subtle spectral changes to more obvious electronic or bell-like effects. Modulation of the sine wave frequency via an envelope is also possible.

Ringmodulator			
Carrier F	Waveform	Rate	Depth >
Env Att	Env Dec	Env Depth	RM Amt >
			Bypass <

Carrier F (Carrier Freq) [1, ... , 1000 Hz]

Controls the basic frequency of the sine wave oscillator. The range is from 1 Hz, to 1000 Hz.

Waveform **[Sine, Squ, Saw Up, Saw Down, Triangle, Random]**

Selects the waveform the LFO will use to modulate the sine wave.

Rate (LFO Rate) [0.01, ... , 40 Hz]

Controls the rate at which the LFO modulates the sine wave oscillator.

Depth (LFO Depth) [0, ... , 127]

Sets the modulation depth. This controls the amount of deviation from the basic frequency the sine wave oscillator will produce.

Env Att (Env Attack) [0, ... , 127]

When the sine wave is modulated by the envelope follower, this control sets the speed at which the envelope follower reacts to increasing signals.

Env Dec (Env Decay) [0, ... , 127]

When the sine wave is modulated by the envelope follower, this control sets the speed at which the envelope follower reacts to decreasing signals.

Env Depth [0, ... , 127]

Sets the depth of modulation of the sine wave by the envelope follower.

RM Amt (RM Amount) [0, ... , 127]

Controls the volume of the ring modulator effect.

Bypass [Off, On]

This mutes and unmutes the effect.

Softclip

The Softclip module permits the loudness of a signal to be boosted without introducing digital overload distorion (hard clipping). Even with signals which are already at the maximum level, an apparent loudness increase can be produced. In addition, the sound takes on a warmer, more "analog" character.

Softclip		
Drive	Output	Bypass

Drive [0, ... , 100%]

Adjusts the intensity of the effect.

Output [- 186.6, ... , 0 dB]

If you wish to alter the sound of a signal and not its loudness, you can use this control to counteract the volume increase produced by the effect.

Bypass [Off, On]

This mutes and unmutes the effect.

Stereo Expander

The Stereo Expander increases or decreases the perceived "width" of a stereo image. The processing is mono-compatible and frequency-independent.

Stereo Expander	
Amount	Bypass

Amount [- 100, ... , 100]

Adjusts the intensity of the effect. Positive values produce a widening of the stereo image; negative values cause it to become narrower.

Bypass [Off, On]

This mutes and unmutes the effect.

Tube Processor

This classic effect distorts the signal with the characteristics of an overdriven tube amplifier. A highpass filter limits the effect to the higher frequencies if desired. Because the *drive* control increases the overall signal level, an *output* control is provided at the output to attenuate the signal as required.

Tube Processor			
Highpass	Drive	Color	Lowpass >
Output			Bypass <

Highpass (Pre EQ Highpass) [10, ... , 1.000 Hz]

Use this filter to limit the distortion to the upper frequencies.

Drive [0, ... , 127]

Controls the amount of distortion. The level of the overall signal is also increased along with the distortion.

Color [0, ... , 127]

Adjusts the tone quality of the tube emulator from soft to hard.

Lowpass (Post EQ Low) [0, ... , 24.000 Hz]

Permits removal of high frequencies from the signal after it has been distorted.

Output [0, ... , 127]

Because distortion processing increases the level of the signal, you can attenuate it with the output gain control as necessary.

Bypass [Off, On]

This mutes and unmutes the effect.

Compressor

The compressor modifies the dynamics of a sound by decreasing the volume level of the signal when it becomes louder. This means that the overall level of the sound can be set higher, with the net result that the level of softer sections is increased. The compressor operates by monitoring the level of the input signal. An adjustable threshold level determines the point at which compression begins. Attack and release controls determine how quickly the compressor responds when the threshold is exceeded and when the input signal level falls back below the threshold respectively. The ratio control determines the intensity of the compression – i.e., the ratio of input level change to output level change. The gain control adjusts the level of the compressed signal.

Compressor			
Attack	Release	Theshold	Ratio >
Gain			Bypass <

Attack [1, ... , 100 ms]

The *attack* time is the compressor's reaction time; the time it takes for it to respond to an increas in level above the threshold.

Release [1, ... ,1000 ms]

This is the time after the signal falls back below the threshold that compression is no longer active.

Threshold [- 60, ... , - 0 dB]

Sets the input signal level above which compression begins.

Ratio [1:1, ..., inf:1]

The *ratio* adjusts the degree of compression for signals that exceed the threshold level. Compression is shown as a relative value. 1:1 means no compression. 3:1 means that a rise of +3dB in the input signal results only in a +1dB increase at the output.

Gain [- 168, ... , 18 dB]

This increases the overall level of the signal, resulting in a fuller sound. Adjust the volume with the *gain* control.

Bypass [Off, On]

This mutes and unmutes the effect.

Expander

The Expander influences the dynamics of a sound by making quiet passages even quieter while leaving other passages alone. Thus the overall dynamic range of the signal increases. You can use this to alter the way an instrument sound decays - for example, to alter a looped drum pattern as it fades out, or to blend undesirable low-level background noise into the signal noise floor. The Expander is provided in both mono and stereo versions. Threshold sets the level at which the effect begins to process the signal. The attack and release controls determine how quickly the effect engages or disengages when the threshold is crossed. Ratio controls the degree of expansion - how much the volume range is increased by the effect.

Expander				
Attack	Decay	Theshold	Ratio	>
Gain			Bypass	<

Attack [1, ... , 100 ms]

The *attack* time (in milleseconds) is the expanders's reaction time - the time it takes it to respond when the level falls below the threshold.

Decay [1, ... ,1000 ms]

Adjusts the speed with which the expander returns to its original level after the input signal exceeds the threshold.

Threshold [- 80, ... , 0 dB]

Sets the input signal level below which expanding engages.

Ratio [1:1, ..., inf:1]

The *ratio* adjusts the compression rate for signals that falls below the threshold level. The compression is displayed as a relation value. 1:1 means that there is no expansion. 3:1e.g. means that a an attenuation of -1dB of the input signal results in -3dB at the output. The maxium Ratio is 10:1.

Gain [- 168, ... , 18 dB]

The output level of the expander can be adjusted and amplified up to +18dB.

Bypass [Off, On]

This mutes and unmutes the effect.

Limiter

This effect is similar to compression and likewise modifies the dynamics of a signal. The level of louder passages is decreased. This means that the overall level of the sound can be set higher, with the net result that the level of softer sections is increased. The limiter operates by monitoring the level of the input signal. An adjustable threshold level determines the point at which limiting begins. Attack and release controls determine how quickly the limiter responds when the threshold is exceeded and when the input signal level falls back below the threshold, respectively. The ratio control determines the intensity of the limiting – i.e., the ratio of input level change to output level change. The gain control adjusts the overall level of the output signal.

Limiter				
Attack	Release	Theshold	Ratio	>
Gain			Bypass	<

Attack [1, ... , 100 ms]

The *attack* time (in milleseconds) is the limiter's reaction time - the time it takes it to respond to a level spike.

Release [1, ... ,1000 ms]

This is the time (in milliseconds) after the signal falls back under the threshold that limitting is no longer active.

Threshold [- 60, ... , 0 dB]

Sets the input signal level above which limitting begins.

Ratio [1:1, ..., inf:1]

The *ratio* adjusts the reduction rate for signals that exceed the threshold level. The compression is displayed as a relation value. 1:1 means that there is no reduction. 3:1 means that a an increase of +3dB of the input signal results only in +1dB of gain in the output.

Gain [- 168, ... , 18 dB]

This increases the overall level of the signal, resulting in a fuller sound. Adjust the volume increase with the Gain control.

Bypass [Off, On]

This mutes and unmutes the effect.

Gate

A *gate* is just what its name says - a gate, or door. When it is open, signals pass freely. When closed, signals are blocked. A typical gate will open when a specific volume threshold is reached, and remain open for a certain specified time. If the threshold is not reached again during that time, it will close. A gate can serve several purposes, such as muting the signal to avoid background noise (noise gate, instrument separation etc.) or to clean up the trailing out of an instrument sound.

Gate			
Attack	Hold	Release	Theshold>
Floor	Hyst	Gain	Bypass <

Attack [1, ... , 100 ms]

Amount of time for the gate to open once the input signal level has exceeded the threshold.

Hold [1, ... ,1000 ms]

Minimum amount of time the gate will stay open once it is triggered (i.e., minimum delay between end of attack and start of release).

Release [1, ... ,1000 ms]

Amount of time gate takes to fully close once it begins to close (i.e., once the input signal level falls below the threshold, and after any remaining hold time has elapsed).

Threshold [- 96, ... , 0 dB]

Sets the input signal level in dB above which the gate will open. This is the turn-on (upper) threshold, shown in the display as a yellow point. The turn-off (lower) threshold is adjusted in tandem with the turn-on threshold, with an offset determined by the Hysteresis control. With the Stereo Gate, opening of the gate is determined by the louder of the two channels.

Floor [-9 6, ... , 0 dB]

Sets the level to which gate gain drops when the gate closes. The gate can thus be set so that it does not close completely.

Hyst (Hysteresis) [- 10, ... , 0 dB]

Adjusts the difference between the turn-on (upper) and turn-off (lower) thresholds. The turn-off threshold appears as a red point in the display. It can be set up to 10 dB below the turn-on threshold.

Gain [- 168, ... , 18 dB]

With the *gain* control you can increase the gate output level by up to 18dB.

Bypass [Off, On]

This mutes and unmutes the effect.

Dynamics

This module unites an expander, a compressor and a limiter, thereby permitting you to use all three of these effects at one time without needing to use more than one effects slot. For each of the three effects there is a corresponding submenu containing the effect's parameters. These parameters are identical to those of the expander, compressor and limiter effects described earlier in this section and are therefore not described here. The only difference is the Bypass option:

Bypass [Off,On]

Permits each of the three effects to be individually switched on or off.

Sources Menu (Preset Sources)

The effect settings for an Instrument are usually stored within the presets for the respective Instrument.

This means that in *Multi* Mode you have to specify the Instrument whose settings you want to use. Because in *Multi* Mode only two Insert effects can be loaded simultaneously, it is not possible to load the effects from all Instrument presets.

Switching to another Instrument while working in Multi Mode would reload this Instrument's effects and thereby replace the current effect settings. There may, however, be times when you are happy with the current effect settings but would like to change an Instrument.

That is why in *Multi* Mode there are two different options for Insert and Aux effects:

a) When you load a preset, the Insert effects stored within it are automatically loaded as well. The current Insert effects will be replaced when you switch to another Instrument preset.

b) When you load a preset, the Insert effects stored within it are *ignored*. So even when changing Instruments, the currently selected Insert effects remain the same.

```
Preset Sources
 Inserts:   Multi      AuxFX:    Multi
```

Inserts: [Multi, Slot1, Slot2, Slot3, Slot4]

You can choose if you want to use the settings from the *Multi* configuration or from one of its Instruments for the Insert Effects. In the second case, specify the respective Instrument by selecting the slot into which it has been loaded.

Aux FX: [Multi, Slot1, Slot2, Slot3, Slot4]

You can choose if you want to use the settings from the *Multi* configuration or from one of its Instruments for the Aux Effects. In the second case, specify the respective Instrument by selecting the slot into which it has been loaded.

The FX Bypass Menu – Muting Effects

The *FX Bypass* Button (18) allows you to mute all effects. Keeping the button pressed for a few seconds will open the corresponding menu, where you can mute the two Insert effects and/or the Aux effects individually.

```
FX Bypass
 Ins1Byp   Ins2Byp    AuxByp   Enabled
```

Ins1Byp [Off, On]

Mutes and unmutes Insert-Slots 1.

Ins2Byp [Off, On]

Mutes and unmutes Insert Slot 2.

AuxByp [Off, On]

Mutes and unmutes all Aux effects.

***Bypass All* [Disabled, Enabled]**

Mutes and unmutes all effects.

The System Menu

Parameter

Open the System Menu by pressing the *System* Button (14).

Here you will find the parameters for the global Noah configuration and for managing MIDI Echo, MIDI Clock, MIDI Controllers and Memory.

The parameters are organized into four sub-menus: MIDI, Controls, Device and Memory.

```
System
  >MIDI    >Controls   >Device    >Memory
```

MIDI sub-menu

Here you can configure the MIDI Echo. You can also select a MIDI Clock source and assign the Clock to an output.

```
MIDI
 MIDIecho  USBecho   ClockSrc  ClockOut
```

MIDIecho (MIDI In Echo)
[none / MIDIOut / USB Out / MIDI/USB]

Controls pass-through of MIDI messages received via the MIDI input to one or both of the MIDI outputs (None / Midi Out / USB Out / MIDI/USB).

USBecho (USB In Echo)
[none / MIDIOut / USB Out / MIDI/USB]

Controls pass-through of MIDI messages received via the USB MIDI input to one or both of the MIDI outputs (None / Midi Out / USB Out / MIDI/USB).

ClockSrc (Clock Source) [Internal, MIDI In, USB In]

Here you can choose if you want the tempo for the Arpeggiators and the Step Sequencer to be generated internally. If not, select the MIDI input from which the external MIDI Clock should be received.

ClockOut [On, Off]

Here you can choose if the MIDI Clock generated by an Arpeggiator or Step sequencer should be sent to Noah's MIDI output or not.

Controls sub-menu

Lets you set specify how Noah handles MIDI Controller and SysEx messages.

```
Controls
   Send     Receive
```

Send [none, CC , SysEx, CC/SysEx]

Enables or disables sending of MIDI controller (CC) and System Exclusive (SysEx) messages .

Receive [none, CC , SysEx, CC/SysEx]

Enables or disables receive of MIDI controller (CC) and System Exclusive (SysEx) messages.

Device sub-menu (System Device)

Here you can set Noah's startup configuration, and the word clock source. In the upper right corner you can see which version of the operating system you are currently using.

System/Device V 1.00
Wake Up NoahID ClkSrc Locked

WakeUp [Default, Last]

You can choose if you want Noah to boot the original configuration (Default) or the last configuration you used.

The Default configuration refers to the currently loaded Module, Instruments and presets. If you have changed any of the presets of the original configuration, **WakeUp** will load these changed presets.

NoahID [0-127]

If you are using several Noah units, you can assign different ID numbers to them. The software will use these ID numbers to address the different Noah units.

Future versions of the software will include support for several Noah units.

ClkSrc [Internal, Ext BNC]

Controls whether Noah generates its own wordclock (Internal) or synchronizes to that of another digital audio device whose wordclock output is connected to Noah's BNC wordclock input (3).

***Display: Wordclock Sync Status* [Locked, No Lock]**

Indicates whether Noah is able to synchronize to an external wordclock (*Locked*) or not (*No Lock*). If *Internal* is selected under **ClkSrc**, this display will always indicate *Locked*.

Memory sub-menu

Displays the amount of free space in Noah's internal storage or in an inserted Compact Flash Card, depending upon whether you select *Internal Flash* or *Compact Flash* for display.

Memory
>Internal Flash >Compact Flash

Internal Flash / Compact Flash

Both menus have the same structure.

Internal Flash
Free: *Value* kb (Optimize)

Free [display only]

Displays the amount of free space.

(Optimize)

Pushing the associated rotary control causes Noah to attempt an optimization of its memory contents. This can free up additional space for storage of presets.

The Utility Menu

Press the *Utility* Button (15) to open the *Utility* menu. Here you will find the options for managing presets.

The parameters of the Utility menu can be found in the *Preset* sub-menu.

Utility
>Presets

Presets sub-menu

This section provides only a brief description of the available options.

For more information see: *Operating Noah: Presets*.

Edit presetlist of > <

Use the Control Wheel (11) to select the preset file you want to edit. The functions you can access from the lower display line will be applied to the selected file.

Add Bank Menu

Add Bank			< Factory >
(Write)	(Back)	(Exit)	

Use this menu to create a new bank within a previously selected preset file.

First, enter the name for the new bank in the upper right corner of the display. Use the Control Wheel (11) to choose the desired characters.

For more information see: *Operating Noah: Presets*.

(Write)

Create the new bank by pressing the Continuous Controller (5) below (*Write)*.

(Back)

Press the Continuous Controller (5) below (*Back*) to leave the *Add Bank* menu and return to the menu one level above.

(Exit)

Press the Continuous Controller (5) below (*Exit*) to leave the *Utility* menu.

Del Bank Menu (Delete Bank)

Delete Bank			M---:
(Write)	(Back)	(Exit)	

Use this menu to delete a bank from a previously selected preset file. Use the Control Wheel (11) to select the bank you want to delete in the upper right corner of the display.

For more information see: *Operating Noah: Presets*.

(Delete)

Delete the selected bank by pressing the Continuous Controller (5) below (*Delete)*.

(Back)

Press the Continuous Controller (5) below (*Back*) to leave the *Delete Bank* menu and return to the menu one level above.

(Exit)

Press the Continuous Controller (5) below (*Exit*) to leave the *Utility* menu.

Ren Bank Menu (Rename Bank)

```
Rename Bank            <              >
 (Write)    (Back)    (Exit)
```

Use this menu to rename a bank in a previously selected preset file. Edit the name in the upper right corner of the display. Use the Control Wheel (11) to select the desired characters.

For more information see: *Operating Noah: Presets*.

(Write)

Rename the bank by pressing the Continuous Controller (5) below (*Write)*.

(Back)

Press the Continuous Controller (5) below (*Back*) to leave the *Delete Bank* menu and return to the menu one level above.

(Exit)

Press the Continuous Controller (5) below (*Exit*) to leave the *Utility* menu.

Del Pres Menu (Delete Preset)

```
Delete Preset   P---:
 (Write)    (Back)    (Exit)
```

Use this menu to delete a preset from a previously selected preset file. The display will show the bank that contains the currently loaded preset.

Use the Control Wheel (11) to select the preset you want to delete in the upper right corner of the display.

For more information see: *Operating Noah: Presets*.

(Delete)

To delete the preset, press the Continuous Controller (5) below (*Delete)*.

(Back)

Press the Continuous Controller (5) below (*Back*) to leave the *Delete Bank* menu and return to the menu one level above.

(Exit)

Press the Continuous Controller (5) below (*Exit*) to leave the *Utility* menu.

Arpeggiator

Introduction

The Arpeggiator presents you with tremendous possibilities:

- Use it to effortlessly (and interactively) generate melodic sequences in real time – with myriad control options, all of which can be adjusted at any time to yield practically endless variations – for direct use, to stimulate your own musical imagination, or to advance the frontiers of musical-acoustic research!

- Even for accomplished keyboardists, for whom keyboard technique is not an issue, the Arpeggiator is a *fabulous* labor-saving device!

- The Arpeggiator is a highly dependable tool for bringing *any* relationship (musical *or* non-musical) to a speedy end. It's therefore also a sure-fire vehicle for getting your solo career (musical *or* non-musical) off to an early start!

But seriously ...

No idea what an arpeggiator is? In simplest terms, it's a device which *captures* chords (or short sequences of notes) that you play into it, and then *scans* the captured notes, periodically sending them out one at a time, thus generating *arpeggios* from your input.

The results of this simple process can be annoying and banal, or truly esoteric. Quite a lot depends upon the sounds you apply it to and the contexts you use it in. We won't presume to prescribe these things for you. All options are open. The Arpeggiator includes enough features and flexibility to enable you reach *any* extreme!

Many features of the arpeggiator are easier to understand intuitively than by reading about them. If you can't make heads or tails of something you read here – just go ahead and try it out. You can't damage anything. Then come back here and reread the section in question and it'll probably make perfect sense.

If you're just getting acquainted with the Arpeggiator, and if you already have a basic idea of what an arpeggiator is, there's nothing at all wrong with going straight into hands-on experimentation – you can refer back to the manual to go more in-depth or for clarification of details as needed. But *do* read the manual sometime – there are features here you wouldn't want to miss out on!

Overview

Basics

This is a MIDI arpeggiator device:

- It is driven by MIDI note events on its **MIDI** input, which it *captures* in its internal chord buffer.

- It continually *scans* this chord buffer according to the current settings and produces MIDI note events as output.

> (Note: in this text, MIDI note event and other input is referred to as coming from a MIDI keyboard, although the arpeggiator can of course be driven by *any* live or sequenced source of note events.)

The arpeggiator is monophonic. It puts out one note at a time, no overlapping notes and no chords.

MIDI Message Handling

The handling of received MIDI messages depends on their type. In general:

- Received **note-on messages** are captured in the chord buffer (up to sixteen at any one time).

- Received **note-off messages** may or may not cause the matching note-on messages to be removed from the chord buffer, depending upon the current settings.

- *Received* note messages are *not* echoed to the output while the arpeggiator is running. The note events which the arpeggiator sends out are primarily those which the arpeggiator itself *generates* via scanning of the captured chord.

- All other received *channel messages* (e.g., modwheel, pitch bend, etc.) are echoed directly to the output at all times and have no particular pre-defined effect upon the arpeggiator.

Control via the MIDI Keyboard

A range of eight **MIDI** note numbers can optionally be dedicated to real-time control of a selected set of arpeggiator functions via the MIDI input (see **Menu Control** and **Menu KeybCtrl**) . This permits these controls to be used more effectively in real time, as the response via the MIDI keyboard.

Timing and Synchronization

The arpeggiator's timing resolution is 24 clocks or pulses per quarter-note (PPQN). The note-on and note-off events it produces are aligned to these time increments.

The arpeggiator does not have its own clock output. It can, however, be synchronized to a sequencer or other device by selecting an external MIDI clock as the sync source in the MIDI menu.

Activating the Arpeggiator

You can open up to four instances of the Arpeggiator, so each slot can have it's own.

You can now select an Arpeggiator as the Trigger Source for an instrument slot. In Edit Mode, select the Arpeggiator (Arpeg) from the Menu MIDI / *Instrument* / Trigger (*Single* mode) or MIDI / Devices / Slot 1 (2,3,4) / Trigger 1 (2,3,4) (*Multi* mode). It is the first menu item in the bottom line.

Parameter

You can now select an Arpeggiator as the Trigger Source for an instrument slot. In Edit Mode, select the Arpeggiator (Arpeg) from the Menu MIDI / *Instrument* / Trigger (*Single* mode) or MIDI / Devices / Slot 1 (2,3,4) / Trigger 1 (2,3,4) (*Multi* mode). It is the first menu item in the bottom line.

After selecting the arpeggiator, push the rotary controller (5) below the selection to access the arpeggiator's parameters.

The Arpeggiator parameters are distrubted among 8 menus (**Control, Scan, Capture, Note, Output, LFO, KeybCtrl, OutTimin**) and their sub-menus:

```
Arpeggiator
 >Control   >Scan    >Capture   >Note   >
 >Output     >LFO    >KeybCtrl >OutTimin<
```

Control Menu

This menu contains the functions that you can switch on or off with the associatd controls (5). These are essentially the "running" controls. Viewed as a group, they produce immediate, "big" changes. This distinguishes them from *settings* which can likewise be adjusted at any time, but whose effects are less "major" and not always immediately audible.

Each of the functions can also be actuated from the MIDI keyboard (see KeybCtrl Menu). This is generally a better way to go, as the response of these functions to MIDI control is virtually instantaneous. Press one of the Continuous Controllers (5) to see the currently associated MIDI note numbers. Change this association with the **CtrlBase** parameter in the **KeybCtrl** Menu.

```
Control
  (Run)    (Clear)   (Hold )   (HTrnsp )>
(ScanDir)  (Rescan)                     <
```

Run/Stop

This is the On/Off button – it alternately activates and deactivates the arpeggiator. Stopping the arpeggiator clears the chord buffer and terminates the current output note, if any. Note that when the arpeggiator is not running, it passes *all* received MIDI events directly through to its output.

Clear

Pressing this button "empties" the arpeggiator (i.e., clears the chord buffer), permitting a completely new chord to be captured. The arpeggiator continues to run as before.

> **Clear** also deactivates **Hold** and **HTrnsp** if they are active (see below).

Hold

This function freezes the chord buffer, locking the current captured chord into the arpeggiator. Incoming MIDI note events are no longer captured, nor can they cause the notes which have already been captured to be removed. Instead, received MIDI note events are passed directly through to the output. This lets you "accompany" the arpeggiator live.

> Once activated, **Hold** can be deactivated only via **Clear** or **Run/Stop**.
>
> Note: **Hold** is also activated automatically whenever **Htrnsp** (see below) is activated.

Htrnsp

Activating **Htransp** will instantly activate **Hold** (see above) if it is not already active, thus freezing the chord buffer. While **Htransp** is active, the arpeggiator output can be "live-transposed" up or down (simple semitone transpose) from the MIDI keyboard.

The transpose produced by playing any note on the keyboard is equal to the offset of this note relative to middle C (MIDI 60). While **Htransp** is active, the keyboard has no effect upon the captured chord, other than transposing it as described.

However, **Htransp** (unlike **Hold**) can be activated and deactivated freely. When **Htransp** is deactivated, **Hold** remains active, as does the last transpose value applied under **Htransp**. This allows switching between transposition and accompaniment of the frozen (and still-transposed) chord.

The transpose produced under **Htransp** is cleared via **Clear** or **Run/Stop**, and is thus always zero whenever **Hold** or **Htransp** is first activated.

ScanDir

> Note: this function is effective *only* when **Scan Mode** is set to **Fwd-Rev**.

This function causes a reversal of the instantaneous arpeggiator scan "direction".

For example, when the arpeggiator is producing notes going up the scale/keyboard, **ScanDir** will cause it to reverse and go down the scale instead, beginning with the next output note – without affecting the output timing in any way.

Rescan

This function "restarts" the arpeggiator scan each time it is actuated. The next arpeggiator output note is the one which should appear "first", based on the current **ScanMode** and other settings (for example, the lowest note in the current captured chord).

Scan Menu

Scan			
Mode	Pattern	RndDepth	>
AutorScn	RscnMode	RscnLen	<

Mode (Scan Mode) [NNumber, NOrder]

This control selects the basic method used to scan the captured chord and determine the next note to be played. The currently selected scan pattern (see below) produces a specific variation on the selected scan mode. The scan mode setting can be changed at any time.

NNumber (Note Number): Scanning of the captured chord is done on the basis of note numbers – e.g., from lowest note to highest note.

NOrder (Note Order): Scanning of the captured chord is done on the basis of the time sequence in which the notes were captured.

Pattern (Scan Pattern) [**Forward / Reverse / Fwd-Rev / Random**]

These controls determine the specific way in which the captured chord is scanned. These settings work in tandem with the **Mode** setting. The **Pattern** setting can be changed at any time.

Forward: The captured chord is scanned in order of increasing note number (**Scan Mode** set to **Note Number**) or in forward time sequence (**Scan Mode** set to **Note Order**).

Reverse: The captured chord is scanned in order of decreasing note number (**Scan Mode** set to **Note Number**) or in reverse time sequence (**Scan Mode** set to **Note Order**).

Fwd-Rev: Scanning of the captured chord scan alternates between **Forward** and **Reverse** as described above, reversing itself each time it reaches the "end" of a scan (highest/lowest or first/last note). The notes at either "end" of a scan are not repeated when the direction reverses (i.e., these notes get played only once, not twice).

Random: Chord scan follows a random pattern. The **Depth** control, which is enabled when you select Random, sets the degree or range of randomness.

Random scan behaves differently depending on the **Scan Mode** setting.

- If **Scan Mode** is set to **Note Number**, then **Random** scan is basically a variation on normal **Fwd-Rev.** The scan proceeds in single steps from one note number to the next without skipping over any notes, but reverses its direction at random. With **Random Depth** set to minimum, this mode is in fact equivalent to **Fwd-Rev.** With **Random Depth** set to maximum, the scan reverses itself after almost every note (and thus tends to "stick in place", continually alternating between two notes).

- If **Scan Mode** is set to **Note Order**, then **Random** scan selects output notes randomly from among all possible output notes, taking the captured chord and all other settings (including **Octave Extend** – see **Output**) into consideration. The **Depth** control confines this selection to a specific range of "scan steps" away from the previous note in either direction. With **RandomDepth** set to minimum, the arpeggiator "sticks" on a single note. With **RandomDepth** set to maximum, the arpeggiator may select any note within 16 scan steps of the previous note – in effect, almost totally random.

RndDepth (Random Depth) [0, ... , 100 %]

This parameter determines the degree of randomness in Scan mode (see previous comments re: *Random*).

AutoRScn (Auto ReScan) [Off, On]

If this function is enabled, then the number of notes in the arpeggio is maintained according to the ReScan Length parameter.This is useful for imposing a specific rhythmic count or "loop length" on the output which does not depend upon the number of captured notes or other scan settings.

The auto rescan counter is reset whenever new notes or chords are played on the MIDI keyboard (see also **Rescan Mode** below) – however, *not* while **Hold** or **HTrns** is active. It is also reset whenever the manual **RESCAN** function is actuated (via the graphical button or the corresponding MIDI key).

The auto rescan counter can also serve as a resynchronization source for the LFO.

RscnMode (ReScan Mode) [NewChords, NewNotes]

This settings specifies whether the **AutoReScan** counter should be reset whenever any new note is played on the MIDI keyboard **(NewNotes**) or only when new *chords* are played (**NewChords**).

The difference between these two choices is that with **NewChords**, only the first note which is played *after all keys have been released* (e.g., the first note of a new chord) will cause the counter to be reset, whereas with **NewNotes**, every note which is played will cause the counter to be reset, even if other keys are still being held down.

For example: Using **Normal** Capture mode, you can hold down the keys of a chord, and from time to time release individual keys or hold down new ones to modify the arpeggio while it plays. With **Rescan Mode**: **NewChords**, adding of new notes can be done at any time without changing the rhythm or shifting the "downbeat" of the resulting arpeggio. With **Rescan Mode**: **NewNotes**, each new note you play redefines the downbeat to that point in time, cutting the current "bar" short and immediately beginning a new one – thus allowing you to redefine the dowbeat as you go, follow time-signature changes on the fly, etc.

Note that the **Mode** also affects keyboard-driven resynchronization of the LFO (when the option *NewChords, NewNotes* is activated – see **LFO -Resync Menu**).

RscnLen (ReScan Length) [NewChords, NewNotes]

If **Auto ReScan** enabled, this parameter limits the number of notes to scan to the value set here.

Capture Menu

Capture	
CaptMode	Ext2Len

CaptMode (Capture Mode) [Normal, Auto, Extend1, Extend2]

These settings control the way in which received MIDI notes are *captured* in (and removed from) the chord buffer. One of the four possible modes is always active. In all capture modes, the arpeggiator keeps track of the order in which captured notes arrived, in addition to recording note number (of course) and velocity for each note. Thus, the use of both **Note Number** and **Note Order** scan modes is always possible, regardless of the **Capture Mode** setting.

The capture mode setting can be changed at any time. In some cases, after doing this, you may need to use **Clear** to empty the chord buffer completely (e.g., if you change from **Auto** to **Normal** after releasing all keys).

Normal: In this mode, captured notes remain in the chord buffer only for as long as the corresponding key is held down. The arpeggiator output pattern varies dynamically as keys are released. When no keys are being held down, the arpeggiator produces no output.

Auto: Under **Auto**, captured notes remain in the chord buffer indefinitely, even after all held keys have been released. The arpeggio continues to play as if all keys which have been played were still being held. New notes continue to be captured as long as at least one key is still being held. The first note or chord which is played following the release of all keys begins a new capture "session", at the same time clearing all previously captured notes.

Extend 1: Notes are added to the chord buffer as they are played and remain in the chord buffer indefinitely. This capture mode permits notes to be added one at a time. **Extend** mode thus makes it easy to create melodic arpeggios – a particular note can appear multiple times at different points within the arpeggio.

Note capture continues until the chord buffer is full (sixteen captured notes). The **Clear** button must be used to clear the chord buffer and silence the arpeggiator, or to capture a new chord.

> Note that **Scan Mode** (see previous section) must be set to **Note Order** in order to have the arpeggiator play the notes back in the order in which they were captured.

Extend 2: Same as **Extend 1,** but new notes push old notes out of the chord buffer once the specified note capture limit ***n*** (adjustable via **Ext2Len**) has been reached. Note capture can thus continue indefinitely – the chord buffer always contains (at most) the last *n* notes captured. Therefore, the rhythm of the arpeggiator output also remains fixed as new input notes are captured, once the limit has been reached. Via **Ext2Len** the note capture limit can be set to any value between 2 and 16 notes and can be adjusted while the arpeggiator is running.

Ext2Len (Extend2 Length) [2, ... , 16]

Configures the maximum number of notes to maintain in the capture buffer in Extend2 Mode.

Note Menu

Note				
NoteLen	GateDur	On=	Off=	>
VelMode	LFOMod		MaxVel	<

NoteLen (Note Length)

[**8/1, 4/1, 2/1,1/1, 1/2 dot, 1/2, 1/2 trpl, 1/4 dot, 1/4, 1/4trpl, 1/8 dot, 1/8, 1/8 trpl, 1/16 dot, 1/16, 1/16 trpl, 1/32, 1/32 trpl, 1 Clock**]

Adjusts the duration (length) of the notes played back.

This setting also directly influences the LFO speed if the **Speed Type - Clock** option is activated in the **LFO** menu (see **LFO** menu).

GateDur (Gate Duration) [0, ... , 100 %]

This sets the proportion of each arpeggiator beat during which the output note remains in the gate-on (sustain) state. Adjustment of this setting through its full range from minimum to maximum produces a gradual change in the arpeggiator output from staccato to legato. The number of clocks for gate-on and gate-off phases, respectively, are displayed for reference – these values cannot be adjusted directly.

Note that this control will have no effect at the minimum from *Note Length*, since in that case, the only available possibility is one clock for gate-on and one for gate-off.

On= / Off= (Note On Width, Note Off Width) [*display only*]

Dispalys the number of clocks between note-on and note-off according to the configured Note Length and Gate Duration parameters.

VelMode (Velocity Mode) [Original, Replace]

These settings provide control over the note-on velocity of arpeggiator output notes (Note-On-Velocity).

Original / Replace: One of these two options is always in effect:

- When **Original** is selected, output notes have the note-on velocity of the captured note from which they were generated (and the other velocity controls described below have no effect).

- When **Replace** is selected, the velocity values of the captured notes are ignored. Instead, output note velocity is determined by the following controls. It is possible to switch back and forth freely between these two options – the actual captured notes are not affected in any way and the original velocity values are always available.

This does not affect the already-processed notes, whose original dynamics can be recalled at any time.

LFO Mod (Lfo Modulation) [0, ... , 127]

This setting regulates modulation of output note-on velocity by the arpeggiator's built-in LFO. This setting is effective only when **Velocity Mode: Replace** is selected (see above).

LFO modulation causes the velocity values of arpeggiator output notes to vary over time. It functions by decreasing velocities relative to the **Max Velocity** value (see following). In effect, the **LFO Modulation** control sets the minimum modulated velocity.

For example:

- At the maximum **LFO Modulation** setting, velocity values vary between the **Max Velocity** setting and the absolute minimum possible MIDI velocity value of 1.

- At the "halfway" setting, by contrast, velocity values still reach **Max Velocity** at the high end of the modulation swing, but at the low end, they go only half as far down – i.e., to roughly one half of **Max Velocity**.

MaxVel (MaxVelocity) [1, ... , 127]

Adjustable between 1 and 127. This setting is effective only when **Velocity Mode: Replace** is selected (see above).

- When **LFO Modulation** (see below) is set to zero, this setting *directly* specifies the velocity of output notes, which is then constant.

- Otherwise, it sets the maximum velocity value which modulated output notes may have. Modulation by the LFO results in time-varying velocity values which are lower than this maximum.

(By the way – the **MaxVelocity** control is *another* excellent candidate for assignment to a MIDI controller.)

Output Menu

Output			
OctvExt	Repeat	SweepTr	>
DropNorm	Drop Alt	Sw: 3	RestNote<

OctvExt (Octave Extend) [0, ... , 127]

Produces cyclical upward transposition of the arpeggiator output by one or more octaves. The transpose amount is automatically "stepped" by one octave each time the arpeggiator completes a scan of the captured chord in the current scan direction. The captured chord is thus effectively extended into additional higher octaves as if the actual notes in the captured chord had been duplicated in those octaves. Setting **Octave Extend** to zero disables it.

"Stepping" of the **Octave Extend** transpose amount is always done in a manner which is consistent with the selected scan pattern. Assuming that **Octave Extend** is enabled (i.e, it is set to 1 or higher):

- With **Scan Pattern** set to **Forward**, output transpose is stepped upward by one octave following each pass through the captured chord until the until the scan in the highest octave (as specified by the **Octave Extend** setting) is complete. Transpose is then reset to 0 and the cycle repeats.

- With **Scan Pattern** set to **Reverse** , output transpose is stepped downward by one octave following each pass until a scan with a transpose of 0 is complete. Transpose is then reset to the highest octave (as specified by the **Octave Extend** setting) and the cycle repeats.

- With **Scan Pattern** set to **Fwd-Rev**, the scan direction is not reversed upon completion of a single forward scan of the captured chord, as would normally occur. Instead, output transpose is stepped upward by one octave and another forward scan is done. This repeats until the forward scan in the highest octave is complete – at this point, the scan direction reverses and the reverse scan is done, still in the highest octave. Subsequently, output transpose is stepped downward by one octave each time a reverse scan is complete (likewise, with no scan direction reverse) until a scan with a transpose of 0 is is complete. The scan direction then switches again to forward and the entire cycle repeats.

- With **Random** scan patterns, the **Octave Extend** setting correspondingly extends (by the specified number of octaves) the set of possible output notes which the random scan can produce – again, as if the actual notes in the captured chord have been replicated in higher octaves.

Repeat (Note Repeat) [0, ... , 7]

When set to values other than zero, this setting causes the arpeggiator to repeat each output note for the specified number of additional beats before proceeding to scan the captured chord for a new note. **Repeat** works with all scan modes and scan patterns.

SweepTr (Sweep Transpose) [0, ... , 127]

This function can be used to produce a dynamically variable "chord inversion" style of upward transpose. Progressively higher settings cause the currently lowest *output* note to be transposed upward by one or more full octaves so that it becomes the highest output note, in effect rolling or "sweeping" the output pattern up the keyboard one note at a time – but without altering its "chord value".

Sweep Transpose works with all scan modes and scan patterns. The control is ranged for a maximum transpose of four octaves and automatically bases itself with respect to the lowest note actually being played on the keyboard at any time. (Hint: the **Sweep Transpose** control is a *natural* for assignment to a performance controller or another MIDI Controller message for "live" adjustment from the keyboard.)

DropNorm (Dropout Norm) [0, ... , 100%]

These controls can be left set to zero for normal use. Increasing the **DropNorm** setting away from zero results in an increasingly probability that an output note will "drop out", or not appear, on any given beat. At the maximum setting, note output is completely suppressed. Scan timing and sequencing is otherwise not affected – these continue to function according to scan mode and scan pattern settings, **Note Repeat**, etc., as though the dropped notes had been played normally. **DropNorm** works with all scan modes and scan patterns.

Drop Alt (Dropout Alt) [0, ... , 100%]

The two **DropNorm** controls are identical in their effect. The **Norm** control is the one whose setting is normally applied. The **Alt** control setting applies only while holding down the MIDI keyboard key assigned to the **Note Dropout ALT** function (see **KeybCtrl Menu**).

This can be used for a variety of effects. The simplest one is to set the **Norm** control all the way to the right (100% dropout) and the **Alt** control full left. With these settings, arpeggiator output notes appear continously while the **Note Dropout Alt** key is held down – but *only* then.

Or, with the opposite settings, the **Dropout Alt** key becomes a "manual rest" or mute key, blocking arpeggiator output as long as it is held down (but not affecting its timing in any way).

More generally, the **Dropout Alt** key can be used to switch instantly back and forth between *any* two note dropout rates.

Sw: [*display only*]

This displays the note used for the **Dropout Alt** function, which results from setting the **Control Base** Note in the **KeybCtrl** Menu.

Rest Note [C-2, ... , G8]

This control permits a specific MIDI input note to be designated as a "rest" note. Whenever the arpeggiator lands on this note during a scan of the captured chord, it inserts a rest – i.e., no output note is generated for this scan step. The inserted rest extends to include repeat notes, if any (see **Repeat**).

The **Rest Note** feature can be used to produce syncopated arpeggios in **Note Order** scan mode (hint: this is easiest with one of the **Extend** capture modes).

The **Rest Note** setting can be changed freely while the arpeggiator is running, with the result that rests appear at different points in the output. Setting **Rest Note** to **G#8 (=OFF)** disables the rest note function.

The scan mode can be switched to **Note Number** while the rest note function is active without any ill effects. This merely results in all rests (if there is more than one) occurring consecutively, since they are (of course) all generated from notes having the same note number.

LFO Menu

These settings control various parameters of the built-in LFO, which can be used to modulate output note-on velocity.

> Note: **>Note/Velocity Mode** must be set to **Replace**, *and* **LFOMod** (in the same menu) must be set higher than minimum, in order for the controls in the **LFO** menu to have any effect.

LFO

Waveform >SpeedType Phase >Resync

Waveform [Triangle / Square / SawUp / SawDown / Random]

Five different LFO waveforms are available: Square, Sawtooth Up, Sawtooth Down, Triangle and Random.

Note that the LFO is applied to velocity in a negative direction. That is, higher (i.e., more positive) instantaneous LFO waveform values correspond to *lower* output note-on velocities. Thus, Sawtooth Up causes velocity to decrease gradually with time and then jump back up to maximum.

SpeedType [Clock, Freq]

Parameter and sub-menu

Two methods of LFO speed control are available:

Clock: Activating this option in the **SpeedType/Clock** submenu permits the LFO speed to be specified in terms of a number of whole arpeggiator beats and individual clocks (fractional beats). Under **Clock**, the LFO waveform goes through a complete cycle over precisely the specified amount of "rhythmical" time.

With this option, LFO speed is based upon current tempo and beat and adjusts itself accordingly if either the beat length (**Note Length**) or the tempo is changed.

Freq: This permits LFO frequency to be specified directly (in Hz) in the **Speedtype/Freq**sub-menu. With this option, LFO speed is completely unrelated to beat settings or tempo and is unaffected by changes to either of these.

Clock sub-menu (Beat/Clk)

If under **Speed Type** the option **Clock** has been set, you can open this sub-menu by pressing the corresponding Continuous Controller (5).

Beat/Clk

Beats Clocks

Beats (Speed Beats) [1, ... , 128]

Speed of the LFO (oscillation rate) in terms of a multiple of the value configured under >Note / NoteLen.

Clock (Speed Clocks) [0, ... , 96]

Fine tuning of the LFO oscillation rate.

Freq sub-menu (Freq Hz)

If under **Speed Type** the option **Freq** has been set, you can open this sub-menu by pressing the corresponding Continuous Controller (5).

Freq Hz

Freq

Freq (Frequency) [0, ... , 20 Hz]

This permits LFO frequency to be specified directly (in Hz). With this option, LFO speed is completely unrelated to beat settings or tempo and is unaffected by changes to either of these.

***

Phase [- 180, ... , 180]

This setting adjusts the starting phase of the LFO – i.e., the point in its waveform to which the LFO springs when it is resynchronized.

Tip: "Good" values for this setting depend upon the selected waveform (see the list above). Setting it to zero causes the LFO to resync to a "zero-crossing" point, which is not always the most interesting starting point:

- For **Saw Up** and **Saw Down** waveforms, a setting of 180° or -180° restarts the waveform at one "end", so that the modulation "ramps" up or down starting from one extreme of its range.

- Similarly, **Triangle** waveforms are resynchronized to positive and negative "peaks" with phase settings of 90° and -90°, respectively.

- By contrast, the **Square** waveform has only two values (+Max and –Max). Therefore, it doesn't "ramp" at all, but instead (as applied to note-on velocity) produces simple loud/soft rhythmic accenting – the **PHASE** setting merely affects the "positioning" of this rhythm.

- Finally, the **Phase** setting has no effect upon the Random waveform, which maintains a single constant (but random) value over the duration of each LFO waveform cycle.

Resync [Resync Off, Resync On]

Parameter and sub-menu

The LFO can be restarted or *resynchronized* by various internal and external events to help produce controlled (or out-of-control) rhythmic effects. Upon „Resync On", the LFO waveform jumps to the point determined by the **Phase** setting (above).

There are four resync sources which can be enabled singly or in any combination in the **Resync** sub-menu. A switch **Resync** permits LFO resync enable/disable without the need to enable/disable the individual sources.

Resync sub-menu

Resync

Internal Manual AutoRSync NoteChrd

Internal [Off, On]

This indicates an "internal" scan restart. That is: in the course of normal scanning, the arpeggiator has once again returned to the "start" of its output pattern – e.g., the lowest output note. The timing of this event depends entirely upon the captured chord and all relevant scan settings. Correspondingly, it varies dynamically as the chord and/or settings are changed.

It should be mentioned that the arpeggiator, as a device which responds dynamically to its input and its settings, does not actually "know" in advance when an internal scan restart will occur, but merely detects this *when* it occurs. This means that LFO resync in response to internal scan restart occurs just barely too late to have an effect upon the "first" or scan restart note sent out by the arpeggiator – instead, the resynchronized LFO is applied starting with the *subsequent* output note. This is not an issue with the other LFO resync sources.

Manual [Off, On]

This indicates a scan restart triggered via the **Rescan** button or the corresponding MIDI key. When this source is enabled, the LFO is resynchronized whenever either of these occurs.

AutoRSync (Auto Resync) [Off, On]

This indicates a scan restart triggered by the **Auto ReScan** beat counter (see **>Scan/Auto ReScan**). When this source is enabled, the LFO is resynchronized periodically, according to the number of beats specified for **Auto ReScan**.

Note that the **Auto ReScan** beat counter can be used to trigger LFO resync *regardless* of whether **Auto ReScan** is itself currently enabled. Likewise, the options affecting the restarting of this counter (see **>Scan/Auto ReScan**) also remain in effect when **Auto ReScan** is disabled, and thus affect the resynchronization of the LFO under this option.

Note/Chord [Off, On]

When this source is enabled, the LFO is resynchronized in response to activity on the MIDI keyboard – i.e., whenever any new note is played on the keyboard, or *only* when new *chords* are played (as determined by the **>Scan – ReScan Mode**).

However, this source is not effective while **HOLD** or **HTrnsp** is active, but is automatically disabled for the duration. Thus, live accompaniment or transposition of a "held" arpeggio via the MIDI keyboard is possible without upsetting the rhythm of the LFO modulation.

KeybCtrl Menu

This group of settings provides control over the mapping of arpeggiator functions to the MIDI keyboard and the echoing of MIDI events from the arpeggiator input to its output.

Eight "performance" controls are assigned to a range of eight contiguous MIDI note numbers, referred to as the **MIDI Control Zone**.

The assigned controls include all parameters in the **Control** group (**Run/Stop, Clear, Hold, HTrnsp, Scan Dir** and **ReScan**). This permits these controls to be used more effectively in real time, as the response via the MIDI keyboard is virtually instantaneous (in contrast to the corresponding graphical buttons in the Noah Remote software).

The MIDI Control Zone also contains two additional functions: **Manual Clocking** and **Note Dropout Alt**.

The MIDI Control Zone includes the **CtrlBase** key and the next seven keys above it (see table following). MIDI notes within this range are used exclusively for control of arpeggiator functions and are *not* captured in the arpeggiator chord buffer, nor are they echoed to the arpeggiator MIDI output. All notes above and below this range are handled normally.

The MIDI Control Zone can be positioned wherever desired on the MIDI keyboard via the **CtrlBase** setting (see below).

The mapping of functions to keys *within* the MIDI Control Zone is fixed, as shown in the following table. The layout is "optimized" for use with the base key set to **C** – typically at the lower end of the keyboard, to permit control with the left hand while playing with the right hand:

Key Position *	Assigned Function
0	Hold
1	Run/Stop
2	HTrnsp
3	Clear
4	Note Dropout Alt
5	Scan Dir
6	Rescan
7	Manual Clocking

(*relative to **CtrlBase** key)

```
KeybCtrl
 CtrlBase  NoteTrans   Thru
```

CtrlBase (Control Base Note) [Off, On]

This setting permits the MIDI Control Zone to be positioned as desired on the MIDI keyboard – or to be moved completely out of the way, if it is not wanted.

NoteTrans (Note Transpose) [-12, ... , 12]

To compensate for the loss of nearly an octave of the MIDI keyboard when the MIDI Control Zone is mapped into the active keyboard range, it is possible to "pre-transpose" incoming note events up or down by a full octave to permit a different range of notes to be used with the arpeggiator, if desired.

> For example, with the MIDI Control Zone positioned at the left end of the keyboard, setting **NoteTrans** to **-12** restores access to the notes in the lowest octave of the keyboard for performance use. This is accompanied, of course, by a corresponding "sacrifice" of note range at the high end of the keyboard.

Thru (Thru Disabled) [Off, On]

Normally, all MIDI messages other than note events are echoed directly to the arpeggiator output – as are note events as well, when the arpeggiator **Hold** function is active, or when the arpeggiator is not running.

In some situations, however, you may not want this. The **Thru** switch is provided for these situations – activating it suppresses *all* echoing of MIDI events from input to output.

OutTiming Menu

These settings provide additional "special-purpose" control over the timing of output notes generated by the arpeggiator. They can be changed at any time.

OutTiming			
ManClk	Tap:	OffBeats	OffClks

ManClk (Manual Clock only) [No, Yes]

This option can normally be left switched off. Activating it disables the arpeggiator's free-running tempo clock (whether internal or synchronized to external MIDI), freezing the arpeggiator in time. Arpeggiator clocking (i.e., stepping) is now manual only, via the MIDI key assigned to the **Manual Clocking** function (as indicated with **Tap**).

Tap [*display only*]

This shows the current value of the assigned manClk MIDI note, which results from the selected CtrlBase note.

OffBeats (Offset Beats) [0, ..., 96]

This is an output delay setting which can be left set to zero for normal use. Non-zero values cause the arpeggiator output to drop "behind" the beat by the specified number, by delaying the start of each scan accordingly.

The **Offset** function can be useful when two (or more) synchronized arpeggiators are being used together, connected to the same MIDI note source. One arpeggiator is used with **Offset** set to 0, the other with a non-zero **Offset** setting. The second arpeggiator produces output whose timing is delayed with respect to that of the first. This can be merely a delayed version of the output from the first arpeggiator – if all other settings on both arpeggiators are identical – or it can be radically different.

OffClks (Offset Clocks)[0, ..., 96]

Setting of the output delay to fractional-beat values can be done via **OffClks**. Among other possibilities, this permits an arpeggiator which is synchronized to a MIDI clock stream from a sequencer to drop its notes "between" or somewhat behind the beat instead of directly on it.

> Note that the specified offset is applied anew on every scan restart, regardless of whether the restart is triggered manually (**Rescan**), via the beat counter, or in response to new note input from the keyboard – or if it is an "internal" restart detected and signalled by the arpeggiator each time it works its way back around to the "first" output note.

The Step Sequencer

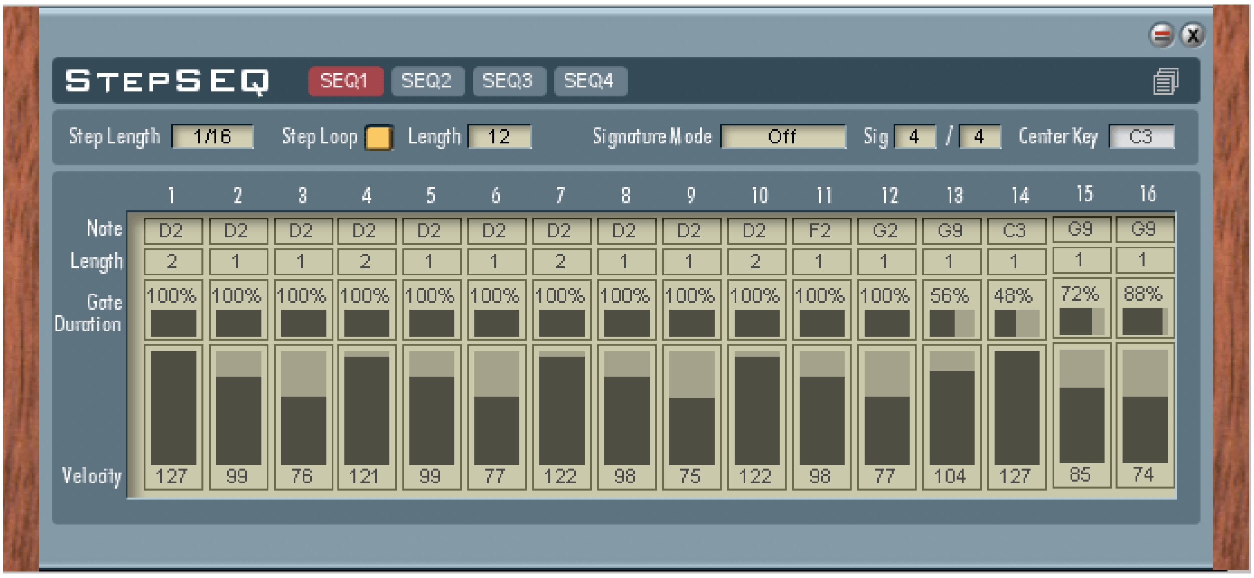
Each of the 4 instrument slots can be addressed not only by an external keyboard or MIDI sequencer, but also by Noah's onboard Arpeggiator or Step Sequencer. You can assign an independent *instance* of the Step Sequencer to each slot.

In order to have an instrument slot be driven by the step sequencer, you must (in Edit mode) select the step sequencer for this slot: in Single mode, via the menu MIDI / Instrument / Trigger, or in Multi mode via the menu MIDI / Devices / Slot 1 (or 2, 3, 4) / Trigger 1 (or 2, 3, 4). This setting is located in the first positon of the lower line in the appropriate menu.

To access the step sequencer's parameters, select the step sequencer and then push the rotary controller (5) which is below it.

The Step Sequencer lets you create unison sequences, or loops. A sequence consists of a maximum of 16 steps in which each step has an adjustable length.

When it receives a MIDI note, the Step Sequencer starts playback at the speed set in the MIDI Manager. The sequence is transposed according to the note received.



Parameters

The global parameters for the Step Sequencer are located in the Global menu. Parameters for the individual steps are in the Steps menu.

```
StepSequencer
 >Global    >Steps
```

Global Menu

```
Global
 StepLen      Loop     LoopLen          >
    SignMode        4 /  4         CKey
```

StepLen (Step Length)

[8/1, 4/1, 2/1,1/1, 1/2 dot, 1/2 1/2 trpl, 1/4 dot, 1/4, 1/4 trpl, 1/8 dot, 1/8, 1/8 trpl, 1/16 dot, 1/16, 1/16 trpl, 1/32, 1/32 trpl, 1 Clock]

Selects the overall resolution for the Step Sequencer (that is, the smallest required note length). For example, choose 1/16 if your shortest step is a 16th note.

Values of the form x/1 are longer than a beat. 4/1, then, indicates 4 beats per step. Values with a „trpl“ are triplets. 1/4trpl indicates a resolution of four triplets, or six steps per bar. (in 4/4). Values with a „dot“ indicate a dotted note value. Therefore, 1/4dot stands for a dotted quarter-note (= 3 eighth notes).

Set the actual length of the individual steps under *Length* per Step, where you can enter a whole number multiple of the value configured here for each step.

Loop [0, ... , 127]

Switches loop playback on or off.

LoopLen (Loop Length) [1, ... , 16]

Select here the number of steps to include in the loop (continuous loop).

Signature Mode (Signature Mode) [Off, AutoRestart, AutoStop]

Signature (time signature) mode specifies the start/stop behavior of loop playback through the following options:

Off: Signature settings have no effect.

Auto Restart: The Step Sequencer starts from the beginning using the bar/beat settings configured under *Signature*. This is independent of whether Step Loop is enabled or not.

Auto Stop: The Step Sequencer stops at the end of the bar configured under *Signature*, even if the total of the step lengths adjusted under *Length* exceeds the bar length.

x / y (Time Signature) [1, ... , 16 / 2,4,8,16]

These two values define the number of beats in the loop (bar), and the value of a beat (see *Signature Mode*). They represent the time signature of the loop.

CKey (Center Key) [0, ... , 127]

This is the MIDI note number that starts the sequence without any transposition. When notes other than this are received, the sequence is transposed according to the note received relative to this one.

Steps Menu

Steps				
Length01	GateD01	Note01	Vel01	>
...				
Length16	GateD16	Note16	Vel16	<

Length 01 - 16 [0, ... , 127]

The length (duration) of the step as a multiple of the resolution as configured under Step Length. For example, if the Step Length is 1/16, a value of 1 means the step will have a duration of a 1/16th note. A value of 2 indicates a step duration of two 1/16th notes, or an 1/8th note.

GateD01 - 16 (GateDuration) [0, ... , 100 %]

This is the length (duration) of the note itself as opposed to the length of the step. Therefore if the step length is 1/16 and the *Gate Duration* is set to 50, the note will last for 1/32nd note (50% of a 16th note) and the remaining time will be played as a rest.

Note01 - 16 [0, ... , 127]

Sets the MIDI note for each step.

See the notes under *Center Key.*

Vel 01 - 16 (Velocity01 - 16) [0, ... , 127]

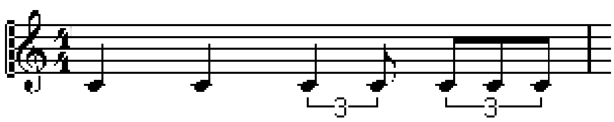
Sets the MIDI Velocity value (keystroke intensity) for each step.

You can also control device parameters such as the filter cutoff of the Minimax or Lightwave's Pan with Velocity values.

Example Sequence Configuration

The following example should help clarify the use of the *Step Length*, *Length*, and *Signature Mode* parameters.

We'll assume that the sequence has the following rhythmic form and that a loop is the equivalent of a bar.



The smallest note value that occurs is an eighth note triplet. Therefore, set the *Global* value for *Step Length* to 1/8trpl. Because the phrase is only seven notes long, you'll need only seven of the 16 steps. Therefore, set the *Length* to 7 steps and enable *Step Loop*. Next, set all notes to the desired pitches (in this example, C3 for all steps).

Give the first two steps a *Length* of 3 (because a quarter-note = 3 eighth-note triplets). Give the next note a length of 2, and the last four notes a length of 1.

The Step Sequencer will now play the seven notes as a continuous loop and will ignore the subsequent, unused steps, since the first seven correspond to the length of a bar.

Alternatively you could set the *Signature* to 4/4, and *Signature Mode* to *Auto Restart*. With these settings, too, the 7-note sequence will play back as a loop.

Lightwave

Introduction

With Lightwave you now posses a highly sophisticated synthesizer that lets you create whole new worlds of sound easily and intuitively–from simple synth and layered sounds to complex, dynamic stereo sounds.

To produce its sounds, Lightwave employs two wavetable oscillators equipped with the wave shaping technology of the legendary Prophet VS. The oscillator signals are combined in the mix section, and routed as desired through two 12dB multi-mode filters. The filters can be configured either in series or in parallel.

In series mode the filters combine to produce slopes of up to 12dB or 24dB/octave if the filters are of the same type. Use filters of different types to create new, complex filter types. In parallel mode you can send the signal from each oscillator to its own filter for creating layered sounds.

-

Two independent pan modulators in the amplifier section make the creation of spacious stereo effects child's play.

With Lightwave you can apply modulation to all important parameters. In addition to the two LFOs and the multi-stage envelope generator you can use almost any MIDI controller as a modulation source.

To help you understand the parameters we urge you to read the Lightwave chapter in the Online Manual as there the structure and function of the instrument is clearly described with illustrations and graphics.

Parameters

To access Lightwave's parameters, open the Lightwave menu (single mode) or Slots/Lightwave (Multi mode).

Lightwave's operating parameters are distributed across 5 menus (Osc, Mix, Vcf, Amp, Mod).

Lightwave				
>OSC	>Mix	>Vcf	>Amp	>
>Mod				<

Osc Menu (Oscillators)

The Osc Menu parameters are found in 3 sub-menus (Common, Osc 1, Osc 2).

Oscillators		
>Common	>Osc 1	>Osc 2

Common sub-menu

Common			
Coarse	Fine	PM Source	PM Depth>
PB Range	PortaMode	PortaTime	Single <

Coarse (Coarse Tune) [-64, ..., 63]

Controls the pitch of the oscillator. Coarse adjusts the pitch in semitones.

Fine (Fine Tune) [- 99, ..., 99]

Controls the pitch of the oscillator. in cents (1cent = 1/100 of a semitone).

PM Source (Pitch Mod Source)
[Off, LFO1, LFO2, LFO1+2, LFO 1*2, LFO1*MW, LFO1*AT, LFO2*MW, LFO2*AT, Filter Env, Amp Env, Free Env+, Free Env-, Keyfollow, Velocity, Aftertouch, Mod.Wheel]

Selects the pitch modulation source.

PM Depth (Pitch Mod Depth) [-64, ... , 63]

This parameter controls the intensity and direction of the Pitch modulation for both oscillators.

PB Range (Pitch Bend Range) [0, ..., 24]

Controls the deflection of the pitch by the pitchwheel. The range is adjustable from 0 to 24 semitones. The pitchwheel has no effect, of course, if the value here is set to zero.

Porta Mode (Porta/Gliss Mode)
[Off, Portamento, Glissando, fing.Porta, fing.Gliss]

When Portamento or Glissando is switched on, the pitch will either glide (Portamento), or progress in a stepwise sequence (Glissando), from one note to the next over a fixed time period (adjustable - see next section).

You can adjust this parameter to ***Off, Portamento, Glissando, fingered Portamento (fing.Porta.)*** or ***fingered Glissando (fing.Gliss.)***.

Note that Portamento/Glissando is only effective when playing in a legato style.

PortaTime (Porta/Gliss Time) [0, ... , 127]

Used in conjunction with the Port/Gliss option, above. This sets the amount of time to glide or gliss from one note to the next.

Single (Single Mode) [Off, On]

Forces the instrument to operate in single-voice mode, regardless of how many voices are actually currently loaded. Guarantees proper performance of solo sounds with portamento.

Osc 1 / 2 sub-menu

Oscillators 1 and 2 are wavetable oscillators equipped with the waveshaping technology of the Prophet VS. This is the oscillator also used by CreamWare's successful Vectron synthesizer.

For each oscillator you can select from a pool of 128 waveforms. A Grunge control adjusts the tone quality from soft and warm, to hard and bright (overtone-rich). The pitch modulation is individually adjustable for each oscillator.

OSC 1			
Waveform	Coarse	Fine	Grunge >
PM Source	PM Depth		<

Waveform [*128 Waveforms*]

Use this control to select one of the 128 available waveforms. The Noise waveform is unusual in that its sonic character is influenced by the Coarse/Fine controls. This is not normally the case with noise waveforms.

Coarse (Coarse Tune) [-64, ..., 63]

Controls the pitch of the oscillator. Coarse adjusts the pitch in semitones.

Fine (Fine Tune) [- 99, ..., 99]

Controls the pitch of the oscillator. Fine adjusts it in cents (1cent = 1/100 of a semitone).

Grunge [0, ..., 127]

Adjusts the timbre of the signal from soft and warm to hard and bright by adding overtones. For the most part the overtones are created by increasing the distortion caused by aliasing, which, in this case, achieves a desirable effect.

PM Source (Pitch Mod Source)
[Off, LFO1, LFO2, LFO1+2, LFO 1*2, LFO1*MW, LFO1*AT, LFO2*MW, LFO2*AT, Filter Env, Amp Env, Free Env+, Free Env-, Keyfollow, Velocity, Aftertouch, Mod.Wheel]

Selects the modulation source for the respective oscillators.

PM Depth (Pitch Mod Depth) [-64, ... , 63]

Selects the pitch modulation source and controls the intensity and direction of the modulation for the respective oscillators.

Mix Menu

As you would expect, the Mix section combines the oscillator output signals. You can also apply modulation to the volume level of each oscillator. The balance controls let you route proportions of each oscillator signal to the two filters. At the middle value, the signal is sent to the two filters in equal proportion. At theminimum value the signal is sent only to Filter1, and at the maximum value, only to Filter2. Depending on the configuration of the filters–serial or parallel–and the selected filter types, innumerable sonic variations are possible. A Gain control sets the level of the pre-filtered oscillator mix signal.

```
Mix
  >Osc 1    >Osc 2   Mix Gain
```

Mix Gain [-0, ..., 127]

Mix Gain sets the overall volume level of the combined signals before it is sent to the filter section.

Additional Mix menu parameters are located in the Osc 1 and Osc 2 sub-menus.

Osc 1 / 2 sub-menu

```
Osc 1
  Level   LM Source  LM Depth      >
 Balance  BM Source  BM Depth      <
```

Level [0, ..., 127]

Basic volume levels of osillators 1 or 2.

LM Source (Level Mod Source)
[Off, LFO1, LFO2, LFO1+2, LFO 1*2, LFO1*MW, LFO1*AT, LFO2*MW, LFO2*AT, Filter Env, Amp Env, Free Env+, Free Env-, Keyfollow, Velocity, Aftertouch, Mod.Wheel]

Selects the modulation source for modulating the levels of oscillators 1 or 2.

LM Depth (Level Mod Depth) [-64, ... , 63]

Controls the depth and direction of the level modulation.

Balance [-64, ..., 63]

Controls the routing of the OSC1 and OSC2 (see Mix menu).

BM Source (Balance Mod Source)
[Off, LFO1, LFO2, LFO1+2, LFO 1*2, LFO1*MW, LFO1*AT, LFO2*MW, LFO2*AT, Filter Env, Amp Env, Free Env+, Free Env-, Keyfollow, Velocity, Aftertouch, Mod.Wheel]

Selects the modulation source for controlling the signal distribution from the oscillators to filters 1 and 2.

BM Depth (Balance Mod Depth) [-64, ... , 63]

Controls the depth and direction of the balance modulation.

Vcf Menu

The Filter section provides two multimode filters with a slope of 12dB/octave and adjustable resonance. Each filter is configurable as lowpass, highpass, or bandpass. A Thru function switches the filters to bypass. The filters can be configured either in series or in parallel. Cutoff and Resonance can be modulated seperately. The two filters share a common envelope generator.

```
Vcf
 >Common    Vcf 1    >Vcf 2  >Envelope
```

Common sub-menu

```
Common
   Mode      Link                      >
 CF ModS   CF ModD   Res ModS  Res ModD<
```

Mode (Vcf Mode) [serial, parallel]

Switches the filters to a serial or parallel configuration. If you set the two filters to the same filter type in serial configuration the effect of the filter is summed, and the slope increases to 24dB/octave.

Link (Vcf Link) [Off, On]

This option links the Filter1 and Filter2 controls so that they operate in tandem. This is especially useful when the filters are configured in series to create, in effect, a single 24dB/octave filter.

Cf ModS (CF Mod Source)
[Off, LFO1, LFO2, LFO1+2, LFO 1*2, LFO1*MW, LFO1*AT, LFO2*MW, LFO2*AT, Filter Env, Amp Env, Free Env+, Free Env-, Keyfollow, Velocity, Aftertouch, Mod.Wheel]

Selects the modulation source for cutoff frequency.

CF ModD (CF Mod Depth) [-64, ... , 63]

This parameter controls the intensity and direction of the CF modulation.

Res ModS (Res Mod Source)
[Off, LFO1, LFO2, LFO1+2, LFO 1*2, LFO1*MW, LFO1*AT, LFO2*MW, LFO2*AT, Filter Env, Amp Env, Free Env+, Free Env-, Keyfollow, Velocity, Aftertouch, Mod.Wheel]

Selects the modulation source for resonance modulation.

Res ModD (Res Mod Depth) [-64, ... , 63]

This parameter controls the intensity and direction of the Resonance modulation.

Vcf 1 / 2 sub-menu

```
Vcf 1
Filter Type                            >
  Cutoff     Res       KeyF   Env Depth>
 CF1 ModS  CF1 ModD  Res1ModS  Res1ModD<
```

Filter Type [LPF, HPF, BPF, Thru]

The filters are switchable to operate as highpass, bandpass, or lowpass filter. When set to Thru the filters are bypassed.

Cutoff (Cutoff Frequency) [0, ..., 127]

Sets the Cutoff frequency. The Cutoff frequency is the frequency above which the filter begins to act on the signal.

Res (Resonance) [0, ..., 127]

Controls the degree of Resonance. Increasing the resonance reinforces the frequencies lying near the cut-off frequency. At high resonance values, the filter oscillates producing a sine wave at the cutoff frequency.

KeyF (Keyfollow) [-200, ..., 200]

This parameter allows the filter cutoff to track the keyboard through the MVC. The keyfollow mid-point is fixed at MIDI note #64 (E3). At this note, the cutoff frequency will always stand at its original value, regardless of the key follow setting. When keyfollow is set to 100%, the cutoff frequency will adjust to maintain its frequency relationship to the pitch across the entire keyboard. At a setting of 50%, the cutoff frequency ratio will be lowered by 50% per octave above E3, and raised 50% per octave below E3. A value of 0% means no keyfollow modulation, and the cutoff frequency remains fixed. If key follow is set to 200%, the cutoff frequency increases or decreases twice as fast as the pitch of the note played. Negative values invert the sense of the modulation - higher notes yield a lower cutoff frequency and vice versa.

Env Depth (Envelope Depth) [-64, ... , 63]

Adjusts intensity and direction of envelope-modulation. Modulation can be positive or negative (inverted).

Link (Vcf Link) [Off, On]

Enable this option to couple the settings of Filter 1 and Filter 2. Use this option when you want to operate the filters in serial mode, as you would to implement a 24dB/Octave filter.

CF1/2 ModS (CF1/2 Mod Source)

[Off, LFO1, LFO2, LFO1+2, LFO 1*2, LFO1*MW, LFO1*AT, LFO2*MW, LFO2*AT, Filter Env, Amp Env, Free Env+, Free Env-, Keyfollow, Velocity, Aftertouch, Mod.Wheel]

Selects the modulation source for the filter cutoff frequency.

CF1/2 ModD (CF1/2 Mod Depth) [-64, ... , 63]

Adjusts the intensity and direction of the Cutoff modulation.

Res1/2 ModS (Res1/2 Mod Source)

[Off, LFO1, LFO2, LFO1+2, LFO 1*2, LFO1*MW, LFO1*AT, LFO2*MW, LFO2*AT, Filter Env, Amp Env, Free Env+, Free Env-, Keyfollow, Velocity, Aftertouch, Mod.Wheel]

Selects the modulation source for modulating the resonance of the respective filter.

Res1/2 ModD (Res1/2 Mod Depth) [-64, ... , 63]

Adjusts the intensity and direction of the resonance modulation.

Envelope sub-menu (VCF Envelope)

```
Vcf Envelope
  Attack     Decay     Sustain    Release >
Time Keyf  Time Vel  Lev Vel              <
```

Attack [0, ..., 127]

The attack time. When a gate signal is received, the attack segment starts, and continues for the slope time at which point the maximum level is reached.

Decay [0, ..., 127]

The decay time. The envelope enters the decay segment when the attack phase completes, and the modulation falls back to the sustain level. The time needed for this is the decay time. A decay will not be heard if the sustain level is set to maximum.

Sustain [0, ..., 127]

The sustain level. This is the level at which the modulation signal will be held as long as the gate is open. When the gate closes, the release segment immediately follows.

Release [0, ..., 127]

The release time. When the envelope generator receives a gate-off signal, it jumps immediately from its current phase to the release stage. When the change from one segment to another takes place, the release time will be adjusted to account for the volume level at that time.

Time Keyf [-200, ..., 200] (Time Keyfollow)

Adjusts the times of all segments of the envelope. Both the intensity and direction of the modulation effect through MIDI-Note-Number is set by this value. Negative values shorten times, and positive values lengthen them.

Time Vel [-200, ..., 200]

Adjusts the times of the three envelope segments. Both the intensity and direction of the modulation effect through velocity is set by this value. Negative values shorten times, and positive values lengthen them.

Level Vel [-200, ..., 200]

Adjusts the levels of all segments of the envelope. The value here (0..max) controls the intensity of the modulation of the levels by velocity.

Amp Menu (Amplifier)

The Amp section consists of two modulatable pan controls and an amplifier with its own envelope generator. Pan1 is permanently assigned to the output from Filter1, and Pan2 to Filter2. This implies that Pan1 has an effect only when the filters are switched to parallel mode. Or, to put it another way, Pan2 alone is used in serial mode. Volume controls the overall signal level.

```
Amplifier
  >Pan 1    >Pan 2  >Envelope   Volume
```

Volume [0, ..., 127]

The overall volume can be set here. When playing with high polyphony distortion can occur. Turn down the volume to avoid this.

Pan 1 / 2 sub-menu

```
Pan 1
  Pan 1    Pan1ModS  Pan1ModD
```

Pan 1/2 (Panorama 1/2) [-64, ... , 63]

Controls the position of the respective signal in the stereo field. Pan1 controls the signal from Filter1, and Pan2 the signal from Filter2.

Pan 1/2 ModS (Pan 1/2 Mod Source)
[Off, LFO1, LFO2, LFO1+2, LFO 1*2, LFO1*MW, LFO1*AT, LFO2*MW, LFO2*AT, Filter Env, Amp Env, Free Env+, Free Env-, Keyfollow, Velocity, Aftertouch, Mod.Wheel]

Selects the modulation source for Panorama Position.

Res1/2 ModD (Res1/2 Mod Depth) [-64, ... , 63]

Adjusts the intensity and direction of the Panorama modulation.

Envelope sub-menu

```
Amp Envelope
 -Attack   Decay    Sustain   Release >
Time Keyf  Time Vel  Lev Vel          <
```

Attack [0, ..., 127]

The attack time. When a gate signal is received, the attack segment starts, and continues for the slope time at which point the maximum level is reached.

Decay [0, ..., 127]

The decay time. The envelope enters the decay segment when the attack phase completes, and the modulation falls back to the sustain level. The time needed for this is the decay time. A decay will not be heard if the sustain level is set to maximum.

Sustain [0, ..., 127]

The sustain level. This is the level at which the modulation signal will be held as long as the gate is open. When the gate closes, the release segment immediately follows.

Release [0, ..., 127]

The release time. When the envelope generator receives a gate-off signal, it jumps immediately from its current phase to the release stage. When the change from one segment to another takes place, the release time will be adjusted to account for the volume level at that time.

Time Keyf [-200, ..., 200] (Time Keyfollow)

Adjusts the times of all segments of the envelope. Both the intensity and direction of the modulation effect through MIDI-Note-Number is set by this value. Negative values shorten times, and positive values lengthen them.

Time Vel [-200, ..., 200]

Adjusts the times of the three envelope segments. Both the intensity and direction of the modulation effect through velocity is set by this value. Negative values shorten times, and positive values lengthen them.

Level Vel [-200, ..., 200]

Adjusts the levels of all segments of the envelope. The value here (0..max) controls the intensity of the modulation of the levels by velocity.

Mod Menu (Modulation)

Modulation		
>Lfo 1	>Lfo 2	>Free Env

Lfo 1 / 2 sub-menu

Two full-featured LFOs serve as possible modulation sources for several parameters in the synthesizer. The LFOs are monophonic, and can be synchronized to a MIDI clock.

Lfo 1			
Waveform	Rate	Retrigger	InitPhase>
Delay	Fade In	Fade Out	Rate Keyf>
RateModS1	RateModD1	RateModS2	RateModD2 >
Lev ModS	Lev ModD	Clock	Note <

Waveform **[Sine, Square, Saw Up, Saw Down, Triangle, Random]**

Selects the desired waveform.

Rate [0, ... , 400 Hz]

The frequency/rate of the modulation.

Retrigger [Off, On]

Determines whether the signal will run continuously, or be restarted at its initial phase setting each time a new note is played.

InitPhase (Initial Phase) [-180, ..., 180]

Determines the position within the waveform (phase) at which the signal will start when a gate signal is received. Retrigger must be enabled for this to take effect.

Delay [0, ..., 127]

Delays the onset of the modulation. The range is from 0 to 20 seconds.

Fade In [0, ..., 127]

When a gate signal is received, the modulation will gradually build to maximum at the time set here. A gate signal must be connected. The range is from 0 to 20 seconds.

Between the fade in and the fade out, the modulation remains at maximum.

Fade Out [0, ..., 127]

When a gate off signal is received, the modulation will gradually fade to 0 at the time set here. A gate signal must be connected. The range is from 0 to 20 seconds.

Rate Keyf (Rate Keyfollow) [-200, ..., 200]

This parameter allows the Rate (frequency) to track the keyboard. The keyfollow mid-point is fixed at MIDI note #64 (E3). At this note, the cutoff frequency will always stand at its original value, regardless of the key follow setting. When keyfollow is set to 100%, the LFO frequency adjusts to maintain its relationship to the pitch across the entire keyboard as it follows the pitch. At a setting of 50%, the frequency ratio will be lowered by 50% per octave above E3, and raised 50% per octave below E3. A value of 0% means there is no keyfollow modulation, and the LFO rate remains fixed.

RateModS1/2 (Rate Mod Source 1/2)
[Off, LFO1, LFO2, LFO1+2, LFO 1*2, LFO1*MW, LFO1*AT, LFO2*MW, LFO2*AT, Filter Env, Amp Env, Free Env+, Free Env-, Keyfollow, Velocity, Aftertouch, Mod.Wheel]

Selects the source for LFO Rate modulation.

RateModD1/2 (Rate Mod Depth 1/2) [-64, ..., 63]

Controls the rate modulation depth and direction of the LFO.

Lev ModS (Level Mod Source)
[Off, LFO1, LFO2, LFO1+2, LFO 1*2, LFO1*MW, LFO1*AT, LFO2*MW, LFO2*AT, Filter Env, Amp Env, Free Env+, Free Env-, Keyfollow, Velocity, Aftertouch, Mod.Wheel]

Controls the level modulation depth and direction of the LFO. Select the modulation source from the associated drop-down list.

Lev ModD (Level Mod Depth) [-64, ..., 63]

Adjusts the intensity and direction of the modulation.

Clock (MIDI Clock) [Off, On]

Switches on MIDI clock synchronization.

Note (Note Length)
[1/32 trpl, 1/32, 1/16 trpl, 1/16, 1/16 dot, 1/8 trpl, 1/8, 1/8 dot, 1/4 trpl, 1/4, 1/4 dot, 1/2 trpl, 1/2, 1/2 dot, 1/1]

Select a note value for the MIDI Clock speed here.

Free Env sub-menu

```
Free Envelope
  Attack    Decay    Sustain   Release >
Time Keyf  Time Vel  Lev Vel           >
Att Slope Dec Slope                    <
```

Attack [0, ..., 127]

The attack time. When a gate signal is received, the attack segment starts, and continues for the slope time at which point the maximum level is reached.

Decay [0, ..., 127]

The decay time. The envelope enters the decay segment when the attack phase completes, and the modulation falls back to the sustain level. The time needed for this is the decay time. A decay will not be heard if the sustain level is set to maximum.

Sustain [0, ..., 127]

The sustain level. This is the level at which the modulation signal will be held as long as the gate is open. When the gate closes, the release segment immediately follows.

Release [0, ..., 127]

The release time. When the envelope generator receives a gate-off signal, it jumps immediately from its current phase to the release stage. When the change from one segment to another takes place, the release time will be adjusted to account for the volume level at that time. ausgehend vom letzten Level.

Time Keyf [-200, ..., 200]

Adjusts the times of all segments of the envelope. Both the intensity and direction of the modulation effect through MIDI-Note-Number is set by this value. Negative values shorten times, and positive values lengthen them.

Time Vel [-200, ..., 200]

Adjusts the times of the three envelope segments. Both the intensity and direction of the modulation effect through velocity is set by this value. Negative values shorten times, and positive values lengthen them.

Level Vel [-200, ..., 200]

Adjusts the levels of all segments of the envelope. The value here (0..max) controls the intensity of the modulation of the levels by velocity.

Att Slope (Attack Slope) [0, ..., 127]

Adjusts the slope curve for the attack phase. The curve is continuously adjustable from linear to logarithmic.

Dec Slope (Decay Slope) [0, ..., 127]

Adjusts the slope curve for the decay/release phases. The curve is continuously adjustable from a linear to an exponential fade out.

Six-String (optional)

Introduction

What does the Six-String give you?

Synthesizer developers are always looking for interesting new points of departure for the production of sounds. What analog systems have to offer; how other digital systems work; and, of course, how natural systems function -the developer investigates all these things when considering a new direction. With regard to the string model the developer sooner or later discovers that this apparently simple natural physical system is much more complex than he or she might have originally thought.

Some very bright minds, employing various approaches, have developed ingenious methods for producing convincing plucked-string emulations that can be computed and played in real-time. Two current techniques that work quite well are Karplus-Strong synthesis (named after the two developers of the algorithm) and the Mass/Spring model. However, none of these methods have duplicated exactly the physical characteristics of a ”real” string. Instead, they use mathematical ”tricks”, such as incorporating delay lines, to produce the tonal behavior of the virtual string.

Thanks to a completely new mathematical approach, the Six-String offers for the first time a true one-to-one emulation of a physical string. This algorithm allows for string characteristics to be specified comprehensively, exactly duplicating its real-world counterpart. Definable characteristics include inertia, material flexibility, string diameter, tension, and so on. Of course, with such a complex algorithm not all the parameters have to be specified by the user at the lowest levels, as the interactions among them could lead to confusing phenomena. Therefore, some values in the Six-String have been pre-configured and made available to the user as presets. For example, the user can make simultaneous adjustments for complete string sets, rather than having to define each string individually.

Some values, such as rigidity (inertia) or elasticity are adjustable only within meaningful or useful limits. The string sets can be fine tuned later, or even altered to such an extent that the resulting spectrum is reminiscent more of a bell than a typical string.

The Six-String allows for extremely versatile plucked-string synthesis, from the simulation of nylon or steel strings through to metal bars and even wood blocks. And because the Six-String was developed to imitate guitars in particular, additional parameters specific to guitars, such as those used to emulate acoustic guitar bodies or electric pickups, are also incorporated.

Synthesis Structure

As mentioned earlier, the synthesis structure of the Six-String begins with the definition of the strings (or the choice of a string set) and the basic decision whether to emulate an acoustic or electric guitar. These early choices are crucial to the subsequent structure of the model. Depending on which model you choose, the main screen (in the *Noah Remote* software) displays one guitar type or the other. Ultimately what you see on the screen bears a direct relationship to what you should expect to hear.

The Acoustic model

With the acoustic model a simulation of the guitar body, using three band-pass filters to produce the most important resonant frequencies, immediately follows the string. This is ”only” an approximation; an actual guitar body exhibits somewhat more complex resonant behavior. The solution implemented here offers a very good, and at the same time, computationally economical way to incorporate the necessary resonant body. The vibrations of the strings are taken from one or two ”dry” positions and then combined with the body signal.

The Electric model

The structure of the electric model is similar to the acoustic model, but without the simulation of a resonant body. Instead, virtual guitar pickups follow the string in the algorithm. Here, too, a filter is used to approximate the pickup behavior—in this case a lowpass filter with adjustable resonance.

With the electric model, too, the string vibrations are picked up from one or two positions. The signal is then sent to the pickup filter. After the pickups you have the option of routing the electric guitar signal to a virtual amplifier. The amplifier has a fairly complex architecture with pre- and post-equalizers and an integrated analog tube emulator. The amplifier is also useful with a real guitar, producing authentic amp sounds - from smooth and round, to heavily distorted.

Presets

As usual with CreamWare instruments, the Six-String features a Preset List with which you can administer and recall sound settings. In addition to the the basic preset list, the Six-String adds *Pluck,* and *Body* preset sub-lists. Presets in these lists can easily be made globally available once you have established the settings. As soon as you have created a preset within a preset sub-list it is added into the list of selectable presets you can call up through the respective drop-down lists. The presets in these lists are not referenced, however; the *values only* are passed to the device. In other words, when you change a preset in a preset sub-list it does not have an effect on the primary presets supplied with the Six-String.

Parameter

To access the Six-String parameters open the Six-String menu (Single mode) or the Slots/Six-String menu in Edit mode.

The Six-String parameters are distributed among 11 menus (Type/Vol, Strings, Excit, Pluck, Damping, Microph, Pickup, Body, Slap, Pitch, Control):

SixString			
>Type/Vol	>Strings	>Excit.	>Pluck
>Damping	>Microph.	>Pickup	>Body
>Slap	>Pitch	>Control	

Type/Vol Menu

Type/Vol		
GitType	MainVol	Level Met

GitType [Acoustic / Electric]

Choose either the *Acoustic* or the *Electric* model.

MainVol (Main Volume) [0 - 127]

This control adjusts the overall volume of the guitar model. The volume can be further modified by settings in the effects section downstream.

When using the Electric model the amplifier can provide yet another substantial increase in volume.

Level Met (Level Meter)

This provides a level meter display to monitor the signal.

Strings Menu

Strings			
Strngset	SkipHarm	BoostHrm	Elast >
Inertia			

Strngset [Parameter see below]

You can select from among various supplied string sets:

Bass Nylon
(105 mm Mensur - 0,114/0,079/0,063/0,047)

Bass Double
(130 mm Mensur - 0,114/0,079/0,063/0,047)

Bass Steel
(81 mm Mensur - 01,106/0,07/0,056/0,043)

Guitar Electric
(65 mm Mensur 0,042/0,032/0,024, 0,016, 0,011, 0,009)

Guitar Jazz
(65 mm Mensur- 0,059/0,044/0,036/0,026/0,017/0,013)

Guitar Nylon
(65 mm Mensur - 0,045/0,036/0,028/0,037/0,029/0,023)

Guitar Western
(65 mm Mensur - 0,05/0,04/0,03/0,022/0,014/0,011)

MonoString (0,1)

The settings for the string sets contain string-specific parameters that are not user-accessible. Also, the Damping parameter is set to values typical for the strings.

SkipHarm (Skip Harmonics) [1-70]

The basic string model computes the first 70 partial tones in the normal harmonic sequence. However, the Skip Harm adjustment allows you to compute only every second partial from a specified starting point. This makes it possible to generate partials above the standard 70^{th} partial normally computed.

Example: With *Skip Harm.* set to 50, even-numbered partials only are computed above the first 50 in the normal sequence—that is, the sequence after the 50^{th} becomes 52, 54, 56 and so on. Because you always have 70 available, this means that partials up to the 90^{th} partial will be computed. This is of particular value with bass tones in which the upper partials remain within the audible range. With higher notes the upper partials lie above the audible spectrum and it makes no difference whether the 90^{th} partial is computed or not.

BoostHrm (Boost Harmonics) [0 – 10.0]

When you omit partial tones using Skip Harm you can use Boost either to compensate for the missing harmonics or simply to add more highs. Note that the result will sound more synthetic with increasing Boost Hrm values.

Elast (Elasticity) [0 – 127]

This value represents the flexibility of the string material. This parameter is defined such that the string set responds normally at a value of 0. Increasing the value reduces the elasticity and the result begins to sound more and more metallic.

Inertia [0 – 127]

This parameter represents the moment of inertia and is derived from the string geometry. A value of 0 indicates a normal (correct) value for the string. As you increase the value the string becomes more rigid and "mutates" gradually from a string to a rod.

Excit Menu

The parameters in this menu configure the excitation charistics of the string model.

Excitation			
ExcLevel	Level Met	ExcPos	Vel->Pos

ExcLevel (Excitation Level) [0 – 127]

This parameter controls the strength or force of the excitation. This is a global setting that can be configured in detail in the Pluck section.

Level Met (Level Meter)

This menu item provides a display of the signal level.

ExcPos (Excitation Pos) [0 – 127]

Adjusts the position of the virtual pick. Adjustable from the bridge (0) to half the length of the string (127).

Vel->Pos (Velocity> Pos) [-64 – +63]

Determines whether or how strongly the strength of the plectrum stroke (velocity) modulates the position of the plectrum. Positive values shift the plectrum to the right towards the bridge, negative values to the left. The maximum modulation depth depends on the basic position of the plectrum. In any case, with full modulation the position shifts to the maximum deflection—either to the bridge or to the center position.

Pluck Menu

For a string to vibrate it must first be excited. This excitation is usually a short impulse produced by an agent such as a fingertip, fingernail or plectrum (pick). The agents differ in various aspects. For example, a fingertip will rub longer against the string than a plectrum which engages the string only very briefly. Because it is very difficult to define the simulation of a fingertip in a mathematical formula that can be steplessly changed to a plectrum, some other excitation formula had to be implemented. The excitation component therefore consists in principle of an AD (attack-decay) envelope in which the excitation comprises basic energy (Global) and noise energy (Pluck). The times of this curve are freely adjustable and can be modulated by velocity. Therefore a range of excitation characteristics is possible, all of which can be dynamically controlled through velocity.

Pluck			
Type	>Noise	>Envelope	>Level

Type [Preset]

Opens the preset list. These presets contain values for Global and Pluck.

Noise sub-menu

These parameters configure the characteristics of the noise portion of the string excitation.

Noise	
HiCut	LowCut

HiCut (Filter High Cut) [20 – 16.000 Hz]

Frequencies of the noise component can be attenuated by means of two filters. The HighCut filter cuts frequencies above the adjusted frequency. It operates with a slope of 12 dB/octave.

LowCut (Filter Low Cut) [20 – 16.000 Hz]

The LowCut filter attenuates all frequencies below the adjusted frequency value. It also operates with a 12 dB/ octave slope.

Envelope sub-menu

These parameters control the envelope of the string excitation.

Envelope			
PlckAtt	PlckDec	Vel->Att	Vel->Dec

PlckAtt (Pluck Attack) [0 – 100 ms]

Controls the rise rate of the excitation A/D curve. The control ranges from 0 to 100 ms.

PlckDec (Pluck Decay) [0 – 100 ms]

Controls the fall rate of the excitation A/D curve. The control range is from 0 to 100 ms.

Vel->Att (Velocity>Attack) [-64 – +63]

The attack time can be modulated by the strength of the excitation. It is possible to extend or shorten the attack time by a factor of 100 at maximum velocity.

> **Example:** With an attack time of 10 ms and Vel on Attack set to maximum positive, the attack time at maximum velocity will be 1000 ms and 10 ms at minimum velocity. With Vel set to maximum negative, the attack time at maximum velocity will be 0.1 ms.

Vel->Dec (Velocity>Decay) [-64 – +63]

The decay time can be modulated by the excitation strength. The decay time can be extended or shortened by a factor of 100 at maximum velocity.

Level sub-menu

Level		
Gl Lev	Noise	Vel>Nois

Gl Lev (Global Level) [0 – 127]

Controls the basic excitation energy.

Noise (Noise Level) [0 – 127]

Controls the energy of the noise component.

Vel->Nois (Velocity>Noise) [0 – 127]

If the control is set to 0, the noise portion is always present with a level independent of the MIDI velocity at the adjusted volume. As you increase the value the noise component decreases with low velocity values and increases with high values.

Damping Menu

After its initial excitation, a string vibrates without an additional supply of energy. The vibration decreases over time due to friction with the air and energy absorption at the nut and bridge. The volume of the signal falls over time, but not proportionally for all frequencies. High frequencies generally fade away more quickly than lower frequencies whose oscillations contain substantially more energy to be absorbed by friction. *Damping* modifies this behavior.

Damping			
Decay	Release	HiDamp	Vel>Damp

Decay [0 – 127]

The frequency-independent energy loss of the string—controls the overall decay time for a string while a key is pressed.

Release [0 – 127]

Controls the frequency-independent energy absorption when the key is released.

HiDamp (High Damp) [0 – 127]

Controls *frequency-dependent* damping of the string. The higher the value, the more quickly high frequencies relative to the fundamental fade away. Moderate settings produce a natural harmonic attenuation while others produce effects of strongly damped strings.

Vel>Damp (Velocity>Damp) [-64 – +63]

Adjusts the influence of MIDI velocity over harmonic damping. In the center position velocity has no influence. Turning the controller clockwise increases the damping of overtones by velocity, while turning it to the left reduces the damping.

Microph Menu (Microphone)

The Six-String emulates the acquisition of the string sound using virtual microphones or pickups.

Microphone			
LinkPos	PickUp1	PickUp2	Stereo >
PUp1 Lev	PUp2 Lev		

LinkPos (Link Positions) [Off / On]

With the Pickup Link switch in the upper left corner you can synchronize the positions of the two microphones. When the option is enabled only a single microphone is displayed.

PickUp1 (PickUp1 Position) [0 – 127]

You can place two virtual microphones to pick up the guitar signal. You can position each microphone independently anywhere between the bridge (0) and the middle of the strings (127), and each microphone can have its own volume level.

PickUp2 (PickUp2 Position) [0 – 127]

Adjust the volumes of the respective pickups.

Stereo (Stereo Spread) [0 – 127]

With the acoustic model this control adjusts the stereo spread of the two microphones. In the 0 position both mics are merged to mono. At the Max position pickup 1 is routed hard left and pickup 2 hard right.

Important: With the acoustic model, the stereo width control is disabled when the pickups are linked.

PUp1 Lev (PickUp 1 Level) [0 – 127]

Adjusts the relative volume level of pickup/mic 1.

PUp2 Lev (PickUp 2 Level) [0 – 127]

Adjusts the relative volume level of pickup/mic 2.

Pickup Menu

With Electric model, a simulation of the pickups, rather than a guitar body, is available. The pickups are implemented with a highpass, and, more important, a resonant lowpass filter. The highpass filter ensures that low frequencies can be cut when desired to prevent boominess from deep frequecies. The lowpass filter ensures that higher frequencies can be dampened, exactly as with physical pickups. Of course, different pickups with different characteristics are available. Most pickups, for example, exhibit a critical resonant frequency - yet another frequency you can adjust.

PickUp			
	Type	Active	>Edit

Type [Preset]

Select here one of the pickup presets.

Active [Off / On]

Enables/disables pickup simulation.

Edit sub-menu

Edit		
HPF	LPF	LPFRes

HPF (Highpass Freq) [20 – 20.000 Hz]

Sets the frequency below which the signal components are attenuated. The slope of the attenuation is 12dB/octave.

LPF (Lowpass Freq) [20 – 20.000 Hz]

Sets the frequency above which the signal is attenuated. The slope of the attenuation is 12dB/octave.

Frequency values of approximately 2kHz produce warm, soft tones. At 3kHz, the tone is somewhat brighter, and at 4kHz even more brilliant. Settings above 5kHz produce a distinctly bright, sharp tone quality. Adjust the resonance afterward to suit your taste.

LPFRes (Lowpass Res) [0 – 127]

Controls the amount of reinforcement of the frequencies lying near the cutoff frequency in the lowpass filter.

Body Menu

With the acoustic model a group of controls labeled **Body** is available. The controls produce an approximate emulation of a guitar body by means of three bandpass filters. This arrangement lets you define three frequency ranges to characterize a particular guitar body. One of the most important resonant frequencies is the *Helmholtz frequency*, known as such because it is the frequency at which a Helmholtz resonator resonates. A Helmholtz resonator can be described as a tube closed at one end. When air inside the tube is excited it produces an oscillation, or steady pitch, at the Helmholtz frequency. A familiar example of this is when you blow across the top of an empty bottle to produce a tone. Now transfer this idea to the action of the strings across the bridge and to the guitar body; the air inside likewise resonates at the guitar's Helmholtz frequency. The exact frequency depends on the size of the resonant cavity. With steel-string guitars the frequency is approximately 55 Hz. Classical guitars resonate at 103.8 Hz and flamenco guitars usually somewhere between 92.5 and 98 Hz.

Other resonant frequencies are a function of the construction of the guitar body and normally range below 1000 Hz.

As mentioned earlier, simulating a guitar body following accurate scientific principles is only one possibility. The model also permits settings that a physical body cannot produce, thereby extending the range of sonic possibilities once again.

```
Body
   Type     Active     Relation     >Edit
```

Type (Body Type) [Preset]

Click on the text field next to the Body section designation to open a list from which you can select a body type.

Active (Body Active) [Off / On]

Enable or disable the body simulation.

Relation [-64 – +63]

Sets the string to body signal ratio. At the minimum setting only the string signal component is produced, and at maximum only the body component.

The body component is monophonic (as is appropriate) and mixed evenly to both channels.

Edit sub-menu (Settings)

```
Settings
   Freq1     Level1     Freq2     Level2     >
   Freq3     Level3                          <
```

Freq1-3 (Resonance Freq1 - 3) [20 – 20.000 Hz]

Enter a frequency at which the body is to produce a resonance.

Level1-3 (Resonance Level 1- 3) [- 0dB]

Enter the respective volumes of the individual resonant frequencies.

Note that you can quite easily generate undesirable distortion by adjusting the filter frequencies very close to each other, and/or by setting high levels. Also, the volume depends on the stimulation of the guitar body, that is, how much of the respective resonant frequency range is present in the string signal.

Slap Menu

The Slap section controls the behavior of the string with regard to the fingerboard. You can adjust the distance of the strings from the frets, and also the vertical deflection of the string when it is plucked. The material of the fingerboard and frets can also be adjusted to six different types. This section therefore controls the amount of impact of the string on the frets/fingerboard which, of course, changes the resulting sound dramatically. This behavior is always coupled to Velocity.

Slap			
Type	Strength	Distance	Deflect

Type (Slap Type) [Hard1/Hard2/Hard3/ Soft1/Soft2/ Soft3]

Selects a fret type. The hardness of the fret material is adjustable from Hard 1, 2, 3 to Soft 1, 2, 3. While the Hard types are appropriate for simulating metal frets, the softer types allow for softer materials, such as is the case with a fretless fingerboard in which the string strikes wood rather than metal.

Strength [0 – 127]

To use this parameter effectively you must know that pickup #2 is used to implement the slap function. Therefore the position of the pickup relative to the strings, and the amount of signal energy at that point are important. The Strength control limits this energy so that it is sufficient to produce the Slap but not so strong that the string strikes too much on the fingerboard.

> To simulate a fretless instrument, set the distance to a low value, select a Soft fret type, turn the Velocity down somewhat, and experiment with the Strength control until you achieve the desired result.

Distance [0 – 127]

Adjusts the distance of the strings to the fingerboard.

Deflection [0 – 127]

This setting describes the vertical deflection of the string as a function of Velocity. The higher the value, the greater the deflection. The text field always indicates the resulting deflection.

Pitch Menu

Pitch		
>Envelope	>LFO	>Global

Envelope sub-menu

The Six-String includes an A/D pitch envelope to modulate pitch. This is used to create various effects, such has having the initial pitch start a little lower or higher as it does frequently in the real world. A Slap Bass is a good example. A convincing slap effect results only if the tension of the string increases, with a corresponding increase in pitch, at the onset of the note and then quickly drops back down to normal. The curve can also be used to produce performance-controlled slap deflections. For this a Threshold parameter is available that starts the envelope from a specific velocity value.

Envelope			
EnvAtt	EnvDec	Vel>Env	EnvDepth>
Treshold	TKF Key	TKF Int	

EnvAtt (Attack Time) [0 – 1000 ms]

Sets the time the envelope takes to reach its maximum value.

EnvDec (Decay Time) [0 – 1000 ms]

Sets the time the envelope takes to return to 0 from its maximum value.

Vel>Env (Velocity>Enve) [0 – 127]

Controls how strongly velocity affects the envelope. At a value of 0, velocity has no influence. With higher values, the envelope is modulated more strongly. Small values produce weak envelope modulation; large values produce strong modulation.

EnvDepth (Envelope Depth) [0 – 127]

Controls the strength with which the envelope modulates the pitch. The value can be positive or negative. With positive values the pitch rises and then drops back to its original pitch. With negative values the pitch falls and then returns to 0.

Tres (Treshold) [0 – 127]

Sets a MIDI velocity value at which the envelope will start. The effect of the envelope curve will only become audible if this value is reached or exceeded.

TKF Key (TFK Center Key) [C-2 – G8]

The expansion or contraction of the envelope (in time) can be linked to the note played (*key follow*). The note selected here is the note at which there is no expansion or contraction. The envelopes of notes above or below this note are expanded or contracted, much like KF Sens would do.

TKF Int (TKF Intensity) [-100 – +100 %]

Controls the compression/expansion of the envelope (decrease/increase of overall envelope time) as a function of the note played. Positive values expand the envelopes of notes above the KF key and compress the envelopes of notes below it. Conversely, negative values expand the envelopes of notes below the KF key and compress the envelopes of notes above it.

You can use this parameter to adjust low notes to take longer to reach the normal pitch, and high notes to reach it very quickly, as is the case with real strings in the real world.

LFO sub-menu

LFO
LFO Freq AT Level MW Level Gl Lev

LFO Freq (Frequency) [0 - 150 Hz]

Sets the frequency of the LFO.

AT Level (Aftertouch Level) [0 - 127]

Controls the increase in intensity of the LFO through aftertouch.

MW Level (ModWheel Level) [0 - 127]

Controls the increase in intensity of the LFO through the Modulation Wheel.

Gl Lev (Global Level) [0 - 127]

Sets a permanent (fixed) LFO pitch modulation level.

Global sub-menu

Global
Coarse Fine AT>Pitch

Coarse (Coarse Tune) [-12 – +12 semitones]

Sets the overall tuning of the Six-String within a range of -12 to +12 semitones.

Fine (Fine Tune) [-100 – +100 Cent]

Adjusts the fine tuning within a range of +/- 100 cents.

AT>Pitch (Aftertouch>Pitch) [-12 – +12]

Allows Aftertouch to control pitch modulation. The value represents the maximum deflection with full Aftertouch (in semitones).

You can use this to simulate a tremolo arm ("whammy bar") very nicely without having to remove your hands from the keyboard to move the Pitchbend Wheel.

Control Menu

Control			
Single	PortMode	PortTime	PW Range

Single (Single Voice Mode) [Off / On]

Diese Option schaltet den Six-String in den Monomode (single voice).

PortMode (Portamento Mode) [Off/Portamento/Glissando/FingPorta/FingGliss]

When Portamento or Glissando is switched on, the pitch will either glide (Portamento), or progress in a stepwise fashion (Glissando), from one note to the next over a fixed time period (adjustable - see next section).

You can adjust this parameter to *Off, Portamento, Glissando, fingered portamento (Fing. Portameto)* or *fingered glissando (Fing. Glissando)*.

Note that Portamento/Glissando is effective only when playing in a legato style.

PortTime (Portamento Time) [0 – 127]

Used in conjunction with the Port/Gliss option, above. This sets the amount of time to glide or gliss from one note to the next.

PW Range (PitchWheelRange) [0 – 12 semitones]

Sets the maximum value in semitones of the pitch modulation controlled by the Pitch Wheel.

B-2003

Introduction

The B-2003 is a drawbar organ in the tradition of the unique Hammond B3™. All the features of the original have been precisely modeled: 92 tone wheels with full polyphonic performance, key clicks and percussion, scanner vibrato, overdrive, and a rotary speaker (Leslie™) effect.

To better understand the various parameters we recommend you also read the B-2003 chapter in the online manual as the structure is more easily grasped through the graphics and illustrations.

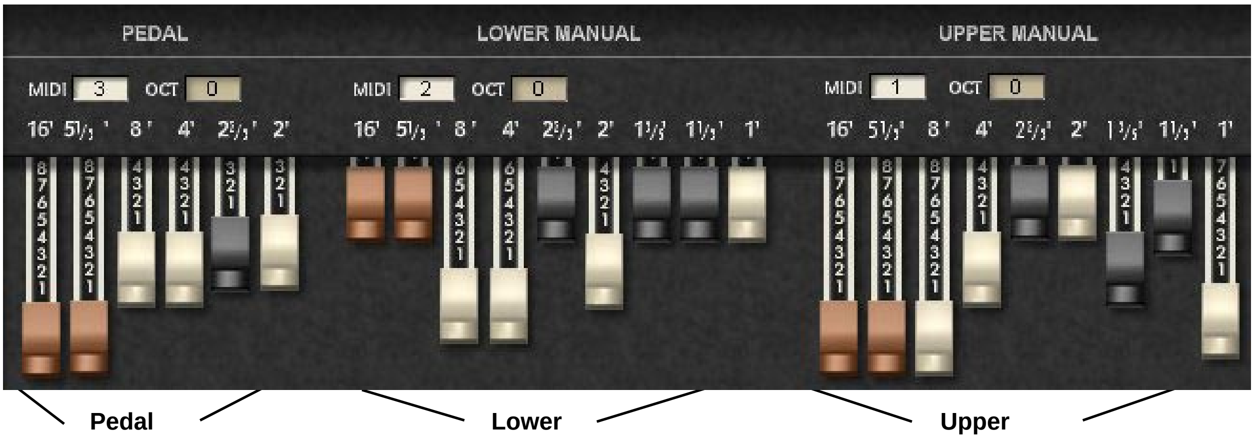
Parameter

To access the B-2003 parameters, open the B-2003 menu (single mode) or the Slots/B-2003 menu (multi mode).

The B-2003's parameters are distributed throughout 8 menus and their sub-menus: **Pedal, Lower, Upper, Perc&KC, Modeling, Effects, External, MIDI**):

```
B-2003
  >Pedal    >Lower    >Upper   >Perc&KC>
>Modeling  >Effects >External   >Midi  <
```

Drawbars



The Hammond B3™ has two keyboards (upper and lower manuals) and foot pedals for bass. The B-2003 lets you reserve sections of a keyboard, or different keyboards, by MIDI channel or address, so you can use several keyboards or keyboard zones to implement the same functionality (see Midi menu).

The sound produced by each section (pedal, lower manual, upper manual) is controlled by individual *drawbars*. Each drawbar corresponds to a specific frequency range. The label on a drawbar (a number followed by an apostrophe) comes from pipe organ technology and simply refers to the length of the pipes required to produce the sound. Longer pipes (higher numbers) produce deeper octaves.

The designations 8', 4', 2' and 1' (white) correspond to the fundamental pitch, and the 2nd, 4th and 8th harmonics respectively. The designations 2 1/2', 1 3/5' and 1 1/3' (black) correspond to the 3rd, 5th, and 6th harmonics. The designations 16' and 5 1/3' (brown) correspond to half of the fundamental and half the third harmonic respectively.

For the pedal range, the upper three drawbars are eliminated.

Pedal Menu

Each drawbar can be pulled out in nine steps (0-8) to regulate the relative volume of the respective frequency range.

```
Pedal
  16'      5 1/3'     8'        4'   >
  2 2/3'     2'                      <
```

16’	**(Drawbar 16’)**	**[0 - 8]**
5 1/3’	**(Drawbar 5 1/3’)**	**[0 - 8]**
8’	**(Drawbar 8’)**	**[0 - 8]**
4’	**(Drawbar 4’)**	**[0 - 8]**
2 2/3’	**(Drawbar 2 2/3’)**	**[0 - 8]**
2’	**(Drawbar 2’)**	**[0 - 8]**

Upper Menu

Each drawbar can be pulled out in nine steps (0-8) to regulate the relative volume of the respective frequency range.

```
Upper Manual
  16'      5 1/3'     8'        4'   >
  2 2/3'     2'     1 3/5'   1 1/3'  >
    1'                               <
```

16’	**(Drawbar 16’)**	**[0 - 8]**
5 1/3’	**(Drawbar 5 1/3’)**	**[0 - 8]**
8’	**(Drawbar 8’)**	**[0 - 8]**
4’	**(Drawbar 4’)**	**[0 - 8]**
2 2/3’	**(Drawbar 2 2/3’)**	**[0 - 8]**
2’	**(Drawbar 2’)**	**[0 - 8]**
1 3/5’	**(Drawbar 1 3/5’)**	**[0 - 8]**
1 1/3’	**(Drawbar 11/3’)**	**[0 - 8]**
1’	**(Drawbar 1’)**	**[0 - 8]**

Lower Menu

Each drawbar can be drawn out in nine steps (0-8) to regulate the relative volume of the respective frequency range.

```
Lower Manual
  16'      5 1/3'     8'        4'   >
  2 2/3'     2'     1 3/5'   1 1/3'  >
    1'                               <
```

16’	**(Drawbar 16’)**	**[0 - 8]**
5 1/3’	**(Drawbar 5 1/3’)**	**[0 - 8]**
8’	**(Drawbar 8’)**	**[0 - 8]**
4’	**(Drawbar 4’)**	**[0 - 8]**
2 2/3’	**(Drawbar 2 2/3’)**	**[0 - 8]**
2’	**(Drawbar 2’)**	**[0 - 8]**
1 3/5’	**(Drawbar 1 3/5’)**	**[0 - 8]**
1 1/3’	**(Drawbar 11/3’)**	**[0 - 8]**
1’	**(Drawbar 1’)**	**[0 - 8]**

Perc&KC Menu

Here you'll find the sub-menus for the Percussion and Key Click effect, and the Swell parameter.

Perc & Key Click
 >Perc KeyClick Swell

Key Click **[0 – 127]**

The electro-mechanical keyboard of the Hammond B3™ produces a clicking or snapping which was not (at first) necessarily considered desirable. However, over time this has been accepted as one of the characteristic elements of the classic Hammond sound. This control adjusts the level of the clicking noise.

Swell **[0 – 127]**

Controls the overall volume of the B-2003.

Perc sub-menu

The Hammond Percussion effect is a patented circuit that changes the attack of a played note by adding an additional tone.

The percussion effect applies only to the upper manual.

The percussion effect occurs only if no other key is in play. When playing legato style, only the first note will exhibit the percussion component.

Percussion
 Harmonic Level Decay On/Off

Harmonic **[Sub/ Sub-3 / Fund / 1st / 2nd / 3rd / 4th / 5th / 6th]**

Controls the frequency of the percussion effect, or the pitch of the added tone. The possible pitches available correspond to the drawbars.

Level **[0 – 127]**

Controls the strength of the percussion effect by controlling the level of the added tone.

Decay **[0 – 127]**

Controls the length of the percussion effect, or the duration of the added tone.

On/Off (Perc On/Off) **[Off, On]**

Switches the percussion effect *On* or *Off.*

Modeling Menu

This menu permits you to adjust the parameters of the ToneWheel and Drawbar emulation as well as those related to sound attack and envelope.

Modeling Parameters
>ToneWheel>Drawbars >Envelope

ToneWheel sub-menu (Tone Wheels)

Tone Wheels
Conditon Tuning

Condition **[0 – 127]**

The tone wheels of a genuine Hammond organ are subject to wear, which impairs the sound quality. This control lets you adjust the virtual condition of the tone wheels from 'brand-new' (*NEW*) to 'in need of repair' (*REPAIR*).

Tuning **[-1.00 – 1.00]**

This rotary control adjusts the overall tuning steplessly within a range of +/- 1 tone.

Drawbars sub-menu

Drawbars
 Leakage Distortion

Leakage **[0 – 127]**

With some Hammond organs you can still hear overtones in the background even if the drawbars are fully pushed in. This arises from cross modulation in the circuits. This parameter lets you include this phenomenon in the instrument's behavior.

Distortion **[0 – 127]**

This control lets you add even more distortion (above and beyond the *Drive* parameter for the tube emulator) to the degree you set here.

Envelope sub-menu

The B-2003 can respond to your dynamic playing style differently than the original, and the signal envelope can also be altered.

Envelope

Attack Release Velocity

Attack (Attack Time) [0 – 127]

Specifies an attack time for the signal. At the minimum setting, the signal fades in softly, while it plays at full strength immediately in the maximum setting.

Release (Release Time) [0 – 127]

This control specifies the signal's release time. In the maximum position the signal is cut off immediately when the key is released, while in the minimum position it is allowed to ring out gradually.

Velocity [Off / On]

With this rocker switch set to *On*, the volume of the signal is controlled by how hard you strike the keys (Velocity control).

Effects Menu

Here you'll find sub-menus for the effects vibrato, tube emulation, tone control, and rotor.

Effects

>Vibrato >Drive >Tone >Rotor

Vibrato sub-menu

Vibrato

Lower Upper Type

Lower / Upper (Lower / Upper On/Off) [Off / On]

Vibrato (periodic pitch variation) can be switched on or off for each manual separately.

Type [C-1 / V-1 / C-2 / V-2 / C-3 / V-3]

The rotary switch selects the type and strength of the effect. The V1, V2, and V3 positions produce only vibrato in increasing degrees. The positions C1, C2 and C3 add a chorus effect to the vibrato. C3 is the position most favored in jazz or rock.

Drive sub-menu

Drive

Drive On/Off Output Swell

Drive [0 – 127]

Controls the degree of emulated tube distortion.

On/Off (Drive On/Off) [Off / On]

Enables or disables the emulated tube distortion of the B-2003. This is a reproduction of the amplifier overdrive in the original Leslie cabinet.

Output [0 – 127]

Because the distortion changes the overall volume level, you can compensate for the change with this control.

Swell [0, ..., 127]

Controls the overall volume of the B-2003. Swell comes before the tube distortion circuit. Therefore, when Drive is activated, the Swell setting also affects the degree of distortion produced by that circuit.

Tone sub-menu

There are two controls to adjust the overall tone quality of the instrument.

Tone	
Treble	Bass

Treble **[-63 - +63]**

Controls the level of high frequencies in the output.

Bass **[-63 - +63]**

Controls the level of low and mid frequencies in the output.

Rotor sub-menu

The original Leslie speaker operated by creating periodic modulations through mechanical rotation of the speakers. The B-2003 emulates this in software.

Here you can adjust several parameters related to the Leslie emulation. Many Leslie cabinets had a speaker for the lows (bass) and a horn for the highs (treble). These rotated differently, and this behavior can be controlled in detail in the B-2003.

Rotor			
>Horn	>Bass	>Mic	>Control

Horn sub-menu

Horn				
Slow	Fast	Accel	Brake	>
Tone				<

Slow (Slow Speed) **[0 – 40 Hz]**

Specifies the speed of the horn speaker when >Control/ Speed is set to *Slow.*

Fast (Fast Speed) **[0 – 40 Hz]**

Specifies the speed of the horn speaker when >Control/ Speed is set to *Fast.*

Accel (Acceleration) **[0 – 30 s]**

Sets the amount of time it takes thehorn speaker to reach the new speed when the speed (>Control/Speed) is switched from Slow to Fast.

Brake **[0 – 30 s]**

Sets the amount of time it takes the horn speaker to slow down when the speed (>Control/Speed) is switched from Fast to Slow.

Tone (Horn Tone) **[0 – 127]**

This control permits alteration of the tone color in such a way that the resonances produced by the Rotor are shifted – the Rotor "sings" with a brighter or darker tone.

Bass sub-menu

Bass				
Slow	Fast	Accel	Brake	>
Tone				<

Slow (Slow Speed) **[0 – 40 Hz]**

Specifies the speed of the bass speaker (woofer) when >Control/Speed is set to *Slow.*

Fast (Fast Speed) **[0 – 40 Hz]**

Specifies the speed of the bass speaker (woofer) when >Control/Speed is set to *Fast.*

Accel (Acceleration) **[0 – 30 s]**

Sets the amount of time it takes the bass speaker to reach the new speed when the speed (>Control/Speed) is switched from *Slow* to *Fast*.

Brake **[0 – 30 s]**

Sets the amount of time it takes the bass speaker to slow down when the speed (>Control/Speed) is switched from Fast to Slow.

Tone (Horn Tone) **[0 – 127]**

This control permits alteration of the tone color in such a way that the resonances produced by the Rotor are shifted – the Rotor "sings" with a brighter or darker tone.

RotorSpd **(Rotor Speed)** **[Slow / Fast]**

Switches the speed of the rotor (Leslie speaker rotation) between slow and fast.

Micro sub-menu

The B-2003 emulates our perception of the bass and treble Leslie speakers with seperate microphones.

Microphones	
Spread	Balance

Spread **[0 – 127]**

This parameter allows you to control the "breadth" of the horn sound, as if you were varying the physical separation of two microphones which are picking up its sound.

Balance **[-63 - +63]**

This control establishes the volume relationship between the treble and bass speakers. In the full left position only the treble speaker is heard, and at full right, only the bass. In the center position the two are at equal levels.

Control sub-menu

Rotor Control				
Speed	On/Off	MW/AT	Sel	Threshold

Speed **[Slow / Fast]**

Switches the speed of the rotor (Leslie speaker rotation) between slow and fast.

On/Off (Rotor On/Off) **[Off, On]**

Switches the rotor effect *On* or *Off*.

MW/AT Sel [Mod Wheel, Aftertouch] MW/AT Select

Alternatively, you can use one of two controllers. Both ModWheel and Channel Aftertouch (Pressure) are available. The setting of the MW/AT parameter determines which of the two is used.

Threshold [0, ..., 127] Threshold

When a controller is used, this parameter sets the controller value at which the rotor speed switches.

External Menu

An external signal can be passed through the effects (Vibrato, Drawbar Distortion, Drive, Tone and Rotor) of the B-2003.

The Vibrato switch of the Lower manual also switches this effect on and off for the external signal.

External	
Level	Ext On/Off

Level (Input Level) [0 – 127]

Adjusts the level of the external signal.

Ext On/Off (External On/Off) [Off,On]

Switches the external signal on or off.

Midi Menu

Here you specify the MIDI channel for each section (Upper manual, Lower manual, Pedal)

```
Midi Channels
Pedal Ch  Lower Ch  Upper Ch        >
Lower Split          Upper Split    >
  Pedal Oct   Lower Oct   Upper Oct
```

Pedal Ch (Pedal Channel) [1- 16, omni]

MIDI channel for the Pedal section.

> **The Pedal section functions only when it has its own MIDI channel assigned to it, or when a split between Lower and Pedal has been set up.**

Lower Ch (Lower Channel) [1- 16, omni]

MIDI channel for the Lower manual.

> **Lower functions only when it has its own MIDI channel assigned to it, or when a split between Upper and Lower has been set up.**

Upper Ch (Upper Channel) [1- 16, omni]

MIDI channel for the Upper manual.

> **The MIDI channel of the upper manual is the master MIDI channel for the B-2003 – this is also the channel on which it responds to controller messages.**

Lower Split [C-2, ..., G8] Lower Split Key

Applying the same MIDI channel setting to neighboring sections activates a split between those sections. This permits any two neighboring sections, or all three sections, to be played from different zones of a single keyboard. The **Lower Split** control sets the boundary between **Pedal** and **Lower** keyboard zones when a split between these two sections is activated.

Upper Split [C-2, ..., G8] Upper Split Key

Sets the boundary between **Lower** and **Upper** keyboard zones when a split between these two sections is activated.

Pedal Oct [-1, ..., +1] Pedal Octave

Permits transposition of the Pedal section by one octave up or down.

Lower Oct [-1, ..., +1] Lower Octave

Permits transposition of the Lower section by one octave up or down.

Upper Oct [-1, ..., +1] Upper Octave

Permits transposition of the Upper section by one octave up or down.

Minimax

Introduction

What's so Max about the Mini?

Just what is it about the Minimax?

Today you would probably not expect too much in terms of sound from a synthesizer with 3 oscillators and noise, a mixer, followed by a filter with an amplifier and two envelope generators. But we're not talking about just any synthesizer, we're talking about the synthesizer!

Minimax is an emulation of the probably best-known of all vintage synths. It's simple structure makes it easier to learn sound creation techniques than on many other synthesizer models. And its sound is famous. The oscillators are strong and powerful, the filter pack and envelope generators blindingly fast.

Many have tried to duplicate these characteristics. However, attempts by both hardware and software manufacturers have failed. With hardware, the different electronic components often used changed the sound. And in software only some parts of the original, such as the filter, were successfully emulated. Other elements, like the oscillators and envelopes, were imported from outside sources.

CreamWare knows how difficult a true emulation is. From today's perspective, the Miniscope and Miniscope MkII must be considered as great synthesizers in their own right, but not as emulations of the original.

Minimax is different than anything that's come before. Minimax does not just emulate parts of the instrument - Minimax is the instrument. Where there have been limits to the hardware, such as original components no longer being available, and where software has not dared to go, there goes Minimax.

With the Minimax we modeled all important sound elements on the original circuit diagrams. And throughout the process we constantly recalibrated our design to conform with the original. Still the reproduction was not exact. Calibration and adjustment was only part of the work we had to do.

We had yet another hurdle to jump before we could claim success. The original synth produces frequencies that, had we relied on the usual algorithms, would have lead to aliasing. And those familiar with aliasing know what an unpleasant noise it is to have in a signal. The modules used in the Minimax are built on newly-developed algorithms free from aliasing. For this reason Minimax easily handles even the wildest filter, FM, or oscillator modulations. And the beauty of it is - Minimax always sounds like the original. Minimax represents the pinnacle of synthesis algorithm development.

Structure and Overview

Like the original, Minimax has an oscillator section with three oscillators. Oscillator 3 can also serve as an LFO. The oscillator signals are mixed with noise or an external signal, if desired, in the mixer section. From the mixer, the combined signal is sent to an amplifier and a filter, each of which has an associated envelope generator with adjustable attack, release and sustain.

The quality of the algorithms is readily apparent in all sections of the synthesizer. The full-range oscillators produce signals with extended highs giving you soaring sound with great vitality and energy. We also paid careful attention to the saturation behavior of the mixer section, both for internal and external signals. Saturation gives the sound more ability to penetrate. More power.

The filter, in particular, benefits from the new algorithms. Although previous filters had resonance, it was never as convincing analog resonance. With digital filters designers had to make sure that aliasing would not develop during filter sweeps. Therefore, many filters limit the resonance range (or do not allow it to fully open) to prevent the signal from exceeding the aliasing threshold (the Nyquist frequency, half the sample rate). Filter sweeps with these filters are useful, but are less lively than they could be. The Minimax filter is completely convincing all down the line. The filter has a unique liveliness and produces all the resonance and distortion you could ever wish for in a filter. Resonance can be fully opened up and sweeps are free of aliasing - even beyond half the sampling frequency.

Finally, the envelope generators need not shrink from comparison with their analog counterparts either. Not only are they fast - they reproduce the overall behavior of the original with great accuracy.

Although the goal of the development of Minimax was to create an absolutely faithful reproduction, we couldn't resist adding a few enhancements. The envelopes have adjustable velocity and switchable trigger behavior, and the normal low-note priority can be switched instead to Last-Note-Priority.

Our love of detail is also apparent in the graphic front panel. The controls were treated with particular care so that they correspond exactly to the originals. If you still have any of your old sound sheets, you can set them up on the Minimax and then store them as presets!

To better understand the various parameters we recommend you also read the Minimax chapter in the online manual as the structure is more easily grasped through the graphics and illustrations.

Parameters

To access the Minimax parameters open the Minimax menu (single mode) in Edit mode or the Slots/Minimax menu (multi mode).

The Minimax parameters are available in 5 menus (Osc, Mixer, Filter, Loudness, Control).

```
Minimax
   >OSC      >Mixer    >Filter   >Loudness>
 >Control                                <
```

Osc Menu (Oscillator Bank)

The oscillator bank consists of three oscillators. Each oscillator has controls to adjust the octave and select the waveform. Oscillators 2 and 3 also have Frequency controls for detuning. Oscillator 3 can also serve as a modulation source in which case it ignores frequency input from the keyboard and functions as an LFO. The LFO rate is then controlled by the Range and Frequency controls. Pitch modulation can be turned on or off with a switch.

The oscillator parameters are located in 4 sub-menus: Osc1, Osc2, Osc3, and Pitch.

```
OSC
  >Osc1      >Osc2      >Osc3      >Pitch
```

Osc 1 -3 sub-menus (Oscillator-1 - 3)

```
Osc3                    Osc3 Range    On
  Range    Frequency  Waveform  Control
```

Range **[LO,32',16',8',4',2']**

Sets the octave of the oscillators. The values 32' and 16' are low octaves suitable for bass while 8' and 4' are good for lead voices. To use oscillator 3 as an LFO, select the value Lo.

Frequency **[-64, ..., 63] (nur Osc2 und Osc3)**

Adjusts the tuning of oscillators 2 and 3 with respect to oscillator 1. The scaling indicates the amount of detune in intervals.The maximum adjustment is about 9 semitones.

Waveform **[Triangle,TriSaw*, Saw*, Square, WideRect, NarrowRect]**

Selects the waveform for an oscillator. Each oscillator can produce one of six possible waveforms: triangle (Triangle), a mix of triangle and sawtooth (TriSaw), sawtooth up (Saw), square (Squ), wide pulse (WideRect), and narrow pulse (NarrowRect).

*Oscillator 3 generates a sawtooth down wave (SawDown) in place of the triangle/sawtooth combination wave. The sawtooth up wave in Osc3 is accordingly called SawUp.

Control [On, Off] (Osc3 only)

This switch sets oscillator 3 to ignore keyboard frequency control. You can then use oscillator 3 to provide a fixed frequency as a sonic element, or to serve as an LFO. When using oscillator 3 as an LFO, adjust the coarse frequency with the Range control and the fine tuning with the Frequency control.

Pitch sub-menu

```
Pitch
   Tune     Osc Mod    Pitch AT   PB Range
```

Tune [-64, ..., 63]

This control adjusts the overall tuning of the instrument. The range is +/- 2.5 semitones.

Pitch Mod (Pitch Modulation) [Off, On]

Enables or disables pitch modulation for the 3 oscillators.The source configured under Control/ModWheel/ModMix serves as the modulation signal. This can be Osc3, Noise, or a combination of both.The intensity is controlled by the modulation wheel and related parameters.

Pitch AT (Pitch Aftertouch) [-64, ..., 63]

Controls the intensity and direction (positive or negative) of the pitch modulation for the 3 oscillators. This parameter lets you control pitch modulation using MIDI aftertouch.

PB Range (Pitch Bend Range) [0, ... , 24]

Sets the maximum pitch bend deflection.

Mixer Menu

In this section the signals are mixed before being sent to the filter section. Altogether you have 5 possible signals available: Osc1-3, Noise, and an External signal. Each source has parameters to control the level and switch it on or off. External signals can be amplified to the point of distortion for use as a sonic element.

The Mixer parameters are distrubted in three sub-menus (Osc 1-3, Noise, and External):

```
Mixer
 >Osc 1-3   >Noise  >External
```

Osc 1-3 sub-menu

```
Osc 1-3             Osc1 Volume    95
 Osc1 Vol  Osc1 Sw   Osc2 Vol  Osc2 Sw
 Osc3 Vol  Osc3   Sw
```

Osc1-3 Vol (Osc1-3 Volume) [0, ...,127]

Controls the volume levels of the respective oscillators.

Osc1-3 Sw (Osc1-3 Switch) [Off, On]

Switches the respective oscillators on or off.

Noise sub-menu

```
Noise
  Volume   Noise Sw    Type
```

Volume [0, ..., 127]

Controls the level of the noise signal. When noise is used as a modulation source, the volume level has no effect on the intensity of the modulation.

Noise Sw (Noise Switch) [Off, On]

Switches the Noise Generator on or off.

Type (Noise Type) [White, Pink]

Selects the noise color. When noise is used as a modulation source, the color has an effect on the modulation.

External sub-menu

```
External
 Ext Vol    Ext Sw   Feedback   Ext Src
```

Volume (External Volume) [0, ..., 127]

The volume level of the external signal. For this to have an effect, a sound source must be connected to the Audio input of the Minimax.

Ext Sw (External Switch) [Off, On]

Switches the external sound source on or off.

Feedback [Off, On]

With Feedbakc switched on, the synthesizer's output is fed back to the External Input. You can amplify the level of the external signal to the point of distortion for use as a sound element. External Volume controls the level of the feedback and/or distortion. In this mode, any signal connected to the External input is ignored.

Ext Src (External Source)
[Slot1, Slot2, Slot3, Slot4, Analog, USB]

Select the mixer channel strip for external signal source here.

Filter Menu

Along with the envelope, the filter controls changes in the tone color. The filter is a 24dB/octave low-pass filter. Frequencies beneath the Cutoff frequency pass freely, while frequencies above are attenuated by 24dB/octave. There's probably not a lot to say about this filter other than it is generally considered to be one of the best sounding filters ever put in a synthesizer. The Minimax confirms this.

The Filter menu parameters are distributed in 3 sub-menus (Vcf, Envelope, and Mod):

Filter

>Vcf >Envelope >Mod

Vcf sub-menu

Vcf

Cutoff Emphasis Contour

Cutoff (Cutoff Frequency) [0, ..., 127]

The cutoff frequency is the frequency above which the spectrum is cut—overtones are attenuated. This control lets you adjust the cutoff frequency manually.

Emphasis [0, ..., 127]

This is the resonance control. Resonance results from the coupling of the filter output to the filter input, thereby reinforcing the frequencies lying near the cutoff frequency. At the maximum setting the filter begins to oscillate, producing a pure sine tone at the cutoff frequency. You can use emphasis as a possible sixth sound source.

Contour (Amount of Contour) [0, ..., 127]

Adjusts the intensity of the envelope modulation. The cut-off frequency follows the progression (contour) of the envelope with the degree of change controlled by this setting. At the beginning and end points of the envelope the cutoff is the adjusted Cutoff Frequency.

Envelope sub-menu (Filter Envelope)

Filter Envelope

Attack Decay Sustain Velocity

Attack [0, ..., 127]

Duration of the first envelope segment. In the attack phase the cutoff frequency increases to its maximum value during the time configured here. Adjust the amount of the increase with the Amount of Contour control. The maximum value is determined by the adjusted cutoff frequency and the Amount of Contour setting.

Decay [0, ..., 127]

Duration of the second envelope segment. In the decay phase the signal falls to the sustain level during the time configured here. If Decay is enabled under Control/ TrigMode the adjusted Release time is assumed here.

Sustain [0, ..., 127]

The envelope's third segment. This is the level held after the decay segment completes. The actual cutoff frequency is determined by this value and the Amount of Contour setting.

Velocity [0, ..., 127]

The amount of modulation of the envelope as influenced by key velocity. This setting controls the envelope peak level between minimum and maximum depending on how hard you strike the keys on your keyboard. This controls how much your playing style (hard or soft) influences the tone quality.

Mod sub-menu (Modulation)

Mod
Vcf Mod Vcf Kybd1 Vcf Kybd2 Vcf AT

Vcf Mod (Vcf Modulation) [Off, On]

Enables additional filter modulation. The Modulation Mix section serves as the modulation source (in the Control/ ModWheel menu). The signal can come from Oscillator 3, Noise, or a combination of both. The Modulation Wheel controls the intensity of the modulation.

Vcf Kybd1-2 (Vcf Kybd Ctrl 1-2) [Off, On]

These two switches enable cutoff Keyfollow in two stages.When the first switch is on, the cutoff frequency follows the keyboard at a rate of 1/3 octave cutoff frequency change per keyboard octave. With switch two (only) on, the cutoff follows at 2/3 octave per keyboard octave. With both switches on, the cutoff frequency changes an octave for every keyboard octave change.

Vcf AT (Vcf Aftertouch) [-64, ..., 63]

Intensity and direction of filter cutoff modulation. Permits generation of filter sweeps controlled by aftertouch.

Loudness Menu

Along with the envelope generator, the amplifier controls the progression of the volume level of the sound.

The Loudness parameters are located in two sub-menus (Amp, and Envelope):

Loudness
>Amp >Envelope

Amp sub-menu (Amplifier)

Amplifier
Volume Amp AT

Volume (Amp Volume) [0, ..., 127]

Overall volume level of the synthesizer.

Amp AT (Amp Aftertouch) [-64, ..., 63]

Intensity and direction of volume modulation. This option is particularly interesting in connection with the internal feedback feature. The amount of overdrive can be controlled via aftertouch.

Envelope sub-menu (Loudness Envelope)

Loudness Enveloper
Attack Decay Sustain Velocity

Attack [0, ..., 127]

The first envelope segment. During the attack phase of the envelope the volume reaches its maximum level. This setting adjusts how quickly the level is reached.

Decay [0, ..., 127]

Duration of the second envelope segment. In the decay phase the signal falls to the sustain level during the time configured here. If Decay is enabled under Control/ Env&Kybd the adjusted Release time is assumed here.

Sustain [0, ..., 127]

The third envelope segment. This is the volume level at which the sound is held at the end of the decay segment.

Velocity [0, ..., 127]

The amount of modulation of the envelope as influenced by key velocity. This setting controls the envelope peak level between minimum and maximum depending on how hard you strike the keys on your keyboard. This controls how much your playing style (hard or soft) influences the volume level of the sound.

Control Menu (Controllers)

This section covers the general settings and provides some tips regarding modulation techniques.

The Control parameters are located in three sub-menus (ModWheel, Glide, and TrigMode):

Controllers		
>ModWheel	>Glide	>TrigMode

ModWheel sub-menu

ModWheel		
MW Int	MWW Offs	Mod Mix

MW Int (MWheel Intensity) [0, ..., 127]

Maximum intensity of the modulation of the oscillators and filter producible by the modulation wheel.

MW Offs (MWheel Offset) [0, ..., 127]

The fundamental intensity of the modulation signal applied to the oscillator bank and the filter with the modulation wheel at zero. The modulation wheel increases the intensity according to the offset and intensity settings.

Mod Mix (Modulation Mix) [0, ..., 127]

Controls the proportion of oscillator 3 and noise in the modulation signal. The resulting signal is used to modulate the oscillator bank and filter. To hear the result you must have modulation switched on and the modulation wheel set to produce some intensity.

Glide sub-menu

Glide	
Glide	Glide T

Glide [Off, On]

Enables the Glide function. When glide is on one note glides into the next (glissando) at an adjustable speed (see Glide Time, below).

Glide T (Glide Time) [0, ..., 127]

Configures the time it takes to glide from one note to the next when the Glide function is enabled.

TrigMode sub-menu (Triggers&Modes)

Trigger&Modes			
Env Decay	Env Retrig	Single	Low Note

Env Decay [Off, On]

Switches the decay times of the envelopes to the release time. The release times are then controlled with the decay controls. If Decay is off, the minimum release time is used.

EnvRetrig (Env Retrig Mode) [Off, On]

Switches the retrigger mode of the envelope generators from legato to retrig. When EnvRetrig is on, the envelopes trigger every time a new key is pressed, even if you are playing a legato style (that is, when you play a new note before you release the previous one).

With Low Note on and EnvRetrig off, the behavior of the Minimax is the same as that of the original. For some sounds, such as pads and sequencer sounds, it may be appropriate to switch Low Note off and EnvRetrig on.

Single (Single Mode) [Off, On]

Forces the instrument to operate in single-voice mode, regardless of how many voices are actually currently loaded. Guarantees proper performance of solo sounds with portamento.

Low Note (Lo Note Priority) [Off, On]

Enabling Low Note On gives low notes a priority over high notes. A higher note cannot displace a low note. When switched off, the last-played note will always have priority.

Vectron Player

Overview

With the Vectron Player you can play any of the presets available for the optional Vectron synthesizer. This puts yet another tone generation technique–vector synthesis– at your disposal. After loading a Vectron preset you can change a (limited) number of parameters. The Player's preset manager then lets you save these changes as a new preset.

Vector Synthesis

The synthesis technique employed in the Vectron is based on a configuration of four oscillators with special *vector control*, a low pass filter, and a complex modulation matrix. This arrangement can produce extremely lifelike spectral progressions. Conceptually, each oscillator occupies one of the four corners of a square *vector field*. The volume of each oscillator is modulated by marking a point within the vector field (the vector position). This point can be positioned statically, or modulated dynamically by various modulation sources. A special multi-segment vector envelope creates complex movements of the vector position within the vector field, and even offers a loop function whereby the envelope becomes the functional equivalent of a complex LFO.

If you are interested in further information regarding generation of sounds in the Vectron, you'll find a detailed manual for the SFP version of the Vectron in the download area of our Web site.

Parameter

Modifications to the oscillator settings are not possible with the Player.

```
Vectron Player
 Main Vol    Cutoff      Res         >Mod
```

Main Vol (Main Volume) [0, ... , 127]

Sets the overall volume of the Vectron Player.

Cutoff (Filter Cutoff) [0, ... , 127]

Adjusts the frequency above which the low pass filter begins to attenuate the higher frequencies. The slope of the Vectron filter is fixed at 24db/octave.

Res (Filter Resonance) [0, ... , 127]

Controls the filter's resonance. Frequencies around the cutoff frequency are reinforced as this value increases.

Mod sub-menu

```
Mod
JoyStickX JoyStickY JoySReturn          >
   ModW    Mod WRet                     <
```

The Joystick

With the Joystick you can modulate the relative volumes of the individual oscillators in real time. You can also choose whether the joystick output value returns to a central null position or to another pre-determined position after it is released.

If, in the current preset, the vector position is already modulated by the vector envelope or the LFO, the joystick modulation will have only a limited additional effect.

JoyStickX (Joystick Value X) [-64, ... , 63]

Joystick X (horizontal) coordinate

JoyStickY (Joystick Value Y) [-64, ... , 63]

Joystick Y (vertical) coordinate

JoySRet (Joystick Return) [Off, On]

On: The joystick springs back to its original position when released.

Off: The joystick remains in its current position when released.

The Return-To position of the joystick is editable only via the Remote software.

ModW (ModW Preset Val) [0, ... , 127]

Many Vectron presets use the modulation wheel to control pitch modulation. MIDI controller 01 is assigned to this control, so that you can perform the same control using the modwheel of your controller keyboard.

ModWRet (ModWheel Return) [Off,On]

On: The modwheel springs back to its original position when released.

Off: The modwheel remains in its current position when released.

The Return-To position of the modwheel is editable only via the Remote software.

Sequential Circuits™ Pro-One (optional)

Introduction

Structure and Overview

From today's perspective the structure of the Pro One could be called "classic". For sound production it features two multi-function oscillators capable of producing multiple waveforms – simultaneously! The oscillator signals are combined with white noise or an external signal in the mixer section and then sent to a 24dB lowpass filter with resonance, and finally to an amplifier. Two envelope generators, each with adjustable attack, decay, sustain and release, are available to modulate both the filter and the amplifier.

A modulation matrix lets you interconnect the filter envelope, oscillator B, and the LFO to achieve a wide range of modulation effects. The combinations of modulation sources and targets available through the selector switches rank this capability as one of the Pro One's notable highlights.

Another performance feature, Auto-Repeat, can be used to gate both internal and external signals. Or you can use the envelope follower with adjustable threshold to gate the signal by analyzing an external signal.

Through the use of special *Circuit Modeling* procedures, the audio signal is rendered faithful to the original – completely free of aliasing. And those familiar with the concept of aliasing know how unnatural it sounds when recreating an analog signal. The various possibilities for modulation and the resulting sounds benefit from this procedure. Wild modulations can produce signals, such as the distortions and sidebands that develop from frequency modulation, that can easily fill the full range of the audio spectrum. Signal rendition free of aliasing is absolutely essential for reproducing such spectra.

Other components of the synthesizer also profit from circuit modeling. Because the oscillators reproduce the full bandwidth, they produce more highs. The floating effect when oscillators are slightly detuned has even more liveliness when more highs are available. You can also use filter resonance to produce distortions; filter FM (frequency modulation) is possible thanks to the high-quality algorithms. The analog character of the original is captured completely.

Although being true to the original was the highest design goal during the development of the Pro One, we couldn't help adding a few enhancements. The envelope generators of the plug-in can be modulated by MIDI velocity, the envelope follower has an adjustable threshold, and we added support for MIDI clock control.

If you still have original patch sheets you can transfer them to the plug-in's graphic surface and then (finally) save the settings as presets!

To better understand the various parameters we recommend you also read the Pro-One chapter in the online manual as the structure is more easily grasped through the graphics and illustrations.

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Parameter

To access the Pro-One parameters open the Pro-One menu (single mode) or the Slots/Pro-One menu (multi mode).

The Pro-One parameters are available in 13 menus (Osc, Mixer, Filter, Amp, Lfo, Mod, Glide, Mode, Global, EnvFol):

PRO-ONE				
>OSC	>Mixer	>Filter	>Amp	>
>Lfo	>Mod	>Glide	>Mode	>
>Global	>Env Fol			<

Osc Menu

The Osc parameters are distrubted in two sub-menus (OscA, Osc B):

Oscillators	
>Osc A	>Osc B

Except for a few differences, oscillators A and B are identical. Selector switches for each oscillator enable or disable the various waveforms. For oscillator A the available waveshapes are sawtooth and pulse; for oscillator B, sawtooth, triangle, and pulse. Because all waveforms can be enabled simultaneously, a mixture of all five oscillator waveshapes is possible. Pulse width can be adjusted manually, or modulated by another signal. If using oscillator B as a modulation source, it is disconnected from keyboard frequency control. Oscillator B can also be switched to operate at low frequencies to serve as an LFO. Oscillator A can be synchronized to oscillator B.

Osc A sub-menu

OscA			
Octave	Freq	Sync/Off	>
Saw	Square	PulsWdth	<

Octave [0, 1, 2, 3]

Sets the octave base for the oscillators. Available values are 0, 1, 2, and 3. Use 0 and 1 for bass instruments, and 2 or 3 for lead sounds.

Freq (Frequency) [-64, ..., 63]

Use this control to detune the oscillators. Mixing the two oscillator signals will produce a livlelier sound. The range extends to an octave.

Sync/Off (Sync On/Off) [Off, Sync]

Enables or disables hard sync of oscillator A to oscillator B. Hard sync restarts oscillator A's waveform at the beginning of each cycle of oscillator B. In effect, this transfers the pitch of oscillator B to oscillator A. Depending on the ocatve, frequency setting, and frequency modulation of oscillator A various spectral effects can be produced.

Saw (Saw Enable) [Off, On]

Switches the sawtooth wave shape on or off.

Square (Square Enable) [Off, On]

Switches the pulse (Square) wave shape on or off.

PulsWdth (Pulse Width) [0, ..., 127]

Manual adjustment of the pulse width. To hear the effect, the pulse waveshape must be switched on. The range extends from approximately 5% to 100% of the total period. See the Modulation section for information on pulse width modulation.

Osc B sub-menu

Osc B			
Octave	Freq	Lofr/Nrm	Kybd/Off>
Saw	Triangle	Square	PulsWdth<

Octave [0, 1, 2, 3]

Sets the octave base for the oscillators. Available values are 0, 1, 2, and 3. Use 0 and 1 for bass instruments, and 2 or 3 for lead sounds.

Freq (Frequency) [-64, ..., 63]

Use this control to detune the oscillators. Mixing the two oscillator signals will produce a livlelier sound. The range extends to an octave.

Lofr/Nrm (LowFrequ/Normal) [Normal, Lofreq]

Scales the frequency range of oscillator B. When set to Lofreq, the oscillator cycles much more slowly, below the audio range, allowing it to serve as an additional LFO in the modulation matrix.

Kybd/Off (Keyboard On/Off) [Off, Kybd]

Disconnects oscillator B from keyboard control. The oscillator no longer follows the keyboard, but operates at the frequency fixed by the Octave, Frequency amd Lofreq controls. This allows it to be used as, for example, a modulation source with adjustable frequency.

SawEnable (Saw Enable) [Off, On]

Switches the sawtooth wave shape on or off.

TriEnable (Triangle Enable) [Off, On]

Because oscillator B can also be used as an LFO, it provides a triangle wave (especially useful in a LFO). This button switches the triangle wave on or off.

SquEnable (Square Enable) [Off, On]

Switches the pulse (Square) wave shape on or off.

PulsWdth (Pulse Width) [0, ..., 127]

Manual adjustment of the pulse width. To hear the effect, the pulse waveshape must be switched on. The range extends from approximately 5% to 100% of the total period. See the Modulation section for information on pulse width modulation.

Mixer Menu

Here the audio signals are mixed before going to the filter. Oscillators A and B each have their own volume controls, while a third controls the level of either the noise or an external signal, depending on which is selected in the Envelope Follower section. The noise signal is always „white“ noise.

> In the original the external audio signal is switched on when the envelope follower is enabled. In the plug-in, enabling the envelope follower and enabling the external signal are separate functions to add flexibility. Find more about this in the Envelope Follower section.

Mixer		
OscA Lev	OscB Lev	NoiseLev

OscA Lev (Osc A Level) [0, ..., 127] Osc A Level

Controls the levels of oscillators A.

OscB Lev (Osc B Level) [0, ..., 127]

Controls the levels of oscillators B.

NoiseLev (Noise/Ext Level) [0, ..., 127]

Controls the levels of the noise or external signal.

Filter Menu

The filter section consists of a modulatable low-pass filter with resonance that can be tuned to the point of oscillation. Together with the envelope generator and other modulation sources, it provides dynamic low-pass filtering. The filter has a slope of 24dB/octave. Frequencies below the cutoff frequency remain unaffected, hence the designation „low-pass“. Frequencies lying above the cut-off are attenuated at a slope of 24dB/octave.

Resonance is implemented by feeding the filter’s output back into the input, reinforcing the frequencies surrounding the cutoff frequency. Thanks to the Circuit Modeling process the filter duplicates the peculiarities of the original. The cutoff settings and resonance behavior display the typical Pro One character. For modulation, the filter uses its own envelope generator, the keyboard, and sources configured in the modulation matrix.

```
Filter
   >Vcf    >Envelope
```

VCF sub-menu

```
Vcf
  Cutoff      Res      Env Amt    Kybd Amt
```

Cutoff [0, ..., 127]

The Cutoff Frequency is the frequency above which the audio spectrum is attenuated, reducing the strength of the overtones. This control provides a manual adjustment of the base Cutoff Frequency.

Res (Resonance) [0, ..., 127]

Resonance results from feeding the filter’s output back into the input, thereby reinforcing the frequencies lying near the cutoff frequency. At full resonance, the filter oscillates, producing a sine wave at the cutoff frequency. For this reason, the filter can also be used as another sound source.

Env Amt (Envelope Amount) [0, ..., 127]

The intensity of the envelope signal. The cutoff follows the changing envelope level according to the adjusted intensity. The beginning and end points of the envelope lie at the adjusted base cutoff frequency.

Kybd Amt (Keyboard Amount) [0, ..., 127]

Controls the influence of the keyboard position over the cutoff frequency. With Keyboard Amount set to 7, the influence over the frequency is 100% - that is, the cutoff frequency doubles with each octave increase in key position.

Envelope sub-menu

```
Filter Envelope
  Attack    Decay    Sustain   Release >
Velocity                               <
```

Attack (Vcf Env Attack) [0, ..., 127]

The duration of the first envelope segment. The envelope level rises to its maximum value during the time adjusted here. The intensity of the effect is goverened by the Envelope Amount parameter. The actual maximum is determined by the Cutoff frequency and Envelope Amount settings.

Decay (Vcf Env Decay) [0, ..., 127]

The duration of the second envelope segment. In the decay phase the envelope falls to the Sustain level.

Sustain (Vcf Env Sustain) [0, ..., 127]

The third segment of the envelope. This is the *level* the envelope holds after the decay phase.

Release (Vcf Env Release) [0, ..., 127]

During the release phase the envelope falls to its minimum level. This value is the amount of time it takes to fall to minimum. The actual level it falls to is the adjusted base cutoff frequency.

Velocity (Vcf Env Velocity) [0, ..., 127]

The amount of modulation of all envelope levels through keystroke intensity. The levels of the envelope are modulated between minimum and maximum according to the adjusted modulation strength. This influences the tone quality through keyboard activity.

Amp Menu

With the help of an envelope generator, the amplifier determines the changing level of the volume of the audio signal. An ADSR envelope generator is provided to modulate the amplifier. A Volume control is also a component of this section.

Amplifier	
>Envelope	Volume

Envelope sub-menu

Envelope			
Attack	Decay	Sustain	Release >
Velocity			<

Attack (Amp Env Attack) [0, ..., 127]

The duration of the first envelope segment. The volume level rises to its maximum value during the time adjusted here.

Decay (Amp Env Decay) [0, ..., 127]

The duration of the second envelope segment. In the decay phase the envelope falls to the Sustain level during the *time* adjusted here. If Decay is enabled in the Controllers section, then the time set here is overridden.

Sustain (Amp Env Sustain) [0, ..., 127]

The third segment of the envelope. This is the volume level held after the decay phase.

Release (Amp Env Release) [0, ..., 127]

The fourth envelope segment, active only if the Decay switch is On. The envelope falls back to minimum (silence) during the realease phase. This control adjusts the amount of time it takes to fall to zero.

Velocity (Amp Env Velocity) [0, ..., 127]

The amount of modulation of all envelope levels through keystroke intensity. The levels of the envelope are modulated between minimum and maximum according to the adjusted modulation strength. This influences the volume level through keyboard activity.

Volume [0, ..., 127]

The synthesizer's overall volume level.

Lfo Menu

Like oscillator B, the Pro One's LFO provides three waveforms selectable via three switches. All, or any combination of the three waveforms can be active simultaneously leading to some very interesting results. The target and intensity of the LFO is configured in the modulation matrix.

Lfo				
Frequency	SawEnable	TriEnable	SquEnable	>
Retr/Fre	Phase	MIDI/Hertz	Note Val	

Frequency [0, ..., 127]

Adjusts the frequency of the LFO.

SawEnable [Off, On]

Switches the sawtooth wave on or off.

TriEnable (Triangle Enable) [Off, On]

Switches the triangle wave on or off.

SquEnable (Square Enable) [Off, On]

Switches the pulse (Square) wave on or off.

Rtr/Free (Retrig/Free Run) [Free Run, Retrig]

Select between *Retrig* and *Free Run* modes. In the Free Run mode, the LFO runs continuously. In Restart mode, the LFO resyncs to the waveform phase position (restarted) whenever a key is struck on the MIDI keyboard.

Phase (Initial Phase) [- 180°, ..., 180°]

Adjusts the phase position at which the LFO resyncs when a key is struck and the LFO is in Restart mode.

MIDI/Hz (LFO MIDI Hertz) [Hertz, MIDI]

Selects between manual or MIDI control of the LFO frequency.

Note Len (Note Length) [1/1, ½ dot, ½, ½ trpl, ¼ dot, ¼, ¼ trpl, 1/8 dot, 1/8, 1/8 trpl, 1/16 dot, 1/16, 1/16 trpl, 1/32, 1/32 trpl]

If MIDI is enabled, select a note length. The length of the note corresponds to a single period of the LFO.

Mod Menu (Modulation)

The Pro One has three modulation sources – filter envelope, oscillator B, and LFO – and five modulatable targets – oscillator A, oscillator B, oscillator A pulse width, oscillator B pulse width, and filter cutoff frequency. You can connect the modulation sources to the targets directly, or route them through the modulation wheel. The basic intensity of the modulation is adjusted at each source. If the modulation involves the modulation wheel, then the wheel also influences the final modulation depth. The filter envelope generator and oscillator B are polyphonic modulation sources; they operate per voice. The LFOs are monophonic; they operate on the overall sound (all voices simultaneously).

Modulation			
>From	>To	>Wheel	>Aftertch

From sub-menu

In this section you establish the basic intensities of the modulation sources and decide if the signal goes directly to the target or first to the modulation wheel for additional control over the target modulation.

From			
Fil Env	FE Route	Osc B	OscRoute>
Lfo	LfoRoute		<

Fil Env (Filter Env Amount) [0, ..., 127]

Sets the modulation depth of the filter envelope for all selected targets.

FE Route (Filter Env Route) [Direct, Wheel]

Selects the the bus for the modulation signal. The signal can be sent directly to the targets (Direct), or to the modulation wheel (Wheel) for additional manual control of the target modulation.

Osc B (Osc B Amount) [0, ..., 127]

Sets the modulation depth of oscillator B for all selected targets.

OscRoute (Osc B Route) [Direct, Wheel]

Selects the the bus for the modulation signal. The signal can be sent directly to the targets (Direct), or to the modulation wheel (Wheel) for additional manual control of the target modulation.

Lfo (Lfo Amount) [0, ..., 127]

Sets the modulation depth of the LFO for all selected targets.

LfoRoute (Lfo Route) [Direct, Wheel]

Selects the the bus for the modulation signal. The signal can be sent directly to the targets (Direct), or to the modulation wheel (Wheel) for additional manual control of the target modulation.

To sub-menu

In this section you connect the targets to the modulation busses. In the Off position, modulation is disabled.

To			
OscA Frq	Osc A PW	OscB Frq	Osc B PW >
Filter			<

OscA Frq (Osc A Frequency) [Direct, Off, Wheel]

Here you choose whether the Oscillator A's frenquency is modulated at all, directly, or through the modulation wheel.

Osc A PW (Osc A Pulse Width) [Direct, Off, Wheel]

Here you choose whether the Oscillator A's Pulse Width is modulated at all, directly, or through the modulation wheel.

OscB Frq (Osc B Frequency) [Direct, Off, Wheel]

Here you choose whether the Oscillator B's frenquency is modulated at all, directly, or through the modulation wheel.

Osc B PW (Osc A Pulse Width) [Direct, Off, Wheel]

Here you choose whether the Oscillator B's Pulse Width is modulated at all, directly, or through the modulation wheel.

Filter (Filter Cutoff) [Direct, Off, Wheel]

Here you choose whether thecutoff frequency of the filter is modulated at all, directly, or through the modulation wheel.

Wheel sub-menu (Wheel Mod)

The intensity of the modulation signal in the Wheel bus is controlled by the modulation wheel. In the original synth, this was the only control over this bus. In our version we added modulation intensity and offset for even more control. The pitch intensity through the Bender is similarly adjustable.

Wheel Mod
PB Range MW Int MW Offs

PB Range (Pitch Bend Range) [0, ..., 24]

Adjusts the maximum pitch deflection of the Pitchbend wheel in semitones. The range is from 0 to 24 semitones.

MW Int (MW Intensity) [0, ..., 127]

Sets the maximum intensity of the modulation signal for the wheel bus. This is the intensity produced when the wheel is fully open.

MW Offs (MW Offset) [0, ..., 127]

Fundamental intensity of the modulation signal for the wheel bus (with wheel at the 0 position). The modulation wheel increases the modulation beyond the basic level depending on the offset and intensity settings.

For these settings to produce modulation with a basic intensity the signal must be switched to the wheel bus. Then select the target and adjust the settings to produce the desired effect.

Aftertch sub-menu (Aftertouch)

The original Pro One filter had only a CV (control voltage) In for external control of the filter. This modulation is now controlled by MIDI aftertouch. The pitches of the oscillators can also be modulated by aftertouch. The intensity of aftertouch is individually adjustable for each target.

Aftertouch
Pitch AT AT Route FilterAT

Pitch AT (Pitch AT Amount) [-64, ..., 63]

Intensity and polarity of the pitch modulation of the oscillators by aftertouch. This lets you control pitchbend with key pressure.

AT Route (Pitch AT Route) [Osc B, Osc A+B, Osc B]

Selector switches direct the modulation to oscillator A, oscillator B, or both.

FilterAT (Filter AT Amount) [-64, ..., 63]

Intensity and polarity of the filter cutoff frequency modulation by aftertouch. This lets you apply filter sweeps using key pressure (aftertouch).

Glide Menu

With Glide enabled the pitch glides from one note to the next at a preset rate. Glide has two modes: *Auto mode* in which glide operates only on notes played with a legato style; and *Normal mode*, in which the pitch always glides whenever a new key is played.

Glide	
Rate	Auto/Nrm

Rate [0, ..., 127]

The time it takes for the pitch to glide from one note to the next. The Glide function does not have to be specifically activated – it is only necessary to enter a glide rate to enable it.

Auto/Nrm (Auto/Normal) [Normal/Auto]

Selects the Glide mode. In Auto mode only notes played with a legato style glide from one to the next. In Normal mode, each new note activates the Glide function.

To play a typical lead sound using Glide, you should be in Single mode. **Only in this mode does the Pro One plug-in behave exactly like the original.** If Single mode is switched off, polyphonic glide is also possible. Single mode is an extension found in the Global section.

Mode Menu

The settings in this section essentially define the trigger behavior of the envelope generators. Apart from the *Trigger* mode which determines how the envelopes start when a new key is played, there is also *Auto Repeat*, which retriggers the envelope with the clock or envelope follower, and a *Drone* mode, in which all played notes freeze and/or stop playing.

Mode		
Ret/Norm	Rep/Norm	Dron/Off

Ret/Norm (Retrig/Normal) [Normal, Retrigger]

Establishes the trigger behavior. In *Normal* mode the envelope is not re-started when stealing voices. This means that legato passages, in which the envlope is not triggered with each new note, is possible. In *Retrig* mode the envelope re-starts regardless of voice-stealing. This mode is appropriate for percussive sounds, for example, in which each new note retains its attack whether played legato or not.

In the original Pro One, 'low note priority' applied in Normal mode, and 'last note priority' in Retrig mode. In the plug-in version the priority settings were separated from the trigger mode for added flexibility. Note priority is now independently adjustable and located in the Global section.

Rep/Norm (Repeat/Normal) [Normal, Repeat]

In the *Normal* position the envelope responds as you would expect, with each key struck on the keyboard. By enabling *Repeat/External* you can have the envelopes trigger automatically via the LFO clock, the MIDI clock, or the Envelope Follower. To use the LFO clock, select *LFO* under Clock settings. Select *MIDI* to use the MIDI clock (16th notes). In either case, the Envelope Follower will be used for envelope triggering if it has been enabled. For the envelope follower, an audio signal must be present and the threshold adjusted appropriately. The envelope will then start when the threshold has been exceeded, and continue until if falls below the threshold level. Because automatic triggering takes place only when a key is struck on the keyboard, „gating“ individual tones or whole chords is possible. Remember to adapt the envelope times to the tempo.

Dron/Off (Drone On/Off) [Off, Drone]

You can think of Drone as a manual MIDI sustain pedal that you can turn on with this switch. If you enable Drone before or while playing a note, the note will be held, or sustained. This also applies to chords. If the adjusted polyphony is exceeded, notes are dumped. Drone behaves exactly like a MIDI sustain pedal.

Both Repeat and Drone modes were improved in the plug-in such that polyphonic play is possible. Therefore Repeat and Drone now function only when a key is pressed.

Global Menu

This section contains the parameters that apply to the entire instrument. You can also select the MIDI clock or the LFO to serve as master clock.

Global

Tune	NotePrio	Single	MIDI/Lfo

Tune (Master Tune) [-64, ..., 63]

This control changes the pitch (tuning) of the entire instrument. The range is +/- 5 semitones.

NotePrio (Note Priority) [Last Note, Low Note]

Switches between low-note and last-note priority. When set to low-note priority, high notes will be 'stolen' before lower notes when maximum polyphony is exceeded. When set to last-note priority, notes played earlier will be turned off in favor of the notes played later. Interesting effects develop through the interaction of voice stealing with envelope triggering as configured by the trigger modes in the ***Mode*** menu.

> In the original Pro One, low-note priority automatically applied in Normal mode, and last-note priority in Retrig mode. Of course, this behaviour can also be configured in the plug-in version with the appropriate settings.

Single (Single Mode) [Poly, Single]

Enables correct single voice administration regardless of how many voices are active. This ensures the correct production of sounds when using solo and/or glide.

MIDI/Lfo (Clock Source) [Lfo, MIDI]

Here you choose whether the Repeat function synchronize to the LFO or the MIDI clock.

Env Fol Menu (Env Follower)

The Envelope Follower analyzes an incoming audio signal and derives an envelope from it. The Pro One uses this signal to produce a gate for triggering the envelopes with Repeat/External switched on The threshold – the level at which the gate opens or closes – is adjustable.

Env Follower

Ext/Nois	EnvF/Off	Thresh	Ext Src

Ext/Nois (External/Noise) [Noise, External]

Selects the signal at the Noise/Ext control in the mixer. Originally in the Pro One switching on Repeat/External in the envelope follower automatically assigned the external signal to the third volume controller. This control can now be configured for either Noise or the External signal.

EnvF/Off (Env F On/Off) [Off, EnvF]

Enables the Envelope Follower. Several things happen in the Pro One when you switch on the envelope follower. If Repeat/External is switched on, the envelopes are triggered by the envelope follower gate. That is, when the threshold is exceeded the envelope starts and holds at the sustain level as long as the gate is open. When the signal falls below the threshold the envelope enters the release phase.

Thresh (Threshold) [0, ..., 127]

Here you set the level at which the envelope follower produces a gate signal. When the incoming signal rises above the level a gate is produced, and remains open until the signal falls below the threshold. Adjust the level so that, for example, certain level peaks in a drum loop produce a gate signal. The gate can then, in Repeat/External mode, trigger the envelope generator. Use this to produce all kinds of interesting rhythmic effects depending on the incoming raw material.

Ext Src (External Source)

[Slot1, Slot2, Slot3, Slot4, Analog, USB]

Select the external source here.

Vocodizer

Introduction

With the Vocodizer, Noah offers an extremely flexible vocoder. From its freely configurable and assignable filters, to individual level and pan controls for each synthesis filter output, the vocoder part of the Vocodizer gives you not only what you would expect in a classic vocoder–it extends the concept significantly. Switchable Voiced/ Unvoiced Detection allows you to optimize the voicing intelligence depending on the input signal.

When the Vocodizer is used, it occupies its own slot, in the same manner as a synthesizer. Its analysis and synthesis inputs can be connected to any desired signal source. Typically, the analog input or a USB audio input is used as the analysis source, while the synthesizer input is connected to a synthesizer in one of the parallel slots. Naturally, an external synthesizer – for example, brought in via a USB audio input – is also possible as a signal source for the synthesis input.

How does a Vocoder actually work?

A typical vocoder contains two filter banks: an *analysis* bank and a *synthesis* bank. As the name implies, the analysis bank takes the incoming voice signals and analyzes them to extract frequency and timing information, creating a "signature" for each filter in the bank. These signatures are later applied to an identically tuned filter bank in the synthesis section, and used to process a secnd, synthesized, signal (or, it could be an entirely unrelated external input signal). In both filter sections, the filters are divided into an equal number of frequency bands.

After the analysis phase, the vocoder examines each individual analysis filter output with an envelope follower to extract dynamic level information. The envelope follower then generates control signals corresponding to the changing volume levels.

The component signals from the synthesis filter bank (whose filters are tuned to the same "signatures" as the analysis filters) are multiplied by the control signals from the envelope follower. In other words, the dynamic level progression of the signals filtered by the synthesis section match exactly the dynamics of the filtered signals of the original voice input. Finally, the synthesized signal components are mixed to produce the final output. The result is vocoder output that produces a synthesized sound with the character and articulation of the original input voice. The vocoder is often used to produce the "talking synthesizer" effect made popular by artists such as Stevie Wonder.

To guarantee the highest quality results, the Vocodizer implements Voiced/Unvoiced Detection. During the analysis phase, the signal is examined for its tonal vs. noise content. Vowels, such as "A" are identified as tonal (voiced), and consonants such as "S" as noise (unvoiced). Depending on the results of this analysis, either the synthesizer signal or a noise signal is passed on to the synthesis filter bank. A noise signal (noise, in this case, referring to a signal containing all frequencies in appropriate proportions) can help to reproduce the sibilant portions of the original signal . Often the synthesized results produced by a traditional vocoder do not contain enough high frequencies. The Vocodizer's "noise substitution" strategy is able to produce more convincing results.

If you want, you can use the sibilant portions of the original signal for the unvoiced source instead of a random noise signal. In this case, the vocoder section filters the input signal in such a way that only the higher frequency components remain, letting you use the original "S" sounds to replace noise as the unvoiced source. In most cases, though, the broadband noise signal is a more appropriate source for unvoiced signal components, as the output already has a synthetic quality to it, and the original signal, if used, tends to stand out too much. But this option can come in handy sometimes, and in the long run it's all a matter of taste and appropriateness for the situation.

Presets

The Vocodizer stores different parameter groups in separate preset lists. This lets you manage, for example, the Matrix presets completely independent of the vocoder presets.

The following preset lists are available:

Vocoder: Stores and recalls settings specific to the Vocoder section.

> **A few settings are not stored, however. For example, settings related to input signal levels are not stored in presets, as it would be self-defeating to do so. Usually you will be experimenting with different presets to process the same input signal. If the input level changed each time you loaded a new preset, the process would be unnecessarily clumsy.**

Settings *not* stored in presets include: Analysis/Synthesis Input gains, Input Insert Effects, Solo, Voiced Source, and Master Insert Effects.

> **Note that all parameters are stored with Multi, and will be restored accordingly when you reload the setup.**

FilterMix (FltrSet sub-menu / Presets) : These presets store the filter settings. Along with the frequency, they also store the volume and pan positions of the individual filters. Depending on the number of filter bands, and their proximity to each other (in terms of frequency) the bandwidth of each filter is also important, and is also saved in the preset.

Matrix (Matrix sub-menu / Presets): The Matrix also has its own preset list. The positions of each of the 22 routing switches are stored in the Matrix presets.

Parameter

To access the Vocodizer parameters, open the Vocodizer menu (single mode) or Slots/Vocodizer (multi mode) while in edit mode.

The Vocodizer parameters are distributed in 8 menus (**Analysis, FltrSet, Matrix, Synthesis, Sources, InGainAn, InGainSy,** and **Output**) and their sub-menus:

```
Vocodizer
>Analysis  >FltrSet  >Matrix >Synthesis>
>Sources>  InGainAn >InGainSy   >Output<
```

Analysis Menu

The settings in this section control the behavior of the Envelope Follower (EnvF) . You can make adjustments independently for each filter type–low pass (LPF), band pass (BPF) , and high pass (HPF). In each case the parameters are the same.

```
Analysis
 >LPF EnvF >BPFsEnvF >HPF EnvF   >Gains
```

LPF EnvF sub-menu

(**LowPassFilter Envelope Follower)**

```
LPF EnvF
  Attack   Release
```

Attack **(LPF EnvF Attack) [1 - 250 ms]**

Adjusts the rate with which the Envelope Follower responds to rising signal levels.

Release **(LPF EnvF Release) [10 - 500 ms]**

Adjusts the rate with which the Envelope Follower responds to falling signal levels.

BPF EnvF sub-menu

(**BandPassFilter Envelope Follower)**

BPFs EnvF
Attack Release

Attack (BPF EnvF Attack) [1 - 250 ms]

Adjusts the rate with which the Envelope Follower responds to rising signal levels.

Release (BPF EnvF Release) [10 - 500 ms]

Adjusts the rate with which the Envelope Follower responds to falling signal levels.

HPF EnvF sub-menu

(HighPassFilter Envelope Follower)

HPF EnvF
Attack Release

Attack (HPF EnvF Attack) [1 - 250 ms]

Adjusts the rate with which the Envelope Follower responds to rising signal levels.

Release (HPF EnvF Release) [10 - 500 ms]

Adjusts the rate with which the Envelope Follower responds to falling signal levels.

Gains sub-menu

Adjusts the output level of the Envelope Follower signals. When you adjust the gain, you are indirectly controlling the overall weighting of the low pass, band pass, and high pass section.

Gains
LBF Lev BBF Lev HBF Lev

LPF Lev (Lowpass Level) [0 - 127]

Output level of the Lowpass Envelope Follower.

BPF Lev (Bandpass Level) [0 - 127]

Output level of the Bandpass Envelope Followers.

HPF Lev (Highpass Level) [0 - 127]

Output level of the Highpass Envelope Followers.

FltrSet Menu (Filter Settings)

FilterSettings
Preset: >Edit

Preset

Select a preset for the filter settings.

Edit sub-menu

Edit
AnalyseQ LinkFltQ SynthQ >Freq

AnalyseQ (Analyse Filter Q) [0 - 127]

Adjusts the Quality of the bandpass filters for the analysis filter bank. Note: The closer the individual AnalyzeQ and SynthQ bandpass filter frequencies lie to each other, the narrower they should be adjusted.

LinkFltQ (Link Filter Q) [Off / On]

Couples the two controls (AnalyseQ and SynthQ) so they operate together.

SynthQ (Synth Filter Q) [0 - 127]

Adjusts the Quality of the bandpass filters for the synthesis filter bank. Note: The closer the individual AnalyzeQ and SynthQ bandpass filter frequencies lie to each other, the narrower they should be adjusted.

Freq sub-menu (Frequencies)

Here you can adjust the frequencies of each filter individually. To adjust a filter's frequency, click on its frequency value and enter the new value with your keyboard. The new value applies to each respective analysis and synthesis filter.

After modulation by the Envelope Follower, the 22 synthesis filters in the synthesis bank are mixed down to stereo through an internal mixer. You can set each filter output's stereo pan position individually to create some very pleasing and spacious stereo effects.

If you also modulate the frequency or pitch of the synthesized sound, you can create some very striking stereo effects.

Because you can control the volume levels of each individual filter output, extremely detailed control of the vocoder effect is possible.

Frequencies			
LPF Frq	BPF01Frq	BPF02Frq	BPF03Frq>
BPF04Frq	BPF05Frq	BPF06Frq	BPF07Frq>
BPF08Frq	BPF09Frq	BPF10Frq	BPF11Frq>
BPF12Frq	BPF13Frq	BPF14Frq	BPF15Frq>
BPF16Frq	BPF17Frq	BPF18Frq	BPF19Frq>
BPF20Frq	HPF Frq		

LPF Frq (LPF Frequency) [0 - 24.000 Hz]

Lowpass filter frequency.

BPF01Frq-BPF20Frq (BPF 01-20 Frequency) [0 - 24.000Hz]

Frequencies of bandpass filters 1 - 20.

HPF Frq (HPF Frequency) [0 - 24.000 Hz]

Highpass filter frequency.

Matrix Menu

With the Vocodizer's Matrix you can route the control signals of the Envelope Follower to any arbitrary synthesis filter. You can also route the control signal of a particular analysis filter to multiple synthesis filters. The Matrix lets you accomplish a variety of effects, from "simple" formant displacement to a complete inversion of the filter assignments.

You can also change the frequencies of individual filters in the Matrix. However, note that each analysis and synthesis filter pair is always adjusted to the same frequencie.

Matrix	
Preset:	>Edit

Preset

Here you can select a Matrix preset.

Edit sub-menu

Matrix			
LPF	BPF01	BPF02	BPF03 >
BPF04	BPF05	BPF06	BPF07 >
BPF08	BPF09	BPF10	BPF11 >
BPF12	BPF13	BPF14	BPF15 >
BPF16	BPF17	BPF18	BPF19 >
BPF20	HPF		<

LPF (LPF ctrl by) [LPF, BPF01, ..., BPF20, HPF]

Select the matrix assignment for the lowpass filter.

BPF01 (BPF 1 ctrl by) [LPF, BPF01, ..., BPF20, HPF]

...

BPF20 (BPF 20 ctrl by) [LPF, BPF01, ..., BPF20, HPF]

Select the matrix assignment for the bandpass filters1-20.

HPF (HPF ctrl by) [LPF, BPF01, ..., BPF20, HPF]

Select the matrix assignment for the highpass filter.

Synthesis Menu

Synthesis			
>V/U Detect	>UnvSource	>Level	>Pan

V/U Det sub-menu (V/U Detection)

The Voiced/Unvoiced (V/U) section feature examines the analysis input signal for its tonal and noise content to determine which is dominant.

V/U Detection, as we have said, determines whether the input signal contains noise (like a sibilant "s" sound) or tone (like a spoken "a", or other vowel). Depending on the content, one of two possible signals is sent to the synthesis filter bank: Unvoiced source (usually noise) or Voiced source (the filtered internal synth or external audio source).

V/U Detection			
Type	Thresh	Hysth	Time

Type [Default / Easy]

The Detection algorithm can operate in one of two modes depending on some specific signal criteria.

Default: In standard mode the Vocodizer splits the signal's energy content into high and low frequency components by means of two filters, and then analyzes the two components separately. By adjusting the Threshold setting you can determine how much high frequency content triggers the detection circuit to identify the signal as unvoiced. However, at the same time, the Vocodizer also examines the spectrum of the low frequency component. The Vocodizer therefore considers two conditions: Only if the high frequencies contain sufficient energy AND the low frequencies do not, will the Vocodizer interpret the signal as being unvoiced.

> Example: A pure "s" sound contains sufficient high frequencies to satisfy the requirements of the first criterion, but the lower frequency energy is not sufficient to qualify it as a tonal sound (second criterion). The Vocodizer interprets this signal to be unvoiced.
>
> With the same settings, the detection circuit now encounters a spoken "k" sound. As before, the high frequency component is determined to be sufficiently high in energy to qualify it as an unvoiced signal. But this time there is also enough tonal energy in the low frequencies to override the initial determination, and the Vocodizer interprets the signal to be voiced. You can adjust the relationship of the weighting of the two criteria with the Hyst (Hysteresis) setting.

Easy: As an alternative to the standard mode in which the Vocodizer analyses both upper and lower frequency ranges, you can use a simplified detection mode in which only the high frequencies are examined to determine the signal characteristic (voiced or unvoiced).

In this mode it is sufficient only for the high frequency content to exceed the threshold for the signal to be identified as unvoiced. In Alternate mode the signal is determined to be unvoiced more frequently than in Standard mode, and the intelligibility in some cases improves because, along with the consonants like "k" and "s", some of the other overtone-rich parts of the signal are also identified as unvoiced.

Tresh (Threshold) [-96dB – 0dB]

Sets the level at which the high frequency energy triggers the detection circuit to identify the signal as unvoiced.

> **When set to minimum, none of the signal will be classed as unvoiced, and the noise source will never replace the original signal in the output. And, as you would expect, setting this parameter to maximum results in the entire signal being identified as unvoiced. In this case, the noise source replaces the entire signal.**

Hysth (Hystheresis) [0 – 127]

Hysteresis describes the difference in threshold values for the upper and lower frequency ranges. At 0, the same level criterion applies to both ranges. With increasing hysteresis values, less energy is required in the lower frequency range to defeat the unvoiced detection in the upper range. Therefore, as you increase the hysteresis value, signals with sibilance are less likely to be identified as unvoiced leading to fewer replacements of the synthesized signal with noise.

> **The Hyst setting applies only in Standard mode as Alternate mode employs only a single threshold value.**

Time [0.1 – 20 ms]

Depending on the result of the V/U detection, the synthesis filter is fed either with noise (unvoiced) or the synthesized signal (voiced). However, the change from one to the other does not take place immediately; rather, the signals are cross-faded. The Time setting adjusts the duration of the crossfade in milliseconds. With higher values, noise is fed into the synthesis section more slowly after unvoiced detection.

UnvSource sub-menu

Here you select the source signal to use for passages determined by the detection circuit to be unvoiced.

UnvSource		
Type	Level	LoCutFrq

Type (Source Type) [FiltOrig/PinkNois/WhitNois]

Selects one of the following possible sources: WhitNois (White Noise), PinkNois (Pink Noise), and FiltOrig (Filt. Original).

Level (UnvSource Level) [- 0 dB]

Adjusts the level of the unvoiced source signal.

LoCutFrq (LowCut Frequency) [20 - 20.000 Hz]

If you have chosen FiltOrig under Type, this adjusts the frequency below which the signal is attenuated.

Level sub-menu

Level			
LPF Lev	BPF01Lev	BPF02Lev	BPF03Lev>
BPF04Lev	BPF05Lev	BPF06Lev	BPF07Lev>
BPF08Lev	BPF09Lev	BPF10Lev	BPF11Lev>
BPF12Lev	BPF13Lev	BPF14Lev	BPF15Lev>
BPF16Lev	BPF17Lev	BPF18Lev	BPF19Lev>
BPF20Lev	HPF Lev		<

LPF Lev (LPF Level) [0 - 127]

Lowpass filter level.

BPF01Lev - BPF20Lev (BPF 01 Level - BPF 20 Level) [0 - 127]

Bandpass filter level.

HPF Lev (HPF Level) [0 - 127]

Highpass filter level.

Pan sub-menu

Pan			
LPF Pan	BPF01Pan	BPF02Pan	BPF03Pan>
BPF04Pan	BPF05Pan	BPF06Pan	BPF07Pan>
BPF08Pan	BPF09Pan	BPF10Pan	BPF11Pan>
BPF12Pan	BPF13Pan	BPF14Pan	BPF15Pan>
BPF16Pan	BPF17Pan	BPF18Pan	BPF19Pan>
BPF20Pan	HPF Pan		<

LPF Pan [-64, ..., 63]

Lowpass filter pan position.

BPF01Pan - BPF20Pan (BPF 01 Pan - BPF 20 Pan) [-64, ..., 63]

Pan positions of the individual bandpass filters.

HPF Pan [-64, ..., 63]

Highpass filter pan position.

Sources Menu (Input Sources)

Here you can select signal sources for the analysis and synthesis inputs.

Input Sources
AnalysIn: (Input) SynthIn: (Input)

AnalysIn: [Slot1 / Slot2 / Slot3 / Slot4 / Analog /USB]

Selects the source from which the analysis input obtains its signal.

SynthIn: [Slot1 / Slot2 / Slot3 / Slot4 / Analog /USB]

Selects the source from which the synthesis input obtains its signal.

InGainAn Menu (Analysis)

In this section you can control the input levels of the Analysis inputs.

InGainAn
InGainAn (Meter) Margin Reset

InGainAn (Analyse In Gain) [-24dB]

Adjusts the level of the analysis input signal with up to 24dB of amplification. Use the level meters to help set a satisfactory level.

Pay close attention to your levels. In particular, avoid letting signals overload the input. It's not so much that strong signals create audible problems with distortion; the bigger problem is that they influence the controls to produce unexpected results.

Meter (Display)

Displays the current level of the signal at the Analysis inputs.

Margin (AnalyseIn Margin) [dB]

Displays the highest signal level reached so far. Use the Margin Reset button to set the margin display back.

Reset (Reset Margin AnIn)[Set]

Resets the Margin display.

InGainSy Menu (Synthesis)

In this section you can control the input levels of the Synthesis inputs.

Synthesis			
InGainSy	(Meter)	Margin	Reset

InGainSy (Synthese In Gain) [-24 dB]

Here you ajust the level of the synthesis input signal with amplification of up to 24dB. Use the level meters to help set the correct level.

> **Again, pay close attention to the level, especially avoiding signals that are too strong. Such signals will produce unexpected high frequency components in the synthesis signal.**

Meter (display)

Displays the current level of the Synthesis inputs.

Margin (SynthesisInMargin) [dB]

Displays the highest signal level reached so far. Use the Margin Reset button to set the margin display back.

Reset (Reset Margin SyIn) [Set]

Resets the Margin display.

Output Menu

In diesem Menü regeln Sie die Ausgangslautstärke des Vocodizers.

Output			
Main Vol	(Meter)	Margin	Reset

Main Vol (MasterVolume) [0 - 127]

Regeln Sie hier die Grundlautstärke des Vocodizers.

Meter (display)

Displays the current level of the Vocodizer's output signal.

Margin (Margin Out) [dB]

Displays the highest signal level reached so far. Use the Margin Reset button to set the margin display back.

Reset (Reset Margin Out) [Set]

Resets the Margin display.

Interpole

Introduction

Interpole is an exciting Stereo Filter offering a wealth of possibilities for the treatment and processing of audio signals. An integrated ***envelope follower*** and ***low frequency oscillator (LFO)*** modulate the filter in ways guaranteed to give your sounds new life. With its two-channel implementation of all sections and a special LFO link mode, the filter can produce compelling frequency- and level-dependent stereo effects. Interpole is a great tool for creating exciting soundscapes from ordinary digital sounds, simple mono recordings, or dull sample loops.

The key element is the filter itself, which is nothing less than the ***24 dB Lowpass cascading filter*** of the probably best-known of all vintage synths -- still considered to be one of the best sounding filters ever. It's to the special characteristics of this filter that Interpole owes its unique liveliness and warmth.

To retain the analog character of the filter while providing a variety of sonic possibilities CreamWare's ***Circuit Modeling process*** has been employed. This has resulted in the highest-fidelity, aliasing-free algorithm by which the analog character of the filter is fully maintained.

So what are you waiting for? Let Interpole add analog warmth to your recordings, wake-up your sample loop with wild filter modulations, or create a stereo field for a synthesizer pad. It's all possible!

Guitarists can also use Interpole as a real-time plugin. Process your funk guitar with AutoWah or make your electric bass sound like a synthesizer. It's all easy for Interpole.

Your CreamWare Team hopes your work with Interpole will be both creative and enjoyable!

To better understand the various parameters we recommend you also read the Interpole chapter in the online manual as the structure is more easily grasped through the graphics and illustrations.

Structure and Overview

Interpole is a stereo effect consisting of two identical sections, left and right, for the processing of any audio signal. Each section consists of an envelope follower *or* ADSR envelope generator, an LFO, a lowpass filter, and a VCA. The two sections can operate independently or in tandem in link mode so that the filter can be used to process either two mono signals or one stereo signal.

Interpole is more than a simple effect: think of it as a synthesizer in which the oscillators are replaced by external signals. The envelopes and LFOs control the filter and the VCA just as in a synthesizer.

One of Interpole's most important features is the envelope section. The envelope section can operate in one of two modes: ”Env” or ”Gate”. In Env mode the section implements an envelope follower. Gate mode implements an attack-decay-sustain-release (ADSR) envelope generator which can be triggered by a Threshold or via MIDI. The envelope can modulate filter frequency, amplitude, and LFO rate.

No less important than the envelope stage is the LFO. It features six modulation waveforms, and synchronization to MIDI clock if desired. Individually adjustable intensities are available for filter frequency and amplitude modulation. In *link* mode, the LFO switches the signal from the first channel to the second, with optional modulation inversion on the second channel, to produce stereo-filter autopan effects. By restarting the LFO with MIDI triggers or by modulating the LFO rate with an envelope, the LFO modulation becomes even more complex and lively.

But it is the filter that performs the most important task: the actual signal processing. In addition to the standard cutoff and resonance parameters, the filter also provides a *Drive* parameter to introduce distortion that produces effects from gentle saturation to heavy distortion with strong resonance—everything you could ask for in a filter. The filter is controlled by the envelope and/or the LFO. With appropriate configuration all kinds of exciting stereo filter effects can be produced.

A ”voltage controlled” amplifier module (VCA) follows the filter. The amplifier is also controlled by the envelope and LFO. Envelope control can be switched off so that the LFO/VCA can serve as a tremolo or autopanner.

Thanks to the Circuit Modeling process, the audio is faithfully rendered and free of aliasing. Those familiar with aliasing know that it produces unpleasant and unnatural sounding analog signals. Interpole's various modulation effects and the resulting audio quality profit from this unique process. Often various distortions or sidebands fill out the full range of the spectrum as a result of heavy or wild modulations (such as frequency modulation). Alias-free rendition is absolutely necessary for producing such spectra. After working for a while with new sounds you will come to appreciate this characteristic.

Parameter

To access the Interpoleparameters open the Interpole menu (single mode) or the Slots/Interpole menu (multi mode).

The Interpole parameters are available in 4 menus (**Channel 1, Channel 2, Link, Sources**) and their sub-menus.

Interpole			
>Channel 1	>Channel 2	>Link	>Sources

Channel 1 / Channel 2 Menu

Because the left and right channels are almost identical, the following function descriptions apply to both channels unless otherwise indicated.

For each of the left and right channels there is an envelope follower or an ADSR envelope, an LFO, the filter, and the VCA. With the parameters in the Link menu the left-channel signals and control positions can be partially transferred to the right channel to simplify the handling of stereo signals.

Channel 1		
>Env	>Lfo	>Filter

Env sub-menu (Envelope)

As mentioned earlier, the envelope section operates in two modes: Envelope Follower or ADSR Envelope. The controls are arranged such that, so far as is possible, they apply to both modes of operation. However, depending on the mode, other functions—or perhaps no function—are assigned to the controls.

The Envelope Follower is enabled when the **Gate Mode** switch is set to the *Envelope* position. The **Input Ext/Int** switch then selects the input signal for the envelope follower (see the *Mode* chapter). The envelope follower examines the amplitude of the input signal and converts it to a signal for use elsewhere as a modulation signal. The *Sensitivity* control adjusts an optimal level for the modulation. The *Attack* and *Decay* settings adjust the accuracy with which the envelope follower tracks the input signal. In this operating mode, the *Sustain* and *Release* settings are not used.

The ADSR envelope is enabled when the **Gate Mode** switch is set to *Gate.* The gate, which controls the ADSR envelope, is triggered either by a MIDI note-on event or by the Threshold. Select the respective mode with the **Trigger Mode** switch.

In MIDI mode, the gate is open as long as a MIDI key is held. In Threshold mode the gate is open while the control signal remains above the adjusted threshold level, and closes when the signal falls below it. Select the input signal for the threshold to use with the **Input Ext/Int** switch. *Sensitivity* controls the threshold, *Attack* and *Decay* adjust the response times, and *Sustain* sets the level at which the gate remains open. In Threshold mode all controls are used. In MIDI mode the sensitivity control is disabled.

Envelope			
Bypass	Input	Gate	Trigger >
Sens	*VU Meter*		>
Attack	Decay	Sustain	Release <

Bypass (Bypass/Effect) [Effect, Bypass]

This switch enables or disables the effect for the respective channel. With Bypass enabled, the signal is routed directly from the input to the output, bypassing the effect.

Input (Input Ext/Int) [Internal, External]

Interpole has both *internal* and *external* inputs. The signal from the internal inputs passes through the entire signal chain and arrives at the filter to be processed. The signal at the *external* inputs serves a different purpose as a signal only to be analyzed by the envelope follower section. This switch selects which signal to send to the envelope follower. The envelope follower converts the amplitude changes to control signals used for changing the cutoff frequency or amplification. When set to *Internal*, the signal that is processed by the filter is the same signal analyzed by the envelope follower. When set to *External*, the signal used for control functions is different than the one processed by the filter. You can therefore use an external signal to process the frequency spectrum of another signal through filter modulation.

Gate (Gate Mode) [Envelope, Gate]

Selects the basic operating mode of the envelope section. When *Env* (Envelope Follower) mode is enabled, the envelope section supplies a continuous modulation signal derived from an analysis of the input signal. When *Gate* is selected, an ADSR envelope triggered by either an adjustable threshold or a MIDI note-on instruction supplies the modulation signal.

Trigger (Trigger Mode) [Sens, MIDI]

This switch selects the Gate's operating mode, that is, whether it is to be triggered by a MIDI note-on message or by an adjustable threshold level. *MIDI* selects MIDI note-on as the trigger, and *Sens* selects threshold. With MIDI selected, any incoming MIDI note will trigger the envelope. Just make sure the correct MIDI channel is configured. In Threshold mode one of two signals, Internal or External, triggers the envelope.

Sens (Sensitivity) [0, ... , 127]

Envelope Mode:

Controls the intensity of the of the envelope follower's modulation signal. Using the VU Meter beside it you can ascertain the level supplied by the envelope follower. If the envelope follower is overridden, reduce the sensitivity somewhat.

Gate Mode:

The Threshold value above which the gate opens and below which it closes.

Attack [0, ... , 127]

Envelope Mode:

Controls the time the envelope follower takes to respond to rising levels of the input signal.

Gate Mode:

The attack time. When the envelope receives a gate signal the modulation signal increases to maximum during the time configured here.

Decay[0, ... , 127]

Envelope Mod:

Controls the time the envelope follower takes to respond to falling levels of the input signal.

Gate Mod:

The Decay time. Once the attack phase has completed, the modulation signal falls to the sustain level for the duration configured here. The Decay will have an audible effect only if the sustain level is not the same as the maximum level.

Sustain [0, ... , 127]

Gate Mode only**:**

The Sustain level—the level at which the signal remains after the attack phase and while the gate is open. When the gate closes, the envelope directly enters the release phase.

Release [Off, On]

Gate Mode only**:**

When set to *On*, the release phase assumes the *Decay* time as the release time. When the envelope generator receives a Gate Off signal, it immediately enters the release phase, and the envelope completes with the adjusted release time from the previous level. In the *Off* position, the release is set to minimum, at which release is disabled.

LFO sub-menu

The LFO parameters provide another playground for sound manipulation. Six different modulating waveforms are available. The rate can be set manually or controlled by the MIDI clock, and further modulated in principal by the envelope. In addition, the LFO can be restarted via MIDI to retain suitable sync with a song. Filter frequency and amplitude are independently adjustable. In Link mode the LFO switches from the first channel to the second (see **Link Menu**). By inverting the modulation on the second channel, interesting stereo filter and autopan effects develop.

LFO				
Sync	Note Len	Retrig	Phase	>
EnvSweep	Rate	Waveform		>
VCF	VCA			<

Sync (Sync Mode) [Off, MIDI]

Synchronizes the LFO oscillation to MIDI clock. To adjust the speed, select a tempo and a note value. When switched to *MIDI*, synchronization is enabled. In the *Off* position the speed (rate) is set manually.

Note Len (Note Length) [1/1, 1/2dot, 1/2, 1/2trpl, 1/4dot, 1/4, 1/4trpl, 1/8dot, 1/8, 1/8trpl, 1/16dot, 1/16, 1/16trpl, 1/32, 1/32trpl]

When MIDI synchronization is enabled the speed of the LFO is controlled by Note values. A complete oscillation corresponds to the note duration.

Retrig (Retrigger Mode) [Off, MIDI]

Restarts the LFO with each MIDI note-on message. The starting phase position is determined by the *Initial Phase* setting. If MIDI is enabled, any MIDI note will restart the LFO waveform. Make sure the correct MIDI channel is selected.

Phase (Initital Phase) [-180, ... , 180

Determines the position (phase) at which the waveform starts when a MIDI note-on message is received and Retrig is set to MIDI.

EnvSweep (Envelope Sweep) [-64, ..., 63]

Adjusts the intensity of the envelope that modulates the speed of the LFO. The rate of the LFO increases or decreases following the envelope within a range governed by the intensity setting. The process starts and ends at the adjusted rate. Both positive and negative modulation is possible.

Rate [0.01, ... , 100.00 Hz]

Sets the basic (unmodulated) LFO oscillation rate.

Waveform [Sine, Square, Saw Up, Saw Down,Triangle, Random]

Select here one of six waveforms.

VCF (VCF Modulation) [0, ... , 127]

Controls the intensity of the frequency modulation.

VCA (VCA Modulation) [0, ... , 127]

Controls the intensity of the amplitude modulation.

Filter sub-menu

The filter is included in the signal processing chain along with the envelope generator and the LFO. The filter is a 24dB/octave lowpass type also known from the cascading filter of thr probably best-known of all vintage synths. Frequencies below the cutoff remain unprocessed (hence the designation ”lowpass” filter). Frequencies above the cutoff frequency are attenuated at a rate of 24dB/octave. Not much needs to be said about the renowned Moog filter, other than it is generally considered to be one of the best sounding filters ever to be implemented in a synthesizer. And now, in Interpole. The filter also provides the characteristic *drive* parameter whereby it can be intentionally be coaxed into over-modulation.

Filter				
Drive	Env Sweep	Cutoff	Resonance	>
VCA				<

Drive [-64, ..., 63]

Controls the filter’s input level. Higher values produce more distortion.

EnvSweep (Envelope Sweep) [-64, ..., 63]

Controls the influence of the envelope over the cutoff frequency. The cutoff follows the envelope curve to produce dynamic filter effects. The beginning and end points of the curve produce the cutoff frequency as set in the *Cutoff Frequency* field. Both positive and negative modulation is possible.

Cutoff [0, ... , 127]

The cutoff frequency is the frequency above which the spectrum is attenuated (overtones are cut). Set the frequency manually here.

Res (Resonance) [0, ... , 127]

Resonance results from coupling the filter output with the input such that frequencies surrounding the cutoff frequency are reinforced. At full resonance the filter oscillates producing a sine wave at the adjusted cutoff frequency. This oscillation occurs even if there is no input signal, so the filter can also be used as a signal source.

VCA (VCA Mode) [Out, In]

The "voltage-controlled" amplifier (VCA) is the last unit in Interpole's processing chain. The VCA follows immediately after the filter, and is controlled by the envelope and the LFO. Envelope control is optional to allow the LFO to perform alone as a tremolo or autopan.

Switches modulation of the amplifier by the envelope section on or off. In the *In* position, the envelope is active. If the envelope is in Envelope Follower mode, the intensity of the modulation is controlled by Sensitivity. With the envelope in ADSR mode the modulation is always at maximum. In the *Out* position envelope control is disabled, although the amplifier can still be modulated by the LFO.

Link Menu

The left-channel signals and control positions can be partially transferred to the right channel to simplify the handling of stereo signals.

```
Link
 Link LFO LinkMode  Link Env  Link Fil
```

Link LFO [Off, On]

When Link is enabled the left channel is switched to the right, making the handling of stereo signals easy and the production of interesting stereo effects possible.

LinkMode (LFO Link Mode) [Normal, Inverse]

When Link is enabled, the right channel's LFO signal is derived from the left channel's signal. *Invert* produces interesting stereo effects by inverting the phase of the right channel. The effect ranges from simple Autopan to more complex stereo effects through filter modulation. Left and right channel link must be enabled (see previous section).

Link Env (Link Envelope) [Off, On]

With Link, the left channel modulation signal is also used by the right channel to facilitate the handling of stereo audio signals.

Link Fil (Link Filter) [Off, On]

With Link, the left channel filter setting is also used by the right channel to facilitate the handling of stereo audio signals.

Sources Menu (Source Select)

Interpole has both *internal* and *external* inputs. The signal from the internal inputs passes through the entire signal chain and arrives at the filter to be processed. The signal at the *external* inputs serves a different purpose as a signal only to be analyzed by the envelope follower section. This switch selects which signal to send to the envelope follower.

```
Sources
 Input     External     Bypass
```

Input [Slot1, Slot2, Slot3, Slot4, Analog, USB]

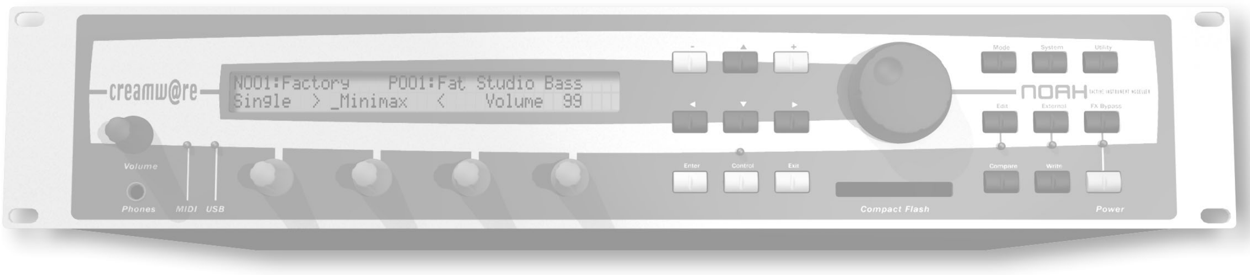
Selects the mixer channel output to connect to the internal Interpole input.

External [Slot1, Slot2, Slot3, Slot4, Analog, USB]

Selects the mixer channel output to connect to the external Interpole input.

Bypass [Off, On] Bypass All

Bypass for the complete effect.



USER'S MANUAL

APPENDIX

Appendix

Noah - Technical Specifications

Noah	2 Instruments simultaneous
Noah EX	Up to 4 instruments simultaneous
Sample rate	44.1 kHz (Master & WordClock Slave)
Number of channels	2 Inputs und 10 Outputs

Analog Stereo Input

Unbalanced	2 x Phone Jack 6,3mm
Input impedance	> 100kOhm
Gain switch = Low	
Input sensitivity	-10 dBV (nominal)
Maximum input level	0 dBV (0dBFs)
Gain switch = High	
Input sensitivity	-24 dBV (nominal)
Maximum input level	-11 dBV (0dBFs)

Analog Stereo Output

Unbalanced	2 x Phone Jack 6,3mm
Output impedance	300 Ohm
Output level	-10 dBV (nominal)
Maximum output level	1,6 dBV (0dBFs)
Headphone	Stereo Phone Jack 6,3mm, > 1 W @ 4 Ohm

Digital Inputs and Outputs

TOS-Link Out	ADAT Out switchable S/P-Dif Out
MIDI	DIN-5-Pol, In, Out, Through
Foot switch	1 Phone Jack 6,3mm, with 2 Switches
USB-Port	Full Speed Rev. 1.1
WordClock In	BNC, 75 Ohm @ 44,1 kHz
Compact Flash Adapter	

General

Power consumption	50 W
Case	19'', 2 HU, 48,1 x 30,0 x 8,9 cm
Weight	3,8 kg

The Hotline

Software Problems

As mentioned above, our products work flawlessly with most computers if the issues in this document are properly attended to. In addition, via the support area of our website:

http://www.creamware.com

we will constantly publish solutions for new problems as they are discovered. If you have Internet access, please check the information which is posted there before contacting us directly. The latest information and trouble-shooting assistance will always appear there first.

If you have checked all of the information presented here and on the Web site and you are still unable to solve your problem, there are various ways to contact us directly for advice.

> **Once again, however, we request that you recheck all of the information presented above *before* doing so!**

In any event, only *registered users* are entitled to direct technical support, so please register now!

If you write or email us, be sure to provide us with all required information about your system. You will find a form below to assist you with this. If you call us, please have this information ready to give to the technical support representative. It's best to first fill out the form completely and then either send it to us or have it ready at hand when you call us.

Be sure to provide us with *all* required information about your system ...

1. I have tried all suggestions given in this document: Yes

2. CONFIGURATION

Processor:

Type:

Hard drive(s):

Graphics card (which slot / IRQ / driver version?):

RAM:

CD writer:

SCSI controller (which slot / IRQ?):

CD-ROM:

other internal devices/cards (which slot / IRQ?):

3. CreamWare products:

Noah-Version:

OS-Version (see System / Device menu):

Version Noah Remote Software:

4. Connected Devices:

Mixers:

Synthesizers:

Synchronizers:

Samplers:

Recorders (DAT / ADAT etc..):

Other (Atari, MAC, MIDI patchbays etc..):

5. Installed Software

Operating system:

Sequencer software:

Audio applications:

Other:

6. Description of the problem

When and where does it appear?

Can it be made to recur via a specific series of actions? How?

Which parts of the program are involved (modules / devices)? *Which* devices? *How* are they connected?

There are four ways to reach our support department:

In the USA and Canada:

email: support@creamware.com

Fax: (604) 435-9937

Phone: (604) 435-5158

Mail: CreamWare US Inc.
6879 Russel Avenue
Burnaby, B.C.
V5J 4R8
Canada

All other countries:

email: support@creamware.de

Fax: (++49) 22 41 - 59 58 57

Phone: (++49) 2241 - 59 58-12

Mail: CreamWare Datentechnik GmbH
Support
Wilhelm-Ostwald-Strasse 0/K1
53721 Siegburg, Germany

But for now – enough hints about possible problems. As an experienced computer user, you are no doubt well aware that neither software nor hardware which is one-hundred percent perfect exists. We at CreamWare strive continually to improve our products, and we welcome your criticism and suggestions.

Having said that – we hope (and expect) that you won't encounter problems with our products, and we wish you all the best in working creatively with our products!!!

Sincerely,

Your CreamWare team!

Garantie

Before sending in your Creamware product for warranty support, please call the Support office to obtain an RMA number for your card. Testing and repair of hardware which is sent to us without making prior arrangements is given a lower priority and can take correspondingly longer.

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Index

A

A/D curve 111
Acoustic model 107
ADAT 154
ADAT Output 19
ALT fader 92
Amount 77
Amount of Contour 127
Analog In Left / Right 19
Analog Out Left / Right 19
Analog Stereo Input 154
Analog Stereo Output 154
analysis phase 140
articulation 140
Attack 72, 78, 79, 127, 149, 150
attack 104
Aufbau 124
AUTO HOLD 90
AUTO RESCAN 89, 96
Auto ReScan 89
Auto Resync 94
Autopan 71, 152
AutoRestart 98
AutoRScn 89
AutoRSync 94
AutoStop 98

B

basic method 88
bass 121, 122
Bass Double 109
Bass Nylon 109
Bass Steel 109
Beats 93
Belegungen der Slots 16
Bend Range 125
Body 108
Boost 73
Bypass 149
Bypass/Effect 149

C

Capture 90
CAPTURE MODE 90
Capture Mode 90
captured notes 90
Carrier Frequency 76
Case 154
Center Key 98
Channel 2 149
channel messages 86
chord buffer 86, 90
chord value 92
Circuit Modeling process 148
Classical guitars 113
CLEAR 87, 90, 95
Clock 93
Coarse L/R 70, 71
Color 75, 78
Compact Flash Adapter 154
Compact Flash Slot 18
Compare Button 18
Compressor M/S 78
Continuous Controllers 17
continuous loop 98
Contour 127
Control Base Note 95
Control Button 17
Control Wheel 18
Cross Feedback 56
CTRL ZONE BASE 95
CtrlBase 95
Cutoff 72, 130, 151
cutoff frequency 152

D

Damp 76
Decay 72, 128, 149, 150
decay 104
Depth
53, 60, 61, 62, 63, 64, 65, 66, 67, 68, 71, 72, 76
Disclaimer 157
Display 17
Distortion 74, 75
dotted note value 97
dotted quarter note 97
Down 17
Drive 73, 75, 78, 148, 151
Drop Alt 92
DropNorm 92
Dropout Alt 92
Dropout Norm 92

E

Edit Button 18
Einleitung 131, 148
Electric model 108
Emphasis 127
Ensemble 60
Enter Button 17
Env 148, 150
Env Attack 76
Env Decay 72, 76
Env Depth 72, 76
Env Sweep 151
envelope follower 148
Envelope Sweep 151
Exit Button 18
Ext2Len 90
EXTEND 92
EXTEND 1 90
Extend 2 90
Extend2 Length 90
External Button 18
External Input Volume 126

F

FB 70
Feedback 53, 56, 60, 61, 64, 65, 71
Feedback L/R 54
Feedback On 126
Filter 128, 151
filter banks 140
Filter Type 72
FilterMix 141
Fine L/R 70, 71
Fing. Glissando 116
Fing. Portameto 116
fingerboard 114
Fingered Glissando (fG) 100
Fingered Portamento (fP) 100
first note 88
flamenco guitars 113
Floor 80
Foot switch 154
Footswitch 19
FORWARD 91
Forward 88
Free Run 135
free-running tempo clock 96
freezing the chord buffer 88
Freq 59, 93
Freq Hz 93
Frequency 93, 125, 132, 133
Frequency Shift L/R 69
FWD-REV 88, 91
Fwd-Rev 88
FX Bypass Button 18

G

Gain 59, 75, 78, 79, 80
Gain Switch 19
Gain switch 154
Gate 148, 150
Gate Duration 90
Gate Mode 150
gate-off phases 90
gate-on phases 90
GateDur 90
General 154
Glide 129
Glide On 128
Glissando 116
Glissando (G) 100
Global 98, 110
Graphic EQ 59
guitar body 107
Guitar Electric 109
Guitar Jazz 109
Guitar Nylon 109
Guitar Western 109

H

Hammond 117
Headphone 154
Helmholtz frequency 113
Hexa Chorus 62
Hi Damp 54, 55, 56, 74
Hi Level 61, 64
HiDamp 62
HiDamp Filter 57
highest note 88
Highpass 75, 78
Highpass Filter 57
HOLD 87, 88, 94, 95
Hold 80
HOLD/TRANS 88, 94, 95
Hotline 155
HTrnsp 87
Htrnsp 88
Hz 93

I

Initital Phase 151
Input Ext/Int 149
Input sensitivity 154
Inputs 154
Insert-Effekte 58
INTERN RESCAN 94
internal chord buffer 86
Introduction 107, 124, 140

J

jazz 120
JoySRet 130
Joystick 130
Joystick Return 130
JoyStickY 130

K

Karplus-Strong synthesis 107
KeybCtrl 95
Keyboard Control 128

L

last note 88
Left 17
legato 90
Length 98
length 97
Leslie 117, 121
Level 55
Level L/C/R 55
LFO 93, 115, 148
LFO Depth 74
LFO frequency 93
LFO Link Mode 152
LFO Mod 91
LFO MODULATION 91
LFO modulation 94
Lfo Modulation 91
LFO Rate 74
Limiter 79
Link Env 152
Link Envelope 152
Link Fil 152
Link Filter 152
Link LFO 152
Link Mode 152
Link to Left Shift 69
Lo Damp 56
Lo Damp L/R 54
LoDamp
13, 61, 62, 63, 64, 65, 66, 67
Loop Active 98
loop length 89
low frequency oscillator 148
Low Level 61, 64
Low Note On 129
lowest note 88
Lowpass 75
lowpass 151
Lowpass Moog cascading filter 148

M

Manual 94
MANUAL CLK ONLY 96
MANUAL CLOCK 96
Manual Clock 96
Mass/Spring model 107
Master Chorus 60
Master Flanger 63
Matrix 141
MAX VELOCITY 91
MaxVelocity 91
MIDI 148, 149, 154
MIDI 60 88
MIDI Control Zone 95
MIDI controller 91
MIDI In / Out / Thru 19
MIDI input 86
MIDI note events 86
MIDI THRU OFF 95
MIDI-LED 17
Miniscope 124
Miniscope MkII 124
Mixer 126
Mod 130
Mode 88, 90
Mode Button 18
modulation matrix 130
Modulation Mix 129
Modulation Offset 129
ModW 130
ModWheel 130
modwheel 86
ModWheel Return 130
ModWRet 130
monophonic 86
MonoString 109
multi-segment vector envelope 130
MultiView 46

N

NEW CHORDS 89
NEW NOTES 89
NewChords 89
NewNotes 89
NNumber 88
NOrder 88
NORM fader 92
NORMAL 89, 90
Note 90
note capture limit 90
NOTE DROPOUT SHIFT 92, 95
Note Length 151
NOTE NUMBER 88, 90, 92
Note Number 88
Note Off Width 90
Note On Width 90
NOTE ORDER 88, 90, 92
Note Order 88
Note Repeat 92
Note Transpose 95
note-off events 87
note-off messages 86
note-on events 87
note-on messages 86
note-on velocity 91
Note/Chord 94
NoteLen 90
NoteTrans 95

O

Octave 132, 133
OCTAVE EXTEND 91
Octave Extend 91
OctvExt 91
Off 98, 150
OffBeats 96
OffClks 96
Offset Beats 96
Offset Clocks 96
On 150
On/Off button 87
Original 91
Oscillator Modulation 125
Output 75, 78, 91
Output impedance 154
Output level 154
output note 89, 90
output pattern 94
Outputs 154
OutTiming 96
Overdrive 75
overtones 119, 151
Overview 124, 148

P

Pan 55
Pattern 88
peaks 94
Phase
53, 60, 61, 63, 66, 67, 71, 72, 76, 94, 151
Phase Invert 65
Phones 17
pickup filter 108
Pickup Link 112
pipes 117
pitch bend 86
plectrum stroke 110
Pluck 108, 110
Plus / Minus Buttons 17
Portamento (P) 100
positions 108
Power Button 18
Power consumption 154
PPQN 87
Presets 108
pulses per quarter-note (PPQN) 87

Q

Q 59, 75

R

RANDOM 88, 91, 93
Random Depth 89
Random Flanger 65
Range 125
range 117
Rate
53, 60, 61, 62, 63, 64, 65, 66, 67, 68, 71, 72, 76, 151
Ratio 78, 79
Release 78, 79, 80, 149, 150
release 104
Repeat 92
REPEATS / NOTE 92
Replace 91
REPLACE VELOCITY 91, 93
Res 130
RESCAN 89, 95, 96
Rescan 88
ReScan Length 89
Resonance 72, 76, 152
Resonator 76
Rest Note 92
Restart 135
Resync 94
RESYNC AR UPON 94
Resync Options 94
resync sources 94
RESYNC UPON 89
resynchronization 89
Retrig On 129

Retrigger 151
Retrigger Mode 151
Reverb 57
REVERSE 91
Reverse 88
rhythm 90
Right 17
Ringmodulator 76
RndDepth 89
rock 120
Room Size 57
RscnLen 89
RUN 87
RUN/STOP 95

S

"S" sounds 140
Sample Rate 74
Saw Down waveform 94
Saw Up waveform 94
Sawtooth Down 93
Sawtooth Up 93
SCAN DIR 95
SCAN MODE 88
Scan Mode 88
SCAN PATTERN 91
Scan Pattern 88
ScanDir 88
semitone transpose 88
Sens 150
Sensitivity 150
SEQ 1 - 4 97
settings 87
Shape 60, 67, 76
signal LED 150
Signature 98
signature 140
Signature Mode 98
Sine 93
slope 130
Softclip 77
source 86
Space Flanger 66
speaker 121
Speed 70, 71
Speed Beats 93
Speed Clocks 93
Speed Type 93
Split Freq 60, 61, 63, 64
Spread 62
SSB Phaser 69
staccato 90
steel-string guitars 113
Step Lag 67
Step Length 98
Step Rate 67
stereo audio signals 152
Stereo Delay 55
Stereo EQ 59
Stereo Expander 77
Steuerung 87
strength 120
string sets 109
string vibrations 108
Structure 124
Sustain 127, 128, 149, 150
sustain 90, 104
Sw: 92
SWEEP TRANSPOSE 92
Sweep Transpose 92
Sync 151
Sync Mode 151
Synthesis 130
System Button 18

T

talking synthesizer 140
Tap 96
Technical Specifications 154
Technische Daten 154
Tempo 45
Threshold 78, 79, 80, 148, 150
Time Keyfollow 103
timing resolution 87
transpose 88
transposition 91
treble 121, 122
Tremolo 72
tremolo arm 115
Triangle 93
Trigger 150
Trigger Mode 150
TrigOut 44, 46
triplet 98
triplets 97
Tune 125
type 120

U

Übersicht 86
Up 17
USB Port 19
USB-LED 17
USB-Port 154
USE ORIG VELOCITY 91
Utility 18

V

Values 97
VCA 148, 151, 152
VCA Mode 152
VCA Modulation 151
VCF 151
VCF Modulation 151
vector control 130
vector envelope 130
vector field 130
vector position 130
Vector Synthesis 130
VelMode 90
Velocity 127, 128
velocity 110
Velocity: 90
Vocoder 141
Volume 17, 126, 128, 129, 130

W

Warranty 157
Waveform
60, 63, 67, 68, 71, 72, 76, 93, 125, 132, 133, 151
waveforms 93
Weight 154
whammy bar 115
Word Clock Input 19
WordClock 154
Write Button 18