SUPER 6

OWNER'S MANUAL
VERSION [1.4] FEBRUARY 2024

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12 VOICE POLYPHONIC BINAURAL ANALOG-HYBRID
SYNTHESIZER WITH SUPER-WAVE TECHNOLOGY

SUPER 6

UDO SUPER 6 — OWNER'S MANUAL

VERSION 1.4 · FEBRUARY 2024

SUPPORT & DOWNLOADS:
UDO-AUDIO.COM

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⚠ IMPORTANT SAFETY INSTRUCTIONS ⚠

THE FOLLOWING SAFETY REQUIREMENTS MUST BE ADHERED TO FOR PREVENTION OF FIRE, ELECTRIC SHOCK OR INJURY:

1. Read all the instructions before using the musical instrument.

2. Do not disassemble or modify the musical instrument.

3. Never attempt to repair this device or replace parts. If repair or part replacement should become necessary, you must contact your dealer. There are no user-serviceable parts inside the musical instrument.

4. Never place the musical instrument in an unstable location. A musical instrument set may fall, causing serious personal injury. Many injuries, particularly to children, can be avoided by taking simple precautions such as:
   - Only using cabinets or stands that can safely support the musical instrument and have an adequate load rating
   - Ensuring that the musical instrument is level and stable before use
   - Ensuring the musical instrument is not overhanging the edge of supporting furniture, which could cause the musical instrument to topple
   - Not placing the musical instrument on tall furniture (for example, cupboards or bookcases) without anchoring both the furniture and the musical instrument to a suitable support
   - Not placing the musical instrument on cloth or other materials that may be located between the musical instrument and supporting furniture or stand
   - Educating children about the dangers of climbing on furniture to reach the musical instrument

5. Do not use or store the musical instrument in the following types of locations:
   - Locations exposed to rain
   - Locations of excessive dust
   - Locations subject to heavy vibration
   - Locations of extremely high temperature (such as in direct sunlight, near heating equipment, or on a device that generates heat, or near naked flames or candles)
   - Near moisture (such as in a bathroom, near a sink, or on a wet floor) or in locations of high humidity

6. Do not stand on the musical instrument, or place heavy objects on it.

7. Do not drop the musical instrument.

8. The musical instrument should only be powered from an electrical outlet which provides a voltage within the ratings of the instrument and provides 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.

9. Only use the power cord included with the device. Do not attempt to modify or disassemble the power cord. If replacing the fuse in the power cord, always replace it with a fuse of the same type.

10. Do not place heavy or sharp objects on the power cord, as this could damage the power cord and render it unsafe. 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 your dealer.

11. Do not place any containers which contain liquids on or near the musical instrument.

12. 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 your dealer.

13. Do not use the musical instrument, disconnect from the electrical supply and contact your dealer if any other serious malfunction is suspected, for example by:
    - The musical instrument becoming wet (by rain, etc.)
    - The musical instrument becoming hot
    - Generation of smoke or an unusual smell
    - Repeated abnormal behaviour
    - Visible damage to the enclosure, for example large dents or holes in the enclosure
14. If the musical instrument is to be used by children, the children must always be supervised by an adult.
15. Ensure that the connected cables are organised and managed in a safe manner, and do not cause an electrical or trip hazard.
16. When you need to transport the musical instrument, package it in the box (including padding) that it came in, otherwise damage during transport could occur.
17. Unplug the power supply from the outlet when left unused for long periods of time or during lightning storms.

Electrical Specifications

Rated input voltage:	90~240VAC
Rated input frequency:	47-63Hz
Power consumption:	50W
Fuse type:	2A T-type

Note

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.

Caution

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 UDO Audio Ltd. could void the user's authority granted by the FCC to operate the equipment.

ACKNOWLEDGEMENTS

Nick Batt, Jasmine Butt, Ben Crosland, Axel Hartmann, Mike Hiegemann, Alexandros Liarokapis, Kester Limb, Dan Parks, Frank Rüffel, Chloe Smith, Ayumu Suzuki and Gaz Williams.

The UDO Team

Anthony Gillan, George Hearn, Magnus Hearn, Levi Morgans, Callum Mulholland, Will Plowman, Leonid Uljanov and Meggie Wood.

Manual revision and illustrations by Mike Hiegemann.

INTRODUCTION

Hello and welcome to your new UDO synthesizer! I'm honoured to have been able to craft this instrument and place it in your hands. Synthesizers have been a love and passion in my life since a young age. It has always been a dream of mine to produce an instrument like the Super 6 and it has been realised by an international collaboration of talented musicians, engineers, designers and people like yourself who have supported this venture.

The Super 6 is a result of years of hard work and many iterations of development. My mission with the Super 6 has been to harmonise what I love about archetypal electronic instrument design with modern, novel synthesis technologies that excel at generating spatially dynamic results.

The architecture leverages the vibrancy of a true-stereo analog signal path, driving it with extremely high sample rate, spectrally versatile, digital audio and presents you with straightforward, expressive controls of superior mechanical build quality.

I would love to see this instrument with the wear and tear of many years of use. Do not be afraid to use it for what it was made for. Experiment, play, take it with you, learn it and hopefully love it like we do.

UDO are dedicated to making powerful and accessible musical instruments, and we hope you'll take much simple joy from the Super 6. We have brought it to you, and now the most significant part of the journey is in your hands.

George Hearn,
Director UDO Audio Ltd

The UDO Super 6 is a 12-voice polyphonic synthesizer based on a hybrid of analog and digital technologies. By combining the aesthetics and sonic character of vintage-era classics with state-of-the-art synthesis technology, it was designed to be flexible, powerful and, above all, *immediate* – providing you with gorgeous sound!

DDS Oscillators

Direct Digital Synthesis is the signal generation method employed by both oscillator cores of the Super 6. At its centre is a clock signal running three orders of magnitude higher than typical audio sample rates. This clock increments a counter through thousands of indices in your chosen waveform, generating samples once every 20-billionths of a second and interpolating between them.

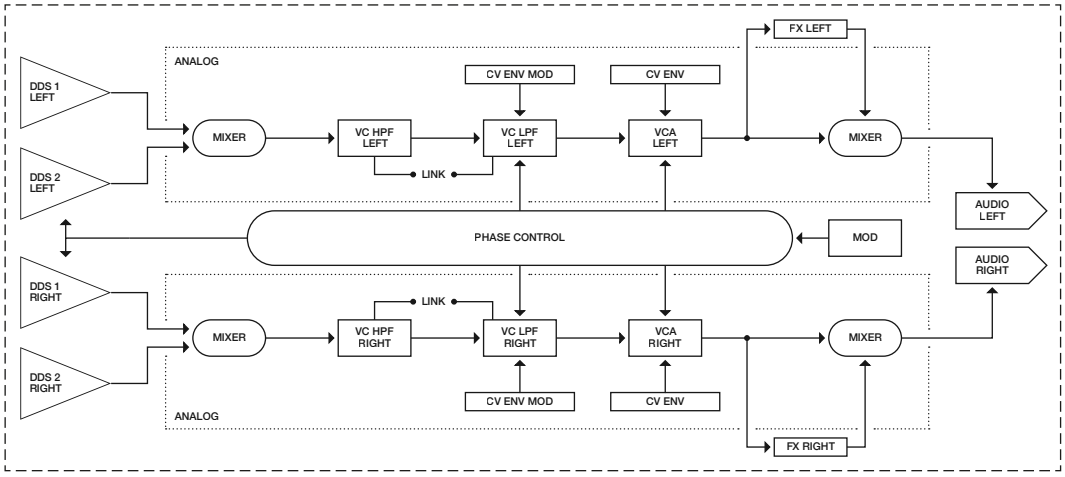
Each numerically controlled oscillator then uses its own DAC, also running at the same high sample rate, to convert the samples to analog voltages before being filtered by a preliminary analog low-pass filtering stage.

The extremely high sample-rate to output-frequency ratio provides DDS oscillators with the advantage of superior phase precision and natural-sounding frequency modulation. Importantly, it also allows us to avoid using the severe band-limiting, or “anti-aliasing”, of typical lower-frequency digital methods. This means our oscillators are easily capable of generating frequency content far above the limits of the human auditory system, as is the norm with analog oscillator synthesis.

What is Binaural Synthesis?

In binaural mode, the Super 6 features a true stereo signal path in which its twelve voices are twinned to form six stereo ‘Super’ voices. Consequently, the left and right channels (and your ears) are each assigned a complete synthesizer voice.

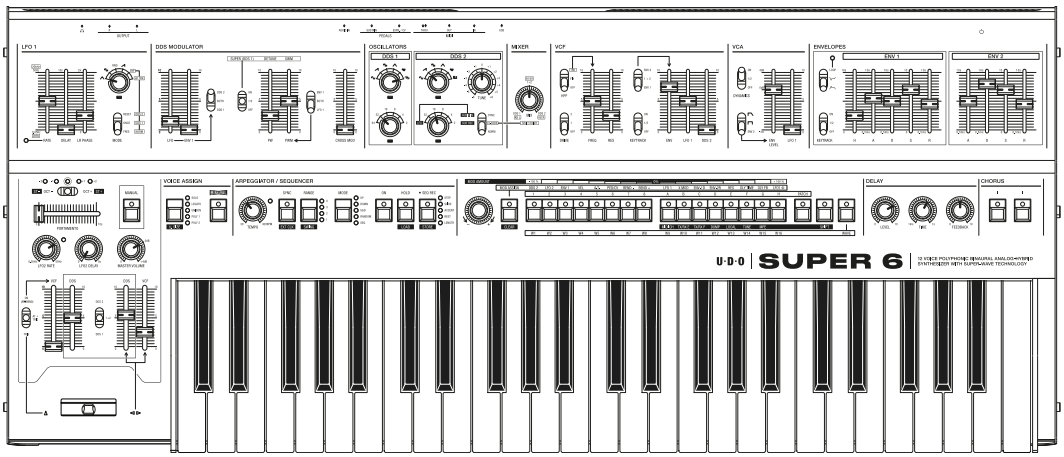
Starting with the stereo oscillators, parameters of both channels of each ‘super voice’ may also be independently controlled, enabling you to create gorgeous stereo images. The effect on the sound ranges from subtle to extreme stereo movement, resulting in an enhanced sense of spatial positioning compared to conventional monaural signal chains.



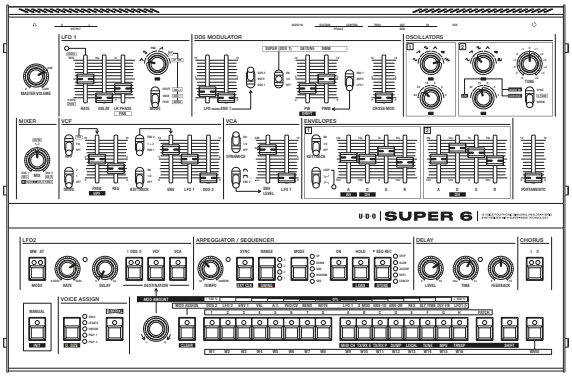
The signal path of the Super 6.

QUICK START

The Super 6 was designed for hands-on performance and experimentation. Not only do we think it is a great sounding instrument, but it also provides you with a pleasing immediacy when interacting with the sound. All of the Super 6's primary controls are accessible directly from the front panel, making it an incredibly intuitive instrument that will spark your creativity in the studio and on stage.



The front panel of the keyboard model.



The front panel of the desktop model.

Exploration and experimentation are downright encouraged, so feel free to dive straight in and start creating your sounds. The best way to learn about how the Super 6 works is to get involved! You can always come back later and read more about each of the Super 6's sound shaping tools in the subsequent paragraphs of this manual. We hope you enjoy playing and tweaking the Super 6 as much as we do!

Setting up Your Super 6

Follow the steps below to setup your brand-new synthesizer:

1. Plug the power cable into the power connector on the rear panel of the Super 6.
2. Use unbalanced 1/4-inch audio cables to connect the outputs labelled **OUTPUT L/R** to your mixer or audio interface, or connect headphones to the headphone output on the rear panel.
3. Turn on the Super 6.
4. Set the **MASTER VOLUME** control to about 0 dB.
5. Play some notes or chords and adjust the levels on your mixer or audio interface.

See pages 24-27 for a full overview of the Super 6's connections.

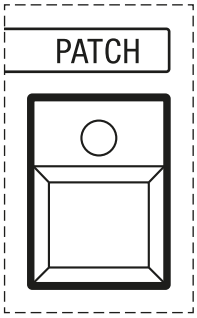
Patches

A patch is a stored set of parameters that determine the characteristics of a sound.

A total of 128 patches are accessible from the front panel. They are organised in 16 banks (**A1**-**H2**) featuring 8 patches each. You can edit these or use the dedicated memory slots to store your own patches.

Loading a Patch

First of all, make sure you are in patch mode. Patch mode is accessed by pressing the button labelled **PATCH**.

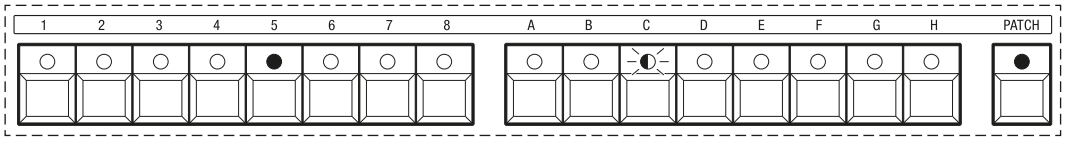


The patch mode button.

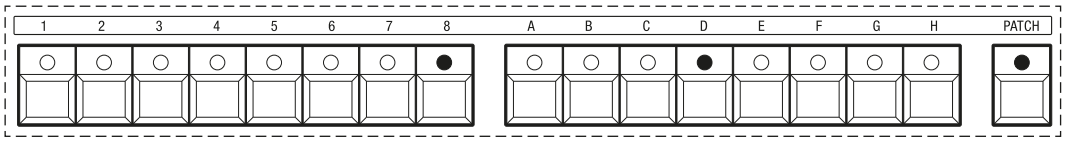
In patch mode, each lettered button (**A**–**H**) allows you to select two banks: **A1** and **A2**, **B1** and **B2**, etc. A lit LED indicates one of the first banks (**A1**-**H1**) is selected, while a flashing LED indicates one of the second banks (**A2**-**H2**) is selected. To toggle between banks with the same letter, simply press the corresponding button repeatedly.

Finally, pressing a numbered button (**1**-**8**) loads one of eight patches within a selected bank.

The LEDs of buttons **1**-**8**, and **A**-**H** indicate which patch is loaded. In the first example, patch **5** from bank **C2** s loaded:



In the second example, patch **8** from bank **D1** is loaded:



To load a different patch from the same bank:

- Press one of the seven numbered buttons (**1**-**8**) that are currently unlit. Its LED will then light up.

If the LED of one of the select buttons does not light up, this indicates that the corresponding memory location is empty.

To load a patch from a different first bank:

1. Press one of the seven lettered buttons (**A**-**H**) that are currently unlit. Its LED will light up, indicating that one of the first banks (**A1**-**H1**) has been selected.

2. Press one of the numbered buttons (**1**-**8**). Its LED will then light up.

Changing the bank does not load a new patch. A new patch is loaded only by pressing one of the numbered buttons after a bank has been selected.

To load a patch from a second bank:

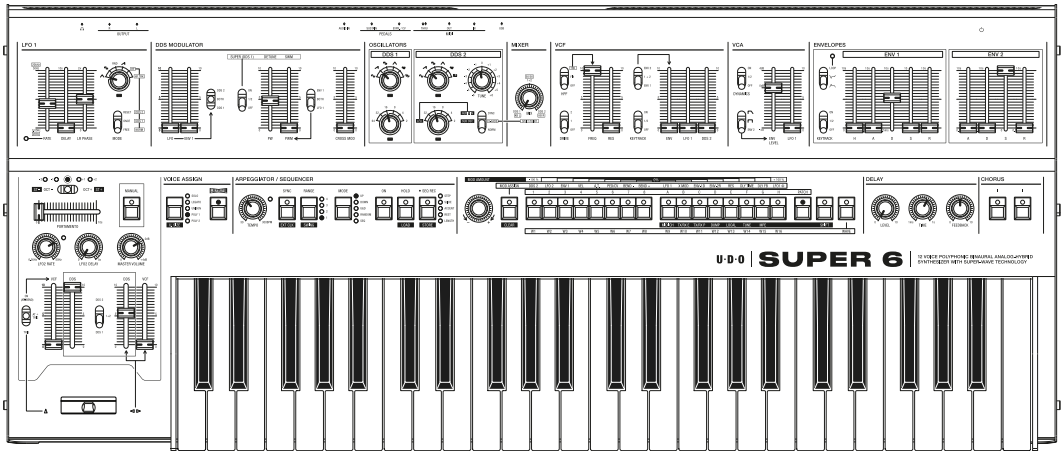
1. Press either the currently lit lettered button again or one of the other lettered buttons (**A**-**H**) twice. Its LED will start flashing, indicating that one of the second banks (**A2**-**H2**) has been selected.

2. Press one of the numbered buttons (**1**-**8**). Its LED will then light up.

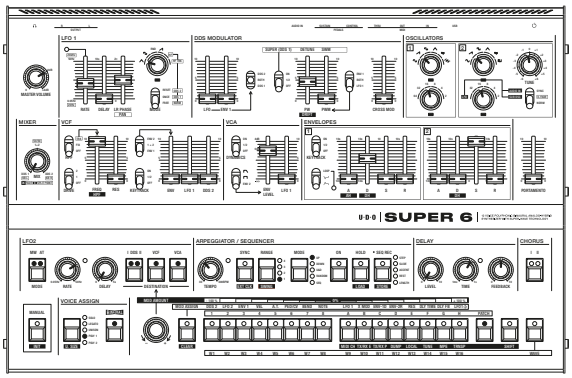
*After loading a patch from one of the second banks (**A2**-**H2**), you remain in select mode for those banks until you load a patch from one of the first banks (**A1**-**H1**) and vice versa.*

Starting from the Init Patch

While stored patches can serve as great starting points, sometimes it may be useful to start from scratch when trying to create a new sound. For this purpose you can load the so-called ‘init patch’, which contains a single oscillator set to a sawtooth wave, among other basic settings.



The init patch settings on the keyboard model.



The init patch settings on the desktop model.

To load the init patch:

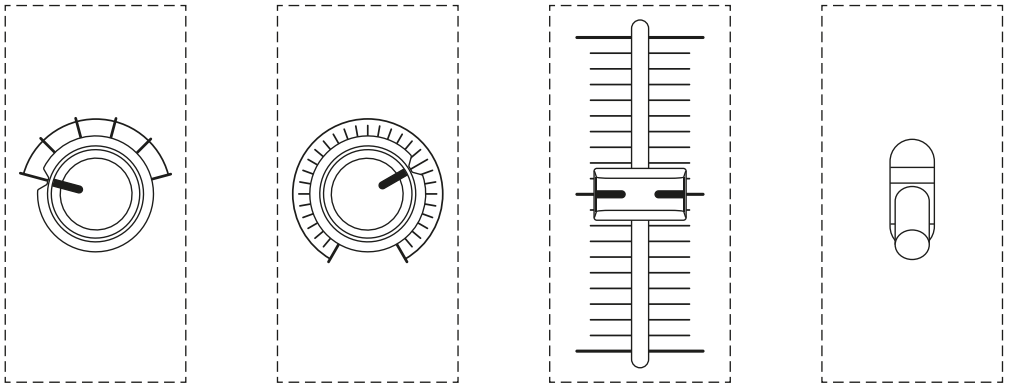
- Hold **SHIFT** and press the **MANUAL** button to load the init patch.

The init patch is loaded by default when the Super 6 is powered on.

Editing a Patch

The Super 6 was designed to allow for quick and enjoyable patch editing. The instrument's control panel is split into two horizontal halves. The lower half contains numerous performance controls, while the top raised panel primarily consists of sound shaping controls.

There are four main types of control elements that can impact a sound:



Rotary switch. Rotary control. Fader. Toggle switch.

Editing a patch is as simple as turning rotary controls, moving faders and toggling switches. Any gesture applied to a control element will have an immediate effect on the sound.

As soon as a patch-related parameter is changed, the LED of the currently lit numbered button (**1**-**8**) will start flashing, indicating that you are in edit mode. Keep experimenting, and once you have created a sound you like, it's time to save it.

Storing a Patch

To store a patch:

1. After editing a patch, press one of the lettered buttons (**A**-**H**) once or twice to select either one of the first banks (**A1**-**H1**) or one of the second banks (**A2**-**H2**). Its LED then lights up or starts flashing.

2. Press and hold one of the numbered buttons (**1**-**8**) for 3 seconds. The LEDs of buttons **1**-**8** and **A**-**H** will flash once to indicate that the patch has been stored.

Storing a patch will overwrite the patch previously stored to that location.

Comparing an Edited With a Stored Patch

Before saving a patch, it can be helpful to first ensure that you are not overwriting a patch that you still have good use for.

To compare an edited with a stored patch:

1. In edit mode, press the numbered button (**1**-**8**) where you want to store the edited patch.

   If you want to select a patch from another bank, first select the bank by pressing the respective button (**A**-**H**) once or twice.

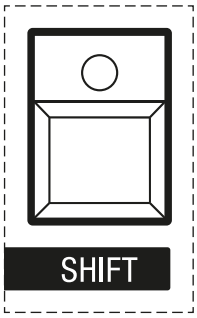
2. Press the same numbered button again to return to the patch you just edited.

There's More to It Than That: Shift Mode

The **SHIFT** button provides access to either shift mode parameters such as the global settings (see pages 102-104) or secondary parameter functions, labelled in inverse colours below the primary parameter names.

- To switch to shift mode, press and release the **SHIFT** button. Its LED will then start flashing to indicate you are in shift mode.

- To temporarily access the secondary function of a patch-related parameter (such as **HPF** cutoff frequency, **DRIFT** or **DECAY HOLD**), press and hold the **SHIFT** button while moving the corresponding control. In this case, shift mode is exited when the **SHIFT** button is released.



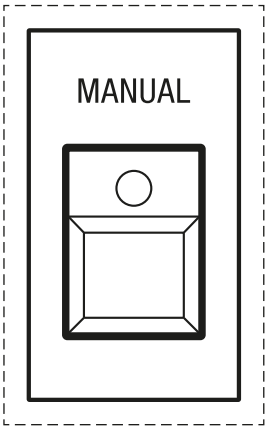
The shift mode button.

What You See Is What You Get: Manual Mode

Entering manual mode ignores the current patch settings and prompts the Super 6 to respond to the actual front panel control settings. This is a great way to better understand how each control affects the sound. In addition, it can also be a source of unexpected results!

To enter manual mode:

1. Press the **MANUAL** button.
2. To return to patch mode, simply press the **MANUAL** button again or press the **PATCH** button.



The manual button.

Level Up!

The Super 6 was designed to be played and tweaked in real-time, and we encourage you to do just that on your journey of finding and creating new sounds. After all, this is by far the best way to learn and fully understand the potential of your new instrument.

The following chapters of this manual provide a more detailed explanation of the Super 6's features and capabilities.

For information on how to adapt the instrument to your specific environment, such as using it alongside other MIDI instruments or to controlling it from a digital audio workstation (DAW) or an external sequencer, we recommend reading the paragraphs on global settings, connections and MPE control.

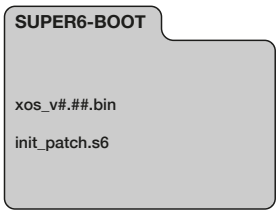
Throughout the manual you will also find some useful hints and tips to help you become familiar with the Super 6 and its sound shaping parameters.

Enjoy!

UPDATING THE FIRMWARE

Follow the steps below to update the firmware of your Super 6:

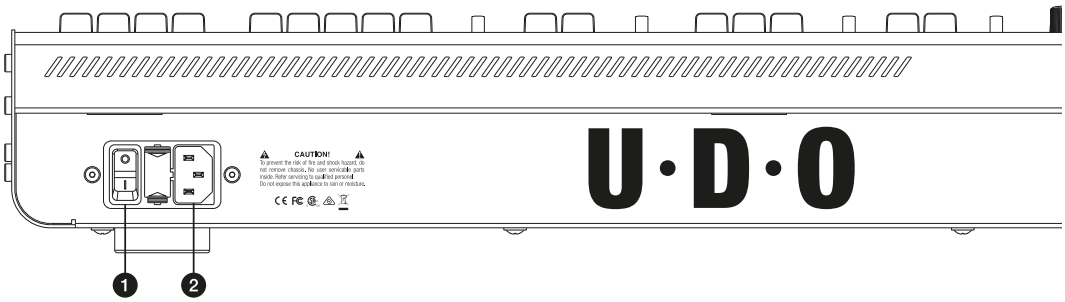
1. Enter bootloader mode:
   - Turn off the Super 6 and wait a few seconds.
   - Whilst holding down the **SHIFT** button, turn on the Super 6 and continue to hold the **SHIFT** button.
   - Progress LEDs will cycle through buttons **1**-**8** and **A**-**H** while the LED of the **SHIFT** button is flashing. (Make sure this is the case, restart step 2 if not.)
   - Release the **SHIFT** button.
2. Connect the Super 6 to your computer using a USB cable.
3. The Super 6's boot drive appears as a disk drive named **SUPER6-BOOT** on your computer.
4. Delete the firmware file 'xos_v*.**.bin' from the **SUPER6-BOOT** drive. Make sure to empty the trash if you are a macOS user, or the update won't be possible.
5. Copy the latest firmware file 'xos_v*.**.bin' from your computer to the **SUPER6-BOOT** drive. If asked if you want to copy files without properties, choose 'yes'. Please note that this process may take a few minutes. Do not turn off the instrument or unplug the USB cable during the transfer!
6. When the transfer is complete, disconnect the USB cable.
7. Turn off the Super 6, wait a few seconds and turn it on again. The instrument reboots and you will see progress LEDs, followed by a combination of briefly lit lettered select buttons. Each time you turn on the Super 6, this combination of lit buttons will indicate the currently installed firmware version.



The contents of the Super 6's boot drive.

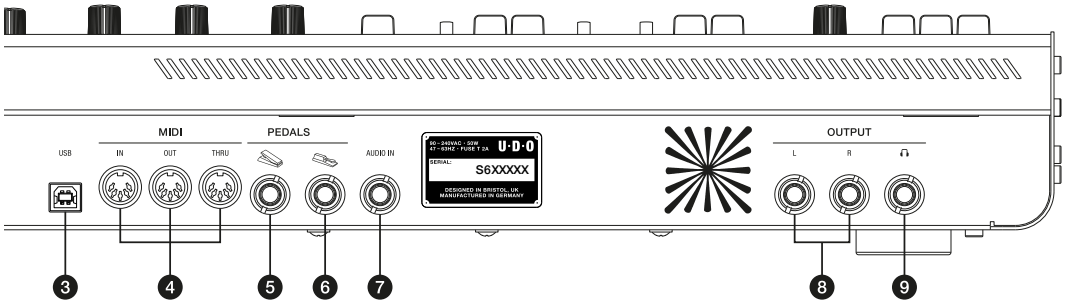
The latest firmware release can be downloaded from udo-audio.com/downloads.

Keyboard Model



1. Power Switch: Use this switch to power cycle the Super 6.

2. Power Connector: The AC power connector accepts a standard, grounded IEC power cord.



3. USB Port: Connect the Super 6 to your computer using the included USB cable for bidirectional MIDI communication, patch, waveform and sequence file management, and firmware updates. The Super 6 does not require any drivers to interface with a computer.

4. MIDI In, Out and Thru Ports: Standard 5-pin MIDI DIN connectors.

5. Expression Pedal Input: Connect an expression pedal to this input to add dynamics to your live performance. There are a variety of options for using an expression pedal, since it is an assignable modulation source in the Super 6's modulation matrix. This input accepts any standard expression pedal that features a TRS (Tip-Ring-Sleeve) connector and operates with a linear potentiometer over a range of 0 to +5 volts.

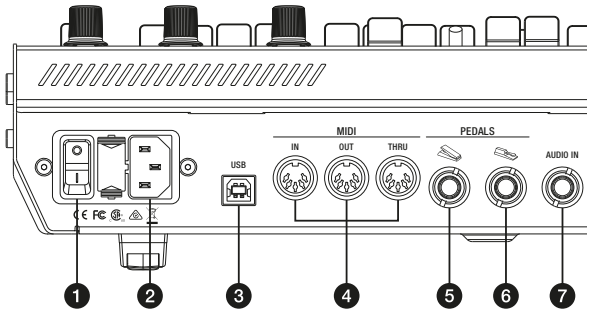
6. Sustain Pedal Input: Connect a footswitch to this input to sustain notes during your performance. Upon power cycling, the Super 6 will automatically detect the polarity of the connected pedal. Note that the state of the sustain pedal at power-up is taken as its 'off' state.

7. Audio In: This input allows you to process an external audio signal. The external audio signal will enter the signal path through the channel of DDS 2 before passing through the mixer, the analog filters, and the chorus and delay effects. The audio input accepts a stereo 1/4 inch jack.

8. Main Audio Outputs (Left and Right): The Super 6 is capable of gorgeous stereo sounds. Connect both outputs to your mixer or audio interface using unbalanced 1/4 inch cables. If you only connect the left output to your mixer or audio interface, the left and right signals will be summed to mono. Refer to page 93 on how to achieve a true monaural signal.

9. Headphone Output: Connect a 1/4 inch stereo headphone jack to this output. The overall volume of the headphone output is controlled by the **MASTER VOLUME** control.

Desktop Model



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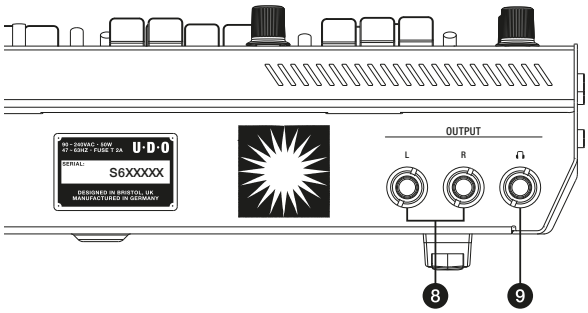

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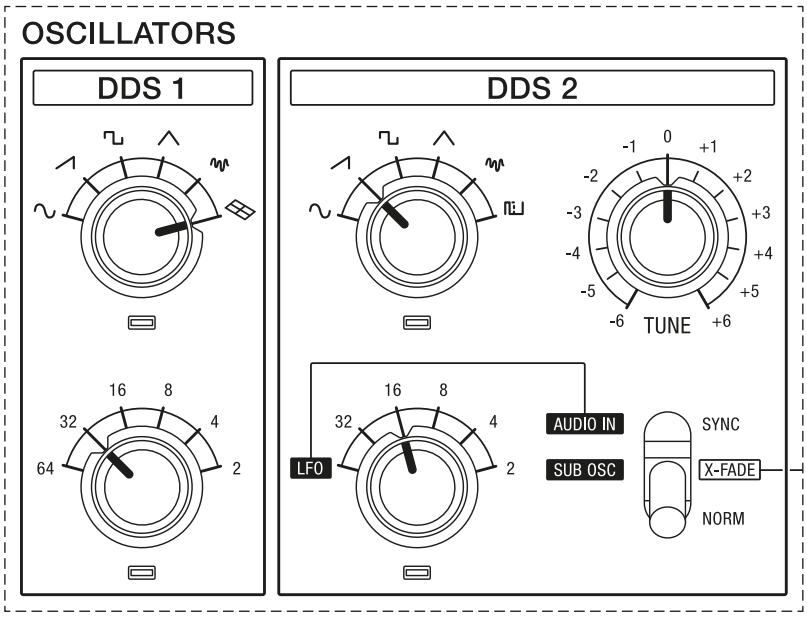
9. Headphone Output: Connect a 1/4 inch stereo headphone jack to this output. The overall volume of the headphone output is controlled by the **MASTER VOLUME** control.

In this chapter we will explore the Super 6's sound design capabilities by explaining the functionality of all patch-specific front panel controls related to the manipulation of sound.

Oscillators

Oscillators belong to the most basic and essential building blocks of a synthesizer. Without them, you could neither hear a sound nor shape or modulate what is generating an audio signal.



The oscillator section.

The Super 6's primary sound sources are its two FPGA-based oscillators (DDS 1 and DDS 2), which are capable of generating classic analog waveforms.

A **sine wave** contains only the first harmonic, the fundamental, which is why it is considered the purest waveform. It is ideal for 'glassy' sounds, an added fundamental and non-dissonant cross or ring modulation effects.

A **sawtooth wave** contains both odd and even harmonics and is bright sounding. It can be used for creating brass, bass and string sounds.

Square and pulse waves contain a wide range of odd harmonics. They sound hollow and can be used for reed-like sounds or basses. Apply pulse width modulation to use a pulse wave for swirling string sounds.

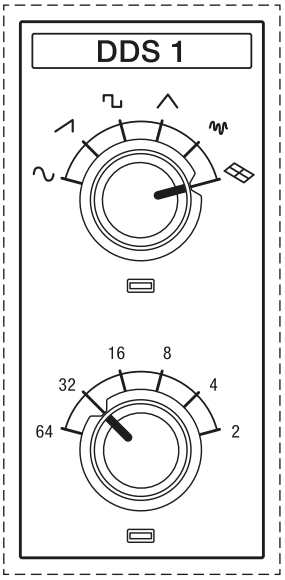
A **triangle wave** contains only odd harmonics and sounds very soft. It is particularly suitable for generating flute, organ or vocal sounds.

White noise contains all frequencies and is the most common noise waveform. It is useful for creating wind or percussive sounds.

In addition, DDS 1 offers up to 32 digital-sounding alternative waveforms organised in two groups. They are user definable (see page 109) and allow for a sheer unlimited range of sounds.

DDS 1 Parameters

DDS 1 features an FPGA-based super waveform oscillator core. It consists of a centroid oscillator and six 'sister' oscillators that can be dynamically de-phased in the stereo field by enabling one of the two super modes in the DDS Modulator section. Essentially, this means that DDS1 contains seven free-running oscillators, which give the Super 6 its characteristically rich and wide sound. See pages 61-63 for more details.



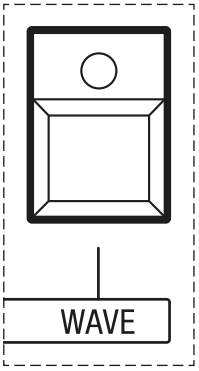
DDS1 controls.

WAVEFORM: Use this rotary switch to select a waveform for DDS 1. You can either select one of the classic waveforms such as sine, sawtooth, square, triangle and white noise or, when this control is set to the rightmost position, a digital-sounding alternative waveform.

RANGE: Use this rotary switch to adjust the coarse frequency of DDS 1 over a range of 64 to 2 feet.

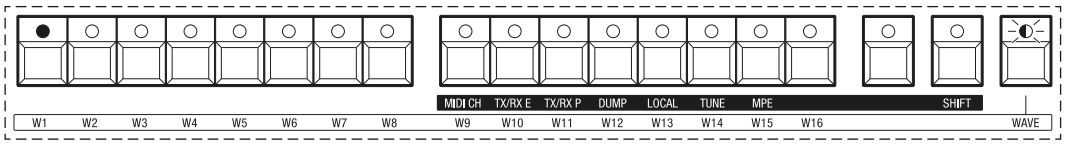
Loading Alternative Waveforms for DDS 1

To load alternative waveforms for DDS 1, make sure you are in wave mode, which is accessed by pressing the button labelled **WAVE** or by moving the DDS 1 waveform rotary switch to the rightmost position.



The wave mode button.

Notice how the **WAVE** button label is surrounded by a box, which also contains the names of the functions the select buttons **1**-**8** and **A**-**H** take over in wave mode: **W1** to **W16**, which stands for waveform 1 to 16.



The waveform select buttons.

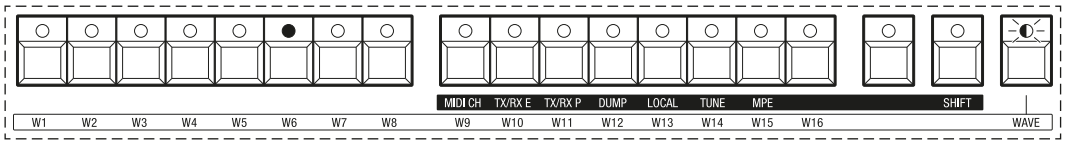
The alternative waveforms are organised in two groups of 16 waveforms each, giving you access to a total of 32 alternative waveforms.

You can use each of the select buttons (**W1**-**W16**) to load two alternative waveforms: **W1** and **W17**, **W2** and **W18**, etc. For example, toggling between alternative waveforms 8 and 24 is achieved by repeatedly pressing button **W8**.

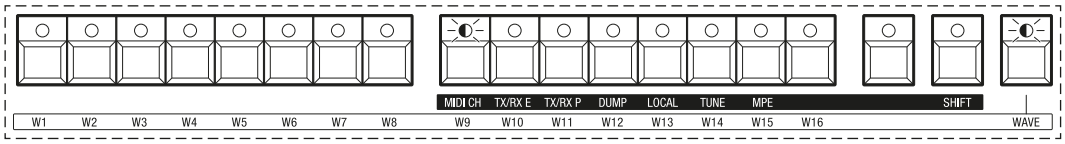
When you load a waveform from the first group of 16 waveforms (1-16), the corresponding LED lights up. If you load a waveform from the second group of 16 waveforms (17-32), the LED will start flashing.

After loading an alternative waveform from the second group, you remain in selection mode for that group until you load an alternative waveform from the first group again, and vice versa.

In the first example, waveform **6** is loaded:



In the second example, waveform **25** is loaded:



To load alternative waveforms:

1. Press **WAVE** or move the DDS 1 waveform rotary switch to the rightmost position. The LED of the **WAVE** button will start flashing, indicating that you are now in waveform selection mode.

2. Play some notes and press one of the select buttons (**W1**-**W16**) once or twice to load a waveform.

3. Press the **PATCH** button to exit wave mode.

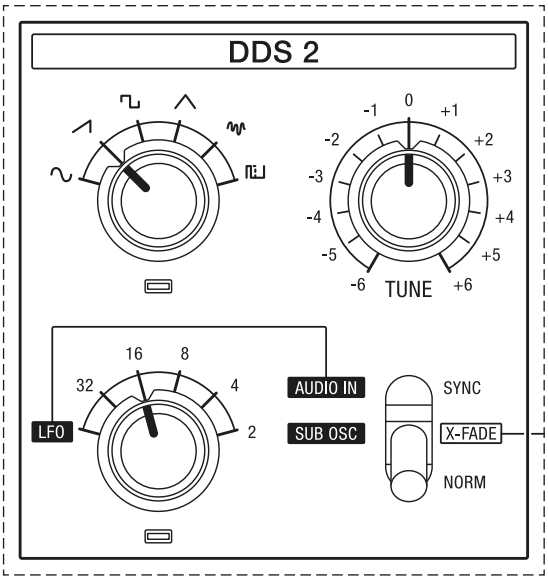
Each patch remembers the alternative waveforms it was saved with. Even if you replaced all alternative waveforms the Super 6 is shipped with, the factory patches wouldn't change.

When you switch from patch to manual mode, the alternative waveform from the previously loaded patch will be retained for DDS 1.

DDS 2 Parameters

DDS 2 features an FPGA-based oscillator core running at a very high sample rate and provides you with six classic waveforms. Unlike DDS 1, which uses sampled waveforms, DDS 2 has an algorithmic core and thus behaves in a subtly different way.

The phase of the DDS 2 waveform is reset to zero with each note played to allow for binaural pitch and pulse width modulation via LFO 1. The latter wouldn't be possible if DDS 2 was operating as a free-running oscillator.



DDS 2 controls.

WAVEFORM: Use this rotary switch to select a waveform for DDS 2. You can choose from the following classic waveforms: sine, sawtooth, square, triangle, white noise or pulse.

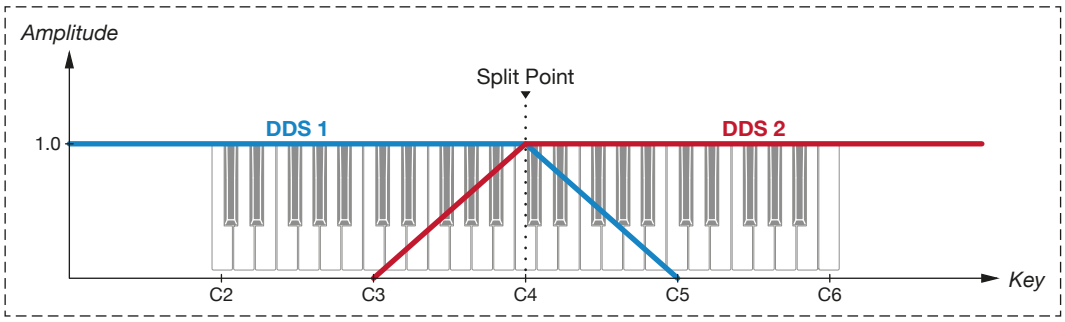
RANGE: Use this rotary switch to adjust the coarse frequency of DDS 1 over a range of 32 to 2 feet. If you turn this control to the leftmost position, DDS 2 will act as an additional LFO. In this mode, you can also enable a sub-oscillator. For more details see page 35.

TUNE: This control allows you to fine-tune the frequency of DDS 2 over a range of +/-10 semitones. You can use this control to either slightly detune DDS 2 relative to DDS 1 or to create intervals such as fourths or fifths.

The toggle switch in the DDS 2 section allows you to choose from three different modes:

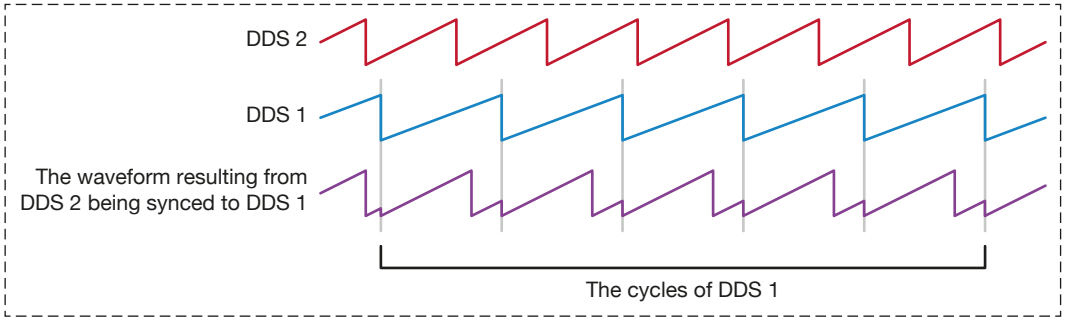
- **NORM:** This is the default mode for DDS 2, in which it behaves like a normal oscillator.

- **X-FADE:** When this mode is enabled, you can crossfade between the signals of DDS 1 and DDS 2 relative to an adjustable split point on the keyboard. The crossfade between the signal of both oscillators covers an interval of two octaves. The split point can be adjusted using the **MIX** control (see page 39).



Crossfading the signals from DDS 1 and DDS 2 across the keyboard.

- **SYNC:** Selecting this option, also known as ‘hard sync’, forces DDS 2 to restart its duty cycle each time DDS 1’s duty cycle begins. By setting the frequency of DDS 2 to a higher pitch than DDS 1, you can create complex and harmonically rich timbres, especially if you modulate the pitch of DDS 2 with an envelope. A famous example of a typical hard-sync patch is the lead sound in Daft Punk’s “Robot Rock”.

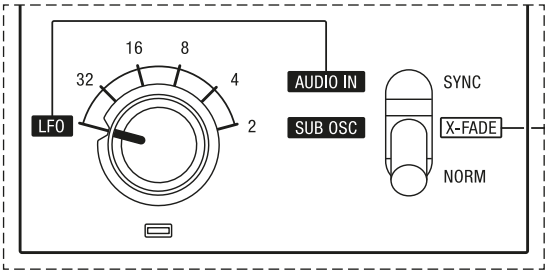


The duty cycle of DDS 2 synchronised to DDS 1 with both oscillators set to a sawtooth wave.

*When you are programming a non-binaural patch, make sure to manually disable binaural mode again after you toggled to **SYNC** via **X-FADE**. Since there are six pairs of VCAs for the mixer, binaural mode is always be enabled when you select X-FADE mode.*

Using DDS 2 as an LFO

As mentioned above, you can also use DDS 2 as an LFO. To do so, turn the **RANGE** control to the leftmost position marked **LFO**.



DDS 2 set to LFO mode.

When DDS 2 is used as an LFO, its signal is no longer routed to the audio path. In this mode, the **TUNE** control determines the frequency over a range of approximately 0.1 to 100 Hz. As in default mode, the LFO waveform can be selected with the **WAVEFORM** rotary switch.

In LFO mode, DDS 2 offers two more waveforms than LFO 1, namely sine and pulse. If you pulse width modulate DDS 2 in the DDS Modulator section or modulate its frequency via LFO 1, DDS 2 can become a very complex and dynamic LFO, which allows for interesting modulation results.

You can route DDS 2 in LFO or default mode to all available modulation destinations via the modulation matrix. See pages 85-91 for more information on how to use the modulation matrix.

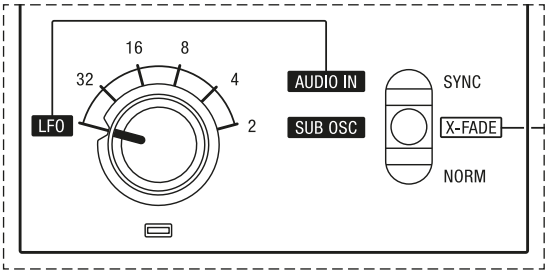
Enabling the Sub-Oscillator

Enabling the sub-oscillator is useful for adding more bottom end to your sounds. In this mode, the audio signal from the sub-oscillator replaces the audio signal from DDS 2. The pitch of the sub-oscillator is locked one octave below the frequency of DDS 1. The **WAVEFORM** and **TUNE** controls have no effect on the sub-oscillator.

The waveform of the sub-oscillator is a square wave that allows you to add harmonically rich bass tones to your sounds.

To enable the sub-oscillator:

1. Turn the **RANGE** control to the leftmost position marked **LFO**.
2. Flip the mode toggle switch to the middle position marked **SUB OSC**.



Enabled sub-oscillator.

You can adjust the level of the sub-oscillator in the mixer section. When the sub-oscillator is enabled, the **MIX** control blends between the audio signals from DDS 1 and the sub-oscillator.

With the sub-oscillator enabled, DDS 2 can still be used as an additional LFO.

Processing an External Audio Signal

In addition to the two options described above, you can also use the DDS 2 module to process an external audio signal:

1. Connect an external audio source to the audio input on the rear side of the Super 6 using a stereo 1/4 inch jack.

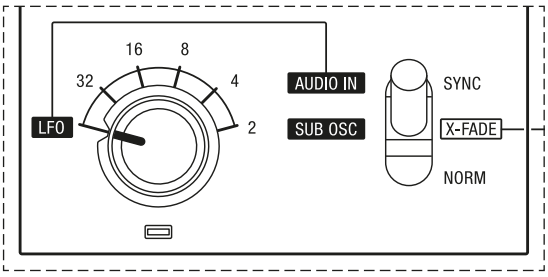
2. Press down a key or engage the hold mode. Turn the **MIX** rotary control fully clockwise so that only the audio signal of DDS 2 is heard.

3. Turn the **RANGE** control to the leftmost position marked **LFO**.

4. Flip the mode toggle switch to the upper position marked **AUDIO IN**.

5. If you still hear an audio signal on top of the external audio signal, ensure that the signal of LFO 1 in either high-frequency mode is not routed to the channel of DDS 2. When LFO 1 is set to one of the high-frequency modes and the LFO 1 mode toggle switch is set to **DDS 2**, the signal from LFO 1 will be mixed with the external audio signal that the Super 6 receives via the DDS 2 channel.

*When you feed an audio signal into the signal path of the Super 6, it will trigger a gate signal as soon as it passes a certain threshold level, unless you select **POLY 2** in the voice assign section.*



Allowing an external audio signal to be processed.

As you may have noticed, these settings enable DDS 2's LFO mode. In this mode, the audio signal of DDS 2 is bypassed to make way for an external audio signal that is routed into the mixer via the DDS 2 channel.

You can control the balance between the audio signals from DDS 1 and an external audio source using the **MIX** control in the mixer section (see page 38 for more details). Here the external audio signal simply replaces that of DDS 2. After the mixer, the external audio signal is passed through the voltage controlled filter (VCF), voltage controlled amplifier (VCA) and effects section, all of which allow you to further modify and process the incoming signal.

To adjust the input gain of an external audio signal:

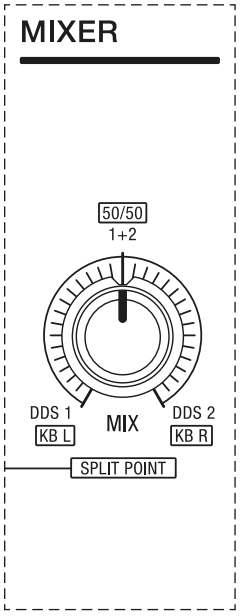
1. Press **SHIFT**.
2. Turn the **MIX** control to adjust the input gain.
3. Press **SHIFT** again to return to editing your sound.

If you don't manually adjust the input gain, the gain will default to a line input level that will make an external audio signal as loud as the Super 6's oscillator signals.

The audio input also features a gate trigger detector so that the envelopes and LFO 1 in reset mode are triggered when an incoming audio signal reaches a threshold determined by the input gain level. The LED to the left of envelope 1 indicates when an external audio signal passes the threshold. It lights up every time a trigger impulse is received.

Mixer

In the mixer section you can control the balance between the audio signals from DDS 1 and DDS 2, DDS 1 and the sub-oscillator or DDS 1 and an external audio signal. When DDS 2 is set to X-FADE mode, you can adjust the split point in this section.



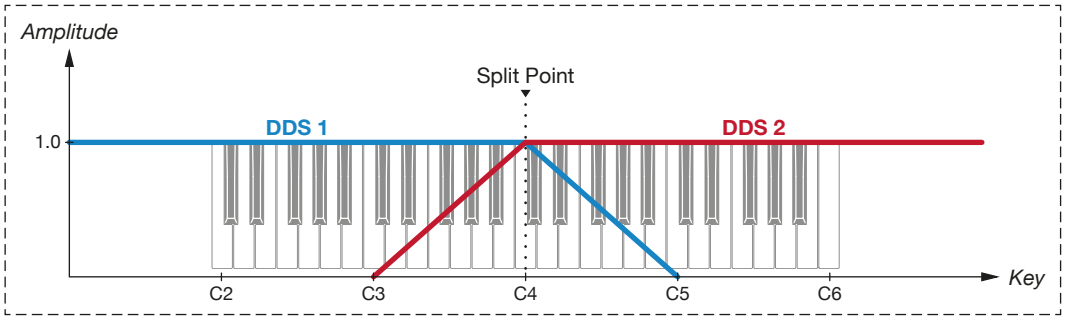
The mixer section.

MIX: At 12 o'clock, the signals from both audio sources are equally balanced. If set to the leftmost position, only the audio signal from DDS 1 will be audible. Likewise, only the audio signal from DDS 2 or that of the sub-oscillator is audible if you turn the control to the rightmost position.

When DDS 2 is set to X-FADE mode (see page 33), you can use the **MIX** control to set the split point on the keyboard relative to which the audio signals from DDS 1 and DDS 2 or DDS 1 and an external source are crossfaded.

At a 12 o'clock setting the split point is set to middle C (C4). Since the crossfade between the signal of both oscillators covers an interval of two octaves, the crossfade ranges from C3 to C5 at this setting.

The further you turn the **MIX** control counterclockwise, the further the split point will move down to the left half of the keyboard. Likewise, turning the **MIX** control clockwise will move the split point up to the right half of the keyboard.



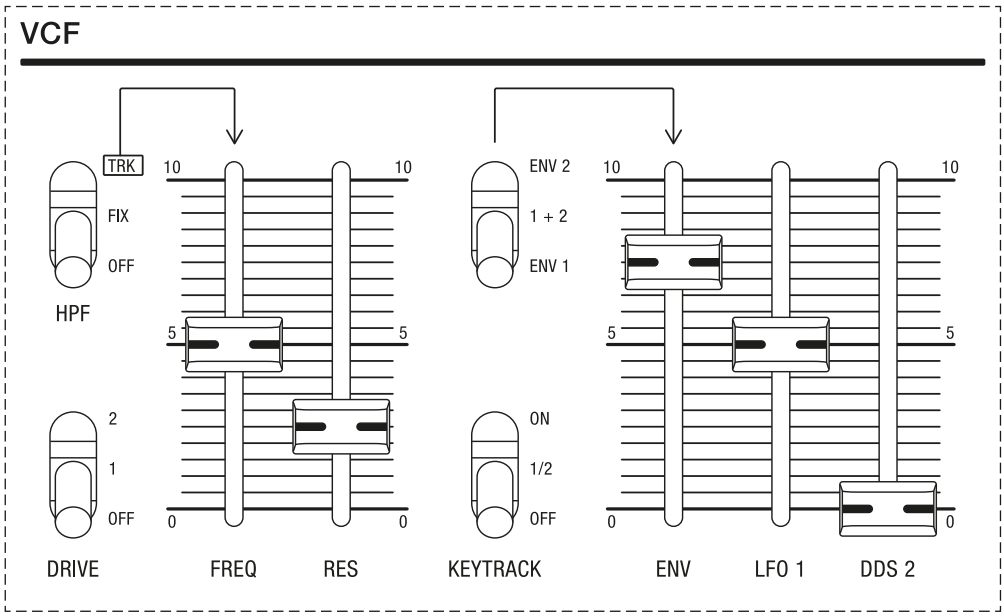
Crossfading the signals from DDS 1 and DDS 2 across the keyboard.

Since there are six pairs of VCAs for the mixer, binaural mode is always be enabled when you select X-FADE mode.

VCF (Voltage Controlled Filter)

The filter section is integral to the instrument's unique sonic character and allows you to shape the sound of the oscillators by modifying the harmonic content of their signals.

The Super 6's main filter is a 4-pole, 24 dB per octave, analog resonant low-pass filter that uses a classic polysynth filter design by Sound Semiconductor (SSI). It is preceded in the signal chain by an analog high-pass filter (HPF) that can either be turned off, set to a fixed frequency, or track the low-pass filter cutoff frequency for band-pass operation.



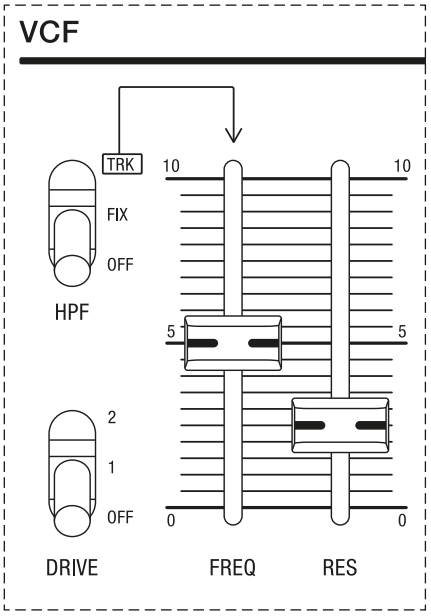
The filter section.

The **low-pass filter** (**LPF**) subtracts frequency content above its cutoff frequency. The frequency content below the cutoff frequency remains unaffected, meaning the lows will pass through.

The **high-pass filter** (**HPF**) subtracts frequency content below its cutoff frequency. The frequency content above the cutoff frequency remains unaffected, meaning the highs will pass through. Using the high-pass filter can be useful for removing 'muddy' low ends from pad sounds, particularly in recording or mixing situations.

The Super 6 offers numerous options for modulating the low-pass filter's cutoff frequency, making the filter section extremely versatile and suitable for a wide variety of sounds.

Overall, the filter section of the Super 6 is divided into two: In the left half you can adjust basic filter settings. The right half of the filter section contains controls that affect how the low-pass filter is modulated by different modulation sources and how it responds to key tracking.



Left half of the filter section.

The **DRIVE** toggle switch allows you to determine whether and to what extent the low-pass filter signal is overdriven:

- **OFF:** This setting results in a clean filter signal.
- **1:** This setting adds subtle saturation with resonance compensation.
- **2:** This setting adds a healthy dose of overdrive.

The **HPF** toggle switch allows you to choose from three different high-pass filter modes:

- **OFF:** With this setting, the high-pass filter is turned off
- **FIX:** With this setting, the high-pass filter removes a fixed amount of low frequency content from the sound at around 500 Hz. This can be useful for eliminating excessive bass content from polyphonic sounds.
- **TRK:** With this setting, the low-pass and high-pass filter modes are combined into a bandpass filter that only allows a frequency band to pass as set by the cutoff frequency.

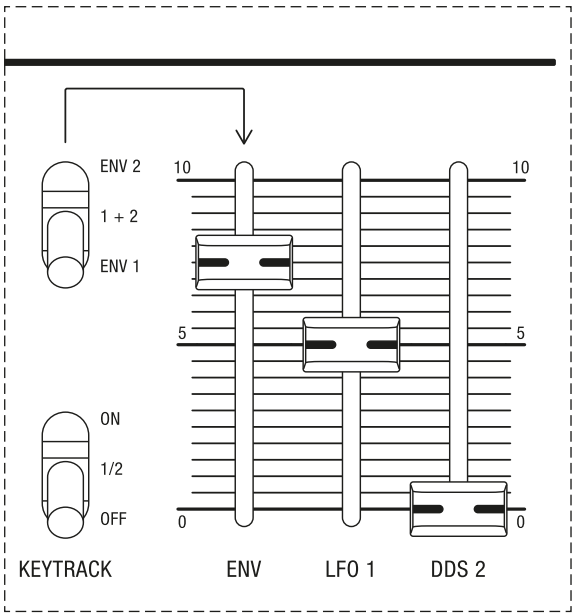
FREQ: This fader controls the cutoff frequency of the low-pass filter.

HPF: In shift mode, you can use the **FREQ** fader to independently control the high-pass filter's cutoff frequency if you set the **HPF** toggle switch to **FIX**.

RES: This fader controls the amount of resonance the low-pass filter responds with. As you increase the amount of resonance, the frequencies around the cutoff frequency set by the **FREQ** fader will be emphasised and more pronounced.

*The Super 6's low-pass filter is designed to respond to key tracking in a musical manner. This in turn determines how much you can open the filter with the **LPF** fader. You can use the remaining headroom if you modulate the low-pass filter's cutoff frequency with an envelope or an expression pedal via the modulation matrix.*

*The low-pass filter can be driven into self-oscillation if you set **RES** to its highest value. In this case, the filter generates a pitch determined by the cutoff frequency and a timbre that sounds like a sine wave. You can also use the keyboard to control or rather play the low-pass filter's pitch if you enable key tracking.*



Right half of the filter section.

ENV: This fader controls the amount by which either or both envelopes modulate the low-pass filter's cutoff frequency over time. See pages 46-53 to learn more about the envelope generators.

Use the upper toggle switch to select the envelope modulation source:

- **ENV 1:** With this setting, envelope 1 acts as the modulation source.
- **1 + 2:** With this setting, both envelopes act as modulation sources.
- **ENV 2:** With this setting, envelope 2 acts as the modulation source.

LFO 1: This fader controls the amount by which LFO 1 modulates the low-pass filter's cutoff frequency.

DDS 2: This fader controls the amount by which DDS 2 modulates the low-pass filter's cutoff frequency. The result can range from subtle textures to complex, experimental timbres.

The **KEYTRACK** toggle switch allows you to determine whether and to what extent the low-pass filter's cutoff frequency responds relative to the pitch of the notes played on the keyboard:

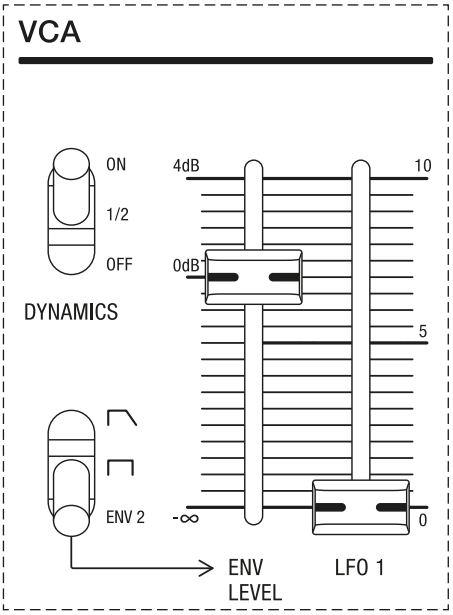
- **OFF:** With this setting, the low-pass filter's cutoff frequency is unaffected by the pitch of the notes played on the keyboard.
- **1/2:** With this setting, the low-pass filter's cutoff frequency follows the keyboard pitch in quarter tone steps, resulting in brighter sounding higher notes. This is how acoustic instruments typically behave, so this setting can be useful for creating more natural-sounding timbres.
- **ON:** With this setting, the low-pass filter's cutoff frequency follows the keyboard pitch in semitones as you move up the keyboard. This is useful when using the low-pass filter in self-oscillating mode, as the pitch generated by the filter then precisely follows the intervals you play on the keyboard, allowing you to play the filter like an oscillator.

Since there are six pairs of VCAs for the mixer, binaural mode is always enabled when you select one of the key tracking modes.

VCA (Voltage Controlled Amplifier)

After being shaped by the filter, the audio signal passes on to the voltage-controlled amplifier (VCA). The Super 6's VCA section can be used to further shape the sound by modifying and modulating its volume.

By default, the level generated by the VCA is controlled by envelope 2 (ENV 2), which gives you control over the attack, decay, sustain, and release stages. On pages 44-45 you can read more about how the VCA level can be modulated by the dedicated envelope. Alternatively, the VCA level can be controlled by one of two fixed envelopes, allowing the second envelope to be freed up for other tasks if desired.



The VCA section.

ENV LEVEL: This fader controls the amount by which envelope 2 or one of the fixed envelopes modulates the VCA level over time.

LFO 1: This fader controls the amount by which LFO 1 modulates the VCA level. This parameter is useful for tremolos, as the volume increases and decreases according to the rate of LFO 1. Use a triangle wave for smooth tremolo effects and a square wave for abrupt tremolo effects.

The envelope selector toggle switch allows you to choose between three types of VCA envelopes:

- **Lower position:** With this setting, envelope 2 acts as the modulation source. This is the default setting, which means that the second envelope is usually responsible for modulating the VCA level.
- **Middle position:** With this setting, the first of the fixed envelopes acts as the modulation source. Its attack, decay and release stages have no duration, meaning it acts as a simple on/off envelope or gate.

- **Upper position:** With this setting, the second of the fixed envelopes acts as the modulation source. Its attack and decay stages have no duration, but it does feature a release stage. Use this envelope when you want to free up envelope 2 for other modulation tasks, but prefer your sound to fade out after playing a note.

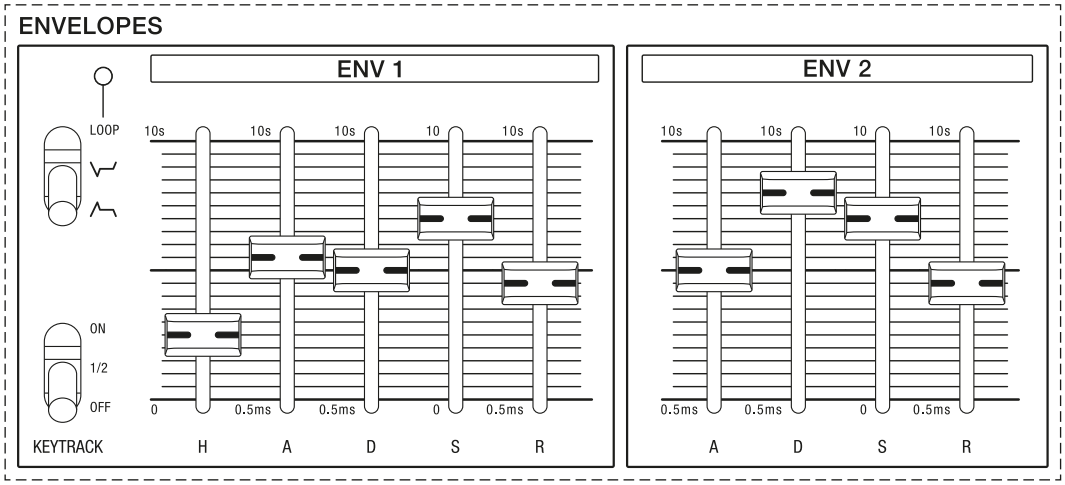
The **DYNAMICS** toggle switch allows you to determine whether and to what extent the VCA level and the low-pass filter will respond to keyboard velocity:

- **OFF:** With this setting, the VCA level and the low-pass filter remain unaffected by keyboard velocity.
- **1/2:** With this setting, the VCA level and the low-pass filter respond to keyboard velocity with half intensity. The harder you hit a key, the louder and brighter the sound becomes. Use this setting when you want the velocity effect to be subtle.
- **ON:** With this setting, the VCA level and the low-pass filter respond to keyboard velocity with full intensity. The harder you hit a key, the louder and brighter the sound becomes. Use this setting when you want velocity to have a significant impact, for example if you wish to emulate the behaviour of acoustic stringed instruments.

*Depending on the level of signals fed into the VCA section, high **ENV LEVEL** settings may cause the amplified output signal to be clipped or distorted regardless of the **MASTER VOLUME** setting. Simply reduce the **ENV LEVEL** settings to avoid unwanted clipping artifacts and to ensure a clean signal at the end of the signal chain.*

Envelopes

By using envelope generators we can specify how a sound evolves over time. Typically, envelope generators are mapped to filters and amplifiers to change the harmonic content and the overall volume of a sound through multiple stages.

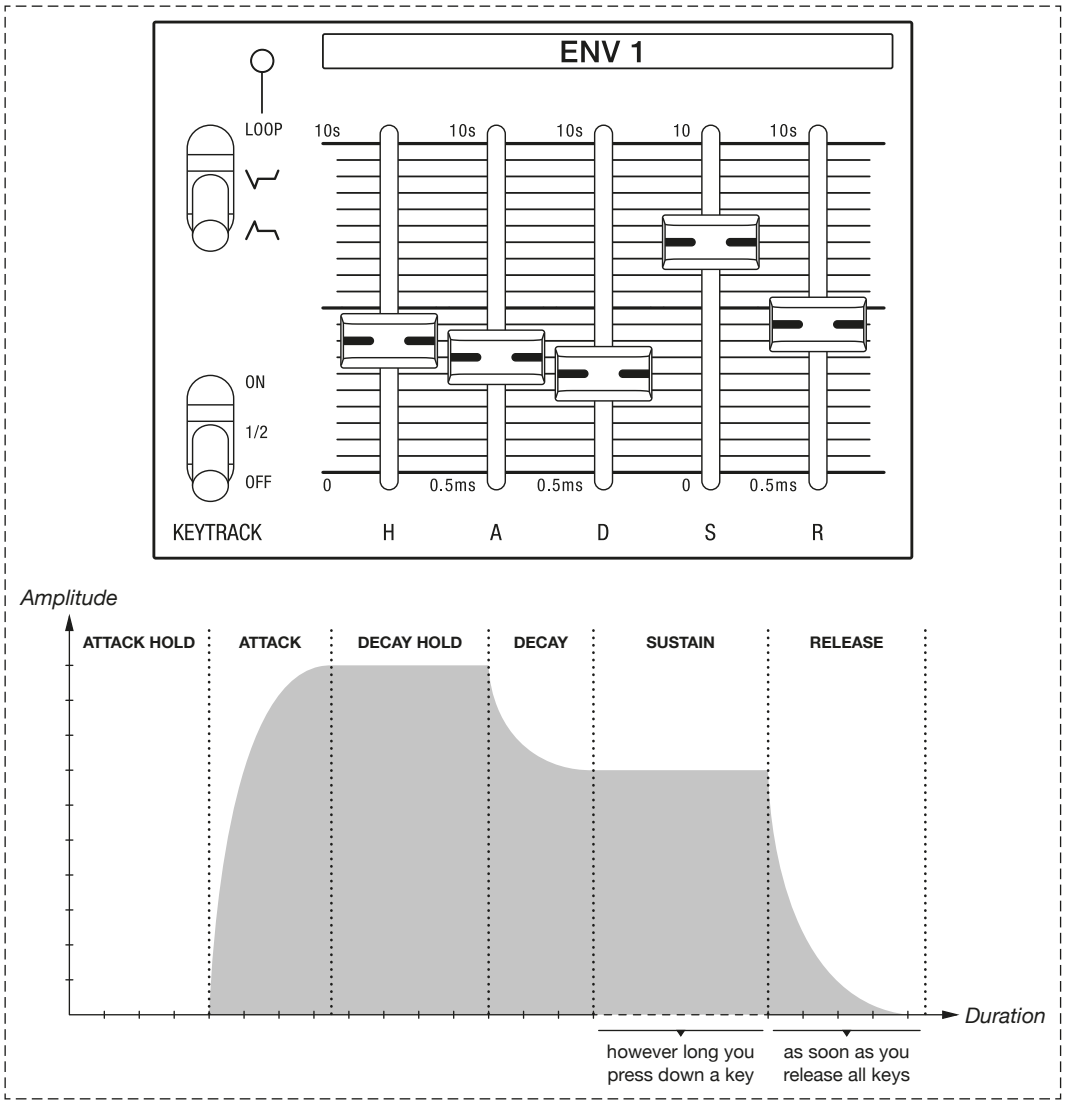


The envelope section.

The Super 6's envelopes can be mapped to multiple destinations including the low-pass filter's cutoff frequency (see page 42), the VCA level (see pages 44-45) and oscillator-specific modulation destinations (see pages 60-63).

Both envelopes contain five stages known as attack, decay hold, decay, sustain and release. In addition, envelope 1 features a hold stage that can be used to delay the moment when the attack stage begins after hitting a key.

ENV 1 (Envelope 1)



Envelope 1 setting and a diagram of the resulting envelope shape.

A(TTACK) H(OLD): In shift mode, you can use the **ATTACK** fader to adjust the time it takes for the attack stage to begin after hitting a key. The attack hold stage can be as long as 10 seconds. At its minimum setting this parameter has no effect, i.e. the envelope then behaves as if it had only four or five stages (attack, decay, sustain and release or attack, decay hold, decay, sustain and release).

On the desktop model, you can access this parameter when you enable shift mode. You can then use the **ATTACK** fader to adjust the duration of the attack hold stage.

A(TTACK): This fader determines the duration of the envelope's attack stage. The higher the setting, the slower the attack time and the longer it takes for the envelope to reach its maximum level. The attack stage can be as short as 1 millisecond or as long as 10 seconds.

D(ECAY) H(OLD): In shift mode, you can use the **DECAY** fader to adjust the time it takes for the decay stage to begin after the attack stage reached its peak. The decay hold stage can be as long as 10 seconds. At its minimum setting this parameter has no effect, i.e. the envelope then behaves as if it had only four or five stages (attack, decay, sustain and release or attack hold, attack, decay, sustain and release).

D(ECAY): This fader determines the duration of the envelope's decay stage. The higher the setting, the longer it takes for the envelope to travel from its maximum level to the level determined by the sustain fader. The decay stage can be as short as 1 millisecond or as long as 10 seconds.

S(USTAIN): This fader determines what level the envelope will be held at when you hold a note past the decay stage. This is the only envelope parameter that is not tied to a duration but to a level. The duration of the sustain stage always depends on how long you hold a note.

R(ELEASE): This fader determines the duration of the envelope's release stage. The higher the setting, the slower the release time and the longer it takes for the envelope to fade out after you release a key. The release stage can be as short as 1 millisecond or as long as 10 seconds.

The **KEYTRACK** toggle switch allows you to determine whether and to what extent envelope 1's decay and release stages respond relative to the pitch of the notes played on the keyboard:

- **OFF:** With this setting, the duration of the envelope's decay and release stages is unaffected by the pitch of the notes played on the keyboard.
- **1/2:** With this setting, the time it takes for the envelope to cycle through its decay and release stages decreases in quarter tone steps as you move up the keyboard.
- **ON:** With this setting, the time it takes for the envelope to cycle through its decay and release stages decreases in semitones as you move up the keyboard.

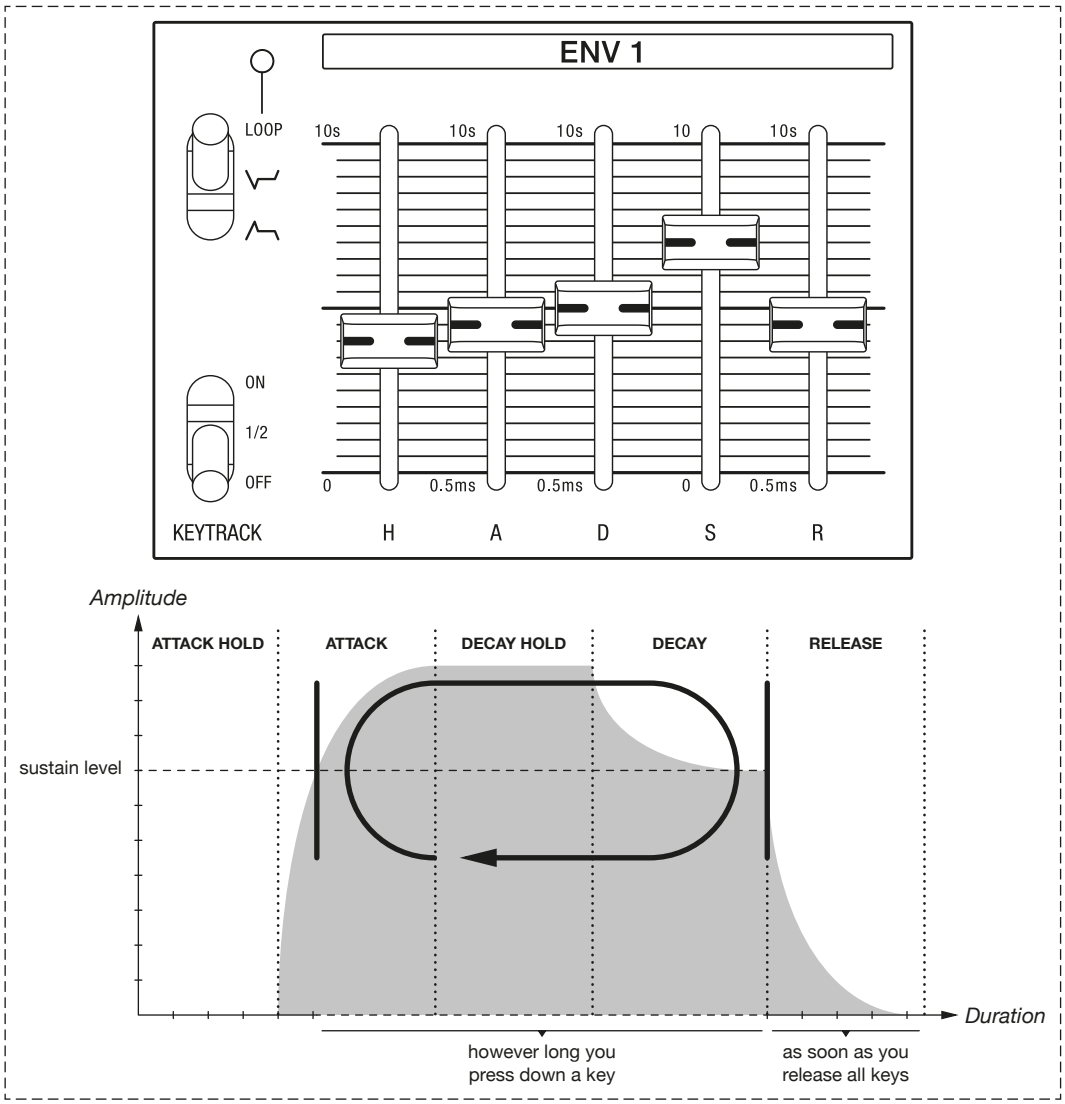
If you don't notice any noticeable key tracking effect when envelope 1 is set to inverted mode, consider that the effect the inverted envelope has on its modulation destination is the opposite of envelope 1 set to default mode.

The upper toggle switch (on the desktop model the lower one) allows you to choose between three types of envelope behaviours:

- **Lower position:** With this setting, envelope 1 is in default mode.

- **Middle position:** With this setting, the shape of envelope 1 is inverted. An envelope that is ramping up during its attack stage will now ramp down. The effect on the modulation destination is the opposite of envelope 1 modulating in default mode.
- **Upper position:** With this setting, envelope 1 operates in loop mode. Instead of being triggered just once, the attack, decay hold and decay stages are repeated indefinitely until you release a key. As soon as you release a key, the release stage is triggered. The rate at which the looped envelope is repeated is indicated by the LED graphically linked to the **LOOP** label.

In loop mode, the sustain setting determines from what level the envelope rises at the beginning of the attack stage and to what level it falls at the end of the decay stage.



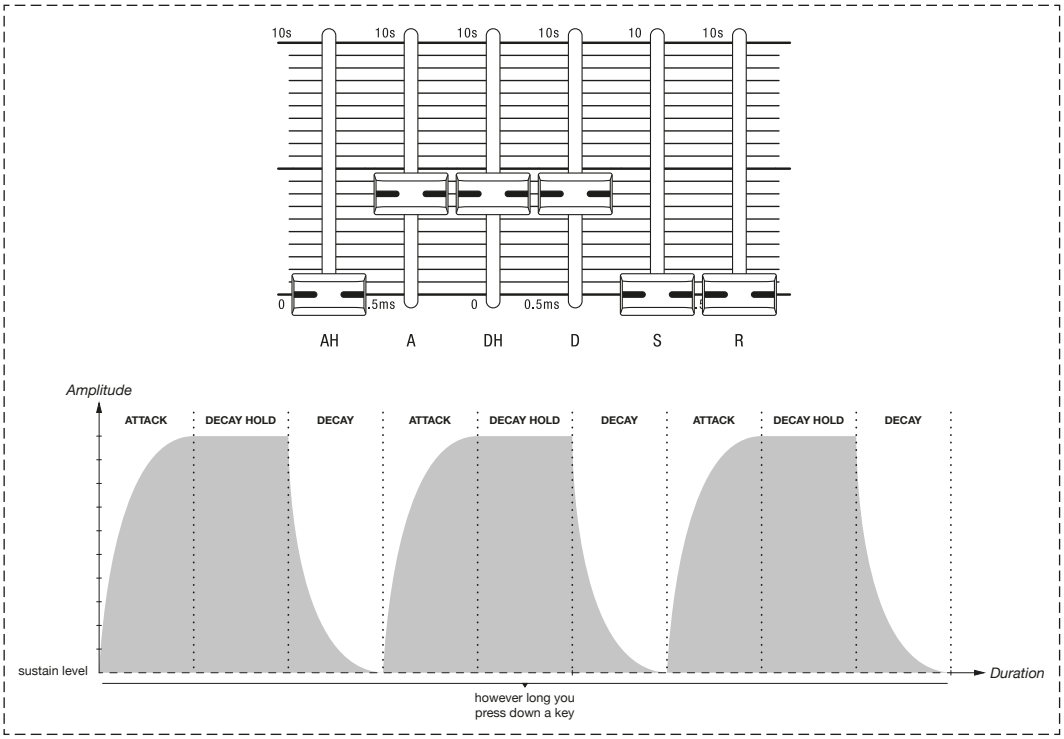
Envelope 1 in loop mode.

In loop mode you can use envelope 1 as an additional LFO, which can even be key tracked. Low attack, decay hold, and decay settings can generate sonic results that resemble frequency modulation.

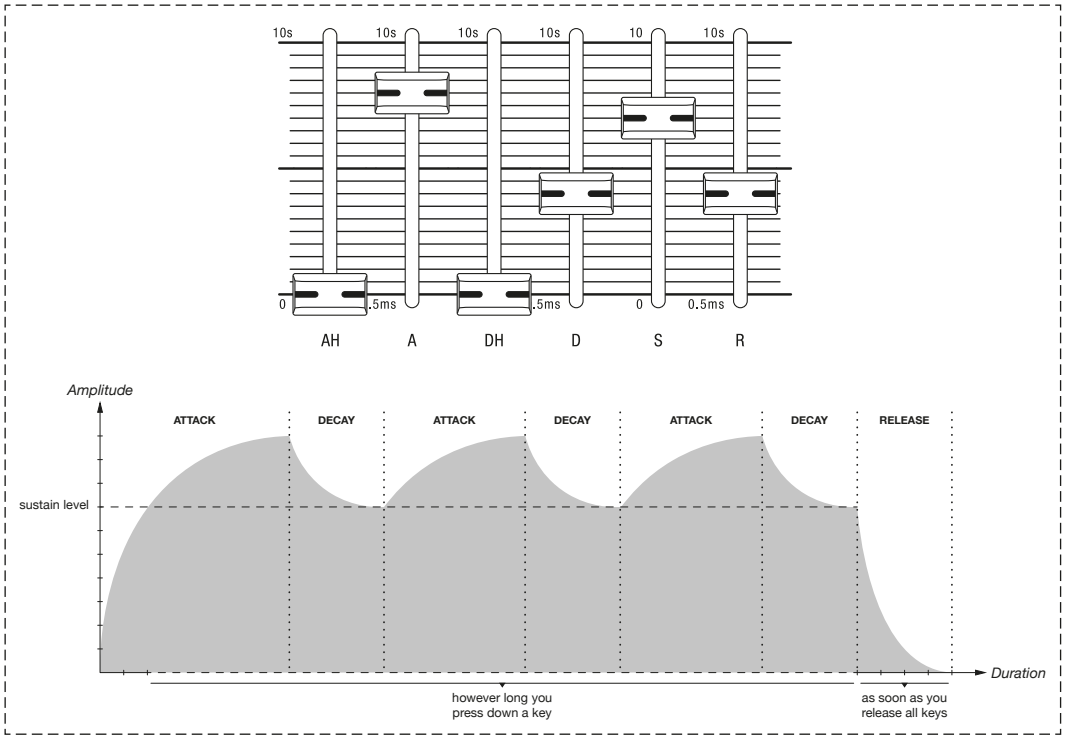
Creating Periodic Modulation Shapes With Envelope 1 in Loop Mode

Below are some examples that illustrate what periodic modulation shapes you can create with envelope 1 in loop mode. The repeated shapes are based on the looped segment of envelope 1, namely the attack, decay hold and decay stages.

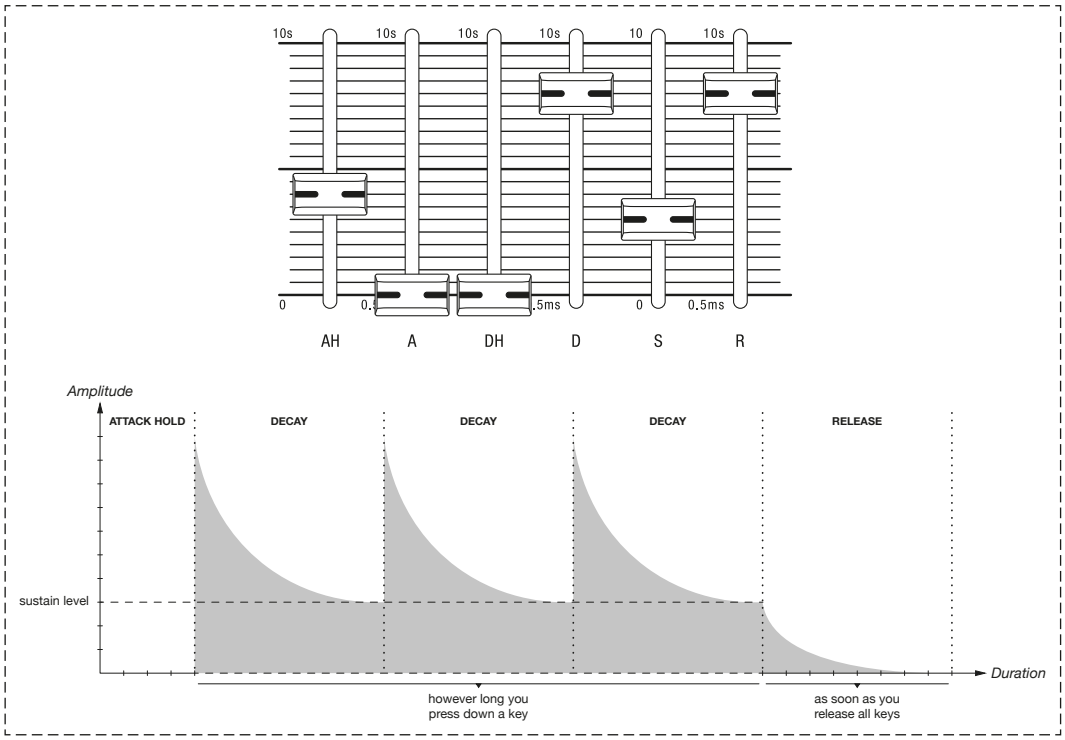
Note that the **ATTACK HOLD** and **DECAY HOLD** faders shown in the figures below have been added for illustrative purposes only. Your Super 6 is not missing any parts!



Example 1.

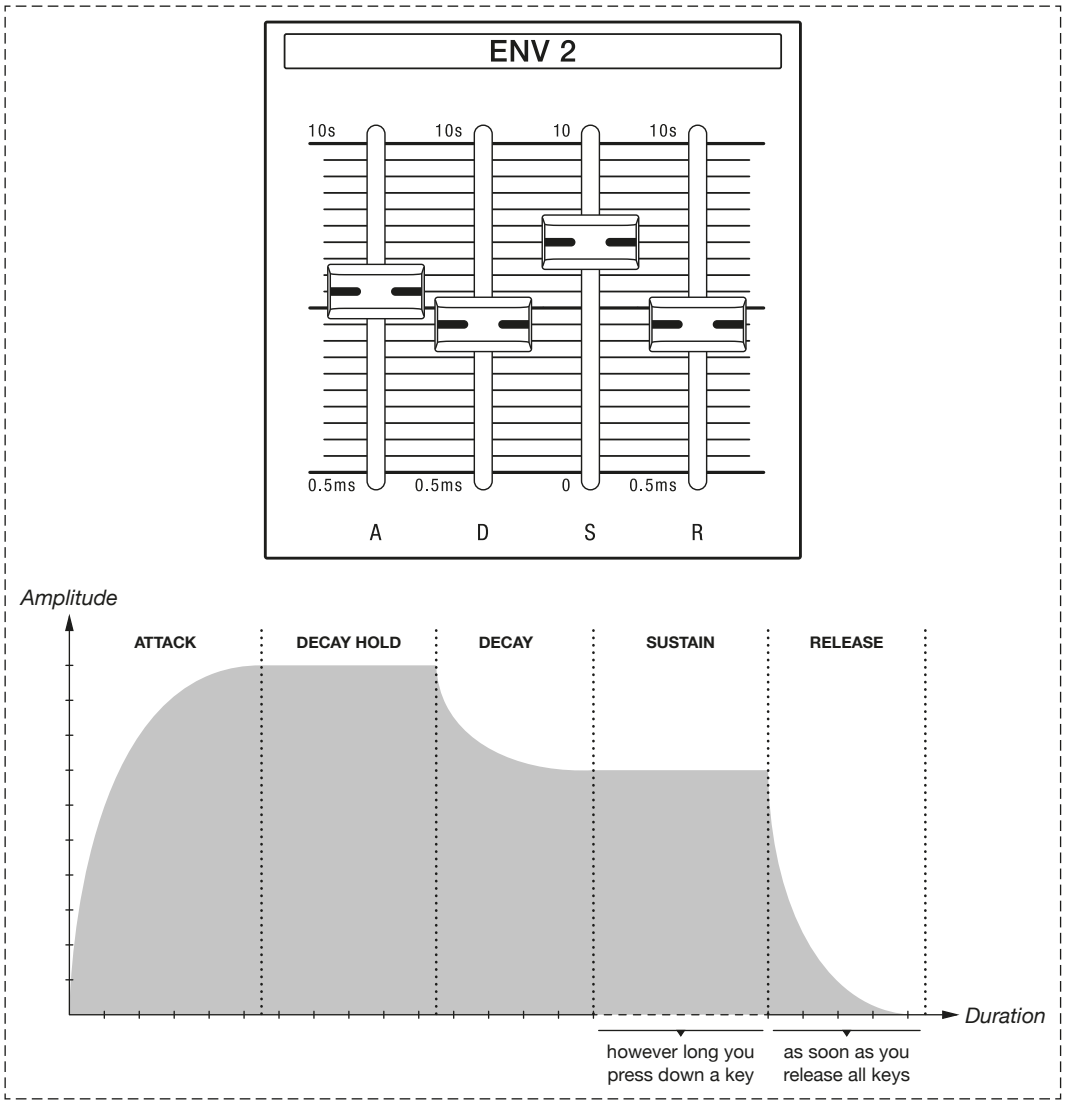


Example 2.



Example 3.

ENV 2 (Envelope 2)



Envelope 2 setting and a diagram of the resulting envelope shape.

A(TTACK): This fader determines the duration of the envelope's attack stage. The higher the setting, the slower the attack time and the longer it takes for the envelope to reach its maximum level. The attack stage can be as short as 1 millisecond or as long as 10 seconds.

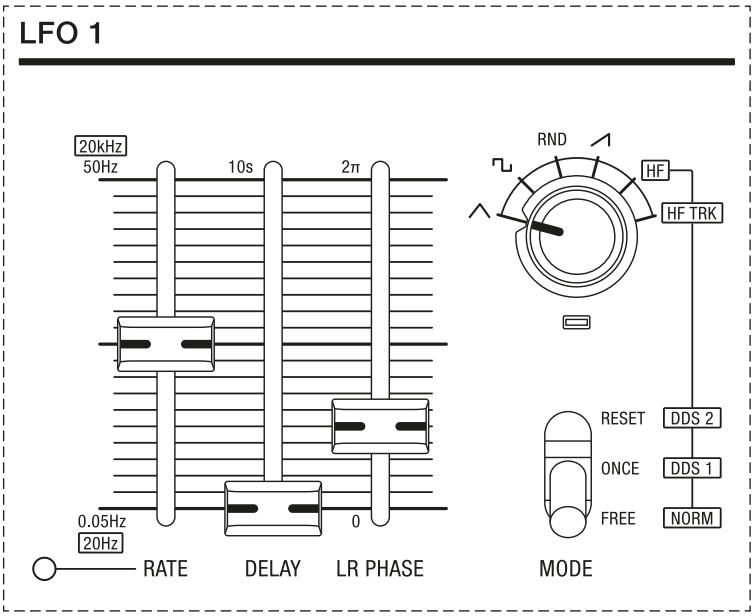
D(ECAY) H(OLD): In shift mode, you can use the **DECAY** fader to adjust the time it takes for the decay stage to begin after the attack stage reached its peak. The decay hold stage can be as long as 10 seconds. At its minimum setting this parameter has no effect, i.e. the envelope then behaves as if it had only four stages (attack, decay, sustain and release).

D(ECAY): This fader determines the duration of the envelope's decay stage. The higher the setting, the longer it takes for the envelope to travel from its maximum level to the level determined by the sustain fader. The decay stage can be as short as 1 millisecond or as long as 10 seconds.

S(USTAIN): This fader determines what level the envelope will be held at when you hold a note past the decay stage. This is the only envelope parameter that is not tied to a duration but to a level. The duration of the sustain stage always depends on how long you hold a note.

R(ELEASE): This fader determines the duration of the envelope's release stage. The higher the setting, the slower the release time and the longer it takes for the envelope to fade out after you release a key. The release stage can be as short as 1 millisecond or as long as 10 seconds.

LFO 1 (Low Frequency Oscillator 1)



LFO 1 controls.

An LFO, or low frequency oscillator, is an oscillator that generates frequencies below the range of human hearing. In default mode, LFO 1 can be used to modulate the frequency of the oscillators to create vibrato effects or to modulate the VCA level to create tremolo effects.

LFO 1 can also be set to high frequencies, allowing you to use it either as a third oscillator, as a drone, or for audio rate modulation.

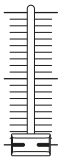
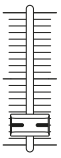
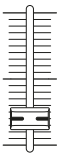
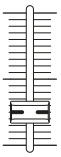
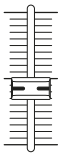
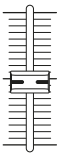
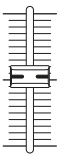
LFO 1 essentially consists of six individual LFOs: one for each of the six 'super voices'. In 12-voice non-binaural mode, each of these six LFOs is shared by two voices.

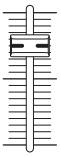
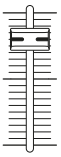
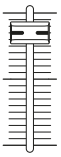
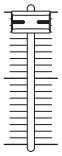
Modulation Parameters

RATE: This fader controls the rate of LFO 1. The LED at the top left of the LFO 1 section provides a visual indication of the rate.

If you enable the **SYNC** option in the arpeggiator and sequencer section, the rate of LFO 1 will be synchronised to either the internal clock as set by the **TEMPO** control (see page 94) or to an external MIDI clock signal (see pages 94-95). When synchronised, you can use the **RATE** fader to adjust the duty cycle duration of LFO 1's waveform in clock divisions relative to the internal or external tempo.

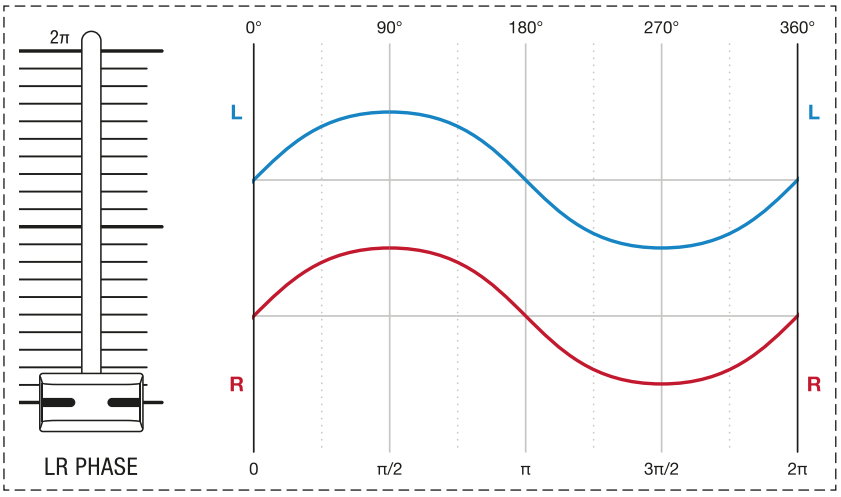
The following table lists the clock divider settings for the synchronised LFO 1 rate:

Setting	Timing Division	LFO 1 Cycle Duration
	8 whole notes	32 beats
	4 whole notes	16 beats
	2 whole notes	8 beats
	Whole note	4 beats
	1/2 note	2 beats
	Dotted 1/4 note	1 1/2 beats
	1/2 note triplet	1/3 of 4 beats
	1/4 note	1 beat
	Dotted 1/8 note	3/4 of 1 beat
	1/4 note triplet	1/3 of 2 beats
	1/8 note	1/2 of 1 beat

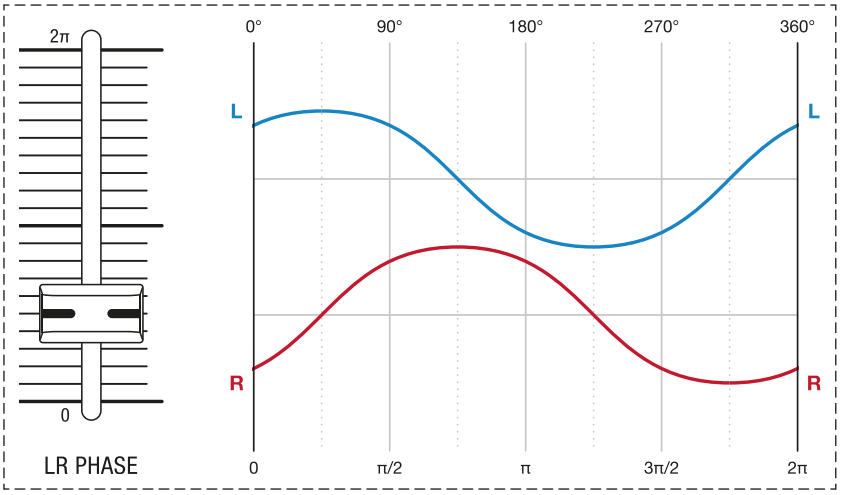
Setting	Timing Division	LFO 1 Cycle Duration
	Dotted 1/16 note	3/8 of 1 beat
	1/8 note triplet	1/3 of 1 beat
	1/16 note	1/4 of 1 beat
	Dotted 1/32 note	3/16 of 1 beat
	1/16 note triplet	1/6 of 1 beat

DELAY: This fader determines the time it takes for the LFO modulation to start affecting the sound as soon as you play a note.

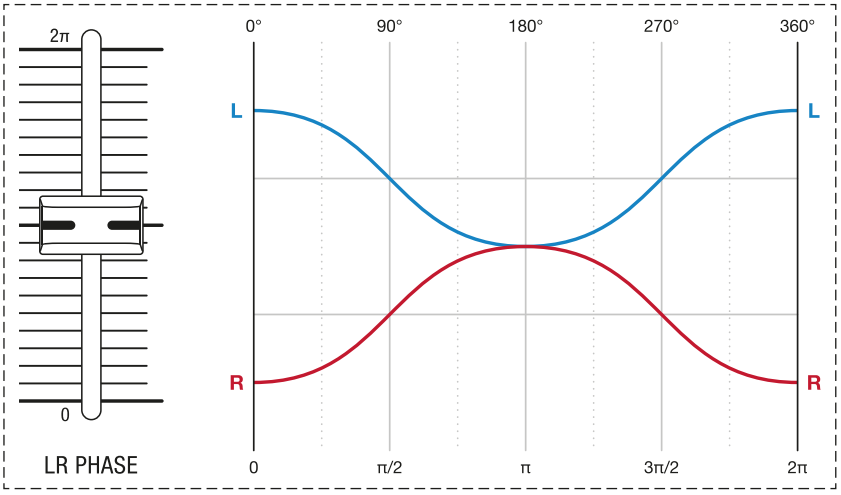
LR PHASE: In binaural mode, this fader controls the left-right channel phase relationship, in other words the effect of LFO 1 on the stereo field. With this single control, you can induce complex stereo modulations of the low-pass filter's cutoff frequency, the VCA level, and DDS 2's pitch and pulse width.



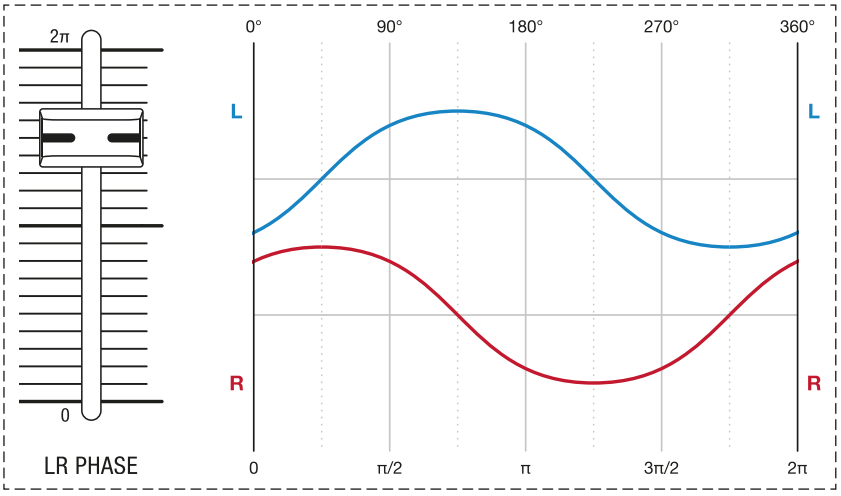
The left-right phase when **LR PHASE** is set to 0% (0).



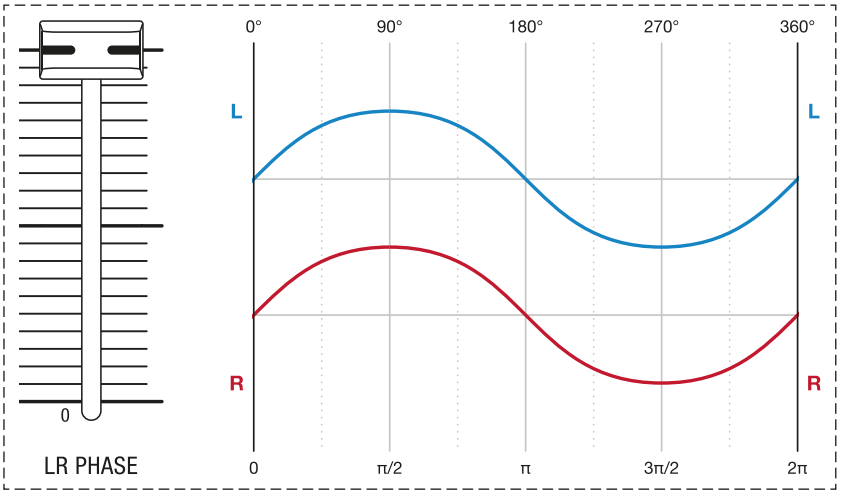
The left-right phase when **LR PHASE** is set to 25% (π/2).



The left-right phase when **LR PHASE** is set to 50% (π).



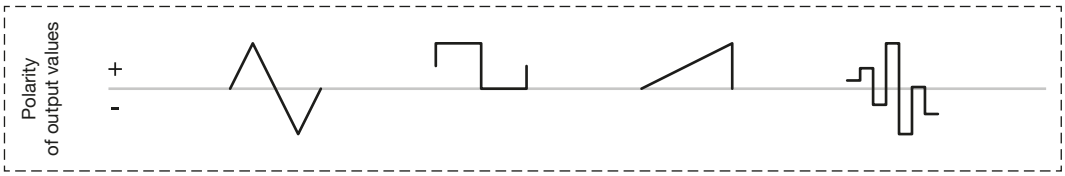
The left-right phase when **LR PHASE** is set to 75% (3π/2).



The left-right phase when **LR PHASE** is set to 100% (2π).

PAN: When binaural mode is disabled, the **LR PHASE** fader turns into a pan spread control, allowing you to adjust how far the non-binaural voices are spread in the stereo panorama. At its lowest setting, all voices are almost centred. At its highest setting, all voices are alternately hard-panned between the left and right channels.

WAVEFORM: In low-frequency mode, this rotary switch allows you to choose from four different waveforms: triangle, square, random or sawtooth.



LFO 1 waveforms.

- **Triangle** can be used to produce vibrato effects as it alternates equally between positive and negative values. This is a bipolar waveform.
- **Square** and **sawtooth** generate positive values that allow for pulsating sounds or modulations. The square wave can also be used to create trill-like effects at higher rates.
- **Random** produces random positive or negative values for the duration of one duty cycle. This waveform can be used to create either subtle movements or wild effects. With **RATE** set to its highest value, sample & hold will generate a white noise signal.
- **HF:** Selecting this option enables high-frequency mode, which allows for rates between 20 Hz and 20 kHz. In this mode, LFO 1 can be used either as a drone or as a constant modulation source for audio rate modulation. By default, LFO 1 is set to a sine wave in this mode.

- **HF TRK:** Selecting this option enables high-frequency mode with key tracking. In this mode, LFO 1 can be used either as a third oscillator or as a dynamic modulation source for audio rate modulation. The tuning of LFO 1 can be matched to that of DDS 1 and DDS 2 with the **RATE** fader. By default, LFO 1 is set to a sine wave in this mode.

To extend LFO 1's capabilities, you can employ a cunning trick and make use of the Super 6's 'Battwave' modification. This allows you to use the waveform that DDS 1 is currently set to (including the alternative DDS 1 waveforms) as LFO 1's waveform. Note that this doesn't include DDS 1's noise waveform.

*Press the **SHIFT** button and at the same time, move LFO 1's rotary control one position in any direction. The waveform, copied from DDS 1, will be used either in the high frequency modes or in low frequency mode depending on whether the position you move the rotary control one position to, is a high frequency mode or low frequency mode. Switching the rotary switch once more, without holding **SHIFT**, will resume normal operation of LFO with its standard waveforms. This method offers you a broad palette of alternate oscillator flavours to use as LFO 1's waveform. Deluxe.*

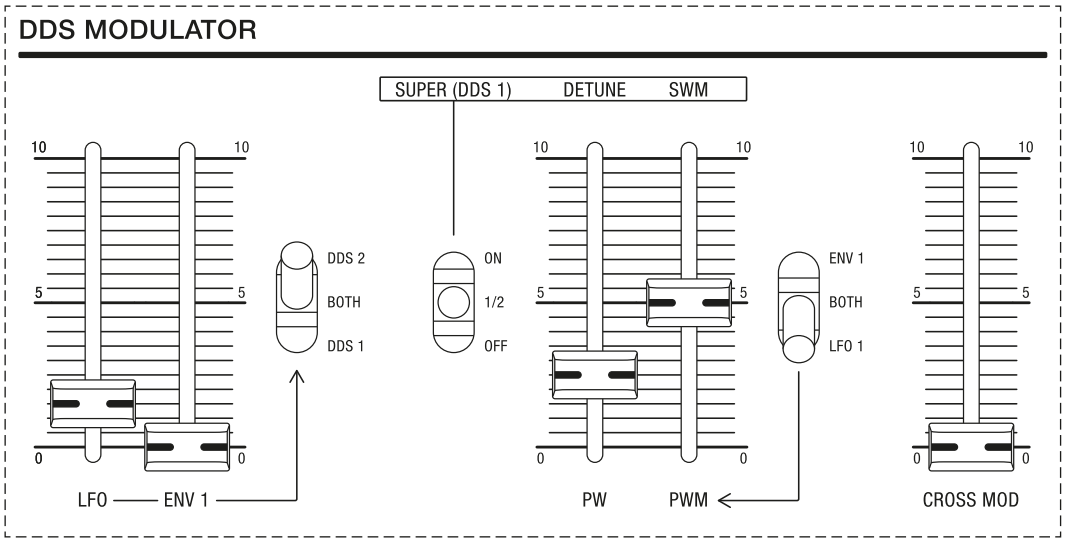
The **MODE** toggle switch allows you to choose between three types of LFO behaviours:

- **FREE:** With this setting, LFO 1 is free-running.
- **ONCE:** With this setting, LFO1 only goes through one duty cycle when you play a note. In this mode, LFO 1 can also be used as a simple envelope whose shape is determined by the selected waveform.
- **RESET:** With this setting, the phase of LFO 1 will be reset each time you play a note.

When LFO 1 is set to one of the two high-frequency modes, you can use the **MODE** toggle switch to select one of the following modes:

- **NORM:** With this setting, LFO 1 acts as a modulation source in high-frequency mode.
- **DDS 1:** With this setting, the audio signal from LFO 1 in high-frequency mode is routed to the channel of DDS 1. The output signals from LFO 1 and DDS 1 are summed, so you can crossfade between this summed signal and the audio signal from DDS 2 in the mixer section.
- **DDS 2:** With this setting, the audio signal from LFO 1 in high-frequency mode is routed to the channel of DDS 2. The output signals from LFO 1 and DDS 2 are summed, so you can crossfade between this summed signal and the audio signal from DDS 1 in the mixer section.

DDS Modulator

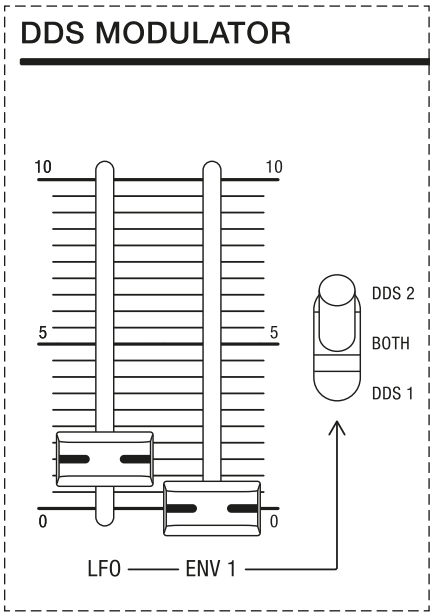


The DDS Modulator section.

This section provides dedicated controls for modulating the Super 6's oscillators in numerous ways to add tonal variety to your sounds.

The first subsection contains controls for modulating the pitch of one or both oscillators. The second subsection allows you to set and modulate parameters specific to both DDS 1 and the pulse wave of DDS 2. The third subsection controls the amount of cross modulation applied.

Modulation Parameters



The pitch modulation controls in the DDS Modulator section.

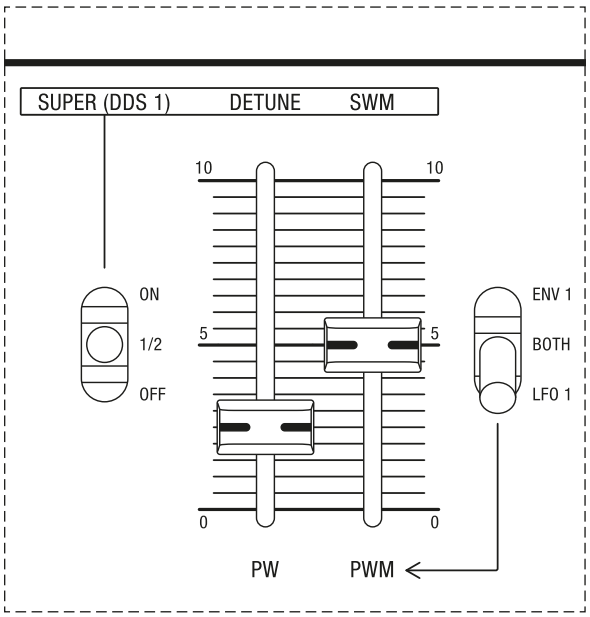
LFO 1: This fader controls the amount of pitch modulation by LFO 1.

ENV 1: This fader controls the amount of pitch modulation by envelope 1.

The oscillator selector toggle switch allows you to select the modulation destination for pitch modulation controlled by LFO 1 and/or envelope 1:

- **DDS 1:** With this setting, the modulation is mapped to DDS 1.
- **1 + 2:** With this setting, the modulation is mapped to both oscillators.
- **DDS 2:** With this setting, the modulation is mapped to DDS 2.

Binaural pitch modulation controlled by LFO 1 can only be achieved with DDS 2 because its phase is reset to zero each time you play a note, whereas DDS 1 is free-running. When an oscillator is free-running, there is no fixed starting point from which to offset its phase.

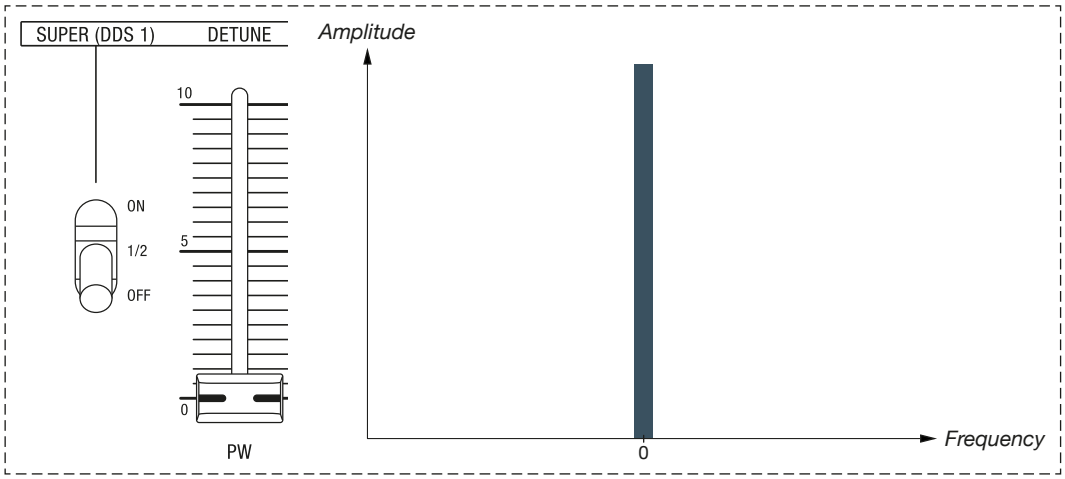


The super mode, super wave modulation and pulse width controls in the DDS Modulator section.

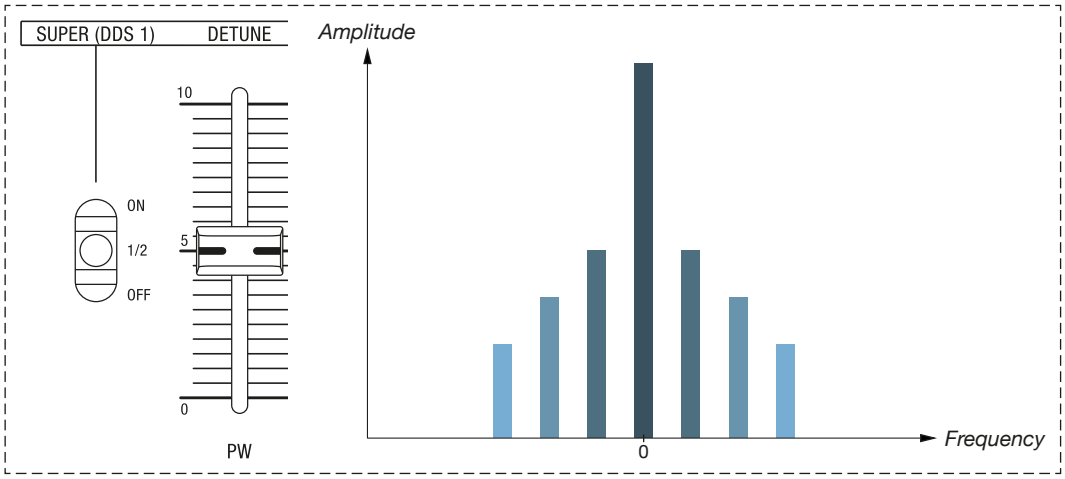
The **SUPER (DDS 1)** toggle switch enables you to activate super mode. Super mode is a unique feature that takes advantage of the Super 6's stereo signal path and the fact that DDS 1 contains seven free-running oscillators: a centroid oscillator and six 'sister' oscillators. In both available super modes, DDS 1's six 'sister' oscillators can be dynamically de-phased in the stereo field, resulting in wide and thick single oscillator sounds. You have the choice between three options:

- **OFF:** Selecting this option disables super mode. In this mode, the **DETUNE** and **SWM** parameters have no effect.

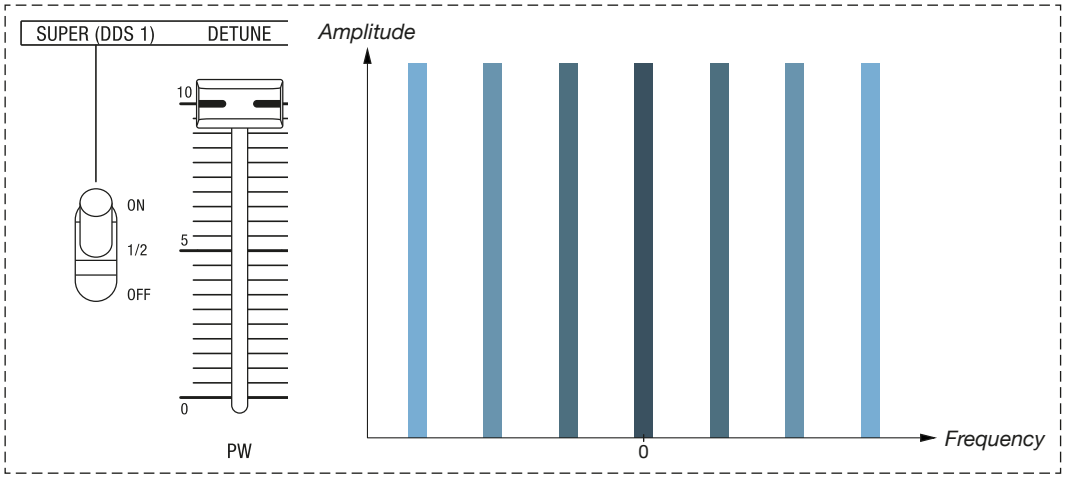
- **1/2:** Selecting this option enables super mode for DDS 1 at half depth. In addition, the phase of the DDS 1 waveforms is reset with each key press. This is useful for flanging sounds when **DETUNE** is set to higher values, or for punchy sounds when **DETUNE** is set to zero.
- **ON:** Selecting this option enables super mode for DDS 1 at full depth.



DDS 1's centroid oscillator. When super mode is off, only this oscillator generates a sound.



DDS 1's centroid oscillator and its six sister oscillators spread to both sides when super mode is set to 1/2 and the detune parameter is set to 50%.



DDS 1's centroid oscillator and its six sister oscillators spread to both sides when super mode is enabled at full depth and the detune parameter is set to 100%.

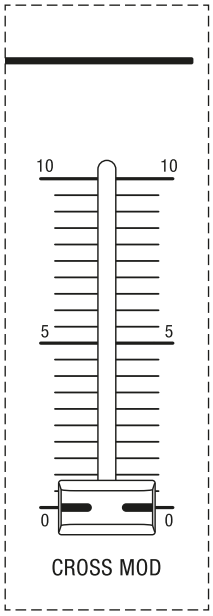
PW/DETUNE: This fader controls the pulse width of DDS 2's pulse wave. With super mode enabled, this fader also controls the amount of detune spread applied to DDS 1. Increasing the amount noticeably thickens the sound of DDS 1 as stacked versions of the same waveform are detuned.

DRIFT: In shift mode, the **PW**/**DETUNE** fader controls the amount of randomisation applied to various parameters, including the pitch of the oscillators, filters and envelopes, among others. Use **DRIFT** to either add a pleasing amount of oscillator movement or to achieve extreme detuning.

PWM/SWM: This fader controls the amount of pulse width modulation (**PWM**) applied to DDS 2's pulse wave. If super mode is engaged, this fader also controls the amount of super wave modulation (**SWM**). Super wave modulation determines how much modulation depth is applied to the DDS 1 detune spread modulation.

The toggle switch to the right allows you to select the modulation source for pulse width modulation (**PWM**) and super modulation (**SWM**):

- **LFO 1:** With this setting, LFO 1 acts as the modulation source.
- **BOTH:** With this setting, LFO 1 and ENV 1 act as the modulation sources.
- **ENV 1:** With this setting, envelope 1 acts as the modulation source.



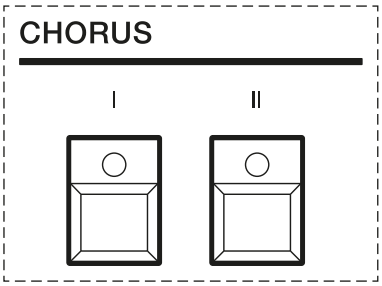
The cross modulation fader in the DDS Modulator section.

CROSS MOD: This fader controls the amount of cross modulation. By default, DDS 2 modulates DDS 1 with exponential FM. Use this parameter to create complex or bell-like timbres.

Enabling hard sync reverses the default cross modulation routing, resulting in DDS 2 being frequency modulated by DDS 1. Since in this case DDS 2 is forced to restart its duty cycle with DDS 1 while DDS 1 is responsible for the fundamental frequency, any changes applied to DDS 2 will alter the harmonic content of the sound, the depth of which is controlled with the ***CROSS MOD*** *fader. This allows for results similar to wave folding or phase modulation.*

The Super 6 features two 24-bit effects to add the finishing touch to your sounds: A classic dual-mode stereo chorus as well as a stereo delay that can be modulated and synchronised to the arpeggiator and sequencer or to an external clock source. The effects are routed in series, with the chorus being the first and the delay being the last in the signal chain.

Chorus



The chorus section.

The Super 6's chorus effect is a simple and effective design that can be used to thicken your sound even further.

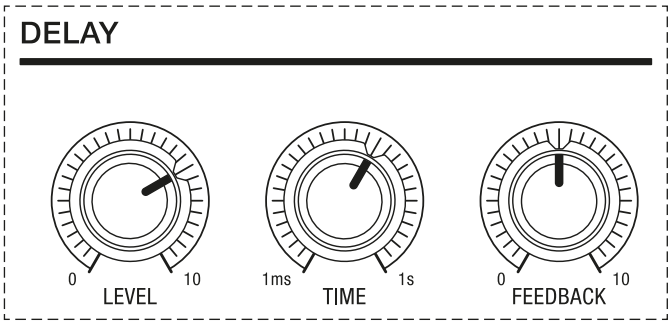
Chorus I: Selecting this option activates a subtle chorus effect.

Chorus II: Selecting this option activates a denser chorus effect that is modulated at a higher rate.

Chorus I + II: Enabling both options at the same time creates a third and even more intense chorus effect, similar to the distinctive ensemble effect used in vintage string machines.

Intense chorus effects can be useful when you only use one oscillator for a patch and therefore need a tool to thicken the overall sound. A subtle chorus is useful for adding some movement to an already rich sounding patch.

Delay



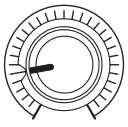
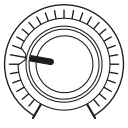
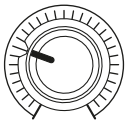
The delay section.

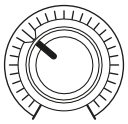
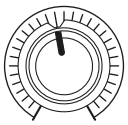
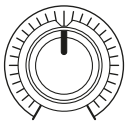
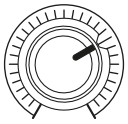
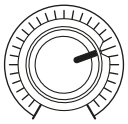
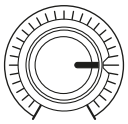
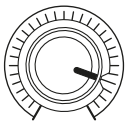
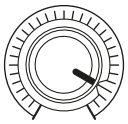
LEVEL: This control allows you to adjust to what degree the delay signal is mixed with the source signal. Higher settings result in a wet effect mix, while lower settings emphasise the dry signal.

TIME: This control allows you to adjust the delay time over a range of 1 millisecond to 1 second. On the desktop model, the LED at the top right of this control provides a visual indication of the delay time.

If you enable the **SYNC** option in the arpeggiator and sequencer section, the delay time will be synchronised to either the internal clock as set by the **TEMPO** control (see page 94) or an external MIDI clock signal (see pages 94-95). When synchronised, the **TIME** control enables you to adjust the delay time in clock divisions that are relative to the internal or external tempo.

The following table lists the clock divider settings for the synchronised delay time:

Setting	Timing Division	Delay Time
	1/32 note triplet	1/12 of 1 beat
	Dotted 1/64 note	3/32 of 1 beat
	1/32 note	1/8 of 1 beat
	1/16 note triplet	1/6 of 1 beat

Setting	Timing Division	Delay Time
	Dotted 1/32 note	3/16 of 1 beat
	1/16 note	1/4 of 1 beat
	1/8 note triplet	1/3 of 1 beat
	Dotted 1/16 note	3/8 of 1 beat
	1/8 note	1/2 of 1 beat
	1/4 note triplet	1/3 of 2 beats
	Dotted 1/8 note	3/4 of 1 beat
	1/4 note	1 beat
	1/2 note triplet	1/3 of 4 beats
	Dotted 1/4 note	1 1/2 beats
	1/2 note	2 beats
	Whole note triplet	1/3 of 8 beats

FEEDBACK: This control allows you to adjust how long the delay signal is repeated. Low settings result in fewer repeats, which is useful for creating slapback effects in conjunction with short delay times. When this control is turned fully clockwise, the delay signal is repeated indefinitely with no decay or degradation.

Delay Freeze

The Super 6 also features a delay freeze function that enables you to create sound-on-sound loops while you are performing. Follow the steps below to access this feature:

1. Turn off the Super 6.
2. Connect a dual footswitch to the sustain pedal input or use a stereo splitter (tip = sustain, ring = freeze) to add delay freeze. Alternatively, connect a sustain pedal “one click” into the sustain pedal input.
3. Power cycle the Super 6. The connected footswitch will auto-calibrate for polarity when the instrument is switched on.

To add notes or chords to the delay loop, release the footswitch while playing. Once you hold down the footswitch, new notes will no longer be added to the delay loop while its current content loops endlessly.

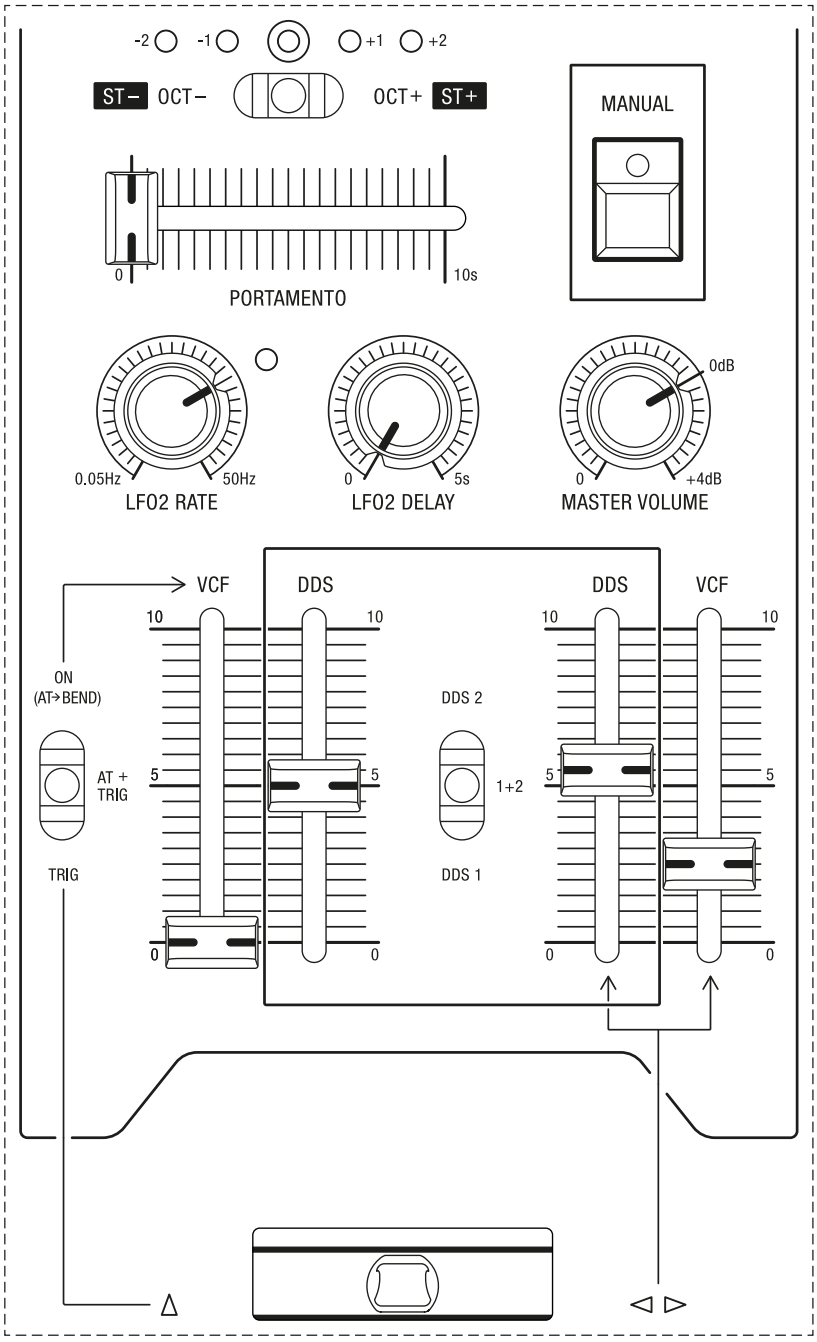
Use the **LEVEL** control to set the level of the delay loop, the **TIME** control to set the length of the delay loop and the **FEEDBACK** control to ensure that looped notes are repeated at a constant level. Delay freeze works best with long delay times and moderate amounts of delay feedback.

When using either a stereo splitter for two footswitches or one dual footswitch, the right pedal will be assigned to the delay freeze function while the left pedal will operate as a regular sustain pedal.

You can also control the delay freeze function via MIDI. It sends and receives MIDI data via CC69.

PERFORMANCE CONTROL SECTION (KEYBOARD MODEL)

Conceived as a true performance instrument that will greatly enhance your expressiveness, the Super 6 features a comprehensive performance control section in addition to its a responsive 49-note Fatar keyboard that is velocity sensitive and responds to aftertouch.

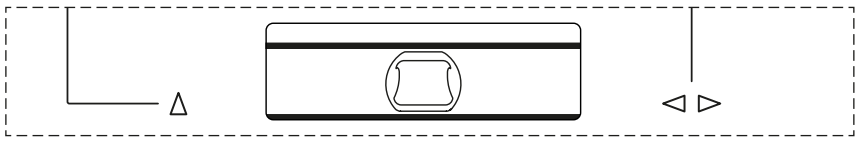


The performance control section of the keyboard model.

The performance control section allows for a number of different modulations and features immediate and versatile controls that can be easily accessed and adjusted while playing.

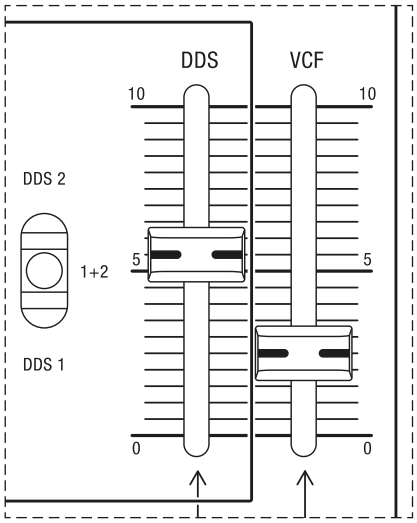
The Bender

The bender can be used to modulate both the pitch of the oscillators and the cutoff frequency of the low-pass filter. It responds to horizontal (left/right) as well as vertical (upwards) movements.



The bender.

A pressure pad is used to measure the vertically applied force and convert it to a modulation amount. The lever is not designed to be pushed up in the same way it can be moved sideways.



The section relevant to the bender assignment.

DDS: This fader controls how much the bender affects the pitch of the oscillators. The maximum pitch-bend range is one octave.

The oscillator selector toggle switch determines which oscillator is affected by the pitch modulation controlled by horizontal bender movements. You have the choice between three options:

- **DDS 1:** With this setting, only the pitch of the first oscillator will be affected by the bender.
- **1 + 2:** With this setting, the pitch of both oscillators will be affected by the bender.
- **DDS 2:** With this setting, only the pitch of the second oscillator will be affected by the bender.

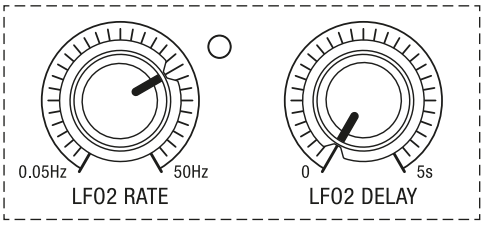
VCF: This fader controls how much the bender affects the cutoff frequency of the low-pass filter. With this fader set to the highest position, the filter can be fully opened or fully closed when the bender is moved horizontally.

LFO 2 (Low Frequency Oscillator 2)

The second LFO generates a triangle wave and can be used to modulate the frequency of the oscillators to produce a vibrato effect, or to modulate the low-pass filter's cutoff frequency for periodic harmonic changes.

LFO 2 can be triggered by pushing the bender upwards or by applying aftertouch. In addition, it can be switched on permanently. The type and amount of modulation controlled by LFO 2 is determined by the faders **DDS** and **VCF** located below LFO 2's primary controls.

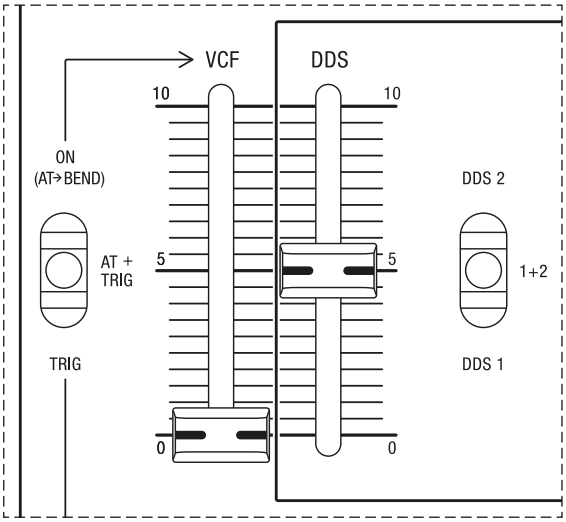
LFO 2 essentially contains twelve individual LFO modules that can be polyphonically modulated to have different rates per voice in both binaural and non-binaural modes. If, for example, you use ENV 1 to modulate the rate of LFO 2, the rate of LFO 2 will be modulated according to the envelope's settings upon each key press. The phase of the twelve LFO modules can be synchronised by toggling the the leftmost toggle switch in the performance control section.



The primary LFO 2 controls.

LFO 2 RATE: This control determines the rate of LFO 2. The LED at the top right of this control provides a visual indication of the rate.

LFO 2 DELAY: This control determines the time it takes for the LFO modulation to start affecting the sound as soon as you play a note.



The LFO 2 trigger and destination controls.

The leftmost toggle switch determines how LFO 2 is triggered:

- **TRIG:** With this setting, pushing the bender upwards will trigger LFO 2.
- **AT + TRIG:** With this setting, both aftertouch and pushing the bender upwards will trigger LFO 2. If you use the bender and apply pressure to a key at the same time, only the gesture with the greater effect on triggering LFO 2 will control the modulation depth.
- **ON (AT -> BEND):** With this setting, LFO 2 is permanently on. Additionally, aftertouch is now set to trigger the same modulations controlled by horizontal bender movements.

VCF: This fader controls the amount by which LFO 2 modulates the low-pass filter's cutoff frequency.

DDS: This fader controls the amount by which LFO 2 modulates the pitch of the oscillators.

The oscillator selector toggle switch determines which oscillator is affected by the pitch modulation controlled by LFO 2. You have the choice between three options:

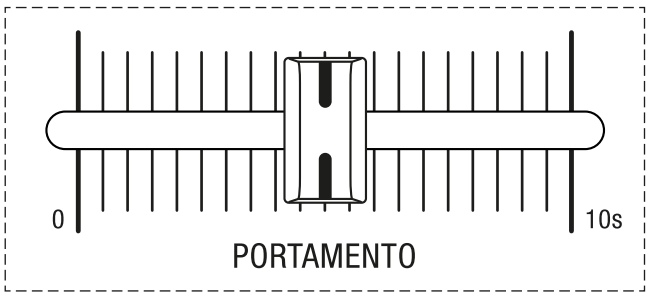
- **DDS 1:** With this setting, only the pitch of the first oscillator will be affected by LFO 2.
- **1 + 2:** With this setting, the pitch of both oscillators will be affected by LFO 2.
- **DDS 2:** With this setting, only the pitch of the second oscillator will be affected by LFO 2.

Portamento

When portamento is enabled, the pitches of the notes you play slide from one note to another. The higher the portamento time, the longer it takes for a note to slide to the pitch of the following note.

The portamento time is also determined by the intervals between the played notes. Smaller intervals result in faster pitch slides, while larger intervals result in slower pitch slides.

When you change chords, each note slides at a different rate depending on the pitch of each note and the intervals between the triggered voices.



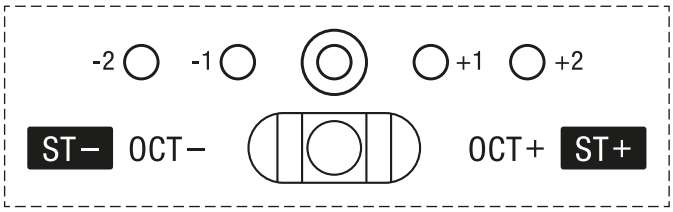
The portamento fader.

PORTAMENTO: This fader controls the amount of time it takes to slide from one pitch to the next. When set to the leftmost position, portamento will have no effect. When set to the rightmost position, the portamento time for an octave interval is 10 seconds.

Octave Selector & Transpose Function

The Super 6's octave selector toggle switch allows you to switch octaves over a range of five octaves. The switch is spring-loaded, allowing for expressive use when playing notes.

The currently selected octave is indicated by the lit LEDs above the toggle switch, with **+2** being the highest octave and **-2** being the lowest.



The octave selector toggle switch.

In shift mode, you can use the octave selector toggle switch to transpose the pitch globally by +/- 12 semitones. As you transpose up, the two LEDs on the right start flashing. As you transpose down, the two LEDs on the left start flashing.

If you adjust the global transpose setting, the middle octave LED will continue to flash even after exiting shift mode to indicate that the default global tuning has been changed.

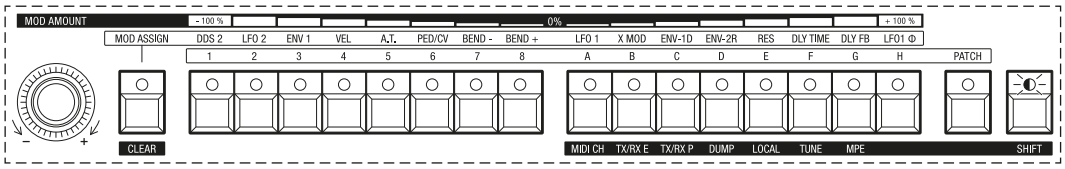
Global Fine Tune

In addition to the transpose function, you can also fine-tune the Super 6 globally. To adjust the global fine-tuning, press **SHIFT** and then rotate the **MOD AMOUNT** encoder.

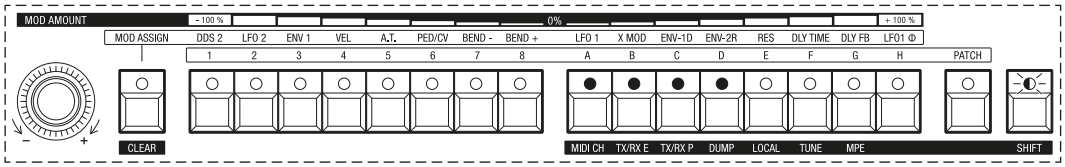
Turning the **MOD AMOUNT** encoder clockwise increases the frequency. Turning the control counter-clockwise decreases the frequency. You can adjust the global fine-tuning over a range of +/- 100 cents.

As soon as you touch or move the **MOD AMOUNT** encoder, the LEDs of buttons **1**-**8** and **A**-**H** indicate the current fine-tune setting. The printed top row above buttons **1**-**8** and **A**-**H** serves as a legend for the current fine-tune setting, indicated by the lit LEDs.

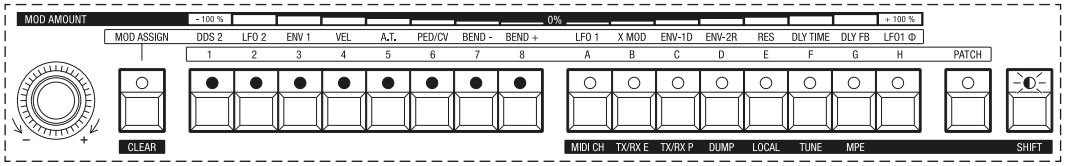
In the first example, the lit LEDs of buttons **1**-**8** and **A**-**H** indicate a global fine-tune setting of 0 cents or 440 Hz, which is the default:



In the second example, the lit LEDs of buttons **1**-**8** and **A**-**H** indicate a global fine-tune setting of +50 cents:

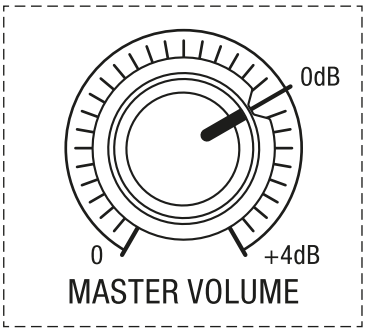


In the third example, the lit LEDs of buttons **1**-**8** and **A**-**H** indicate a global fine-tune setting of -100 cents:



If you set the global fine-tune to anything other than 440 Hz, the middle octave LED will continue to flash even after exiting shift mode to indicate that the default global tuning has been changed.

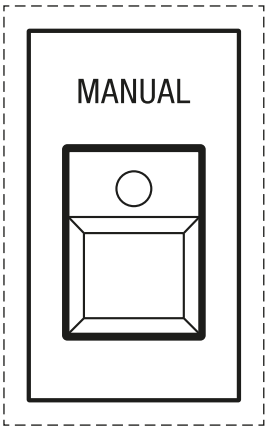
Master Volume



The master volume control.

MASTER VOLUME: This parameter controls the Super 6's master volume as well as the headphone volume. Turning the control fully clockwise increases the volume to a maximum of +4 decibels. This is the only control whose setting is not stored with a patch.

Manual Mode



The manual button.

MANUAL: Press this button to enter manual mode, which ignores the current patch settings and prompts the Super 6 to respond to the actual front panel control settings. To return to patch mode, simply press the **MANUAL** button again or press the **PATCH** button.

ADDITIONAL CONTROLS & PARAMETERS (DESKTOP MODEL)

Pitch Bend Control

You can use an external MIDI controller's pitch bend wheel to control the pitch of the oscillators by following the steps below:

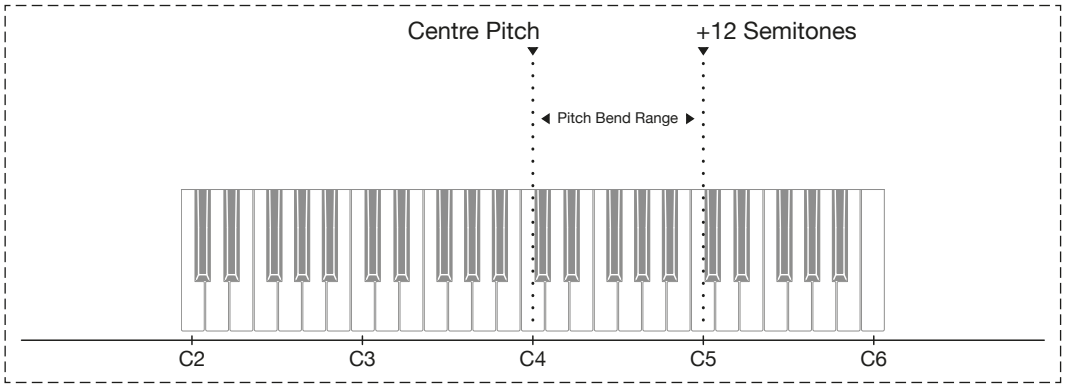
1. Press the **MOD ASSIGN** button. Its LED will flash, indicating that you have entered the modulation matrix.

2. Press and hold the button labelled **BEND** to select a pitch bend wheel as the modulation source.

3. While holding the **BEND** button, select the note that you would like the bend to reach by referring to C4 as the centre pitch. For example, if you press D4, the bend range is 2 semitones, so you can bend from A#3 to D4 when playing a C4. If you press C5 while holding down the **BEND** button, the bend range is one octave.

4. Once you have selected your preferred pitch bend range, release the **BEND** button.

5. Press the **MOD ASSIGN** button until its LED turns off to exit the modulation matrix.



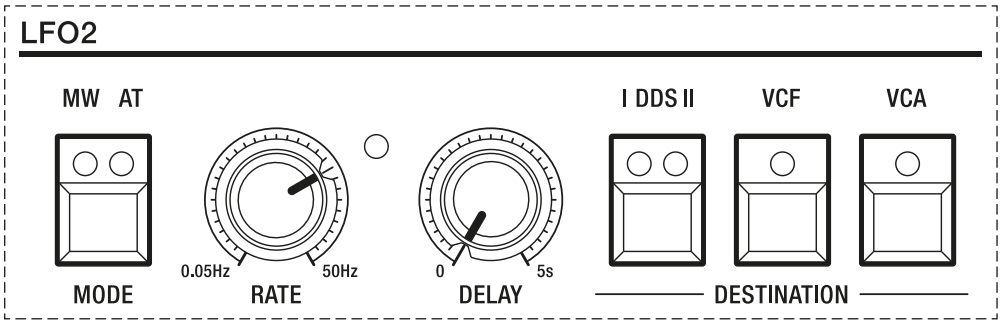
Using a 49-note keyboard as a reference for defining the pitch bend range.

The maximum pitch bend range is one octave or 12 semitones.

LFO 2 (Low Frequency Oscillator 2)

The second LFO generates a triangle wave and can be used to modulate the frequency of the oscillators to produce a vibrato effect, to modulate the low-pass filter's cutoff frequency for periodic harmonic changes, or to modulate the VCA level to create tremolo style effects.

LFO 2 essentially contains twelve individual LFO modules that can be polyphonically modulated to have different rates per voice in both binaural and non-binaural modes. If, for example, you use ENV 1 to modulate the rate of LFO 2, the rate of LFO 2 will be modulated according to the envelope's settings upon each key press. The phase of the twelve LFO modules can be synchronised by toggling the the leftmost toggle switch in the performance control section.



The LFO 2 section.

Modulation Parameters

MODE: This button allows you to determine how LFO 2 modulations will be triggered from an external MIDI controller that features aftertouch and a modulation wheel:

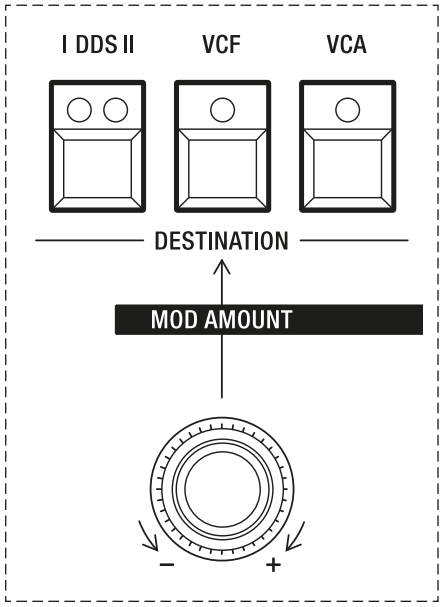
- **MW:** With this setting, a modulation wheel will trigger LFO 2.
- **MW + AT:** With this setting, both a modulation wheel and aftertouch will trigger LFO 2. If you use a modulation wheel and apply pressure to a key at the same time, only the gesture with the greater effect on triggering LFO 2 will control the modulation depth.
- **NONE:** When no LED is lit, LFO 2 is permanently on. Additionally, aftertouch is now set to trigger the same modulations controlled by a pitch bend wheel.

RATE: This control determines the rate of LFO 2. The LED at the top right of this control provides a visual indication of the rate.

DELAY: This control determines the time it takes for the LFO modulation to start affecting the sound as soon as you play a note.

Assigning Modulation Destinations to LFO 2

On the right side of the LFO 2 section you can determine which modulation destination should be affected by LFO 2. Notice how an arrow is pointing from the **MOD AMOUNT** encoder to the modulation destinations in the LFO 2 section. You can use this control to dial in the modulation depth that should be applied to each of the four destinations.



The modulation destinations in the LFO 2 section and the **MOD AMOUNT** control.

To assign a modulation destination to LFO 2:

1. Press and hold any of the three modulation destination buttons in the LFO 2 section (**I DDS II**, **VCF** or **VCA**) to select a modulation destination. After holding any of these buttons for more than 2 seconds, its LED will begin flashing, indicating that a modulation mapping has been created. Note that when you press the **I DDS II** button you can toggle between three options: DDS 1, DDS 2 or both oscillators.

2. While holding down one of the modulation destination buttons, turn the **MOD AMOUNT** encoder to dial in the modulation depth you would like to apply. You can adjust the modulation depth over a range of -100% (negative modulation amount) to +100% (positive modulation amount). The LEDs of buttons **1**-**8** and **A**-**H** indicate the respective setting. The printed top row above buttons **1**-**8** and **A**-**H** serves as a legend for the modulation amount, indicated by the lit LEDs.

3. Release the modulation destination button. Its LED will stop flashing.

You can also use the modulation matrix to map more modulation destinations to LFO 2. See pages 85-91 for more details on how to use the modulation matrix.

The three modulation destination buttons don't only allow you to create modulation mappings. You can also use them to instantly determine whether a modulation destination should be affected by LFO 2 or not.

I DDS II: This button allows you to determine which oscillator is affected by the pitch modulation controlled by LFO 2. You have the choice between three options::

- **I:** With this setting, only the pitch of the first oscillator will be affected by LFO 2.
- **II:** With this setting, only the pitch of the second oscillator will be affected by LFO 2.
- **I + II:** With this setting, the pitch of both oscillators will be affected by LFO 2.

VCF: This button allows you to determine whether the low-pass filter's cutoff frequency will be affected by LFO 2.

VCA: This button allows you to determine whether the VCA level will be affected by LFO 2.

You may also toggle the single modulation destination buttons on and off during your playing to modify the impact of LFO 2 in real-time.

Direct Modulation Wheel Assignment

A modulation wheel (CC# 1) can be directly mapped to various parameters, rather than being limited to control the amount of LFO 2 modulation.

Follow the steps below to map the modulation wheel of your MIDI controller to a modulation destination:

1. Press and hold the LFO 2 **MODE** button to enter panel assign mode for the modulation wheel. The **MW** LED starts flashing.
2. While holding down the **MODE** button, move a fader or rotary control on the front panel. An LED scroll across buttons **1**-**8** and **A**-**H** will indicate a mapping lock.
3. Turn the **MOD AMOUNT** encoder to dial in the modulation depth. The LEDs of buttons **1**-**8** and **A**-**H** will indicate your setting.
4. Release the LFO 2 **MODE** button.

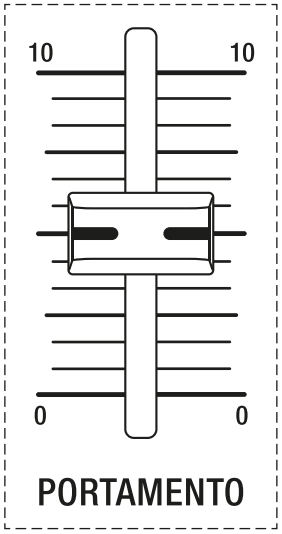
*This feature is most useful when LFO 2 is set to be permanently 'on' (**MW** and **AT** LEDs off). In this mode, LFO 2 will be triggered independently of a modulation wheel and can be conveniently assigned to a modulation destination by using the **I DDS II**, **VCF** and **VCA** buttons.*

Portamento

When portamento is enabled, the pitches of the notes you play slide from one note to another. The higher the portamento time, the longer it takes for a note to slide to the pitch of the following note.

The portamento time is also determined by the intervals between the played notes. Smaller intervals result in faster pitch slides, while larger intervals result in slower pitch slides.

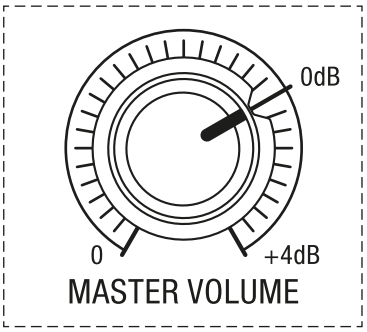
When you change chords, each note slides at a different rate depending on the pitch of each note and the intervals between the triggered voices.



The portamento fader.

PORTAMENTO: This fader controls the amount of time it takes to slide from one pitch to the next. When set to the leftmost position, portamento will have no effect. When set to the rightmost position, the portamento time for an octave interval is 10 seconds.

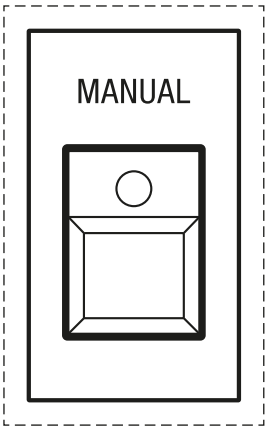
Master Volume



The master volume control.

MASTER VOLUME: This parameter controls the Super 6's master volume as well as the headphone volume. Turning the control fully clockwise increases the volume to a maximum of +4 decibels. This is the only control whose setting is not stored with a patch.

Manual Mode



The manual button.

MANUAL: Press this button to enter manual mode, which ignores the current patch settings and prompts the Super 6 to respond to the actual front panel control settings. To return to patch mode, simply press the **MANUAL** button again or press the **PATCH** button.

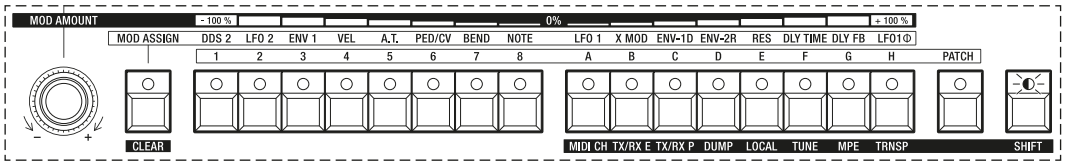
Global Fine Tune

You can also fine-tune the Super 6 globally. To adjust the global fine-tuning, press **SHIFT** and then rotate the **MOD AMOUNT** encoder.

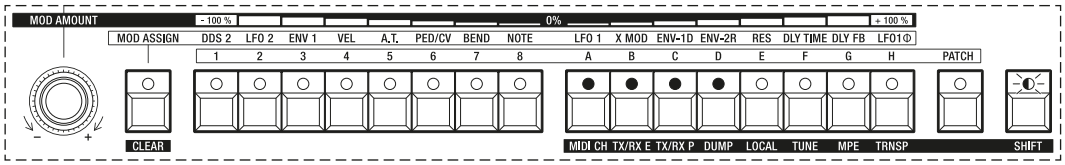
Turning the **MOD AMOUNT** encoder clockwise increases the frequency. Turning the control counter-clockwise decreases the frequency. You can adjust the global fine-tuning over a range of +/- 100 cents.

As soon as you touch or move the **MOD AMOUNT** encoder, the LEDs of buttons **1**-**8** and **A**-**H** indicate the current fine-tune setting. The printed top row above buttons **1**-**8** and **A**-**H** serves as a legend for the current fine-tune setting, indicated by the lit LEDs.

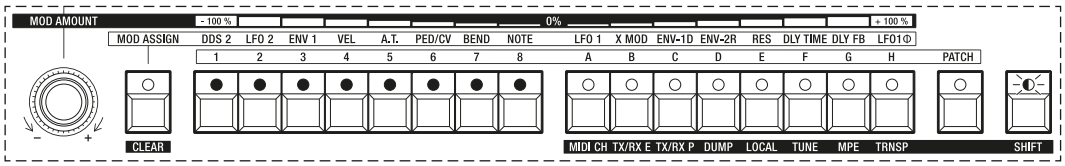
In the first example, the lit LEDs of buttons **1**-**8** and **A**-**H** indicate a global fine-tune setting of 0 cents or 440 Hz, which is the default:



In the second example, the lit LEDs of buttons **1**-**8** and **A**-**H** indicate a global fine-tune setting of +50 cents:



In the third example, the lit LEDs of buttons **1**-**8** and **A**-**H** indicate a global fine-tune setting of -100 cents:

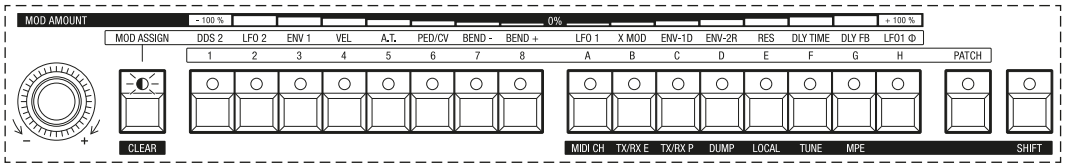


If you set the global fine-tune to anything other than 440 Hz, the middle octave LED will continue to flash even after exiting shift mode to indicate that the default global tuning has been changed.

USING THE MODULATION MATRIX

The top raised panel and the performance control section provide you with numerous options for assigning a variety of modulation sources to several different modulation destinations. You can go beyond these possibilities by using the modulation matrix.

The modulation matrix is accessed via the front panel section which contains the numbered and lettered select buttons and there are two different methods of creating modulation mappings that are described below.



The front panel section relevant to the modulation matrix.

Before we dive into the different ways you can create modulation mappings, it's important to note how buttons **1**-**8** and **A**-**H** behave in the context of the modulation matrix.

The modulation matrix is entered by pressing the **MOD ASSIGN** button. Its LED will then flash, indicating that you are now in modulation assign mode. In this mode, buttons **1**-**8** represent eight modulation sources, while buttons **A**-**H** represent eight modulation destinations.

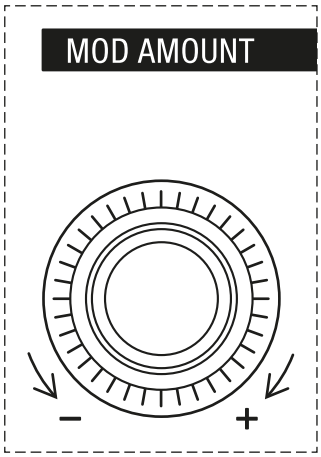
The following modulation sources and destinations are available to you on the keyboard model:

	Modulation Source		**Modulation Destination**
1	DDS 2	A	LFO 1 Rate
2	LFO 2	B	Cross Modulation
3	Envelope 1	C	Envelope 1 Decay
4	Velocity	D	Envelope 2 Release
5	Aftertouch	E	Filter Resonance
6	Expression Pedal/CV	F	Delay Time
7	Bender pushed to the left (-)	G	Delay Feedback
8	Bender pushed to the right (+)	H	LR Phase

The following modulation sources and destinations are available to you on the desktop model:

	Modulation Source		Modulation Destination
1	DDS 2	A	LFO 1 Rate
2	LFO 2	B	Cross Modulation
3	Envelope 1	C	Envelope 1 Decay
4	Velocity	D	Envelope 2 Release
5	Aftertouch	E	Filter Resonance
6	Expression Pedal/CV	F	Delay Time
7	Pitch Bend Wheel	G	Delay Feedback
8	Note Number	H	LR Phase

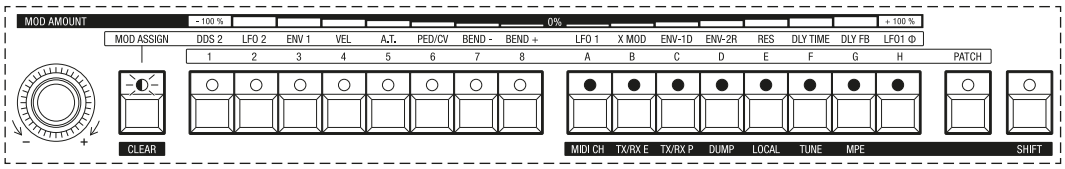
The modulation amount, i.e. the depth at which a modulation source modulates a modulation destination, is determined by using the **MOD AMOUNT** encoder.



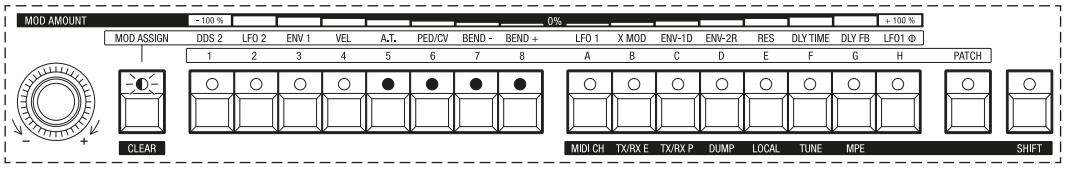
The endless encoder used for dialling in the modulation amount.

By turning the **MOD AMOUNT** encoder, you can adjust the modulation depth over a range of -100% (negative modulation amount) to +100% (positive modulation amount). The LEDs of buttons **1**-**8** and **A**-**H** indicate the respective setting. The printed top row above buttons **1**-**8** and **A**-**H** serves as a legend for the modulation amount, indicated by the lit LEDs.

In the first example, the lit LEDs of buttons **1**-**8** and **A**-**H** indicate a modulation amount of +100%:



In the second example, the lit LEDs of buttons **1**-**8** and **A**-**H** indicate a modulation amount of -50%:



Matrix Destination Mappings

This method of creating modulation mappings allows you to assign any of the eight fixed sources to any of the eight fixed destinations with individual modulation amounts:

1. Press the **MOD ASSIGN** button. Its LED will start flashing, indicating that you have entered the modulation matrix.

2. Flashing LEDs of buttons **1**-**8** indicate which modulation sources are actively modulating, while flashing LEDs of buttons **A**-**H** indicate which modulation destinations are currently modulated.

3. To create a modulation mapping by first selecting a source, press any modulation source button (**1**-**8**). Its LED will then light up. If this source is actively modulating any destination in the matrix, the LEDs of the respective buttons (**A**-**H**) will flash.

   Press any destination button to create a mapping. The LEDs of the source and destination buttons will then light up continuously, indicating the mapping lock.

4. To create a modulation mapping by first selecting a destination, press any modulation destination button (**A**-**H**). Its LED will then light up. If this destination is currently modulated by any source in the matrix, the LEDs of the respective buttons (**1**-**8**) will flash.

   Press any source button to create a mapping. The LEDs of the source and destination buttons will then light up continuously, indicating the mapping lock.

5. After creating a modulation mapping, use the **MOD AMOUNT** encoder to dial in the modulation amount. The LEDs of buttons **1**-**8** and **A**-**H** indicate the respective setting.

6. To return to the initial ‘view’ of the modulation matrix as described in step 2, press the **MOD ASSIGN** button. Otherwise, press any source or destination button to create a new mapping or to edit an existing mapping.

7. Press either the **MOD ASSIGN** button once more or the **PATCH** button to exit the modulation matrix and return to patch mode.

*The modulation sources represented by buttons **1**-**8** can also modulate destinations other than those represented by buttons **A**-**H** (see “Direct Parameter Mappings”).*

Direct Parameter Mappings

This alternative method of creating modulation mappings allows you to assign any modulation source present in the modulation matrix to front panel parameters that are not present in the modulation matrix:

1. Press the **MOD ASSIGN** button. Its LED will start flashing, indicating that you have entered the modulation matrix.

2. Flashing LEDs of buttons **1**-**8** indicate which modulation sources are actively modulating, while flashing LEDs of buttons **A**-**H** indicate which modulation destinations are currently modulated.

3. To create a direct parameter mapping, press and hold any of the eight modulation source buttons (**1**-**8**).

4. While holding the selected modulation source button, move any patch-related parameter on the front panel you wish to modulate. An LED scroll across buttons **1**-**8** and **A**-**H** indicates a mapping lock.

5. After creating a modulation mapping, use the **MOD AMOUNT** encoder to dial in the modulation amount. The LEDs of buttons **1**-**8** and **A**-**H** indicate the respective setting.

6. To return to the initial ‘view’ of the modulation matrix as described in step 2, press the **MOD ASSIGN** button. Otherwise, press and hold one of the eight modulation source buttons to create a new mapping.

7. Press either the **MOD ASSIGN** button once more or the **PATCH** button to exit the modulation matrix and return to patch mode.

*When the modulation sources represented by buttons **1**-**8** are modulating destinations outside the modulation matrix, the respective LEDs are flashing as well.*

If a parameter cannot be assigned as a modulation destination, there will be no LED scroll. In general, only parameters represented by continuous controls are assignable. This excludes toggle and rotary switches.

The following table lists the modulation destinations available when direct parameter mappings are created:

	Modulation Destination		Modulation Destination
1	DDS 2 Tune	12	Envelope 1 Release
2	Oscillator Mix	13	Envelope 2 Attack
3	VCF Cutoff Frequency	14	Envelope 2 Decay
4	VCF Envelope Amount	15	Envelope 2 Sustain
5	VCF LFO 1 Amount	16	LFO 1 Delay
6	VCF DDS 2 Amount	17	LFO 2 Rate
7	VCA Envelope Level	18	LFO 2 Delay
8	VCA LFO 1 Amount	19	DDS Modulator LFO 1 Amount
9	Envelope 1 Hold	20	DDS 2 Pulse Width / DDS 1 Detune
10	Envelope 1 Attack	21	Portamento Time
11	Envelope 1 Sustain	22	Delay Level

Clearing Modulation Mappings

To clear all modulation mappings:

1. Press **MOD ASSIGN** to enter the modulation matrix.
2. Hold **SHIFT** and press the **MOD ASSIGN** button.
3. An LED scroll across buttons **1**-**8** and **A**-**H** indicates that all modulation assignments have been cleared.

To clear all modulation mappings from a particular modulation source:

1. Press **MOD ASSIGN** to enter the modulation matrix.
2. Press the button assigned to the desired modulation source.
3. Hold **SHIFT** and press the **MOD ASSIGN** button.
4. An LED scroll across buttons **1**-**8** and **A**-**H** indicates that all modulations controlled by the selected modulation source have been cleared.

To clear all modulation mappings to a particular modulation destination:

1. Press **MOD ASSIGN** to enter the modulation matrix.
2. Press the button assigned to the desired modulation destination.
3. Hold **SHIFT** and press the **MOD ASSIGN** button.
4. An LED scroll across buttons **1**-**8** and **A**-**H** indicates that all modulations assigned to the selected modulation destination have been cleared.

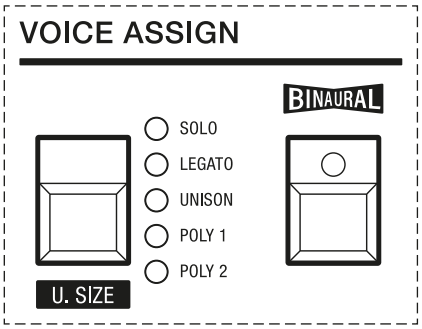
Clear Modulation Mappings Convenience Function (Desktop Only)

The **MOD AMOUNT** encoder's push function allows you to instantly clear modulation mappings in three different ways:

- Push the **MOD AMOUNT** upon entering the modulation matrix to clear all modulation mappings.
- Push the **MOD AMOUNT** encoder when a modulation source is selected to clear all modulations controlled by the selected modulation source.
- Push the **MOD AMOUNT** encoder when a modulation destination is selected to clear all modulations assigned to the selected modulation destination.

VOICE ASSIGN

In the voice assign section you can specify how the Super 6's voices are distributed when a note is played. Here you have the choice between two polyphonic and two monophonic modes, as well as various unison modes and the option to turn binaural mode on or off.



The voice assign section.

The mode button to the left allows you to choose from five different voice assign modes:

- **SOLO:** In solo mode, the Super 6 behaves like a monophonic synthesizer, meaning that only one note can be played at a time. With each note played, the envelopes are retriggered.
- **LEGATO:** In legato mode, the Super 6 behaves like a monophonic synthesizer, meaning that only one note can be played at a time. This mode differs from solo mode in that each time a note is played while playing the legato style, the envelopes are not retriggered.
- **UNISON:** In unison mode, the Super 6's voices are stacked for massive monophonic sounds. You can use the **U.SIZE** parameter to specify how the voices are stacked.
- **POLY 1:** This mode gives you full polyphony. The total number of voices depends on whether binaural mode is activated or not. When new notes are played, the release stages of all notes overlap, allowing smooth transitions between notes. This is the instrument's default mode.
- **POLY 2:** This mode gives you full polyphony. The total number of voices depends on whether binaural mode is activated or not. The release stage of overlapping notes is curtailed in this mode.

U. SIZE: Press **SHIFT** and the mode button to access the unison size settings, which allow you to determine how the voices are stacked in unison mode. The number of lit LEDs indicate how the voices are going to be handled:

- **1 LED:** With this setting, 3 binaural voices will be stacked.
- **2 LEDs:** With this setting, 6 binaural voices will be stacked.
- **3 LEDs:** With this setting, 6 binaural voices will be stacked as an octave.
- **4 LEDs:** With this setting, 6 binaural voices will be stacked as an octave and a fifth.
- **5 LEDs:** With this setting, 6 binaural voices will be stacked as a major chord.

BINAURAL: Use this button to turn binaural mode on or off. The Super 6 defaults to binaural mode, in which its 12 voices are twinned to form six stereo ‘super voices’.

The left and right channels are assigned a complete synthesizer voice. Starting with the stereo oscillators, parameters of both channels of each ‘super voice’ may be independently controlled, enabling you to create gorgeous stereo images.

When binaural mode is disabled, the Super 6 switches to a monaural signal path with 12 voices.

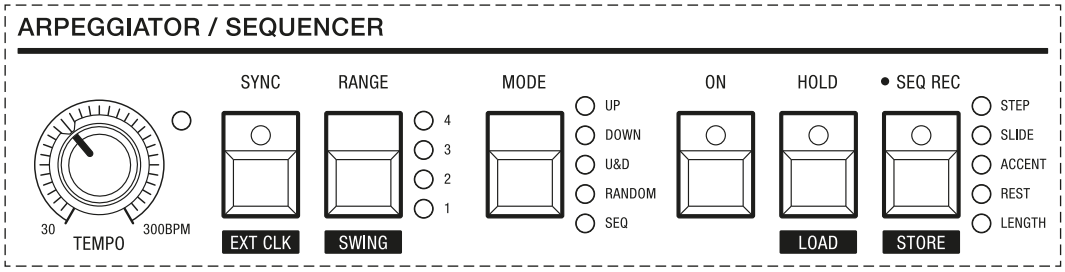
In non-binaural mode, the **LR PHASE** fader in the LFO 1 section turns into a pan spread control (**PAN**), allowing you to adjust how far the non-binaural voices are spread in the stereo panorama. At its lowest setting, all voices are almost centred. At its highest setting, all voices are alternately hard-panned between the left and right channels.

To achieve a true monaural signal:

1. Connect only the left audio output to your mixer or audio interface.
2. Disable binaural mode in either voice assign mode **POLY 1** or **POLY 2**.
3. Once the Super 6 is set to non-binaural mode, set **LR PHASE** (or **PAN** in this mode) to 100%. If you set **PAN** below 100% in non-binaural mode, this will result in phase cancellation every time the signals of both audio channels ‘overlap’.

ARPEGGIATOR & SEQUENCER

The Super 6 features a flexible arpeggiator as well as a simple yet powerful 64-step sequencer.



The arpeggiator and sequencer section.

Clock Parameters

TEMPO: This control allows you to adjust the playback speed of the arpeggiator or sequencer. The tempo can be as slow as 30 BPM or as fast as 300 BPM. The LED at the top right is your visual click track, indicating the current tempo by flashing according to the set tempo rate. When the Super 6 is synchronised to an external MIDI clock source, the **TEMPO** control allows you to adjust the playback speed in clock divisions that are relative to the external tempo, for example quarter notes, eighth notes or quarter note triplets.

SWING: Press **SHIFT** and then the **RANGE** button to access the current swing setting. You can choose from five different swing settings, the first of which is off. When set to 1, the amount of swing is subtle. Set to the other extreme (4), the swing will be very pronounced. Try using different swing amounts to find the best rhythmic feel for your arpeggio or sequence.

SYNC: With this option enabled, the rate of LFO 1 and the delay time will be synchronised to the playback speed of the arpeggiator or sequencer as set by the **TEMPO** control or an external clock source. You can then use the **RATE** fader in the LFO 1 sections and the **TIME** control in the delay section to adjust the corresponding parameter values in clock divisions. If this option is disabled, LFO 1 and the delay time will be free-running.

EXT CLK: Press **SHIFT** and then the **SYNC** button to enter the external clock settings. Once you have accessed the external clock settings, you can enable the following options:

- **BUTTON 1:** MIDI Clock Transmit. When enabled (LED lit), MIDI clock signals are transmitted.
- **BUTTON 2:** MIDI Clock Receive. When enabled (LED lit), MIDI clock signals are received. In addition, the Super 6 will respond to MIDI Start/Stop messages.

- **BUTTON 3:** MIDI Note Output. When enabled (LED lit), the arpeggiator and sequencer will output MIDI notes during playback. Please note that Local Control needs to be enabled for this option.
- **BUTTON 4:** MIDI Stop Message Receive. When enabled (LED lit), each of the two **HOLD** buttons "arms" the sequencer while the latter waits for a MIDI start message to start playback. In this mode, the sequencer will stop playback when a MIDI Stop message is received, even if the external MIDI clock is still running.

  When this option is disabled (LED flashing), each of the two **HOLD** buttons will start and stop the sequencer asynchronously. A MIDI Start message will "snap" the sequence back into the correct timing, but a MIDI Stop message will be ignored if the external MIDI clock is still running.

  In both modes, transport control is achieved via Note On/Off messages. This gives you easy control over when a sequence should start or stop, and also allows you to transpose a sequence.

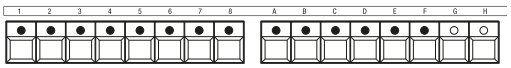
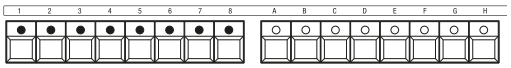
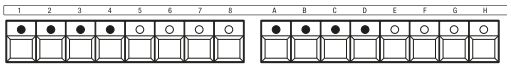
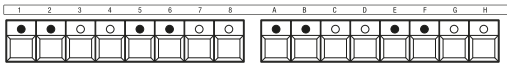
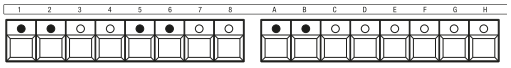
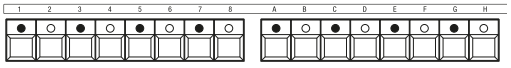
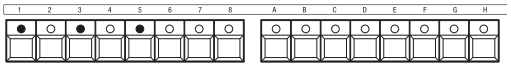
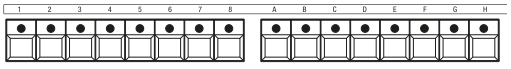
To exit the external clock settings, press **SHIFT** again.

Please note that you can either enable option 1 or option 2. As soon as you enable one of these options, the other is automatically disabled.

If you enable MIDI Clock Receive, the arpeggiator or sequencer won't respond until a MIDI clock signal is received.

The Super 6 automatically switches to MIDI Clock Receive when it detects an external MIDI clock signal. You can manually disable this setting after a MIDI clock signal was detected, or choose not to send MIDI clock signals from an external sequencer or a DAW to the Super 6 to enable arpeggiator or sequencer playback regardless of external clock signals.

When the arpeggiator or sequencer is synchronised to an external MIDI clock source, you can use the **TEMPO** control to set the playback speed in clock divisions relative to the external tempo while the LEDs of buttons **1**-**8** and **A**-**H** will indicate the according settings. The following table lists the clock divisions that can be dialed in using the **TEMPO** control:

Setting	Timing Division	Tempo
	Whole note	BPM/4
	1/2 note	BPM/2
	1/4 note	BPM
	1/8 note	BPM x 2
	1/8 note triplet	BPM x 3
	1/16 note	BPM x 4
	1/16 note triplet	BPM x 6
	1/32 note	BPM x 8

Arpeggiator Mode

When you turn on the arpeggiator and play a chord, the arpeggiator will generate an arpeggio based on its settings and the notes you hold. There are many ways to change the way the currently held chord is arpeggiated.

ON: Press this button to turn the arpeggiator on or off. When the LED of this button is lit, the arpeggiator is active, unless **MODE** is set to **SEQ**, in which case sequencer mode is selected.

RANGE: In arpeggiator mode, this button allows you to choose from four different octave settings:

- **1:** The notes you actually hold on the keyboard will be arpeggiated.
- **2:** The notes you actually hold on the keyboard and the corresponding notes in the octave above will be arpeggiated.
- **3:** The notes you actually hold on the keyboard and the corresponding notes in the two octaves above will be arpeggiated.
- **4:** The notes you actually hold on the keyboard and the corresponding notes in the three octaves above will be arpeggiated.

MODE: This button allows you to choose from five different playback modes:

- **UP:** The arpeggio moves from the lowest note to the highest note.
- **DOWN:** The arpeggio moves from the highest note to the lowest note.
- **U&D:** The arpeggio moves from the lowest note to the highest note and back to the lowest note.
- **RANDOM:** All held notes will be arpeggiated in random order.
- **SEQ:** With this option selected, sequencer mode is enabled. For more details on the sequencer see pages 98-101.

If you change playback modes while the arpeggiator is on, sequencer mode will be skipped, allowing you to smoothly cycle through the four arpeggiator playback modes without interruptions.

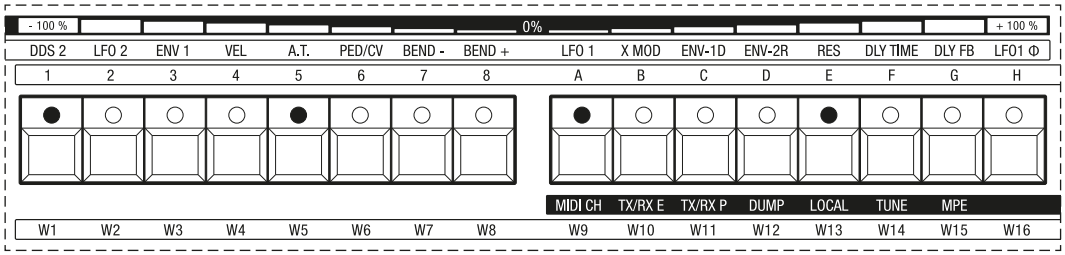
HOLD: If you activate this function while the arpeggiator is switched on, the arpeggio will continue to play without you having to hold down a key. As soon as you release all keys and play a new chord, a new arpeggio is generated.

Sequencer Mode

In sequencer mode, the Super 6 allows up to 64 steps to be recorded with programmable step, slide, accent, rest and sequence length settings.

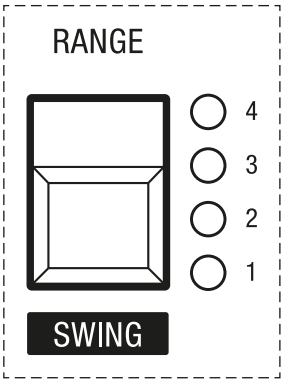
Up to 16 sequences can be stored and recalled. Although a sequence can be linked to a patch when saved, the independent sequencer memory allows you to try out different sequences with each patch.

Once sequencer mode is enabled, buttons **1**-**8** and **A**-**H** turn into a series of 16 steps, with each of the buttons representing a step in the sequence.



The numbered and lettered buttons in sequencer mode with steps 1, 5, 9 and 13 activated.

The sequence represented by the 16 step buttons is divided into 4 pages so that all 64 steps can be displayed. Which page you are on is indicated by the four LEDs next to the **RANGE** button.



The range select button.

During recording and playback, the sequencer automatically jumps to the next page if the sequence is longer than 16 steps.

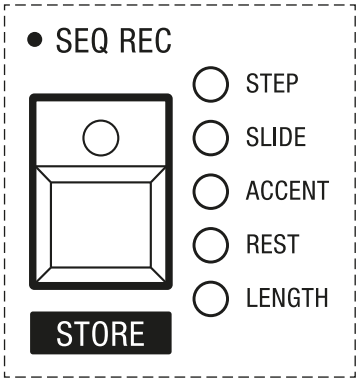
ON: Press this button to turn the sequencer on or off. When the LED of this button is lit, the sequencer is active, unless **MODE** is set to one of the arpeggiator playback modes, in which case arpeggiator mode is selected.

RANGE: In sequencer mode, this button enables you to jump to the following pages of the step sequence:

- **1:** The 16 buttons display page 1 of the sequence: steps 1-16. This is the default page when you enter sequencer mode.
- **2:** The 16 buttons display page 2 of the sequence: steps 17-32.
- **3:** The 16 buttons display page 3 of the sequence: steps 33-48.
- **4:** The 16 buttons display page 4 of the sequence: steps 49-64.

MODE: Selecting the option **SEQ** enables sequencer mode. For all other playback modes related to the arpeggiator, see page 97.

SEQ REC: Pressing this button once, arms the sequencer for recording. Its LED will then flash, indicating that you are now in recording mode. By repeatedly pressing this button, you can step through the five available sequencer tracks. As you select one of the options, the sequence of steps represented by the 16 buttons will update accordingly.



The **SEQ REC** button and the five track LEDs that indicate which track is currently selected.

These are the sequencer tracks you can select using the **SEQ REC** button:

- **STEP:** This track is selected by default when you enter sequence recording mode. It allows you to record notes or chords step by step. Before you can start recording, press the step button you wish to start from – in most cases this will be the first step. The respective LED will then start flashing to indicate that this step is ready for recording. Steps are recorded as soon as you start playing notes. A note or chord is recorded after you released all keys. The sequencer then advances to the next step. Recorded steps are represented by lit LEDs.

If you want to edit or re-record a step, simply press the corresponding step button. Its LED will start flashing, indicating that the sequencer is now waiting for you to play a new note or chord. Once you've done so, the LED will stop flashing and the sequencer will advance to the next step.

- **SLIDE:** This track allows you to specify which steps are tied together. When portamento is set to a non-zero value, a pitch slide occurs between tied notes of different pitches. To tie steps together, press adjacent step buttons. For example, if you would like to tie steps 3 and 4 together, press buttons **3** and **4** while on page 1 of the sequence. Active ties are indicated by lit LEDs.
- **ACCENT:** This track allows you to specify which steps should be accented. With accents you can emphasise the level and brightness of notes or chords if the **DYNAMICS** switch in the VCA section is set to either **1/2** or **ON**. This is useful for adding dynamic variety to your sequence. Active accents are indicated by lit LEDs.
- **REST:** This track allows you to specify which steps to omit. An active rest causes the note or chord you recorded to that step to be skipped. Active rests are indicated by unlit LEDs.
- **LENGTH:** This track allows you to define the length of a sequence. First use the **RANGE** button to select the page on which the last step of the sequence should be. Then press the button corresponding to the step you wish to be the last step in the sequence. For example, if you would like your sequence to be eight steps long, make sure you are on page 1 and then press button **8**. To indicate active steps, the LED representing the last step in the sequence and all steps before it light up. After the last step is triggered, the sequence starts over.

*When you're in recording mode and **TRACK** is set to **STEP**, you can also record the position of the bender as you play notes.*

*When the sequencer is switched on and **SEQ REC** is flashing, you can use the remaining voices, i.e. the voices not used for the current sequence, to play along with the sequence.*

HOLD: If you activate this function, the sequence can be transposed according to the notes you play. Transposition is relative to middle C (C4). For example, if you play a note above middle C, the sequence is transposed above middle C by this interval.

Clearing a Sequence

In case you would like to start from scratch again, use the following shortcut for clearing a sequence:

1. Hold the **SEQ REC** button.
2. Press the **MOD ASSIGN** button.

Loading and Storing Sequences

LOAD: Hold **SHIFT** and press the **HOLD** button to load a sequence. The LEDs of buttons **1**-**8** and **A**-**H** indicate which of the 16 available sequences is currently selected. Press any of the other 15 buttons to load a different sequence.

Let's say you would like to load sequence 12:

1. Hold **SHIFT** and press the **HOLD** button.
2. Press button **D**. Its LED will light up to indicate that sequence 12 is now loaded.

Why not spend some time loading different sequences to try out which one works best with the current patch?

STORE: Hold **SHIFT** and press the **SEQ REC** button to store a sequence. The LEDs of buttons **1**-**8** and **A**-**H** indicate which of the 16 available sequences is currently selected. Press any of these buttons for 3 seconds to store the current sequence.

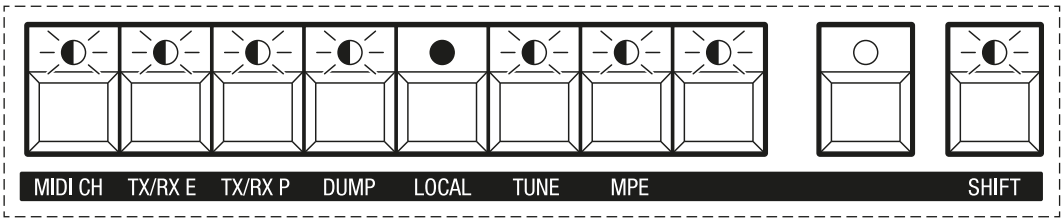
Let's say you would like to store your sequence to memory location 4:

1. Hold **SHIFT** and press the **SEQ REC** button.
2. Press and hold button **4** for 3 seconds. The LEDs of buttons **1**-**8** and **A**-**H** will flash once to indicate that your sequence is now saved.

GLOBAL SETTINGS

The global settings allow you to define how the Super 6 behaves and responds on a global level, i.e. regardless of individual performance or patch settings. To access the global settings, simply press the **SHIFT** button. If you press **SHIFT** again, you exit the global settings.

Any changes applied to the global settings will be automatically saved after a few seconds of no interaction with the instrument.



The global settings section.

MIDI CH: Press this button to specify which MIDI channel the Super 6 should respond to. The LEDs of buttons **1**-**8** and **A**-**H** indicate which of the 16 available MIDI channels is currently selected. Press any of the other 15 buttons to select a different MIDI channel.

Do not connect a USB cable when using the MIDI DIN ports. The Super 6 does not support simultaneous DIN and USB MIDI.

The Super 6 is capable of receiving polyphonic aftertouch via MIDI. The default mapping is the same as for channel pressure or monophonic aftertouch, meaning that polyphonic aftertouch will either control the amount of LFO 2 or the amount of upwards pitch bending (when LFO 2 is set to be permanently 'on').

*To allow polyphonic aftertouch to be mapped through the modulation matrix only, set LFO 2 to be permanently 'on' by toggling the leftmost switch in the keyboard model's performance control section to **ON (AT -> BEND)** or by pressing the desktop model's LFO 2 **MODE** button until both of its LEDs are unlit (options **MW** and **AT** disabled).*

TX/RX E: Press this button to determine how parameter changes are sent and received via MIDI. You can combine the following options:

- **BUTTON 1:** MIDI CC Transmit. When enabled (LED lit), parameter changes are transmitted as continuous controller messages.
- **BUTTON 2:** MIDI CC Receive. When enabled (LED lit), parameter changes are received as continuous controller messages.
- **BUTTON 3:** NRPN mode. When enabled (LED lit), parameter changes are transmitted as unregistered parameter number messages at a resolution of 14 bits if MIDI CC Transmit is also enabled.

To exit this parameter, press **TX/RX E** again.

When MIDI CC Receive is enabled, the Super 6 will always respond to both parameter changes sent in 7 and 14 bit resolution.

TX/RX P: Press this button to determine whether program change messages should be transmitted and received via MIDI. You can combine the following options:

- **BUTTON 1:** When enabled (LED lit), program change messages are transmitted.
- **BUTTON 2:** When enabled (LED lit), program change messages are received.

To exit this parameter, press **TX/RX P** again.

DUMP: Reserved for future use.

LOCAL: This button toggles Local Control on or off. Disabling Local Control can be useful if you want to control external MIDI devices or avoid MIDI data loops while recording in a DAW.

- **ON:** When Local Control is on (LED lit), the Super 6's front panel controls and keyboard are connected to the internal sound engine.
- **OFF:** When Local Control is off (LED not lit), the Super 6's front panel controls and keyboard have no effect on its sound engine.

With Local Control disabled and MIDI CC Transmit and Receive enabled, MIDI data is still transmitted through the MIDI outputs.

TUNE: Press this button to auto-tune the Super 6's filters for calibration purposes. The LEDs of Buttons **1**-**8** and **A**-**H** will indicate the progress of the auto-tune process from left to right until 12 LEDs are lit. As soon as the filter calibration is complete, the LEDs will turn off.

MPE: Press this button to enter the Super 6's MIDI Polyphonic Expression (MPE) settings. See pages 105-106 for more details. You can combine the following options:

- **BUTTON 1:** When enabled (LED lit), the Super 6 will respond to incoming MIDI messages sent from an MPE controller.
- **BUTTON 2:** When enabled (LED lit), a dead zone for polyphonic aftertouch is active, which defines a starting value from which polyphonic aftertouch affects the sound. This allows you to play a note without immediately triggering polyphonic aftertouch on MPE controllers that don't provide a dead zone setting for this purpose. Enabling this option is recommended for use with ROLI controllers or the Osmose.
- **BUTTON 3:** When enabled (LED lit), the Y-axis sensitivity (assigned to CC74) is reduced. This essentially limits the modulation scope controlled by front/back movements in case this gestural dimension turns out to be too sensitive in relation to the fully available modulation range. Enabling this option is recommended for use with the LinnStrument or the Osmose.

To exit this parameter, press **MPE** again.

MPE settings are stored per patch, allowing you to customise each patch to your MPE controller.

TRNSP (Desktop model only): Press this button to adjust the global transpose setting. Once you press the **TRNSP** button, you can use the **MOD AMOUNT** encoder to transpose the pitch globally by +/- 12 semitones. The current transpose setting is indicated by the LEDs of buttons **1**-**8** and **A**-**H**. When no transposition was applied, no LED is flashing. As you transpose up, the LEDs of buttons **A**-**H** start flashing. As you transpose down, the LEDs of buttons **1**-**8** start flashing.

GLOBAL RESET: If your Super 6 is not behaving as expected, you can reset the global parameters to their default settings:

1. Turn on the Super 6.
2. Press and hold the **MANUAL** button for 5 seconds until all LEDs turn on and off again.
3. The auto-tune process starts automatically. Release the **MANUAL** button and wait for the auto-tune process to complete.

MPE SUPPORT

When you enable MPE mode, the Super 6 will respond to incoming MIDI messages sent from an MPE controller via an individual MIDI channel per note. MPE mappings are an extension of standard MIDI mappings like pitch bend and channel aftertouch, but polyphonic. Therefore, dedicated controls for the corresponding settings are already present on the Super 6's front panel.

The Super 6's response to the five gestural dimensions of MPE controllers can be adjusted in the following ways:

- **Note On Velocity:** For this MIDI message to take effect in MPE mode, set the **DYNAMICS** switch in the VCA section to either **1/2** or **ON**.
- **Note Off Velocity:** For this MIDI message to take effect in MPE mode, set the **DYNAMICS** switch in the VCA section to either **1/2** or **ON**.
- **X-Axis/Polyphonic Pitch Bend:** The Super 6's response to X-axis gestures or polyphonic pitch bend data is determined by the bender settings. Use the rightmost **DDS** fader and the oscillator selector toggle switch in the performance control section to specify how much polyphonic pitch bend should affect the pitch of the oscillators and which oscillators should respond to pitch bend messages.
- **Y-Axis/Polyphonic Timbre Control:** As per MPE standard, Y-axis gestures are sent via MIDI CC message 74. Since CC74 is assigned to the filter cutoff frequency, the Y-axis data sent from an MPE controller controls the brightness of a sound.
- **Z-Axis/Polyphonic Pressure:** The Super 6's response to Z-axis gestures or polyphonic pressure data is determined by the internal aftertouch settings. Use the aftertouch-related controls in the performance control section or the modulation source **AT** in the modulation matrix to specify what should be modulated by your MPE controller's polyphonic pressure.

Since MPE mode requires individual note articulation, the control voltages must be different for each triggered voice. The analog hardware only allows up to six notes to have different control voltages for most parameters, so MPE mode is limited to six voices.

Recommended MPE Settings

On the LinnStrument:

1. Choose a bend range of 48 semitones.
2. Set Y-axis mode (CC74) to “Relative”.
3. Set Z-axis to “Channel Pressure” or “Polyphonic Pressure”.

On ROLI Controllers:

1. Set Y-axis mode (CC74) to “Relative”.
2. Set Z-axis to “Channel Pressure” or “Polyphonic Pressure”.

On the Osmose:

1. Set the external MIDI mode to “MPE” (default).
2. In the “Sensitivity” menu, set bending to “1/12” as a starting point. Then adjust to what feels natural for the current patch.
3. In the “Adjust” menu, set “Aftertouch” to send Channel Pressure MIDI messages (“Ch. Press”).
4. In the “Adjust” menu, set “Pressure” to either
   - send MIDI messages via CC74 (default) with a minimum value of 64 and a maximum value adjusted to what feels natural for the current patch,

     or:

   - send MIDI messages via CC4 (Foot Controller) with a minimum value of 0. Use the modulation matrix on the Super 6 to assign **PED/CV** to the desired control(s).

It may be necessary to adjust the sensitivities, minimum/maximum values and curvatures depending on the type of sound you are working with.

FILE MANAGEMENT

By connecting your Super 6 to a computer, you can easily access and organise performances, patches, alternative waveforms and sequences stored on the instrument. This is useful for sharing and backing up related files, as well as freeing up the Super 6's internal storage.

Follow the steps below to unlock the Super 6's patch drive:

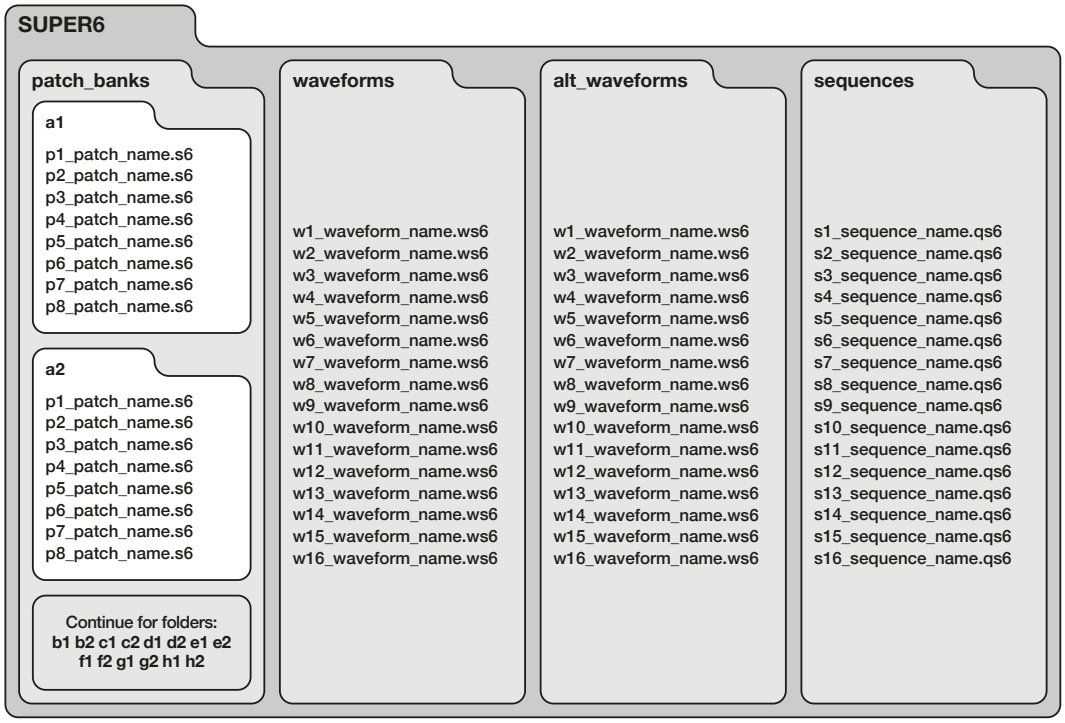
1. Turn off the Super 6 and wait a few seconds.
2. Whilst holding down the **PATCH** button, turn on the Super 6 and continue to hold the **PATCH** button.
3. The LEDs of buttons **1**-**8** and **A**-**H** light up one by one to indicate the loading progress.
4. Once the LEDs indicate that the Super 6 is in patch mode, release the **PATCH** button.

Once you then connect the Super 6 to your computer using a USB cable, the Super 6's patch drive will appear on your computer as a disk drive named **SUPER6**, which can be accessed like any other flash drive.

The Super 6's patch drive contains four folders:

- 'patch_banks',
- 'waveforms' (the first group of 16 alternative waveforms),
- 'alt_waveforms' (the second group of 16 alternative waveforms) and
- 'sequences'.

The folder 'patch_banks' contains up to 16 subfolders: One for each bank of patches. Within each of the bank folders 'a1' to 'h2' up to 8 patch files are stored. The folders 'waveforms' and 'alt_waveforms' each contain up to 16 waveform files. The folder 'sequences' contains up to 16 sequence files.



The file structure of the Super 6's patch drive.

File Name Convention

The first character of each file – **p**, **w** or **s** – is used to prefix patch, waveform and sequence files. The second character is used to define the memory location of the patch, waveform or sequence. For patches use numbers 1-8. For alternative waveforms and sequences use numbers 1-16.

The first character of each file – **p**, **w** or **s** – is used to prefix patch, waveform and sequence files. The second character is used to indicate the memory location of the patch, waveform or sequence. Use numbers 1-8 for patches. Use numbers 1-16 for alternative waveforms and sequences.

Patch files: p1_patch_name.s6

Waveform files: w1_waveform_name.ws6

Sequence files: s1_sequence_name.qs6

After the prefix, the number and an underscore, patch, waveform and sequence names can be freely defined to make it easier for you to identify the files. However, you should avoid spaces and use underscores instead.

Loading Patches Stored to Your Computer

1. Follow steps 1-4 on page 107 to unlock the patch drive.
2. Connect the Super 6 to your computer using a USB cable.
3. The Super 6's patch drive appears as a disk drive named **SUPER6** on your computer.
4. Click on the icon of the **SUPER6** drive.
5. Navigate to the folder 'patch_banks'.
6. Open the desired bank folder (**a1**-**h2**).
7. Copy and paste the patch files you would like to transfer to the Super 6 to the bank folder you have selected in the previous step.
8. If necessary, edit the name prefix of the patch files you copied and pasted, so that it matches the desired patch location. Make sure to manually delete the patches you would like to replace in the selected folder if the names of the new patches are not identical to the names of the old patches. Empty the trash on your computer so that the files are indeed deleted from the **SUPER6** drive.

Loading Waveforms Stored to Your Computer

1. Follow steps 1-4 on page 107 to unlock the patch drive.
2. Connect the Super 6 to your computer using a USB cable.
3. The Super 6's patch drive appears as a disk drive named **SUPER6** on your computer.
4. Click on the icon of the **SUPER6** drive.
5. Navigate to the folder 'waveforms' or 'alt_waveforms'.
6. Copy and paste the waveform files you would like to transfer to the Super 6 to the folder you selected in the previous step.
7. If necessary, edit the name prefix of the waveform files you copied and pasted, so that it matches the desired waveform location. Make sure to manually delete the waveforms you would like to replace if the names of the new waveforms are not identical to the names of the old waveforms. Empty the trash on your computer so that the files are indeed deleted from the **SUPER6** drive.

UDO will periodically release more waveform packs which can be downloaded from udo-audio.com/downloads*.*

Loading Sequences Stored to Your Computer

1. Follow steps 1-4 on page 107 to unlock the patch drive.
2. Connect the Super 6 to your computer using a USB cable.
3. The Super 6's patch drive appears as a disk drive named **SUPER6** on your computer.
4. Click on the icon of the **SUPER6** drive.
5. Navigate to the folder 'sequences'.
6. Copy and paste the sequence files you would like to transfer to the Super 6 to the folder you selected in the previous step.
7. If necessary, edit the name prefix of the sequence files you copied and pasted, so that it matches the desired sequence location. Make sure to manually delete the sequences you would like to replace if the names of the new sequences are not identical to the names of the old sequences. Empty the trash on your computer so that the files are indeed deleted from the **SUPER6** drive.

Backing up Patches to Your Computer

1. Connect the Super 6 to your computer using a USB cable.
2. The Super 6's drive appears as a disk drive named **SUPER6** on your computer.
3. Click on the icon of the **SUPER6** drive.
4. Click on the folder 'patch_banks_a' or 'patch_banks_b' and copy and paste it to your computer's hard drive. You may also navigate to one of the 16 bank folders (**a1**-**h2**) or a single patch file within those folders to copy and paste it to your computer's hard drive.

Backing up Waveforms to Your Computer

1. Connect the Super 6 to your computer using a USB cable.
2. The Super 6's drive appears as a disk drive named **SUPER6** on your computer.
3. Click on the icon of the **SUPER6** drive.
4. Click on the folder 'waveforms' or 'alt_waveforms' and copy and paste it to your computer's hard drive. You may also navigate to a single waveform file within those folders to copy and paste it to your computer's hard drive.

Backing up Sequences to Your Computer

1. Connect the Super 6 to your computer using a USB cable.
2. The Super 6's drive appears as a disk drive named **SUPER6** on your computer.
3. Click on the icon of the **SUPER6** drive.
4. Click on the folder 'sequences' and copy and paste it to your computer's hard drive. You may also navigate to a single sequence file to copy and paste it to your computer's hard drive.

Changing the Init Patch

Store the patch you would like to be the new init patch as ‘init_patch.s6’ on your computer. You may insert any additional information between the name ‘init_patch’ and the file extension ‘s6’. For example, you could name it ‘init_patch_binaural_pad.s6’.

After naming and saving your custom init patch, copy it to the Super 6’s boot drive:

1. Enter bootloader mode:
    - Turn off the Super 6 and wait a few seconds.
    - Whilst holding down the **SHIFT** button, turn on the Super 6 and continue to hold the **SHIFT** button.
    - Progress LEDs will cycle through buttons **1**-**8** and **A**-**H** while the LED of the **SHIFT** button is flashing. (Make sure this is the case, restart step 2 if not.)
    - Release the **SHIFT** button.
2. Connect the Super 6 to your computer using a USB cable.
3. The Super 6’s boot drive appears as a disk drive named **SUPER6-BOOT** on your computer.
4. Delete the init patch file ‘init_patch.s6’ from the **SUPER6-BOOT** drive. Make sure to empty the trash if you are a macOS user, or the update won’t be possible.
5. Copy the desired init patch file you saved as ‘init_patch.s6’ from your computer to the **SUPER6-BOOT** drive. If asked if you want to copy files without properties, choose ‘yes’. Do not turn off the instrument or unplug the USB cable during the transfer!
6. When the transfer is complete, disconnect the USB cable.
7. Turn off the Super 6, wait a few seconds and turn it on again.

If no init patch file is stored on the boot drive, the Super 6 will load the last active patch upon power cycle. If no patch file is stored on the patch drive, the Super 6 will start in manual mode.

Storing New Waveform Packs to the Super 6

Additional waveforms packs can easily be stored on the Super 6's patch drive:

1. Download additional waveform packs to your computer from udo-audio.com/downloads.
2. Extract the .zip file on your computer (not on the Super 6 itself) and rename the extracted folder to 'waveforms' or 'alt_waveforms', depending on which folder you would like to replace.
3. Follow steps 1-4 on page 107 to unlock the patch drive.
4. Connect the Super 6 to your computer using a USB cable.
5. The Super 6's patch drive appears as a disk drive named **SUPER6** on your computer.
6. Click on the icon of the **SUPER6** drive.
7. Delete the waveform folder you would like to replace and make sure to empty the trash if you are a macOS user.
8. Copy the new waveform folder to the **SUPER6** drive. If asked if you want to copy files without properties, choose 'yes'. Do not turn off the instrument or unplug the USB cable during the transfer!
9. When the transfer is complete, disconnect the USB cable.
10. Turn off the Super 6, wait a few seconds and turn it on again.

Creating Your Own Alternative Waveforms

If you would like to create your own alternative waveforms for DDS 1, make sure that the files meet the following standards:

- 16-bit signed integer format samples
- Normalised, single-cycle waveform with 4096 points (8192 bytes)
- Bandlimited at sampling frequency/8 (Nyquist/4), i.e. frequency content above 512 Hz in your 4096 point waveform should be removed
- Binary file containing no header data and file extension '.ws6'

Using the Sequencer for Chord Memory

You can play any chord with one finger only by making creative use of the sequencer:

1. Use the **MODE** button in the arpeggiator and sequencer section to select the option **SEQ**.

2. Press the **SEQ REC** button to enable note recording.

3. Press the first step button (patch select button **1**) to start the recording from the first step onwards. Its LED will start flashing.

4. Play the desired chord on the keyboard. After you released all keys, the LED of first step will become solidly lit to indicate that you have recorded a chord to the first step.

5. Press the **SEQ REC** button again to select the **SLIDE** track.

6. Activate the slide option for the first step. Its LED will become solidly lit.

7. Press the **SEQ REC** button three more times to select the **LENGTH** track.

8. Set the sequence length to one step only by pressing the first step button (patch select button **1**). Its LED will become solidly lit. You have now created a step sequence with one chord on one step that is infinitely tied to itself. The latter frees you of any sequence duration constraints, allowing you to play and hold the recorded chord like any other note or chord.

9. Press the **SEQ REC** button again to exit track selection. Its LED will become unlit.

10. Press the **ON** button in in the arpeggiator and sequencer section to be able to trigger the sequence/chord with any key press.

11. Turn the **TEMPO** control all the way clockwise to ensure that the sequence/chord will be triggered as expected, even when you play very fast.

Setting up the Super 6 as a MIDI Device in a DAW

Follow the steps below to use your Super 6 in conjunction with a DAW:

1. Make sure you only connect the Super 6 via either MIDI DIN or USB. The Super 6 does not support simultaneous DIN and USB MIDI.

2. In the preferences of your DAW, ensure that MIDI data is being received by and sent to the Super 6.

3. If you would like to sync the Super 6's arpeggiator and sequencer to your DAW, ensure that your DAW sends a MIDI clock signal to the Super 6 and enable External Clock Receive on the Super 6.

4. Disable Local Control on the Super 6 to avoid MIDI data loops during playback and recording.

5. Create a MIDI track for the Super 6. Ensure that MIDI data is sent to the MIDI device **SUPER 6** (if connected via USB) or to the MIDI interface to which the Super 6 is connected.

6. Create an audio track for the Super 6 and route the audio signal from the MIDI track to the audio track. Set the audio channels to the stereo inputs of your audio interface the outputs of the Super 6 are connected to.

CHEAT SHEET

The following tables provide an overview of all shortcuts, each parameter's secondary function, the various functions of the select buttons and the global settings.

Shortcuts

Combination	Function
SHIFT + **MANUAL**	Load init patch
Press and hold button **1**-**8** in patch mode	Store patch
SHIFT + turn LFO 1 **WAVEFORM** rotary control	Use DDS 1 waveform as LFO 1 waveform
Press and hold any of the buttons **1**-**8** in mod assign mode + move parameter control	Map a fixed modulation source to a parameter not available in the modulation matrix
SHIFT + **MOD ASSIGN** after entering the modulation matrix	Clear all modulation mappings
SHIFT + **MOD ASSIGN** after selecting a modulation source (buttons **1**-**8**)	Clear all mappings from that modulation source
SHIFT + **MOD ASSIGN** after selecting a modulation destination (buttons **A**-**H**)	Clear all mappings to that modulation destination
SEQ REC + **MOD ASSIGN**	Clear sequence
Press and hold **MANUAL** for 5 seconds	Reset global settings

Secondary Parameter Functions

Parameter	Mode	Function
DDS 2 Mode	DDS 2 Range = LFO	LFO Mode Norm = Sub-oscillator off Sub Osc = Sub-oscillator Audio In = External audio signal
LR Phase	Binaural off	Pan Spread
LFO 1 Mode	LFO 1 Wave = HF/HF TRK	LFO 1 Routing Norm = LFO 1 as audio rate modulation source DDS 1 = LFO 1 signal through DDS 1 channel DDS 2 = LFO 1 signal through DDS 2 channel
PW/Detune	Shift	Drift
Envelope 1 Attack	Shift	Envelope 1 Attack Hold
Envelope 1 Decay	Shift	Envelope 1 Decay Hold
Envelope 2 Decay	Shift	Envelope 2 Decay Hold
Detune	Shift	Performance Detune +/- 7 semitones
Voice Assign Mode	Shift	Unison Size 1 = Three binaural voices stacked 2 = Six binaural voices stacked 3 = Six binaural voices stacked as octave 4 = Six binaural voices stacked as fifth + octave 5 = Six binaural voices stacked as major chord
Arpeggiator Range	Shift	Swing Amount 0 LEDs = No swing 1 LED = Swing setting 1 2 LEDs = Swing setting 2 3 LEDs = Swing setting 3 4 LEDs = Swing setting 4
Hold	Shift	Load sequence 1-16 (buttons **1**-**8**, **A**-**H**)
Seq Rec	Shift	Store sequence 1-16 (buttons **1**-**8**, **A**-**H**)

Multifunctional Select buttons

Buttons	Mode	Function
1-**8**	Patch	Load patch 1-8
1-**8**	Mod Assign	Select modulation source
A-**H**	Patch	Select patch bank A1-H2
A-**H**	Mod Assign	Select modulation destination
1-**8**, **A**-**H**	Wave	Load alternative waveform 1-32
1-**8**, **A**-**H**	Sequencer	Step programming
1-**8**, **A**-**H**	Load sequence	Load sequence 1-16
1-**8**, **A**-**H**	Store sequence	Store sequence 1-16

Global Settings (Shift Mode)

Parameter	Function
MIDI CH	Set MIDI channel 1-16 (buttons **1**-**8**, **A**-**H**)
TX/RX E	1 = MIDI CC Transmit on/off 2 = MIDI CC Receive on/off 3 = NRPN Mode on/off
TX/RX P	1 = Program Change Messages Transmit on/off 2 = Program Change Messages Receive on/off
DUMP	Reserved for future use
LOCAL	Local Control on/off
TUNE	Filter Auto-Tune
MPE	1 = MPE Mode on/off 2 = Polyphonic Aftertouch Dead Zone on/off 3 = Reduce Y-Axis Sensitivity on/off
EXT CLK	External Clock Settings 1 = MIDI Clock Transmit on/off 2 = MIDI Clock Receive on/off 3 = MIDI Note Output on/off 4 = MIDI Stop Message Receive on/off
Octave selector toggle switch (keyboard model)	Global Transpose +/- 12 semitones
MOD AMOUNT	Global Fine Tuning +/- 100 cents

MIDI SPECIFICATIONS

System Real-Time Messages

Control Function	Transmit	Receive
MIDI Timing Clock	Yes	Yes
Start	Yes	Yes
Stop	Yes	Yes

Channel Messages

Control Function	Transmit	Receive
Note Off	Yes	Yes
Note On	Yes	Yes
Polyphonic Key Pressure	No	Yes
Control Change	See “Global Settings” (page 103)	See “Global Settings” (page 103)
Program Change	See “Global Settings” (page 103)	See “Global Settings” (page 103)
Channel Pressure	Yes	Yes
Pitch Bend	Yes	Yes

Continuous Controller Messages

The table below lists the continuous controller messages (CCs) that are mapped to the controls of the Super 6. These messages are transmitted and/or received dependent on **TX/RX E** configuration in the global settings (see page 103).

CC #	Value Range	Parameter Name
0	0-127	Bank Select
1	0-127	Modulation Lever
2	-	-
3	0-127	Tempo
4	0-127	Foot Controller
5	0-127	Portamento Time
6	0-127	Data Entry MSB
7	0-127	VCA Envelope Level
8	-	-
9	-	-
10	-	-
11	0-127	Expression
12	0-127	Delay Time
13	0-127	Delay Feedback
14	0-15	Sequence Load
15	-	-
16	0 = Triangle 21 = Square 43 = Random 64 = Saw 85 = HF 107 = HF TRK	LFO 1 Waveform/HF Mode
17	0-127	LFO 1 Rate
18	0-127	LFO 1 Delay
19	0-127	LFO 1 LR Phase

CC #	Value Range	Parameter Name
20	0 = Free/Norm 43 = Once/DDS 1 85 = Reset/DDS 2	LFO 1 Mode
21	0-127	DDS LFO 1 Amount
22	0-127	DDS Env 1 Amount
23	0 = DDS 1 43 = Both 85 = DDS 2	DDS Modulator Destination
24	0 = Off 43 = 1/2 85 = On	Super Mode
25	0-127	PW/Detune
26	0-127	PWM/SWM
27	0 = LFO 1 43 = Both 85 = ENV 1	PWM/SWM Source
28	0-127	Cross Modulation
29	0 = Sine 21 = Saw 43 = Square 64 = Triangle 85 = Noise 107 = Alternative Waveform	DDS 1 Waveform
30	0 = 64’ 21 = 32’ 43 = 16’ 64 = 8’ 85 = 4’ 107 = 2’	DDS 1 Range
31	0 = Sine 21 = Saw 43 = Square 64 = Triangle 85 = Noise 107 = Pulse	DDS 2 Waveform
32	0-127	Envelope 1 Decay Hold
33	0-127	Envelope 2 Decay Hold
34	0 = LFO 21 = 32’ 43 = 16’ 64 = 8’ 85 = 4’ 107 = 2’	DDS 2 Range

CC #	Value Range	Parameter Name
35	0-127	DDS 2 Tune
36	0 = Norm 43 = X-Fade/Sub Osc 85 = Sync/Audio In	DDS 2 Mode
37	0-127	Oscillator Mix/Split Point
38	0-127	LSB for Control 6 (Data Entry)
39	-	-
40	0 = Off 43 = Fix 85 = Trk	VCF HPF Mode
41	0 = Off 43 = 1 85 = 2	VCF Drive
42	-	-
43	0 = Off 43 = 1/2 85 = On	VCF Keytrack
44	0 = Env 1 43 = 1 + 2 85 = Env 2	VCF Envelope Source
45	0-127	VCF Envelope Amount
46	0-127	VCF LFO 1 Amount
47	0-127	VCF DDS 2 Amount
48	0 = Off 43 = 1/2 85 = On	VCA Dynamics
49	0 = Env 2 43 = Fixed Env 1 85 = Fixed Env 2	VCA Envelope Mode
50	0 = Normal 43 = Inverted 85 = Loop	Envelope 1 Mode
51	0 = Off 43 = 1/2 85 = On	Envelope 1 Keytrack
52	0-127	Envelope 1 Attack Hold
53	0-127	Envelope 1 Attack

CC #	Value Range	Parameter Name
54	0-127	Envelope 1 Decay
55	0-127	Envelope 1 Sustain
56	0-127	Envelope 1 Release
57	0-127	Envelope 2 Decay
58	0-127	Envelope 2 Sustain
59	0 = Off 64 = On	Manual Mode
60	0 = Trig 43 = AT + Trig 85 = On	LFO 2 Trigger Source
61	0 = DDS 1 43 = 1 + 2 85 = DDS 2	Performance Control Destination
62	0-127	LFO 2 Rate
63	0-127	LFO 2 Delay
64	0 = Off 64 = On	Sustain Pedal
65	-	-
66	-	-
67	0 = -2 26 = -1 51 = 0 77 = +1 102 = +2	Octave Select
68	-	-
69	0 = Off 64 = On	Delay Freeze
70	0-127	DDS LFO 2 Amount
71	0-127	VCF Resonance
72	0-127	Envelope 2 Release
73	0-127	Envelope 2 Attack
74	0-127	VCF Cutoff Frequency
75	0-127	VCF LFO 2 Amount

CC #	Value Range	Parameter Name
76	0-127	DDS Pitch Bend Amount
77	0-127	VCF Pitch Bend Amount
78	0 = Poly 2 26 = Poly 1 51 = Unison 77 = Legato 102 = Solo	Voice Assign Mode
79	0 = 3 voices 26 = 6 voices 51 = 6 voices, octave 77 = 6 voices, octave & fifth 102 = 6 voices, major chord	Unison Size
80	0 = Off 64 = On	Binaural Mode
81	0 = Off 64 = On	Clock Sync
82	0 = 1 octave 32 = 2 octaves 64 = 3 octaves 96 = 4 octaves	Arpeggiator Range
83	0 = Swing 0 26 = Swing 1 51 = Swing 2 77 = Swing 3 102 = Swing 4	Arpeggiator/Sequencer Swing
84	0 = Off 64 = On	Arpeggiator/Sequencer External Clock
85	0 = Up 26 = Down 51 = Up & Down 77 = Random 102 = Sequencer	Arpeggiator/Sequencer Mode
86	0 = Off 64 = On	Arpeggiator/Sequencer On/Off
87	0 = Off 64 = On	Arpeggiator/Sequencer Hold
88	-	-
89	-	-
90	-	-
91	0-127	Delay Level
92	0-127	VCA LFO 1 Amount

CC #	Value Range	Parameter Name
93	0 = Off 32 = Chorus 1 64 = Chorus 2 96 = Chorus 1 & 2	Chorus
94	0-127	Drift
95	0-127	HPF Fix Cutoff Frequency
96	-	Data Increment
97	-	Data Decrement
98	0-127	Non-Registered Parameter Number (NRPN) - LSB
99	0-127	Non-Registered Parameter Number (NRPN) - MSB
100	0-127	Registered Parameter Number (RPN) - LSB
101	0-127	Registered Parameter Number (RPN) - MSB
102	-	-
103	-	-
104	-	-
105	-	-
106	-	-
107	-	-
108	-	-
109	-	-
110	-	-
111	-	-
112	-	-
113	-	-
114	-	-
115	-	-
116	-	-
117	-	-

CC #	Value Range	Parameter Name
118	-	-
119	-	-
120	0	All Sound Off
121	0	Reset All Controllers
122	0 = Off 64 = On	Local Control On/Off
123	0	All Notes Off
124	0	Omni Mode Off
125	0	Omni Mode On
126	0	Mono Mode On
127	0	Poly Mode On

Registered Parameter Numbers

The table below lists the registered parameter numbers (RPNs) that are mapped to the parameters of the Super 6. These messages are transmitted and/or received dependent on **TX/RX E** configuration in the global settings (see page 103).

RPN	RPN MSB (CC# 101)	RPN LSB (CC# 100)	Data Entry Value	Parameter Name
0	00H	00H	MSB = +/- 12 semitones	Pitch Bend Sensitivity
1	00H	01H	00H 00H = -100 cents 40H 00H = A440 7FH 7FH = +100 cents	Channel Fine Tuning
2	00H	02H	Only MSB used 00H = -12 semitones 40H = A440 7FH = +12 semitones	Channel Coarse Tuning

Non-Registered Parameter Numbers

The table below lists the non-registered parameter numbers (NRPNs) that are mapped to the global and patch-related parameters of the Super 6. These messages are transmitted and/or received dependent on **TX/RX E** configuration in the global settings (see page 103).

Global Parameters

NRPN	NRPN MSB (CC# 99)	NRPN LSB (CC# 98)	Value Range	Parameter Name
2051	10H	03H	0 = MIDI Channel 1 15 = MIDI Channel 16	MIDI Channel
2052	10H	04H	0 = Off 1 = On	MIDI Clock Transmit
2053	10H	05H	0 = Off 1 = On	MIDI Clock Receive
2055	10H	07H	0 = Off 1 = On	MIDI Program Change Transmit
2056	10H	08H	0 = Off 1 = On	MIDI Program Change Receive

Patch-Related Parameters

NRPN	NRPN MSB (CC# 99)	NRPN LSB (CC# 98)	Value Range	Parameter Name
1024	-	-	-	-
1025	-	-	-	-
1026	-	-	-	-
1027	08H	03H	0-16383	Tempo
1028	-	-	-	-
1029	08H	05H	0-16383	Portamento Time
1030	-	-	-	-
1031	08H	07H	0-16383	VCA Envelope Level
1032	-	-	-	-
1033	-	-	-	-
1034	-	-	-	-
1035	-	-	-	-
1036	08H	0CH	0-16383	Delay Time
1037	08H	0DH	0-16383	Delay Feedback
1038	-	-	-	-
1039	-	-	-	-
1040	-	-	-	-
1041	08H	11H	0-16383	LFO 1 Rate
1042	08H	12H	0-16383	LFO 1 Delay
1043	08H	13H	0-16383	LFO 1 LR Phase
1044	-	-	-	-
1045	08H	15H	0-16383	DDS LFO 1 Amount
1046	08H	16H	0-16383	DDS Env 1 Amount
1047	-	-	-	-

NRPN	NRPN MSB (CC# 99)	NRPN LSB (CC# 98)	Value Range	Parameter Name
1048	-	-	-	-
1049	08H	19H	0-16383	PW/Detune
1050	08H	1AH	0-16383	PWM/SWM
1051	-	-	-	-
1052	08H	1CH	0-16383	Cross Modulation
1053	-	-	-	-
1054	-	-	-	-
1055	-	-	-	-
1056	08H	20H	0-16383	Envelope 1 Decay Hold
1057	08H	21H	0-16383	Envelope 2 Decay Hold
1058	-	-	-	-
1059	08H	23H	0-16383	DDS 2 Tune
1060	-	-	-	-
1061	08H	25H	0-16383	Oscillator Mix/Split Point
1062	-	-	-	-
1063	-	-	-	-
1064	-	-	-	-
1065	-	-	-	-
1066	-	-	-	-
1067	-	-	-	-
1068	-	-	-	-
1069	08H	2DH	0-16383	VCF Envelope Amount
1070	08H	2EH	0-16383	VCF LFO 1 Amount
1071	08H	2FH	0-16383	VCF DDS 2 Amount
1072	-	-	-	-
1073	-	-	-	-

NRPN	NRPN MSB (CC# 99)	NRPN LSB (CC# 98)	Value Range	Parameter Name
1074	-	-	-	-
1075	-	-	-	-
1076	08H	34H	0-16383	Envelope 1 Attack Hold
1077	08H	35H	0-16383	Envelope 1 Attack
1078	08H	36H	0-16383	Envelope 1 Decay
1079	08H	37H	0-16383	Envelope 1 Sustain
1080	08H	38H	0-16383	Envelope 1 Release
1081	08H	39H	0-16383	Envelope 2 Decay
1082	08H	3AH	0-16383	Envelope 2 Sustain
1083	-	-	-	-
1084	-	-	-	-
1085	-	-	-	-
1086	08H	3EH	0-16383	LFO 2 Rate
1087	08H	3FH	0-16383	LFO 2 Delay
1088	-	-	-	-
1089	-	-	-	-
1090	-	-	-	-
1091	-	-	-	-
1092	-	-	-	-
1093	-	-	-	-
1094	08H	46H	0-16383	DDS LFO 2 Amount
1095	08H	47H	0-16383	VCF Resonance
1096	08H	48H	0-16383	Envelope 2 Release
1097	08H	49H	0-16383	Envelope 2 Attack
1098	08H	4AH	0-16383	VCF Cutoff Frequency
1099	08H	4BH	0-16383	VCF LFO 2 Amount

NRPN	NRPN MSB (CC# 99)	NRPN LSB (CC# 98)	Value Range	Parameter Name
1100	08H	4CH	0-16383	DDS Pitch Bend Amount
1101	08H	4DH	0-16383	VCF Pitch Bend Amount
1102	-	-	-	-
1103	-	-	-	-
1104	-	-	-	-
1105	-	-	-	-
1106	-	-	-	-
1107	-	-	-	-
1108	-	-	-	-
1109	-	-	-	-
1110	-	-	-	-
1111	-	-	-	-
1112	-	-	-	-
1113	-	-	-	-
1114	-	-	-	-
1115	08H	5BH	0-16383	Delay Level
1116	08H	5CH	0-16383	VCA LFO 1 Amount
1117	-	-	-	-
1118	08H	5EH	0-16383	Drift
1119	08H	5FH	0-16383	HPF Cutoff Frequency

Please see UDO's support site for the most up to date MIDI specification.

GLOSSARY

The following list provides brief explanations of key terms printed on the Super 6's front panel as well as basic synthesis terminology used throughout this manual.

Aftertouch (AT): Aftertouch is a keyboard expression feature that allows you to modulate a sound via key pressure.

Arpeggiator: An arpeggiator generates arpeggios based on a chord you hold. An arpeggio is a 'broken chord' in which the notes of a chord are played one after the other in an order determined by the current playback mode setting.

Band-Pass Filter: A band-pass filter is a combination of a low-pass and a high-pass filter. It subtracts spectral content above and below its cutoff frequencies.

Bender: A bender is a performance controller that can be moved along two axes: horizontal (left/right) and vertical (up). The corresponding gestures can impact the sound individually.

Binaural: The Latin term 'binaural' literally means 'with both ears'. In binaural mode, the 12 voices of the Super 6 are twinned to form six stereo 'super voices'. The effect on the sound ranges from subtle to extreme stereo movement and an enhanced sense of spatial positioning compared to conventional monaural signal chains.

Clock Signal: A clock signal acts like a metronome. Typically, a square wave oscillating between high and low states at a constant frequency is used to sync instruments or parameters like sequencers, LFOs or time-based effects. An 'external' clock signal is fed into your instrument from another device, such as your DAW.

Clock Sync: This function allows you to synchronise modules of a system, such as an arpeggiator, a sequencer, LFOs, and time-based effects, to an internal or external clock signal. When synchronised, parameters like LFO rate or delay time will respond at a rate that is relative to the clock signal. Using different clock divisions (quarter notes, eighth notes, etc.) for each parameter allows you to create complex rhythmic effects.

Cross Modulation (CROSS MOD, X MOD): Cross modulation is a type of frequency modulation (FM) the result of which depends on the frequency ratio between both oscillators. It can be used to create complex, clangourous or bell-like timbres.

Cutoff Frequency: This filter parameter allows you to set the point at which the filter begins to subtract frequencies from the oscillators' signals to shape the sound.

Delay Freeze: The delay freeze function turns the Super 6's delay effect into a basic looper, allowing you to create sound-on-sound loops you can play along with.

Detune Spread: In super mode, this parameter determines the degree to which the six 'sister' oscillators of DDS 1 are detuned and stereo spread relative to DDS 1's centroid oscillator.

Direct Digital Synthesis Oscillator (DDS): Direct Digital Synthesis is the signal generation method employed by both oscillator cores of the Super 6. At its centre is a clock signal running three orders of magnitude higher than typical audio sample rates. This clock increments a counter through thousands of indices in your chosen waveform, generating samples once every 20-billionths of a second and interpolating between them. Each numerically controlled oscillator then uses its own DAC, also running at the same high sample rate, to convert the samples to analog voltages before being filtered by a preliminary analog low-pass filtering stage.

Drive: This parameter determines the degree to which the input signal of the Super 6's filter circuitry is overdriven. Its three settings allow you to choose between a pure signal, a gentle level-boost with resonance compensation and a hard saturation.

Envelope (ENV): An envelope is a modulation source that defines how the signal or parameter it modulates evolves over time. Most envelope generators contain four stages that allow you to define the envelope's shape: attack, decay, sustain and release (ADSR). The Super 6's envelopes also feature a decay hold stage that determines the time it takes for the decay stage to begin after the attack stage reached its peak. In addition, the first envelope features an attack hold stage that determines the time it takes for the attack stage to begin after hitting a key.

High-Pass Filter (HPF): A high-pass filter subtracts frequency content below its cutoff frequency. The frequency content above the cutoff frequency remains unaffected, meaning the highs will pass through. Use this type of filter to make sounds thinner or brighter by reducing bass frequencies.

Keyboard Tracking (KEYTRACK, TRK, NOTE): Keyboard tracking is a type of modulation that uses the MIDI note number as a modulation source. Whatever is tied to keyboard tracking will respond relative to the pitch of the notes you play. In the modulation matrix of the desktop model, keyboard tracking is freely assignable via the **NOTE** button.

Left-Right Phase (LR PHASE): This parameter controls the left-right channel phase relationship of the Super 6's binaural sound engine, in other words the effect of LFO 1 on the stereo field.

Loop: A loop is essentially a repetition of a recording or shape, meaning once the end is reached, whatever is looped will start all over again. The Super 6 features a loop option for the first envelope and also allows you to create loops with the Delay Freeze function.

Low Frequency Oscillator (LFO): An LFO is an oscillator that generates frequencies below the range of human hearing. It can be used to modulate pitch to create vibrato effects or to modulate the VCA level to create tremolo effects. The Super 6's first LFO can also be set to higher frequencies, allowing it to be used as a third oscillator, for drone effects, or for audio-rate modulations.

Low-Pass Filter: A low-pass filter subtracts frequency content above its cutoff frequency. The frequency content below the cutoff frequency remains unaffected, meaning the lows will pass through. Use this type of filter to make the sound warmer or to emphasise bass frequencies.

MIDI: Musical Instrument Digital Interface. MIDI is a standardised protocol that allows various devices from different manufacturers to communicate with each other. This not only includes instruments but also computers and several types of controllers.

Mixer (MIX): The mixer allows you to adjust the level of each oscillator in relation to the other.

Modulation (MOD): Modulation is the process of affecting a destination signal or parameter with a source signal. For example, you can have an LFO control the behaviour of an oscillator's frequency, or an envelope control the volume of a sound. Common modulation sources include LFOs, envelopes, regular oscillators, and performance controls like aftertouch and velocity.

MPE: MIDI Polyphonic Expression. This is a standardised protocol that allows sound engines or synthesizers to be played via dedicated MPE controllers. MPE controllers emulate the complex articulation one might find in an acoustic instrument's individual notes. Each pad or key of an MPE controller allows for simultaneous gestures across different axes (pressure, left/right, up/down) that will alter how each individual note is articulated while a pad or key is held. How hard you hit a pad or key and in what manner you have released it will also have an impact on the sound of the note.

Oscillator: Oscillators belong to the most basic and essential building blocks of a synthesizer. Without them, you could neither hear a sound nor shape or modulate what is generating an audio signal. Both of the Super 6's oscillators generate classic waveforms such as sine, triangle, sawtooth and square. In addition, the first oscillator (DDS 1) offers up to 32 digital sounding alternative waveforms.

Patch: A patch is a stored set of parameters which determine a sound's characteristics. A total of 128 patches can be stored in the Super 6's internal memory. They are organised in 16 banks featuring 8 patches each.

Portamento: Portamento is a pitch-sliding effect between consecutive notes. The higher the portamento time, the longer it takes for a note to slide to the pitch of the following note. The portamento time is also determined by the intervals between the played notes: smaller intervals result in faster pitch slides, while larger intervals result in slower pitch slides.

Pulse Width (PW): The pulse width marks the duration a pulse signal is 'on'. It is commonly measured in percentages of a duty cycle. A duty cycle of 50% produces a square wave, meaning that the pulse signal is on for as long as it's off per duty cycle. Changing the on/off ratio alters the harmonic content, and thus changes the timbre. The sound of a pulse wave that has a duty cycle of more or less than 50% is thinner than that of a square wave and bears a nasal character. At a duty cycle of 0% or 100% there is no audible sound, as there is no change in amplitude that constitutes oscillation.

Pulse Width Modulation (PWM): Pulse width modulation affects how the pulse width changes over time while you are holding a note. The pulse width can be modulated by a modulation source such as an LFO or an envelope, resulting in a thicker or more harmonically interesting sound.

Resonance: This filter parameter emphasised the frequencies around the cutoff frequency. The Super 6's low-pass filter can be driven into self-oscillation if you set the resonance to its highest value. In this case, the filter generates a pitch determined by the cutoff frequency and a timbre that sounds like a sine wave.

Sequencer: A sequencer is a modulation source that acts like a recording and playback device, sending control signals to a variety of parameters per step, the smallest unit of a sequence. The Super 6's sequencer allows recording of up to 64 steps and was primarily designed for recording and editing note events.

Split Point: The split point is a note on the Super 6's keyboard relative to which the audio signals of DDS 1 and DDS 2 or DDS 1 and an external audio signal are be crossfaded if you enable X-Fade mode.

Sub-Oscillator: A sub-oscillator is an oscillator with a fixed waveform that is an octave or more below the frequency of the oscillator to which it is tied. In the case of the Super 6, the enabled sub-oscillator replaces the audio signal from DDS 2. Its waveform is a square wave pitch-locked one octave below the frequency of DDS 1.

Super Mode: Super mode is a unique feature that takes advantage of the Super 6's stereo signal path. In both available super modes, DDS 1 can be dynamically de-phased in the stereo field, resulting in a thick, wide sound from a single oscillator.

Super Wave Modulation (SWM): When super mode is engaged, super wave modulation determines how much modulation depth is applied to the DDS 1 detune spread modulation.

Swing: Swing is a rhythmic variation in which the first and second consecutive notes of a two-part beat pattern are alternately lengthened and shortened. The Super 6 offers five different swing settings when you engage arpeggiator or sequencer playback, ranging from none to pronounced. Use swing to make your pattern 'bounce'.

Sync: Also known as ‘hard sync’, this function forces DDS 2 to restart its duty cycle each time DDS 1’s duty cycle begins. By setting the frequency of DDS 2 to a higher pitch than DDS 1, you can create complex and harmonically rich timbres, especially if you modulate the pitch of DDS 2, for instance with an envelope.

Velocity (VEL): Keyboard velocity allows a sound to respond dynamically when you hit a key. For example, if velocity controls the behaviour of the VCA, the softer you play, the quieter the sound will be. Conversely, the harder you hit the keys, the louder the sound.

Voltage Controlled Amplifier (VCA): A voltage-controlled amplifier controls volume of a sound. On the Super 6, envelope 2 is mapped to the VCA level by default. You can use this envelope to determine how a sound’s volume evolves over time.

Voltage Controlled Filter (VCF): This is the module that gave subtractive synthesis its name. The voltage-controlled filter is integral to the Super 6’s sonic character, shaping the sound of the oscillators by subtracting frequencies from their signals.

Waveform: A waveform describes the shape of a signal produced by an oscillator. Classic analog waveforms include shapes such as sine, sawtooth, square, pulse, triangle and white noise. A sine wave contains only the first harmonic, the fundamental, which is why it is considered the purest waveform. It is ideal for ‘glassy’ sounds or an added fundamental. A sawtooth wave contains both odd and even harmonics and is bright sounding. It can be used for creating brass, bass and string sounds. Square and pulse waves contain a wide range of odd harmonics. They sound hollow and can be used for reed-like sounds or basses. If you apply pulse width modulation to use a pulse wave, it can also be used for swirling string sounds. A triangle wave contains only odd harmonics and sounds very soft. It is particularly suitable for generating flute, organ or vocal sounds. White noise contains all frequencies and is the most common noise waveform. It is useful for creating wind or percussive sounds.

X-Fade: This function allows you to crossfade between the signals of DDS 1 and DDS 2 relative to an adjustable split point on the keyboard. The crossfade between the output signal of both oscillators covers a range of two octaves.

SUPPORT INFORMATION

If you are experiencing any issues with your Super 6, contact our technical support at support@udo-audio.com.

Please provide the following information when you get in touch with us:

- Instrument name
- Serial number
- Firmware version
- Purchase date (new or used) and location (country, dealer)

If you haven't already done so, make sure to register your product through our website.

You may also visit our FAQ section or user forum to check if your question has already been answered.

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