SUPER GEMINI / SUPER 8

OWNER'S MANUAL
VERSION [1.0] MARCH 2024

U·D·O

SUPER GEMINI

SUPER 8

UDO SUPER GEMINI + UDO SUPER 8
OWNER'S MANUAL

VERSION 1.0 · MARCH 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:	90~240 VAC
Rated input frequency:	47-63 Hz
Power consumption:	50 W
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

Ben Crosland, Anthony Gillan, Axel Hartmann, Ann Hsu, Alexandros Liarokapis, Hazel Mills, Frank Rüffel, Klaus Weber and Angell Yu.

The UDO Team

Phillip Anthony, Ben Charlton, Chris Hayes, George Hearn, Magnus Hearn, Mike Hiegemann, Ben Miles and Jamie Tucker.

Manual and illustrations by Mike Hiegemann.

INTRODUCTION

Hello and welcome to your new UDO synthesizer. What sits beneath your fingers is an instrument crafted in recent times yet having benefitted from a long history of players and pioneers, of engineers and artists, technicians and tinkerers over several generations.

There is a magic in synthesizers, a detail between the lines, in the space between the traces of copper, and the complex interactions of code and fabrications of metal. If we are lucky, we can find ourselves in a state of flow as we play, comfortably putting aside the challenges of our inner world and the wider world beyond, and fully focus on the unique joy of what we find ourselves engaged with. If we are luckier still, we can share this moment with others. Yet, if you are one of those who have come to own this synthesizer, you have probably already learned the healing and restorative qualities of creativity as its own reward.

The creators of what you have before you understand the power of play, and what you play now while not a simple instrument, works hard to conceal its complexities and its secrets and presents you only with straightforward expressive controls of a superior mechanical quality to encourage you to seek and explore.

A journey into sound now awaits you and it is a journey with no wrong turns, and no dead-ends. It is an adventure of spectral dynamics, of glittering frequencies and shattering subharmonics, of comfort and discord, expectations and surprises – and accompanying you always will be your loyal workmate and trusted companion, your brand new UDO synthesizer.

I hope you learn to love playing this instrument as much as I do, and when you find that moment, that sound, that glimpse of feeling or emotion that transcends your moment in time, then we have truly achieved something good together.

George Hearn,
Director UDO Audio Ltd

The UDO Super Gemini and Super 8 are polyphonic, bi-timbral synthesizers based on a mixture of analog and digital technologies, with the Super Gemini offering 20 voices and the Super 8 16 voices. Both instruments combine the aesthetics and sonic character of vintage classics with state-of-the-art synthesis technology and were designed to be flexible, powerful and immediate and, above all, to provide you with gorgeous sounds times two!

DDS Oscillators

Direct Digital Synthesis is the signal generation method employed by both oscillator cores of the Super Gemini and Super 8. 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.

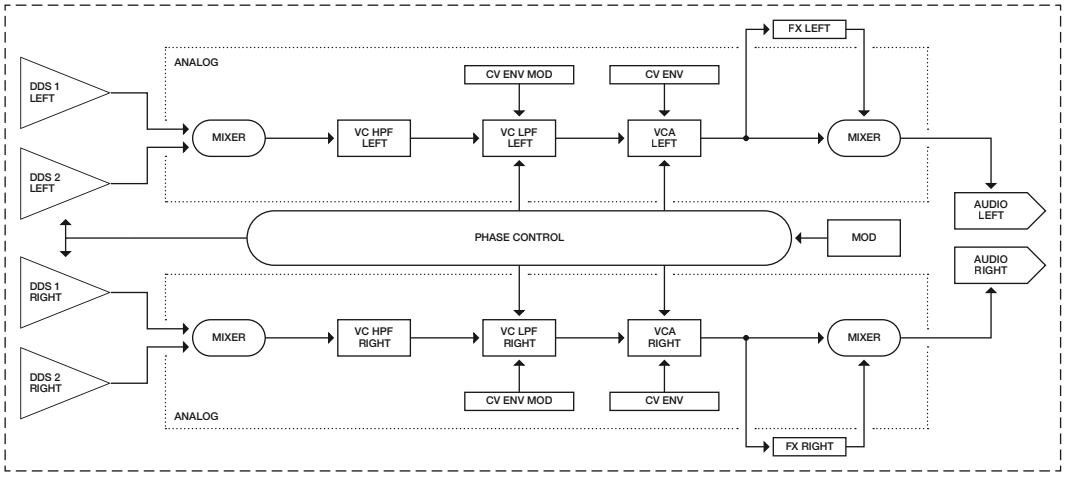
Each numerically controlled oscillator then uses its own DAC, also running at the same high sample rate, to convert the samples to analog voltages before being filtered by a preliminary analog low-pass filtering stage.

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 Gemini and Super 8 feature a true stereo signal path in which its 20 (Super Gemini) or 16 (Super 8) voices are twinned to form either ten (Super Gemini) or eight (Super 8) stereo “super voices” in single mode or five (Super Gemini) or four (Super 8) stereo “super voices” in dual or split mode. Consequently, the left and right channels (and your ears) are each assigned a complete synthesizer voice per layer.

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.

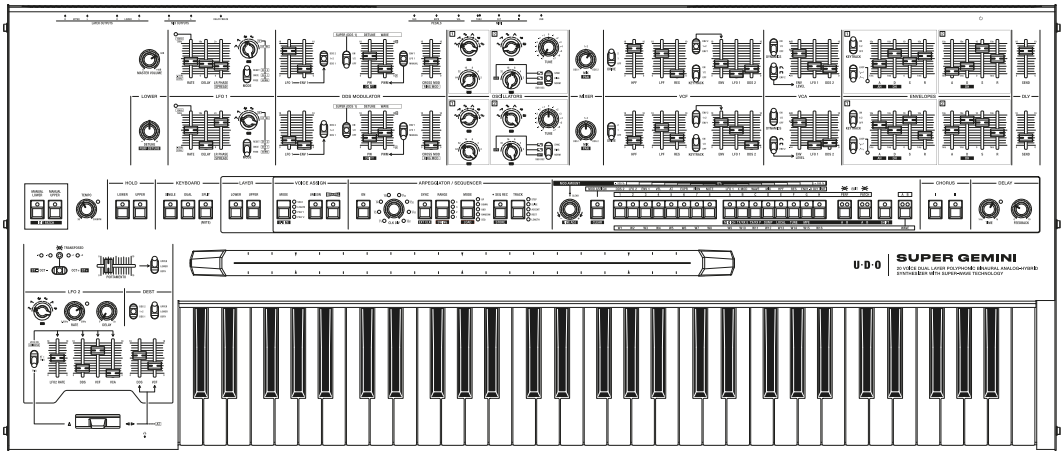


The signal path per layer.

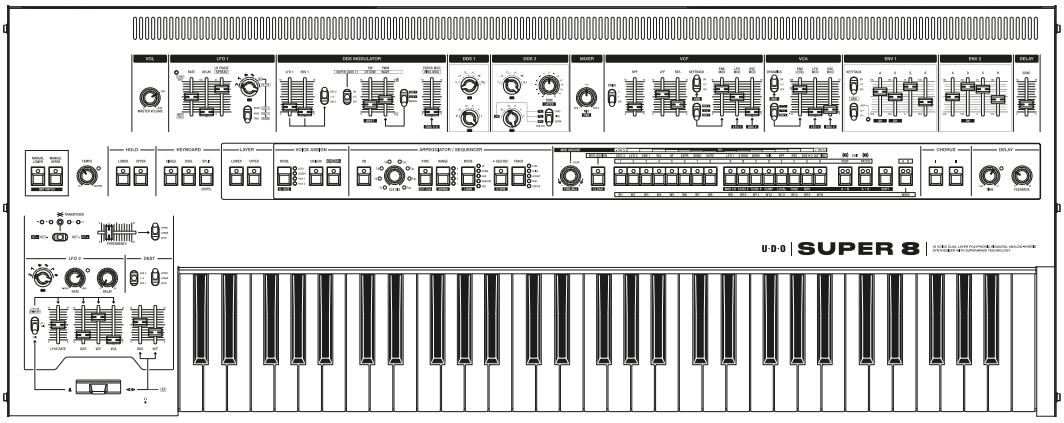
QUICK START

The Super Gemini and the Super 8 were designed for hands-on performance and experimentation. Not only do we think both instruments sound great, but they also offer a pleasant immediacy when interacting with the sound. All of the primary controls are accessible directly from the front panel, making programming an incredibly intuitive experience that will spark your creativity in the studio and on stage.

Apart from the number of available voices, the Super Gemini and the Super 8 are identical, both in terms of the synthesis engine used and the connections. Hence, this manual covers both instruments. If a function or parameter is only available on one of the two instruments, this is indicated.



The front panel of the Super Gemini.



The front panel of the Super 8.

The upper part of the front panel features individual sound shaping controls for each of the two layers, allowing you to program and edit patches. The Super Gemini has two individual rows of controls for each layer, allowing simultaneous editing, while the Super 8 has one row of controls used for both layers.

The bottom row of the front panel contains shared controls for both layers, which you can use to load a patch into a layer or change the arpeggiator, sequencer or effect settings for each layer, among other things.

To the left of the 61-note velocity-sensitive Fatar keyboard with polyphonic aftertouch is the performance control section, which contains performance-related elements such as the bender, transpose and portamento controls as well as a second LFO. To further increase the expressiveness of your performance, the Super Gemini is also equipped with a custom-designed ribbon controller.

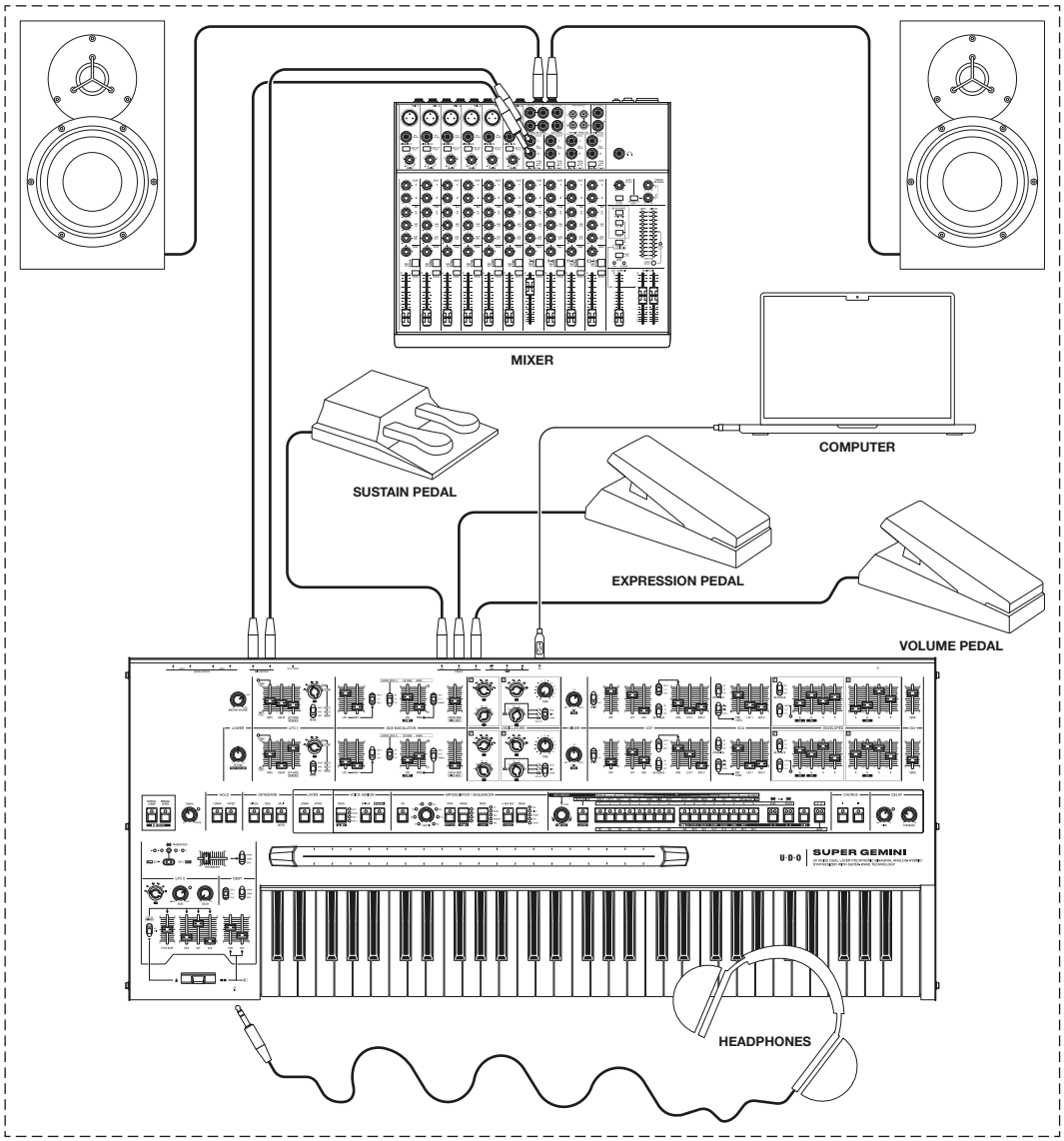
Exploration and experimentation are downright encouraged, so feel free to get started right away and create your own sounds. The best way to find out how your new instrument works is to get involved! You can always come back later and read more about the individual sound shaping tools in the following chapters of this manual. We hope you enjoy playing and tweaking your new instrument as much as we do!

Setting up Your Instrument

Follow the steps below to set up your brand new synthesizer:

1. Plug the power cable into the power connector on the rear panel.
2. Use unbalanced 1/4-inch audio cables to connect the outputs labelled **MIX OUTPUT** to your mixer or audio interface, or connect headphones to the headphone output on the front left.
3. Turn on the instrument.
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.

On pages 28-29 you will find a complete overview of all connections.



One of many ways to set up your instrument, using the Super Gemini as an example.

Performances

A performance comprises a complete instrument setup, consisting of two layers – an upper and a lower layer – each with its own patch (see pages 20-25) and overarching parameter settings (see pages 79-83).

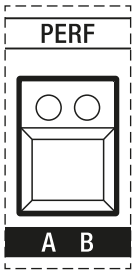
A performance is always loaded because it provides the framework for individual sounds or patches. It can consist of one patch in single mode, two stacked patches in dual mode, or two patches assigned to the left and right halves of the keyboard in split mode. See pages 82-83 for more details on each of these modes.

*The **MASTER VOLUME** setting is not part of a performance and must always be adjusted manually.*

A total of 128 performances are accessible from the front panel. They are organised in two groups (**A** and **B**) of 8 banks (**A**-**H**) featuring 8 performances each. You can edit these or use the dedicated memory slots to store your own performances.

Loading a Performance

First of all, make sure you are in performance mode. The instrument defaults to this mode when you turn it on, but otherwise it is activated by pressing the button labelled **PERF**.

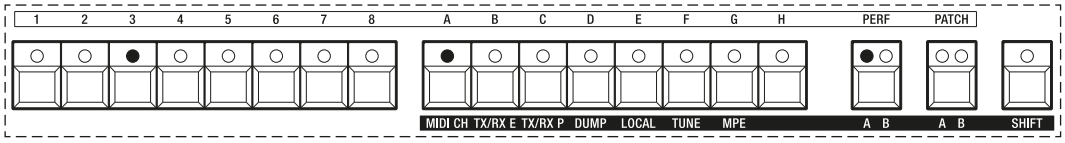


The performance mode button.

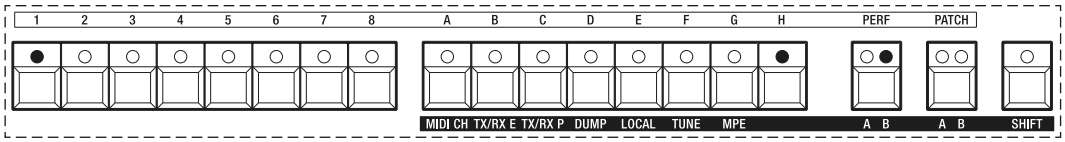
Pressing **PERF** gives you access to group **A**, indicated by the lit LED above the letter **A**. To access the performances of group **B**, press **SHIFT** and then press **PERF**. The LED above the letter **B** will now light up.

Each lettered button (**A**–**H**) allows you to select a bank. Pressing a numbered button (**1**-**8**) loads one of eight performances within a selected bank.

The LEDs of buttons **1**-**8**, **A**-**H** and **PERF** indicate which performance is loaded. In the first example, performance **3** from bank **A** of group **A** is loaded:



In the second example, performance **1** from bank **H** of group **B** is loaded:



To load a different performance 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 performance from a different bank:

1. Press one of the seven lettered buttons (**A**-**H**) that are currently unlit. Its LED will then light up, indicating that a different bank 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 performance. A new performance is loaded only by pressing one of the numbered buttons after a bank has been selected.

To load a performance from a different group:

1. Press the **PERF** button to select group **A**, or press the **SHIFT** button and then press **PERF** to select group **B**.
2. Press one of the lettered buttons (**A**-**H**) to select a bank. Its LED will then light up.
3. Press one of the numbered buttons (**1**-**8**). Its LED will then light up.

Editing a Performance

A performance is edited as soon as you alter any parameter except **MASTER VOLUME**. When a setting is changed, the LED of the currently lit numbered button (**1**-**8**) will start flashing, indicating you are in edit mode.

Storing a Performance

To store a performance:

1. Press the **PERF** button to select group **A**, or press the **SHIFT** button and then press **PERF** to select group **B**.
2. Press one of the lettered buttons (**A**-**H**) to select a bank. Its LED will then light up.
3. 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 performance has been stored.

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

When you store a performance, you are essentially saving a snapshot that contains copies of the patches in each layer. As a result, any changes applied to any of the layers are also saved without changing the original patches.

If you want to use a specific layer of a performance as part of other performances, you need to save it as a patch. This makes the patch banks the place to store your favourite or most important sounds. See page 24 on how to store patches.

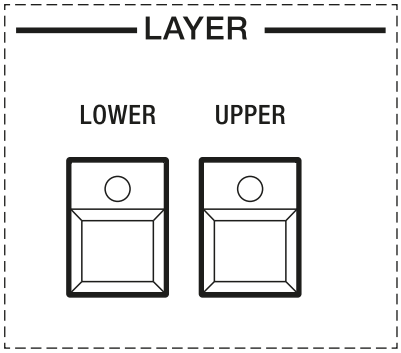
Patches

A patch is a stored set of parameters that determine the characteristics of a sound. Patches are each loaded into one of the layers of a performance.

A total of 256 patches are accessible from the front panel. They are organised in two groups (**A** and **B**) of 16 banks (**A1**-**H2**) featuring 8 patches each. You can edit these or use the dedicated memory slots to store your own patches.

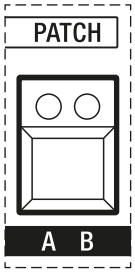
Loading a Patch

First of all, select the layer you want to load a patch into by using the buttons in the layer section. Pressing the **UPPER** button selects the upper layer, pressing the **LOWER** button selects the lower layer.



The layer section.

After selecting a layer, make sure you are in patch mode. Patch mode is accessed by pressing the button labelled **PATCH**.



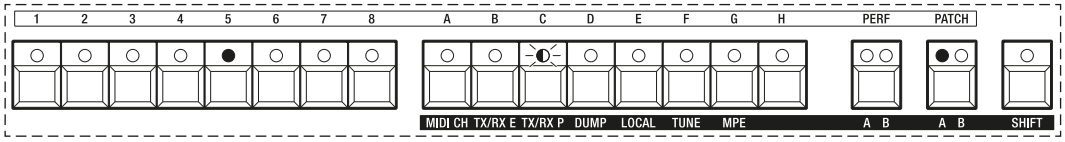
The patch mode button.

Loading a patch is similar to loading a performance: Pressing **PATCH** gives you access to group **A**, indicated by the lit LED above the letter **A**. To access the performances of group **B**, press **SHIFT** and then press **PATCH**. The LED above the letter **B** will now light up.

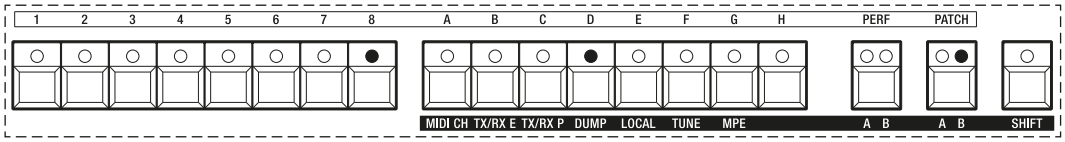
Each lettered button (**A**–**H**) allows you to select two banks: **A1** and **A2**, **B1** and **B2**, etc. A lit LED indicates one of the first banks (**A1**-**H1**) is selected, while a flashing LED indicates one of the second banks (**A2**-**H2**) is selected. To toggle between banks with the same 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**, **A**-**H** and **PATCH** indicate which patch is loaded. In the first example, patch **5** from bank **C2** of group **A** is loaded:



In the second example, patch **8** from bank **D1** of group **B** 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.*

To load a patch from a different group:

1. Press the **PATCH** button to select group **A**, or press the **SHIFT** button and then press **PATCH** to select group **B**.

2. 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.

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

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.

To load the init patch:

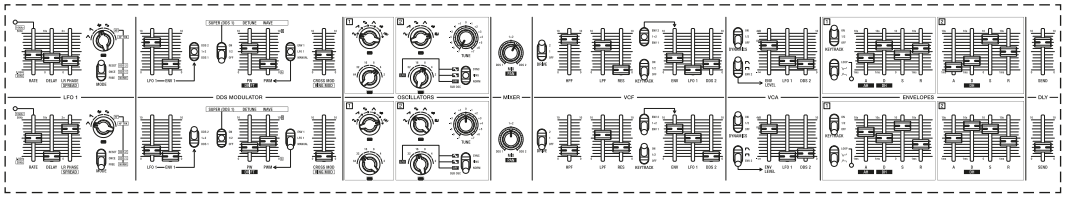
- Press **SHIFT** and then press either the **MANUAL UPPER** button to load the init patch into the upper layer, or the **MANUAL LOWER** button to load the init patch into the lower layer.

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

If you load the init patch into one of the layers, the performance will switch to single mode.

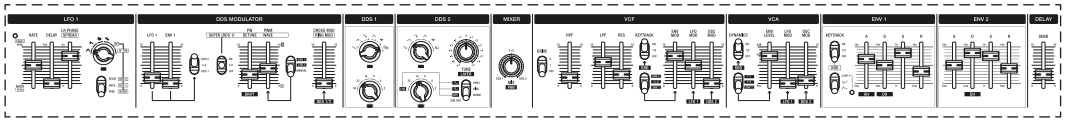
Editing a Patch

The sound shaping controls for each of the two layers are located in the upper part of the front panel. On the Super Gemini, the top row contains the controls for the upper layer, while the second row contains the controls for the lower layer.



The sound shaping controls for both layers on the Super Gemini.

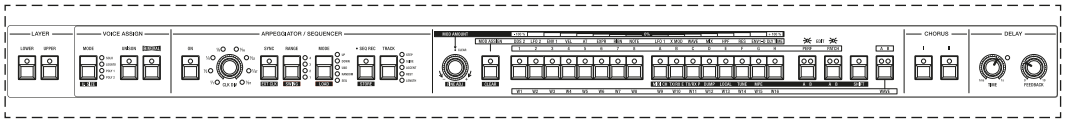
On the Super 8, both layers are edited using the controls in the top row, and you can decide which layer you want them to affect by using the layer select buttons **UPPER** and **LOWER** in the layer section.



The sound shaping controls for both layers on the Super 8.

The framed sections in the bottom row, starting with the layer section, contain patch-related controls that are shared for both layers. This front panel area includes the voice assign, arpeggiator and sequencer sections, the multifunctional select buttons, as well as the chorus and delay effects.

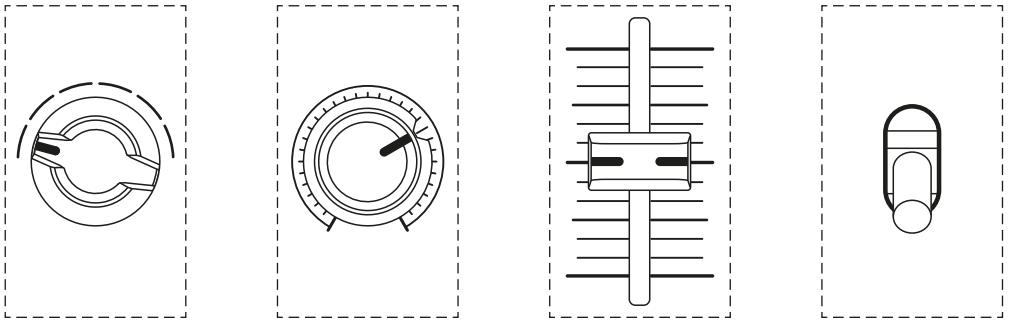
You can toggle between the layer-specific settings controlled in these sections by pressing the **UPPER** and **LOWER** buttons in the layer section.



The boxed front panel area that contains shared controls for both layers.

In addition to these sections, the performance control section to the left of the keyboard allows you to further shape your sounds by using a second LFO, adding portamento and determining what patch parameters of each layer should be affected by pitch bend and polyphonic aftertouch. See pages 68-74 for more details.

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 in the upper or lower layer, press the **PATCH** button to select group **A**, or press the **SHIFT** button and then press **PATCH** to select group **B**.

2. 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.

3. 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. If you want to select a patch from another group, select the desired group by pressing the **PATCH** button either alone or in combination with the **SHIFT** button before selecting a bank.

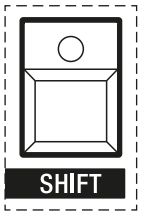
2. Press the same numbered button again to return to the patch you just edited.

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 99-101) 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 **DRIFT**, **PAN**, 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.

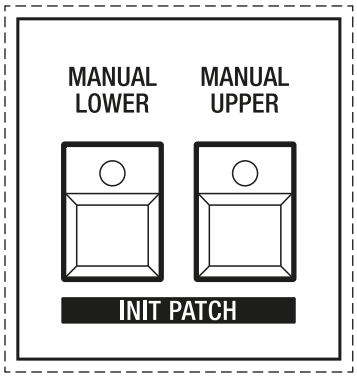
What You See Is What You Get: Manual Mode

Entering manual mode ignores the current patch settings and prompts the currently selected layer 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 either the **MANUAL UPPER** button for the upper layer or the **MANUAL LOWER** button for the lower layer.

2. To return to performance or patch mode, simply press one of the numbered patch or performance select buttons.



The manual mode buttons for both layers.

Level Up!

The Super Gemini and Super 8 were 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 understand the potential of your new instrument.

All functions and capabilities are explained in more detail in the following chapters of this manual.

For information on adapting the instrument to your specific environment, for example to use it alongside other MIDI instruments or to control it from a digital audio workstation (DAW) or an external sequencer, we recommend that you read the paragraphs on global settings and connections.

Throughout this manual you will also find some useful hints and tips to help you familiarise yourself with the instrument and its sound shaping parameters.

Enjoy!

UPDATING THE FIRMWARE

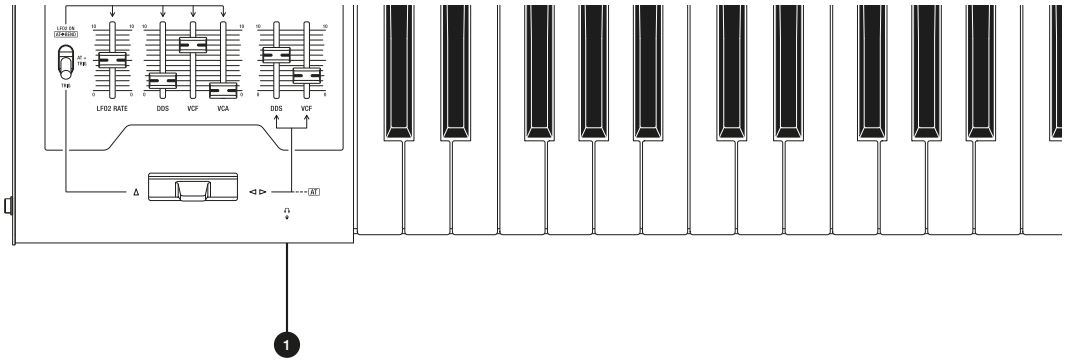
Follow the steps below to update the firmware of your instrument:

1. Unlock the instrument's drive:
   - Turn off the instrument and wait a few seconds.
   - Whilst holding down the **PATCH** button, turn on the instrument and continue to hold the **PATCH** button.
   - The LEDs of buttons **1**-**8** and **A**-**H** light up one by one to indicate the loading progress. Then all LEDs extinguish and the envelope LED(s) will start flashing. (If this is not the case, repeat step 2.)
   - Release the **PATCH** button.
2. Connect the instrument to your computer using a USB cable.
3. The instrument appears on your computer as a drive named **GEMINI** or **SUPER8**.
4. Click on the icon of the instrument's drive. There should be no other ".bin" file on the drive, but if there is, delete it and make sure you empty the trash if you are a macOS user, or the update will not be possible.
5. Copy the firmware file "gem_v*.**.bin" from your computer to the instrument's 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 instrument, wait a few seconds and turn it on again. The instrument reboots after several LED sequences on the LFO LED(s), followed by a combination of briefly lit lettered select buttons. Each time you turn on the instrument, this combination of lit buttons will indicate the currently installed firmware version.

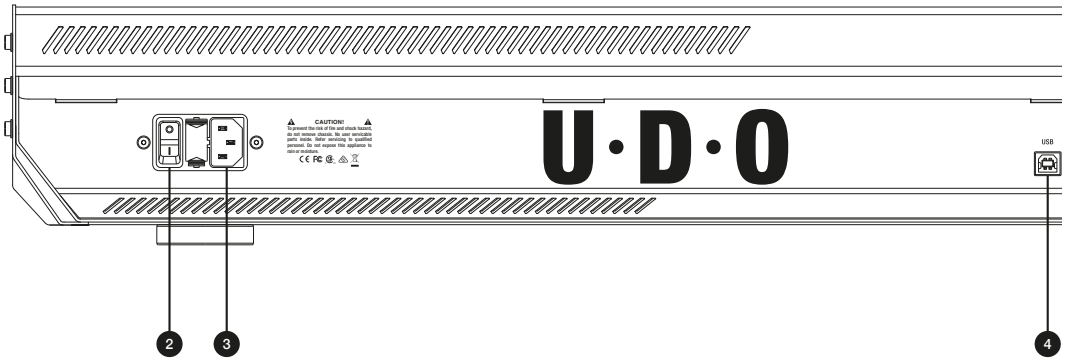
If the instrument's drive is unlocked, you will not be able to play it, but can only manage or change the files stored on it.

The latest firmware version can be downloaded at udo-audio.com/downloads.

CONNECTIONS



1. Headphone Outputs: Connect a 6.35 mm or a 3.5 mm stereo headphone jack to the respective outputs on the left front of the instrument. The overall volume of the headphone output is controlled by the **MASTER VOLUME** control.

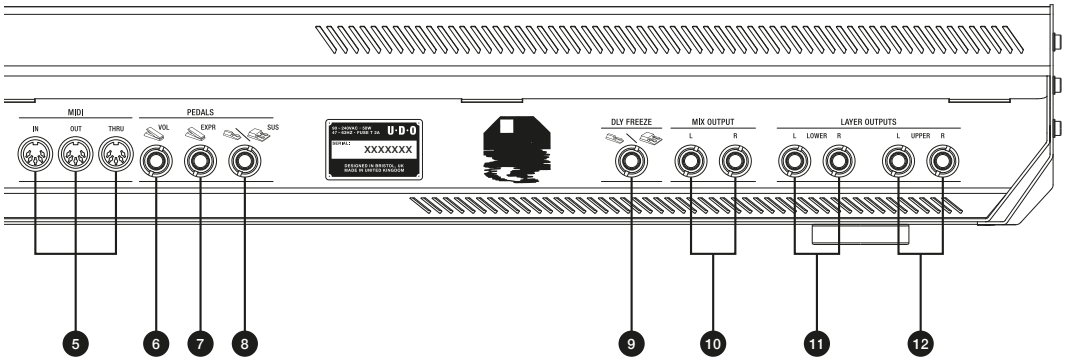


2. Power Switch: Use this switch to power cycle the instrument.

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

4. USB Port: Connect the instrument to your computer using the included USB cable for bidirectional MIDI communication, file management and firmware updates. The instrument does not require any drivers to interface with a computer.

Do not connect a USB cable when using the MIDI DIN ports. The simultaneous use of DIN and USB-MIDI is not supported.



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

6. Volume Pedal Input: Connect an expression pedal to this input to use it as a volume pedal. 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.

7. 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 instrument'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.

8. Sustain Pedal Input: Connect a single or a dual footswitch to this input to sustain notes during your performance. A single footswitch sustains upper layer notes. When a dual footswitch is connected, the left pedal sustains the upper layer notes while the right pedal sustains the lower layer notes. Upon power cycling, the instrument 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.

9. Delay Freeze: Connect a single or dual footswitch to this input to create sound-on-sound loops, the length and depth of which you can control in the delay section. If a single footswitch is used, it will control the upper layer delay freeze function. If a dual footswitch is connected, the left pedal controls the upper layer delay freeze and the right pedal controls the lower layer delay freeze. For more information on delay freeze, see page 67.

10. Mix Output (Left and Right): This pair of connectors outputs the upper and lower layer audio signals. Connect both outputs to your mixer or audio interface using unbalanced 6.35 mm audio jack cables.

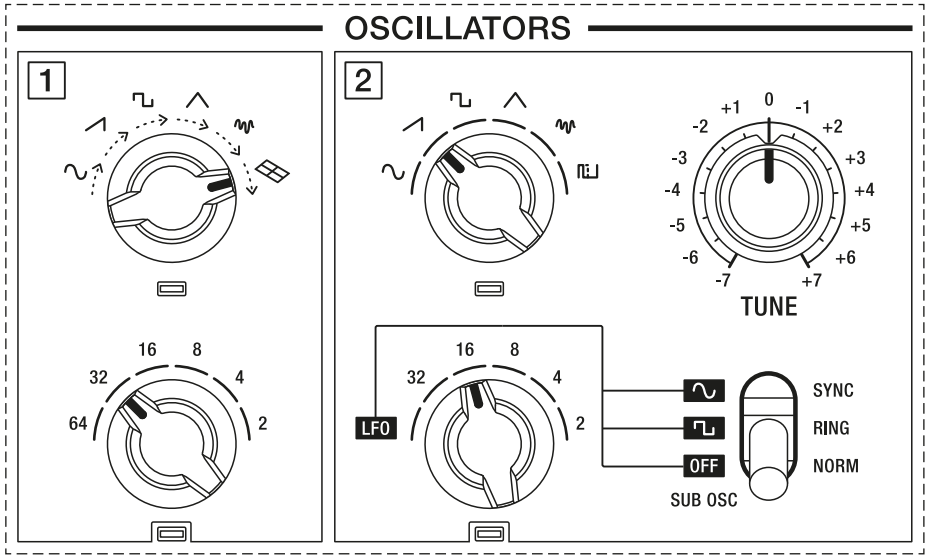
11. Lower Layer Output (Left and Right): This pair of connectors outputs the lower layer audio signal, allowing you to mix and process it individually. Connect both outputs to your mixer or audio interface using unbalanced 6.35 mm audio jack cables.

12. Upper Layer Output (Left and Right): This pair of connectors outputs the upper layer audio signal, allowing you to mix and process it individually. Connect both outputs to your mixer or audio interface using unbalanced 6.35 mm audio jack cables.

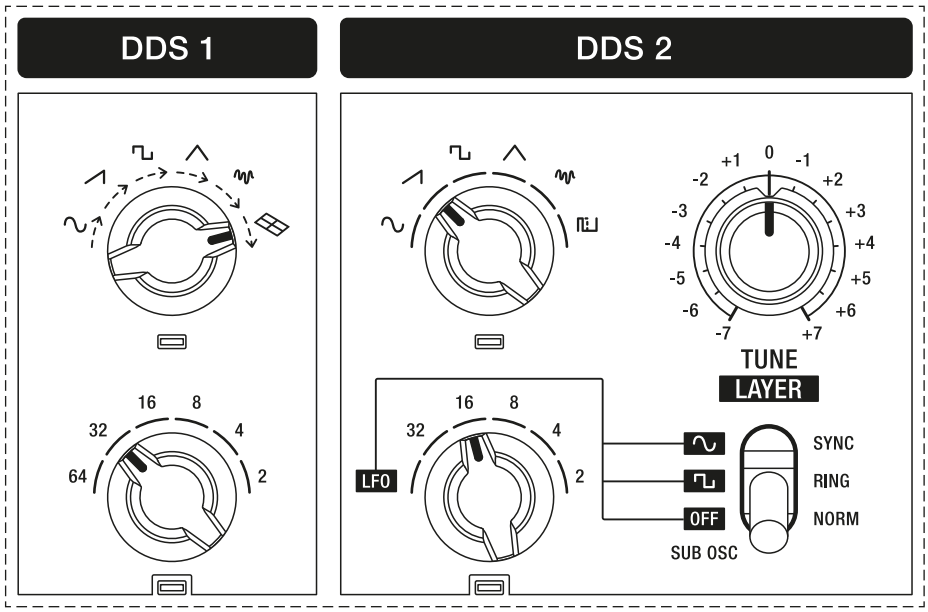
This chapter explores the sound design capabilities of the Super Gemini and Super 8 by explaining the functionality of all patch-specific parameters related to the manipulation of sound. They are identical for both layers.

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 of the Super Gemini.



The oscillator section of the Super 8.

The primary sound sources are the 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 an additional fundamental tone 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.

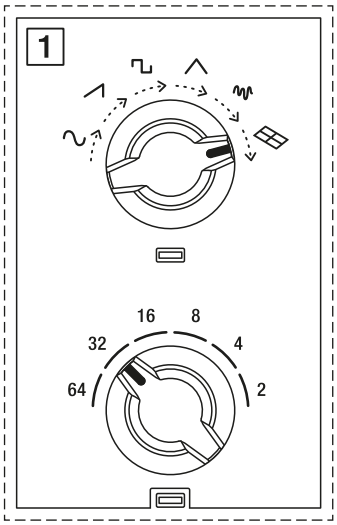
A **triangle wave** contains only odd harmonics and sounds very soft. It is particularly suitable for generating flute or organ sounds.

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

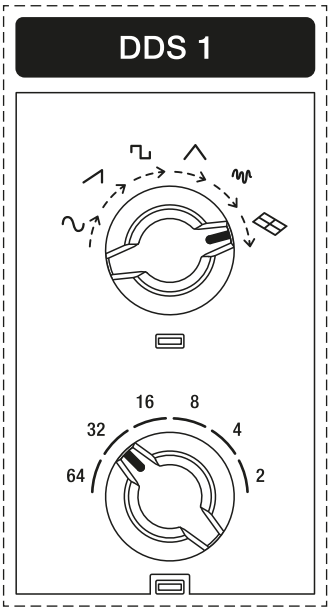
In addition, DDS 1 offers up to 32 digital-sounding alternative waveforms organised in two groups. They are user definable (see page 109) and allow for a sheer unlimited range of sounds.

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 activating one of the two super modes in the DDS Modulator section. Essentially, this means that DDS1 contains seven free-running oscillators, which give the instrument its characteristically rich and wide sound. See pages 61-62 for more details.



The DDS 1 controls of the Super Gemini.

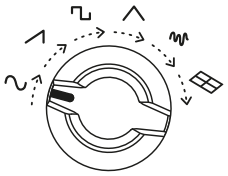
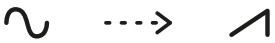
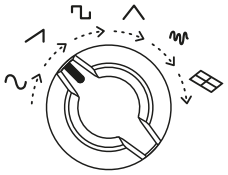
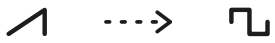
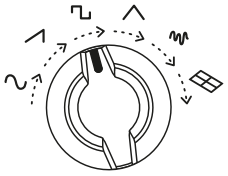
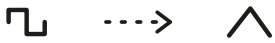
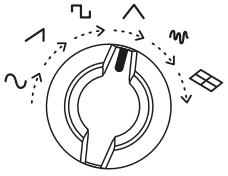
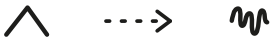
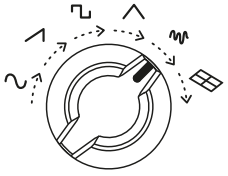
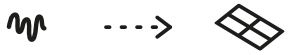
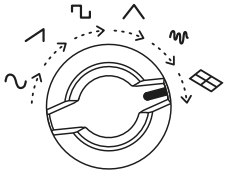
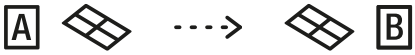


The DDS 1 controls of the Super 8.

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.

As indicated by the dashed arrows below the waveform icons, you can also morph between two adjacent waveforms using the **PWM**/**WAVE** fader in the DDS Modulator section (see page 62). When the **WAVEFORM** control is set to the rightmost position, use the **PWM**/**WAVE** fader to morph between the currently loaded alternative waveforms assigned to channels **A** and **B**.

The following table lists the available wave modulation options:

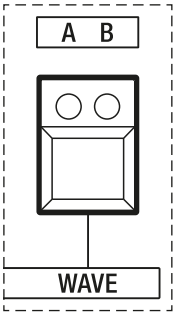
DDS 1 Waveform Setting	Wave Modulation
	
	
	
	
	
	

*When moving the **WAVEFORM** control to the rightmost position, you will hear the alternative waveform assigned to channel **A**. The waveform assigned to channel **B** is made audible by setting the **PWM**/**WAVE** fader to the highest position.*

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.

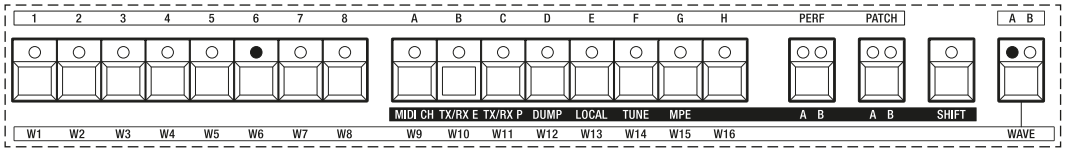
The alternative waveforms are organised in two groups of 16 waveforms each, giving you access to a total of 32 alternative waveforms. You can choose which channel (**A** or **B**) to assign one of these waveforms to by pressing the **WAVE** button.

Once you have selected a channel, you can use the select buttons **1**-**8** and **A**-**H**, which are labelled **W1**-**W16** (waveform 1 to 16) in this mode, 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**.

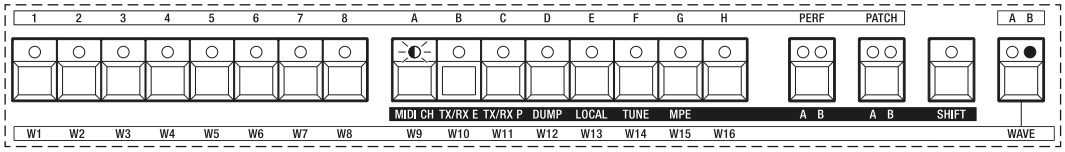
If you load a waveform from the first group of 16 waveforms (1-16), the corresponding LED lights up. If you load a waveform from the second group of 16 waveforms (17-32), the LED starts to flash.

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.

The LEDs of buttons **W1**-**W16** indicate which waveform is loaded, while the two LEDs of the **WAVE** button indicate which channel it is assigned to. In the first example, waveform **6** is loaded and assigned to channel **A**:



In the second example, waveform **25** is loaded and assigned to channel **B**:



To load alternative waveforms:

1. Press **WAVE** or move the DDS 1 waveform rotary switch to the rightmost position. LED **A** will light up, indicating that you are now in wave selection mode for channel **A**.

2. Play some notes and press one of the select buttons (**W1**-**W16**) once or twice to load the waveform you would like to assign to channel **A**.

3. Press **WAVE** again. LED **B** will light up, indicating that you are now in wave selection mode for channel **B**.

4. Play some notes and press one of the select buttons (**W1**-**W16**) once or twice to load the waveform you would like to assign to channel **B**.

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

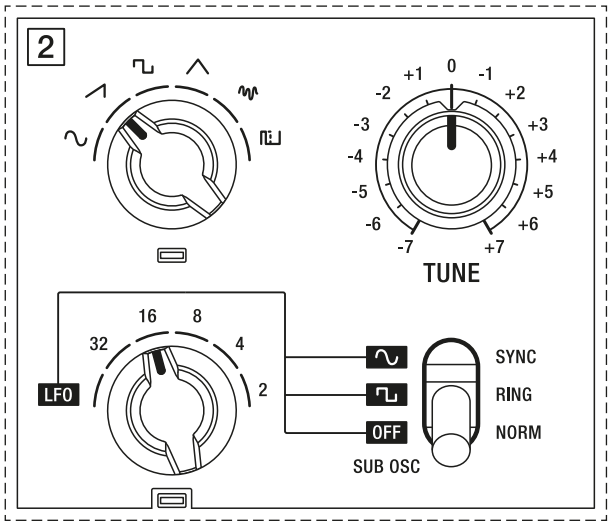
Each patch remembers the alternative waveforms it was saved with. Even if you were to replace all the alternative waveforms the instrument is shipped with, the factory patches would not change.

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

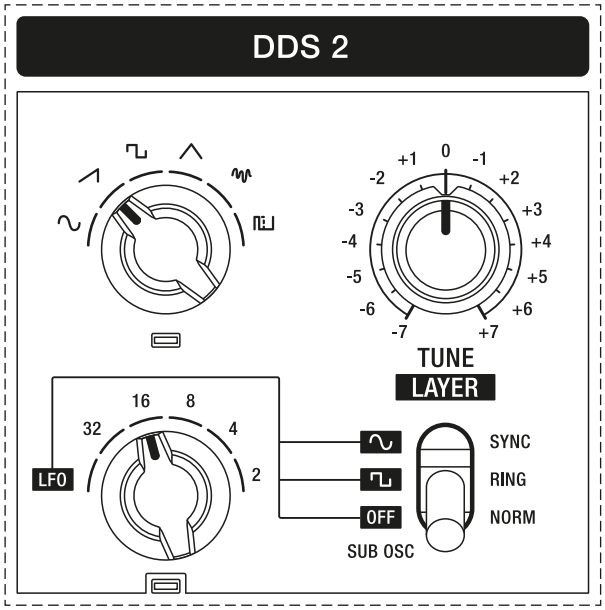
DDS 2 Parameters

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.

In contrast to DDS 1, the phase of the DDS 2 waveform is reset to zero each time a note is played, which allows for binaural pitch and pulse width modulation by LFO 1. This would not be possible with a free-running oscillator such as DDS 1, as there would be no fixed starting point from which the phase could be offset.



The DDS 2 controls of the Super Gemini.



The DDS 2 controls of the Super 8.

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.

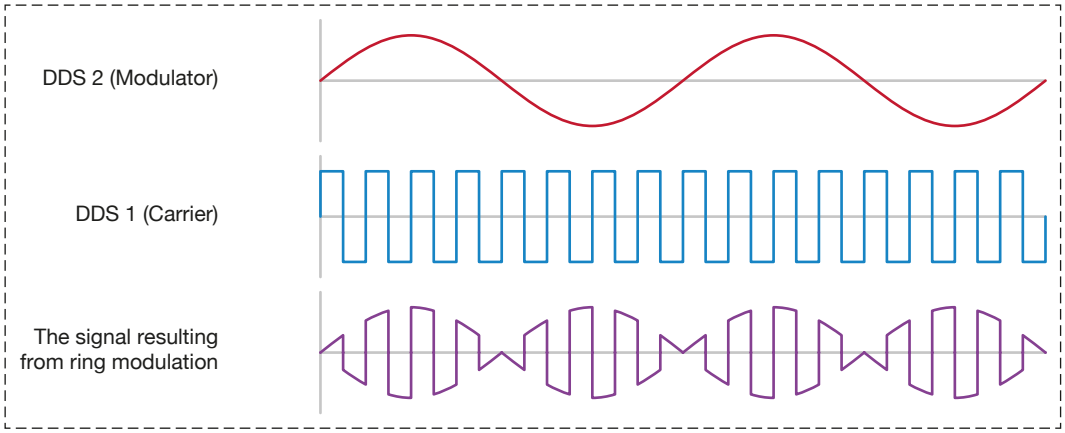
RANGE: Use this rotary switch to adjust the coarse frequency of DDS 1 over a range of 32 to 2 feet. If you turn this control to the leftmost position, DDS 2 will act as an additional LFO. In this mode, you can also enable a sub-oscillator. See page 39 for more details.

TUNE: This control allows you to fine-tune the frequency of DDS 2 over a range of +/-7 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.

LAYER: This parameter is only available on the Super 8. In shift mode, the **TUNE** control determines the fine tuning setting for the each layer. Turning the control clockwise increases the frequency. Turning the control counter-clockwise decreases the frequency. You can adjust the fine tuning over a range of +/- 7 semitones.

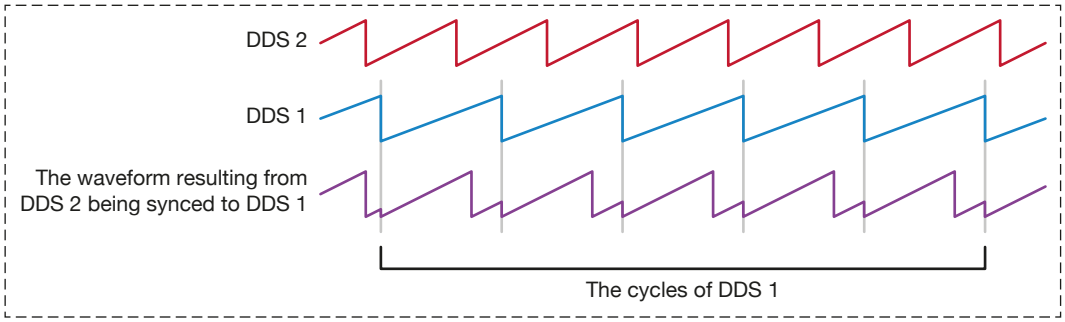
The toggle switch in the DDS 2 section allows you to choose from three different modes:

- **NORM:** This is the default mode for DDS 2, in which it behaves like a normal oscillator.
- **RING:** Selecting this option activates the ring modulator. Ring modulation combines two signals, a carrier (DDS 1) and a modulator (DDS 2), and outputs their sum and difference while subtracting the frequencies of the original signals. This can result in clangorous tones, especially when applied to harmonically rich waveforms such as sawtooth or square. The output signal of the ring modulator is passed through the channel of DDS 2. To fade in the ring modulated signal, turn the **MIX** control clockwise.



Ring modulation between DDS 1 and DDS 2. In this example, DDS 1 is set to a square wave at 8 times the frequency of DDS 1, which is set to a sine wave.

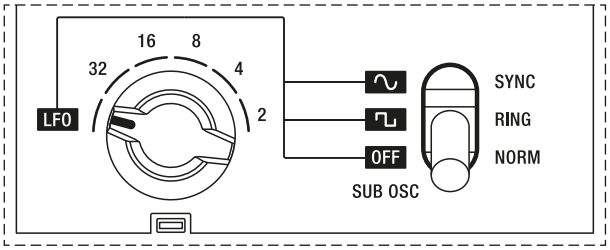
- **SYNC:** Selecting this option, also known as “hard sync”, forces DDS 2 to restart its duty cycle each time the duty cycle of DDS 1 begins. By setting the frequency of DDS 2 to a higher pitch than DDS 1, you can create complex and harmonically rich timbres, especially if you modulate the pitch of 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.

Using DDS 2 as an LFO

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



DDS 2 set to LFO mode.

When DDS 2 is used as an LFO, its signal is no longer fed into 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 using 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 that enables interesting modulation results.

You can route DDS 2 in LFO or default mode to all available modulation destinations via the modulation matrix. See pages 84-89 for more information on how to use the modulation matrix.

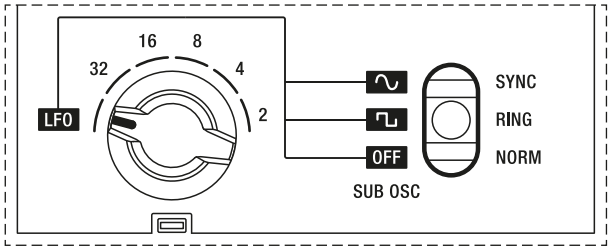
Using the Sub-Oscillator

Activating 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.

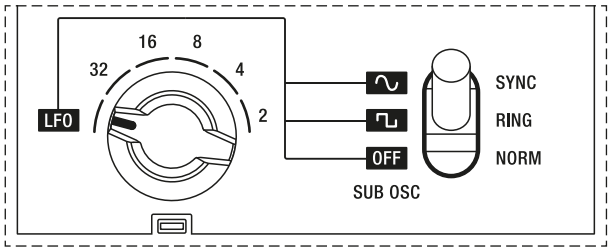
The waveform of the sub-oscillator can be either a square wave or a sine wave. Select the square wave to add harmonically rich bass tones to your sound. Select the sine wave if you want to subtly boost the lower harmonics.

To activate the sub-oscillator:

1. Turn the **RANGE** control to the leftmost position labelled **LFO**.
2. Flip the mode toggle switch to either the middle position to select the square wave sub-oscillator or the upper position to select the sine wave sub-oscillator.



The sub-oscillator set to square wave.



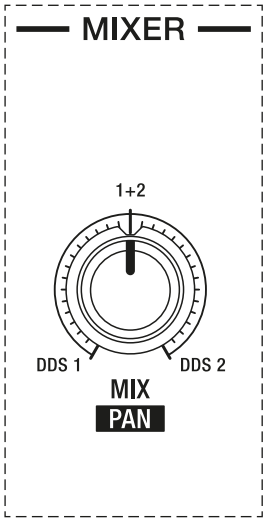
The sub-oscillator set to sine wave.

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

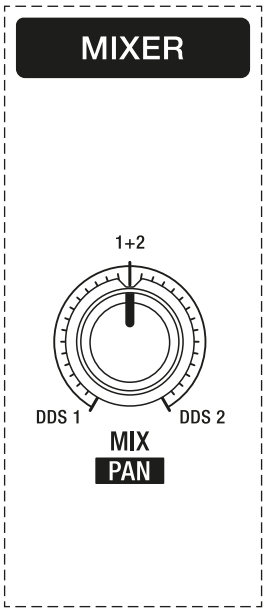
When the sub-oscillator is activated, DDS 2 can still be used as an additional LFO.

Mixer

In the mixer section you can control the balance between the audio signals from DDS 1 and DDS 2 or DDS 1 and the sub-oscillator.



The mixer section of the Super Gemini.



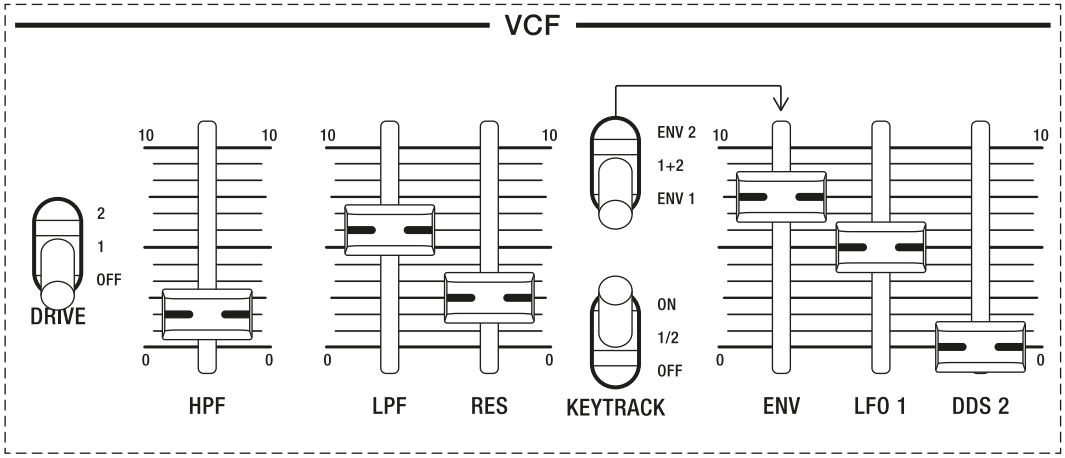
The mixer section of the Super 8.

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 can be heard. Likewise, only the audio signal from DDS 2 or that of the sub-oscillator can be heard if you turn the control to the rightmost position.

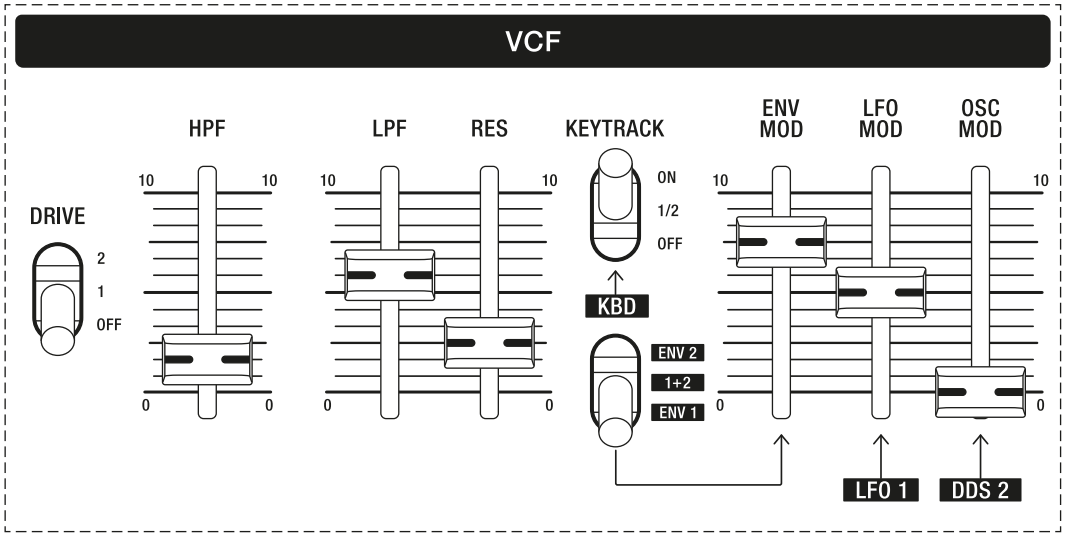
PAN: In shift mode, the **MIX** parameter becomes a pan control, allowing you to position a layer within the stereo field. At 12 o'clock, the layer is centred. If set to the leftmost position, the layer is hard-panned to the left. If set to the rightmost position, the layer is hard-panned to the right.

VCF (Voltage Controlled Filter)

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



The filter section of the Super Gemini.



The filter section of the Super 8.

The main filter is a 4-pole, 24 dB per octave, analog resonant low-pass filter that uses a classic polysynth filter design from Sound Semiconductor (SSI). It is preceded in the signal chain by a 1-pole, 6 dB per octave analog high-pass filter.

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.

There are numerous ways to modulate the cutoff frequency of the low-pass filter, making the filter section extremely versatile and suitable for a wide range of sounds.

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

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.

HPF: This fader controls the cutoff frequency of the high-pass filter. This filter can be useful for removing “muddy” low ends from pad sounds, particularly in recording or mixing situations.

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

RES: This fader controls the amount of resonance. If you increase the resonance, the frequencies around the cutoff frequency set with the **LPF** fader will be emphasised and more pronounced.

*The low-pass filter can be driven into self-oscillation if you set **RES** to the 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 control or play the pitch of the low-pass filter using the keyboard if you activate key tracking.*

ENV/ENV MOD: This fader controls the amount by which either or both envelopes modulate the cutoff frequency of the low-pass filter over time. You can find out more about the envelope generators on pages 46-53.

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/LFO MOD: This fader controls the amount by which LFO 1 modulates the cutoff frequency of the low-pass filter.

DDS 2/OSC MOD: This fader controls the amount by which DDS 2 modulates the cutoff frequency of the low-pass filter. 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 cutoff frequency of the low-pass filter responds relative to the pitch of the notes played on the keyboard:

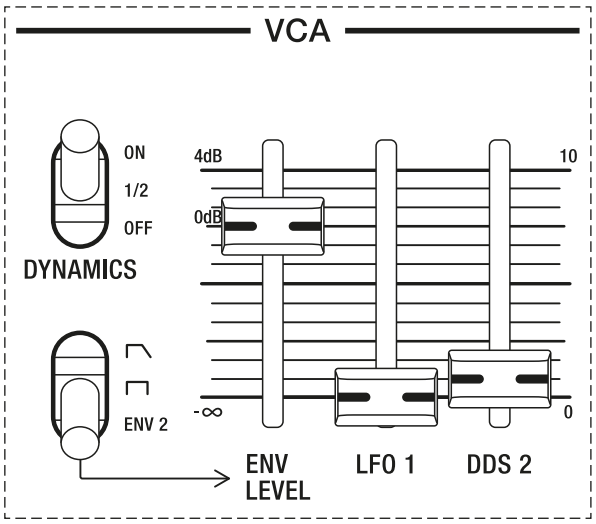
- **OFF:** With this setting, the cutoff frequency of the low-pass filter is unaffected by the pitch of the notes played on the keyboard.
- **1/2:** With this setting, the cutoff frequency of the low-pass filter follows the keyboard pitch in quarter tone steps, resulting in brighter sounding higher notes.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

*The low-pass filter is designed to respond to key tracking in a musical way. This in turn determines how far you can open the filter with the **LPF** fader. You can utilise the remaining headroom by modulating the cutoff frequency of the low-pass filter with an envelope or an expression pedal via the modulation matrix.*

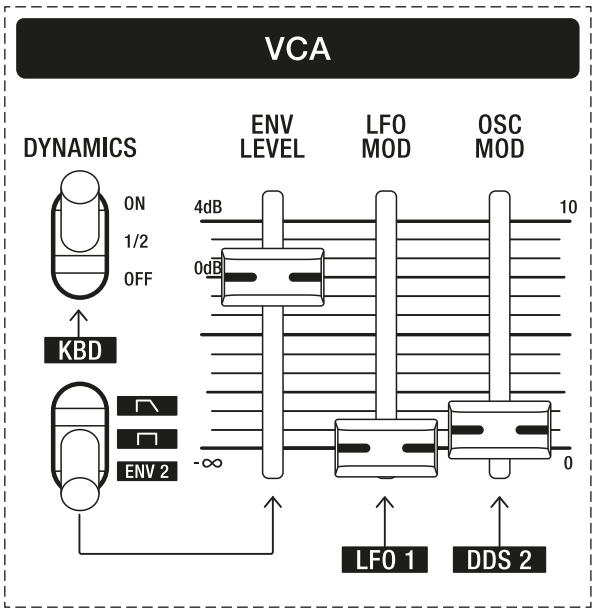
VCA (Voltage Controlled Amplifier)

After the audio signal has been shaped by the filter, it is passed on to the voltage-controlled amplifier (VCA). The 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. You can find out more about how the VCA level can be modulated by the dedicated envelope on pages 52-53. Alternatively, the VCA level can also be controlled by one of the two fixed envelopes so that the second envelope is available for other tasks if required.



The VCA section of the Super Gemini.



The VCA section of the Super 8.

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/LFO MOD: 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.

DDS 2/OSC MOD: This fader controls the amount by which DDS 2 modulates the VCA level. The resulting amplitude modulation can generate anything from subtle textures to clangourous timbres.

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

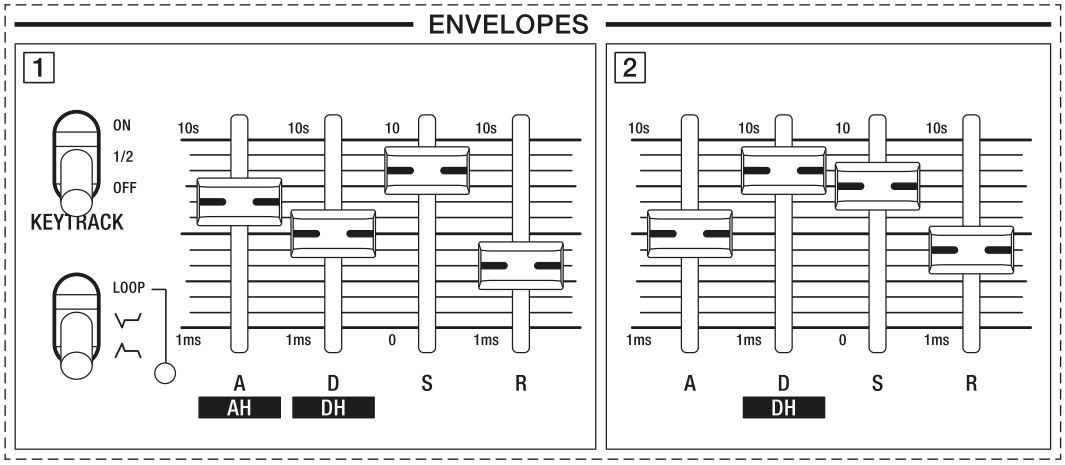
- **Lower position:** With this setting, envelope 2 acts as the modulation source. This is the default setting, which means that the second envelope is usually used for modulating the VCA level.
- **Middle position:** With this setting, the first of the fixed envelopes acts as the modulation source. Its attack, decay and release stages have no duration, i.e. it functions 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 has a release stage. Use this envelope if you want to release envelope 2 for other modulation tasks but want your sound to decay 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:

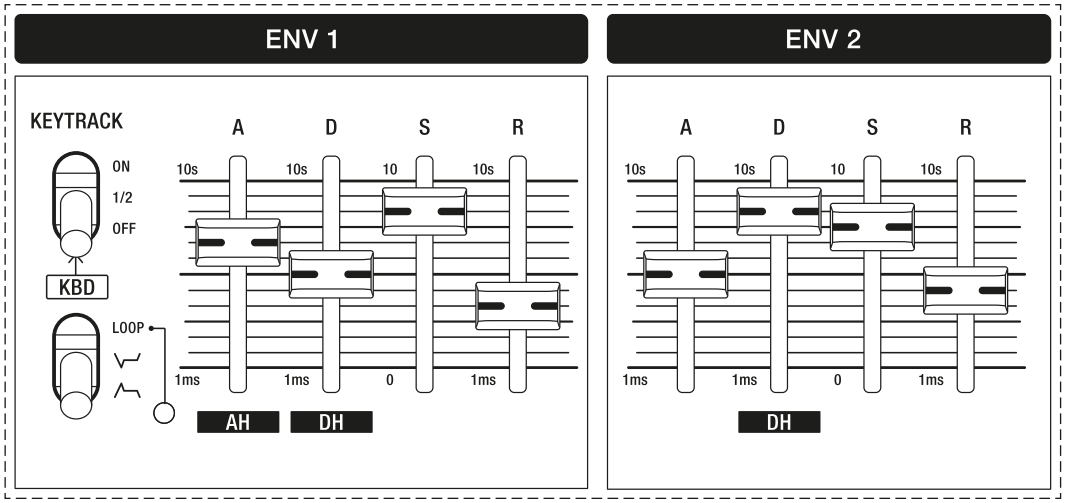
- **OFF:** With this setting, the VCA level and the low-pass filter remain unaffected by keyboard velocity.
- **1/2:** With this setting, the VCA level and the low-pass filter respond with half intensity to keyboard velocity. The harder you hit a key, the louder and brighter the sound becomes. Use this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

Envelopes

With the help of envelope generators, we can determine how a sound develops over time. Typically, envelope generators are assigned to filters and amplifiers in order to change the harmonic content and the overall volume of a sound over several stages.



The envelope section of the Super Gemini.

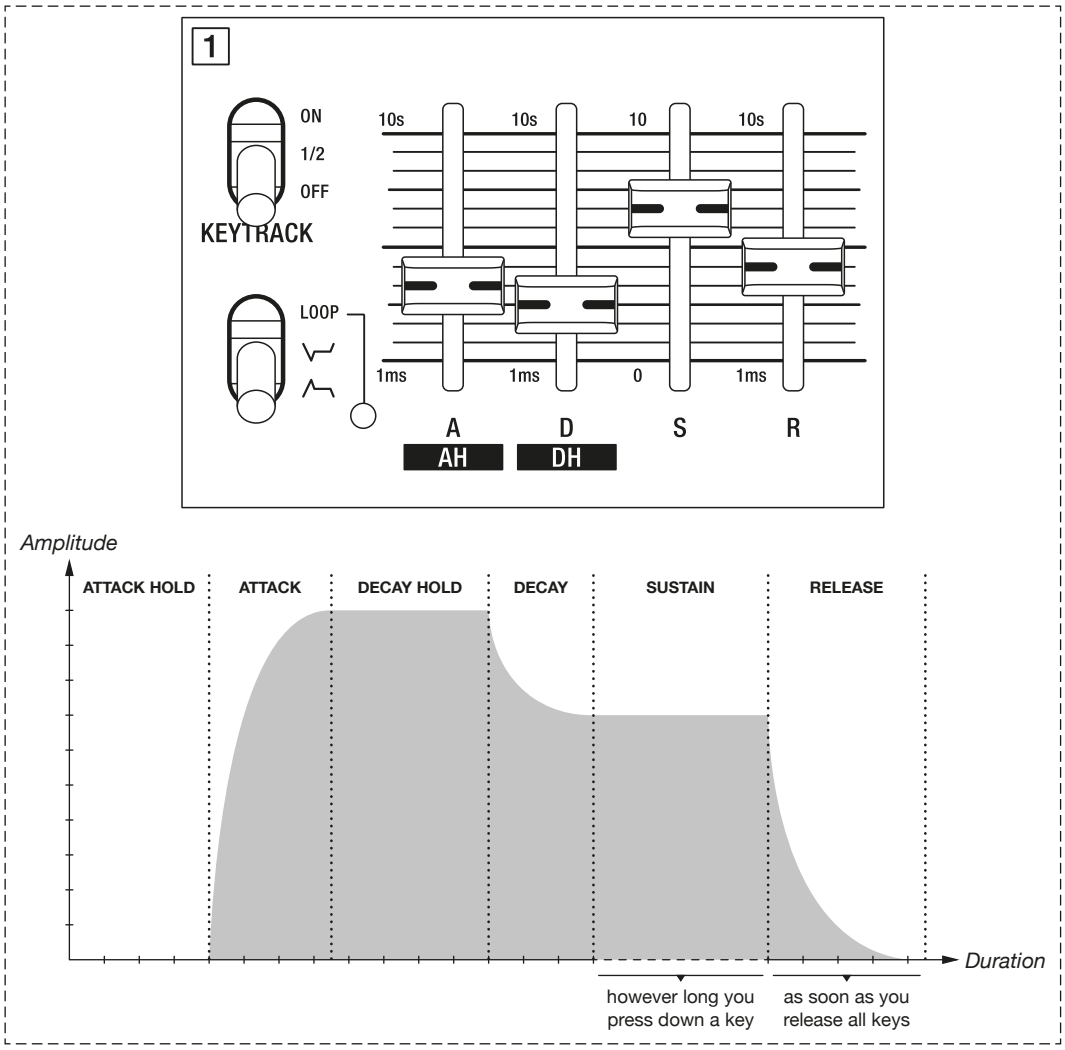


The envelope section of the Super 8.

The envelopes can be mapped to various modulation destinations, including the cutoff frequency of the low-pass filter (see page 42), the VCA level (see page 45), the waveforms of DDS 1 and the pulse width of DDS 2 (see page 62).

Both envelopes contain five stages known as attack, decay hold, decay, sustain and release. In addition, the first envelope features a hold stage that can be used to delay the start of the attack stage after a key is hit.

ENV 1 (Envelope 1)



Setting of envelope 2 and a diagram of the resulting envelope shape.

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

A(TTACK): This fader determines the duration of the envelope's attack stage. The higher the value, 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 only had 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 value, 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.

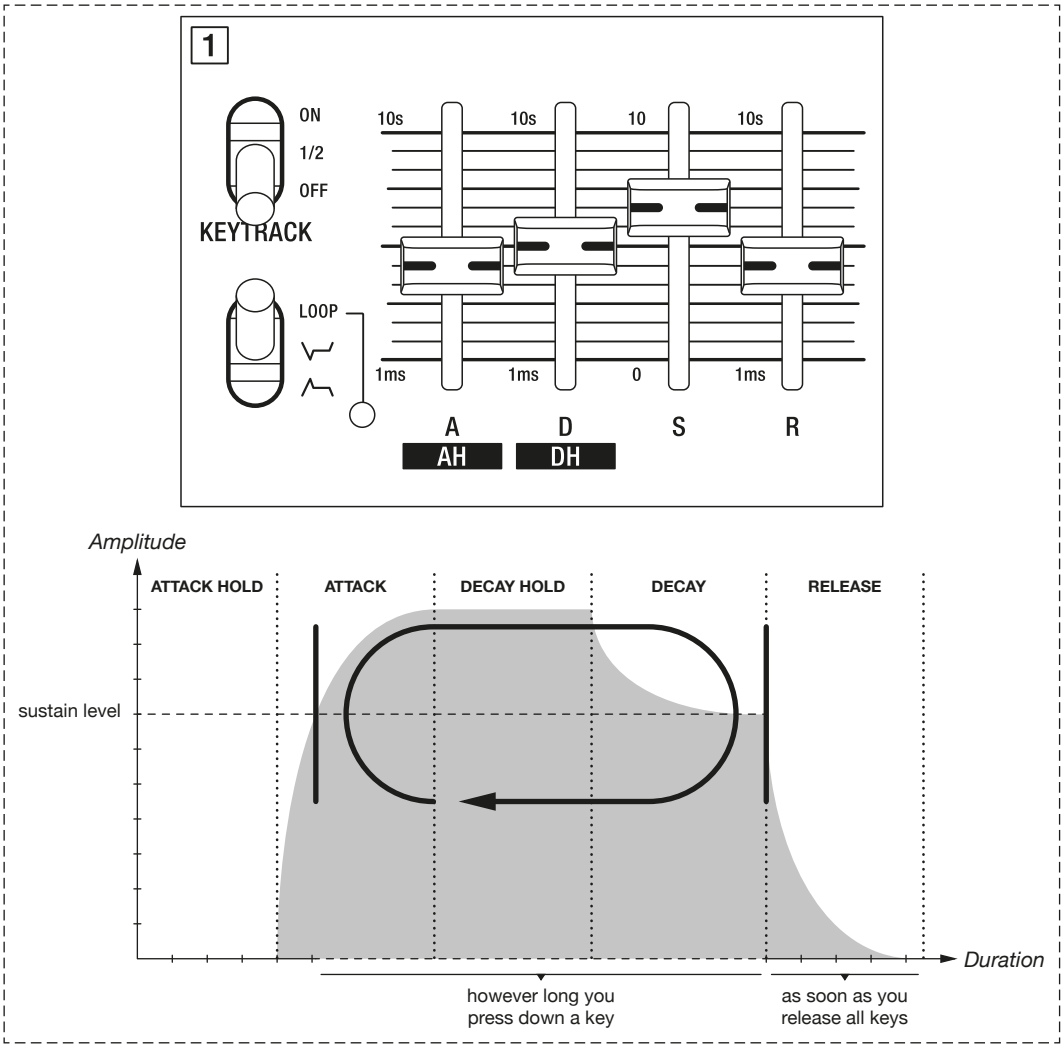
S(USTAIN): This fader determines the level at which the envelope is held when you hold a note beyond the decay stage. This is the only envelope parameter that is linked to a level and not to a duration. The duration of the sustain stage always depends on how long you hold a note.

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

The bottom toggle switch 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 infinitely until you release a key. As soon as you release a key, the release stage is triggered. The speed at which the looped envelope is repeated is indicated by the LED that is graphically linked to the **LOOP** label.

In loop mode, the sustain setting determines the level from which the envelope rises at the beginning of the attack stage and the level to which 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 controlled via key tracking. Low attack, decay hold and decay settings can generate sonic results similar to frequency modulation.

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:

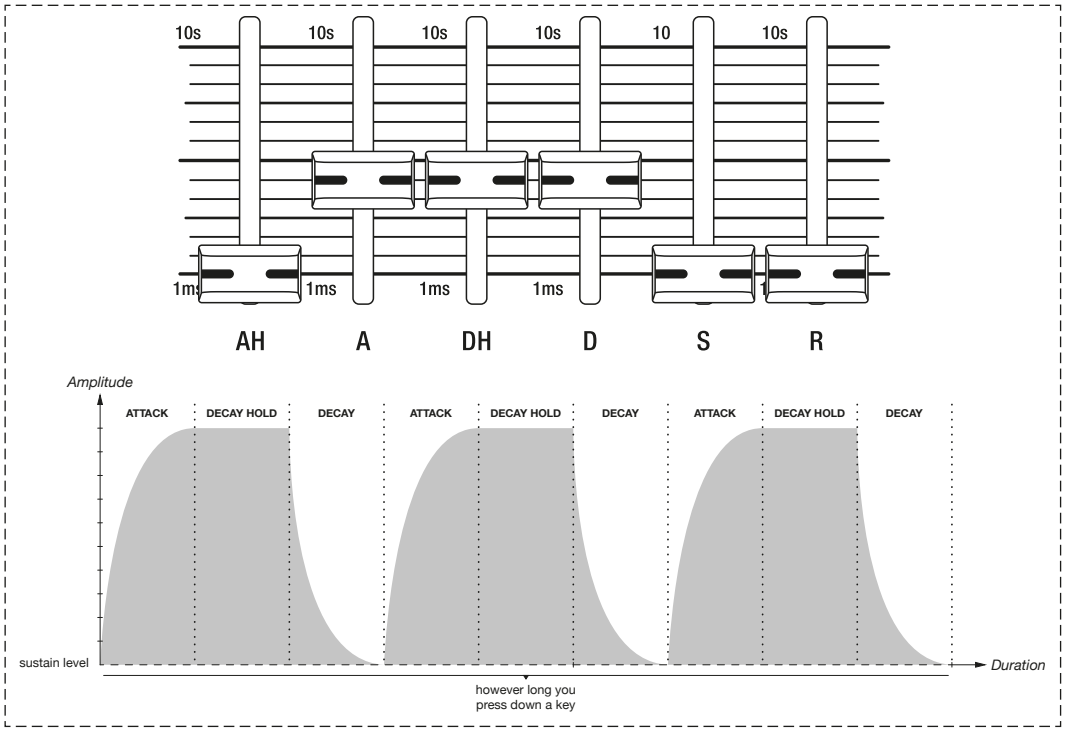
- **OFF:** With this setting, the duration of the envelope's decay and release stages is unaffected by the pitch of the notes played on the keyboard.
- **1/2:** With this setting, the time it takes for the envelope to cycle through its decay and release stages decreases in quarter tone steps as you move 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 a key tracking effect when envelope 1 is set to inverted mode, consider that the effect of the inverted envelope on its modulation destination is the opposite of envelope 1 in default mode.

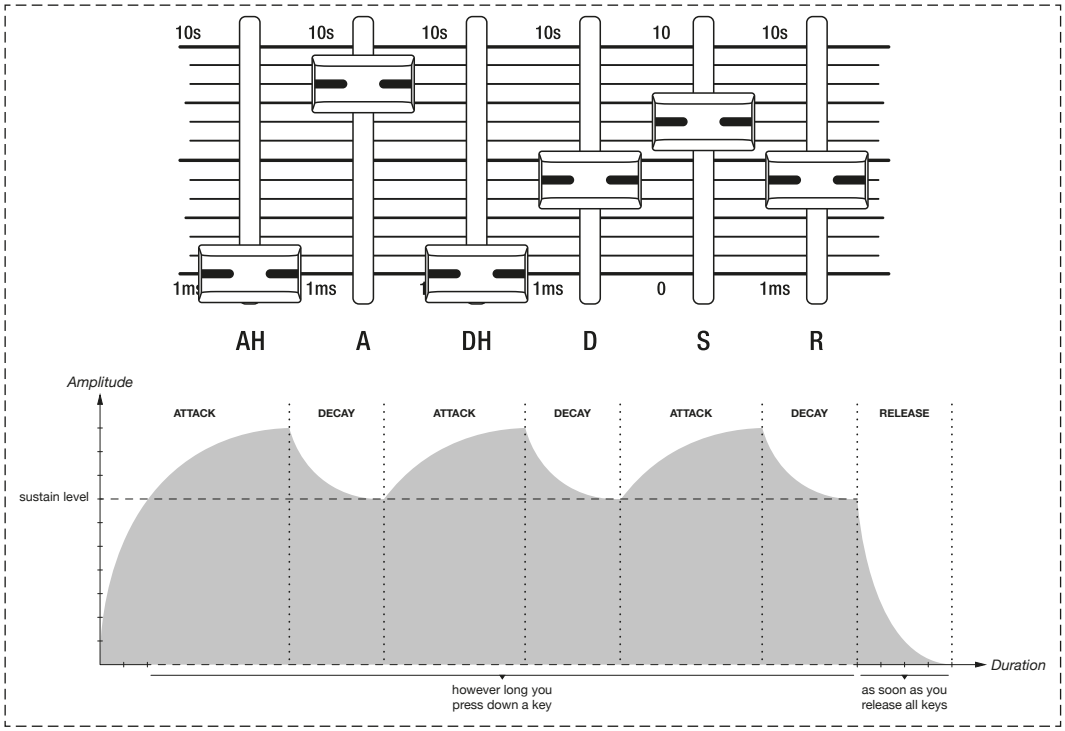
Creating Periodic Modulation Shapes With Envelope 1 in Loop Mode

Below you will find some examples that illustrate which 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 its attack, decay hold and decay stages.

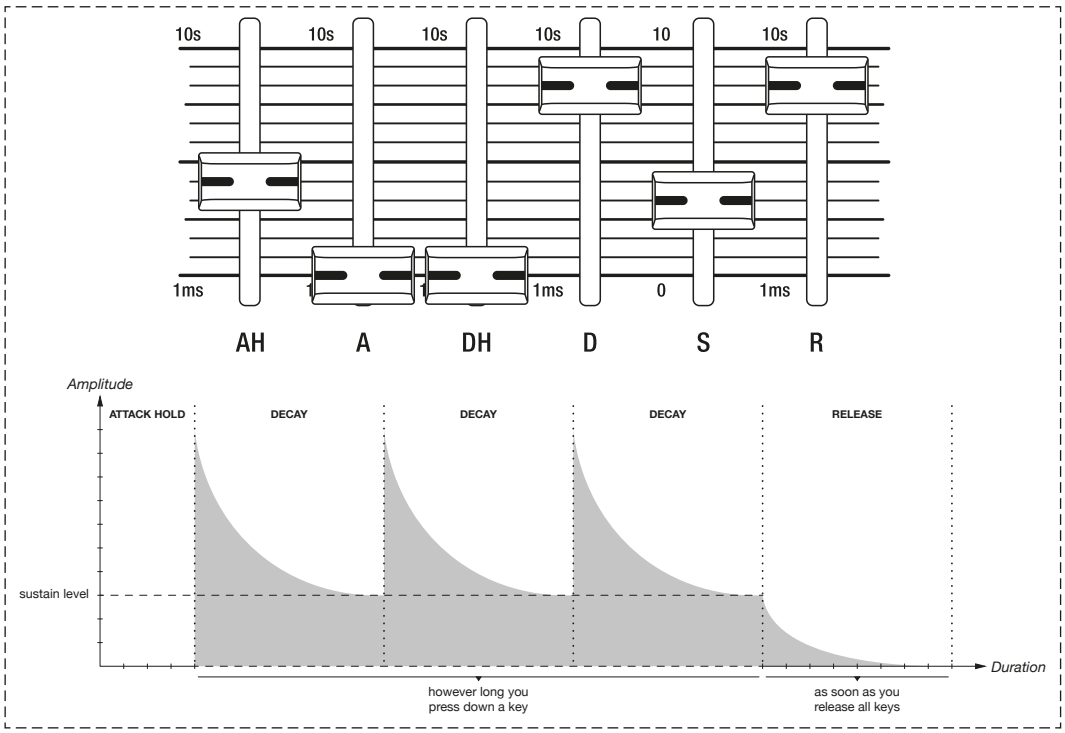
Please note that the **ATTACK HOLD** and **DECAY HOLD** faders shown in the illustrations below have been added for illustrative purposes only. Your instrument is not missing any parts!



Example 1.

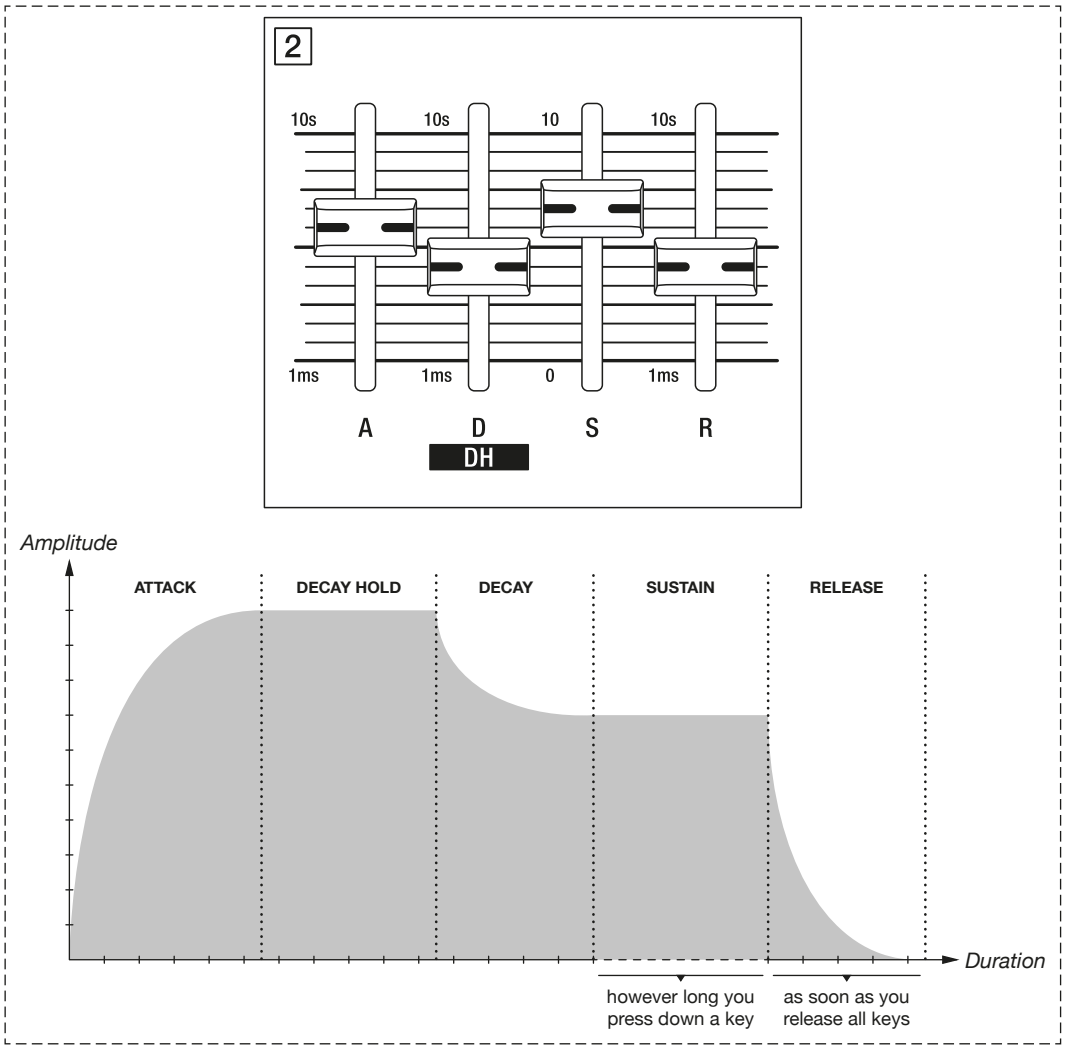


Example 2.



Example 3.

ENV 2 (Envelope 2)



Setting of envelope 2 and a diagram of the resulting envelope shape.

A(TTACK): This fader determines the duration of the envelope's attack stage. The higher the value, 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 only had four stages (attack, decay, sustain and release).

D(ECAY): This fader determines the duration of the envelope's decay stage. The higher the value, 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.

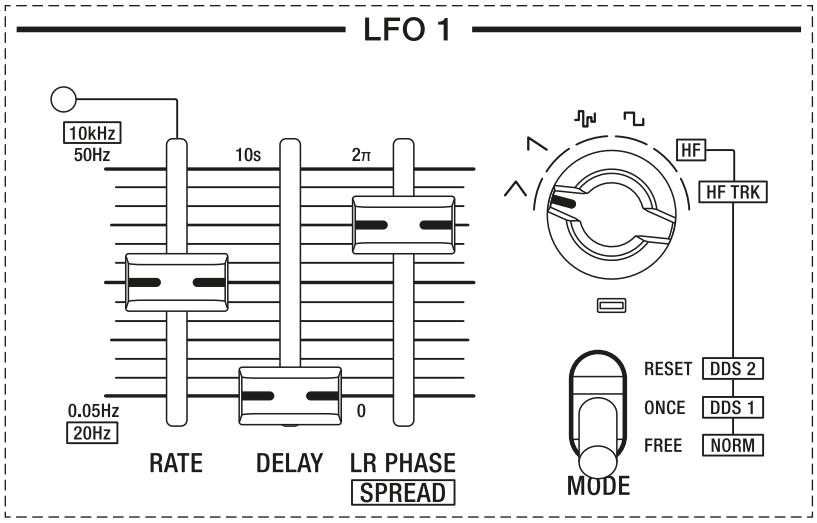
S(USTAIN): This fader determines the level at which the envelope is held when you hold a note beyond the decay stage. This is the only envelope parameter that is linked to a level and not to a duration. The duration of the sustain stage always depends on how long you hold a note.

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

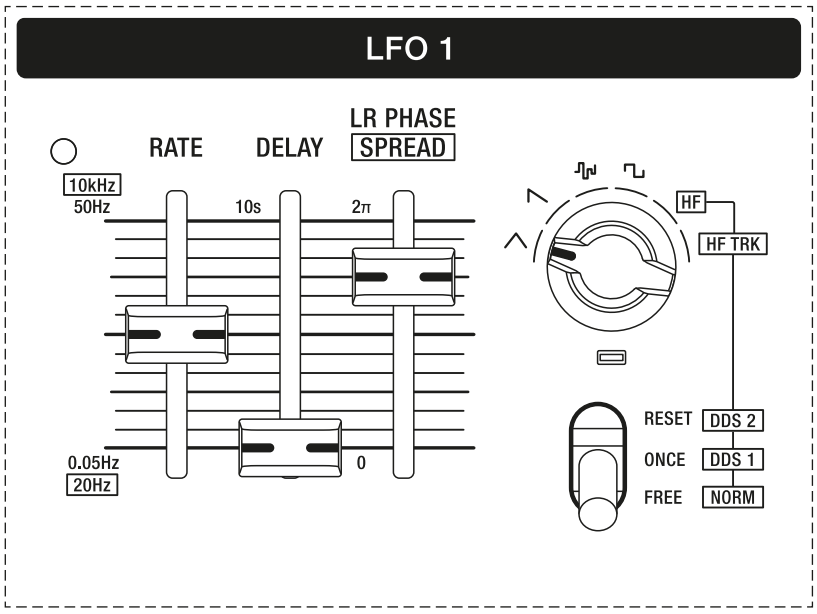
LFO 1 (Low Frequency Oscillator 1)

An LFO (low frequency oscillator) is an oscillator that generates frequencies below the human hearing range. In default mode, LFO 1 can be used to modulate the pitch 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 so that you can use it either as a third oscillator, as a drone or to modulate at audio rates.



The LFO 1 section of the Super Gemini.



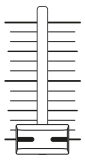
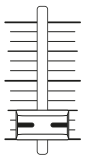
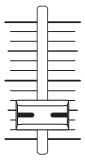
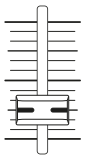
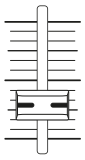
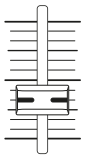
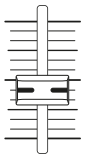
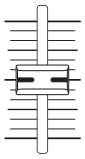
The LFO 1 section of the Super 8.

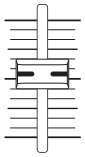
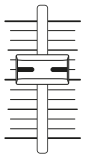
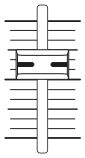
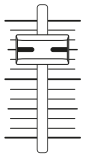
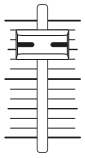
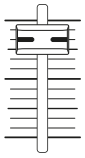
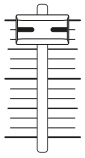
LFO 1 essentially consists of ten (Super Gemini) or eight (Super 8) individual LFOs: one for each of the ten or eight "super voices". In non-binaural mode, each of these LFOs is shared by two voices.

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 79) or to an external MIDI clock signal (see pages 92-93). When synchronised, you can use the **RATE** fader to set the duration of the duty cycle of the LFO 1 waveform in clock divisions relative to the internal or external tempo.

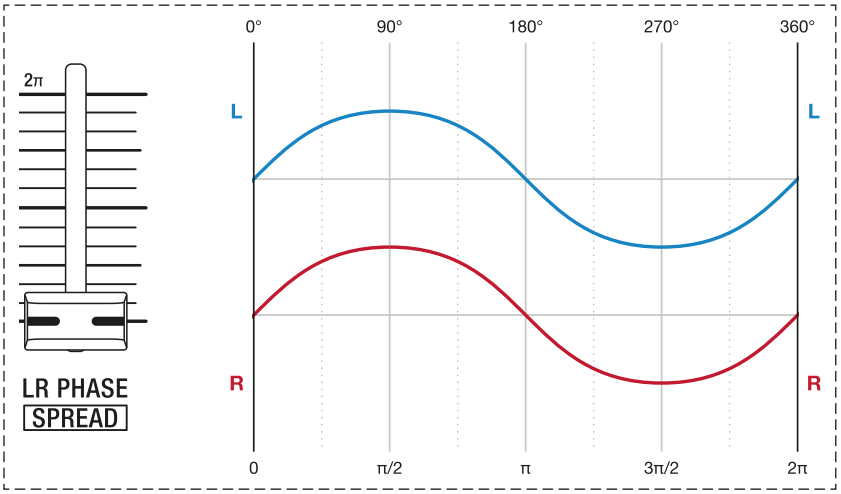
The following table lists the clock divider settings for the synchronised LFO 1 rate:

Setting	Clock Division	LFO 1 Duty 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

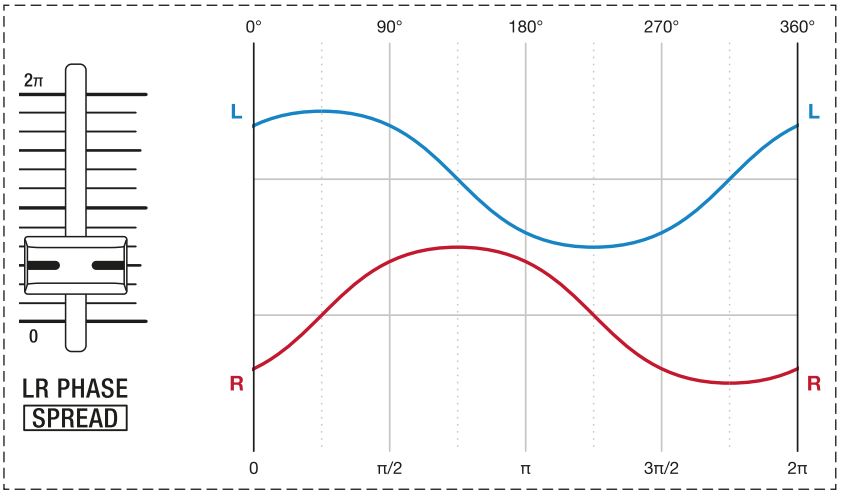
Setting	Clock Division	LFO 1 Duty Cycle Duration
	Dotted 1/8 note	3/4 of 1 beat
	1/4 note triplet	1/3 of 2 beats
	1/8 note	1/2 of 1 beat
	Dotted 1/16 note	3/8 of 1 beat
	1/8 note triplet	1/3 of 1 beat
	1/16 note	1/4 of 1 beat
	Dotted 1/32 note	3/16 of 1 beat
	1/16 note triplet	1/6 of 1 beat

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

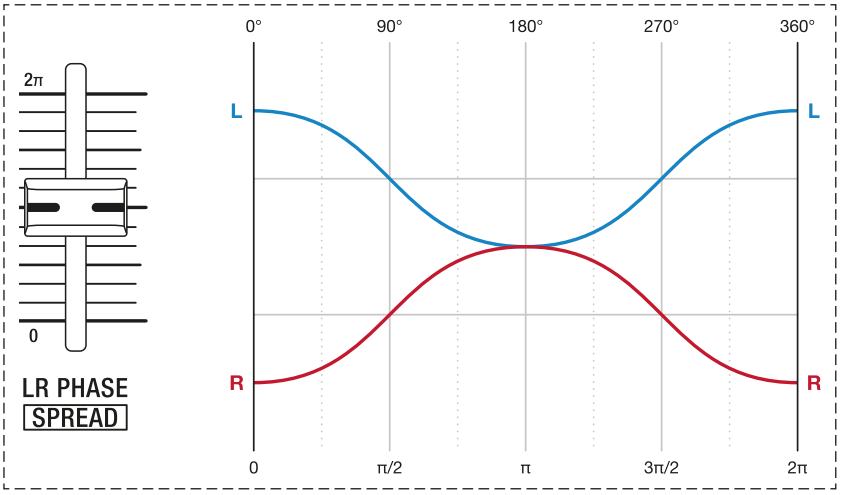
LR PHASE: In binaural mode, this fader controls the left-right channel phase relationship, in other words the effect of LFO 1 on the stereo field. With this single control, you can create complex stereo modulations of the cutoff frequency of the low-pass filter, the VCA level, and the pitch and pulse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



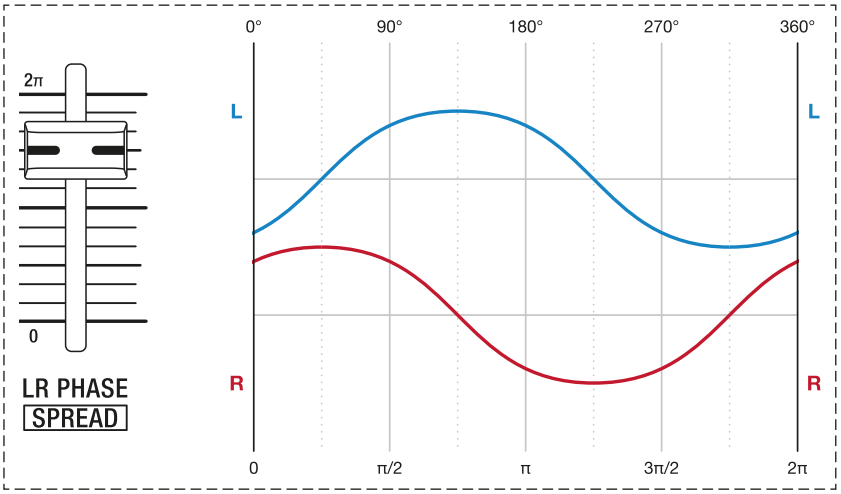
The left-right phase when **LR PHASE** is set to 0% (0).



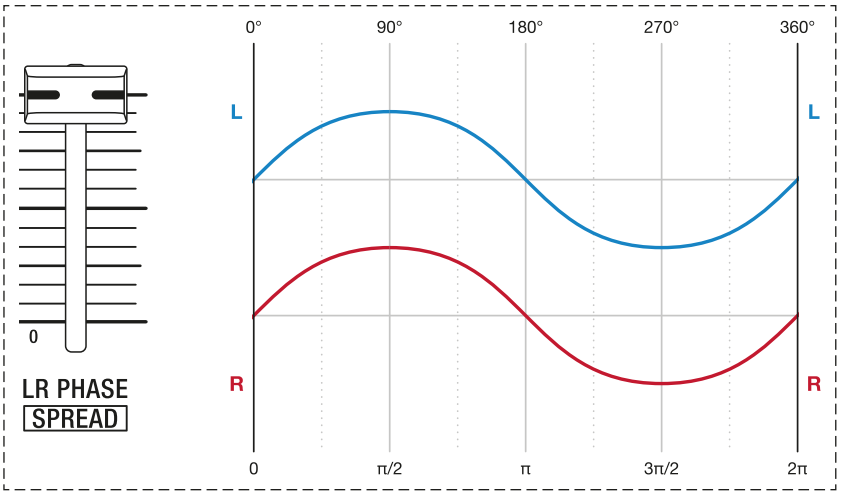
The left-right phase when **LR PHASE** is set to 25% (π/2).



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



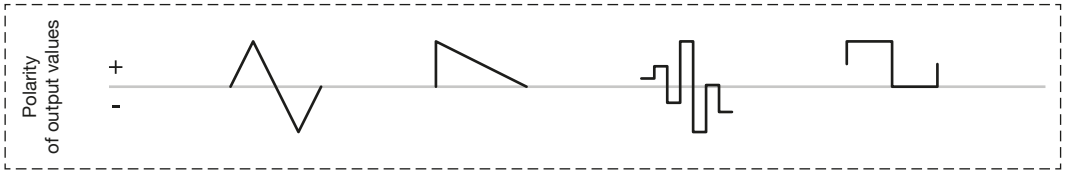
The left-right phase when **LR PHASE** is set to 75% (3π/2).



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

SPREAD: In non-binaural mode and with voice assign mode **POLY 1** or **POLY 2** selected, the **LR PHASE** fader becomes a pan spread control. At the lowest setting, all voices are centred. At the highest setting, all voices are alternately hard-panned between the left and right channels.

WAVEFORM: In low-frequency mode, you can use this rotary switch to choose between four different waveforms: triangle, reverse sawtooth, sample & hold or square.



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.
- **Reverse sawtooth** and **square** 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.
- **Sample & hold** 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 the highest value, sample & hold will generate a white noise signal.
- **HF:** Selecting this option activates the high frequency mode with 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 modulations at audio rates. By default, LFO 1 is set to a sine wave in this mode.
- **HF TRK:** Selecting this option activates the 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 modulations at audio rates. The pitch of LFO 1 can be matched to that of DDS 1 and DDS 2 using the **RATE** control. By default, LFO 1 is set to a sine wave in this mode.

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

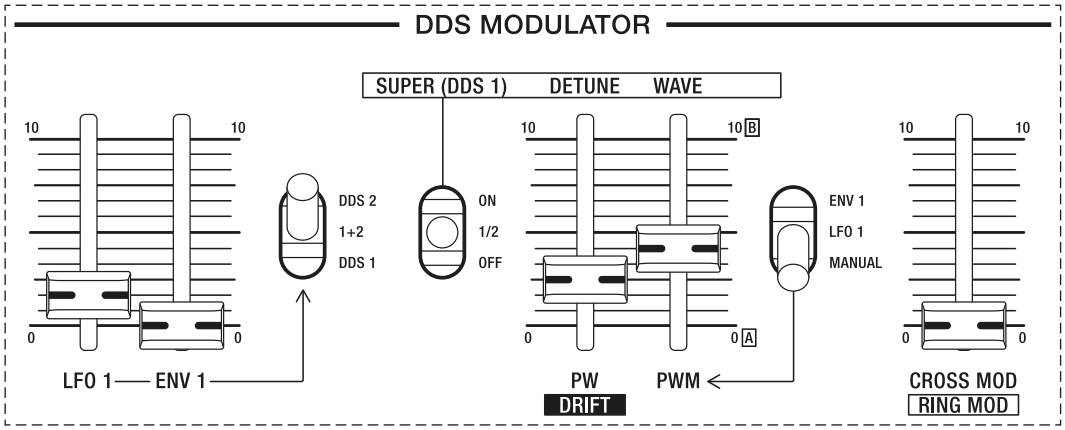
- **FREE:** With this setting, LFO 1 is free-running.
- **ONCE:** With this setting, LFO1 only runs through one duty cycle when you play a note. In this mode, LFO 1 can also be used as a simple envelope whose shape is determined by the selected waveform.
- **RESET:** With this setting, the phase of LFO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

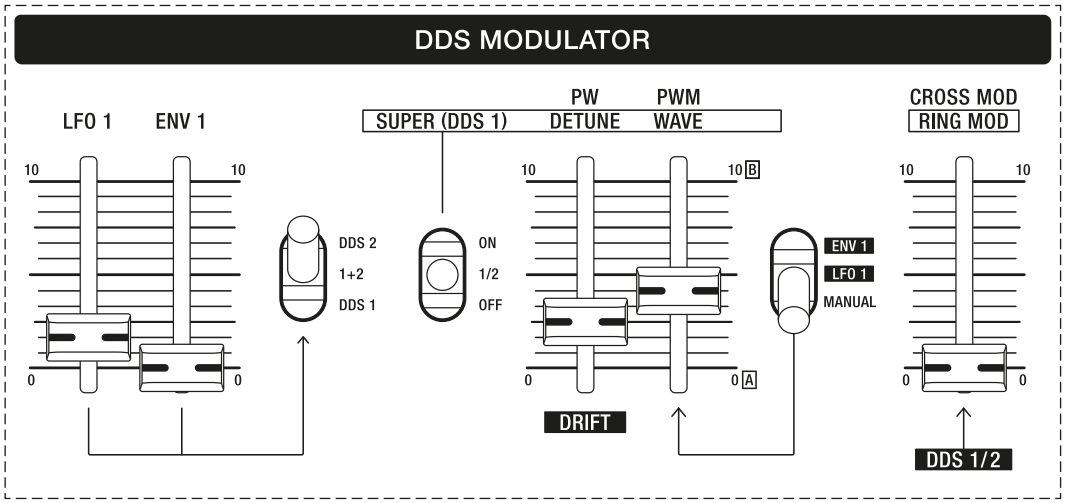
- **NORM:** With this setting, LFO 1 acts as a modulation source in high frequency mode.
- **DDS 1:** With this setting, the audio signal from LFO 1 in high frequency mode is routed to the channel of DDS 1. The output signals from LFO 1 and DDS 1 are summed so that you can crossfade between this summed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

DDS Modulator

The DDS Modulator section offers dedicated controls for modulating the oscillators to give your sounds more tonal variety. It is divided into three subsections: The first contains controls for pitch modulation, the second covers modulation parameters specific to both DDS 1 and the pulse wave of DDS 2, and the third consists of a cross modulation control.



The DDS Modulator section of the Super Gemini.



The DDS Modulator section of the Super 8.

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

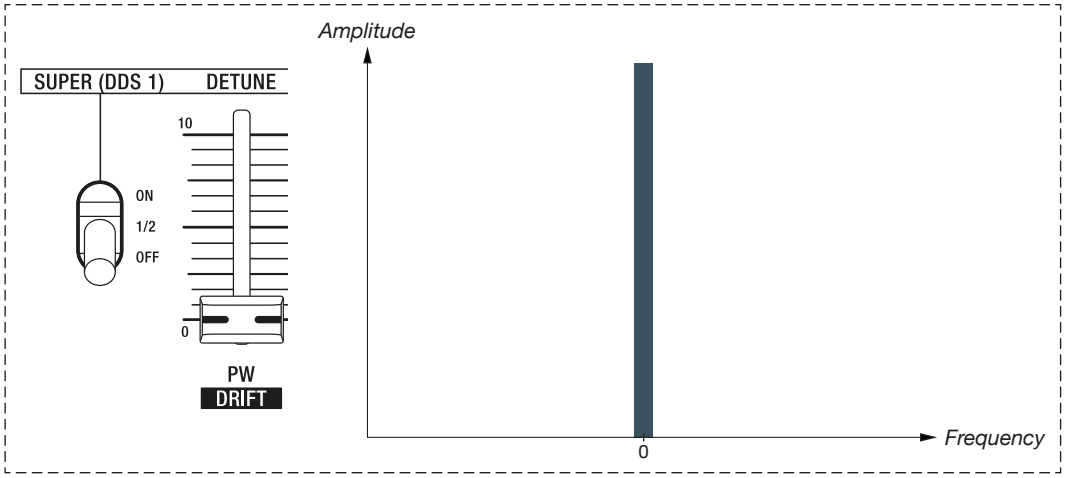
ENV 1: This fader controls the amount of pitch modulation by envelope 1.

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

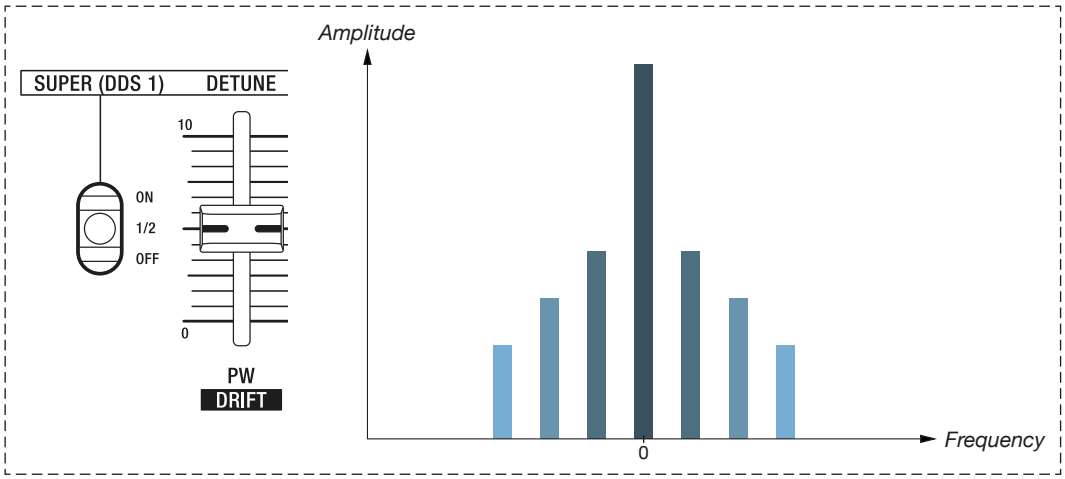
- **DDS 1:** With this setting, the modulation is mapped to DDS 1.
- **1 + 2:** With this setting, the modulation is mapped to both oscillators.
- **DDS 2:** With this setting, the modulation is mapped to DDS 2.

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

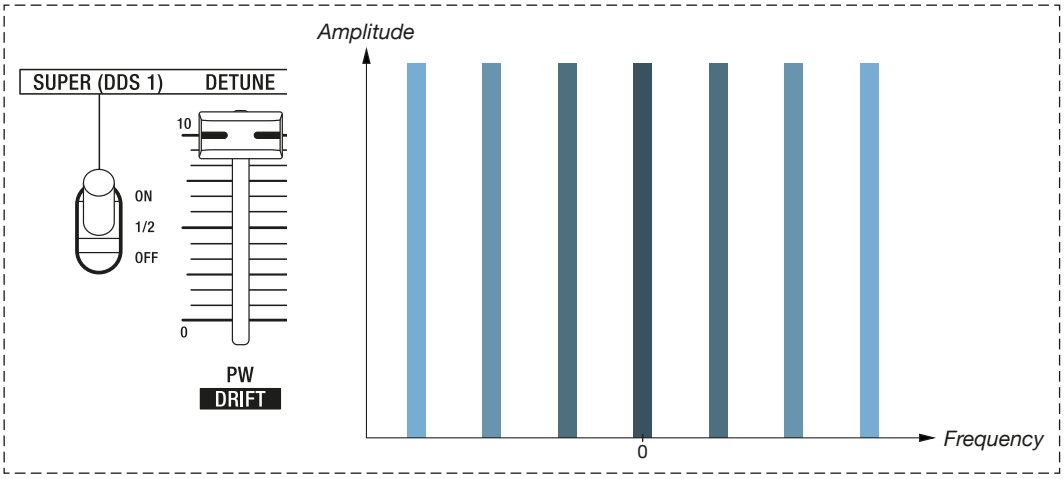
- **OFF:** Selecting this option deactivates super mode. The **DETUNE** parameter has no effect in this mode.
- **1/2:** Selecting this option activates super mode for DDS 1 at half depth. In addition, the phase of the DDS 1 waveform is reset each time a note is played. This is useful for flanging sounds when **DETUNE** is set to higher values or for punchy sounds when **DETUNE** is set to zero.
- **ON:** Selecting this option activates super mode for DDS 1 at full depth.



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



The centroid oscillator of DDS 1 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%.



The centroid oscillator of DDS 1 and its six sister oscillators spread to both sides when super mode is activated 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. When Super mode is activated, it also controls the amount of detune spread applied to DDS 1. As you increase the amount, the sound of the DDS 1 becomes noticeably thicker 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, to emulate unstable vintage synths.

PWM/WAVE: This fader controls the amount of pulse width modulation applied to the pulse wave of DDS 2. In addition, it controls the amount of wave modulation applied to DDS 1. The latter allows you to morph either between two adjacent DDS 1 waveforms or between the currently loaded alternative waveforms assigned to channels **A** and **B** (see pages 33-34). You can use this function to create a large number of hybrid waveforms.

At the lowest position (**A**) you will hear either the current DDS 1 waveform selected by the **WAVEFORM** control, or the alternative waveform assigned to channel **A** when the **WAVEFORM** control is at its rightmost position.

At the highest position (**B**) you will hear either the DDS 1 waveform to the right of the waveform currently selected by the **WAVEFORM** control, or the alternative waveform assigned to channel **B** when the **WAVEFORM** control is at its rightmost position.

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

- **MANUAL:** With this setting, you can manually morph either between two adjacent DDS 1 waveforms or between the currently loaded alternative waveforms assigned to channels **A** and **B**. If DDS 2 is set to pulse wave, this fader also controls the pulse width of DDS 2.
- **LFO 1:** With this setting, LFO 1 acts as the modulation source.
- **ENV 1:** With this setting, envelope 1 acts as the modulation source.

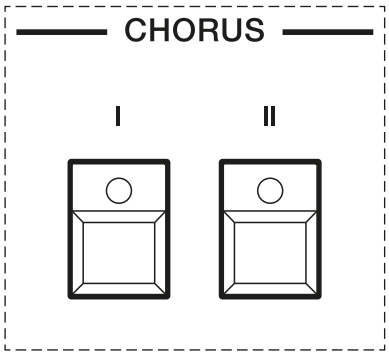
CROSS MOD: This fader controls the amount of cross modulation. By default, DDS 2 modulates DDS 1 with exponential FM. Use this parameter to create complex or bell-like timbres.

*Activating 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 by the **CROSS MOD** fader. This allows similar results to be achieved as with wave folding or phase modulation.*

*If DDS 2 is set to **RING** mode, the **CROSS MOD** fader applies additional cross modulation to the carrier (DDS 1).*

The Super Gemini and Super 8 feature two 24-bit effects that you can use to add the finishing touches to your sounds: A classic dual-mode stereo chorus and 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 first and the delay second in the signal chain.

Chorus



The chorus section.

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

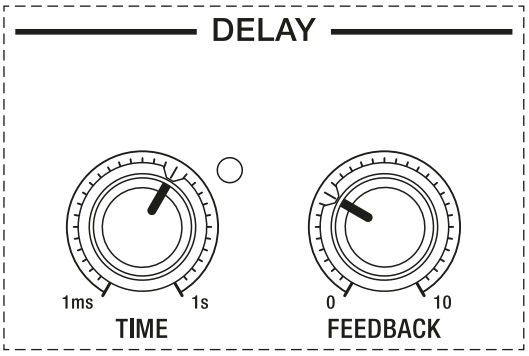
Chorus I: Selecting this option activates a subtle chorus effect.

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

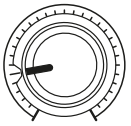
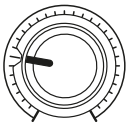
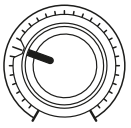
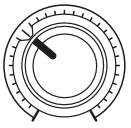


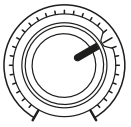
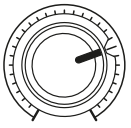
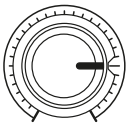
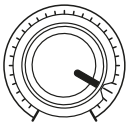
The delay section.

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

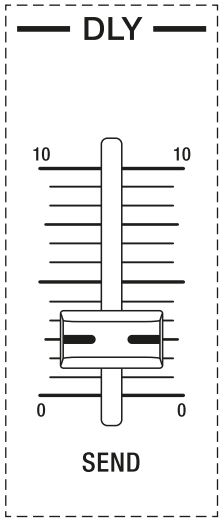
If you enable the **SYNC** option in the arpeggiator and sequencer section, the delay time will be synchronised to either the internal clock as set by the **TEMPO** control (see page 79) or an external MIDI clock signal (see pages 92-93). When synchronised, the **TIME** control enables you to adjust the delay time in clock divisions relative to the internal or external tempo.

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

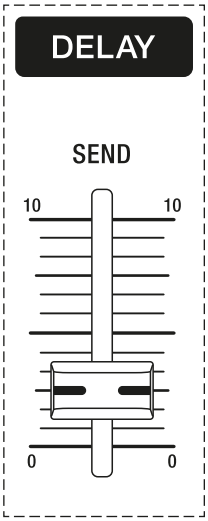
Setting	Clock Division	Delay Time
	1/32 note triplet	1/12 of 1 beat
	Dotted 1/64 note	3/32 of 1 beat
	1/32 note	1/8 of 1 beat
	1/16 note triplet	1/6 of 1 beat
	Dotted 1/32 note	3/16 of 1 beat

Setting	Clock Division	Delay Time
	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 repetitions, 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.



The delay send fader of the Super Gemini.



The delay send fader of the Super 8.

SEND: This fader controls how much of the output signal from each layer is fed into the delay effect. Higher settings result in a wet effect mix, while lower settings emphasise the dry signal.

Delay Freeze

The delay freeze function enables you to create sound-on-sound loops while you are performing. To access this feature:

1. Turn off the instrument.
2. Connect a single or dual footswitch to the delay freeze input on the rear panel. If a single footswitch is used, it will control the upper layer delay freeze function. If a dual footswitch is connected, the left pedal controls the upper layer delay freeze and the right pedal controls the lower layer delay freeze.
3. Turn on the instrument. The connected footswitch will auto-calibrate for polarity when the instrument is switched on.

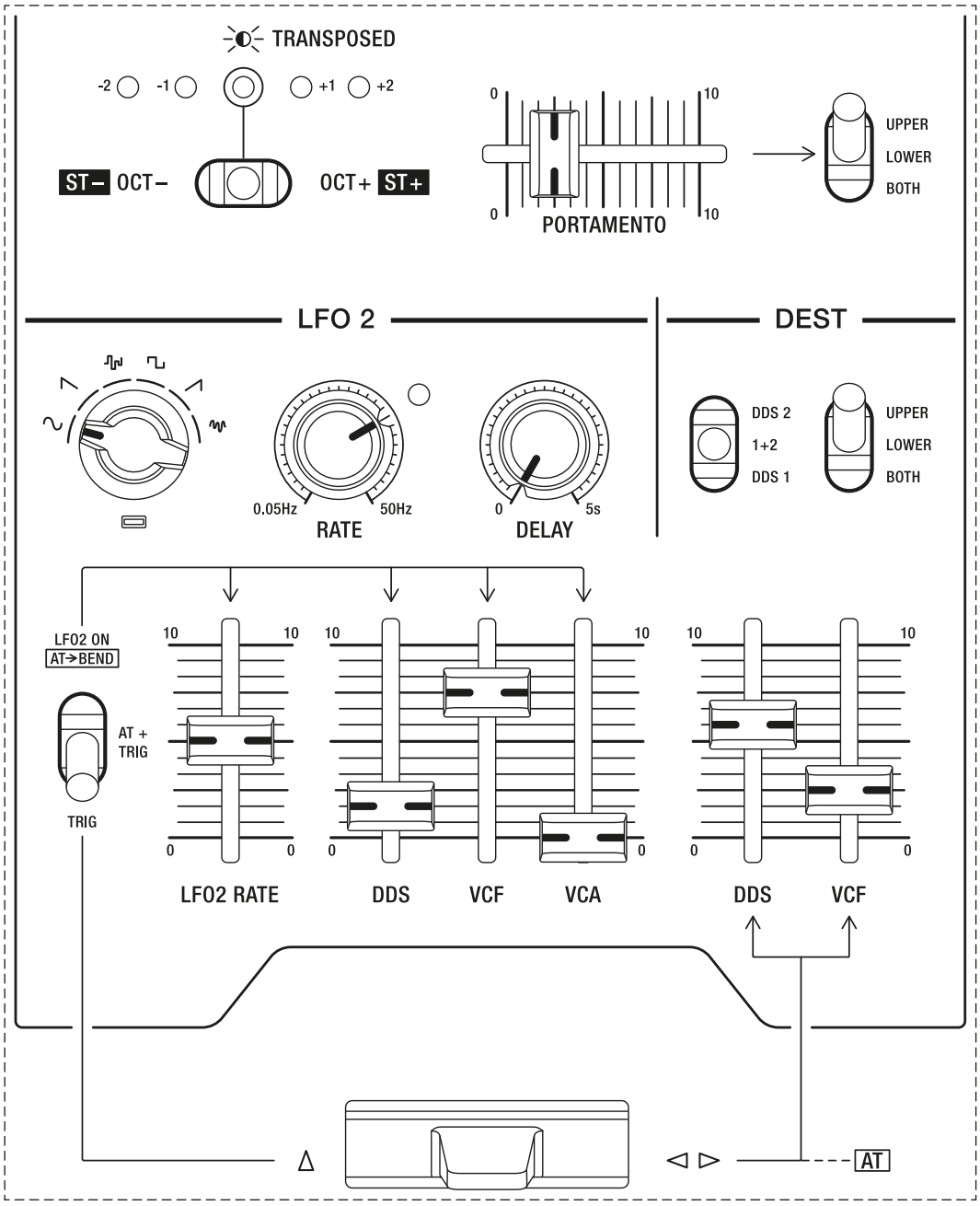
To add notes or chords to the delay loop, release the footswitch while playing. Once you hold down the footswitch, new notes will no longer be added to the delay loop while its current content loops endlessly.

Use the **SEND** 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.

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

PERFORMANCE CONTROL SECTION

In addition to the 61-note velocity-sensitive Fatar keyboard with polyphonic aftertouch, the Super Gemini and Super 8 also feature an extensive performance control section. Settings in the performance control section can affect both layers either individually or together.

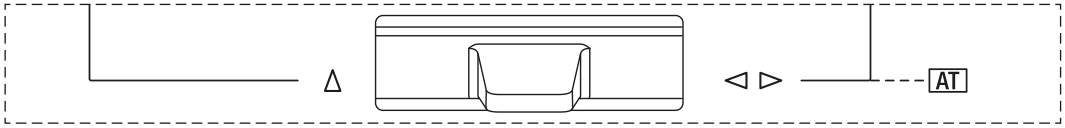


The performance control section.

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

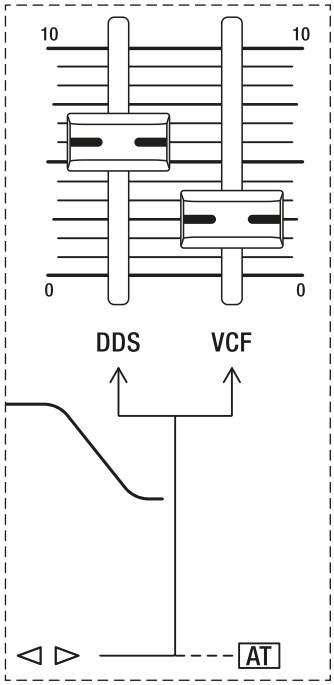
The Bender

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 for the bender assignment.

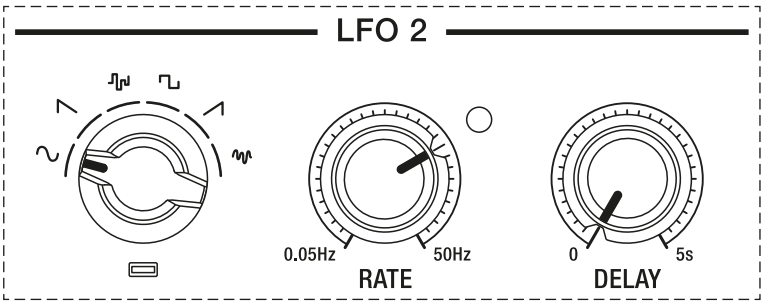
DDS: This fader controls how much the bender affects the pitch of the oscillators. The maximum pitch-bend range is one octave. The oscillator selector toggle switch above this control determines which oscillator is affected by pitch modulation (see page 72).

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)

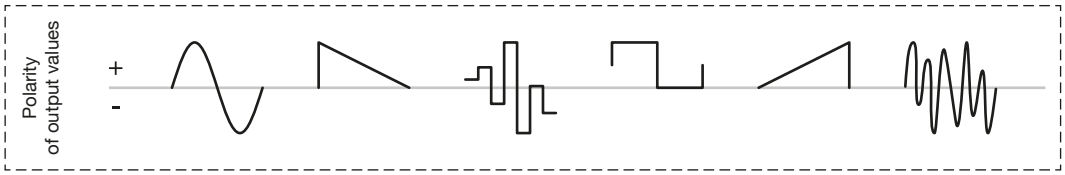
The second LFO generates six waveforms and can be used to modulate the pitch of the oscillators to produce a vibrato effect, to modulate the cutoff frequency of the low-pass filter for periodic harmonic changes, or to modulate the VCA level to create tremolo style effects.

LFO 2 can be triggered by pushing the bender upwards or by aftertouch. It can also be switched on permanently. The type and depth of the modulation controlled by LFO 2 is determined by the **DDS**, **VCF** and **VCA** faders, which are located below the primary controls of LFO 2.



The primary LFO 2 controls.

WAVEFORM: Use this rotary switch to choose between six different waveforms: sine, reverse sawtooth, sample & hold, square, sawtooth or noise.

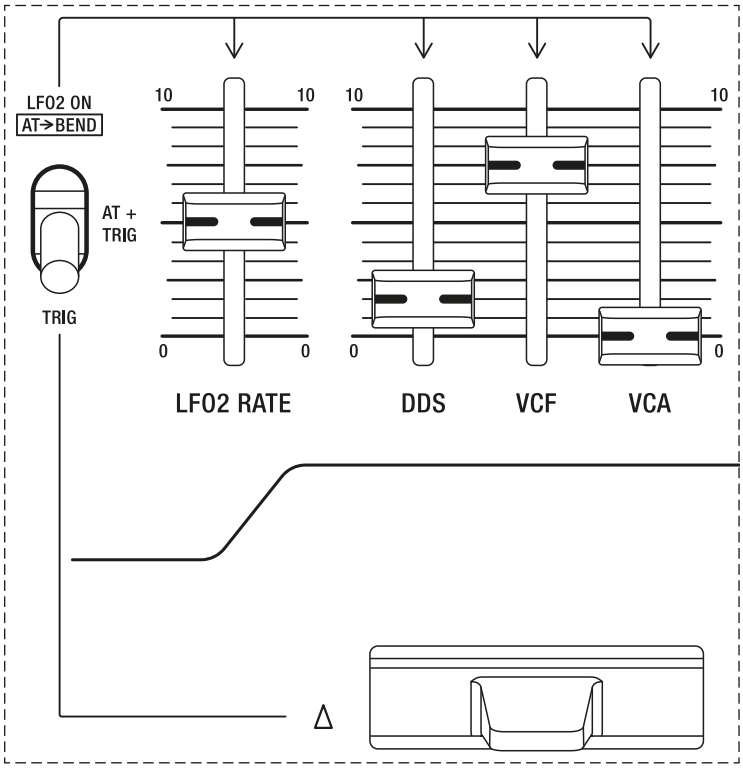


LFO 2 waveforms.

- **Sine** can be used to produce vibrato effects as it alternates equally between positive and negative values. This is a bipolar waveform.
- **Reverse sawtooth**, **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.
- **Sample & hold** produces random positive or negative values for the duration of one duty cycle. This waveforms can be used to create either subtle movements or wild effects.
- **Noise** creates a white noise signal that can be used to create grainy textures.

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.



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 simultaneously apply pressure to a key, only the gesture with the greater effect on the triggering of LFO 2 controls 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.

LFO 2 RATE: This fader determines how much vertical bender movements and/or aftertouch affect the rate of LFO 2. The harder you press the bender or a key, the more the rate increases.

DDS: This fader controls the amount by which LFO 2 modulates the pitch of the oscillators. The oscillator selector toggle switch in the destination section determines which oscillator is affected by pitch modulation.

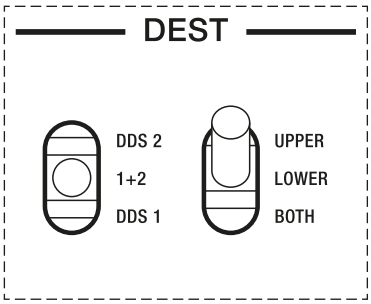
VCF: This fader controls the amount by which LFO 2 modulates the cutoff frequency of the low-pass filter.

VCA: This fader controls the amount by which LFO 2 modulates the VCA level.

When selected as a modulation desination, LFO 2 responds polyphonically. For example, if you use envelope 1 to modulate the rate of LFO 2, each time you play a note, the rate of LFO 2 will be modulated according to the envelope shape. The phase of the per-voice LFOs can be synchronised again by toggling the leftmost toggle switch in the performance control section.

Oscillator and Layer Selection

In the destination section you can specify which oscillator to modulate per layer and select the layer to which you would like to apply the modulation settings for the bender and LFO 2.



The destination section.

The oscillator selector toggle switch determines which oscillator is affected by the pitch modulation controlled by horizontal bender movements and 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 the bender and LFO 2.
- **1 + 2:** With this setting, the pitch of both oscillators will be affected by the bender and LFO 2.
- **DDS 2:** With this setting, only the pitch of the second oscillator will be affected by the bender and LFO 2.

Use the layer selector toggle switch to select the layer for which you would like to adjust modulation settings related to the bender and LFO 2. You have the choice between three options:

- **BOTH:** Select this option if you want to create the same bender and LFO 2 specific modulation settings for both layers. If you toggle to this position from either **UPPER** or **LOWER**, the current settings for the upper layer are applied to both layers.
- **LOWER:** Select this option if you want to create bender and LFO 2 specific modulation settings for the lower layer.
- **UPPER:** Select this option if you want to create bender and LFO 2 specific modulation settings for the upper layer.

Suppose you would like individual pitch-bend ranges for two pad sounds layered in dual mode. A pitch-bend range of one octave (+/- 12 semitones) should be applied to the oscillators of the upper layer and a pitch-bend range of a fifth (+/- 7 semitones) should be applied to the oscillators of the lower layer:

1. Toggle the layer selector switch to **UPPER**.
2. Toggle the oscillator selector switch to **1 + 2**.
3. Move the rightmost **DDS** fader to the highest position.
4. Toggle the layer selector switch to **LOWER**.
5. Toggle the oscillator selector switch to **1 + 2**.
6. Move the rightmost **DDS** fader to the position where the pitch-bend range equals a fifth.

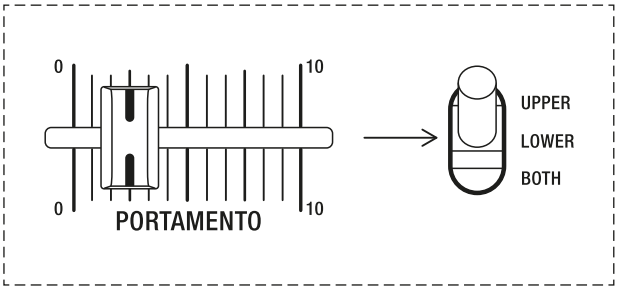
Moving the bender horizontally to its extreme positions bends the pitch of the upper layer oscillators by an octave, while bending the pitch of the lower layer oscillators by a fifth. This allows you to introduce complex harmonic relations by moving just one performance control. After tweaking the settings to your liking, make sure to save either the performance or both patches.

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.

Use the layer selector toggle switch to select the layer for which you would like to adjust the portamento setting. You have the choice between three options:

- **BOTH:** Select this option if you want the same portamento setting for both layers. If you toggle to this position from either **UPPER** or **LOWER**, the current setting for the upper layer is applied to both layers.
- **LOWER:** Select this option if you want to adjust the portamento setting for the lower layer.
- **UPPER:** Select this option if you want to adjust the portamento setting for the upper layer.

Suppose you would like individual portamento settings for two monophonic lead sounds layered in dual mode:

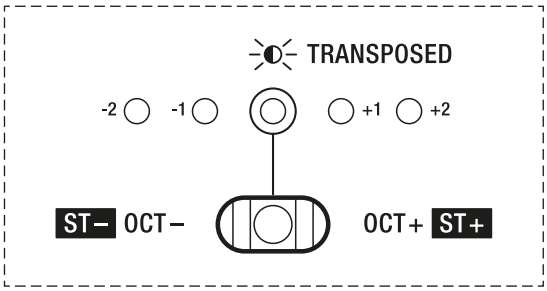
1. Toggle the layer selector switch to **UPPER**.
2. Adjust the portamento time to taste.
3. Toggle the layer selector switch to **LOWER**.
4. Adjust the portamento time to taste.

When playing legato style, the pitches of both sounds slide at different rates, causing both layers to be momentarily out of tune with each other, which in turn fattens up the overall sound. After tweaking the settings to your liking, make sure to save either the performance or both patches.

Octave Selector & Transpose Function

The octave selector toggle switch allows you to switch octaves over a range of five octaves. The switch is spring-loaded and enables 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.

Use the layer selector toggle switch next to the **PORTAMENTO** fader to select the layer for which you would like to adjust the octave setting. You have the choice between three options:

- **BOTH:** Select this option if you want the same octave setting for both layers. If you toggle to this position from either **UPPER** or **LOWER**, the current setting for the upper layer is applied to both layers.
- **LOWER:** Select this option if you want to adjust the octave setting for the lower layer.
- **UPPER:** Select this option if you want to adjust the octave setting for the upper layer.

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.

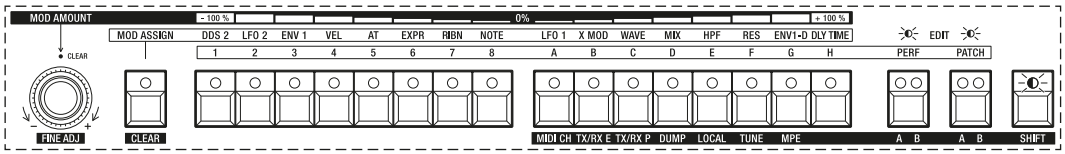
Global Fine Tuning

In addition to the transpose function, you can also fine-tune the instrument globally. To adjust the global fine tuning, press **SHIFT** and then turn the **MOD AMOUNT** encoder, which is called **FINE ADJ** in shift mode.

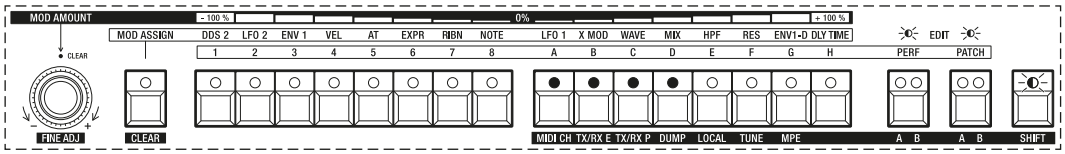
Turning the **FINE ADJ** encoder clockwise increases the frequency. Turning the control counter-clockwise decreases the frequency. You can adjust the global fine tuning over a range of +/- 100 cents. Pressing the **FINE ADJ** encoder clears the current tuning and resets it to 440 Hz.

As soon as you touch or move the **FINE ADJ** encoder, the lit LEDs of buttons **1**-**8** and **A**-**H** indicate the current setting, while the printed top row above buttons **1**-**8** and **A**-**H** serves as a legend for the fine tuning.

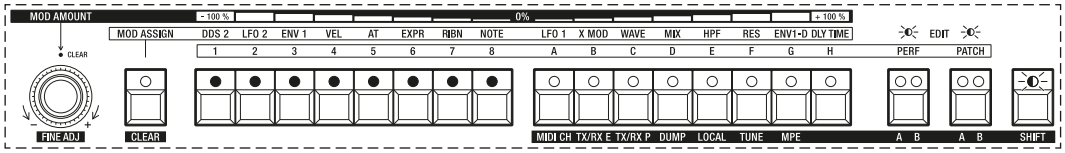
In the first example, the lit LEDs of buttons **1**-**8** and **A**-**H** indicate a global fine tuning of 0 cents or 440 Hz, which is the default:



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



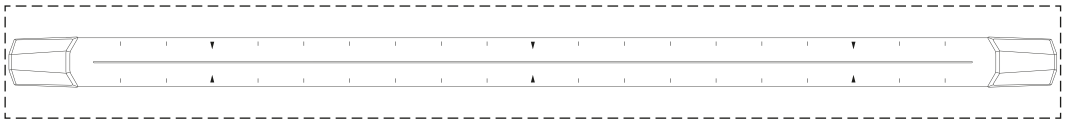
In the third example, the lit LEDs of buttons **1**-**8** and **A**-**H** indicate a global fine tuning of -100 cents:



RIBBON CONTROLLER (SUPER GEMINI)

The Super Gemini is also equipped with a custom-designed ribbon controller to further enhance the expressiveness of your performance. By default, the ribbon controller is mapped to control the pitch of the oscillators. The scope of the pitch-bend range depends on your finger position:

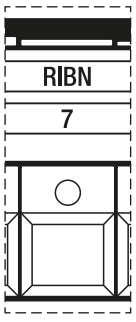
- To achieve the maximum range of a pitch bend upwards, place your finger at the bottom of the ribbon controller and slide it to the top.
- To achieve the maximum range of a pitch bend downwards, place your finger at the top of the ribbon controller and slide it down.



The ribbon controller.

Pitch bends controlled by the ribbon controller only affect notes held on the keyboard. During the release stage of the envelopes, the ribbon controller no longer affects the pitch of the notes.

You can also map the ribbon controller to various modulation destinations using the modulation matrix. See pages 84-89 for more information on how to use the modulation matrix.



The ribbon controller as a modulation source in the modulation matrix.

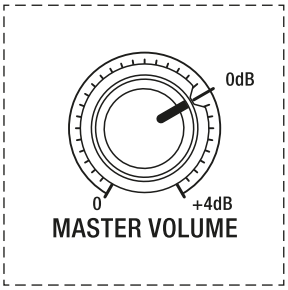
When you assign the ribbon controller to any modulation destination using the modulation matrix, the ribbon controller no longer controls the pitch of the oscillators. If you clear all modulation mappings to the ribbon controller as a modulation source in the modulation matrix (see page 89), pitch control is enabled again.

The ribbon controller always sends a coarse/fine pair of MIDI CC messages: CC2 (MSB) and CC34 (LSB). When the ribbon controller is used in default mode to control the pitch of the oscillators, it does not receive MIDI CC messages. However, it sends and receives MIDI CC messages when used as a modulation source in the modulation matrix.

ADDITIONAL CONTROLS & PARAMETERS

In addition to the patch-related parameters, the Super Gemini and Super 8 also feature overarching controls and parameters that either affect the instrument as a whole or are relevant for setting up a performance.

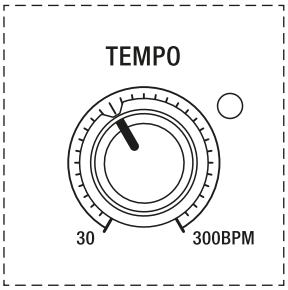
Master Volume



The master volume control.

MASTER VOLUME: This parameter controls the instrument'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 performance or patch.

Tempo



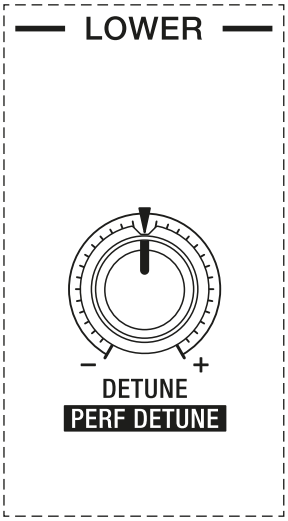
The tempo control.

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 instrument is synchronised to an external MIDI clock source, the **TEMPO** control has no effect. In this case, use the **CLK DIV** control in the arpeggiator and sequencer section to set the playback speed in clock divisions relative to the external tempo.*

Lower Layer Detune (Super Gemini)

This parameter is only available on the Super Gemini. You can use it to either slightly detune the lower layer relative to the upper layer to thicken a dual mode performance, or to detune an entire performance to compensate for pitch offsets caused by cross modulation. Its setting is stored with a performance.



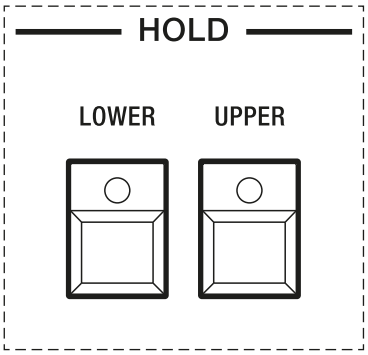
The lower layer detune control.

DETUNE: This control determines the fine tuning setting for the lower layer. Turning the control clockwise increases the frequency. Turning the control counter-clockwise decreases the frequency. You can adjust the fine tuning over a range of +/- 7 semitones.

PERF DETUNE: In shift mode, the **DETUNE** control allows you to detune a performance. Turning the control clockwise increases the frequency. Turning the control counter-clockwise decreases the frequency. You can adjust the tuning over a range of +/- 7 semitones.

*You can achieve both functions on the Super 8 by using the **TUNE** control of DDS 2 in shift mode.*

Hold



The hold mode buttons for both layers.

UPPER: With this option enabled, the notes of the upper layer will continue to be sustained after you release all keys.

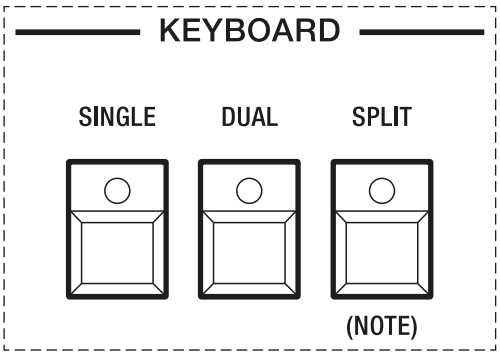
LOWER: With this option enabled, the notes of the lower layer will continue to be sustained after you release all keys.

If you activate the hold function of one of the layers while the arpeggiator is switched on, the arpeggio of the respective layer 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.

If you activate the hold function of one of the layers while the sequencer is switched on, the sequence of the respective layer can be transposed according to the keys you play.

Keyboard

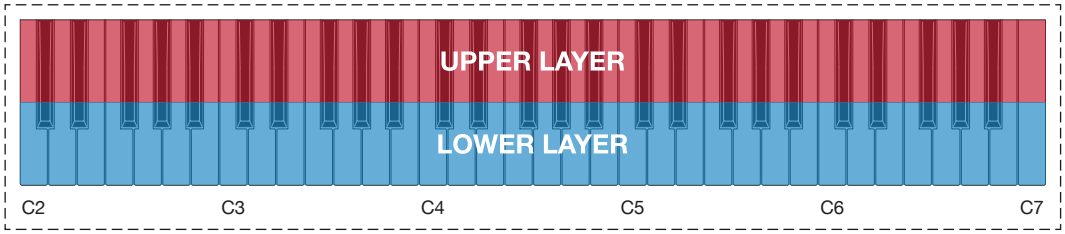
In the keyboard section you can set the mode of a performance.



The keyboard section.

SINGLE: In single mode, only one layer with 20-voice (Super Gemini) or 16-voice (Super 8) polyphony is active. In binaural mode, the number of voices is halved to ten (Super Gemini) or eight (Super 8) voices. By default, single mode is set to the upper layer. You can switch between layers by using the buttons **UPPER** and **LOWER** in the layer section.

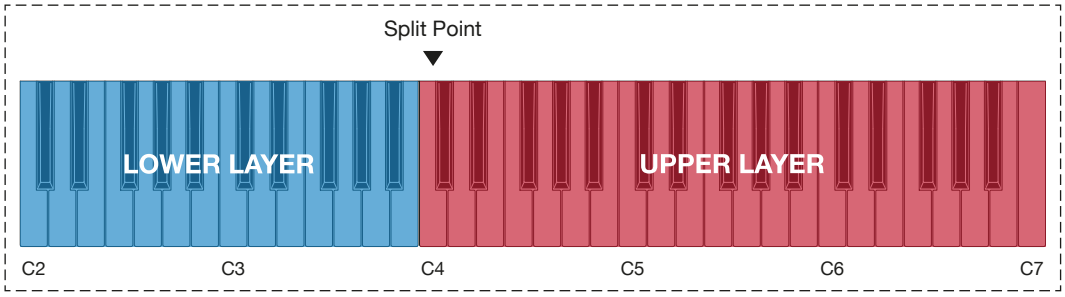
DUAL: In this mode, the upper and lower layers are stacked, with the 20 (Super Gemini) or 16 (Super 8) available voices split evenly between the two. If you activate binaural mode for a layer, its respective voice-count will be reduced to five (Super Gemini) or four (Super 8).



Dual mode.

SPLIT: In split mode, the upper and lower layers are mapped either side of a split point on the keyboard, with the 20 (Super Gemini) or 16 (Super 8) available voices split evenly between the two. If a layer is set to binaural mode, it will be limited to five (Super Gemini) or four voices (Super 8). The upper layer is assigned to all keys above and including the designated split point, while the lower layer is assigned to all keys below.

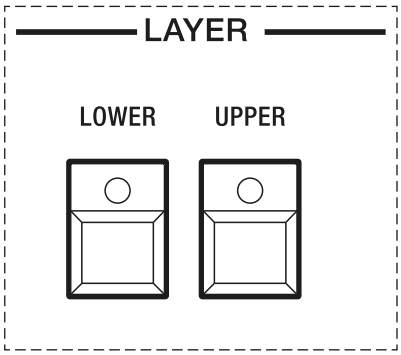
Hold **SPLIT** and play a note to set the split point. The split point is the first note of the upper layer. By default it is set to C4.



Split mode.

In dual and split modes each layer will be directed to its respective outputs. Since single mode uses both layers for full polyphony, you will need to use the mix outputs to achieve the same result.

Layer



The layer section.

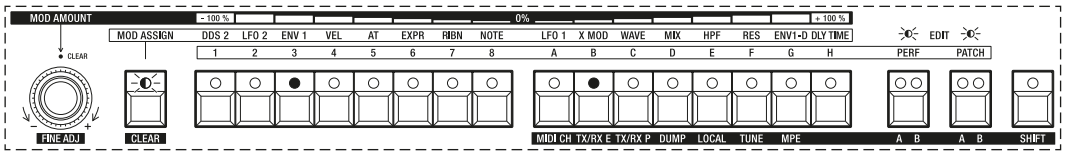
You can use the select buttons in the layer section to switch between the patch settings of the individual layers.

UPPER: Press this button to load a patch into the upper layer or to access the patch-related settings for the upper layer.

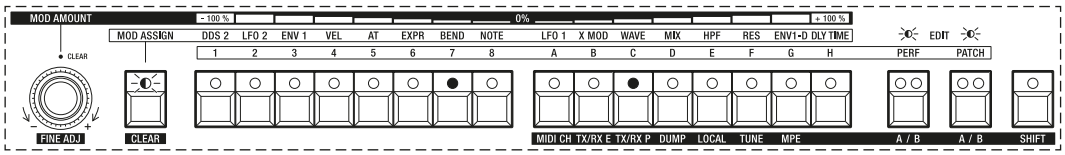
LOWER: Press this button to load a patch into the lower layer or to access the patch-related settings for the lower layer.

USING THE MODULATION MATRIX

The top panel and the performance control section offer you numerous options for assigning a variety of modulation sources to different modulation destinations. You can go beyond these options by using the modulation matrix of each layer. There are two methods for creating modulation assignments, which are described below.



The front panel section relevant for the modulation matrix of the Super Gemini, including the ribbon controller as a freely assignable modulation source (**RIBN**).



The front panel section relevant for the modulation matrix of the Super 8, including the bender as a freely assignable modulation source (**BEND**).

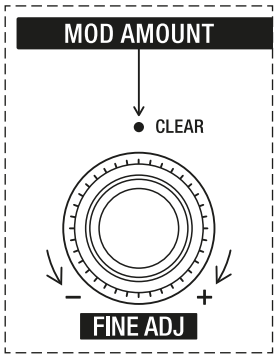
The modulation matrix is entered by pressing the **MOD ASSIGN** button. Its LED then flashes to indicate that you are in modulation assignment mode. In this mode, buttons **1**-**8** represent eight modulation sources, while buttons **A**-**H** represent eight modulation destinations.

The following modulation sources and destinations are available:

	Modulation Source
1	DDS 2
2	LFO 2
3	Envelope 1
4	Velocity
5	Aftertouch
6	Expression Pedal/CV
7	Ribbon Controller (Super Gemini) Bender (Super 8)
8	Note Number

	Modulation Destination
A	LFO 1 Rate
B	Cross Modulation
C	Wave Modulation
D	Oscillator Mix
E	High-Pass Filter Cutoff Frequency
F	Low-Pass Filter Resonance
G	Envelope 1 Decay
H	Delay Time

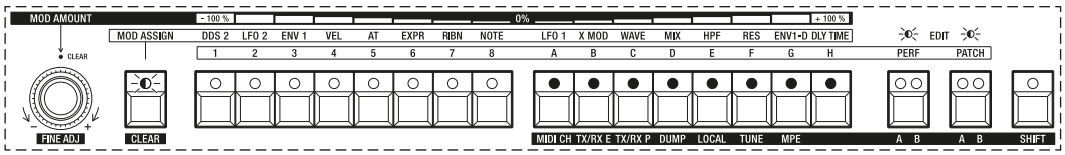
The modulation amount, i.e. the depth at which a modulation destination is modulated, is set by turning the **MOD AMOUNT** encoder.



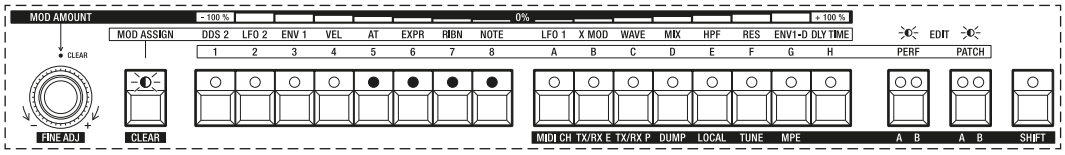
The endless encoder used for dialling in the modulation amount.

You can adjust the modulation amount over a range of -100% (negative modulation amount) to +100% (positive modulation amount). The lit LEDs of buttons **1**-**8** and **A**-**H** indicate the respective setting, while the printed top row above buttons **1**-**8** and **A**-**H** serves as a legend for the modulation amount.

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

With this method of creating modulation mappings, you can assign each of the eight fixed sources to each of the eight fixed destinations with individual modulation amounts:

1. Press the **MOD ASSIGN** button. Its LED then flashes to indicate 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 **PERF** or **PATCH** button to exit the modulation matrix and return to performance or 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

With this alternative method of creating modulation mappings, you can assign each modulation source in the modulation matrix to front panel parameters that are not available in the modulation matrix:

1. Press the **MOD ASSIGN** button. Its LED then flashes to indicate 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 **PERF** or **PATCH** button to exit the modulation matrix and return to performance or 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.

*Already existing modulation mappings will not be duplicated via the direct parameter mapping method. For example, since **VCF** is already a selectable modulation destination for LFO 2 in the performance control section, you cannot map LFO 2 to the cutoff frequency of the low-pass filter using the modulation matrix.*

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

	Modulation Destination		Modulation Destination
1	DDS 2 Tune	13	Envelope 2 Decay
2	Low-Pass Filter Cutoff Frequency	14	Envelope 2 Sustain
3	VCF Envelope Amount	15	Envelope 2 Release
4	VCF LFO 1 Amount	16	LFO 1 Delay
5	VCF DDS 2 Amount	17	LFO 1 LR Phase
6	VCA Envelope Level	18	LFO 2 Rate
7	VCA LFO 1 Amount	19	LFO 2 Delay
8	VCA DDS 2 Amount	20	DDS Modulator LFO 1 Amount
9	Envelope 1 Attack	21	DDS 2 Pulse Width/DDS 1 Detune
10	Envelope 1 Sustain	22	Portamento Time
11	Envelope 1 Release	23	Delay Send
12	Envelope 2 Attack	24	Delay Feedback

Clearing Modulation Mappings

To clear all modulation mappings:

1. Press **MOD ASSIGN** to enter the modulation matrix.
2. Press the **MOD AMOUNT** encoder or press **SHIFT** and then 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:

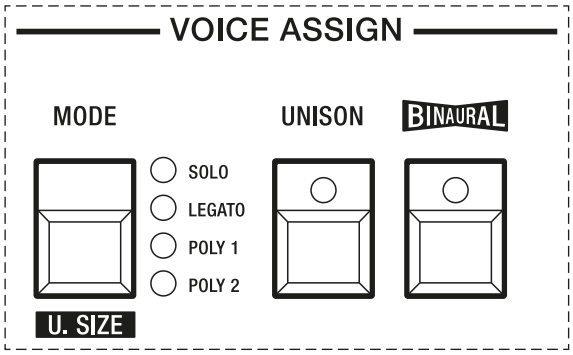
1. Press **MOD ASSIGN** to enter the modulation matrix.
2. Press the button assigned to the desired modulation source.
3. Press the **MOD AMOUNT** encoder or press **SHIFT** and then press the **MOD ASSIGN** button.
4. An LED scroll across buttons **1**-**8** and **A**-**H** indicates that all modulations 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. Press the **MOD AMOUNT** encoder or press **SHIFT** and then 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.

VOICE ASSIGN

In the voice assign section you can specify for each layer how the instrument'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.

MODE: This button allows you to choose from four different voice assign modes:

- **SOLO:** In solo mode, the instrument 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 instrument 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.
- **POLY 1:** This mode gives you full polyphony per layer. The total number of voices depends on whether you are in single, dual or split mode and 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 per layer. The total number of voices depends on whether you are in single, dual or split mode and whether binaural mode is activated or not. The release stage of overlapping notes is curtailed in this mode.

UNISON: In unison mode, the instrument's voices are stacked for massive sounds. Enabling unison in solo and legato modes stacks all available voices for each note played. In both polyphonic modes, the total number of available stacked voices is divided by the number of notes currently being played.

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:** Half of all available voices are stacked.
- **2 LEDs:** All available voices are stacked.
- **3 LEDs:** All available voices are stacked as an octave.
- **4 LEDs:** All available voices are stacked as octave and fifth.

BINAURAL: Use this button to turn binaural mode on or off. The Super Gemini and the Super 8 default to binaural mode, in which their 20 (Super Gemini) or 16 (Super 8) voices are twinned to form either ten (Super Gemini) or eight (Super 8) stereo “super voices” in single mode or five (Super Gemini) or four (Super 8) stereo “super voices” in dual or split mode.

If binaural mode is deactivated, the instrument switches to a monaural signal path with either 20 (Super Gemini) or 16 (Super 8) voices in single mode or ten (Super Gemini) or eight (Super 8) voices in dual or split mode.

In non-binaural mode, the **LR PHASE** fader in the LFO 1 section becomes a pan spread control (**SPREAD**). At the lowest setting, all voices are centred. At the highest setting, all voices are alternately hard-panned between the left and right channels.

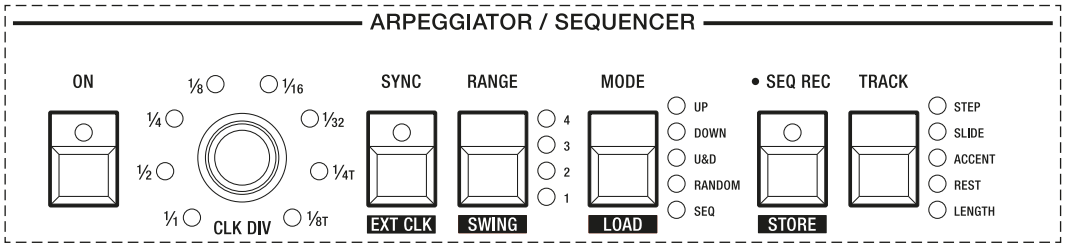
*In **SOLO** and **LEGATO** modes, non-binaural voices are not panned. In these instances, **SPREAD** only controls phase offsets for LFO 1.*

To generate a true monaural signal for each layer:

1. Press **BINAURAL** to deactivate binaural mode.
2. Set **LR PHASE** (or **SPREAD** in this mode) to zero.

ARPEGGIATOR & SEQUENCER

The Super Gemini and Super 8 feature a flexible arpeggiator and a simple but powerful 64-step sequencer. By combining different arpeggios and sequences in dual or split mode, you can create complex rhythm patterns.



The arpeggiator and sequencer section.

Clock Parameters

CLK DIV: This control allows you to set the playback speed in clock divisions relative to the tempo set with the **TEMPO** control or an external MIDI clock source. The currently selected clock division, for example quarter notes, eighth notes or quarter note triplets, is indicated by a lit LED. Pressing the **CLK DIV** encoder resets the clock division to 1/16.

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 of LFO 1 and the **TIME** control in the delay section to set 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. When the arpeggiator or sequencer is synchronised to an external MIDI clock source, you can use the **CLK DIV** control to set the playback speed in clock divisions relative to the external tempo. If you enable MIDI Clock Receive, the **TEMPO** control will have no effect.

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 instrument will respond to MIDI Start/Stop messages.
- **BUTTON 3:** Reserved for future use.

- **BUTTON 4:** MIDI Stop Message Receive. When enabled (LED lit), each of the two **HOLD** buttons "arms" the sequencer of the respective layer while it 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 starts and stops the sequencer of the respective layer 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 will not respond until a MIDI clock signal is received.

The instrument 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 instrument, which then allows you to enable arpeggiator or sequencer playback regardless of external clock signals.

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.

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 activated. See pages 95-98 for more details on the sequencer.

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.

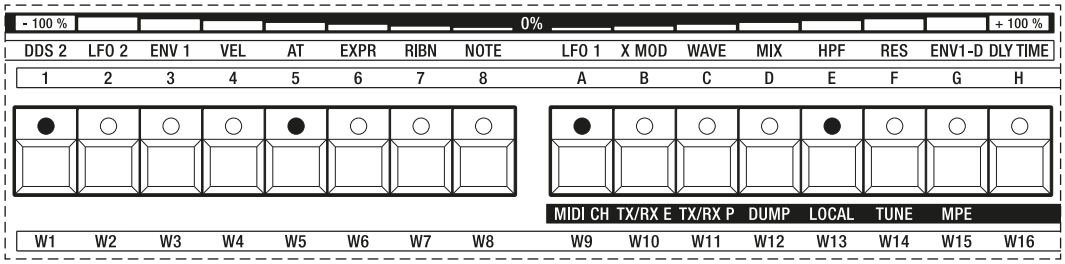
If you activate the hold function of one of the layers while the arpeggiator is switched on, the arpeggio of the respective layer will continue to play without you having to hold down a key. As soon as you release all keys and play a new chord, a new arpeggio is generated.

Sequencer Mode

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

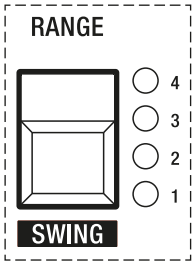
Up to 16 sequences can be stored and recalled. Although a sequence can be linked to a patch when saved, the independent sequencer memory allows you to try out different sequences with each patch.

Once sequencer mode is activated, 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.

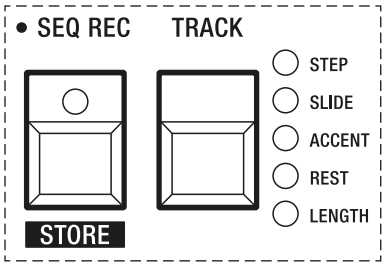
RANGE: In sequencer mode, this button enables you to jump to the following pages of the step sequence:

- **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 the other playback modes related to the arpeggiator, see page 94.

SEQ REC: Pressing this button arms the sequencer for recording. Its LED will then flash, indicating that you are now in recording mode.

TRACK: This button allows you to 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 **TRACK** button LEDs that indicate which track is currently selected.

- **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 have 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 of each layer 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 of the last step in the sequence and the LEDs of all steps before it light up. As soon as the last step has been triggered, the sequence starts again from the beginning.

*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.*

If you activate the hold function of one of the layers, the sequence of the respective layer 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: Press **SHIFT** and then press the **MODE** 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. Press **SHIFT** and then press the **MODE** 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: Press **SHIFT** and then 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.

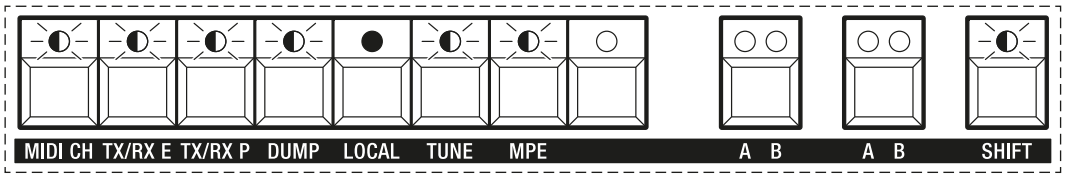
Let's say you would like to store your sequence to memory location 4:

1. Press **SHIFT** and then 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 instrument 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 will exit the global settings.

Any changes applied to the global settings will be automatically saved after 30 seconds of no interaction with the instrument.



The global settings section.

MIDI CH: Press this button to specify which base MIDI channel the instrument 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.

The MIDI channel selected here is assigned to the upper layer, while the MIDI channel one number higher is assigned to the lower layer. For example, if you set the base MIDI channel to 5, MIDI channel 5 will be assigned to the upper layer and MIDI channel 6 to the lower layer.

Do not connect a USB cable when using the MIDI DIN ports. The simultaneous use of DIN and USB-MIDI is not supported.

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 instrument 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 instrument's front panel controls and keyboard are connected to the internal sound engine.
- **OFF:** When Local Control is off (LED not lit), the instrument'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 calibrate the filters. 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: Reserved for future use.

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

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

Saved data such as performances, patches, alternative waveforms and sequences are not affected by a global reset.

FILE MANAGEMENT

When you connect your instrument to a computer, you can easily access and organise the saved performances, patches, alternative waveforms and sequences. This is useful for sharing and backing up related files, as well as freeing up the instrument's internal storage.

Follow the steps below to unlock the instrument's drive:

1. Turn off the instrument and wait a few seconds.
2. Whilst holding down the **PATCH** button, turn on the instrument 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. Then all LEDs extinguish and the envelope LED(s) will start flashing. (If this is not the case, repeat step 2.)
4. Release the **PATCH** button.

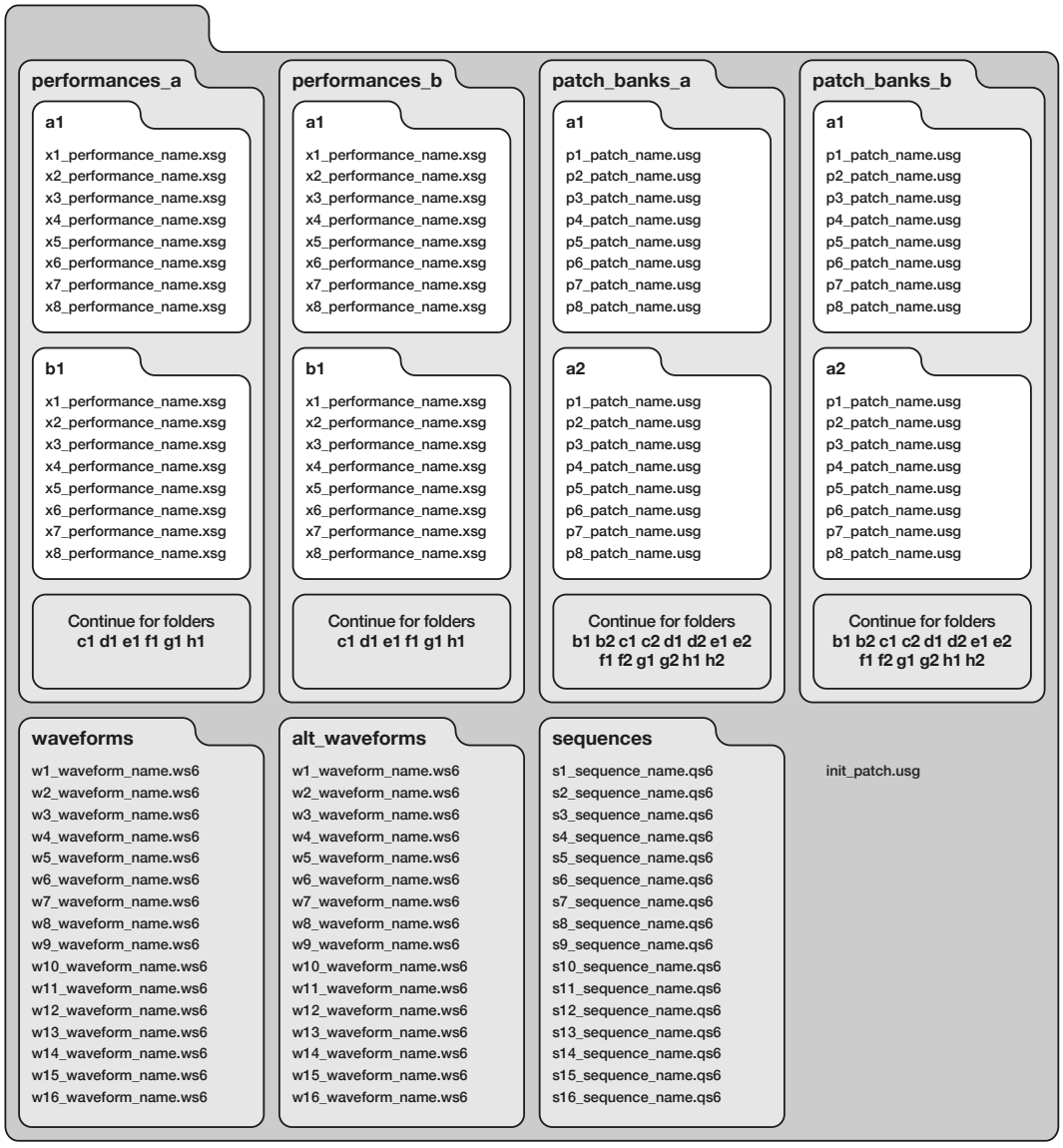
If you then connect the instrument to your computer using a USB cable, it will appear on your computer as a drive called **GEMINI** or **SUPER8**, which you can access like any other flash drive.

The instrument's drive contains the init patch file and seven folders:

- "performances_a" (the first group of 64 performances),
- "performances_b" (the second group of 64 performances),
- "patch_banks_a" (the first group of 128 patches),
- "patch_banks_b" (the second group of 128 patches),
- "waveforms" (the first group of 16 alternative waveforms),
- "alt_waveforms" (the second group of 16 alternative waveforms) and
- "sequences".

The folders "performances_a" and "performances_b" each contain up to 8 subfolders: One for each bank of performances. Within each of the bank folders "a1" to "h1" up to 8 performance files are stored. The folders "patches_a" and "patches_b" each contain 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.

If the instrument's drive is unlocked, you will not be able to play it, but can only manage or change the files stored on it.



The file structure of the Super Gemini and the Super 8 drive.

File Name Convention

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

Performance files: x1_performance_name.xsg

Patch files: p1_patch_name.usg

Waveform files: w1_waveform_name.ws6

Sequence files: s1_sequence_name.qs6

After the prefix, the number and an underscore, performance, 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.

Importing Performances From Your Computer

1. Follow steps 1-4 on page 102 to unlock the instrument's drive.
2. Connect the instrument to your computer using a USB cable.
3. The instrument appears on your computer as a drive named **GEMINI** or **SUPER8**.
4. Click on the icon of the instrument's drive.
5. Navigate to the folder "performances_a" or "performances_b".
6. Open the desired bank folder ("a1"-"h1").
7. Copy the performance files you would like to transfer to the instrument and paste them into the folder you selected in the previous step.
8. If necessary, change the name prefix of the copied and pasted performance files so that it matches the desired location. Make sure that you manually delete the performance files to be replaced in the selected folder if the names of the new performance files do not match the names of the old performance files. Empty the trash on your computer so that the files are actually deleted from the instrument's drive.

Importing Patches From Your Computer

1. Follow steps 1-4 on page 102 to unlock the instrument's drive.
2. Connect the instrument to your computer using a USB cable.
3. The instrument appears on your computer as a drive named **GEMINI** or **SUPER8**.
4. Click on the icon of the instrument's drive.
5. Navigate to the folder "patch_banks_a" or "patch_banks_b".
6. Open the desired bank folder ("a1"-"h2").
7. Copy the patch files you would like to transfer to the instrument and paste them into the folder you selected in the previous step.
8. If necessary, change the name prefix of the copied and pasted patch files so that it matches the desired location. Make sure that you manually delete the patch files to be replaced in the selected folder if the names of the new patch files do not match the names of the old patch files. Empty the trash on your computer so that the files are actually deleted from the instrument's drive.

Importing Waveforms From Your Computer

1. Follow steps 1-4 on page 102 to unlock the instrument's drive.
2. Connect the instrument to your computer using a USB cable.
3. The instrument appears on your computer as a drive named **GEMINI** or **SUPER8**.
4. Click on the icon of the instrument's drive.
5. Navigate to the folder "waveforms" or "alt_waveforms".
6. Copy the waveform files you would like to transfer to the instrument and paste them into the folder you selected in the previous step.
7. If necessary, change the name prefix of the copied and pasted waveform files so that it matches the desired location. Make sure that you manually delete the waveform files to be replaced in the selected folder if the names of the new waveform files do not match the names of the old waveform files. Empty the trash on your computer so that the files are actually deleted from the instrument's drive.

UDO will periodically release further waveform packs, which can be downloaded at udo-audio.com/downloads.

Importing Sequences From Your Computer

1. Follow steps 1-4 on page 102 to unlock the instrument's drive.
2. Connect the instrument to your computer using a USB cable.
3. The instrument appears on your computer as a drive named **GEMINI** or **SUPER8**.
4. Click on the icon of the instrument's drive.
5. Navigate to the folder "sequences".
6. Copy the sequence files you would like to transfer to the instrument and paste them into the folder you selected in the previous step.
7. If necessary, change the name prefix of the copied and pasted sequence files so that it matches the desired location. Make sure that you manually delete the sequence files to be replaced in the selected folder if the names of the new sequence files do not match the names of the old sequence files. Empty the trash on your computer so that the files are actually deleted from the instrument's drive.

Backing up Performances on Your Computer

1. Connect the instrument to your computer using a USB cable.
2. The instrument appears on your computer as a drive named **GEMINI** or **SUPER8**.
3. Click on the icon of the instrument's drive.
4. Click on the folder "performances_a" or "performances_b" and copy it to your computer's hard drive. You can also navigate to one of the 8 bank folders ("a1"-"h1") or to a single performance file within these folders to copy it to your computer's hard drive.

Backing up Patches on Your Computer

1. Connect the instrument to your computer using a USB cable.
2. The instrument appears on your computer as a drive named **GEMINI** or **SUPER8**.
3. Click on the icon of the instrument's drive.
4. Click on the folder "patch_banks_a" or "patch_banks_b" and copy it to your computer's hard drive. You can also navigate to one of the 16 bank folders ("a1"-"h2") or to a single patch file within these folders to copy it to your computer's hard drive.

Backing up Waveforms on Your Computer

1. Connect the instrument to your computer using a USB cable.
2. The instrument appears on your computer as a drive named **GEMINI** or **SUPER8**.
3. Click on the icon of the instrument's drive.
4. Click on the folder "waveforms" or "alt_waveforms" and copy it to your computer's hard drive. You can also navigate to a single waveform file within these folders to copy it to your computer's hard drive.

Backing up Sequences on Your Computer

1. Connect the instrument to your computer using a USB cable.
2. The instrument appears on your computer as a drive named **GEMINI** or **SUPER8**.
3. Click on the icon of the instrument's drive.
4. Click on the folder "sequences" and copy it to your computer's hard drive. You can also navigate to a single sequence file to copy 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.usg” on your computer. You may insert any additional information between the name “init_patch” and the file extension “usg”. For example, you could name it “init_patch_binaural_extravaganza.usg”.

Once you have named and saved your customised init patch, copy it to the drive of the Super Gemini or Super 8:

1. Follow steps 1-4 on page 102 to unlock the instrument’s drive.
2. Connect the instrument to your computer using a USB cable.
3. The instrument appears on your computer as a drive named **GEMINI** or **SUPER8**.
4. Click on the icon of the instrument’s drive.
5. Delete the init patch file “init_patch.usg” from the instrument’s drive and make sure to empty the trash if you are a macOS user.
6. Copy the desired init patch file from your computer to the instrument’s 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!
7. When the transfer is complete, disconnect the USB cable.
8. Turn off the instrument, wait a few seconds and turn it on again.

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

Importing New Waveform Packs

Additional waveform packs can easily be stored on the drive of the Super Gemini or Super 8:

1. Download additional waveform packs to your computer from udo-audio.com/downloads.
2. Extract the .zip file on your computer 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 102 to unlock the instrument’s drive.
4. Connect the instrument to your computer using a USB cable.
5. The instrument appears on your computer as a drive named **GEMINI** or **SUPER8**.
6. Click on the icon of the instrument’s drive.
7. Delete the waveform folder you would like to replace from the instrument’s drive and make sure to empty the trash if you are a macOS user.
8. Copy the new waveform folder from your computer to the instrument’s 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 instrument, 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”

Setting up Your Instrument as a MIDI Device in a DAW

Follow the steps below to use your Super Gemini or Super 8 in conjunction with a DAW:

1. Make sure you only connect the instrument via either MIDI DIN or USB. The simultaneous use of DIN and USB-MIDI is not supported.

2. In the preferences of your DAW, ensure that MIDI data is received from and sent to the Super Gemini or Super 8.

3. If you would like to sync the arpeggiator and sequencer with your DAW, make sure that your DAW sends a MIDI clock signal to the instrument. Also enable External Clock Receive on the instrument.

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

5. Create two MIDI tracks for both layers: one for the upper layer and one for the lower layer.

    - For both MIDI tracks, make sure that MIDI data is sent to the MIDI device **SUPER GEMINI** or **SUPER 8** (if connected via USB) or to the MIDI interface to which either instrument is connected.

    - For the upper layer MIDI track, set the MIDI channel to 1 or your preferred channel.

    - For the lower layer MIDI track, set the MIDI channel to 2 or one channel above the channel you selected for the upper layer.

6. Create two audio tracks for both layers: one for the upper layer and one for the lower layer.

    - Route the audio signal from each MIDI track to the respective audio track.

    - For the upper layer audio track, set the audio channels to the stereo inputs of your audio interface the upper layer outputs of the instrument are connected to.

    - For the lower layer audio track, set the audio channels to the stereo inputs of your audio interface the lower layer outputs of the instrument are connected to.

If you only use the mix outputs, you only need to create one audio track that receives its audio signal from the stereo inputs of your audio interface to which the mix outputs of the instrument are connected.

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 + **PERF**	Select performance group B
SHIFT + **PATCH**	Select patch group B
SHIFT + **MANUAL UPPER** or **LOWER**	Load init patch into the upper/lower layer
Press and hold button **1**-**8** in performance mode	Store performance
Press and hold button **1**-**8** in patch mode	Store patch
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
Press and hold **SPLIT** + note on keyboard	Set split point
SEQ REC + **MOD ASSIGN**	Clear sequence
Press and hold either **MANUAL UPPER** or **MANUAL LOWER** for 5 seconds	Reset global settings

Secondary Parameter Functions

Parameter	Mode	Function
DDS 2 Mode	DDS 2 Range = LFO	Sub-Oscillator Off = Sub-oscillator off Square = Square wave sub-oscillator Sine = Sine wave sub-oscillator
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
Mix	Shift	Pan
Envelope 1 Attack	Shift	Envelope 1 Attack Hold
Envelope 1 Decay	Shift	Envelope 1 Decay Hold
Envelope 2 Decay	Shift	Envelope 2 Decay Hold
DDS 2 Tune	Shift	Layer Detune +/- 7 semitones (Super 8 only)
Detune	Shift	Performance Detune +/- 7 semitones (Super Gemini only)
Voice Assign Mode	Shift	Unison Size 1 = Half of all available voices 2 = All available voices 3 = Octave 4 = Fifth + octave
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
Arpeggiator Mode	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**	Performance	Load performance 1-8
1-**8**	Patch	Load patch 1-8
1-**8**	Mod Assign	Select modulation source
A-**H**	Performance	Select performance bank A-H
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	Reserved for future use
EXT CLK	External Clock Settings 1 = MIDI Clock Transmit on/off 2 = MIDI Clock Receive on/off 3 = Reserved for future use 4 = MIDI Stop Message Receive on/off
Octave selector toggle switch	Global Transpose +/- 12 semitones
FINE ADJ	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	Yes	Yes
Control Change	See “Global Settings” (pages 99-100)	See “Global Settings” (pages 99-100)
Program Change	See “Global Settings” (page 100)	See “Global Settings” (page 100)
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 front panel controls. These messages are transmitted and/or received according to the **TX/RX E** configuration in the global settings (see pages 99-100).

CC#	Value Range	Parameter Name
0	0-127	Bank Select
1	0-127	Modulation Lever
2	0-127	Ribbon Coarse
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	0-127	Mod Amount/Fine Adjust
10	0-127	Pan
11	0-127	Expression
12	0-127	Delay Time
13	0-127	Delay Feedback
14	0-15	Sequence Load
15	-	-
16	0 = Triangle 21 = Rev Sawtooth 43 = Random 64 = Square 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/Pan Spread

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 Envelope 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/Wave Modulation
27	0 = Manual 43 = LFO 1 85 = ENV 1	PWM/Wave Modulation Source
28	0-127	Cross Modulation
29	0 = Sine 21 = Sawtooth 43 = Square 64 = Triangle 85 = White 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 = Sawtooth 43 = Square 64 = Triangle 85 = White Noise 107 = Pulse	DDS 2 Waveform
32	0-127	Envelope 1 Decay Hold
33	0-127	Envelope 2 Decay Hold
34	0-127	Ribbon Fine
35	0-127	DDS 2 Tune
36	0 = Norm/Sub Osc Off 43 = Ring/Sub Osc Square 85 = Sync/Sub Osc Sine	DDS 2 Mode

CC#	Value Range	Parameter Name
37	0-127	Oscillator Mix
38	0-127	LSB for Control 6 (Data Entry)
39	-	-
40	-	-
41	0 = Off 43 = 1 85 = 2	VCF Drive
42	0-127	VCA LFO 2 Amount
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 Envelope 1 85 = Fixed Envelope 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
54	0-127	Envelope 1 Decay
55	0-127	Envelope 1 Sustain
56	0-127	Envelope 1 Release
57	0-127	Envelope 2 Decay

CC#	Value Range	Parameter Name
58	0-127	Envelope 2 Sustain
59	64 = On	Manual Mode
60	0 = Trig 43 = AT + Trig 85 = LFO 2 On/AT->Bend	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	0 = Both 43 = Lower 85 = Upper	Portamento Layer Select
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
76	0-127	DDS Pitch Bend Amount
77	0-127	VCF Pitch Bend Amount

CC#	Value Range	Parameter Name
78	0 = Solo 32 = Legato 64 = Poly 1 96 = Poly 2	Voice Assign Mode
79	1 = Half of all available voices 2 = All available voices 3 = Octave 4 = Fifth + octave	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 Send
92	0-127	VCA LFO 1 Amount
93	0 = Off 32 = Chorus 1 64 = Chorus 2 96 = Chorus 1 & 2	Chorus
94	0-127	Drift

CC#	Value Range	Parameter Name
95	0-127	HPF 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	0 = Both 43 = Lower 85 = Upper	Modulation Layer
105	0 = Sine 21 = Rev Sawtooth 43 = Sample & Hold 64 = Square 85 = Sawtooth 107 = Noise	LFO 2 Waveform
106	0 = LFO 21 = 32’ 43 = 16’ 64 = 8’ 85 = 4’ 107 = 2’	DDS 2 Range
107	0 = Whole notes 16 = Half notes 32 = Quarter notes 48 = Eighth notes 64 = Sixteenth notes 70 = Thirty-second notes 96 = Quarter note triplets 102 = Eighth note triplets	Clock Divider
108	0-127	Lower Layer Detune (Super Gemini) Layer Detune (Super 8)
109	0-127	VCA DDS 2 Amount
110	0-127	LFO 2 Rate Modulation
111	0-127	Performance Detune (Super 8: Receive only)
112	-	-

CC#	Value Range	Parameter Name
113	-	-
114	-	-
115	-	-
116	-	-
117	-	-
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 front panel controls. These messages are transmitted and/or received according to the **TX/RX E** configuration in the global settings (see pages 99-100).

RPN	RPN MSB (CC101)	RPN LSB (CC100)	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. These messages are transmitted and/or received according to the **TX/RX E** configuration in the global settings (see pages 99-100).

Global Parameters

NRPN	NRPN MSB (CC99)	NRPN LSB (CC98)	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 (CC99)	NRPN LSB (CC98)	Value Range	Parameter Name
1024	-	-	-	-
1025	08H	01H	0-16383	Modulation Lever
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	08H	09H	0-16383	Mod Amount/Fine Adjust
1034	08H	0AH	0-16383	Pan
1035	08H	0BH	0-16383	Expression
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/Spread
1044	-	-	-	-
1045	08H	15H	0-16383	DDS LFO 1 Amount
1046	08H	16H	0-16383	DDS Envelope 1 Amount
1047	-	-	-	-
1048	-	-	-	-

NRPN	NRPN MSB (CC99)	NRPN LSB (CC98)	Value Range	Parameter Name
1049	08H	19H	0-16383	PW/Detune
1050	08H	1AH	0-16383	PWM/Wave Modulation
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
1062	-	-	-	-
1063	-	-	-	-
1064	-	-	-	-
1065	-	-	-	-
1066	08H	2AH	0-16383	VCA LFO 2 Amount
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	-	-	-	-
1074	-	-	-	-

NRPN	NRPN MSB (CC99)	NRPN LSB (CC98)	Value Range	Parameter Name
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
1100	08H	4CH	0-16383	DDS Pitch Bend Amount

NRPN	NRPN MSB (CC99)	NRPN LSB (CC98)	Value Range	Parameter Name
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 Send
1116	08H	5CH	0-16383	VCA LFO 1 Amount
1117	-	-	-	-
1118	08H	5EH	0-16383	Drift
1119	08H	5FH	0-16383	HPF Cutoff Frequency
1120	-	-	-	-
1121	-	-	-	-
1122	-	-	-	-
1123	-	-	-	-
1124	-	-	-	-
1125	-	-	-	-
1126	-	-	-	-

NRPN	NRPN MSB (CC99)	NRPN LSB (CC98)	Value Range	Parameter Name
1127	-	-	-	-
1128	-	-	-	-
1129	-	-	-	-
1130	-	-	-	-
1131	-	-	-	-
1132	08H	6CH	0-16383	Lower Layer Detune (Super Gemini) Layer Detune (Super 8)
1133	08H	6DH	0-16383	VCA DDS 2 Amount
1134	08H	6EH	0-16383	LFO 2 Rate Modulation
1135	08H	6FH	0-16383	Performance Detune (Super 8: Receive only)

Head to our downloads page for the most up to date MIDI specification.

GLOSSARY

The following list contains brief explanations of the key terms printed on the front panel of the Super Gemini and Super 8, as well as the 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.

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, all available voices are twinned to form ten (Super Gemini) or eight (Super 8) stereo “super voices” in single mode, or five (Super Gemini) or four (Super 8) stereo “super voices” in dual or split mode. 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.

Bi-timbrality: A bi-timbral synthesizer is an instrument that can generate two different sounds simultaneously, each controlled by a different MIDI channel.

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 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 the centroid oscillator of DDS 1.

Direct Digital Synthesis Oscillator (DDS): Direct Digital Synthesis is the signal generation method employed by both oscillator cores. 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 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.

Dual Mode: In this mode, the upper and lower layers are stacked, with the 20 (Super Gemini) or 16 (Super 8) available voices split evenly between the two. If you activate binaural mode for a layer, its respective voice-count will be reduced to five (Super Gemini) or four (Super 8).

Envelope (ENV): An envelope is a modulation source that defines how the signal or parameter it modulates develops over time. Most envelope generators contain four stages that allow you to define the shape of the envelope: attack, decay, sustain and release (ADSR). The envelopes of the Super Gemini and the Super 8 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, keyboard tracking is freely assignable via the **NOTE** button.

Layer: In a bi- or multi-timbral synthesizer, a layer is essentially a placeholder for a patch. The Super Gemini and the Super 8 allow you to play two layers at once, either in dual or in split mode. Each of the two layers contains its own patch. The up to 256 patches are shared between both layers.

Left-Right Phase (LR PHASE): This parameter controls the phase relationship between the left and right channels of the binaural sound engine, i.e. 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 Gemini and the Super 8 feature a loop option for the first envelope and also allow 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 first LFO of the Super Gemini and Super 8 can also be set to high frequencies so that it can be used as a third oscillator, for drone effects or for modulations at audio rates.

Low-Pass Filter (LPF): 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.

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. The oscillators of the Super Gemini and Super 8 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 256 patches can be stored in the internal memory. They are organised in two groups (A and B) of 16 banks featuring 8 patches each.

Performance: Essentially, a performance is a snapshot of the entire instrument, consisting of both layers and the settings for the performance control section (including LFO 2) as well as the global controls, with the exception of **MASTER VOLUME**. A total of 128 performances can be stored in the internal memory. They are organised in two groups (A and B) of 8 banks featuring 8 performances 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 %, no sound can be heard as there is no change in amplitude to generate an 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 emphasises the frequencies around the cutoff frequency. The low-pass filter of the Super Gemini and Super 8 can be driven into self-oscillation if you set the resonance to the highest value. In this case, the filter generates a pitch determined by the cutoff frequency and a timbre that sounds like a sine wave.

Ring Modulation (RING MOD, RING): Essentially, ring modulation is a type of amplitude modulation as well as a form of frequency mixing. It combines two signals, a carrier and a modulator, and outputs their sum and difference while subtracting the frequencies of the original signals. The frequencies resulting from the sum and difference are called sidebands. If the modulator is a sine wave with a frequency of 1,000 Hz and the carrier is a sine wave with a frequency of 500 Hz, the ring modulator will output the sum 1,500 Hz (1,000 plus 500) and the difference 500 Hz (1,000 minus 500). In case the frequencies of the modulator and carrier are identical, only one sideband will be generated, as the difference would always be 0 Hz. When you apply ring modulation to harmonically rich waveforms, such as sawtooth or pulse, rather clangourous sounds will be generated due to the discordant harmonic relationships between the frequencies.

Sequencer: A sequencer is a modulation source that functions like a recording and playback device and sends control signals to a variety of parameters per step, the smallest unit of a sequence. The sequencer of the Super Gemini and Super 8 allows the recording of up to 64 steps and was primarily designed for the recording and editing of note events.

Single Mode: In single mode, only one layer with 20-voice (Super Gemini) or 16-voice (Super 8) polyphony is active. In binaural mode, the number of voices is halved to ten (Super Gemini) or eight (Super 8) voices.

Split Mode: In split mode, the upper and lower layers are mapped either side of a split point on the keyboard, with the 20 (Super Gemini) or 16 (Super 8) available voices split evenly between the two. If a layer is set to binaural mode, it will be limited to five (Super Gemini) or four voices (Super 8). The upper layer is assigned to all keys above and including the designated split point, while the lower layer is assigned to all keys below.

Sub-Oscillator: A sub-oscillator is an oscillator with a fixed waveform that is one octave or more below the frequency of the oscillator to which it is tied. In the case of the Super Gemini and the Super 8, the activated sub-oscillator replaces the audio signal of DDS 2. Its waveform is either a sine wave or a square wave that is one octave below the frequency of DDS 1.

Super Mode: Super mode is a unique feature that takes advantage of the stereo signal path of the Super Gemini and Super 8. 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.

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. Swing is used to make a rhythm pattern “bounce”.

Sync: Also known as “hard sync”, this function forces DDS 2 to restart its duty cycle each time the duty cycle of DDS 1 begins. By setting the frequency of DDS 2 to a higher pitch than DDS 1, you can create complex and harmonically rich timbres.

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 Gemini and Super 8, envelope 2 is mapped to the VCA level by default. You can use this envelope to determine how the volume of a sound develops over time.

Voltage Controlled Filter (VCF): This is the module that gave subtractive synthesis its name. The voltage-controlled filters are an essential part of the sonic character of the Super Gemini and Super 8, shaping the sound of the oscillators by subtracting frequencies from their signals.

Waveform: A waveform describes the shape of a signal generated 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 an additional fundamental tone. A sawtooth wave contains both odd and even harmonics and is bright sounding. It can be used to produce 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. A triangle wave contains only odd harmonics and sounds very soft. It is particularly suitable for creating flute or organ sounds. White noise contains all frequencies and is the most common noise waveform. It is useful for creating wind or percussive sounds.

SUPPORT INFORMATION

If you are experiencing any issues with your Super Gemini or Super 8, please contact our technical support at support@udo-audio.com.

Please provide the following information when you contact 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 our user forum to check whether your question has already been answered.

SUPER GEMINI

SUPER 8

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