

PLANKTON ELECTRONICS  
SPICE USER MANUAL

rev0

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## WARRANTY

Plankton Electronics warrants this product to be free of any defect in the manufacturing or materials for a period of one year from the date of purchase. This warranty does not cover any damage or malfunction caused by incorrect use – such as, but not limited to, power cables connected backwards, excessive voltage levels, or exposure to extreme temperature or moisture levels.

The warranty covers replacement or repair, as decided by Plankton Electronics. Please, contact our customer service for a return authorization before sending the module (support@planktonelectronics.com). The cost of sending a module back for servicing is paid for by the customer.

Plankton Electronics implies and accepts no responsibility for harm to person or apparatus caused through operation of this product.

## ACKNOWLEDGMENTS

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## ABOUT THIS MANUAL

**SPICE** is modular distortion unit. It has 6 different functions that can work independently or as a group. Think about different modules prepatched between them. Some of the functions, like the feedback, can not be used on its own, they depend on the other functions.

**SPICE** can be tricky to use. The first operation is simple. Sound destruction can be achieved very fast by playing the controls randomly. To extract subtler effects from it, a deeper knowledge is needed. Unless you're an experienced user, reading this **manual**, watching the **Youtube** videos at and reproducing the patches of the **metapatch** are mandatory steps to master the machine.

Metapatch link: [spice.metapatch.com](http://spice.metapatch.com)

## POWER

**SPICE** needs a 15V 1000mA DC center positive power supply. Be sure that you use the correct power supply. Don't mess it up!

As an **eurorack** module it needs 195mA on the +12V rail and 160mA on the -12V one.

## RACKING

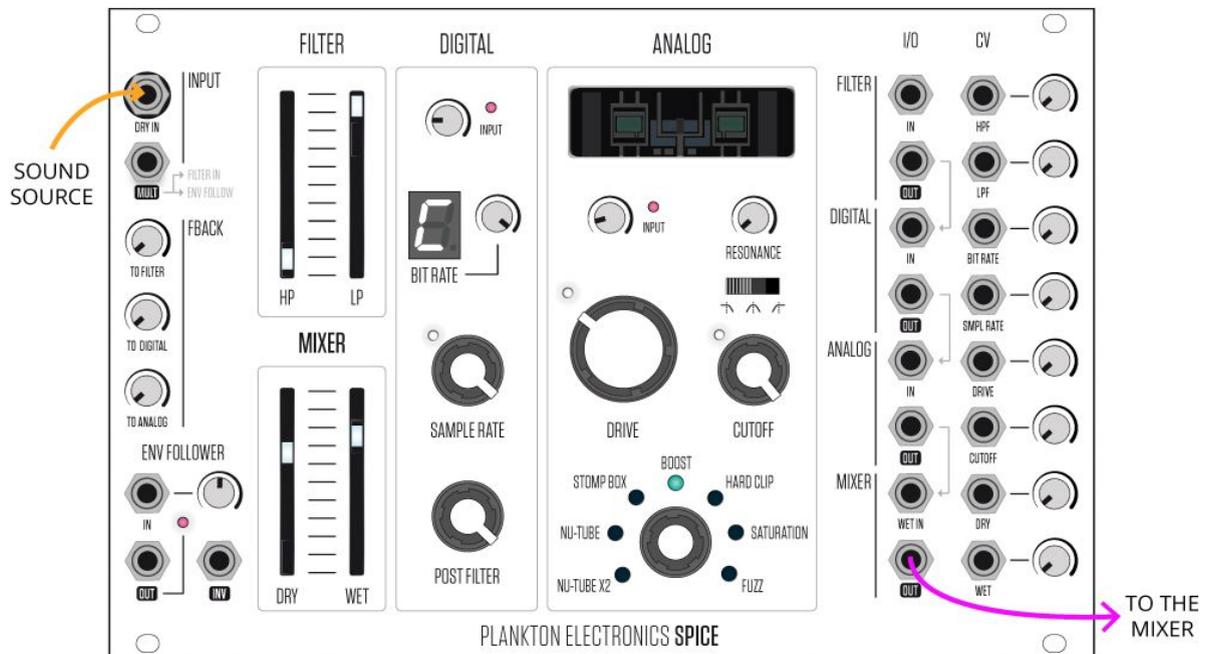
**SPICE** can be used in a desktop format or as a eurorack module. If you want to rack it follow these steps:

1. Turn OFF the unit and unplug the power cable.
2. Unscrew the 4 screws on the corner.
3. Pull the unit from the case. This can be tricky. If you use a tool be careful to not to scratch the panel. Use it only on the eurorack screw holes.
4. Disconnect the power cable from the case.
5. With the eurorack case turned OFF connect the eurorack cable to the eurorack PSU.
6. Screw the panel to the rack.
7. Turn ON the power. Patch.

## FIRST STEPS

Before reading the manual you will like to play the unit. This chapter may add some light to the first contact with the machine.

First of all set up the unit like in the following picture. (You can also find this patch at: <http://spice.metapatch.com/patch/9492>)



If you have patched it like in the picture you have to listen to some sounds now. Some notes about this initial patch:

- Adjust the gain ("INPUT") potentiometers on the "DIGITAL" and "ANALOG" sections so they start to blink. If you're using an eurorack signal they will start to blink very quickly. With lower signals, like a guitar, you will need to turn up the signal a lot.
- The "HP" fader must be at its lower position while the "LP" at its higher position. This way you're not filtering the sound.
- All the "FBACK" potentiometers must be turned down.
- The "BITRATE", "SAMPLE RATE" and "POST FILTER" potentiometers on the "DIGITAL" section must be fully open.
- The filter on the "ANALOG" section must be opened and on the LPF setting (switch to the left).

Now follow this **steps** (go back before moving to the next step) :

- Turn down the "BITRATE" and "SAMPLE RATE" potentiometers.
- Move the "DRY" and "WET" faders and listen to the difference between the clean and processed signals
- Select the different analog distortion circuits.
- Bypass the digital section by connecting a patch cable from the "FILTER OUT" to the "ANALOG IN".
- Add some resonance with the "BOOST" or "SATURATION" circuits and mix with the result with the "DRY" signal.
- Try the different "ANALOG" filter settings (LP, BP and HP)
- Turn up the "HP" fader and see how it affect the different distortion circuits.
- Use a drum beat and connect the inverted output "INV" of the "ENV FOLLOWER" to the the "DRIVE" CV input (turn up the attenuator).
- Repeat the above with the different CV inputs. Try the standard output "OUT" of the "ENV FOLLOWER".
- Patch the "ANALOG" "OUT" to one of the "CUTOFF" CV input. Turn up the resonance.
- Invert the "ANALOG" and "DIGITAL" sections: [spice.metapatch.com/patch/9489](https://spice.metapatch.com/patch/9489)
- Turn up the "DRIVE" on the "ANALOG" section and turn up the different "FBACK" potentiometers. At this point try these steps:
  - Experiment with the different "ANALOG" circuits.
  - Change the "FILTER" parameters.
  - Change the gain ("INPUT") of the "DIGITAL" and "ANALOG" sections.

Have you had fun? This is just the beginning. Let's see how the machine works.

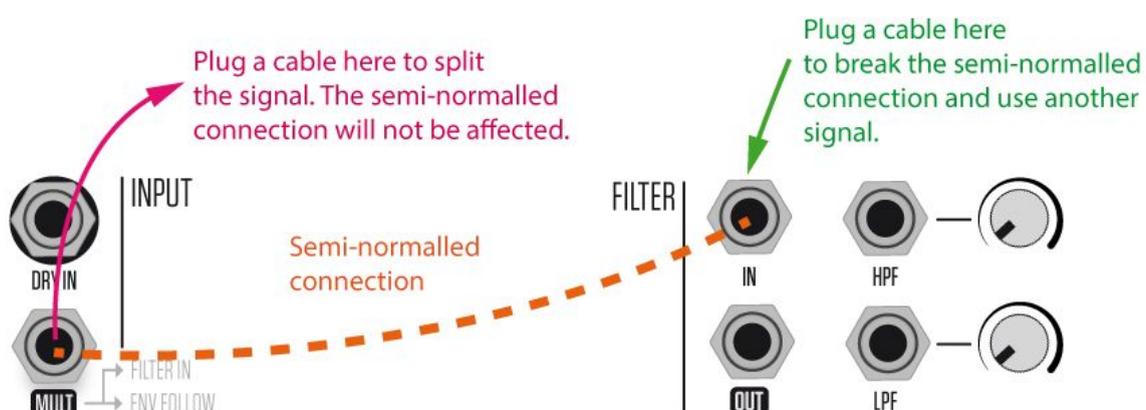
## OVERVIEW

**SPICE** has different functions prepatched between them. These functions are (as labeled in the unit):

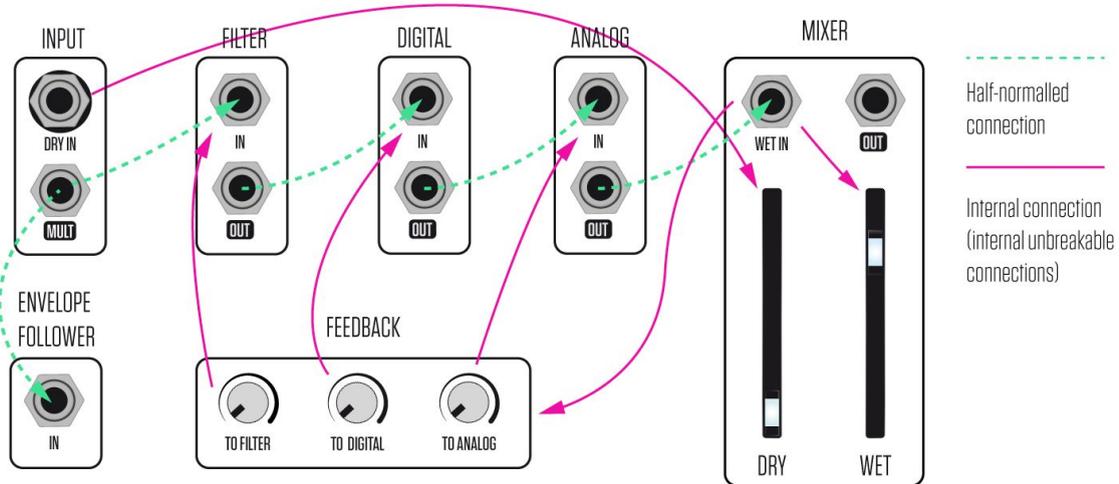
- **FILTER:** a prefilter used to shape the sound being fed to the distortion stages. It has a high pass and a low pass filters with 2 CV inputs.
- **DIGITAL:** a low-fi digital section with fractional bit rate reduction, sample rate reducing, bit muting, bit inverting, bit swapping, and a low pass filter. CV inputs for the bit rate and sample rate functions.
- **ANALOG:** 7 distortion circuits followed by a resonant state variable filter. 2 CV inputs for the "DRIVE" and the "CUTOFF" controls.
- **MIXER:** a two channel VCA to mix the DRY and WET signals with CV inputs.
- **ENV FOLLOWER:** An envelope follower with normal and inverted inputs. It takes an audio input and convert it to an envelope.
- **FBACK:** a triple feedback section that takes signal of the "WET IN" back to the "FILTER", the "DIGITAL" and the "ANALOG" sections.

## PATCHING SYSTEM

**SPICE** is pre-patched (or half-normalled). This means that some inputs and outputs are already internally connected. You can break any of these connections by connecting a cable into the input (destination) of the half-normalled connection. By connecting a cable at the output (source) you are splitting the signal so that output would be available for another use while the normalled connection will remain the same.



The half-normalled routing is highlighted in the machine with the grey text. Below there is a schematic with all the connections.



There are some internal connections (pink) that can't be broken. These are:

- The "DRY IN". It goes directly to "DRY" VCA of the "MIXER" section. It's splitted to the "MULT" output to be processed to any of the sections (or external modules).
- The whole "FBACK" section. It take the audio from the "WET IN" and sends it the input of the "FILTER", "DIGITAL" and "ANALOG" sections.



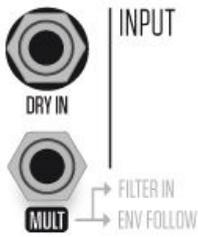
*It's important to understand that SPICE can be used in many different ways. The prepatched system is just one of the possible configurations. Think about SPICE as 5 different modules that can be patched freely, independently or as whole, with or without other modules. Think out of the box.*

## LED SYSTEM

There are two LED colors on SPICE. White and Pink:

- **White** LEDs indicate a **position**, an state or a CV voltage. The 7 LEDs around the encoder indicate the selected analog circuit. The LEDs on the "HP", "LP", "DRY" and "WET" faders plus the "SAMPLE RATE", "DRIVE" and "CUTOFF" ones indicate a voltage. This voltage is the sum of the potentiometer and its related CV input.
- **Pink** LEDs indicate an **audio** level. In the case of the the "ENV FOLLOWER" LED, it indicates the audio level. **This LED must blink** in order to have a proper envelope at the output. The "INPUT" LEDs at the "DIGITAL" and "ANALOG" sections blink when the signal clips the input. They have to start to blink a little to have a proper audio level.

## INPUT



The "INPUT" section is an easy one. Usually you will connect the sound source to the "DRY IN" connector. This will go directly to the "DRY" VCA in the "MIXER" section. This way the DRY and WET signals can be mixed at the end of the processing chain.

The signal is also sent to the "MULT" output. From here the signal is seminormalled to the "ENV FOLLOWER" and the "FILTER" inputs.



*An different use of this section could be: Connect the sound source to the "DIGITAL" "IN" and then to the INPUT section. From "FILTER" "OUT" to "ANALOG" "IN". This way you will mix a digitized signal with it's distorted version in the "MIXER" section.*

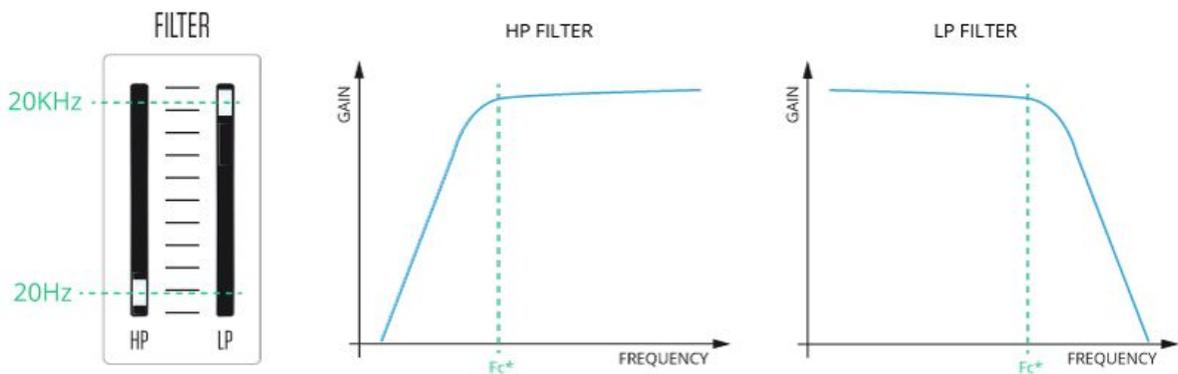
## FILTER

The "FILTER" section is formed by two 12dB filters in series. A high-pass filter and a low pass filter plus its related CV inputs and attenuators. This section is made to provide a prefiltering stage before the clipping sections. A proper filtering stage will determine the desired distorted sound.



**NEWBIE NOTE:** "a high-pass filter (HPF) is an electronic filter that passes signals with a frequency higher than the cutoff frequency and attenuates signals with frequencies lower than the cutoff frequency" ([Wikipedia wisdom](#)).

A low-pass filter passes signals lower than the cutoff frequency and attenuates signals with higher frequencies.



$F_c$  = Cutoff frequency. It is determined by the fader position + the voltage at the CV input.

**By default the “HP” fader will be at its lower position (about 20Hz) and the “LP” at its top position (about 20KHz). At these positions they will not filter any sound.**



*The prefiltering stage is very important to get decent results at the distortion stage. It is a basic tool to squeeze the machine and get the best of it. The most common use is to roll off the bass frequencies before using intense distortions, like the “FUZZ” circuit. A subtler effect is achieved by removing the high frequencies, reducing the total harmonic content at the end of the process. This will help to fit the track in a mix.*



*An different use of the filter is to patch it after the “DIGITAL” or the “ANALOG” sections. You can use it to remove some of the generated frequencies at these stages. Note the difference between using it before and after the distortion stages.*



*By connecting the “INV” output of the “ENV FOLLOWER” to the “HP” CV input you will get interesting results with drum loops. The HP will open and close according to the rhythm of the loop. Move the fader and the CV attenuator to get the best results from it.*

## DIGITAL

The “DIGITAL” section is a 12 bit **lo-fi** audio processing where the signal is digitized and processed in 5 different ways: fractional bit rate adjust, sample rate reducing, bit mutting, bit swapping and bit inverting.



**NEWBIE NOTE:** *“A digital representation expresses the audio waveform as a sequence of symbols, usually binary numbers” ([Wikipedia wisdom](#)). Zeros and ones. The audio signal is digitized with an analog-to-digital converter (ADC), processed inside the microprocessor and sent back to the analog circuit through the digital-to-analog converter (DAC).*

## INPUT

The input potentiometer controls the pre-amplifier stage of the “DIGITAL” section. It must be adjusted where the clipping led starts to blink. At this point there is no clipping yet. For eurorack signal it should be close to the zero point. For weak signals like an electric guitar, it should be turned up.



*If you’re rolling off some frequencies on the previous “FILTER” stage you will need to readjust the “INPUT” potentiometer to get the optimal gain.*

The main two functions of the "DIGITAL" section are the "BIT RATE" and the "SAMPLE RATE". They are accessible via the two potentiometers and its CV inputs.



### FRACTIONAL BIT RATE

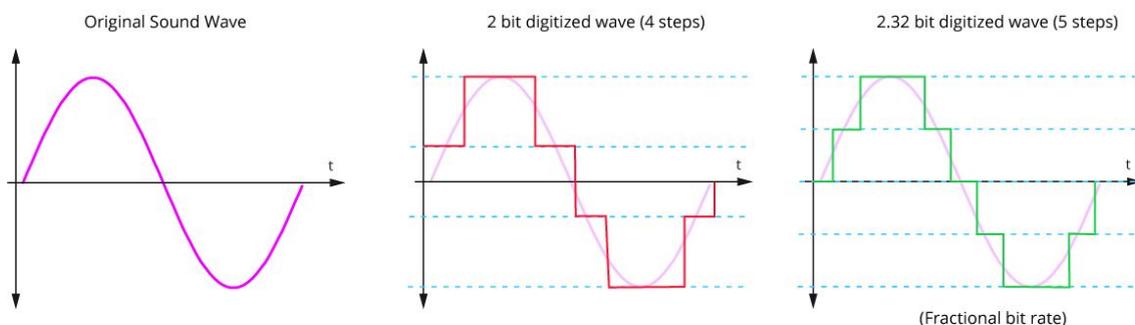
The bit rate is the number of bits of information on each sample. In the audio realm this is traduced on how many steps we have to define the amplitude of the signal. A bit rate of 1 means that there are only two states available, while a bit rate of 12 means that there are 4096 steps.

The "BIT RATE" potentiometer and its CV input, changes the number of bits of the digitized waveform. The 7 segment display will show how many bits are used: 1, 2, 3, 4, 5, 6, 7, 8, 9, 10 ("A"), 11 ("b") and 12 ("C"). The dot "." is reserved for special functions.

Usually, the systems that has the bit rate reducing feature only offer the number of steps determined by the current bit rate. This means that for 12 bits you should have 2, 4, 8, 16, 32, 64, 128, 256, 512, 1024, 2048 and 4096 steps available. SPICE uses a different system called "**fractional bit rate**". This means that when going from the bit 1 to the bit 2 there is a middle step (1.5) that uses 3 steps if sampling. From the bit 2 (4 steps) to the 3 (8 steps) there are several steps (5, 6, 7). And go on.



*For practical reasons it makes no sense to cover all the available steps. At highest bit rates no one can tell the difference between 1024 and 1050 steps. We kept the fractional bit rate from the bits 1 to 8.*





The 7 segment display will show only entire numbers. If the step 1.5 is selected the display will show a "1". Similarly, from the bit 2 to the bit 2.8 (7 steps) the display will show a "2".



When using the bit rate reduction function a lot of audio information will be lost. Processing complex sounds with it will result in a noisy signal. Unless you're looking for extreme noise we recommend to use this feature (and the other bit features) with purest or simplest signals that can be processed later.



An example of the above: connect a triangle wave of an oscillator directly to the digital section. Take the output and filter into a VCF and to a VCA after that. Go back to the SPICE and process the result with the "FILTER", "ANALOG" and "MIXER" sections.

Demo patch: <http://spice.metapatch.com/patch/9493> ("Triangle Wave Crushing")

Do you want to add some complexity? Use the "DIGITAL" potentiometer of the "FBACK" section to send the result to digitized waveform and create a complex feedback loop between modules.



A variation on the previous patch could be to use the whole SPICE as (an almost) a synth. Instead of using external VCFs, and VCAs you can use the ones on the SPICE. You just need a basic wave and shape it with the help of external envelopes.

Demo patch: <http://spice.metapatch.com/patch/9494> ("Almost a synth")

If you have patched the previous examples you can see how SPICE can be used in many different ways. Don't stick to the post-processing tool!

## SAMPLE RATE

Like the bit rate, the sample rate divides the audio signal in multiple samples. Each one of this samples is represented by a value of 12 bits (or less if you have reduced the bit rate).

At the maximum position the sample rate potentiometer will have a frequency resolution of about 15 KHz. This resolution together with the 12 bits makes the digital section a lo-fi processing system. (A CD audio has a resolution of 22 KHz and 16 bit).



If you play white noise through the digital section and turn down the "SAMPLE RATE" potentiometer, you will listen how the sample rate works very clearly.

## ENCODER

Before going to the remaining digital functions we need to understand how the encoder works. You can find this section on the "Quick Manual":

The encoder can be used to control both the Analog and the Digital sections:

### Analog

This is the **initial mode** (or analog mode) of the encoder. By turning it you can select one of the 7 distortion circuits of the Analog section. The internal memory of the SPICE will remember the last selected state when the machine is powered OFF.

### Digital

The digital circuit has 2 main functions: **bit processing** and **sample rate reduction**. The bit processing includes **bit rate reduction, bit muting, bit inverting** and **bit swapping**. The bit rate will always be selected with the potentiometer close to the 7 segment display and the "BIT RATE" CV input. The three other available options are controlled through the encoder:

- The encoder function is set in the analog mode by default . In this mode the 7 segment display will not blink.

- By clicking on the encoder twice (**double click**) you will enter the **Bit Mute mode**. The 7 segment will blink. By turning the encoder knob you will be able to select the different bit numbers (from "0" to "b"). With **a single click** you will mute or un-mute each bit. A **dot** will appear on the muted bit. You can mute as many bits as you want. A **long click** will reset the Bit Mute mode to its initial state where no bits are muted.

- By clicking on the encoder with a **double click** a second time you will enter the **Bit Invert mode**. It works as the Bit Mute mode. The blinking rate will be faster.

- By clicking on the encoder with a **double click** a third time you will enter the **Bit swap mode**. In this mode you will not see numbers but different symbols. In this mode **different bits will be swapped**. For example in the first step the bit number 4 will be swapped for the bit number 6. You can make as many swaps as you want by doing a single click. A long click will completely reset the mode. The full table of swaps can be found in the full manual.

- By clicking on the encoder with a **double click** a fourth time you will go back to the **initial mode** and the encoder will be used to select the distortion circuits. If any of the digital modes is activated (i.e. a bit mute at bit number 6) a **dot** will be displayed on the 7 segment display. By doing a **long click** it will **reset all the digital modes** to its initial state. Turning OFF the unit will not reset the changed digital modes.

					
	Rotate	Single Click	Double Click	Long Click	Display
INITIAL STATE	Circuit Select	Nothing	 Go to the next mode	Reset all bit modes	Bit Rate (1 to C)
BIT MUTE	Bit Select	Mute current bit		Reset current mode	Bit to mute
BIT INVERT		Invert current bit			Bit to invert
BIT SWAP	Swap Select	Apply current swap		Current Swap	

## BIT MUTING

The process of bit muting consists on muting one of the 12 bits. This means that the muted bit will be equal to 0 no matter how is the sampled wave. Due the lost of information the sonic result will be the addition of noise and sound artifacts.

Some notes about the bit mutting:

- *The display will show which bit is to be muted. The bit "0" is the less important while the bit "b" is the most important. This means that by muting the higher bit the audible effect will be more dramatic.*
- *By muting all the bits there will be no sound at all.*
- *Muting the bits one by one starting from "0" and going up has the same effect as reducing the bit rate*
- *Note that the bits are expressed here with its number, from "0" to "b". In the bit rate section the 7 segment displays shows the total number of bits used, this is from "1" to "C".*

## BIT INVERTING

The process of bit inverting consists on inverting one of the 12 bits. This means that if the inverted bit has a value of "0" it will be "1" and vice versa.

Some notes about the bit inverting:

- *Like in the bit muting function, the display will show which bit is to be muted.*
- *By inverting all the bits there will be an inverted copy of the wave.*
- *The sound effect will be close the bit muting effect but with some differences.*
- *Note that the bits are expressed here from "0" to "b" while in the bit rate section they are named by its natural number (from "1" to "C").*

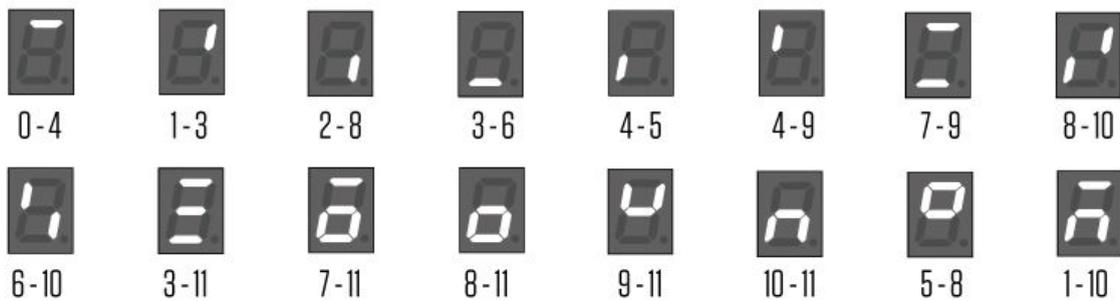
## BIT SWAPPING

The process of bit swapping consists on changing the value of one bit for the value of another bit. If we have the 12 bit sequence "011100011111" and we have a swap between the bits 0 and 11 the result will be "111100011110".



The bit swapping function has 16 available swaps. You can select as many as you can, none or all of them. The results can vary a lot from one to another. There are more than 65.000 possible combinations.

On the next table you can see all the available swaps:



## POST FILTER



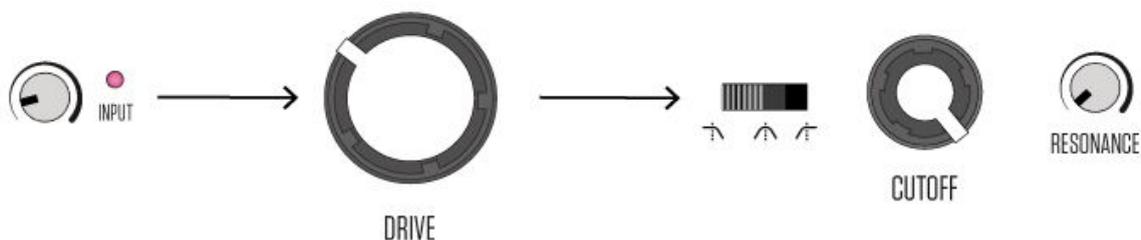
POST FILTER

Finally, after the digital process there is a simple low pass filter that can be used to remove the high frequency content of the digital processing.

The "DIGITAL" section can add a lot of noise, specially on the high frequencies. By removing them you can smooth the overall signal before sending it to the next stage.

## ANALOG

The "ANALOG" section is the **core** of SPICE. It is formed by a preamplifier, a distortion stage with 7 different analog circuits and state variable resonant filter.



## INPUT

The input potentiometer controls the pre-amplifier stage of the "ANALOG" section. It must be adjusted where the clipping led starts to blink. At this point the analog circuits will work the best. For eurorack signal it should be close to the zero point. For weak signals like an electric guitar, it should be turned up.

## **ANALOG CIRCUITS**

There are 7 analog circuits:

### **BOOST**

A clean circuit. It is used when you want to use the filter without any distortion. At high gains it will clip the signal.

Very useful to filter the digitized signal, make some compressor like effects, or use the feedback circuit with maximum gains.

### **SATURATION**

A soft clip saturator. It adds a bit of harmonics to warm up your signals.

This is a classical design that uses 2 LEDs in the amplifier feedback loop to clip the signal in a smooth and rounded shape. At low gains it will not clip.

### **HARD CLIP**

Another classic circuit that distorts the signal but more intensely. It uses a pair of diodes to clip the signal after the gain stage. The result is a clipped signal even at lower gains.

### **NU-TUBE**

A tube distortion from the *Korg Nu-tube* vacuum fluorescent display. Amazing tube like distortion. When the signal is distorted the light on the Nu-tube will turn off.

At lowest gains it will not distort the signal. Ideal for tube like controlled saturation.

### **NU-TUBE X2**

Both channels of the Nu-tube for a higher gain tube distortion. At lower gains it will distort

### **FUZZ**

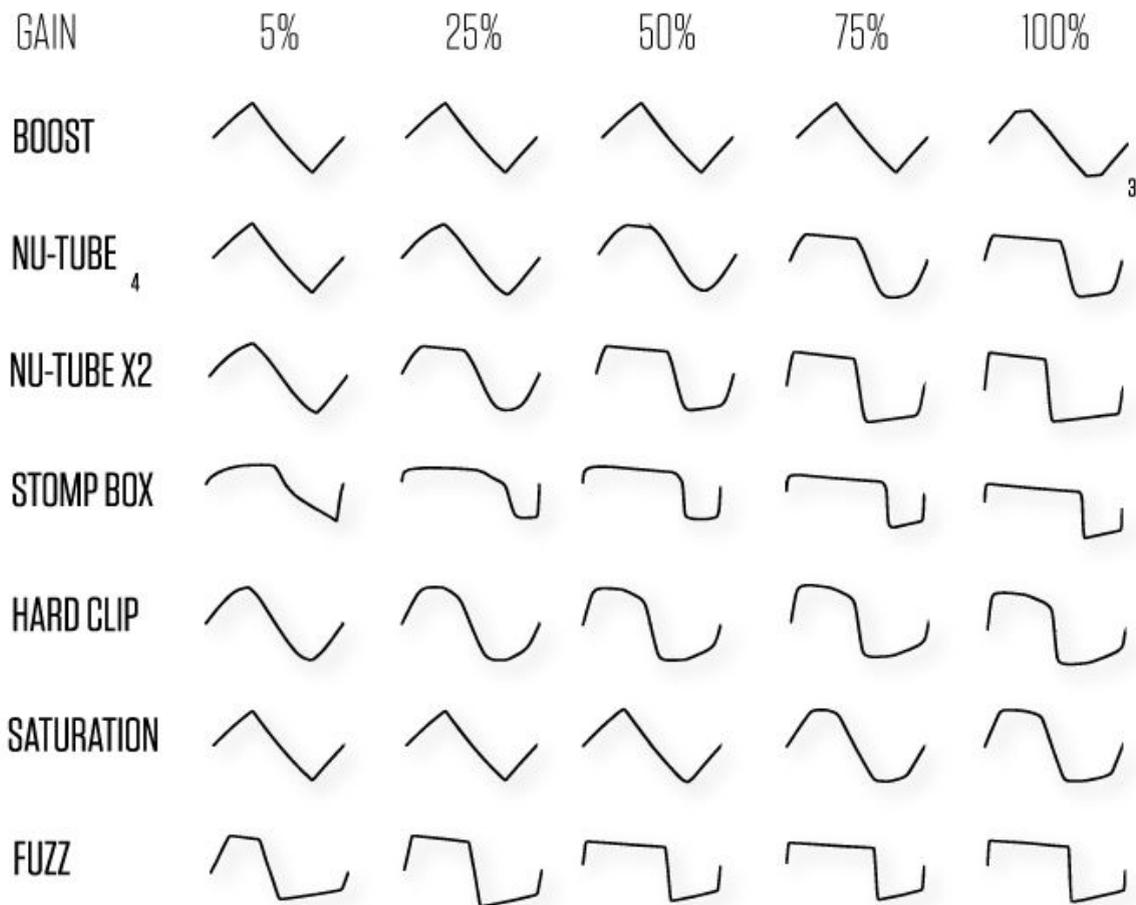
A discrete circuit based on transistors and a Germanium diode. A fuzz circuit for strong asymmetrical clipped effects.

### **STOMP BOX**

A filtered high gain distortion based on a very popular guitar pedal. It uses several filters, a TL061 classic opamp and clipping diodes for a hard clip distortion.

Below there is a chart with a triangle wave processed with the different analog circuits:

## SPICE ANALOG CIRCUITS CHART<sub>1,2</sub>



1 Based on a +/-4V triangle wave at the Analog input

2 Waves are normalised. Gain is not showed.

3 Clipping may occur at higher gains

4 Symetry can vary depending on the bias trimmer (accessible at the rear of the PCB)



The different analog circuits will not only shape the waves in a different way but also other aspects of the audio like amplitude, frequency, and transient response. Try different kinds of sounds with the different circuits to familiarize yourself with all the circuits.



We recommend to skip the Digital section to learn how the Analog section works and affect the sound. Use this simple patch to start: <http://spice.metapatch.com/patch/9488> ("No Digital")

## STATE VARIABLE FILTER

The last step of the "ANALOG" section is the VCF. It is a resonant state variable filter with low-pass, band-pass and high-pass outputs. The resonant circuit is based on the classical SEM synth VCF.



The VCF can be used for many purposes: roll off the high end harmonics generated with the distortion, keep only the high frequency content of a signal to enhance the DRY signal, shape a distorted wave with an envelope, create movement with the envelope follower, send a resonant audio to the "DIGITAL" section, etc.



*A possible use of the filter is to use the "BP" to add some mid-frequency harmonics to a flat synth signal. Instead of distorting the signal too much we add just a small amount of saturation and sum the result to the "DRY" signal. <http://spice.metapatch.com/patch/9495> ("Band Pass Spicing").*



*Another possible patch may be created connecting the output of the "ANALOG" section to the "CUTOFF" CV input. This will create a different form of feedback or X-modulation.: <http://spice.metapatch.com/patch/9496> ("Xmodulation")*

## MIXER

The "MIXER" consists of two VCA channels that control the level of the "DRY IN" and "WET IN" signals. The two signals are mixed together and are sent to the "MIXER OUT". The amplitude of each signal depends on the sum of the fader position and the voltage applied at the CV input of each channel.



**NEWBIE NOTE:** "A variable-gain or voltage-controlled amplifier is an electronic amplifier that varies its gain depending on a control voltage (often abbreviated CV). VCAs have many applications, including audio level compression, synthesizers and amplitude modulation." ([Wikipedia wisdom](#)).



*The "MIXER" section can be used in many ways. The most common way is to use it setting a static value for each channel. But using an external envelope or the outputs of the "ENV FOLLOWER" will give a new dimension to your sound processing techniques.*



*If the desired result is a highly distorted signal, the way to go is to turn up the "WET" fader and turn down the "DRY" one. But mixing both signals is a must if the goal is a subtler effect.*

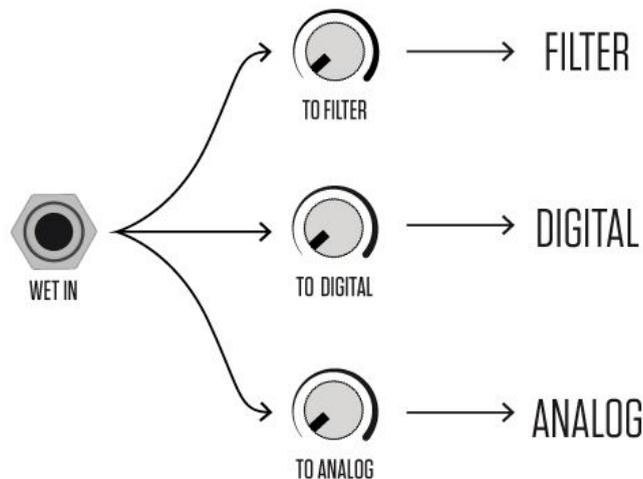
## FEEDBACK

The feedback section (labelled "FBCK") consists of an internal routing (non breakable and non patchable) where the signal at the "WET IN" is sent back to the input of the 3 other sections: "FILTER", "DIGITAL" and "ANALOG".

This situation creates **audio feedback**. The results are unpredictable and they will depend on the input level at the "WET IN", the input levels at each section, the kind of sound used and settings at each section. Mainly it will provide some doses of chaos, distortion and resonance.



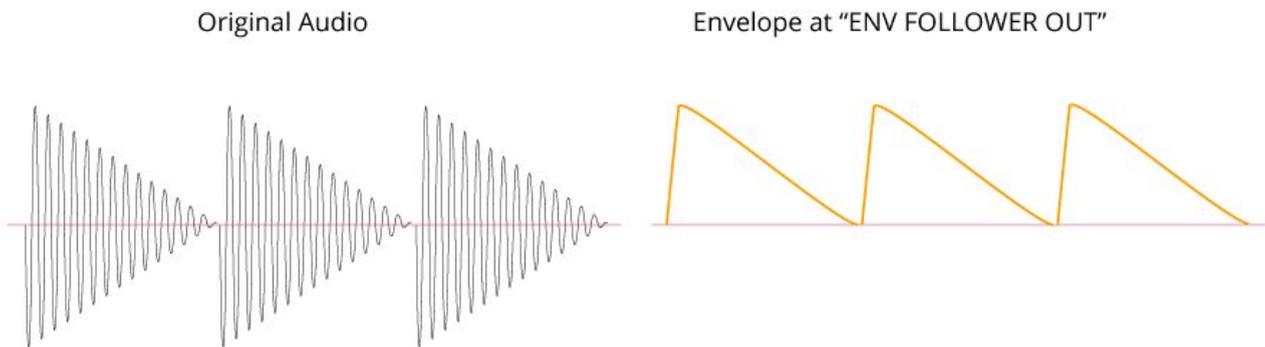
*It is **very important** to keep the "FBCK" potentiometers at the minimum position when not using the function. A tiny increase on its position may affect totally the behaviour of a whole section.*



*The signal sent to the "FBCK" will come from the "WET IN". This is not relative to the "WET" fader position, just by the signal previous to it. In a non-patched situation this means that the output of the "ANALOG" section will be sent directly to the feedback.*

## ENVELOPE FOLLOWER

The envelope follower “ENV FOLLOWER” is a circuit that converts the amplitude of a signal into an envelope. The resulting envelope is used to control the CV inputs.



*The potentiometer of the “ENV FOLLOWER” sets the amplitude of the signal before being converted to an envelope. For best results the light must blink ON and OFF.*



*The inverted output “INV” inverts the signal. This means that when at the “OUT” there is a rising voltage, at the “INV” there is a falling voltage. The level at the “INV” is always negative. This means that to use it, the potentiometers related to the CV inputs must not be at its minimum position to benefit from the “INV” envelope.*



*Despite of being a tiny section in the corner of the unit, the envelope follower is a **powerful** function. When applied to the different CV inputs it will give **movement** to the processed sound. It will make the difference between a static sound and a dynamic one.*



*The “ENV FOLLOWER” input is prepatched to the “DRY IN”. This means that it will respond to the same signal that is being processed. But other signals different from the input one can be used to generate envelopes. I.e.: a bass drum can generate a rhythmic envelope which will be used to shape a synth line or a pad.*



*One of our most loved patches is to use the envelope follower to turn down the “DRIVE”. This way a compressor like effect is generated. Use it with the Nu-tube distortion and a drum loop. You can mix it with the “DYM” signal to make parallel compression. <http://spice.metapatch.com/patch/9497> (“Pump the beat”)*

## FINAL NOTES

If you have followed all the manual and patched the proposed examples, you may noted that **SPICE** is a powerful unit that can be used in many different ways. From an immediate distortion unit, like a guitar pedal, to a more sophisticated way to generate your own sounds and textures from basic sound waves.

There is a long way to explore all the possibilities that this machine offers. We hope that you will share your patches with us. We will also publish new demos, videos and patches. Remember to follow us in the these channels for news, updates and demos:

[www.youtube.com/PlanktonElectronics](http://www.youtube.com/PlanktonElectronics)  
[www.facebook.com/PlanktonElectronics/](http://www.facebook.com/PlanktonElectronics/)  
[www.instagram.com/planktonelectronics/](http://www.instagram.com/planktonelectronics/)  
<http://spice.metapatch.com/>

**Thank you for trusting us and happy patching!**

*THE PLANKTON ELECTRONICS TEAM*