Eurorack Owners Manual

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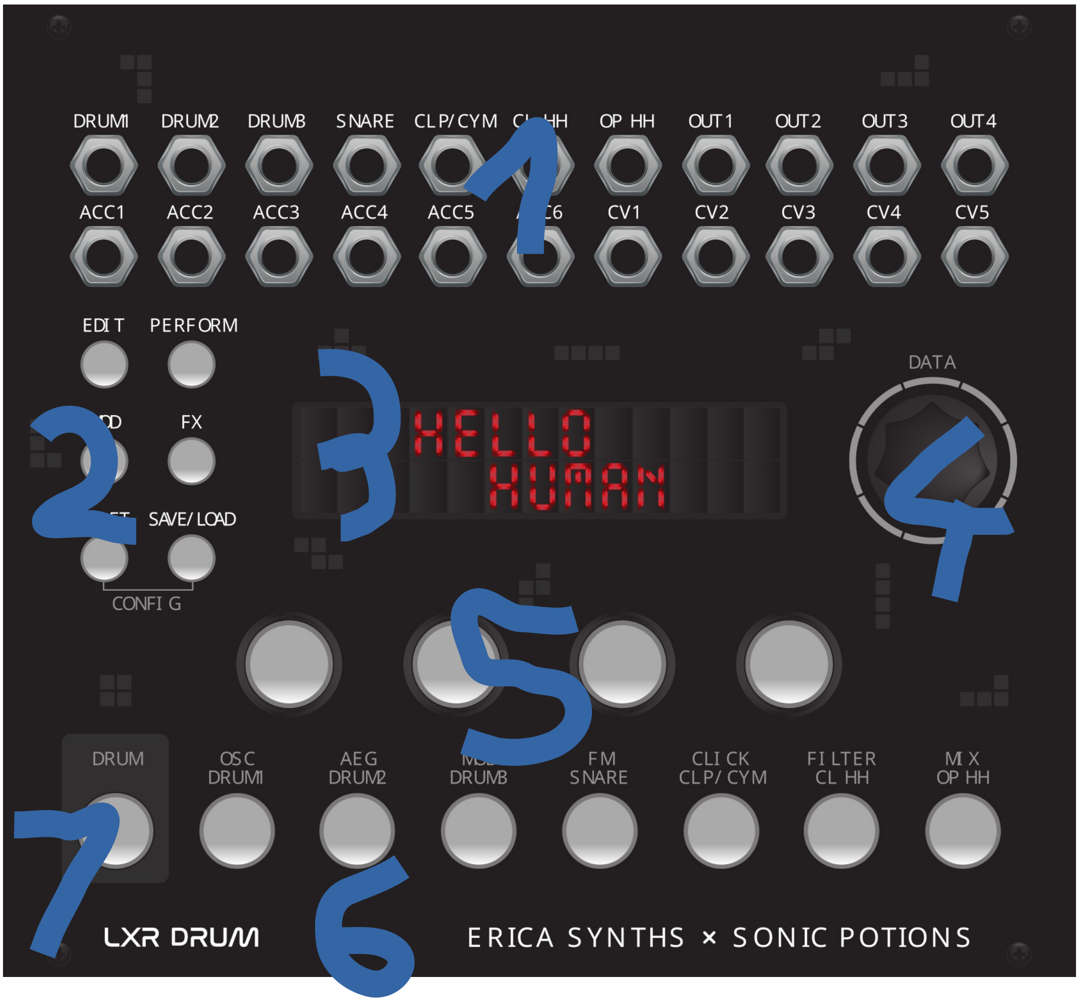
1. Intro

Thank you for purchasing the Erica Synths X Sonic Potions LXR-02 eurorack module!

The LXR is a full fledged digital drum generator. Its sound engine provides 6 different instruments, each with over 30 parameters to tweak. It can produce a wide variety of sounds, ranging from classic analogue emulations to crunchy digital mayhem.

2. Overview

In this first chapter we will focus on the physical appearance of the LXR, describing the front panel controls as well as the connection jacks. Further, the basic menu navigation is explained.



1) IO Jacks
In the first row, we have 7 trigger in jacks (5V) to trigger the voices and the 4 audio outs.
The second row has 6 accent inputs (5V) and 5 freely assignable CV (-5V to 5V) inputs.

2) MENU Buttons
Select the operating mode of the module

3) Display
The display is used to show parameter values from the selected menu page.

4) Encoder
Navigate through the menu.

5) Endless Potentiometers
The 4 pots are used to edit the values shown in the display above.

6) SELECT buttons
Switch voices and select sound parameter sections.

7) DRUM Button
Used to switch the active voice together with the select buttons.

2.1. Menu Navigation

The upper row of the display will show the short name of the parameter. The bottom row shows the value of the parameter. You can change the value using one of the 4 knobs below the display.

- The encoder selects a menu parameter. Capital letters highlight the selected parameter. In our example picture above, the 'RES' parameter is selected.
- If there are more than 4 entries in the current menu, the next page will be shown if you scroll over the screen boundary. A '>', '*' or '<' sign in the right upper corner indicates if there is more than 1 page available and which page is active. The pages of a menu can be toggled by pressing the MENU button again.

If you push the encoder the detail page is shown

On the detail page you can see the full name of the parameter and change its value using the encoder. This is good for fine adjustments, where the knob control is too coarse. By pushing the encoder again, you return to the normal menu mode.

3. Memory management

3.1. SD Card

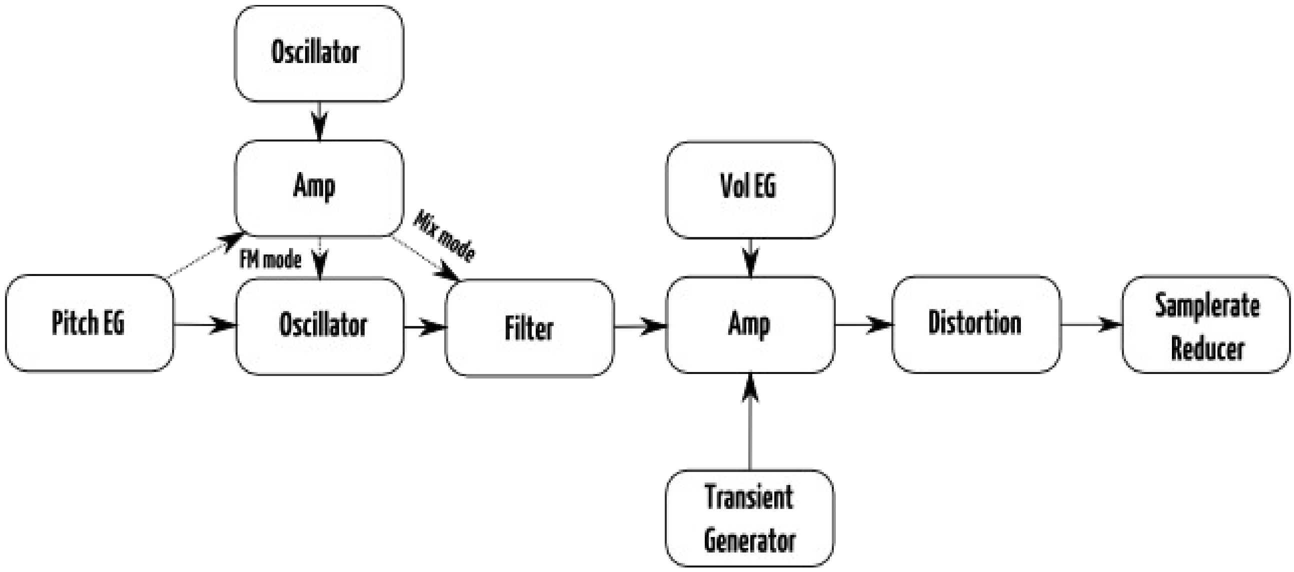
The memory card is used as non volatile storage. Without a memory card you are not able to save your work!

3.2. Kits

A kit contains the instrument data for the 6 Voices. All synthesis parameters are stored here. Kits are stored as .SND binary files inside the Project00 folder on the SD card.

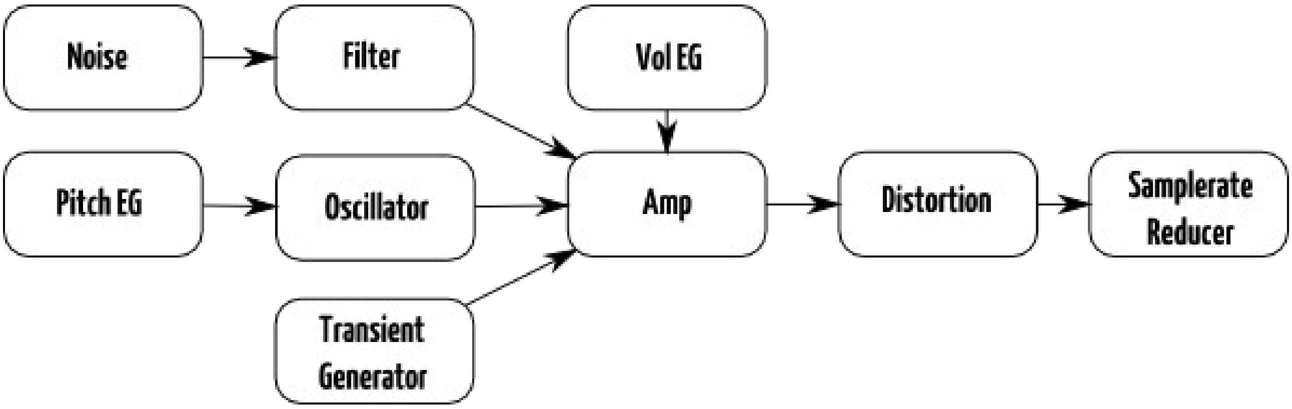
4. Voices

The LXR offers 6 voices. They are optimized for different kinds of drum sounds. The type of a voice can not be changed. There are 3 drum voices, a subtractive clap/snare voice, a FM percussion voice and a hi-hat voice. In this chapter a diagram of the synthesis structure is shown for each voice type, then the parameters are explained. Let's take a closer look at the voice types.



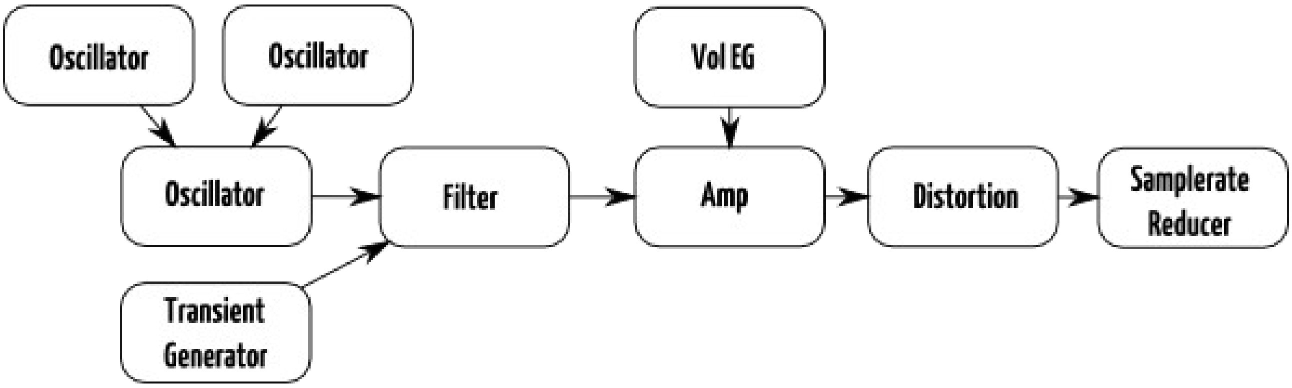
4.1. Snare voice (voice 4)

This voice is good for snare-drum and clap sounds. A noise source and a pitched oscillator can be mixed. Only the noise source is routed trough the filter. There is no FM capability on this voice.



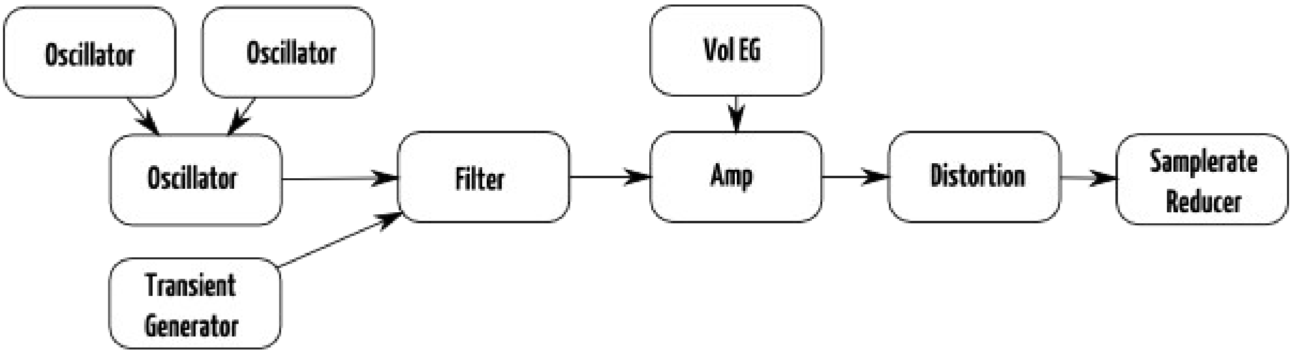
4.2. Cymbal/Clap (voice 5)

This voice uses a 3 operator FM to generate the sound. The main oscillator is modulated by 2 modulation oscillators. It's great for metallic and noisy sounds!



4.3. Hihat voice (6)

The hihat voice is nearly identical to the Cymbal voice, but it offers 2 different decay times for the amplitude envelope. The voice is shared between voices 6 and 7, each using one of the 2 available decay times. This allows to play open and closed hihats. The closed hat (voice 6) will choke the open hat (voice 7).



4.4. Voice parameter menu sections

The synthesis parameters of each voice can be altered in voice **EDIT** mode. The synthesis engine is grouped into 7 different sections that can be selected with the **DRUM** + **SELECT** buttons. In this section each of those 7 menu pages and their parameters are described.

4.4.1. Oscillator Page (OSC)

Provides access to the main oscillator parameters. Frequency and waveform of the main oscillator can be set here.

Drum 1-3, Clap, Hihat

Displayed name	Name	Description
coa	Coarse tune	Coarse tuning of the main oscillator in semitones
fin	Fine tune	Fine tuning of the main oscillator. +/- 50 cent
wav	Waveform	The waveform of the main oscillator
pwm	Pulsewidth	If the PWM waveform is selected, its duty cycle can be adjusted here

Snare

Displayed name	Name	Description
coa	Coarse tune	Coarse tuning of the main oscillator in semitones
fin	Fine tune	Fine tuning of the main oscillator. +/- 50 cent
noi	Noise frequency	Coarse tuning of the noise oscillator.
mix	Oscillator mix	Mix ratio between oscillator and noise source
wav	Waveform	The waveform of the main oscillator
pwm	Pulsewidth	If the PWM waveform is selected, its duty cycle can be adjusted here

4.4.2. Amplitude Envelope Page (AEG)

Each voice has an amplitude envelope. The common parameters are attack and decay time as well as the slope of the curve.

Drum 1-3

Displayed name	Name	Description
att	Amplitude envelope attack	Rise time of the envelope.
dec	Amplitude envelope decay	Fall time of the envelope
slp	Amplitude envelope slope	Variable slope from exp to linear to logarithmic.

Snare/Cymbal

Displayed name	Name	Description
att	Amplitude envelope attack	Rise time of the envelope
dec	Amplitude envelope decay	Fall time of the envelope
rpt	Repeat count	Number of retriggers.
slp	Amplitude envelope slope	Variable slope from exp to linear to logarithmic.

HiHat

Displayed name	Name	Description
att	Amplitude envelope attack	Rise time of the envelope
D1 / D2	Closed / Open hihat decay time	Fall time of the envelope for the hihats
slp	Amplitude envelope slope	Variable slope from exp to lin to log

4.4.3. Modulation Page (MOD)

If the selected voice offers a second envelope, it's parameters can be found here. The modulation page also contains the accent velocity modulation. Here you can turn the accent to volume modulation on and off.

Drum 1-3, Snare

Displayed name	Name	Description
dec	Modulation envelope decay	Fall time of the envelope.
slp	Modulation envelope slope	Variable slope from exp to linear to logarithmic.
mod	Modulation envelope amount	Controls how strong the envelope modulates it's target The envelope is hardwired to the pitch. This parameter controls the pitch modulation intensity. In FM mode the EG is additionally hardwired to the FM amount.
vol	Accent to volume modulation	Can be either On or Off. If activated the accent signal controls the voice volume.

Clap,Hihat

Displayed name	Name	Description
vol	Velocity to volume modulation	Can be either On or Off. If activated the accent signal controls the voice volume.

4.4.4. Frequency Modulation Page (FM)

The FM page hosts the frequency, waveform and modulation amount settings for the FM oscillators.

Displayed name	Name	Description
amt	Modulation amount [mix ratio]	FM mode: modulation amount Mix mode: Mix ratio of the 2 OSCs
frq	Frequency of the FM OSC	Coarse tuning of the FM oscillator in semitones.
wav	Waveform of the FM OSC	The waveform of the FM oscillator.
mod	Oscillator mode	FM mode: the main OSC is modulated by the FM OSC Mix mode: The main OSC and FM OSC are mixed.

Snare

Voice 4 offers no FM capabilities at the moment, so the Page is empty.

Clap, Hihat

Displayed name	Name	Description
f1	Frequency 1	Frequency of the first modulation oscillator.
f2	Frequency 2	Frequency of the second modulation oscillator.
g1	Gain 1	Gain of the first modulation oscillator.
g2	Gain 2	Gain of the second modulation oscillator.
wav	Waveform 1	Waveform of the first modulation oscillator.
wav	Waveform 2	Waveform of the second modulation oscillator.

4.4.5. Transient Generator Page (Click)

The transient generator parameters are the waveform, volume and frequency of an additional short attack transient that can be mixed to the voice output.

Displayed name	Name	Description
wav	Transient wave shape	Selects the transient sample to play.
vol	Transient volume	Volume of the transient sample.
frq	Transient frequency	Frequency of the transient sample.

4.4.6. Filter Page (FIL)

All parameters of the filter are on this page. The available parameters are frequency, resonance, type and drive.

Displayed name	Name	Description
frq	Filter frequency	Changes the cut off frequency of the filter.
res	Filter resonance	Adjusts the filter resonance.
typ	Filter type	Selects the filter characteristic.
drv	Filter drive	Increases the input gain of the filter .

4.4.7. Mixer Page (Mix)

The mixer page provides access to volume, panning, routing, voice FX and sequencer track length.

Displayed name	Name	Description
vol	Voice volume	Adjusts the volume of the voice.
pan	Voice panning	Voice panning. Only active if output is set to a stereo channel.
sr	Sample rate	Sample rate decimation
drv	Drive	Soft clipping amount
out	Output	Selects the hardware audio out for this voice. The voice can either be routed to one of the 2 stereo channels or use the 4 outputs as individual mono channels or the FX bus.

5. Menu

There are different menus for different tasks. Voice editing, performance or modulation and FX routing. The main modes are accessible with the 6 **MENU** buttons.

5.1. EDIT mode

The voice **EDIT** mode is used to modify the sound parameters for each voice.

- The parameters of the voice are divided into several pages which can be selected using the 7 **SELECT** buttons (6).
- While the **DRUM** button (2) is held, the 7 **SELECT** buttons (6) select the active voice, which is indicated by a lit LED.
- Each voice type has its own set of parameters, but the overall page structure is the same for all voices.
- Parameters in the display can be edited using the 4 knobs below.

Parameter editing

As an example lets take a look at the filter page:

```
Frq  res  typ  drv
 32  120   LP    0
```

To change the frequency of the filter of voice 3

- Press ' **DRUM** + **SELECT** Button 3' to select the voice.
- Press ' **SELECT** button 6' to show the filter page.
- Use the first knob to adjust the 'frq' parameter.

5.2. Performance Mode

This mode is designed to jam live. You can manually trigger the voices using the **SELECT** buttons and access the morph and global sample rate FX. There is no voice editing possible in this mode.

5.2.1. Basic performance menu

In performance mode the display shows a set of 3 parameters.

Displayed name	full name	function
mrp	Morph amount	Ratio between the original sound and the morph target.
mtg	Morph target	Select the kit number of the morph target.
sr	Global samplerate	A global sample rate reduction effect.

5.2.2. Morph

The morph feature allows you to morph from one preset sound to another. You can use any kit as a morph target. The Morph amount value controls the ratio between the original and the target sound. As the morph parameter is increased, the current sound is gradually transformed into the selected morph target sound.

Did you know?

If morphing arbitrary presets is too drastic for you, try modifying your favorite pattern just a little bit and save it to a new location. Now you can control all tweaked parameters at once!

5.3. Modulation Menu

5.3.1. Mod Matrix

Each voice has a 3 slot mod matrix where the CVs, accents and LFOs can be routed to any parameter of the synth engine. The mod matrix can be accessed with the **MOD** button.

The voice can be selected by holding the **DRUM** button and pushing one of the **SELECT** buttons while the mod matrix menu is active.

Displayed name	full name	function
src	Modulation Source	There are 17 modulation sources to choose from: 5 CVs from the input jacks 6 LFOs 6 Accent signals
amt	Modulation Amount	Modulation strength 0% to 100%
dst	Modulation destination	Target sound parameter list from the current voice

The modulation is bipolar around the current value of the parameter. So for a full range modulation the target parameters value is best set to 50%.

The mod matrix has a special mod target called "**v/o**". Using this target, the incoming Cvs are scaled to allow tonal play of the voices using a 1V/Oct CV source. Amount has to be set to 100% for exact tracking. In 1V/Oct mode negative CVs are ignored. This gives you a 5 octave range.

5.3.2. LFO

There are 6 low frequency oscillator. They are basically the same as the audio oscillators, but run at a lower maximum rate.

The LFO menu is accessed by pressing **SHIFT** + **MOD**.

The 6 LFOs can be selected by holding the **DRUM** button and pushing one of the **SELECT** buttons while the LFO menu is active.

Unlike the audio oscillators you can not hear the LFOs directly. They are used to alter other parameter values over time. For example if you want the filter of a voice to slowly open and close again, you can use a sine lfo to modulate the filter cutoff.

Displayed name	Name	Description
frq	LFO frequency	Manual LFO rate control. Only available when sync is turned off!
off	Phase offset	When the LFO is retriggered, it is reset to the selected phase offset.
wav	LFO waveform	Select the LFO waveform. • Sine (sin) • Triangle (tri) • Saw up (sup) • Saw down (sdn) • Square (sqr) • Random (rnd) • Exponential saw up (xup) • Exponential saw down (xdn)
rtg	LFO retrigger	Allows the LFO to be retriggered from different sequencer tracks. Select the voice that should retrigger the LFO (v1-v6) and it will reset its phase whenever a note is played on the selected track.

Did you know?

You can use the LFO as an additional envelope using the retrigger feature. If set to the same voice as the modulation destination, the LFO will restart on every played note. You just have to set the LFO frequency so that one cycle of the LFO is slightly longer than the amplitude envelope, so it won't start a new cycle while the note is still playing. Great for filter decay envelopes on juice synth bass souds!

5.4. FX Mode

The LXR provides a digital FX processor. Only 1 effect can be used at a time. Currently there are 4 FX available: Drive, Ringmodulator, Compressor and Delay.

You can use the FX processor, by setting the output routing on the mixer page to “FX”. So the voice will be routed to the FX bus before going to the output. So you can select which of the voices will be processed by the FX processor.

5.4.1. Menu location

The FX menu is accessed by pushing **SHIFT** + **FX**

5.4.2. Drive

The drive FX is a collection of different distortion algorithms. It offers 3 different types of saturation. In contrast to the drive parameter of the single voice filter and mixer pages, this FX processes all signals in a single distortion unit. This leads to interesting inter modulation between the input signals.

Drive Types

The 4th parameter “type” selects the drive algorithm to use.

- **Tub –** Adds tube Saturation
- **Fld –** A wavefolder distortion
- **Clp –** A rodent inspired hardclip pedal emulation

Parameters

Displayed name	Name	Description
typ	FX Type	Drive
out	Output routing	Selects the Audio output the FX signal is mixed to.
d/w	Dry / Wet ratio	Mixing ratio between the original and the FX signal.

typ	Drive Type	Select the drive type (Tube, Fold, Clip)
drv	Drive strength	Input gain of the distortion unit
col	Colour	Tube Bias — Fold Offset — Clip Feedback
ton	Tone	Tone Control Filter
vol	Volume	Output volume of the distortion

The feedback parameter of the Clip processor will generate a tone when no input is present. This is the normal behavior, as the effect is quite similar to the feedback sound a microphone in front of a speaker produces. While a drum pattern is playing, it gives nice intermodulation, but without input a clean tone is produced. Just pull down the master volume if you stop the sequencer when you are using full feedback or use an external VCA.

5.4.3. Ringmodulator

The ringmodulator multiplies the input signal with a second oscillator signal. Ever wanted to hear a Dalek beatboxing? Then the ringmod is your friend!

Parameters

Displayed name	Name	Description
typ	FX Type	Ring
out	Output routing	Selects the Audio output the FX signal is mixed to.
d/w	Dry / Wet ratio	Mixing ratio between the original and the FX signal.
wav	Waveform	Modulation Waveform
frq	Frequency	Frequency of the modulation waveform.

5.4.4. Compressor

A compressor is a dynamic processor that is able to reduce the volume of loud sounds and boost the volume of quiet sounds, thus reducing the dynamic range of the signal.

Parameters

Displayed name	Name	Description
typ	FX Type	Compressor
out	Output routing	Selects the Audio output the FX signal is mixed to.
d/w	Dry / Wet ratio	Mixing ratio between the original and the FX signal.

rat	Compression ratio	1:1 up to 8:1 in dB. Strength of the compression. For example a ratio of 2:1 will attenuate the volume of a signal 2dB above the treshold down to 1dB above the treshold.
atk	Attack	Time it takes for the compressor to kick in. 0.3 to 100ms range
dec	Decay	Time it takes to return to normal gain, after the input falls below the treshold. 25 to 200ms range.
tre	Treshold	The treshold sets the level at which the compression effect sets in. -96 to 0dB
gai	Makeup Gain	Compression lowers the volume. Use the output gain to make up for the difference.

5.4.5. Delay

Straight forward delay FX. Not much to say here. It's a classic!

Try using the range control for very short delay times and LFO modulation to get some flange action going!

Parameters

Displayed name	Name	Description
typ	FX Type	Delay
out	Output routing	Selects the Audio output the FX signal is mixed to.
d/w	Dry / Wet ratio	Mixing ratio between the original and the FX signal.
typ	Delay type	The 4th parameter "type" selects the delay algorithm to use. > Del – Mono delay > Pp – Stereo ping pong delay. The feedback alternates between the left and right channel. Pan must be set to a non zero value to work!
tim	Delay time	Adjust the delay time in the range specified by the range parameter.
rng	Time range	Sets the range for the delay time dial. The low range is 1ms to 20ms The high range is 20ms to 0.7sec
ton	tone	Frequency of the lowpass filter in the feedback path.
fbk	feedback	Controls the number of repeats.

5.5. Loading and saving

When the LXR is turned on, the last used kit is automatically loaded from the SD card to the machines memory.

After this initial load, all changes made are temporary. If another kit is selected, all changes to the current kit are lost.

To make these changes permanent, you have to save the kit to commit them to the SD card.

The **SAVE/LOAD** button will bring up the kit load menu. Press **LOAD/SAVE** again, to bring up the save menu.

5.5.1. Display description

Load: Kit
[1] Name

- The first row of the display gives information about the active mode (load or save)
- The second row chooses the active preset no. and displays the name of the selected preset.
- You confirm the selection by pushing the encoder button to load/save a kit.

5.5.2. Menu navigation

- Use the encoder to navigate trough the menu
- push the encoder to confirm your selection

Quick naming scheme

To speed up the name entry on the save page you can also use the knobs below the display. If the cursor is in the name area the knobs have the following functions:

- The first knob selects the cursor position
- The second knob switches between upper and lower case letters.
- The third knob scrolls through the available default ASCII characters.

Kit Loading

- Press **LOAD/SAVE**
- Select a new kit with the encoder.

- To load the selected kit push the encoder.
- The kit sound will play immediately
- unsaved changes to the previous kit are lost

Kit Saving

- Press **SHIFT** + **LOAD/SAVE**
- Select the preset number you want to save your kit to.
- Push the encoder
- You will be asked : ”Edit Name? [y/N]”
- if you don’t want to change the name, push the encoder and you kit is saved.

- if you want to change the name, use the encoder to navigate to [Y/n] and push it.
- Navigate to the letters of the name.
- You can change the letters if you push the encoder and turn it or refer to the quick naming scheme
- If you are happy with the name, navigate the cursor to the 'ok' button and push the encoder down to save the kit.

5.6. Global settings menu

The settings menu contains the global configuration of the synth. Settings will be saved as soon as you exit the menu.

5.6.1. Menu location

- Press the **SHIFT** + **LOAD/SAVE** buttons
- The **SHIFT** + **LOAD/SAVE** button starts to flash to indicate that the global settings menu is active.

5.6.2. Menu options

Displayed name	Name	Description
ssv	Screensaver	Turn the screensaver on or off
cv	Global CV Save	Save the CV routing globally or on a per kit basis

5.6.3. Screensaver

The display provided with the kit is an OLED display. To expand the lifetime of the OLED display and to avoid burn in, the firmware provides a screensaver for the display. After no control is touched for some minutes, the display will show the screensaver. As soon as any control is touched, the display will show the menu again. The screensaver can be deactivated in the settings menu.

6. Synth modules

In this chapter we will have a closer look at the synth modules used in the different voices.

6.1. Oscillator

The oscillators provide 6 different waveforms:

- Sine
- Triangle
- Saw
- Rectangle
- Noise
- PWM

The waveforms are classic analogue waveforms realized using bandlimited wavetable oscillators.

6.2. Filter

The filter is a 2 pole (12dB) state variable filter (SVF). It's used to shape the harmonic content of the sound. For example a hihat sound consists of high frequencies and no low frequencies, so you want to use a highpass filter. A clap has lots of mid sounds, here a bandpass filter is useful.

There are 3 parameters that control the filter response. Frequency, resonance and drive.

6.2.1. Frequency

The frequency determines the operating point of the filter in the spectrum. For example a low pass filter will cut off all frequencies above its set cutoff frequency (all frequencies below it pass the filter unaltered – hence the name).

6.2.2. Resonance

The resonance controls the feedback path of the filter. With higher resonance settings the frequencies around the operating point (i.e. the set frequency) will be amplified more and more.

It becomes clear if we look at the filter amplitude response plots in the next section.

6.2.3. Drive

Controls how 'hot' the filter is driven. More drive yields in more distortion. In the normal operating range, when drive is set to 0, the filter is quite clean and nearly linear. With higher input levels/drive settings, a soft clipper as well as the slew rate limit of the integrators comes into effect. Low settings will only affect the resonance peaks, higher settings will distort the whole signal. Since the soft clipper is scaling down excessive peaks in the signal, the audible resonance is reduced with higher drive settings.

6.2.4. Filter types

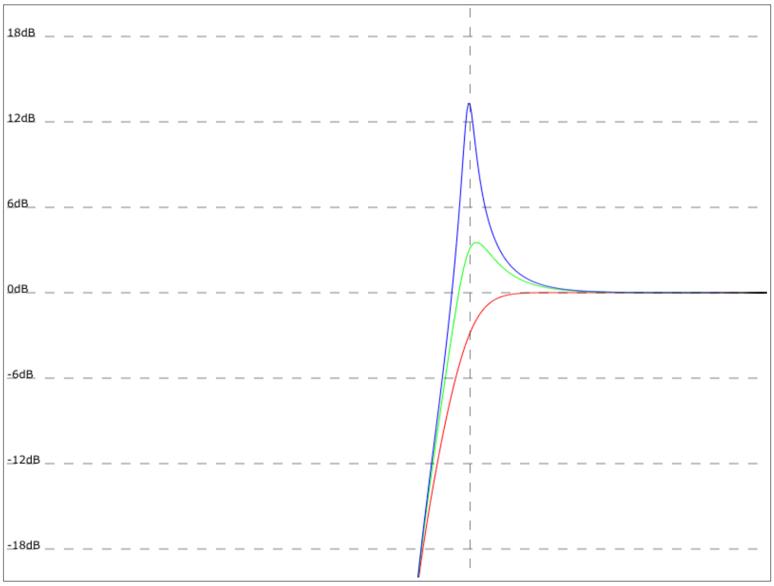
There are several different filter types available, each with it's own characteristics. Let's have a closer look at each one of them. The filters are plotted with 3 different resonance settings. No resonance, medium and high. The cut off frequency is the same in every plot and marked by the vertical dashed line. The x axis shows the frequency from 0Hz to 22kHz, the y axis shows the gain for the specific frequency - Both on a logarithmic scale.

Lowpass

The lowpass filter removes high frequencies from the signal. All frequencies above the cut off frequency are reduced gradually.

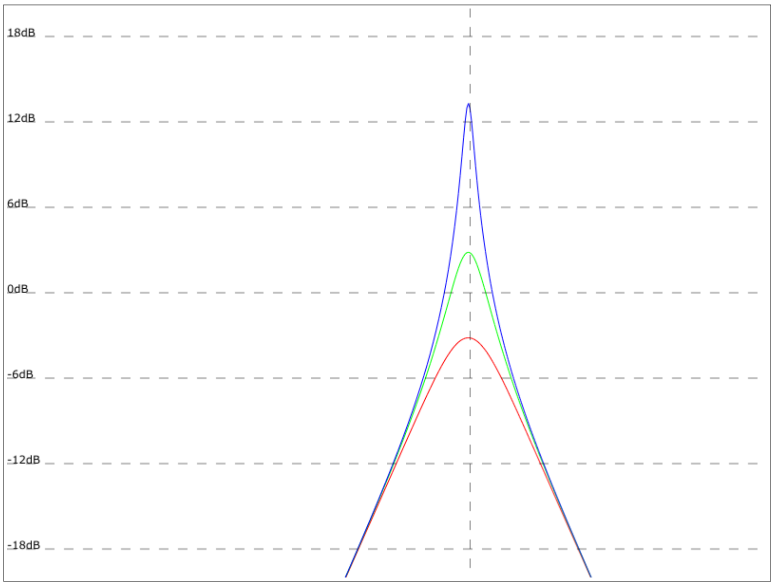
Highpass

The highpass filter removes low frequencies from the signal. All frequencies below the cutoff frequency are reduced gradually.



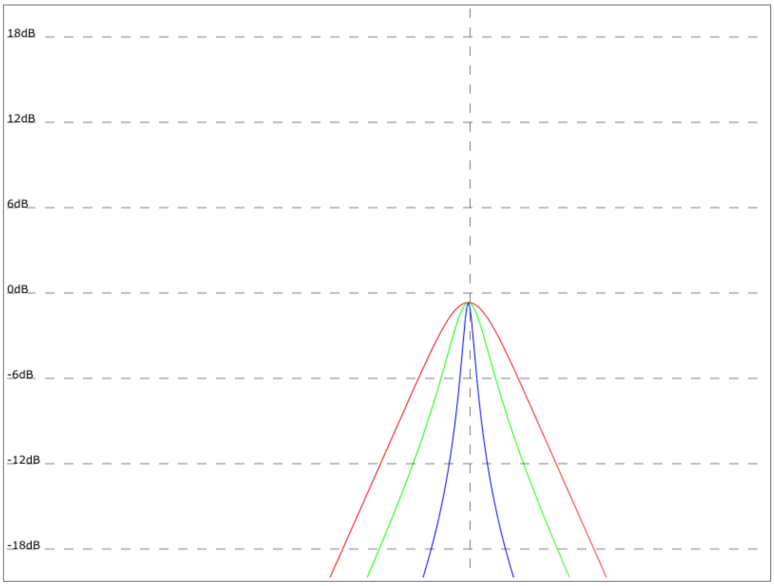
Bandpass

The bandpass filter removes frequencies above and below the set cutoff frequency.



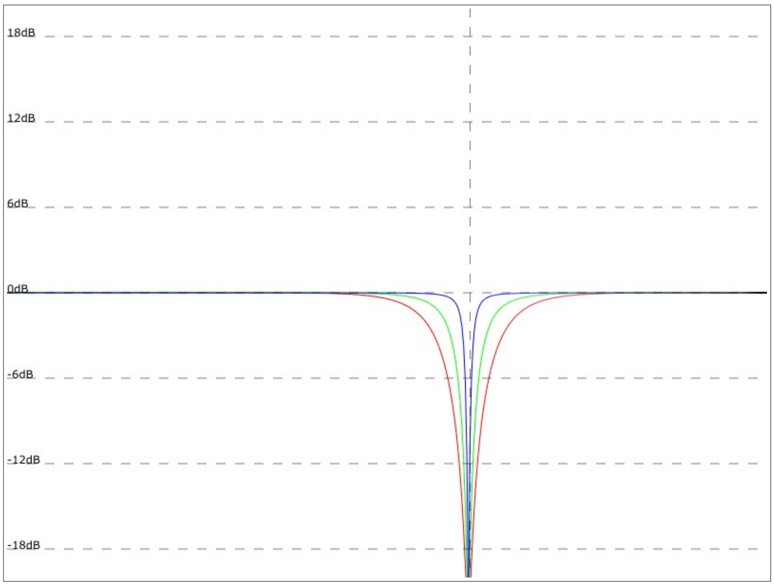
Unit gain bandpass

The unit gain bandpass is a scaled version of the normal bandpass filter. The gain is always adjusted to have its maximum around 0dB gain. Unlike the normal bandpass, where the resonance controls the amplitude of the peak gain, it controls the width of the passband.



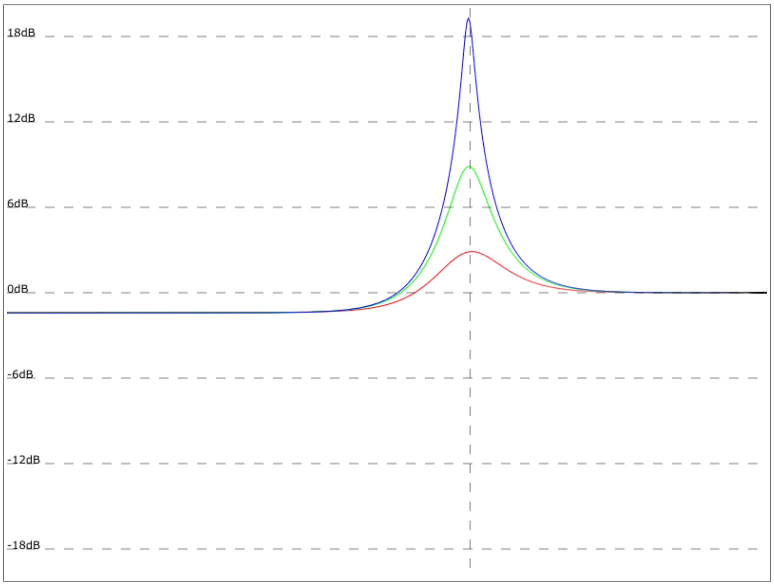
Notch

The notch filter removes frequencies around the set filter frequency. Again, the resonance controls the width of the stopband.



Peak

The peak filter amplifies frequencies around the filter frequency, but lets the other frequencies pass nearly unaltered. The resonance controls the amount of amplification.

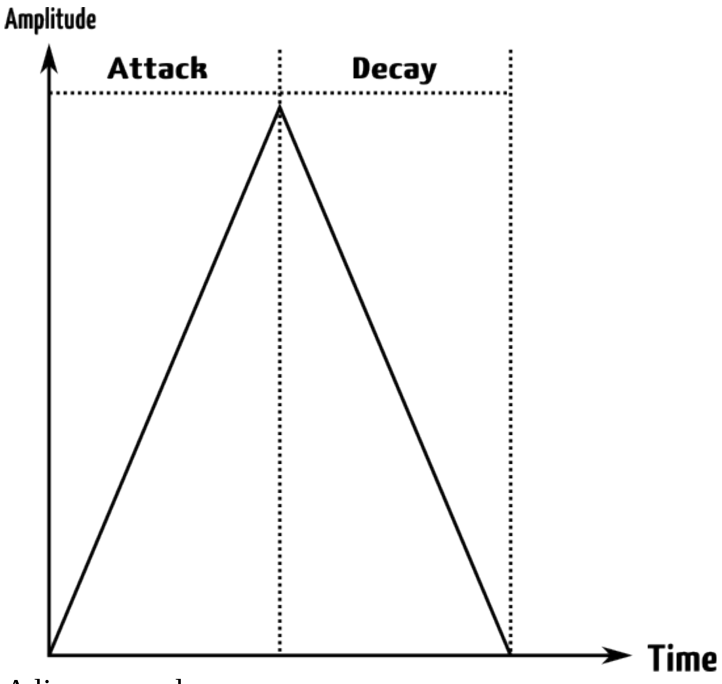


LP2 Type

This type is a special case. It replaces the SVF model with a simpler implementation of the SVF topology with much more digital resonance. Use it for those screaming acid sounds!

6.3. Envelopes

Envelopes are used to generate a varying control signal that can control other synthesis parameters. Whenever a voice is triggered, the envelope is restarted. The signal rises with the speed selected by the attack setting until it reaches the maximum amplitude. Then it will fall back down to zero, with the speed set by the decay parameter.



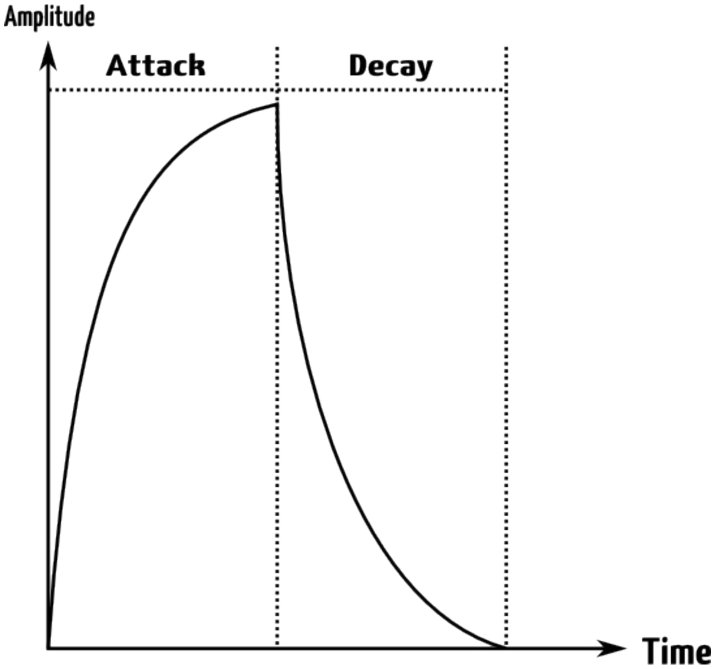
A linear envelope curve

6.3.1. Attack/Decay times

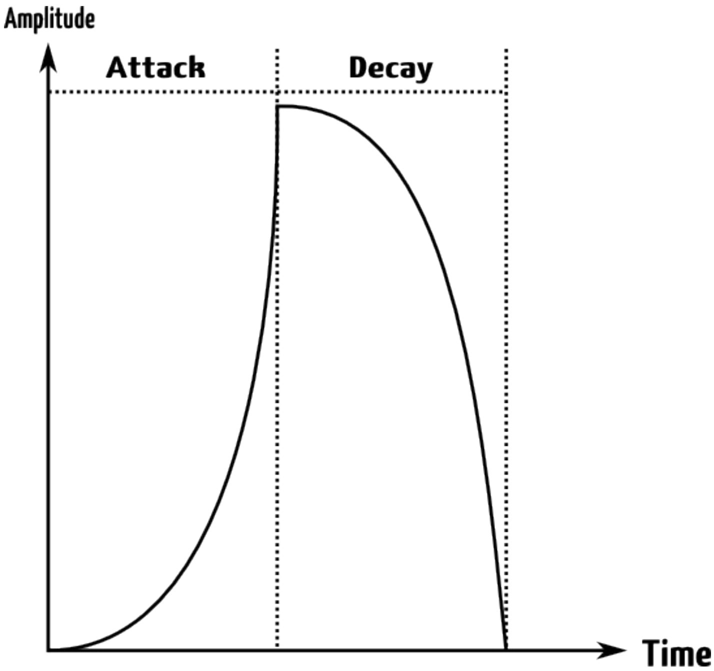
These 2 parameters control the duration of the attack and decay stage of the envelope. A higher value equals longer time. If the attack time is set to zero, the envelope will directly start in the decay stage.

6.3.2. Slope

The slope parameter controls the shape of the generated curve. A setting of 63 will give you a linear envelope as seen above. Lower settings will result in an exponential curve, higher values result in a logarithmic curve.



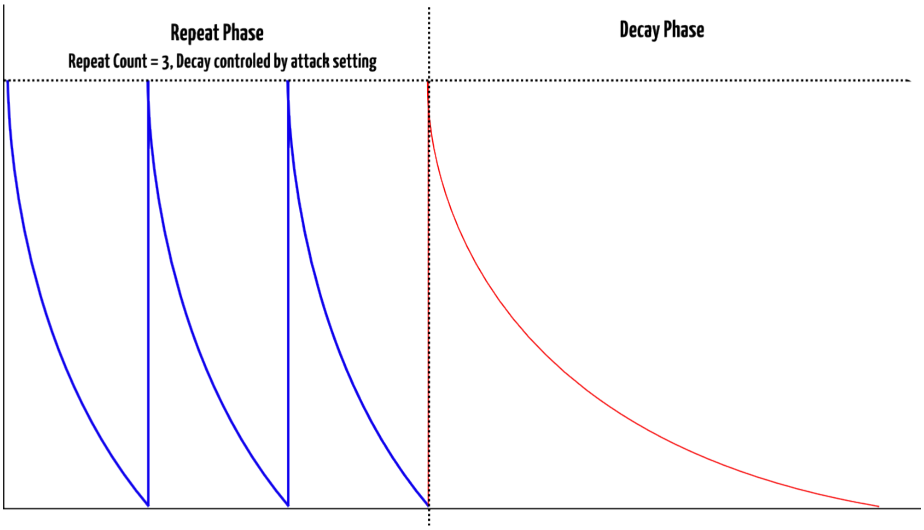
Curve with a low slope setting. Exponential decay stage.



Curve with a high slope setting. Logarithmic decay stage.

6.3.3. Repeat

A special feature of the envelope to simulate claps and other sounds with a fuzzy/rattling attack.
If the repeat count is set to a value greater than zero, the envelope is in repeat mode.
The slow onset of the attack stage is replaced by the repeat stage in which the envelope is retriggered rapidly. The attack parameter controls the duration of the complete repeat phase.



6.4. Transient generator

The transient generator is used to shape the beginning of a sound. Since the perception of sounds is heavily influenced by their first few milliseconds, this feature is useful to spice up or vary the sound character of an instrument.

There are different modes the transient generator uses to shape the attack

6.4.1. Snappy mode

The snappy mode is the first entry in the selection list, shown in the menu as 'Snp'
To make the attack of the sound more 'snappy' an additional pitch envelope is used.

- The volume parameter controls the modulation depth
- The frequency parameter controls the decay time

The envelope is quite fast and modulates the pitch of the oscillators resulting in a nice, controllable snap or click sound.

6.4.2. Offset mode

The second entry is the offset mode with the menu name 'Off'.
In this mode the start phase of the oscillators can be adjusted with the volume parameter. The frequency parameter has no effect.

If the volume is set to zero, the oscillators will be reset to a zero crossing of the waveform on each trigger. If the volume parameter is set to its maximum value, the waveform will start on the position with the highest amplitude, generating a loud pop. This results in the volume parameter controlling the intensity of the initial pop.

6.4.3. Sample mode

A short transient sample is mixed to the sound. They are short (~50ms) 8-bit ROM samples that are played as a one shot whenever the sound is triggered. Different samples can be selected.

6.4.4. Parameters

The parameters of the transient generator include the PCM waveform to use, the playback frequency and the volume. The ROM samples can't be changed by the user on the fly. They are hard coded into the firmware.

A list of the included samples:

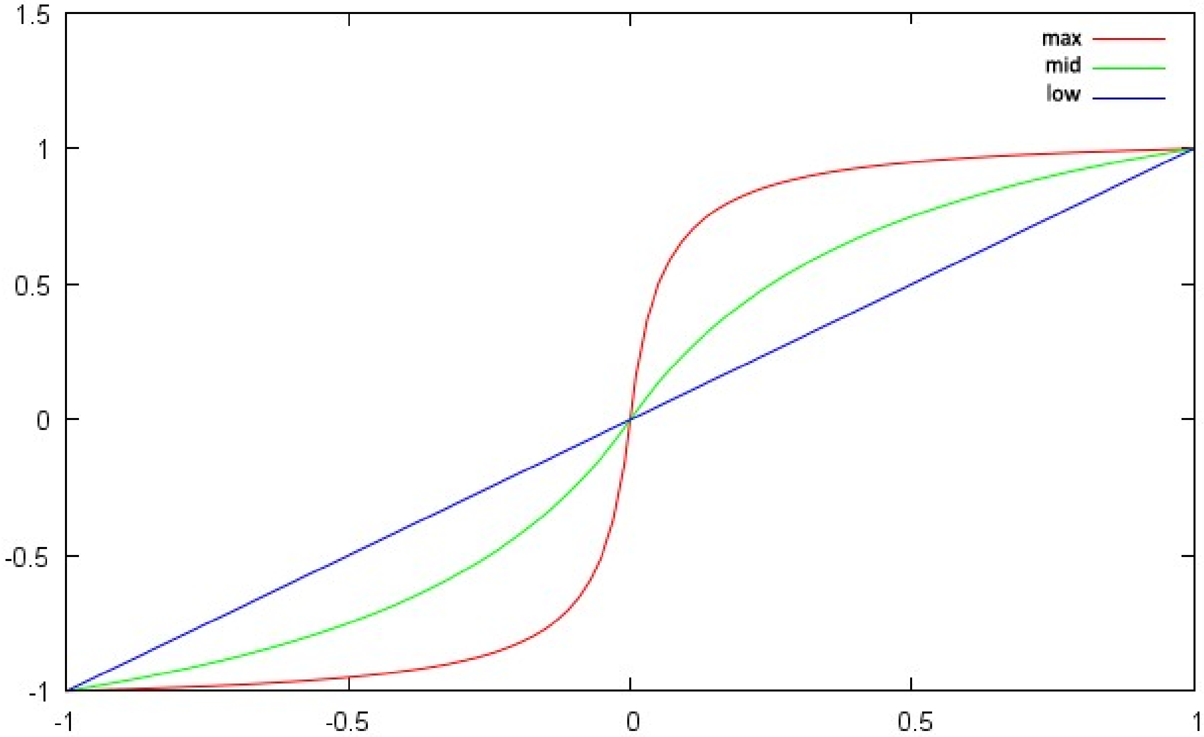
Displayed name	Name	Description
Snp	Snappy Mode	
Off	Offset Mode	
Transient Samples		
Clk	Click	
Ck2	Click 2	
Tik	tick	Some kind of tick sound[i]
Kik	Acoustic kick	An acoustic kick drum[ii]
Rim	Rim shot	A rim shot sound[iii]
Drp	Drip	A dripping sound made by a human being.[iv]
Hat	Hat	Sampled off of Zildjian A-Custom Hi-Hats[v]
Clp	Clap	808 clap[vi]
Ki2	Kick 2	A kick sample
Sna	Snare	A snare sample
Tom	Tom	A tom sample (sort of)
Sp2	snap	A finger snap

6.5. Sample Rate Reduction

The sample rate reducer can gradually reduce the sample rate from 44kHz to zero. Reducing the sample rate will give more of a lo-fi sound. The more it is reduced, the more odd harmonics will be added to the sound as overtones are folded down at the Nyquist frequency until the sound is totally destroyed.

6.6. Distortion

The distortion unit on the mixer page is a variable waveshaper. It can be set from no distortion, over soft saturation up to hard clipping.



6.7. Accent modulation

The accent can be used as a modulation source and is an important tool to bring some life to your static patches. The accent can modulate 2 destinations simultaneously.

- The voice volume.
- Any parameter of the voice.

By default the velocity is modulating the voice volume. This hard wired connection can be turned off in the modulation menu with the 'Vol' on/off parameter

6.8. Output Routing

Each voice can be freely routed to the 4 different output jacks. They act either as 4 individual mono outs (pan parameter has no effect) or as 2 pairs of stereo outputs. There is an additional FX bus available as routing target as well. Have a look at the FX section of the manual for more info.

7. Firmware update

The software for the LXR can be updated using the memory card.

7.1. Update procedure

- Copy the latest LXRE_V2_xxx.img to the root of the SD card.
- Delete any old firmware .img files from the SD card.
- Push and hold the encoder button while turning on the device.
- "LXR Bootloader" will pop up in the display
- The LXR will check if the firmware image is valid and has the proper checksum
- If the check is passed, the update will be installed.
- Do not turn off the power during installation!
- Once done, the LXR will prompt you to reboot and the new firmware will be running.

7.2. Bootloader error messages

- **LXR*.img Not Valid!**
  File doesn't seem to be a valid firmware file. File header not matching
- **LXR*.img Wrong size!**
  Filesize differs from the size specified in the header. Probably an incomplete download.
- **Wrong Checksum**
  File corrupted. File content does not match the checksum. Probably a download error.

8. Introduction to drum synthesis

This chapter will introduce you to the very basics of drum synthesis. Of course this is just a small pointer in the direction of some classic drum sounds. There are much more sounds to discover than the well known drum sounds of the analogue classics. Nonetheless we will mostly focus on generating those classic sounds, because they give a good starting point for further explorations.

8.1. Kicks

The basis of nearly every kick sound is the simple combination of an oscillator and a decaying envelope modulating its pitch. The first 3 voices are perfect for this!

Oscillator

A good starting point is a sine oscillator with a pitch around 30 to 35.

Amplitude Envelope

To decide how to set up the envelope, let's think about the kick sound. A kick has a sudden impact, so we want no attack (set to zero) since the sound starts with its maximum amplitude.

The decay controls the length of the kick sound. For the average kick a decay in the range 20-30 and a slope around 25 (exponential) gives a nice, not too short, not too long kick. For those booming 808 kicks you may want to set the decay higher.

Pitch Envelope

To get some 'oooomph' and punch into your sound you need to set up the pitch envelope.
Try a decay and slope setting similar to the amplitude envelope as a start. All three parameters, decay, slope and modulation amount, have a great impact on the sound of the kick.

Click

The attack sound is an important part of a kick. There is no general rule here. Try different transient generator settings for various clicks and pops or even slightly increasing the amplitude attack to give some softer bass drums.

Tips and tricks

- You can further shape the attack of the sound using the FM oscillator.
- You want even more clicks in the attack? Why not enhance the click with a peak filter?

8.2. Snare

The snare is a 2 component sound. There is the tonal part from the drum body and the noise part from the rattles.

Voice 4 of the LXR is designed with these 2 parts in mind. You have an oscillator for the tonal part and a noise generator with filter for the noise part. Since in most of the cases you want a highpass filtered noise with an unfiltered drum body, only the noise generator is routed trough the filter.

For most snare sounds you want the noise part to be louder than the tonal part, so try setting the mix parameter on the oscillator page to a value around 100.

Tonal part

The tonal part is quite similar to the kick but not as pronounced. The typical snare does not need a loud click, nor that much modulation depth for the pitch envelope. Additionally, since a snare is smaller than a bass drum, the frequency of the oscillator has to be higher than on the kick.

Noise part

For most cases highpass filtered noise with a moderate resonance setting is sufficient to get good results.

Amplitude envelope

We want a sharp percussion sound, so the attack will stay at zero.
For the decay it depends on what kind of snare you are after.

- Longer decay times with a very exponential slope setting as low as 5 to 10 will give you a short hit sound with a nicely decaying noise tail. The tail acts a little bit like a room reverb on the perception of the snare sound and gives more natural results.
- Shorter decay times with a linear or exponential slope will give a very dry, direct snare.

Tips and tricks

- Not every snare needs tone and noise. Try using only the tonal part with a high pitch modulation envelope to get the classy Kraftwerk zap.

8.3. Clap

A clap sound is made from noise. What's special about a clap, is the fuzzy attack of the sound. Imagine a few people clapping a rhythm. They will never all clap at exactly the same time. Each of them will have small variations in their timing, some will clap a little bit earlier, some a bit later.

Amplitude Envelope

To simulate this behaviour we will use the repeat feature of the amplitude envelope. To get the classic clap sound, a setting of 3-4 repeats is good. Since the timing differences for the clap are very small, a short repeat time has to be chosen. An attack value below 10 is good.
Now you will have a short burst in the beginning of the sound.

For the decay we need an exponential slope. A value below 10 is recommended.

Oscillator

A single white noise oscillator set to its maximum frequency.

Filter

BP filter 60-80 with high resonance settings gives good results.

8.4. Hihat

Oscillator

For hihats a complex, metallic noise spectrum is needed. Just use the 3 oscillators with high frequencies and modulation amounts to get a spectrum you like. Sometimes a simple white noise oscillator can give nice hats, too.

Amplitude Envelope

For hihats you have to set 2 decay times. One for the closed and one for the open hihat. Both are quite short, but the open hihat decay should be longer than the closed. Use a very exponential slope below 10 for hat sounds.

Filter

Use a highpass filter to remove all low frequencies. A high frequency around 120 is used.

8.5. Cymbal ride

Cymbals are similar to hats. The difference comes from the envelope and filter settings.

Oscillator

The oscillator settings can be similar to those of the hihat sound.

Amplitude Envelope

The only real difference to the hihat envelope is the decay time. We also use a strong exponential slope, but a much longer decay is needed. It is also recommended to lower the volume of the voice otherwise the cymbal sound may be too loud.

Filter

A bandpass filter with high resonance settings and a frequency above 110 gives good results.

LFO

The cymbal sounds really benefits from a LFO modulation the filter frequency. Use a low modulation frequency and a sine wave with a gentle modulation depth so the different strikes won't sound exactly the same and the frequency spectrum of the sound changes a little bit over time.

8.6. Bells

8.6.1. Realistic bells

Realistic bells are best achieved using FM oscillators. As long as you have a complex noisy, metallic spectrum, you just need to add an exponential decay amplitude envelope and a bandpass filter to get different kinds of bells. The sample rate reducer is also very useful to make the sound more metallic.

8.6.2. 808 style cowbell

The 808 cowbell consists of 2 detuned rectangle oscillators send through a bandpass filter.

Recommended voice: drum voice 1-3

FM Page

- set mode to mix. The main and FM oscillators are now mixed together.
- Set amount to 63. This sets the mix ratio between the 2 oscillators to 50% each.
- Select a rectangle waveform
- set frequency to something around 78

OSC page

- Select rectangle wave.
- Set frequency to 71

Amplitude envelope

- Attack 0
- very exponential slope around 2 to 5
- depending on the slope setting a short decay in the range of 25 to 50

Filter

- Use a bandpass filter
- medium resonance around 70
- cutoff frequency around 92

8.7. More info about drum sound design

There are a lot of good sources about drum synthesis on the web. Two exceptionally good readings are:

- The Sound on Sound synth secrets series has a lot of good articles for different drum sounds
- The owners manual of the Waldorf Attack synthesizer contains a lot of HowTos on drum synthesis.

Use your search engine of choice to find both documents.

9. TECHNICAL INFORMATION

9.1. Electrical Specification

Main outs

level: 10Vpp

Output impedance: 100 Ω unbalanced

Power

+12v- 88mA - peak 160mA

-12v - 10mA - - peak 12mA

CV Range

-5V to 5V

Trigger amplitude

5V

9.2. Hardware

2 x 20 OLED screen

4 × 3.5mm audio out jacks

7 x 3.5mm trigger in jacks

6 x 3.5mm accent in jacks

5 x 3.5mm CV input jacks

44.1 kHz, 16-Bit D/A converter

4 x endless pot

1 x push encoder

9.3. Physical Specification

Dimensions: 28HP, Module depth 35mm

Do not expose instrument to temperatures over +50℃ or below -20°C

i Sound by patchen
July 14th, 2005
http://www.freesound.org/people/patchen/sounds/4102/

STEREO ATIK household percussion by 'patchen' released under a creative commons attribution license

ii Acoustic kick
http://www.freesound.org/people/KEVOY/sounds/82279/
Post processed with EQ, compressor and amplitude curves by Julian Schmidt
This work is licensed under the Creative Commons 0 License.

iii TR-909 JGB pack » rs03.wav
http://www.freesound.org/people/altemark/sounds/26672/
altemark
December 2nd, 2006

Sampled by Janne G:son Berg from his old 909. Cut up and organized by me. Here is the original readme.txt Janne distributed with the wav:

Sampled in one session from my (now sold) 909. 24 bit, 44.1 kHz. Feel free to use the samples for whatever you like. If you use any of the included patterns, please send me an mp3 of the whole song or a physical copy (contact me for further details).
/Janne G:son Berg 2005

This work is licensed under the Attribution License.

iv http://www.freesound.org/people/Neotone/sounds/75344/
Drips » Drip2.wav
Neotone
July 10th, 2009

A dripping sound made by a human being.
This work is licensed under the Creative Commons 0 License.

v High Quality Acoustic Percussion Samples » Zildjian A Custom Hi-Hat Cymbals Pedal Chic.WAV
http://www.freesound.org/people/pjcohen/sounds/45668/
pjcohen
December 30th, 2007

Sampled off of Zildjian A-Custom Hi-Hats

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vi Provided by shiftr