Trademarks

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Disclaimer

Novation has taken all possible steps to ensure that the information given here is both correct and complete. In no event can Novation accept any liability or responsibility for any loss or damage to the owner of the equipment, any third party, or any equipment which may result from use of this manual or the equipment which it describes. The information provided in this document may be modified at any time without prior warning. Specifications and appearance may differ from those listed and illustrated.

IMPORTANT SAFETY INSTRUCTIONS

1. Read these instructions.
2. Keep these instructions.
3. Heed all warnings.
4. Follow all instructions.
5. Clean only with dry cloth.
6. Do not install near any heat sources such as radiators, heat registers, stoves, or other apparatus (including amplifiers) that produce heat.
7. Do not defeat the safety purpose of the polarized or grounding-type plug. A polarized plug has two blades with one wider than the other. A grounding type plug has two blades and a third grounding prong. The wide blade or the third prong are provided for your safety. If the provided plug does not fit into your outlet, consult an electrician for replacement of the obsolete outlet.
8. Protect the power cord from being walked on or pinched particularly at plugs, convenience receptacles, and the point where they exit from the apparatus.
9. Only use attachments/accessories specified by the manufacturer.
10. Use only with the cart, stand, tripod, bracket, or table specified by the manufacturer, or sold with the apparatus. When a cart is used, use caution when moving the cart/apparatus combination to avoid injury from tip-over.

11. Unplug this apparatus during lightning storms or when unused for long periods of time.
12. Refer all servicing to qualified service personnel. Servicing is required when the apparatus has been damaged in any way, such as power-supply cord or plug is damaged, liquid has been spilled or objects have fallen into the apparatus, the apparatus has been exposed to rain or moisture, does not operate normally, or has been dropped.
No naked flames, such as lighted candles, should be placed on the apparatus.

WARNING: Excessive sound pressure levels from earphones and headphones can cause hearing loss.

WARNING: This equipment must only be connected to USB 1.1 or 2.0 type ports.

ENVIRONMENTAL DECLARATION

This device complies with part 15 of the FCC Rules. Operation is subject to the following two conditions: (1) This device may not cause harmful interference, and (2) this device must accept any interference received, including interference that may cause undesired operation.

For USA
To the User:

1. **Do not modify this unit!** This product, when installed as indicated in the instructions contained in this manual, meets FCC requirements. Modifications not expressly approved by Novation may void your authority, granted by the FCC, to use this product.

2. **Important:** This product satisfies FCC regulations when high quality shielded cables are used to connect with other equipment. Failure to use high quality shielded cables or to follow the installation instructions within this manual may cause magnetic interference with appliances such as radios and televisions and void your FCC authorization to use this product in the USA.

3. Note: This equipment has been tested and found to comply with the limits for a Class B digital device, pursuant to part 15 of the FCC Rules. These limits are designed to provide reasonable protection against harmful interference in a residential installation. This equipment generates, uses and can radiate radio frequency energy and, if not installed and used in accordance with the instructions, may cause harmful interference to radio communications. However, there is no guarantee that interference will not occur in a particular installation. If this equipment does cause harmful interference to radio or television reception, which can be determined by turning the equipment off and on, the user is encouraged to try to correct the interference by one or more of the following measures:

• Reorient or relocate the receiving antenna.
• Increase the separation between the equipment and receiver.
• Connect the equipment into an outlet on a circuit different from that to which the receiver is connected.
• Consult the dealer or an experienced radio/TV technician for help.

For Canada
To the User:
This Class B digital apparatus complies with Canadian ICES-003

Cet appareil numérique de la classe B est conforme à la norme NMB-003 du Canada.

RoHS Notice

Focusrite Audio Engineering Limited has conformed and this product conforms, where applicable, to the European Union's Directive 2002/95/EC on Restrictions of Hazardous Substances (RoHS) as well as the following sections of California law which refer to RoHS, namely sections 25214.10, 25214.10.2, and 58012, Health and Safety Code; Section 42475.2, Public Resources Code.

CAUTION:
The normal operation of this product may be affected by a strong electrostatic discharge (ESD). In the event of this happening, simply reset the unit by powering off and on again. Normal operation should return.

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CONTENTS

INTRODUCTION

Thank you for purchasing the MiniNova synthesizer. The MiniNova is a powerful compact digital synthesizer equally at home in live performance or a recording environment.

NOTE: The MiniNova is capable of generating audio with a large dynamic range, the extremes of which can cause damage to loudspeakers or other components, and also to your hearing!

KEY FEATURES:

- Full polyphony, with up to 18 voices
- Classic analogue synth waveforms
- 36 wavetables
- 14 filter types
- Built-in digital FX section with compression, panning, EQ, reverb, delay, distortion, chorus and gator effects
- Four assignable rotary controls for immediate access to up to 24 primary sonic parameters
- 8 performance pads for arpeggiator control and adding expression while playing
- 12-band Vocoder with dynamic gooseneck microphone (supplied)
- VocalTune processor
- 37-note velocity-sensitive keyboard
- MIDI input and output
- LCD display

The following features are available in conjunction with the appropriate MiniNova/Novation software (downloadable):

- MiniNova Editor (VST™, AU™, RTAS™ plug-in) for DAW
- Mac/Windows-based librarian software for management of patches

ABOUT THIS MANUAL

We don't know whether you've got years of experience with electronic keyboards, or if this is your very first synth. In all probability, you're somewhere between the two. So we've tried to make this manual as helpful as possible for all types of user, and this inevitably means that more experienced users will want to skip over certain parts of it, while relative novices will want to avoid certain parts of it until they're confident they've mastered the basics.

However, there are a few general points that are useful to know about before you continue reading this manual. We've adopted some graphical conventions within the text, which we hope all types of user will find helpful in navigating through the information to find what they need to know quickly:

Abbreviations, conventions, etc.
As the four rotary controls in the **PERFORM** area of the control panel are referred to repeatedly throughout the manual, we've abbreviated them to **RCn**, where **n** is a number between 1 and 4, referring to the control in question.

Where top panel controls or rear panel connectors are referred to, we've used a number thus: [x] to cross-reference to the top panel diagram, and thus: {x} to cross-reference to the rear panel diagram. (See page 4 and page 5)

We've used **BOLD CAPS** to name top panel controls or rear panel connectors. We've used `LCD dot-matrix text` to denote text which appears on the LCD at the beginning of each parameter description and within the parameter tables, but **Bold** to indicate this text within the main manual paragraphs.

Tips

These do what it says on the tin: we include bits of advice, relevant to the topic being discussed that should simplify setting up the MiniNova to do what you want. It's not mandatory that you follow them, but generally they should make life easier.

Extra Info

These are additions to the text that will be of interest to the more advanced user and can generally be avoided by the novice. They are intended to provide a clarification or explanation of a particular area of operation.

Performance Parameter

MiniNova has a fantastic degree of flexibility in tailoring sounds, as you will see in the second part of this manual, where each individual parameter available in the menu system is described. However, to avoid having to navigate menus during live performance, we have made the most useful and commonly-needed parameters immediately available for adjustment by the four rotary controls in the **PERFORM** area of the control panel. We have clearly indicated these parameters within the parameter descriptions.

WHAT'S IN THE BOX?

The MiniNova has been carefully packed in the factory and the packaging was designed to withstand rough handling. Should the unit appear to have been damaged in transit, do not discard any of the packing material and notify your music dealer.

Save all the packing materials for future use if you ever need to ship the unit again.

Please check the list below against the contents of the packaging. If any items are missing or damaged, contact the Novation dealer or distributor where you purchased the unit.

- MiniNova synthesizer
- Gooseneck microphone
- DC power supply unit (PSU)
- USB cable
- Software download card and Warranty Registration card
- This manual

Registering your MiniNova

It is important to register your MiniNova on-line using the Software download and warranty registration card. Apart from validating your manufacturer's warranty, you will also then be able to download the additional software that you are entitled to as a MiniNova purchaser. The card also contains codes you will need to enter in the on-line forms on our website to download the software, but before you attempt to do this, warranty registration is required.

Power requirements

The MiniNova is shipped with a 9 V DC, 900 mA power supply. The centre pin of the coaxial connector is the positive (+ve) side of the supply. The MiniNova can either be powered by this AC-to-DC mains adaptor, or by the USB connection to a computer. To obtain the best possible audio performance from the MiniNova we recommend using the supplied adaptor.

There are two versions of the PSU, your MiniNova will be supplied with the one appropriate to your country. The PSU comes with detachable adaptors; use the one that fits your country's AC outlets. When powering the MiniNova from the mains PSU, please ensure that your local AC supply is within the range of voltages required by the adaptor – i.e., 100 to 240 VAC - BEFORE you plug it into the mains.

We strongly recommend that you only use the supplied PSU. Failure to do so will invalidate your warranty. Power supplies for your Novation product can be purchased form your music dealer if you have lost yours.

If powering the MiniNova via the USB connection you should be aware that although the USB specification agreed by the IT industry states that a USB port should be able to supply 0.5 A at 5 V, some computers - particularly laptops – are unable to supply this current. Unreliable operation of the synth will result in such a case. When powering the MiniNova from a laptop's USB port, it is strongly recommended that the laptop is powered from AC mains rather than its internal battery.

HARDWARE OVERVIEW

Top view – controls

1. 37-note (3 octaves) keyboard with velocity sensing.

2. **PITCH** and **MOD** wheels: The **PITCH** wheel is mechanically biased to return to the centre position when released.

SELECT/EDIT section

3. Custom 2-row x 8-character LCD dot matrix display for patch selection and menu access. The LCD also incorporates a bargraph meter showing audio input signal level, tempo indication in BPM and other status information.

4. **TYPE/GENRE** selector: Use this to select a subset of available patches.

5. **SORT** switch: allows you to order your set of patches by patch number or alphabetically by name.

6. **DATA** detented rotary control: Used in patch selection, and to alter parameter values within menus.

7. **PAGE** ◀ and ▶ buttons: these are used to step forwards and backwards between menu pages.

8. **MENU/BACK** button: Press to enter menu system; within the menu system, pressing again will jump back to the previous menu level. A 'long' press (> 1 sec) will exit the menu system entirely.

9. **OK** button: Used within the menu system for navigation (goes to next menu level), and to confirm data entry.

10. **SAVE** button: Used to save alterations to patches.

11. Patch ◀ and ▶: dedicated buttons to scroll through the currently available patches. Pressing both buttons simultaneously for at least one second enters DEMO mode.

PERFORM section

12. **Rotary controls**: 4 rotary "Tweak" controls for parameter adjustment. The function of each control is determined by the setting of the PERFORM ROW selector [13]. (Use of a rotary control in the manual text is indicated by 'RCn', where n is the number of the control; e.g., 'RC1' refers to rotary control 1).

13. **Perform Row selector**: This 6-way switch determines the functions of the four rotary controls [12]. An LED indicates the currently selected row, and the parameters then available for adjustment are printed on the MiniNova's top panel. Moving the switch allows you to select any row of the table printed on the panel. The first two rows assign the Tweak controls to parameters which have been factory-selected by the Novation programming team for each Patch, giving you immediate access to the most useful and striking sonic variations.

14. **FILTER**: this is a large rotary control intended to aid more expressive performance when playing live. It always adjusts Filter 1's cut-off frequency.

PAD section

15. **PADS** 1 to 8: a set of eight backlit, multicolour, pressure-sensitive pads, which may be used in two primary ways – Animate or Arpeggiate. Additionally, in conjunction with the FAVORITE button [17], they may be used as "Quick Load" buttons to recall preferred patches.

16. **ANIMATE/ARPEGGIATE** switch: A 2-position switch (spring-loaded to return-to-centre), which assigns the pads [15] to act as Animate controls or Arpeggiator pads.

17. **FAVORITE** button: used to store and recall preferred patches in conjunction with the eight pads [15].

18. **HOLD** button: modifies the action of a pad [15] in Animate mode by "locking" it in an "On" state.

ARP section

19. **ON**: backlit button to switch the Arpeggiator on and off. When selected 'On', the eight pads [15] enter Arpeggiator mode and the Arpeggiator LED in the Pads section illuminates.

20. **LATCH** button: applies the Arpeggiator effect to the last note(s) played continuously, until a subsequent key is pressed. LATCH can be pre-selected so that it is effective as soon as the Arpeggiator is enabled.

21. **TEMPO** control: sets the tempo of the Arpeggiator pattern being played. An adjacent LED blinks to give a visual indication of the tempo, and the actual BPM value is displayed on the LCD.

Misc

22. **Dynamic Mic Input**: an XLR socket for the connection of the supplied gooseneck microphone, or alternative dynamic microphone (i.e., a mic not requiring phantom power to operate). The mic can be used with the MiniNova's vocoder and VocalTune features, or routed to the audio outputs. This input is overridden when a jack plug is plugged in to EXT IN {8} on the rear panel.

23. **MASTER VOLUME**: the level control for the main audio outputs and the headphone output.

24. **OCTAVE +** and **–** buttons: these transpose the keyboard up or down one octave each time they are pressed. Associated multi-colour LEDs confirm that a transposition has been applied.
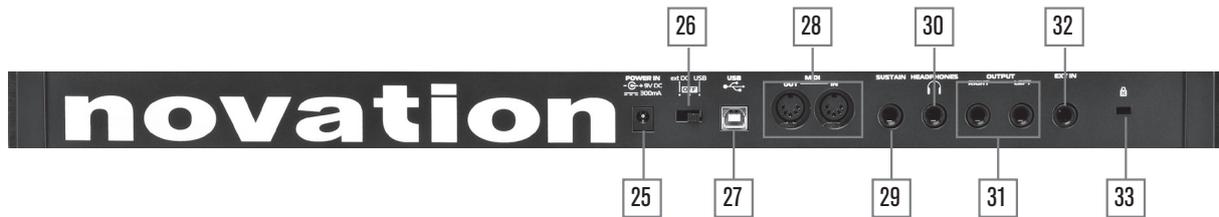
Rear view – connections

25. **DC** power connector: standard 2.2 mm socket for connecting the external **9 V DC PSU** (supplied). See "Power requirements" on page 3.

26. On/off switch: 3-position switch:

| POSITION	ACTION
ext DC	Enables external 9 V DC input
OFF	Off
USB	Enables power via USB port

27. **USB port**: Type B USB Type 1.1 (2.0-compatible) socket for connection to PC or Mac

28. **MIDI connectors**: standard MIDI In/Out sockets (5-pin DINs)

29. **Sustain pedal socket**: 2-pole (mono) ¼" jack socket for connection of a sustain pedal. Both NO (Normally Open) and NC (Normally Closed) pedal types are compatible; if the pedal is connected when the MiniNova is powered on, the type will be automatically sensed during boot-up (provided your foot is not on the pedal!). See "Parameter: Footswitch configuration" on page 13 for more information.

30. **Headphones socket**: 3-pole ¼" jack socket for stereo headphones. Phones volume is adjusted by the MASTER VOLUME control [23].

31. **OUTPUT LEFT** and **RIGHT**: 2 x ¼" jack sockets carrying main stereo output. Outputs are unbalanced, at +5 dBu maximum level.

32. **EXT IN**: ¼" jack socket for external instrument or line level audio inputs. This input overrides an XLR connector plugged into the Dynamic Mic Input [22] on the top panel. Input is balanced, and can accept a maximum input level of 0 dBu. The sensitivity of the input can be adjusted via the menu system (see "Parameter: Input Gain" on page 12).

33. **Kensington Lock Port**: to secure your synthesizer.

GETTING STARTED

Standalone and computer operation – a foreword

The MiniNova may be used as a standalone synthesizer, with or without MIDI connections to/from other sound modules or keyboards. It may also be connected - via its USB port – to a computer (Windows or Mac) running a DAW application. The MiniNova can then be controlled entirely from the computer by using the MiniNova Editor plug-in. The MiniNova Librarian is a separate software application, which greatly assists in the organising, saving and recall of patches.

MiniNova connected to a computer via USB.
USB transmits MIDI to and from the computer
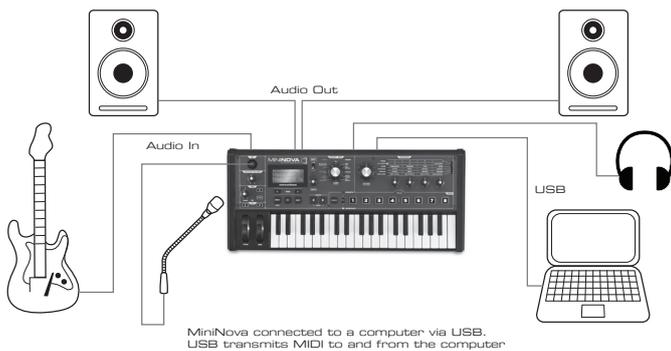
The various ways of connecting the MiniNova to accommodate the various methods of working are covered in the documentation supplied with the MiniNova Editor and MiniNova Librarian software packages. The Installers for this software, and the related USB drivers may be downloaded from http://novationmusic.com/support.

When using the MiniNova with the MiniNova Editor, an **EDITOR** flag is displayed on the LCD to confirm the connection. Also note that a **USB** flag is displayed when MiniNova is connected to a computer via USB, and valid data interchange has been established.

STANDALONE OPERATION – AUDIO AND MIDI CONNECTIONS

The simplest and quickest way of getting started with the MiniNova is to connect the two rear panel jack sockets marked **OUTPUT LEFT** and **RIGHT** {31} to the inputs of a stereo amplifier, audio mixer, powered speakers, third-party computer sound card or other means of monitoring the output.

Note: the MiniNova is not a computer MIDI interface. MIDI can be transmitted between the MiniNova synth and computer via the USB connection, but MIDI cannot be transferred between the computer and external equipment via the MiniNova's MIDI DIN ports.

If using the MiniNova with other sound modules, connect **MIDI OUT** {28} on the MiniNova to **MIDI IN** on the first sound module, and daisy-chain further modules in the usual way. If using the MiniNova with a master keyboard, connect the controller's **MIDI OUT** to **MIDI IN** on the MiniNova, and ensure that the master keyboard is set to MIDI channel 1 (the MiniNova's default channel).

With the amplifier or mixer off or muted, connect the AC adaptor to the MiniNova {25}, and plug it into the AC mains. Turn the MiniNova on by moving the rear panel switch {26} to **ext DC**. After completing its boot sequence, the LCD display will indicate the Patch which has been loaded. If the **TYPE/GENRE** knob has not been moved since the last power-off, this will be the last Patch used. If the **TYPE/GENRE** knob has been moved, the Patch loaded will be the lowest-numbered (or lowest alphabetically-sorted, depending on the setting of the **SORT** switch) in the selected Type or Genre.

Turn on the mixer/amplifier/powered speakers, and turn up the **Master Volume** control [23] until you have a healthy sound level from the speakers when you play the keyboard.

Using headphones

Instead of speakers via an amplifier and/or an audio mixer, you may wish to use a pair of stereo headphones. These may be plugged into the rear panel headphone output socket {30}. The main outputs are still active when headphones are plugged in. The MASTER LEVEL control [23] also adjusts headphone level.

NOTE: The MiniNova headphone amplifier is capable of outputting a high signal level; please take care when setting the volume.

A WORD ABOUT MENU NAVIGATION

The MiniNova has been designed to give the player maximum control over sound character and system operation with the minimum of hassle. The menu system is always entered by pressing the **MENU** button [8]. The menu system consists of six individual menus:

```
Audio In
Global
Arp
Chord
Edit
Dump
```

Step between the menus with the PAGE ◀ and ▶ buttons [7], and press **OK** [9] to enter the desired menu. Use the **PAGE** buttons again to access the parameter you wish to change; use the **DATA** control [6] to alter the parameter value.

The menu system can be exited by pressing the **MENU/BACK** button again; otherwise, it will automatically time-out after a short period, and the screen will revert to display the currently-loaded Patch information.

Scrolling through Patches

Your MiniNova comes pre-loaded with a set of factory Patches, which may be auditioned at any time, providing you are not in the menu system. The Patches are arranged as 3 banks (A to C), each with 128 patches (000 to 127). Banks A and B come pre-loaded with a full set of factory Patches, while Bank C contains 128 copies of an Initial Patch, which you can either overwrite, or use as the basis for creating your own sounds. With the **TYPE/GENRE** selector [4] set to **ALL**, either rotate the **DATA** control [6], or use the **PATCH ◀** and **▶** buttons [11] to step through the patches. The new sound is loaded as soon as the patch data shows in the display.

The Patch set can be browsed in either bank and numerical order, or alphabetically by name, according the setting of the **SORT** switch [5].

Searching through Types or Genres

Apart from being arranged in 3 banks, the patches are also categorised for you according to the type of sound; this makes finding suitable sounds much easier. Each patch belongs to both a Genre and a Type; the Genre broadly indicates the musical area for which the patch might be suitable, the Type alternatively arranges the patches by sonic characteristics. Use the **TYPE/GENRE** control to select the Type or Genre in which you are interested.

Once the Type or Genre has been specified, the patch set can again be browsed either in numerical or alphabetical order.

The Genres and Types are listed below:

TYPES	GENRES
All	
Vocoder/VocalTune	Rock/Pop
Bass	R&B/Hip Hop
Keyboard/Lead	Dubstep
Pad/Strings	House/Techno
Arp/Movement	D&B/Breaks
Classic Synth	

Using the FAVORITE button to load Patches

You can assign up to eight of your favourite Patches to the eight Performance Pads, and then quickly reload them without needing to search through the whole Patch list.

Assigning a Patch to a Pad

With the Patch already loaded, press and hold the **FAVORITE** button [17], and simultaneously press and hold a Pad button. The display will show AssignIn, with a 3 second countdown timer. After 3 seconds, the display changes to Favorite Assigned, and the Patch is now assigned to that Pad. Note that the Pad turns red to confirm the assignment.

Loading a Patch from a Pad

Press and hold the **FAVORITE** button; all the Pads will flash blue (unless the currently-loaded Patch is one that was previously assigned to a Pad, in which case the Pad shows a steady red). While they are flashing, press the Pad which has the Patch you want assigned to it, and that Patch will now be loaded. The LCD will confirm the new Patch by name.

Demo Mode

Press the two **PATCH ◀** and **▶** buttons [11] simultaneously, and the MiniNova will enter Demo Mode. Using any control will cause a brief description its function to be displayed on the LCD screen. Note that none of the controls (except master volume) or keyboard are active in Demo Mode.

MODIFYING SOUNDS - USING THE PERFORMANCE CONTROLS

The MiniNova is equipped with a set of controls specifically designed for use in live performance. These allow you to modify the sound of the loaded patch in a variety of interesting and sometimes startling ways!

These controls are found in the **PERFORM**, **PADS** and **ARP** areas of the control panel (see items 12 - 21 on page 4).

Parameter Controls

While playing live, it is often desirable to manually adjust some aspect or other of the sound – i.e., "tweak" a particular parameter. Although the MiniNova's design allows you to access all the parameters defining a particular sound, it is useful if the most important parameters that you need while playing live are readily available, on a convenient set of controls. These are the four rotary controls to the right of the control panel, see item 12 on page 4.

Use these four knobs in conjunction with the **Perform Row Selector switch** [13]. An LED will illuminate to show you which of the six banks of available parameters the knobs are assigned to. Note that Rows 3 to 6 always control the same parameters, regardless of the patch you have loaded – though the actual effect of the control will quite likely sound different! Rows 1 and 2 place the four knobs into "Tweak" mode, where the parameters they control vary with the patch (see below).

Don't worry too much at this stage what words like "Resonance" and "Sustain" mean – all these (and many other) terms are explained in much greater detail further on in the manual. Just try to become familiar with the actual sonic effect you hear when you adjust each of the parameters in turn, for different categories of patch.

ℹ The four knobs used for "tweaking" will almost never be in the correct position relative to the value of the parameters they control which are stored as part of the currently-loaded Patch. For example, in the Patch A000 ("BassIsWet DC"), the value of the Filter Envelope Decay Time parameter is 27. If the Tweak control for this (RC2 in Row 4) is set to – say – 2 o'clock, the knob position implies a completely different value. The LCD display includes two arrows which tell you which way to turn the knob to get the knob position to "match" the stored parameter value. As long as **Pot Pickup** is set **On** (in the Global Menu), the knob will have no effect until both arrows are off. If **Pot Pickup** is **Off**, turning the knob will immediately alter the parameter, which may produce an audible "jump". See page 13 for more information on Pot Pickup.
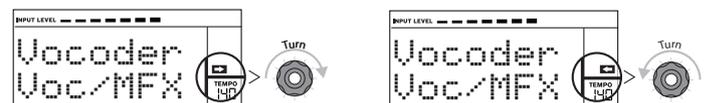
Rows 1 and 2 – Tweak and (FX) Tweak controls

With either Rows 1 or 2 selected, the knobs will have a different effect depending on the patch loaded. This is because the actual assignment of the controls forms part of the Patch. You will find that all the factory Patches have some Tweak Controls pre-assigned, but you can change their function or add others if you wish.

The best way of understanding the Tweak controls is to load a patch and play with them. Try loading the Patch "Synchromatic 1 PS", which can be found in the Arp/Movement TYPE*. Select the **TWEAK** row with the **Perform Row Selector** switch [13]. As you play, adjust each of the four **TWEAK** controls in turn to hear their effect. You will find that you can introduce further variations to the sound. Now select the **(FX) TWEAK** row; you'll find that the **TWEAK** controls now do something different and the sound can be modified in other ways – in this case, by altering the audio effects processing applied to the sound.

The important point to grasp here is that the effect of each **TWEAK** control on the sound is specific to the patch. With different patches loaded, the **TWEAK** controls will alter different sonic characteristics.

NOTE: RC4 is preset to control the level of FX level when Row 2 ((FX) TWEAK) is selected. However, this can be changed in the TWEAK submenu of the EDIT menu.

* You will be able to find this – or any Patch you know by name – more quickly by setting SORT to A-Z and scrolling through the listed Patches alphabetically.

Rows 3 to 6 – Fixed Tweak controls

The function of the four rotary controls is predetermined when any of Rows 3 to 6 is selected. The table below lists the functions, and tells you where to look in the User Guide to find out more information on the parameter controlled in each case.

> ℗ Full details of the parameters each of the Tweak controls in Rows 3 to 6 are available at the page number indicated in the table below.

Row	Group	RC1		RC2		RC3		RC4	
		Parameter	More Info?	Parameter	More Info?	Parameter	More Info?	Parameter	More Info?
3	Filter	Resonance	F1Res page 17	Tracking	F1Track page 17	Type	F1Type page 17	Drive	F1Damnt page 17
4	Filter Envelope	Attack	FltAtt page 22	Decay	FltDec page 22	Sustain	FltSus page 22	Amount	F1Env2 page 17
5	Amplitude Envelope	Attack	AmpAtt page 20	Decay	AmpDec page 20	Sustain	AmpSus page 20	Release	AmpRel page 21
6	Oscillator	Osc1 Virtual Sync	O1VSync page 14	Osc 1 Density	O1Dense page 15	Osc 2 Virtual Sync	O2VSync page 14	Osc 2 Density	O2Dense page 15

The Filter knob

Adjusting the frequency of the synth's primary filter (Filter 1) is probably the most commonly-used method of sound modification. For this reason, Filter 1 Frequency has its own dedicated control in the form of a large rotary control [14] next to the parameter controls. Experiment with different types of patch to hear how changing the filter frequency alters the characteristic of different types of sound.

Using the Pads as performance controls

The eight Pads below the parameter controls have a number of functions on the MiniNova. In this section, we are concerned only with their use as performance controls. To enable the Pads for performance use, set the **ANIMATE/ARPEGGIATE** switch [16] to **ANIMATE**.

Like the **TWEAK** controls, the precise effect that each Pad will have on the characteristic of the sound is Patch-dependent. Again, the best way of understanding what they can do is to load a Patch and play with them. Load the Patch "Cry4Moon DF" - which can be found in the Keyboard/Lead TYPE* - and lightly touch each of the pads in turn while playing normally. You'll find that when you touch a pad, something distinctive happens to the sound. Try loading different types of Patch to see what effect the Pads have in each. Note that not all Patches have all eight Pads assigned.

Later in the manual, you will discover how to reassign the Pads to make specific parameter changes to any given patch, and that these assignments remain with the patch for your future use.

* You will be able to find this – or any Patch you know by name – more quickly by setting SORT to A-Z and scrolling through the listed Patches alphabetically.

The Arpeggiator

The MiniNova has a powerful Arpeggiator feature which allows arpeggios of varying complexity and rhythm to be played and manipulated in real-time. If a single key is pressed, the note will be retriggered by the Arpeggiator. If you play a chord, the Arpeggiator identifies its notes and plays them individually in sequence (this is termed an arpeggio pattern or 'arp sequence'); thus if you play a C major triad, the selected notes will be C, E and G.

The MiniNova Arpeggiator is enabled by pressing the **ARP ON** button [19]; its backlight will confirm and the eight Pads will turn red. Holding a note down will repeat the note in the sequence, and you will see the Pads' illumination changing to purple as the pattern progresses. Initially all enabled beats in the sequence are sounded, but if you press a Pad, the beat corresponding to that Pad's position will now be omitted from the sequence, generating a rhythmic pattern. The 'deselected' Pads will not illuminate. A 'deselected' Pad may be re-enabled by tapping it a second time.

Arpeggiator operation in the MiniNova is controlled by the three **ARP** buttons [19], [20] & [21]: **ON**, **LATCH** and **TEMPO**. The **ON** button enables or disables the Arpeggiator.

The **LATCH** button plays the currently selected arp sequence repeatedly without the keys being held. **LATCH** can also be pressed before the Arpeggiator is enabled. When the Arpeggiator is enabled, the MiniNova will immediately play the arp sequence defined by the last set of notes played, and will do so indefinitely. The tempo of the arp sequence is set by the **TEMPO** control; you can make the sequence play faster or slower by altering this. See page 13 for further details.

The Vocoder

Your MiniNova comes with a Vocoder section, which allows you to create some really great sounds by combining synth sounds with either a voice or another instrument such as a guitar.

To use the Vocoder, first connect a microphone (one is supplied with your MiniNova) to the **MIC** socket [22] on the top panel. Alternatively, you can plug a guitar or other instrument into the **EXT IN** socket {8} on the rear panel (this will disconnect the mic socket). Next, you need to set the audio gain of the mic or instrument. To do this, press **MENU** [8], select **Audio In** using the **DATA** wheel [6], then press **OK** [9]. This will open the menu system, and **Audio In** is the first menu displayed. The first menu item in the Audio Menu is Input Gain (InptGain); adjust the input gain with the **DATA** wheel [6] while noting the signal level as displayed at the top of the LCD screen as a horizontal bargraph meter. Ensure that the loudest audio level does not cause the **OVER** segment to illuminate.

Set the **TYPE/GENRE** control [4] to **VOCODER/VOCALTUNE**, and select a patch from the available subset. Now hold one or more keys down and sing into the microphone (or play the instrument connected at **EXT IN**). You will hear the sound of the synth, modified by the external audio input. As with any other patch, you can alter various parameters with the **FILTER** and four rotary encoders in the **PERFORM** section, or use the Animate functions as described above.

As with all the other performance controls, we recommend that there is no substitute for experimentation to get an understanding of how the various controls interact.

> ℹ Note that two of the factory Vocoder Patches, "Aaah 1" (B073) and "Aaah 2" (B074), do not make use of the built-in microphone. Although these use the MiniNova's Vocoder functions, they use fixed formants which are stored with the Patches.

Pitch and Mod wheels

The MiniNova is fitted with a standard pair of synthesizer control wheels adjacent to the keyboard, **PITCH** and **MOD** (Modulation). The **PITCH** control is spring-loaded and always returns to the centre position.

Moving **PITCH** will always raise or lower the pitch of the note(s) being played. The range of operation can be set via the menu system, from a semitone to an octave, in semitone steps.

The **MOD** wheel's precise function varies with the Patch loaded; it is used in general to add expression or various elements to a synthesised sound. A common use is to add vibrato to a sound; another is to control the speed of a "virtual" rotary speaker.

It is possible to assign the **MOD** wheel to control any parameter constituting the sound – or a combination of parameters simultaneously. This topic is discussed in more detail elsewhere in the manual. See "What is Legato?" on page 21.

Octave Shift

These two backlit buttons [24] transpose the keyboard up or down one octave each time they are pressed, to a maximum of four octaves. The colour displayed by the buttons indicates the number of octaves shifted: when both LEDs are off (the default state), the lowest note on the keyboard is one octave below Middle C.

Middle C
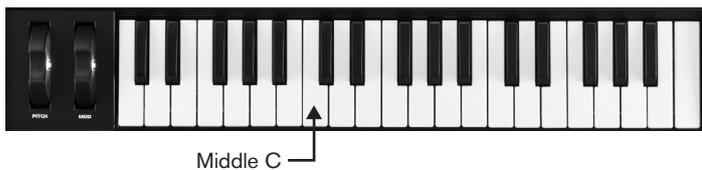
SHIFT	COLOUR
(no buttons pressed)	LEDs off
± 1 octave	Red
± 2 octaves	Magenta
± 3 octaves	Purple
± 4 octaves	Blue

Normal keyboard pitch may be reinstated at any time by pressing both Octave buttons together.

Storing a Patch

Novation have worked hard to create a really useful and great-sounding set of factory Patches, and we are sure that many of them will meet your needs without further modification. However, the scope for altering – or creating completely new - sounds in the MiniNova is almost limitless, and when you have done so, you will probably want to save the sounds for future use.

It is possible to store or write your own patches directly into the MiniNova without using the MiniNova Editor and Librarian software applications. Once any of the parameters of a Patch have been changed, the **SAVE** flag will illuminate in the LCD, to remind you that you are no longer working with an unmodified Patch. To save the modified Patch:

1. Press the **SAVE** button [10], which will display the name that the Patch had when it was first loaded.

NOTE: The Memory Protect function is active by default, so you are likely to see the words

Memory Protect! flash on-screen. It will not be possible to save a modified version of the current patch, without turning this option off. See "Parameter: Memory Protection" on page 12.
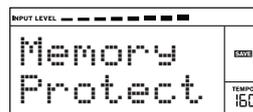
Note that the screen will prompt you for a new name for the modified version (Name?), and that the current name will be offered as a suggestion, with the first character flashing. Use the **DATA** control [6] or the **PATCH** ◀ and ▶ buttons [11] to select a different alphanumeric character.

- Use the **PAGE** ◀ and ▶ buttons [7] to move to the next character, and continue in this manner until the new name has been entered.
- Press **SAVE** again. You will now be prompted to choose the location where the new Patch is to be saved. The location of the original Patch will be offered as the default; if you choose this, the original patch data will be overwritten. Use the **DATA** control [6], or **PATCH** ◀ and ▶ buttons [11] to select a different location. Note that Bank C (128 locations) has been left empty for you to save your own patches; this avoids overwriting any of the original versions.
- Press **SAVE** again, and you will now be prompted to choose the **TYPE** Category that will allow MiniNova's sorting system to retrieve it. Use the **DATA** control to select the most appropriate one, and press **SAVE** again.
- You will finally be prompted to choose the **GENRE** for filing purposes. Use the **DATA** control to select the most appropriate, and press **SAVE** again.
- The screen will now confirm the new Patch with the message Patch Saved. Note that whichever location is chosen for the new Patch, any Patch data already saved to that location will be lost.

NOTE**:** A faster method of managing patches (writing, loading, renaming, reordering etc.) is by using the downloadable MiniNova Librarian. This can be downloaded free of charge from www.novationmusic.com/support.

Updating the MiniNova's Operating System

OS update files will be available from time to time at www.novationmusic.com/support in the form of a MIDI SysEx file. The update procedure requires the MiniNova to be connected via USB to a computer which has first had the necessary USB drivers installed. Full instructions on performing the update will be supplied with the download.

SYNTHESIS TUTORIAL

This section covers the subject of sound generation in more detail and discusses the various basic features available in the MiniNova's sound generation and processing blocks.

It is recommended that this chapter is read carefully if analogue sound synthesis is an unfamiliar subject. Users familiar with this subject can skip this chapter and move on to the next chapter.

To gain an understanding of how a synthesizer generates sound it is helpful to have an appreciation of the components that make up a sound, both musical and non-musical.

The only way that a sound may be detected is by air vibrating the eardrum in a regular, periodic manner. The brain interprets these vibrations (very accurately) into one of an infinite number of different types of sound.

Remarkably, any sound may be described in terms of just three properties, and all sounds always have them. They are:
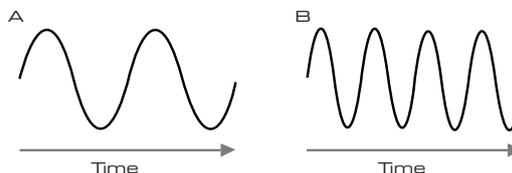
- Pitch
- Tone
- Volume

What makes one sound different from another is the relative magnitudes of the three properties as initially present in the sound, and how the properties change over the duration of the sound.

With a musical synthesizer, we deliberately set out to have precise control over these three properties and, in particular, how they can be changed during the "lifetime" of the sound. The properties are often given different names: Volume may be referred to as Amplitude, Loudness or Level, Pitch as Frequency and Tone as Timbre.

Pitch

As stated, sound is perceived by air vibrating the ear drum. The pitch of the sound is determined by how fast the vibrations are. For an adult human, the slowest vibration perceived as sound is about twenty times a second, which the brain interprets as a bass type sound; the fastest is many thousands of times a second, which the brain interprets as an high treble type sound.

If the number of peaks in the two waveforms (vibrations) are counted, it will be seen that there are exactly twice as many peaks in Wave B as in Wave A. (Wave B is actually an octave higher in pitch than Wave A). It is the number of vibrations in a given period that determines the pitch of a sound. This is the reason that pitch is sometimes referred to as frequency. It is the number of waveform peaks counted during a given period of time which defines the pitch, or frequency.

Tone

Musical sounds consist of several different, related pitches occurring simultaneously. The lowest is referred to as the 'fundamental' pitch and corresponds to the perceived note of the sound. Other pitches making up the sound which are related to the fundamental in simple mathematical ratios are called harmonics. The relative loudness of each harmonic as compared to the loudness of the fundamental determines the overall tone or 'timbre' of the sound.

Consider two instruments such as a harpsichord and a piano playing the same note on the keyboard and at equal volume. Despite having the same volume and pitch, the instruments still sound distinctly different. This is because the different note-making mechanisms of the two instruments generate different sets of harmonics; the harmonics present in a piano sound are different to those found in a harpsichord sound.

Volume

Volume, which is often referred to as the amplitude or loudness of the sound is determined by how large the vibrations are. Very simply, listening to a piano from a metre away would sound louder than if it were fifty metres away.
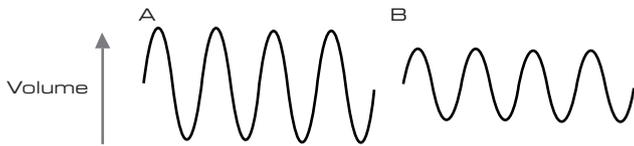
Having shown that just three elements may define any sound, these elements now have to be related to a Musical synthesizer. It is logical that a different section of the Synthesizer 'synthesizes' (or creates) these different elements.

One section of the synthesizer, the Oscillators, provide raw waveform signals which define the pitch of the sound along with its raw harmonic content (tone). These signals are then mixed together in a section called the Mixer, an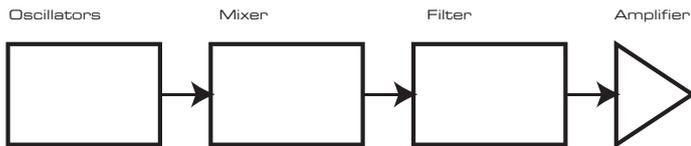d the resulting mixture is then fed into a section called the Filter. This makes further alterations to the tone of the sound, by removing (filtering) or enhancing certain of the harmonics. Lastly, the filtered signal is fed into the Amplifier, which determines the final volume of the sound.

Additional synthesizer sections - LFOs and Envelopes - provide further ways of altering the pitch, tone and volume of a sound by interacting with the Oscillators, Filter and Amplifier, providing changes in the character of the sound which can evolve over time. Because LFOs' and Envelopes' only purpose is to control (modulate) the other synthesizer sections, they are commonly known as 'modulators'.

These various synthesizer sections will now be covered in more detail.

The Oscillators And Mixer

The Oscillator is really the heartbeat of the Synthesizer. It generates an electronic wave (which creates the vibrations when eventually fed to a loudspeaker). This Waveform is produced at a controllable musical pitch, initially determined by the note played on the keyboard or contained in a received MIDI note message. The initial distinctive tone or timbre of the waveform is actually determined by the waveform's shape.

Many years ago, pioneers of musical synthesis discovered that just a few distinctive waveforms contained many of the most useful harmonics for making musical sounds. The names of these waves reflect their actual shape when viewed on an instrument called an Oscilloscope, and these are: Sine waves, Square waves, Sawtooth waves, Triangle waves and Noise.

Each waveform shape (except noise) has a specific set of musically-related harmonics which can be manipulated by further sections of the synthesizer.

The diagrams below show how these waveforms look on an oscilloscope, and illustrate the relative levels of their harmonics. Remember, it is the relative levels of the various harmonics present in a waveform which determine the tone of the final sound.

Sine Waves
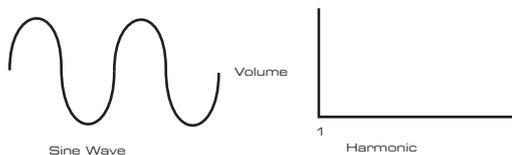
These possess just a single harmonic. A sine waveform produces the "purest" sound because it only has its single pitch (frequency).

Triangle Waves
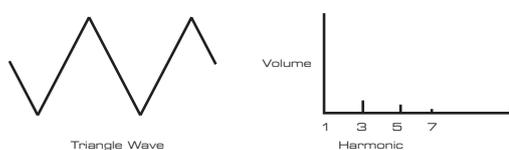
These contain only odd harmonics. The volume of each decreases as the square of its position in the harmonic series. For example, the 5th harmonic has a volume 1/25th of the volume of the fundamental.

Sawtooth Waves
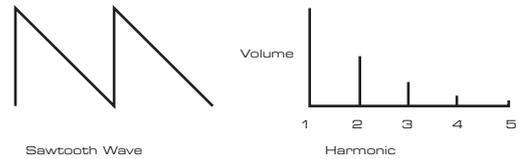
These are rich in harmonics, and containing both even and odd harmonics of the fundamental frequency. The volume of each is inversely proportional to its position in the harmonic series.

Square / Pulse Waves
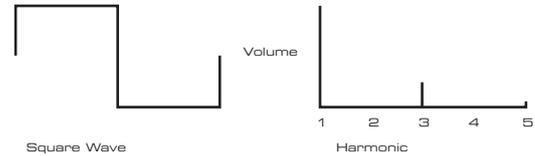
These only have odd harmonics, which are at the same volume as the odd harmonics in a sawtooth wave.
It will be noticed that the square waveform spends equal amounts of time in its 'high' state and its 'low' state. This ratio is known as the 'duty cycle'. A square wave always has a duty cycle of 50% which means it is 'high' for half the cycle and 'low' for the other half.
In the MiniNova, it is possible to adjust the duty cycle of the basic square waveform to produce a waveform which is more 'rectangular' in shape. These are often known as Pulse waveforms. As the waveform becomes more and more rectangular, more even harmonics are introduced and the waveform changes its character, becoming more 'nasal' sounding.

The width of the pulse waveform (the 'Pulse Width') can be altered dynamically by a modulator, which results in the harmonic content of the waveform constantly changing. This can give the waveform a very 'fat' quality when the pulse width is altered at a moderate rate.
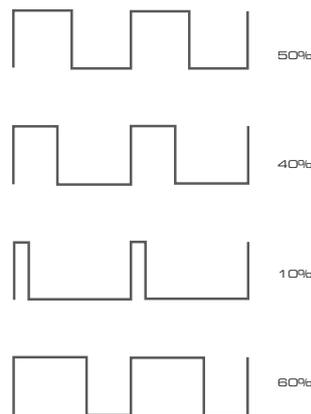
It does not make any difference to how a pulse waveform sounds whether the duty cycle is 40% or 60%, since the waveform is just "inverted" and the harmonic content is exactly the same.

Noise Waves

These are basically random signals, and have no one fundamental frequency (and therefore no pitch property). All frequencies are at the same volume. Because they possess no pitch, noise signals are often useful for creating sound effects and percussion type sounds.

Digital Waveforms

In addition to the traditional types of Oscillator waveforms detailed above, the MiniNova also offers a set of carefully selected, digitally-generated waveforms containing useful harmonic elements normally difficult to produce using traditional Oscillators.

Wavetables

A "wavetable" is essentially a group of digital waveforms. The MiniNova's 36 wavetables each contain 9 separate digital waveforms. The benefit of a wavetable is that consecutive waveforms in the wavetable can be blended. Some of the MiniNova's wavetables contain waveforms with similar harmonic content, while others contain waveforms with greatly differing harmonic content. Wavetables come alive when the 'wavetable index' – the position within the wavetable - is modulated, resulting in a sound that continually changes character, either smoothly or abruptly.

9 Waves make up a wave table
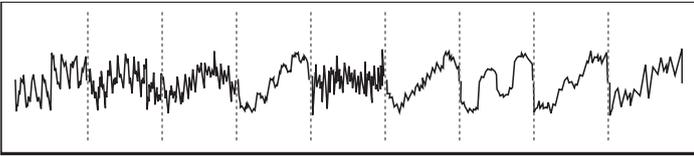
Ring Modulation

A Ring Modulator is a sound generator that takes signals from two of the MiniNova's oscillators and effectively "multiplies" them together. The MiniNova has 2 Ring Modulators, one takes Osc 1 and Osc 3 as inputs, and the other takes Osc 2 and Osc 3. The resulting output depends on the various frequencies and harmonic content present in each of the two oscillator signals, and will consist of a series of sum and difference frequencies as well as the frequencies present in the original signals.
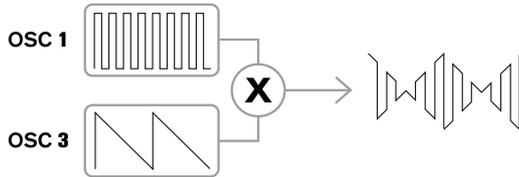
The Mixer

To extend the range of sounds that may be produced, typical analogue synthesizers have more than one Oscillator. By using multiple Oscillators to create a sound, it is possible to achieve very interesting harmonic mixes. It is also possible to slightly detune individual Oscillators against each other, which creates a very warm, 'fat' sound. The MiniNova's Mixer allows mixing of three independent Oscillators, a separate Noise Oscillator and two Ring Modulator sources.
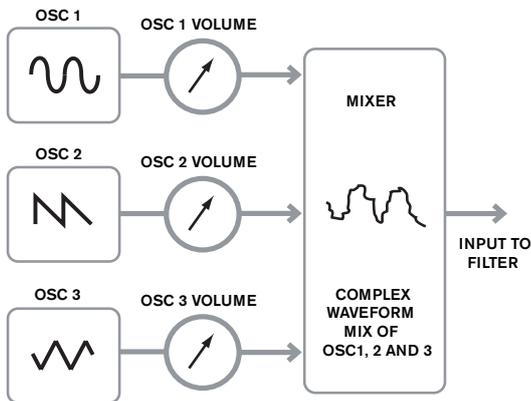
The Filter

The MiniNova is a subtractive music synthesizer. Subtractive implies that part of the sound is subtracted somewhere in the synthesis process.

The Oscillators provide the raw waveforms with plenty of harmonic content and the Filter section subtracts some of the harmonics in a controlled manner.

14 types of Filter are available on the MiniNova, though these are varieties of three basic filter types: Low Pass, Band Pass and High Pass. The type of Filter most commonly found on synthesizers is the Low Pass type. With a Low Pass Filter, a cut-off point (or cut-off frequency) is chosen and any frequencies below the point are passed, and frequencies above are filtered out. The setting of the Filter Frequency parameter dictates the point below which frequencies are removed. This process of removing harmonics from the waveforms has the effect of changing the sound's character or timbre. When the Frequency parameter is at maximum, the filter is completely "open" and no frequencies are removed from the raw Oscillator waveforms.

In practice, there is a gradual (rather than a sudden) reduction in the volume of the harmonics above the cut-off point of a Low Pass Filter. How rapidly these harmonics reduce in volume as frequency increases above the cut-off point is determined by the Filter's slope. The slope is measured in 'volume units per octave'. Since Volume is measured in decibels, this slope is usually quoted as so many decibels per octave (dB/oct). Typical values are 12 dB/oct and 24 dB/oct. The higher the number, the greater the rejection of harmonics above the cut-off point, and the more pronounced the filtering effect.

A further important parameter of the Filter is its Resonance. Frequencies at the cut-off point may be increased in volume by the Filter Resonance control. This is useful for emphasizing certain harmonics of the sound.

As Resonance is increased, a whistling-like quality will be introduced to the sound passing through the filter. When set to very high levels, Resonance actually causes the filter to self-oscillate whenever a signal is being passed through it. The resulting whistling tone being produced is actually a pure sine wave, the pitch of which depends on the setting of the Frequency control (the filter's cut-off point). This resonance-produced sine wave can

actually be used for some sounds as an additional sound source if wished.

The diagram below shows the response of a typical low pass filter. Frequencies above the cut-off point are reduced in volume.
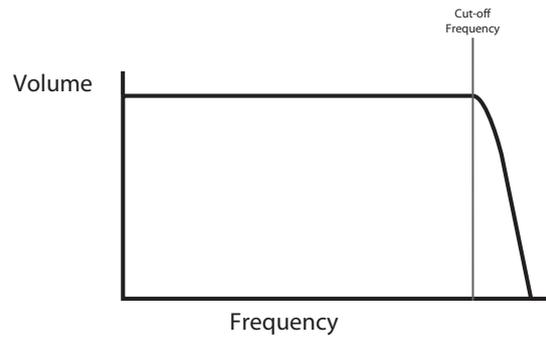
When resonance is added, frequencies at the cut-off point are boosted in volume.
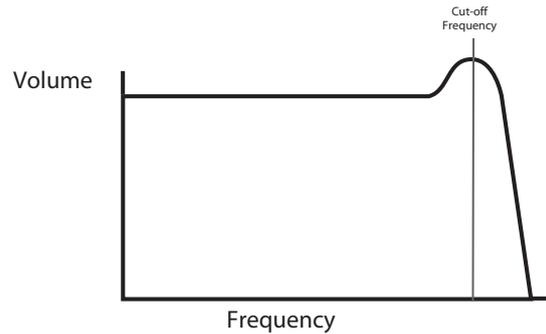
In addition to the traditional Low Pass Filter type, there are also High Pass and Band Pass types. The type of Filter used is selected with the Filter Type parameter.

A High Pass Filter is similar to a Low Pass Filter, but works in the "opposite sense", so that frequencies below the cut-off point are removed. Frequencies above the cut-off point are passed. When the Filter Frequency parameter is set to zero, the filter is completely open and no frequencies are removed from the raw Oscillator waveforms.
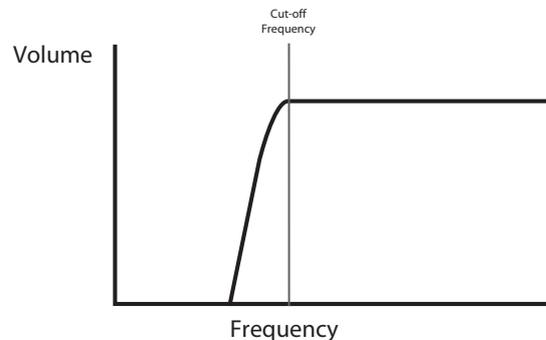
When a Band Pass Filter is used, only a narrow band of frequencies centered around the cut- off point are passed. Frequencies above and below the band are removed. It is not possible to fully open this type of Filter, and allow all frequencies to pass.
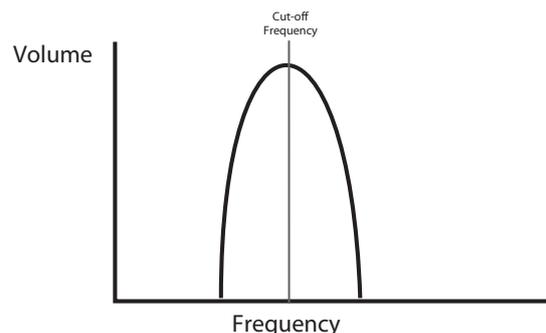
Envelopes And Amplifier

In earlier paragraphs, the synthesis of the pitch and the timbre of a sound were described. The next part of the Synthesis Tutorial describes how the volume of the sound is controlled. The volume of a note created by a musical instrument often varies greatly over the duration

of the note, according to the type of instrument.

For example, a note played on an Organ quickly attains full volume when a key is pressed. It stays at full volume until the key is released, at which point the volume level falls instantly to zero.
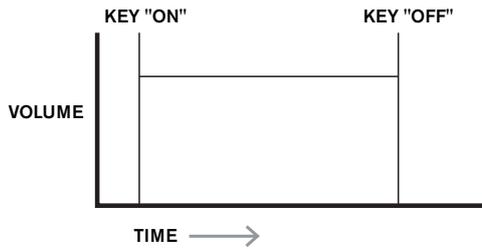
A Piano note quickly attains full volume after a key is pressed, and gradually falls in volume to zero after several seconds, even if the key is held.
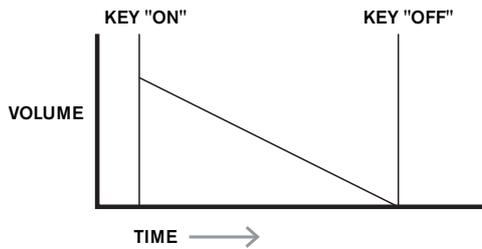
A String Section emulation only attains full volume gradually when a key is pressed. It remains at full volume while the key is held down, but once the key is released, the volume falls zero fairly slowly.

In an analogue synthesizer, changes to a sound's character which occur over the duration of a note are controlled by a section called an Envelope Generator. The MiniNova has 6 Envelope Generators (called Env 1 to Env 6). Env 1 is always related to an Amplifier, which controls the note's amplitude – i.e., the volume of the sound - when the note is played.

Each envelope generator has four main controls which are used to adjust the shape of the envelope.

Attack Time
Adjusts the time it takes after a key is pressed for the volume to climb from zero to full volume. It can be used to create a sound with a slow fade-in.
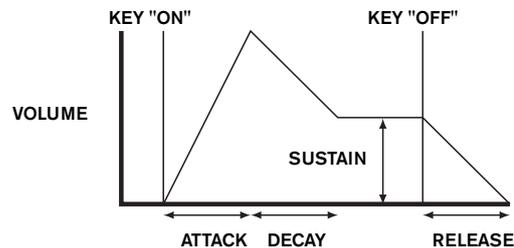
Decay Time
Adjusts the time it takes for the volume to fall from its initial full volume to the level set by the Sustain control while a key is held down.

Sustain Level
This is unlike the other Envelope controls in that it sets a level rather than a period of time. It sets the volume level that the envelope remains at while the key is held down, after the Decay Time has expired.

Release Time
Adjusts the time it takes for the volume to fall from the Sustain level to zero once the key is released. It can be used to create sounds that have a "fade-out" quality.

A typical synthesizer will have one or more envelopes. One envelope is always applied to the amplifier to shape the volume of each note played. Additional envelopes can be used to dynamically alter other sections of the synthesizer during the lifetime of each note.

The MiniNova's second Envelope Generator (Env 2) is used to modify the filter cut-off frequency over the lifetime of a note.

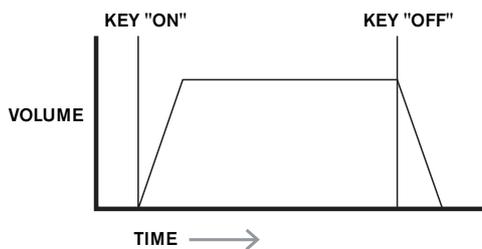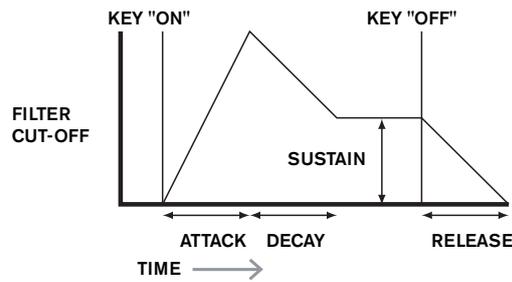In the MiniNova, Envelope Generators 3 to 6 can be used for special purposes, such as modulating the Wavetable index or FX levels.

LFOs

Like the Envelope Generators, the LFO section of a synthesizer is a Modulator. Thus instead of being a part of the sound synthesis itself, it is used to change (or modulate) other sections of the synthesizer. For example, an LFO can be used to alter Oscillator pitch, or Filter cut-off frequency.

Most musical instruments produce sounds that vary over time both in volume and in pitch and timbre. Sometimes these variations can be quite subtle, but still contribute greatly towards characterising the final sound.

Whereas an Envelope is used to control a one-off modulation during the lifetime of a single note, LFOs modulate by using a repeating cyclic waveform or pattern. As discussed earlier, Oscillators produce a constant waveform which can take the shape of a repeating sine wave, triangle wave etc. LFOs produce waveforms in a similar way, but normally at a frequency which is too low to produce a sound that the human ear could perceive. (In fact, LFO stands for Low Frequency Oscillator.)

As with an Envelope, the waveforms generated by the LFOs may be fed to other parts of the synthesizer to create the desired changes over time – or 'movements' - to the sound. The MiniNova has three independent LFOs, which may be used to modulate different synthesizer sections and can run at different speeds.

A typical waveshape for an LFO would be a Triangle wave.
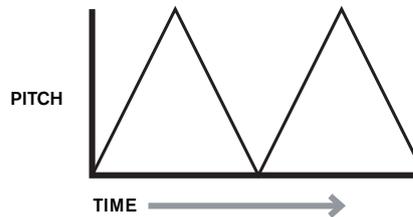
Imagine this very low frequency wave being applied to an Oscillator's pitch. The result is that the pitch of the Oscillator slowly rises and falls above and below its original pitch. This would simulate, for example, a violinist moving a finger up and down the string of the instrument whilst it is being bowed. This subtle up and down movement of pitch is referred to as the 'Vibrato' effect.

Alternatively, if the same LFO signal were to modulate the Filter cut-off frequency instead of the Oscillator pitch, a familiar wobbling effect known as 'wah- wah' would be result. As well as setting up various sections of the synthesizer to be modulated by LFOs, additional Envelopes may also be used as modulators at the same time. Clearly, the more Oscillators, Filters, Envelopes and LFOs there are in a synthesizer, the more powerful it is.

Summary

A synthesizer can be broken down into five main sound generating or sound modifying (modulating) blocks.

1. Oscillators that generate waveforms at a various pitches.
2. A Mixer that mixes the outputs from the Oscillators together.
3. Filters that remove certain harmonics, changing the character or timbre of the sound.
4. An Amplifier controlled by an Envelope generator, which alters the volume of a sound over time when a note is played.
5. LFOs and Envelopes that can be used to modulate any of the above.

Much of the enjoyment to be had with a Synthesizer is with experimenting with the factory preset sounds and creating new ones. There is no substitute for 'hands on' experience. Experiments with adjusting the MiniNova's many parameters will eventually lead to a fuller understanding of how the various controls alter and help shape new sounds. Armed with the knowledge in this chapter, and an understanding of what is actually happening in the machine when tweaks to the knobs and switches are made, the process of creating new and exciting sounds will become easy - Have fun.

SIMPLIFIED MININOVA BLOCK DIAGRAM
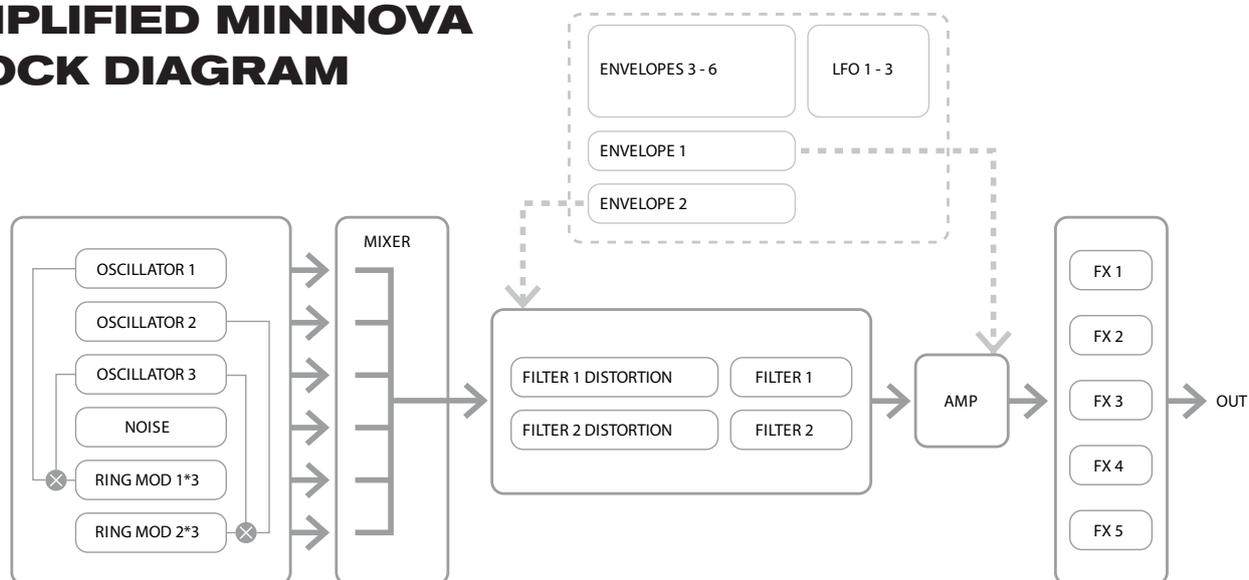
SYNTH MENUS – REFERENCE SECTION

This part of the User Guide gives you a detailed description of every parameter available for adjustment in the MiniNova. As previously explained, all adjustments to patches – other than those made via the controls in the **Perform** and **Pads** sections of the top panel – are made via the MiniNova's comprehensive menu structure. The menus also include "System" or setup options, such as patch dumping, keyboard setup, and so on.

The structure is "context-sensitive" – this means that you will be offered a range of options which is dependent on what it is you are trying to do.

The menu system is always entered by pressing the **MENU** button [8]. The menu system consists of six individual menus:

```
Audio In
Global
Arp
Chord
Edit
Dump
```

Step between the menus with the **PAGE** ◀ and ▶ buttons [7], and press **OK** [9] to enter the desired menu. Use the **PAGE** buttons again to access the parameter you wish to change; use the **DATA** control [6] to alter the parameter value.

The menu system can be exited by pressing the **MENU/BACK** button again; otherwise, it will automatically time-out after a short period, and the screen will revert to display the currently-loaded Patch information.

NOTE: The Default values shown for each parameter apply to `Initial` patches; other factory Patches will have different values as part of the Patch definition.

Top Menu: Audio In

Parameter:	**Input Gain**
Displayed as:	InptGain
Default value:	+20 dB
Range of adjustment:	-10 dB to +65 dB, Off

This control adjusts gain for the Audio input. The gain is displayed directly in dBs. As the gain is increased, the signal at the input will be seen on the bargraph meter at the top of the LCD display. The gain should be adjusted so that the meter peaks two or three segments below the right-most on the loudest passages. The meter also includes an OVER flag; aim to set your signal level so that this never comes on! Note that if InptGain is set to Off, the audio input is inoperative.
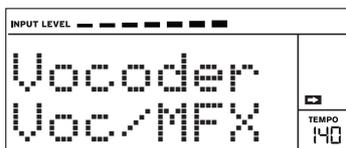
Parameter:	**Input FX level**
Displayed as:	InputFX
Default value:	0
Range of adjustment:	0 – 127

This parameter adjusts the amount of the input signal sent to the FX processor for the currently selected Patch.

Top Menu: Global

Parameter:	**Operating System Version**
Displayed As:	OS Ver

Displays the firmware version currently installed in your MiniNova. You may need to know this in the event of a technical issue arising, or to check whether a newer version is available from the Novation website.

Parameter:	**Memory Protection**
Displayed As:	Protect
Default value:	On
Range of adjustment:	On, Off

This is a safety feature, used to prevent accidental erasure of memories and loss of data. When set to **On**, writing Patches or Global data into memory will be prevented, and a brief warning message (Memory Protect!) shown on the MiniNova's display. It is recommended that Protect is left **On** unless Patches are being edited for storing in memory, or a System Exclusive dump from a computer is to be received.

Parameter:	**Local Control On/Off**
Displayed As:	Local
Default value:	On
Range of adjustment:	On, Off

This control determines whether the MiniNova is to be played from its own keyboard, or to respond to MIDI control from an external device, such as a MIDI sequencer or master keyboard. Set **Local** to **On** to use the keyboard, and to **Off** if you are going to control the synth externally via MIDI or use the MiniNova's keyboard as a master keyboard. When **Off** is selected, a **LOCAL OFF** flag appears in the LCD display.
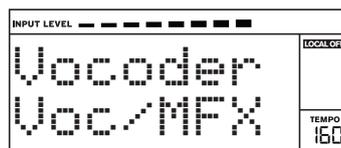
A primary use of Local Control On/Off is to avoid unwanted MIDI loops through external equipment. When set to **Off**, the MiniNova's keyboard and all other controls still transmit MIDI messages from the MIDI OUT port. If any external equipment is set to re-transmit MIDI back to the MiniNova, then the synth will still operate. This will avoid notes sounding twice, a reduction in polyphony or any other unpredictable effects.

Parameter:	**Assign MIDI Channel**
Displayed As:	`MIDI Ch`
Default value:	1
Range of adjustment:	1-16

The MIDI protocol provides 16 channels allowing up to 16 devices to co-exist on a MIDI network, if each is assigned to operate on a different MIDI channel. MIDI Ch lets you set the MiniNova to receive and transmit MIDI data on a particular channel, so that it can interface correctly with external equipment.

Parameter:	**Master Fine Tuning**
Displayed As:	`TuneCent`
Default value:	0
Range of adjustment:	-50 to +50

This control adjusts the frequencies of all the Oscillators by the same small amount, allowing you to fine-tune the whole synth to another instrument if necessary. The increments are cents (1/100 of a semitone), and thus setting the value to ±50 tunes the synth to a quarter-tone midway between two semitones. A setting of zero tunes the keyboard with the A above middle C at 440 Hz – i.e., standard Concert Pitch.

Parameter:	**Key Transposition**
Displayed As:	`Transpse`
Default value:	0
Range of adjustment:	-24 to +24

Transpose is a very useful global setting which "shifts" the whole keyboard one semitone at a time up or down. It differs from oscillator tuning in that it modifies the control data from the keyboard rather than the actual oscillators. Thus setting Transpose to +4 means that you can play with other instruments in the actual key of E major, but only need to play white notes, as if you were playing in C major.

Parameter:	**Pot Pickup (Rotary Perform Control value matching)**
Displayed As:	`PotPckup`
Default value:	Off
Range of adjustment:	On, Off

Operates with the four rotary **PERFORM** controls and the **FILTER** knob to match the parameter value stored in the Patch to the Tweak control's position. If **PotPckup** is set **On**, the rotary control will have no effect until its level matches that stored in the Patch, thus avoiding sudden changes in parameter value. Also, the display shows `->Pickup` until the value is reached. With **PotPckup** set **Off**, the parameter value will change as soon as the control is turned.

Parameter:	**Keyboard Velocity**
Displayed As:	`VelCurve`
Default value:	Normal
Range of adjustment:	Low, Normal, High, Switch, Fixed 4 to 127

Selects the MIDI NoteOn Velocity value which relates the Velocity response of the keys to the force applied to them as they are played. The values 4 to 127 correspond to the actual Velocity values. **Normal** is the default setting, and should be acceptable for most playing styles.

> Use **Low** if playing with heavy touch, and **High** if you have a lighter touch. **Switch** is useful for accenting a change in touch where a lighter touch will output a velocity value of 90 and a heavier touch will output a value of 127. Try different curves to suit your individual playing style(s).

Parameter:	**Footswitch configuration**
Displayed As:	`FootSwth`
Default value:	Auto
Range of adjustment:	Auto, N/Open, N/Closed

A sustain footswitch (pedal) can be connected to the MiniNova via the **SUSTAIN** pedal socket {29}. Ascertain whether your sustain pedal is of the Normally-open or Normally-closed type, and set this parameter to suit. If you are unsure which it is, connect the footswitch with the MiniNova unpowered, and then power it on (without your foot on the pedal!) Provided the default setting of **Auto** is still selected, the polarity will now be correctly sensed.

Parameter:	**Clock Source**
Displayed As:	`ClkSourc`
Default value:	Internal
Range of adjustment:	Internal, USB, MIDI, Auto

The MiniNova uses a master MIDI clock in order to set the tempo (rate) of the arpeggiator and to provide a time base for synchronisation to an overall tempo. This clock may be derived internally or provided by an external device able to transmit MIDI clock. The **ClkSourc** setting determines whether the MiniNova's tempo-synchronised features (Arpeggiator, Chorus Sync, Delay Sync, Gator Sync, LFO Delay Sync, LFO Rate Sync & Pan Rate Sync) will follow the tempo of an external MIDI clock source or follow the tempo set by the **TEMPO** knob [21].

- **Internal** – the MiniNova will synchronise to the internal MIDI clock irrespective of what external MIDI clock sources may be present.
- **USB** – synchronisation will only be set to the external MIDI clock received via the USB connection. If no clock is detected, the tempo "flywheels" to the last-known clock rate.
- **Midi** – synchronisation will only be to an external MIDI clock connected to the MIDI input socket. If no clock is detected, the tempo "flywheels" to the last-known clock rate.
- **Auto** – when no external MIDI clock source is present the MiniNova will default to the internal MIDI clock. Tempo (BPM) will be that set by the **TEMPO** knob. If an external MIDI clock is present the MiniNova will synchronise to it.

When set to any of the external MIDI clock sources the tempo will be at the MIDI Clock rate received from the external source (e.g., a sequencer). Make sure the external sequencer is set to transmit MIDI Clock. If unsure of the procedure, consult the sequencer manual for details.

Most sequencers do not transmit MIDI Clock while they are stopped. Synchronisation of the MiniNova to MIDI Clock will only be possible while the sequencer is actually recording or playing. In the absence of an external clock, the tempo will flywheel and will assume the last known incoming MIDI Clock value.

Parameter:	**Wheel Illumination**
Displayed As:	`WhelLeds`
Default value:	On
Range of adjustment:	On, Off

The **PITCH** and **MOD** wheels [2] are internally illuminated; this setting allows them to be turned on or off.

Parameter:	**MiniNova Power Save**
Displayed As:	`PwrSave`
Default value:	On
Range of adjustment:	On, Off, 10 mins

This is an energy-saving option. Setting **PwrSave** to **On** will make the MiniNova turn off (saving the current settings) when the computer goes into sleep mode. This only applies if it is powered via the USB connection. If set to **10 mins**, the keyboard will turn off after that period regardless of how it is powered. In either case, pressing any key will restore power. If set to **Off**, the keyboard will remain on.

Top Menu: Arp

Parameter:	**Arpeggiator Rate Sync**
Displayed As:	`ArpSync`
Default value:	16th.
Range of adjustment:	See Sync Values table "Sync Values Table" on page 34

This parameter effectively determines the beat of the arp sequence, based on the current tempo. See "Parameter: Clock Source" on page 13.

Parameter:	**Arpeggiator Gate Time**
Displayed As:	`Arp Gate`
Default value:	64
Range of adjustment:	1 to 127

This parameter sets the basic duration of the notes played by the Arpeggiator (though this will be further amended by both the **Arp Pttn** and **Arp Sync** settings). The lower the parameter value, the shorter the duration of the note played. At its maximum value, one note in the sequence is immediately followed by the next without a gap. At the default value of 64, the note duration is exactly half the beat interval (based on the current tempo), and each note is followed by a rest of equal length.

Parameter:	**Arpeggiator Mode**
Displayed As:	`Arp Mode`
Default value:	Up
Range of adjustment:	See **Arp Mode table** "Arp Mode Table" on page 38

When enabled, the Arpeggiator will play all notes held down in a sequence which is determined by the **Arp Mode** parameter. The third column of the table describes the nature of the sequence in each case.

Parameter:	**Arpeggiator Octaves**
Displayed As:	`Arp Octv`
Default value:	1
Range of adjustment:	1 to 4

This setting adds upper octaves to the arp sequence. If **Arp Octv** is set to 2, the sequence is played as normal, then immediately played again an octave higher. Higher values of **Arp Octv** extend this process by adding additional higher octaves. **Arp Octv** values greater than 1 have the effect of doubling, tripling, etc., the length of the sequence. The additional notes added duplicate the complete original sequence, but octave-shifted. Thus a four-note sequence played with **Arp Octv** set to 1, will consist of eight notes when **Arp Octv** is set to 2.

Parameter:	**Arpeggiator Pattern**
Displayed As:	Arp Pttn
Default value:	Arp Edit
Range of adjustment:	Arp Edit, UN pat 2 to 33

On the MiniNova, Arpeggiator sequences may be user-configured to be up to eight notes in length by setting **Arp Pttn** to **Arp Edit**. In this setting you can edit the Arp sequence using the eight Pads in **ARPEGGIATE** mode. It is only possible to modify an Arp sequence with the Pads when **Arp Pttn** is set to **Arp Edit**.

UN pat 2 to 33 are pre-assigned Arp patterns of various lengths (greater than eight notes) and timings, and are derived from the UltraNova. These are not modifiable.

> t You should spend some time experimenting with different combinations of **Arp Mode** and **Arp Pttn**. Some patterns work better in certain Modes.

Parameter:	**Arpeggiator Length**
Displayed As:	ArpLen
Default value:	8
Range of adjustment:	1 to 8

This parameter is only available when **Arp Pttn** is set to **Arp Edit**. This parameter represents the number of steps making up the sequence.

Parameter:	**Arpeggiator Swing**
Displayed As:	ArpSwing
Default value:	50
Range of adjustment:	1 to 100

This parameter is only available when **Arp Pttn** is set to **Arp Edit**. If this parameter is set to something other than its default value of 50, some further interesting rhythmic effects can be obtained. Higher values of Swing lengthen the interval between odd and even notes, while the even-to-odd intervals are correspondingly shortened. Lower values have the opposite effect. This is an effect which is easier to experiment with than describe!

Top Menu: Chord

The MiniNova's Chorder is a very useful feature which permits you to play chords containing up to ten notes by pressing a single key. The resulting chord uses the lowest note played as its root; all the other notes in the chord will be above the root.

Parameter:	**Chord Mode**
Displayed As:	ChrdMode
Default value:	Off
Range of adjustment:	On, Off

Turns Chord mode On or Off.

Parameter:	**Chord Transposition**
Displayed As:	ChrdTrns
Default value:	0
Range of adjustment:	-11 to +11

The transposition control is calibrated in semitone intervals, and the pitch of the chord can be shifted up to 11 semitones, either up or down.

Parameter:	**Save Chord**
Displayed As:	SaveChrd

To save a chord, first set **ChrdMode** to **On** and then select this menu option (**SaveChrd**). The bottom line of the display will show OK?; press the **OK** button [9]. The display will change to PlayChrd; you should now play the chord you want to capture; you can play it in any key or inversion you wish. Then press the the **OK** button. After a short delay, the display will confirm the action by with Chord SAVED!
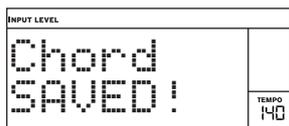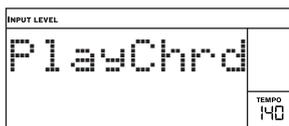
> i Note that the Arpeggiator precedes the Chorder in the MiniNova's synth engine. This has the consequence that if both the Arpeggiator and Chorder are in use, the entire chord resulting from each key press will be arpeggiated.

Top Menu: Edit

This menu is where you can modify the sound of a Patch, or create a new one from first principles. The Edit menu is divided further submenus as follows:

Tweaks
Osc
Mixer
Filter
Voice
Env
LFO
ModMatrx
Effects
Vox Tune
Vocoder

Edit menu - Submenu 1: Tweaks

Parameter:	**Tweak Number**
Displayed As:	Tweak n (where n is 1 to 8)
Default value:	*(unassigned)*
Range of adjustment:	See Tweak Parameters table on page 36.

Use the **PAGE** ◀ and ▶ buttons [7] to select which of the eight Tweak controls you wish to configure, and the **DATA** control [6] to select the parameter which the selected Tweak control will vary.

Edit menu - Submenu 2: Osc

With this submenu, it is first necessary to select the oscillator whose parameters are to be adjusted. This selection is made with the **PAGE** ◀ and ▶ buttons [7].

Displayed As:	Osc n (where n is 1 to 3)
Default value:	Osc 1
Range of adjustment:	Osc 1 to 3, OscComn

The MiniNova has three identical oscillators and a noise source; these are the synth's sound generators.

Per-Oscillator parameters

In the following parameter descriptions, the text refers to Oscillator 1; however, it applies equally to whichever oscillator is selected. A separate set of parameters applicable to all three oscillators is available when the Oscillator submenu is selected to **OscComn** (see "Common Oscillator parameters" on page 15).

Parameter:	**Coarse tuning**
Displayed As:	O1Semi
Default value:	0
Range of adjustment:	-64 to +63

This parameter sets the basic per-oscillator tuning. Incrementing its value by 1 shifts the pitch of every note on the keyboard up by one semitone *for the selected oscillator only*, thus setting it to +12 effectively shifts the oscillator tuning up one octave. Negative values detune in the same manner. See also "Parameter: Key Transposition" on page 13.

Parameter:	**Fine tuning**
Displayed As:	O1Cents
Default value:	0
Range of adjustment:	-50 to +50

This parameter lets you make finer adjustments to the tuning. The increments are cents (1/100 of a semitone), and thus setting the value to ±50 tunes the oscillator to a quartertone midway between two semitones.

Parameter:	**Virtual Oscillator Sync**
Displayed As:	O1VSync
Default value:	0
Range of adjustment:	0 to 127

Oscillator Sync is a technique of using an additional "virtual" oscillator to add harmonics to the first, by using the virtual oscillator's waveform to retrigger that of the first. This technique produces an interesting range of sonic effects. The nature of the resulting sound varies as the parameter value is altered because the virtual oscillator frequency increases as a multiple of the main oscillator frequency as the parameter value increases. When the **Vsync** value is a multiple of 16, the virtual oscillator frequency is a musical harmonic of the main oscillator frequency. The overall effect is a transposition of the oscillator that moves up the harmonic series, with values in between multiples of 16 producing more discordant effects.

VSync = 0

VSync = 5

VSync = 16

O1VSync can be also adjusted directly from Row 6 of the **PERFORM** section of the control panel with Tweak Control **RC1**.

O2VSync can be also adjusted directly from Row 6 of the **PERFORM** section of the control panel with Tweak Control **RC3**.

To get the best out of **Vsync**, try modulating it using an LFO. Alternatively, select Row 6 in the **PERFORM** section, and vary it while playing with Tweak Control **RC1**.

Parameter:	**Oscillator Waveform**
Displayed As:	O1Wave
Default value:	Sawtooth
Range of adjustment:	See **Waveforms** table on page 34.

This selects the oscillator's waveform from a range of 72 options. As well as analogue synth type waveforms like sine, square, sawtooth, pulse and 9 ratios of sawtooth/pulse mix, there are various digital waveforms and 36 wavetables consisting of nine individual waveforms per wavetable, plus two audio input sources.

Two audio sources are included in the Waveform table; although the MiniNova only has a single audio input (**AudInL/M**), **AudioInR** is included for compatibility with UltraNova Patches.

If audio input sources are selected, then any additional oscillator parameters will have no effect on the sound. The audio input will be used as the source for subsequent manipulation (e.g., filters, modulation, etc).

When the external input is selected as an oscillator source it is really being selected in place of that oscillator and fed through the synth's signal path from this point. To hear the audio input when selected as an oscillator source, a note must be played on the keyboard.

It is possible to create a MIDI gate effect on vocals using audio inputs as the source.

Parameter:	**Pulse Width/Wave Table Index**
Displayed As:	O1PW/Idx
Default value:	0
Range of adjustment:	-64 to 63

This control has two functions, depending on the waveform selected by **O1Wave**. With pulse waveforms, it varies the pulse width of the oscillator output. This basic effect can most easily be heard by adjusting this parameter with **O1Wave** set to **PW**; you will note how the harmonic content varies and at high settings the sound becomes quite thin and metallic. A pulse wave is essentially an asymmetric square wave; when set to zero, the waveform is a normal square wave. (See page 9.) This parameter has a different function if the oscillator waveform is set to be one of the 36 Wave Tables (see **O1Wave** above). Each Wave Table consists of nine related waveforms, and the setting of **O1PW/ Idx** determines which is in use. The total parameter value range of 128 is divided into 9 (approximately) equal segments of 14 value units, so setting the value to anything between -64 and -50 will generate the first of the 9 waveforms, -49 to -35 the second, and so on. See also the Wave Table Interpolation parameter (**O1WTInt**), which can be used to introduce further variation in the way wavetables are used.

Parameter:	**Hardness**
Displayed As:	O1Hard
Default value:	127
Range of adjustment:	0 to 127

The Hardness parameter modifies the harmonic content of the waveform, reducing the level of the upper harmonics as the value is decreased. Its effect is akin to that of a lowpass filter, but operates at oscillator level. You will note it has no effect on a sine waveform, as this is the one waveform with no harmonics.

Parameter:	**Density**
Displayed As:	O1Dense
Default value:	0
Range of adjustment:	0 to 127

The density parameter effectively adds copies of the oscillator waveform to itself. Up to eight additional virtual oscillators are used for this, depending on the value of the parameter. This produces a "thicker" sound at low to medium values, but if the virtual oscillators are detuned slightly (see **O1DnsDtn** below), a more interesting effect is obtained.

O1Dense can be also adjusted directly from Row 6 of the **PERFORM** section of the control panel with Tweak Control **RC2**.

O2Dense can be also adjusted directly from Row 6 of the **PERFORM** section of the control panel with Tweak Control **RC4**.

Parameter:	**Density Detuning**
Displayed As:	O1DnsDtn
Default value:	0
Range of adjustment:	0 to 127

This parameter should be used in conjunction with the **Density** control. It detunes the virtual density oscillators, and you will notice not only a thicker sound, but the effect of beating as well.

The Density and Density Detune parameters can be used to "thicken" the sound, and simulate the effect of adding additional Voices. The Unison and Unison Detune parameters in the Voice Menu can be used to create a very similar effect, but using Density and Density Detune have the advantage of not needing to use additional Voices, which are finite in number.

Parameter:	**Pitch Wheel Range**
Displayed As:	O1PtchWh
Default value:	+12
Range of adjustment:	-12 to +12

The pitch wheel can vary the oscillator pitch by up to an octave, up or down. The units are in semitones, so with a value of +12, moving the pitchwheel up increases the pitch of the notes being played by one octave, and moving it down takes them down an octave. Setting the parameter to a negative value has the effect of reversing the operating sense of the pitch wheel. You will find that many of the factory patches have this parameter set to +2, allowing a pitch wheel range of ±1 tone. It is worth noting that (like all per-oscillator parameters) the value can be set independently for each oscillator.

Parameter:	**Wave Table Interpolation**
Displayed As:	O1WTInt
Default value:	127
Range of adjustment:	0 to 127

This parameter sets how smooth the transition is between adjacent waveforms in the same wavetable. A value of 127 will create a very smooth transition, with the adjacent waveforms blending together. With a value of zero the transitions will be abrupt and obvious. With a high **O1WInt** value set, it is possible to retain a mix of adjacent waveforms if the modulation value remains fixed. When modulating the wavetable Index (via LFO, etc.), the wavetable interpolation parameter sets how smooth (or not!) the transition is.

Common Oscillator parameters

The remaining parameters in the Oscillator Menu are common to all 3 oscillators. They are available when **Oscillator Number** is set to **OscComn**.

Parameter:	**Vibrato Depth**
Displayed As:	ModVib
Default value:	0
Range of adjustment:	0 to 127

Adding vibrato to an oscillator modulates (or varies) the pitch of the note cyclically, adding a "wobble" to the tone. This parameter determines the vibrato depth, and hence how obvious the "wobble" is. The mod wheel is used to apply vibrato, with the **ModVib** parameter value representing the maximum depth of vibrato that can be obtained with the mod wheel in its fully 'up' position. On the MiniNova, **VibMod** and **MVibRate** are common parameters that affect all oscillators and do not require the use of the LFO section.

Parameter:	**Vibrato Rate**
Displayed As:	MVibRate
Default value:	65
Range of adjustment:	0 to 127

This parameter sets the rate (or frequency) of the vibrato from very slow (value=0) to very fast (value=127).

Parameter:	**Oscillator Drift**
Displayed As:	OscDrift
Default value:	0
Range of adjustment:	0 to 127

When the three oscillators are set to the same tuning, their waveforms are perfectly synchronised. Old analogue synthesizers were unable to stay perfectly in tune, and Oscillator Drift 'emulates' this imperfection by applying a controlled amount of detuning so that the oscillators are slightly out of tune with each other. This adds a "fuller" character to the sound.

Parameter:	**Oscillator Phase**
Displayed As:	OscPhase
Default value:	0deg
Range of adjustment:	Free, 0deg to 357deg

This adjusts the point in the waveform at which the oscillators start, and is adjustable in 3 degree increments over one whole waveform cycle (360°). The effect of this is to add a slight "click" or "edge" to the start of the note, as the instantaneous output voltage when the key is pressed is not zero. Setting the parameter to 90deg or 269deg produces the most obvious effect. With the parameter set to 0deg, the oscillators always start precisely in step. If Free is set, the phase relationship of the waveforms is unrelated to when a key is pressed.

Parameter:	**Single Fixed Note**
Displayed As:	FixNote
Default value:	Off
Range of adjustment:	Off, C#-2 to G8

Some sounds need not be chromatically-dependent. Examples would be certain percussion sounds (e.g., bass drums), and sound effects, such as a laser gun. It is possible to assign a fixed note to a patch, such that playing any key on the keyboard generates the same sound. The pitch on which the sound is based may be any semitone note in a range of over ten octaves. With the parameter set Off, the keyboard behaves as normal. With it set to any other value, every key plays the sound at the pitch corresponding to the value.

Parameter:	**Noise Source Type**
Displayed As:	NoiseTyp
Default value:	White
Range of adjustment:	White, High, Band, HiBand

In addition to the three main oscillators, the MiniNova also has a noise generator. White noise is defined as a signal with "equal power at all frequencies", and is a familiar "hissing" sound. Restricting the bandwidth of the noise generator alters the characteristic of the "hiss", and the other three options for this parameter apply such filtering. Note that the noise generator has its own input to the mixer, and in order to hear it in isolation, its input will need to be turned up and the oscillator inputs turned down. (See "Parameter: Noise Source Level" on page 16.)

Edit menu - Submenu 3: Mixer

The outputs of the three oscillators and the noise source are passed to a simple audio mixer, where their individual contributions to the overall sound output can be adjusted. Most of the factory Patches use either two, or all three oscillators, but with their outputs summed in various combinations of levels. A total of 6 inputs and two FX sends are available for adjustment.

> As with any other audio mixer, don't be tempted to turn all the inputs up. The mixer should be used to balance sounds. If multiple sources are in use, then each input setting should be about halfway – about 64 or so, and the more inputs you are using, the more careful you need to be. If you get this wrong, you risk internal signal clipping, which will sound extremely unpleasant.

Parameter:	**Oscillator 1 Level**
Displayed As:	O1Level
Default value:	127
Range of adjustment:	0 to 127

This parameter sets the amount of Oscillator 1's signal present in the overall sound.

Parameter:	**Oscillator 2 Level**
Displayed As:	O2Level
Default value:	0
Range of adjustment:	0 to 127

This parameter sets the initial amount of Oscillator 2's signal present in the overall sound.

Parameter:	**Oscillator 3 Level**
Displayed As:	O3Level
Default value:	0
Range of adjustment:	0 to 127

This parameter sets the initial amount of Oscillator 3's signal present in the overall sound.

Parameter:	**Ring Modulator Level (Oscs. 1 * 3)**
Displayed As:	RM1*3Lvl
Default value:	0
Range of adjustment:	0 to 127

In its simplest form, a Ring Modulator is a processing block with two inputs and one output, which effectively "multiplies" the two input signals together. Depending on the relative frequencies and harmonic content of the two inputs, the resulting output will contain a series of sum and difference frequencies as well as the fundamentals. The MiniNova has two Ring Modulators; both use Oscillator 3 as one input, one combines this with Oscillator 1, the other with Oscillator 2. The Ring Modulator outputs are available as two additional inputs to the mixer, controlled by **RM1*3Lvl** and **RM2*3Lvl**. The parameter controlled by **RM1*3Lvl** sets the amount of the Osc. 1 * 3 Ring Modulator output present in the overall sound.

> Try the following settings to get a good idea of what a Ring Modulator sounds like. In the Mixer Menu, turn down the levels of Oscs 1, 2 & 3 and turn up **RM1*3Lvl**. Then go to the Oscillator Menu. Set Osc3 to an interval of +5, +7 or +12 semitones above Osc1 and the sound will be harmonically pleasing.
> Changing the pitch of Osc 1 to other semitone values creates discordant, but interesting sounds. **O1 Cents** can be varied to introduce a 'beating' effect.

Parameter:	**Ring Modulator Level (Oscs. 2 * 3)**
Displayed As:	RM2*3Lvl
Default value:	0
Range of adjustment:	0 to 127

The parameter controlled by RM2*3Lvl sets the amount of the Osc. 2 * 3 Ring Modulator output present in the overall sound.

Parameter:	**Noise Source Level**
Displayed As:	NoiseLvl
Default value:	0
Range of adjustment:	0 to 127

This parameter sets the amount of Noise present in the overall sound.

Parameter:	**Pre-FX level send**
Displayed As:	PreFXLvl
Default value:	0 dB
Range of adjustment:	-12 dB to +18 dB

The summed mixer inputs are routed through the FX block (even if no effects are active) at a level determined by PreFXLvl. This control should be adjusted with care to avoid overloading the FX processing.

Parameter:	**Post-FX level send**
Displayed As:	PostFXLv
Default value:	0 dB
Range of adjustment:	-12 dB to +18 dB

This parameter adjusts the level returned from the output of the FX processor. Thus both **PreFXLvl** and **PostFXLv** will alter the signal level even when all FX slots in the FX block are bypassed.

> **PreFXLvl** and **PostFXLvl** are critical controls and incorrect adjustment can produce clipping in the FX processing section and elsewhere. It is always a good idea to set up the FX parameters you think you need first (see "What is Legato?" on page 21), and then increase these two parameters carefully until you get the amount of FX you're after.

Edit menu - Submenu 4: Filter

With this submenu, it is first necessary to select the filter whose parameters are to be adjusted.

Displayed As:	Filter n (where n is 1 or 2)
Default value:	Filter 1
Range of adjustment:	Filter 1, Filter 2, FiltrCmn

The MiniNova has two identical filter sections, which modify the harmonic content of the oscillators' outputs. They can be thought of as elaborate tone controls, with the additional ability of being dynamically controllable by other parts of the synth. A total of 8 parameters per filter are available for adjustment. Note that some parameters are common to both filters (found in the **FiltrCmn** submenu). It is possible to use the two filter blocks together, placing them in various series/parallel configurations, by adjustment of the common parameter **FRouting**.

Per-Filter parameters

Filter 1 is used as the example in the descriptions which follow, but the two are identical in operation, except where indicated.

Parameter: **Filter frequency**
Displayed As: `F1Freq`
Default value: 127
Range of adjustment: 0 to 127

This parameter sets the frequency at which the filter type selected by **F1Type** operates. In the case of hi-pass or low-pass filters, it is the "cut-off" frequency; for band-pass filters, it is the "centre" frequency. Sweeping the filter manually will impose a "hard-to-soft" characteristic on almost any sound.

> **i** If Filter Frequency Link is set **On** (see **FreqLink** below), **F2Freq** assumes a different function:
> Parameter: **Filter 2 frequency offset**
> Displayed As: `Fq1<>Fq2`
> Default value: +63
> Range of adjustment: -64 to +63
> See "Parameter: Filter Frequency Link" on page 19 for more information.

Parameter: **Filter resonance**
Displayed As: `F1Res`
Default value: 0
Range of adjustment: 0 to 127

This parameter adds gain to the signal in a narrow band of frequencies around the frequency set by **F1Freq**. It can accentuate the swept-filter effect considerably. Increasing the resonance parameter is very good for enhancing modulation of the cut-off frequency, creating a very edgy sound. Increasing Resonance also accentuates the action of the Filter Frequency parameter, so that as the **FILTER** knob [14] is moved, a more pronounced effect is obtained.

> **P** `F1Res` can be also adjusted directly from Row 3 of the **PERFORM** section of the control panel with Tweak Control **RC1**.

> **i** If Filter Resonance Link is set **On** (see **ResLink** page 19), the values of filter resonance for Filters 1 and 2 become equal and are varied by either control.
> Parameter: **Filter 1 & 2 resonance**
> Displayed As: `F1&F2Res`
> Default value: not applicable
> Range of adjustment: 0 to 127

Parameter: **Filter control by Envelope 2**
Displayed As: `F1Env2`
Default value: 0
Range of adjustment: 0 to 127

The filter's action may be triggered by Envelope Generator 2. Envelope 2's own menu provides comprehensive control over precisely how this shape of the envelope is derived, see "Filter Envelope" on page 22. F1Env2 lets you control the "depth" and "direction" of this external control; the higher the value, the greater the range of frequencies over which the filter will sweep. Positive and negative values make the filter sweep in opposite directions, but the audible result of this will be further modified by the filter type in use.

> **P** `F1Env2` can be also adjusted directly from Row 4 of the **PERFORM** section of the control panel with Tweak Control **RC4**.

Parameter: **Filter tracking**
Displayed As: `F1Track`
Default value: 127
Range of adjustment: 0 to 127

The pitch of the note played can be made to alter the cut-off frequency of the filter. At the maximum value (127), this frequency moves in semitone steps with the notes played on the keyboard – i.e., the filter tracks the pitch changes in a 1:1 ratio (e.g., when playing two notes an octave apart, the filter cut off frequency will also change by one octave). At minimum setting (value 0), the filter frequency remains constant, whatever note(s) are played on the keyboard.

> **P** `F1Track` can be also adjusted directly from Row 3 of the **PERFORM** section of the control panel with Tweak Control **RC2**.

Parameter: **Filter type**
Displayed As: `F1Type`
Default value: LP24
Range of adjustment: See **Filter Table** on page 38

The MiniNova filter sections offer 14 different types of filter: four hi-pass and four low-pass (with different slopes), and 6 band-pass filters of various types. Each filter type differentiates between frequency bands in a different way, rejecting some frequencies and passing others, and thus each imposes a subtly different character on the sound.

> **P** `F1Type` can be also adjusted directly from Row 3 of the **PERFORM** section of the control panel with Tweak Control **RC3**.

Parameter: **Drive Amount**
Displayed As: `F1DAmnt`
Default value: 0
Range of adjustment: 0 to 127

The filter section includes a dedicated drive (or distortion) generator; this parameter adjusts the degree of distortion treatment applied to the signal. The basic 'type' of drive added is set by **F1DType** (see below). The drive is added pre the filter (but see below).

> **P** `F1DAmnt` can be also adjusted directly from Row 3 of the **PERFORM** section of the control panel with Tweak Control **RC4**.

> **t** Filter Drive is always added before the filter, and therefore the filter frequency affects the amount of drive you hear. If you want to filter your sound before it is treated by the drive processor, try settings similar to the following:
>
> | PARAMETER | IN MENU | VALUE |
> |-----------|---------|-------|
> | FRouting | FiltrCmn | Series |
> | FBalance | FiltrCmn | 63 |
> | F1DAmnt | Filter 1 | 0 |
> | F2DAmnt | Filter 2 | As required |

Parameter: **Drive Type**
Displayed As: `F1DType`
Default value: Diode
Range of adjustment: Diode, Valve, Clipper, XOver, Rectify, BitsDown, RateDown

The drive processor for each filter is located immediately before the filter section itself. The type of drive (or distortion) generated can be selected with the **F1DType** parameter.

Parameter: **Filter Q Normalisation**
Displayed As: `F1QNorm`
Default value: 64
Range of adjustment: 0 to 127

This parameter alters the bandwidth of the peak created by the resonance control **F1Res**. The value of **F1Res** has to be set to something other than zero for this parameter to have any effect. This feature enables the Filter section to emulate many of the filter responses found on various classic analogue and digital synths.
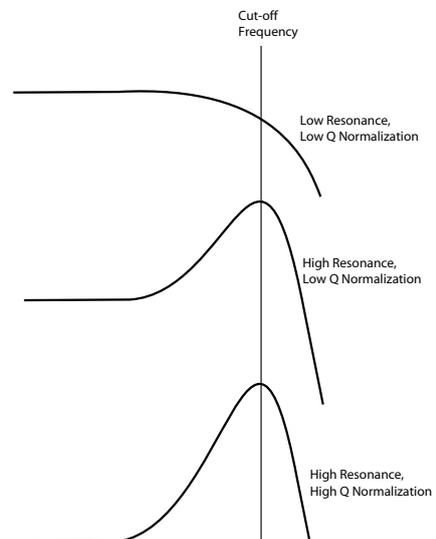
Common Filter parameters

With **Filter number** set to **FiltrCmn**, the displayed parameters in the Filter Menu are common to both filters.

Parameter: **Filter balance**
Displayed As: `FBalance`
Default value: -64
Range of adjustment: -64 to +63

MiniNova's two filter sections may be used simultaneously, but configured in different ways (see **FRouting** below). Low-pass and band-pass filters could be combined in parallel to create speech-like sounds (see page 19). For configurations using both filters, **FBalance** lets you mix the outputs of the two filter sections together in whatever combination you want. The minimum parameter value of -64 represents maximum output from Filter 1 and no output from Filter 2, and the maximum value of +63 represents maximum output from Filter 2 and no output from Filter 1. With a value of 0, the outputs of the two filter sections are mixed in equal proportion.

Parameter:	**Filter Routing**
Displayed As:	FRouting
Default value:	Parallel
Range of adjustment:	Bypass, Single, Series, Parallel, Paral2, Drum

MiniNova provides five possible combinations of the two filter blocks, plus bypass. Single mode uses Filter 1 only, the other modes interconnect the two filter sections in various ways.

Bypass - No filiters in circuit

Single - Filter 1 only

Series - Filter 1 feeds Filter 2, but output is still derived from Filter Balance control

Parallel - Filter sections are driven with the same input signal, and their output mix is adjusted by the Filter Balance parameter.

Parallel 2 As Parallel Mode, but Filter 1 is driven by Osc 3 and Noise source, remaining sources feed Filter 2.

Drum - As Parallel Mode 2, but Filter 1's output is added to Filter 2's input signals.

Note that Paral2 and Drum modes differ in an important respect from the others in that the Filter 1 and Filter 2 are fed from different sources. This allows the noise source and Osc 3 to be filtered in a different way from Oscillators 1 and 2 and the Ring Modulator Outputs, an important requirement when creating certain percussive sounds.

Two examples of Filter routings…

…a Notch Filter:

F1Type:	LP
F2Type:	HP
F1Routing:	Parallel

…a wideband bandpass filter

F1Type:	HP
F2Type:	LP
F1Routing:	Series

Parameter:	**Filter Frequency Link**
Displayed As:	FreqLink
Default value:	Off
Range of adjustment:	Off or On

Setting **FreqLink** to **On** creates a relationship between the frequencies of the two Filter sections, and re-assigns the function of **F2Freq** for Filter 2 from Frequency to Frequency Offset (see **F1Freq**, above). Filter 2's offset is relative to Filter 1's frequency.
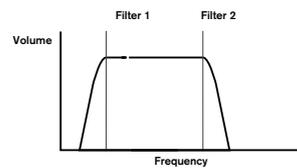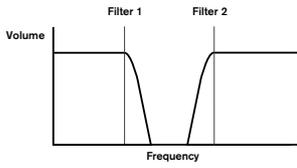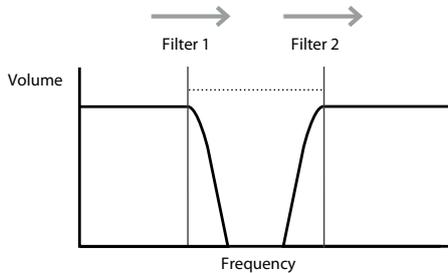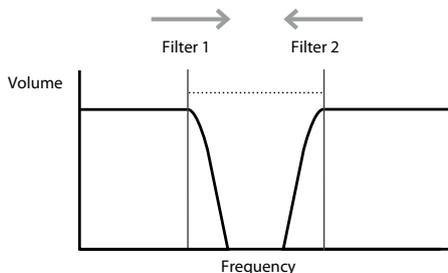

An increase in Filter 1's frequency will also increase Filter 2's frequency
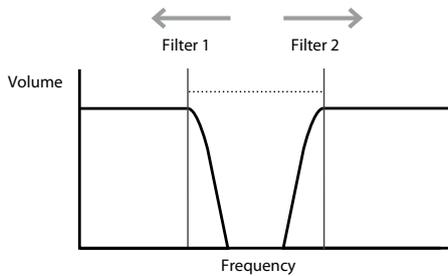
An increase in Filter 2's frequency will decrease Filter 1's frequency

A decrease in Filter 2's frequency will increase Filter 1's frequency

Parameter:	**Resonance Link**
Displayed As:	ResLink
Default value:	Off
Range of adjustment:	Off or On

Setting **ResLink** to **On** applies the same Resonance parameter value to both Filter 1 and Filter 2. The Filter Resonance control (**F1Res**) affects both filters, regardless of which filter is currently selected for adjustment.

Edit menu - Submenu 5: Voice

The MiniNova is a multi-voice, polyphonic synthesizer, which basically means you can play chords on the keyboard, and every note you hold down will sound. Each note is termed a 'voice', and the MiniNova's DSP engine is sufficiently powerful to ensure that you will always run out of fingers before you run out of voices! However, if you are controlling the MiniNova from a MIDI sequencer, it is theoretically possible to run out (there are a maximum of 18 voices internally). Although this is likely to happen only rarely, users may occasionally observe this phenomenon, which is termed 'voice stealing'.

The alternative to polyphonic voicing is mono. With mono voicing, only one note sounds at a time; pressing a second key while holding the first down will cancel the first and play the second – and so on. The last note played is always the only one that you hear. All the early synths were mono, and if you are trying to emulate a 1970s analogue synth, you may wish to set the voicing to mono as the mode imposes a certain restriction on playing style that will add to authenticity.

In addition to selecting polyphonic or mono voicing, the Voice menu also lets you set the portamento and other related voicing parameters.

Parameter:	**Unison Voices**
Displayed As:	Unison
Default value:	Off
Range of adjustment:	Off, 2, 3, 4

Unison can be used to "thicken" the sound by assigning additional voices (up to 4 in total) for each note. Be aware that the "reservoir" of voices is finite and with multiple voices assigned, polyphony is accordingly reduced. With 4 voices per note, a four-note chord approaches the MiniNova's limit, and if further notes are added to the chord, "voice stealing" is implemented and the initial note(s) played may be cancelled.

t
If the limitation on polyphony imposed by Unison Voices is restrictive, a similar effect can be obtained by using multiple oscillators and adjusting their Density and Detune parameters. In fact, most of the factory patches use Density and Detune rather than Unison to achieve their thickening effect.

Parameter:	**Unison Detune**
Displayed As:	UniDTune
Default value:	25
Range of adjustment:	0 to 127

Unison Detune applies only when **Unison Voices** is set to something other than **Off**. The parameter determines how much each voice is detuned relative to the others; you will be able to hear a difference in the sound of the same note with different numbers of voices even if **Unison Detune** is set to zero, but the sound gets more interesting as it is increased in value.

t
Changing the settings of **Unison Voices** or **Unison Detune** while holding a note down has no effect on the sound. The new settings will only be effective when a fresh note is played.

Parameter:	**Portamento Time**
Displayed As:	PortTime
Default value:	Off
Range of adjustment:	Off, 1 to 127

With Portamento active, notes played sequentially glide from one to the next, rather than immediately jumping to the desired note pitch. The synth remembers the last note played and the glide will start from that note even after the key has been released. The **PortTime** is the duration of the glide, and a value of 115 equates to approximately 1 second. Portamento is primarily intended for use in a mono Mode (see **PortMode** below), where it is particularly effective. It can also be used in a Poly mode, but its operation can be unpredictable, particularly when chords are played. Note that **PreGlide** must be set to zero in order for Portamento to be operative.

Parameter:	**Portamento Mode**
Displayed As:	PortMode
Default value:	Expo
Range of adjustment:	Expo or Linear

This sets the 'shape' of the Portamento and **PreGlide** (see following page) transitions from one note to the next. In **Linear** mode, the glide alters the pitch evenly between the previous note and that being played. In **Expo** mode, the pitch changes more rapidly at first, and then approaches the 'target' note more slowly, i.e., exponentially.

PortMode = Linear

PortMode = Expo
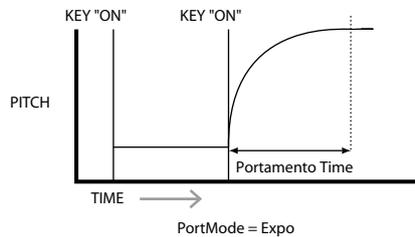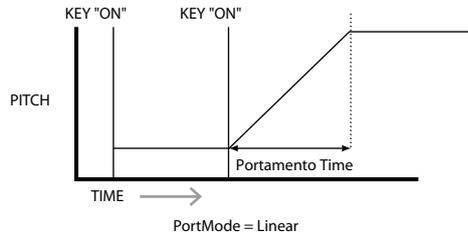
Parameter:	**Pre-Glide**
Displayed As:	PreGlide
Default value:	0
Range of adjustment:	-12 to +12

PreGlide takes priority over Portamento, though it does use the **PortTime** parameter to set its duration. **PreGlide** is calibrated in semitones, and each note played will actually begin a on a chromatically-related note up to an octave above (value = +12) or below (value = -12) the note corresponding to the key pressed, and glide towards the 'target' note. This differs from Portamento in that, e.g., two notes played in sequence will each have their own **PreGlide**, related to the notes played, and there will be no glide 'between' the notes.

> t   Although the use of Portamento is not recommended in Poly modes when playing more than one note at a time, this restriction does not apply to **PreGlide**, which can be very effective with full chords.

Parameter:	**Polyphony Mode**
Displayed As:	PolyMode
Default value:	Poly1
Range of adjustment:	Mono, MonoAG, Poly1, Poly2, Mono2

As the names imply, three of the possible modes are mono and two are polyphonic.

1. **Mono** – this is standard monophonic mode; only one note sounds at a time, and the "last played" rule applies.
2. **MonoAG** – AG stands for Auto-Glide. This is an alternative mono mode, which differs from Mono in the way Portamento and Pre-Glide work. In Mono mode, Portamento and PreGlide apply both if notes are played separately, or in a legato style (when one note is played when another is already held down). In MonoAG mode, Portamento and Pre-Glide only work if the keys are played in a legato style; playing notes separately produces no glide effect.
3. **Poly1** – in this polyphonic mode successively playing the same note(s) uses separate voices and the notes are therefore 'stacked', so the sound gets louder as more notes are played. The effect will only be evident on patches with a long amplitude release time.
4. **Poly2** – in this alternative mode, successively playing the same note(s) uses the original voices, so the volume increase inherent in Poly1 mode is avoided.
5. **Mono 2** – this differs from Mono in the way the Attack phases of the Envelopes are triggered. In Mono mode, when playing Legato style, the envelopes are only triggered once, by the initial key press. In Mono 2 mode, every key press will re-trigger all the Envelopes.

Edit menu - Submenu 6:   Env

The MiniNova provides a great deal of flexibility in the use of envelopes in sound creation, based on the familiar ADSR concept.
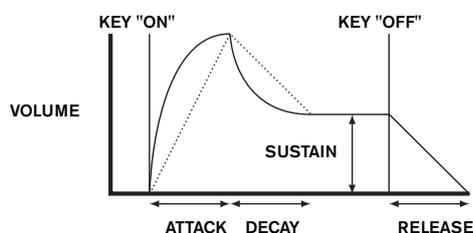
The ADSR envelope can be most easily visualised by considering the amplitude (volume) of a note over time. The envelope describing the "lifetime" of a note can be split into four distinct phases, and adjustments are provided for each of these:

*   **Attack** – the time it takes for the note to increase from zero (e.g., when the key is pressed) to its maximum level. A long attack time produces a "fade-in" effect.
*   **Decay** – the time it takes for the note to drop in level from the maximum value reached at the end of the attack phase to a new level, defined by the Sustain parameter.
*   **Sustain** – this is an amplitude value, and represents the volume of the note after the initial attack and decay phases – i.e., while holding the key down. Setting a low value of Sustain can give a very short, percussive effect (providing the attack and decay times are short).
*   **Release** – This is the time it takes for the note's volume to drop back to zero after the key is released. A high value of Release will cause the sound to remain audible (though diminishing in volume) after the key is released.
    Although the above discusses ADSR in terms of volume, note that the MiniNova is equipped with six separate envelope generators, allowing control of other synth blocks as well as amplitude – e.g., filters, oscillators, etc. Note that envelope generators 1 and 2 are dedicated to Amplitude and Filter control respectively, and are referred to as **Amp Env** and **Fltr Env**. A total of 16 parameters per envelope is available for adjustment.

With this submenu, it is first necessary to select the envelope whose parameters are to be adjusted:

Displayed As:	xxx Env or Env n (see range below)
Default value:	Amp Env
Range of adjustment:	Amp Env, Fltr Env, Env 3, Env 4, Env 5, Env 6

Amplitude Envelope
The following parameters apply only to the Amplitude Envelope, and will be available if **Env n** (above) is set to **Amp Env**.

Parameter:	**Amplitude Attack Time**
Displayed As:	AmpAtt
Default value:	2
Range of adjustment:	0 to 127

This parameter sets the note's attack time. With a value of 0 the note is at its maximum level immediately the key is pressed; with a value of 127, the note takes over 20 seconds to reach its maximum level. At the mid-setting (64), the time is approx. 250 ms (provided **Amplitude Attack Slope** (AmpAtSlp) has a value of zero).

> P   AmpAtt can be also adjusted directly from Row 5 of the **PERFORM** section of the control panel with Tweak Control **RC1**.

Parameter:	**Amplitude Decay Time**
Displayed As:	AmpDec
Default value:	90
Range of adjustment:	0 to 127

This parameter sets the notes' decay time. Decay time only has meaning if **AmpSus** (see below) is set to less than 127, as the Decay phase will be inaudible if the sustain level is the same as the level reached during the attack phase. At the mid-setting (64), the time is approx. 150 ms (provided **AmpDcSlp** has a value of 127).

> P   AmpDec can be also adjusted directly from Row 5 of the **PERFORM** section of the control panel with Tweak Control **RC2**.

Parameter:	**Amplitude Sustain Level**
Displayed As:	AmpSus
Default value:	127
Range of adjustment:	0 to 127

The value of the Sustain parameter sets the volume of the note after the completion of the decay phase. Setting a low value will obviously have the effect of emphasising the start of the note; setting it to zero will render the note silent after the decay phase has elapsed.

> P   AmpSus can be also adjusted directly from Row 5 of the **PERFORM** section of the control panel with Tweak Control **RC3**.

Parameter:	**Amplitude Release Time**
Displayed As:	AmpRel
Default value:	40
Range of adjustment:	0 to 127

Many sounds acquire some of their character from the notes remaining audible after the key is released; this "hanging" or "fade-out" effect, with the note gently dying away naturally (as with many real instruments) can be very effective. A setting of 64 gives a Release Time of approx. 360 ms. The MiniNova has a maximum release time of over 20 seconds (with AmpRel set to 127), but shorter times will probably be more useful! Note that the relationship between the parameter value and Release Time is not linear.

> (P) AmpRel can be also adjusted directly from Row 5 of the **PERFORM** section of the control panel with Tweak Control **RC4**.

> (t) Note that when playing polyphonically with sounds having long release times, it is possible for 'Voice Stealing' to occur. This means that some notes still sounding (in their Release phase) might suddenly cut off when other notes are played. This is more likely to happen when multiple Voices are in use.
> See "Parameter: Unison Voices" on page 19 for more information on this topic.

Parameter:	**Amplitude Velocity**
Displayed As:	AmpVeloc
Default value:	0
Range of adjustment:	-64 to +63

AmpVeloc does not modify the shape of the ADSR amplitude envelope in any way, but adds touch sensitivity to overall volume, so that with positive parameter values, the harder you play the keys, the louder will be the sound. With AmpVeloc set to zero, the volume is the same regardless of how the keys are played. The relationship between the velocity at which a note is played and volume is determined by the value. Note that negative values have the inverse effect.

> (t) For the most "natural" playing style, try setting Amplitude Velocity to about +40.

Parameter:	**Amplitude Envelope Repeat**
Displayed As:	AmpRept
Default value:	Off
Range of adjustment:	Off, 1 to 126, KeyOff

By using **Amplitude Repeat**, it is possible to repeat the Attack and Decay phases of the envelope before the Sustain phase is initiated. This can produce an interesting "stuttering" effect at the start of the note if the Attack and Decay times are set appropriately. The value of the **Repeat** parameter (from 1 to 126) is the actual number of repeats, so that if you set it to, e.g., **3**, you will hear a total of four attack/decay phases of the envelope – the initial one, plus three repeats. If set to **Off** there are not repeats. The maximum setting of **KeyOff** generates an infinite number of repeats.

Parameter:	**Amplitude Touch Trigger**
Displayed As:	AmpTTrig
Default value:	Off
Range of adjustment:	Off, T1ReTrig….T8ReTrig

You will have noticed that the MiniNova's eight Performance Pads are touch-sensitive. The Pads can be used in real time to provide creative control over the sound, which is especially useful when playing live.

Amplitude Touch Trigger assigns any of the Pads to act as a re-trigger button – as soon as the assignment is made, the Pad illuminates. When the Pad is touched, the amplitude envelope is re-triggered. After making the assignment, in order to use the feature it is necessary to put the Pads into Animate mode (see "Using the Pads as performance controls" on page 7) .

Parameter:	**Amplitude Multi-trigger**
Displayed As:	AmpMTrig
Default value:	Re-Trig
Range of adjustment:	Legato, Re-Trig

When this parameter is set to **Re-Trig**, each note played will trigger its full ADSR amplitude envelope, even if other keys are held down. In **Legato** mode, only the first key to be pressed will produce a note with the full envelope, all subsequent notes will omit the attack and decay phases, and sound only from the start of the Sustain phase. "Legato" literally means "smoothly", and this mode aids this style of playing.

It is important to appreciate that for the Legato mode to be operative, mono voicing must be selected – it will not work with polyphonic voicing. See "Edit menu - Submenu 5: Voice" on page 19.

> (i) **What is Legato?**
> As stated above, the musical term Legato means "smoothly". A Legato keyboard style is one where at least two notes overlap. This means that as you play the melody, you keep the previous (or an earlier) note sounding as you play another note. Once that note is sounding, you then release the earlier note.
>
> Legato style playing is relevant to some of the MiniNova's sonic possibilities. In the case of **Amplitude Multi-Trigger**, for example, it is important to appreciate that the envelope will re-trigger if any 'gap' is left between notes.

Parameter:	**Amplitude Attack Slope**
Displayed As:	AmpAtSlp
Default value:	0
Range of adjustment:	0 to 127

This parameter controls the "shape" of the attack characteristic. With a value of 0, the volume increases linearly during the attack phase – that is to say, increases by equal amounts in equal time intervals. A non-linear attack characteristic may be selected as an alternative, where the volume increases more rapidly at first. The diagram below illustrates this.

Parameter:	**Amplitude Decay Slope**
Displayed As:	AmpDcSlp
Default value:	127
Range of adjustment:	0 to 127

This parameter applies the same function as Amplitude Attack Slope to the Decay phase of the envelope. With a value of 0, the volume drops linearly from the maximum value to that defined by the Sustain parameter, but setting Decay Slope to a higher value will cause the volume to reduce more rapidly initially. The diagram below illustrates this:
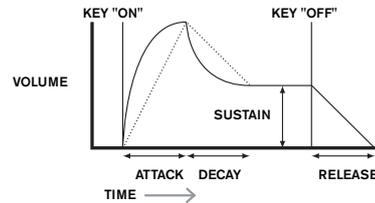
Parameter:	**Amplitude Attack Track**
Displayed As:	AmpAtTk
Default value:	0
Range of adjustment:	-64 to +63

This parameter relates a note's attack time to its position on the keyboard. When **Amplitude Attack Track** has a positive value, the attack time of a note decreases the higher up the keyboard it is played. Conversely, lower notes have a longer attack time. This aids in simulating the effect of a real stringed instrument (such as a grand piano), where the mass of the strings on the lower notes have a slower response time when struck. When a negative value is applied, the relationships are reversed.

Parameter:	**Amplitude Decay Track**
Displayed As:	AmpDecTk
Default value:	0
Range of adjustment:	-64 to +63

This parameter works in exactly the same way as **Attack Track**, except that it is the Decay time of a note that becomes dependent on its position on the keyboard.

Parameter:	**Amplitude Sustain Rate**
Displayed As:	AmpSusRt
Default value:	Flat
Range of adjustment:	-64 to -1, Flat, +1 to +63

With this parameter set to **Flat**, the volume during the Sustain phase of the envelope remains constant. Additional variations to a note's character can be obtained by causing the note to become louder or quieter while the key is held. A positive value of **Sustain Rate** will cause the volume to increase during the Sustain phase, and it will continue to do so until maximum level is reached. The parameter controls the rate at which the note increases volume, and the higher the value, the faster the rate of increase. Any Release time set will act as normal when the key is released, whether or not the maximum volume has been reached. If a negative value is set, the volume during the Sustain phase drops, and if the key is not released, the note will eventually become inaudible.
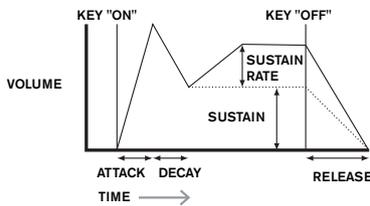
> **t** Lower values (positive or negative) of **Amplitude Sustain Rate** are generally more useful.

Parameter: **Amplitude Sustain Time**
Displayed As: AmpSusTm
Default value: KeyOff
Range of adjustment: 0 to 126, KeyOff

This parameter sets the duration of the Sustain phase. With a value of **KeyOff**, the note will remain audible continuously until the key is released (unless a negative value of **Sustain Rate** has been applied to reduce its volume). Any other value of **Sustain Time** will cut the note off automatically after a pre-determined time if the key is still held down. **Release Time** still applies if the key is released sooner. A value of 126 sets the Sustain time to approx. 10 seconds, while values around 60 set it to about 1 second.
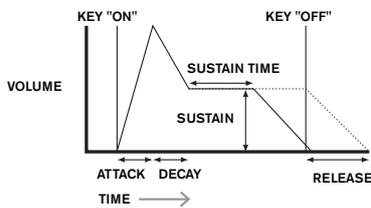
Parameter: **Amplitude Level Track**
Displayed As: AmpLvlTk
Default value: 0
Range of adjustment: -64 to +63

This parameter works in a similar way to the other "tracking" parameters **Attack Track** and **Decay Track**, but it is the volume of the note which is changed, according to the interval between it and the Level Track Note (see below). With a positive value, notes higher than the Track Note get progressively louder the further from the Track Note they are, and vice-versa. With a negative value, notes higher than the Track Note get progressively quieter the further from the Track Note they are, and again, vice-versa. Note that this volume modification is applied to all phases of the amplitude envelope equally; it is the overall volume of the note which changes with **Amp Level Track**. The effect should be used sparingly; low values have a better effect.

> **i** Note that although **Amplitude Level Track** appears to operate in a very similar manner to **Amplitude Attack Track** and **Amplitude Decay Track**, only **Amplitude Level Track** uses a user-definable note as the reference (set by **Level Track Note**), above which, for positive values, notes get louder, and below which they get softer. With negative values, the inverse relationship will apply.

Common Envelope Parameter
Parameter: **Level Track Note**
Displayed As: LvlTkNte
Default value: C3
Range of adjustment: C-2 to G8

This parameter is common to all the envelopes. This sets the reference note used for all Level Track parameters, including Amp Level Track. When active, this parameter increases the volume for notes above the chosen Track Note, and reduces it for notes below it. C 3, the default value, is Middle C on the keyboard; this is the C one octave above the lowest note on the keyboard (also C), providing no **OCTAVE** buttons [24] are selected.

Filter Envelope
The following parameters apply only to the Filter Envelope, and will be available if **Env n** (page 20) is set to **Fltr Env**.

The 16 parameters available for adjustment with the Filter Envelope closely match those for the Amplitude Envelope. Whereas the Amplitude Envelope is concerned with modifying the sound's amplitude, the Filter Envelope gives you "dynamic" filtering, by establishing a relationship between the filter section and ADSR Filter Envelope, resulting in the filter frequency being varied by the shape of the envelope.

> **i** In order to hear the effect of any of the Filter Envelope parameters, you will first need to go to the Filter Menus and set up some filtering. Then set **F1Env2** or **F2Env2** to an initial value of approx. +30 and ensure that the filter is not fully open – i.e., set **F1Freq** to mid-range.

Parameter: **Filter Attack Time**
Displayed As: FltAtt
Default value: 2
Range of adjustment: 0 to 127

This parameter sets how the filter section acts during the note's Attack phase. The higher the value, the longer it takes for the filter to react during this phase.

> **P** FltAtt can be also adjusted directly from Row 4 of the **PERFORM** section of the control panel with Tweak Control **RC1**.

Parameter: **Filter Decay Time**
Displayed As: FltDec
Default value: 75
Range of adjustment: 0 to 127

This parameter sets how the filter section acts during the note's Decay phase. Again, the higher the parameter value, the longer the period for which filtering is applied.

> **P** FltDec can be also adjusted directly from Row 4 of the **PERFORM** section of the control panel with Tweak Control **RC2**.

Parameter: **Filter Sustain Level**
Displayed As: FltSus
Default value: 35
Range of adjustment: 0 to 127

The frequency of the filter (cut-off or centre, depending on filter type) "settles" at a value set by the **Filter Sustain Level**. Thus, once the Attack and Decay stages of the envelope are completed, the harmonic content that will be most evident in the sound will be determined by this parameter. Remember that if the filter frequency parameter (as set in the **Filter** Menu) is set at a too low or too high a value, the envelope's effect will be limited.

> **P** FltSus can be also adjusted directly from Row 4 of the **PERFORM** section of the control panel with Tweak Control **RC3**.

Parameter: **Filter Release Time**
Displayed As: FltRel
Default value: 45
Range of adjustment: 0 to 127

As Filter Release is increased in value, the note undergoes increasingly more filter action once the key is released.

> **t** Note that the **Amplitude Release** time (adjusted in the Amplitude Envelope submenu) must be set sufficiently high to produce an audible "fade-out" before the effect of filtering on the "tail" of the note is evident.

Parameter: **Filter Velocity**
Displayed As: FltVeloc
Default value: 0
Range of adjustment: -64 to +63

As **Amplitude Velocity** adds touch sensitivity to volume, so **Filter Velocity** can be set to make filter action touch sensitive. With positive parameter values, the harder you play the keys, the greater will be the effect of the filter. With **Filter Velocity** set to zero, the sound's characteristics are the same regardless of how the keys are played. Note that negative values have the inverse effect.

Parameter: **Filter Repeat**
Displayed As: FltRept
Default value: Off
Range of adjustment: Off, 1-126, KeyOff

When Filter Repeat is set to a value other than Off, the Attack and Decay phases of the envelope are repeated before the Sustain phase is initiated. This has a similar effect to **Amplitude Repeat** and use of either or both repeat parameters can create some quite striking sounds.

Parameter: **Filter Touch Trigger**
Displayed As: FltTTrig
Default value: Off
Range of adjustment: Off, T1ReTrig…T8ReTrig, T1Triggr…T8Triggr, T1Enable…T8Enable

Unlike **Amplitude Touch Trigger**, **Filter Touch Trigger** has three options per Pad control: **Trigger**, **Re-trigger** and **Enable**. However, as with **Amplitude Touch Trigger**, it is necessary to enable the **ANIMATE** mode for the Pads to be operative (see "Using the Pads as performance controls" on page 7).

1. **Re-Trigger** – acts in a similar way to **Amplitude Re-Trigger**, except that it is the filter action which is re-triggered by touching the selected Pad. The note plays as normal when the key is pressed, pressing the Pad re-triggers the entire envelope.
2. **Trigger** - in this mode, the envelope-triggered filter action is not initiated by pressing a key, and the note will initially sound with no envelope acting on the filter. Pressing the Pad (while the key is pressed) will trigger the filter envelope.
3. **Enable** – in this mode, the envelope-triggered filter action is initiated by the keyboard, but only while the Pad is being pressed. Thus you can very easily flip between the sound with and without the action of the envelope on the filter.

Parameter:	**Filter Multi-trigger**
Displayed As:	FltMTrig
Default value:	Re-Trig
Range of adjustment:	Re-Trig or Legato

This operates in a very similar way to **Amplitude Multi-trigger**. When set to **Re-Trig**, each note played will trigger its full ADSR envelope, even if other keys are held down. With the envelope applied to the filter section, this means that the effect of any envelope-triggered filtering will be heard on every note. When set to **Legato**, only the first key to be pressed will produce a note with the full envelope and thus produce any filtering effect. All subsequent notes will lack any dynamic filtering. Remember that for the Legato mode to be operative, mono voicing must be selected – it will not work with polyphonic voicing. See "Edit menu - Submenu 5: Voice" on page 19.

> ℹ️ See "What is Legato?" on page 21 for more details about Legato style.

Parameter:	**Filter Attack Slope**
Displayed As:	FltAtSlp
Default value:	0
Range of adjustment:	0 to 127

This parameter controls the "shape" of the attack characteristic as applied to the filters. With a value of zero, any filtering effect applied to the Attack phase increases linearly – that is to say, increases by equal amounts in equal time intervals. A non-linear attack characteristic may be selected as an alternative, where the filter effect increases more rapidly at first.

Parameter:	**Filter Decay Slope**
Displayed As:	FltDcSlp
Default value:	127
Range of adjustment:	0 to 127

This corresponds to **Filter Attack Slope** in the same manner that **Amplitude Decay Slope** corresponds to **Amplitude Attack Slope**. Thus the linearity of the reaction of the filter section during the Decay phase of the envelope can be varied, from linear to a more exponential slope, where any filter effect is more pronounced during the first part of the Decay phase.

Parameter:	**Filter Attack Track**
Displayed As:	FltAtTk
Default value:	0
Range of adjustment:	-64 to +63

Like **Amplitude Attack Track**, this parameter relates a note's attack time to its position on the keyboard. When **Filter Attack Track** has a positive value, the filtering effect during the Attack phase of a note is shortened as you go up the keyboard. Conversely, lower notes have their attack time increased. When a negative value is applied, the relationships are reversed.

Parameter:	**Filter Decay Track**
Displayed As:	FltDecTk
Default value:	0
Range of adjustment:	-64 to +63

This parameter works in exactly the same way as **Attack Track**, except that it is the filter effect during the Decay phase of a note that becomes dependent on its keyboard position.

Parameter:	**Filter Sustain Rate**
Displayed As:	FltSusRt
Default value:	Flat
Range of adjustment:	-64 to -1, Flat, 1 to 63

With a value of Flat, filter frequency remains constant during the Sustain phase of the note. If **Filter Sustain Rate** is given a positive value, the filter frequency continues to increase during the Sustain phase, thus the character of the note continues altering audibly for longer. With low values of **Filter Sustain Rate**, the change is slow, and increases in rapidity as the value is increased. With negative values, the filter frequency decreases during the Sustain phase. See "Parameter: Amplitude Sustain Rate" on page 21 for an illustration.

Parameter:	**Filter Sustain Time**
Displayed As:	FltSusTm
Default value:	KeyOff
Range of adjustment:	0 – 126, KeyOff

This parameter also applies to the Sustain phase, and sets how long any envelope-triggered filtering remains active. When set to **KeyOff,** the filtering remains applied continuously until the key is released. Any lower value of **Sustain Time** will cause the filtering effect to suddenly stop before the note ends, and you will be left with release

phase of the envelope. This does, of course, only occur if the **Amplitude Sustain Time** is longer than the Filter Sustain Time, otherwise the note will stop sounding altogether before the filter has cut off.

Parameter:	**Filter Level Track**
Displayed As:	FltLvlTk
Default value:	0
Range of adjustment:	-64 to +63

This parameter works in a similar way to the other "tracking" parameters, but it is the depth with which the envelope is applied to the filter which alters, in relation to the interval between the note played and the **Level Track Note** (see below). With a positive value, the envelope-triggered filtering effect becomes progressively more pronounced for notes higher than the **Track Note** the further from the **Track Note** they are, and vice-versa. With a negative value, notes higher than the **Track Note** undergo progressively less filtering the further from the **Track Note** they are, and again, vice-versa.

Parameter:	**Level Track Note**
Displayed As:	LvlTkNte
Default value:	C3
Range of adjustment:	C-2 to G8

This parameter is common to all the envelopes. See "Parameter: Amplitude Level Track" on page 22.

Envelopes 3 to 6

In addition to dedicated Amplitude and Filter envelopes, the MiniNova is equipped with four further assignable envelopes, Envelopes 3 to 6. These envelopes have virtually the same set of parameters as the Amplitude and Filter envelopes, but they can be assigned at will to control many other synth functions, including most oscillator parameters, filters, EQ and panning among others. These parameters will be available if **Env n** (page 20) is set to **Env 3** to **Env 6** .

The assignment of Envelopes 3 to 6 to other synth parameters is performed in the Modulation Matrix (**ModMatrx**) Menu (see "What is Legato?" on page 21 for full details). In order to audition their effects, you must first open the **ModMatrx** Menu and set a **Mod Slot Source** to **Env3** and the Destination to a parameter of your choice (e.g., Global Oscillator Pitch – **0123Ptch**).

The arrangement of parameters for Envelopes 3 to 6 is identical, and the arrangement closely follows that of Envelopes 1 and 2 (Amplitude and Filters). Although denoted as Envelope 3, the parameter summaries below apply equally to Envelopes 4, 5 & 6, so are not repeated.

The actual function of Envelopes 3 to 6 will obviously depend on what they are routed to control in the Modulation Matrix Menu. However, the derivation of the envelope parameters themselves follow those already described for the Amplitude and Filter envelopes, with the exception of the **Delay** parameter (e.g., **E3Delay**), whose function is described below.

Parameter:	**Envelope 3 Attack Time**
Displayed As:	E3Att
Default value:	10
Range of adjustment:	0 to 127

Parameter:	**Envelope 3 Decay Time**
Displayed As:	E3Dec
Default value:	70
Range of adjustment:	0 to 127

Parameter:	**Envelope 3 Sustain Level**
Displayed As:	E3Sus
Default value:	64
Range of adjustment:	0 to 127

Parameter:	**Envelope 3 Release Time**
Displayed As:	E3Rel
Default value:	40
Range of adjustment:	0 to 127

Parameter:	**Envelope 3 Delay**
Displayed As:	E3Delay
Default value:	0
Range of adjustment:	0 to 127

This parameter delays the start of the entire envelope. When a key is pressed, its note sounds normally, with Envelopes 1 and 2 acting as they are programmed. But any further modulation effects triggered by Envelopes 3 to 6 will be delayed by a time set by the **Delay** parameter. The maximum value of 127 represents a delay of 10 seconds, while a value of about 60~70 represents a delay of approximately 1 second.

Parameter:	**Envelope 3 Repeat**
Displayed As:	E3Rept
Default value:	Off
Range of adjustment:	Off, 1 to 126, KeyOff

Parameter: **Envelope 3 Touch Trigger**
Displayed As: E3TTrig
Default value: Off
Range of adjustment: Off, T1ReTrig…T8ReTrig, T1Triggr…T8Triggr,
T1Enable…T8Enable

Parameter: **Envelope 3 Multi-trigger**
Displayed As: E3MTrig
Default value: Re-Trig
Range of adjustment: Re-Trig or Legato

Parameter: **Envelope 3 Attack Slope**
Displayed As: E3AtSlp
Default value: 0
Range of adjustment: 0 to 127

Parameter: **Envelope 3 Decay Slope**
Displayed As: E3DcSlp
Default value: 127
Range of adjustment: 0 to 127

Parameter: **Envelope 3 Attack Track**
Displayed As: E3AtTk
Default value: 0
Range of adjustment: 0 to 127

Parameter: **Envelope 3 Decay Track**
Displayed As: E3DecTk
Default value: 0
Range of adjustment: -64 to +63

Parameter: **Envelope 3 Sustain Rate**
Displayed As: E3SusRt
Default value: Flat
Range of adjustment: -64 to -1, Flat, +1 to +63

Parameter: **Envelope 3 Sustain time**
Displayed As: E3SusTm
Default value: 0
Range of adjustment: 0 to 126, KeyOff

Parameter: **Envelope 3 Level Track**
Displayed As: E3LvlTk
Default value: 0
Range of adjustment: -64 to +63

Parameter: **Level Track Note**
Displayed As: LvlTkNte
Default value: C3
Range of adjustment: C-2 to G8
This parameter is common to all the envelopes.
See "Parameter: Amplitude Level Track" on page 22.

Edit menu - Submenu 7: LFO

The MiniNova has three separate Low Frequency Oscillators (LFOs). These are designated LFO1, 2 and 3, are identical in terms of features, and can be used freely to modify many other synth parameters, such as oscillator pitch or level, filters, panning, etc.

The assignment of LFOs 1 to 3 to other synth parameters is performed in the Modulation Matrix Menu (see "What is Legato?" on page 21 for full details). In order to audition their effects, you should first open the Modulation Matrix Menu and set a Modulation Slot's Source to Lfo1+/- or Lfo1+* and the Destination to a parameter of your choice. Note also that the Depth control on this menu determines the amount of LFO modulation applied to the Destination parameter, and increasing this value will have a different effect depending on what the Destination parameter is, but can generally be taken to mean "more effect". The interpretation of negative values of Depth will also depend on the chosen Destination parameter.

*Selecting **Lfo1+** as the source makes the LFO vary the controlled parameter in a positive sense (i.e., increasing) only. Selecting it as **Lfo1+/-** varies it in both a positive and a negative sense.

With this submenu, it is first necessary to select the LFO whose parameters are to be adjusted:

Displayed As: LFO n (where n is 1 to 3)
Default value: LFO 1
Range of adjustment: LFO 1, LFO 2, LFO 3

A total of 12 parameters per LFO is available for adjustment. Because the three LFOs are identical, only the functions of LFO1 are described.

Parameter: **LFO 1 Rate**
Displayed As: L1Rate
Default value: 68
Range of adjustment: 0 to 127

Rate is the LFO's frequency. A value of zero stops the LFO, and most musical effects are likely to use values in the 40~70 range, though higher or lower values may be appropriate for certain sound effects.

> When the LFO Rate is set to zero, the LFO is "stopped", but will still apply an offset to the parameter it is modulating of a magnitude dependent on where it stopped in its cycle.
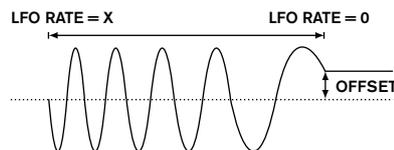
Parameter: **LFO 1 Rate Sync**
Displayed As: L1Sync
Default value: Off
Range of adjustment: See Sync Values Table on page 34.
This control allows the frequency of the LFO to be synchronised to an internal/external MIDI clock. When set to **Off**, the LFOs run at a frequency set by the **L1Rate** parameter. At all other settings **L1Rate** becomes inoperative, and the LFO rate is determined by **L1Sync**, which in turn is derived from the MIDI clock. When using internal MIDI clock, the rate can be set using the **TEMPO** control [21].

Parameter: **LFO 1 Waveform**
Displayed As: L1Wave
Default value: Sine
Range of adjustment: See LFO Waveform Table on page 34.

The MiniNova's LFOs are able to generate not only the familiar sine, sawtooth, triangle and square waveforms for modulation purposes, but are also able to produce a wide range of preset sequences of various lengths and random waveforms. A common use of an LFO is to modulate the main oscillator(s), and with many of the sequenced waveforms, setting the **Depth** parameter in the Modulation Matrix Menu to either 30 or 36 (see table) will ensure that the resulting oscillator pitches will be musically associated in some way.

Parameter: **LFO 1 Phase**
Displayed As: L1Phase
Default value: Free
Range of adjustment: Free, 0deg -357deg
This control is only active if **L1KSync** (same menu) is set **On**. It determines the start point of the LFO waveform when the key is pressed. A complete waveform has 360°, and the control's increments are in 3° steps. Thus a half-way setting (180deg) will cause the modulating waveform to start at half-way through its cycle.
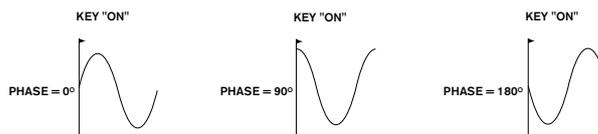
Parameter: **LFO 1 Slew**
Displayed As: L1Slew
Default value: Off
Range of adjustment: Off, 1 to 127
Slew has the effect of modifying the shape of the LFO waveform. Sharp edges become less sharp as **Slew** is increased. The effect of this can be easily observed by selecting Square as the LFO waveform and setting the rate fairly low so that the output when a key is pressed alternates between just two tones. Increasing the value of **Slew** will cause the transition between the two tones to become a "glide" rather than a sharp change. This is caused by the vertical edges of the square LFO waveform being slewed.

Note that **Slew** has an effect on all LFO waveforms, including sine. The effect of LFO Slew differs somewhat with different LFO waveforms. As **Slew** is increased, the time taken to reach maximum amplitude is increased, and can ultimately result in it never being achieved at all, though the setting at which this point is reached will vary with waveform.

SQUARE WAVE
NO SLEW

SMALL SLEW VALUE

LARGE SLEW VALUE
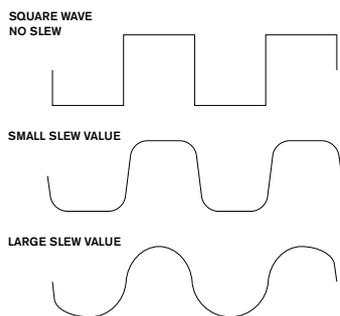
Parameter: **LFO 1 Key Sync On/Off**
Displayed As: L1KSync
Default value: Off
Range of adjustment: Off or On

Each LFO runs continuously, 'in the background'. If the **Key Sync** setting is **Off**, there is no way of predicting where the waveform will be when a key is pressed. Consecutive presses of a key will inevitably produce varying results. Setting **Key Sync** to **On** re-starts the LFO at the same point on the waveform every time a key is pressed. The actual point is set by the **Phase** parameter (**L1Phase**).

KEY SYNC = OFF

NOTE 1

NOTE 2

KEY SYNC = ON

NOTE 1

NOTE 2
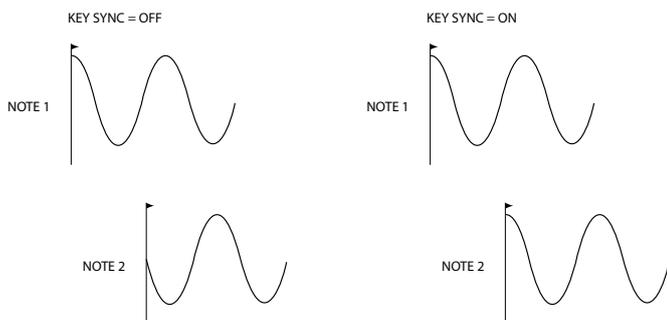
Parameter: **LFO 1 Common Sync**
Displayed As: L1Comm
Default value: Off
Range of adjustment: Off or On

When LFOs are in use for pitch modulation (their most common application), **Common Sync** is only applicable to polyphonic voices. It ensures that the phase of the LFO waveform is synchronised for every note being played. When to set **Off**, there is no such synchronisation, and playing a second note while one is already pressed will result in an unsynchronised sound as the modulations will be out of time.

COMMON SYNC = OFF

NOTE 1

NOTE 2

COMMON SYNC = ON

NOTE 1

NOTE 2

Set **LFO Common Sync** to **On** for an emulation of early analogue polyphonic synths.

Parameter: **LFO 1 One-Shot**
Displayed As: L1OneSht
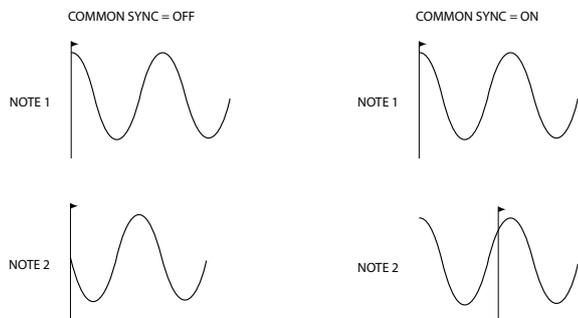Default value: Off
Range of adjustment: Off or On

As its name suggests, setting this parameter to **On** causes the LFO to generate just a single cycle of its waveform. Note that a full waveform cycle is always generated regardless of the setting of LFO Phase; if LFO Phase is set to 90°, the one-shot waveform will start at the 90° point, execute a full cycle, and end at 90°.

Parameter: **LFO 1 Delay**
Displayed As: L1Delay
Default value: 0
Range of adjustment: 0 to 127

LFO Delay is a time parameter whose function is determined by **L1InOut** (see below).

Parameter: **LFO 1 Delay Sync**
Displayed As: L1DSync
Default value: Off
Range of adjustment: See **Sync Values Table** at page 34.

When this parameter is set to Off, the LFO delay is controlled by the **Delay** parameter (**L1Delay**). At all other settings L1Delay becomes inoperative, and the LFO delay is derived from the internal/external MIDI clock.

Parameter: **LFO 1 Fade In/Fade Out**
Displayed As: L1InOut
Default value: FadeIn
Range of adjustment: FadeIn, FadeOut, GateIn, GateOut

The function of the four possible settings of **L1InOut** are as follows:

1.   **FadeIn** - the LFO's modulation is gradually increased over the time period set by the **Delay** parameter (**L1Delay**).
2.   **GateIn** – the onset of the LFO's modulation is delayed by the time period set by **L1Delay** parameter, and then starts immediately at full level.
3.   **FadeOut** – the LFO's modulation is gradually decreased over the time period set by the **L1Delay** parameter, leaving the note with no LFO modulation.
4.   **GateOut** – the note is fully modulated by the LFO for the time period set by the **L1Delay**. At this time, the modulation stops abruptly.

Parameter: **LFO 1 Delay Trigger**
Displayed As: L1DTrig
Default value: Legato
Range of adjustment: Legato or Re-Trig

This parameter works in conjunction with the Fade/Gate parameters set by **L1InOut**. In **Re-Trig** mode, every note played has its own delay time, as set by L1Delay (or MIDI clock if **L1Dsync** is active). In **Legato** mode, it is only the first note of a legato-style passage which triggers the delay – i.e., second and subsequent notes do not re-trigger the Delay function. For the **Legato** setting of **Delay Trigger** to be operative, mono voicing must be selected – it will not work with polyphonic voicing. See "Edit menu - Submenu 5: Voice" on page 19.

See "What is Legato?" on page 21 for more details about Legato style.

Edit menu - Submenu 8:   ModMatrx

The heart of a versatile synthesizer lies in the ability to interconnect the various controllers, sound generators and processing blocks such that one is controlling – or "modulating" - another, in as many ways as possible. MiniNova provides tremendous flexibility of control routing, and there is a dedicated menu for this, the Modulation Matrix Menu (**ModMatrx**).

NUMBER 20

NUMBER 1

SOURCE 1

SOURCE 2

TOUCH SELECT

DESTINATION

DEPTH
+
−
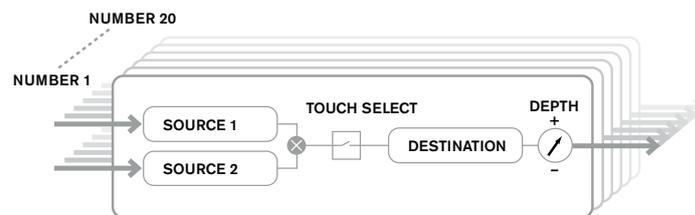
The menu may be visualised as a system for connecting controlling sources to a specific area of the synth. Each such connection assignment is termed a *slot*, and there are 20 such slots, accessed by **ModSlt** (see below). Each slot defines how one or two control sources are routed to a controlled parameter. The routing possibilities available in each of the 20 slots are identical, and the control description below is applicable to all of them.

By 'variable', we mean that it is not just the routing of a controlling source to a controlled parameter which is defined in each slot, but also the "magnitude" of the control. Thus the 'amount' of control – or control 'range' – used is up to you.

By 'additive' we mean that a parameter may be varied by more than one source, if wished. Each slot allows for two sources to be routed to a parameter, and their effects are *multiplied* together. This means that if either of them is at zero, there will be no modulation. However, there is no reason why you can't have further slots routing these or other sources to the same parameter. In this case, the control signals from different slots "add" to produce the overall effect.
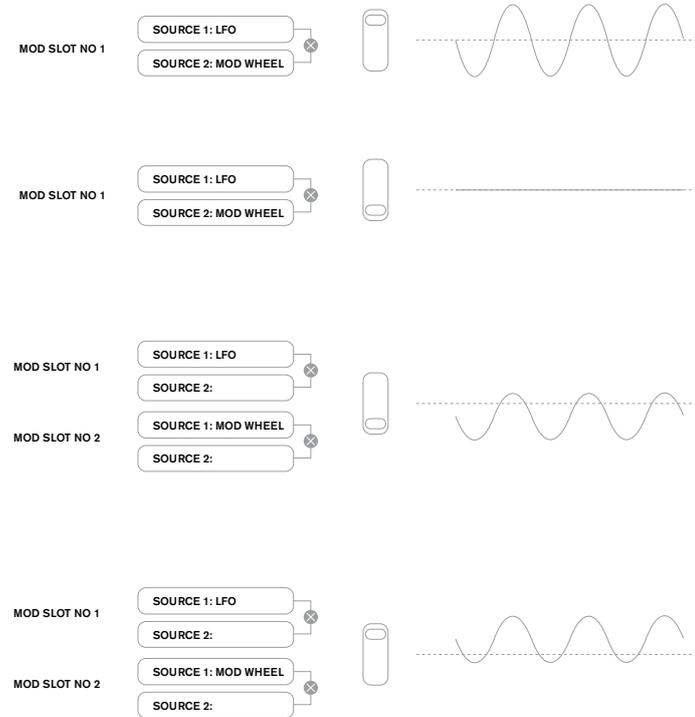
In addition, the Modulation Matrix Menu lets you assign the Pads as additional controllers, so long as the Animate mode is enabled (see "Using the Pads as performance controls" on page 7).

With this submenu, it is first necessary to select the Modulation Slot whose parameters are to be adjusted:

Displayed As:	ModSltn (where n is 1 to 20)
Default value:	ModSlt1
Range of adjustment:	ModSlt1…ModSlt20

The Modulation Matrix has 20 'slots' ('mod slots'), each defining a routing assignment of one (or two) sources to a destination. All the slots have the same selection of sources and destinations and any or all can be used. The same source can control multiple destinations, and one destination can be controlled by multiple sources.

Because the 20 Modulation Slots are identical, only the functions of Slot 1 are described.

Parameter:	**First Source**
Displayed As:	Source 1
Default value:	Direct
Range of adjustment:	See **Modulation Matrix Sources** Table on page 35.

This selects a control source (modulator), which will be routed to the destination set by **Destin**. Setting both **Source1** and **Source2** to **Direct** means no modulation is defined.

Parameter:	**Second Source**
Displayed As:	Source 2
Default value:	Direct
Range of adjustment:	See **Modulation Matrix Sources** Table on page 35.

This selects a second control source for the chosen destination. If only one source per

patch is being used, set **Source2** to.

Parameter:	**Touch controller enable**
Displayed As:	TouchSel
Default value:	Off
Range of adjustment:	Touch1….Touch 8

The eight **ANIMATE** Pads can be programmed as touch controllers, so that they initiate a change to a parameter value (defined by **Destin**, see below) when pressed. Note that Animate mode must be enabled for the Pads to be active. **ANIMATE** pads will illuminate purple if a controller has been assigned. See "Using the Pads as performance controls" on page 7 for more details regarding using the Pads. Note that when both a Pad and other sources (**Source1 and/or Source2**) are assigned in the same slot, the Pad acts as a switch for the other sources, whose effect will only be heard when the Pad is pressed.

Parameter:	**Destination**
Displayed As:	Destin
Default value:	O123Ptch
Range of adjustment:	See **Mod Matrix Destination** Table on page 36.

This sets which MiniNova parameter is to be controlled by the selected source (or sources) in the current matrix configuration. The range of possibilities comprises:

- Parameters that directly affect the sound:
    - four parameters per oscillator
    - global pitch (**0123Ptch**)
    - the six mixer inputs from the oscillators, noise source and ring modulators, plus the mixer output level
    - per-filter drive amount, frequency and resonance, plus filter balance
    - 34 assorted FX parameters including chorus, delay, EQ etc.
    - 3 Vocoder parameters
    - Vocal tuning pitch shift
- Parameters that can also act as modulating sources (thus permitting recursive modulation):
    - LFO 1 to 3 rate
    - the Decay phases of the Amplitude Envelope (Env1Dec) and Filter Envelope (Env2Dec)

Parameter:	**Depth**
Displayed As:	Depth
Default value:	0
Range of adjustment:	-64 to +63

The **Depth** control sets the level of the control being applied to the Destination – i.e., the parameter being modulated. If both Source1 and Source2 are active in the current slot, **Depth** controls their combined effect.

Edit menu - Submenu 9: Effects

The MiniNova comes equipped with a comprehensive set of DSP-based effects processors, which can be applied to both the synth sound and any audio applied to the MiniNova's audio inputs.

The FX section comprises five processing *slots*, each of which may be "loaded" with an FX processor from a pool of devices that includes panning, equalisation, compression, delay, chorus, distortion, reverb and gator effects. In addition to the slots, controls are also provided for global FX parameters like panning, FX level, FX feedback, etc.

The FX controls are accessed from the **Effects** submenu. This provides six options: **PanRoute** and **FXSlot1** to **FXSlot5**. **PanRoute** provides selection of panning and slot configuration. Entering FXSlot1 to FXSlot5 allows you to choose the FX device and its associated parameters for each of the five slots.

The following parameters apply only to the **PanRoute** option:

Parameter: **Pan Position**
Displayed As: PanPosn
Default value: 0
Range of adjustment: -64 to +63

This is the main manual pan control and positions the dry (pre-FX) synth sound/Input audio in the stereo image between the left and right outputs. Negative values of PanPosn move the sound to the left and positive values to the right. Note that some FX (e.g., reverb, chorus) are inherently stereo, and these are added post-panning. Thus if you are using a sound that uses FX such as these, PanPosn will appear to not entirely localise the sound fully left or right at its extreme settings.

Parameter: **Pan Rate**
Displayed As: PanRate
Default value: 40
Range of adjustment: 0 to 127

Automatic panning is also possible, and the Pan section has a dedicated sinewave LFO which controls this. The **PanRate** parameter controls the LFO frequency, and thus how rapidly the sound moves between left and right and back again. With a value of 40, the sound takes approx. 3 seconds to complete a full cycle, and the range of control allows extremely slow or extremely fast panning.

> **t** For the most effective results with **Pan Rate**, ensure that **PanPosn** is set 0 (i.e., centre panning)

Parameter: **Pan Sync**
Displayed As: PanSync
Default value: Off
Range of adjustment: See Sync Values Table at page 34.

Auto-panning rate may be synchronised to the internal or external MIDI clock, using a wide variety of tempos.

Parameter: **Pan Depth**
Displayed As: PanDepth
Default value: 0
Range of adjustment: 0 to 127

This control determines the amount of image shift applied by the auto-panner. At its maximum value of 127, the auto-panner will pan the sound both fully left and fully right; lower values will pan less extremely, with the sound remaining more centrally located. The auto-panner is effectively off when the parameter value is zero (but the "manual" pan parameter **PanPosn** is still operative).

Parameter: **FX Slot Routing**
Displayed As: FXRouting
Default value: 1
Range of adjustment: 0 to 7

This parameter lets you configure the interconnection of the FX slots. The five slots may be interconnected in serial, in parallel, or in various combinations of serial and parallel.

FXRouting = 0
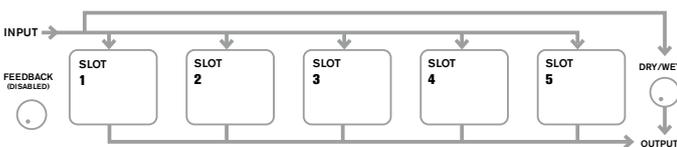
FXRouting = 1
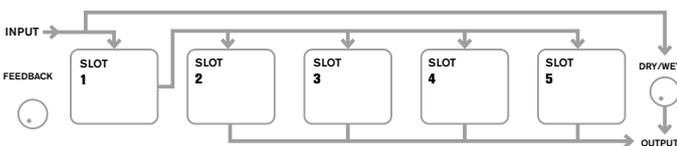
FXRouting = 2
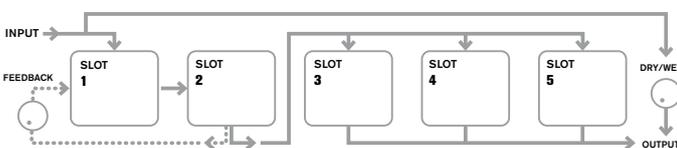
FXRouting = 3
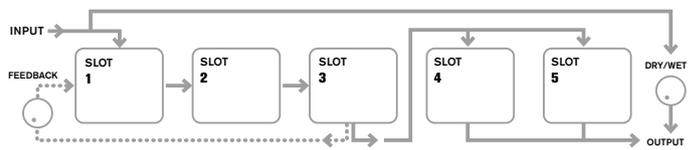
FXRouting = 4
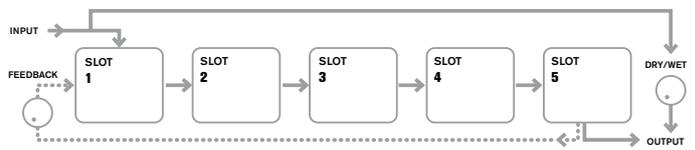
FXRouting = 5
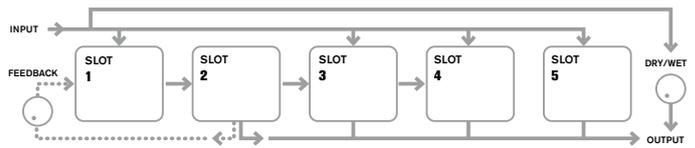
FXRouting = 6
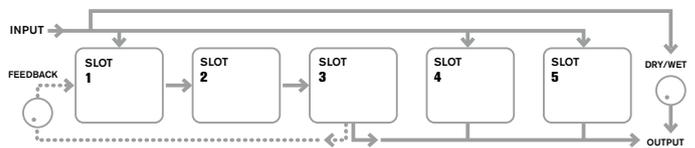
FXRouting = 7
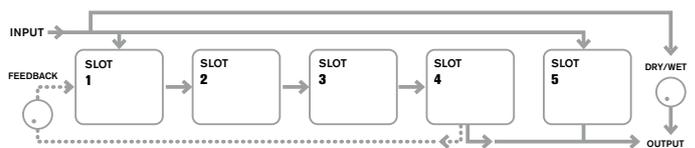
Parameter: **Effect feedback**
Displayed As: FXFedback
Default value: 0
Range of adjustment: 0 to 127

This parameter controls how much signal is fed back to the input of the effects chain from its output. The FX Slot from which the feedback is derived varies with the FX Routing configuration in use – see diagrams above. However, with all routing configurations, the feedback is added back into the chain at FX Slot 1. Note that not all configurations employ feedback.

FX Slots

Each of the FX slot options (accessed from the initial **Effects** submenu**)** are identical, and may be loaded with one of the various FX processors available. The parameter descriptions following refer to the first FX slot; operation of the other four are identical.

> **t** The FX types can be categorised in various ways: some are time-based (chorus, delay), others are static (EQ, distortion). Some should be used as an FX send/return loop (implying a parallel connection), others as an insert (implying a serial connection). Depending on the synth sound itself and the actual effects being used, some configurations will clearly work better than others. When using multiple effects, try a few different interconnections to see which works best.

Parameter: **FX1 Type**
Displayed As: FX1 Type
Default value: Bypass
Range of adjustment: See **Effects Types** Table on page 38.

The table shows the "pool" of FX devices available. As DSP capacity is finite, each device in the list may only be loaded into one slot, and once it has been loaded, it will no longer appear in the list of available processors for the other slots. You will see that multiples of most of the FX devices are provided, to allow the most creative use of the FX.

Parameter: **Effect Amount**
Displayed As: `FX1 Amnt`
Default value: 64
Range of adjustment: 0 to 127

The precise function of this parameter depends on which FX device is loaded into the slot. See the table below for a summary.

FX TYPE	ADJUSTED PARAMETER
Compressor	Level
EQ	Level
Distortion	Amount or bit/sample rate reduction
Delay	Send and Return levels
Chorus	Level
Reverb	Send and Return levels
Gator	Level

The remaining parameters available for adjustment in the **FXSLOTn** submenu are determined by which effects device has been loaded into the slot. A slot which has no FX device loaded has no further menu options available.

Each FX device has its own menu; these are described in turn below. All references to FX1 may be taken as being equally applicable to the other four FX slots.

EQ Menu

The equaliser is a three-band 'swept' type, with cut/boost and frequency controls for each band. The LF and HF sections are second-order (slope of 12 dB/octave) shelving filters, and the MF section is a bell-response filter.

> ℹ Note that **FX1 Amnt** parameter should be set to 127 for the full range of cut or boost (±12 dB) to be available. Lower settings of **FX1 Amnt** will result less cut or boost being applied at the minimum or maximum values of the EQ Level parameters

Parameter: **LF Cut/Boost**
Displayed As: `EQBasLvl`
Default value: 0
Range of adjustment: -64 to +63

This parameter controls the LF response of the equaliser; a value of **0** gives a flat response in the LF region, positive values will give a rise in the LF response – i.e., more bass, and negative values will have the opposite effect. The range of adjustment is ±12 dB (with **FX1 Amnt** set to **127**).

Parameter: **MF Cut/Boost**
Displayed As: `EQMidLvl`
Default value: 0
Range of adjustment: -64 to +63

This parameter controls the MF response of the equaliser; a value of 0 gives a flat response in the MF region, positive values will give a rise in the MF response – i.e., more mid-frequencies (the voice region of the audio spectrum), and negative values will accordingly reduce the MF response. The range of adjustment is ±12 dB (with **FX1 Amnt** set to 127).

Parameter: **HF Cut/Boost**
Displayed As: `EQTrbLvl`
Default value: 0
Range of adjustment: -64 to +63

This parameter controls the HF response of the equaliser; a value of **0** gives a flat response in the HF region, positive values will give a rise in the HF response – i.e., more treble, and negative values less treble. The range of adjustment is ±12 dB (with **FX1 Amnt** set to **127**).

Parameter: **LF Frequency**
Displayed As: `EQBasFre`
Default value: 64
Range of adjustment: 0 to 127

The equaliser is a "swept" type, which means that in addition to being able to increase or decrease the treble, mid or bass, you can also control the frequency band over which the Cut/Boost controls are effective – i.e., just what is meant by "bass", "mid" and "treble". This gives you much more accurate control over the frequency response. Increasing the value of **EQBasFre** increases the frequency below which **EQBasLvl** is effective, so in general, **EQBasLvl** will have more effect on the sound the higher the value of **EQBasFre**. Decreasing the value of **EQBasFre** will lower the frequency below which the cut/boost control is effective with a value of **0** corresponding to approx. 140 Hz . The maximum value of **127** corresponds to about 880 Hz and the default value of **64** to about 500 Hz.

Parameter: **MF Frequency**
Displayed As: `EQMidFre`
Default value: 64
Range of adjustment: 0 to 127

Increasing the value of this parameter increases the "centre" frequency of the MF response. The centre frequency is the one that gets the maximum amount of cut or boost as you adjust **EQMidLvl**, and this control will have a proportionately decreasing effect on frequencies above and below the centre frequency. The range of adjustment is from 440 Hz (value = **0**) to 2.2 kHz (value = **127**). The default value of **64** corresponds to approximately 1.2 kHz.

Parameter: **HF Frequency**
Displayed As: `EQTrbFre`
Default value: 64
Range of adjustment: 0 to 127

Decreasing the value of **EQTrbFre** decreases the frequency above which **EQTrbLvl** is effective, so in general, **EQTrbLvl** will have more effect on the sound the lower the value of **EQTrbFre**. Increasing the value of **EQTrbFre** will raise the frequency above which the cut/boost control is effective, with a value of **127** corresponding to about 4.4 kHz. A value of **0** corresponds to about 650 Hz and the default value of **64** to about 2 kHz.

Compressor Menu

Two compressor devices are available. Their facilities are identical; the example below illustrates Compressor 1.

The compressors can be used to reduce the dynamic range of the synth sound (or the external audio input), which gives the effect of "thickening" the sound and/or giving it more "punch" or impact. They are particularly effective on sounds with a strong percussive content.

Parameter: **Compression Ratio**
Displayed As: `C1Ratio`
Default value: 1.0
Range of adjustment: 1.0 to 13.7 (0.1 steps)

With the minimum value of **1.0** set, the compressor has no effect as 1.0 means that every change in input level results in an equal change in output level. The parameter sets the degree to which sounds which are louder than the level set by the Threshold level parameter get reduced in volume. If the Ratio is set to **2.0**, a change in input level results in a change in output level of only half the magnitude, thus the overall dynamic range of the signal is reduced. The higher the setting of Compression Ratio, the more compression is applied to those parts of the sound which are above the Threshold level.

Parameter: **Threshold Level**
Displayed As: `C1Thrsh`
Default value: -16
Range of adjustment: -60 to 0

Threshold defines the signal level at which compressor action starts. Signals below the threshold (i.e. the quieter parts of the sound) are unaltered, but signals exceeding the threshold (the louder sections) are reduced in level - in the ratio set with **C1Ratio** - resulting in an overall reduction of the dynamic range of the sound. Note that the value of the parameter approximately represents the actual analogue signal level – i.e., the number of dBs below the maximum digital clip level of 0 dB.

> ℹ Note that any alteration in volume resulting from compressor action has nothing to do with how the output level of the synth is set. Whether you are using the MiniNova's **MASTER VOLUME** control or an Expression pedal to control your overall volume, any compression in the FX section is applied 'before' these volume control methods, and thus will remain constant.

Parameter: **Attack Time**
Displayed As: `C1Attack`
Default value: 0
Range of adjustment: 0 to 127

The **Attack Time** parameter determines how fast the compressor applies gain reduction to a signal exceeding the threshold. With percussive sounds - such as struck drums or plucked bass - it may be desirable to compress the main envelope of the sound while retaining the distinctive front edge or "attack phase" of the sound. A low value gives a fast attack time, and compression will be applied to the front edge of the signal. High values give slow response times, and percussive leading edges will not be compressed, to give a "punchier" sound. The range of attack times available is from 0.1 ms to 100 ms.

Parameter: **Release Time**
Displayed As: `C1Rel`
Default value: 64
Range of adjustment: 0 to 127

This parameter should be adjusted in conjunction with the **Hold Time** parameter (see **C1Hold** below). **Release Time** determines the time period over which the gain reduction is removed (resulting in no compression) after the completion of the **Hold Time**. Low values give a short **Release Time**, high values a long one. The range of release times available is from 25 ms to 1 second.

Parameter: **Hold Time**
Displayed As: `C1Hold`
Default value: 32
Range of adjustment: 0 to 127

Hold Time determines for how long any gain reduction applied to a signal exceeding the Threshold Level remains applied after the signal level drops below the **Threshold Level**.

At the end of the **Hold Time**, the amount of gain reduction is reduced over the **Release Time**. Low values give a short **Hold Time**, high values a long one. The range of hold times available is from 2.5 ms to 500 ms.

> **t** Compressor times are of particular importance with repetitive, rhythmic sounds. For example, setting too short a **Hold Time** may result in audible "pumping" of background noise between notes, which can be quite unpleasant. **Hold**, **Release** and **Attack Times** are usually best adjusted in conjunction with one another, by ear, to obtain an optimum effect with the particular sound you are using.

Parameter: **Auto Gain**
Displayed As: C1Gain
Default value: 127
Range of adjustment: 0 to 127

A consequence of compression is that the overall volume of the sound may be reduced. The MiniNova's compressors automatically "make up" for this loss of level, and ensure that the level of the compressed signal remains as near as possible to that of the input. Auto **Gain** provides additional gain, which may be useful in situations where heavy compression is used.

Distortion Menu

Distortion is usually regarded as something undesirable, and although we all take great pains most of the time to avoid it, there are circumstances when adding some carefully-controlled distortion gives you exactly the sound you're after.

Distortion arises when a signal is passed through a non-linear channel of some kind, the non-linearity producing alterations to the waveform which we hear as distortion. The nature of the circuitry exhibiting the non-linearity dictates the precise nature of the distortion. The MiniNova's distortion algorithms are able to simulate various types of non-linear circuitry, with results ranging from a slight thickening of the sound to something really quite nasty.

> **t** Care should be taken when selecting different Distortion Types, as the same setting of the **FX1 Amnt** parameter will produce very different volumes depending on the Distortion **Type** in use.

The MiniNova has two distortion effect devices. These may be loaded into any two FX slots. Their facilities are identical; the example below illustrates Distortion 1.

Parameter: **Distortion Type**
Displayed As: Dist1Typ
Default value: Diode
Range of adjustment: Diode, Valve, Clipper, XOver, Rectify, BitsDown, RateDown
                         (see below)

- **Diode** - Simulation of analogue circuitry producing distortion where the waveform is progressively "squared-off" as the amount of distortion is increased.
- **Valve** - Simulation of analogue circuitry producing distortion similar to **Diode**, but at extreme settings alternate half-cycles of the waveform are inverted.
- **Clipper** - Simulation of a digital overload.
- **XOver** - Simulation of the crossover distortion generated by bi-polar analogue circuitry, e.g., amplifier output stages.
- **Rectify** - All negative-going half-cycles are inverted, simulating the effect of rectification.
- **BitsDown** - Reproduces the "grainy" quality associated with lower bit rates, as found in older digital devices.
- **RateDown** - Gives the effect of reduced definition and HF loss, similar to the use of a low sample rate.

Parameter: **Distortion Compensation**
Displayed As: Dist1Typ
Default value: 100
Range of adjustment: 0 to 127

Distortion Compensation only has an effect on **Diode** and **Valve** distortion types. Increasing compensation reduces the harshness of the distortion effect.

Delay Menu

The Delay FX processor produces one or more repetitions of the note played. Although the two are intimately related in an acoustic sense, delay should not be confused with reverb in terms of an effect. Think of delay simply as "Echo".

The MiniNova has two delay processors. Their facilities are identical; the example below illustrates Delay 1.

Parameter: **Delay Time**
Displayed As: Dly1Time
Default value: 64
Range of adjustment: 0 to 127

This parameter sets the basic delay time. With **Dly1Sync** (see below) set to Off, the note played will be repeated after a fixed time. Higher values correspond to a longer delay, with

the maximum value of 127 equating to approx. 700 ms. If **Delay Time** is varied (either manually or via modulation), while a note is being played, pitch shifting will result. See also **Delay Slew**.

Parameter: **Delay Sync**
Displayed As: Dly1Sync
Default value: Off
Range of adjustment: See **Sync Values** Table on page 34.

Delay time may synchronised to the internal or external MIDI clock, using a wide variety of tempo dividers/multipliers to produce delays from about 5 ms to 1 second.

> **i** Be aware that the total delay time available is finite. Using large tempo divisions at a very slow tempo rate may exceed the delay time limit.

Parameter: **Delay Feedback**
Displayed As: Dly1Fbck
Default value: 64
Range of adjustment: 0 to 127

The output of the delay processor is connected back to the input, at a reduced level; **Dly1Fbck** sets the level. This results in multiple echoes, as the delayed signal is further repeated. With **Dly1Fbck** set to zero, no delayed signal at all is fed back, so only a single echo results. As you increase the value, you will hear more echoes for each note, though they still die away in volume. Setting the control in the centre of its range (**64**) results in about 5 or 6 audible echoes; at the maximum setting, the repetitions will still be audible after a minute or more.

Parameter: **Delay Left-Right Ratio**
Displayed As: Dly1L/R
Default value: 1/1
Range of adjustment: 1/1, 4/3, 3/4, 3/2, 2/3, 2/1,1/2, 3/1, 1/3, 4/1, 1/4,
                         1/OFF, OFF/1

The value of this parameter is a ratio, and determines how each delayed note is distributed between the left and right outputs. Setting **Dly1L/R** to the default **1/1** value places all echoes centrally in the stereo image. With other values, the larger number represents the delay time, and an echo will be produced at this time in one channel only, depending whether the larger number is to the left of the slash or the right. It will be accompanied by a faster echo in the other channel, at a time defined by the ratio of the two numbers. Values with **OFF** to one side of the slash result in all the echoes being in one channel only.

> **i** The **PanPosn** parameter (the first parameter in the **PanRoute** submenu) sets the overall stereo placement of both the initial note and its delayed repetitions, and takes precedence. This means, for example, that if you select **1/OFF** as the **L/R Ratio**, so that all the echoes are on the left, these echoes will gradually diminish if you set a positive value of **PanPosn**, which pans the signal to the right. When **PanPosn** is at **+63** (fully right), you will hear no echoes at all. However, all of this only applies to FX Slot 1, when **FXRouting** is set to **1**! With other FX Slots and/or slot configurations, you may find that the panning works slightly differently.

Parameter: **Delay Stereo Image Width**
Displayed As: Dly1Widt
Default value: 127
Range of adjustment: 0 to 127

The **Delay Stereo Image** Width parameter is only really relevant to settings of **Delay Left-Right Ratio** which result in the echoes being split across the stereo image. With its default value of 127, any stereo placement of delayed signals will be fully left and fully right. Decreasing the value of **Dly1Wdth** reduces the width of the stereo image and panned echoes are at an intermediate position between centre and fully left or right.

Parameter: **Delay Slew Rate**
Displayed As: Dly1Slew
Default value: Off
Range of adjustment: Off, 1 to 127

Delay Slew Rate only has an effect on the sound when the **Delay Time** is being modulated. Modulating delay time produces pitch-shifting. With DSP-generated delays, very fast changes of delay time are possible, but these may produce unwelcome effects, including digital glitching and clicks. **Delay Slew Rate** effectively slows the applied modulation down, so that any such glitches resulting from trying to change the delay time too rapidly may be avoided. The default value of **Off** corresponds to the maximum rate of change, and the delay time will attempt to follow any modulation accurately. Higher values will produce a smoother effect.

Reverb Menu

The Reverb algorithms add the effect of an acoustic space to a sound. Unlike delay, reverberation is created by generating a dense set of delayed signals, typically with different phase relationships and equalisations applied to re-create what happens to sound in a real acoustic space.

The MiniNova has two reverb processors. Their facilities are identical; the example below illustrates Reverb 1.

Parameter: **Reverb Type**
Displayed As: `Rvb1Type`
Default value: LrgHall
Range of adjustment: Chamber, Small Room, Large Room, Small Hall, Large Hall, Great Hall

MiniNova provides six different reverb algorithms, designed to simulate the reflections which occur in rooms and halls of various sizes.

Parameter: **Reverb Decay**
Displayed As: `Rvb1Dec`
Default value: 90
Range of adjustment: 0 to 127

The **Reverb Decay** parameter sets the basic reverb time of the selected space. It can be thought of as setting the size of the room.

Chorus Menu

Chorus is an effect produced by mixing a continuously delayed version of the signal with the original. The characteristic swirling effect is produced by the Chorus processor's own LFO making very small changes in the delays. The changing delay also produces the effect of multiple voices, some of which are pitch-shifted; this adds to the effect.

The Chorus processor can also be configured as a Phaser, where varying phase shift is applied to the signal in specific frequency bands, and the result remixed with the original signal. The familiar 'swishing' effect is the result.

The MiniNova has four Chorus processors. Their facilities are identical; the example below illustrates Chorus 1. Note that although the parameters are named 'Chorus', they are all effective in both Chorus and Phaser modes.

Parameter: **Chorus Type**
Displayed As: `Ch1Type`
Default value: Chorus
Range of adjustment: Phaser or Chorus
Configures the FX processor as either a Chorus or Phaser.

Parameter: **Chorus Speed**
Displayed As: `Ch1Rate`
Default value: 20
Range of adjustment: 0 to 127

The **Chorus Rate** parameter controls the frequency of the Chorus processor's dedicated LFO. Lower values give a lower frequency, and hence a sound whose characteristic changes more gradually. A slow rate is generally more effective.

Parameter: **Chorus Sync**
Displayed As: `Ch1Sync`
Default value: Off
Range of adjustment: See **Sync Values** Table on page 34.
Chorus Rate may synchronised to the internal or external MIDI clock, using a wide variety of tempos.

Parameter: **Chorus Feedback**
Displayed As: `Ch1Fbck`
Default value: 10
Range of adjustment: -64 to +63
The Chorus processor has its own feedback path between output and input, and a certain amount of feedback will usually need to be applied to get an effective sound. Higher values will generally be needed when Phaser mode is selected. Negative values of Feedback mean that the signal being fed back is phase-reversed.

Parameter: **Chorus Depth**
Displayed As: `Ch1Depth`
Default value: 64
Range of adjustment: 0 to 127
The **Depth** parameter determines the amount of LFO modulation applied to the Chorus delay time, and thus the overall depth of the effect. A value of zero produces no effect.

Parameter: **Chorus Delay**
Displayed As: `Ch1Delay`
Default value: 64
Range of adjustment: 0 to 127
Chorus Delay is the actual delay that is used to generate the chorus/phaser effect. Dynamically altering this parameter will produce some interesting effects, though the difference in sound between different static settings is not marked, unless **Chorus Feedback** is at a high value. The overall effect of **Chorus Delay** is more pronounced in **Phaser** mode.

Modulating **Chorus Delay** with an LFO gives a much richer, dual chorus effect.

Gator Menu

The built-in Gator is a very powerful Novation effect. In essence, it is similar to a Noise Gate, triggered by a repeating pattern derived from the internal or external MIDI clock. This breaks a note up rhythmically. One of six patterns is available by setting the **Gator Mode** parameter; the base patterns have 16 steps, but by combining these in various ways, the Gator Mode setting produces longer, more complex patterns.

The Gator is compatible with Patches prepared on the Novation UltraNova. The UltraNova allows the user to freely create and edit 32-step patterns, including definition of per-step volume, and to save these patterns as part of a Patch. As UltraNova Patches are fully compatible with the MiniNova, these Gator patterns will be replayed correctly if they are imported into a MiniNova.

Gator patterns in the MiniNova may be edited "off-line" using MiniNova Editor software.

Note that for the Gator to have its full effect, the **FX Amount** setting for the Slot into which it is loaded needs to be at maximum - 127. In addition to this, the **FX Routing** configuration will also have a bearing on its audibility.

Parameter: **Gator On/Off**
Displayed As: `GtOn/Off`
Default value: On
Range of adjustment: Off or On
This switches the Gator effect on or off.

Parameter: **Gator Latch**
Displayed As: `GtLatch`
Default value: On
Range of adjustment: Off or On
With the **Latch Off**, a note sounds only while its key is pressed. With **Latch On**, pressing a key will cause the note, modified by its Gator pattern, to sound continuously. It is cancelled by setting **GtLatch** to **Off** again.

Parameter: **Gator Rate Sync**
Displayed As: `GtRSync`
Default value: 8th
Range of adjustment: See **Sync Values** Table on page 34.
The clock driving the Gator's trigger is derived from the MiniNova's master tempo clock and the BPM may be adjusted by the **ARP TEMPO** control [21]. **Gator Rate** may synchronised to the internal or external MIDI clock, using a wide variety of tempos.

Parameter: **Gator Key Sync**
Displayed As: `GtKSync`
Default value: On
Range of adjustment: Off or On
When **Key Sync** is **On**, every time you press a key, the Gator pattern restarts at its beginning.
With Key Sync Off, the pattern continues independently in the background.

Parameter: **Gator Edge Slew**
Displayed As: `GtSlew`
Default value: 16
Range of adjustment: 0 to 127
Gator Edge Slew controls the rise time of the triggering clock. This in turn controls how fast the gate opens and closes and thus whether the note has a sharp attack or a slight 'fade-in" and "fade-out". Higher values of **GtSlew** lengthen the rise time, and thus slow down the gate response.

Parameter: **Gator Hold**
Displayed As: `GtHold`
Default value: 64
Range of adjustment: 0 to 127
The **Gator Hold** parameter controls how long the **Noise Gate** is open for once it has been triggered, and thus the duration of the note heard. Note that this parameter is independent of the clock tempo or **Gator Rate Sync** parameter, and that the note duration set by **GtHold** is constant, whatever speed the pattern is running at.

Parameter: **Gator Left-Right Delay**
Displayed As: `GtL/RDel`
Default value: 0
Range of adjustment: -64 to +63
To enhance the effect of sequenced patterns further, the Gator includes a dedicated delay processor. When set to zero, the notes in the pattern are centrally located in the stereo image. With positive values, the notes are panned hard left and a delayed repetition of the note is panned hard right. The value of the parameter controls the delay time. With negative values, a pre-echo results (an echo which precedes the note). The stereo imaging is the same, with the timed pattern note itself on the left and the pre-echo on the right.
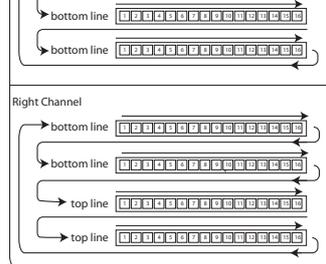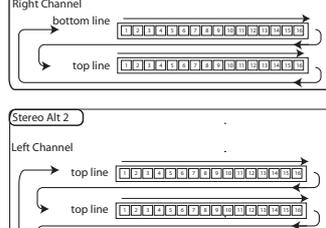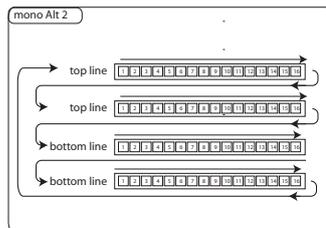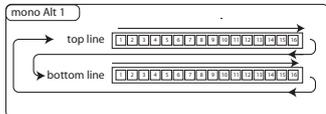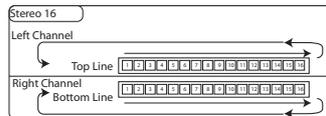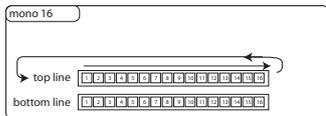
Parameter: **Gator Mode**
Displayed As: `GtMode`
Default value: `Mono16`
Range of adjustment: see Gator Modes Table on page 38.

The Mode parameter allows you to select one of 6 methods of combining the two sets of 16-step Groups, {A} and {B}. Three of the modes are mono, and three are stereo, in which notes in Set {A} are routed to the Left output and those in Set {B} to the Right output.

> The main FX Pan parameters in the first submenu of the Effects menu will override the stereo Gator Modes. The stereo modes will only operate as described if the main FX Pan controls are set centrally.

Edit menu -Submenu 10: `VoxTune`

Parameter: **VocalTune Mode**
Displayed As: `VT Mode`
Default value: Off
Range of adjustment: Off, ScalCorr, KBCtrl, Pitch

VocalTune is a powerful MiniNova feature, which lets you alter the pitch of a signal at the Audio/Mic Input (e.g., your voice through the MiniNova's mic). There are three methods of providing the musical scale that VocalTune uses as a reference when changing the pitch of the audio signal.

- **ScalCorr** - Scale Correction. A fixed scale is selected with the **VT Scale** parameter (below), and a key with **VT Key**. This setting will set the pitch of the Mic input to match that scale.
- **KBCtrl** – Keyboard Control. The keyboard sets the guide pitch based on the last note(s) played. If you play a chord, the audio input will assume the pitch of the nearest note in the chord.
- **Pitch** – Pitch Shift. Adds a fixed amount of pitch shift to the incoming audio. The amount of shift is set using the **PtchShft** parameter. Additional real-time pitch shifting can be controlled using the Pitch Wheel (range being set using the **BendShft** parameter).

Parameter: **VocalTune Scale**
Displayed As: `VT Scale`
Default value: Played
Range of adjustment: Played, Chrmatic, Major, RelMinor, HarMinor, MelMinor

When in Scale Correction mode (with **VT Mode** set to ScalCorr) you can select the scale that Vocal Tune uses as a reference. If **VT Scale** is set to **Played,** the VocalTune will reference the notes in the chord that has been played most recently.

> The more notes in the last chord, the more notes VocalTune has to snap to. A three-note triad doesn't give great results.

> Try working out all the notes that make up a simple melody and play them all at once as a chord. Then, if you sing the melody, VocalTune will snap your vocal only to those notes.

Parameter: **VocalTune Key**
Displayed As: `VT Key`
Default value: C
Range of adjustment: C to B (standard 12-note scale)

Sets the key in which Vocal Tune operates (with **VT Mode** set to ScalCorr and **VT Scale** not set to **Played**).

Parameter: **VocalTune Speed**
Displayed As: `VT Speed`
Default value: 64
Range of adjustment: 0 to 127

Sets the time for Vocal Tune to adjust the pitch of the incoming audio to the target pitch. A value is **0** is slow and **127** is fast.

Parameter: **VocalTune Routing**
Displayed As: `VTInsert`
Default value: PreFX
Range of adjustment: PreFilt, PostFilt, PreFX

This parameter controls the routing of the Vocal Tune output within the synth.

- **PreFilt** – Pre Filter; inserts the pitch-shifted audio (before the filter) into the same mixer audio channel as the Oscillator. The vocal signal will therefore only be heard when a key is pressed (or when a MIDI Note On command is received).
- **PostFilt** – Post Filter; inserts the pitch-shifted audio (after the filter) into the same mixer audio channel as the Oscillator. The vocal signal will still only occur when a key is pressed (or actioned via a MIDI Note On command).
- **PreFX** – Inserts the pitch-shifted audio directly into the FX stage of the MiniNova. With this setting, there is no need to press a key to hear the vocal.

Parameter: **VocalTune Output Level**
Displayed As: `VT Level`
Default value: 127
Range of adjustment: 0 to 127

VT Level sets the output level of the pitch-shifted audio.

Parameter: **VocalTune Vibrato Level**
Displayed As: `VibAmont`
Default value: 0
Range of adjustment: -12 to +12

The **VocalTune** feature incorporates a vibrato effect, which adds additional authenticity to the pitch-shifted audio. **VibAmont** sets the amount of vibrato applied to the pitch-shifted audio.

Parameter: **VocalTune Vibrato Level Via MOD Wheel**
Displayed As: `VibModWl`
Default value: 0
Range of adjustment: -12 to +12

In addition to **VibAmont**, you have the ability to alter the amount of vibrato applied to the pitch-shifted audio in real-time, using the MOD Wheel. **VibModWl** sets the range that can be applied.

Parameter: **VocalTune Vibrato Rate**
Displayed As: `VibRate`
Default value: 80
Range of adjustment: 0 to 127

The rate (speed) of the vibrato applied to both **VibAmont** and **VibModWl**.

Parameter: **VocalTune Pitch Shift**
Displayed As: `PtchShft`
Default value: 0
Range of adjustment: -24 to +24

VocalTune applies both fixed and dynamic pitch shifting. **PtchShft** sets the amount of fixed pitch shift that will be applied to the incoming audio signal. This will be in addition to any pitch shift applied as a result of VocalTune being used to alter the pitch of an incoming audio signal in real time (e.g., **VTMode** settings of **ScalCorr** and **KBCntl**). **PtchShft** intervals are in semitones.

Parameter: **VocalTune Pitch Wheel Range**
Displayed As: `BendShft`
Default value: 12
Range of adjustment: -24 to +24

BendShft sets the range of additional pitch shift available from use of the **Pitch** Wheel. Bend Shift intervals are also in semitones. **VT Modes ScalCorr** and **KBCntl** apply additional correction prior to the Bend Shift stage.

Parameter: VocalTune Gate Threshold
Displayed As: GateThr
Default value: -50
Range of adjustment: -96 to 0

The input channel of the VocalTune feature includes a Noise Gate to help reject unwanted microphone noise. Set **GateThr** to suit the incoming audio source. Values are in dBs.

Parameter: VocalTune Gate Release Time
Displayed As: GateRel
Default value: 64
Range of adjustment: 0 to 127

This parameter sets how long the gate remains open after the signal level has dropped below the value set by **GateThr**. The default value of **64** should suffice for many purposes, but longer or shorter times may be more suitable for certain types of material.

Edit menu - Submenu 11: Vocoder

A Vocoder is a device which analyzes selected frequencies present in an audio signal (called a Modulator) and superimposes these frequencies onto another sound (called the Carrier). It does this by feeding the Modulator signal into a bank of band pass filters. Each of these filters (12 of them in the MiniNova) covers a particular band in the audio spectrum, and the filter bank thus "splits" the audio signal into 12 separate frequency bands. The result of this arrangement is that the spectral content – i.e. the "character" of the audio signal is "imposed" upon the synth sound, and what you hear is a synth sound simulating the audio input (typically a vocal).

The final character of the vocoded sound will depend greatly on the harmonics present in the synth sound used as the Carrier. Patches very rich in harmonics (for example using Sawtooth Waves) will generally give the best results.

Typically, the Modulator signal used by a Vocoder would be a human voice speaking or singing into a microphone. This creates the distinctive robotic or 'talky' like sounds which have recently returned to popularity and are now being used in lots of current music genres. Bear in mind however, that the Modulator signal need not be restricted to human speech. Other types of Modulator signal can be used (for example, an electric guitar or drums) and can often give quite unexpected and interesting results.

The most common way of using the Vocoder is with the dynamic gooseneck mic supplied with the MiniNova (or any other dynamic mic) plugged into the top panel XLR socket [22]. Alternatively, the Modulator signals may be from an instrument or other source connected to the EXT IN socket {32}, located on the rear panel, but remember that a jack plug connected at this input will override the top panel XLR input. The Modulator input to the Vocoder is always mono.

The pitch of the final vocoded sound will depend on the notes that the Carrier (the currently selected Patch) is playing. Notes can either be played on the MiniNova's keyboard or received via MIDI from an external keyboard or sequencer. Both Carrier and Modulator signals must be present simultaneously for the Vocoder effect to work, so notes must be played while the Modulator signal is present. The Vocoder is enabled by selecting a Patch of Type **VOCODER/MIC FX** with the **TYPE/GENRE** knob [4], and controlled from the **VOCODER** submenu.

Submenu: Vocoder
Parameter: Vocoder On/Off
Displayed As: On/Off
Default value: Off
Range of adjustment: On or Off

Enables/disables the Vocoder function.

Parameter: Vocoder Level
Displayed As: VocodLvl
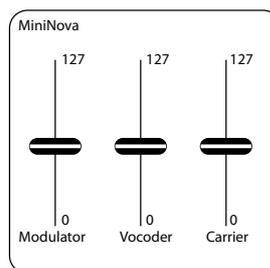**Default value:** 0
Range of adjustment: 0 to 127

Characteristic Vocoder sounds are obtained by blending the Vocoder output with one or other of the two source signals. MiniNova lets you mix the output of the vocoder with either the Modulator signal or the Carrier signal, or both. VocodLvl adjusts the level of the Vocoder output in this mix.

Parameter: Carrier Level
Displayed As: CarriLvl
Default value: 0
Range of adjustment: 0 to 127

CarriLvl adjusts the level of the Carrier signal (the currently selected synth Patch) in the Vocoder output mix.

Parameter: Modulator Level
Displayed As: ModulLvl
Default value: 0
Range of adjustment: 0 to 127

ModulLvl adjusts the level of the microphone (or other external input) that is mixed with the Vocoder output signal.

Parameter: Vocoder Width
Displayed As: VocWidth
Default value: 127
Range of adjustment: 0 to 127

The outputs of each Vocoder filter band are routed to the Left and Right channels alternately to produce a stereo image with good depth. Decreasing the value of **Width** will progressively route all the filter outputs to both outputs, so with **Width** set to zero, the Vocoder output will be in mono and centrally located in the stereo image.

Parameter: Vocoder Mode
Displayed As: VocMode
Default value: Normal
Range of adjustment: Normal, AllMax

The Normal setting produces standard vocoder operation. The modulator signal (usually the microphone input) is analysed to produce driving levels for the **vocoder** carrier synthesis bands. Use this mode if you want the typical 'talking robot' type of sound.

If **VocMode** is set to **AllMax,** no analysis is performed. All the carrier synthesisis bands are set to a high level and this allows the vocoder to be used as a powerful multi-filter effect. Used in conjuction with the other vocoder parameters, in particular **Resonate**, **VocShift** and **VocSpred** (see below), effects ranging from subtle stereo comb filtering and phasing to strange bell like textures can be found. Experiment!

Parameter: Vocoder Freeze Mode
Displayed As: VocFreez
Default value: Off
Range of adjustment: Off or On

With **VocFreez** set to **Off**, normal vocoder operation is available. In this mode, the modulator input (normally the mic) will be constantly analysed by the **vocoder**.

If **VocFreez** is set to **On**, the current levels of the **vocoder** modulator analysis filters will be frozen, and stored. (Imagine taking a single frame from a film as an analogy.) This can be used to 'capture' the mic signal. Factory Patches 'Aaah1' (B073) and 'Aaah2' (B074) use this freeze mode. Note that the frozen formant is stored as part of the Patch data.

Parameter: Vocoder Shift
Displayed As: VocShift
Default value: 0
Range of adjustment: -64 to +63

The **VocShift** parameter changes how the **vocoder modulator** analysis filter band frequencies are mapped to the **carrier** synthesis band frequencies. It offsets **the whole** of the analysis bands by the same amount relative to the synthesis bands. A positive value shifts the **carrier** bands up the frequency spectrum, whereas negative values shifts them downwards.

NORMAL - SHIFT AND SPREAD AT ZERO

SHIFT (+ve)

SHIFT (-ve)
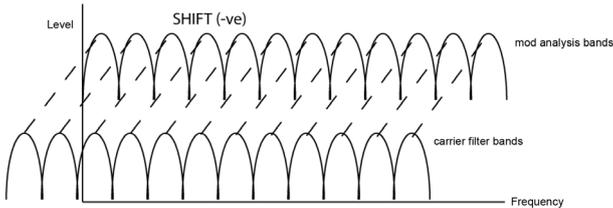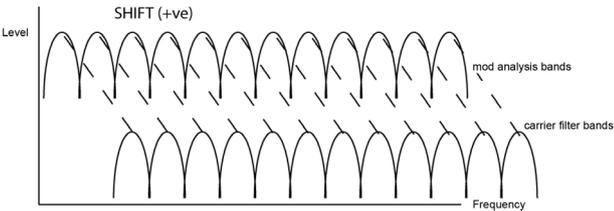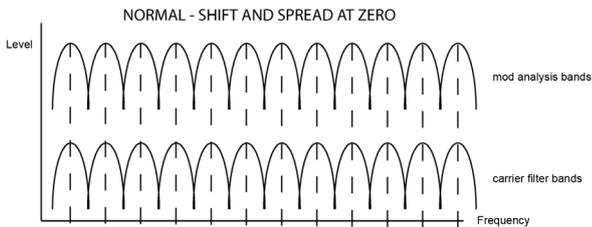
Parameter:	**Vocoder Spread**
Displayed As:	`VocSpred`
Default value:	0
Range of adjustment:	-64 to +63

VocSpred further modifies how the **vocoder modulator** analysis filter band frequencies are mapped to the **carrier** synthesis band frequencies. It increases or decreases the range of frequencies involved (think of 'stretching' and 'shrinking'). Positive values of **VocSpred** **stretch how the frequencies are mapped**, negative values have the opposite effect.

SPREAD (+ve)

SPREAD (-ve)
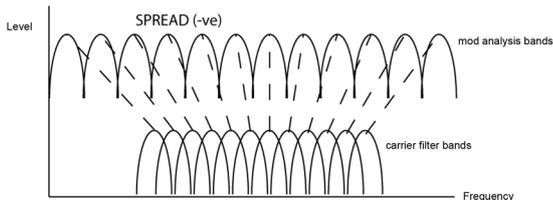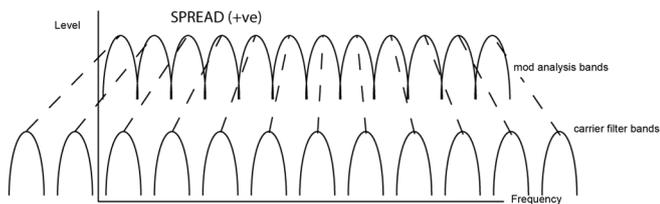
> Both **VocShift** and **VocSpred** drastically alter the tonal output of the **vocoder**. Changing them widely from their default values may have a detrimental effect on the intelligibility of the **vocoder** output, but they are very useful creative tools. Note that both are also mod slot destinations in the **Modulation Matrix**. Great 'moving' vocoder sounds can be achieved by using these destinations.

Parameter:	**Vocoder Resonance**
Displayed As:	`Resonate`
Default value:	0
Range of adjustment:	0 to 127

Resonate sets the amount of resonance that the vocoder synthesis filter bands have. More resonance gives a ringing sound to the vocoder output. Less resonance gives a drier sound.

Parameter:	**Vocoder Decay**
Displayed As:	`VocDecay`
Default value:	0
Range of adjustment:	0 to 127

Controls how long the analysis bands take to close once their threshold has been exceeded. Short decay times aid intelligibility of the **vocoder**. Longer release times are useful for more creative vocoder effects.

Parameter:	**Vocoder Sibilance Type**
Displayed As:	`SibType`
Default value:	HighPass
Range of adjustment:	HighPass or Noise

In the default setting of **HighPass**, sibilance is extracted from the Modulator signal (i.e. the vocalist's natural voice) by filtering. This setting will allow some of the Modulator signal to be heard. If you want to add some sibilance to the vocoded vocals, but the performer's voice just isn't that sibilant naturally, you can artificially simulate sibilance by selecting **Noise** as **Sibilance Type**. This will add a small level of noise to the Modulator signal and the **vocoder** will treat the additional HF content in the same manner as it would natural sibilance.

Parameter:	**Vocoder Sibilance Level**
Displayed As:	`SibLevel`
Default value:	40
Range of adjustment:	0 to 127

This parameter determines the amount sibilance that will be present in the final vocoded signal, and can make the Vocoder emphasise the explosive 'S' and 'T' sounds found in speech. Sibilance can be added to give the Vocoder to give a more distinctive sound and to make vocoded vocals more intelligible.

Parameter:	**Vocoder Noise Gate Threshold**
Displayed As:	`GateThr`
Default value:	-96
Range of adjustment:	-96 to 0

The Modulator signal (from the microphone or external audio input) has a noise gate in the signal path to reject unwanted low level signals. **GateThr** sets the threshold of the gate. This is a valuable feature when using the Vocoder in live performance as it will help prevent Vocoder action being triggered by extraneous sounds picked up by the microphone. The calibration is approximately in dBs below internal clip level (0 dB).

Parameter:	**Vocoder Noise Gate Release Time**
Displayed As:	`GateRel`
Default value:	0
Range of adjustment:	0 to 127

GateRel sets the release time of the **Noise Gate**; that is, how long the **Gate** stays open after the Modulator signal level drops below the level set by **GateThr** (i.e., how long the mic remains live after you stop singing).

Top Menu: Dump

The final menu is where you transfer Patch and other data between the MiniNova and a MIDI-enabled device (hardware or software) that can store MIDI SysEx data.

Parameter:	**Dump Current Patch**
Displayed As:	`DmpCrPch`

On pressing the **OK** button while **DmpCrPch OK?** is displayed, the currently-loaded Patch (i.e., all the current synth Patch parameters) will be transmitted via both the USB and MIDI OUT ports. You can alternatively press **MENU/BACK** if you decide not to go ahead with the dump.

Parameter:	**Set Bank**
Displayed As:	`Set Bank`

Use the **DATA** knob to select Bank A, B or C; on pressing **OK**, you will be asked to confirm whether you want to go ahead and dump the Patch data for all patches in the currently-selected bank.

Parameter:	**Set Patch to dump**
Displayed As:	`SetPatch`

This option lets you dump any Patch in the MiniNova – not necessarily the currently-loaded one. The name of the Patch to dump is displayed on the second row of the LCD. Use the **DATA** knob to select the Patch to dump by name, then use the PAGE ▶ button to select the next menu option:

Parameter:	**Dump Selected Patch**
Displayed As:	`DumpPtch`

Press OK to dump the Patch selected by SetPatch.

Parameter:	**Dump All Patches**
Displayed As:	`Dump All`

Pressing OK while this screen is displayed will dump all 384 Patches (128 x 3 banks). This dump will not include the MiniNova's Global settings (see below).

Parameter:	**Dump Global settings**
Displayed As:	`DumpGlobal`

This function is the complement to **Dump All**; the current Global settings (i.e., audio levels, transposition settings, etc.) will be dumped as a separate write procedure.

Waveform Table

DISPLAY	FORM
Sine	Sine
Triangle	Triangle
Sawtooth	Sawtooth
Saw9:1PW	Sawtooth Pulse Width 9:1 Ratio
Saw8:2PW	Sawtooth Pulse Width 8:2 Ratio
Saw7:3PW	Sawtooth Pulse Width 7:3 Ratio
Saw6:4PW	Sawtooth Pulse Width 6:4 Ratio
Saw5:5PW	Sawtooth Pulse Width 5:5 Ratio
Saw4:6PW	Sawtooth Pulse Width 4:6 Ratio
Saw3:7PW	Sawtooth Pulse Width 3:7 Ratio
Saw2:8PW	Sawtooth Pulse Width 2:8 Ratio
Saw1:9PW	Sawtooth Pulse Width 1:9 Ratio
PW	Pulse Width
Square	Square
BassCamp	Camp Bass
Bass_FM	Frequency Modulated Bass
EP_Dull	Dull Electric Piano
EP_Bell	Bell Electric Piano
Clav	Clavinova
DoubReed	Double Reed
Retro	Retro
StrnMch1	String Machine 1
StrnMch2	String Machine 2
Organ_1	Organ 1
Organ_2	Organ 2
EvilOrg	Evil Organ
HiStuff	High Stuff
Bell_FM1	Frequency Modulated Bell 1
Bell_FM2	Frequency Modulated Bell 2
DigBell1	Digital Bell 1
DigBell2	Digital Bell 2
DigBell3	Digital Bell 3
DigBell4	Digital Bell 4
DigiPad	Digital Pad
Wtable 1	Wavetable 1
Wtable	Wavetable ....
Wtable	Wavetable ....
Wtable36	Wavetable 36
AudioInL/M	Left Audio Input (or Gooseneck Microphone)
AudioInR	Right Audio input

Sync Values Table

DISPLAY	DETAILS	CHORUS SYNC / LFO RATE SYNC / LFO DELAY SYNC / PAN SYNC	ARP SYNC / GATOR SYNC / FX DELAY SYNC
32nd T	48 cycles per 1 bar	✔	✔
32nd	32 cycles per 1 bar	✔	✔
16th T	24 cycles per 1 bar	✔	✔
16th	16 cycles per 1 bar	✔	✔
8th T	12 cycles per 1 bar	✔	✔
16th D	8 cycles per 3 beats / 32 cycles per 3 bars	✔	
8th	8 cycles per 1 bar	✔	✔
4th T	6 cycles per 1 bar	✔	✔
8th D	4 cycles per 3 beats / 16 cycles per 3 bars	✔	✔
4th	4 cycles per 1 bar	✔	✔
1 + 1/3	3 cycles per 1 bar	✔	✔
4th D	2 cycles per 3 beats / 8 cycles per 3 bars	✔	✔
2nd	2 cycles per 1 bar	✔	✔
2 + 2/3	3 cycles per 2 bars	✔	✔
3 Beats	1 cycle per 3 beats / 4 cycles per 3 bars	✔	✔
4 Beats	1 cycles per 1 bar	✔	✔
5 + 1/3	3 cycles per 2 bars	✔	✔
6 Beats	1 cycle per 6 beats / 2 cycles per 3 bars	✔	✔
8 Beats	1 cycle per 2 bars	✔	✔
10 + 2/3	3 cycles per 4 bars	✔	
12 Beats	1 cycle per 12 beats /1 cycle per 3 bars	✔	
13 + 1/3	3 cycles per 10 bars	✔	
16 Beats	1 cycle per 4 bars	✔	
18 Beats	1 cycle per 18 beats /2 cycles per 9 bars	✔	
18 + 2/3	3 cycles per 8 bars	✔	
20 Beats	1 cycle per 5 bars	✔	
21 + 1/3	3 cycles per 16 bars	✔	
24 Beats	1 cycle per 6 bars	✔	
28 Beats	1 cycle per 7 bars	✔	
30 Beats	2 cycles per 15 bars	✔	
32 Beats	1 cycle per 8 bars	✔	
36 Beats	1 cycle per 9 bars	✔	
42 Beats	2 cycles per 21 bars	✔	
48 Beats	1 cycle per 12 bars	✔	
64 Beats	1 cycle per 16 bars	✔	

LFO Waveforms Table

DISPLAY	WAVEFORM	EXTRA INFO
Sine	Traditional LFO shapes	
Triangle		
Sawtooth		
Square		
Rand S/H		Jumps to random values every cycle of the LFO
Time S/H		Jumps to min and max value each held for a random amount of time
PianoEnv		A curved sawtooth shape
Seq 1	These are sequences that jump to different values, holding each for a sixteenth of the LFO cycle rate.	
Seq 2		
Seq 3		
Seq 4		
Seq 5		
Seq 6		
Seq 7		
Altern 1	These are sequences that jump between a minimum and a maximum value, each value held for a varying interval of time.	
Altern 2		
Altern 3		
Altern 4		
Altern 5		
Altern 6		
Altern 7		
Altern 8		
Chromat	These are "melodic" sequences of various kinds. When modulating oscillator pitch, to obtain chromatic results, set Modulation Depth to either ±30 or ±36.	
Major		
Major 7		
Minor 7		
MinArp 1		
MinArp 2		
Diminish		
DecMinor		
Minor3rd		
Pedal		
4ths		
4ths x12		
1625 Maj		
1625 Min		
2511		

Modulation Matrix Sources Table

DISPLAY	SOURCE	COMMENTS
Direct		No modulation source selected.
ModWheel	Mod Wheel	Mod Wheel is the controller.
AftTouch	Aftertouch	Modulation is proportional to the pressure applied to a key while it is held down. (Monophonic aftertouch).*
Express	Expression pedal	An external foot pedal provides the control.
Velocity	Key velocity	Modulation is proprtional to hard the key is played.
Keyboard	Key position	Modulation is proportional to key position.
Lfo1+	LFO 1	'+' = LFO increases value of controlled parameter in a positive sense only. '+/-' = LFO increases and decreases value of controlled parameter equally.
Lfo1+/-		
Lfo2+	LFO 2	
Lfo2+/-		
Lfo3+	LFO 3	
Lfo3+/-		
Env1Amp Env2Filt Env3 - Env6	Envelopes 1 to 6	All six envelopes are triggered by a keypress, and any/all may be used to vary parameters over time. Note that Env1 and Env 2 are "hard-wired" to control Amplitude and Filter parameters, but are still available to control other parameters.
AudInEnv	Audio Input Envelope	Output of Envelope Follower in Mic/Audio Input signal path.

* Note that the MiniNova keyboard does not send Aftertouch data, but the synth engine will correctly respond to any Aftertouch data received via MIDI (via DIN or USB).

Modulation Matrix Destinations Table

DISPLAY	DESTINATION	COMMENTS
	Oscillators:	
O123Ptch	Global oscillator pitch	All Oscillators: Pitch Transpose
O1Pitch	Per-oscillator pitch	Oscillator 1: Pitch Transpose
O2Pitch		Oscillator 2: Pitch Transpose
O3Pitch		Oscillator 3: Pitch Transpose
O1Vsync	Per-oscillator Variable Sync	Oscillator 1: Virtual Sync
O2Vsync		Oscillator 2: Virtual Sync
O3Vsync		Oscillator 3: Virtual Sync
O1PW/Idx	Per-oscillator pulse width/ Wave Table Index	Oscillator 1: Pulsewidth / Wavetable Index
O2PW/Idx		Oscillator 2: Pulsewidth / Wavetable Index
O3PW/Idx		Oscillator 3: Pulsewidth / Wavetable Index
O1Hard	Per-oscillator hardness	Oscillator 1: Hardness
O2Hard		Oscillator 2: Hardness
O3Hard		Oscillator 3: Hardness
	Mixers:	
O1Level	Mixer input levels	Mixer: Oscillator 1 Level
O2Level		Mixer: Oscillator 2 Level
O3Level		Mixer: Oscillator 3 Level
NoiseLvl		Mixer: Noise Level
RM1*3Lvl		Mixer: Ring Mod 1*3 Level
RM2*3Lvl		Mixer: Ring Mod 2*3 Level
	Filters:	
F1DAmnt	Pre-filter distortion, per-filter	Filter 1: Distortion Amount
F2DAmnt	Filter 2: Distortion Amount	
F1Freq	Per-filter frequency	Filter 1: Frequency
F2Freq		Filter 2: Frequency
F1Res	Per-filter resonance	Filter 1: Resonance
F2Res		Filter 2: Resonance
FBalance	Filter 1/Filter 2 balance	Filter Balance
	LFOs:	
L1Rate	Per-LFO frequency	LFO 1: Rate
L2Rate		LFO 2: Rate
L3Rate		LFO 3: Rate
	Envelopes:	
Env1Dec	Envelope Decay Time	Envelope 1 (Amp): Decay Time
Env2Dec		Envelope 2 (Filter): Decay Time
	FX:	
FX1Amnt		FX1: FX Amount
FX2Amnt		FX2: FX Amount
FX3Amnt		FX3: FX Amount
FX4Amnt		FX4: FX Amount
FX5Amnt		FX5: FX Amount
FXFedbac		FX: FX Feedback
FXWetLvl		FX: Wet Level
Ch1Rate	Chorus parameters	Chorus 1: Rate
Ch1Depth		Chorus 1: Depth
Ch1Delay		Chorus 1: Delay
Ch1Fback		Chorus 1: Feedback
Ch2Rate		Chorus 2: Rate
Ch2Depth		Chorus 2: Depth
Ch2Delay		Chorus 2: Delay
Ch2Fback		Chorus 2: Feedback
Ch3Rate		Chorus 3: Rate
Ch3Depth		Chorus 3: Depth
Ch3Delay		Chorus 3: Delay
Ch3Fback		Chorus 3: Feedback
Ch4Rate		Chorus 4: Rate
Ch4Depth		Chorus 4: Depth
Ch4Delay		Chorus 4: Delay
Ch4Fback		Chorus 4: Feedback
Dly1Time	Delay parameters	Delay 1: Delay Time
Dly1Fbak		Delay 1: Feedback
Dly2Time		Delay 2: Delay Time
Dly2Fbak		Delay 2: Feedback
EQBasLvl	EQ settings	EQ: Bass Level
EQBasFrq		EQ: Bass Frequency
EQMidLvl		EQ: Mid Level
EQMidFrq		EQ: Mid Frequency
EQTrbLvl		EQ: Treble Level
EQTrbFrq		EQ: Treble Frequency
PanPosn	Pan Position	Pan: Pan Position
VocShift	Vocoder Shift	
VocSpred	Vocoder Spread	
VocRes	Vocoder Resonance	
PreFXLvl	Pre FX Level	Mixer output level
PitShift	Pitch Shift	Controls dynamic pitch shifting in Vocal Tuning processor

Tweak Parameters Table

DISPLAY	AREA	DETAIL

PortTime		Voice: Portamento Time
FXWetLvl		FX: Wet Level
PstFXLvl		Mixer: Post FX Level
PanPosn		FX: Pan Position
UniDetune		Voice: Unison Detune
	Oscillators:	
O1WTInt	Oscillator 1 parameters	Oscillator 1: Wavetable Interpolation
O1Pw/Idx		Oscillator 1: Pulsewidth / Wavetable Index
O1VSync		Oscillator 1: Virtual Sync
O1Hard		Oscillator 1: Hardness
O1Dense		Oscillator 1: Density
O1DnsDtn		Oscillator 1: Density Detune
O1Semi		Oscillator 1: Semitone Transpose
O1Cents		Oscillator 1: Cents Transpose
O2WTInt	Oscillator 2 parameters	Oscillator 2: Wavetable Interpolation
O2Pw/Idx		Oscillator 2: Pulsewidth / Wavetable Index
O2Vsync		Oscillator 2: Virtual Sync
O2Hard		Oscillator 2: Hardness
O2Dense		Oscillator 2: Density
O2DnsDtn		Oscillator 2: Density Detune
O2Semi		Oscillator 2: Semitone Transpose
O2Cents		Oscillator 2: Cents Transpose

Tweak Parameters Table - Continued

O3WTInt	Oscillator 3 parameters	Oscillator 3: Wavetable Interpolation
O3Pw/Idx		Oscillator 3: Pulsewidth / Wavetable Index
O3Vsync		Oscillator 3: Virtual Sync
O3Hard		Oscillator 3: Hardness
O3Dense		Oscillator 3: Density
O3DnsDtn		Oscillator 3: Density Detune
O3Semi		Oscillator 3: Semitone Transpose
O3Cents		Oscillator 3: Cents Transpose
	Mixer:	
O1Level		Mixer: Oscillator 1 Level
O2Level		Mixer: Oscillator 2 Level
O3Level		Mixer: Oscillator 3 Level
RM1*3Lvl		Mixer: Ring Mod 1*3 Level
RM2*3Lvl		Mixer: Ring Mod 2*3 Level
NoiseLvl		Mixer: Noise Level
	Filters:	
Fbalance		Filter Balance
F1Freq		Filter 1: Frequency
F1Res		Filter 1: Resonance
F1DAmnt		Filter 1: Distortion Amount
F1Track		Filter 1: Keyboard Tracking
F2Freq		Filter 2: Frequency
F2Res		Filter 2: Resonance
F2DAmnt		Filter 2: Distortion Amount
F2Track		Filter 2: Keyboard Tracking
F1Env2		Filter 1: Envelope 2 Amount
F2Env2		Filter 2: Envelope 2 Amount
	Envelope 1:	
AmpAtt		Envelope 1 (Amp): Attack Time
AmpDec		Envelope 1 (Amp): Decay Time
AmpSus		Envelope 1 (Amp): Sustain Level
AmpRel		Envelope 1 (Amp): Release Time
	Envelope 2:	
FltAtt		Envelope 2 (Filter): Attack Time
FltDec		Envelope 2 (Filter): Decay Time
FltSus		Envelope 2 (Filter): Sustain Level
FltRel		Envelope 2 (Filter): Release Time
	Envelope 3:	
E3Delay		Envelope 3: Delay
E3Att		Envelope 3: Attack Time
E3Dec		Envelope 3: Decay Time
E3Sus		Envelope 3: Sustain Level
E3Rel		Envelope 3: Release Time
	LFOs:	
L1Rate		LFO 1: Rate
L1RSync		LFO 1: Sync Rate
L1Slew		LFO 1: Slew Amount
L2Rate		LFO 2: Rate
L2RSync		LFO 2: Sync Rate
L2Slew		LFO 2: Slew Amount
L3Rate		LFO 3: Rate
L3RSync		LFO 3: Sync Rate
L3Slew		LFO 3: Slew Amount

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	FX:	
FX1Amnt		FX1: FX Amount
FX2Amnt		FX2: FX Amount
FX3Amnt		FX3: FX Amount
FX4Amnt		FX4: FX Amount
FX5Amnt		FX5: FX Amount
FXFedbck		FX: FX Feedback
Dst1Lvl	Distortion	Distortion: Distortion 1 Level
Dst2Lvl		Distortion: Distortion 1 Level
Dly1Time	Delay parameters	Delay 1: Delay Time
Dly1Sync		Delay 1: Delay Sync Time
Dly1Fbck		Delay 1: Feedback
Dly1Slew		Delay 1: Slew Amount
Dly2Time		Delay 2: Delay Time
Dly2Sync		Delay 2: Delay Sync Time
Dly2Fbck		Delay 2: Feedback
Dly2Slew		Delay 2: Slew Amount
Ch1Rate	Chorus parameters	Chorus 1: Rate
Ch1Fbck		Chorus 1: Feedback
Ch1Depth		Chorus 1: Depth
Ch1Delay		Chorus 1: Delay
Ch2Rate		Chorus 2: Rate
Ch2Fbck		Chorus 2: Feedback
Ch2Depth		Chorus 2: Depth
Ch2Delay		Chorus 2: Delay
Ch3Rate		Chorus 3: Rate
Ch3Fbck		Chorus 3: Feedback
Ch3Depth		Chorus 3: Depth
Ch3Delay		Chorus 3: Delay
Ch4Rate		Chorus 4: Rate
Ch4Fbck		Chorus 4: Feedback
Ch4Depth		Chorus 4: Depth
Ch4Delay		Chorus 4: Delay
GtSlew	Gator parameters	Gator: Slew Amount
GtDecay		Gator: Decay Time
GtL/RDel		Gator: Left/Right Delay Time
ArpGTime	Arpeggiator parameters	Arpeggiator: Gate Time
ArpSwing		Arpeggiator: Swing
	Modulation Depth:	
M1Depth		Modulation Matrix: Slot 1 Depth
M...Depth		Modulation Matrix: Slot ... Depth
M20Depth		Modulation Matrix: Slot 20 Depth

Filter Table

DISPLAYED AS	DESCRIPTION
LP6NoRes	Lo-pass, 6 dB/oct, no resonance
LP12	Lo-pass, 12 dB/oct
LP18	Lo-pass, 18 dB/oct
LP24	Lo-pass, 24 dB/oct
BP6⁄\6	Symmetric Band-pass, 6 dB/oct
BP12⁄\12	Symmetric Band-pass, 12 dB/oct
BP6⁄\12	Asymmetric Band-pass, 6 dB/oct (hi-pass), 12 dB/oct (lo-pass)
BP12⁄\6	Asymmetric Band-pass, 12 dB/oct (hi-pass), 6 dB/oct (lo-pass)
BP6⁄\18	Asymmetric Band-pass, 6 dB/oct (hi-pass), 18 dB/oct (lo-pass)
BP18⁄\6	Asymmetric Band-pass, 18 dB/oct (hi-pass), 6 dB/oct (lo-pass)
HP6NoRes	Hi-pass, 6 dB/oct, no resonance
HP12	Hi-pass, 12 dB/oct
HP18	Hi-pass, 18 dB/oct
HP24	Hi-pass, 24 dB/oct

Arp Mode Table

DISPLAYED AS	DESCRIPTION	COMMENTS
Up	Ascending	Sequence begins with lowest note played
Down	Descending	Sequence begins with highest note played
Chord	"Polyphonic" mode	All keys held are played simultaneously as a chord
UpDown	Ascend/descend	Sequence alternates
UpDown2		As UpDown, but lowest and highest notes are played twice
Random	Random	The keys held are played in a continuously-varying random sequence
Played	Key order	Sequence comprises notes in the order in which they are played

Gator Modes Table

DISPLAY	MODE	DESCRIPTION
Mono16	16-note mono	16-note mono sequence: {A}
MonoAlt1	32-note mono	32-note mono sequence: {AB}
MonoAlt2	2 x 32-note mono	2 x 16-note sequences, each repeated: {AABB}
Stereo16	16-note stereo	2 x 16-note sequences simultaneously, {A} L, {B} R
SterAlt1	16-note stereo	2 x 16-note sequences simultaneously: {A} L, {B} R, {A} R, {B} L
SterAlt2	16-note stereo	As SterAlt1, but each sequence pair is repeated
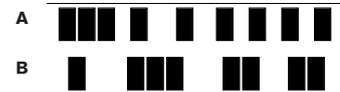
Effects Type Table

DISPLAY	EFFECT	COMMENTS
Bypass	-	No effects enabled
EQ	Equalisation	3-band sweep EQ
Compres1 Compres2	Compression	Compressor with variable threshold and ratio, and variable ADSR
Distort1 Distort2	Overdrive	Adds distortion effects
Delay1 Delay2	Delay line (Echo)	Single and multiple echos
Reverb1 Reverb2	Reverberation	Hall and room simulation
Chorus1 Chorus2 Chorus3 Chorus4	Chorus & Phasing	Time-domain effects
Gator	Gator	8-level, 32-step sequencer

FIRMWARE UPDATES

It is possible to install firmware updates into the MiniNova. To check whether updates are available and to find out how to perform this operation, visit the Novation website (www.novationmusic.com/support).