Operation Manual
Attack Percussion Synthesizer

www.waldorfmusic.de

Programming:	Wolfram Franke, Giuliano Orsini, Stefan Stenzel
Graphic Design:	Design Box
Manual/ Diagrams:	Holger "Tsching" Steinbrink

Sound design and beta testing:

Holger Bahr, Raymund Beyer, Mert Ergün, Wolfram Franke, Christian Halten, Axel Hartmann, Jörg Hüttner, Arnd Kaiser, Dirk Krause, Till Kopper, Peter Kuhlmann, Sascha Kujawa, Dominik De Leon, Hubertus Maas, Dr. Georg Müller, Michael Johnson, Mark Pulver, Jörg Schaaf, Marcus-S. Schröder, Henry Stamerjohann, Sven Steglich, Holger Steinbrink, Stefan Stenzel, Wolfgang Thums

Many thanks to Wolfgang Düren, Achim Flor, Yvan Grabit, Achim Lenzgen, Amanda Pulver, Manfred Rürup, Stefan Scheffler, Karl Steinberg and Lu Wangard.

Content

Waldorf Attack Percussion Synthesizer

Introduction

Thank you for purchasing the Waldorf Attack software synthesizer. The Attack combines many exciting possibilities for analog drum and percussion production with the advantages of a software plug-in.

The Waldorf Attack offers a straightforward sound structure that faithfully emulates a wide range of percussion sounds, from those of well-known classic drum or rhythm generators to new, unique drum sounds such as bass drums and snare drums over shakers as well as heavily modulating synth effects. In addition, the Attack offers ways to create amazing bass and lead sounds.

To understand how to create these sounds, you should know a little about how the classic drum machines worked. Basically, those drum machines had special analog circuitry for each drum sound they could produce. These analog circuits were often built in a very strange manner, sometimes consisting only of a handful of filters without any kind of oscillator or envelope, or perhaps simply a noise generator with one envelope. Each was made to create just one particular drum sound in a more or less authentic way. But it was just this assertiveness and distinctiveness of character which are responsible for the fame and popularity of these sounds that endures today.

For this Millennium, the Attack will continue this story!

Besides the regular manual, the following sections offer some insights as to how particular sounds were built, and give information about how to achieve similar results with the Waldorf Attack.

System Requirements for Windows

In order to be able to use the Waldorf Attack, you will need at least:

- A PC with a Pentium III / AMD Athlon 600 processor or better
- 64 MB free RAM
- Windows 2000 / XP
- VST 2.0 compatible host application supporting virtual instruments, such as Cubase VST 3.7 or higher, Cubase SX/ SL/ SE or Nuendo 1.5 or higher. This must be correctly installed on your computer.

Please also observe the system requirements of your host application!

Installation under Windows

Proceed as follows to install the Attack:

1. Start your computer and launch your operating system.
2. Insert the Attack CD-ROM into your CD-ROM drive.

If you have enabled the Autostart function in Windows, the Installer will start automatically and you can proceed with Step 6 below. If not, please proceed as follows:

3. Launch the Explorer or open the “My Computer” window.
4. Double click on the icon for the CD-ROM drive that holds the Attack-CD-ROM.
5. Double click on the Attack Installer icon. This launches a special installation program.
6. Follow the on-screen instructions

The Attack CD-ROM is your verification that you have purchased the program. Please store it in a safe place. If you lose it, you have lost the Program.

Please note the “Read Me” and “Important Changes” text files on the Attack CD-ROM, which contain additional information and/or any changes.

System Requirements for Mac OS

In order to be able to use the Waldorf Attack, you will need at least:

- Macintosh with 400 MHz G4 PowerPC processor or better
- Mac OS X 10.3.9 or newer
- 128 MB free RAM

or

- Macintosh with 1.5 GHz Intel CoreSolo processor or better
- Mac OS X 10.4 or newer
- 128 MB of free RAM.
- A VST 2.0 compatible host application that supports virtual instruments, such as Cubase SX / SL / SE or Nuendo. This must be correctly installed on your computer.

or

- An AudioUnit 2.0 compatible host application that supports AudioUnit Instruments, such as Apple Logic or GarageBand. This must be correctly installed on your computer.

Please also observe the system requirements of your host application!

Installation under Mac OS

Proceed as follows to install the Attack:

1. Quit all other applications so that you return to the Finder. Disable any system activity monitoring software or extension, in particular anti-virus software. Then insert the Attack CD into your computer's CD-ROM drive.

2. If you need to, double click on the Attack icon to open the CD window.

3. Double click on the Attack Installer icon to load the installation software. Follow the instructions on the screen.

The Attack CD-ROM is your verification that you have purchased the program. Please store it in a safe place. If you lose it, you have lost the Program.

Please note the "Read Me" and "Important Changes" text files on the Attack CD-ROM, which contain additional information and/or any changes.

Using the Attack as VST or AudioUnit Instrument

For users of VST or AudioUnit (Macintosh only) compatible host applications, Waldorf offers the Attack as a completely plug-in-based synthesizer that fully utilizes Steinberg's VST and Apple's AudioUnit interface technology. If you wish to use it in a VST or AudioUnit compatible host application, please consult its documentation to find out what you have to do to set up the plug-in.

Using the Attack

You can play the Attack via MIDI like any other instrument, and record what you play on a MIDI track. To make sure, press a key on your MIDI keyboard. You should now hear a sound. If you don't hear anything, determine first whether your host application receives MIDI data at all. If you notice considerable latency times (delay) when you play the Attack via MIDI, please read the chapter "Playing the Attack Live". For more information about MIDI functionality, refer to page 7 of this manual.

- If you don't want to play the Attack exclusively from the integrated screen keyboard, make sure that your host application receives MIDI data that you generate from your external MIDI Master keyboard.
- To activate the Attack as a virtual instrument please consult the corresponding documentation of your host application.

Audio Channels of the Attack

The audio signals that are created by a virtual instrument are automatically routed to the Channel Mixer of your host application. Open this mixer. For each open Attack plug-in, you find six stereo channel strips in the Channel Mixer. These are named „Atk 1L", „Atk 1R" through „Atk 6L" and „Atk 6R" (note that some hosts only show the names of the left channels). By using the **Output** parameter in the Amplifier section, you can assign the audio signals to any of the output channels. Find more information on page 21 of this manual.

Using the Channel Mixer, you can comfortably mix the signals created with the Attack and treat them in the same manner as audio tracks. You can apply EQ, effects, or other external studio gear, and process the audio in diverse ways.

If needed, you can transform any or all Attack tracks into an audio file. To do this, simply use the "Export Audio" function of your host application. Please find more information in the corresponding documentation.

To get the most out of the Attack, please make sure that you use the latest software version of your host application.

Loading Banks and Kits

An Attack (drum-) Kit consists of 24 different sounds. The Attack comes with several preset kits, created by renowned sound designers.

A Bank consists of 16 complete kits, each with 24 sounds.

You can load complete banks or single kits. For information on loading, please read the corresponding manual section of your host application.

When you use the Attack as AudioUnit on the Macintosh, you can only play the first kit of a loaded bank. You can, however, copy individual sounds from one of the other 15 kits to the first kit through the Preset menu, if you have loaded a bank that contains 16 kits.

Storing Banks and Kits

In the Attack, you can save both single kits or a complete bank.

You can save complete banks or single kits. For information on saving, please read the corresponding manual section of your host application.

When you save your song or project file, the following information is saved with it:

- The number of Attack modules used in the song.
- Which banks and kits were used.
- The changed settings of edited sounds are also stored.

However, if you wish to use the edited version of a program in another song, then you must save that kit manually before.

Importing VST or AudioUnit Presets into the corresponding plug-in format (Macintosh only)

VST and AudioUnit use different formats to save the plug-in data. If you had created a kit in a VST host and want to use it in an AudioUnit host (or vice versa), please proceed as follows:

- Save the kit in the original host as a single Preset or Program. In VST hosts, those files typically have the suffix „fxp", in AudioUnit hosts they are called „aupreset".
- Quit that host and start the other host that supports the corresponding plug-in format. Open an instance of the plug-in.
- Select „Import File..." in the „Edit"-menu of the plug-in editor.
- Select the file to import from the Open dialog.
- Confirm the selection with OK.

The Preset or Program is loaded and can be played as usual.

Selection of a Kit

To select a kit from a bank, please read the corresponding chapter in the manual of your host software.

AudioUnits (Macintosh only) only support the first kit natively. The other kits of a bank are stored and loaded anyway but you should avoid to use them because AudioUnits don't support anything like „Program Changes".

Selection and Naming of Sounds

To select a sound from a kit, please proceed as follows:

- Click on the Sound selector button on the left hand side of the plug-in.
- You will hear the chosen sound. If you want to disable this, hold down the shift button on your computer keyboard while selecting the sound.

To (re-) name a sound, please proceed as follows:

- Hold down the [Alt] key on your computer keyboard while selecting a sound. Now you can enter a new name.

Playing the Attack Live

If you notice considerable latency times (delay) when you play the Attack via MIDI, then the sound/audio card that you use and/or its driver is the source of the problem. If the problem occurs within your system and recording by playing in real time is important to you, then it is recommended that you replace your sound card and its driver with a fast audio card and an ASIO driver (or CoreAudio driver for Mac OS X) that have been optimized for the shortest possible latency time. When playing back Attack data from a MIDI Track, latency does not occur.

The Solo Function of the Attack

With the Solo button in the upper left corner of the plug-in, you can listen to a sound by itself. This is helpful if you want to listen to one particular sound while triggering different sounds with the sequencer. All other sounds are muted when this function is active. The Solo button lights up in red when the function is active. Of course you can choose different sounds while soloing, and only the selected sound will be heard. To deactivate the Solo function, simply click the button again.

Functions of the Edit Menu

The Attack offers several different functions for convenient organization of kits and sounds. In the edit menu in the upper left corner of the plug-in you can choose the following functions:

- **Import File...** (Macintosh only) imports a VST or AudioUnit program.
- **Copy Sound** copies the sound into the buffer.
- **Paste Sound** puts the copied sound into any desired place. This is useful when creating variations of one particular sound.

Please take into consideration that the buffer can hold only one sound at a time. As soon as a second sound is copied with the **Copy Sound** function, the previously selected sound is erased from the buffer. After inserting a sound with **Paste Sound**, the sound is still in the buffer and can be inserted into yet another position.

- **Copy Kit** copies the kit into the buffer.
- **Paste Kit** inserts the copied kit at any position. This is useful when exchanging kits between different songs.
- With **Compare Sound** you can compare the edited sound with the original. As soon as the Compare function is activated, the sound changes back to its original settings and the Compare function in the edit menu gets a flag. Reselecting the function brings up the edited version again.
- With **Compare Kit** you can compare the edited drum kit with the original. As soon as the Compare function is activated, the kit changes to its original settings and the Compare function in the edit menu gets a flag. Reselecting the function brings up the edited version again.
- **Recall Sound** restores the sound to its original saved settings. Use this function if you're not happy with the edited sound.
- **Recall Kit** restores the drum kit to its original saved settings. Use this function if you're not happy with the edited drum kit.

Functions of the Preset Menu

The Preset menu in the upper left corner of the plug-in offers different entries with suggested default preferences for the various drum sounds and kits as well as two random functions. Select the corresponding entry when you know what sort of drum sound you wish to program.

- **Init Sound** restores the selected sound to its basic settings.
- **Random Sound** creates a random sound. With this parameter you easily can create new and possibly interesting sounds.
- **Init Kit** restores the drum kit to its basic settings.
- **Random Kit** creates a random drum kit. With this parameter you easily can create new and interesting kits.

MIDI Functions

The MIDI interface of the Attack is identical in its most aspects to the interface of most common synthesizers. When the MIDI connection is set up, the Attack can be played over the key range from C1 to G9 if required. The 24 sounds can be played through MIDI notes C1 to B2, thus affording 1 sound per key. This is a common layout for a drum and percussion synthesiser, making it possible to play the sounds on the keyboard next to each other.

Additionally, the upper 12 sounds of the Attack can be played melodically and polyphonically on the keyboard on MIDI Channels 1 through 12. This is because the Attack is capable of producing other sounds such as basses or leads with its synthesis functions. These sounds call for playing over a greater key range than 1 key. For these 12 sounds, the key range from C3 to G9 is available for melodic playing.

Please note that playing sounds melodically is only possible for the upper 12 sounds on MIDI channels 1 to 12.

Playing the Attack Polyphonically

The Attack offers a 5 octave keyboard. If you choose one of the upper 12 sounds with the sound select button, the Attack's keyboard and pitch bender appears on-screen, allowing you to play the selected sound melodically with your mouse on that keyboard or via a connected MIDI keyboard. As soon as you select one of the lower 12 sounds, the keyboard becomes invisible.

MIDI Control of the Attack

All functions of the Attack can be controlled by MIDI controller data. The sounds 1 through 12 receive MIDI controllers from #12 to #59, while sounds 13 through 24 receive #72 to #119. This is only possible when transmitting on channels 1 through 12. Channel 16 is reserved for controlling both delays with controllers. As an addendum to this manual, you'll find a table showing all the controller assignments.

Example: You want to distort a snare drum on Sound 2 dynamically. Therefore, you create controller data on MIDI channel 2 (= Sound 2) for the Drive parameter (MIDI controller #39). At the same time you want to bring up ring modulation for a bass on Sound 16. To do this you create a MIDI controller #89 on MIDI channel 4 (= Sound 16). If you want to change the parameter of Delay 1, use the corresponding MIDI controller on MIDI channel 16.

You can control the knobs of the Attack with an external MIDI controller unit (a hardware fader or knob box) or a MIDI master keyboard. Additional MIDI controller data can be created graphically or numerically on an editor (e.g. the List or Controller Editor of Cubase VST). The corresponding MIDI controllers and their assignments can be found in the table on page xx in this manual.

Please note that control changes edit the sound instantly, just like automation that uses system exclusive data.

Special Function Modulation Wheel

The modulation wheel of your master keyboard (MIDI controller 1) is only available for sounds 13 through 24 on the MIDI Channels 1 through 12. It controls Cutoff, in a range beginning at the selected value to the maximum, which is useful for bass and lead sounds.

Controlling the Attack through the Novation RemoteSL

Certain parameters of the Attack can be controlled through the Novation RemoteSL. The RemoteSL offers 24 encoders, potentiometers and sliders that can be mapped on up to 72 different parameters.

As you can immediately see, 72 parameters is by far not enough to control every parameter of each of the 24 sounds. Therefore, we mapped the 9 most important parameters of the first 8 sounds on the RemoteSL. These parameters are:

RemoteSL Controls	Attack Parameter
Page 1 Encoders 1...8	Oscillator 1 Pitch Sound 1...8
Page 1 Pots 1...8	Oscillator 1 FM Sound 1...8
Page 1 Sliders 1...8	Envelope 2 Decay Sound 1...8
Page 2 Encoders 1...8	Oscillator 2 Pitch Sound 1...8
Page 2 Pots 1...8	Oscillator 1 FM Envelope Sound 1...8
Page 2 Sliders 1...8	Envelope 1 Decay Sound 1...8
Page 3 Encoders 1...8	Filter Drive Sound 1...8
Page 3 Pots 1...8	Filter Cutoff Sound 1...8
Page 3 Sliders 1...8	Amplifier Volume Sound 1...8

The host application must support the Novation RemoteSL and its „AutoMap“ feature to be able to control the Attack. Please read the documentation of the corresponding host application.

Polyphony of the Attack

The Attack has up to 64 voices. The number of available voices depends on available processor power. The number of voices can easily be set on the plug-in user interface. All you have to do is use the mouse button to increase or decrease the value in the voice display.

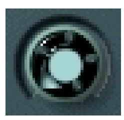
Keep in mind that each additional voice demands additional calculating power from your computer. Try to set the number of voices to a sensible value, especially if you simultaneously use other plug-ins and if you also play back audio tracks with your host application.

Hint: Even if no voice is sounding, the Attack creates a little processor load because of the two stereo delays which have to be calculated.

The Controls

Simply use the mouse to set the Attack’s controls.

- Dials: To set this type of control, click on it, hold down the mouse button and drag the mouse around the dial in a circle. Dragging in a larger circle increases the value resolution that is available. The Attack display simultaneously shows the actual value of the corresponding parameters.



Cubase VST 5.0 offers you a choice between different mouse movements for parameter setting. Find more information in your Cubase VST documentation.

- 

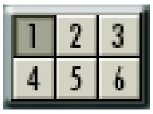
Sound select button: By clicking a Sound select button once, you select the corresponding sound.

- 

Display: If your turn a dial, its parameter name and numeric value is shown in the display. If you move the cursor over a parameter without clicking, the current numeric value is displayed.

- 

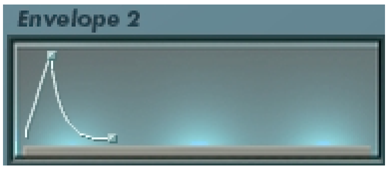
Switches: By clicking a switch once, you switch the corresponding function on or off.

- 

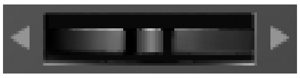
Button group: By clicking a button of a group, you select the corresponding function.

- 

Value selector: Position the mouse pointer on the value, hold down the mouse button and drag up or down. If you hold down the [Shift] key while changing the value, you will get another value scale area.

- 

Envelope graphic display: Click on one of the handles and drag the mouse to continuously and smoothly change the envelope parameters, or click into any envelope phase to let its value jump there.

- 
Pitch bender: To change the pitch, click on the bender and drag the mouse to the left or right. The pitch bender snaps back into its center position as soon as you let go of the mouse button.

- Attack logo: If you click on the Attack logo, a window opens with program information and MIDI controller documentation.

- 
Automatic keyboard: This integrated keyboard pops up automatically. For more information, refer to the chapter “The automatic keyboard”.

Key Combinations

- If you hold down the [Strg] key (on the PC) or [Command] key (on a Macintosh computer) when you click on a control, its value is automatically set to its default (e.g. Cutoff is set to 100%, Env to 0%).
- You can increase the resolution of a dial by holding down the [Shift] key on your computer keyboard when use the “linear knob mode”. If a dial is set to “circle mode” (so that you change a value by dragging around the dial in a circle), then you can temporarily change to “linear knob mode” by holding down [Alt].

The "Automatic" Keyboard

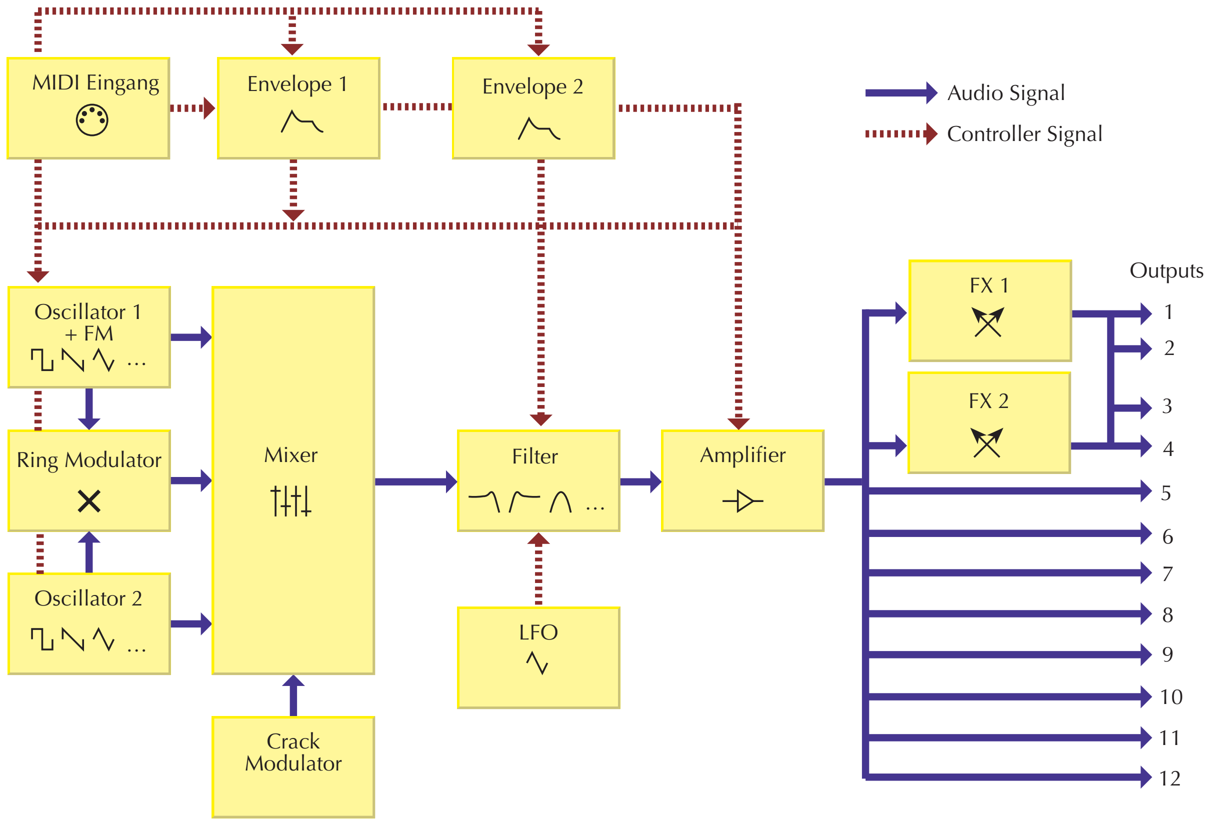
The Attack has a 5 octave keyboard in its user interface. When selecting one of the upper 12 sounds, the Attack's keyboard screen will appear. You can play these sounds melodically through your MIDI keyboard or by clicking on the displayed keyboard with your mouse. When selecting one of the lower 12 sounds, the keyboard is hidden.

The User Interface of the Attack

The user interface of the Attack is split into useful segments for easy access to different parameters. Though the Attack is primarily designed for creating drum and percussion sounds, the setup is reminiscent of a synthesiser offering subtractive synthesis. To the left you can see 24 Sound Select buttons in the form of a stylized keyboard positioned vertically. With these buttons you can select the sounds of a drum kit. Next appear parameter groups for both Oscillators, the Mixer, the Filter, the Amplifier, the Delay effects, the Crack Modulator and settings for both Envelopes. A display as well as some pop-up menus round out the programming interface.

Due to its flexible synthesis structure, the Attack is capable of creating classic synthesiser sounds, especially basses and leads.

In the chart you see a graphical presentation of the signal flow within the Attack. Following this chart, the parameter groups and their functions are explained in detail.



Oscillator Section

The Attack is equipped with two oscillators that have almost identical functionality. Oscillator 1 additionally contains an integrated FM (frequency modulation) section. The following explanation is valid for both oscillators.

Pitch
0.007984...<20000 Hz

Sets the pitch of the oscillator over a very wide range. This is necessary to reproduce drum and percussion sounds.

Semitone
C-11...E 10

This parameter works with **Pitch**, setting the pitch of the oscillator in semitone steps. This is useful for melodic sounds. Lead and Solo sounds sound interesting when you set one oscillator to e.g. a fourth (+5 semitones).

Detune
-50...+50

This parameter also works with **Pitch**, fine-tuning the oscillator in cents. The audible result of detuned oscillators is a chorus or flanger effect. Use a positive setting for one oscillator and an equivalent negative setting for another.

Shape (Waveform)
Tri, Sine, Pulse, Saw, S&H, Noise, Sample 1-3

Sets the type of waveform to be generated by the oscillator. The parameter is called **Shape** instead of “waveform”, because it doesn’t necessarily set only classic synthesizer waveforms, but it also generates noise, sample&hold and three samples. However, the term “waveform” is used interchangeably throughout this manual. The following shapes are currently available:

- Triangle selects the triangular waveform. The triangle mainly consists of odd harmonics with very low magnitudes. It is perfect for nearly all drum and percussion sounds. The *Sine* waveform consists of the fundamental frequency only. It has no harmonics at all. With a sine wave you can create clean bass drums and snares. The sine wave is also perfect for FM. *Pulse* selects the pulse waveform. This waveform produces a hollow / metallic sound and is perfect for bass drums and snare drums.
- *Saw* selects the sawtooth waveform. A sawtooth wave has all the harmonics of the fundamental frequency in descending magnitude. This waveform is pleasing to the ear. You can use it for bass and lead sounds.
- S&H (Sample&Hold): S&H samples a random value and holds it. **Pitch** selects the time of this process. S&H is perfect for industrial-type FX sounds and as an FM source.

- Noise is a fundamental source for any kind of analog-type percussion, especially hi-hats and snare drums. Also, wind and other sound effects can be created by using noise. If noise is selected, you can “color” it with the **Pitch** knob. Negative values create pink noise (fewer high frequencies), positive values a blue noise (fewer low frequencies).

The next three buttons don´t set an oscillator waveform; instead, you can choose a sample for further programming.

- Sample 1 contains a closed hi-hat. It can be processed with Pitch and FM and all other synthesis functions.
- Sample 2 contains an open hi-hat. It can be processed with Pitch and FM and all other synthesis functions.
- Sample 3 contains a crash cymbal. It can be processed with Pitch and FM and all other synthesis functions.

The square, sine, triangle and sawtooth waveforms always start at full amplitude to create a necessary start click. This is characteristic of drum and percussion sounds. To avoid this click, simply open up the attack value for Envelope 2 by a small amount. When creating typical synthesizer sounds you might like to avoid the flanging effect caused by detuned oscillators. This effect is caused by the fixed phase of the waveforms. To do this, assign a short pitch envelope to one of the oscillators.

Pitch Env (Envelope) *-100%...100%*

Sets the amount of pitch modulation from Envelope 1 or 2 (depending on the settings). Positive amounts will raise the pitch when envelope modulation is applied. Negative amounts will lower the pitch when envelope modulation is applied. Use this parameter to create time-dependent pitch changes.

Pitch Env is one of the most important parameters in drum programming because it simulates the character of a real drum.

Pitch Vel (Velocity) *-100%...100%*

Determines the amount of influence the selected envelope has on the pitch, based on key velocity. This parameter works similarly to the **Pitch Env** parameter, with the difference that its intensity is velocity based. Use this feature to give a more expressive character to the sound. When you hit the keys smoothly, only minimal modulation is applied. When you hit harder, the modulation amount also gets stronger.

The overall modulation applied to the pitch modulation is calculated as the sum of both the **Pitch Env** and **Pitch Vel** parameters. Therefore you should always bear this total in mind, especially when pitch does not behave as you expect. You can also create interesting effects by setting one parameter to a positive and the other to a negative amount.

FM (Frequency Modulation)

Sets the amount of frequency modulation that is applied to Oscillator 1 by Oscillator 2. The sound will get more metallic and sometimes even drift out of tune. Triangle waves, sine waves and noise are especially suited for FM. To change the frequency modulation dynamically, use an envelope or velocity. The FM range of the Attack is very wide, so that you can generate nearly chaotic FM out of periodic waveforms like sine waves. This is necessary to create hi-hats.

If you use noise as FM source, the sound will become more tonal when you use higher **FM** settings. To create a vibrato, set Oscillator 2 to a deep pitched triangle waveform and use very low FM settings. Playing this sound at low octaves creates a wobble effect.

FM background hints: The frequency modulation of the Attack modulates the phase of Oscillator 1 with the amplitude of Oscillator 2. This effect can be very strong, such that there can be phase overflows by a factor of 8. This creates noisy waveforms – perfect for drum sounds. Lower FM settings generate many different spectra of a metallic character. A FM envelope can change the metallic character to chaotic FM, also necessary for drum sounds. Another point to observe is that FM into the Attack is scaled linearly.

FM Env

-100%...100%

Sets the amount of FM with Envelope 1 or 2 (depending on the settings). Positive amounts will raise FM when envelope modulation is applied. Negative amounts will lower FM when envelope modulation is applied. Use this parameter to create time-dependent FM changes.

FM Vel

-100%...100%

Determines the amount of influence the selected envelope has on FM, based on key velocity. This parameter works similarly to the **FM Env** parameter, but with the difference that its intensity is velocity based. Use this feature to give a more expressive character to the sound. When you hit the keys smoothly, only minimal modulation is applied. When you hit harder, the modulation amount also gets stronger.

The overall modulation applied to FM is calculated as the sum of both the **FM Env** and **FM Vel** parameters. Therefore you should always bear this total in mind, especially when FM does not behave as you expect. You can also create interesting effects by setting one parameter to a positive and the other to a negative amount.

Mixer Section

In the Mixer, you control the volumes of the two oscillators. Ring modulation and Crack modulation can be applied optionally to extend the Attack´s tonal range.

Osc 1 *0%...100%*

Volume of Oscillator 1.

Osc 2 *0%...100%*

Volume of Oscillator 2.

If the sum of all mixer signals (Osc 1, Osc 2 and Ring Modulator) is higher than 100%, filter saturation will be attained. At this point Resonance doesn´t make any volume difference, as it does in the normal filtering process. Use this phenomenon for additional sound manipulation.

Env (Envelope) *-100%...100%*

This controls the influence of Envelope 1 or 2 on the Oscillator 2 level. With positive settings, the level is increased by the modulation of the envelope, and with negative settings, the level is decreased. Use this parameter to change the volume of Oscillator 2 independently over time.

Vel (Velocity) *-100%...100%*

Determines the amount of influence the selected envelope has on level of Oscillator 2, based on key velocity. This parameter works similarly to the **Env** parameter, with the difference that its intensity is velocity based. When you hit the keys smoothly, the level of Oscillator 2 only rises minimally. When you hit harder, the level will rise higher.

The overall modulation applied to Oscillator 2 level is calculated as the sum of both the **Env** and **Vel** parameters. Therefore you should always bear this total in mind, especially when the level does not behave as you expect. You can also create interesting effects by setting one parameter to a positive and the other to a negative amount.

Rmod (Ring Modulation) *0%...100%*

Volume of the ring modulation between Oscillators 1 and 2. From a technical point of view, ring modulation is the multiplication of two oscillators' signals. The result of this operation is a waveform that contains the sums and the differences of the source frequency components. Since ring modulation generates disharmonic components, it can be used to add metallic distorted sound characteristics. This is useful when generating crashes or cowbells. Please note that in a complex waveform all harmonic components behave like interacting sine waves, resulting in a wide spectral range of the ring modulated sound.

Ring modulation can result in unwanted low frequencies when the pitches of Oscillators 1 and 2 don't differ very much. This is logical because, for example, when you use one oscillator set to 100 Hz and the second set to 101 Hz, the resulting ring modulation is 201 Hz and 1 Hz, and 1 Hz is very low.

Crack *0%...100%*

Fades in the Crack Modulator. The Crack Modulator was designed especially for creating hand clap sounds. Technically it is an amplitude modulation using a sawtooth waveform. The speed and the number of waveforms can be chosen. After transmitting its intended modulation, the Crack Modulator resumes emission of a constant signal.

The Crack Modulator superimposes its effect on all other mixer signals (Oscillator 1 and 2, Ring Modulator).

Crack Speed *1Hz...5000Hz*

Determines the frequency of the Crack Modulator.

Crack Length *1 Cycles...∞ Cycles*

Determines the number of modulations the Crack Modulator creates.

> To program an authentic sounding handclap, set Crack Speed to 105 Hz and Crack Length to 3 cycles.

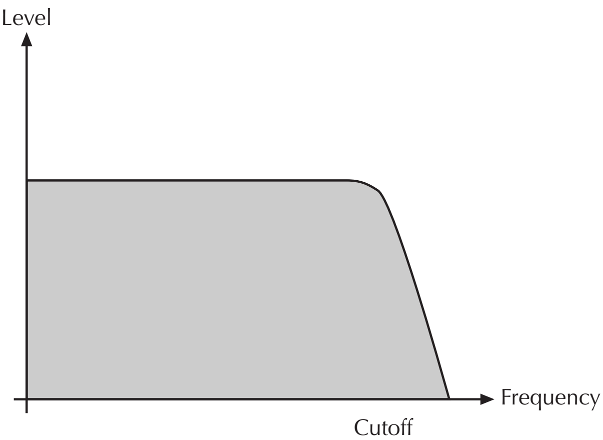
Filter Section

Once the audio signal leaves the mixer, it is sent to the filters. The Attack offers a multimode filter with different filter types.

In a subtractive synthesizer a filter is a component that have significant influence on sound characteristics. But the Attack was designed to make drum and percussion sounds, for which the filter is used merely for fine tuning the sound. Yet you can also create bass and lead sounds which definitely require a filter.

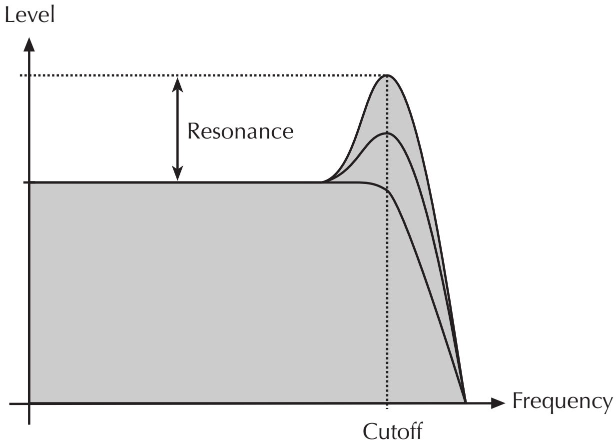
For now, we'll explain the basic function of a filter, discussing the type used most commonly in synthesizers: the low pass filter.

The low pass filter type dampens frequencies that lie above a specified cutoff frequency. Frequencies below this threshold are barely affected. The frequency below the cutoff point is called the pass band range, and the frequencies above are called the stop band range. The Attack´s filter dampens frequencies in the stop band with a certain slope. The slope is 12dB per octave. This means that the level of a frequency that lies an octave above the cutoff point will be 12dB less than those frequencies of the signal that fall into the pass band. The following picture shows the basic principle of a low pass filter:



Low pass filter principle

The Attack´s filter also features a resonance parameter. Resonance in the context of a low, band or high pass filter means that a narrow frequency band around the cutoff point is emphasized. The following picture shows the effect of the resonance parameter on the filter's frequency curve:



Low pass filter resonance

If the resonance is raised to a great enough extent, then the filter will begin self-oscillation, i.e. the filter generates an audible sine wave even when it does not receive an incoming signal.

Beside the low pass filter, the Attack offers the following filter types:

Type (Filter Types)

The following filter types are available when selected with the mouse:

-  12 dB Low Pass Filter. Use this type if you want to create sounds with a typical audible filtered character; for example, bass or lead sounds. With a low pass filter you can shape bass drums or snare drums.
-  12 dB Hi Pass Filter. This type is useful to thin out a sound's bass frequencies. This may also give interesting results in conjunction with cutoff frequency modulation. By doing this you can "fly-in" a sound, starting with its high harmonics and then coming up to its full frequency range. You can cut the bass and mid range of high frequency drum sounds like hi-hats or crashes.
-  12 dB Band Pass Filter. This type removes frequencies both below and above the cutoff point. As a result, the sound character gets thinner. Use these filter types when programming effect and percussion-like sounds.
-  12 dB Notch Filter. This type is the opposite of the band pass type. It dampens frequencies around the cutoff point. Frequencies below or above the cutoff point are passed through. Use this filter type for programming effect sounds. Resonance has no great influence because it raises the frequencies that the notch filter dampens. You can hear a little effect of phase changes, but not to any great degree.
-  EQ Lo- or Hi- Shelf Filter: Unlike a normal synthesiser filter, this filter type works like an equalizer. When the resonance parameter is set below 50%, the filter works as Hi Shelf, raising high frequencies up to 12 db. Values over 50% cause the filter to work as Lo Shelf. Now, deep frequencies can be raised up to 12db. The Cutoff knob sets the center frequency of the Shelf EQ.

-  EQ Bell Type Function: Like the EQ Lo- or Hi-Shelf, this filter type has an equalizer function. The Resonance knob serves as gain, raising or lowering the frequency set with the Cutoff knob up to 12db.

Cutoff

11.56 Hz...18794 Hz

Controls the cutoff frequency for the low pass and high pass filter types, and the center frequency for the band pass and notch filter types. It has a special function for the EQ Type Filter.

- When the low pass type is selected, all frequencies above the cutoff frequency are dampened.
- When the high pass type is selected, all frequencies below the cutoff frequency are dampened.
- When the band pass type is selected, only frequencies near the cutoff setting will be passed through.
- When the notch type is selected, the frequencies near the cutoff frequency are dampened.
- When EQ Lo- or Hi-Shelf is selected, the cutoff knob selects the start frequency of the Shelf EQ.
- When EQ Bell Type is selected, the frequencies near the cutoff frequency are boosted or dampened with the Resonance parameter.

Resonance

0%...100%

Controls the emphasis of the frequencies around the cutoff point (exept the EQ filter types). Use lower values to give more brilliance to the sound. At higher values the sound acquires a typical filter character with a strong boost around the cutoff frequency. When the setting is raised to maximum, the filter starts to self-oscillate, generating a pure sine wave. This feature can be used to create typical solo sounds.

Env (Filter Envelope Amount)

-100%...100%

Determines the amount of influence the selected envelope has on cutoff frequency. For positive settings, the filter cutoff frequency is increased by the modulation of the envelope, and for negative settings, the cutoff frequency is decreased. Use this parameter to change the timbre of the sound over time. Sounds with a hard attack usually have a positive envelope amount that makes the start phase bright and then closes the filter for a darker sustain phase.

Vel (Filter Velocity)

-100%...100%

Determines the amount of influence the selected envelope has on the cutoff frequency, based on key velocity. This parameter works similarly to the **Filter Env** parameter, with the difference that its intensity is velocity based. Use this feature to give a more expressive character to the sound. When you hit the keys smoothly, only minimal modulation is applied. When you hit harder, the modulation amount also gets stronger.

The overall modulation applied to the filter’s cutoff frequency is calculated as the sum of both the **Filter Env** and **Filter Vel** parameters. Therefore you should always bear this total in mind, especially when the filter does not behave as you expect. You can also create interesting effects by setting one parameter to a positive and the other to a negative amount.

Drive
0 dB...54 dB

Determines the amount of saturation that is added to the signal. If set to 0, no saturation will be added or, in other words, the signal will remain clean. Lower values will add some harmonics to the signal, resulting in a warm character. Increasing the value will bring in more and more distortion, suitable for harder lead sounds and effects.

Sync

Makes the low frequency oscillator (LFO) sync the filter modulation either to the key press or to the tempo of the host application (assuming that it is capable of sending the needed timing information to the Attack). When you click on the Sync button, a pop-up menu will appear to select the different settings.

- **Off** lets the LFO run freely.
- **Key** makes the triangle waveform of the LFO start at the maximum amplitude whenever a key is pressed.
- **8 Bars** to **1/64** syncs the triangle waveform of the LFO in musical note values.
- **1/2. Bars** to **1/64.** syncs the triangle waveform of the LFO to dotted note values.
- **1/1t Bars** to **1/64t** syncs the triangle waveform of the LFO to third note values.

With positive **Mod Depth** values the LFO starts at maximum amplitude, and inversely, with negative values it starts at minimum amplitude.

Mod Speed
S&H...1000 Hz

The integrated Low Frequency Oscillator (LFO) creates a Triangle wave to modulate filter frequency. **Mod Speed** determines the frequency of the LFO. At low values, it might take several minutes for the LFO to perform a complete cycle, while higher values are in the audible range. A setting of *S&H* creates a random value which will be held for the length of a tone (independent from the **Sync** settings).

Mod Depth
-100%...100%

Determines the amount of filter frequency modulation by the LFO.

Amplifier Section

This section is found near the end of the Attack's signal routing, followed by the Delay FX section. Its purpose is to set the volume of the sound.

It is important to know that Envelope 2 always controls the amplifier level.

Additionally the Amplifier section offers sound output, panning and effect send levels.

Output *1...6*

The Attack offers 6 stereophonic audio outputs. This parameter sets the audio output of the selected sound. You can use this function to assign different effects or EQ settings to sounds of a kit.

Please take into account that Delay 1 only appears on stereo output 1, and Delay 2 only on stereo output 2. Outputs 3 to 6 are always dry.

Volume *-∞ dB...0 dB*

Sets the output volume of the selected sound.

Vel (Velocity) *-100%...100%*

Specifies by how much volume will be affected by keyboard velocity. Use this feature to give more expression to the sound. With a setting of 0, velocity will have no effect on volume. For positive settings, the volume rises with higher velocities. This is the most commonly used setting. With negative settings, the volume decreases at higher velocities. This gives an aypical character suitable for effect sounds. The maximum volume is always set with the **Volume** parameter.

Pan (Panning) *100%L...Center...100%R*

Determines the position in the stereo panorama. When the setting is to the left, the sound is panned far left; when the setting is to the right, it is panned far right. If you want to situate the sound in the middle of the stereo panorama, use the *Center* setting. Because the delay effects are independent from the pan settings, the higher the proportion of delayed signal in your mix, the less pronounced the panning effect wil be.

Mix *100:0%...0:100%*

This parameter controls the volume ratio between the original signal and the effect output of the integrated delay effects. If set to *100:0*, the dry signal is sent to the outputs only, so that no effects can be heard. Higher values will increase the effect signal. At the maximum setting, *0:100*, the pure effected signal will be heard.

XOR Group *Off, 1, 2, 3*

Assigns the selected instrument to one of the three XOR groups. When several instruments of the same XOR group receive MIDI notes, the sounding instrument will be muted by the succeeding one. Use this function to program realistic hi-hats, especially if only the open or closed hat should be heard. This parameter can also be used to create monophonic synth sounds. When you set the sync parameter in the filter section to *Off* and assign different instruments to the same XOR group, the LFO stays in sync nonetheless, since the same voice is always triggered.

Delay Effect Sections

The Waldorf Attack offers two independent Modulation Delay effects. These effects are tied to outputs 1 and 2. When a sound is routed to 1, delay 1 is selected automatically, and similarly delay 2 appears at output 2.

A delay creates repeats of the input signal. The delay time can be set either in milliseconds or in musical note values if Sync is selected (this assumes that your host application can send tempo information to the plug-in). The maximum delay time is 2 seconds. Additionally, the delay can be modulated.

Time *0ms...2000ms*

Sets the length of the delay tap in milliseconds or note values depending on the **Sync** setting (if your host application supports VST 2.0).

Sync

Syncs the delay time to the tempo of the host application (assuming that it is capable of sending needed timing information to the Attack). When you click on the (delay-) **Sync** button, a pop-up menu will appear to select the different settings.

- **Off** allows no synchronisation. **Time** can be set in milliseconds.
- **2 Bars** to **1/64** syncs the delay time in musical note values.
- **1/2. Bars** to **1/64.** syncs the delay time in dotted note values.
- **1/1t Bars** to **1/64t** syncs the the delay time in third note values.

Cubase VST offers an internal synchronisation with version 3.7 (PC) and 4.1 (MAC) or higher, offering more precision than MIDI clock.

Feedback *-100%...100%*

Controls the amount of signal that is routed back into the delay line. Lower feedback values will by definition produce fewer echoes than higher values. Very short delay time settings can be colored by feedback settings.

Spread *50:100%...100:50%*

Spreads the left and right delay output to half of the delay time maximum. Settings from *50:100%* or *100:50%* create a typical ping pong delay.

Speed *0.01 Hz...10 Hz*

The delay effect can be modulated in its delay time with the integrated low frequency oscillator (LFO). **Speed** determines the frequency of the LFO.

Depth *0%...100%*

Controls the modulation depth when delay time is changed by means of the LFO. The depth ranges from no delay to the amount set by the **Time** parameter.

Lo Cut

Dampens the lower frequencies of the signal generated by the delay effect. The filter is inserted before the feedback loop, thus dampening each successive step. Lower values suppress deeper frequencies from the feedback. In conjunction with the **Hi Cut** Parameter, the delay effect can thus be narrowed to a certain frequency range.

Hi Cut

Dampens the higher frequencies of the signal generated by the delay effect. The filter is inserted before the feedback loop, thus dampening each successive step. This creates the typical dulled effect familiar from natural delays. Lower values suppress deeper frequencies from the feedback. In conjunction with the **Lo Cut** Parameter, the delay effect can thus be narrowed to a certain frequency range.

Envelopes

The envelopes create a control signal that varies with time. They are used, for example, to modulate the filter settings or the level of a sound within a given period of time. The envelopes are structured identically, and offer Attack, Decay, Shape and Release. Attack and Decay can also be graphically edited. If you press a key, the envelope is started. The envelope parameter has the following functions:

- **Attack** defines the period of time for the envelope to rise to its maximum.
- **Decay** defines the time for the envelope to fall to 0.
- **Shape** controls the shape of the Decay and Release phases. You can shade from exponential to linear to inverse exponential, or to a combination of exponential and inverse exponential (cosine like).
- After you let go of the key, **Release** defines the period of time it takes for the envelope to sink back to zero. If you turn Release fully left, this parameter is ignored. This is the most useful setting for drum sounds.

Both envelopes can be edited quickly and easily with the mouse. Editing is simplified by the graphic changes you see in the corresponding function.

To edit, click on the respective handle and drag in the desired direction. The changes and their parameter names are visible in the display. Keep the following in mind:

Envelopes 1 and 2 are structured identically. Attack and Decay are time-dependent parameters, which is why they can only be moved horizontally.

Envelope 1

Envelope 1 allows you to manipulate different sound parameters. The intensity of modulation is controlled with the corresponding **Env** or **Vel** parameter.

Envelope 2

Envelope 2 is structured identically to Envelope 1, but is pre-routed to the amplifier level.

Appendix

Programming Drum Sounds

To understand how to create drum sounds, you should know a little about how the classic drum machines worked. The following sections give some insights how particular sounds were built and information how to achieve similar results on the Waldorf Attack.

Roland TR-808 Bass Drum

On the Roland TR-808, this sound was made by one filter with a high resonance setting,

triggered by a short impulse. Two controls were provided to adjust the bass drum: "Tone" was used to set the pitch by changing the filter's cutoff frequency, and "Decay" was used to set its resonance, which in turn controlled the decay rate.

On the Attack, you could use the filter´s self-oscillation by triggering it with a short noise impulse made by the second oscillator, whose volume you can control by an envelope.

But a better way is to use Oscillator 1 playing a sine wave, and by changing the initial click impulse by means of the filter.

Roland TR-909 Bass Drum

The Roland TR-909 used an oscillator and a noisy click, controlled by three envelopes, to create a bass drum sound. The oscillator played a sine wave whose pitch was controlled by an envelope and the "Tune" control. The rate of the envelope’s decay was not adjustable.

This oscillator signal was routed to an amplifier with an envelope whose "Decay" parameter adjusted the decay rate of the envelope. The second part of the bass drum sound was made with a short impulse and a low pass filtered noise generator, both summed and routed into another envelope that controlled their output volume. The "Attack" parameter controlled the overall pulse/noise level, and the decay rate of the envelope was not adjustable.

On the Attack, you can make this sound as follows: Oscillator 1 plays a sine wave, and Envelope 2 is used to modulate its pitch. This means that the pitch of the oscillator becomes higher or lower depending on the setting of the Decay parameter of Envelope 2, but this slight variation doesn't affect the drum sound once it has been set up.

The noise of the impulse can be ignored, because it is low pass filtered anyway. But how do we create an impulse with the Attack? The answer is simply to use a square wave with a very low pitch setting for Oscillator 2, and to control its level with a very short envelope. Now we have an impulse. This impulse is low pass filtered afterwards with a slightly resonating filter, preferrably set to around 5000Hz with a resonance of around 18%.

With the Oscillator 1 Pitch and Pitch Env controls you can adjust the sound of the bass drum, while Envelope 2 Decay controls its length.

Simmons SDS-5 Bass Drum

The Simmons SDS-5 bass drum consists of an oscillator and a noise generator, both routed into a low pass filter and an amplifier. An envelope controls the oscillator pitch, the filter cutoff, and the amplifier volume. The envelope has a decay shape that is in-between exponential and linear.

The oscillator plays a triangle wave whose pitch is controlled by a "Tune" control and a "Bend" parameter that controls the influence of the amp envelope to oscillator pitch.

A "Noise - Tone" parameter controls the mix between the oscillator and the noise generator.

A "Noise" parameter controls the filter cutoff. (Very confusing, isn't it?)

A "Decay" parameter controls the envelope decay rate.

A "Click - Drum" parameter controls the most important aspect of the Simmons drums: the mix between the original signal from the pad trigger microphone and the triggered drum sound.

On the Attack, you can make this sound as follows:

Oscillator 1 plays a triangle or sine wave pitched at around 30Hz, and Envelope 2 is used to modulate its pitch. Use the "Vel" control to simulate the velocity-dependent pitch bend amount that you would find on the SDS-5. You can simulate the click by setting FM Env to a medium value, with Envelope 1 set to a very short decay. Oscillator 2 generates noise, and the pitch is set to center. The Filter Cutoff can vary between 100Hz and 5000Hz, and Vel should be set at 25% or so. Filter Resonance should be set to 10%. Envelope 2 should be set to an almost linear shape. Use Osc 1 and Osc 2 Level to adjust the mix of tone and noise, and use Osc 1 FM Env to vary the click strength.

Roland TR-808 Snare Drum

On the Roland TR-808, the snare drum was made of two resonating filters and a noise generator with high pass filtering. The "Tone" parameter controlled the output mix from the first and the second filters, while "Snappy" controlled the volume of the noise generator. The noise generator was routed through a separate envelope and a high pass filter.

On the Attack, you can make this sound as follows:

Oscillator 1 plays a sine wave at around 150Hz, and you can use a little FM to disturb the periodic character of the sine wave. This trick makes the oscillator sound thicker, almost as if two oscillators were running at once.

Oscillator 2 generates noise, and you should use Pitch to high pass filter it.

In the Mixer, turn up Osc 1 to 50% and Osc 2 Env to 50%, set to Envelope 1.

Set Envelope 1 to a shorter decay phase than Envelope 2.

Use the filter with a low pass setting and add a little resonance to emphasize the high frequency range.

Roland TR-909 Snare Drum

The TR-909 Snare Drum was made with two oscillators and two filters for noise. The two oscillators started in phase but were slightly detuned, and one of the oscillators was modulated a bit by a pitch envelope. The "Tune" parameter controlled the basic pitch of the two oscillators. The noise was split in two parts: there is always some low pass filtered noise during the whole snare drum sound, while a high pass filtered sound is routed through another envelope whose level can be controlled by the "Snappy" parameter.

One Attack sound doesn't feature as many different modules as the TR-909 snare drum had. One solution can be to use two sounds, one emulating the first oscillator and the low pass filtered noise, and the other emulating the second oscillator plus the high pass filtered noise. You will have to play the two simultaneously in your track, but this shouldn't be a problem because the Attack has sample-exact timing.

However, you can re-create the TR-909 snare drum with just one Attack sound instead, by doing the following:

Set up Oscillator 1 to play a sine wave, modulate its pitch slightly with Envelope 2, and add a little FM to it - around 0.1 to 0.5%. When you set the second oscillator to produce noise, you will hear that the sine wave gets smeared, which means that you are not hearing an exact tone any more. This already sounds very close to two slightly detuned oscillators and a low pass filtered noise. Now you only need the "Snappy" part, which is added simply by using Envelope 1 to modulate Oscillator 2's mix level. You can high pass filter the noise with the Pitch control, but in fact the result is already quite similar without doing so. If you want a little more punch, use the Drive control carefully until you can hear a slight distortion at the beginning of the sound.

Another variation can be heard in the sound library that comes with the Attack. This one uses a very low noise signal level that is boosted greatly behind the high pass filter. The reason for this is that Oscillator 2 plays the tone of the snare drum while the high pass filter dampens this tone heavily. To raise it back up to a good volume, Drive boosts it to a normal level.

Simmons SDS-5 Snare Drum

The Simmons SDS-5 Snare Drum module was laid out identically to the Bass Drum module. However, a number of parameters were set in a different way internally to create snare drum sounds.

When you want to create Simmons snare drum sounds on the Attack, just keep in mind that you should use a very short envelope to frequency modulate the first oscillator, set the envelopes to almost linear shapes, and use the "Vel" control for all envelope modulations.

TR-808 Side Stick

The TR-808 Side Stick (called RS on the 808, which stands for Rim Shot) sound is very tricky: although it consists of only two oscillators running through an amplifier and a high pass filter, the sound is very complex. This comes from the fact that one oscillator seems to “cut” the other oscillator and that the VCA is used to add high harmonics. How Roland did it is something only they and maybe a handful of people know. If you happen to be one of these people, let us know!

If you want a sound of this type from the Attack, use the representative sound from the library instead of trying to simulate it on your own. Look at the parameters and try to find out why it sounds quite close. A couple of hints: Crack is used with a very high frequency setting doing amplitude modulation on the oscillators’ summed signal, and Drive is used to add further harmonics by distorting the signal.

TR-909 Side Stick

The TR-909 Side Stick is made of 3 resonating band pass filters that are triggered by a short impulse. Behind the band pass filter cluster there is a distortion unit, followed by a VCA with an envelope and a high pass filter.

Its specific sound comes from the cutoff frequencies, the resonance, and the volumes of the trigger impulses of the three band passes. These settings are:

* 500Hz, 20ms decay, full volume

* 222Hz, 45ms decay, half volume

* 1000Hz, 5ms decay, full volume

Now, the Attack doesn't have three band passes plus a high pass filter, but there's a way to simulate the architecture with the Attack.

What produces a resonating band pass filter? Nothing more than a sine wave. So, why not just use two oscillators producing two sine waves, plus a high pass filter that uses the lowest frequency setting as the third sine wave generator. Thus the filter will include both oscillator signals and add its own resonance to the sum.

The high pass filter is therefore set to 222Hz, with a resonance of 100%. Oscillator 1 produces the 500Hz sine wave, while Oscillator 2 is set to a 1000Hz sine wave, but is controlled by a very short Envelope 1 set to around 75% to produce the 5ms signal. The fact that oscillator 1 plays longer than 20ms can be ignored, because it's not that noticable. Don't add it with full volume, however; set it only to a level of around 25%. This comes into play because there is an additional high pass filter on the original TR-909 Side Stick that dampens lower frequencies.

Finally, add a good amount of Drive (around 30dB) to the signal, set Envelope 2 Decay to 45ms, and you will have the sharp attack of the original sound.

TR-909 Hand Claps

TR-909 hand claps are made using the same signal routing as in the original TR-808. However, due to the differing parts and internal parameter settings that were used in the TR-909, the TR-909's hand clap sounded different. Essentially, the "Crack" (or as Roland called it "Sawtooth Envelope") was clearer, and the reverb effect was longer.

Hi-hats

For hi-hats, we don't use references to classic drum machines, although there is a quite good emulation in the TR-808 set included in the sound library. Hi-hats can be made in various ways:

The simplest method is to use the built-in samples of the Attack. However, those are provided just in case you don't have time to "synthesize" a good-sounding hi-hat.

If you like a really "vintage" sound, use a high pass filtered noise. This gives the very archaic hi-hat sound that was used by many drum machine companies for years. One of the last examples of this sound was the good old Roland CR-78.

If you want more sophisticated results, use FM. Don't use noise as FM source, but instead use a sine or triangle waveform with a very high pitch. The modulated oscillator can be set either to square or sine. The FM of the Attack has a maximum amount of around 8 waveform cycles, which results in heavy but tonal noise. When you don't use a static FM but change the amount by an envelope, the sound gets really exciting. The noisy FM effect changes over time, resulting in a very lively hi-hat sound. You will probably have to experiment with the settings of Oscillator 2 Pitch and FM Env, but the results are very much worth the work.

A good rule of thumb is to start with the Open Hi-hat sound, and copy that sound to the location for the Closed Hi-hat. Making a hi-hat sound shorter almost always succeeds, but making a short hi-hat longer may result in an unwanted characteristics. Also, don't forget to set the sounds to the same XOR Group so that they cut off one another.

Cymbals

With cymbals, the situation is similar to hi-hats. A sample is provided for an authentic crash cymbal, though you can achieve more interesting and unique results using filtered noise or FM.

Ride cymbals are more difficult to create, and their sound is so special that you might wish to use a good sampler or sample player to generate those sounds. If you want to create your own ride cymbal sounds anyway, you might come up with interesting results using FM and ring modulation.

Toms

Tonal percussion instruments can be created easily. Just set one oscillator to produce a sine or a triangle wave, modulate its pitch by an envelope, and set up the second oscillator to create either the attack noise or the resonance skin. When you want to do the latter, just copy the settings of the first oscillator and change the pitch or the envelope depth a little.

Also, it might be interesting to remove a little of the "tone" from Oscillator 1 by applying FM from the second oscillator producing noise. Note that a short envelope used for FM creates astounding drum stick hit sounds. Furthermore, you can high pass filter the result to get more punch and less tone into the sound.

Congas

Congas can be made by using a sine wave oscillator, with a very short envelope controlling the FM amount of Oscillator 2 producing noise. This, together with a medium fast attack on Envelope 2, creates very authentic conga sounds.

Muted or slapped congas can be made by increasing the basic FM amount a little and using a high pass filter to dampen the "tone".

Shakers and Maracas

Both are made with noise, either unfiltered or used to frequency modulate Oscillator 1 in order to create strong colorization. A high pass filter can be applied to remove some low end.

The difference between shakers and maracas from the synthesist's viewpoint is that a shaker has a longer attack and decay phase than maracas. Of course the sound depends a great deal on how you play, so don't forget to set up velocity-based changes to the amplifier.

Claves and Woodblock

Claves and woodblock sounds are also very similar. They both consist of very short sine or triangle waveforms. A woodblock is lower in frequency, and you can add the second oscillator to produce a different frequency. Claves should be made with only one sine oscillator and a very short envelope.

TR-808 Cowbell

You are waiting for this one, aren't you? The TR-808 Cowbell is made of two square oscillators, one oscillating at 540Hz, the other oscillating at 800Hz. The attack phase of the envelope is emphasized heavily to create the strong click. Afterwards, the summed signal is sent through a band pass filter and an envelope that stops abruptly.

A funny side note: on the TR-808, the square oscillators were the same that were used for the cymbal and hi-hat sounds. However, those sounds used a cluster of six detuned square oscillators with different band pass and high pass filter settings.

Concluding Remarks

We hope that this manual has helped you to familiarize yourself with the interesting world of analog drum sound synthesis.

Since 1989, Waldorf has cultivated a tradition of synthesizers, and continues to develop a wide range of different units based on powerful sound generation systems. If you like the sound character and expressiveness of the Attack, we invite you to drop by to visit our web site at:

http://www.waldorfmusic.de

We wish you a lot of creative fun with your Attack!

Glossary

Aftertouch

The majority of contemporary keyboards are capable of generating aftertouch messages. On this type of keyboard, when you press harder on a key you are already holding down, a MIDI aftertouch message is generated. This feature makes sounds even more expressive (e.g. through vibrato).

Aliasing

Aliasing is an audible side effect arising in digital systems as soon as a signal contains harmonics higher than half the sampling frequency.

Amplifier

An amplifier is a component that influences the volume level of a sound via a control signal. This control signal is often generated by an envelope or an LFO.

Attack

An envelope parameter. "Attack" is a term that describes the ascent rate of an envelope from its starting point to where it reaches its highest value. The Attack phase is initiated immediately after a trigger signal is received, i.e. after you play a note on the keyboard.

Band Pass Filter

A band pass filter allows only those frequencies around the cutoff frequency to pass. Frequencies both below and above the cutoff point are dampened.

Band Stop Filter (Notch Filter)

A band stop filter does the opposite of a band pass filter, i.e. it dampens only the frequencies around the cutoff point and lets all other frequencies pass through.

Clipping

Clipping is a sort of distortion that occurs when a signal exceeds its maximum value. The curve of a clipped signal is dependent upon the system in which the clipping takes place. In the analog domain, clipping effectively limits the signal to its maximum level. In the digital domain, clipping is similar to a numerical overflow, and so the polarity of the signal’s portion that exceeds the maximum level is negated.

Controller (Control-Change)

You can automate all Attack parameters using MIDI controller messages. This lets you create interesting sound transformations in real time. Controller data are directly created when you use the corresponding dials, and can be recorded in your sequencer program. You can also graphically create MIDI controller data in the respective Editor of your program. (Please read the manual of your host application for more information). A list of all available MIDI controllers and their functions can be found at the end of this manual.

Cutoff

See Filter Cutoff Frequency.

CV

CV is the abbreviation for control voltage. In analog synthesizers, control voltages are used to control sound parameters like pitch, cutoff frequency etc. To get a tremolo effect, for example, the output signal of a LFO must be routed to the CV input of one or more oscillators.

Decay

"Decay" describes the descent rate of an envelope once the Attack phase has reached its zenith and the envelope drops to the level defined for the Sustain value.

Envelope

An envelope is used to modulate a sound-shaping component within a given time frame so that the sound is changed in some manner. For instance, an envelope that modulates the cutoff frequency of a filter opens and closes this filter so that some of the signal's frequencies are filtered out. An envelope is started via a trigger, usually a fixed trigger. Normally, the trigger is a MIDI note. The classic envelope consists of four individually variable phases: Attack, Decay, Sustain and Release. This sequence is called an ADSR envelope. Attack, Decay and Release are time or slope values, and Sustain is a variable volume level. Once an incoming trigger is received, the envelope runs through the Attack and Decay phases until it reaches the programmed Sustain level. This level remains constant until the trigger is terminated. The envelope then initiates the Release phase until it reaches the programmed minimum value.

Filter

A filter is a component that allows some of a signal's frequencies to pass through it and dampens other frequencies. The most important aspect of a filter is the filter cutoff frequency. Filters generally come in four categories: low pass, high pass, band pass, and band stop. A low pass filter dampens all frequencies above the cutoff frequency. A high pass filter in turn dampens the frequencies below the cutoff. The band pass filter allows only those frequencies around the cutoff frequency to pass, while all others are dampened. A band stop filter does just the opposite, i.e. it dampens only the frequencies around the cutoff frequency. The most common type is the low pass filter.

Filter Cutoff Frequency

The filter cutoff frequency is a significant factor for filters. A low pass filter, for example, dampens the portion of the signal that lies above this frequency. Frequencies below this value are allowed to pass through without being processed.

Gate

The term "gate" has different meanings in a technical context. Like a real gate, it describes something that can be open or closed, or - to use a technical term - active or inactive. A gate in the sense of a device is a unit that prevents a signal from passing through under specific conditions. For example, in a noise gate a signal is cut off when its level falls above a predetermined threshold. Gate stands also for a control signal of analog synthesizer systems. A keyboard generates an active gate signal as long as a key is held down. When the key is released, the gate signal becomes inactive again. An envelope generator can use this signal for triggering purposes, and as a result a VCA unit can be controlled.

High Pass Filter

A high pass filter dampens all frequencies below its cutoff frequency. Frequencies above the cutoff point are not affected.

LFO

LFO is an acronym for low-frequency oscillator. The LFO generates a periodic oscillation at a low frequency and features variable waveshapes. Similar to an envelope, an LFO can be used to modulate a sound-shaping component.

Low Pass Filter

Synthesizers are often equipped with a low pass filter. It dampens all frequencies above its cutoff frequency. Frequencies below the cutoff point are not affected.

MIDI

The acronym MIDI stands for "musical instrument digital interface." It was developed in the early '80s so that diverse types of electronic musical instruments by different manufacturers could interact. At the time a communications standard for heterogeneous devices did not exist, so MIDI was a significant advance. It made it possible to link all devices with one another through simple, uniform connections. Essentially, this is how MIDI works: One sender is connected to one or several receivers. For instance, if you want to use a computer to play the Pulse, then the computer is the sender and the Pulse acts as the receiver. With a few exceptions, the majority of MIDI devices are equipped with two or three ports for this purpose: MIDI In, MIDI Out and in some cases MIDI Thru. The sender transfers data to the receiver via the MIDI Out jack. Data are sent via a cable to the receiver's MIDI In jack. MIDI Thru has a special function. It allows the sender to transmit to several receivers. It routes the incoming signal to the next device without modifying it. Another device is simply connected to this jack, thus creating a chain through which the sender can address a number of receivers. Of course it is desirable for the sender to be able to address each device individually. Consequently, there is a rule which is applied to ensure each device responds accordingly.

MIDI Channel

This is a very important element of most messages. A receiver can only respond to incoming messages if its receive channel is set to the same channel as the one the sender is using to transmit data. Subsequently, the sender can address specific receivers individually. MIDI Channels 1 through 16 are available for this purpose.

MIDI Clock

The MIDI Clock message determines the tempo of a piece of music. It serves to synchronize processes based on time.

Modulation

A modulation influences or changes a sound-shaping component via a modulation source. Modulation sources include envelopes, LFOs or MIDI messages. The modulation destination is a sound-shaping component such as a filter or a VCA.

Note On / Note Off

This is the most important MIDI message. It determines the pitch and velocity of every generated note. The time of arrival is simultaneously the start time of the note. Its pitch is derived from the note number, which lies between 0 and 127. The velocity lies between 1 and 127. A value of 0 for velocity is similar to "Note Off".

Panning

Panning is the process of changing the signal's position within the stereo panorama.

Pitch Bend

A MIDI message. Although pitch bend messages are similar in function to control change messages, they are a distinct type of message. The reason for this distinction is that the resolution of a pitch bend message is substantially higher than that of a conventional controller message. The ear is exceptionally sensitive to deviations in pitch, so the higher resolution is used because it relays pitch information more accurately.

Program Change

These are MIDI messages used to switch between sound programs. Program numbers 1 through 128 can be selected via program change messages.

Release

An envelope parameter. The term "release" describes the descent rate of an envelope to its minimum value after a trigger is terminated. The Release phase begins immediately after the trigger is terminated, regardless of the envelope's current status. For instance, the Release phase may be initiated during the Attack phase.

Resonance

Resonance is an important filter parameter. It emphasizes a narrow bandwidth around the filter cutoff frequency by amplifying these frequencies. This is one of the most popular methods of manipulating sounds. If you substantially increase the resonance, i.e, to a level where the filter begins self-oscillation, then it will generate a relatively clean sine waveform.

Sustain

An envelope parameter. The term "sustain" describes the level of an envelope that remains constant after it has run through the Attack and Decay phases. Sustain lasts until the trigger is terminated.

System Exclusive Data

System exclusive data allow access to the heart of a MIDI device. They enable access to data and functions that no other MIDI messages are able to address. "Exclusive" in this context means that these data pertain only to one device type or model. Every device has unique system exclusive data. The most common applications for SysEx data include transfer of entire memories and complete control of a device via a computer.

Trigger

A trigger is a signal that activates events. Trigger signals are very diverse. For instance, a MIDI note or an audio signal can be used as a trigger. The events a trigger can initiate are also very diverse. A common application for a trigger is its use to start an envelope.

VCA

VCA is the acronym for voltage-controlled amplifier. A VCA is a component that influences the volume level of a sound via a control voltage. This is often generated by an envelope or an LFO.

VCF

VCF is the acronym for voltage-controlled filter. It is a filter component that allows you to manipulate the filter parameters via control voltages.

Volume

The term describes a sound's output level.

MIDI Controller List

Please be sure to note:

Sounds 1 to 12 recognize MIDI controllers #12 to #59 on MIDI channels 1 to 12.

Sounds 13 to 24 recognize MIDI controllers #72 to #119 on MIDI channels 1 to 12.

Both Delay sections recognize MIDI controllers only on MIDI channel 16. Delay 1 recognizes MIDI controllers #12 to #18, and Delay 2 recognizes MIDI controllers #72 to #78.

Parameter	MIDI Controller Sound 1-12	MIDI Controller Sound 13-24	Parameter	MIDI Controller Sound 1-12	MIDI Controller Sound 13-24
Modwheel	1	1	Drive	39	99
Volume	7	7	Filter Env	40	100
Osc 1 Shape	12	72	Filter Env Select	41	101
Osc 1 Pitch	13	73	Filter Env Vel.	42	102
Osc 1 PitchEnv	14	74	Filter Mod Sync	43	103
Osc 1 PEnvVel	15	75	Filter Mod Speed	44	104
Osc 1 PEnv Select	16	76	Filter Mod Depth	45	105
Osc 1 FM	17	77	Output	46	106
Osc 1 FM Env	18	78	Amp Volume	47	107
Osc 1 FM EnvVel	19	79	Amp Velocity	48	108
Osc 1 FMEnv Select	20	80	Pan	49	109
Osc 2 Shape	21	81	Delay Mix	50	110
Osc 2 Pitch	22	82	XOR Group	51	111
Osc 2 PitchEnv	23	83	Env 1 Attack	52	112
Osc 2 PEnvVel	24	84	Env 1 Decay	53	113
Osc 2 PEnv Select	25	85	Env 1 Shape	54	114
Crack Speed	26	86	Env 1 Release	55	115
Crack Length	27	87	Env 2 Attack	56	116
Osc 1 Level	28	88	Env 2 Decay	57	117
RingMod Level	29	89	Env 2 Shape	58	118
Osc 2 Level	30	90	Env 2 Release	59	119
Osc 2 LevelEnv	31	91	Delay Time	12	72
Osc 2 LevelEnv Vel	33	93	Delay Sync	13	73
Osc 2 Level Env Sel.	34	94	Delay Feedback	14	74

Crack Level	35	95	Delay Spread	15	75
Filter Type	36	96	Delay Mod Depth	16	76
Cutoff	37	97	Delay Lo Cut	17	77
Resonance	38	98	Delay Hi Cut	18	78

Controller List for Novation RemoteSL

RemoteSL Controls	**Attack Parameter**
Page 1 Encoders 1...8	Oscillator 1 Pitch Sound 1...8
Page 1 Pots 1...8	Oscillator 1 FM Sound 1...8
Page 1 Sliders 1...8	Envelope 2 Decay Sound 1...8
Page 2 Encoders 1...8	Oscillator 2 Pitch Sound 1...8
Page 2 Pots 1...8	Oscillator 1 FM Envelope Sound 1...8
Page 2 Sliders 1...8	Envelope 1 Decay Sound 1...8
Page 3 Encoders 1...8	Filter Drive Sound 1...8
Page 3 Pots 1...8	Filter Cutoff Sound 1...8
Page 3 Sliders 1...8	Amplifier Volume Sound 1...8