

# FABRIKAT

Synthetic ambiance, sonic deconstruction and spectral choirs. FABRIKAT is a 100% inorganic, experimental multi-effects processor featuring 16 algorithms focusing on granular synthesis and modified sample playback. Among its algorithms you'll find granular time stretching, pitch shifting time stretching, freeze and shuffling effects, glitching sample playback and a beat repeater. FABRIKAT is aimed for use both as a guitar pedal and as a tabletop device as some of its algorithms greatly reward live parameter adjustment.

FABRIKAT has been developed in tandem with the BandOrg FORM workshop series. Many thanks to BandOrg for funding the software development. Additionally many thanks to Jo Tandrevold, Sigbjørn Håland, Stephan Meidell, Bjørn Ognøy and Kent Sommer for their input on the algorithms and hardware. FABRIKAT is dedicated to Frode Mjøs Johnsen.

*The new FABRIKAT v3 differs slightly to the older versions. Other than the new impedance switch (see internal parameters) these differences does not affect the audio, but the way the pedal is operated.*

## PARAMETERS

**MIX:** Sets the dry/wet balance from 100% dry to 100% processed audio.

**VOL:** Sets the master volume for the pedal. Unity gain is at 12 o'clock. Max gain is +16dB.

**EQ:** Lo/hi frequency emphasis at min/max. Flat response at noon. Interacts with the feedback knob.

**FEEB:** Sets the amount of resynthesis feedback across the digital signal processor (DSP) and the EQ control. Adding feedback yields different effects for different algorithms. Ascending/descending pitch shifting, chaotic stacking and synthetic decay are some possibilities. Excessive runaway feedback will make the bypass LED flicker as a warning when the pedal is bypassed.

**Right footswitch:** This is your bypass switch. Holding the switch longer than 500 ms will only momentarily change its status.

**[1 2 3 4 5 6 7 8] & [1 2]:** This rotary encoder is your algorithm selector. FABRIKAT has 16 algorithms divided into 2 banks. Jumping between algorithms will erase the content of the DSP memory buffer. Jumping from algorithm 8 to 1 will switch effect bank. Thus the rotary encoder has 16 settings (2 x 8).

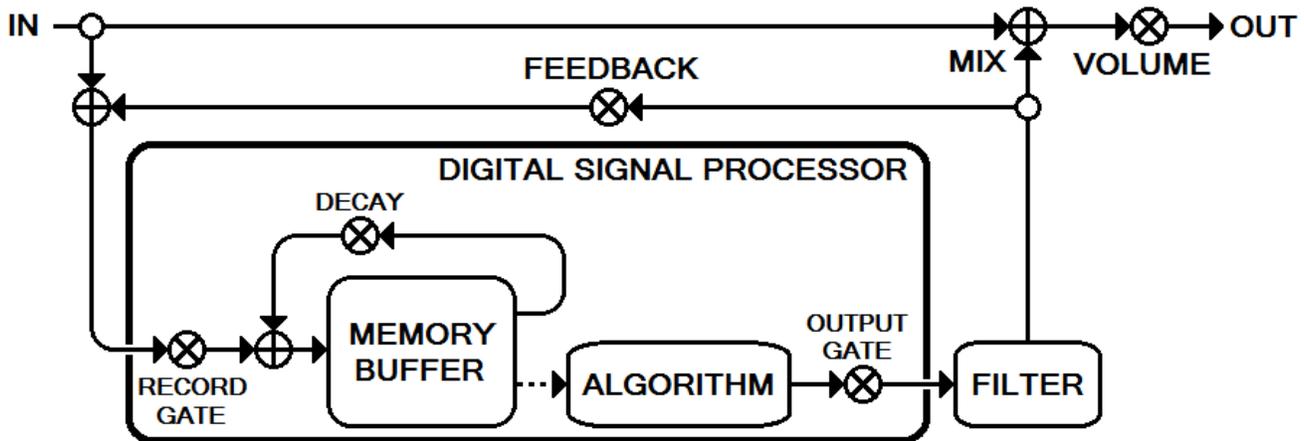
**P1 & P2:** The functions of these knobs are depend on the selected algorithm.

**EXP socket & P1 / P2 switch:** The EXP socket lets you connect an expression pedal or a control voltage source (0 to 3.3V). The EXP input can be mapped to either the P1 or P2 knob by adjusting the toggle switch. In the center position EXP is inactive.

**Left footswitch:** Activating the left footswitch overrides the LOOP knob, replacing its setting by that of the infinity-toggle. This allows you to swap between different LOOP settings on the fly. Holding the switch longer than 500 ms will only momentarily change the LOOP setting. More on the infinity-toggle below.

**LOOP & Infinity toggle:** Common for all the algorithms is that they store audio in a memory buffer. This audio is played back in a manipulated manner unique to the specific algorithm. What LOOP does is decide whether new audio will be recorded to the buffer, how long that audio will remain in the buffer and if the DSP will output audio. Think of it as an odd delay pedal where LOOP controls the input volume, output volume and number of repeats with one knob:

Min: Audio is written to the buffer, but the DSP output is muted.  
 Min to 9 o'clock: Audio is written to the buffer and the DSP outputs audio.  
 9 o'clock to 3 o'clock: Turning the knob clockwise stores audio for longer durations.  
 3 o'clock to max: Audio is stored indefinitely. *Same as the lower position of the infinity-toggle.*  
 Max: Audio is stored undisturbed indefinitely as the DSP input is muted. *Same as the upper position of the infinity-toggle.*



## INTERNAL PARAMETERS

There are two internal miniature switches. **Z/100** reduces the input impedance to 10kOhm. **PAD** adds a -10dB input pad (damping). These may be useful when running line level signals through the pedal. Note that this will change the unity position of the volume knob. The unlabeled trimpot is associated with the expression input and should not be adjusted.

## ALGORITHMS

When exploring a new algorithm it's recommended to start off with FB at min and VOL, MIX, EQ and LOOP at noon. Make some noise and turn LOOP to max. Play around with P1 and P2 to get a lay of the land. 'Buffer size' parameters are destructive, meaning that adjusting the parameter permanently alters the content of the buffer. All other P1 and P2 parameters are non-destructive.

Algorithm 1 through 5 on bank 2 are variable speed sample playback systems. These work similarly to manipulating the playback speed of a tape recorder. E.g. "+1 octave" refer to audio being played back at twice the regular speed, causing a +1 octave pitch transposition. These algorithms can be combined with the FB knob to create ascending/descending pitch shifting effects.

The remaining 11 programs (all of bank 1 + algorithms 6, 7 and 8 of bank 2) are granular synthesizers. This is a method that breaks audio into short snippets (or grains) and rearrange them to create various effects. Using a granular approach you can perform time stretching and scrubbing without changing the signals pitch. However granular synthesis affects the character/timbre of your audio through its "patchwork" process.

## **Bank 1: (B1)**

### **1. Time Stretch with Buffer Size**

P1 : Buffer size. From 100 to 1000 ms. Similar to a delay time knob.

P2 : Time stretching speed. At the middle position the audio is frozen. Moving the knob away from the middle position increases the playback speed. Clockwise yields forward playback while counterclockwise yields reverse playback. At the min/max positions the audio is played back at the original speed.

### **2. Time Stretch with Pitch Shifting (1000 ms buffer)**

P1 : Grain pitch transposition. From -1 octave to +1 octave.

P2 : Time stretching speed. Same as last algorithm.

### **3. Time Stretch with Grain Size (1000 ms buffer)**

P1 : Grain size. From 10 to 200 ms. Adjusts the timbre of the time stretching effect. Shorter grain sizes will skew lower frequencies.

P2 : Time stretching speed. Same as last algorithm.

### **4. Time Stretch Pendulum with Grain Size (1000 ms buffer)**

P1 : Grain size. Same as last algorithm.

P2 : Time stretching speed. Alternates between stretching your audio forward and in reverse. At the minimum position your audio is frozen. Max yields regular speed playback.

### **5. Time Stretch Random Pendulum (1000 ms buffer)**

P1 : Random direction generation rate. From 0 to 10 Hz. Alternates between stretching your audio forward or in reverse in a random sequence.

P2 : Time stretching speed. Same as last algorithm.

### **6. Freeze/Scrub with Spread (1000 ms buffer)**

P1 : Grain randomization. Adjusts the fluidity of the freeze effect. Static at min.

P2 : Sets the window position. For performing manual sample scrubbing/time stretching.

### **7. Freeze/Scrub with Pitch Shifting (1000 ms buffer)**

P1 : Grain pitch transposition. From -1 octave to +1 octave.

P2 : Sets the window position. Same as last algorithm.

### **8. Beat Repeater (1000 ms buffer)**

P1 : Window/grain size. The size of the memory buffer segment that is continuously repeated. From 10 to 1000 ms. The beat repeater is a non-randomized single grain system giving it a harsh and static quality.

P2 : Sets the window position. Same as last algorithm.

## **Bank 2: (B2)**

### **1. Sample Playback – 1 Head Manual Sweep**

P1 : Buffer size. From 100 to 1000 ms.

P2 : Adjustable playback speed/pitch. At min/max you get +1 octave in reverse/forward. The middle position causes a tape stop.

### **2. Sample Playback – 1 Head Octave Steps**

P1 : Buffer size. From 100 to 1000 ms.

P2 : Adjustable playback speed/octave. This knob is locked into 9 steps:

+1 Rev, 0 Rev, -1 Rev, -2 Rev, Tape Stop, -2 Forw, -1 Forw, 0 Forw, +1 Forw.

### 3. Sample Playback – 2 Heads Manual Sweep (1000 ms)

P1 : Adjustable playback speed/pitch for head 1. Similar to algorithm 1.

P2 : Adjustable playback speed/pitch for head 2.

### 4. Sample Playback – 2 Head Octave Steps (1000 ms)

P1 : Adjustable playback speed/octave for head 1. Similar to algorithm 2.

P2 : Adjustable playback speed/octave for head 2. This algorithm has a slightly harsher voicing than the other sample playback systems.

### 5. Sample Playback – 1 Head Random Octave/Direction

P1 : Buffer size. From 100 to 1000 ms.

P2 : Random octave and direction generation rate. From 0 to 40 Hz. +/- 1 octave range.

### 6. Shuffle Delay

P1 : Delay time (buffer size). From 100 to 1000 ms.

P2 : At min this algorithm behaves as a regular delay. Clockwise adjustment increases the amount of shuffling. Use the FB knob for accumulative shuffling.

### 7. Shuffler

P1 : Buffer size. From 100 to 1000 ms.

P2 : Grain size. From 20 to 400 ms. The content of the buffer is randomly rearranged. Small grains skew low frequencies while large grains transform shuffler into a delay-like effect.

### 8. Shuffle/Mask (250 ms)

P1 : Grain masking ratio. Counterclockwise adjustment will increase the probability for a grain to be masked (muted). At min every grain is masked while at max none are masked.

P2 : Grain size. From 10 to 200 ms.

## TECHNICAL SPECIFICATIONS

Input Impedance	1M $\Omega$
Output Impedance	<1k $\Omega$
Power supply	9 VDC center negative (normal BOSS/Ibanez/1Spot power supply) Does not support battery operation
Current Draw	100 mA
Dimensions	125 x 95 x 55 mm
Weight	~440 g