PULSAR·23
ORGANISMIC DRUM MACHINE

USER
MANUAL

Pulsar-23 is a complex device with many non-obvious functions and capabilities. To unleash its full potential, it is strongly recommended to read these instructions.

GENERAL OVERVIEW

Pulsar-23 is a multi-functional analog synthesizer and generator of complex rhythmic patterns.

The Pulsar consists of 23 different modules, including four flexible sound generators with completely different structures, four envelope generators, four looper-recorders, a clock generator with dividers, a controlled chaos generator, an LFO, a two-channel CV-controlled effects processor, distortion, two controlled amplifiers, an inverter, a controlled inverter and two controlled analog switches.
In addition to these 23 main units, the Pulsar also contains 13 auxiliary units, such as a four-channel MIDI to CV converter, a noise generator, four attenuators, two dynamic CV generators with sensory control, two impulse converters and single passive electronic components for live circuit bending.

Pulsar-23 can be used for the synthesis of percussion instruments and rhythms, bass and melodic lines, effects and sound landscapes, as well as a source of control voltage. The Pulsar functions in three different modes: stand-alone, MIDI control and CV control. Moreover, all the above features and control modes can work simultaneously in any proportion or combination. Additionally, Pulsar offers live circuit bending capabilities and the use of the artist's body conductivity to create patches and cross modulations.

Pulsar continues the line of organismic synthesizers begun by LYRA-8, but now in the area of percussion instruments.

ORGANISMIC SYNTHESIZER

"Organismic" means that some of the principles of how living systems—organisms—operate, form the basis of development:
• Everything can interact with everything, forming multiple feedback loops, resulting in very complex behavior of the system, even with a simple set of its constituent elements.
• The blurring of the functions of organs and parts, allowing them to be interpreted differently, depending on which context and connection they work in.
• The absence of a rigid linear structure, where there is a clearly leading head and the tail strictly following it. Any part of the body can become for a while both leading and driven.
• The resulting behavior is a dynamic equilibrium spontaneously formed between the interacting parts of the living system.

These principles are most clearly expressed in neurosystems (brain) and systems built on their basis (for example, human society).

Consider how these principles were implemented in Pulsar:

The Pulsar is a semi-modular system, where each unit has an input, output and several available control points that can control the processing. The audio and control signals operate in the same voltage range of 0-10 volts, and the inputs and outputs are organized in such a way that the audio signal can be a control signal, and the control signal can be used as an audio source. For example, you can use the bass drum channel as an LFO, the LFO as an additional sound oscillator, the clock generator as a source of percussive sounds, and the bass synth channel as a clock source for the loopers. This lets you build many different structures, including paradoxical ones.

You can connect the inputs and outputs of the Pulsar in any combination, without worrying that something will be damaged or done "incorrectly". At the same time, due to the smart organization of the input and output impedance of the connection points, several signals connected together will be automatically mixed, and the points that can work as input and output (for example, triggering envelope generators) will figure out on their own what is connected to them and start to either receive the signal, or send it, or they will begin mutual modulation if a point with the same behavior is connected.
All inputs and outputs of the Pulsar are ready for integration into a Eurorack system and are protected from overloads. This means you can do all kinds of experiments with various sound equipment, without the danger of damaging the instrument. The permissible voltage range connected to the inputs of the Pulsar is -20 to +20 volts. However, the effective operation of the input is limited to a range of 0 to 10 volts.

Without any patching, Pulsar-23 is a drum machine with a sequencer with a conventional linear structure: clock generator -> looper -> sound modules -> FX -> output. The full capabilities of the Pulsar are revealed when you start connecting modules to each other, creating control and modulation channels. Since the number and depth of interactions are completely under your control, a smooth transition from classical analog drum synthesis to abstract noise and similar things is possible.

The functions of many Pulsar modules are blurry and can, with different settings and controls, move from one area of sound synthesis to another.
A key feature of the envelope generators and sound modules is that they recognize sustain, i.e. the duration of pressing the sensor or key on the MIDI keyboard. Thus, with a short press, we get a percussive sound, the character of which will depend on the length of the press. When holding the sensor or key for a long time, we get a tonal or noise sound, depending on the synthesis module used and its settings. Thanks to this, your drum line can suddenly turn into noise or drone. Also, Pulsar-23 can be used as a powerful and unusual monophonic synthesizer controlled by MIDI and/or CV.
The LFO and SHAOS modules can be used as sound generators, and in general any voltage source in Pulsar can be considered a sound source, processing it in various ways and mixing it into the common mix or using it separately. Just like any audio output, you can use it as a source of control voltage or modulation. The Pulsar invites you to experiment with an open mind, free from the dogma of what is what.

ALLIGATOR CLIPS CONNECTION SYSTEM

When I started developing Pulsar, my intention was to make its structure as open as possible and to put patch points everywhere it made sense. And that meant not worrying too much about the number of such points. But each plug is a place on the PCB and an added cost. When the number of connectors exceeds 100 (there are 119 in Pulsar) this becomes a very significant factor and can significantly increase the size and cost of the device. All existing solutions were either space-consuming and expensive, or unreliable, which is not acceptable for an instrument of this class. And then I arrived at the idea of using specially-made vertical pins and alligator clips.

The benefits of this solution are as follows:
• Saving space. Mounting a pin requires only a few square millimeters on a PCB.
• Low manufacturing cost.
• Extreme reliability, since with such a simple design there is simply nothing to break. It's just a ribbed metal pin to make it easy to attach the clips.
• You can connect several clips to one pin, hereby multiplying or combining signals.
• Two alligator clips can be connected together if you don't have enough cable length.

• You can carry out a lot of experiments by connecting alligator clips to various radio components, parts of electronic circuits (for example, an old radio), touch plates and even take two forks with connected alligator clips, plug them into a cucumber and listen to how it sounds in the snare drum synthesis chain:). All of these connections will become parts of the Pulsar circuit, which is very sensitive to such inclusions.
• You can easily connect to connectors such as jack, mini-jack and banana by simply attaching an alligator clip to the signal pin of the connector.
• Ready-made cables with alligator clips are easy to buy and they are significantly cheaper than regular audio cables.

This non-standard solution offers so many advantages that we settled on it, despite its unconventional nature.

At the same time, we made sure you can also connect to the usual formats — the Pulsar has eight freely assignable 3.5 mm mini-jacks and six freely assignable 1/4-inch jacks. MIDI input is implemented with a standard DIN connector.

LIVE CIRCUIT BENDING

A number of Pulsar patch points are more than just CV control inputs. Some of the points commonly used for circuit bending have been brought out, which allows you to wedge into the circuit, changing its behavior on the fly. You can use single electronic components, such as a resistor, capacitor, diode or transistor, including them in a control or modulation circuit to get different behaviors and sound.
The patch-point design and specially calculated input impedance make it easy to use the artist's body as a connecting cable. By touching various contacts and closing them together during the performance, you can create quick and dynamic changes in the sound and behavior of the Pulsar. Since the resistance to contact with the skin strongly depends on the pressure, you can easily and intuitively change the modulation depth with simple hand movements, controlling several points at once.

SOUND MIXING CONCEPT

When developing Pulsar, I found it pointless to do internal mixing in stereo. In order to get a good stereo picture in a drum machine, you must either have many different instruments diluted in the panorama (usually percussive sounds), or use individual spatial effects superimposed on individual sounds. In Pulsar, instead of making a lot of specific, little controlled sound generators (tom1, tom2, cowbell, clap etc.) like in classical drum machines, there are only four yet powerful and flexible generators. The sounds of three of them, focused on the synthesis of bass drum, snare drum and bass, are usually placed in the center. Therefore, it turns out that there is nothing special to pan.
As a result, it was decided to make the summing bus and the main audio output monophonic, but at the same time provide the opportunity for full external mixing, where you can create a good stereo picture using external spatial effects.
Therefore, everything that can produce sound in Pulsar has a separate output contact. These contacts can be assigned to any of the six 1/4-inch jacks or any of the mini-jacks, allowing separate sounds to be processed by external stereo effects and fed to an external stereo mixer, or channel-recorded in a DAW for further processing and information.
The output contacts of the sound generators and the "send" to the FX processor are located before the volume knob, so you can easily exclude any sound from the main mix and assign it to external mixing and processing.
The effects processor also has separate output pins for each of the two channels and can produce a stereo signal that you can use for further stereo mixing.

MIDI

The Pulsar has significant capabilities to be controlled via MIDI.
MIDI controls have:
• Dynamic trigger of all four synthesis modules, taking into account velocity.
• The BASS module recognizes pitch bender (the range is +/- 12 semitones) and portamento (CC05).
• MIDI controllers can be assigned to SHAPE and WARP synthesis parameters of the BASS module.
• The Pulsar can receive MIDI clock which can synchronize the array of clock dividers and the looper\recorder. To do this, turn the INT MIDI switch of the clock section to the MIDI position.
• There are four MIDI to CV converters (MIDI CV block) capable of converting messages from MIDI controllers and keyboards into CVs. The converter outputs can then be connected to any Pulsar input contacts, which will provide MIDI control or automation of the functions connected to these contacts.

The intention was to make the MIDI implementation as user-friendly as possible. Next to each function that has MIDI automation there is a learn (LRN) button, which makes it easy to assign a MIDI controller there. To do this, press the LRN button next to the desired function and turn the midi-controller or press the key on the midi-keyboard that you want to assign there. The Pulsar will remember the channel number, key or controller number (CC) and will remember them even when the power is off.

Altogether there are 12 parameters that can be automated via MIDI:
Triggers of the 4 synthesis modules, 4 freely assignable MIDI to CV converters, SHAPE, WARP, Portamento and Pitchbender functions of the BASS module.

To assign a MIDI controller (a key or a continuous controller) to a synthesis module (BD, SD, HHT etc.) hold the LRN button of this module and press the desired key or turn the controller.

The pitchbender and portamento functions are firmly fixed to the corresponding midi-controllers. The receiving midi-channel number of these functions is the same as the midi-channel number of the keyboard assigned to BASS. Portamento can only be adjusted via MIDI.

The synthesis modules and MIDI to CV converter automatically recognize the key and the continuous controller (CC). In case a MIDI keyboard key is assigned, the velocity value of the pressed key will be transmitted. If a controller is assigned, the position of the controller is transmitted.

You can control the synthesis module trigger with a continuous controller instead of by pressing a key. This will give you many unusual features, such as drawing the attack and signal decline in a DAW. To do this, use the LRN button of the desired module and assign a continuous controller to the module you need.

You can assign MIDI keys to a MIDI to CV converter, this will give you the opportunity to rhythmically control various synthesis parameters (for example, a filter) from a MIDI keyboard and apply functions such as quantization, which are difficult to use with a continuous controller.

If any key is assigned to the first channel of the MIDI to CV converter with the same MIDI channel as the MIDI keyboard assigned to the BASS module, the key tracking signal of the

BASS module will be sent to the output of the first channel of the converter. The CV on this output will be proportional to the note number that is currently played by the BASS module. This function is very useful if, for example, you want the cutoff frequency of the filter to follow the pitch of the note being played. To activate this function, simply simultaneously press the LRN button of the first channel of the converter and any key on the MIDI bass keyboard. The first channel of the converter is marked KTR (key tracking).

Please remember that the looper-recorder has a clock up-sampling process that needs a little time for setting. The first few dozens of milliseconds after MIDI clock start can be inaccurate. To avoid this, the timespan between stop and start on your DAW or sequencer that feeds MIDI-clock to Pulsar, has to be at least 5 seconds (then the up-sampling system stops to wait for the next clock pulse and saves the previous value that will be used after clock start).
A second option is to use the LRST pin for alignment with the clock divider.

When using MIDI-clock synchronization, we strongly recommend connecting the LRST pin to 0.25 out of the clock divider to ensure a perfect sync.

Pressing simultaneously the SHAPE and WARP LRN buttons will stop all MIDI triggering. You can use it as a MIDI panic button if your MIDI system freezes.

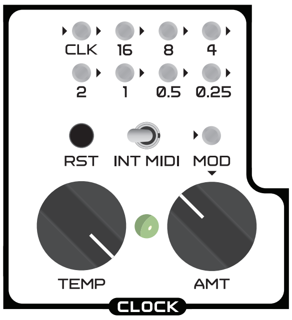
GENERAL LABELING PRINCIPLES

Contact function marking:

The arrows connecting the various elements indicate their relationship and show the direction of the signal flow.

MAIN MODULES

MASTER **CLOCK** GENERATOR



The basic structure of the Pulsar begins with a master clock generator. When creating a patch, you can make the source of the clock something else and even have several sources at the same time. The clock generator sets the tempo of your rhythmic pattern, as well as the looper/recorder's loop length, which is four measures in four quarters or 128 ticks of the clock. One tick equals a 32nd note. At the same time, you can freely choose what constitutes one quarter and the length of your measure, since there is no quantization by default in the looper.

TEMP knob. Sets the generator clock frequency, which varies from one to two hundred hertz, thus determining the looper's loop length from several minutes to less than one second. The green LED flashes once a quarter and makes a bright flash at the beginning of the loop (based on the fact that the loop length is four measures in four quarters).

MOD pin. CV input modulating and controlling the clock frequency. The useful voltage swing, as with all modulation inputs, is 0 to 10 volts.

AMT knob. Determines the amount of influence of CV applied to the MOD pin on the clock frequency.

INT MIDI switch. Allows you to select the clock source. In the INT (internal) position, the source is the Pulsar clock generator. In the MIDI position, the source is an external MIDI clock. With the switch in the middle position, the internal clock is disabled and the array of dividers and the looper stop. Now, when voltage pulses with an amplitude of at least 3 volts (nominal amplitude 0 to 10 volts) are applied to the CLK pin, they will be perceived as a clock signal and the array of dividers together with the looper will be synchronized to it. So you can synchronize the Pulsar with a Eurorack system or any device that generates a clock in the given voltage range. The frequency of the supplied external analog clock should be calculated for 128 ticks per full length of the array of dividers and looper. If the switch is in the INT or MIDI position, you can pick up the clock signal from the CLK pin or the outputs of the clock divider to synchronize external equipment to the Pulsar.

CLK pin. The output of the clock generator or the clock input of the frequency dividers and looper/recorder.
With the switch set to INT or MID, the pin works as the output. With the switch in the middle position, it functions as input. (see "INT MIDI Switch")

16, 8, 4, 2, 1, 0.5, 0.25 pins. The outputs of the array of binary clock dividers. A very powerful tool for creating rhythms and controlling various Pulsar modules. The number under the pins represents the note duration that this output gives. The looper's loop length is equal to the length of the array of clock dividers.

RST (reset) button. Resets the array of clock dividers and resets the looper to the beginning of the loop. Used to set the start of the loop and to synchronize the dividers and looper. It is recommended that you press the RST button before starting loop recording. This will guarantee synchronization between the looper and the dividers.

- You can connect several outputs of the divider together. The signals will automatically be summed and you will get a complex rhythmic pattern based on this sum.

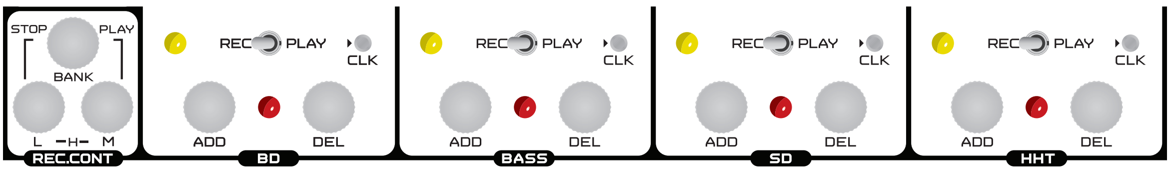
- You can use the signal from the outputs of the divider to create the sound of a metronome. To do this, connect the desired output of the divider (if the loop length is 4 steps of 4 quarters to obtain 1/4 duration, use output 2) with the input of one of the attenuators (for the possibility of adjusting the volume of the metronome) and connect the attenuator output to MIX IN input.

- To quickly get the metronome sound, simply touch one finger of your left hand to output 2, and the other to the MIX IN input. The circuit will be created using body conductivity.

- You can use outputs 16, 8, 4, 2, etc. to create a line of hi-hat, bass drum, etc. with equal notes. To convert a rectangular signal into short pulses, which form the distinct sound of the drums, use one of the impulse converters (see the corresponding section).

- You can use one or several summed outputs of the clock divider array to modulate the clock frequency, thus creating shuffle and more complex uneven ripples.

LOOPER-RECORDER



One of the features of the Pulsar is that it has no conventional sequencer, and instead a looper-recorder (LR) designed by the author. The main idea behind the LR is to enable quick and convenient creation and editing of live grooves "on the fly", flexible improvisation options during performance, and the possibility of experimenting with loops of different lengths and speeds on the different LR channels.

Pulsar's LR has 4 independent channels, each of which can have its own, independent playback speed. In order to be able to switch between different rhythmic variations, there are 4 banks of loops, each of which contains a set of 4 loops (a loop for each synthesis module). You can switch banks anywhere in the loop, creating rhythmic variety.

Unlike a sequencer, which records the time of the trigger event, the LR is essentially a virtual tape that constantly records all manipulations with the ADD and DEL sensors, taking into account the velocity specified by the sensors of the REC.CONT (recorder control) section. It does not record manipulations of the knobs and switches, or incoming MIDI events and CV signals. In line with the concept of Pulsar, LR is not a piece of code in the general processor that controls everything. Instead, it's an independent module executed on a separate micro-controller that deals only with this specific task. Thanks to this, it provides excellent stable behavior without delays and glitches, similar to the behavior of analog and mechanical devices.

While LR has a basic quantize option for individual LR channels, it does not allow for step-by-step editing of a rhythmic pattern. In our opinion, any portable sequencer with a miniature screen is significantly inferior in terms of ease of use and functionality compared to

a sequencer and an editor based on a personal computer. Therefore, if you need precise control over each beat, we suggest using a computer sequencer (Cubase, Ableton, Protools etc.) connected to the Pulsar via the MIDI interface. Rather, the capabilities of LR are concentrated on high quality and making live performances more convenient. To achieve this, data processing and storage are carried out with a resolution that significantly exceeds standard requirements. For example, the frequency of polling sensors and outputting a signal to sound-synthesizing modules is 110 kHz, which is 2.5 times higher than the standard frequency of digital audio and 36 (!) times higher than the MIDI frequency interface. The recording resolution is 96 events per pulse of the clock signal, which for a standard situation is 192 events per sixteenth note. In combination with the author's analog circuitry of capacitive sensors, providing a response time of 0.01 milliseconds, this gives Pulsar an almost instantaneous response to touching the sensor and a full-fledged “live" playing experience, characteristic of acoustic instruments.
It is also worth noting that the LR architecture allows you to record the duration of notes and dynamic changes within a single note arising from manipulations with REC.CONT sensors.

The speed of the virtual LR tape is determined by the frequency of the incoming clock. The loop length is fixed and equal to 128 clock pulses. However, the loop length can be made shorter if a forced restart of the LR is done more than once for 128 clock pulses. This will be described in detail below, when explaining the LRST function. One clock pulse is a 32nd note (1/32 of measure). To ensure a high resolution of the recording, LR upsamples the incoming clock, multiplying its frequency 96 times. The upsampling procedure imposes certain specifics on the LR operation. In particular, if you want to modulate or change the clock during performance and at the same time preserve the synchronization of the LR and the dividers, you must use the synchronization pin LRST, which will be described below.
By default, the clock for LR is taken from the internal clock generator, which can either work independently or receive a MIDI clock (selected by the INT MIDI switch).
The Pulsar's LR has four independent channels (one for each sound generator), each of which has its own independent memory and can have its own independent clock. This provides a unique opportunity to play various LR channels at different speeds, creating complex rhythmic patterns and going beyond repeating patterns. An individual clock is fed to the CLK pins (see below).
Control the LR by using the capacitive sensors that are triggered by the touch of a finger or any conductive object that has sufficient capacity.

LR memory is volatile. When the Pulsar is turned off, the contents of the memory are reset. It is not possible to upload the contents of LR into external memory or download data into LR.

ADD sensors — Add notes to LR loops. Unlike most drum machines, Pulsar recognizes and records not only the start time of a note, but also its length, which blurs the line between the synthesis of percussion sounds and ordinary synthesis, allowing you to make a smooth transition from rhythmic parts to noise and drone soundscapes.
If the REC PLAY switch is in the REC position, ADD will operate in overdub mode, i.e. new notes will be superimposed on already recorded notes. The velocity of recorded notes can be determined by the sensors L (low) and M (middle). By default, the velocity of recorded notes is maximum.
If the REC PLAY switch is in PLAY mode, touching the ADD sensor will play notes on top of those recorded in the LR, but the contents of the loop will not change.
If the REC PLAY switch is in the middle position (MUTE) or the clock is stopped, the ADD sensor will simply trigger the corresponding sound generator.

DEL sensors – Erase notes from LR loops.

If the REC PLAY switch is in the REC position, the DEL sensor deletes notes from the corresponding loop.
If the REC PLAY switch is in the PLAY position, the DEL sensor will mute the recorded notes without changing the contents of the loop.

When in the REC mode, holding DEL while touching ADD to play, will record a new part, while the already recorded notes will be erased (punch-in record mode). If you do the same in PLAY mode, the performance will be played on top of the muted recorded part. In this case, the contents of the loop will not change.

REC PLAY switch – determines the operation mode of the LR channel.
REC – recording in a loop is done while playing back already recorded notes. PLAY – plays back recorded part. Middle position mutes the loop, but it keeps moving according to the incoming clock.

CLK pins are individual clock inputs for each LR channel.
If you connect something with a low output resistance, the internal clock will be automatically turned off and replaced by the incoming signal. Synchronization occurs along the rising edge when the signal exceeds the level of 2 volts. CLK inputs can work with any type of signal: digital, analog, periodic, aperiodic, noise, etc. Thanks to hardware and software protection, any reasonable experimentation with CLK inputs will not cause the LR to freeze or crash. This opens up great opportunities for creating aleatoric compositions and experimental rhythms.

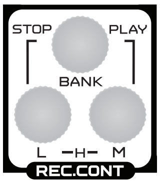
TRIG (trigger) pin – LR channel output and envelope generator input.

LRST pin (looper restart) – applying a positive impulse (rising edge) to this pin will cause LR to restart from the zero position.
This function is needed for synchronizing the LR with the array of clock dividers, and it can be used for shortening the loop length. Naturally, this pin can also be used for all kinds of experiments.
For tight LR synchronization with the array of clock dividers, connect the LRST pin to the 0.25 divider output. This connection will ensure the synchronization of the LR and the divider in the case of modulation of the clock frequency, restart of LR from different positions, etc. Such synchronization may be necessary if you generate part of the rhythmic behavior (for example, control filters and other synthesis parameters) with the divider, and the other part plays from the looper/recorder. Also, such synchronisation is strongly recommended if you use MIDI-clock. Actually, if you have no intention to make asynchronous beats, it's better to keep the LRST and 0.25 pins always connected.

You can make the LR loop length shorter if you connect the LRST to the lower output of the divider (0.5, 1, 2, 4, etc.)

Please remember that the LR has the clock up-sampling process that needs a little time for setting. The first few dozens of milliseconds after MIDI clock start can be inaccurate. To avoid this, the timespan between stop and start on your DAW or sequencer that feeds MIDI-clock to Pulsar, has to be at least 5 seconds (then the up-sampling system stops to wait for the next clock pulse and saves the previous value that will be used after clock start).
A second option is to use the LRST pin for alignment with the clock divider.

BLOCK **REC.CONT** (RECORDER CONTROL)



This is the looper/recorder control unit. It consists of three multi-function sensors: L, M, BANK,

L M sensors — allow you to set the velocity of recorded or played notes.
L (low) — lowest velocity.
M (middle) — medium velocity.
L + M (high) — simultaneous pressing will cause maximum velocity.
In REC mode, the L M sensors let you set the volume of notes recorded by the ADD sensor. If neither L nor M are pressed, the recording velocity is maximum.

In PLAY mode, the L M sensors let you set the volume of notes played by the ADD sensor and modify the volume of notes recorded in the LR.

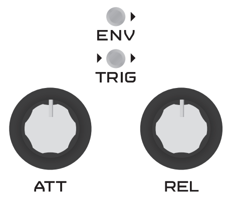
BANK sensor in combination with the ADD or DEL sensors — switches the LR banks.
Each of the four LR channels is also associated with one of the four banks of loops.
To switch bank: while holding BANK, press ADD or DEL of the desired channel.
The activated bank is indicated by a lit yellow LED.
It is possible to partially or fully copy the contents of banks into each other. To do this, hold BANK and press ADD + DEL of the channel you want to copy to. The bank will switch over while copying the contents of the previous bank to the selected bank. Copying occurs on the fly, i.e. without stopping playback and does not interrupt the play process.
Copying will be performed only on those channels where the recording mode is turned on. Copying occurs exactly as long as the combination of three buttons is held. If you need to copy the entire loop, you must hold this combination for at least the length of the entire loop of the looper. You can also copy the loop partially, holding the combination only for the desired fragment. This allows you to mix fragments from different banks. When copying, the previous contents of the bank are erased.

BANK + L (stop) — stops the LR.

BANK + M (start) + one of the ADD or DEL sensors — starts or restarts the LR from the specified position. The LR loop is divided into 8 equal parts. Each part is associated with one of the 8 ADD and DEL sensors. By pressing BANK + M and the desired sensor, you can start the loop from that section. If LR is already running, it will restart from the specified position. This allows for non-integer measures and changing rhythmic patterns.

BANK + L + M + one of the sensors (ADD or DEL) of the desired LR channel — will quantize the content of the channel to 16 notes. The LR will stop when this operation is performed. To start it again, use the start function. For the quantize function to work correctly, please make sure that the LR is aligned with the clock dividers!! To do this, press the RST button on the CLOCK module before recording, and\or connect the LRST pin to the 0.25 output of the clock divider, and use a metronome during the recording!

AR ENVELOPE GENERATORS OF SYNTHESIS MODULES



All synthesis modules (BD, BASS, SD, HHT) have the same envelope generators with two control parameters — Attack (ATT) and Release (REL). The input of the generator is connected to the output of the looper/recorder, the output of the MIDI converter and the TRIG pin. All three sources of the trigger signal can be used simultaneously, but the envelope generator will respond only the strongest signal. For example, if one source sends 2V, a second sends 5V and a third 7V, the EG will respond to the 7V source. If that signal stops, the EG starts to respond to the 5V source instead. If that turns off, it will respond to the 2V source.

ATT (attack) knob — adjusts the attack of a given drum sound.

REL (release) knob — adjusts the release of a drum sound.

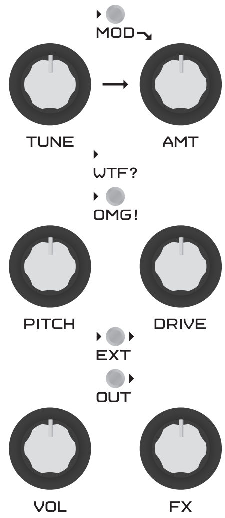
TRIG (trigger) pin — LR channel output and envelope generator input.

ENV (envelope) pin — Envelope generator output.

You can use any of the channels of the looper/recorder and the envelope generator attached to it, regardless of its synthesis module. For example, you can create a CV signal with complex behavior, using an EG. This complex CV signal you can get from the ENV pin.

Like many of Pulsar's inputs, TRIG inputs are touch sensitive. Try to touch the TRIG pins with one hand and the clock divider outputs, SHAOS, LFO outputs, etc. with the other.

BD (BASS DRUM) SYNTHESIS MODULE



Designed for synthesizing the sound of a bass drum. Like other synthesis modules, it can be used to generate a fairly wide range of sounds.
It should be noted that all synthesis modules have a different structure and that their sound palettes do not intersect for this reason.

TUNE knob – controls the pitch of the BD.

MOD (modulation) pin – is the input modulating the pitch of a BD. This input has a linear (volt-hertz) relationship.

AMT (amount) knob – adjusts the amount of influence of the MOD signal on the pitch.

WTF? pin – circuit bending node for the pitch modulator.

OMG! pin – circuit bending node for the triangle waveform generator that forms the base of BD synthesis.

PITCH knob – controls the modulator that generates the pitch jump at the beginning of a sound, characteristic for a bass drum. This knob adjusts the decay rate and the depth of the modulation.

DRIVE knob – controls the BD waveform. When rotated clockwise, the waveform changes from triangle to sine to square.

EXT (external) pin – is the input for processing an external signal through the BD synthesis circuit. Located before the waveshaper/distortion that makes the waveform.

OUT pin – is the output of the BD synthesis module. Located before the VOLUME knob.

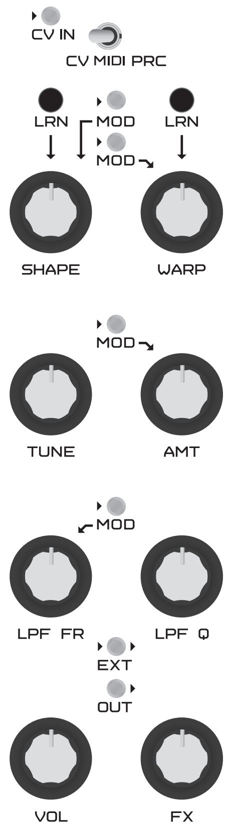
VOL (volume) knob – BD volume control

FX knob – adjusts the send level of the BD signal to the effects processor. It takes the signal before the VOL knob (pre-fader). This way you can send a signal to the effects processor while the BD volume in the main mix is low or zero.

Try modulating the BD pitch with noise (use NOISE pin).

You can create an additional pitch drop typical of a hip-hop style BD by connecting the ENV and MOD pins and setting the AMT knob.

BASS SYNTHESIS MODULE



The BASS synthesis module is a powerful monophonic synthesizer with two modes of operation — classic monophonic synthesis and percussion synthesis of the author's architecture. This allows you to create a wide range of sounds, such as bass, lead, various types of low-frequency and high-frequency percussion, and sound effects. In monophonic synthesis mode, the pitch can be controlled by MIDI and CV (standard 1V/octave logarithmic dependence).
The synthesis module has a hybrid architecture: a digitally controlled oscillator (DCO) followed by an analog processing chain containing a low-frequency resonance filter (LPF) with saturation mode, a voltage-controlled amplifier (VCA) and an envelope generator.
A DCO is built on unique authoring algorithms of pure mathematical synthesis making the sound and behavior of DCO close to analog. Most digital synthesizers use wavetables to generate waveforms, which gives the sound a characteristic digital deadness and stillness caused by the fact that we essentially have a rompler that reproduces the same thing all the time, and not a synthesizer that has natural living breath and many small nuances. The DCO in Pulsar does not contain any tables and instead generates waveforms through special recursive equations, computing with very high accuracy (32 bit with floating point). This makes our DCO sensitive to the smallest changes in control signals and gives the sound a vibrant breath inherent to analog synthesis.
In addition, all DCO controls, except for MIDI, occur through analog circuits, which also brings its behavior closer to analog.

CV IN pin is a logarithmic (volt-octave) input that allows you to control the pitch of a note with a standard CV signal. It has an input voltage range of 0-4 volts (four octaves).

CV MIDI PRC switch — selects the DCO mode of operation. CV — CV control. MIDI — MIDI control. PRC — synthesis of percussive sounds.

SHAPE knob — is a DCO synthesis parameter that controls the waveform. When turning clockwise, the level of harmonics increases.

MOD pin associated with SHAPE: CV control of the SHAPE parameter.

WARP knob — sets the amount of loading of the waveshaper, located after the oscillator.

MOD pin associated with WARP: CV control of the WARP parameter.

In PRC (percussion) mode, the SHAPE and WARP knobs control many synthesis parameters at once, a clear description of which would take up too much space. Therefore, it is easier to explore this experimentally.

TUNE knob — oscillator pitch settings. In MIDI mode it has a range of +/- 1/2 tones, in other modes it has a range of 5 octaves.

MOD pin associated with the AMT knob: CV control of the phase modulation of the DCO. In PRC mode it becomes sidechain input.

AMT (amount) knob — determines depth of modulation by the signal received on the MOD pin.

LPF FR (low-pass filter frequency) knob — controls the cutoff frequency of the resonant low-pass filter.

MOD pin associated with the LPF FR knob: CV control of the filter cutoff frequency.

LPF Q knob — controls the resonance level of the low-pass filter.

EXT (external) pin is the input for processing an external signal through the BASS synthesis circuit. Located before the low pass filter.

OUT pin — output of the BASS synthesis module. Located before the VOLUME knob.

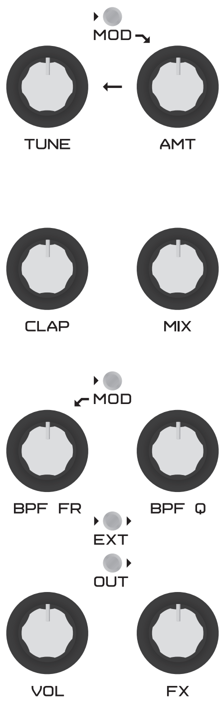
VOL (volume) knob — BASS volume control.

FX knob — adjusts the send level of the BASS signal to the effects processor. It takes the signal before the VOL knob (pre-fader). This way you can send a signal to the effects processor while the BASS volume in the main mix is low or zero.

Regardless of DCO mode (CV MIDI PRC), this module can always be triggered via MIDI. In CV and PRC mode, only one note on the MIDI keyboard can be associated with the module using the learn function (LRN button above the sensors of the BASS module). In MIDI mode, it will be a chromatic MIDI keyboard. If, in PRC mode, the MIDI channel tied to BASS does not coincide with the MIDI channels of the rest of the synthesis modules, it also works as chromatic MIDI keyboard where you can change the pitch of the note.

You can control the portamento parameter through the standard portamento midi controller (CC05).

If you assign any key of the MIDI keyboard assigned to BASS to the first channel of the MIDI to CV converter, the key tracking signal associated with this MIDI keyboard appears at the output of the converter.



SD (SNARE DRUM AND CLAP) SYNTHESIS MODULE

The SD synthesis module is focused on producing the sound of a snare drum and a clap, while remaining, like all Pulsar sound modules, flexible enough to synthesize a wide palette of sounds that goes far beyond the classical snare drum. The heart of this module is a noise generator of original design with a controlled spectrum, which largely determines the characteristic sound of the Pulsar snare drum.

TUNE knob — adjusts the spectrum of the noise generator.

MOD (near the TUNE and AMT knobs) pin — CV input for controlling the noise spectrum.

AMT (amount) knob — sets the amount of influence of the CV received on the MOD pin.

CLAP knob — creates a clap sound by splitting the attack of the sound.

MIX knob — determines the balance between pink and spectral noise.

BPF FR (band-pass filter frequency) knob — determines the cutoff frequency of the band-pass filter.

MOD (near the BPF FR knob) pin is the CV input that controls the BPF cutoff frequency.

BPF Q (band-pass filter resonance) knob — determines the level of self-oscillation of the filter.

EXT (external) pin is the input for processing an external signal through the SD synthesis circuit. Located before the band-pass filter.

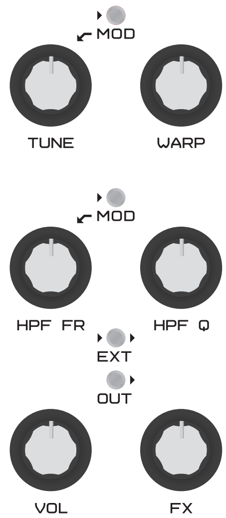
OUT pin is the output of the SD synthesis module. Located before the VOLUME knob.

VOL (volume) knob — SD volume control.

FX knob — adjusts send level of the SD signal to the effects processor. Takes the signal before the VOL knob (pre-fader). This way you can send a signal to the effects processor while the SD volume in the main mix is low or zero.

Use a band-pass filter on the verge of self-oscillation (set by the BPF Q knob) to create the characteristic resonance of the snare drum body.

HHT (HI-HAT) SYNTHESIS MODULE



A module designed to synthesize the sound of a hi-hat, cymbal or shaker.

TUNE knob — adjusts the spectrum of the noise generator.

MOD (near the TUNE knob) pin — CV input controlling the noise spectrum.

WARP knob is a waveshaper that changes the noise spectrum.

HPF FR (high-pass filter frequency) knob — determines the cutoff frequency of the high-pass filter.

MOD (near the HPF FR knob) pin is the CV input that controls the HPF cutoff frequency.

HPF Q (high-pass filter resonance) knob — determines the amount of self-oscillation of the filter.

EXT (external) pin is the input for processing an external signal through the HHT synthesis circuit. Located before the high-pass filter.

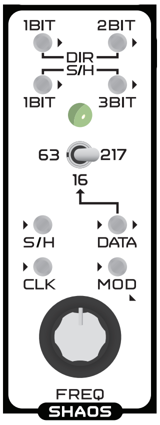
OUT pin is the output of the HHT synthesis module. Located before the VOLUME knob.

VOL (volume) knob — HHT volume control.

FX knob — adjusts send level of the HHT signal to the effects processor. Takes the signal before the VOL knob (pre-fader). This way you can send a signal to the effects processor while the HHT volume in the main mix is low or zero.

To create the sound of a shaker, set the WARP knob to more than 60% and make a soft attack with the ATT knob.

SHAOS PSEUDO-RANDOM GENERATOR



This is a generator of complex pseudo-random signals of the author's design, built on the basis of shift registers. Hence the name SHIFT + CHAOS = SHAOS. It consists of a clock generator, a shift register with a feedback circuit generating a pseudo-random sequence, along with a sample and hold unit that allows sampling from a pseudo-random sequence, synchronized to an external signal.

FREQ (frequency) knob — controls the clock frequency of the SHAOS synthesis.

MOD (modulation) pin — CV input that controls the clock frequency (FREQ parameter).

CLK (clock) pin — input intended for connecting an external clock. When a source with a low output resistance is connected to this pin, the internal clock is automatically turned off and replaced with the external one.

S/H (sample and hold) pin — input for external pulses. With each impulse applied to the S/H pin, the pseudo-random seq will be sampled and held. The S/H function will be synchronized to impulses applied to the S/H pin. If nothing is connected to the S/H pin, sampling and holding are performed in sync with the internal clock of the module. When a source with a low output resistance is connected to this pin, the internal clock is automatically turned off and replaced by the external clock signal.

63 16 217 switch — determines the length of the pseudo-random sequence. It can be equal to 63, 16 or 217 pulses of the internal or external clock.

DATA pin — can be used to write a sequence to the memory of shift registers. When the switch 63 16 217 is in position 16, the shift registers turn into a cyclic memory, into which you can write a short sequence of states using the DATA pin. In the simplest case, connect a wire to this pin and short-circuit it to the +10V and GND contacts, or simply move the switch from the edge positions to the center (16). Doing this records various sequences in the cyclic memory, which will be played in sync with the incoming clock.

1BIT DIR pin is the 1bit output of the pseudo-random sequence without sample and hold (direct). It works regardless of the incoming sync signal sample and hold. The signal has two states (1bit resolution).

2BIT DIR pin — 2bit output of the pseudo-random sequence without sample and hold (direct). It works regardless of the incoming sync signal sample and hold. The signal has four states (2bit resolution).

1BIT S/H pin — 1bit output of the pseudo-random sequence with sample and hold. The output signal is synchronized to the incoming sync signal applied to the S/H pin. The signal has two states (1bit resolution).

3BIT S/H pin — 3bit output of the pseudo-random sequence with sample and hold. The output signal is synchronized to the incoming sync signal applied to the S/H pin. The signal has eight states (3bit resolution).

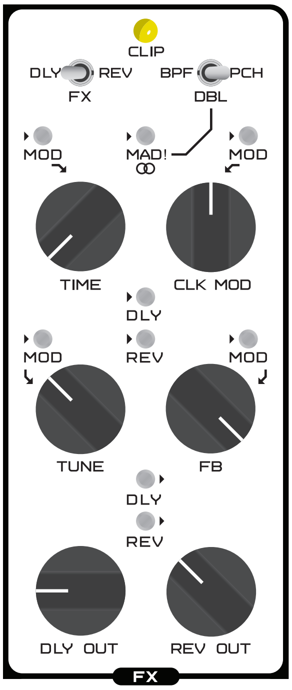
- All SHAOS outputs are shifted relative to each other and produce different sequences.

- If you want SHAOS to work in sync with the master clock, connect the CLK input to the desired output of the master clock divider array.

- If a signal with a frequency that is not related to the internal SHAOS clock, for example from the LFO generator, is fed to the S/H input, the length of the pseudo-random sequence will increase significantly, theoretically to infinity.

- If the SHAOS clock is very fast, the generator will start working in the audio range and can be used to synthesize complex waveforms. A very fast clock can be obtained using the square wave output of the LFO module operating in HI (HIGH) mode.

FX EFFECT PROCESSOR



Pulsar contains a two-channel effects processor. The first channel is different types of delay, the second is reverb. FX has three operating modes:
BPF (band-pass filter) – Channel 1 is a 1-tap delay with an adjustable band-pass filter. Channel 2 is Classic Hall.
DBL (double) – Channel 1 is a 2-tap delay. Channel 2 is a variation of the Classic Hall.
PCH (pitch) – Channel 1 is a 1-tap delay with adjustable pitch shifter in the feedback. Channel 2 is Hall with pitch shifter in the feedback. Both pitch shifters are adjustable within +\- 1 octave. Setting pitch shifters takes place in opposite directions, i.e. when the frequency of one increases, the other decreases.
The FX module has the unique ability to modulate the clock of the entire DSP processor along with the AD DA converters. This means that the speed of the entire processing, together with all the code, can change 7 times! This creates unique sounding effects that cannot be reproduced in this form using a virtual change of the sampling frequency and similar purely software solutions.
The outputs of both channels are mono. However, in DBL mode, a stereo mode is possible when the mono outputs become the left and right channels to which a stereo mix of double delay and reverb is fed. To enable stereo mode, the mode switch must be in the DBL position, and a voltage greater than 5 volts must be applied to the MAD! \ stereo icon \ pin.

DLY REV (delay reverb) switch – determines which of the FX channels the submixer output will be assigned to (FX knobs in the synthesis modules).

BPF DBL PCH switch – determines the operating mode of the FX processor.
BPF (band-pass filter) – Channel 1 is a 1-tap delay with an adjustable band-pass filter. Channel 2 is Classic Hall.
DBL (double) – Channel 1 is a 2-tap delay. Channel 2 is a variation of the classic Hall.
PCH (pitch) – Channel 1 is a 1-tap delay with adjustable pitch shifter in the feedback. Channel 2 is Hall with pitch shifter in the feedback.

CLIP indicator – lights up when overloading the input of the AD converters of the DSP processor.

MAD! / Stereo icon / pin – applying voltage to this input in BPF and PCH modes causes madness in the FX processor; in DBL mode it activates stereo mode. To permanently activate these modes, use the +10V pin.

TIME knob – sets the delay time.

MOD pin associated with the TIME knob – CV input controlling the TIME parameter.

MOD pin associated with the CLK MOD knob – CV input driving the DSP clock.

CLK MOD (clock modulation) knob – This determines the modulation depth of the DSP clock with the signal received at the MOD contact.

TUNE knob — In BPF mode, it adjusts the band pass filter. In DBL mode, it sets the delay time of the second delay line. In PCH mode, it adjusts the pitch interval of the pitch shifter.

MOD pin associated with the TUNE knob — CV input controlling the TUNE parameter.

FB (feedback) knob — sets feedback depth of the delay and the reverb and determines the decay time of reflections respectively.

MOD pin associated with the FB-CV knob — CV input controlling the FB parameter.

DLY (delay) input pin — auxiliary delay input. The signal applied to it will be processed by the delay effect.

REV (reverb) input pin — auxiliary reverb input. The signal applied to it will be processed by the reverb effect.

DLY (delay) output pin — delay output. It can be used for external mixing and for creating non-trivial modulation loops.

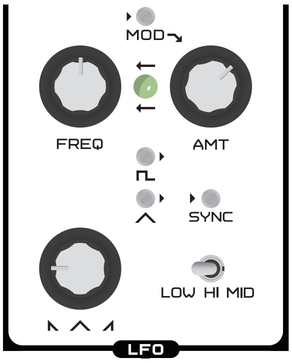
REV (reverb) output pin — reverb output. It can be used for external mixing and for creating non-trivial modulation loops.

DLY OUT knob sets the level of delay effect returning to the main mix.

REV OUT knob sets the level of the reverb effect returning to the main mix.

Try connecting the CLK MOD to the ENV output of the BD module. The DSP sample rate will follow the envelope of the bass drum, creating cool effects.

LFO



FREQ (frequency) knob — sets the frequency of the LFO (low frequency oscillator).

MOD pin — CV input for controlling the frequency of the LFO.

AMT knob — determines the level of influence of the MOD CV on the LFO frequency.

⊓ pin — the square wave output of the LFO.

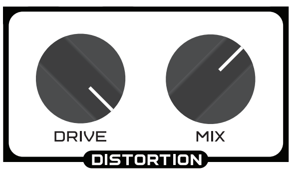
∧ pin — the triangle wave output of the LFO.

SYNC (synchronization) pin — applying a positive impulse to this input causes the LFO to reset to zero. Using this contact, you can sync the LFO to any event in Pulsar's life. For example, this contact can be connected to the TRIG output of one of the synthesis modules or to one of the clock divider outputs. This will synchronize with notes of the selected duration or with the start of the selected drum.

◣ ∧ ◢ knob — sets the output waveform of the LFO to the triangle output pin. Turning this knob provides a smooth transition from falling saw through triangle to rising saw.

LOW HI MID (low high middle) switch — sets the LFO frequency range, which can vary from fractions of hertz to kilohertz.

DISTORTION



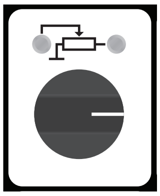
Pulsar contains parallel distortion, to which the main audio mix is fed.

DRIVE knob — determines the distortion drive.

MIX knob — determines the balance of clear and distorted sound.

ADDITIONAL MODULES

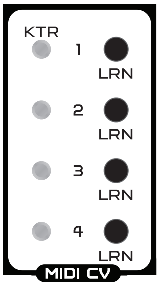
ATTENUATORS



Pulsar contains four assignable attenuators. They are necessary if you need to reduce or control the level of any audio or CV signal. The right contact is the input, the left contact (wire with an arrow) is the output. Using assignable attenuators instead of placing a special attenuator near each CV input allowed us to save a huge amount of space on the Pulsar panel, as well as greatly reduce the price.

To create a metronome, connect the desired output of the clock divider (usually 4 or 2) to the attenuator input, and connect the attenuator output to the MIX IN input.

MIDI TO **CV** CONVERTER



The four-channel MIDI converter allows you to assign four CV outputs to any MIDI controller. To assign an output, simply press the LRN button next to it and turn or press the desired controller or key (send a MIDI message).

The MIDI to CV converter automatically recognizes the key and the continuous controller (CC). In case a MIDI key is assigned, the velocity value of the pressed key will be output. If a controller is assigned, the output will be position of the controller.

Assigning a specific key will give you the opportunity to rhythmically control various synthesis parameters (for example, a filter) from a MIDI keyboard and apply functions such as quantization, which are difficult to use with a continuous controller.

The first output of the converter, marked KTR (key tracking) can generate a CV signal proportionally dependent on the note number playing on the BASS channel. To assign this function, hold the LRN button near the first output and press any key of the MIDI keyboard assigned to the BASS channel. Key tracking is needed if you want to drive a certain parameter, for example the LPF frequency, by note pitch.

INDIVIDUAL COMPONENTS AND IMPULS CONVERTERS

diode is a single radio component, useful for atomic control of signals and circuit bending.

You can modulate the velocity of a trigger output of the looper/recorder with the LFO signal. To do this, connect the left contact of the diode (anode) to the TRIG output, and the right contact of the diode (cathode) to the triangle LFO output. The velocity of selected synthesis module will follow the LFO.

Try to insert the diode in different directions in different modulation and dynamic control circuits.

capacitor is a single radio component, useful for atomic signal control and circuit bending. There are two capacitors, 0.1 mf and 10 mf. The capacitors allow you to cut off the constant component of the signal and form a decay of the desired length for long signals.

Try connecting the low-frequency output of the clock divider through a capacitor to the CV input that controls the filter frequency of one of the modules. For each jump in a rectangular signal, the capacitor will form a smoothly falling envelope, the length of which will depend on the nominal value: 0.1 mf will give a quick decline, 10 mf will be slow.

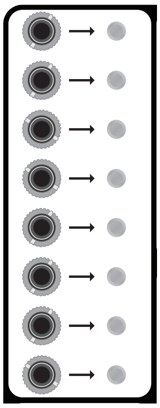
Try inserting a capacitor in various modulation and dynamic control circuits.

By connecting one contact of the capacitor to the ground (GND pin), and the other to a signal, you will create a low-pass filter that cuts off the high-frequency component of the audio signals and softens the attack of the control signals.

pulse converter — designed to convert rectangular signals into short pulses suitable for triggering synthesis modules. Designed to work with the clock divider, but can be used in various experiments.

To get a basic techno rhythm, connect output 2 of the clock divider to the left contact (input) of the first converter, and connect the right contact (output) of the first converter to the TRIG SD input; connect output 4 of the clock divider to the input of the second converter, and connect the output of the second converter to the input of TRIG BD; connect output 16 of the clock divider to the TRIG HHT input.

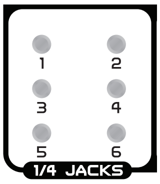
EURORACK — PIN ADAPTER



Pulsar has eight adapters for connecting mini-jacks used in Eurorack to the pins designed to connect the alligator clips. If you need more connections with Eurorack — just attach the alligator clip to the mini-jack TIP.

At least one of the connections between Pulsar and Eurorack modules must be made through the adapter in order to ensure the connection of the ground of the devices. Or you can connect the grounds in any other way.

1/4 INCH **JACK** — PIN ADAPTER

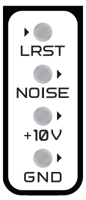


Pulsar has six adapters for connecting the 1/4-inch jack format used in professional audio production, to the contacts intended for the alligator clips. The 1/4-inch jack connectors are located on the rear panel and are numbered in the same way as the adapter pins. If you need more connections with 1/4 inch jack — just attach an alligator clip to the TIP of the jack plug.

At least one of the connections connecting the Pulsar and the external audio system must be made through an adapter in order to ensure the connection of the grounds of the devices. Or you can connect the grounds in any other way.

You can perform external mixing of the Pulsar using stereo panning and spatial effects by connecting the four individual outputs of the synthesis modules and the two outputs of the FX processor through an adapter to an external mixer or DAW audio interface.

INDIVIDUAL SIGNALS



LRST (looper restart) pin — refers to the looper/recorder. Applying a positive impulse (rising edge) to this input will cause the LR to restart from the zero position. This function is needed to synchronize the LR with the clock dividers or to shorten the loop length. Naturally, this contact can also be used for all kinds of experiments.

For tight LR synchronization with the clock dividers, connect the LRST pin to the 0.25 divider output. This connection will ensure the synchronization of the LR and the divider, in the case of modulation of the clock frequency, LR restarting from different positions, etc. Such synchronization may be necessary if you generate part of the rhythmic behavior (for example, control filters and other synthesis parameters) with the divider, and some parts are played from the looper/recorder. Also, such synchronisation is strongly recommended if you use MIDI-clock. Actually, if you have no intention to make asynchronous beats, it's better to keep the LRST and 0.25 pins always connected.

NOISE pin — pink noise output.

+10V pin — 10 volt DC output, protected from overload.

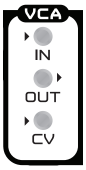
GND (ground) pin — Pulsar ground.

AUXILIARY INPUT TO THE MAIN MIX BUS

MIX IN pin is an audio input for signals that should be added to the main mix.

To create a metronome, connect the output of the clock divider of the desired time signature (usually 4 or 2) to the attenuator input, and connect the attenuator output to the MIX IN.

VCA



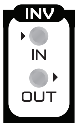
The two VCA modules are two independent CV-controlled amplifiers. They can be used for control and audio signals.

IN pin – controlled signal input.

OUT pin – controlled signal output after VCA processing.

CV pin – control signal input. The voltage at this pin determines the gain of the VCA, which can range from 0 to 1.

INV NON-CONTROLLABLE INVERTER



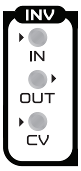
Inverts the incoming signal relatively to the value of +5 volts. It can be used for control and audio signals.

IN pin – inverter input.

OUT pin – inverter output.

How to create a sidechain compressor effect: Connect the ENV BD output to the input of the non-controllable inverter, the inverter output to the CV VCA input, the VCA OUT to the MIX IN input; apply a long signal to the VCA IN, for example, noise from the NOISE pin.

INV CONTROLLED INVERTER



Inverts the incoming trigger signal when a voltage higher than +5 volts is applied to the CV pin. It can only be used for trigger signals since it has a binary output of 0 \ +10V.

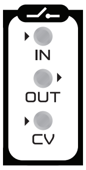
IN pin – inverter input.

OUT pin – inverter output.

CV pin – control voltage input. Applying voltage to this contact will invert the signal applied to IN.

You can use the controlled inverter to control the shift of pulses coming from the clock divider that are used to trigger synthesis modules. For example, you can shift a hi-hat pattern from fourth to unstressed eighth notes. When voltage is applied to the CV pin, the trigger input of the HHT module will begin to respond to the negative edge of the signal, which coincides with unstressed eighths. If you remove the voltage from the CV pin, the HHT beats will return to fourth notes.

CONTROLLED SWITCHES



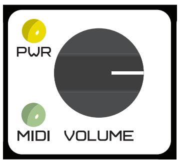
There are two voltage-controlled switches. They can be used for control and audio signals. Applying a voltage of more than +5 volts to the CV pin will turn the switch on.

IN pin — switch input.

OUT pin — switch output.

CV pin — control voltage input. Applying voltage to this pin will short-circuit IN and OUT.

MASTER VOLUME

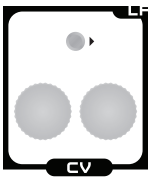


VOLUME knob — adjusts the volume of the main output and headphones output of the Pulsar.

PWR (power) indicator — lit when Pulsar is switched on.

MIDI indicator — Red — a MIDI signal is received, but not assigned anywhere; Green — a MIDI signal is received and is assigned to a function.

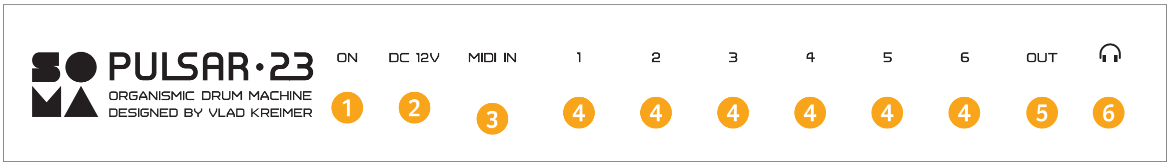
TOUCH-CONTROLLED **CV** GENERATORS



There are two resistive sensors generating CV, each of which can be used for some kind of control, for example, to control the filter cutoff frequency. Unlike the capacitive sensors of the looper/recorder, these sensors work on the conductivity of the skin. Therefore, to activate the sensors, you need to put a finger between the two sensors, closing them through your body. The sensors are dynamic, i.e. they respond to pressure, touch area and skin moisture.

The pin — output CV.

REAR PANEL



1 **Power switch.**

2 **DC IN.** 12 volts, 0.3 amperes, plus in the center. Only a well-stabilized power supply must be used! If the supplied PSU is out of order, we recommend a modern switching mode power supply with a wide input AC range. They have excellent stabilization.

3 **MIDI IN** (5-pin DIN).

4 **Six 1/4-inch connectors** for jack-to-pin adapter (see 1/4 INCH JACK—PIN ADAPTER).

5 **Main audio out.**

6 **Headphones out** (stereo mini-jack 3.5 mm).

SPECIFICATIONS

The range of input and output CV and audio signals . . . . 0 to + 10V
(Pulsar inputs are protected against overloads and can receive signals in the range of -20 + 20 volts for a long time without problems)
Main output voltage swing 2V
1/4 inch jack . . . . 7 total
Mini-jack (for Eurorack connection) . . . . 8 total
MIDI input . . . . standard DIN socket
Headphone output . . . . .3.5 mm mini-jack
Supply voltage . . . . .12 volts (plus in the center)
Consumption current . . . . .0.3 amperes
Only a well-stabilized power supply must be used! If the supplied PSU is out of order, we recommend modern switching mode power supplies with excellent stabilization for replacement !! WATCH THE POLARITY !!
Weight . . . . 4 kg
Dimensions . . . . .380x280x80 mm

WHAT IS IN THE KIT

Pulsar-23
Power Supply 12 V
20 x 65 cm patch cables with alligator clips
10 x 30 cm patch cables with alligator clips
Protection and transportation soft bag

ABBREVIATIONS AND SHORTENINGS

+10V — DC 10 volt
AMT — amount
ATT — attack
BD — bass drum
CLK — clock
DEL — delete
DIR — direct
DLY — delay
ENV — envelope
FB — feedback
FR — frequency
FREQ — frequency
GND — ground
H — high
HHT — hi-hat
L — low
LRST — looper restart
LRN — learn
M — middle
MOD — modulation
OMG! — oh my God!
PRC — percussion
PWR — power
Q — resonance
REC — record
REC.CONT — recorder control
REL — release
REV — reverb
RST — reset
S/H — sample and hold
SD — snare drum
SYNC — synchronization
TRIG — triggering
VOL — volume
WTF? — ...

PULSAR·23 TEAM:

Adam Brewczynski — EU commercial department.
Anastasia Azartsova — top and rear panel design.
Andrzej Slowik — EU production management and control.
Arseniy Vasylenko — web administration.
Evgeny Aleynik — lawyer consulting.
Grigory Ryazanov — industrial construction for mass production.
Grzegorz Lacek — EU management and communications.
Max Bogdanov — management and communication.
Maxim Manakov — development assistance.
Pawel Wieczorek — EU production technologies.
Thomas Lundberg — editor and proofreader.
Valeriy Zaveryaev — manual design and layout.
Viktor Grigoryev — help in design and technology, RU production.
Vitaly Zhidikov — RU commercial department.
Vyacheslav Grigoryev — production technology, RU production director.

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