FRAP MAGNOLIA

8-VOICE ANALOG THRU-ZERO FM SYNTHESIZER

User's Manual version 1

Safety and Warranty

Warning! It is mandatory that you follow the following precautions when operating this electronic musical instrument. If you do not follow all these safety instructions, you will automatically void the instrument's warranty.

1. Read all the instructions (yes, all) before using Magnolia.
2. Do not open, disassemble or modify the instrument.
3. Never attempt to repair this instrument or replace any of its parts. If repair or part replacement should become necessary, you must contact us at support@frap.tools. There are no user-serviceable parts inside the musical instrument.
4. Do not place any containers which contain liquids on or near the instrument: this includes glass of wines, beers, sodas, schnapps, anything!
5. Do not use the instrument:
    a. near water or moisture (sink, showers, rain, lakes, the sea…);
    b. near open flames or heat sources (stove tops, ovens, bonfires…);
    c. in location of high temperature as under direct sunlight or close to heat sources.
    d. in a cold or hot environment (less than 10 °C or more than 40 °C);
    e. in a dusty environment;
    f. in a location subject to heavy vibration.
6. Do not drop the instrument.
7. Do not place any object, animal, or person on top of the instrument.
8. Ensure that the instrument is placed on a stable and flat support, like a professional keyboard stand or a desk, and that it does not topple over or fall, causing serious personal injury.
9. Ensure that the instrument has at least 10 cm of breathing room around it for ventilation and do not place it close to cloths/fabrics of any kind.
10. Do not use the instrument at high volumes for long periods of time.
11. The instrument should only be powered from an electrical outlet providing a voltage within the ratings of the instrument and an earth connection. Connection to any supply voltage outside the rated range, or a supply without an earth connection, can cause permanent damage and serious personal injury.
12. The instrument should come with a proper power cord based on your area: we know that you will use your own power cord anyway, but for the sake of safety please be sure it is earthed. Do not attempt to modify or disassemble the power cord.
13. Do not place heavy or sharp objects on the power cord, as this could damage the power cord and render it unsafe. This refers to any power cord you use in your life. If damage to the power cord is suspected, disconnect it from the electrical outlet if safe to do so, do not use the power cord and contact us.
14. If you need to replace the fuse in the power block, always replace it with a fuse of the same type, and do it only when the instrument is off and the power cord is detached from the instrument.
15. If any other serious malfunction is suspected, do not use the musical instrument, disconnect from the electrical supply, and contact us. Examples of malfunctions are:
    a. the musical instrument becoming wet (by rain, booze etc.);
    b. the musical instrument becoming very hot;

   c. generation of smoke or an unusual smell;
   d. repeated abnormal behaviour;
   e. visible damage to the enclosure, for example large dents or holes in the enclosure.
16. If the musical instrument is used by children, the children must always be supervised by an adult.
17. Ensure that all the connected cables are organized and managed in a safe manner, and do not cause an electrical or personal hazard.
18. When you need to transport the musical instrument, pack it in the box that it came in (including padding), otherwise damage during transport could occur.
19. Unplug the power supply from the outlet when left unused for long periods of time or during lightning storms.
20. Do not allow foreign objects or liquids to enter the musical instrument, as this can cause permanent damage and may result in serious personal injury and possible ignition of the liquid if flammable. If damage from foreign objects or liquids entering the musical instrument is suspected, do not use the musical instrument, disconnect from the electrical outlet and contact us.
21. Remember that this instrument is designed to last. This doesn't mean that every part used will be available in 100 years from the moment it has been produced but that servicing part of it can be achieved in a sustainable way. Electromechanical components as potentiometers, buttons, sliders, etc. are subject to deterioration, as well as some electrical components. There might be a time when you may need to replace these, in the same way as you change tyres on your car, or apply a patch to a blazer: this is called maintenance. Get in touch with us at support@frap.tools if you think you may need any maintenance.

This product is not user serviceable. All servicing should be carried out by qualified personnel only. Please note that any changes or modifications made to this product not expressly approved by Frap Tools srls could void the user's authority granted by the FCC to operate the equipment.

Electrical Specifications

- Rated input voltage: 100/240 VAC
- Rated input frequency: 5060 Hz
- Power consumption: 105 W Max
- Fuse type: Fast 1.25 A

This device has been tested and complies with Part 15 of the FCC Rules. Operation is subject to the following two conditions:

1. This device may not cause harmful interference.
2. This device must accept any interference received, including interference that may cause undesired operation.

Dear Musician,

It is with a lot of gratitude and a bit of trepidation that we at Frap Tools thank you for purchasing Magnolia. Congratulations on your new synthesizer!

Magnolia is our first attempt at designing a keyboard instrument. When we were working on our Brenso complex oscillator, we thought that it would have been interesting to encapsulate the West Coast approach to analog sound design in the format of a polyphonic synthesizer. Later, when our Cunsa filter was finally ready, we had everything we needed, and after a couple more years Magnolia was ready to see the light.

Our main goal was to make an instrument with a good sound and a good playability, an instrument that could fast-track the musician to some sonic territories that would be otherwise harder to reach in the analog domain, and to do so we meticulously designed every circuit from scratch.

We are of course aware that, in the bigger picture, Magnolia is still based on technologies and concepts that are far from being new or groundbreaking (as a matter of fact, they are decades old in the best case). However, we believe that such technologies and concepts are also *classic*, and the power of the classics is their ability to sound familiar and intriguing at the same time to any generation. Everybody is touched, if not captivated, by stories about love, hate, or quest, just like everybody is moved by the sound of an analog synthesizer. Moreover, a classic can be constantly reinterpreted and recontextualized – and that is what we wanted to do.

Magnolia is an important milestone for us, and if we have been able to finish this project, it is because of three kinds of people.

First, all the inventors, engineers, and artists that worked before us since the early days of electronic music. It is on the shoulder of these giants that we are standing, and it is to all those men and women that our greatest acknowledgement must go.

Then, we must thank all the Eurorack musicians that supported Frap Tools over the past ten years: we have been able to grow to this point thanks to their trust and friendship.

And finally, as we said in the beginning, we want to thank you for your purchase: we hope that playing Magnolia will be even just half as rewarding as it was for us making it.

But enough talk for now: keep reading the manual and start playing!

The Frap Tools team

What is Magnolia?

Magnolia is an eight-voice analog thru-zero linear FM synthesizer. Besides the classic analog polysynth sounds, its design allows programming some less conventional timbres closer to a particular family of synthesizer modules known as “complex oscillators.”

The main characteristic of those modules is the use of electronic circuits to create complex waveforms right at the beginning of the audio chain, instead of sculpting “simpler” waveforms like sawtooth or pulse waves. Those circuit are, for example, wave shapers, wave folders, and, most importantly, analog linear frequency modulation (FM), where one oscillator modulates another one at audio rate.

This technique was discovered by John Chowning in the 1970s and popularized by many digital synthesizers of the 1980s, like the Yamaha DX7. However, somewhere along the road FM was also featured in some modular analog oscillators, most notably the Buchla 259 Complex Waveform Generator. While the digital FM polysynths shaped the sound of the pop music of the 80s and 90s, analog complex oscillators created a much more experimental music in the late 70s: maybe less known, but certainly equally as iconic.

Analog linear FM was more rudimentary and less refined than its “original” digital sibling, mainly due to the technical limitations of analog circuits. After a few decades, thanks to more reliable and advanced components, it became possible to achieve much purer and stable sounds even in the analog domain, where our Brenso oscillator has been a prominent figure since 2020.

With the Magnolia, it is thus possible to bring advanced synthesis techniques in the world of analog polysynths and retain their known advantages: polyphony, immediacy of use, powerful sound, and those little unpredictable details generally known as “analog warmth.”

On top of that, a refined set of interaction points and a vast array of modulators aims at making the “keyboard approach” as expressive and immediate as possible.

Before Anything Else

Magnolia is a synthesizer that can make mellow and soothing tones, but it can also go wild very often and very quickly. In every section of its signal path there is at least one point that can distort the signal to create a specific flavor: FM, wave folders, pre-filter saturation, filter resonance, filter FM, global drive… We recommend reading the manual carefully to understand what kind of distortion each individual circuit can add to the sound, and thus be able to know, according to your taste, when to take advantage of them and when not to.

Key Features

- 8-voice polyphony.
- All-analog signal path.
- Analog thru-zero frequency modulation.
- 5-octave, velocity-sensitive keyboard with polyphonic aftertouch and adjustable curves.
- Bi-timbral programs with Morph, Dual, and Split capabilities.

- Two oscillators per voice: an “east coast” one with PWM and a “west coast” one with linear TZFM, wavefolder, and flip sync.
- Continuously variable wave form selector.
- 18 dB/oct high-pass and 24 dB/oct low-pass resonant filters with linear FM.
- Analog pre-filter saturation.
- Three loopable DAHDSR digital envelopes.
- Three digital LFOs with continuously variable wave forms.
- Per-part arpeggiator.
- Per-part 16-step sequencer.
- 200 memory slots for presets.
- Analog global distortion.
- Two global digital effect slots with choruses and delays.
- Hands-on modulation matrix with explicit visual cue.
- Velocity-off modulation source.
- “Polymove,” an expanded polyphonic randomization source.
- A Macro knob assignable to every parameter.
- 16 modulation sources.
- 38 modulation destinations.
- 64 modulation slots per part.
- Independent midi channels.

Acknowledgements

We would like to thank, in alphabetical order, Andrea Di Giorgio, Francesco Gennari, Gianni Proietti, Mattia Rubizzi, Mauro Cavalieri D’Oro, Sam Sakr, Stephan Schmitt, and Tim Davies for their precious feedback during the whole design process.

Magnolia is a design of Frap Tools, formed by Antonio Masiero, Federico Foglia, Giovanni Grandi, Ivan Tonizza, Simone Fabbri.

Manual and illustrations by Giovanni Grandi.

How to read this manual

Structure

The manual groups the information concerning Magnolia in eight chapters.

- Chapter 1 and 2 will get you started with Magnolia and guide you through the basic operations.
- Chapter 3 explains the architecture of Magnolia in terms of signal flow and preset structure.
- Chapter 4 explains all the parameters that create a sound.
- Chapter 5 explains the modulation matrix and its capabilities.
- Chapter 6 explains the arpeggiator.
- Chapters 7 and 8 dive into advanced settings.

Please note that this manual reflects the current firmware version and will inevitably change in the future. Check for updates at frap.tools/magnolia.

Typographic Conventions

- The parameters, functions, and architecture elements are mentioned with Capital Letters.
- The interface components (knobs, buttons, connections) are printed in **THE PANEL'S FONT**.

- A paragraph with light background contains important notices, tips, and performance suggestions.

- Instructions are printed in numbered lists (1, 2, 3…).
- Other numbered items are printed with letters (a, b, c…).

Graphic Conventions

In the various illustrations across the manual, the tiny hands indicate two things: either they point at the interface's elements, or they visually describe an interaction.

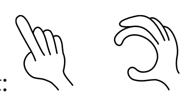
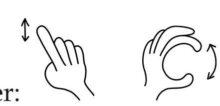
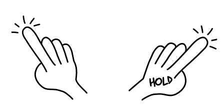
- These hands simply point at the interface element: 
- These mean to turn a knob or move a slider: 
- These mean "push" or "push and hold:" 

Table of Contents

Chapter 1: Panel and Connections

Magnolia allows you to program its sounds with a nearly one-knob-per-function interface, with all the functions explicitly labeled. The following diagram shows you how to read the front panel and the rear connections at first sight.

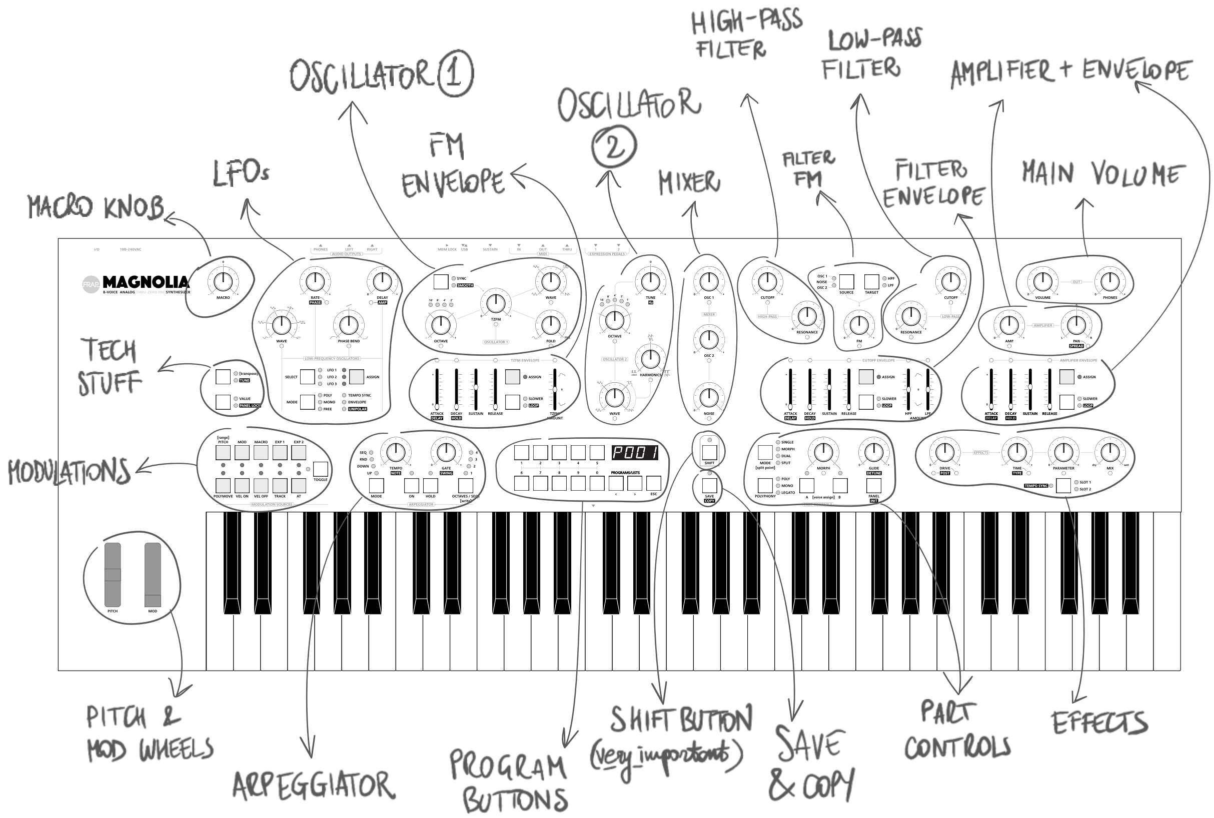


Figure 1: an overview of Magnolia's front panel.

Interacting with the Panel

Magnolia is a digitally controlled analog synthesizer, which allows to store and recall the panel settings as Programs. However, when loading a new Program, the stored panel values will inevitably be different than the current physical knobs' position.

Magnolia's knobs have thus a slightly integrated *jump mode*, which immediately updates the parameter's value to the one set by the knob as soon as it moves.

It is always possible to set the Program to match the current panel layout, on which see Panel Mode on page 4 below.)

The Shift Button

We said "nearly" one-knob-per-function interface because some knobs have a secondary function that you access through the black Shift button. When pushed, its LED will light up and the knobs on the panel will control the parameter with a white text over black background. (The knobs that do not have a secondary function will retain their behavior.)

The **SHIFT** button is located in the central area of the instrument so that you can easily access it with either the left or the right hand.

Its behavior is both momentary and toggled. If you push and hold it, its effect will last as long as you keep the finger pressed. If you push it and immediately release the finger, it will stay on as long as you rotate any knob: push the button a second time to disable the Shift function. After five seconds without any panel interaction, the toggled Shift function will be automatically disengaged anyway.

Chapter 2: Getting Started

Set Up

To set up the Magnolia you will need:

- A keyboard stand or a stable, flat surface.
- A functioning wall socket.
- A PA with jack cables (TS or TRS), or a pair of headphones).

Here is a suggested setup procedure:

1. Gently pull the Magnolia out of the packaging by placing it horizontally on its wider side on a flat surface (like a large table or the floor).
2. Place the Magnolia on a flat and stable surface like a table or a keyboard stand.
3. Connect the provided AC cable to the rear plug and to the wall socket.
4. Connect the keyboard to your PA using one or two TS or TRS cables patched to the main outputs (not included) or connect the headphones to the headphones output. If using a PA, ensure that its volume is at 0.
5. Ensure that Magnolia's master volume and headphones volume are fully counterclockwise.
6. Power the Magnolia through the power switch.
7. Raise the volume on your PA.

Tune Everything

After you have played a few notes, there is a chance that they may have sounded slightly out of tune. That is completely normal, because the Magnolia is an analog synthesizer and thus its tuning may be sensitive to many factors, like temperature changes, prolonged inactivity, long trips, and so on. Just like any other acoustic instrument!

But don't worry, you don't have to open it and tune everything by hand. Instead, there is a handy software routine that does everything automatically.

To enter the tuning routine:

1. Push the **SHIFT** button.
2. Push and hold the **TUNE** button for three seconds.

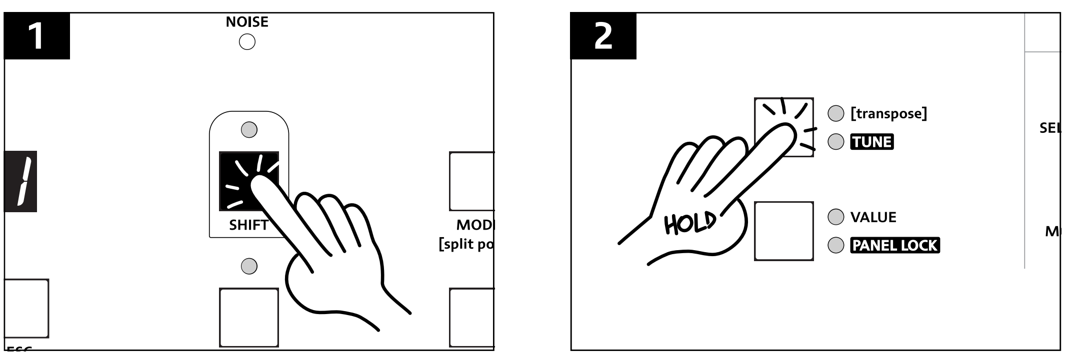


Figure 2: the Autotune procedure.

There is a much longer tuning routine that can be accessed via menu: see below, p. 75.

And now it is time to play!

Select a Program

The Magnolia can have up to 200 preprogrammed sounds. A “sound” is, in reality, a detailed set of instructions that immediately reprogram the synthesizer parameters. Such sets of instructions are called *Programs*.

Upon power-up, Magnolia loads the first program. To change the program within one bank, push any of the numbered buttons (**1** to **0**) or use the arrow keys. If you want to select, for example, program 30, you can:

- Dial “030”.
- Dial “30” and press the right arrow.
- Scroll the programs with the arrow keys.

Only the first memory slots are populated with factory Programs: the other ones are left with an initial patch, and they are meant to store your sounds.

Play a Program

When you change a program, the synthesizer gets instantly reprogrammed. When playing a Program, pay attention to every “interaction point” with the instrument and explore how they change from one Program to another.

- Keyboard: does it play the same sound across all keys, or does it play two different sounds?
- Keys: does anything happen when you play harder or softer? And if you play a key and then push it further down?
- Pitch Bend, Modulation Wheels, Macro knob: what kind of modulation do they do? What range do they have?

These ways of modulating our programs are non-destructive, meaning that they are not modifying the sound parameters and their position is not saved in the program. These interaction points are a sort of “macro controllers” that change more parameters at the same time and make the performance livelier. To change the very nature of the sound, we need to edit a program.

Edit a Program

It is possible to edit every parameter of a program’s sound through the panel knobs, sliders, and buttons, and their role will be discussed in detail in chapter 4, on page 25. However, for the purpose of this introductory tutorial, we will explore just a few of them.

Panel Mode

Since programs are sets of instructions that configure the parameters on the fly, loading a new program will cause the knob position not to match the actual stored parameter.

It is possible to set the Magnolia to immediately play whatever sound the current knobs are defining: to activate the Panel mode, push and hold the **PANEL** button for three seconds.

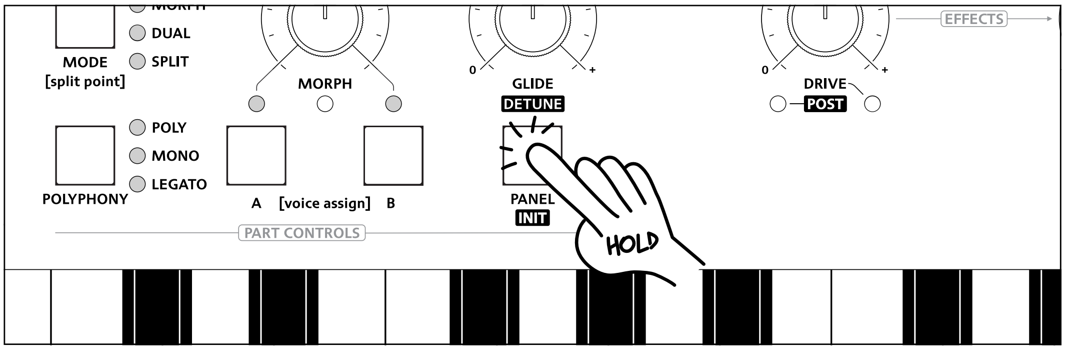


Figure 3: activating Panel Mode.

In Panel mode, the modulations are removed and the shifted parameters are set to 0.

You have now reconfigured whatever program you have currently loaded, but the changes are non-destructive unless you overwrite it. To save the current sound, refer to the section below on saving, comparing, and deleting programs, p. 9.

> Be very careful with Panel mode! You may end up with a configuration that does not output any sound, e.g., with filters or VCAs completely closed.

Load an Initialized Program

An *initialized program* is designed to be a sort of “white canvas” for you to create your sound from scratch. It sounds very dull, because it has been stripped of all the fun things that make a sound interesting. There are two ways to load an initialized program:

a. Load any program after the factory presets.
b. Load any preset and hold the **INIT** button for three seconds.

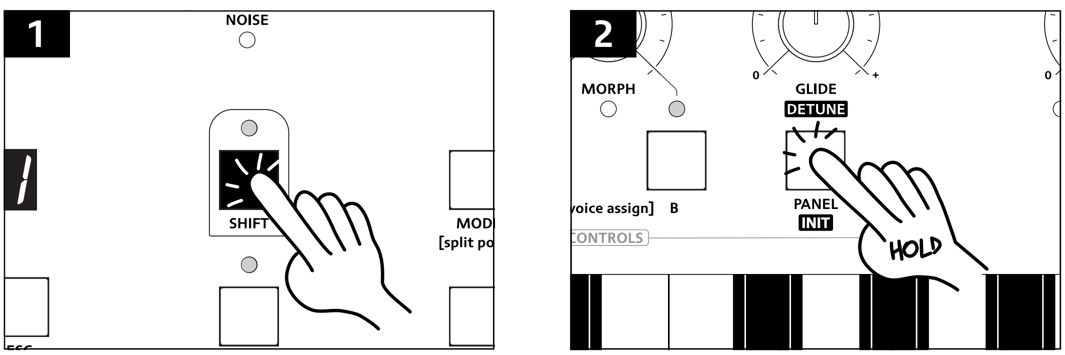


Figure 4: initializing a Program.

First Encounter with FM Synthesis

The initialized program consists of a simple sine wave with no further processing. The sound is coming from Oscillator 1, while Oscillator 2 is silent: the result is pretty dull.

Rotate oscillator ’s **TZFM** knob and hear how it radically changes the timbre. It is Oscillator 2 that is modulating oscillator’s 1 frequency at audio rate. This technique is called frequency modulation (FM), and the final timbre depends on two factors:

- The amount of modulation.
- The tuning of Oscillator 1 and 2.

We have just explored the FM amount, now let us see what happens when we change the tuning: rotate Oscillator 2's **TUNE** knob and hear how the sound becomes more and more dissonant. Take some time to experiment with those two parameters. If you want to listen to how Oscillator 2 sounds, turn up the **OSC 2** knob in the mixer section, but remember: the FM does not affect it.

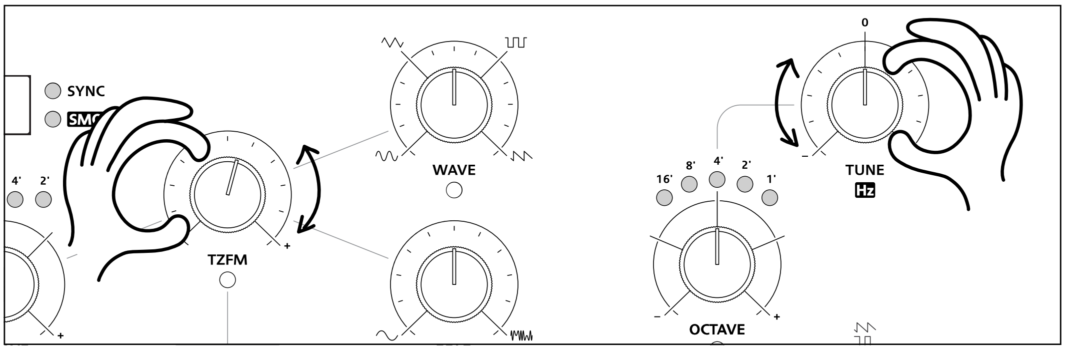


Figure 5: playing with the two main FM parameters.

Now the timbre of Oscillator 1 is much more interesting than before, but it still gets boring after some time. We need to add some motion to it, like any acoustic sound that changes over time.

Return the **FM** knob to a lower level, set the envelope's **ATTACK**, **DECAY**, **SUSTAIN**, and **RELEASE** faders to the following position, and then gradually raise the **TZFM AMOUNT** fader. Now the envelope control will change the FM amount over time whenever you play a key.

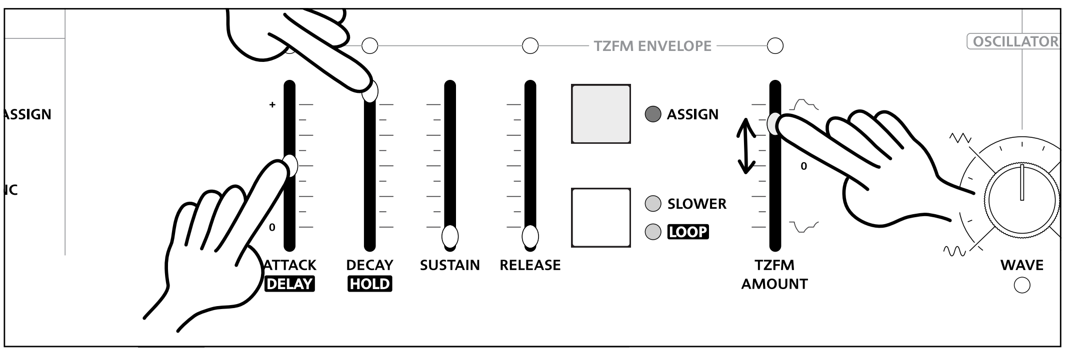


Figure 6: a basic FM envelope.

Now try this: set the **FM** knob to a reasonably high position and then move the **TZFM** modulation fader *below* the central position. You will hear that the envelope will have an opposite effect on the parameter.

Congratulations! This is your first FM patch.

Other oscillator parameters like wave shaping and wave folding allow you to create even more complex timbres. Learn more about them in the *Oscillators* section, from page 25 onwards.

First encounter with Filters

The Magnolia has two filters that process the blend of Oscillator 1, Oscillator 2, and Noise defined by the Mixer. The first in line is the high-pass filter, which removes the lower frequencies. The second one is the low-pass filter, which removes the high ones.

Let us use them to add a different kind of motion to our sound: remove the envelope over the FM so to obtain a static FM sound and then play with the two **CUTOFF** knobs. You will hear that the process goes in the same direction as the FM knob, but the outcome is very different.

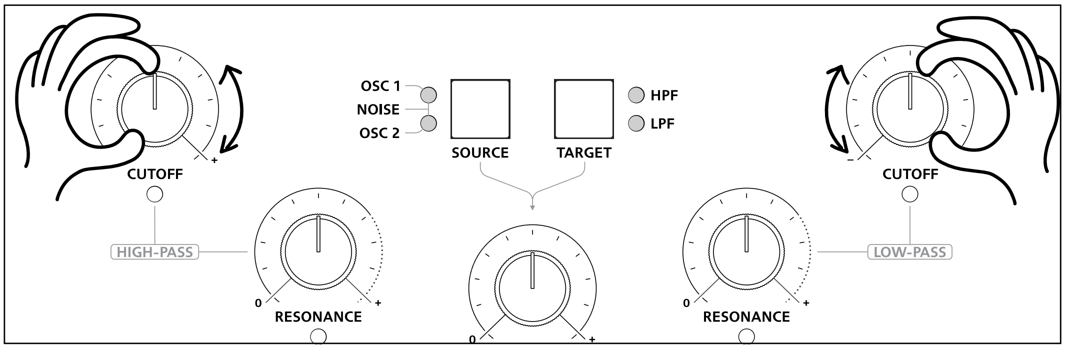


Figure 7: playing with the Cutoff knobs.

Now set the filter envelope to the following settings and then play with the **HPF** and **LPF** modulation faders. The keyboard will automatically open and close the filters as you play the keys.

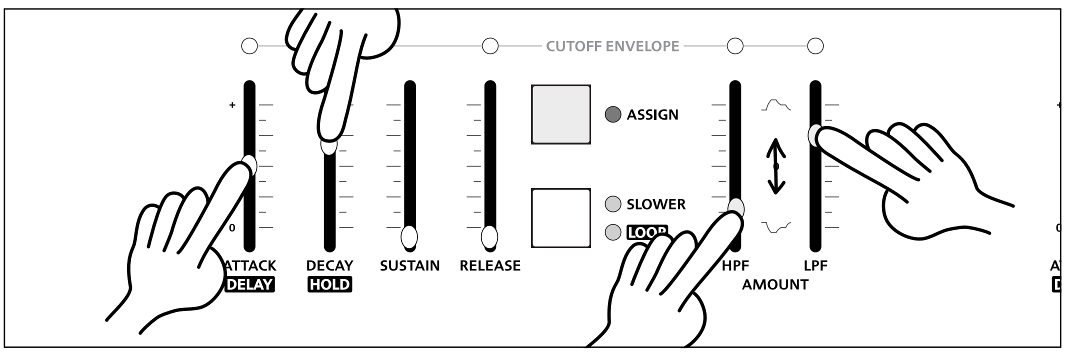


Figure 8: a basic filter envelope.

At this point, try rotating the **RESONANCE** knob. You will experience a sort of whistling sound on top of the oscillator. Experiment with different envelope settings to obtain different “whistle trajectories.”

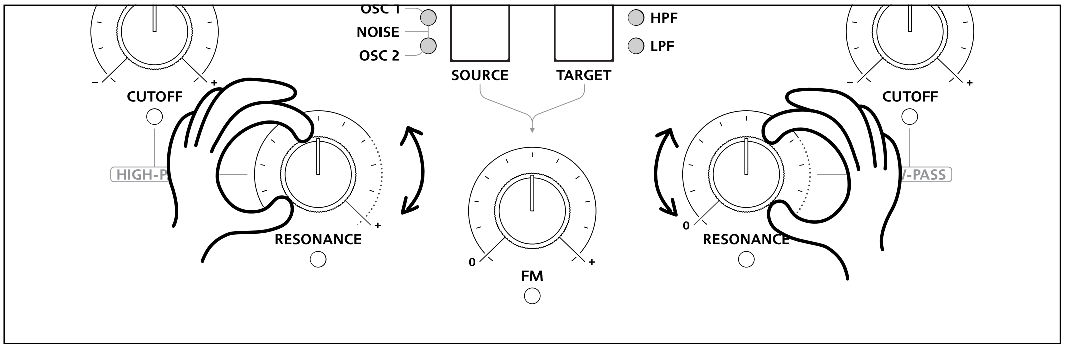


Figure 9: playing with the Resonance controls.

Another filter parameter is the FM: learn more about it in the *Filter FM* section on page 34.

First encounter with the Amplifier

You may have noticed that when you press a key the Magnolia plays, and when you don't it stays silent: it may sound obvious, but it is not! The oscillator on an analog synthesizer keeps spinning and generating waveforms all the time the instrument is on, and it is up to another circuit to cut the sound when we do not need it. Such a circuit is the Amplifier.

On the Magnolia, it is the last section on the right. Try and rotate the **AMP** knob: you will hear the sound increasing and decreasing its volume.

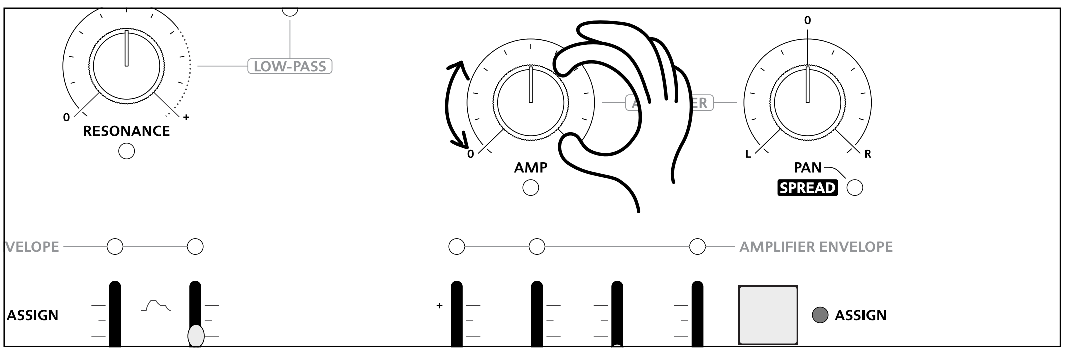


Figure 10: playing with the Amplifier control.

By now, you may have already guessed that the bottom section is the amplifier's envelope, but this time there are no assigned faders: the reason is that this envelope must always stay connected to the Amplifier, otherwise the keyboard won't stop playing! Anyway, experiment with its parameter and hear how they affect the sound. A long attack will have a "fade in" effect, while a short attack, fast decay and low sustain will create a plucked sound.

Another parameter of the Amplifier section is the Pan/Spread, on which see 35 onwards.

First encounter with the Modulations

We have seen how interesting the sound becomes once we add envelopes. An envelope is a *control signal*, like an invisible hand that moves certain knobs for us (the TZFM, the Cutoff frequency, and the VCA of the previous examples).

The envelopes were pre-assigned to those parameters, but it is possible to use these and other "invisible hands" to move nearly every other knob on the panel. Here is an example.

First, turn the **LPF** and **HPF** envelope faders to the central position to remove any modulation over the filter. Feel free to set the low-pass filter **CUTOFF** knob to your favorite setting. Then:

1. Push the AT button (its LED will flash).
2. Rotate the low-pass filter **CUTOFF** knob until it crosses the "noon" position.
3. Rotate it clockwise: you will see a number updating on the screen. Set it to a middle value.
4. Push the AT button again.

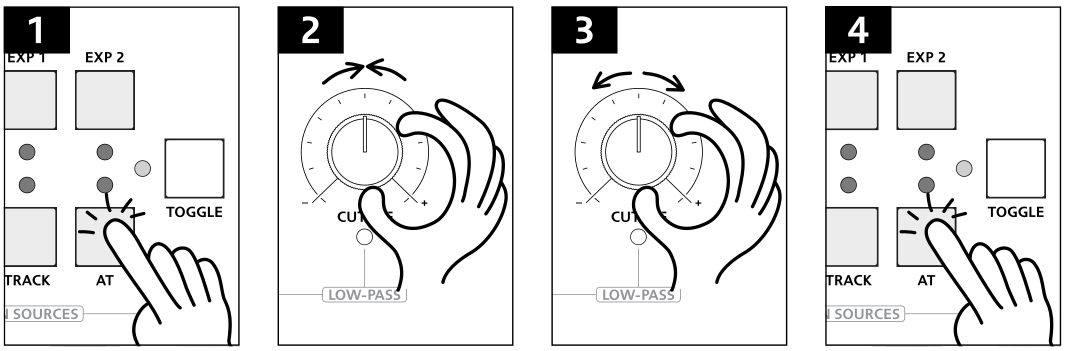


Figure 11: assigning the Aftertouch to the low-pass filter's Cutoff.

At this point, try playing some keys and pushing them a bit below their end point so to activate the polyphonic aftertouch function. You will hear the filter opening as you push the keys.

You have just assigned your first modulation! Feel free to repeat the operation with any other modulation source.

Save, Compare, and Delete Programs.

All the editing that we did is volatile, so if we power off the keyboard it will be gone forever! If we want to be able to recall it, we need to save it on a memory slot.

Save a Program

The Magnolia comes with a memory protection switch on the back to avoid messing up with your setups in a live context, so first we need to take care of it. To be able to save the presets, reach for the memory protection switch on the back and set it to the left (from the player's perspective).

Then, here is the saving procedure:

1. Push the save button. (The display number will flash.)
2. Select the desired group, bank, and program to overwrite. (If you ignore this step, you will overwrite the current program.)
3. Push the save button again to confirm.

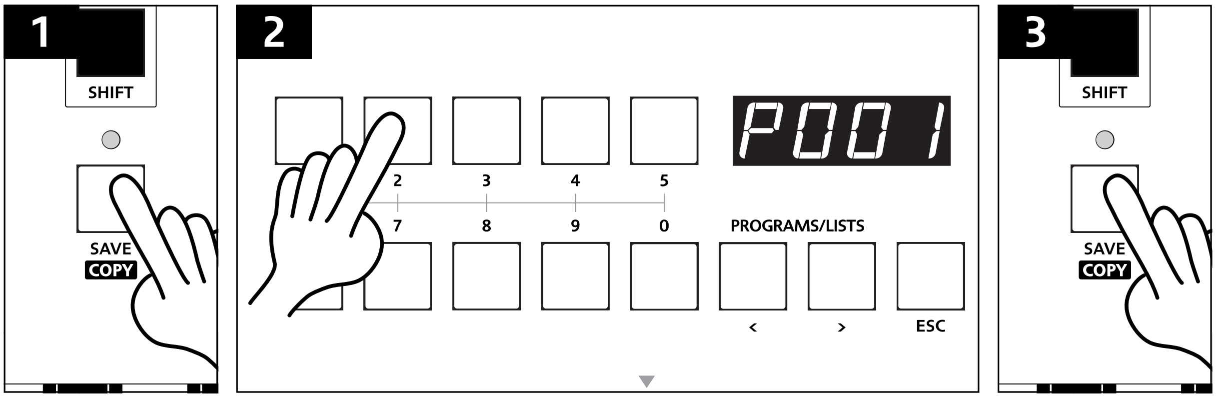


Figure 12: saving a Program.

At this point, the project you selected has become your new sound. Congratulations! The previous sound occupying that memory slot is now lost forever.

Compare Edits

If you play the keyboard during points 2 and 3 of the saving procedure, you will be able to hear the sounds that you are about to overwrite. This feature is useful to ensure that you are not erasing anything important.

If you do not select another program, you will hear the current project *before* your edits while. This allows you to check if your refinements are an actual improvement or not from the saved program – an occurrence that happens very often when sound designing.

Delete a Program

Since there are no "empty" slots on the Magnolia, you do not really delete programs. However, if you really cannot stand a program and want to get rid of it, you may save it as an initialized program, by holding down the **INIT** button for three seconds and then following the saving procedure above.

Chapter 3: Voices, Parts, and Programs

Any synthesizer can play infinite sounds. To change a sound there are two main ways: programming it through the front panel, or, if the synthesizer has a digital memory, recalling a set of instructions that instantly reconfigures the instrument settings without having to tweak every knob by hand.

The Magnolia can work in both ways and in this chapter we will see how. We will start from the smallest element, the *Voice*, then we will see how voices are edited and managed in *Parts*, and finally we will recap how everything is arranged and stored in *Programs*.

Voice Architecture

The Magnolia can play eight notes simultaneously, so it has an 8-voice polyphony. Whenever we turn a knob to change an aspect of our sound, we are thus controlling eight independent monophonic synthesizers, each containing:

- Two oscillators.
- A mixer.
- A high-pass filter.
- A low-pass filter.
- An amplifier.

If these things sound a bit unfamiliar, have a look at the Appendix A on page 77.

The Magnolia also features a noise source that can be mixed with the oscillators and serve other purposes. However, the noise is digitally generated and it is thus not part of the analog voice architecture.

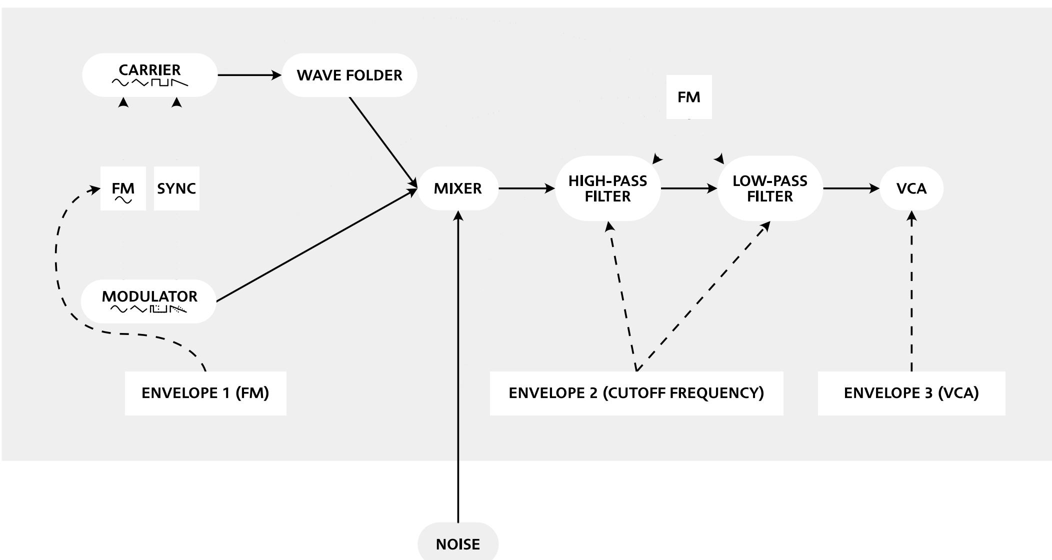


Figure 13: diagram of one synth voice.

Eight voices mean that we can play up to eight notes at once (unless the program is in Dual mode, on which see below p. 13). If we play a nine-note chord, the ninth note will play by “replacing” the first one played. This phenomenon is called “voice stealing” and it is an unavoidable limitation of analog polyphonic synthesizers.

Magnolia’s voice allocation is indeed a bit more refined, as it considers also the pressed keys and the sustain pedal. As it needs to steal a voice, it first looks for unused voices. If all the voices are in use, it looks for the voices that are in their amplifier envelope’s Release stage. If all the voices are still sustained, for example through a sustain pedal, it seeks the voices whose key is no longer pressed. Only if all the keys are pressed it will start stealing the oldest ones. This behavior guarantees a very natural playing style.

Parts and Modes

To change the synthesizer settings and obtain different sounds, we use the controls on the front panel that we will discuss in detail in the next chapter. For now, we will say that if we want to change our sound, we must program all the voices in the same way, so the panel settings must affect all of them simultaneously on the same parameters and by the same amount. This is the most straightforward behavior of an analog polyphonic synthesizer.

However, sometimes we may want to program certain voices differently, like if we had two independent synthesizers with a four-note polyphony each. We’d then be able to play one sound with the left and another with the right hand, or to layer two different sounds on top of each other. This feature is called *bitimbrality* because our instrument would be able to play two sounds (*timbres*) simultaneously. On the Magnolia, this is possible thanks to the *Part* controls.

A program contains two independent panel settings called *Part A* and *Part B*, so it may be said that a program always contains two sounds (or “timbres”).

To edit the sound of a part, push the **PART A** or **B** button: wen a part is selected, the controls on the panel will edit it. It is possible to select both parts at the same time: the second selected part will have a flashing LED, and every parameter change will affect them equally. Still, you may hear only the first selected part according to the Part Mode which we will discuss now.

A Program has four ways of playing the two parts, and you can cycle through them with the **MODE** button. The four modes are Single, Morph, Dual, and Split.

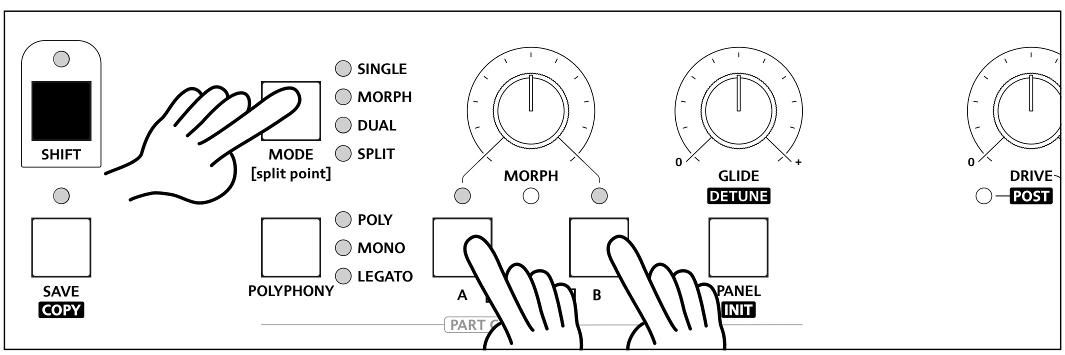


Figure 14: The Part and Mode Buttons.

Single Mode

When in Single mode, the eight voices of polyphony are all programmed to the same settings. Part A and B will effectively work as two “sub-programs” between which you may switch through the **PART** buttons.

> If you select both parts for editing certain parameters simultaneously, remember that only the one selected first will play.

Morph Mode

This mode is identical to the Single mode, but enables the **MORPH** knob that allows you to gradually transform the sound from the settings of part A to the settings of part B.

The Morph parameter is a feature first developed on virtual analog synthesizers, most notably the Clavia Nord Lead and the Yamaha AN1X, but it has been recently implemented also on analog polysynths as well, like the Arturia Polybrute.

The morphing technique consists of changing sound from one part to the other by gradually adjusting all the parameters at once, instead of changing everything in the blink of an eye.

It is not the same as an audio crossfade, since the crossfade only reduces the volume of a sound while simultaneously increasing the other. Morph, on the other hand, is a continuous reconfiguration of all the parameters at the same time, and so it may create infinite new sounds in between.

> The Morph function is active only if the Program is in Morph mode. When selecting this mode, the program will automatically load Part A, and the two Parts will share the same Arpeggiator and LFO settings, see p. 69.

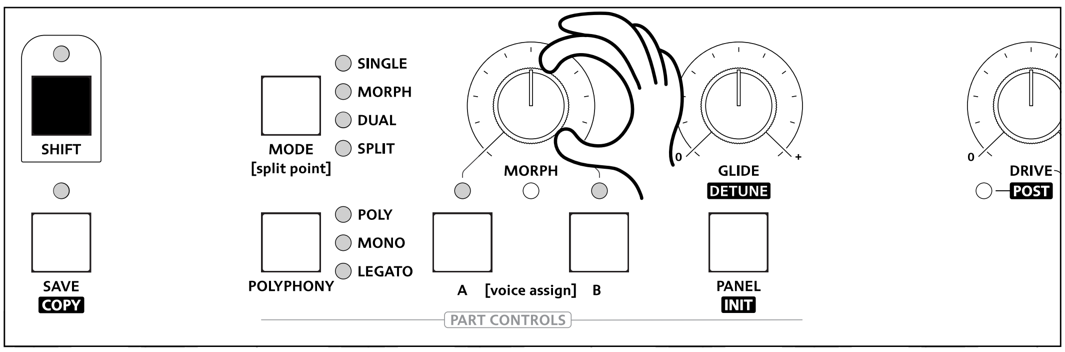


Figure 15: the Morph knob.

Dual Mode

This mode allows you to layer two different sounds on top of each other. A classic example a short bell-like sound and a slow pad.

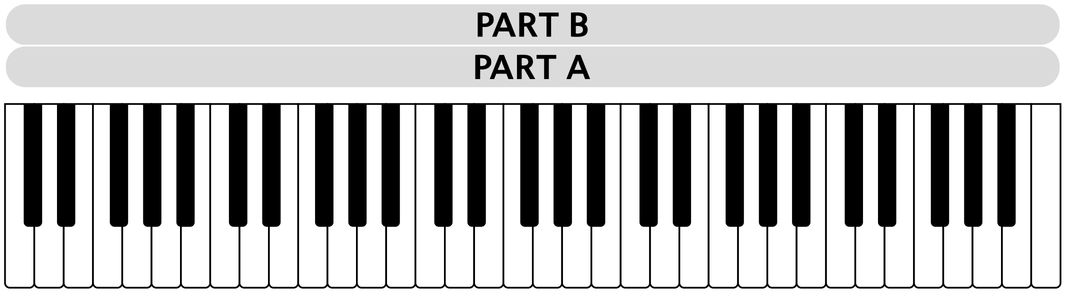


Figure 16: diagram of Part A and B in Dual mode.

Split Mode

This mode allows you to assign two different sounds to two sections of the keyboard. The lower section will play Part A, and the higher section will play Part B.

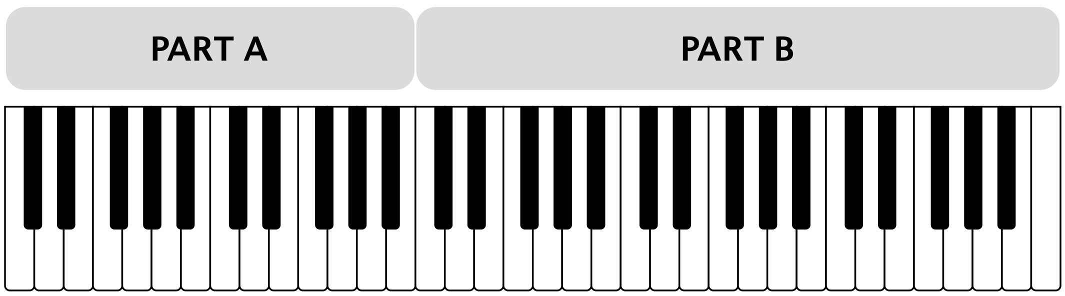


Figure 17: diagram of Part A and B in Split mode.

The default split point of most Programs is C 4, but it is possible to change it through the following procedure:

1. Push and hold the **MODE** button.
2. Press the key corresponding to the first note of the upper part (Part B).

The Split Point setting will be saved in the current Program.

On Part Modes and Polyphony

Since Magnolia's voices are always eight, if we want to play the two parts simultaneously in Dual or Split mode, we must distribute them between part A and part B. The default settings for Dual and Split mode assigns four voices to part A and four voices to part B, which means that, while it is possible to play two sounds at the same time, the number of simultaneously available notes is reduced to four.

However, there are cases in which you may need to distribute the voices differently: Magnolia allows you to freely assign any number of voices between 1 and 8 to each part. You can find the procedure to assign the voices in the section below, on page 16.

Polyphony (Poly/Mono/Legato)

Early synthesizers were monophonic, and a wide gamut of classic synth sounds were developed on these machines. Such sounds are, for example, leads or basses often modeled on instruments that by nature would not (or could not) play chords, and they still require a single-voice architecture to work properly. Just like with flutes or bassoon, a monophonic synth sound allows to play the keyboard with some peculiar techniques otherwise harder to achieve with polysynths, like legatos, trills, and other embellishments.

Despite having an eight-voice polyphony, Magnolia can become a monophonic synthesizer that plays only one note at a time through the **POLYPHONY** button.

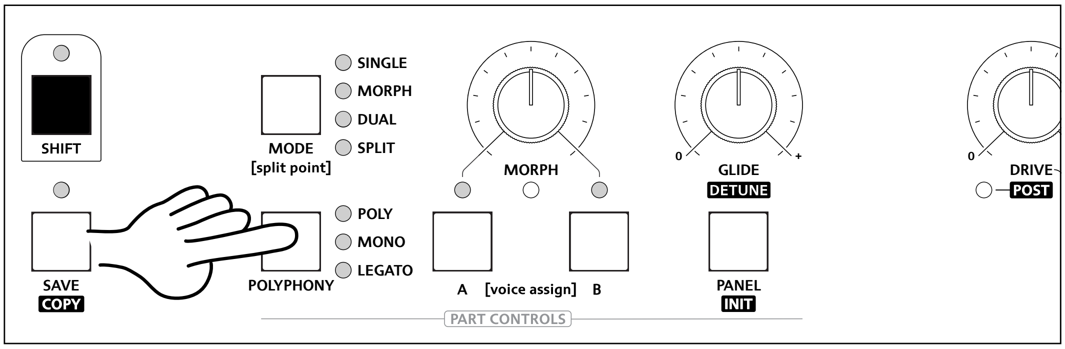


Figure 18: the Polyphony button.

The **POLYPHONY** button has three settings:

a. **POLY**: allows you to play up to eight notes at the same time.
b. **MONO**: the keyboard plays one note only, regardless of how many keys you press.
c. **LEGATO**: same as Mono, but with a different articulation: the envelopes will not fire until all the keys are lifted, much like the hammer-on and pull-off techniques on the guitar.

The differences between Mono and Legato are three: how the envelope retrigs when more than one keys are pressed, how the Glide behaves after all the keys are released, and how the Polymove modulation generates new values. You can read more about it in the respective sections on pp. p. 17, 50, and 62.

The polyphony control affects the program in different ways according to the part settings.

- If the program is set to Single or Morph mode, both parts will share the same polyphony setting.
- If the program is set to Dual or Split, the two parts can have different polyphony settings.

Key Priority in Mono and Legato Parts

On a monophonic synthesizer you play with only one voice, but it may happen that you push more than one key at the same time and the machine must decide which note to output. This decision procedure is called *key priority*, and Magnolia outputs the last played note in order of time.

For example, if you play the C, F, G, and B keys in this order and hold them all down, only the B note will play at the end. If you lift the B finger up, Magnolia will play the G note. If you lift the C finger, Magnolia will still play the G note.

Voice Assign

As we saw earlier (p. 14), there are some circumstances in which we need to manually assign voices to a part. More specifically:

- When a Part is in Mono or Legato, to achieve the Unison or Chord modes (see the next sections)
- When the Program is in Split mode
- When the Arpeggiator is active (see the Arpeggiator section, p. 73).

This is the procedure to distribute the eight available voices across the two parts when your Program and Part settings fit in one of the three cases above:

1. Push the **PART A** or **B** button and hold it.
2. Press the C4 note on the keyboard as many times as the number of voices you want to assign to that part. For example, one tap=one voice; five taps=five voices.

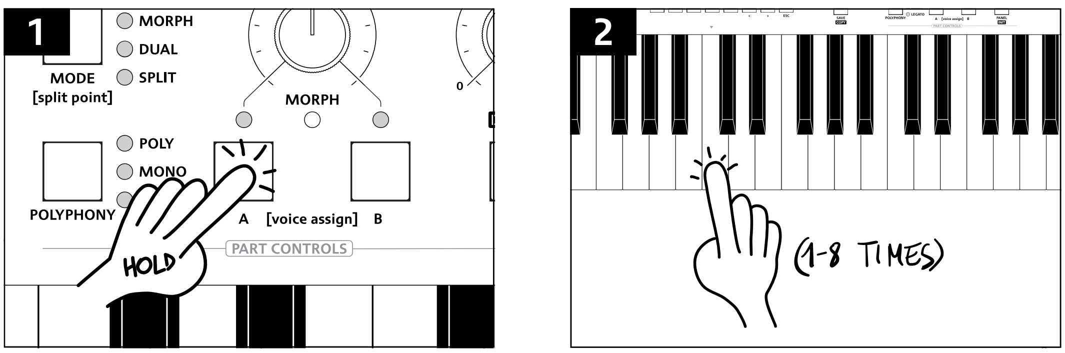


Figure 19: assigning Voices to a Part.

In cases (b) and (c), where both Parts are active simultaneously, assigning voices to a Part will also automatically assign the remaining ones to the other. For example, if we assign six voices to Part A, Part B will have two voices.

> In Split or Dual Mode, the maximum number of voices that can be assigned to a Part is seven, since it is not possible to have a part with zero voices.

Unison Mode

We have said that in Mono or Legato we can play only one note at a time. However, we may not be limited to one single voice! If we want, we can make a monophonic part with up to eight voices, all tracking in unison. The result is a very powerful sound, full of beatings and with an adjustable stereo image, on which see the Pan/Spread section on p. 36.

> In Single and Morph Mode, it is possible to achieve a Unison of eight voices, since only one part is playing at a time. In Dual or Split mode, the Unison can be of up to seven voices, since the other Part must be able to play with at least one note.

Chord Mode

A curious variation of the Unison mode can be achieved by assigning more voices to a monophonic patch *at different intervals*, instead of in unison. This effect will create actual chords on every key, all transposing in parallel motion, much like the famous “chord stabs” effect.

To tune a monophonic patch in chords:

1. Ensure that your Polyphony setting is Mono or Legato.
2. Push the **PART A** or **B** button and hold it.
3. Press the any note on the keyboard as many times as the number of voices you want to assign to that part. The intervals will be calculated on the root note of C4.

For example, pressing C4, E4, G4 will assign to a monophonic patch three voices tuned in a major chord, so *all the keys* will play a major chord based on their note.

In Morph mode, both parts will share the same Unison and Chord settings, if present.

Glide

When you press any key, Magnolia's oscillators get immediately reprogrammed to play the new note, like a guitar string when we change fret. However, sometimes it could be desirable to have a smoother transition from a note to another, like the portamento effect on a cello.

This effect is possible through the Glide parameter, whose knob defines the "sliding time" between the last played note and the new one.

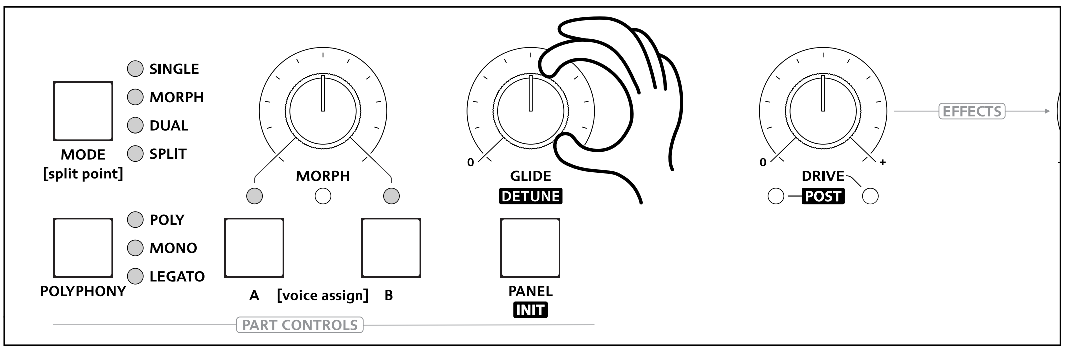


Figure 20: the Glide knob.

When the **GLIDE** knob is fully counterclockwise, the effect is disabled. Rotate it clockwise to gently increase the integration time between the notes. Please note that the higher the note leap is, the longer will be the portamento effect, like if we were sliding with a constant speed throughout the length of a string.

The most canonic use of the Glide parameter is with monophonic patches, but it can have a nice effect also with slow polyphonic pads. However, in those cases it will depend on the voice allocation, so whenever a voice cart is selected to play a note, it will glide from the previously played note. This will cause an effect that may not relate to the keys played on the keyboard.

The Glide parameter is per part, so it is possible to assign it to one part only, leaving the other unaffected. In Dual programs, this will cause a detuning effect at the beginning of every note, which will get progressively more consonant. To assign the same Glide value to both parts, just select them by pushing the two **PART** buttons at once.

Glide in Mono and Legato Parts

The Glide control behaves a bit differently when a Part is set to Mono or Legato.

- When the part is in Mono, the Glide effect always integrates between the last played note and the new one, even if you lift all the fingers for a long time. So if you played a C five minutes ago, and now you play a G, it will slide up from C.
- When the part is in Legato, the Glide effect will operate only between notes played in legato mode, i.e., with at least one key pressed before the new note. When all the keys are released, the new note will not glide.

Transpose

The keyboard is mapped to output MIDI signals from C2 to C7 for each part. The Transpose button allows you to shift the keyboard from a semitone to two octaves up or down.

The Transpose function by default works on the selected Part only, allowing to extend its range (for example, if in a split program we need our Part A to be higher, or our Part B to be lower). However, it is also possible to transpose the whole instrument up or down by selecting both Parts.

1. Select the Part(s) you want to transpose.
2. Push and hold the **TRANSPOSE** button.
3. Press the key corresponding to the note that will replace C4.

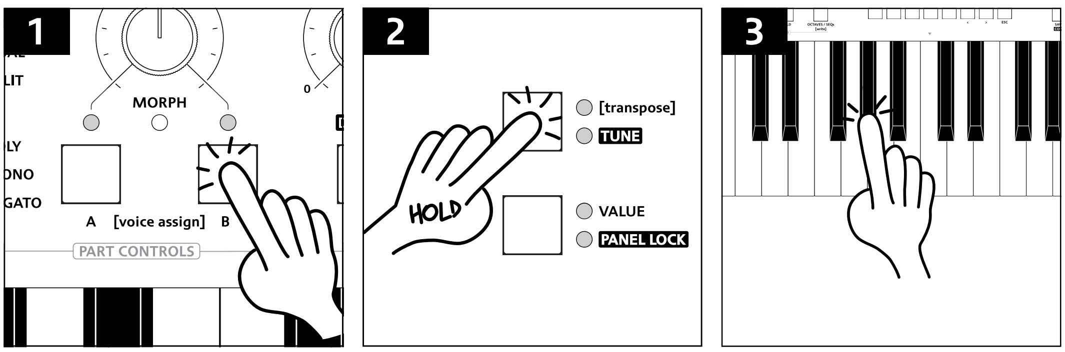


Figure 21: transposing a Part.

For example, pressing C3 will transpose the notes down by one octave. Or, pressing C♯4 will transpose the keyboard up by a semitone.

When the keyboard is transposed, the **TRANSPOSE** LED will be on, and the MIDI notes output by a given part will be shifted accordingly.

In the Sound Design chapter, we will see that the oscillators can be transposed as well, and we will learn the difference that one must know between those two transposition methods (see below on page 25).

Hold Button and Sustain Pedal

The Hold button and the Sustain Pedal (not included) perform a similar task, retaining the Note On signal even after you released the keys. They work on the Sustain segment of the three envelopes, exactly like the keyboard gate of a pressed key.

> If the envelope's Sustain level is set to 0, the pedal and button will not have any effect.

However, their behavior is very different. When the Sustain pedal is pressed, every note played on the keyboard will have an infinite sustain, until the pedal is released. If you add another note, it will pile up to the existing “chord.”

The Hold function, on the other hand, works by holding only the notes played in legato mode, so without lifting all the keys. As long as at least one key is held down, Magnolia will continue adding notes. After releasing all the keys, the previously played notes will continue to sound, but they will be wiped out as a new note is played.

To engage the Hold function, push the **HOLD** button.

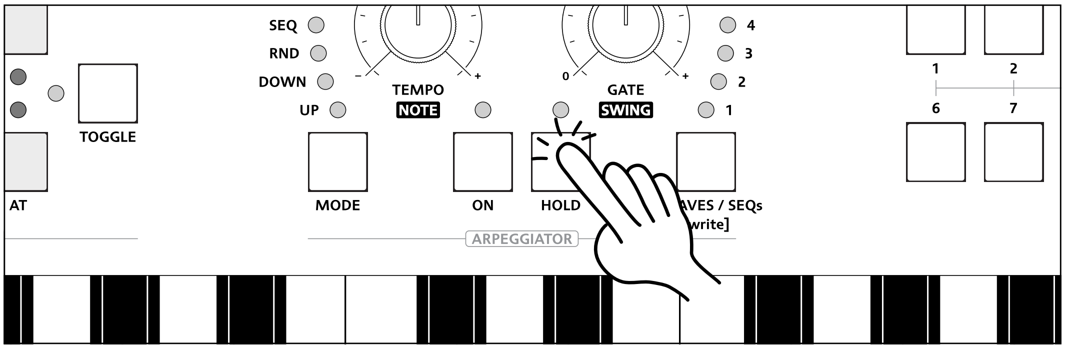


Figure 22: the Hold button.

The Hold button works on Parts A and B individually, so it is possible to activate it on a single part only, or both.

The sustain, by default, works on both parts, but it is possible to disable it for a single part by holding down the Hold button and pushing the Part A or B buttons. The display will show the message SuS 1 when on, and SuS0 when off.

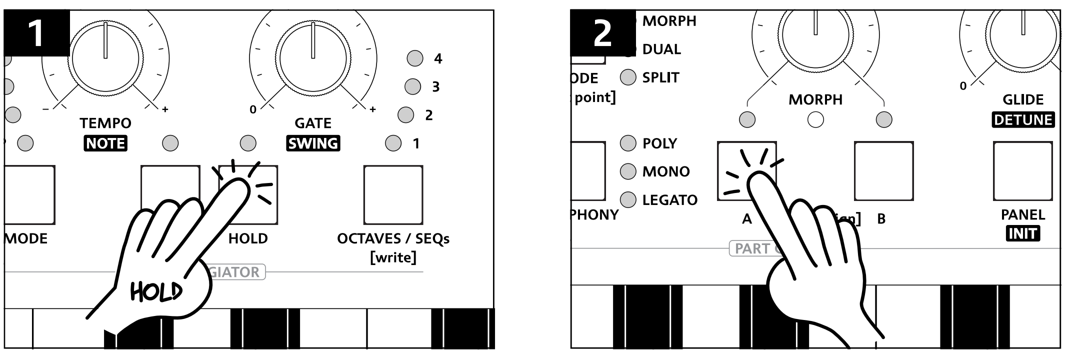


Figure 23: deactivating the Sustain pedal for a Part.

When the Arpeggiator is active, both the hold button and the sustained pedal will cause the arpeggiator to play even after you have released the keys. See below, p. 72.

When the Hold function is active, the Sustain pedal will not have any effect over that Part. This can be useful, for example, in some bi-timbral programs, for example where a part must play a continuous arpeggio through the Hold button and the other Part is a dreamy pad that requires a bit of old-fashioned sustain pedal technique.

Holding the sound is also useful for drone music, for example when a static note or chord is modulated by LFOs or looping envelopes.

Copy and Paste a Part

In the previous chapter, we saw how to save a Program (p. 9), which stores the setting of both parts. However, Magnolia also allows you to save the current part's setting to another Part, either within the current Program or into another one. Here is how to do it:

1. Select the part you want to copy.
2. Hold **SHIFT** and push **COPY**. Magnolia has now copied the current part to its clipboard and has entered the "paste" stage: the display will show PASt.
3. Select the desired Program (Omit this step if you want to overwrite the current Program)
4. If the display still reads PASt, select the Part on which you want to copy the current setting. If not, hold **SHIFT** and push **COPY** to recall the paste function then select the target Part.

While the display shows the PASt sign, Magnolia is ready to paste the part. However, after a few seconds of inactivity or when selecting another Program, the Magnolia will return to the default programming mode. However, the clipboard will still contain the copied part, and it is possible to recall the Paste function by holding **SHIFT** and pushing **COPY** again.

To empty the clipboard and abort the Copy procedure, push **ESC**.

Swap Parts

It is possible to swap Parts A and B within a program. For example, in case you programmed a nice bass sound on part B and, when in Split mode, is in the right hand range instead of the left one. To do so:

1. Select any Part.
2. Hold **SHIFT** and push **COPY**.
3. Select the same Part of point 1.

Programs

All the settings that we saw in this chapter, as well as those that we will see in the following ones, are stored and recalled into Magnolia's memory in the form of Programs.

We already encountered programs during our first steps guide, and we already saw how to load, save, and delete them. In this chapter, we will see the remaining operations that we can do with Programs.

Effects

After designing the two sounds corresponding to parts A and B, you can apply effects like analog distortion, digital chorus, or digital delay to the whole Program. Since they are applied on the common bus, they will affect parts A and B equally. Read more about effects in the Sound Design chapter, p. 37.

Lists

A List is a sort of "speed dial" for Programs: it consists of 99 slots that you can assign to any Program to recall them in your favorite order, for example for a live set.

You can also assign the same program to multiple slots in case you need it various times throughout a live concert. So, for example, the first slot may recall Program 005, the second slot can recall Program 184, the third slot Program 005 again, and so on.

> The programs remain in their original memory location, so if you edit them, the updates will be automatically reflected in the Set List.

Magnolia contains 10 lists. To access the Set List mode, push an arrow button and then immediately push the other one.

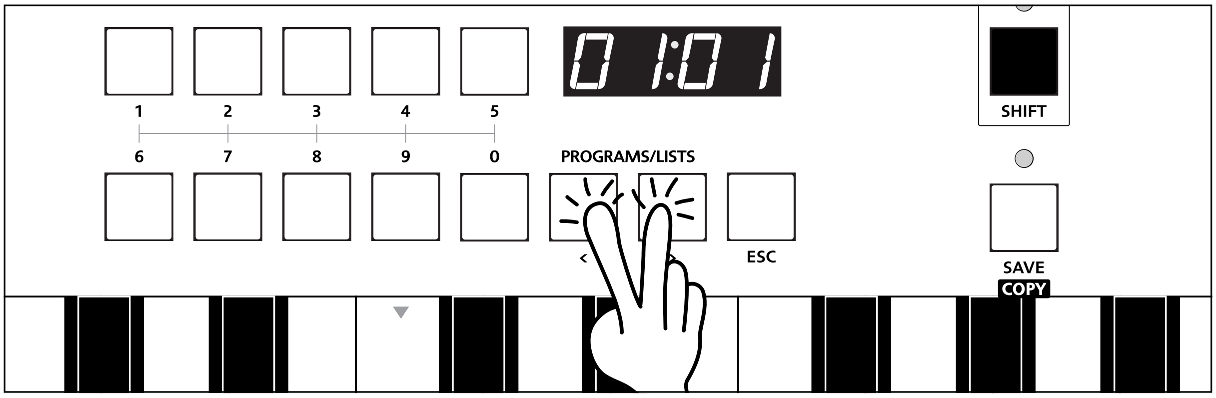


Figure 24: push the two arrow keys to access the Lists.

The Set List view contains two numbers: the first one is the List number (1 to 10) and the second one is the Slot number within that list (1 to 99).

- To change a Set List slot, use the numbered buttons **1**-**0** or the arrow keys.
- To change a Set List bank, push the **SHIFT** button and use the numbered buttons or the arrow keys.

To exit the Set List mode, push the two arrow buttons again.

Here is how to save a program to the Set List. (Remember to remove the memory protection with the switch on Magnolia's back.)

1. Enter Set List mode and select the desired List number.
2. Return to Program Mode and select the desired program, for example 314.
3. Push the save button. (The display number will flash.)
4. Push together the arrow buttons to select the Set List group.
5. Select the slot number.
6. Push the save button again to confirm.

During this saving procedure you will still be able to pre-listen the slot that you are about to overwrite.

> Another use of the Lists is to sort programs, for example: all the leads, all the brass sounds, all the cinematic presets…

Dynamic and Static Value Display

The Magnolia is designed to give a similar experience to a classic analog synthesizer's one, where you turn a knob and listen to what happens. In some cases, however, it could be desirable to have a finer degree of control and maybe even see the precise value of a certain parameter, for example to replicate a "sweet spot" on different patches.

The **VALUE** button allows you to perform those tasks: when activated, all the knobs will show in real-time their value on a scale that goes from 0 to 1000 for unipolar controls, e.g. the Amplifier, and -500 to 500 for bipolar controls, e.g. the Pan.

> The scale of 1000 increments is just an approximation: the controls have a much finer resolution.

To access the Dynamic Value Display:

1. Push the Value button (its LED will light up).
2. Rotate any parameter on the front panel, including the shifted ones.

The Fine Edit mode will stay active either until you push the button again, or for five seconds after you rotated the last knob.

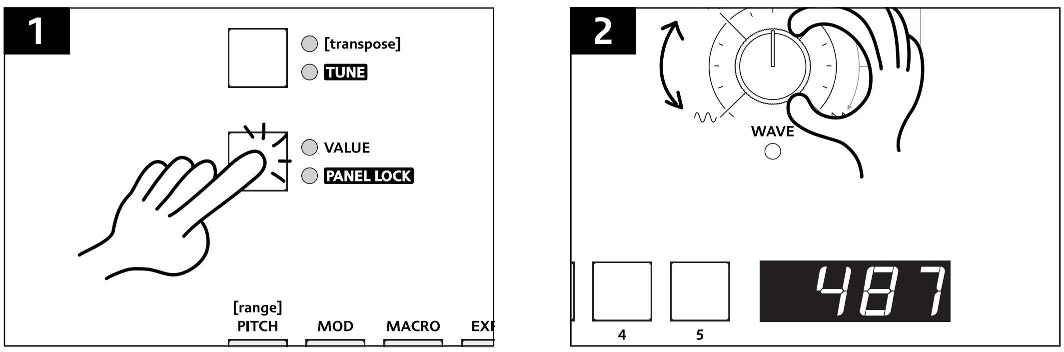


Figure 25: accessing the Fine Edit and Value Display mode.

Another useful function of the Value button is to display a parameter's value without changing it, which is especially useful for reverse-engineering presets. To perform the Static Value Display:

1. Push and hold the **VALUE** button.
2. Rotate any knob.

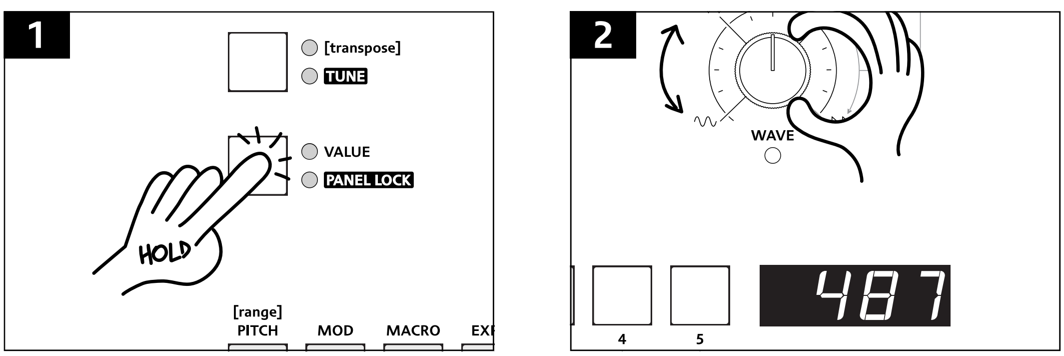


Figure 26: the Static Value Display mode.

The procedure will automatically end as you release the button.

The Static Value Display works also to check the modulation amount without changing it: see p. 46.

Panel Lock

The **PANEL LOCK** button, when engaged, disables every control on the front panel, except for the main Volume and Phones ones. When in Panel Lock, the Magnolia will still respond to MIDI CCs and Program Change messages.

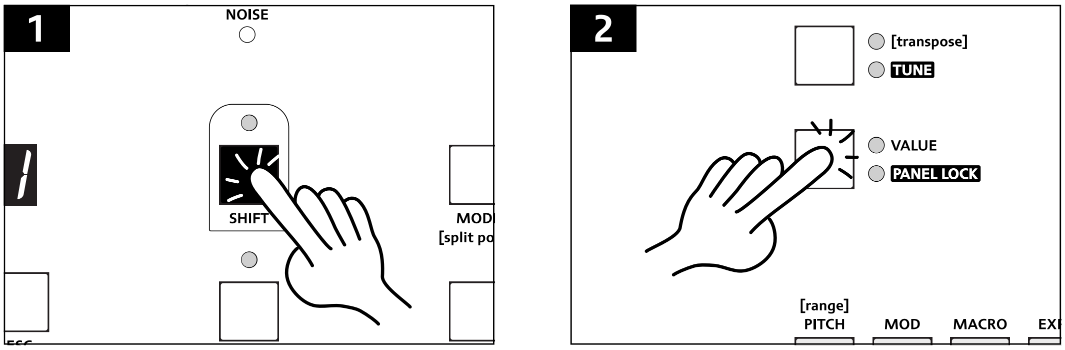


Figure 27: the Panel Lock procedure.

It is particularly useful when you are playing live and do not want to accidentally touch the front panel.

Chapter 4: Sound Design

A sound is defined by how pitch, timbre, and amplitude change over time. Any synthesizer's job is to use their circuits to create such changes: the more changes it can do, the more expressive it is.

The circuits of a synthesizer have various controls, called *parameters*, that are accessible via the many knobs, faders, and buttons that form the interface.

We can group the parameters into four main categories: sound generators, sound treatments, mixers, and modulators. In this chapter, we will deal with the first three, and we will discuss the world of modulations in the following chapter.

Oscillators

> The oscillator is our synthesizer's sound source, like a string for a guitar. If you want to learn more about oscillators, refer to Appendix A, page 80).

The Magnolia has two analog oscillators per voice called Oscillator 1 and Oscillator 2. They are built on the same triangle-core design but have different features: Oscillator 1 is the more experimental one, allowing for advanced synthesis techniques like thru-zero frequency modulation, flip sync, and wave folding; Oscillator 2 is more straightforward and provides an essential tool of subtractive synthesis, the PWM, but with a twist.

Octave Controls (Oscillators 1 and 2)

The main parameter of an oscillator is the frequency, which we mostly control through the keys we press. However, both oscillators allow us to define the range in which they play through the **OCTAVE** controls, which transpose them up or down by octaves.

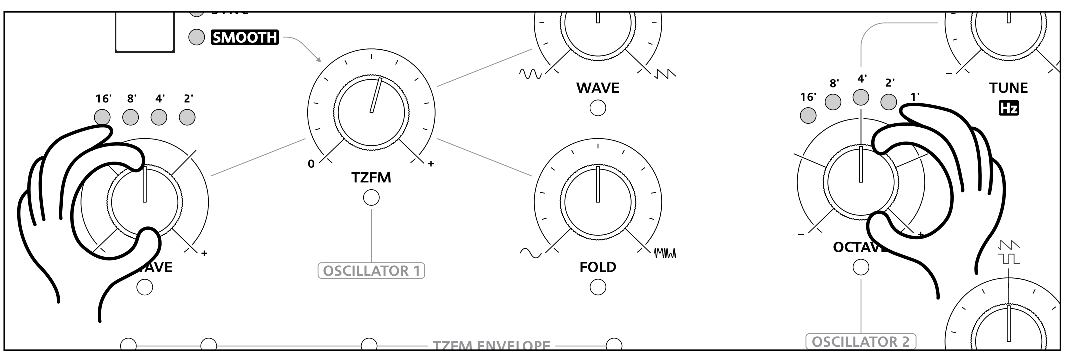


Figure 28: the Octave knobs of Oscillator 1 (left) and 2 (right).

Following the conventional nomenclature that appears in most classic synthesizers, which is derived from the pipe organ stops, the native pitch of an oscillator is 8′, meaning that with this setting, the A 4 key on the keyboard will correspond to the frequency of 440 Hz. 16′ is an octave lower, 4′ an octave higher, 2′ two octaves higher, and so on.

Oscillator 1 has four controls, from 16′ to 2′, while Oscillator 2 has an extra octave up to 1′. This range is very high per se, but it may be very useful when doing FM over Oscillator 1, see below p. 28.

In the introductory chapter on Parts and Programs, we saw that a Part can be transposed as well. Even though sometimes it may sort the same result, there are many differences between these two techniques that it is worth describing in detail:

- The Part Transpose affects the keyboard and the MIDI notes sent out, while the Octave knobs do not.
- The Part Transpose can transpose by any interval between a semitone and two octaves, while the Octave knobs are fixed to octave increments.
- The Part Transpose knob affects everything that responds to keyboard tracking, most notably the filters in tracking mode, while the Octave knobs affect only the oscillator.

To sum up, the Part Transpose is a function related to the performance, much like adding new keys above or below its five octaves, while the Octave knob relates to the sound design only.

Fine Tune/Hz (Oscillator 2)

Oscillator 2 features an additional **FINE** tune control capable of transposing the pitch up to a perfect fifth (7 semitones and 2 cents) higher or lower. This control is continuous, so it is possible to tune the two oscillators to natural intervals.

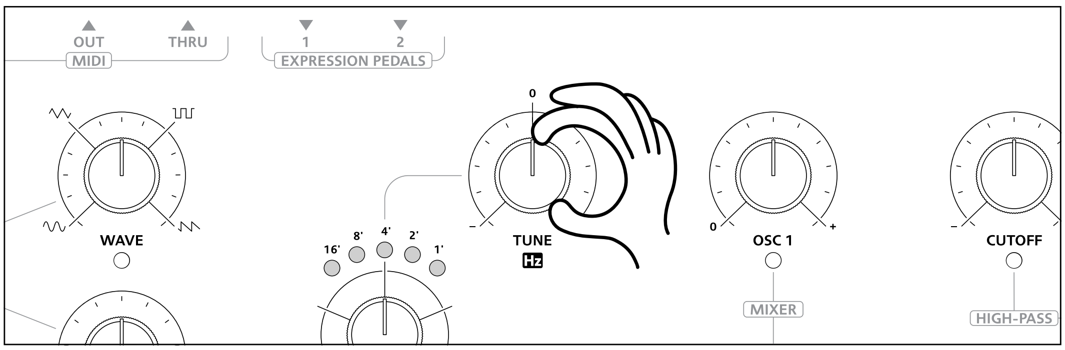


Figure 29: the Tune/Hz knob of Oscillator 2.

Another common use for it is to slightly detune the two oscillators to achieve a rich sound due to the phase cancellations and reinforcements called "beatings." Due to the exponential nature of note frequencies, the beatings of two detuned oscillators will be faster on higher notes and slower on lower ones.

If you desire to have the same beating pattern across all the notes, you must detune the oscillators by Hertz, instead of musical intervals. For this purpose, the knob has a **SHIFT** function called **HZ**: rotate it to detune the oscillator by 1.5 Hz above or below the selected frequency.

The two functions of this knob can be combined.

> When doing subtractive synthesis, the Fine Tune control is used to tune the oscillators to specific intervals or to slightly detune them; when doing FM synthesis, it allows to tune the modulating oscillator to achieve the desired timbre on the Carrier one.

Wave Shape (Oscillator 1 and 2)

The two oscillators produce four different waveforms at the same time, and through the **SHAPE** knob it is possible to continuously morph between them.

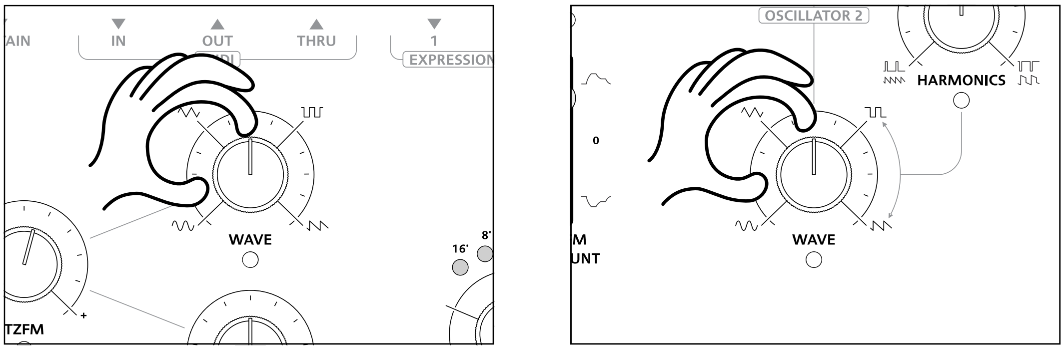


Figure 30: the Wave knob of Oscillator 1 (left) and 2 (right).

Starting at the leftmost position, we have the *sine* wave, which is the purest waveform available. By rotating the knob clockwise, we gradually reach the *triangle* wave, which has slightly more harmonic content. By rotating the knob even further, we reach the *square* wave, which contains only odd harmonics like the triangle, but in a different proportion. Then, when the knob is fully clockwise, we can hear the *sawtooth* wave, which is the richest waveforms of the set, containing odd and even harmonics.

> If you want to select the four "pure" waveforms, use the **SHIFT** button: the knob's behavior will be discrete and will not blend the waveforms.

Harmonics (Oscillator 2)

Oscillator 2 has the same waveform set as Oscillator 1, but it features a Harmonics parameter that allows for further shaping of the sound's timbre.

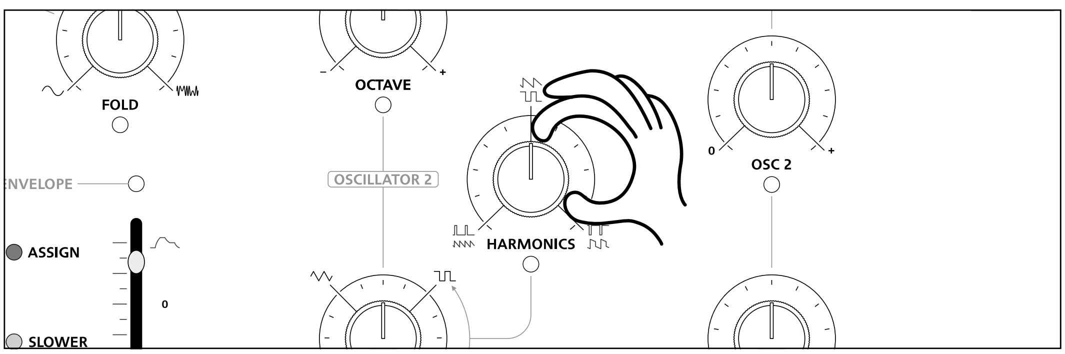


Figure 31: the Harmonics knob of Oscillator 2.

This parameter has an effect only on the square and sawtooth waves. On the square wave, it changes the duty cycle's symmetry, thus creating a pulse wave. The resulting sound will be progressively thinner and richer in even harmonics.

> If this parameter is modulated through a triangle LFO, it will be an actual Pulse-Width Modulation (PWM) that resembles the behavior of two slightly detuned oscillators, see p. 63.

On the sawtooth, it shifts half a waveform up or down: at the leftmost position, it will progressively emphasize the first harmonic, to the point of making the saw wave an octave higher with a slightly lower amplitude; at the rightmost position, it will square up the sawtooth, emphasizing the odd harmonics.

Figure 32: diagram of the Harmonics effects on a square wave (left) and sawtooth wave (right)

You can obtain interesting results by setting the Wave Shape knob between the square and the sawtooth way and modulate their Harmonics at the same time.

Thru-Zero Frequency Modulation (Oscillator 1)

The main parameter of Magnolia's voice architecture is the analog thru-zero linear frequency modulation, or simply TZFM. Through this circuit, the sine wave of Oscillator 2 modulates Oscillator 1's frequency to achieve sounds otherwise unthinkable in the world of analog synthesis. According to the conventional FM terminology, Oscillator 1 act as the *Carrier* and Oscillator 2 as the *Modulator*.

> The modulating waveform of Oscillator 2 is always a sine wave, and rotating the **WAVE** knob will not affect the frequency modulation of Oscillator 1.

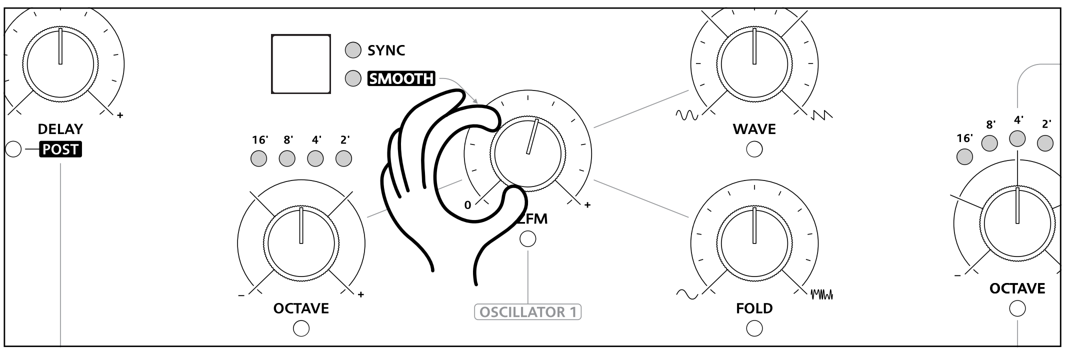


Figure 33: the TZFM knob of Oscillator 1.

Rotate the **TZFM** knob clockwise to increase the amount of modulation and obtain a richer timbre. Then, change the Oscillator 2's **TUNE** and **FINE** controls to create dissonant or consonant sounds.

> Please keep in mind that the Magnolia's FM knob defines the *deviation* in Hz, *not the index*, just as it was originally designed on the Brenso oscillator. As a consequence, the modulation amount will be perceivably higher in the lower range and lower as you play higher notes. If you want to emulate a classic index-based FM as in digital synthesizers, you may want to assign a gentle amount of keyboard tracking to the TZFM parameter or the TZFM Envelope Amount (see pp. 63 and 65). If you are new to these concepts of frequency modulation, please refer to the Core Concept section on p. 82 onwards.

In a classic FM patch, we hear just the Carrier. We say this just to introduce the fact that with FM, we are not much concerned with the Oscillator 2 a sound source, but rather as a modulation source whose effects we appreciate *through* Oscillator 1. However, this is just a general practice, and nothing prevents you from experimenting with the mixer and blend in Oscillator 2 as well.

The combination of modulator frequency and FM amount is responsible for an infinite number of timbers, so keep experimenting!

Since Magnolia's FM is analog, it may cause some slight pitch drift of the carrier at high modulation amounts. Feel free to use the **FINE** Tune knob on Oscillator 2 to compensate for this slight drift and remove or add any beatings.

TZFM Envelope (Oscillator 1)

Classic FM synthesis requires to dynamically control the modulation amount over the carrier to create a timbre that evolves over time. The most important tool for this purpose is an envelope, which is located precisely under Oscillator 1's parameters.

You can read more about envelopes in the Modulation section, on p. 48.

An envelope creates a dynamic control signal with every key stroke, and the **TZFM AMOUNT** slider defines how such a voltage must modulate the **TZFM** knob. Its default position is in the middle: at this point, no modulation is applied to the FM knob.

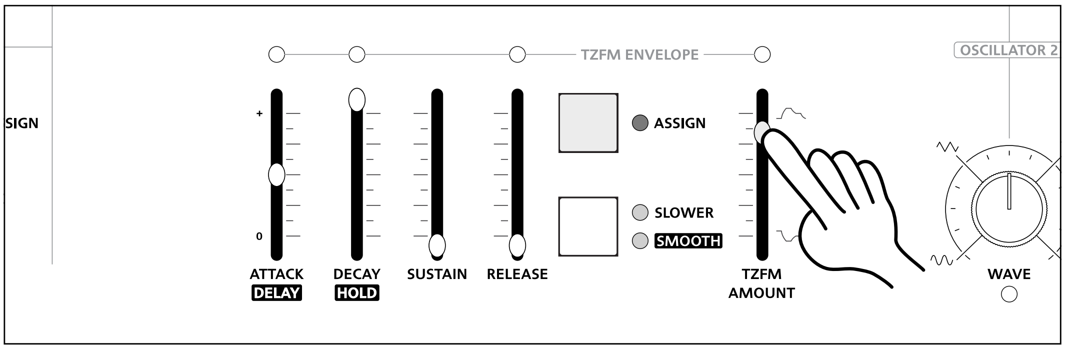


Figure 34: the TZFM envelope amount's fader.

Pushing the **TZFM AMOUNT** slider upwards will apply a positive modulation: pressing a key and firing the envelope will cause the modulation amount to increase and then return to the level set by the knob. If the FM knob is fully clockwise, the envelope will not have any effect.

Pulling the **TZFM AMOUNT** slider down will apply a negative modulation, so the modulation level will decrease and then increase again while returning to the knob value. If the FM knob is fully counterclockwise, the envelope wont' have any effect.

Sync (Oscillator 1)

When Oscillator 1 is synchronized, it will invert its waveform a direction at every cycle of Oscillator 2. As a result, changing its frequency will no longer cause a change in pitch, but rather in timber and overtones, while the actual pitch is determined by Oscillator 2.

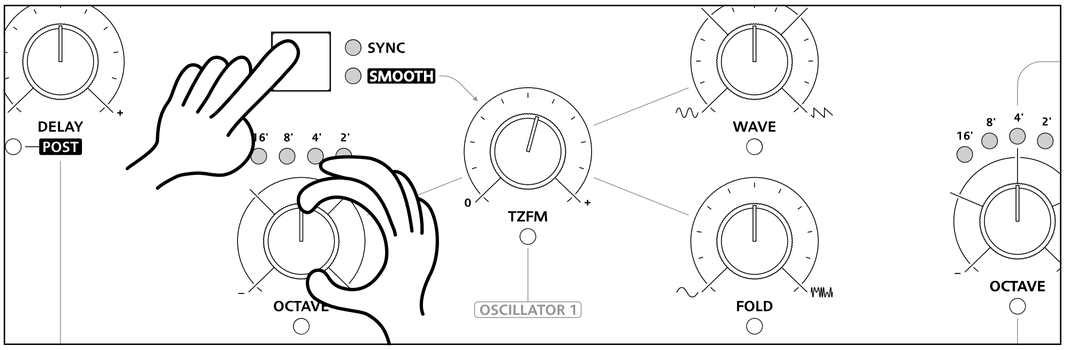


Figure 35: the Sync button and LED (the Octave knob becomes continuous).

To engage the sync mode, push the **SYNC** button in the Oscillator 1 section. At this point, the **TUNE** knob will no longer jump by integer voltages, but instead it will continuously modify the natural pitch between ± 24 semitones.

The practice of changing the synchronized oscillator's pitch is called *sync sweep* and has become a classic synth sound for piercing leads and basses. It can be performed manually or, better, through modulation sources (on which see p. 62).

Smooth FM (Oscillator 1)

When performing FM with carrier wave shapes like the square or the sawtooth ones, some rough crackles may happen at low modulation amounts. This behavior is a natural consequence of analog oscillators and can provide some interesting textures for distorted sounds, but in some other cases, it could be desirable to avoid it.

For this reason, we added a special circuit that uses Oscillator 2 to modulate Oscillator 1's amplitude. In this way, the points where the FM may cause crackles will be automatically suppressed. The result is a smoother and more manageable timbre, with slightly less energy in the low range.

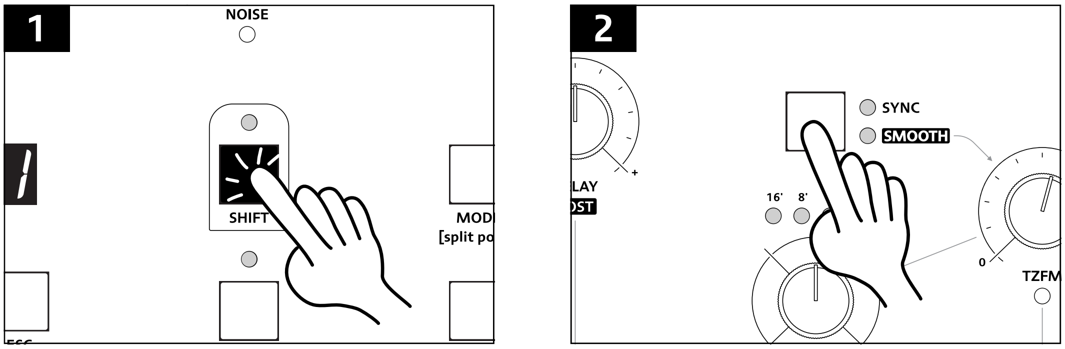


Figure 36: the Smooth button and LED (shifted function of the Sync button).

To engage this behavior, push the **SMOOTH FM** button: feel free to experiment also with triangle and sine waves, but the result may be "thinner" than regular TZFM.

Wave Folder (Oscillator 1)

Oscillator 1 has a parameter called Wave Folder. As the name suggests, it is an amplification circuit that folds the waveform back onto itself after it crosses a certain threshold. Once the wave reaches the end of the first fold, it is flipped back again and again the more you rotate the knob.

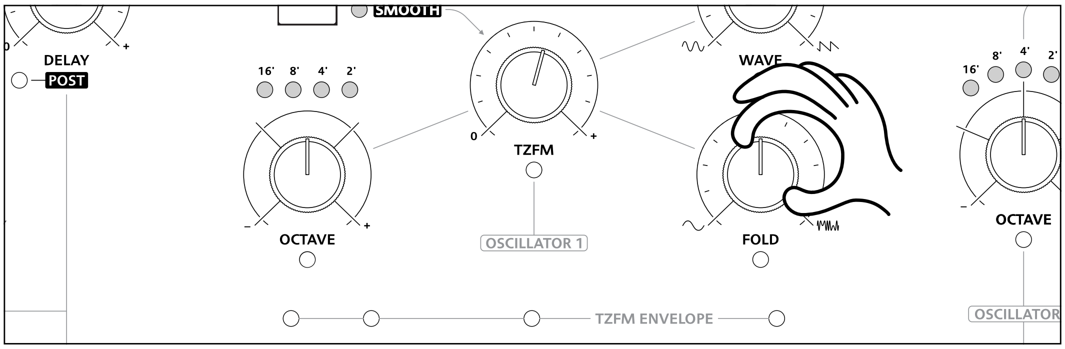


Figure 37: the Fold knob of Oscillator 1.

To increase the level of wave folding, rotate the **FOLD** knob clockwise.

By definition, a wave folder does not work particularly well over a square wave because it is already flat at its top and bottom, and the only result would be a phase inversion. If you want to use this waveform and also the wave folding circuit, consider setting the Wave knob a bit towards the Triangle or Saw waves, so to blend in some “foldable” waveforms.

> Try rotating the **WAVEFORM** knob with a medium or high wave folder level – the effect will be way more dramatic!

Noise

The third and last sound source on the Magnolia is the noise circuit. It outputs a white noise and has no parameters, except for the volume knob in the mixer, on which see the next section.

It can be used by itself as a sound source for sound effects like the “wind” sound, or it can be summed to other sound sources to add a percussive transient or a “breathing” texture. Finally, it can be used as a modulation source over the filter’s Cutoff Frequencies to add a gritty and lo-fi vibe to the sound. This last technique will be discussed in the filter FM section (p. 34).

Mixer

The Mixer defines which sounds will go to the filter section. It consists of three knobs corresponding to Oscillator 1, 2, and the noise source respectively. When a knob is fully counterclockwise, its corresponding sound is completely muted. Rotate the knob clockwise to increase its amplitude.

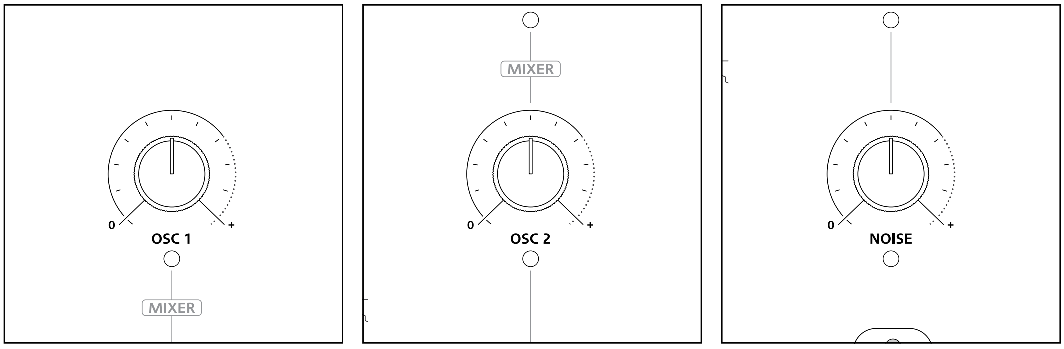


Figure 38: the mixer knobs for Oscillator 1, 2, and Noise.

The mixer has plenty of headroom, but the unity gain position is roughly at two o'clock. After that mark, the sound will gently distort by saturating the mixing bus. Such a saturation may alter the wave shape of the oscillators, especially when using the sine wave through the wave folder. Feel free to experiment with various gain staging!

Please mind that a mixer's headroom must be divided among the active sound sources, so the more they are, the easier it will saturate. In other words, the mixer may not saturate if a single oscillator is at 2 o'clock, but it will if all the sounds are at that level.

Filters

The filter is one of the crucial features of a synthesizer because it gives the sound its final shape. Magnolia contains two filters: a high-pass filter with an 18 dB/oct slope, and a low-pass filter with a 24 dB/oct slope. If you are unfamiliar with the concept of filters, refer to the Core Concepts section at the end of this manual, on page 81 onwards.

The filters are connected in series, meaning that the sound first passes through the high-pass, and then to the low-pass. This configuration allows to combine them to create a band-pass filter. The filters have the same controls. Cutoff, Resonance, and a common FM bus, even though the parameter's behavior is differs according to the different filter designs.

Cutoff

The high-pass filter removes the part of the sound below the cutoff frequency, and the low- pass filter the ones above it. Rotate the knobs clockwise to increase the cutoff frequency and the counterclockwise to reduce it.

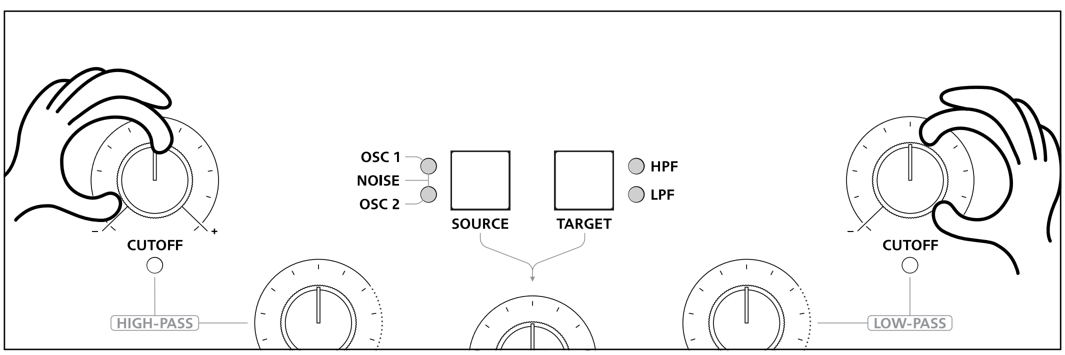


Figure 39: the Cutoff knob (high- and low-pass filter).

The high-pass filter will be open with the knob to the left and closed with the knob to the right, while the low-pass filter will work in a mirrored way, so open when clockwise and closed when counterclockwise.

If the high-pass filter's cutoff frequency is higher than the low-pass filter's one, no sound will pass through. However, if we respect this boundary, we can combine the high-pass and low-pass filters to create a band-pass filter.

A filter, by definition, works by subtracting portions of sound, so it is important to feed it with rich timbres like sawtooth waves, FM sounds, or folded waveforms to hear its effect. Waveforms with fewer overtones like the triangle, or even without overtones like the sine wave, will not produce any remarkable effect when filtered, except for an amplitude reduction.

If a filter has a fixed cutoff frequency, the notes will have a different timbre according to how close or far they are to such frequency. This may or may not be a desirable effect. In those cases where we need a timbral uniformity, so all notes with the same harmonic content, we need to shift the cutoff frequency with every note. This procedure is called *filter tracking* and can be engaged via the **TRACKING** modulation source, on which see page 58 and 63.

Cutoff Envelope

The envelope modulation of the cutoff frequency is one of the most classic parameters of a subtractive synthesizer. This envelope is identical to the FM Envelope that we saw above (p. 29) both in functions and in the way it is hardwired to the filters.

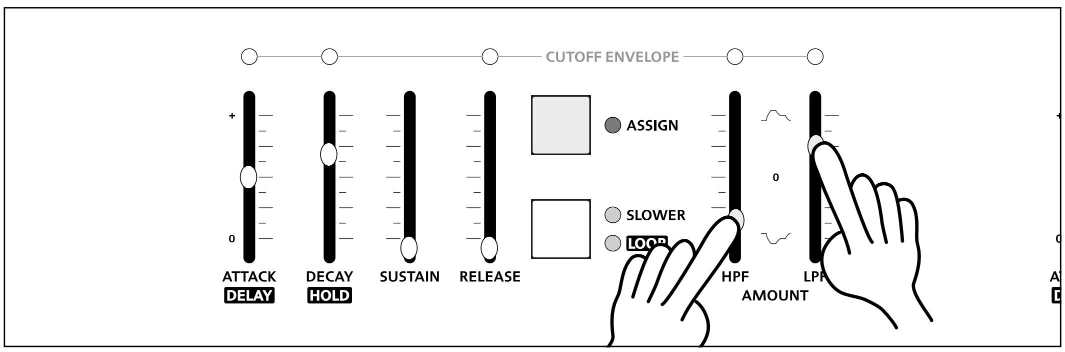


Figure 40: the cutoff envelope amount's faders of the high-pass filter (left) and low-pass filter (right).

Having independent controls over the high-pass and low-pass allows you to modulate only one filter, or even apply "mirrored" modulations, for example by increasing the high-pass' cutoff frequency while decreasing the low-pass' one.

Resonance

The Resonance parameter emphasizes the cutoff frequency by boosting its amplitude. Moderate Resonance settings will cause the cutoff frequency modulation to be more distinguishable, adding a more distinct character to the sound. Try setting the **RESONANCE** knob to noon and turning the cutoff frequency knob. The "filter sweep" will become more satisfying, and the filter function will pass from a gentle regulation to an intrusive timbral feature.

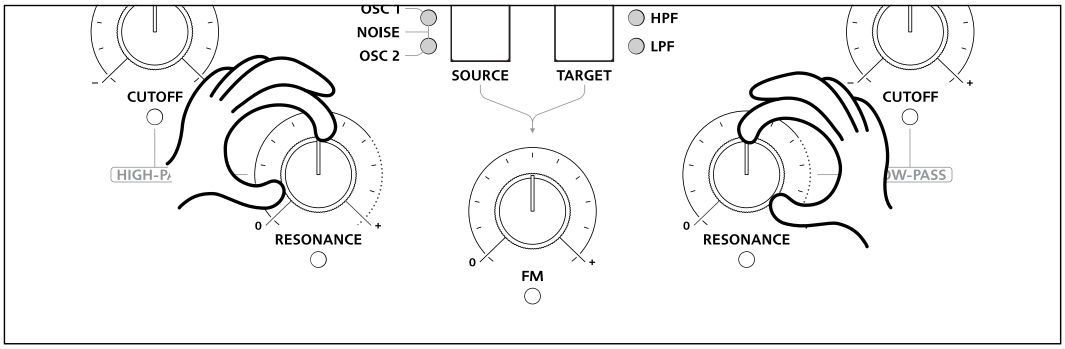


Figure 41: the Resonance knobs (high- and low-pass filters).

When fully clockwise, the Resonance circuit will make the filter "oscillate:" the emphasis on the cutoff frequency will be so high that it will produce an actual tone that can be clearly heard on top of the sound coming from the mixer.

Classic low-pass filters used to lose a lot of bass content at high Resonance settings, but Magnolia's one is capable of retaining all the low frequencies. As a result, the sound is richer but also more prone to distortion. If you want to achieve a more "ladder-like" filter, consider using a non-resonant high-pass filter to scoop away some bass frequencies.

If the filter resonance is past the point of self-oscillation, the filter can "play" a sine wave even without a sound patched to its input.

The two filters have radically different behaviors at high Resonance settings, roughly in the dotted section of the knob's graphic. The low-pass filter produces a relatively pure sine wave, while the high-pass filter generates a louder and more distorted wave, and it also reintroduces some of the filtered frequencies.

Modulating the filter cutoff frequency with the Keyboard Tracking will cause it to move together with the oscillators, on which see pp. 58 and 63.)

Filter FM

Both filters allow you to control their cutoff frequency at audio rate in a linear fashion, similarly to what happens to Oscillator 1 through its FM input. Since we are now modulating a filter, the effect will not be as dramatic as modulating the sound source, but it can create some interesting timbres, nonetheless.

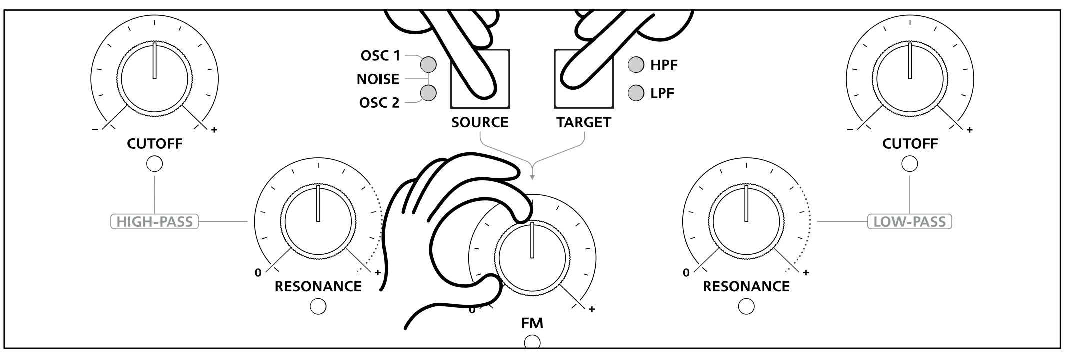


Figure 42: the filter FM section with the Source and Target controls (top) and FM amount (bottom).

The FM section has three parameters:

- **SOURCE**: defines if the frequency modulator should be Oscillator 1, Oscillator 2, or the Noise source.
- **DESTINATION**: defines if the modulator should affect the high-pass filter, the low-pass filter, or both.
- **FM**: defines the modulation amount of the selected source over the selected destination(s).

Filter FM is a slightly less conventional synthesis technique, but it can be found on many synthesizers since the early days since the Minimoog, probably as a byproduct of the modular approach where the distinction between audio-rate and modulation-rate was not so rigid.

Conventional modulation sources are oscillators and noise, which we thus featured on Magnolia, and their usual effects are various kinds of distortions. We recommend experimenting with different settings, but here are a few suggestions to start exploring:

- Use the noise as FM source and apply a gentle amount to the Cutoff frequency with a pinch of Resonance to create a rough sound texture.
- Feed the filter with only one oscillator and use the same as FM source for a peculiar distortion.
- Feed the filter with only one oscillator, but use the other as FM source, to bring in some strange overtones. Then, set the Resonance very high, the modulating oscillator’s frequency to its highest octave, and modulate the cutoff and the FM amount to make the filter “talk”.

Amplifier

The Amplifier stage is what makes the synthesizer stay silent when you are not playing it. An envelope like the FM and Cutoff ones is permanently wired to an amplifier with a full, positive modulation: this device allows Magnolia to output a sound only when a key is pressed.

This section also features controls over the part’s volume, its stereo balance, and even the eight voices’ placement across the stereo image.

Amplifier

The **AMP** knob defines the part’s volume. As opposed to the **MASTER VOLUME** knob, this control affects the Part and is thus saved in a Program. It has two main purposes:

a. It can define the balance between two parts in a Split or Dual program.
b. It can be a modulation target, like for tremolo effects, on which see the following chapter on Modulations.

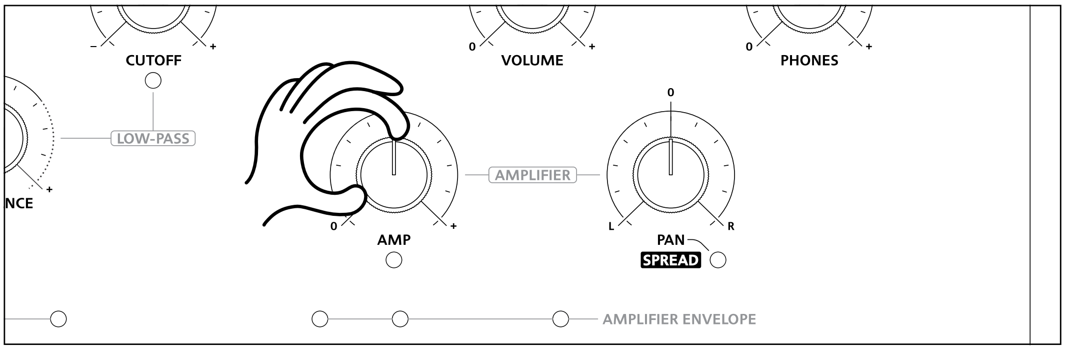


Figure 43: the Amplifier knob.

When the **AMP** knob is fully counterclockwise, the Part is silent. When fully clockwise, the Part is at its maximum volume.

Amplifier Envelope

This envelope is similar to the FM and Cutoff ones except for one element: it does not feature the modulation amount slider. The reasons are that it must be always active on the amplifier, and that it does not need to modulate it negatively, since a sound that is always on *except for* when you press the key is not particularly useful.

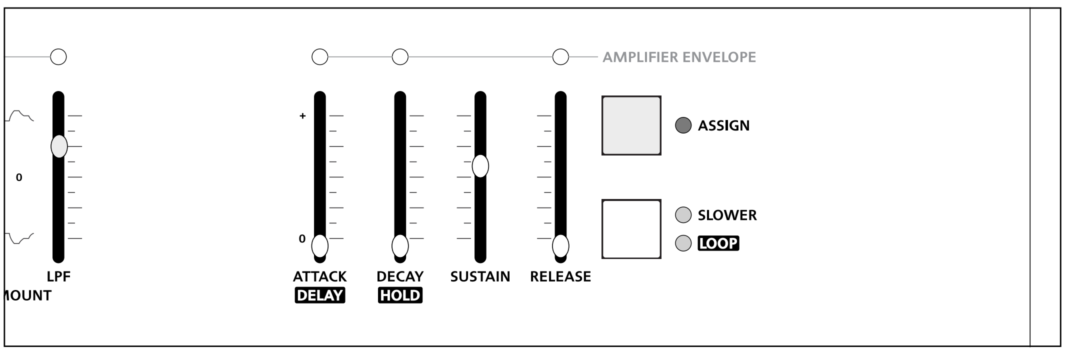


Figure 44: the Amplifier Envelope.

Just like every other envelope, this can be routed to any other modulation target, on which see pp. 41, 48, and 62.

Pan

The Pan knob defines the balance of a Part over the left and right output channels. When the **PAN** knob is at noon, the signal will be equally present on the left and right channels.

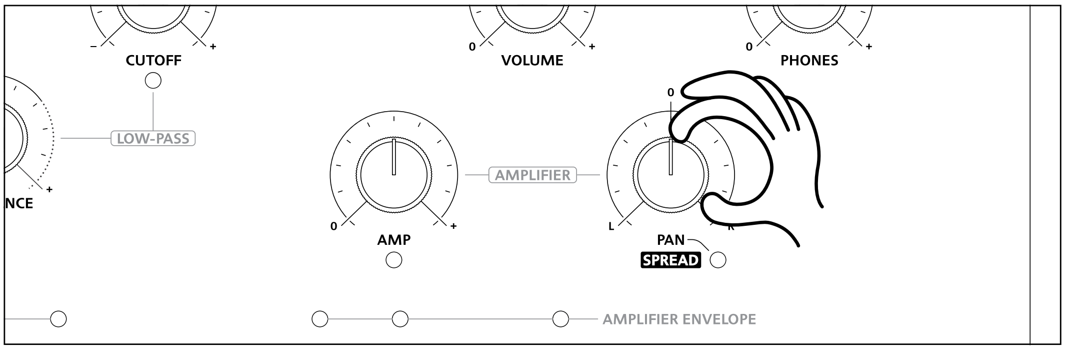


Figure 45: the Pan knob.

Moving it to the left or to the right will shift the balance accordingly, until the sound will come out of a single speaker or headphone pad.

Stereo Spread

This function regulates the stereo placement of the eight voices. When the knob is to the leftmost position, all the voices will be centered to the value set by the Pan parameter.

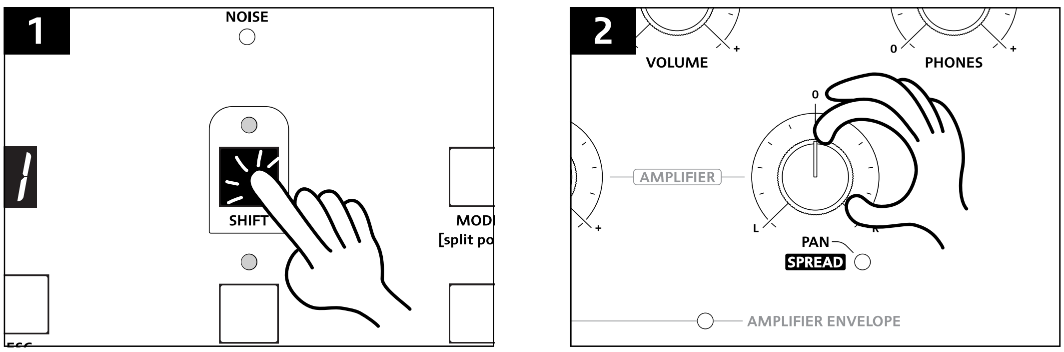


Figure 46: the Spread knob (shifted function of the Pan knob).

Rotating the **SPREAD** knob to the right will gradually spread the voices around the Pan point as they are activated by the keys pressed, in a ping-pong fashion: the first note will be offset to the left, the second one to the right, the third one to the left, and so on.

If the part is centered, it will result in a wider sound for polyphonic pads and monophonic leads in unison or chord mode, it can also be used to give a ping-pong panning effect to plucked arpeggios.

Effects

All the previous parameters belong to the voice cards and can thus have different settings per part. However, Magnolia also features global sound effects that affect all the voices equally.

Their settings are stored in the Program and can be modulated. However, since they affect both parts at the same time, they do not fully react to polyphonic modulations, see below, p. 47.

Distortion

The first one is the Distortion circuit, which can add anything from a soft saturation to a complete sound devastation to both Parts. The Distortion circuit is entirely analog and stereo.

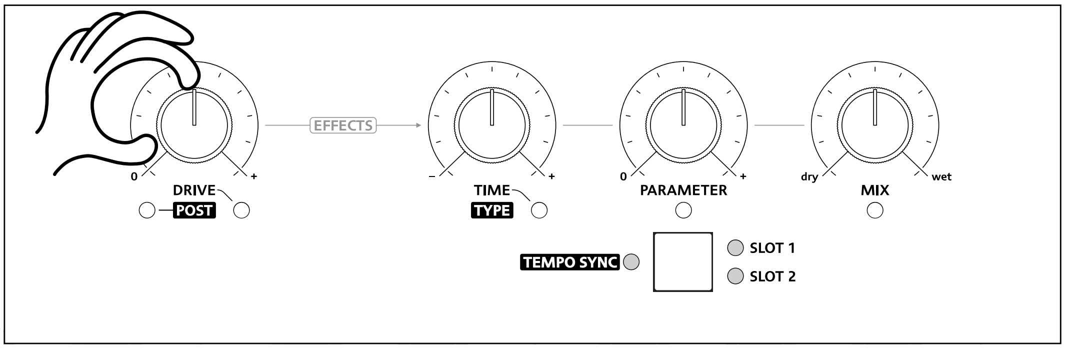


Figure 47: the distortion knob with its Drive and Post controls.

It features two parameters:

- **DRIVE** controls the distortion amount.
- **POST** controls the global volume after the distortion and before the other effects. It can be useful to lower the volume of a heavily distorted sound and balance it with other Programs.

Digital Effects

Every program has two chained effect slots that act as virtual stomp boxes fed one into another.

The **SLOT** button selects the slot to edit, cycling between effect 1 and effect 2. Each slot can have a delay or a chorus effect, so that you can choose your favorite order.

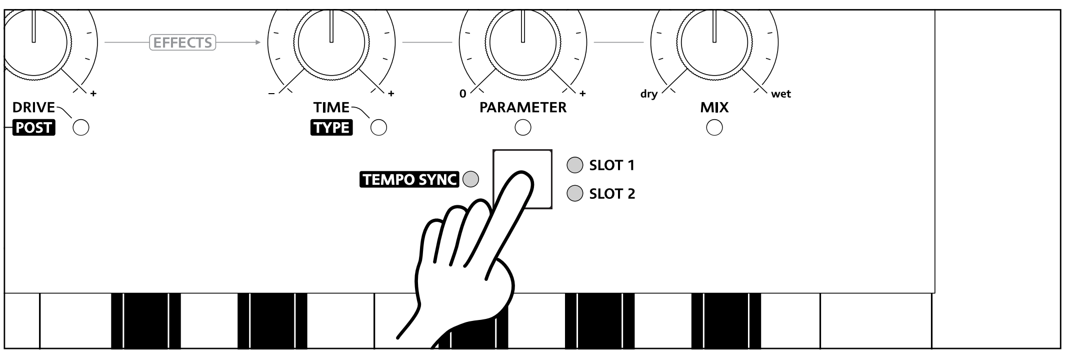


Figure 48: the effect slot selector.

While the general behavior involves using a chorus on Slot 1 and a delay on Slot 2 or vice versa, nothing prevents you from chaining two choruses or two delays for intricate modulation patterns.

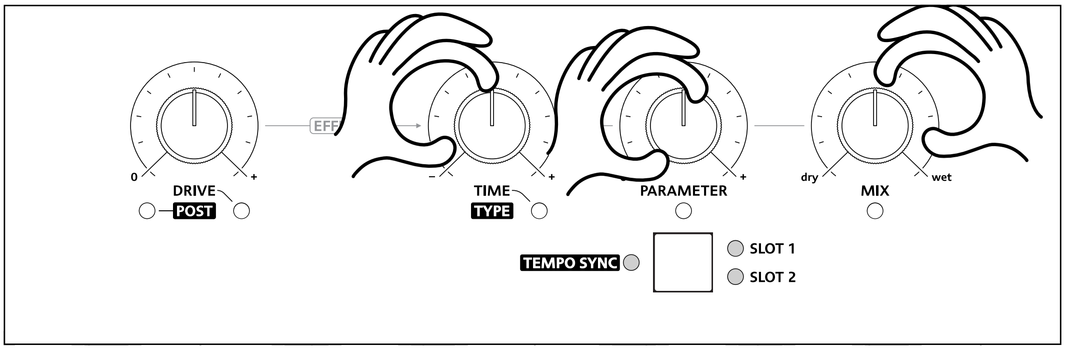


Figure 49: the digital effects' controls.

The effects controls are:

- **TYPE** selects the effect for the selected slot (see Table 1 below).
- **TIME** defines the speed of the selected effect.
- **PARAMETER** depends on the effect Type (see Table 1 below).
- **DRY/WET** selects the balance between the dry sound and the processed one. Since the effects are chained, the Dry sound of the second effect is the output of the first effect.

When the **TEMPO SYNC** function is activated, the Time knob is quantized to musical notes which will be shown on the display as with the arpeggiator's Note parameter, see Table 4, p. 39).

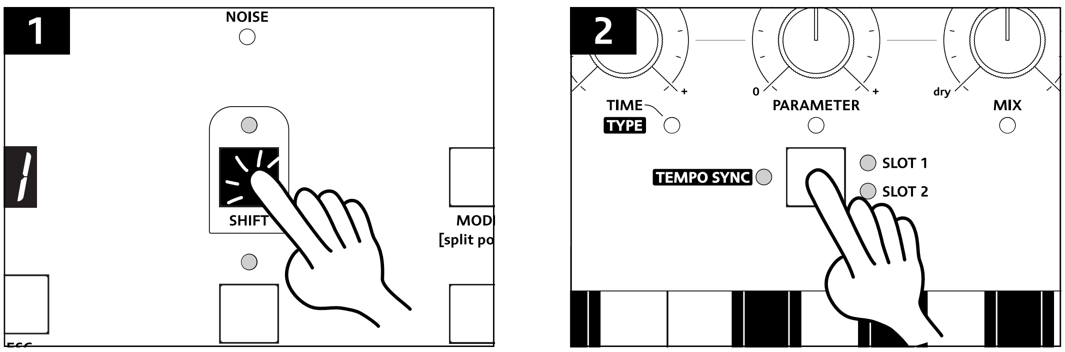


Figure 50: the Tempo Sync function, shifted control of the effect slot selector.

Display	Effect	Description	Param function
cho 1	**Chorus 1**	A classic chorus	Chorus feedback
cho2	**Chorus 2**	An amplitude-modulated chorus	Chorus feedback
dLY 1	**Delay 1**	A short stereo delay	Delay feedback
dLY2	**Delay 2**	A long ping-pong delay	Delay feedback

Table 1: the effects' abbreviations explained.

Chapter 5: Modulations

In the previous chapter, we learned that by tweaking certain knobs we can change the timbre of a sound. In this chapter, we will learn how to automate this manual knob-turning so that our sound can dynamically change over time.

This practice is called *modulation*. We will call *modulation sources* the "invisible hands" that will turn the knobs for us, and *modulation destinations* the knobs that can be automated. On Magnolia, the modulation sources are 16:

- Three envelopes
- Three LFOs
- Velocity On
- Velocity Off
- Aftertouch
- Keyboard Tracking
- Modulation wheel
- Pitch wheel
- Macro knob
- Polymove
- Two expression pedals.

Each of these can modulate most of the parameters that we explored in the previous chapter. To do so, we need to virtually connect sources and destinations, a process that we will call *modulation routing*.

> Envelopes are a bit different because each of them is also permanently connected to a specific parameter. See the Envelopes section below, p. 44.

Magnolia's Modulation Routing

Modulation routing consists of three kinds of operations:

a. Assign modulations.
b. Remove modulations.
c. Monitor modulations (i.e., understand what is going on).

On the Magnolia, these operations are all performed through intuitive button combinations, with no need for "menu diving" or programming skills.

All the parameters that can be modulated (the *destinations*) have a white LED close to their label. Every modulation *source* has a button that allows you to assign it where you need, as well as a red LED.

> To recap, the sources LEDs are red, and the destination LEDs are white, to state that they are a specific kind of operation different than those marked by the orange "system" LEDs.

Envelopes and LFOs are modulation sources whose settings are stored inside a part, and so they have an **ASSIGN** button close to them:

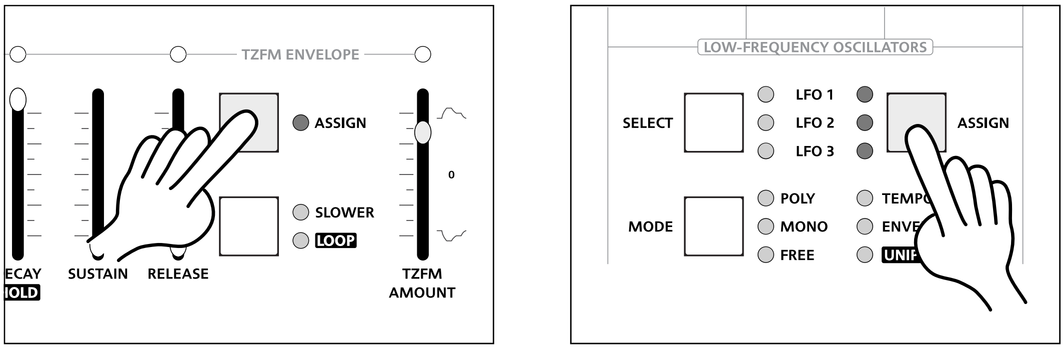


Figure 51: the Assign buttons of Envelopes (left) and LFOs (right).

while every other modulation source, the more “gestural” ones, have a button with the sources name in the Modulation Assign panel section.

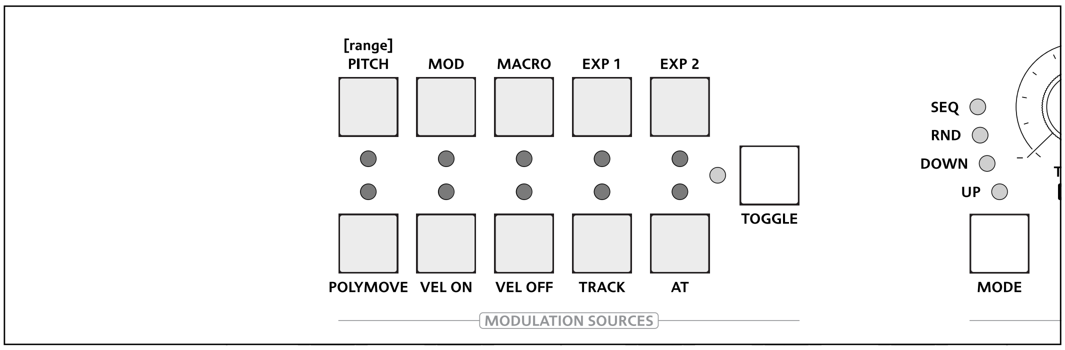


Figure 52: the Assign buttons of the other modulation sources.

The **TOGGLE** button deactivates a modulation (see p. 43).

Assign the Modulations

Assigning a modulation is like connecting a virtual cable between the source and the destination. This is the modulation assign procedure:

1. Push the desired Assign button. Its red LED will start blinking.
2. Move the desired parameter’s knob to the noon position to reach the modulation amount starting point.
3. Rotate it to the right for a positive modulation, or to the left for a negative one. The destination’s white LED will light up.
   (Repeat points 1, 2, and 3 with other sources and destinations if needed, or even just points 2 and 3.)
4. When done, push the last selected Assign button a second time to end the procedure. From this point on, the modulations will be active, and all the modulated parameters will have an active white LED.

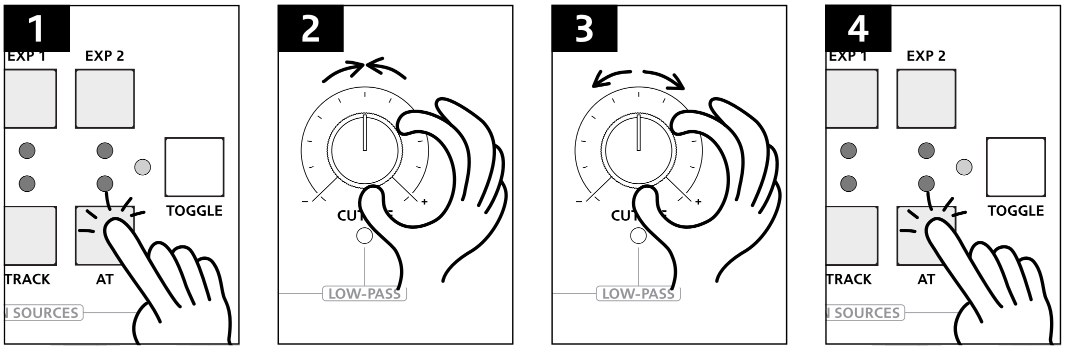


Figure 53: the Modulation assignment procedure.

To change the modulation amount, just repeat the procedure.

This procedure allows you to define the modulation target *and* the modulation amount with just one gesture. Please note two things:

- If you play a note while assigning a modulation, you will hear the modulation effect over the parameter increasing or decreasing in real time. You will not alter the parameter's value stored in the program.
- When assigning a modulation, the parameter's knob defines the modulation amount. At noon, no modulation is applied; rotating the knob clockwise will introduce a *positive* modulation; rotating it counterclockwise will introduce a *negative* modulation. You will see the modulation amount updating in real time on the display as a positive or negative percentage.

> For example, let us say that we want to control the cutoff frequency of the low-pass filter with the modulation wheel. After pushing the **MOD WHEEL** button in the Assign section, we will rotate the LPF **CUTOFF** frequency knob clockwise to make the sound brighter when we will move the wheel up, or counterclockwise to make it darker.

Deactivate the Modulations

The **TOGGLE** button allows you to deactivate an active modulation. However, unlike the assigning procedure, removing a modulation is an operation that may need to be performed in different ways while programming, namely:

a. Removing a single modulation destination from a source.
b. Removing all the modulation destinations from a source.
c. Removing a single modulation source from a destination.
d. Removing all the modulation sources from a destination.

Theoretically, everything could be done with just procedure (a). However, we know that programming a synthesizer is a creative work and we believe that it should pass the shortest time between your idea and its realization, so we added procedures (b), (c), and (d) to speed up your workflow.

> Deactivating a modulation is a non-destructive operation and you can always restore it to its previous value throughout the toggling procedure.

To remove a single destination from a source:

1. Push **TOGGLE**.
2. Push the desired Assign button.

3. Rotate the desired parameter knob.
   (Repeat point 2 and 3 for every desired source or destination if needed.)
4. When done, push the **TOGGLE** button to end the procedure.

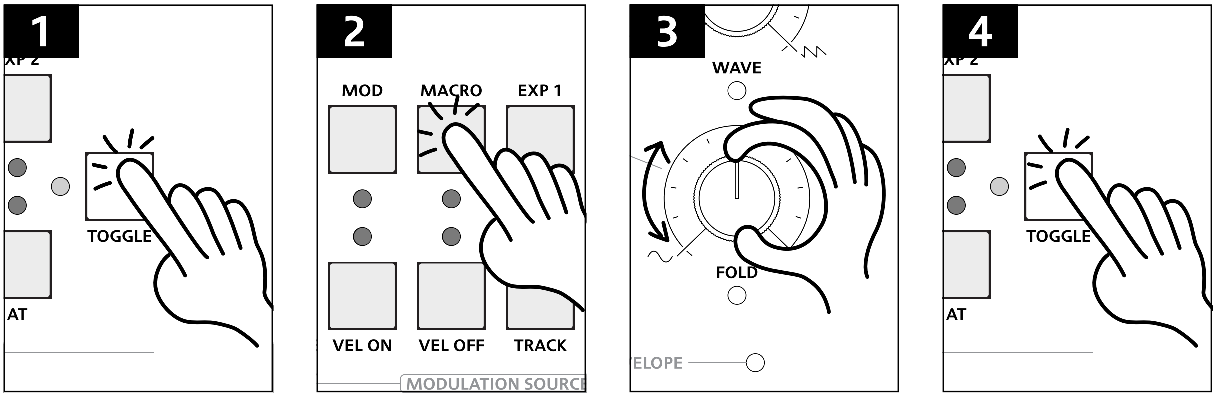


Figure 54: removing a single modulation destination from its source.

During the toggling procedure you can restore a deactivated modulation to its previous settings by repeating points 2 and 3.

For example, let us say that we want to remove the LPF cutoff frequency from the modulation wheel. After pushing the **TOGGLE** button and the **MOD WHEEL** button, we will rotate the **CUTOFF** knob to any direction to deactivate the modulation. If we want to restore it to its exact previous setting, we just have to rotate the knob again to any direction. Otherwise, we can end the toggle procedure and start a new assigning procedure.

To remove all the destinations and "clear up" a source:

1. Push and hold **TOGGLE**.
2. Push the desired Assign button.
3. (Repeat if needed.)

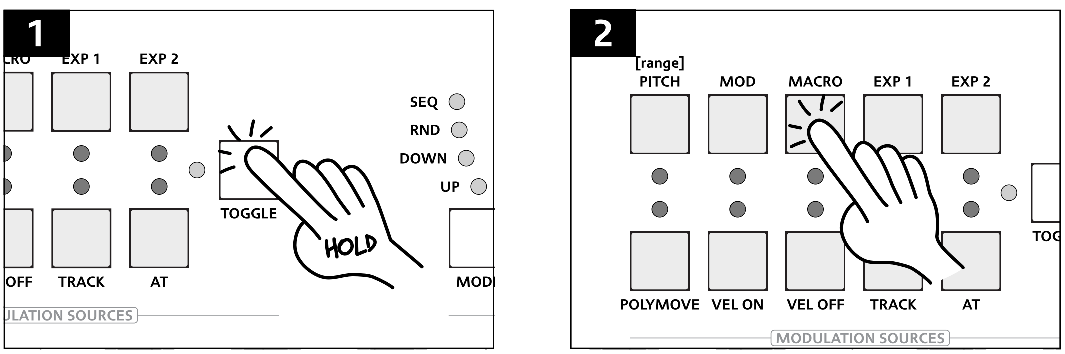


Figure 55: removing all the modulation destinations from their source.

The procedure automatically ends as you release the **TOGGLE** button. During this procedure, you can restore the modulations by pushing the desired Assign button a second time.

However, sometimes you may want to remove one or all the modulation sources that are currently affecting one parameter. It is a very common circumstance, for example when the amplifier is controlled both by the velocity and by an LFO for a tremolo effect.

1. To remove a single source from a destination:
2. Push **TOGGLE**.
3. Move the desired knob or fader. (Only the assigned modulation sources' LEDs will stay on.)
4. Push the Assign button corresponding to the source you want to remove. (Repeat points 2 and 3 with every desired destination.)
5. When done, push the **TOGGLE** button to end the procedure.

During the deactivation procedure you can restore a deactivated modulation to its previous settings by repeating points 2 and 3.

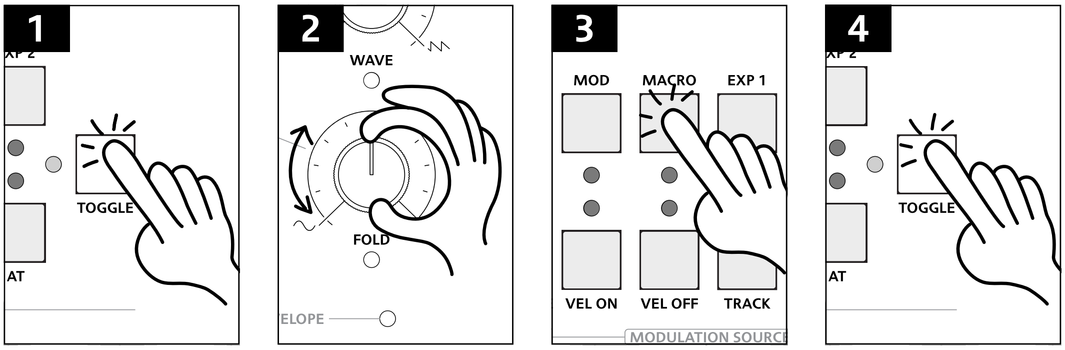


Figure 56: removing a single modulation source from a destination.

To remove all the sources and "clear up" a destination:

1. Push and hold **TOGGLE**.
2. Move the desired knob or faders.
3. (Repeat point 2 if needed.)

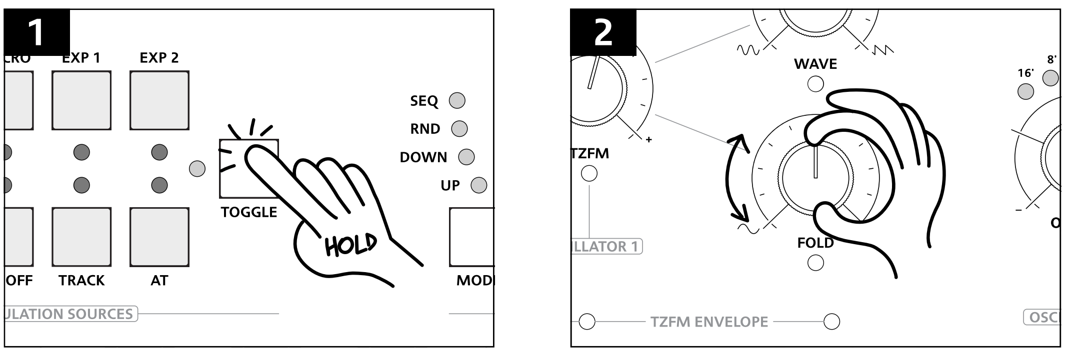


Figure 57: removing all the modulation sources from a destination.

The procedure automatically ends as you release the **TOGGLE** button. During the deactivation procedure you can restore the deactivated modulations to their previous settings by repeating point 2.

Monitor the Modulations

We at Frap Tools come from the modular synthesizer world where a modulation path can be guessed at a glance, because there are actual cables connecting sources and destinations. We wanted to preserve this kind of "physical" approach on the Magnolia, and we did so through the LEDs on the sources and on the destinations.

There are two ways in which you may need to visualize the active modulations:

a. From the *source* point of view. (For example: “how many parameters does the mod wheel control?”)
b. From the *destination* point of view. (For example: “what is modulating the wave folder right now?”)

To visualize the active modulations from the source’s point of view simply push the corresponding Assign button: only the modulated destinations’ LEDs will stay on.

To visualize the active modulations from the destination point of view, push the **TOGGLE** button without holding it, and then rotate the desired control. Only the active modulation sources’ LEDs will stay on.

Visualize the Modulation Amount

To visualize the current modulation amount without changing it, hold **VALUE** during the desired procedure. The display will show a static number, and the operation will not have any effect, much like the Static Value Display that we saw on p. 21.

Some Notes on the Modulation Sources

Every modulation source on Magnolia has unique features that one must keep in mind when programming a sound. Those features are of three kinds: polarity, polyphony, and level. Let us see what it means.

Unipolar and Bipolar Modulation Sources

When a modulation signal is unipolar, it means that its range is positive only, and so it will modulate the destination parameter in one direction from its starting value (above or below, according to the positive or negative amount). Unipolar modulations are useful, for example, to modulate the amplifier, since it is a circuit that usually stays closed and needs to be modulated only towards its open state.

When a control is bipolar, its range is both positive and negative, so the “zero point” is halfway through its whole range. It will thus modulate the destination parameter in two directions: above and below the starting value. Bipolar modulations are common, for example, to achieve a vibrato effect, since we would need to modulate the pitch up and down, as with a violin string.

Unipolar modulations are:

- Envelopes
- Velocity On
- Velocity Off
- Aftertouch
- Mod Wheel
- Expression Pedals

Bipolar modulations are:

- Pitch Wheel
- Polymove
- Macro knob
- Keyboard Tracking (though its assignment is a bit different: see the dedicated section below, p. 58)

The LFOs can be either unipolar or bipolar, see p. 56.

> Please keep in mind this important feature of unipolar and bipolar signals: when assigning a unipolar modulation, we rotate the target knob to reach the maximum level of modulation we need. When assigning a bipolar modulation, we rotate the knob to define the maximum *positive* level of modulation, but we must keep in mind that the modulation source will have an equal negative impact.

Both unipolar and bipolar modulation sources have the same span, but the unipolar ones are entirely above zero and the bipolar ones are around it. For example, if we say that a modulation source has a range of 1, if it is unipolar it will go from 0 to 1; if it is unipolar, from -0,5 to 0,5.

Polyphonic and Monophonic Modulation Sources

Another important distinction to keep in mind is the one between polyphonic and monophonic modulation sources: polyphonic sources can have different values per voice, while monophonic sources modify all the voices simultaneously by the same amount.

Polyphonic modulations are:

- Envelopes
- LFOs
- Velocity On
- Velocity Off
- Aftertouch
- Keyboard Tracking
- Polymove

Monophonic modulations are:

- Mod Wheel
- Expression Pedals
- Pitch Wheel
- Macro knob

Suppose that we play a four-note chord with the Velocity On mapped to the VCA: the keys that we played more heavily will make louder notes, and those we played more softly will make quieter notes. This is a polyphonic modulation because the same parameter can have different values in each voice.

Now, suppose to map the Mod Wheel to the VCA and move it: all the voices will play louder or quieter by the same exact amount. This is a monophonic modulation.

However, this distinction only applies to modulation destinations that are inside a Part; those outside it, i.e., the effects, will treat the polyphonic sources as monophonic by picking the highest available value of them all.

Part and Program Modulation Sources

The final distinction between modulation sources is between those within a Part and those within a Program.

The Part modulations are envelopes and LFOs, because each part can generate different modulation signals at the same time. For example, we can have a short VCA envelope on part A, and a long one on part B. We will call these modulations *Part Modulations*.

All the other modulations are of a higher level, and they generate the same values for both part A and part B. For example, if we strike a key with a high velocity, we generate a single velocity control signal. We can, of course, map it with different degrees to different destinations on parts A and B, but the original modulation signal is the same. We will call these modulations *Program Modulations*.

This distinction is useful to keep in mind when modulating the Morph knob: since it lies at a higher level than the individual parts, the Part modulations cannot modulate it.

Modulation	Polarity	Polyphony	Level
Envelopes	Unipolar	Poly	Part
LFOs	Unipolar/Bipolar	Poly/Mono	Part
Velocity On	Unipolar	Poly	Program
Velocity Off	Unipolar	Poly	Program
Aftertouch	Unipolar	Poly	Program
Keyboard Tracking	Bipolar	Poly	Program
Modulation wheel	Unipolar	Mono	Program
Pitch wheel	Bipolar	Mono	Program
Macro knob	Bipolar	Mono	Program
Polymove	Bipolar	Poly	Program
Expression pedals.	Unipolar	Mono	Program

Table 2: the Modulations' attributes

Envelopes

An envelope is a control signal that can increase or decrease a parameter's value multiple times according to the state of a keyboard's key. A conventional synthesizers envelope consists of four or more stages in which the control signal can go up, down, or remain static.

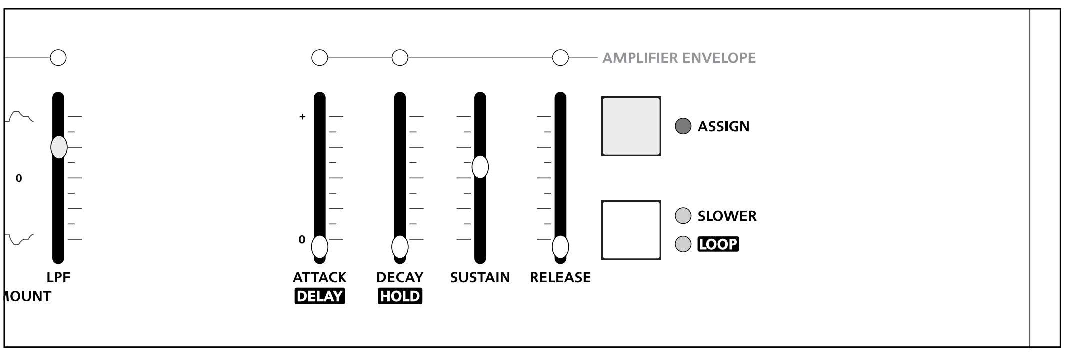


Figure 58: the controls of one envelope.

Magnolia's envelopes have four stages, but they can become six if needed. The four main stages are Attack, Decay, Sustain, and Release, each with its own control fader.

- **ATTACK**: this parameter defines how long does it take to the control signal to go from the lowest to highest point. This stage begins once a key is pressed.
- **DECAY**: defines how long does it take to the control signal to go from the highest point the point defined by the **SUSTAIN** parameter. This stage begins right after the Attack has ended.
- **SUSTAIN**: defines the value that the control signal holds after the Decay stage. This stage begins after the Decay has ended and lasts as long as the key is pressed.

- **RELEASE**: this parameter defines how long does it take for the control signal to go from the Sustain value back to the lowest value of the beginning. This stage begins once a key is released.

An envelope with those four stages is often called ADSR, and it looks like this.

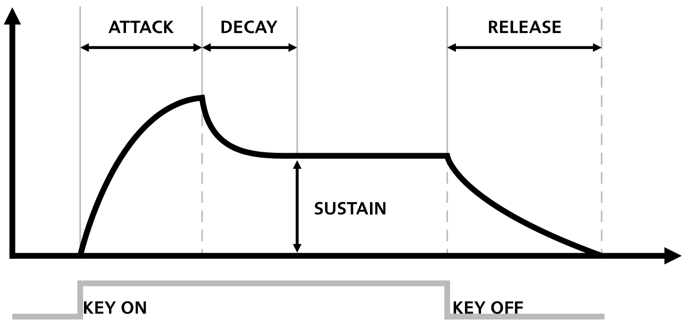


Figure 59: a four-stage envelope.

The Magnolia allows you to add two more stages, called Delay and Hold. The Delay parameter defines a time during which the envelope remains at its lowest value before the attack stage starts. Similarly, the Hold parameter defines a time between the attack and the decay stages during which the envelope stays at its highest value.

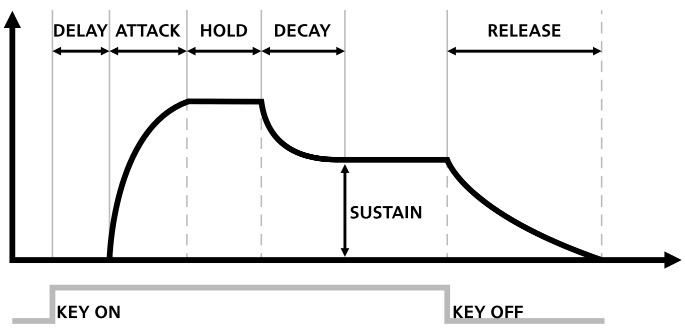


Figure 60: a six-stage envelope.

You can access the **DELAY** and **HOLD** controls by holding **SHIFT/ESC** while moving the **ATTACK** and **DECAY** faders.

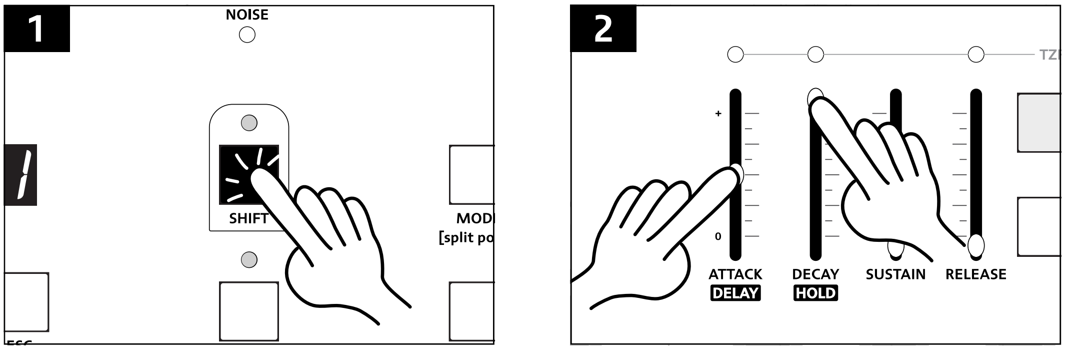


Figure 61: accessing the Delay and Hold stages.

The envelope is, traditionally, a control source that provides a fundamental expressiveness to the sound, since it can define how the timber changes over time when we strike a key, which is like what happens with acoustic keyboard instruments.

> The duration of the three envelopes' Sustain stage depends on how the time a key remains pressed. The Hold button and the Sustain pedal will cause the Sustain stages to be held as long as they are engaged.
>
> When the Arpeggiator is active, the duration of the Sustain stage depends on the Gate Length level.

Envelopes in Mono and Legato Parts

By default, an envelope fires as soon as a key is pressed, whether the Part is polyphonic or monophonic. When the Part is set to Legato, however, the envelope does not restart unless all the keys have been released first. If you thus play a part legato, i.e., by pressing a key without lifting the previous one, the envelope will fire at the beginning of the musical phrase and will develop independently from the notes that you play.

Assign Envelopes

The Magnolia has three envelopes, and that they are the only control sources that have a predetermined modulation target, respectively oscillator FM, filters cutoff, and VCA.

As we saw in the respective sound design chapters, the envelopes feature a fader that defines the modulation amount to the preassigned destination in a bipolar fashion. When pushed up from the central position, the envelope will modulate the preassigned parameter positively, as if turning the knob clockwise and back. When pulled below the central point, the envelope will modulate the parameter negatively, as if rotating the knob counterclockwise and back.

It is still possible to assign the envelope to other destinations, just like any other modulation source, through their **ASSIGN** buttons; however, having a dedicated amount slider for certain modulation targets has two advantages:

a. It makes standard patches easier to program.
b. It makes the envelope amount itself a modulation target, for more advanced programming (see p. 65).

Time Scale

The envelopes have a maximum attack, decay, and release time of 5 seconds. However, since in many times a slower envelope may be useful, the **SLOWER** button extends their range to 20 seconds.

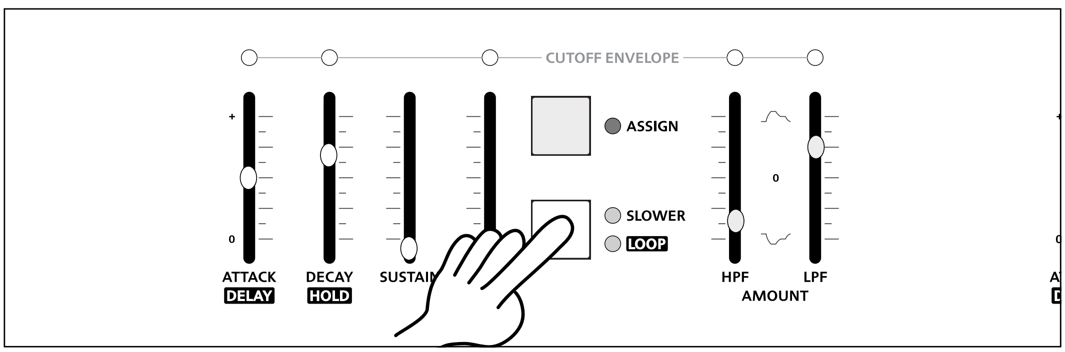


Figure 62: accessing the Slower time scale.

In this way, the default behavior allows for more precision in the faster range, while the Slower version is perfect for lush pads and evolving sounds.

Loop

The envelope is, traditionally, a momentary modulation, meaning that it plays once per note. However, the Magnolia has an alternative behavior that transforms the envelopes into cycling modulations like LFOs: the **LOOP** button.

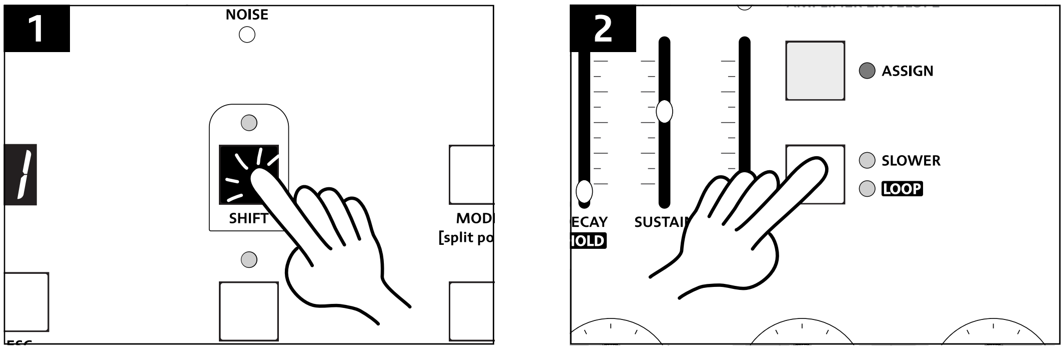


Figure 63: the Envelope Loop, shifted function of the Slower button.

When engaged, the loop mode makes the envelope cycle between the Delay, Attack, Hold, and Decay stages, starting when the key is pressed and stopping when it is released. After the key is released, the Release stage is engaged.

When the envelope is in loop mode the Sustain parameter becomes the modulation depth.

LFOs

An LFO, short for Low Frequency Oscillator, is a continuous modulation signal that cycles between a low and a high value. The Magnolia features three LFOs, all of them with the same settings.

To select the LFO that you want to edit and assign, push the **SELECT** button: an orange LED will light up besides the selected LFO.

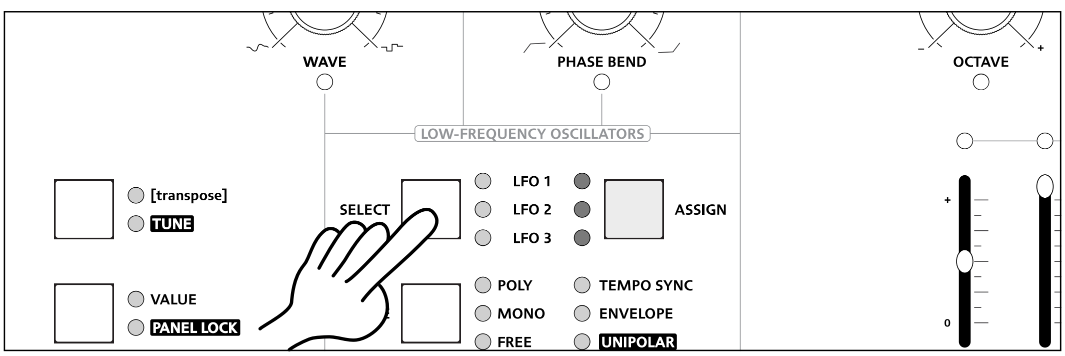


Figure 64: selecting an LFO.

The LFOs feature various controls and modes that will be discussed in detail in the following sections.

Rate

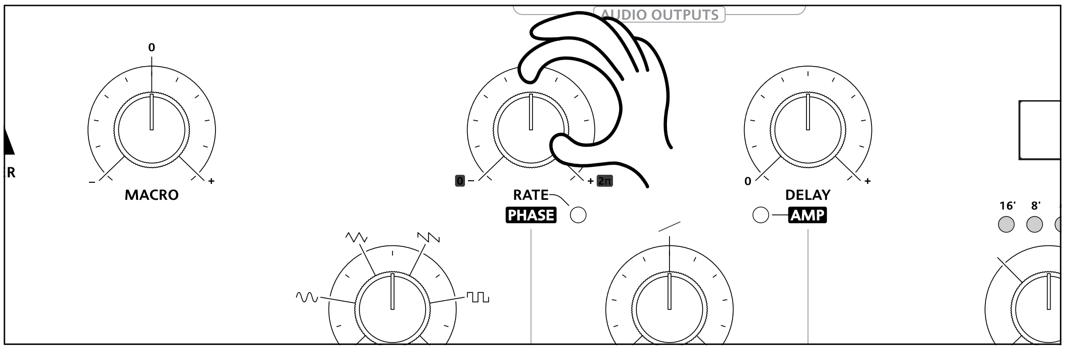


Figure 65: the Rate control.

The **RATE** control defines the speed of the LFO, from 0.1 to 10 Hz. Rotate the knob to the right make the LFO faster, and to the left to make it slower.

Delay

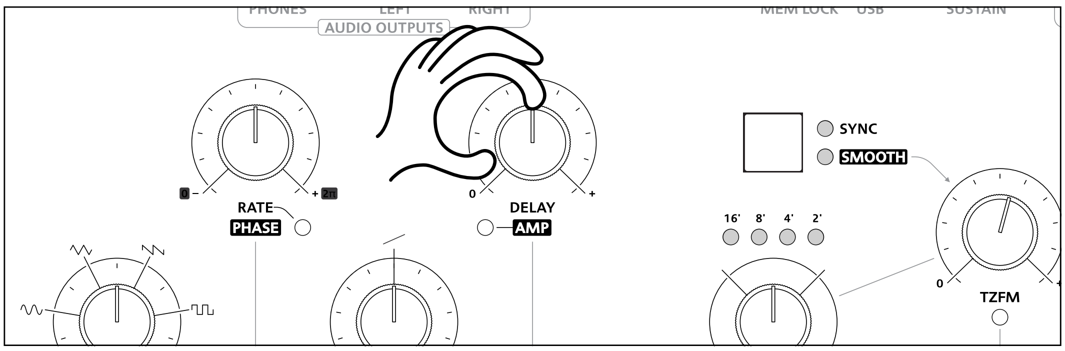


Figure 66: the LFO Delay knob.

The **DELAY** control introduces a “fade-in” effect to the LFO. When you press a key, the LFO gradually increases its effect from zero to the level set by the modulation amount.

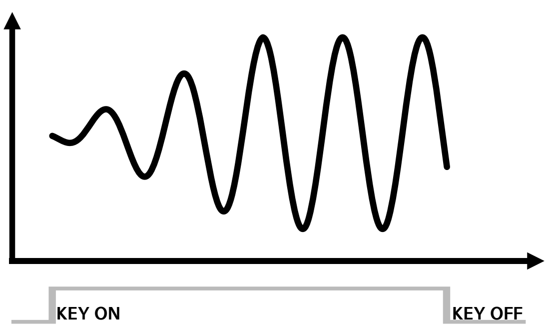


Figure 67: diagram of the Delay knob: the LFO starts quiet and gets progressively higher.

Wave

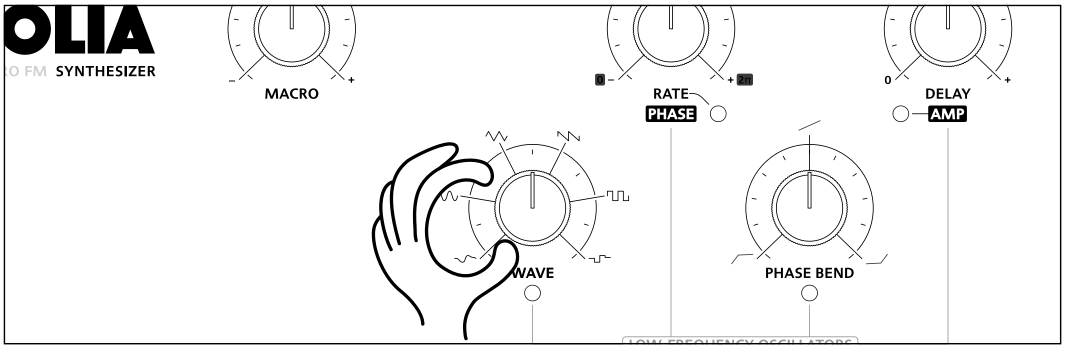


Figure 68: the LFO Wave knob.

The **WAVE** knob defines the LFO's waveform shape, which affects the modulation "trajectory" from the lowest to the highest value. The available waveforms are six:

a. Fluctuating random.
b. Sine.
c. Triangle.
d. Sawtooth.
e. Square.
f. Stepped random.

The various shapes produce different results. Sine, triangle, and fluctuating random LFOs will create "undulating" modulations, while shapes with abrupt transients like sawtooth waves, square waves, or stepped random voltages will have a more rhythmic connotation.

The **WAVE** knob's default behavior is continuous, allowing some hybrid wave shapes. Push **SHIFT** while rotating the knob to change its behavior to discrete, selecting just the pure wave shapes.

Phase Bend

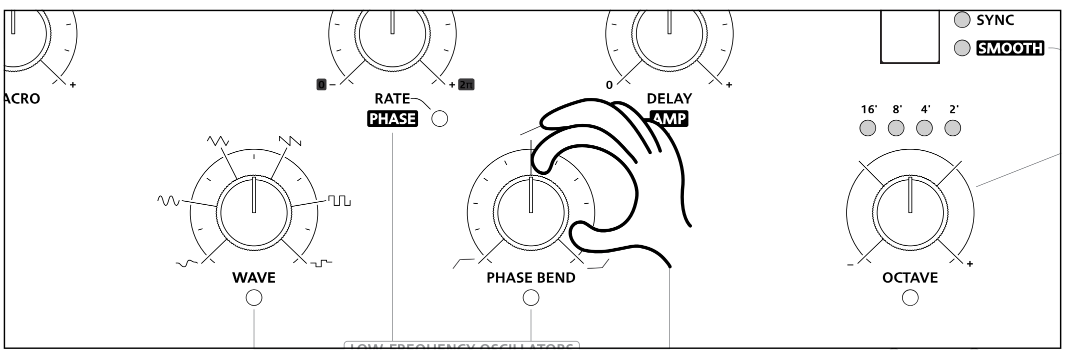


Figure 69: the LFO Phase Bend knob.

The **PHASE BEND** control distorts the wave shape selected by the Wave knob, basically by splitting every straight diagonal line into two segments. The result is an LFO that seems to become exponential or logarithmic.

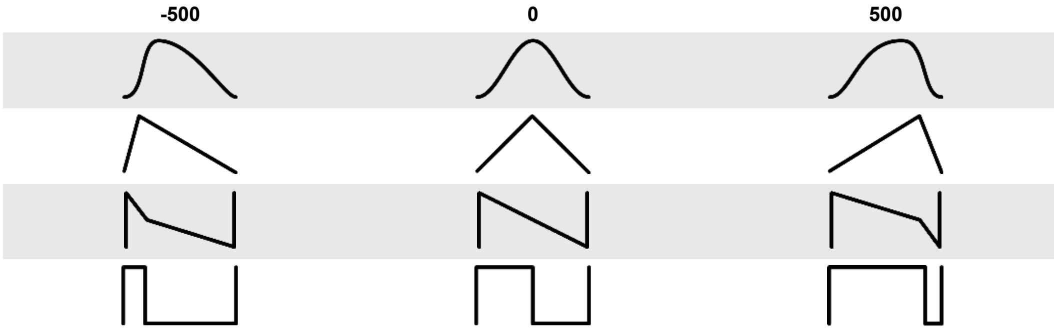


Table 3: Graphical representation of the Phase Bend control on sine, triangle, sawtooth, and square waveforms. The middle column shows no phase bend, the left column maximum negative bend, the right column maximum positive bend.

Phase

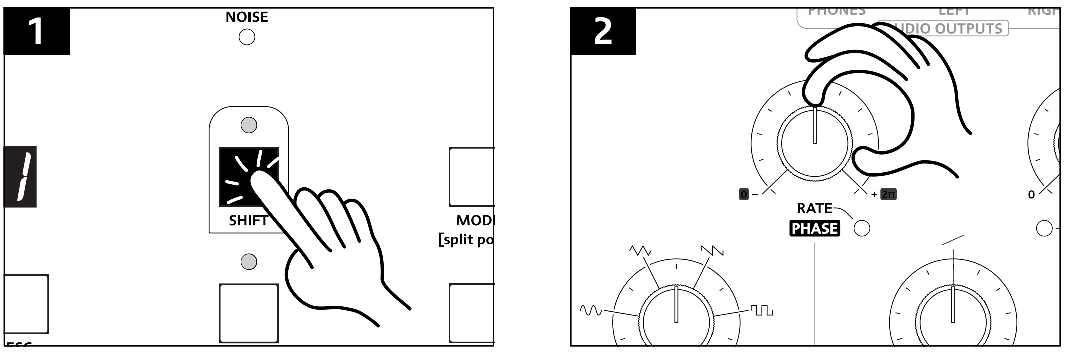


Figure 70: The LFO Phase control.

The **PHASE** control defines the starting point of the LFO. When set at 0, the LFO starts at its "zero point" of a bipolar waveform:

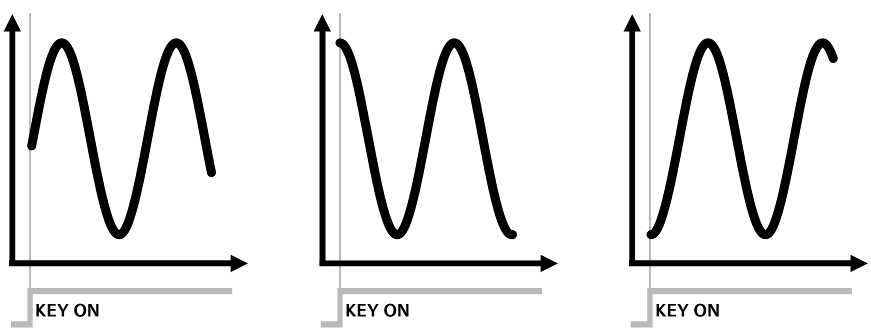


Figure 71: Diagram of three different Phase settings.

Rotating the knob clockwise shifts the starting point further down the waveform:

Please note that the Phase control is calculated on a bipolar waveform: when the LFO is set to unipolar, it will be shifted up, so the starting point will no longer be at 0 on sine and triangle waveforms. If you want a unipolar LFO to start at its lowest value, you may need to apply some phase correction.

Amp

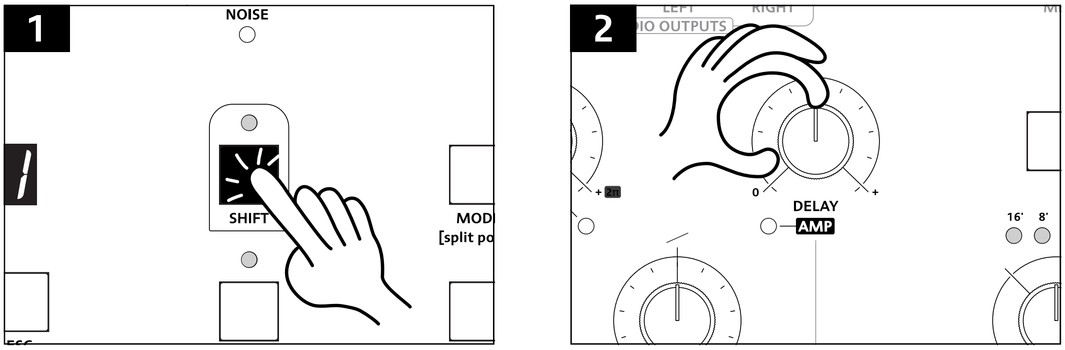


Figure 72: The LFO Delay knob.

The **AMP** control defines the global LFO amplitude. When the knob is set to the right, the LFO uses its full range; rotate the knob counterclockwise to reduce its amplitude, until the LFO modulation is completely inaudible.

The purpose of this parameter is to allow you to bring the LFO in and out by modulating it with another source, most commonly like the modulation wheel, an expression pedal, or the aftertouch. It is a more refined and specific way of dynamically change the LFO depth as opposed to the "pre-packaged" **DELAY** control.

By controlling the Amp parameter through an envelope with a slow attack, no decay, and a high sustain, you will obtain the same effect as the **DELAY** knob.

Please bear in mind that this parameter affects the LFO at its origin, and it will thus have a noticeable impact on every modulation destination.

LFO Modes

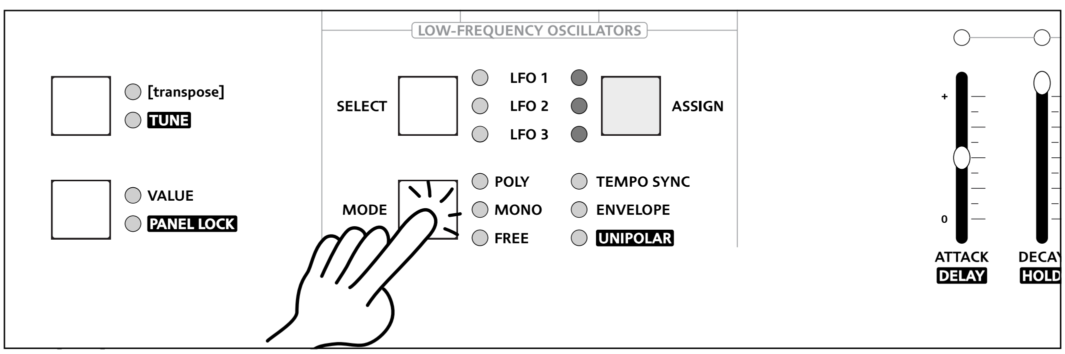


Figure 73: the LFO Modes.

We have seen that having an eight-voice synthesizer is like having eight monosynths. To a certain extent, this also applies to the modulations, so the Magnolia has eight copies of every LFO we see on the front panel. The **MODE** button defines how the LFO behaves on polyphonic patches.

a. **POLY**: every voice has its own LFO that starts when a key is pressed. This mode creates interesting moving pads, since the LFOs will have different phase relations with one another.
b. **MONO**: A single LFO controls all the voices simultaneously: it starts when the first key is pressed and does not retrig with any other note as long as one key is

held. This mode is useful when we want a single modulation to be heard and we want to have a sharp attack.

c. **FREE**: the same free-running LFO controls all the notes and never retrigs. Similar to the previous mode, but with a bit more unpredictability as the LFO will have no correlation with the played notes.

d. **TEMPO SYNC**: in this mode, the LFO is synchronized to the Arpeggiator Tempo (see p. 69). The Rate knob is thus quantized to rhythmic subdivisions of the Clock, which are shown on the display with the same conventions as for the Arpeggiator Notes, see Table 4, p. 71.

e. **ENVELOPE**: the LFO is polyphonic but plays one cycle only, triggered by the keyboard. In this mode, the LFO is exclusively unipolar and starts at the waveform's lowest point. The Phase and Delay knobs will not have any effect.

LFO Polarity

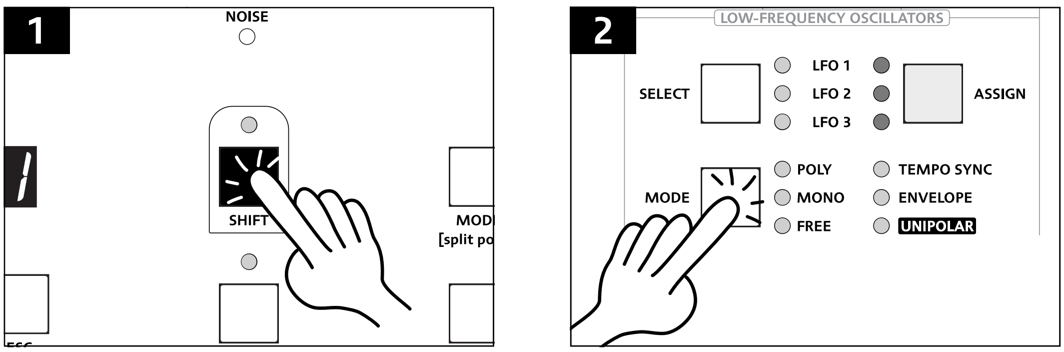


Figure 74: the LFO Unipolar switch.

The LFOs are by default bipolar (unless in Envelope mode). If you need an LFO to be unipolar, use the Unipolar switch: its amplitude will be the same, only offset up.

Velocity On

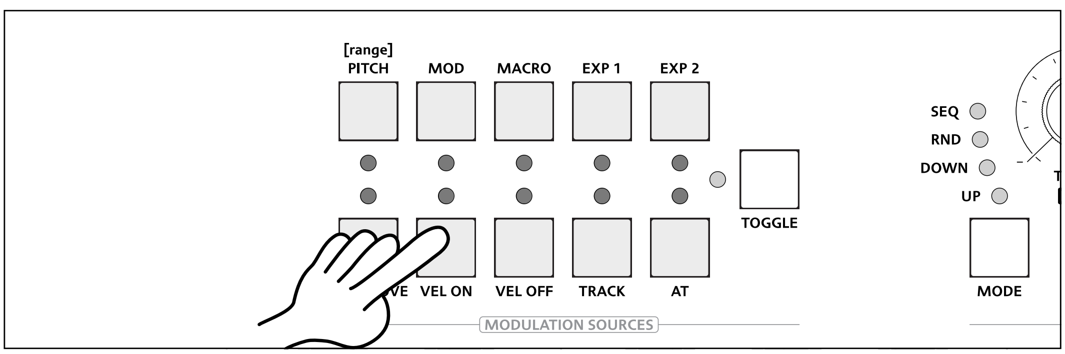


Figure 75: the Velocity Assign button.

Most of the acoustic instruments respond to the players' dynamics, meaning that there is a direct correlation between how strongly you play a note and its loudness and timbre.

Electronic keyboards allow to recreate this behavior through the velocity control, which is a control signal that is high when the key is struck heavily, and low when struck lightly.

Assigning the Velocity On control to parameters like wave folder, cutoff frequency, or the VCA is an efficient way to establish a correspondence between the players articulation and the dynamics of the sound.

A more advanced and expressive technique is assigning the Velocity On button to the envelope modulation amount so that with a soft dynamic the sound will have a small timbral development, and with a heavy dynamic it will change its timbre more dramatically.

Velocity Off

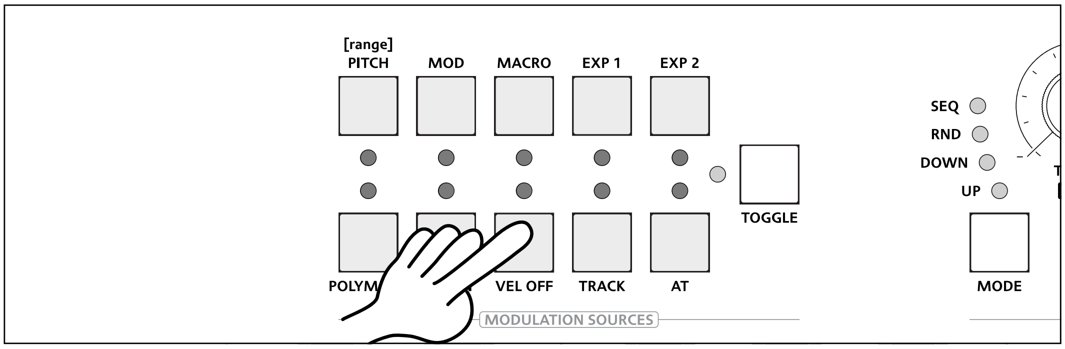


Figure 76: the Velocity Off Assign button.

This modulation source is slightly less common on synthesizers as it measures how fast or slow we release the keys after we pressed them.

Since it acts *after* we played the note, its effects will be audible only if the Amplifier Envelope has a substantially long Release time.

The Velocity Off source can be useful for subtle expressive effects. For example, try assigning it to an envelope's release time with a negative amount: it will cause the sound to have a slow release if you release the keys slowly, and a fast one if you release them fast.

Aftertouch

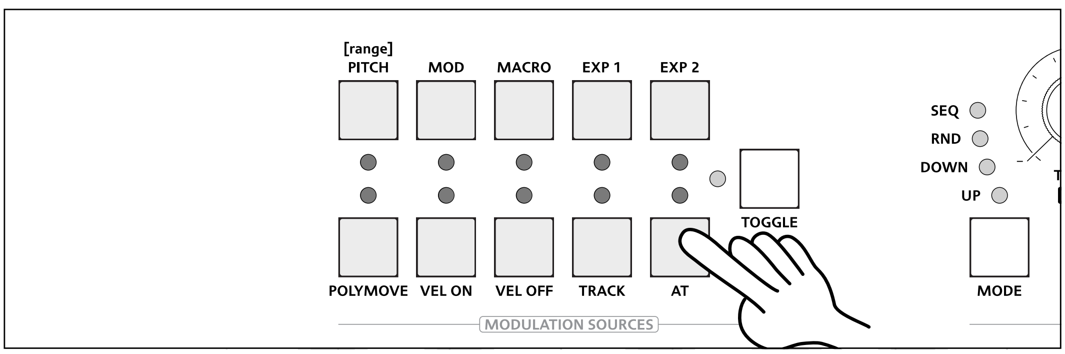


Figure 77: the Aftertouch Assign button.

A drawback of instruments like the piano is that you cannot change the dynamic of a sound after you pressed the key, an effect that would be extremely easy to achieve on instruments like strings or woodwinds.

The Aftertouch control allows this kind of expressiveness by sending a controlled signal when the key is further pressed after it has been struck. A high pressure corresponds to a high value, and vice versa.

Keyboard Tracking

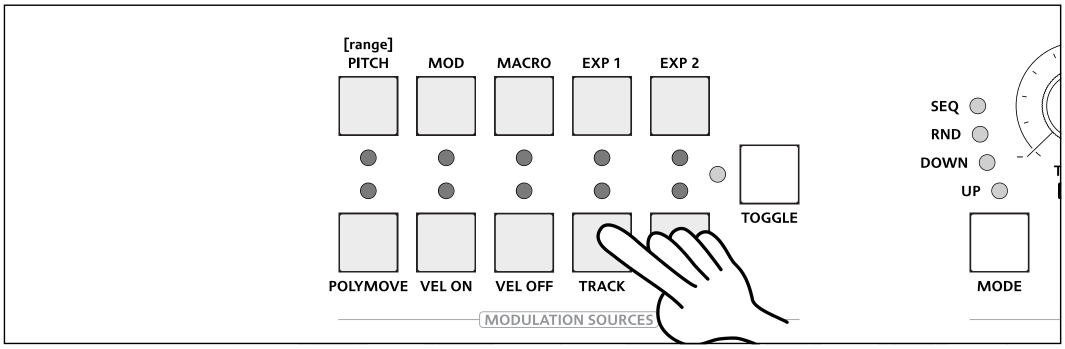


Figure 78: the Keyboard Tracking Assign button.

The Keyboard Tracking control allows you to scale a parameter according to the note that is playing. We already introduced this concept with the filter cutoff frequencies: by making them track along with the oscillator's pitch, it is possible to have a consistent timbre across the instrument's range.

This modulation source is different from all the others because it does not define an absolute modulation amount (as in: "the modulation goes from this point to this one"), but rather a proportion with the keys (as in: "the modulation must grow by this percentage with every key").

In other words, when we assign a Keyboard Track value to a parameter, we do not define the highest positive point, but the percentage according to which it will change in relation to the note played.

On the Magnolia, C 4 is the point where no modulation is active, like the pivot point of a seesaw. With every Keyboard Tracking setting, C 4 will always play the parameter's default value.

The modulation amount is expressed in the usual -100/100 range, except for two parameters: the oscillators' pitch and the filters' cutoff frequency.

The tracking on each filter has a range that goes from -200% to +200%, where 100 is the same tracking as the oscillators, 200 is twice as much, and the same goes for the negative range.

The oscillators have a tracking range that goes from 0 to 200%, where 0 is no tracking and 200 is double tracking (i.e., two octaves in the keyboard space of one octave). In the case of the oscillators, though, the default value (i.e., with no Tracking modulation and no active modulation LED) is 100%, since it is expected that an oscillator tracks the keyboard signal, unless otherwise specified.

> When assigning a keytrack modulation to the Oscillators and Filters, hold down the Shift button to select only the precise values of 0, ±50, ±100, and ±200.

Mod Wheel

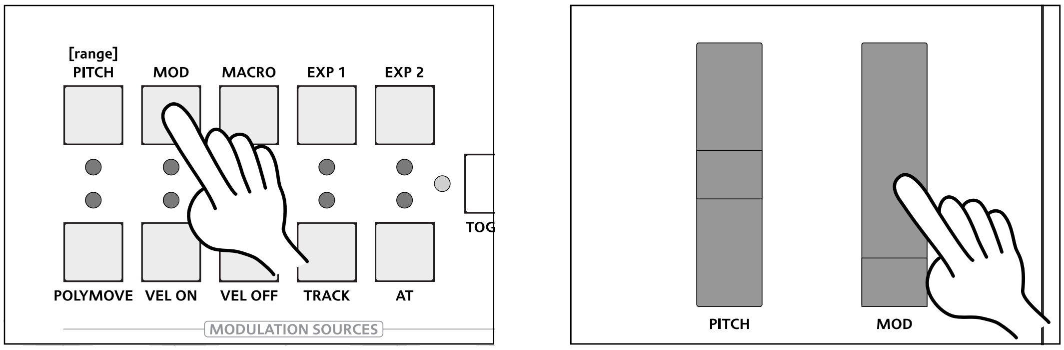


Figure 79: the Mod Wheel Assign button (left) and controller (right).

This tactile controller dates to the early days of monophonic synthesizers: since they were played with just one hand, it was possible to control their articulation with the other.

However, even in the present day when dozens of advanced tactile controllers are available, the mod wheel still holds its place on a keyboard’s interface.

The most common use for the mod wheel, though, is the vibrato depth. This is a case of “nested” modulation because the mod wheel ultimately controls an LFO modulation depth. On Magnolia, this feature can be achieved by modulating the LFO **AMP** parameter. Assigning it to the mixer controls can create a crossfade between Oscillator 1 at the bottom position, and Oscillator 2 at the top position.

Pitch Wheel

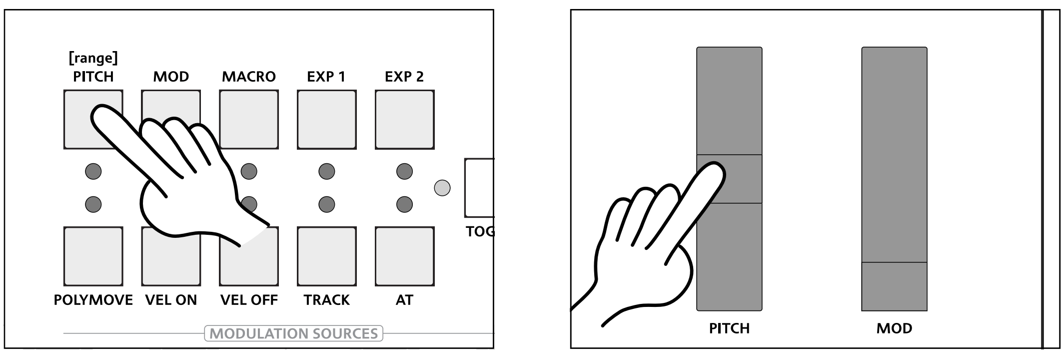


Figure 80: the Pitch Bend Assign button (left) and controller (right).

The Pitch Wheel is a spring-loaded controller that moves up or down and returns to its center position once released. It is generally used to perform pitch bends like on guitars or other fretted stringed instruments.

For this reason, it is the only controller that has a specific Assign procedure for pitch modulation.

Define the Pitch Bend Range

“Bending” a note means gradually shifting its pitch up or down by a desired interval. On the Magnolia, the default setting of most programs is a whole tone up and down, but it is possible to change it to any desired interval.

The procedure of defining the Pitch Wheel range uses the keyboard. It considers C 4 as the reference pitch, so the notes above it define the upward bending target note, and those below it the downward targets.

1. Push and hold the **PITCH** button under the Assign section.
2. Press any key above C 4 it to define the upward bending interval. (For example, C♯ 4 will be one semitone, G 4 a fifth, D 5 a ninth, and so on.)
3. Repeat with lower keys to define the downward bending interval.

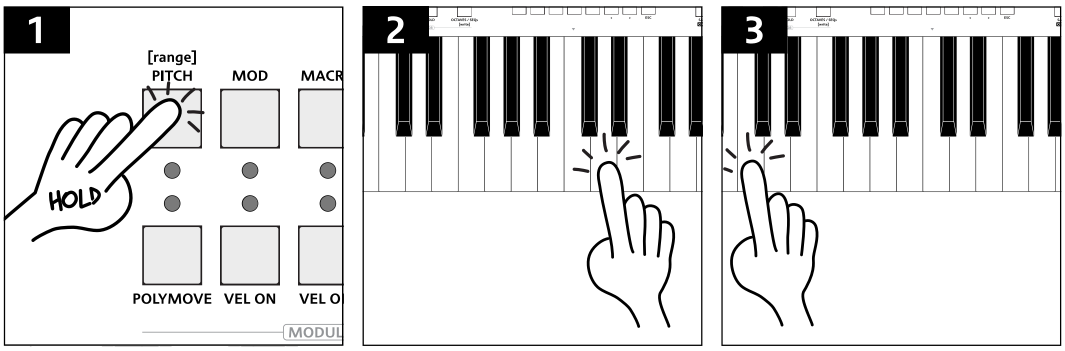


Figure 81: assigning the Pitch Bend range.

Let us say that we want to bend our note by a whole tone up and an octave down, to mimic the “dive bomb” technique of heavy metal guitarists. We will push the pitch wheel button, then the D 4 key (a whole tone higher than C 4), and then the C 3 key (an octave lower).

Press the C4 key two times to deactivate the Pitch Bend function. It is also possible to assign the Pitch Bend only in one direction: first deactivate it, and then define its range only in the direction you need.

> The Pitch Bend function affects both oscillators, but also both filters whenever they are set with tracking values other than 0.

Assign the Pitch Wheel to Other Targets

The Pitch Wheel can be assigned to other modulation targets as well, following the usual Assign procedure.

It is a bipolar controller, so when we assign it to a modulation destination, we are defining its positive behavior only: the negative effect will be mirrored. It is thus not possible to assign two different modulation ranges for the positive and negative sides of the pitch bend on any modulation target except the actual pitch.

However, it is still a valuable tool for expressive performances: for example, we can assign it to the cutoff frequency so that bending a note will also produce a change in timbre.

If we want, we can also deactivate its pitch bend function through the procedure described in the previous section, and then use it only as a modulation source.

Macro Control

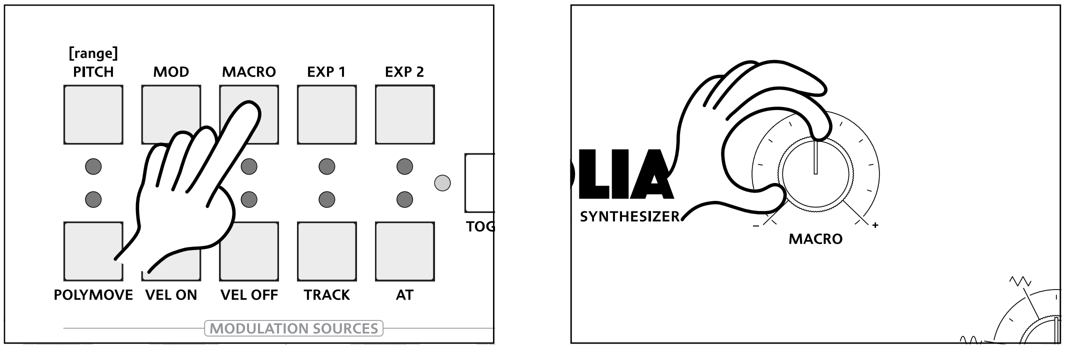


Figure 82: the Macro Assign button (left) and knob (right).

The Macro control is a bipolar knob that can be assigned to any available modulation destination.

Its function is similar to the mod wheel, but different grip, position, and resistance, combined with its bipolar nature, allows for different modulation styles.

Polymove

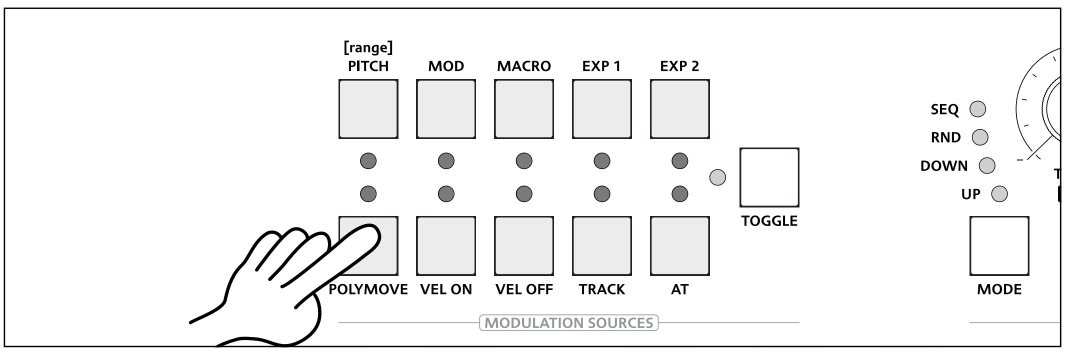


Figure 83: the Polymove Assign button.

The Polymove is a custom modulation source that expands the concept of randomization. Whenever you press a key, Magnolia generates a different random value per each voice and each destination it is assigned to.

For example, let us say that the Polymove is mapped to the low-pass filter cutoff and the wavefolder. If we strike a six-note chord, Magnolia will generate twelve unrelated random values, six for the filter, and six for the wavefolder.

But the Polymove also generates different values *per voice*, which means that, in a Unison mono lead with stacked voices, every voice will have its own set of random values.

For example, if we have a Unison lead with three voices and the Polymove is assigned to the Pan and Wave Shape parameters, when we strike a key Magnolia will generate six different values: two destinations for three voices.

Using it gently on many destinations allows for infinite subtle variations that can animate every patch. Using it heavily can create unpredictable patches where every key generates unexpected sounds.

Polymove in Mono and Legato Parts

The Mono and Legato settings affect the random values generation: when a Part is set to Mono, the Polymove generates new values at every note. When a Part is set to Legato, the Polymove will generate new values only if a key is pressed after all the others are lifted.

Expression Pedals

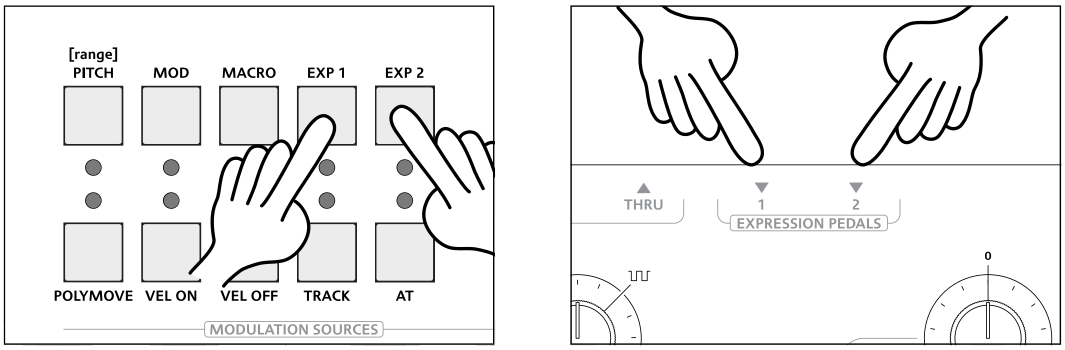


Figure 84: the Expression Pedals Assign buttons (left) and connections (right).

An expression pedal can be thought of as a foot-controlled modulation wheel. It provides a continuous unipolar modulation that can be held in place when lifting the foot from the controller.

Common uses for the foot control are volume, like an organ swell pedal, and filters.

Assigning it to the high-pass and low-pass in the same way will mimic a wah-wah pedal.

Modulation Destinations and Common Techniques

This section will contain a comprehensive list of all the 38 modulation destinations, including their peculiarities and their more conventional uses.

Oscillator Pitch

When we want to modulate our oscillator's pitch, the **OCTAVE** knob defines a continuous range of ±2 octaves. Common uses of pitch modulation are:

- Vibrato through an LFO.
- Adding a "detuned attack" like brass or wind instruments through an envelope.
- Sync Sweep, by modulating Oscillator 1 in **SYNC** mode.
- Manual "pitch bending" through the aftertouch.

Wave Shape

Modulating the wave shape is an excellent way to add motion to our timbre: since the Wave knob morphs between waveforms with progressively more harmonics, its result will resemble opening and closing a low-pass filter, but with a more exotic twist.

Harmonics (Oscillator 2)

The most common use of this parameter is to perform pulse-width modulation when the Oscillator 2 wave is a square wave. The most popular source is an LFO, which produces a "chorusy" effect similar to two detuned oscillators. Another nice PWM source is a gentle amount of Polymove, which can slightly alter the timbre of every note.

When the Wave knob is set to a sawtooth, the Harmonics control have a less conventional behavior, whose most appropriate modulation source is yet to be discovered!

TZFM (Oscillator 1)

We already discussed the importance of this parameter in its dedicated chapter (p. 28), to the point that it features its own pre-patched envelope. Other interesting modulation sources can be LFOs, velocity, and aftertouch, but a valuable one is the keyboard tracking.

In some patches where a static FM amount is desired uniformly across the whole keyboard range, controlling the parameter with a positive keyboard tracking allows the amount to grow proportionately to the pitch.

Fold (Oscillator 1)

Modulating the wave folder is an excellent way of defining the degree of distortion of a sound.

Most modulation sources work great with this parameter, but we would like to suggest the keyboard tracking with an inverted amount. It will cause the wave folder to be more pronounced in the lower range than in the higher range, thus resulting in growling basses and smooth leads.

Using a snappy envelope to rapidly open and close the Fold circuit can create sounds similar to Brenso's Ping circuit.

Mixer

The mixer's volume knobs are an excellent modulation targets for evolving sounds:

- Assigning a modulation source to both oscillators' volumes, but with opposite amounts, will crossfade between them.
- Assigning a snappy envelope or an LFO in envelope mode to the noise amplitude will add to the sound a noisy transient at the beginning, which can emulate the snare of a snare drum, or the breathy attack of a flute, according to the envelope shape.
- Since the mixer is also capable of saturating the filter, an LFO controlling all the sources at the same time will also change the filter drive, a sort of more colored tremolo effect than the standard one over the end-of-chain amplifier.

Cutoff

The filters' cutoff frequency is probably the most modulated parameter in the history of sound synthesis, and it is good with basically any modulation source.

The two filter configurations, though, allow for a more refined uses of the cutoff modulation:

- Applying the same source with opposite amounts to the high-pass and low-pass cutoff frequencies will cause the sound's timbre to expand in the high and low range simultaneously.
- One of the canonic filter modulation sources is the keyboard tracking: an amount of 1 will make the filter track in unison with the oscillators, thus guaranteeing a total timbre consistency across all the notes. Playing with other tracking curves will make the sound predictably darker or brighter across the instrument's extension.
- If the filter is self-oscillating, controlling it with a 100% tracking value will almost fake an additional sine wave oscillator in the mix.
- Assign a 100% keytrack to a low, self-oscillating high-pass filter to add a sub-oscillator.

Resonance

Modulating the resonance must be approached with care, otherwise it may cause some ear-piercing effects.

- A stepped random LFO can add to the sound a subtle rhythmic pulse.
- An inverted tracking curve can add some liquid brightness to the low notes while keeping the higher ones mellower.

Filter FM

If we use the filter FM as a distortion tool, we can mimic the Minimoog behavior by modulating its amount through the mod wheel.

Morph

The Morph knob, being a multiple modulation source itself, bears the most flamboyant results when modulated.

- A pedal modulation allows us to morph between sounds when playing with both hands.
- A dramatic keyboard the tracking can create a program that gradually mutates its sound from the lower register to the higher one with infinite shades in between.
- Velocity On or Aftertouch are capable of morphing between Part A and Part B independently per key.

The Morph parameter cannot be modulated by LFOs or Envelopes, since they are actual Part elements that can be affected by the Morph itself. The only available Modulations are the Program ones, see Table 2, p. 48.

Amplifier

The AMP control is permanently modulated by the Amplifier Envelope, so any modulation applied to it must take place while the envelope is active.

Common modulation sources for the Amplifier modulation are:

- LFOs for a tremolo effect (also with square waves).
- Velocity On, to establish a correlation between how hard you strike a key and how loud the sound will play.
- In Dual Mode, assigning any modulation with opposite amounts to the amplifier of part A and part B will create a crossfade effect.

Pan

Modulating the panorama causes the sound to move between the speakers. Interesting modulation sources can be:

- The LFOs, maybe with another LFO modulating its speed, see below on p. 65).
- The keyboard tracking, which pans the notes according to their pitch, much like a piano stereo microphone technique.
- The Polymove, which will randomize the notes' position in the stereo field. Subtle amounts are good for pads, and extreme settings are nice for ping-ponging fast arpeggiators.

Envelope Attack, Decay, and Release

Modulating the envelope times is a way of dynamically changing the character of a sound. For example, it can create some realistic effects like a palm-muted guitar, or more impossible ones like transforming a pad into a mallet sound.

- Controlling these times with the Velocity On and the Keyboard Tracking will create a sound that resonates more with strong dynamics on lower notes and less with a quiet dynamic or higher notes, much like a guitar string.
- Using the Velocity Off with a negative amount over the release time will create an expressive effect that shortens the decay time if we release the keys abruptly and lengthens it if we release them softly.

Envelope Amount over FM and Cutoff

Modulating a modulation amount might seem odd at first, but it is an advanced technique that can highly improve the expressiveness of a program. It is not like having a second modulation over the same parameter, because instead of being summed, now the modulations are multiplied!

For example, let us say that we have an FM piano sound with the TZFM envelope modulating the FM amount. If we want to make the piano timbre respond to the velocity, we cannot assign the Velocity On straight to the TZFM control, because it would change the default value on which the envelope will be applied, causing an unnatural behavior. Instead, we want to assign the Velocity On to the Envelope Amount, so that soft notes will have less articulation, and hard notes will have more, but all of them will eventually reach the level set by the knob.

In a similar fashion, if we want our FM amount to increase proportionately to the played note, for a more uniform sound across the keyboard, we can assign to the TZFM Envelope Amount a reasonable keyboard tracking modulation as well.

The Envelope amount can also benefit from a mildly inverted keyboard tracking, which will make a sound more evolving and resonating in the lower range and gradually dampen it in the higher range.

LFO Rate

Another beautiful case of nested modulation! Modulating the LFO rate is an excellent way to manually control its speed or add some randomization to an otherwise predictable modulation source.

- If we assign an LFO to the Pan control, we can then assign an expression pedal foot pedal to modulate its rate and create a very rudimentary rotary speaker speed control.

- Using a second LFO to modulate the one that we patched to a sound parameter will create subtle variations on pad sounds.
- Finally, an envelope with a fast attack, a slow decay, and a negative modulation amount will make the LFO start very slow and progressively speed up at every key strike, like if dropping a handful of marbles on the ground.

LFO Shape

Modulating the LFO shape is a rather unusual practice in the world of keyboard synthesizers, because most of the times the LFO shape is a discrete parameter. Since Magnolia allows to continuously blend between waveforms, it is possible to dynamically change this parameter through various modulation sources, for example:

- A second, slow LFO, for evolving and sustained pads.
- Velocity ON or Polymove, to randomly spice up an LFO.

LFO Amp

This parameter, much like the envelope amount, ultimately defines the impact of the LFO over a given destination. Being able to dynamically control it can create a variable modulation amount.

- Using the mod wheel on the LFO amp and assigning the LFO to the pitch allows you to introduce a vibrato effect, much like old school synthesizers.
- A second LFO can bring the modulation in and out.

Drive

The Drive control can be controlled via the modulation pedal to switch between a clean and a distorted sound much like our fellow guitarist friends do. A less conventional modulation could be a part’s LFO, which could create a distorted tremolo effect.

Post Drive

The most common modulation for this target is the same you assigned to the Drive control, but inverted: in this way, you’ll be able to distort a sound and at the same time keep its overall amplitude under control.

Effects Rate

Modulating the rate of a chorus will make it less predictable. Modulating the rate of a delay will have a much more dramatic effect and will create some weird, unpredictable detuning.

Delay 1 is very fast, and modulating its rate with fast tempos may create some nice Karplus-Strong vibes.

Effects Parameter

This parameter has different functions according to the selected effect: the choruses will have a more dramatic effect, and the Delays will increase the number of repetitions. Controlling it with a mod wheel or a pedal will allow you to dynamically control the spatial dimension of your sound.

Effects Dry/Wet

Being able to dynamically control the balance between the dry and the wet sounds is a very dramatic effect, which is useful for transitions or finales. Also in this case, an expression pedal or a mod wheel are nice, but for more frequent modulations of this kind you may want to try also the aftertouch.

Chapter 6: Arpeggiator

Arpeggios are a technique of articulating chords so that the notes play one after another instead of all at the same time.

The arpeggiator is thus a circuit that generates arpeggios out of the chords that are currently played on the keyboard. We can define how fast the arpeggio must go, which pattern should it follow, and on how many octaves we should spread it.

Part Settings

Magnolia has an arpeggiator for each part. The Tempo is common to both Parts, but every other parameter can be different. Feel free to experiment with different settings to achieve weird polyrhythmic parts.

The Arpeggiator works differently according to each program mode. When the program is in Single or Morph mode, the Arpeggiator is active on both parts, because one can morph between the two at any time.

When the program is in Dual or Split mode, the Arpeggiator can be active on Part A, B, or both. To activate the Arpeggiator on a Part, select it through the Part button and push the Arpeggiator's **ON** button. To activate the Arpeggiator on both parts you can either repeat the previous procedure, or you select both parts and then push the **ON** button. (If the Arpeggiator is active on a single part, the first push will activate it on both.)

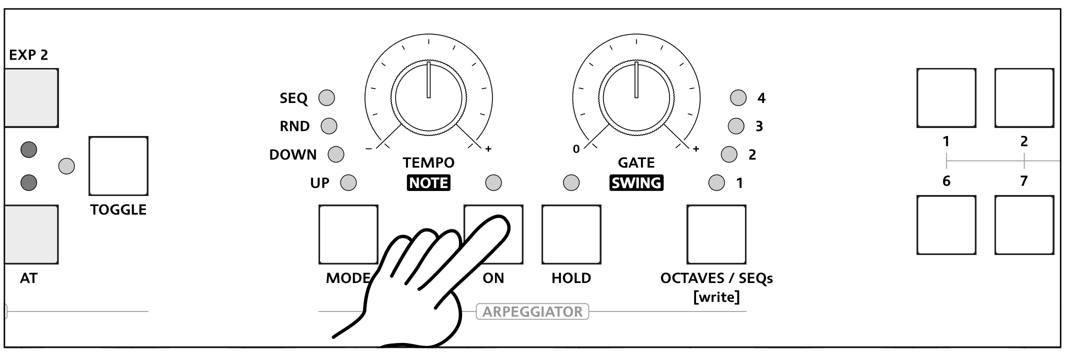


Figure 85: the Arpeggiator On button.

Tempo Settings

The arpeggiator plays its notes according to the clock settings that use the standard Western note duration system. Two parameters allow us to select the arpeggio speed: Tempo and Note Value.

Tempo

The arpeggiator can work with Magnolia's internal clock or with an external MIDI clock coming from the DAW, a sequencer, or another device capable of sending MIDI clock. The default behavior is the internal clock.

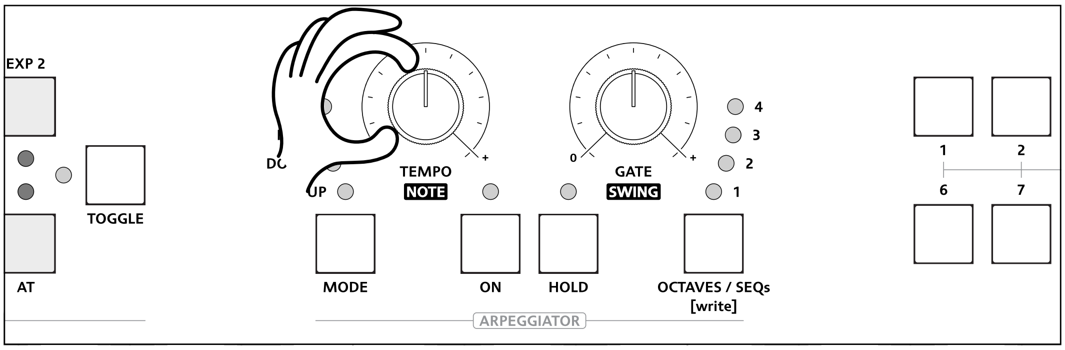


Figure 86: the Arpeggiator Tempo parameter.

The **TEMPO** knob defines the arpeggio's tempo, so it is expressed in BPM, from 30 to 300.

> If the LFO, the Delay, or the Chorus are synchronized to the Tempo, rotating the knob or changing the external MIDI clock rate will affect all of these parameters.

To use a MIDI clock, simply connect Magnolia to a MIDI device (like your DAW) through the MIDI in input: as soon as Magnolia will receive a MIDI clock, it will automatically follow it.

Note Value

This parameter defines the note value of the arpeggio in relation to the clock (either internal or external).

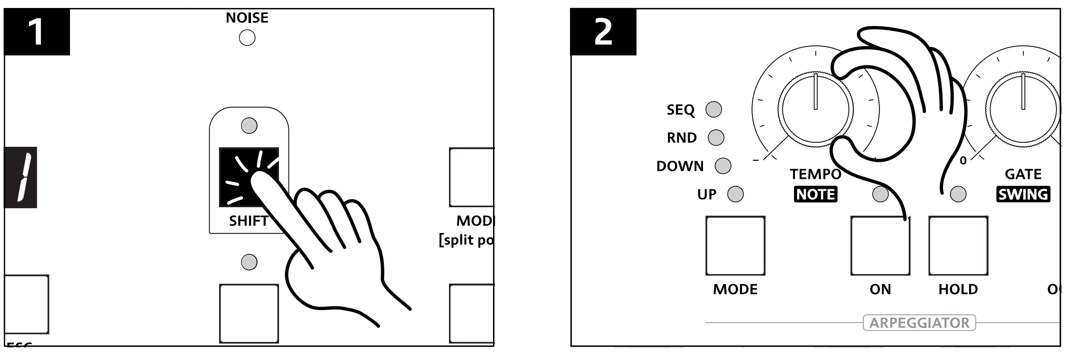


Figure 87: the Arpeggiator Note Value control, shifted function of the Tempo knob.

When rotating the **NOTE** knob, the display will automatically show the corresponding value. The table below sums up the display symbol and also their mathematical relationship to the internal clock.

Symbol	Note duration	Clock-to-note ratio
0.5	Double whole note (breve)	8
1	Whole note (semibreve)	4
2.	Dotted half note	3
2	Half note (minim)	2
2t	Half note triplet	1 ⅓
4.	Dotted quarter note	1 ½
4	Quarter note (crotchet)	1

4t	Quarter note (triplet)	⅔
8.	Dotted eighth note	¾
8	Eighth note (quaver)	½
8t	Eighth note triplet	⅓
16.	Dotted sixteenth note	⅜
16	Sixteenth note (semiquaver)	¼

Table 4: the Arpeggiator Note values.

Pattern Settings

Once we have selected how fast should the arpeggio be, we are left with something fun but a bit boring after a while. The pattern settings allow us to spice the things up a bit.

Mode

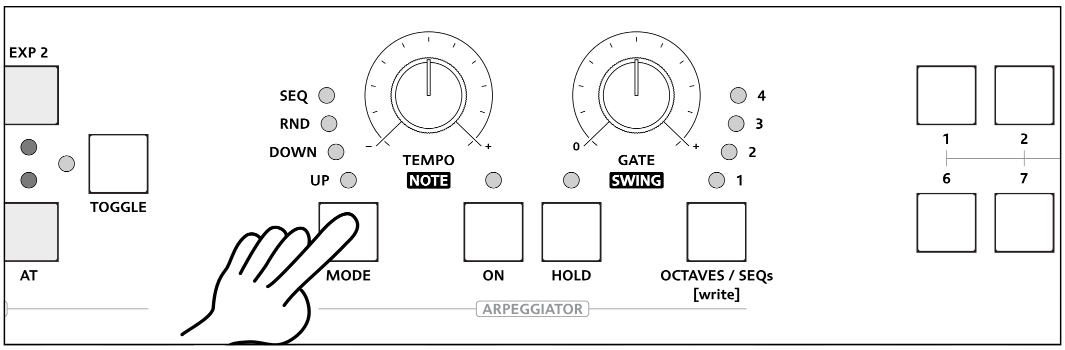


Figure 88: the Arpeggiator Mode selector.

This parameter defines the arpeggio direction, and it has four settings:

a. Up: the arpeggio starts from the lowest note and goes upwards.
b. Down: the arpeggio starts from the highest note and goes downwards.
c. Up and Down: the arpeggio starts from the lowest notes and goes upwards and downwards alternately.
d. Random: the arpeggio plays the notes randomly.

Figure 89: musical representation of the four arpeggio Modes: from top to bottom, Up, Down, Up and Down, Random. The first measure of each line shows the keys played on the keyboard, and the second shows the arpeggiator output.

Octave

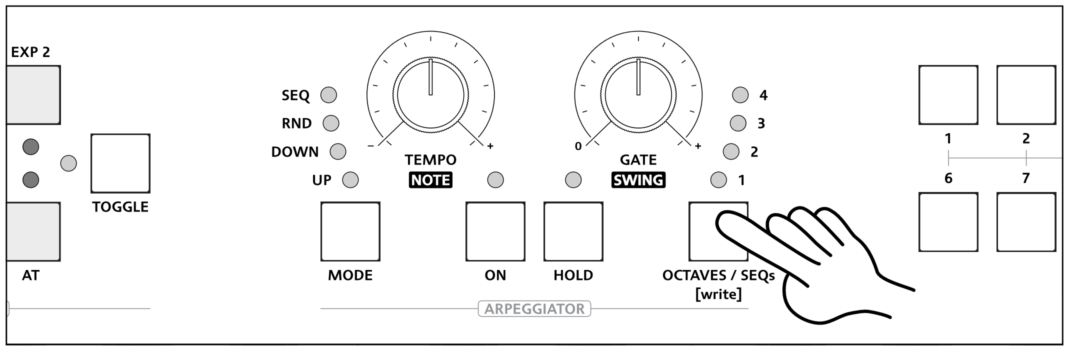


Figure 90: the Octave settings.

This parameter defines on how many octaves the arpeggiator will spread the notes. It has four settings:

a. 1: Only the actual notes played on the keyboard will arpeggiate.
b. 2: The arpeggio will be repeated over two octaves (one higher).
c. 3: The arpeggio will be repeated over three octaves (two higher).
d. 4: The arpeggio will be repeated over four octaves (three higher).

Combining the Direction and Octave parameters will create interesting effects. For example, Up and Down combined with a three-octave spread will create long slurs of notes going up and down. The Random setting, on the other hand, will displace the notes in random octaves for a melodic, yet unpredictable effect.

Figure 91: musical representation of an arpeggio in Down mode over three octaves. The first measure of each line shows the keys played on the keyboard, and the second shows the arpeggiator output.

Hold

The Hold button and the Sustain pedal (p. 18 above) will cause the Arpeggiator to continue playing even after you released the keys, with the usual distinction between the two controllers: the Hold button is toggled, and the Sustain pedal is momentary.

Programs in Dual or Split mode really call for a careful Hold and Sustain assignment, as we discussed in the Hold and Sustain section on p. 21).

Gate Length

When the arpeggiator is active, it does all the job, and we just have to select the note it must play, but this also means that we do not have control over the sustain time of the notes' envelopes: instead, they skip from the Decay straight to the Sustain stage.

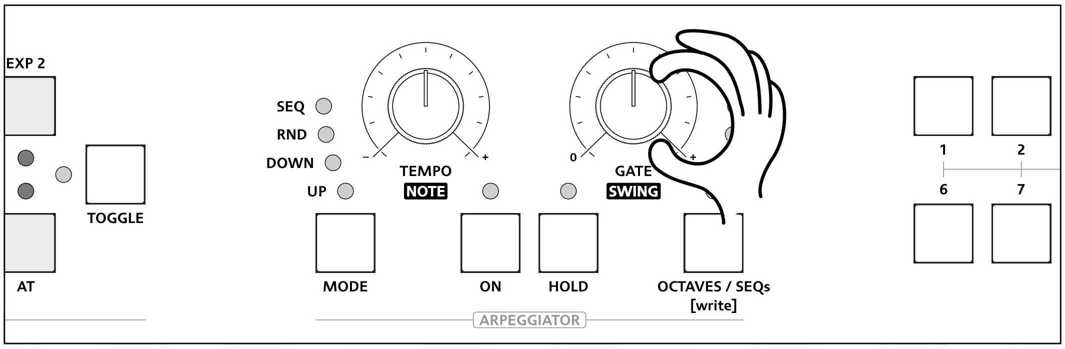


Figure 92: the Gate knob.

However, it is possible to define the Sustain time (or, how long the "imaginary keys" of the Arpeggiator must stay pressed) through the Gate Length control. Rotate it clockwise to increase the gate time, and counterclockwise to reduce it.

Swing

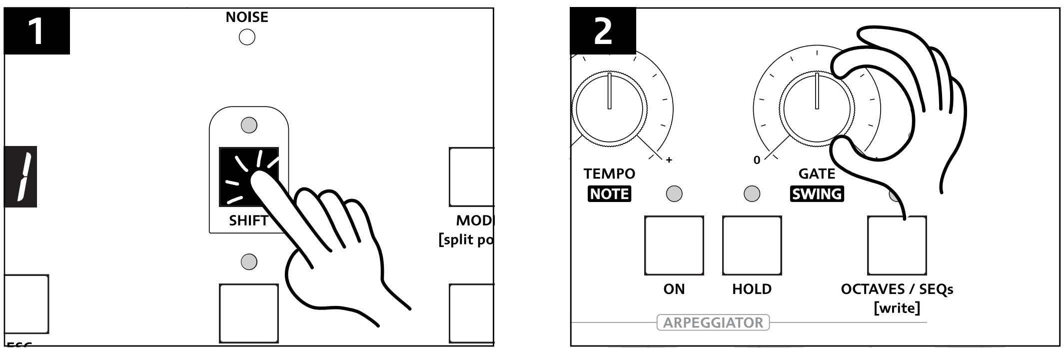


Figure 93: the Swing parameter, shifted function of the Gate knob.

Arpeggios and Polyphony

The Arpeggiator circuit plays one and note at a time, but it does not mean that we *hear* one note at a time. For example, if we play an arpeggio on a harp, as the name suggests, we can still hear all the notes that we played, albeit played in succession.

The same happens when an arpeggiated sound on the Magnolia has a release time longer than the space between two notes: we will hear them overlap, as in a chord. A natural consequence of this phenomenon is that if the release time is extremely long, more than the duration of eight notes, the ninth note will use the voice of the first one, thus creating the "note stealing" phenomenon discussed at the beginning of this manual.

If we plan to play a very staccato arpeggio, we may want to consider setting the part to Mono. For example, when we are playing a program that is set in Dual mode, where a part should play a pad sound and another an arpeggio, we may want to deviate from the classic "four and four" voice assignment (see above, p. 16) and assign to the arpeggiated part only one or two voices, thus having six or seven spare voices for the pad sound to make richer chords.

Sequencer Mode

When the Mode button is set to **SEQ**, the four Octave buttons become four sequence slot that can save sequences of up to 16 steps each.

The four sequences are independent per part and are saved and recalled in a Program.

By default, the sequences are empty. To write a sequence:

1. Select **SEQ** through the **MODE** button.
2. Select the sequence you want to edit through the **OCTAVE/SEQ** button.
3. Hold the **OCTAVE/SEQ** button (The corresponding LED will start blinking intermittently).
4. Write your sequence:
    a. To enter a note, simply play it on the keyboard.
    b. To enter a pause, use the **HOLD** button.

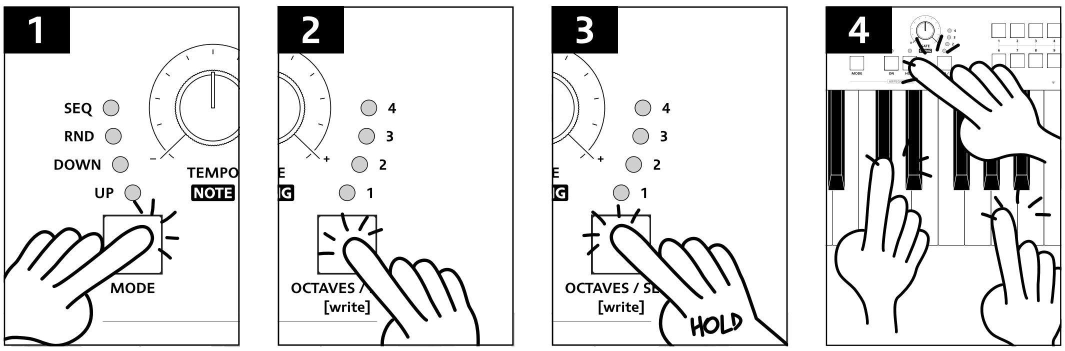


Figure 94: writing a sequence.

You can enter up to a total of 16 notes or pauses. If you exceed the limit, you will start writing a new sequence.

Chapter 7: Global Settings

This chapter discusses settings that affect the Magnolia as a whole and are thus not stored in a single Program. Accessing those settings has a very simple procedure:

1. Push and hold the Shift button.
2. Enter the number combination corresponding to the selected setting.
3. Use the arrow keys to navigate the options.
4. Push Save to store the setting.

Refer to the following table to the options currently available:

Number	Display	Options	Default	Description
1001	MIDI IN PT A	1-16	1	Defines the MIDI IN channel for Part A
1002	MIDI IN PT B	1-16	2	Defines the MIDI IN channel for Part B
1003	MIDI OUT PT A	1-16	1	Defines the MIDI OUT channel for Part A
1004	MIDI OUT PT B	1-16	2	Defines the MIDI OUT channel for Part B
1005	MIDI LOCAL	On, Off	On	Enables or disables the MIDI local controls.
1006	MIDI FWD	On, Off	Off	Forwards the MIDI IN signal to the MIDI OUT if their channel is the same.
2001	CALIBRATION	Yes, No	No	Performs a deep calibration routine that tunes oscillators and filters in a longer and more precise way than the autotune routine.
2002	REF PITCH	415-470	440	Defines the frequency of the A 4 in Hertz

To exit the menu without changing a setting, push **ESC**.

Appendix A: Cheat Sheet

Throughout the following cheat sheet, we followed these two typographic conventions:

- The *plus* sign + between two controls means: "push and hold the first button while interacting with the second control."
- The *comma* sign , means "push and release the first button, then interact with the second control."

Function	Button Combination	Page
Toggle between Programs and Lists	**< + >**	20
Init patch	**PANEL** (hold 3 sec.)	5
Panel mode	**INIT** (hold 3 sec.)	4
Select a part	**PART A/B**	12, 5
Define the polyphony	**POLYPHONY**	14
Voice assign	**PART A/B** + C 4 note (2-8 times)	16
Chord mode	**PART A/B** + Desired note (2-8 times)	16
Visualize a parameter's value without changing it	**VALUE** + Parameter	21
Visualize a parameter's value while changing it	**VALUE**, Parameter	21
Assign a modulation	Assign, Parameter	42
Remove one modulation destination	**TOGGLE**, Parameter, Assign	43
Remove one modulation source	**TOGGLE**, Assign, Parameter	43
Remove all modulation destinations	**TOGGLE** + Assign	43
Remove all modulation sources	**TOGGLE** + Parameter	43
Monitor active modulation destinations	Assign	45
Monitor active modulation sources	**TOGGLE**, Parameter	45
Visualize the modulation amount without changing it	Assign, **VALUE** + Parameter	45
Save a program	**SAVE**, (select bank/group/program,) **SAVE**	9
Compare an edited program	**SAVE**	10
Select a knob's secondary parameter	**SHIFT**	1
Define the Split point	**MODE** + Desired note	14

Table 5: Cheat Sheet.

Appendix B: Sound Synthesis Core Concepts

To hear music, we need some vibrating air, and to make the air vibrate, we need a vibrating body: we can pluck a string, hit two wooden sticks, blow into a pipe, and so on. In the case of electronic instruments, our vibrating body is the loudspeaker (or the headphones).

In the specific case of analog synthesizers, everything, from the sound generation to its articulation to the final audio output through the speaker cone's vibration, is achieved through voltages that change over time.

With our synthesizer, we create a complex, electric signal that makes our cones move back and forth. The variation of voltage over time translates into the extremely complex waveform that we call music. It goes without saying that the more control we have over our voltage, the more expressive and articulated will become our composition's waveform.

Pitch, Timbre, and Overtones

The way a body vibrates generates many pattens of motion at the same time. These patterns are often locked into mathematical relationships. When these patterns set the surrounding air into motion, they make it vibrate in a way that we identify through pitch and timbre. The major vibration is what determines the pitch, and the other smaller vibrations characterize its timbre.

Those smaller frequencies are called *overtones* and are part of every sound except the sine waves, which only have a single overtone–the fundamental. Every other sound has a much more complex spectrum, which translates to a richer timbre.

We tend to distinguish sound timbres as harmonic and inharmonic. This distinction may sound subjective and based solely on taste, with harmonic sounds being "pleasant" and inharmonic "unpleasant", but it is in fact very scientific. The overtones of a harmonic sound are integer multiples of the fundamental frequency, so its multiples by 2, 3 4, 5, 6, 7, and so on. They are all nice and easy round numbers: they are thus called harmonics, and they form something called the "harmonic series". An inharmonic sound, on the other hand, has overtones that aren't integer multiples of the fundamental frequency, and they create an aperiodic waveform, or a waveform whose overtones don't end and begin at the same time. Inharmonic overtones are often called *partials* and can be found in sounds like percussions or bells. The most inharmonic sound is the noise, which is packed with overtones all over the place. It virtually contains every frequency, and so it is so aperiodic that it does not even have a waveform, and thus a discernible pitch.

In the real world there is a blurred line between harmonic and inharmonic sounds, and even the most well-tuned instruments have a degree of inharmonic overtones introduced by their resonant bodies which contribute to their overall timbre. In the electronic world, however, it is possible to create artificial waveforms composed exclusively by harmonic overtones, on which see the Oscillators section below.

Voltage: Audio and Modulation

In analog modular synthesizers, we use voltage both to generate sound and to control it. If a circuit generates a voltage that varies from, say, -5 V to 5 V, 110 times a second, it will make our loudspeaker vibrate as many times, thus producing a sound whose frequency is 110 Hertz (Hz), corresponding to the note A. However, if the same circuit generates a voltage oscillation of 10 Hz, meaning that the voltage varies from -5 to 5 V ten times a second, we wouldn't be able to hear it anymore. It is because its frequency is below our audible range of human beings, which spans from 20 to 20.000Hz.

This brings in a conventional distinction between voltages that we use for sound generation, which provide alternate current in the audio range, and voltages that we use for sound modulation or control, which are often called control voltages, or CV. Generally speaking, on a modular synthesizer we use CV to turn our synthesizer's knobs for us, in a sort of automated way. On Magnolia, those control signals are digitally created by the microprocessor, while the sound voltages are created by the eight analog voice cards.

Sound Generators: Oscillators and Noise

The circuits that generate "fast" alternating voltages, those for sound generation purposes, are called oscillators, because their output is an oscillating voltage. The oscillation is regular, and it is measured in Hertz (Hz), just like the sound wave that will come out of the loudspeakers. In a visual representation of a generic oscillator's output, the voltage will fluctuate up and down, and its path will determine the sound's waveform, with a corresponding effect over the timbre that we will perceive once we will listen to it.

The conventional analog waveforms are three: triangle, square, and sawtooth, and if you will look at them, you will see that they have one thing in common: they consist only of straight lines, either diagonal, horizontal, or vertical.

The timbral difference between those waveforms is very pronounced and depends on their harmonic content (see the section on timbre at page 79 above).

The triangle wave, in its mathematical abstraction, only contains the odd harmonics, so the fundamental, the third harmonic, the fifth, and so on. The amplitude of the higher harmonics is quadratic, according to their harmonic number. So for example, if the fundamental has an amplitude of 1, the third harmonic would have an amplitude of $1/3^2$ so 1/9, the fifth $1/5^2$ so 1/25, and so on.

The square wave has the same harmonics of the triangle wave, but with different intensity: their amplitude decreases in a linear fashion, so 1/3, 1/5, 1/7, and so on.

On many synthesizers, it is possible to vary the symmetry of the square wave, making its upper and lower sides asymmetrical. In such cases, we no longer call it square: instead, it is a pulse wave. A dynamic variation of a pulse wave symmetry is called pulse width modulation, abbreviated to PWM.

By changing the symmetry we also change the proportion and the amplitude of its overtones. As a rule of thumb, we can say that the fraction of the duty cycle determines which overtones are missing from the final wave. For example, our square wave has a 50% duty cycle, which is 1/2 up and 1/2 down. So every second harmonic is missing from the spectrum, and that is what we already knew. If our pulse wave is 1/3 positive, every third harmonic will be missing. If it is 1/4, every fourth, and so on. What if the positive side is higher than the negative one? Well, it is more or less the same, but with inverted phase.

And finally, the sawtooth is the richest waveform available, and contains all the harmonics, still in a linear proportion.

These are the easiest waveforms that we can create in the analog domain. They sound pretty boring per se, but the square and the sawtooth are so rich in harmonics that can often be made more interesting by removing some of them through a particular circuit called filter. This technique is called *subtractive synthesis* because it creates new timbres by subtracting harmonic content.

The final waveform that we must know is the sine wave, which is the simplest one: so simple, in fact, that it only has one harmonic, the fundamental. It is extremely dull per se and, since it does not have any harmonic material to filter, it is not particularly useful for subtractive synthesis. However, it is perfect for other techniques, like additive synthesis, which is beyond the scope of this manual, and FM, which is actually Magnolia's key features.

Filters and Subtractive Synthesis

A filter is a circuit that removes a certain spectral portion from the sound that passes through it. A low-pass filter removes the higher frequencies (and lets the lower ones pass through, hence the name); a high-pass filter removes the low frequencies; a band-pass filter removes both and lets only a middle section pass through. A common band-pass filter design is a high-pass filter and a low-pass filter in series.

Ideal and Realistic Filters

An ideal filter should remove any sound above or below a certain frequency, which is called cutoff frequency. However, in the real world, this is not possible, and many frequencies love to sneak out of the filter even if they are past the cutoff threshold. They are progressively quieter as they move away from the cutoff frequency until finally there are no more of them.

The spectrum's portion between the cutoff frequency and the actual silence is the filter *slope*, and it can be wider or thinner according to the filter design. The width of this filter slope is measured in dB per octave, which indicates how quiet a frequency is compared to its lower octave when both are past the cutoff frequency. For example, if we have a low-pass filter with a 6 dB/oct slope and we set the cutoff frequency at 440 Hz, the frequency of 880 Hz, which is an octave higher, will be 6 dB lower, so half of its amplitude; the frequency of 1760 Hz will be 12 dB lower, and so on until the attenuation is so high that we can no longer perceive that frequency: at that point, we can finally say that the frequency is filtered out. However, 6 dB/oct is quite a generous filter because it lets a lot of sound pass through. It is the kind of filter that you may find on hi-fi equipment since it can perform some degree of "correction" without altering too much the nature of the sound. Other more common filter designs for musical purposes have a 12 dB/oct, 18 dB/Oct, and 24 dB/oct slope, which create a progressively more dramatic filter effect.

However, even a perfect filter slope after the cutoff frequency is impossible in the real world. Every filter design has a zone around the cutoff frequency where the amplitude of every frequency is no longer "full," but neither attenuated with a proper slope yet. This sort of "transition zone" also affects the frequencies below the cutoff frequency, and they might get progressively duller as they approach it.

Filter Tracking

We have said that a filter progressively attenuates all the frequencies past a certain threshold until complete silence. Having a fixed cutoff frequency means that the notes we play with a filtered oscillator will get progressively duller and quieter as we approach and cross the cutoff frequency.

Let us take for example a low-pass filter with a 24 dB/oct slope. We might find a nice filter tone for an A at 110 Hz, but the effects over an A at 220 Hz will be greater, and even more at 440 Hz. The three notes will have a completely different harmonic "color." By making the filter track with the oscillator, instead, we progressively shift the cutoff frequency according to the oscillator's pitch, thus making all the notes have proportionally the same harmonic content.

Beyond Filtering: Q

When used for musical purposes, filters can go beyond removing frequencies: lab precision is not a priority, and musical filters might even add some extra timbre to the sound, as long as it is "pleasant." There are many filter designs, and they differ from one another by the extra flavor they add to the source signal.

One of the most common "additions" is the Q control, also called *emphasis*, *resonance*, *feedback*, and other terms. The resonance emphasizes the cutoff frequency, often by a re-injection of the filtered signal to its input. Since it involves a feedback design, the cutoff frequency can be emphasized so much that it starts oscillating with extreme Q settings. This produces a sine wave whose frequency is the same as the cutoff frequency. When the circuit is filtering an external signal, the new sine is summed to it; however, many filters can also oscillate by themselves, so they can output one of the purest sine waves of the analog domain with no input. If such a filter's cutoff frequency can be controlled with a V/oct signal, a resonant filter can become an actual sine wave oscillator.

Being a filter an electronic circuit, it is subject to the structural limits of its components. It has a certain headroom to accommodate the incoming sound, after which it will saturate or distort the signal. Sometimes this can be a desired effect, and many filters allow you to boost the signal input way past its headroom to saturate the circuit.

FM Synthesis

When an oscillator's frequency is modulated at a sub-audio rate, this generates noticeable fluctuations of the pitch, similar to a vibrato effect. However, when the modulating signal runs at audio rate, the human ear can no longer perceive the vibrato fluctuations, and instead perceives a new timbre.

The result of audio-rate frequency modulation (FM) is a more complex sound whose timbre is a result of the interaction of the two frequencies (that of the oscillator being modulated, which is usually called *carrier*, and that of the *modulator*). The interaction between carrier and modulator generates new frequencies that enrich the sound spectrum, which are called *sidebands*.

Depending on how the modulation is applied to the carrier signal, FM can be exponential or linear. Exponential FM modulates the carrier based on its pitch, i.e., with intervals: a symmetric bipolar signal will thus increase and decrease the carrier frequency by the same interval (for example one octave), according to the modulation amount.

Linear FM, on the other hand, modulates the carrier based on the frequency: the modulator increases and decreases the carrier frequency by the same Hz value, according to the modulation amount.

Exponential FM

In the early days of analog synthesis, frequency modulation was mainly exponential due to the oscillator's design. It was possible to have an "audio-rate vibrato," which created some interesting, metallic timbres, but it had a downside: it was not possible to play melodies.

This happens because the exponential modulation is asymmetrical: if an A=440 Hz waveform is modulated exponentially, and the modulation amount is +/–1 octave, the carrier frequency will oscillate between 220 Hz and 880 Hz, which is 220 Hz below and 440 Hz above the original frequency. Such modulation also causes a shift of the central frequency: in this case, it will be 550 Hz, which is exactly 330 Hz above 220 Hz and below 880 Hz.

This, in turn, will generate a perceived detune of the original pitch, which will be different every time the carrier frequency changes. For example, if we modulate a B4 (493.88 Hz) by +/–1 octave, the perceived pitch will be between 246.94 and 987.77 Hz, so 617.355 Hz, and so on.

Exponential FM generates sidebands that are not equally spaced above and below the carrier frequency, which makes them way harder to calculate, compared to AM and RM sidebands.

Linear FM

The downsides that exponential FM shows in certain applications can be easily overcome with linear FM, which brings into the world of audio synthesis some well-known concepts of electronic communications thanks to the research of John Chowning.

The concept is the same as with Exponential FM: the only difference is how the modulation is applied to the carrier. Instead of shifting its frequency up and down by musical interval, here we use the frequency: our A4 (440 Hz) can now shift up and down by the same amount, like +/–100, 200, 800 Hz. If we used a very slow modulator to create a "linear vibrato", the effect would be odd, as the perceived pitch modulation will be lower going up, and higher going down.

The first consequence of this technique is that we can retain all the original pitch information, without any drift.

The second consequence is that we can have much more harmonic and predictable sidebands since they are equally spaced above and below the carrier frequency.

The number and amplitude of sidebands are proportional to the amount of modulation that is applied to the carrier, which is often called 'deviation': this value defines the difference between the carrier's frequency and the higher or lower frequency that it reaches when modulated. The more deviation, the wider will be the fluctuations of the carrier frequency, and the greater the number of sidebands.

The relation between the deviation and the modulator's frequency, both expressed in Hz, defines the FM Index. (For example, if the modulator's frequency is 200 Hz and the deviation is 400 Hz, the FM index would be 400/200=2). Magnolia, just like the Brenso, has a control for the FM deviation, rather than the index.

In linear FM, the sidebands are the sum and difference of the carrier and integer multiples of the modulator. There can be some cases, however, in which the difference between the carrier and the modulator would provide a negative number, which gives us a “negative frequency”: for example, if the carrier frequency is 150 Hz and the modulator frequency is 200 Hz, the first couple of sidebands would be at 350 Hz and −50 Hz, which is 50 Hz with inverted phase.

In digital synthesis, this is no big deal. However, a conventional analog oscillator stops oscillating whenever it reaches 0 Hz, thus losing these negative sidebands.

For this purpose, new analog oscillators capable of not stalling at 0 Hz were developed, including our Brenso. This technique is called Thru-Zero FM because the oscillator goes “through 0 Hz:” this term, however, is meaningful in the analog domain only, since in digital FM the negative sidebands were always there. As Chris Meyer put it, ‘Through Zero FM is linear FM that just works.’

FM in the analog domain is often an approximate process, because of the difficulty for the analog components to guarantee a precise ratio between carrier and modulator frequencies.

Indices

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