



# USER'S MANUAL

v2.1.6 (October 10, 2022)

# THE CENTRE



# Contents

## I Before Start

1	Software Update	5
2	Calibration	6
2.1	Why Calibrate?	6
2.2	Calibration of music instruments	6
2.3	Calibrating The Centre	6
3	What's In The Box	7
4	Clocks	8
5	BPM	8

## II System Overview

6	Module Overview	10
6.1	Patch	10
6.2	Physical Connectors	10
6.2.1	Rotary Encoder	10
6.2.2	Knobs	10
6.2.3	V/OCT	10
6.2.4	CVY	10
6.2.5	CV	10
6.2.6	VOUT	11
7	Patch	12
7.1	Knobs	12
8	Audio Outputs	13
8.1	Physical Outputs	13
8.2	Virtual Audio Buffers	13
8.3	Quick Configuration of Audio Inputs and Outputs for Modules	13
8.3.1	Visual Output Presentation in Patch View	14
9	CV Internal Outputs	15
10	Module Inputs	16
10.1	Overview	16
10.2	Input Configuration Mode	16
10.2.1	Changing VALUE and CONTROLLER in Input Configuration mode	16
10.2.2	Layout of controls in input menu (Input Configuration)	16
10.3	Individual Component Configuration Mode	16
10.3.1	Layout of controls in input menu (Controller Configuration)	17
11	Pitch Control	18
11.1	Overview	18
11.2	1V/Oct	18
11.3	Note Control in Inputs	18
12	System Settings	19
12.1	Changing System Settings	19

## III Module Reference

13	WTO - Wavetable Oscillator	21
13.1	Mapping of Controls	21
14	VCO - Voltage Controlled Oscillator	24
14.1	Mapping of Controls	24
14.2	Waveforms	24

15	LFS - Low Frequency Shaper	26
15.1	Mapping of Controls	26
15.2	Loading Shapes	26
16	LFS - Shape Editor	28
16.1	Mapping of Controls	28
16.2	Editing Shapes	28
16.3	Position Quantisation	28
17	LFO - Low Frequency Oscillator	29
17.1	Mapping of Controls	29
17.2	Waveforms	29
18	ENV - Envelope Generator	31
18.1	Mapping of Controls	31
18.2	Operation	31
18.3	Envelope Stages	31
18.4	Envelope Types	32
18.5	Trigger Mode	32
19	VCA - Voltage Controlled Amplifier	34
19.1	Mapping of Controls	34
19.2	Operation	34
19.3	Sidechain	34
20	BRM - Balanced Ring Modulator	36
20.1	Mapping of Controls	36
20.2	Operation	36
21	SMP - Sample Player	38
21.1	Mapping of Controls	38
21.2	Operation	38
22	NOI - Noise Generator	40
22.1	Mapping of Controls	40
22.2	Waveforms	40
23	DLY - Delay	42
23.1	Mapping of Controls	42
23.2	Operation	42
24	DST - Distortion	44
24.1	Mapping of Controls	44
24.2	Operation	44
25	VCF - Voltage Controlled Filter	46
25.1	Mapping of Controls	46
25.2	Operation	46
25.3	Tracking	46
26	DRC - Drum Rack	48
26.1	Mapping of Controls	48
26.2	Operation	48
27	CLK - Clock Generator	50
27.1	Mapping of Controls	50
27.2	Operation	50
27.3	External Clock	50
27.4	Clock Output	50

28	GAT - Gate Divider	52
28.1	Operation	52
29	EUC - Euclidean Rhythm Generator	54
29.1	Mapping of Controls	54
29.2	Operation	54
30	PLY - Polyrhythm	57
30.1	Mapping of Controls	57
30.2	Operation	57
31	RNG - Random Note Generator	59
31.1	Mapping of Controls	59
31.2	Operation	59
32	QNT - Quantiser	61
32.1	Mapping of Controls	61
32.2	Musical Scales	61
32.3	Custom Scales	61
33	ARP - Arpeggiator	63
33.1	Mapping of Controls	63
33.2	Operation	63
33.3	Control arpeggiator position from external source	63
34	OUT - Output	66
34.1	Mapping of Controls	66
34.2	Operation	66

# **Part I**

## **Before Start**

# Software Update

Software Update is the most important process to keep your unit updated with the latest features and bug fixes as well as to keep it safe from breaking during update process. Updating process is fairly simple from user perspective although there might be some issues with downloading files because of specific Internet Browser behaviour during downloading.

1. Download latest firmware release from Github [https://github.com/1V-0ct/3318\\_the\\_centre\\_releases/releases](https://github.com/1V-0ct/3318_the_centre_releases/releases)
2. Copy downloaded firmware file into SD Card (make sure that the filename is: 'the\_centre\_v4.fwx' and it is in root directory of your card)
3. Insert SD Card into The Centre
4. Power on The Centre or perform Reset
5. Go to [[System Menu]] and check firmware version.

■ *NOTE: Sometimes Safari, Explorer and other browsers add some extra bits to the filename like '(1)' or '\_1' to indicate that it is different file than already downloaded. Please rename the file to the correct name.*

# Calibration

TL;DR: I came here to calibrate and not read poems... Here is YouTube video that shows Calibration of The Centre:

<https://www.youtube.com/watch?v=uEFr7RkuP7k>

## 2.1 Why Calibrate?

In the ideal world things are well... ideal. Although in real world things are not perfect and this is where term tolerance kicks in. Every electronic component has tolerance. Usually measured in percentage and usually this tolerance is like %1 for resistors and %5 for capacitors. Each circuit contains multiple electronic components and tolerances of those components add together making quite a huge percentage tolerance of the circuit. When circuit is operating the tolerances of this circuit do not change. Tolerance just change component properties at manufacturing stage. Therefore the circuit is always stable but might be slightly off.

Of course there are different ways to deal with the non-ideal circuits. Some of them are expensive (on hardware level) and some of them are quite cheap. One of those methods is calibration.

Calibration allows to establish default values returned by circuit under controlled environment and use them as reference points.

## 2.2 Calibration of music instruments

Many music instruments need calibration. Even tuning the piano or guitar. For digital or hybrid modules like The Centre the calibration process is to connect a well calibrated source of pitch that will provide stable value of voltages at two or more reference points. Usually two points are enough. The software will recalculate all the values and will apply proper algorithms to always generate correct pitch for voltages.

## 2.3 Calibrating The Centre

The Centre needs two voltages separated by 2V. In other words, the centre needs a Control Voltage (CV) for two C notes separated by 2 octaves. Ideally that would be C1 and C3 but many current MIDI keyboards supply only voltages between 0V and 5V which translates to C2 and C8. Yep, there is no standard for V/Oct assignemnt of voltage to notes so at 1V/OCT we assume that C2 is 0V. It does not matter anyway, because every V/Oct input has Octave and Note correction anyway.

To calibrate V/OCT inputs press two middle buttons (button 2 + 3) and it will bring System Menu. From there select "Calibrate" and press encoder down (select).

Now every channel can be calibrated individually or 4 inputs at the same time. Use encoder to select channel or all channels (when calibrating all channels use signal splitter to send CV voltage for calibration (pitch) to all inputs).

Now follow instructions on screen. First send any note from your keyboard except note C (we calibrate by C notes) and press Start (button 1). Now press any low octave C note (lets say C3) wait 10 seconds for next instructions, move two octaves up and send note C5. Wait 10 seconds and your unit is fully calibrated.

Now you can Save your calibration settings and enjoy your fully calibrated unit.

# What's In The Box

Your Eurorack module comes packaged in the box together with:

Accessory bag

There is tiny bag included with your The Centre.

Inside this bag you will find:

1. 8 small caps for knobs
2. 4 small caps for buttons
3. 5 sets of washer + nut

[https://github.com/1V-Oct/3318\\_the\\_centre\\_releases.wiki/images/accessory\\_bag.jpg](https://github.com/1V-Oct/3318_the_centre_releases.wiki/images/accessory_bag.jpg)

The Buttons

You might want to alternate between black and white buttons. It is very easy to change button caps. Just use pliers to remove the cap and put the other one.

Knobs Caps

The 8 "attenuator" knobs are naked because if you want to change face plate those caps sit very tight and they are hard to remove therefore I haven't installed them. It's very easy to install. Just position the pot in centre and put the cap pointing to the top. Thats it.

Washer + Nut Sets

I haven't put all those under big Level knobs. They are not necessary because there is no pressure applied and 4 already secures plate well. If you want all of them attached that's why they are inside this bag.



# BPM

Many modules require clock to ensure synchronisation to given time interval requirements. Whether this is simple Random Note Generator (RNG) or Low Frequency Shaper (LFS) the supplied clock ensures that duration of quarter note in one module equals duration of quarter note in another module. There are multiple standards or just ad-hoc designs defining different number of clock pulses per beat. The most popular one is MIDI standard that established 24 clocks per quarter note (24 PQN).

The Centre by default uses 24 clocks per quarter note (beat) but this value can be changed globally for all modules. That setting can be adjusted in Global Settings and can vary between 24, 12, 6, 4, 3, 2 and 1 clocks per beat.

- *When using MIDI to control The Centre it is recommended to keep 24 CPQN (Clocks Per Quarter Note) to adhere to MIDI standard.*

Beats Per Minute (BPM) is a measure used in electronic music to define time interval of music. Beat in electronic music is equivalent to quarter note and there are 4 quarter notes to bar (4/4 tempo). The Centre by default configures all modules to work at 120BPM. 120 Beats Per Minute that's 120 beats per 60 seconds and in the end one quarter note duration is half second or 500 milliseconds (ms). To change default tempo it is necessary to provide modules with clock input via Clock CV (CLK) on modules inputs. The clock can be submitted via CVY input, MIDI input or generated internally via Clock Module (CLK). By adding CLK module we can generate Clock for other modules derived from user defined tempo.

## **Part II**

# **System Overview**

# Module Overview

The Centre is a multipurpose Eurorack module designed around its primary function being Wavetable Synthesis and incorporating large number of functional modules that can operate on its own or can be connected to create internal patches or presets. The Centre inspired by fully fledged WaveTable Synthesisers in VST format brings that functionality into Eurorack module however with the scalability and flexibility of modular synth.

The Centre can be considered as modular-in-modular as it allows creating instances of virtual modules and making connections between them either through audio or modulation paths. Every module can also be controlled via external CV inputs or physical knobs.

## 6.1 Patch

The Centre revolves around idea of patch (or preset) which is basically a set of different modules cascaded on top of each other connected via shared outputs and inputs and sharing the physical knobs and CV inputs and audio outputs.

## 6.2 Physical Connectors

The Centre features:

- **Rotary Encoder** - used for navigating menu system
- **16 knobs** - 8 Level and 8 Attenuators
- **4 V/OCT inputs** - calibrated inputs taking 1V per Octave to control pitch of multiple modules
- **4 CVY inputs** - low latency gate inputs that result only in binary inputs (gate set and not set)
- **8 CV inputs** - higher latency analog inputs with -10V to +10V range.
- **4 VOUT inputs** (DC coupled inputs capable of outputting audio frequency and very low frequencies for modulating external modules)

### 6.2.1 Rotary Encoder

Rotary Encoder allows navigating through all menu systems. Pressing rotary encoder down executes **[SELECT]** function and rotating encoder is referred to as **[ROTATE]**

### 6.2.2 Knobs

All knobs in The Centre have are fully configurable and the differentiation on panel for **LVL** - *Level* and **ATT** - *Attenuator* is just for aesthetic and visual purpose (easier to navigate through configured patch). Every knob is fully assignable to any parameter in patch that can be controlled via knob assignment.

### 6.2.3 V/OCT

V/OCT inputs are preconfigured and calibrated to take -3V to 8V Control Voltage and control it to music notes by applying standard formula of 1V per Octave.

- 1V per Octave (1V/Oct) is a term of controlling pitch of oscillators in modular synthesisers via control voltage that steps in linear way and every difference of 1 Volt causes difference in pitch of 1 octave.
- The Centre is calibrated that 0V (or no input) results in note C2 (MIDI note 36)

### 6.2.4 CVY

CVY are simple gate inputs with very low latency (with the worst case being the standard length of audio slice - 2.7ms). CVY inputs are good for external clock and gates. To sync module to external clock only CVY gates can be used. CV gates will result with unacceptable latency.

### 6.2.5 CV

CV inputs are standard inputs taking analog signal in range of -10V to +10V. Those inputs have much higher latency than CVY (Gates) inputs and that latency varies at around 10ms being an average.

### 6.2.6 VOUT

VOUT outputs are physical outputs of patch that are propagated by **OUT** module from VOUT1-4 audio buffers in patch.

■ If multiple modules write to the same audio buffer the output is either mixed or overwritten based on module.

# Patch

Patch (also known as preset) is a set of modules that carry following characteristics:

- Connected via audio inputs and outputs
- Connected internally via CV inputs and outputs
- Having set of knobs assigned to control performance
- Having set of assigned external CV inputs (V/OCT, CVY, CV) to control performance from external gear

## 7.1 Knobs

In **PATCH EDIT** screen the knobs perform actions as assigned in Module Inputs. That means the knobs can be assigned to perform arbitrary functions like control Cutoff of VCF, Frequency of LFO or attenuation of external envelope that goes to VCA.

■ When navigating to different screens the knobs are no longer assigned to the inputs but they perform a function on the screen. In the modules section every screen has knobs explained. The mapping of the knobs and external CV inputs can be seen in the **PATCH** screen as seen below:

# Audio Outputs

Audio Outputs is a fundamental mechanism of transferring audio between modules and out of The Centre.

## 8.1 Physical Outputs

There are 4 output channels via **VOUT1** to **VOUT4** 3.5mm jacks that output audio level signal. VOUTs are DC coupled thus capable of sending output of utility modules such as LFS (Low Frequency Shaper - free shape oscillator) or ENV (AHDSR Envelope) or even triggers and gates. Those outputs are filled as a copy VOUT1-4 buffers respectively at the end of processing patch.

■ Modules in patch are processed in order for every time slice. If there are two modules writing to the same **VOUTx** they may either overwrite the content of mix it. It is strongly suggested to use **VBuf - Virtual Buffers** whenever possible and only produce final result to **VOUTx** channel

## 8.2 Virtual Audio Buffers

Virtual Buffer Output called in The Centre by shortened name **VBuf** is a concept of transferring audio inside The Centre between modules without occupying outside outputs. VBuf can be understood as a buffer that stores audio that can be then assigned to another module (or a few modules) and further processed individually.

The most important part of VBuf is that every module within The Centre has own VBuf. Furthermore VBuf Output of a module can be used as VBuf input to other modules without being modified.

★ WTO can output to VBuf and then VCF processes WTO input. At the end MIX module takes input from WTO VBuf (original oscillator signal) and VCF Vbuf (filtered oscillator signal) and MIX module can act as Dry/Wet Mixer for those two signals. In such scenario WTO VBuf (oscillator output signal) is being sent unmodified to two modules (VCF and MIX).

Audio Outputs are integral part of almost every module. Audio output of some modules can be configured to **Stereo** or **Mono** while some modules will output only single channel (Mono). Certain modules that take input of **VBuf** will detect if the input is Mono or Stereo and process it accordingly.

## 8.3 Quick Configuration of Audio Inputs and Outputs for Modules

While navigating patch menu press **[EDIT]** button and enter Input/Output configuration mode. In this mode you can directly configure inputs and outputs of selected module by using **ATT1** to configure primary input of module and use **ATT8** to configure output of module.

When the module is configured to output its Audio to Virtual Buffer it will automatically show in the list of Audio Inputs for other modules configuration. The module will have name in inputs as it's own name.

By pressing **[Inputs]** or **[Set]** buttons we can enter Inputs and Settings of selected module.

■ When having two or more modules of same type they automatically get numbered. The numbers will get automatically assigned in ascending order in list.

SD	TL	*	NO	NAME	*	VCA	0	6
LFS						V		
WTO						VIV		
VCF	WTO					VIV		
VCA	VCF					112		
OUT								
END OF PATCH								
System Inputs			Set			Back		

Figure 1: In this example the audio flows from WTO to VCF then to VCA via Virtual Buffers and finally to VOUT1|2

### 8.3.1 Visual Output Presentation in Patch View

- **1, 2, 3, 4** - monophonic VOUTx direct output (via OUT module)
- **V** - monophonic virtual buffer **VBuf**
- **V|V** - stereophonic virtual buffer **VBuf**
- **1|2, 2|3, 3|4, 4|1** - stereophonic VOUTx direct output (via OUT module)

## CV Internal Outputs

Modules in Thw Centre nopt only output Audio Signal (that can be routed inside the module to other modules or outside of the module). Each module outputs Internal CV Signals that can be routed to other modules and used as CV (Control Voltage) Inputs. See: Module Inputs

★ VCO outputs two CV signals, VCO.OSC (output of oscillator) and VCO.RST (reset signal when phase of oscillator resets). The VCO.RST signal can be used as input of another VCO in Inputs: VCO Hard Sync that resets phase of oscillator. This way two VCOs can run in sync.



# Module Inputs

## 10.1 Overview

Each module in The Centre has configurable set of inputs. Each input is composed out up to 4 components. Those components are either **VALUE** or **CONTROLLER**. The most common input is a triplet Level-Attenuator-CV. Level and Attenuator components can be represented either by VALUE or by assigned CONTROLLER (knob). CV component can be assigned to either external CV input (3.5mm jacks labelled CVY (gates) and CV (Control Voltage)) or to the internal output of another modules.

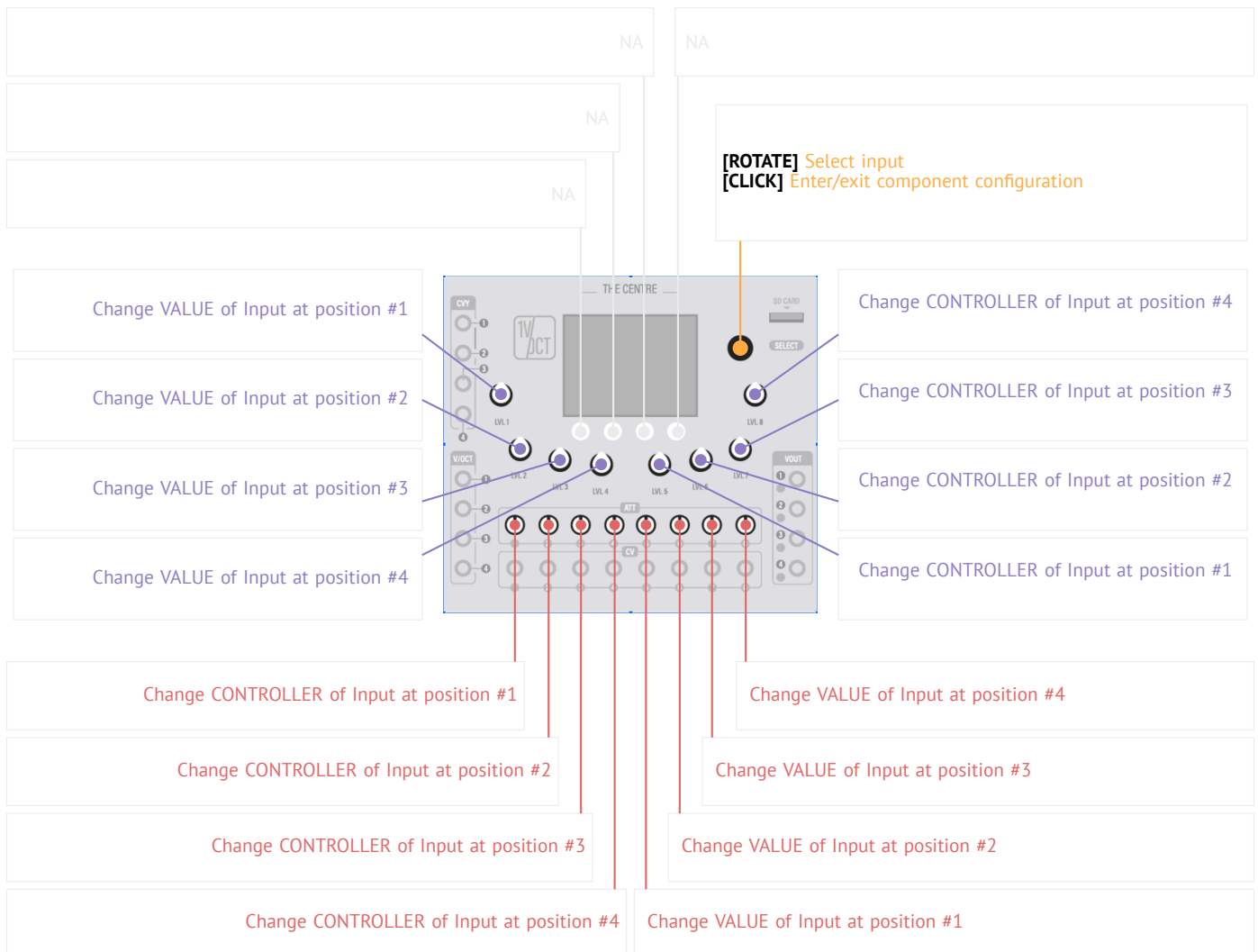
★ A good example of internal CV connection: ENV (AHDSR Envelope) Module and VCA (Voltage Controlled Amplifier) module connected via Input:Modulation of VCA to the output of envelope module Output:ENV.ENV.

## 10.2 Input Configuration Mode

### 10.2.1 Changing VALUE and CONTROLLER in Input Configuration mode

In the Input Screen of each module, we can use encoder (SELECT) to navigate through list of inputs and then use knobs to change VALUE or CONTROLLER of component at the selected Input. Please see below diagram to understand mapping of knobs in the Input Screen.

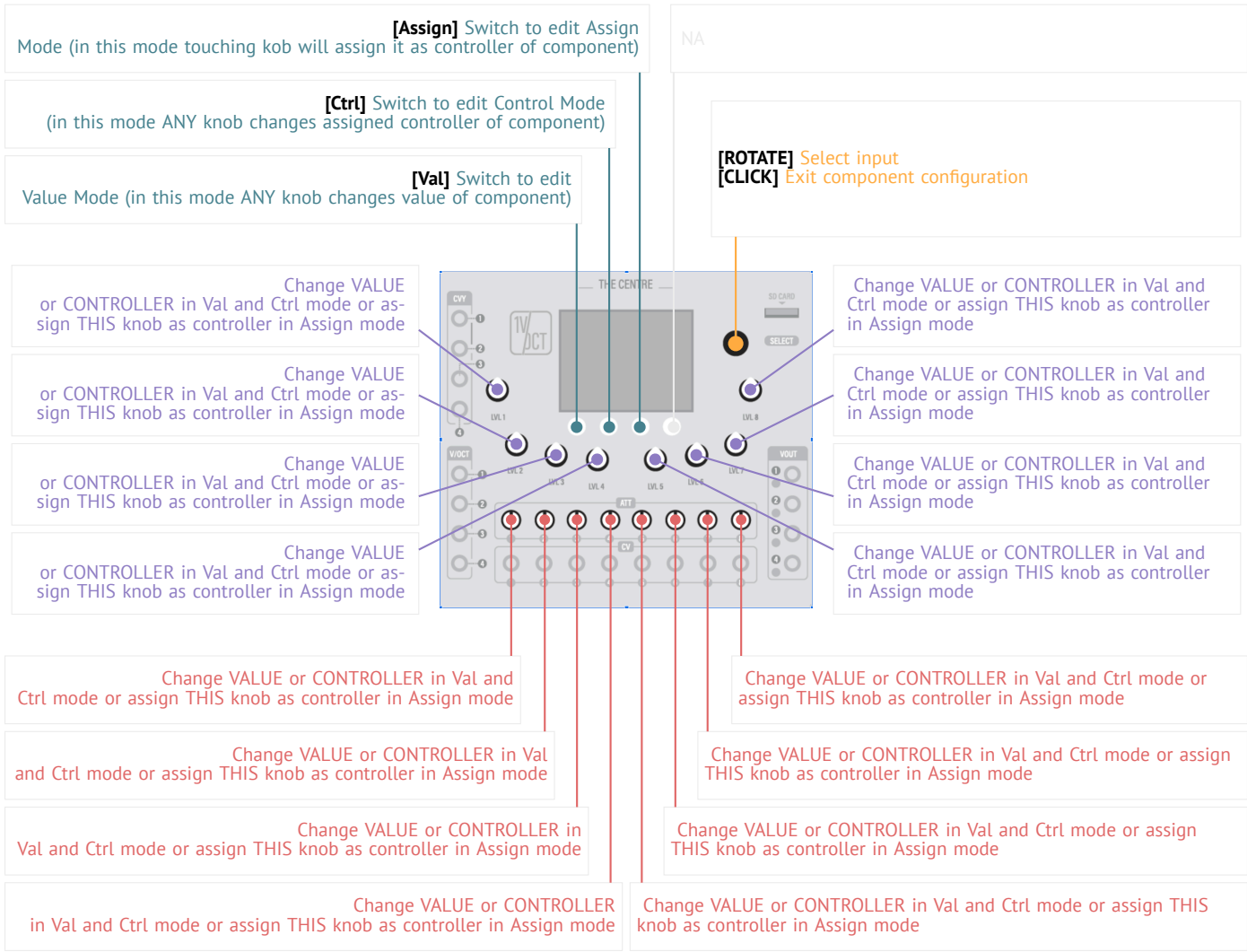
### 10.2.2 Layout of controls in input menu (Input Configuration)



## 10.3 Individual Component Configuration Mode

In Individual Component Configuration Mode each Component of each Input can be configured individually by changing it's Value or assigned Controller.

### 10.3.1 Layout of controls in input menu (Controller Configuration)



## Assigning knobs

Another convenient way of changing knobs assignment is by pressing **Assign** button and switching to individual component setup in Assign Mode. In this mode Encoder (SELECT) navigates through all components of inputs individually and when placed upon Level or Attenuator component it is enough to rotate designated knob slightly to get it assigned to the component.

## Configuring Components

In individual componet configuration pressing buttons Ctrl, Val and Assign switches between operation modes of changing CONTROLLER, VALUE or switching int Assign Mode described above. In those individual mode ANY knob will change either VALUE or CONTROLLER depending on selected mode.

# Pitch Control

## 11.1 Overview

The Centre pitch control affects following modules: WTO, VCO, SMP. Pitch control is generally change of frequency of sound based on configuration of input.

## 11.2 1V/Oct

**1V/Oct** is an abbreviation from "1 Volt per Octave" and is a method of controlling pitch (frequency) of oscillator where increase of Control Voltage (CV) by 1 Volt doubles the frequency of oscillator effectively increasing pitch by one octave.

There is no clear standard what voltage results in what note therefore between manufacturers of equipment there are clear differences on what 0V (zero Volt) results in.

Some suggest that 0V should give "Middle C" note but it is also unclear what Middle C is even in MIDI standard (note 48 or 60 are the most commonly suggested) or maybe the 0V should result in frequency 440Hz which is commonly used as tuning frequency for A note.

The Centre follows Moog standard and defines 0V (zero Volt) for MIDI Note C4 and ultimately frequency of 261.63Hz (assuming concert pitch A4 is 440Hz).

## 11.3 Note Control in Inputs

The Centre Note Control for Note controlled values is configured in Inputs:NOTE:

SD	TL	wto_test	VCO	8	1
Note					
VOCT	OCT	NOTE	FINE		
--	--	--	--		
36	0	0	0.00		
AM					
LVL	ATT	CV	--		
--	--	--			
1.00	1.00	0.00			
Val	Ctrl	Assign	Back		

### VOCT

1V/Oct input for note control. This input can be either set as value (fixed), get input from external modules or synthesizers by V/OCT 1-4 3.5mm input jacks or get input from any other modules CV output. Mostly the output from internal modules should be named .NTE (like: RNG.NTE Random note generator, note output).

■ VOCT is the only dynamically controlled component that controls pitch

### OCT

Tuning note by octaves +/- 8 octaves

### NOTE

Tuning note by semitones +12 semitones (one octave)

### FINE

Tuning note by cents +/- one semitone

# System Settings

## 12.1 Changing System Settings

To enter System Settings go to System Menu by pressing **[EDIT]** and the **[System]** buttons and then select **Settings** from menu.

**Load Last Patch** On system start (reset) it will load last patch that has been manually loaded by user).

**Reverse Encoder** Change encoder behaviour - select from 4 different settings

**Clocks PQN** Number of clocks per quarter note (beat). Default to 24 CPQN as MIDI standard

**V/OCT Quantise** Turn on quantisation of V/OCT incoming pitch to the nearest semitone - nearest MIDI note

**Fast SD Operation** Turns on 4x speed for SD Card access but might be incompatible with some cheap cards.

**Multi Patch** Starts The Centre in Multi Patch mode aka Set

**Auto Remap Pots** Experimental feature - please ignore

**Overlay Timer** Time in seconds for overlays in menus to stay on screen

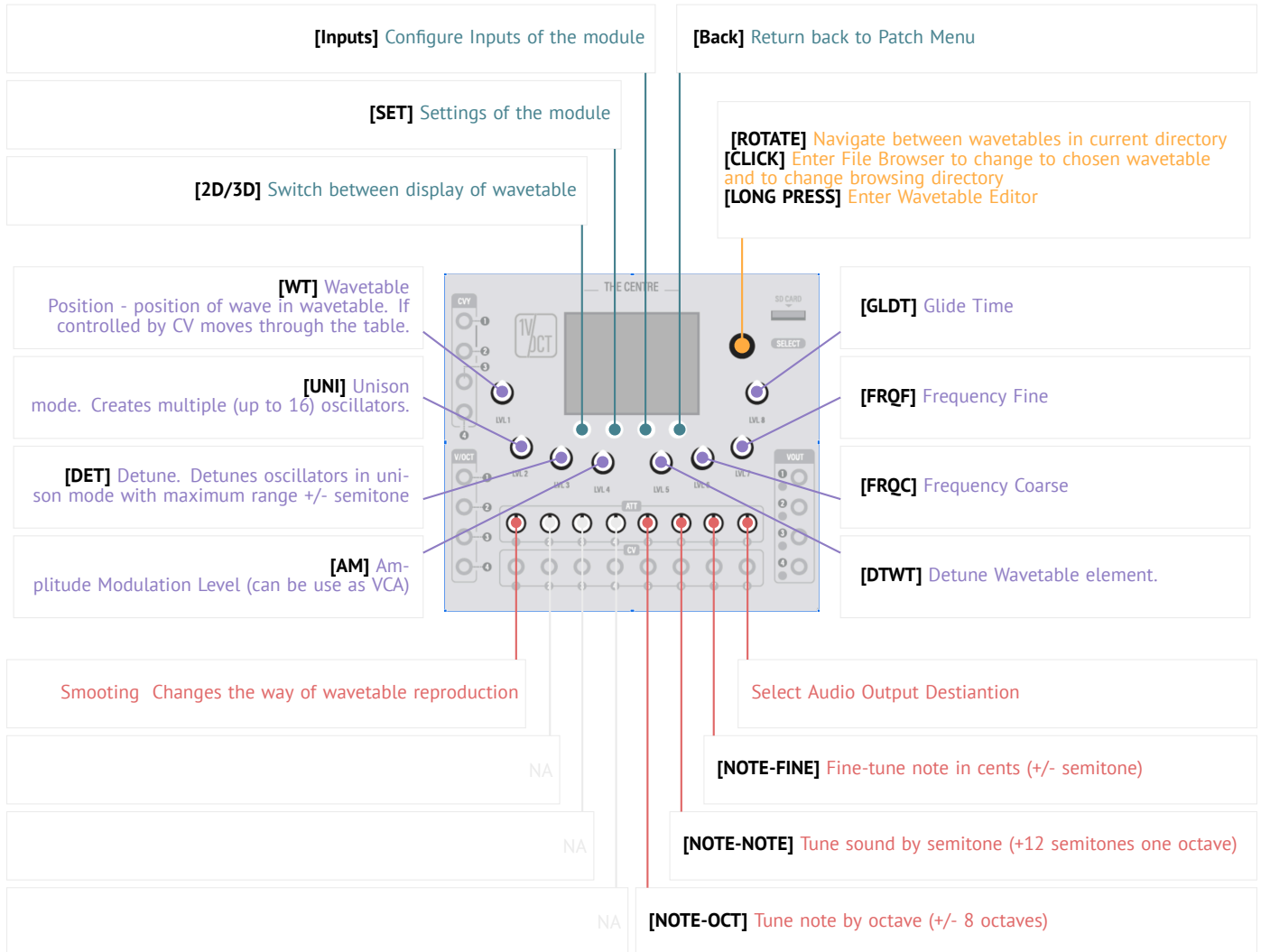
**Part III**

**Module Reference**

# WTO - Wavetable Oscillator

Wavetable Oscillator (WTO) is the main building block of The Centre sound. WTO consists of 1 to 16 oscillators working in Unison mode and reproducing loaded wavetable with possibility to detune pitch of oscillators and detune played waveforms at different positions in wavetable.

## 13.1 Mapping of Controls



# WTO - Wavetable Oscillator

## Settings

### Audio Output *Audio Output of WTO - Stereo*

Selects audio output destination for WTO. All oscillators including Polyphony mode are output through single channel (Mono or Stereo).

See: [Audio Outputs](#)

### Polyphony *Paraphony mode for WTO oscillator*

**1** Paraphony disabled, one pitch control for WTO

**2, 3, 4** Paraphony of 2, 3 or 4 oscillators running different pitch. If oscillator 1 pitch comes from V/OCT1 input then pitch for following oscillators will come from corresponding V/OCT2 - V/OCT4 inputs

### Glide Scaled *Scaling of Glide Time by distance of pitch*

**Off** Glide Time will be same for pitch portamento between any notes

**On** Glide Time will change depending on distance of pitch. Glide Time set by Input Glide Time (see below in Inputs) will be for distance between pitch of one octave. For notes played within same octave will be in fraction of semitones separating pitch of following notes

### Warp Mode *Function used to process waveform during oscillation*

Warp Mode specifies extra function that is used to process waveform according to modulation that is supplied via Input **Warp Mode**. Currently only one function is enabled.

**Off** No processing of waveform

**FM** Warp Mode Input will modulate frequency (FM) of waveform

### NOTE *Pitch control of oscillator*

Note controls pitch or frequency of oscillator.

See: [Pitch Control](#)

**VOCT** 1V/Oct input for note control/

**OCT** Tuning note by octaves +/- 8 octaves

**NOTE** Tuning note by semitones +12 semitones (one octave)

**FINE** Tuning note by cents +/- one semitone

### Unison *Unison mode (1-16 voices)*

Enables unison mode and controls number of oscillators running in parallel.

### Detune *Detune oscillators*

Detunes oscillators in Unison mode by spreading them equally in Detune range. With detune set at maximum the spread range is 2 semi-tones. With odd number of oscillators the centre oscillator will always be following pitch from Note Input. For even number of oscillators there is no oscillator following Note and each oscillator is detuned +/- from the center note.

★ With Unison running two oscillators, detune set to maximum (1.0) and pitch set at note D there is one oscillator playing note C# and one playing note D# (no oscillator is actually playing note C)

## Inputs

# WTO - Wavetable Oscillator

## Inputs

### **Wavetable** *Wavetable position*

Adjusts position of waveform in wavetable. When controlled by CV allows changing texture of sound by scanning through set of waveforms contained in wavetable

### **Reset Osc** *Reset Oscillator CV input*

When connected with CV input and set to high level it resets phase of oscillator to 0. Useful to run oscillators in Sync

### **AM** *Amplitude Modulation*

Amplitude modulation changes sound level. Connected with CV signal of Envelope acts as VCA. Connected with LFO creates ring modulator

### **Detune WT** *Detune Wavetable Position for oscillators in Unison mode*

In Unison mode this parameter will detune position of waveform in wavetable for individual oscillators. Each oscillator will reproduce different waveform thus creating more textured sound.

★ A wavetable with 3 waveforms (triangle, sine, square) and Unison of 3 oscillators and Detune WT parameter set at 1.0, each oscillator will play unique waveform: triangle, sine, square - respectively

### **Frequency Coarse** *Adjust Frequency of oscillator in big steps*

Adjusts frequency of oscillator in range of 0Hz to 100Hz (in frequency mode) or between 1/256 note and 8bar (in BPM/note duration mode)

### **Frequency Fine** *Fine tune frequency of oscillator*

Adjust frequency of oscillator in very small steps

### **Glide Time** *Portamento (pitch glide) time*

Time to glide from one pitch to another upon change of pitch (Note). Glide time is in range of 0s to 1s.

■ This parameter is affected by Setting \*Glide Scaled\*. With Glide Scaled set to OFF the time that it takes to glide between two notes is constant and defined by Glide Time, with Glide Scaled turned ON, time is determined by pitch distance of notes played.

### **Warp Mode** *Modulation parameter for waveform modulator*

Warp Mode is modulation parameter usually via CV to be used with Warp Mode function selected in Settings Warp Mode parameter

## Outputs

### **OUT** *Output of WTO*

Output of wavetable oscillator

### **RST** *Oscillator reset*

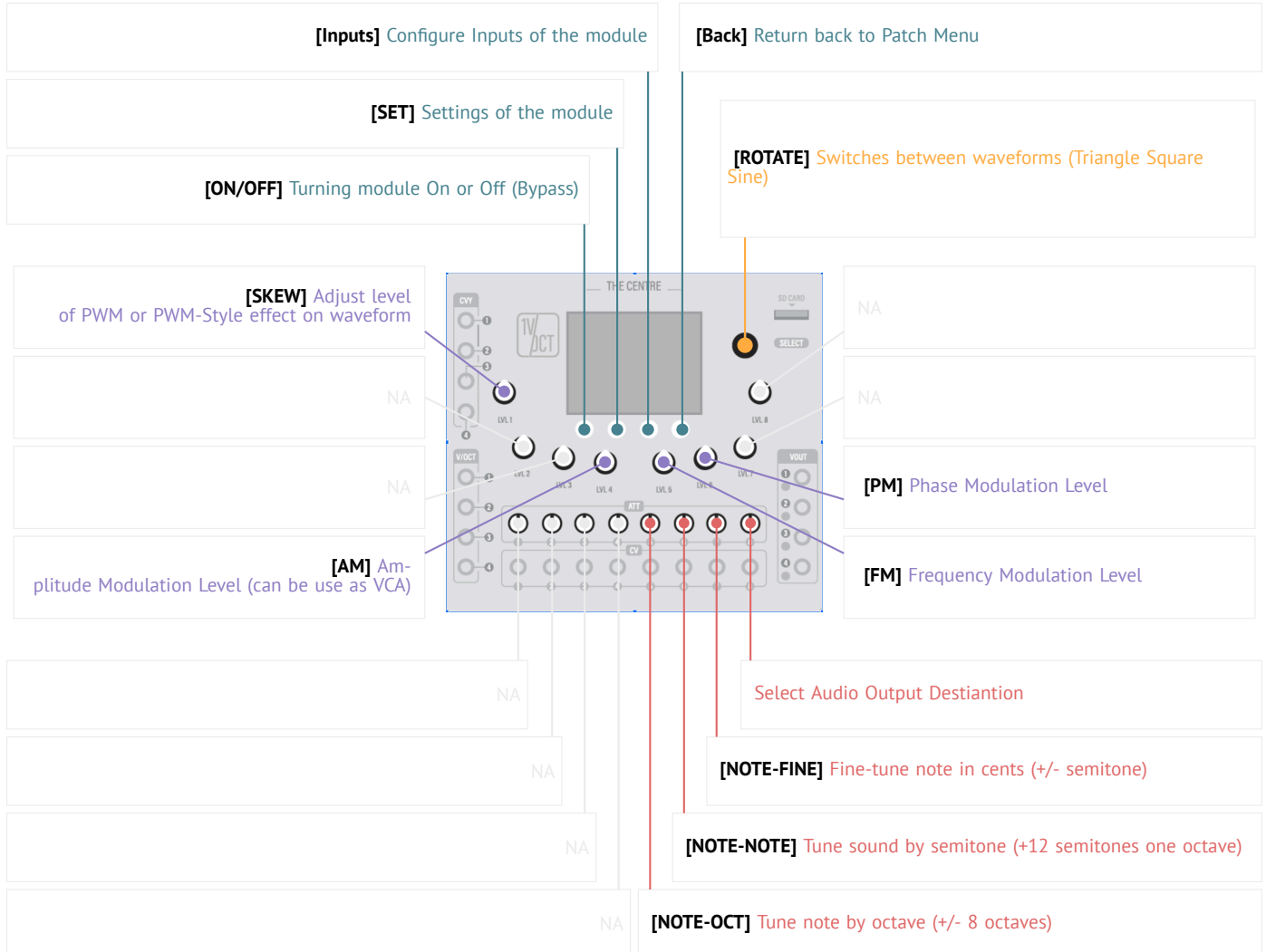
Set to high when oscillator phase resets



# VCO - Voltage Controlled Oscillator

Voltage Controlled Oscillator is a sound source generating oscillations of waveforms.

## 14.1 Mapping of Controls



## 14.2 Waveforms

VCO takes basic shapes: Sine, Triangle and Square and oscillates the single phase of those waveforms at given frequency. Each **Waveform** can be modified by bending it's pivot params with the Input:Skew CV input. Applying Skew parameter to triangle turns it into either *Ramp* or *Saw*. For Square, Skew parameter adjusts PWM (Pulse Width Modulation) which modifies length of square pulse. For Sine, Skew parameter changes the ratio of phase for positive and negative parts of Sine wave.

# VCO - Voltage Controlled Oscillator

## Settings

### Audio Output *Audio Output of VCO*

Selects audio output destination for VCO.

See: [Audio Outputs](#)

### Waveform *Selection of Waveform*

#### Triangle

Standard triangle waveform. The waveform is affected by Input: Skew and changes from Ramp through Triangle to Saw

#### Square

Standard square waveform. The waveform is affected by Input: Skew that modifies PWM (Pulse Width Modulation) of the waveform

#### Sine

Standard sine waveform. The waveform is affected by Input: Skew that modifies it to two assymetric sine phases.

■ Adjusting certain ratios of negative and positive half sines by modifying Skew parameter will add very interestic harmonics that go well with Diode Ladder filter.

### NOTE *Pitch control of oscillator*

Note controls pitch or frequency of oscillator.

See: [Pitch Control](#)

#### VOCT

1V/Oct input for note control/

#### OCT

Tuning note by octaves +/- 8 octaves

#### NOTE

Tuning note by semitones +12 semitones (one octave)

#### FINE

Tuning note by cents +/- one semitone

### AM *Amplitude Modulation*

Amplitude modulation changes sound level. Connected with CV signal of Envelope acts as VCA. Connected with LFO creates ring modulator

### FM *Frequency Modulation*

Frequency modulation is a change of frequency by modulation.

■ Adding a modulator to CV input of FM and having it running at certain ratio of frequency to the actual VCO will create interesting sound texture

### Skew *Modify assymetry of waveform*

Skew affects assymetry of waveform by turning Triangle into Saw or Ram or modyfying PWM of Square wave. See above in Settings: Waveform

### Hard Sync *Reset phase of oscillator*

Resets phase of oscillator and starts oscillating from the beginning of waveform.

★ By connecting two VCOs detuned slightly through VCO1 Ooutput: Reset to VCO2 Input: Hard Sync we can create ver rich texture of simple oscillation

### PM *Phase Modulation*

Phase modulation allows modulator to modify phase of oscillator. Unlike FM (Frequency Modulation) Phase Modulation can operate oscillator in reverse mode by turning its direction.

### OSC *Output of VCO*

Output of oscillator signal

### RST *Oscillator reset*

Set to high when oscillator phase resets

## Inputs

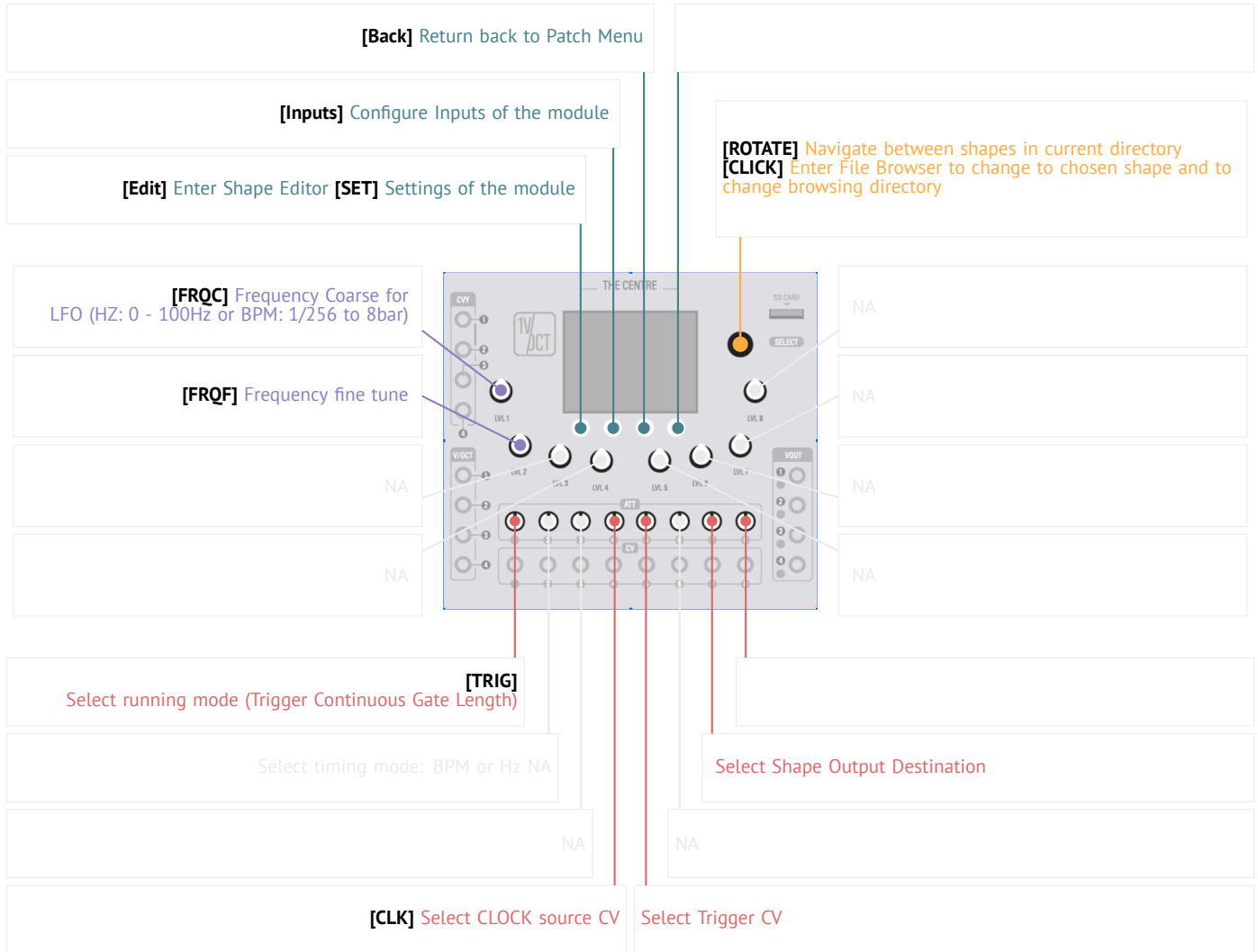
## Outputs

# LFS - Low Frequency Shaper

## LFS Low Frequency Shaper (Low Frequency Oscillator generating irregular shapes)

This module creates low frequency oscillations of irregular shapes. The shapes can be either edited by user or imported from .shp (Serum) and .vitallfo (Vital) formats.

### 15.1 Mapping of Controls



### 15.2 Loading Shapes

To load a shape press **[SELECT]** (Encoder down) and navigate in file browser to directory with shapes and press **[SELECT]** again to load the desired shape. Rotating Encoder cycles through shapes in current directory.

# LFS- Low Frequency Shaper

## Settings

### Running Mode *Select type of running mode of low frequency shape*

- Continuous** Oscillator keeps running and never stops. Gate High (Trigger) signal will reset oscillator phase to 0.
- Trigger** Oscillator runs once upon the gate signal high and runs complete shape once and stops generation when phase of shape ends. Another trigger signal will reset phase to 0.
- Gate length** Oscillator runs as long as gate signal is high. Stops immediately when signal goes to low.

### Timing Mode *Select timing mode for LFS*

Timing for LFS can be either selected to be calculated in Hz for more time based experience or tied to BPM and measured in length of notes or bars.

- BPM** Timing based on note duration
- Hz** Timing based on frequency

### Polarity *Polarity of output*

- Bipolar** Outputs shape in range -5V to +5V. Good for Ring Modulation or FM
- Unipolar** Outputs shape in range 0V to +5V. Good for VCA

## Inputs

### Trigger *Trigger CV input*

Trigger or gate signal that initiates or resets oscillator

### Clock *Clock for timing oscillator's phase duration in BPM mode (otherwise 120BPM is used)*

### Frequency Coarse *Adjust Frequency of oscillator in big steps*

Adjusts frequency of oscillator in range of 0Hz to 100Hz (in frequency mode) or between 1/256 note and 8bar (in BPM/note duration mode)

### Frequency Fine *Fine tune frequency of oscillator*

Adjust frequency of oscillator in very small steps

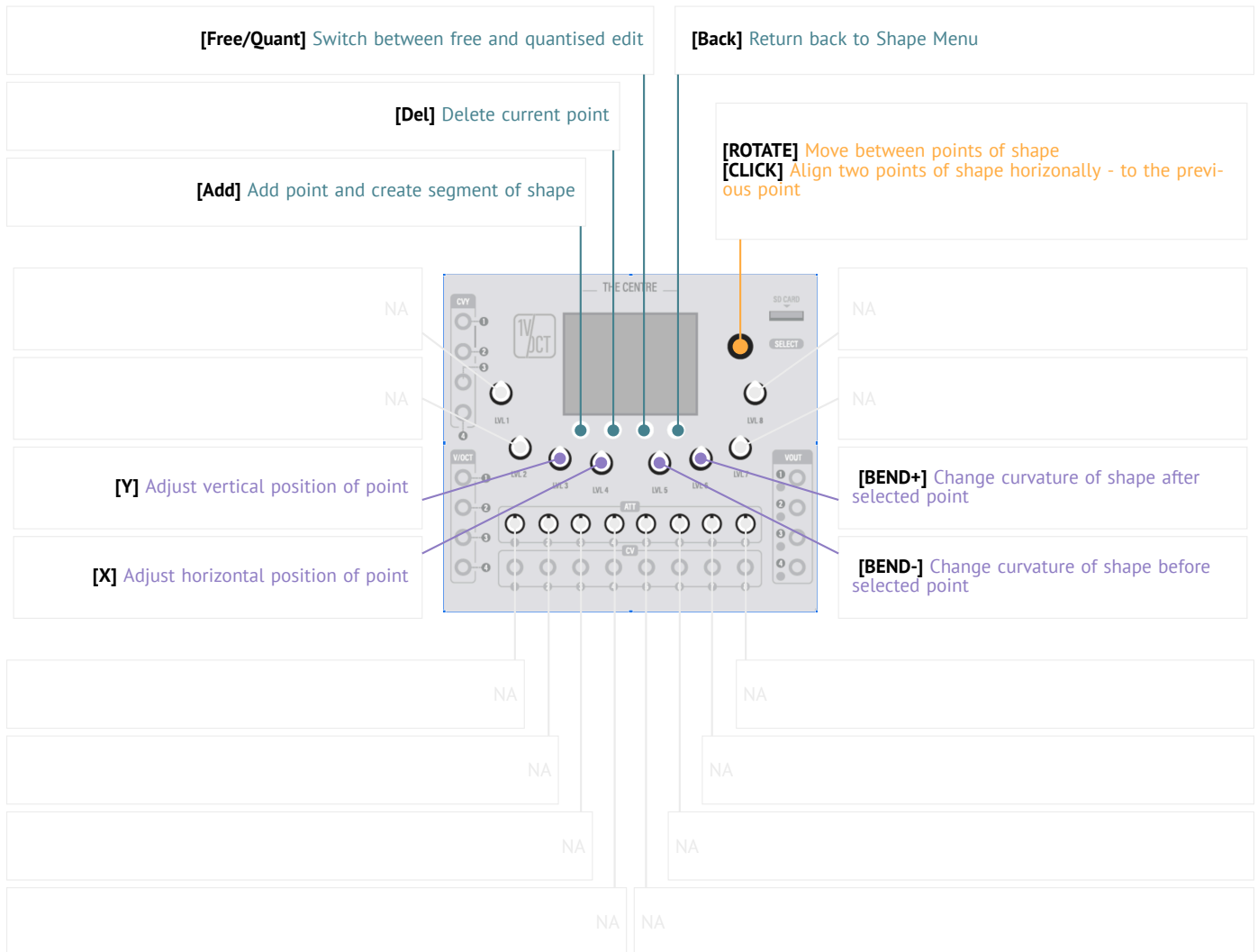
## Outputs

### SHP *Output of*

Output of shape oscillator that can be used to modulate other signals

# LFS - Shape Editor

## 16.1 Mapping of Controls



## 16.2 Editing Shapes

To edit current shape press **Edit** in LFS Screen. Now you can use buttons **Add** and **Del** to add points to the shape (point will be added to between current points - in the middle). Use **LVL3** and **LVL5** to adjust vertical and horizontal position of the point accordingly (NOTE: first and last point horizontal positions cannot be adjusted). **LVL5** will adjust curvature of shape segment prior to selected point and **LVL6** will adjust curvature of segment after selected point.

## 16.3 Position Quantisation

By pressing button **Free** you can change the mode to Free Editing and then by pressing button **Quant** (the same button) you switch mode to Quantised editing where vertical positions are quantised (aligned) to division of 12 (simulate semitones) and horizontal positions are aligned to divisions of 32 to provide alignment simulating notes lengths.

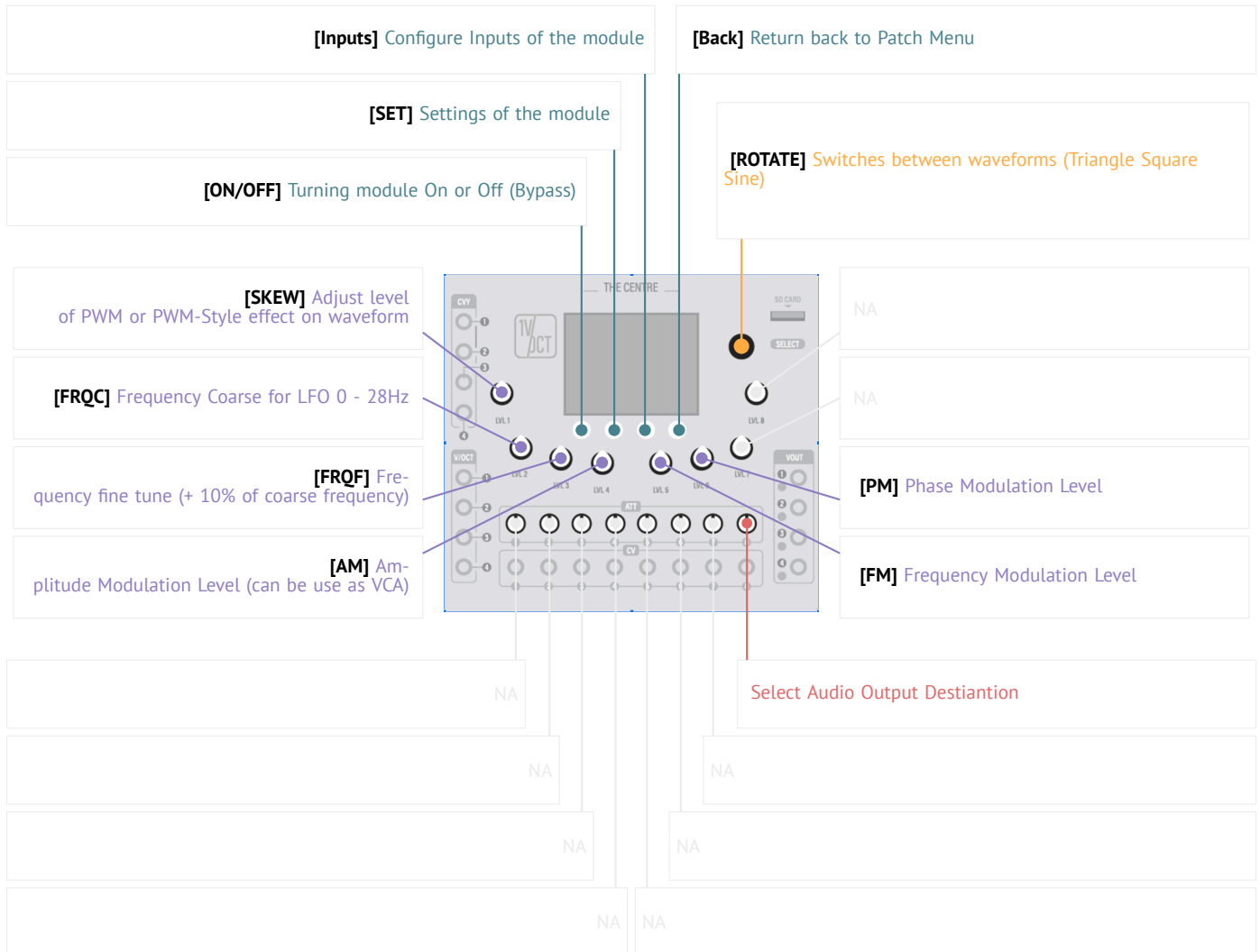
■ Alignment or quantisation does not really align on notes or semitones just helps to have better alignment when the shape is played at the speeds of BPM.

★ Aligned shape can be fed into frequency of VCO or WTO to create arpeggiator effect.

# LFO - Low Frequency Oscillator

Low Frequency Oscillator is a source of modulation that oscillates basic waveforms at very low frequencies (usually below audible level).

## 17.1 Mapping of Controls



## 17.2 Waveforms

LFO takes basic shapes: Sine, Triangle and Square and oscillates the single phase of those waveforms at given frequency. Each **Waveform** can be modified by bending it's pivot params with the Input:Skew CV input. Applying Skew parameter to triangle turns it into either *Ramp* or *Saw*. For Square, Skew parameter adjusts PWM (Pulse Width Modulation) which modifies length of square pulse. For Sine, Skew parameter changes the ratio of phase for positive and negative parts of Sine wave.

# LFO - Low Frequency Oscillator

## Settings

### Audio Output *Audio Output of LFO*

Selects audio output destination for LFO.

See: [Audio Outputs](#)

### Waveform *Selection of Waveform*

#### Triangle

Standard triangle waveform. The waveform is affected by Input: Skew and changes from Ramp through Triangle to Saw

#### Square

Standard square waveform. The waveform is affected by Input: Skew that modifies PWM (Pulse Width Modulation) of the waveform

#### Sine

Standard sine waveform. The waveform is affected by Input: Skew that modifies it to two assymetric sine phases.

■ Adjusting certain ratios of negative and positive half sines by modifying Skew parameter will add very interesting harmonics that go well with Diode Ladder filter.

### NOTE *Pitch control of oscillator*

Note controls pitch or frequency of oscillator.

See: [Pitch Control](#)

#### VOCT

1V/Oct input for note control/

#### OCT

Tuning note by octaves +/- 8 octaves

#### NOTE

Tuning note by semitones +12 semitones (one octave)

#### FINE

Tuning note by cents +/- one semitone

### AM *Amplitude Modulation*

Amplitude modulation changes sound level. Connected with CV signal of Envelope acts as VCA. Connected with LFO creates ring modulator

### FM *Frequency Modulation*

Frequency modulation is a change of frequency by modulation.

■ Adding a modulator to CV input of FM and having it running at certain ratio of frequency to the actual LFO will create interesting sound texture

### Skew *Modify assymetry of waveform*

Skew affects assymetry of waveform by turning Triangle into Saw or Ram or modifying PWM of Square wave. See above in Settings: Waveform

### Hard Sync *Reset phase of oscillator*

Resets phase of oscillator and starts oscillating from the beginning of waveform.

★ By connecting two LFOs detuned slightly through LFO1 Ooutput: Reset to LFO2 Input: Hard Sync we can create ver rich texture of simple oscillation

### PM *Phase Modulation*

Phase modulation allows modulator to modify phase of oscillator. Unlike FM (Frequency Modulation) Phase Modulation can operate oscillator in reverse mode by turning its direction.

### OSC *Output of LFO*

Output of oscillator signal

### RST *Oscillator reset*

Set to high when oscillator phase resets

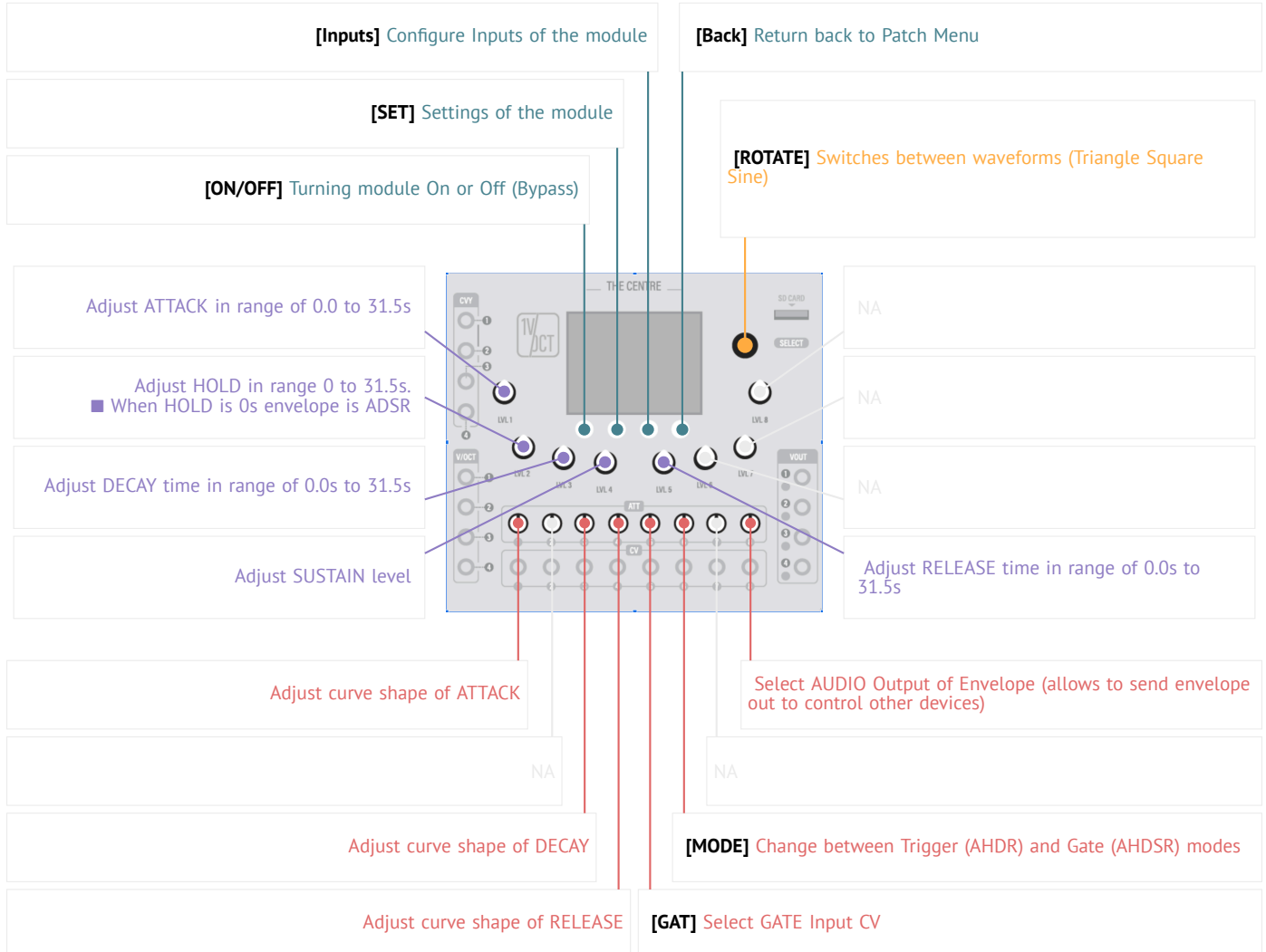
## Inputs

## Outputs

# ENV - Envelope Generator

Envelope Generator produces a modulation signal that in its substance is of raise and decay at the given rates.

## 18.1 Mapping of Controls



## 18.2 Operation

Envelope Generator produces AHDSR (Attack, Hold, Decay, Sustain and Release) or AHDR envelopes that are Control Voltage signals to modulate parameters of modules in the synthesiser (usually, but not limited to, Amplitude of ENV and Cutoff Frequency of VCF).

## 18.3 Envelope Stages

- A: Attack** Attack stage is first and the raising stage of signal from 0 to 100% volume of signal during given period of time.
- H: Hold** Hold stage keeps signal at the highest volume level for a given period of time
- D: Decay** Decay stage determines the length of time the sound volume will be dropping from **Hold** level to **Sustain** level. The Decay stage length is given as period of time.
- S: Sustain** Sustain stage determines the volume level at which sound will keep playing before the gate signal that is relative to played note length keeps high signal. Sustain stage unlike other parameters of envelope is not given in duration of time but rather as a level of volume.
- R: Release** Release is the final stage of sound when it either slowly decays or rapienv ends. Stage is defined as period of time.



# ENV - Envelope Generator

★ If the **Gate** signal length (played note length) is shorter than combined length of **Attack** + **Hold** + **Decay** then some of those stages will be skipped all the way to Sustain stage as the Sustain stage starts when **Gate** signal becomes low. Therefore if **Trigger** signal is applied instead of **Gate** signal then all stages (A+H+D) will be skipped and the envelope will be limited to only Release stage.

## 18.4 Envelope Types

The most popular envelopes out there are ADSR (Attack, Decay, Sustain and Release) - this is envelope designed for Gate signal of variable length that is relative to length of played note. The envelope upon raise of gate signal will execute Attack and Decay stages and will keep amplitude at Sustain level until gate signal turns low (note length ends) and the release stage will help amplitude to continue decay. On the contrary, AD envelopes are designed for Trigger signal and they execute both Attack and Decay stage regardless of note length. The length of note is dictated by length of Attack and Decay stages.

**Hold** stage in **AHDSR** envelope is just extension of typical ADSR envelope where the maximum raised stage (after **Attack**) can be sustained for period of time. With Hold stage period length set to 0s AHDSR envelope becomes standard ADSR envelope.

## 18.5 Trigger Mode

Envelopes can be generated in two modes which can be activated by switching Setting:Mode between **Gate** and **Trigger**.

**Gate** In this mode the envelope is based on the length of provided gate signal being high. When the gate is in high mode the envelope processes through AHD stages and stays in Sustain stage until gate closes (signal turns low) and the Release stage is being processed

**Trigger** In this mode the envelope is always fully processed however without Sustain element. The envelope is beign triggered by high Trigger (Gate) signal and regardless of length of that signal will execute AHDR stages and the Sustain part will be ommited.

★ This mode is ideal to emulate older AD envelopes by setting stages A and D to desired values and every other stage to 0

## Settings

### Audio Output *Output of ENV*

Audio Output of processed signal by ENV. Signal has amplitude modulated by sum of two modulators: Level and Sidechain.

See: [Audio Outputs](#)

### Audio Input *Audio signal to be modulated*

Audio signal (can be any signal) that will have amlitude (level) modulated with modulator Input: Level

### Mode *Envelope Mode*

Mode switch between AHDSR (with Sustain) and AHDR (no sustain) envelopes

**Gate** The envelope lasts as long as the gate signal is open (high) and sustains volume at the Sustain level

**Trigger** The enevelope only lasts as long as sum of stages AHDR and never keeps note sustained

## Inputs

### **Attack** *Attack Stage*

Attack is the raising first stage of envelope. Duration between 0s and 32s

### **Hold** *Hold Stage*

Hold is the full volume level sustained second stage of envelope. Duration between 0s and 32s

### **Decay** *Decay Stage*

Decay is the falling third stage of envelope going from full level of volume to level set . Duration between 0s and 32s

### **Sustain** *Sustain Stage*

Sustain is the level of the sound after AHD stages at which sound is kept as long as Gate is open (high). Level of sound in decibels

### **Release** *Release Stage*

Release is the final stage where the sound decays to level 0. Duration between 0s and 32s

### **Gate** *Gate CV Source*

Gate or Trigger signal to start and sustain envelope generation

### **Attack Curve** *Exponent of Attack stage*

Attack Curve is the exponent of the attack stage to modify stage to exponential rather than linear

### **Decay Curve** *Exponent of Decay stage*

Decay Curve is the exponent of the Decay stage to modify stage to exponential rather than linear

### **Release Curve** *Exponent of Release stage*

Release Curve is the exponent of the Release stage to modify stage to exponential rather than linear

## Outputs

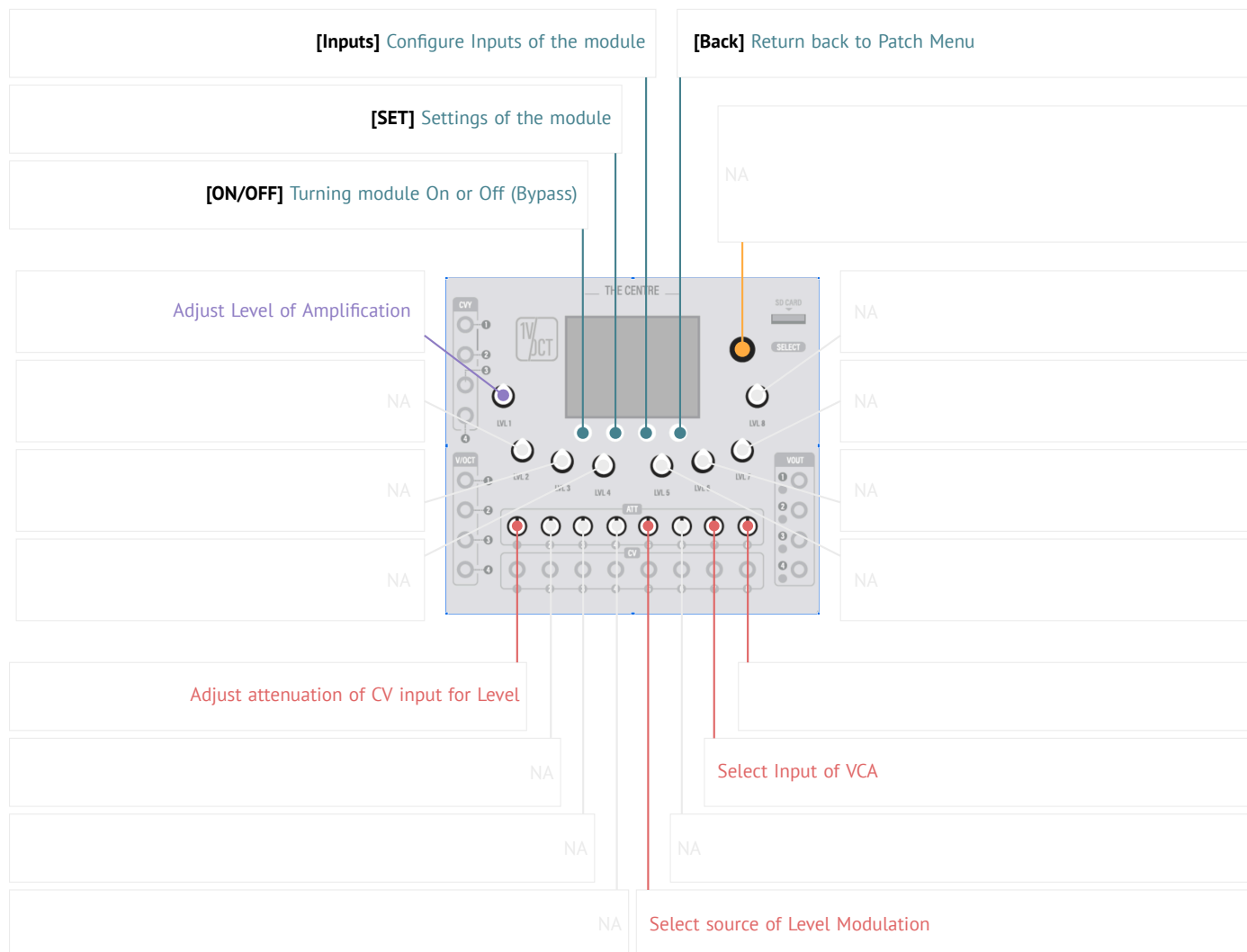
### **Envelope** *Output of Envelope*

Output of Envelope signal to be routed to other modules as a Control Voltage

# VCA - Voltage Controlled Amplifier

Voltage Controlled Amplifier modulates level of signal with level of modulator.

## 19.1 Mapping of Controls



## 19.2 Operation

VCA performs simple Amplitude Modulation of those signal resulting in reduced volume (level) of audio signal. The signal coming in Audio Input is then modulated (attenuated) with a modulator taken from Input:Modulation. In fact VCA operates using principle of Amplitude Modulation however it only takes positive unipolar CV signal to modulate Audio Signal.

★ Usually modulator signal is coming from Envelope Generator (ENV) in from of AD, ADSR or AHDSR envelopes and applied to the uadio signal coming from audio source like oscillator will result in the plucking or raising sound.

## 19.3 Sidechain

Sidechain is the technique to duck (reduce) audio signal when another audio signal needs to be more prominent. The sidechain modulator is an envelope of the prominent signal and is substracted from envelope of audio signal coming to VCA to reduce level of the secondary signal.

★ Sidechaining is used to reduce level of bassline when kickdrum is being played. Very often both kickdrum/bassdrum and bassline share the same low frequencies and if mixed together they result in muffled sound. By loweering the amplitude of bassline at the point kickdrum plays, kickdrum envelope is send to sidechain input of VCA and VCA will reduce level of bassline by the amount of sidechain. This way low frequencies of kickdrum will be heard over the bassline.

# VCA - Voltage Controlled Amplifier

## Settings

### Audio Output *Output of VCA*

Audio Output of processed signal by VCA. Signal has amplitude modulated by sum of two modulators: Level and Sidechain.

See: Audio Outputs

### Audio Input *Audio signal to be modulated*

Audio signal (can be any signal) that will have amplitude (level) modulated with modulator Input: Level

### Level *Modulation of sound level*

Modulation of level of sound. Amplitude modulation with Unipolar (only positive) modulator (usually envelope)

### Sidechain *Opposite modulation of level*

Sidechain is a reverse signal to damp the level of primary signal modulation.

★ Usually used to duck the amplitude of bassline when kick drum comes in to remove interfering frequencies

## Inputs

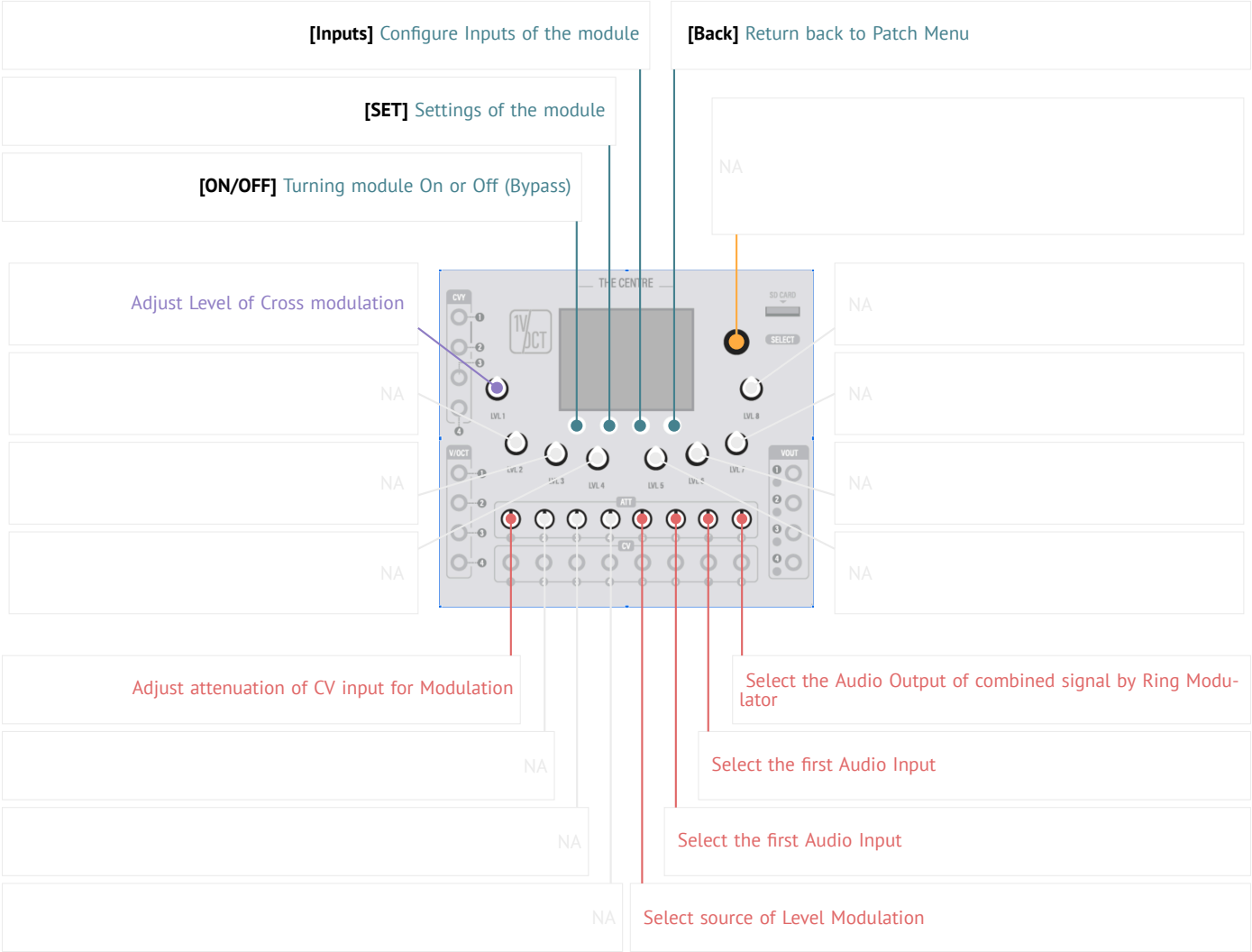
## Outputs

NONE

# BRM - Balanced Ring Modulator

Balanced Ring Modulator takes two Audio signals and mcombines them to gether producing output signal.

## 20.1 Mapping of Controls



## 20.2 Operation

BRM performs cross modulation of two signals by combining Aduio Input 1 with Audio Input 2. The Input:Balance setting is the ratio between those two signals.

# BRM - Balanced Ring Modulator

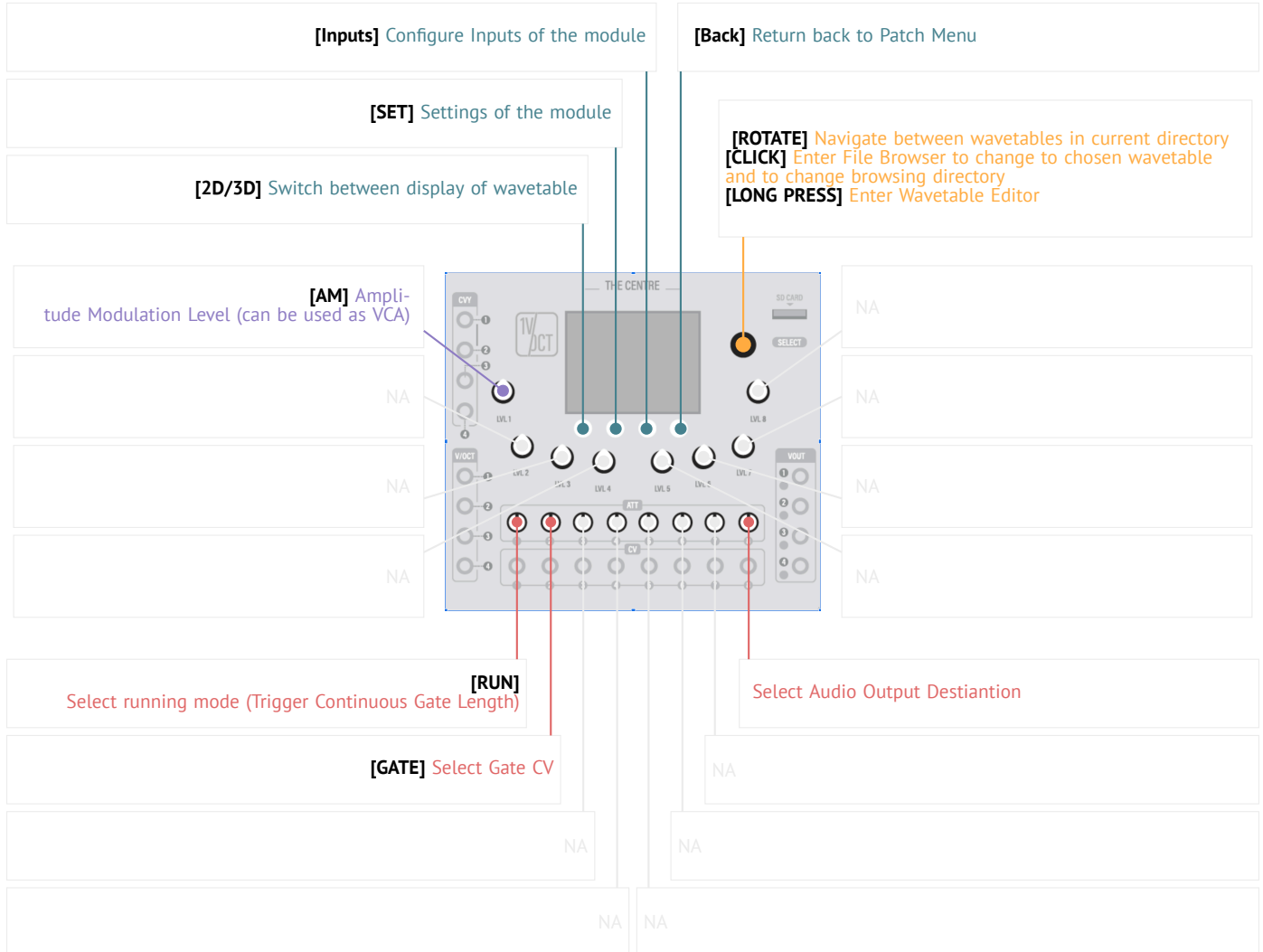
Settings
Inputs
Outputs

- Audio Output** *Output of BRM*  
Audio Output of processed signal by BRM. Signal has amplitude modulated by sum of two modulators: Level and Sidechain.  
See: Audio Outputs
- Audio Input 1** *Input of the first Audio Signal*  
Audio signal (can be any signal) That will be combined with the second signal
- Audio Input 2** *Input of the second Audio Signal*  
Audio signal (can be any signal) That will be combined with the first signal
- Balance** *Balance between signals*  
The ratio between two signals when combining them. The volume level of the signals will be adjusted respectively before combining)
- NONE**

# SMP - Sample Player

Sample Player (SMP) is the sound source generating sound by playing samples at requested pitch.

## 21.1 Mapping of Controls



## 21.2 Operation

Sample Player reproduces samples at given pitch provided by Inputs:Note and triggered by Inputs:Gate. Samples can be loaded from SD Card by pressing encoder (SELECT) down. After the sample is loaded from directory, rotating the encoder allows loading next and previous samples from the same directory.

# SMP - Sample Player

## Settings

### Audio Output *Audio Output of SMP - Stereo*

Selects audio output destination for SMP. All oscillators including Polyphony mode are output through single channel (Mono or Stereo).

See: [Audio Outputs](#)

### Running Mode *Select type of running mode of low frequency shape*

#### Continuous

Sample playback keeps looping. Gate High (Trigger) signal will reset sample playback to start position.

#### Trigger

Sample plays once upon the gate signal high and runs complete shape once and stops at the end position. Another trigger during playback signal will reset playback to start position

#### Gate length

Sample plays as long as gate signal is high. Stops immediately when signal goes to low.

## Inputs

### NOTE *Pitch control of oscillator*

Note controls pitch or frequency of oscillator.

See: [Pitch Control](#)

#### VOCT

1V/Oct input for note control/

#### OCT

Tuning note by octaves +/- 8 octaves

#### NOTE

Tuning note by semitones +12 semitones (one octave)

#### FINE

Tuning note by cents +/- one semitone

### Gate *Gate CV input*

Gate or Trigger signal that initiates or resets playback of sample

### AM *Amplitude Modulation*

Amplitude modulation changes sound level. Connected with CV signal of Envelope acts as VCA. Connected with LFO creates ring modulator

## Outputs

### OUT *Output of SMP*

Output of wavetable oscillator

### RST *Oscillator reset*

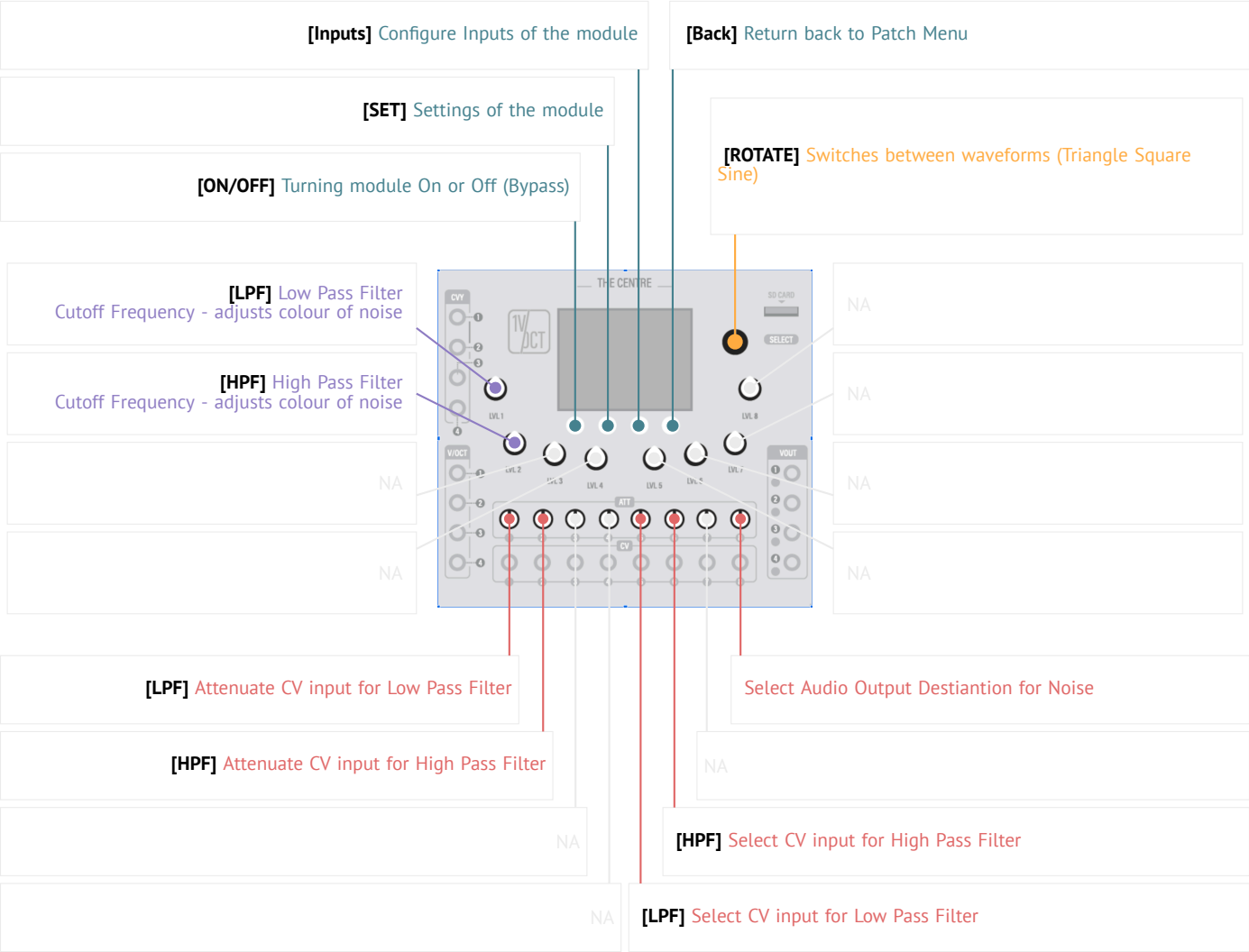
Set to high when oscillator phase resets



# NOI - Noise Generator

Noise Generator is a sound source that produces noise of different colour.

## 22.1 Mapping of Controls



## 22.2 Waveforms

Noise generator produces White Noise as a base for further filtering. With two incorporated filters (Low Pass Filter and High Pass Filter) the colour of noise can be adjusted with the help of CV Inputs.

# NOI - Noise Generator

Settings  
Inputs  
Outputs

**Audio Output** *Audio Output of Noise Generator*  
Selects audio output destination for Noise.  
See: Audio Outputs

**LPF** *Low Pass Filter*  
Low Pass Filter controlled by CV

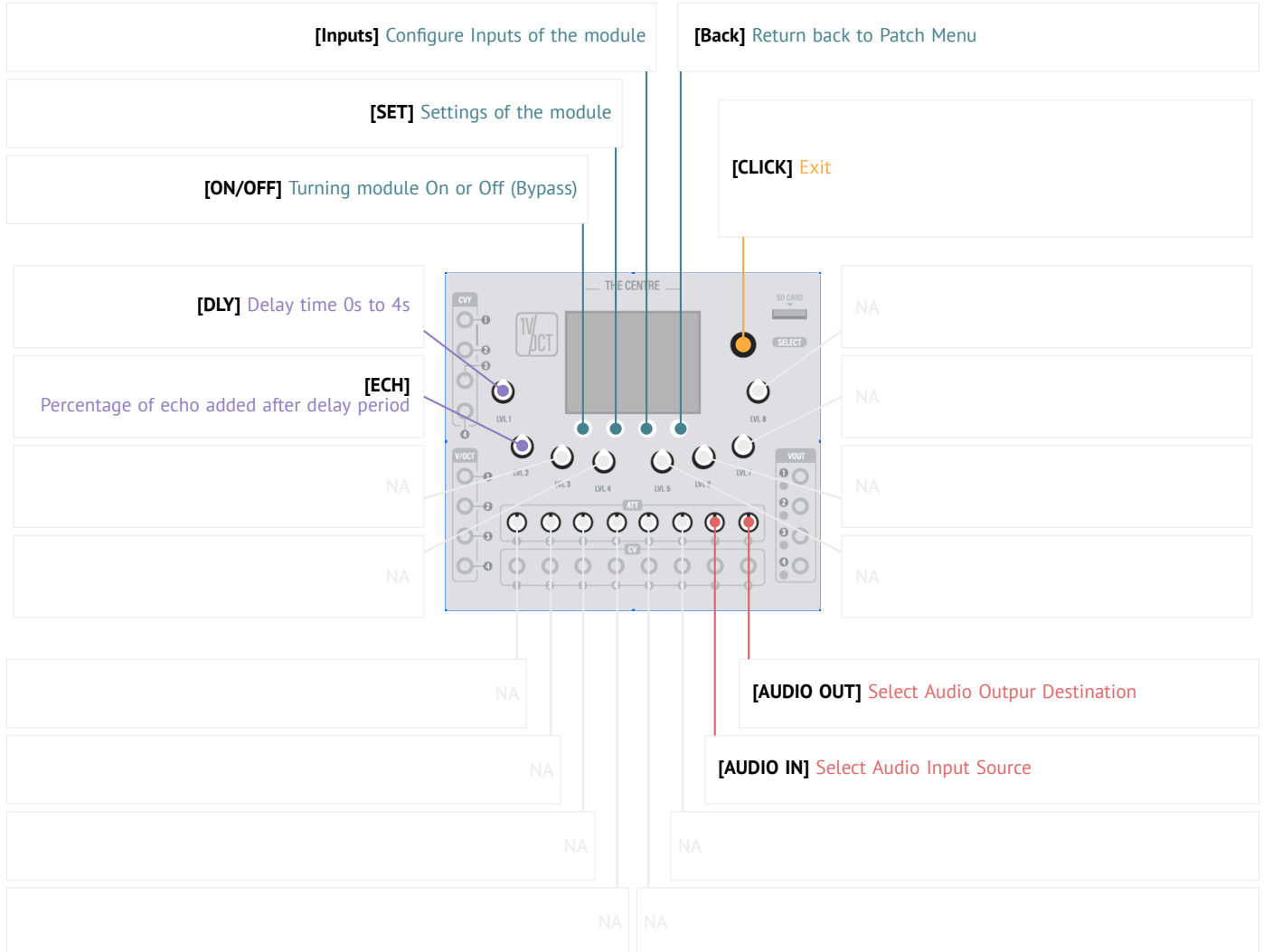
**HPF** *High Pass Filter*  
High Pass Filter controlled by CV

**OSC** *Output of NOI*  
Output of noise signal

## DLY - Delay

Delay (DLY) Effect Processor buffers signal and sums it with buffer to provide delayed echo effect.

## 23.1 Mapping of Controls



## 23.2 Operation

Delay (DLY) Effect Processor takes input signals and buffers it for requested period of time based on Input:Delay parameter and then sums it with buffered signal at attenuated level given by Input:Echo parameter. Delay creates audible echo effect.

# DLY - Delay

## Settings

### Audio Output *Audio Output of Delay*

Audio Output of processed signal by Delay. Audio Output is Audio Input signal summed with buffered delayed signal

See: [Audio Outputs](#)

### Audio Input *Audio Input signal to be delayed*

Audio signal that will be delayed and summed with buffered and attenuated signal to produce echo.

## Inputs

### Delay *Delay Period Length*

Length of Delay period between 0s and 4s. Delay is the size of the buffer.

### Echo *Level of Echo*

Percentage of level of signal to be buffered for further layering

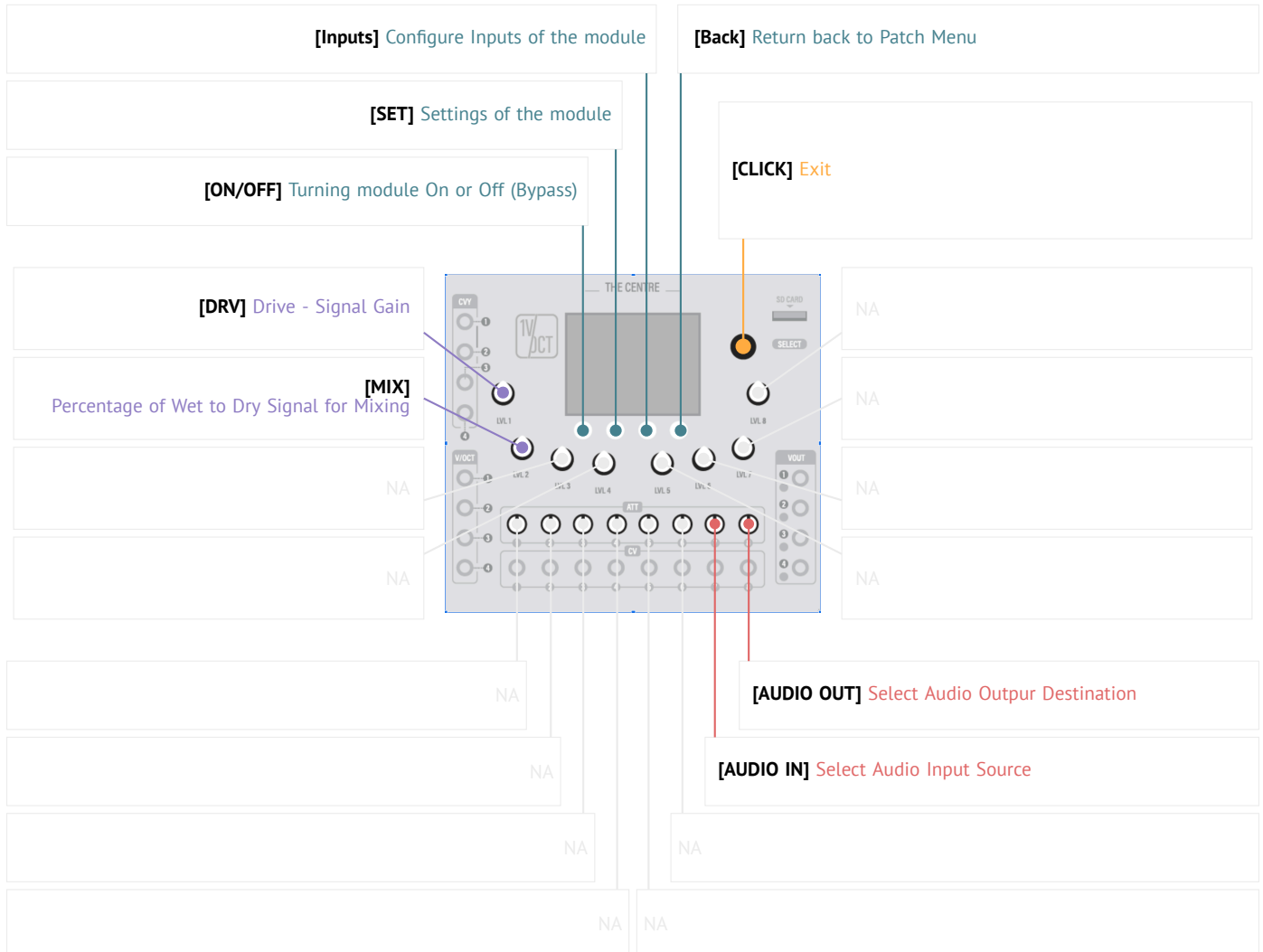
## Outputs

NONE

## DST - Distortion

Distortion (DST) Effect Processor distorts and damages signal to add rough texture.

## 24.1 Mapping of Controls



## 24.2 Operation

Distortion (DST) Effect Processor takes input signal and distorts it by using selected algorithm. Wet Signal (distorted input signal) is then mixed with Dry Signal (unprocessed input signal) by ratio controlled via Input:Mix parameter.

# DST - Distortion

## Settings

### Audio Output *Audio Output of Distortion*

Audio Output of distorted signal. Audio Output is Audio Input signal distorted with selected algorithm  
See: Audio Outputs

### Audio Input *Audio Input signal to be delayed*

Audio signal that will be delayed and summed with buffered and attenuated signal to produce echo.

### Drive *Signal Gain*

Level of gain on Audio Input Signal while processing Distortion making effect more prominent

### Mix *Dry/Wet Mix*

Amount of unprocessed (original) Ainput Signal mixed with Distorted Audio Signal

## Inputs

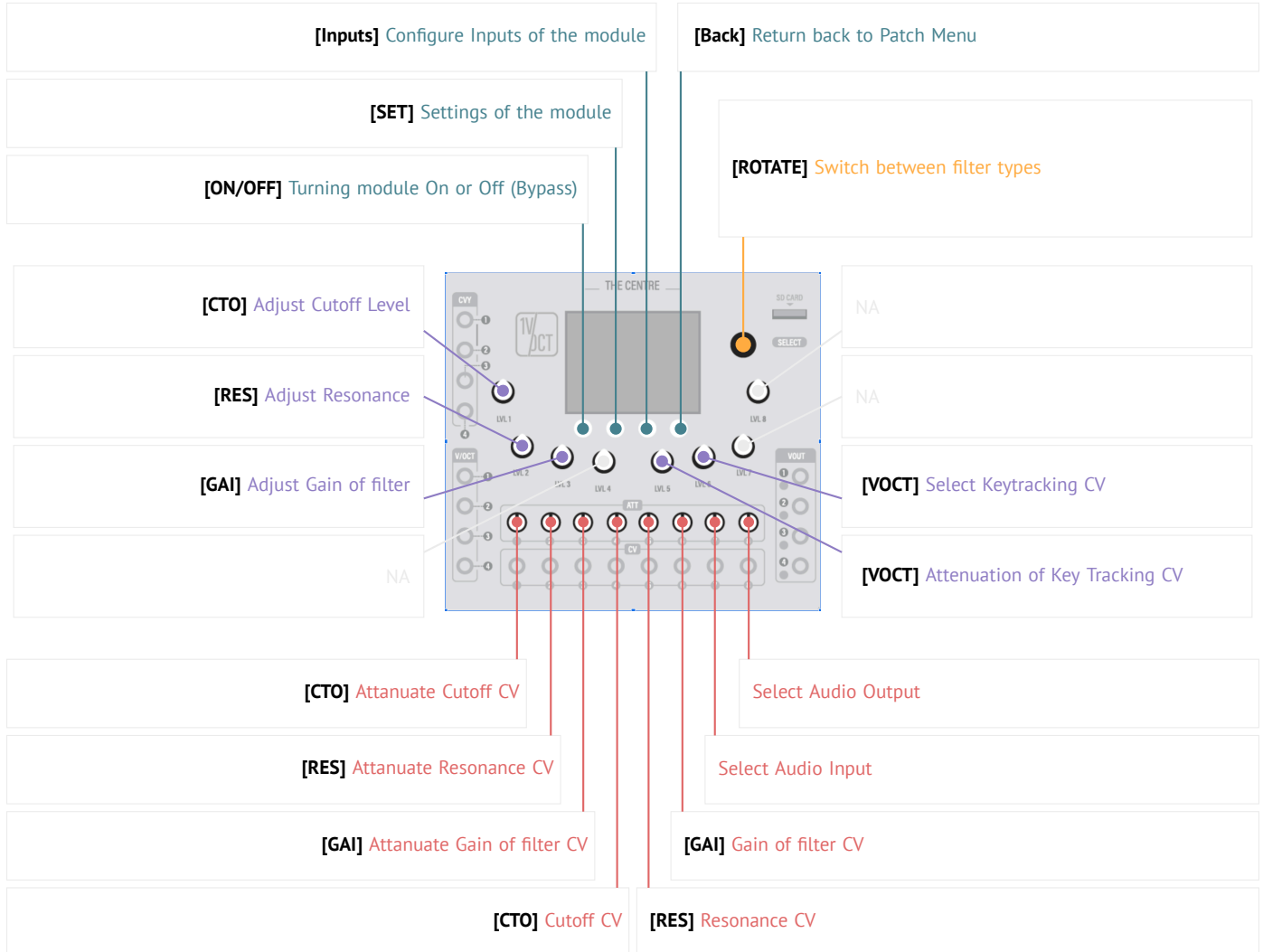
## Outputs

NONE

# VCF - Voltage Controlled Filter

Voltage Controlled Filter (VCF) is a set of digital filters with control of Cutoff, Resonance and Gain.

## 25.1 Mapping of Controls



## 25.2 Operation

Voltage Controlled Filter (VCF) is a set of different filters implemented in digital domain to allow control on limiting frequencies of sound.

## 25.3 Tracking

VCF implements cutoff frequency tracking mechanism that can be controlled in multiple ways however there are 3 standard parameters to control Cutoff Frequency.

Cutoff CV this is part of Input:Cutoff parameter and control voltage either external or internal can be assigned to modulate base cutoff frequency.

Tracking V/OCT and Tracking CV are two voltage controlled parameters that are part of the same input control Input:Tracking. Tracking V/OCT can be assigned to pitch from external (V/OCT jack) or internal (NOTE Output of internal modules) and adjusted with its attenuator. Tracking CV in contrary is to be assigned to CV modulator like Envelope or LFO.

★ Assigning the same V/OCT to Input:Tracking of VCF and to Input:NOTE pitch control of VCO or WTO allows to follow the key of played note creating punchy sounds on change of pitch when using Low Pass filters.

# VCF - Voltage Controlled Filter

## Settings

### Audio Output *Audio Output of VCF*

Selects audio output destination for sound filtered through VCF.

See: [Audio Outputs](#)

### Audio Input *Audio signal to be filtered through VCF*

Audio signal (can be any signal) that will be filtered through different type of filter

### Filter Type *Selection of Filter*

**OP Lowpass, OP Highpass** One Pole filters.

**BQ Lowpass, Highpass, Bandpass, Low Shelf, High Shelf, Peaking, Notch, Allpass** Biquad filters

**DL Ladder** Diode Ladder filter

**MG Ladder** MG Type Ladder filter

**LD Lowpass 12, Highpass 12, Bandpass 12, Lowpass 24, Highpass 24, Bandpass 24** Ladder filters  
12dB and 24dB versions

## Inputs

### NOTE *Pitch control of oscillator*

Note controls pitch or frequency of oscillator.

See: [Pitch Control](#)

**VOCT** 1V/Oct input for note control

**OCT** Tuning note by octaves +/- 8 octaves

**NOTE** Tuning note by semitones +12 semitones (one octave)

**FINE** Tuning note by cents +/- one semitone

### Cutoff *Cutoff Frequency*

Frequency boundary at which filter starts filtering out the signal

### Resonance *Resonance*

A level of suppression or enhancement of signal

### Gain *Pre-Gain of Signal*

Amplification of signal prior to filtering

## Outputs

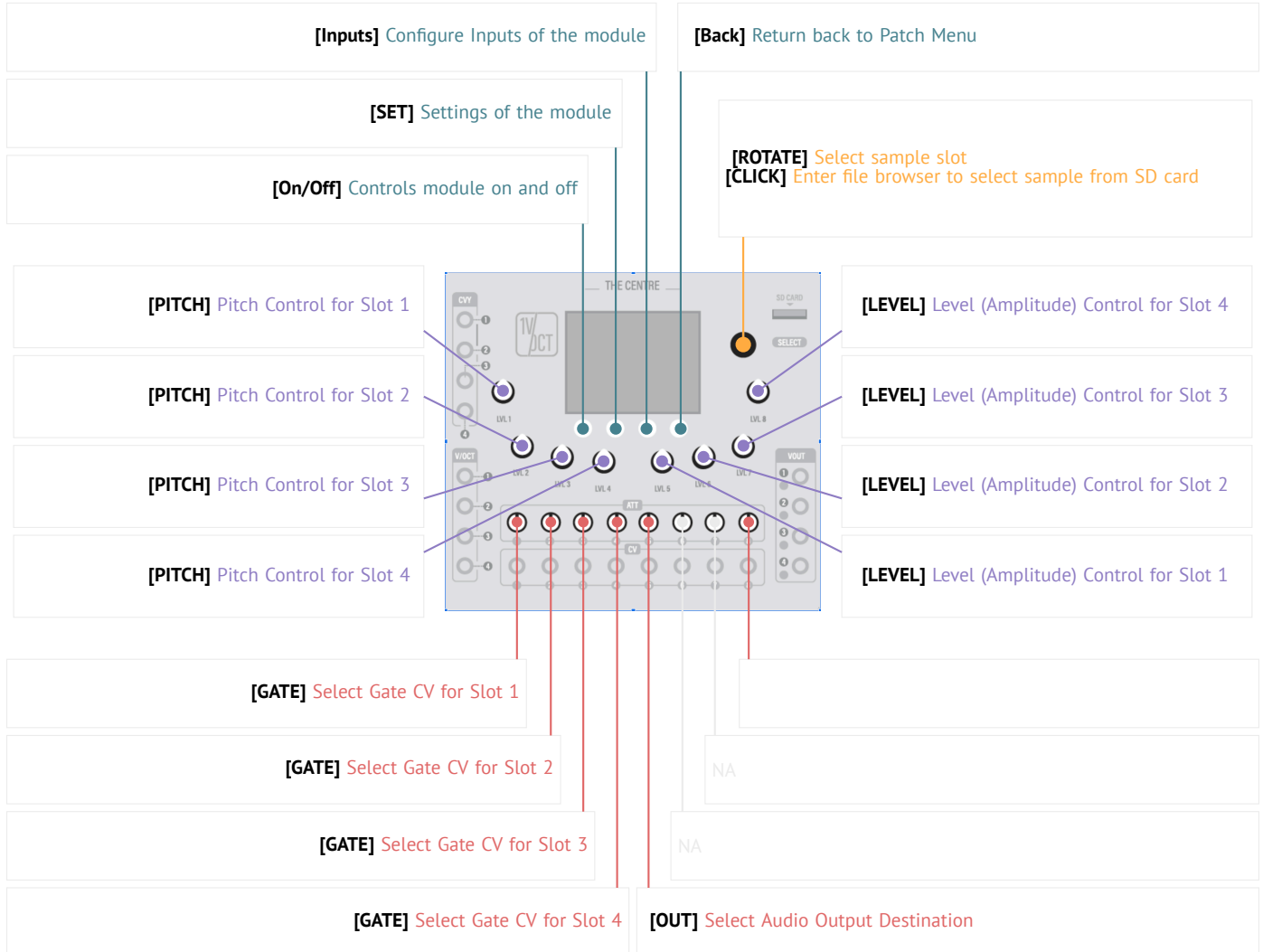
NONE



# DRC - Drum Rack

Drum Rack (DRC) is a 4 slot simple sample player with pitch adjustment and volume level control (amplitude modulation).

## 26.1 Mapping of Controls



## 26.2 Operation

Drum rack consists of 4 slots for loading samples. Upon receiving trigger or gate signal on **GATE** input the sample starts as single shot. There are only two parameters currently to be adjusted and both can be controlled either via setting static value or using CV to modulate them. Those parameters are Volume Level (Amplitude) and Pitch Correction. Pitch correction adjusts pitch of sample by +/- two octaves.

# DRC - Drum Rack

## Settings

### Audio Output *Audio Output of Drum Rack*

Selects audio output destination for DRC. All 4 sample slots are sharing the same audio output where they get downmixed

See: [Audio Outputs](#)

## Inputs

### Gate 1 *Gate CV Input for Slot 1*

Gate or trigger signal that initiates sample playback

### Pitch 1 *Pitch Adjustment for Slot 1*

relative pitch adjustment for sample playback

### Level 1 *Amplitude (Level) for Slot 1*

Level (Amplitude Modulation) for sample playback

### Steps 2 *Number of Steps for Channel 2*

Number of Steps within a channel.

### Beats 2 *Number of Beats for Channel 2*

Number of beats within a channel.

### Offset 2 *Offset of first step for Channel 2*

Offset of first step for channel

### Steps 3 *Number of Steps for Channel 3*

Number of Steps within a channel.

### Beats 3 *Number of Beats for Channel 3*

Number of beats within a channel.

### Offset 3 *Offset of first step for Channel 3*

Offset of first step for channel

### Steps 4 *Number of Steps for Channel 4*

Number of Steps within a channel.

### Beats 4 *Number of Beats for Channel 4*

Number of beats within a channel.

### Offset 4 *Offset of first step for Channel 4*

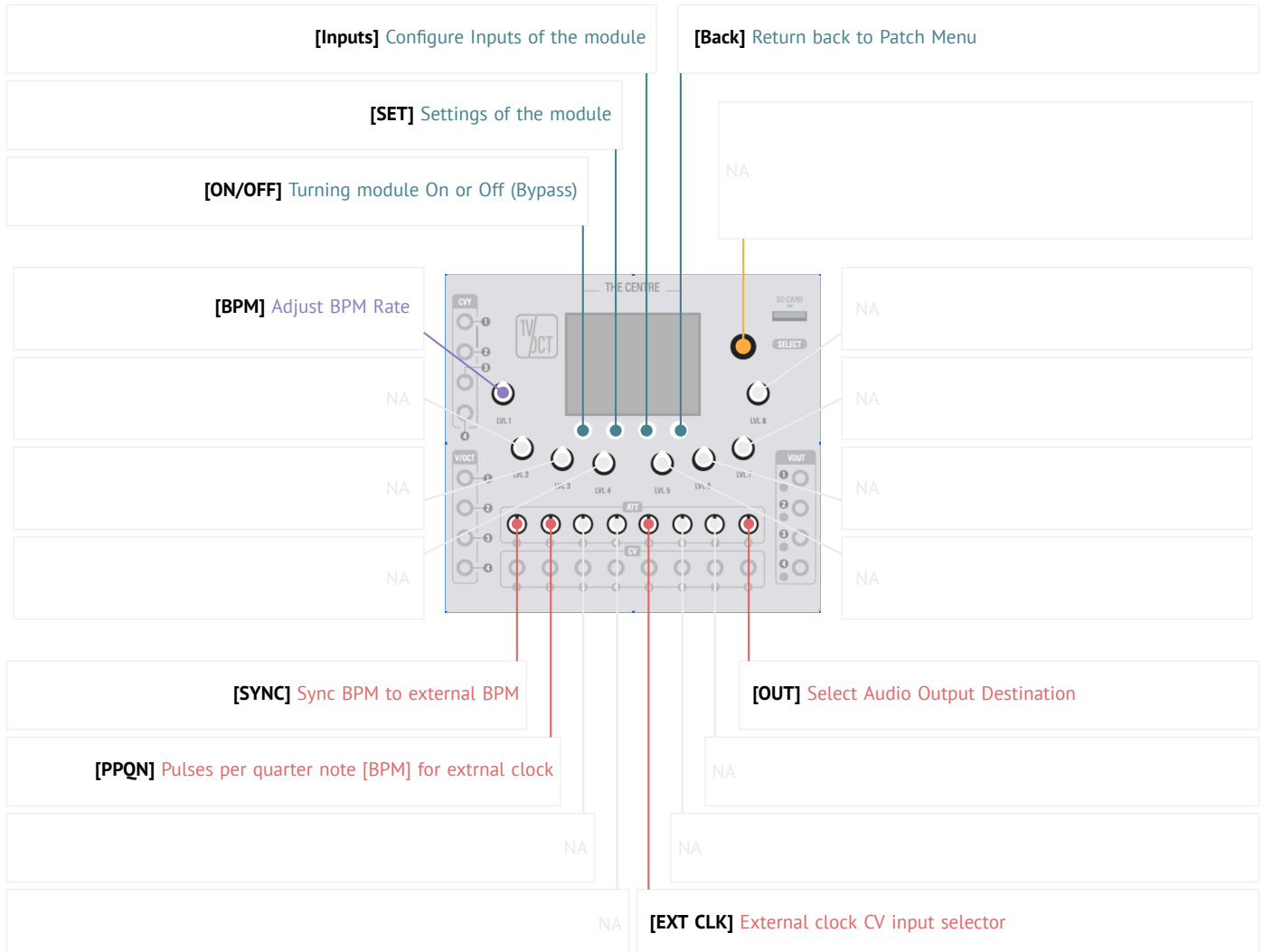
Offset of first step for channel

## Outputs

# CLK - Clock Generator

Clock Generator (CLK) produces clock pulses based on BPM to synchronise modules.

## 27.1 Mapping of Controls



## 27.2 Operation

Clock Generator (CLK) produces steady pulse signal of a rate related to the BPM (Beats Per Minute). Number of generated clocks vary depending on global Setting:Clocks Per Quarter Note (CPQN). BPM determines how many beats are per minute and CPQN determines how many clocks will be generated per beat.

■ MIDI standard sets 24 clocks per quarter note however this value can be modified as different equipment uses different assumptions.

★ Clock should be supplied to modules like LFS - Low Frequency Shaper or PLY - Polyrhythm to change their default timing from 120 BPM to required one set in CLK

## 27.3 External Clock

When connecting Ext Clock Input **[INPUT:CLOCK]** the module will start displaying external clock BPM in accordance to configured CPQN (Clocks Per Quarter Note) in Settings of the module. After turning **[SET:Sync]** to ON the internal BPM can no longer be configured with the **[Input:BPM]** but it will use the value of external clock BPM

■ When using external clock only usable are CVY1-4 inputs as those are low latency inputs. See: Physical Connectors

## 27.4 Clock Output

Clock Module can provide master clock to external modules via Clock Output. Clock Output can be configured via settings to one of VOUT1-4 3.5mm outputs or to VBuf Audio Outputs

# CLK - Clock Generator

## Settings

### **Clock Output** *Clock Output*

Selects audio output destination for CLK. The VOUT1-4 can carry clock out as they are DC coupled. Clock will output square clock signal through the selected output.  
See: [Audio Outputs](#)

### **Sync** *Sync clock with external clock*

Sync will synchronise clock module with external clock's BPM rate.

### **Clocks PQN** *Clocks Per Quarter Note for External Clock*

Selection of clocks for external source of BPM. User needs to select the number of clocks per BPM the external source generates

## Inputs

### **BPM** *Beats Per Minute*

Selection of Beats Per Minute setting

### **Clock** *External Clock Input*

External Clock input used to calculate BPM of external source  
■ Use only with CVY1-4 inputs

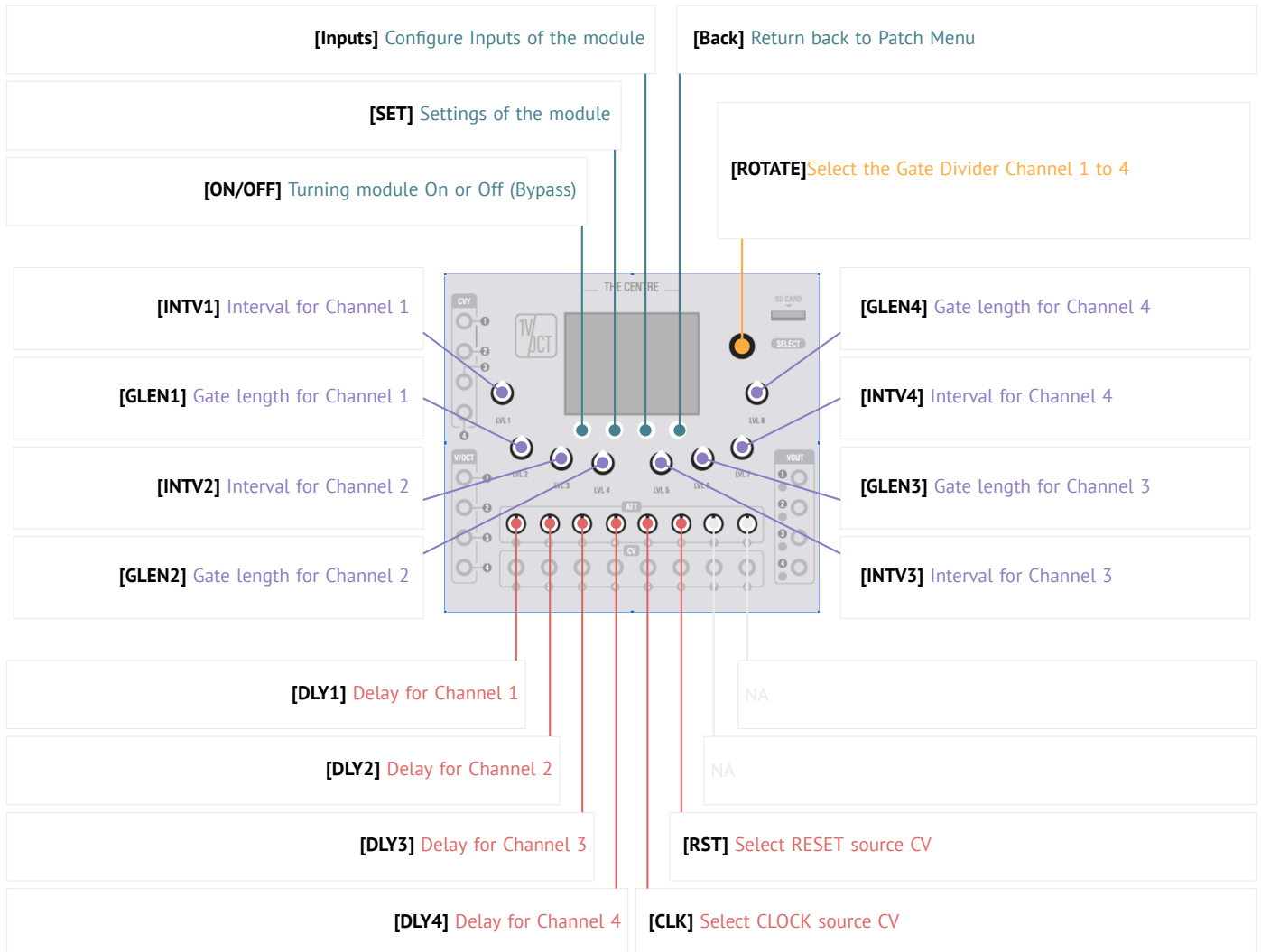
## Outputs

### **Clock** *Clock signal*

Output of generated clock signal relative to BPM

# GAT - Gate Divider

Gate Divider (GAT) divides pulse clock signal (clock) into longer period pulse clock signals.



## 28.1 Operation

Gate Divider (GAT) generates pulse signal (square wave with different phase width) upon dissecting and dividing input pulse signal (usually clock). The input signal triggers counter in 4 channels and based on set parameters creates longer pulse signals with modified width of phase (both negative and positive).

■ Such division can be seen as turning clock signal into evenly spaced notes with evenly spaced rests or evenly spaced notes triggers with particular notes lengths.

★ Divided gates are great source of repetitive patterns to be used as repetitive rhythm generator (drum machine). Triggering drum samples from gate divider and setting gate divider divisions to different notes can create regular beat pattern.

# GAT - Gate Divider

## Settings

**NONE**

## Inputs

**Clock** *Clock CV Source*

Pulse CV source to base divisions on

**Reset** *Reset CV Source*

Reset signal to synchronise position of all dividers by setting all positions to start

**Interval 1** *Generated pulse interval*

Interval between gate divisions whn the gate becomes high

**Gate Len 1** *Length of pulse*

Number of beats within a channel.

**Delay 1** *Delay before triggering pulse*

Delay is the duration before pulse signal turns gate high

**Interval 2** *Generated pulse interval*

Interval between gate divisions whn the gate becomes high

**Gate Len 2** *Length of pulse*

Number of beats within a channel.

**Delay 2** *Delay before triggering pulse*

Delay is the duration before pulse signal turns gate high

**Interval 3** *Generated pulse interval*

Interval between gate divisions whn the gate becomes high

**Gate Len 3** *Length of pulse*

Number of beats within a channel.

**Delay 3** *Delay before triggering pulse*

Delay is the duration before pulse signal turns gate high

**Interval 4** *Generated pulse interval*

Interval between gate divisions whn the gate becomes high

**Gate Len 4** *Length of pulse*

Number of beats within a channel.

**Delay 4** *Delay before triggering pulse*

Delay is the duration before pulse signal turns gate high

## Outputs

**Gate 1** *Gate 1 Output*

Output of Channel 1 pulse signal

**Gate 2** *Gate 1 Output*

Output of Channel 2 pulse signal

**Gate 3** *Gate 1 Output*

Output of Channel 3 pulse signal

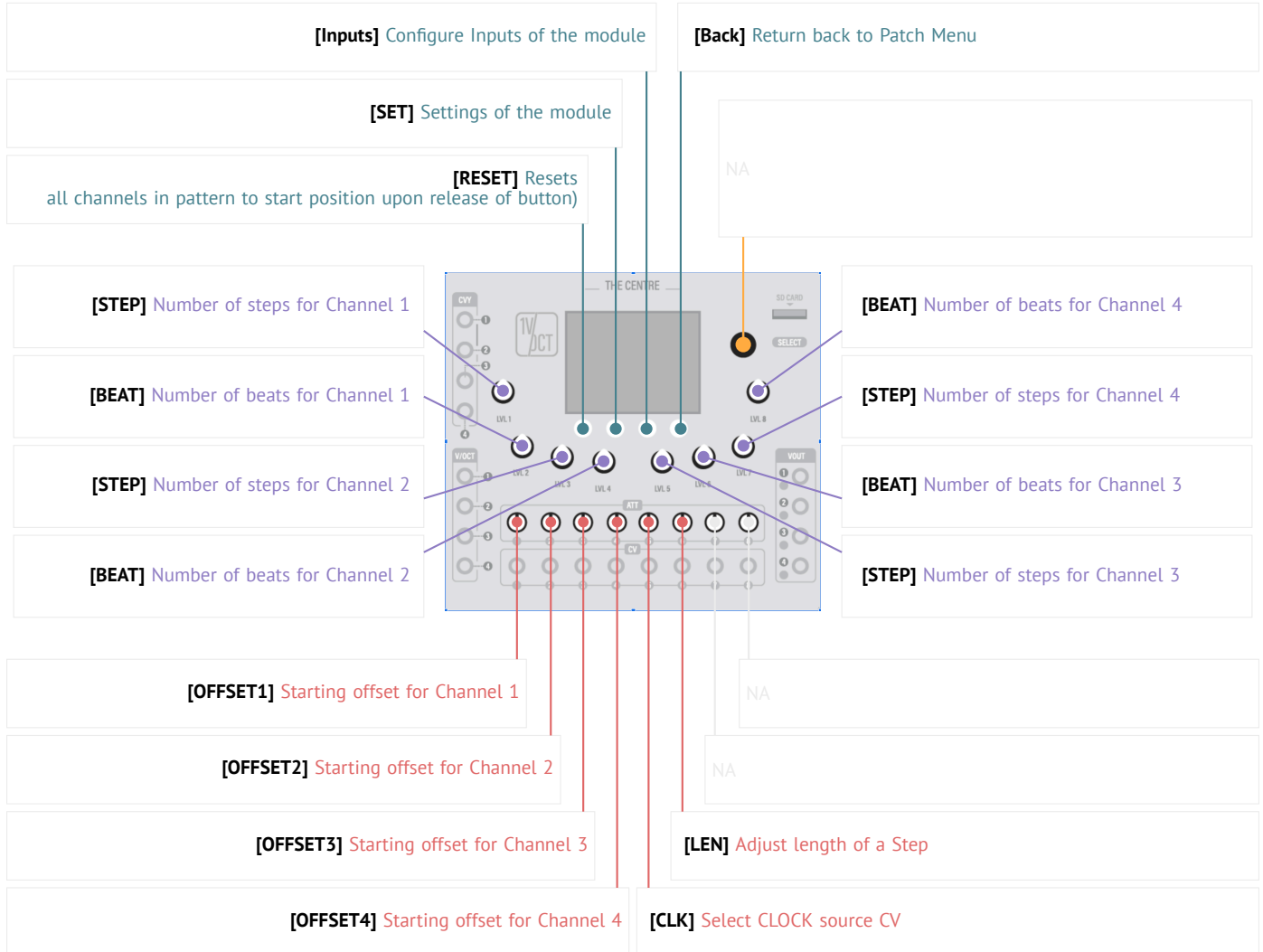
**Gate 4** *Gate 1 Output*

Output of Channel 4 pulse signal

# EUC - Euclidean Rhythm Generator

Euclidean Rhythm Generator (EUC) produces rhythmic pattern based on Euclid's algorithm to divide periods of time into equal parts.

## 29.1 Mapping of Controls



## 29.2 Operation

Polyrhythm generates 4 channels of rhythms by dividing pattern duration into equal steps. Pattern is defined as period of time covering number of bars (4 beats or 4 quarternotes) at given BPM (beats). Number of steps in channel can be controlled via Inputs:Beats X (where X is the number of channel 1 to 4).

- Without clock source the BPM is fixed at 120 BPM the pattern

# EUC - Euclidean Rhythm Generator

## Settings

**Step Length** *Length of single Step*  
Length of single step, patterns are consisting of arbitrary number of equal length

**Clock** *Clock CV Source*  
Clock source to synchronise Poly Rhythm with other modules.

**Reset** *Reset CV Source*  
Reset position for all channels in pattern upon trigger

**Steps 1** *Number of Steps for Channel 1*  
Number of Steps within a channel.

**Beats 1** *Number of Beats for Channel 1*  
Number of beats within a channel.

**Offset 1** *Offset of first step for Channel 1*  
Offset of first step for channel

**Steps 2** *Number of Steps for Channel 2*  
Number of Steps within a channel.

**Beats 2** *Number of Beats for Channel 2*  
Number of beats within a channel.

**Offset 2** *Offset of first step for Channel 2*  
Offset of first step for channel

**Steps 3** *Number of Steps for Channel 3*  
Number of Steps within a channel.

**Beats 3** *Number of Beats for Channel 3*  
Number of beats within a channel.

**Offset 3** *Offset of first step for Channel 3*  
Offset of first step for channel

**Steps 4** *Number of Steps for Channel 4*  
Number of Steps within a channel.

**Beats 4** *Number of Beats for Channel 4*  
Number of beats within a channel.

**Offset 4** *Offset of first step for Channel 4*  
Offset of first step for channel

## Inputs



# Outputs

## **Gate 1** *Gate 1 Output*

Output of Channel 1 setting gate to high upon beat

## **Gate 2** *Gate 2 Output*

Output of Channel 2 setting gate to high upon beat

## **Gate 3** *Gate 3 Output*

Output of Channel 3 setting gate to high upon beat

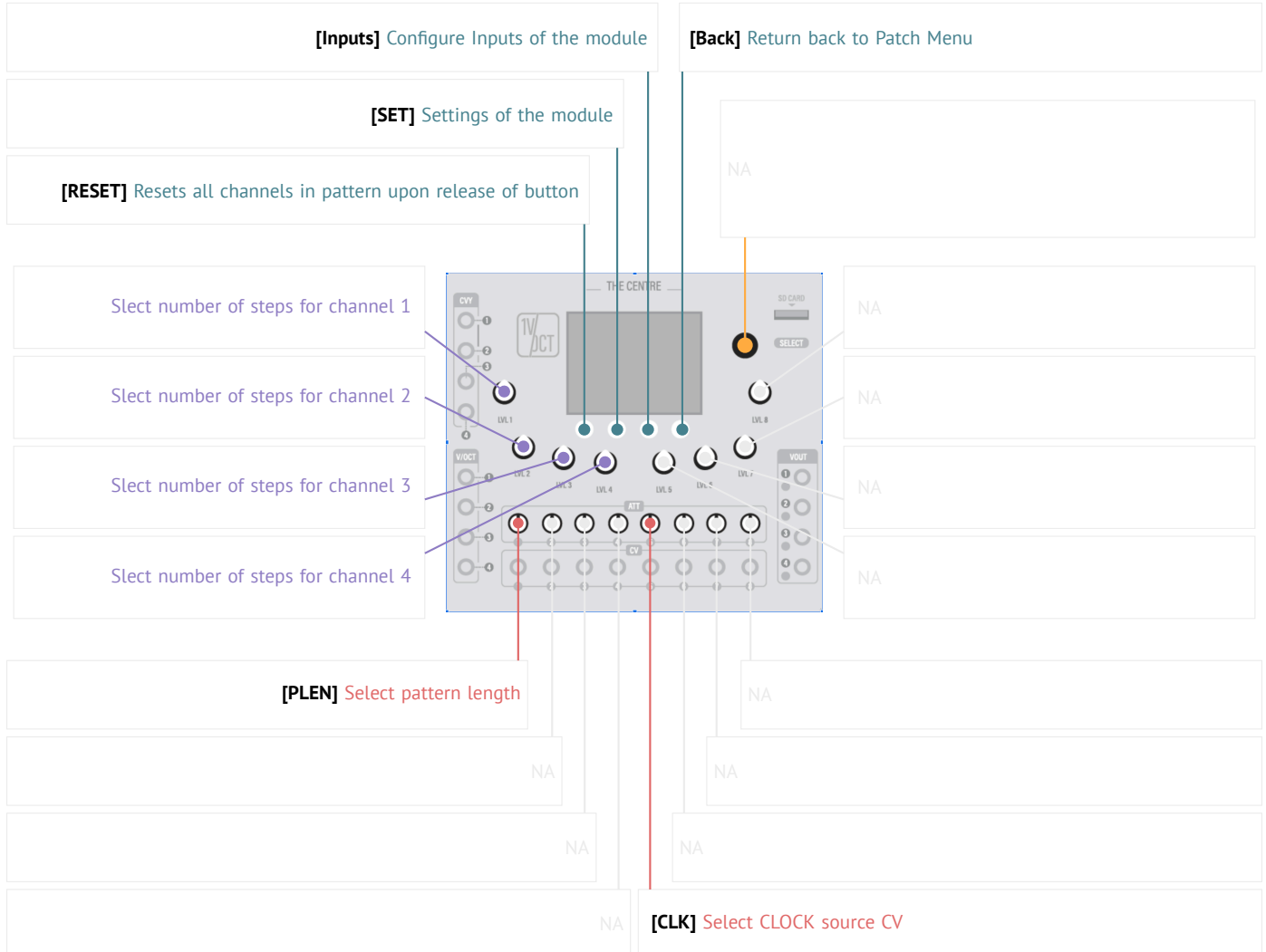
## **Gate 4** *Gate 4 Output*

Output of Channel 4 setting gate to high upon beat

# PLY - Polyrhythm

Polyrhythm (PLY) generates contrasting rhythms within a pattern.

## 30.1 Mapping of Controls



## 30.2 Operation

Polyrhythm generates 4 channels of rhythms by dividing pattern duration into equal number of steps decided by input parameter Inputs:Steps. Pattern is defined as period of time covering number of bars (4 beats or 4 quarternotes) at given BPM (beats). Number of steps in channel can be controlled via Inputs:Beats X (where X is the number of channel 1 to 4).

■ Without clock source the BPM is fixed at 120 BPM the pattern

# PLY - Polyrhythm

## Settings

### Pattern Length *Length of pattern in bars*

Length of pattern in bars. Each pattern is divided into number of steps configured by Inputs:Pulses

### Gate length *Length of gate*

#### Trigger

Trigger only

## Inputs

### Clock *Clock CV Source*

Clock source to synchronise Poly Rhythm with other modules.

### Reset *Reset CV Source*

Reset position for all channels in pattern upon trigger

### Pulses 1 *Number of pulses for channel 1*

Number of steps that the pattern gets divided into and generating a beat at the beginning of each step

### Pulses 2 *Number of pulses for channel 2*

Number of steps that the pattern gets divided into and generating a beat at the beginning of each step

### Pulses 3 *Number of pulses for channel 3*

Number of steps that the pattern gets divided into and generating a beat at the beginning of each step

### Pulses 4 *Number of pulses for channel 4*

Number of steps that the pattern gets divided into and generating a beat at the beginning of each step

## Outputs

### Gate 1 *Gate 1 Output*

Output of Poly Rhythm setting gate to high upon beat

### Gate 2 *Gate 2 Output*

Output of Poly Rhythm setting gate to high upon beat

### Gate 3 *Gate 3 Output*

Output of Poly Rhythm setting gate to high upon beat

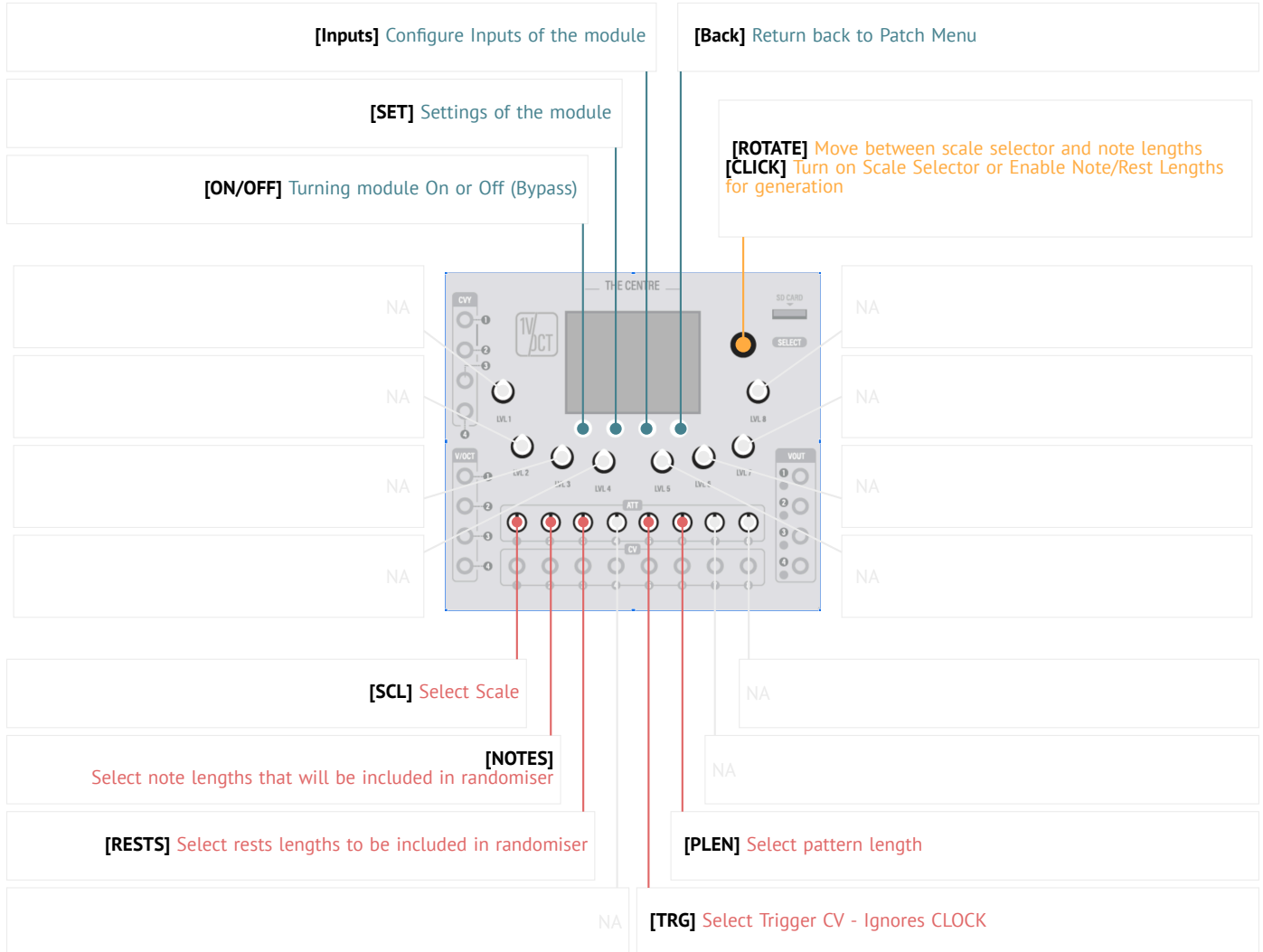
### Gate 4 *Gate 4 Output*

Output of Poly Rhythm setting gate to high upon beat

# RNG - Rendom Note Generator

Random Note Generator (RNG) produces random notes.

## 31.1 Mapping of Controls



## 31.2 Operation

Random Note Generator (RNG) produces random values for output of pitch, gate length and velocity. The randomisation happens every time upon receiving trigger signal at Input:Trigger. The randomisation process can be controlled by selecting range of available notes and rests as well as predefines musical scale to limit generated pitches.

■ Without clock source the BPM is fixed at 120 BPM the pattern

★ The musical scale selection is not necessary as routing output of RNG through QNT (Quantiser) and limit available pitch there gives much better flexibility.

# RNG - Rendom Note Generator

## Settings

### Pattern Length *Length of pattern in bars*

Length of pattern in bars. Each pattern is divided into number of steps configured by Inputs:Pulses

### Gate length *Length of gate*

**Trigger**

Trigger only

## Inputs

### Clock *Clock CV Source*

Clock source to generate notes automatically and measure length of notes

### Trigger *Trigger CV Source*

Triggers generation of next note, if Trigger is not connected notes and rests get generated automatically upon their period ends

## Outputs

### Pitch *Pitch Output of generated note*

Output of generated random pitch limited by setting of selected musical scale

### Gate *Gate Output of generated note*

Output of Poly Rhythm setting gate to high upon beat

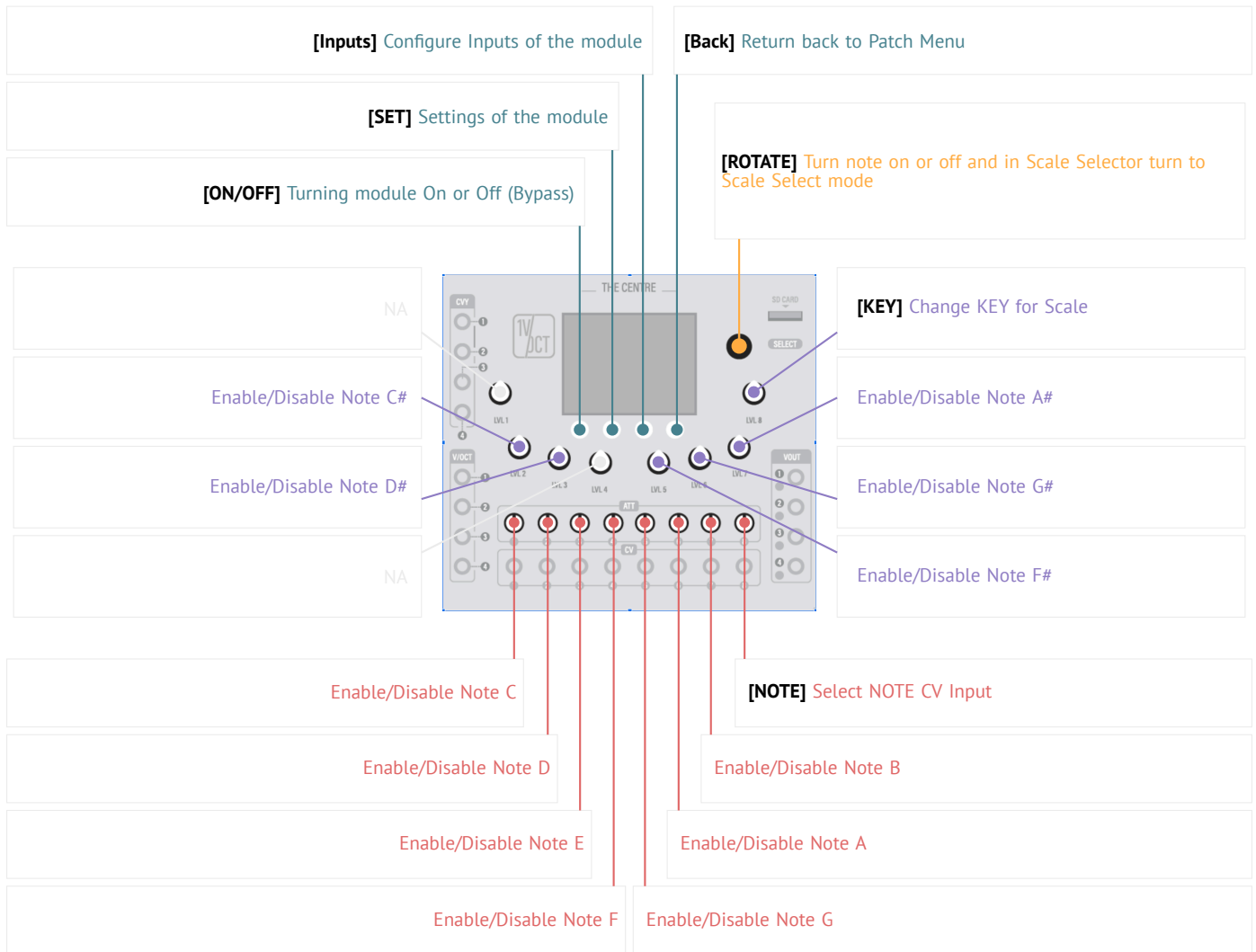
### Velocity *Velocity Output of generated note*

Output of Poly Rhythm setting gate to high upon beat

# QNT - Quantiser

Quantiser (QNT) allows correcting incoming pitch into a selected set of outgoing pitches. Usually it is used to limit number of notes to the notes available only in given musical scale. Quantiser takes pitch from Input:NOTE and finds the closest corresponding pitch in the scale by method of approximation.

## 32.1 Mapping of Controls



## 32.2 Musical Scales

Currently there are only a few musical scales preconfigured in Quantiser. Each preconfigured scale can be also adjusted to the correct **Key** (using LVL8 knob or through settings).

## 32.3 Custom Scales

Custom scale is a set of notes configured by user. By turning any knobs corresponding to notes (as pictured above) the note will be turned ON and OFF. The note configuration can be also done by rotating Encoder [SELECT] and performing Click with Encoder on selected notes.

■ Turning knobs or clicking on notes automatically switches module into Custom Scale.

★ If there is only one note selected in the quantiser, for example note C and the input note is E4 then output note will be C4. However for note G4 the output note will be C5 as C5 is closer to G4 than C4 (approximation).

## Settings

### Mode *Mode of Operation*

**Simple**

Mode of operation in which quantisation is based on selecting notes that the incoming pitch will be aligned to

### Scale *Musical Scale in Simple Mode*

**Custom**

User selected notes

**Chromatic**

Chromatic Scale

**Major/Major Scale Natural Minor/Natural Major Scale Harmonic Minor/Harmonic Major Scale Melodic Minor**

Major/Major Scale Natural Minor/Natural Major Scale Harmonic Minor/Harmonic Major Scale Melodic Minor/Melodic Major Scale

### Key *Change Key for a selected Scale (except for custom scale)* Change Key for a selected Scale (except for custom scale)

## Inputs

### NOTE *Note input for quantisation*

Note input that will be quantised. The incoming note can be adjusted and tuned before quantisation. See: [Pitch Control](#)

**VOCT**

1V/Oct input for note control/

**OCT**

Tuning note by octaves +/- 8 octaves

**NOTE**

Tuning note by semitones +12 semitones (one octave)

**FINE**

Tuning note by cents +/- one semitone

## Outputs

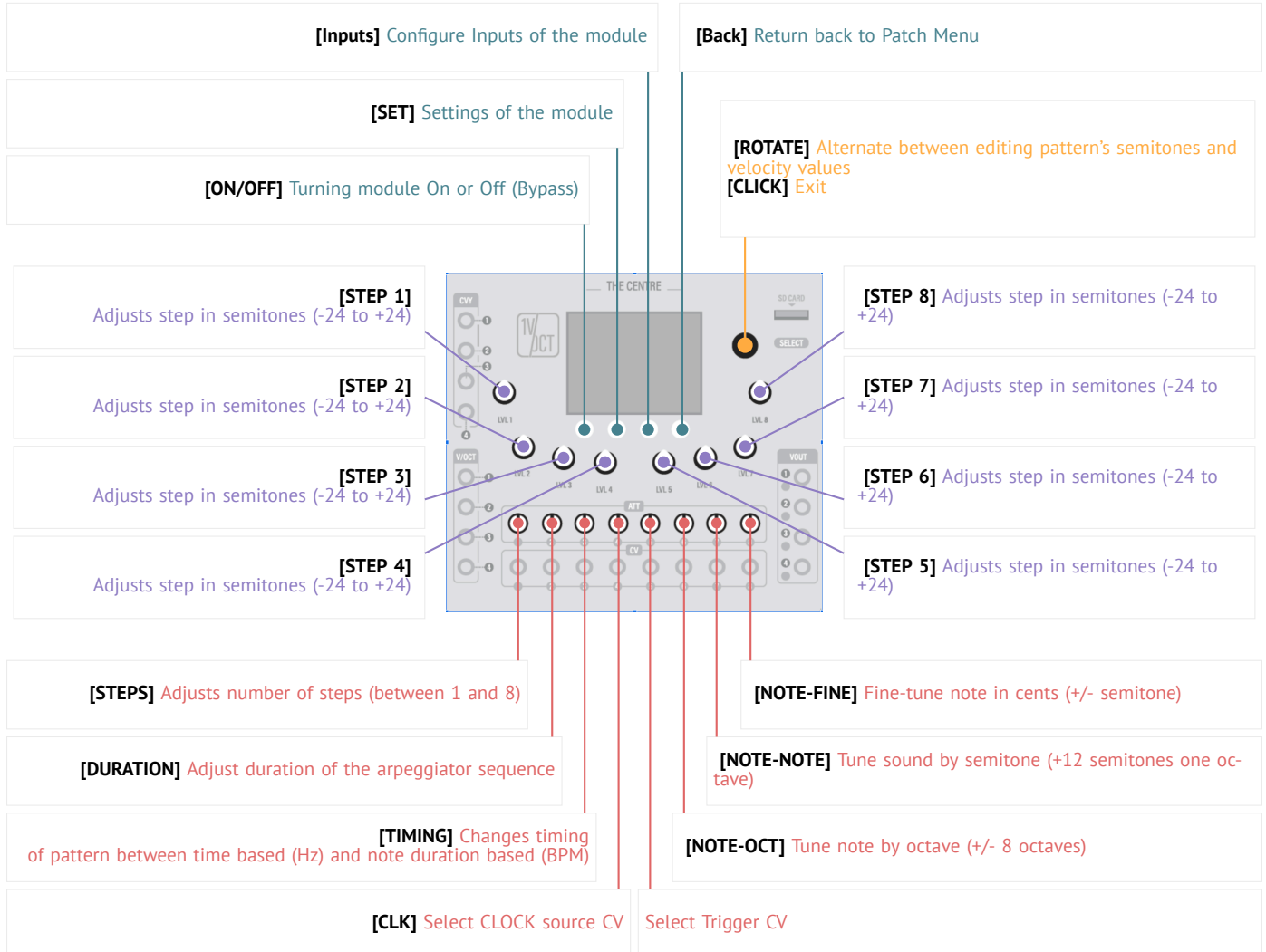
### NOTE *Quantised Note*

Quantised note (note that has been aligned to the closest pitch of quantisation scale)

# ARP - Arpeggiator

Arpeggiator (ARP) cycles through a set of notes defined in short pattern of defined duration.

## 33.1 Mapping of Controls



## 33.2 Operation

Arpeggiator (ARP) is a pitch manipulation utility that creates a repetitive music pattern consisting of up to 8 notes that are defined as a difference in semitones to the played note. The steps can be adjusted with knobs and the number of steps can vary between 1 and 8 steps. Duration of the pattern can be set by specifying duration of whole pattern by frequency of oscillation of pattern in Hertz (Hz) or duration of pattern measured in length of notes or bars based on timing coming from clock source (BPM). Use encoder (SELECT) to change between editing pattern's semitones and velocity values.

- Every step Arpeggiator will generate Gate signal that lasts for the whole duration of the step.
- Velocity value is just an arbitrary value assigned to each step and can be used for modulating any parameter within The Centre.

## 33.3 Control arpeggiator position from external source

Arpeggiator position in default setting increases position with every step. Use Position CV to set position of arpeggiator according to external CV source.

- By setting external CV source, arpeggiator will output only TRIGGER and not GATE signal on GATE Output. Furthermore CLOCK will be ignored and externally controlled position will determine position of arpeggiator.
- Only positive values of CV input will affect position. Negative values will be clipped to position 0. Use either positive CV source or modify source with Attenuation and Level.
- ★ Use LFO TRIANGLE input with ATTENUATOR set to 0.5 and LEVEL set to 0.5 to create TRIANGULAR input for arpeggiator to work



in PING-PONG mode.

★ Change LFO to RAMP or SAW (LFO Skew) and arpeggiator will work either in ascending or descending mode respectively.

# ARP - Arpeggiator

## Settings

**Timing Mode** *Duration of arpeggiator pattern*

**BPM**

Timing based on note duration calculated from BPM which is controlled by steps

**Hz**

Frequency at which pattern repeats

## Inputs

**NOTE** *Base note*

Input note that is a base note for arpeggiator to create sequence.

See: [Pitch Control](#)

**VOCT**

1V/Oct input for note control/

**OCT**

Tuning note by octaves +/- 8 octaves

**NOTE**

Tuning note by semitones +12 semitones (one octave)

**FINE**

Tuning note by cents +/- one semitone

**Clock** *Clock CV Source*

Clock source to synchronise with other modules.

**Reset** *Reset CV Source*

Reset position to step 0

**Duration** *Length of Pattern*

Duration of pattern measured either in frequency (Hz) or note duration (BPM)

**Position** *Position CV Source for pattern step*

External CV to control position of pattern

## Outputs

**NTE** *Pitch Output of generated note*

Output of current step's pitch calculated by adding step's semitones to played note

**GTE** *Gate Output of generated note*

Output of Poly Rhythm setting gate to high upon beat

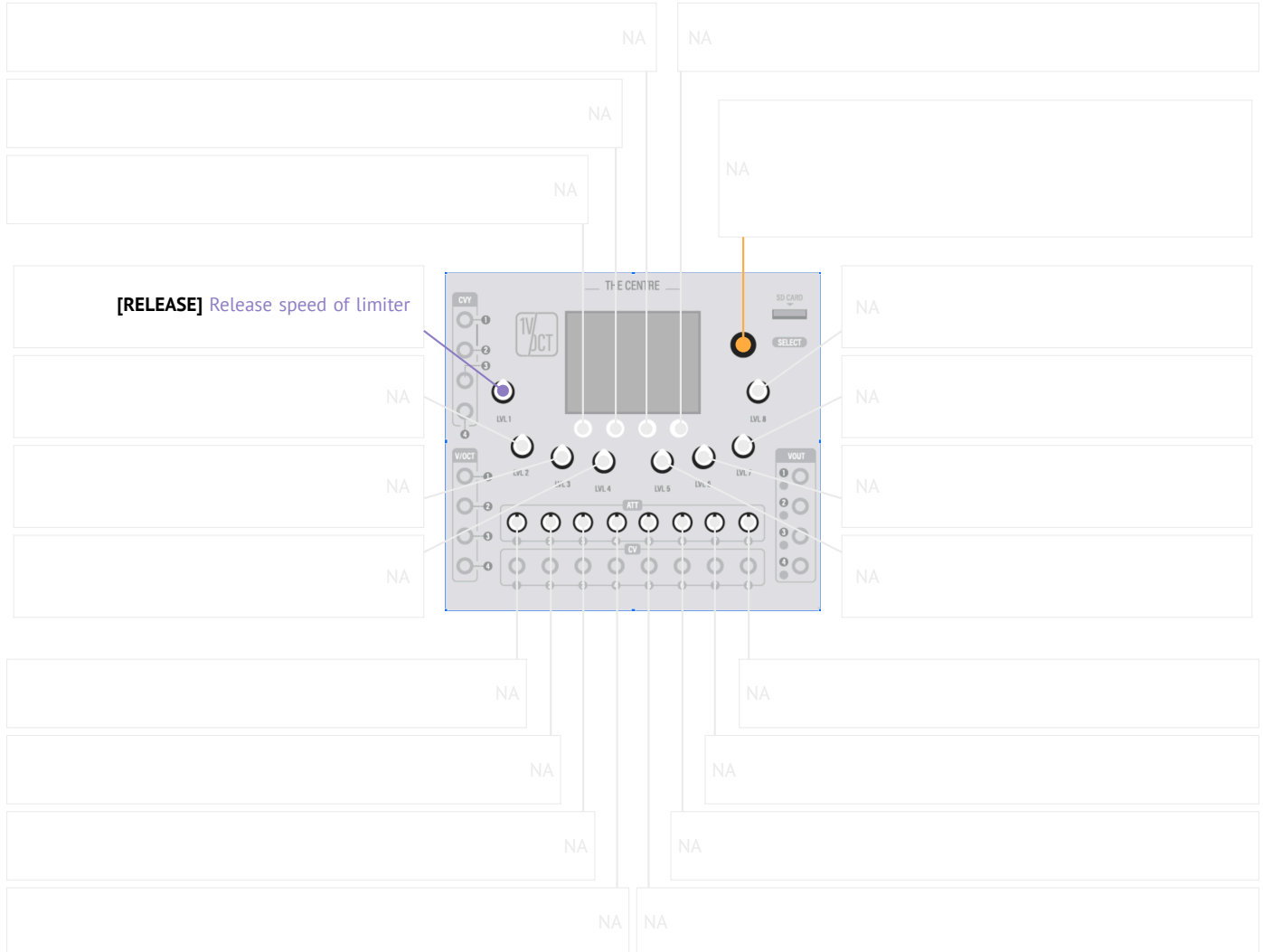
**VEL** *Velocity Output of generated note*

Output of velocity/modulation parameter associated with current step

# OUT - Output

Output (OUT) is a limiter and clipper for oversteered audio levels.

## 34.1 Mapping of Controls



## 34.2 Operation

Output module acts as limiter and when the audio signal exceeds reproducible range it ducks (limits) audio signal with instant reaction (quick Attack mode) and then releases ducking according to **RELEASE** input setting. The high value of Release (short time) will bring the audio level back quickly thus making limiter behave like a siplistic compressor. Longer Release rates (smaller value) will make limiter produce slightly wobbly sound for audio levels that peak above limit.

Outputs Inputs Settings

**Release** *Release for Limiter*  
Release rate for limiter