attack

advanced drum synthesizer

User Manual

English

waldorf

Content

Foreword

Thank you for purchasing the Waldorf Attack Drum Synthesizer for iPad. Attack offers nearly all components to generate analog drum and percussion sounds, combined with a powerful pattern sequencer.

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. The extraordinary Phrase Vocoder generates voice effects based on text input.

We promise you a lot of fun while playing, working and composing with the Attack.

Your Waldorf Team

Hint

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Das Attack Development Team

Software:	Christian Bacaj, Stefan Stenzel
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Version:	1.1, November 2015

Please visit our website **www.waldorfmusic.com** Here you will find information of all our products.

We would like to thank

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Introduction

About this Manual

This manual was written to help you to become familiar with the Attack drum synthesizer. It will also aid experienced users with routine tasks.

To avoid confusion, the terminology in this manual is based on the Attack parameter names. You will find the various terms explained in a glossary at the end of this manual.

We also used a uniform set of symbols to show you topics of particular interest or significance. Important terms are highlighted in bold letters.

Symbols

⚠ **Caution – The comments that follow this symbol will help you avoid errors and malfunctions.**

Info – Additional information on a given topic.

✻ **Example –** Real-world examples to try out.

Highlighted Control Features and Parameters

All of the Attack's buttons, controls and parameters are highlighted in **bold** letters throughout the manual.

Example:

- Tap on the **Keytrack** switch.
- Tap and move the **Cutoff** dial.

The Attack's different parameter pages are illustrated in a depiction of the display.

Basic Operation

Audio Output

Use the volume buttons of your iPad to control the overall level. We recommend to use a suited Class Compliant Audio interface, a headphone or a connected amplifier / loudspeaker system to receive the best sound quality.

MIDI Input

Attack can be played via the internal keyboard. We recommend to connect a suited MIDI keyboard via a Core MIDI iPad interface. You can also send MIDI data via WIFI MIDI as well as a virtual MIDI connection

By using the iPad Camera Connection Kit you can also connect USB Class Compliant keyboards as the Waldorf Blofeld Keyboard or the Zarenbourg.

Preset Selection

Tap on the name of a preset in the center of the Attack Top section to open the Preset list. Here you can choose your favorite Song or drumset (Set) as well as a single sound. Tap onn the corresponding button (e.g. **Load Set**) and close the preset list by tapping on **Close**.

More about loading and saving of Songs and Sets can be found on page 10 of the manual.

Control Elements

To edit a sound you have to change its parameters. Therefore, Attack offers different types of control elements:

Dials

To set a value, tap on the dial, hold down and drag your finger up or down. Double tapping sets the parameter to its default value.

Dials with ring

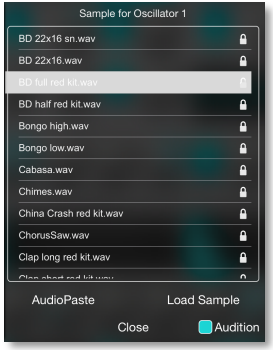
This dial controsl two parameter. One by the inner dial (turquoise color), one by the outer ring (orange color), e.g. for **Cutoff** and **Resonance**. To change the value, tap and hold on the corresponding dial (or on the corresponding point beside the parameter name) and move your finger up or down.

Selection Switches

Selection switches can be easily tapped. The corresponding symbol lits, when activated. Tapping on another symbol deactivates the first selected. Selection switches can be deactivated by tapping again.

Pop-up Menus

Tap on the corresponding parameter to open a pop-up menu, where you can choose the desired option by tapping. Tap on *Close* to close the pop-up menu.

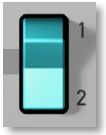


Blades

Tap on the desired blade to select and to play the corresponding sound.

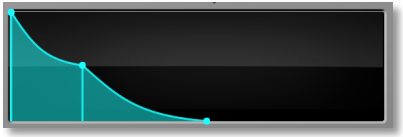
Switches

Switches can be simply tapped. The switch of the corresponding function switches to the respective position.



The Envelope Graphics

Tap on the corresponding graphic and slide it horizontal to change the envelope parameters Attack and Decay.



Pitchbend and Modulation Wheel

To change the value, tap on the wheel and drag it up or down. The Pitchbend wheel snaps automatically back into its center position as soon as you release your finger.

The Virtual Keyboard

Attack provides a virtual keyboard with 77 keys. Tap on a key to hear the corresponding note. The vertical position of the initial touchdown of the key determines the velocity.

Fader (on Mixer Menu Page)

Tap on the corresponding fader and hold it. Move your finger up or down to change the value.

The Control Functions

Overview of Functions

Attack consists of numerous sound-shaping components.

> Is this your first drum synthesizer? Are you curious about sound synthesis? If so, we recommend to read the chapter "Sound Synthesis Basics" in this manual.

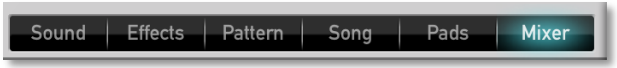
You should know that Attack consists of two different types of components for sound generation and sound shaping:

- Sound synthesis: oscillators, filter, amplifier, effects: These modules represent the audio signal flow. Sound generation actually occurs within the oscillators. It produces different waveforms or samples. The filter then shapes the sound by amplifying (boosting) or attenuating (dampening) certain frequencies. The amplifier is located at the end of the signal chain. It sets the overall volume of the signal and can add some saturation. Additionally, effects can be added to the signal.
- Modulators: The modulators are designed to manipulate or modulate the sound generating components to add dynamics to the sound. The Low-frequency Oscillator (LFO) is designed for periodic or recurring modulations while the envelopes are normally used for modulations that occur once.

The Top Section

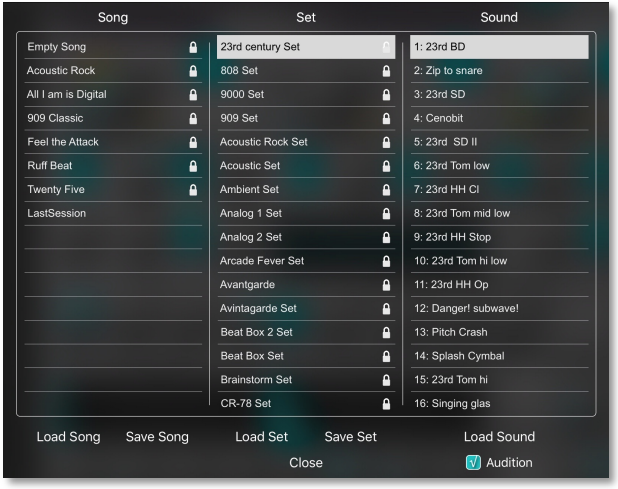
The Top section provides the global overview and includes the following options:

Menu Page Selection



Tap on the desired menu page (**Sound**, **Effects**, **Pattern, Song**, **Pads** oder **Mixer**) to open it. The actual menu page will be lit in turquoise color.

Preset Selection



Tap on the Preset name to open the preset pop-up menu. Here you find three columns for the selection of **Songs,** (Drum-)**Sets** and **Sounds**.

- Tap on the desired set name. The sound list will be updated automatically with the included sounds.
- Tap on the desired **Load** button to load the actual selection. Keep in mind, that the current preset (Song, Set or Sound) will be overwritten with the new one.
- If **Audition** is activated, a selected sound will be played automatically if selected.

Tap on **Close** at the bottom of the window to close it.

The Save Function for Songs and Sets

Tap on **Save Song** to save the current song. Tap on **Save Set** to save the current drum set. A new window opens:

- Tap on the name field to change the name before finally saving the Song or Set.
- Tap on **Save** to finally save the Song or Set.
- Tap on **Cancel** to cancel the process

You can transfer Songs and Sets from your iPad to your computer and vice versa by using the iTunes Folder. A deeper explanation of the iTunes Folder can be found in the manual appendix.

Keep in mind that preset with lock symbol can´t be overwritten.

The Tempo (BPM)

Tap on the tempo display to open a pop-up window. Here you can set the tempo for the step sequencer in BPM (beats per minute). By using the slider or the tap function, you can easily set up the desired tempo. You can also enter the tempo directly by double tapping the tempo value. Tap again on the tempo display to close the window.

The Pattern Display (PTN)

Attack´s pattern sequenzer can hold up to 999 pattern. Tap on the pattern display to open a window for selecting the desired pattern. Use the plus/minus buttons to select the pattern. Tap again on the pattern display to close the window. Alternatively, you can wipe with your finger over the pattern number to change the pattern.

The Transport Controls

Here you find all controls regarding playback and recording of patterns:

- Tap again and again on the **Chain** mode symbol to switch between the three playback modi *Pattern*, *Song* and *Loop*:
  - If *Pattern* mode ist selected, only the current pattern will be played in loop.

    
  - If *Song* mode is activated, all pattern in the song timeline (see "Song Menu Page") will be played in order.

    
  - If *Loop* mode is activated, the selected loop on the Song menu page (see corresponding chapter) will be played back in order.

    
- Tap on the **Record** button to activate the pattern recording via the internal keyboard, the Pads or via a connected MIDI keyboard. Tapping the button again activates the automatic

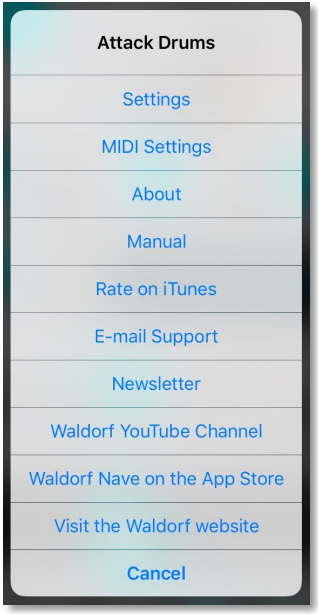
    

quantize function during recording, that moves every incoming note to the next step. To start recording, tap on the **Play** button.

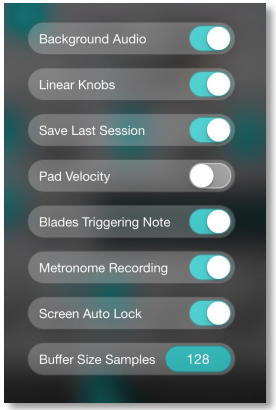
- Tap on the **Play** button to start the playback of the pattern sequencer or the playback of a song, when the **Chain** mode is activated. If the **Record** button is activated, the **Play** button starts the recording. Tap the **Play** button again ends the playback or the recording.

The Information Symbol

Tap on the information symbol to open a pop-up menu with additional functions.



- **Settings** opens a window with further options. To activate or deactivate an option, you need to wipe over the corresponding slider. The following options are available:



  - **Background Audio**: If activated, Attack reacts on incoming MIDI data, when the App is in background. Use **Background Audio** if you want to work with another App while Attack should keep on playing. Keep in mind: Running idle in background still consumes CPU and battery power. So, if you don't need it, switch it off.

Attack is compatible to the **Audio Bus** technology. It is available as input device. If you want to know more about Audio Bus, please visit: www.audiob.us

  - **Linear Knobs**: If activated, you can edit a dial value by tapping on it and move your finger up and down. If **Linear Knobs** is deactivated, you need to move around the dial to edit the corresponding value.
  - **Save Last Session** saves all data automatically (Song and Set with sounds) when the Attack App is closed. It is saved under the name "Last Session" and will be opend automatically, when the Attack App is started again.
  - **Pad Velocity** enables the velocity triggering of the Pads. The harder a pad is tapped, the higher is the generated velocity.
  - **Blades Triggering Note** activates the triggering of notes by tapping on the blades.
  - **Metronome Recording** activates a metronome click, which gives you a rhythmic orientation for manual pattern recording.
  - **Screen Auto Lock** overwrites the iOS settings for the iPad screen. If activated, Attack reacts as in the basic iOS settings. If deactivated, the Attack screen is active until the iPad battery is flat.

- **Buffer Size Samples** selects the buffer size (latency) in samples (64, 128, 256, 512, 1024). When triggering a MIDI note, a more less latency gives you a better realtime feeling. Keep in mind: the more less the buffer size, the higher the CPU load of your iPad. We recommend to use a setting of 64 samples only on the iPad Air 2 or newer.

- **MIDI Settings** opens a screen with different MIDI settings. Here you can e.g. define how Attack reacts on incoming MIDI data. A chart shows all connected MIDI hardware and its corresponding settings.

- The rows **Attack Drums A** and **Attack Drums B** repesents two MIDI data busses of Attack. It offers the usage of 2 x 16 MIDI channels and the dedicated playback of each of the 24 Attack sounds.

- The row **Network Session** refers to the internal MIDI bus of your iPad.

Keep in mind that based on the connected MIDI hardware different input and output columns will be shown. This depends on the configuration of your MIDI hardware.

- In the column **MIDI In to Sound** you can choose, how the corresponding row reacts on incoming MIDI notes. If deactivated, no MIDI notes will be received. If *1...16* is selected, incoming MIDI notes on channels 1 to 16 will be mapped to the sounds 1 to 16. Same ist for *17...24*. Here only the first 8 MIDI channels will be interpreted and mapped. The recommend setting is *1...16* for **Attack Drums A** and *17...24* for **Attack Drums B**, so that you can play all sounds of Attack independently via your MIDI hardware. A further setting is *selected*. Here, the MIDI input is only valid fort he selected sound.

  - In the column **Drums CH10** you can define, that all sound of a set will be mapped on your MIDI keyboard (if activated) or not (if deactivated). If activated, the first sound (mostly a bass drum) will be mapped to MIDI note C2, the second sound to C#2 and so on.
  - In the column **Clock In** you can choose, if Attack reacts on inconing MIDI clock data or not.
  - In the column **Clock Out** you can choose, if Attack sends MIDI clock data to the corresponding MIDI hardware or not.
  - In the column **MIDI Out** you can choose, if Attack sends MIDI notes to the corresponding MIDI hardware or not. The Attack sounds will be mapped automatically to the corresponding MIDI channels (same as for **MIDI In to Sound**).
- **About** opens a window with information about the Attack and the current version number. Tap on this window to close it.
- **Manual** opens a website for downloading the Attack PDF manuals in english and german. To go back to Attack you have to call up the Attack App again.
- **Rate on iTunes** opens the Apple AppStore product page for Attack. Here you can set a ranking for the Attack.
- **E-Mail Support** opens the standard e-mail editor with a predefined support e-mail. Here you can address your issue and send it directly to the Waldorf support e-mail adress.
- **Newsletter** opens a website where you can sign up for the Waldorf newsletter. To go back to Attack you have to call up the Attack App again.
- **Waldorf YouTube Channel** opens the Youtube channel of Waldorf Music. To go back to Attack you have to call up the Attack App again.
- **Waldorf on the App Store** opens the Apple AppStore product page for our Nave wavetable synthesizer. To go back to Attack you have to call up the Attack App again.
- **Visit Waldorf website** opens the Waldorf Music website. To go back to Attack you have to call up the Attack App again.
- **Cancel** closes the information window.

The Sound Menu Page

The user interface of the Attack is split into useful sections for easy access to the different parameters. Though the Attack is primarily designed for creating drum and percussion sounds, the setup is reminiscent of a synthesiser offering subtractive synthesis. Due to its flexible synthesis structure, the Attack is capable of creating classic synthesiser sounds, especially basses and leads.

To the left there are 24 blade buttons in the form of a stylized keyboard positioned vertically. Next appear parameter groups for both Oscillators, the Mixer, the Filter, the Amplifier, the Crack Modulator and settings for both Envelopes. At the bottom of the menu page you will find the pattern sequencer. Alternatively, you can overlay a virtual keyboard.

To get deeper into drum synthesis, we recommend to read the chapter "Sound Synthesis Basics" in the Appendix of this manual.

The Blade Buttons

By tapping on a blade button, you select the corresponding sound. A selected sound lits turquoise. You will also hear the corresponding sound, when tapping on a blade.

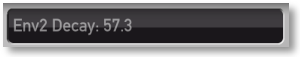
Sound Name Field

Here, the name of the current selected sound is displayed. Tap on the name field to open a keyboard for changing the sound name.

Init / Copy / Paste Buttons

Tap on **Init** to open a pop-up window. "Init Sound" sets all parameters of the actual sound to simple but useful settings. You can also enter a new name for the initialized sound. "Init all sounds" initializes all sounds of the current set. Tap on **Copy** to copy the current sound into buffer. After that, select a desired sound and tap on **Paste** to insert the copied sound. The current sound will be overwritten. To avoid accidently pasting, you have to confirm this process before.

Parameter Name and Value



Here, the name and the value of the current edited parameter is displayed.

The Mute and Solo Buttons

Tap on the **Mute** button (M) to mute the current selected sound. Tap on the **Solo** button (S) to solo the current selected sound. This mutes all other sounds of the drum set. Tap and hold the **Solo** button to deactivate the solo function for all sounds, that are in solo mode.

The 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 (Inner Dial)

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

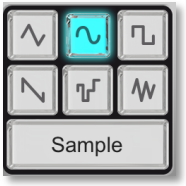
Detune (Outer Dial)

Detune also works with **Pitch** and fine-tunes 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.

- ✻ A low value of *±1* results in a slow and soft flanger effect.
- ✻ Mid-ranged settings of *±5* are perfect for pads and other fat sounding programs.
- ✻ High values of *±12* or above will give a strong detune that can be used for effect sounds.

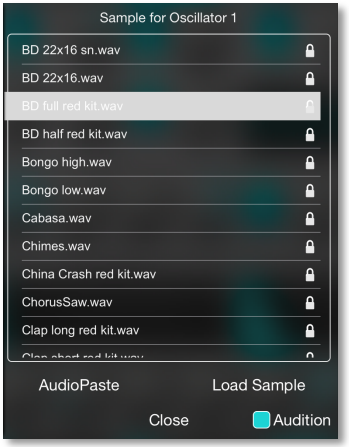
Shape Buttons (Waveform)



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 audio 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.**Sine**: Selects the sine waveform. It 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.
- **Sawtooth**: 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 Sample button doesn´t select an oscillator waveform; instead, you can choose a samples from a list for further programming. Tap on the sample button to open a pop-up menu:



- Here you find an alphabetic list with a couple of samples. Tap on a sample to select it. Tap on **Load Sample** to load it into an oscillator. To abort sample loading, tap on **Close**. The loaded sample will be shown with its name on the Sample button.
- If **Audition** is activated, every sample will be played, when selected.
- **Audio Paste** allows you to import audiofiles from another App on your iPad. If any audiofile is copied to the clipboard, it can be inserted to the end of the sample list by using the **Audio Paste** function.
- The **Delete** function allows you to delete a non-locked sample from the sample list. Wipe your finger from right to left over the sample name and confirm with *Delete*. To abort deleting, tap on any place in sample list.
- You can copy own samples to your iPad and use them within the sample list. Therefore, you need to upload samples via the iTunes folder. Read more in the Appendix of this manual.

The square, sine, triangle and sawtooth waveforms always start at full amplitude to create a necessary start click. This is characteristic for drum and percussion sounds. To avoid this click, simply raise the **Attack** value for Envelope 2.

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.

Keytrack

Determines, how much the pitch of the selected oscillator depends on the MIDI note number. When **Keytrack** is activated, it corresponds to a 1:1 scale, e.g. when an octave is played on the keyboard, the pitch changes for the same amount, based on the settings of **Pitch** and **Detune**

(Pitch) Envelope

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.

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

(Pitch) Velocity

Determines the amount of influence the selected envelope has on the pitch, based on key velocity. This parameter works similarly to the **Pitch Envelope** 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 Envelope** and **Pitch Velocity** 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 wave-

form 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) Envelope

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. Use this parameter to create time-dependent FM changes.

(FM) Velocity

Determines the amount of influence the selected envelope has on FM, based on key velocity. This parameter works similarly to the **FM Envelope** 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 Envelope** and **FM Velocity** 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.

The 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.

Oscillator 1

Volume of Oscillator 1.

Ring Mod

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.

Oscillator 2

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.

(Osc 2) Envelope

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 parame-

ter to change the volume of Oscillator 2 independently over time.

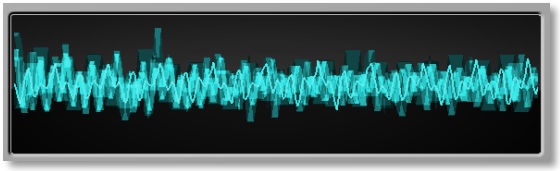
(Osc 2) Velocity

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

The overall modulation applied to Oscillator 2 level is calculated as the sum of both the **Envelope** and **Velocity** 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.

The Oscilloscope

This is a graphic realtime display for the generated audio signal. You can also tap in the oscilloscope display to trigger the actual sound.



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.

Keep in mind, that the Crack modulator superimposes its effect on all other mixer signals (Oscillator 1 and 2, Ring Modulator).

Crack

Fades in the Crack modulator.

Speed

Determines the frequency of the Crack modulator.

Length

Determines the number of modulations the Crack modulator creates.

The Filter Section

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

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.

Cutoff (Inner Dial)

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 Comb and S&H types.

- 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 the Comb type is selected, all frequencies near the cutoff setting will be boosted.
- When the S&H type is selected, the sample rate will be changed.

Resonance (Outer Dial)

Controls the emphasis of the frequencies around the cutoff point (exept for the Comb and S&H filter types). Use lower values to add 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.

Filter Types

The following filter types are available by tapping:

- **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 High Pass Filter**: This type is useful to thin out a sound's bass frequencies. This may also give inte-

resting 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.
- **Comb** differs from the other filter types greatly, because it doesn’t actually damp any part of the signal, but instead add a delayed version of the input signal to the output.

What is a Comb filter? A Comb filter is basically a very short delay that can be controlled in length and feedback. The delay time is so short that you can’t hear its individual taps but a colorization of the original signal created by peaks or holes in the frequency spectrum. The frequency of the colorization is set by the delay length, which is controlled in the Largo through **Cutoff**, while the amount of colorization is set by the Comb filter feedback, which is controlled in the Largo through **Resonance**.

- **S&H Filter**: This is no filter type indeed, but a special function to reduce the sample rate of the oscillator signal. **Cutoff** controls the frequency of the rate. **Resonance** controls the intensity of the hold function. A maximum setting is nearly similar to a high pass filter.

(Filter) Keytrack

Determines, if the cutoff frequency depends on the MIDI note number or not. The reference note for Keytrack is E3, note number 64. If activated, cutoff corresponds to a 1:1 scale, so e.g. when an octave is played on a keyboard the cutoff frequency changes by the same amount.

(Filter) Envelope

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.

(Filter) Velocity

Determines the amount of influence the selected envelope has on the cutoff frequency, based on key velocity. This parameter works similarly to the **Filter Envelope** 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 Envelope** and **Filter Velocity** 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.

Mod Speed

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.

Mod Depth

Determines the amount of filter frequency modulation by the LFO.

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

Sync Switch

Makes the low frequency oscillator (LFO) sync the filter modulation either to the key press or to the tempo of the host application. When activating **Sync**, **Mod Speed** can be set in musical note values (e.g. 1 bar, 1/4 or 1/8 triplet), based on the tempo setting of the Attack.

Drive

Determines the amount of saturation that is added to the signal. If set to *0.0%*, 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 drum sounds and effects.

- **PNP Transistor** generates a distortion based on a bipolar transistor.
- **Diode** generates a typical diode distortion.
- **Tube** simulates the asymmetric distortion of a tube circuit.
- **Hammer** is a sinusoidal waveshaper. It generates FM-like sounds that can be distorted very extremly.

The Envelope Section

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 trigger a note, 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 a trigger, **Release** defines the period of time it takes for the envelope to sink back to zero. Keep in mind, that the release phase only starts, when **Decay** is set to maximum. 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 your finger. Editing is simplified by the graphic changes you see in the corresponding function.

To edit, tap on the respective handle oft he **Attack** or **Decay** parameter and drag in the desired direction.

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 **Envelope** or **Velocity** parameter.

Envelope 2

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

The Amplifier Section

This section is found near the end of the Attack's signal routing, followed by the Effects section. Its main purpose is to set the volume of the sound. Additionally, the Amplifier section offers effect routing, panning and XOR groups.

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

EFX Select

Tap on one of the four EFX buttons to route the current sound to one of the four effect units.

Read more about Attack´s effects in the chapter "The Effects Menu Page".

Reverb

This parameter controls the volume ratio between the original signal and the reverb effect output. If set to *0.0*, the dry signal is sent to the output only so that no reverb effect can be heard. Higher values will increase the reverb effect signal. At maximum setting of *100*, the pure reverb effect signal will be heard.

Reverb can be used as follows:

- If none of the four EFX buttons is active, **Reverb** is used as send level for the reverb effect.
- If one of the four EFX buttons is active, **Reverb** has the same function as the **Reverb** knob on the Effects menu page for the corresponding effect unit. No matter, if you use the **Reverb** knob in the Amplifier section or on the Effects menu page.

- There is a special case, if effect units are chained. **Reverb** works as explained in the last case, except that the other **Reverb** knobs on the Effects menu page works self-sustaining for the chained effects.

Keep in mind, that the Reverb effect on the Effetcs menu page must be activated to can be heard. Read more about the Reverb effect in the chapter "The Effects Menu Page".

Pan (Panning)

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.

Volume

Sets the output volume of the selected sound.

Velocity

Specifies, how much volume will be affected by keyboard velocity. Use this feature to give more expression to the sound. With a setting of *0.0%*, 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.

XOR Groups

Assigns the selected instrument to one of the four XOR groups. When several instruments of the same XOR group receive trigger 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 hi-hat should be heard. This parameter can also be used to create monophonic synth sounds.

The Pattern Sequencer on the Sound Menu Page

The Attack offers an extensive Pattern sequencer for programming/recording rhythmic or melodic/harmonic phrases.

At the bottom of the Sound menu page, you will find the pattern line for the current sound.

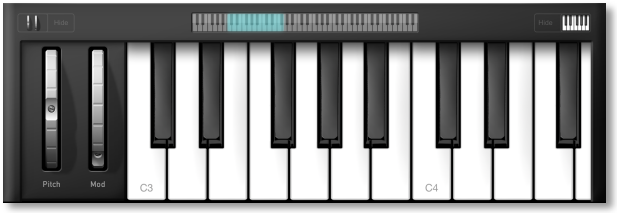
> Here you can enter only monophonic lines. The full functional pattern sequencer can be found on the Pattern menu page. Read more about it in the chapter "The Pattern Menu Page".

Tap on one of the 16 steps to activate (lits in turquoise) or deactivate it. You can easily enter sequences for every of the 24 sounds of the current drum set.

Use the buttons **Soft** (Velocity 32), **Mid** (Velocity 64) and **Hard** (Velocity 127) to pre-select a velocity for the entered steps.

The Virtual Keyboard

Attack offers a virtual keyboard with 77 keys. To activate it, tap on the keyboard symbol right to the pattern sequencer. Tap the symbol again to close the keyboard.



Left to the keyboard, you can find a pitch bend as well as a modulation wheel. You can hide both wheels by tapping on the button above it.

The scroll ribbon above the keyboard can be used to transpose the keys. The range goes from C2 to E8.

The Effects Menu Page

Attack offers an extensive effect section with four effect units, a Reverb effect and a Master compressor.

The Effect Units

Every effect unit (1 to 4) can generate 7 programmable effects.

To hear the effect of an effect unit, you need to route the corresponding sound to it. Therefore, use the **EFX Select** button on the Sound menu page. If no effect unit is selected, you can use the reverb and the compressor anyway.

Tap on the corresponding button to activate the desired effect unit. The button lits, when activated.

Every effect unit offers a **Reverb** dial to send the signal into the common reverb effect.

Use the **Routing** switch to send the signal from e.g. effect unit 1 into effect unit 2 and so on. In this case, the corresponding **Reverb** knobs can be used indepentantly to send the effect signal to the reverb.

✻ **Example for effect chaining –** On the Sound menu page, choose any EFX for the desired sound (for example EFX 2). Effect unit 2 creates a delay effect. Tap on the **Routing** switch to chain effect unit 2 with effect unit 3. Effect unit 3 creates a drive effect. Use the **Reverb** knob of effect unit 2 to send the signal directly from the delay effect into the reverb. The signal will also sent to effect unit 3 for distortion. You can also use the **Reverb** knob of effect unit 3 to send the signal with delay and distortion to the reverb.

The following effects are available for all four effect units:

Tap on the desired effect type name to select it. Based on the selected effect type, different effect parameters are offered.

Delay

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. Additionally, the delay can be modulated.

Delay

Sets the length of the delay tap in milliseconds or note values depending on the **Sync** setting.

Sync

Syncs the delay time to the Attack´s tempo. When activated, the **Delay** parameter can be set in muscial values.

Feedback

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

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

Rate

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

Depth

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.

Low 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 **High Cut** Parameter, the delay effect can thus be narrowed to a certain frequency range.

High 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 **Low Cut** Parameter, the delay effect can thus be narrowed to a certain frequency range.

Mix

This parameter controls the volume ratio between the original signal and the effect output. If set to *0.0%*, the dry signal is sent to the output only so that no effect can be heard. Higher values will increase the effect signal. At maximum setting of *100.0%*, the pure effect signal will be heard.

Equalizer

An equalizer is used to adjust the sound frequencies.

The equalizer contains three bands with the following parameters:

- **Gain** (for all three bands) raises or lowers the volume of the selected frequency
- **Frequency (Frq)** (for all three bands) sets the frequency in Hz (Hertz) respective kHz (Kilohertz) at which the tones will be affected.
- **Q (Filter Quality)** (only mid band) widens or shortens the frequency range.

Drive

Drive distortes the signal by using one of the four selectable drive curves.

Type

- **PNP Transistor** generates a distortion based on a bipolar transistor.
- **Diode** generates a typical diode distortion.
- **Tube** simulates the asymmetric distortion of a tube circuit.
- **Hammer** is a sinus-like wave shaper. Based on the incoming signal and the setting of the **Drive** parameter, FM-like sounds can be created.

Drive

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.

Tone

Dampens the higher frequencies of the drive effect.

Gain

Determines the output volume of the drive effect. Use **Gain** to adjust the overall volume for higher distortions.

Phaser

A phaser is generated by adding a second signal with a different phase. This generates an effect with equally spaced frequency peaks or throughs. The phase changing is controlled by a LFO.

Rate

Sets the LFO speed of the phaser effect.

Depth

Sets the modulation depth of the phaser effect.

Flanger

The flanger effect is very similar to the chorus effect, but features feedback circuitry to feed the generated signal back into the comb filter. This generates a deeper detuning and colorizes the signal. With extreme settings you can hear a whistling sound which is very characteristic of a flanger effect.

Rate

Sets the LFO speed of the flanger effect.

Feedback

Controls the feedback amount of the flanger signal.

Chorus

A chorus effect is generated by using comb filters that generate slightly detuned copies of the input signal and mix it into the output signal. The result sounds like an ensemble of several simultaneous sounds, like a choir as opposed to a single voice; hence the name chorus. The detuning is generated by an internal LFO that can be controlled in speed and depth.

Speed

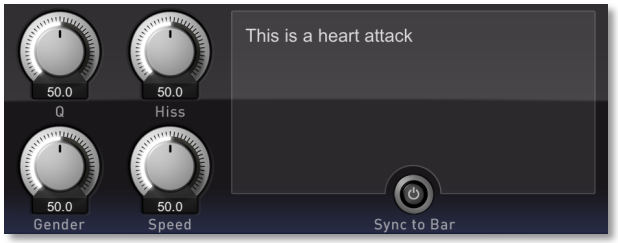
Sets the LFO speed of the chorus effect.

Depth

Sets the modulation depth of the chorus effect.

Phrase Vocoder

The phrase vocoder is an effect that works similar to a classic vocoder. Other than, the vocoder filter bank is not driven by a speech signal, but by written text. Every step of a sound, that uses the phrase vocoder, a new syllable is articulated.



Q

Determines the quality of the formant filter.

Hiss

Determines the level of the noisy components that are added to the signal. This is important for the understandability of some consonants.

Gender

Determines the detected gender of the phrase vocoder. Values from center to left let the signal sound male to monster-like. Values from center to right let the signal sound female, child-like and adds some "Mighty Mouse" effect.

Speed

Determines the speed of the consonant playback. For fast rap voices, lower settings are perfect, while higher settings improve the understandability.

Textfeld

Tap in the text field to open a window, where you can enter words and sentences. The phrase vocoder knows english language only, so in some cases you need to experiment with the notation. Correct spelling is secondary for having fun with this effect.

Sync to Bar

When activated, the phrase vocoder jumps to the next text line, when a new bar is played.

Set up a corresponding envelope with a higher decay, when using the phrase vocoder for a tonal sound. Set up a simple pattern with a step on every 1/8 note. Switch on **Sync to Bar** and enter the desired speech text. Now you should hear the phrase vocoder in action!

Reverb

The reverb effect is probably the most widely used effect in music production. Attack's reverb is an addition to the sound to make it more 3-dimensional and expressive.

To hear the reverb effect, you need to raise the **Reverb** dial on the Sound menu page (if no EFX button is active). If any effect unit is active, you can use the **Reverb** knob in this unit to control the reverb send level.

Predelay

Determines the delay between the direct sound and the reverb effect output. Lower settings connect the reverb more to the original signal while higher settings separate the effect signal to produce a more spatious sound.

Time

Determines the reverb time. Low settings simulate a small room while higher settings simulate a big hall or church.

Color

Determines the spectral colorization of the reverb sound. Negative values dampen the higher frequencies while positive settings dampen the deeper frequencies.

Output

Determines the output level of the reverb effect.

Compressor

The compressor compensates differences in the level of a sound signal by automatically raising lower levels.

The compressor is a common effect that has influence on all sounds of a drum set.

Threshold

Determines, to which signal level the compressor will be working. If the signal falls below the threshold, the amplification takes place.

Ratio

Ratio defines the maximum amplification of the signal in dB.

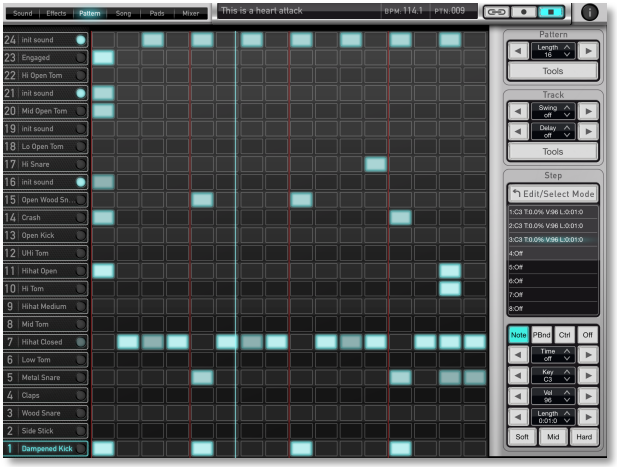
Attack

Determines the timed operation of the compressor. Short attack times cuts the signal transients, so that the signal sounds louder but also flatter. Longer attack times let pass the attack phase of the signal and emphasize the impulsivity.

Release

Determines the return time of the compression effect. The shorter **Release**, the louder the audio signal will sound. At once, the typical compressor pump effect could appear.

The Pattern Menu Page



On this menu page, you find the extensive pattern sequencer with 24 tracks. You can program rhythmic sequences as well as polyphonic chord steps. Besides this, every step can be edited in many ways.

Changes that are made on the Pattern menu page affects also the pattern line on the Sound menu page and vice versa. Extensive editing and polyphonic chord creation is only possible at the Pattern menu page.

Tap and wipe two fingers together from right to left or from left to right to choose the next/last pattern.

Basic Step Programming

The pattern editor contains 16 step fields for each of the 24 sounds. The grid is divided into 16 notes for one bar. The quarter positions are marked for better recognition.

- Tap on an empty step to create a note step. Created steps will be shown as colored rectangles. Tapping on a step field with an existing step will delete this step.
- Tap and hold on a step and move your finger up and down or left and right to create several steps at once. This is a perfect way to create random patterns.

Editing of Patterns

All steps for the 24 sounds are called "Pattern". The pattern field in the rightmost corner offers some further options for the current pattern:

- **Length**: Here you can change the length of a pattern. The minimum length is 4 steps, the maximum 16 steps (default settings). Tap on one of the arrow buttons to change the length by one step. You can also change the desired length by tapping on the **Length** value and move your finger up or down.
- Tap on the **Tools** button to open a pop-up menu with further options:

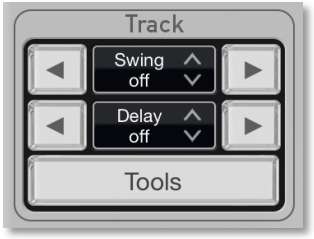
Keep in mind: As long as the **Tools** pop-up menu is active, only the Tools options acan be used for editing.

  - **Previous Pattern** switches back to the last pattern.
  - **Next Pattern** switches to the next Pattern.

- **Copy** copies the current pattern into the memory buffer. Use this function to copy the data from one pattern into another.
- **Paste** pastes the content of the memory buffer into the current pattern. All exiting data is overwritten. To avoid accidently deleting, you have to confirm this process before.
- **Clear** clears all steps of the current pattern. To avoid accidently deleting, you have to confirm this process before.
- **Quantize** quantizes all recorded steps of the current pattern to the next semiquaver step.
- **Double Time** doubles the speed of all steps and copies the steps to avoid gaps.
- **Half Time** halves the speed of all steps and deletes corresponding steps to avoid overlaps.
- **Rotate Left** moves all steps of the pattern to the left by one step.
- **Rotate Right** moves all steps of the pattern to the right by one step.
- **Transpose +1** transponses all steps of the pattern up by one semitone.
- **Transpose -1** transponses all steps of the pattern down by one semitone.
- **Humanize** varies the velocity of all steps to random values near the actual velocity. Tapping again and again creates higher aberrations.
- **Ramp** creates a ramp-like velocity interpolation between the first and the last step. Use this function to create rising or falling gradients. Tapping again and again intensifies the ramp effect.
- **Fade In** creates a fade-in for the velocities of all steps. Tapping again and again intensifies the fade-in effect.
- **Fade Out** creates a fade-out for the velocities of all steps. Tapping again and again intensifies the fade-out effect.

Editing of Tracks

All steps for one sound are called a "Track". The track field in the right section offers some further options for the current selected track:



- **Swing** lets you delay the triggering of the note steps to achieve a "swing" or "shuffle" playback. When **Swing** is set to *50%,* this parameter will not influence the pattern. Higher settings will raise the "swing" when playing back the pattern. Lower values moves the timing forward.
- **Delay** delays the playback of the current track based on the choosen value.
- Tap on the **Tools** button to open a pop-up menu with further options:
  - **Copy** copies the current track into the memory buffer. Use this function to copy the data from one track into another.
  - **Paste** pastes the content of the memory buffer into the current track. All exiting track data is overwritten. To avoid accidently deleting, you have to confirm this process before.
  - **Clear** clears all steps of the current track. To avoid accidently deleting, you have to confirm this process before.
  - **Quantize** quantizes all recorded steps of the current track to the next semiquaver step.
  - **Double Time** doubles the speed of the steps oft he current track and copies the steps to avoid gaps.
  - **Half Time** halves the speed of the steps of the current track and deletes corresponding steps to avoid overlaps.
  - **Transpose +1** transponses all steps of the current track up by one semitone.
  - **Transpose -1** transponses all steps of the current track down by one semitone.
  - **Rotate Left** moves all steps of the current track to the left by one step.
  - **Rotate Right** moves all steps of the current track to the right by one step.
  - **Humanize** varies the velocity of the steps of the current track to random values near the actual velocity. Tapping again and again creates higher aberrations.
  - **Ramp** creates a ramp-like velocity interpolation between the first and the last step of the current track. Use this function to create rising or falling

gradients. Tapping again and again intensifies the ramp effect.

- **Fade In** creates a fade-in for the velocities of all steps of the current track. Tapping again and again intensifies the fade-in effect.
- **Fade Out** creates a fade-out for the velocities of all steps of the current track. Tapping again and again intensifies the fade-out effect.

Fast Edit Mode

Here you will find three predifined velocity buttons for step programming:

- **Soft** creates steps with a velocity of 32.
- **Mid** creates steps with a velocity of 64.
- **Hard** creates steps with a velocity of 127.

If you are in **Edit/Select Mode**, tap on one of the three buttons to change into the **Fast Edit Mode**.

Edit/Select Mode

To enter this mode, simply tap on the area around the Waldorf logo in the right section of the Pattern menu page.

Here you can choose, if you want to enter/delete steps into the grid (Edit Mode) or if you want to select steps for further editing (Select Mode).

Tap on the Edit/Select Mode button, to switch between the two modes. The button lits, if the Select mode is active.

Editing of Steps

The smallest object of a pattern/track is a step. A step can be a trigger event for a note, a controller or a chord. The step section in the right section offers some further options.

To select and edit steps, you need to change from Edit mode to Select mode. Tap on the **Edit/Select Mode** button, until it lits.

Keep in mind, that all changes made in the Step section affects only the selected step respectively new created steps. Other existing steps won´t be changed.

- **Step List**: Here you can define, how much trigger notes will be created for a step. In most cases, you need to create a single note for triggering a drum sound. But you can built up chords by adding up to 7 notes per step. Tap on the desired entry in the list to select it.
- **Step Data** button (Note, PBend, Ctrl, Off): Here you can define, which trigger data is created for a step. You can choose between a Note, pitch bend data (Pbend), MIDI Controller data (Ctrl) or no data (Off). Based on the choosen type, different parameters are available:

  For **Note**:

  - **Time** moves the timing of a note forwards or backwards.
  - **Key** determines the pitch of a note. Use this function to define different pitches for notes in the step list. For example, you can choose C3 for note 1, E3 for note 2 and G3 for note 3 to et a C major chord.
  - **Vel** (Velocity) determines the velocity for a step.
  - **Length** determines the length of a note.

  For **PBnd**:

  - **Time** moves the timing of a pitch bend event forwards or backwards.

- **PitchBend** determines the desired value for the pitch bend event.

For **Ctrl**:

- **Time** moves the timing of a controller event forwards or backwards.
- **ControlNr.** defines the desired MIDI controller number.
- **Value** defines the value for the choosen MIDI controller.

For **Off** there is no parameter. An empty step will created.

- The buttons **Soft**, **Mid** and **Hard** select a pre-defined velocity. Tapping on one of the buttons switches automatically to the Fast Edit Mode.

Drum Pattern Example

- ✻ First of all, load the drum set preset "Fat Set".
- ✻ On the Pattern menu page, set some note steps for "Klick Klick", "Comb Snare" and "HiHat Closed".
- ✻ Activate the Select mode.
- ✻ Select an unaccented hi-hat step and set the velocity (Vel) to a medium value. You can also do so for other unaccented steps.
- ✻ Make sure that no step ist selected!
- ✻ Define a chord in the step editor. Use the first three notes and the key settings of C3, E3 and G3.
- ✻ Note length should be set to 1.
- ✻ Create a step on position 1 for the sound "Saw Chords".
- ✻ By playing the pattern, a C major chord will be triggered.

The Song Menu Page

Here you can combine your patterns to a song.

Basic Functions

In the upper row you find all patterns that are used for your song (Song Timeline). The lower row shows all available patterns (Pattern Bay).

- Drag and drop a pattern from the Pattern Bay section to the desired position of the Song Timeline section. Alternatively, you can add a pattern by tripple tapping.
- To exchange a pattern within the Song Timeline section, drag it to the desired new position.
- Wipe from left to right or vice versa to scroll through the patterns of your song timeline.
- Delete a pattern from the song timeline by easily drag it out of the timeline list.
- You can select one or more patterns in the Song Timeline section by tapping on it. Selected patterns lit in turquoise.
- The loop function (see next page) is for looping defined song pattern.

Playback of a Song

In the Top section, activate the **Chain** function in the transport panel. Start playback. All patterns of the song timeline will be played one after another.

The Loop Function

The loop selection allows you to set a loop start and loop end pattern. This is usesful to check a pattern sequence within the song timeline.

- Loop Start selects the first pattern where the loop will be started.
- Loop End selects the last pattern where the loop will be ended.

The loop duration is shown within the song timeline as coloured bar.

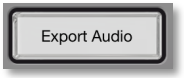
- The Loop mode in the transport control actives the loop.

Function Buttons

The **Function** buttons give you further functions to edit your song.

- Tap on **Clear** to remove all patterns from the Song Timeline section. To avoid accidently deleting, you have to confirm this process before.
- **Cut** cuts out all selected patterns from the Song Timeline section and copies them into the memory buffer.
- **Copy** copies all selected patterns from the Song Timeline section into the memory buffer.
- **Paste** pastes the content of the memory buffer into the song timeline. If no pattern is selected, the content will be copied to the beginning of the song. If one or more patterns are selected, the content will be copied after the last selected pattern.
- Tap on the **Clear All** button to remove all patterns from the Pattern Bay section. To avoid accidently deleting, you have to confirm this process before.

Exporting a Song as Audio file



Tap on the **Export Audio** button to render the current song data as WAV audio file and save it on your iPad.

Before the render process ist started, you can enter a desired song name. After that, tap on "Export Audio" to start the render process.

You can either choose the *AudioCopy* option. Here, the song will also be exported and but additionally copied to your iPad´s clipboard. Use this function, if you want to use your song as sample for one of the oscillators or for using it into another iPad App.

Tap on *Cancel* to cancel the audio export process.

Exported songs can be transfered to your computer via iTunes folder. Read more about this in the chapter "Knowledge about the iTunes Folder" in the Appendix of this manual.

The Pads Menu Page

On this page, all 24 sounds of the current drum set are displayed as playable pads. Use it to jam or to record patterns. In the **Settings** (see page 12) you can define, if the pads will be react on velocity.

The Mixer Menu Page

On this page, all 24 sounds of a drum set will be shown as mixer channels. You can directly edit e.g. volume, panning and effect routing.

Basic Functions

- To trigger a sound, tap on the corresponding blade.
- To change the volume of a sound, tap on the corresponding fader and move it up or down. Tap on a desired position of a fader channel to abruptly change the volume.
- To change the overall volume for all channels, tap on the master fader and move it up or down.
- To change the panning of a sound, tap on the corresponding **Pan** dial and move your finger up or down.
- To mute a sound, tap on the desired **Mute** button (M). Its lit red. Tap on the **Mute** button again to switch mute off.
- To solo a sound, tap on the desired **Solo** button (S). It lits turquoise. All other sounds will be muted automatically. Tap on the **Solo** button again to switch solo mode off. You can switch more than one sound to solo, if desired. Tap and hold one **Solo** button to deactivate the solo function for all sounds, that are in solo mode.
- To route a sound to a XOR group, tap on the corresponding XOR button, until the desired group number occurs (e.g. 2).

- To route a sound into an EFX unit, tap the corresponding EFX button, until the desired FX unit number occurs (e.g. 3).

Sound Synthesis Basics

Oscillators Introduction

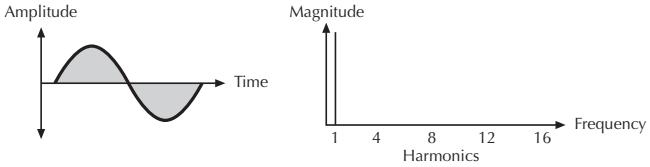
The oscillator is the first building block of a synthesizer. It delivers the signal that is transformed by all other components of the synthesizer. In the early days of electronic synthesis, engineers found that most real acoustic instrument waveforms can be reproduced by using abstracted electronic versions of these waveforms. They weren't the first who came to that conclusion, but they were the first in recreating them electronically and building them into a machine that could be used commercially. What they implemented into his synthesizer were the still well-known waveforms sawtooth and square. For sure, this is only a minimal selection of the endless variety of waveforms, but Attack gives you exactly these waveforms at hand.

Now, you probably know how these waveforms look and sound, but the following chapter gives you a short introduction into the deeper structure of these waveforms.

Let's start with the most basic one.

The Sine Wave

The Sine Wave is the purest tone that can be generated. It consists only of one harmonic, the fundamental, and has no overtones. The following picture shows the sine wave and its frequency representation:



The Sine Wave

There is no acoustic music instrument that generates a pure sine wave; the only instrument that comes close to it is the pitch fork. Therefore, the sine wave sounds a little artificial to the ear. However, the sine wave can be an interesting oscillator waveform to emphasize a certain harmonic while other oscillators are playing more complex waveforms, or as FM source for frequency modulation.

The sine wave is the most basic building block of each waveform. Any waveform can be broken down to several or many sine waves that are arranged with different frequencies and magnitudes. These sine waves are called *partials*. In most waveforms, the partial with the lowest

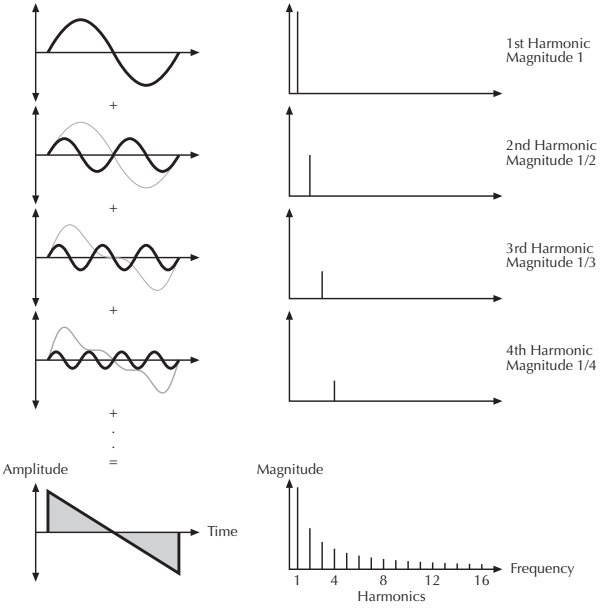
frequency is dominant, meaning that this partial is used by the ear to determine the pitch of the tone. This partial is called *fundamental*. All other partials are called *overtones*. So, the second partial is the first overtone.

Cyclic waveforms such as the waveforms in the Largo only feature sine waves in integer frequency ratios to the fundamental like double frequency, triple frequency and so on. Those partials are called *harmonics* because their frequency is a harmonic multiple of the fundamental.

Confusing? Let's generalize it to cyclic waveforms: a cyclic waveform like sawtooth, square etc. only consists of harmonic partials. The harmonic with the lowest frequency is dominant and therefore called fundamental. All other harmonics are called overtones.

The Sawtooth Wave

The Sawtooth wave is the most popular synthesizer waveform. It consists of all harmonics in which the magnitude of each harmonic descends by the factor of its position. This means that the first harmonic (the fundamental) has full magnitude, the second harmonic has half magnitude, the third harmonic has a third magnitude and so on. The following picture shows how the individual harmonics build up the sawtooth wave:



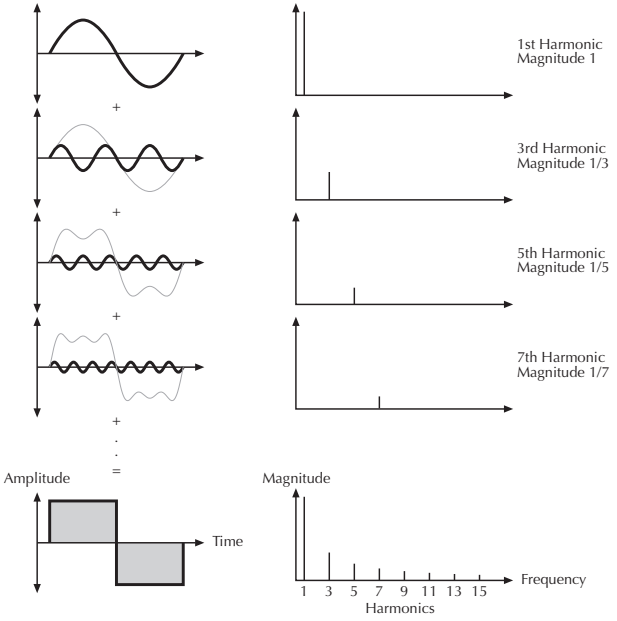
Additive components of the Sawtooth wave

The sawtooth wave was thought as an abstraction of the timbre of string and brass instruments. You can easily understand that when you think of a violin. Imagine a bow pulling the string slightly into one direction. At one

point, the string abruptly comes off the bow and swings back to its original position. The bow is still moved and so it catches the string again and the procedure is repeated. The result is a waveform that looks like a sawtooth. The same is true for a brass instrument. The string in this case are the lips while the bow is the air. The lips are moved by the air to a certain extent and abruptly move back to their original position.

The Square Wave

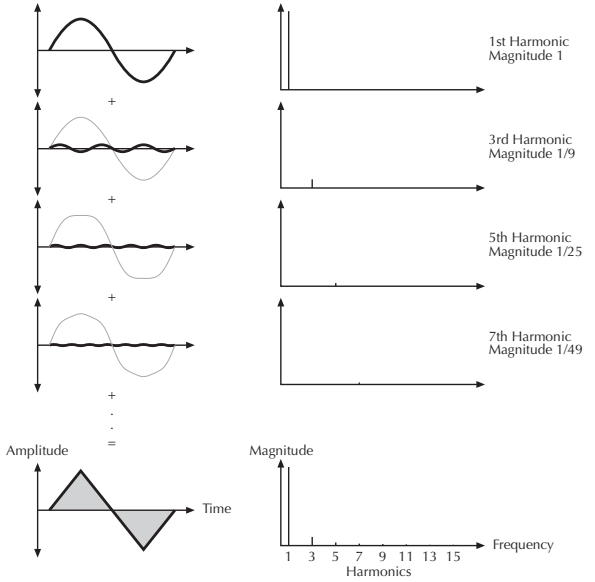
The Square Wave is a pulse waveform with 50% pulse width. This means that the positive part of the waveform has the same length as the negative part. The pulse waveform can have other pulse widths as you will read later. For now, we'll talk about the square wave as a unique waveform. The square wave consists of all odd harmonics in which the magnitude of each harmonic descends by the factor of its position. This means that the first harmonic has full magnitude, the third harmonic has a third magnitude, the fifth harmonic has a fifth magnitude and so on. The following picture shows how the individual harmonics build up the pulse wave:



Additive components of the square wave with 50% pulse width

The Triangle Wave

The Triangle Wave is very similar to the square wave. It is composed of the same harmonics as the square wave, but with different magnitude ratios. The magnitude of each harmonic is divided by the power of its number. This means that the third harmonic's magnitude is a ninth, the fifth harmonic is a twenty-fifth and so on. The following illustration shows the harmonic content:



Additive components of the Triangle wave

The reason why the triangle wave is so popular in classic synthesizers: It could act as a suboscillator wave, to emphasize certain frequencies or to frequency modulate other oscillators.

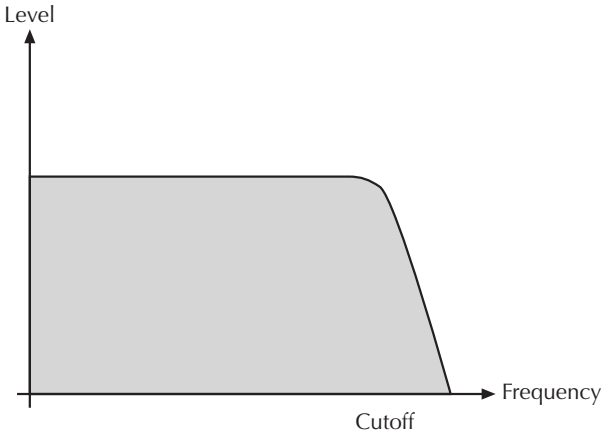
The triangle wave sounds like a woodwind instrument, i.e. a clarinet. It can also be used for mallet instruments like vibraphone, xylophone etc.

Introduction Filter

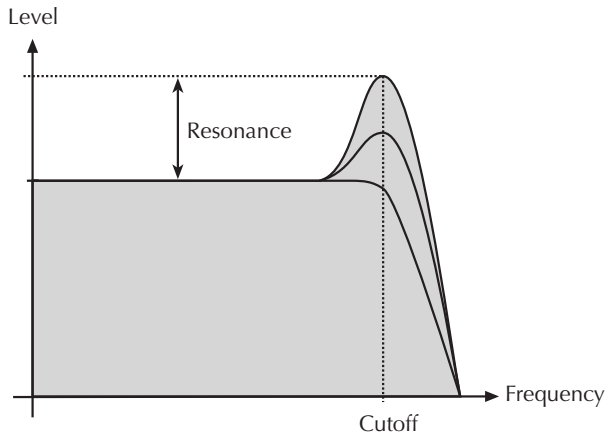
Once the audio signal leaves the oscillator, it is sent to the filter. The filter is a component that has significant influence on Attack's sound characteristics.

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 above a specified cutoff frequency. Frequencies below this threshold are hardly affected. The frequency below the cutoff point is called the pass band range, the frequencies above are called the stop band range. Attack's filter dampens frequencies in the stop band with a certain slope. The following picture shows the basic principle of a low pass filter:



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:



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

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 5000 Hz 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 und 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 und 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 540 Hz, the other oscillating at 800 Hz. 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.

We need more cowbell!

Appendix

Knowledge about the iTunes Folder

This "Folder" is for exchanging data between an iPad App and a computer, where the iTunes software is installed and the corresponding iPad is connected to.

This enables you to save the data, which is produced by Attack (Songs, Sets and Wave files etc.) to your computer.

To exchange data from Attack to a computer or vice versa please proceed as follows:

- Connect your iPad to your computer and start your iTunes application.
- In iTunes, select your iPad in the devices list.
- Click the "Apps" button in the left section and navigate to "File Sharing". Click on the Attack App symbol.
- The documents list in the right area gives you an overview of all available data from Attack, e.g. saved songs (songname.attacksong), saved drum sets (drumsetname.set) and exported audio songs (songname.wav). These files can be moved via drag & drop to a desired folder on your computer. In addition to that you can use the "Save to.. " function at the bottom of the list. If you select a file and click on the "Delete" key, it will be deleted from yor iPad.
- You can also copy files from your computer to the documents list, e.g. WAV files for sample playback. These files can be moved via drag & drop to the documents list. In addition to that, you can use the "Add..." function at the bottom of the list. All data is transfered automatically to your iPad and can be used in Attack.

By the way, data in your iTunes Folder can be easily renamed by clicking on the corresponding file in the list and enter a desired name.

Inter-App Audio with the Attack

Inter-App Audio can connect two Apps directly. The Attack supports this function.

For example, you can use a sequencer App as Steinberg Cubasis to record the audio output of Attack directly into an audio track. Please read the corresponding documentation, how to set up an Inter-App Audio connection.

If Inter-App Audio is active, the Attack shows a transport display which allows you to control basic transport functions of the connected App.

Here you can start playback as well as record of the master App. You can also skip back the song. A timepiece is also shown within the transport display. By tapping on the App symbol of your master App you can easily switch to it.

The Inter-App transport control can be folded, if desired.

Attack is also compatible to **AudioBus** technology. Attack runs as "Input" position. If you want to know more about this technology, please visit: www.audiob.us

Glossary

Amount

Describes to which extent a modulation influences a given parameter.

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.

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 damped.

Clipping

Clipping is a sort of distortion that occurs when a signal exceeds its maximum value. The curve of a clipped signal is dependent of the system where 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 part above the maximum level is negated.

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 programed Sustain level. This level remains constant until the trigger is terminated. The envelope then initiates the Release phase until it reaches the 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, 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 dampens the portion of the signal that lies above this frequency. Frequencies below this value are allowed to pass through without being processed.

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. A low pass filter 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 1980s 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 MIDI keyboard to play Attack, then the keyboard is the sender and Attack acts as the receiver.

With a few exceptions, the majority of MIDI hardware 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 Kanal

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 sets 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 sound-shaping component such as a wave position or filter.

Note on / Note off

This is the most important MIDI message. It sets 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

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

Pitch Bend

Pitch Bend is a MIDI message. Although pitchbend 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 pitchbend message is substantially higher than that of a conventional Controller message. The human ear is exceptionally sensitive to deviations in pitch, so the higher resolution is used because it relays pitchbend information more accurately.

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 imme-

diately 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 amplifing 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.

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.

USB

The Universal Serial Bus (USB) is a serial bus system to connect a computer with an external device. USB equipped devices can be plugged together while active. The recognition is made automatically..

Volume

The term describes a sound's output level.

Product Support

If you have any questions about your Waldorf product, feel free to contact us via one of the four options listed below:

① Send us an email message. This is the most efficient and fastest way to contact us. Your questions will be forwarded immediately to the resident expert and you will quickly receive an answer.

support@waldorfmusic.de

② Send us a letter. It will take a bit longer, but it is just as dependable as an email.

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