# FABRIKAT v2

*The goldplate FABRIKAT has some slight differences compared to the handstamped graphics FABRIKAT. These differences does not affect the audio, but the way the pedal is operated.*

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 FORM2 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.

## 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 ephasis at min/max. Flat response at noon. Interracts with the feedback knob.

**FB:** 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.

**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:** 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.

**B1 / B2 switch:** Chose between the algorithms in bank 1 or 2. After changing bank you'll have to readjust the algorithm selector before the DSP will skip from the currently running algorithm.

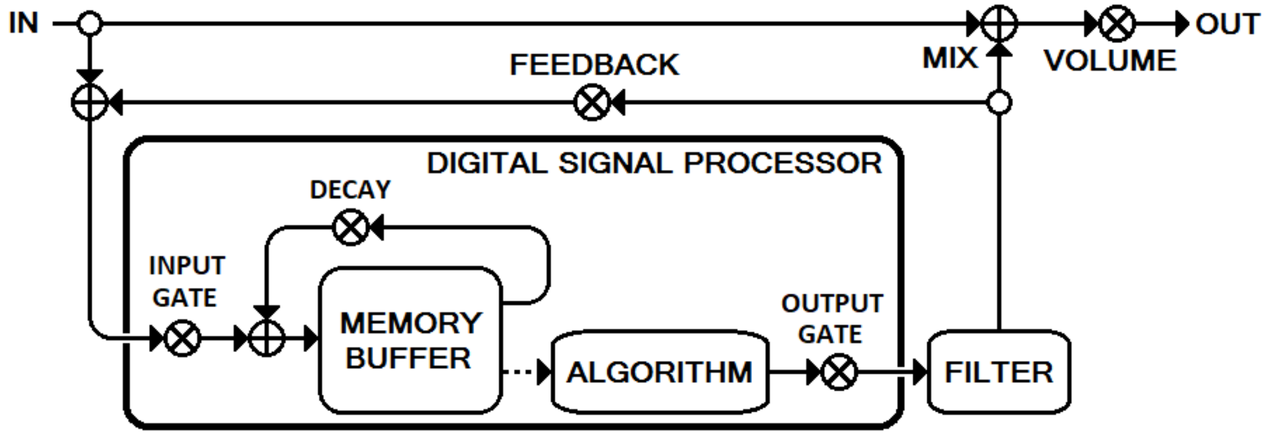
**P1 & P2:** The functions of these knobs are depend on the selected algorithm.

**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.

**L.SW & the left footswitch:** Activating the left footswitch causes the L.SW knob to override the LOOP knob. 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.

**LOOP:** 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'clk: | Audio is written to the buffer and the DSP outputs audio. |
| 9 o'clk to 3 o'clk: | Turning the knob clockwise stores audio for longer durations. |
| 3 o'clk to max: | Audio is stored indefinitely. |
| Max: | Audio is stored undisturbed indefinitely as the DSP input is muted. |




## INTERNAL PARAMETERS

There are two internal miniature switches. **Z/10** reduces the input impedance to 100kOhm. **-9dB** adds a -9dB 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 unlabled trimpot is associated with the expression input and should not be adjusted (nothing interesting will happen).

## 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 (B2) are variable speed sample playback systems. These work similarly to manipulating the playback speed of a tape recorder. E.g. "+1 octave" referres 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 B1 + algorithms 6, 7 and 8 of B2) 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 | 1MΩ |
| Output Impedance | <1kΩ |
| Voltage | 9 VDC center negative (normal BOSS/Ibanez/1Spot power supply)<br>Does not support battery operation |
| Current Draw | ~90 mA |
| Dimensions | 124 x 95 x 52 mm |
| Weight | ~410 g |