

FORM2

FORM2 is a programmable effect pedal built around the Spin FV-1 digital signal processor (DSP). The pedal is made up of a DSP core, a "wrapper" consisting of an analog dry/wet mixer with various parameters and two mappable expression functions for overriding its three digital parameter knobs. FORM2 comes with 24 programs as well as 8 empty program slots.

Analog Audio Circuit

VOLUME - the master volume from the pedal. Unity gain around noon. Maximum gain is +16dB.

MIX - the mix of processed and unaffected audio. From 100% dry to 100% wet audio.

TONE - the cut-off of a lowpass filter affecting only the wet audio. From 500 Hz to 22kHz.

FEEDBACK - the amount of feedback across the DSP and the lowpass filter. An internal trimpot can be tuned to adjust the maximum available amount of feedback.

ENGAGE – your bypass switch utilizing relay-based true bypass. If the stompswitch is held down for longer than 500 ms it will momentarily hold the current mode (bypass or active) and go back to the previous mode when released.

Digital Audio Circuit

I, II and III and programmable knobs. Their functions depend upon the current program. The **PROGRAM** and **BANK** rotary knobs lets you cycle through 8x4 (32) programs. Changing the bank wont change your program until a new program has been chosen.

Expression Functions

The **SWITCH** stompswitch can be mapped to override one of the three programmable knobs. This is done by adjusting the SWITCH toggle. When activated the SWITCH knob will decide the setting for the overridden parameter. Think of it as a parameter preset function. SWITCH can be momentarily (dis)engaged by holding the footswitch down longer than 500 ms.

The **EXP/CV** socket on the backpanel can be connected to an external expression pedal or control voltage source (0 – 3.3V). When an expression pedal is connected it automatically overrides one of the three programmable knobs, set by the EXP/CV toggle. The analog knobs can not be controlled by the EXP/CV or SWITCH functions.

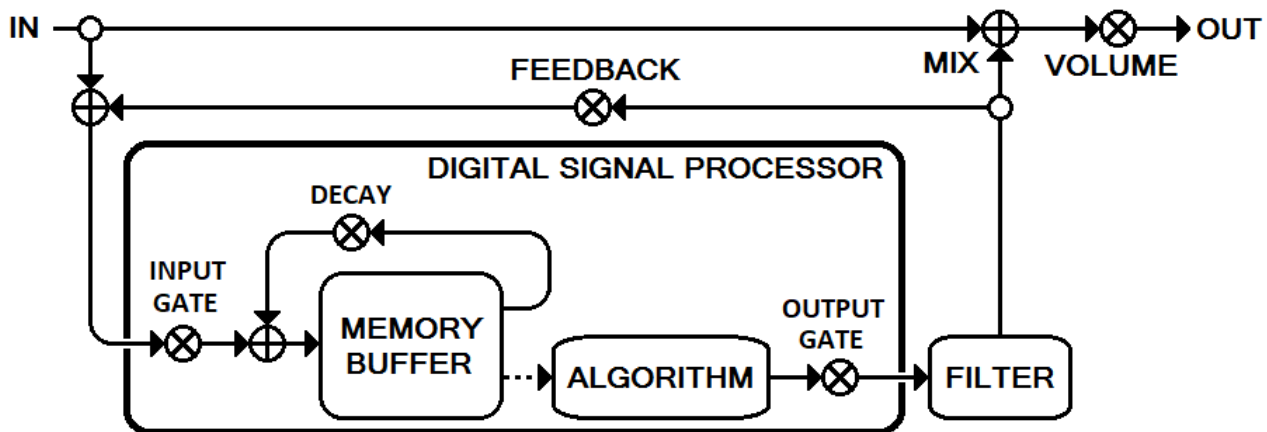
Programming

The DSP core reads programs from ROM chips. Bank 4 is empty, and you may fill it with your own programs. Its associated socket making it easy to insert/remove a chip for uploading purposes. Your ROM chip should be a 24LC32A in an 8-DIP package. It is located on the MAIN board next to the SWITCH stompswitch. Be sure to disconnect the powersupply when inserting/removing the chip.

Programs

LOOP (Bank 1 and 2) - Parameter for knob "I" for all programs in Bank 1 and 2. Common for all these 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.
 Max: Audio is stored undisturbed indefinitely as the DSP input is muted.



Buffer size parameters are destructive, meaning that when you adjust the buffer size you permanently manipulate the content of the memory buffer. All other II / III parameters are non-destructive for bank 1 and 2.

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” refers to audio being played back at twice the regular speed, causing a +1 octave pitch transposition. These algorithms can be combined with the feedback 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: [Time stretch and freeze effects]

1. Time Stretch with Buffer Size

II : Buffer size. From 100 to 1000 ms. Similar to a delay time knob.
 III : 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)

II : Grain pitch transposition. From -1 octave to +1 octave.
 III : Time stretching speed. Same as last algorithm.

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

II : Grain size. From 10 to 200 ms. Adjusts the timbre of the time stretching effect. Shorter grain sizes will skew lower frequencies.
 III : Time stretching speed. Same as last algorithm.

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

II : Grain size. Same as last algorithm.

III : 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)

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

III : Time stretching speed. Same as last algorithm.

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

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

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

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

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

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

8. Beat Repeater (1000 ms buffer)

II : 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.

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

Bank 2: [Sample Playback and Shuffling effects]

1. Sample Playback – 1 Head Manual Sweep

II : Buffer size. From 100 to 1000 ms.

III : 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

II : Buffer size. From 100 to 1000 ms.

III : 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)

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

III : Adjustable playback speed/pitch for head 2.

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

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

III : 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

II : Buffer size. From 100 to 1000 ms.

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

6. Shuffle Delay

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

III : 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

II : Buffer size. From 100 to 1000 ms.

III : 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)

II : 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.

III : Grain size. From 10 to 200 ms.

Bank 3: [Various experimental effects]

1 - Single Sideband / Ring Modulator

I : Coarse frequency control. From a phase shifting tremolo to ring modulation.

II : Fine frequency control

III : Upper / Lower sideband balance. Set to noon for ring modulation. At max/min the upper/lower sideband is isolated leaving you with a linear frequency shifter.

2 - Sample/Hold Lowpass Filter

I : Step rate. 0 to 20 Hz.

II : Filter resonance

III : Filter maximum depth (cutoff frequency)

3 - Random Vibrato

I : Modulation rate and the max available random modulation rate

II : Fixed vs Random modulation rate ratio.

III : Modulation depth

4 - Tape Delay Simulator

I : Delay time. Use feedback to add repeats.

II : Flutter color/voicing. From fluid and clean to noisy and harsh.

III : Flutter Amount

5 - Texture Reverb

I : Reverb decay. The maximum decay time is somewhere close to 10 minutes.

II : Post reverb flutter modulation amount

III : Post reverb saturation amount

6 - Octave Reverb

I : Reverb decay.

II : Dry/Octave balance

III : +/-1 Octave balance

7 - Octave Generator / Delay

I : Delay time. Use feedback to add repeats.

II : Lower octave volume

III : Upper octave volume

8 - Pitch Tremolo

I : Effect rate. 0 to 20 Hz. Jumping between base pitch and a set pitch shift interval.

II : Pitch glissando

III : Pitch jump interval. -12, -7, -5, 0, +7, +12, +19 semitones.

Technical Specifications

Operating Voltage:	9VDC w. center negative polarity (standard Boss, Ibanez, 1Spot PSU)
Current Draw:	90 mA
Input Impedance:	1 M Ω
Output Impedance:	~1k Ω
Dimensions:	250 x 220 x 55 mm
weight:	650 g

Interface

