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.

Compliance Information Statement: Declaration of Compliance procedure

Product Identification: Novation MiniNova
Responsible party: American Music & Sound
Address: 4925 Executive Drive Suite 300
Southaven, MS 38672
Telephone: (800) 431-2609

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 confirmed 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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Audio Units (AU) is a trademark of Apple, Inc.
RTAS is a trademark of Avid, Inc.

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Contents

Important Safety Instructions ........................................ 1
Environmental Declaration ........................................... 1
Copyright and Legal notices ........................................ 1
Introduction .............................................................. 3
Key Features .................................................................. 3
About this manual ........................................................ 3
What’s in the box? .......................................................... 3
Registering your MiniNova .............................................. 3
Power requirements ...................................................... 3
Hardware Overview ...................................................... 4
Top view – controls ....................................................... 4
Rear view – connections ............................................... 5
Getting Started ........................................................... 5
Standalone and computer operation – a foreword .......... 5
Standalone operation – audio and MIDI connections .... 6
Using headphones ......................................................... 6
A word about Menu Navigation ...................................... 6
Scrolling through Patches ............................................. 6
Searching through Types or Genres .................................. 6
Using the FAVORITE button to load Patches .................. 6
Assigning a Patch to a Pad ............................................ 6
Loading a Patch from a Pad ........................................... 6
Demo Mode ................................................................. 6
Modifying Sounds - using the performance controls .... 6
Parameter Controls ...................................................... 6
Rows 1 and 2 – Tweak and (FX) Tweak controls ............. 7
Rows 3 to 6 – Fixed Tweak controls ............................. 7
The Filter knob ......................................................... 7
Using the Pads as performance controls ....................... 7
The Arpeggiator ......................................................... 7
The Vocoder ............................................................ 7
Pitch and Mod wheels ............................................... 7
Octave Shift ........................................................... 8
Storing a Patch ........................................................ 8
Updating the MiniNova’s Operating System .............. 8
Synthesis Tutorial ......................................................... 8
Synth Menus – Reference section ............................... 12
Top Menu: Audio In ................................................... 12
Top Menu: Global ...................................................... 12
Top Menu: Arp .......................................................... 13
Top Menu: Chord ....................................................... 14
Top Menu: Edit .......................................................... 14
Edit menu - Submenu 1: Tweaks ................................. 14
Edit menu - Submenu 2: Osc ....................................... 14
Per-Oscillator parameters ......................................... 14
Common Oscillator parameters ................................. 15
Edit menu - Submenu 3: Mixer ..................................... 16
Edit menu - Submenu 4: Filter ..................................... 16
Per-Filter parameters ............................................... 17
Common Filter parameters ....................................... 17
Edit menu - Submenu 5: Voice .................................... 19
Edit menu - Submenu 6: Env ....................................... 20
Amplitude Envelope ............................................... 20
What is Legato? ....................................................... 21
Common Envelope Parameter ................................... 22
Filter Envelope ........................................................ 22
Envelopes 3 to 6 ....................................................... 23
Edit menu - Submenu 7: LFO ...................................... 24
Edit menu - Submenu 8: ModMatrx ............................ 25
Edit menu - Submenu 9: Effects .................................. 26
EQ Menu ............................................................... 28
Compressor Menu ..................................................... 28
Distortion Menu ....................................................... 29
Delay Menu ............................................................ 29
Reverb Menu .......................................................... 29
Chorus Menu .......................................................... 30
Gator Menu ........................................................... 30
Edit menu - Submenu 10: VoxTune ............................. 31
Edit menu - Submenu 11: Vocoder ............................. 32
Top Menu: Dump ..................................................... 33

Waveform Table ......................................................... 34
Sync Values Table ..................................................... 34
LFO Waveforms Table ............................................... 35
Modulation Matrix Sources Table ............................ 35
Modulation Matrix Destinations Table ....................... 36
Tweak Parameters Table ............................................ 36
Filter Table ............................................................ 38
Arp Mode Table ........................................................ 38
Gator Modes Table ................................................... 38
Effects Type Table .................................................... 38
Firmware updates ..................................................... 39
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, rever, 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.

• MinNova 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/GEND 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].

**ARP section**

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:

<table>
<thead>
<tr>
<th>POSITION</th>
<th>ACTION</th>
</tr>
</thead>
<tbody>
<tr>
<td>ext DC</td>
<td>Enables external 9 V DC input</td>
</tr>
<tr>
<td>OFF</td>
<td>Off</td>
</tr>
<tr>
<td>USB</td>
<td>Enables power via USB port</td>
</tr>
</tbody>
</table>

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.

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 **on**. 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 \(\uparrow\) and \(\downarrow\) 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 \(\uparrow\) and \(\downarrow\) 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:

<table>
<thead>
<tr>
<th>TYPES</th>
<th>GENRES</th>
</tr>
</thead>
<tbody>
<tr>
<td>All</td>
<td></td>
</tr>
<tr>
<td>Vocoder/VocalTune</td>
<td>Rock/Pop</td>
</tr>
<tr>
<td>Bass</td>
<td>R&amp;B/Pop</td>
</tr>
<tr>
<td>Keyboard/Lead</td>
<td>Dubstep</td>
</tr>
<tr>
<td>Pad/Strings</td>
<td>House/Techno</td>
</tr>
<tr>
<td>Arp/Movement</td>
<td>D&amp;B/Breaks</td>
</tr>
<tr>
<td>Classic Synth</td>
<td></td>
</tr>
</tbody>
</table>

### 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 re-use them without having 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 **Assign In**, with a 3 second countdown timer. After 3 seconds, the display changes to **Assign In** 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 \(\uparrow\) and \(\downarrow\) 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 [11]. 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.

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 (“BassSIn Wet 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 “Synchronize 1 PS”, which can be found in the Arp/ModeType menu. 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.

<table>
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</thead>
<tbody>
<tr>
<td>3</td>
<td>Filter</td>
<td>Parameter</td>
<td>Resonance</td>
<td>FILRes</td>
<td>page 17</td>
<td>Tracking</td>
<td>FILTrack</td>
<td>page 17</td>
<td>Type</td>
</tr>
<tr>
<td>4</td>
<td>Filter</td>
<td>Attack</td>
<td>FILATT</td>
<td>page 22</td>
<td>Decay</td>
<td>FILDec</td>
<td>page 22</td>
<td>Sustain</td>
<td>FILSUS</td>
</tr>
<tr>
<td>5</td>
<td>Amplitude</td>
<td>Attack</td>
<td>AMPATT</td>
<td>page 20</td>
<td>Decay</td>
<td>AMPDec</td>
<td>page 20</td>
<td>Sustain</td>
<td>AMPsUS</td>
</tr>
<tr>
<td>6</td>
<td>Oscillator</td>
<td>Osc1</td>
<td>OVS</td>
<td>Sync</td>
<td>page 14</td>
<td>Osc 1 Density</td>
<td>O1Dense</td>
<td>page 15</td>
<td>Osc 2</td>
</tr>
</tbody>
</table>

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 Pad-dependent. Again, the best way of understanding what each Pad can do is to load a Patch and play with them. Lead the Patch “Cry2Moon 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.

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 ‘deslected’ Pads will not illuminate. A ‘deslected’ Pad may be re-enabled by tapping it a second time.

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 (InpGain); 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.

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.
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.

Octave Shift

These two backlight 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.

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.

SYNTHESST TUTORIAL
These contain only odd harmonics. The volume of each decreases as the square of its position in the harmonic series. For example, the 8th harmonic has a volume 1/256th of the volume of the fundamental.

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 distinct 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 8th harmonic has a volume 1/256th of the volume of the fundamental.

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, and 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 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.

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.

50% 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.

40% 60% 10%

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.
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.

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

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.

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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.

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.

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 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 reach its maximum. It can be used to create a sound with a sharp ‘on’ attack.

- **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.

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 violinst 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.
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 I and H 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.

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: Transpose
Default value: 0
Range of adjustment: -24 to +24
This setting adds upper octaves to the arp sequence. If Octv is set to 2, a four-note sequence played with greater than 1 have the effect of doubling, tripling, etc., the length of the sequence. The sequence is played as normal, then immediately played again an octave higher. Higher values of Octv is set to 2.

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 parameter changes in value. Also, the display shows -OPickup 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.

Parameter: Footswitch configuration
Displayed As: FootSwUph
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: Clock Sourc
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 Clock Sourc 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: Auto
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.

Parameter: Arpeggiator Rate Sync
Displayed As: Arp synagogue
Default value: 16th.
Range of adjustment: See Sync Values table “Sync Values Table” on page 34
This parameter effectively determines the beat of the arpeg sequence, based on the current tempo. See “Parameter: Clock Source” on page 19.

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 Ptlh 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 Oclv
Default value: 1 to 4
This setting adds upper octaves to the arpeg sequence. If Arp Oclv is set to 2, the sequence is played as normal, then immediately played again an octave higher. Higher values of Arp Oclv extend this process by adding additional higher octaves. Arp Oclv values greater than 1 have the effect of doubling, tripling, etc., the length of the sequence. The additional octaves added duplicate the complete original sequence, but octave-shifted. Thus a four-note sequence played with Arp Oclv set to 1, will consist of eight notes when Arp Oclv is set to 2.
Parameter: Arpeggiator Pattern
Displayed As: Arr Ptn
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 Ptn 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 Ptn 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.

Parameter: Arpeggiator Length
Displayed As: ArpLen
Default value: 8
Range of adjustment: 1 to 8
This parameter is only available when Arp Ptn 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 Ptn 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 MinNova's Chord 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: Fine tuning
Displayed As: 01Centz
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: 01USync
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.

Edit menu - Submenu 1: Tweaks
Parameter: Tweak Number
Displayed As: Tweak n
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].

Parameter: Coarse tuning
Displayed As: 01Semi
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: 01Centz
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.

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.
A pulse wave is essentially an asymmetric square wave; when set to zero, the harmonic content varies and at high settings the sound becomes quite thin and most easily be heard by adjusting this parameter with pulse waveforms, it varies the pulse width of the oscillator output. This basic effect can be also heard by adjusting the parameter value range of 128 is divided into 9 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 (O1WInt), which can be used to introduce further variation in the way wavetables are used.

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.

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.

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.

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: NoiseType
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 8 inputs and two FX sends are available for adjustment.

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*3Lv1
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*3Lv1 and RM2*3Lv1. The parameter controlled by RM1*3Lv1 sets the amount of the Osc. 1 * 3 Ring Modulator output present in the overall sound.

Parameter: Ring Modulator Level (Oscs. 2 * 3)
Displayed As: RM2*3Lv1
Default value: 0
Range of adjustment: 0 to 127
The parameter controlled by RM2*3Lv1 sets the amount of the Osc. 2 * 3 Ring Modulator output present in the overall sound.

Parameter: Noise Source Level
Displayed As: NoiseLv1
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: PreFXLv1
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 PreFXLv1. This control should be adjusted with care to avoid overloading the FX processing.

Parameter: Post-FX level send
Displayed As: PostFXLv1
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 PreFXLv1 and PostFXLv1 will alter the signal level even when all FX slots in the FX block are bypassed.

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

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're using, the more careful you need to be. If you get this wrong, you risk internal signal clipping, which will sound extremely unpleasant.
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 "center" frequency. Sweeping the filter manually will impose a "hard-to-soft" characteristic on almost any sound.

If Filter Frequency Link is set On (see FreqLink below), F2Freq assumes a different function:
Parameter:  Filter 2 frequency offset
Displayed As:  F41<>F4x2
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.

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
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.

Parameter:  Filter control by Envelope 2
Displayed As:  F1Env
Default value:  0
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.

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

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.

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:

<table>
<thead>
<tr>
<th>PARAMETER</th>
<th>IN MENU</th>
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</tr>
</thead>
<tbody>
<tr>
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<td>FiltrCmn</td>
<td>Series</td>
</tr>
<tr>
<td>FBalance</td>
<td>FiltrCmn</td>
<td>63</td>
</tr>
<tr>
<td>F1DAntl</td>
<td>Filter 1</td>
<td>0</td>
</tr>
<tr>
<td>F2DAntl</td>
<td>Filter 2</td>
<td>As required</td>
</tr>
</tbody>
</table>

Parameter:  Drive Amount
Displayed As:  F1DAntl
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).

F1DAntl can be also adjusted directly from Row 3 of the PERFORM section of the control panel with Tweak Control RC4.

FRouting can also be set to FiltrCmn.

Parameter:  Filter Q Normalisation
Displayed As:  F1QNor
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.

Parameter:  Drive Amount
Displayed As:  Drive Amount
Default value:  F1DAntl
Range of adjustment:  0 to 127
The type of drive (or distortion) generated can be selected with the F1DType parameter.

Parameter:  Filter structure
Displayed As:  F1Struct
Default value:  0
Range of adjustment:  0 to 127
The kind of structure of the filter is set by this parameter. This parameter lets you select the type of filter sections used, as well as the relationship between filters in the MiniNova.

The MiniNova filter section offers 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.

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:

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<td>Filter 2</td>
<td>As required</td>
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</table>

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.

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:

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Parameter:  Drive Amount
Displayed As:  F1DAntl
Default value:  0
Range of adjustment:  0 to 127
The type of drive (or distortion) generated can be selected with the F1DType parameter.

F1DAntl can be also adjusted directly from Row 3 of the PERFORM section of the control panel with Tweak Control RC4.

FRouting can also be set to FiltrCmn.

Parameter:  Filter Q Normalisation
Displayed As:  F1QNor
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.

Parameter:  Drive Amount
Displayed As:  Drive Amount
Default value:  F1DAntl
Range of adjustment:  0 to 127
The type of drive (or distortion) generated can be selected with the F1DType parameter.

Parameter:  Filter structure
Displayed As:  F1Struct
Default value:  0
Range of adjustment:  0 to 127
The kind of structure of the filter is set by this parameter. This parameter lets you select the type of filter sections used, as well as the relationship between filters in the MiniNova.

The MiniNova filter section offers 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.

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:

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</tbody>
</table>

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.

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:

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</table>

Parameter:  Drive Amount
Displayed As:  F1DAntl
Default value:  0
Range of adjustment:  0 to 127
The type of drive (or distortion) generated can be selected with the F1DType parameter.

F1DAntl can be also adjusted directly from Row 3 of the PERFORM section of the control panel with Tweak Control RC4.

FRouting can also be set to FiltrCmn.

Parameter:  Filter Q Normalisation
Displayed As:  F1QNor
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.

Parameter:  Drive Amount
Displayed As:  Drive Amount
Default value:  F1DAntl
Range of adjustment:  0 to 127
The type of drive (or distortion) generated can be selected with the F1DType parameter.

Parameter:  Filter structure
Displayed As:  F1Struct
Default value:  0
Range of adjustment:  0 to 127
The kind of structure of the filter is set by this parameter. This parameter lets you select the type of filter sections used, as well as the relationship between filters in the MiniNova.

The MiniNova filter section offers 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.
Parameters:
- **Filter Routing**
  - Displayed As: FilterRouting
  - 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 filters 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:  

<table>
<thead>
<tr>
<th>F1Type:</th>
<th>LP</th>
</tr>
</thead>
<tbody>
<tr>
<td>F2Type:</td>
<td>HP</td>
</tr>
<tr>
<td>F1Routin9:</td>
<td>Parallel</td>
</tr>
<tr>
<td>F2Routin9:</td>
<td>Series</td>
</tr>
</tbody>
</table>

...a wideband bandpass filter  

<table>
<thead>
<tr>
<th>F1Type:</th>
<th>HP</th>
</tr>
</thead>
<tbody>
<tr>
<td>F2Type:</td>
<td>LP</td>
</tr>
<tr>
<td>F1Routin9:</td>
<td>Parallel</td>
</tr>
<tr>
<td>F2Routin9:</td>
<td>Series</td>
</tr>
</tbody>
</table>

F1Res is set to zero, but the sound gets more interesting as it is increased (see following page) transitions from below), where must be set to zero.

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

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

Volume

Frequency

Filter 1

Filter 2

Portamento Time

Default value: Off

Range of adjustment: Off 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.

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:  

UniDTune

Displayed As: UniDTune

Default value: 25

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.

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.

Changing the settings of UniDTune applied 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.
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 sub-menu, it is first necessary to select the envelope whose parameters are to be adjusted:

**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 (AmpAtSlo) has a value of zero).

**Amplitude Decay Time**

**Parameter:** AmpDec

**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 AmpDecSlo has a value of 127).

**Amplitude Sustain Level**

**Parameter:** AmpSus

**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.

**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.
Amplitude Release Time
Displayed As: RnReRl
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.

Amplitude Multi-Trigger
Displayed As: RnHRTr;1
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.

AmpReL can be also adjusted directly from Row 5 of the PERFORM section of the control panel with Tweak Control RC4.

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.

Amplitude Velocity
Displayed As: RnVlEloc
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.

For the most “natural” playing style, try setting Amplitude Velocity to about +40.

Amplitude Envelope Repeat
Displayed As: RnReP;
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 no repeats. The maximum setting of KeyOff generates an infinite number of repeats.

Amplitude Touch Trigger
Displayed As: RnTTr;1
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 Attack Slope
Displayed As: RnApStTk
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: RnApDecTk
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:

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 Sustain Rate
Displayed As: RnApSusRt
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.

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.
The effect should be used sparingly; low values have a better effect.

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: LuLThNnLe
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.

In order to hear the effect of any of the Filter Envelope parameters, you will first need to go to the Filter Menu 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.

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.

Filter Decay Time
Parameter:
Display 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.

This parameter can be also adjusted directly from Row 4 of the PERFORM section of the control panel with Tweak Control RC1.

Filter Sustain Level
Parameter:
Display 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.

This parameter can be also adjusted directly from Row 4 of the PERFORM section of the control panel with Tweak Control RC2.

Filter Re-trigger
Parameter:
Display As: FltRRep
Default value: Off
Range of adjustment: -64 to +63

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

This parameter can be also adjusted directly from Row 4 of the PERFORM section of the control panel with Tweak Control RC3.

Filter Velocity
Parameter:
Display 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.

Filter Repeat
Parameter:
Display As: FltRRep
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.

Filter Touch Trigger
Parameter:
Display As: FltTrg
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: F14Trig19
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.

---

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

Parameter: Filter Attack Slope
Displayed As: F14AtSlp
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: F14DecSlp
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: F14AttTrk
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: F14DecTrk
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: F14Susrat
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: F14SusTim
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 Sustain Rate
Displayed As: F14SusTim
Default value: KeyOff
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: LVTNote
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.

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**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 – 0123Pich).

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: E3At
Default value: 0
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: E3Del
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 1 second.

Parameter: Envelope 3 Repeat
Displayed As: E3Repeat
Default value: Off
Range of adjustment: Off, 1 to 126, KeyOff

23
The MiniNova has three separate Low Frequency Oscillators (LFOs). These are designated LFO1, 2 and 3, and 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.

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

- **LFO1 Rate**
  - Displayed As: L1Rate
  - Default value: Off
  - Range of adjustment: On, Off
  - Default value: Off
  - 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.
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.

NOTE 1

**Slew** is only applicable to polyphonic voices. It ensures that the phase of the LFO is synchronised for every note being played. When to set **LFO 1 Common Sync** to **On** re-starts the LFO at **On** setting of **L1Phase** (or MIDI clock **COMMON SYNC = OFF**). In **On** mode, it is only the first note of a Legato style – the onset of the LFO’s modulation is delayed by the time period set by **L1Delay** parameter, leaving the note with no LFO modulation.

NOTE 2

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

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

1. **Fadeln** - the LFO’s modulation is gradually increased over the time period set by **L1Delay** parameter.
2. **GateOut** – the note is fully modulated by the LFO for the time period set by the **L1Delay** parameter.
3. **GateIn** – the note is fully modulated by the LFO for the time period set by the **L1Delay** parameter. At this time, the modulation stops abruptly.
4. **L1Delay** - 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°.

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**).

The menu may be visualised as a system for connecting controlling sources to a specific area of theynth. 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.

**Edit menu - Submenu 8: ModMatrx**

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

1. **Fadeln** - the LFO’s modulation is gradually increased over the time period set by **L1Delay** parameter.
2. **GateOut** – the note is fully modulated by the LFO for the time period set by the **L1Delay** parameter.
3. **GateIn** – the note is fully modulated by the LFO for the time period set by the **L1Delay** parameter. At this time, the modulation stops abruptly.
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The Modulation Matrix is both variable and additive. What do we mean by a ‘variable’ and ‘additive’ matrix?

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.

You need to be careful when setting up patches like this to ensure that the combined effect of all the controllers acting simultaneously still creates the sound that you want.

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:

- Parameter: Displayed As: Default value: Range of adjustment: ModSlt1n (where n is 1 to 20) ModSlt1 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.

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: Displayed As: Default value: Range of adjustment: Source1 ModSlt1n Off 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 (0123Pitch)
  - 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 (Env1/Env2) and Filter Envelope (Env2/Env2)

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.

Depth effectively defines the "amount" by which the controlled parameter varies when under modulation control. Think of it as the "range" of control. It also determines the "sense" or polarity of the control – positive Depth will increase the value of the controlled parameter and negative Depth will decrease it, for the same control input. Note that having defined source and destination in a patch, no modulation will occur until the Depth control is set to something other than zero.

With both sources set to Direct and TouchSel set to Off, the Depth control becomes a "manual" modulation control which will always affect whichever parameter is set as the Destination.

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 FXSlt1 to FXSlt5. PanRoute provides selection of panning and slot configuration. Entering FXSlt1 to FXSlt5 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

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

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

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.

**Parameter: Pan Depth**
- Displayed As: PanDefTh
- 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: FXRoutin9
- 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.

- **FX Routing = 0**
  - FX Slot 1
  - FX Slot 2
  - FX Slot 3
  - FX Slot 4
  - FX Slot 5

- **FX Routing = 1**
  - FX Slot 1
  - FX Slot 2
  - FX Slot 3
  - FX Slot 4
  - FX Slot 5

- **FX Routing = 2**
  - FX Slot 1
  - FX Slot 2
  - FX Slot 3
  - FX Slot 4
  - FX Slot 5

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.

<table>
<thead>
<tr>
<th>FX TYPE</th>
<th>ADJUSTED PARAMETER</th>
</tr>
</thead>
<tbody>
<tr>
<td>Compressor</td>
<td>Level</td>
</tr>
<tr>
<td>EQ</td>
<td>Level</td>
</tr>
<tr>
<td>Distortion</td>
<td>Amount or bit/sample rate reduction</td>
</tr>
<tr>
<td>Delay</td>
<td>Send and Return levels</td>
</tr>
<tr>
<td>Chorus</td>
<td>Level</td>
</tr>
<tr>
<td>Reverb</td>
<td>Send and Return levels</td>
</tr>
<tr>
<td>Gater</td>
<td>Level</td>
</tr>
</tbody>
</table>

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.

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 treble, 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: EQBasFr  
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 EQBasFr increases the frequency below which EQBasLvl is effective, so in general, EQBasLvl will have more effect on the sound the higher the value of EQBasFr. Decreasing the value of EQBasLvl will lower the frequency below which the cut/boost control is effective with a value of 0 corresponding to approx. 140 Hz. The value of 127 corresponds to about 880 Hz and the default value of 64 to about 500 Hz.

Parameter: MF Frequency  
Displayed As: EQMidFr  
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: EQTrbFr  
Default value: 64  
Range of adjustment: 0 to 127

Decreasing the value of EQTrbFr decreases the frequency above which EQTrbLvl is effective, so in general, EQTrbLvl will have more effect on the sound the lower the value of EQTrbFr. Increasing the value of EQTrbFr 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: CIRatio  
Default value: 1.0  
Range of adjustment: 0.1 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: CIThresh  
Default value: 0  
Range of adjustment: 0 to 127

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 CIRatio - 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.

Parameter: Attack Time  
Displayed As: CIRate  
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: CIRel  
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.

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.

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.

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 Type
- **Diode** - Simulation of analogue circuitry producing distortion where the waveform is progressively "squashed" 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 non-going-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.

Distortion Compensation
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.

Delay Time
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.

Distortion Menu

Distortion Type

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

Distortion Menu
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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 "squashed" as the amount of distortion is increased.
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**Parameter:** Delay Sync  
**Displayed As:** Dly1Sinc  
**Default value:** Off  
**Range of adjustment:** 0 to 127

Delay time may be 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.

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

Delay Left-Right Ratio
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.

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

Delay Stereo Image Width
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.

**Parameter:** Delay Stereo Image Width  
**Displayed As:** Dly1Wdth  
**Default value:** 127  
**Range of adjustment:** 0 to 127

Delay Slew Rate
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 unwanted 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.

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

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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: **RvbtTyp**
Default value: LrgHll
Range of adjustment: Chamber, Small Room, Large Room, Small Hall, Large Hall, Great Hall

The 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: **RvbdDec**
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: **Ch1Typ**
Default value: Chorus
Range of adjustment: Phaser or Chorus

Configures the FX processor as either a Chorus or Phaser.

Parameter: **Chorus Speed**
Displayed As: **Ch1Rat**
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: **Ch1Snc**
Default value: Off
Range of adjustment: See Sync Values Table on page 34.

**Chorus Rate** may be synchronised to the internal or external MIDI clock, using a wide variety of tempos.

Parameter: **Chorus Feedback**
Displayed As: **Ch1Fbc**
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: **Ch1Del**
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: **GlOn/Off**
Default value: On
Range of adjustment: Off or On

This switches the Gator effect on or off.

Parameter: **Gator Latch**
Displayed As: **GlLatch**
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 **GlLatch** to Off again.

Parameter: **Gator Rate Sync**
Displayed As: **GlrSyc**
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 be synchronised to the internal or external MIDI clock, using a wide variety of tempos.

Parameter: **Gator Key Sync**
Displayed As: **GksSyc**
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: **GleSlew**
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 **GleSlew** lengthen the rise time, and thus slow down the gate response.

Parameter: **Gator Hold**
Displayed As: **GltHol**
Default value: 64
Range of adjustment: 0 to 127

**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 **GltHol** is constant, whatever speed the pattern is running at.

Parameter: **Gator Left-Right Delay**
Displayed As: **GlR/Rd**
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: VocalTune Mode
Displayed As: UT 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 PitchShift parameter. Additional real-time pitch shifting can be controlled using the Pitch Wheel (range being set using the BendShift parameter).

Parameter: VocalTune Scale
Displayed As: UT Scale
Default value: Played
Range of adjustment: Played, Chromatic, Major, RelMinor, HarMinor, MelMinor
When in Scale Correction mode (with VT Mode set to ScalCorr) you can select the scale that VocalTune 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.

Parameter: VocalTune Pitch Shift
Displayed As: PitchShift
Default value: 0
Range of adjustment: -24 to +24
VocalTune applies both fixed and dynamic pitch shifting. PitchShift 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 KBControl). PitchShift intervals are in semitones.

Parameter: VocalTune Pitch Wheel Range
Displayed As: BendShift
Default value: 12
Range of adjustment: -24 to +24
BendShift 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 KBControl apply additional correction prior to the Bend Shift stage.

Parameter: VocalTune Key
Displayed As: UT Key
Default value: C
Range of adjustment: C to B (standard 12-note scale)
Sets the key in which VocalTune operates (with VT Mode set to ScalCorr and VT Scale not set to Played).

Parameter: VocalTune Speed
Displayed As: UT Speed
Default value: 64
Range of adjustment: 0 to 127
Sets the time for VocalTune to adjust the pitch of the incoming audio to the target pitch. A value of 0 is slow and 127 is fast.

Parameter: VocalTune Routing
Displayed As: UTInsert
Default value: PreFX
Range of adjustment: PreFilt, PostFilt, PreFX
This parameter controls the routing of the VocalTune 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: VibRate
Default value: 80
Range of adjustment: 0 to 127
The rate (speed) of the vibrato applied to both VibAmont and VibModWI can be applied.

Parameter: VocalTune Vibrato Rate
Displayed As: VibRate
Default value: 80
Range of adjustment: 0 to 127
VibAmont and VibModWI sets the range that can be applied.

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. VibModWI sets the range that can be applied.

Parameter: VocalTune Pitch Shift
Displayed As: PitchShift
Default value: 0
Range of adjustment: -24 to +24
VocalTune applies both fixed and dynamic pitch shifting. PitchShift 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 KBControl). PitchShift intervals are in semitones.

Parameter: VocalTune Pitch Wheel Range
Displayed As: BendShift
Default value: 12
Range of adjustment: -24 to +24
BendShift 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 KBControl apply additional correction prior to the Bend Shift stage.

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.

**VocalTune Mode**
- **VoxTune**
- **VT Scale**
- **VocalTune Scale**
- **VT Mode**
- **VocalTune Key**
- **VocalTune Speed**
- **VocalTune Routing**
- **VocalTune Output Level**
- **VocalTune Vibrato Level**
- **VocalTune Vibrato Rate**
- **VocalTune Vibrato Level Via MOD Wheel**
- **VocalTune Pitch Shift**
- **VocalTune Pitch Wheel Range**

**Edit menu - Submenu 10: VoxTune**

**Parameter:** VocalTune Mode
**Displayed As:** UT 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 PitchShift parameter. Additional real-time pitch shifting can be controlled using the Pitch Wheel (range being set using the BendShift parameter).

**Parameter:** VocalTune Scale
**Displayed As:** UT Scale
**Default value:** Played
**Range of adjustment:** Played, Chromatic, Major, RelMinor, HarMinor, MelMinor
When in Scale Correction mode (with VT Mode set to ScalCorr) you can select the scale that VocalTune 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.

**Parameter:** VocalTune Pitch Shift
**Displayed As:** PitchShift
**Default value:** 0
**Range of adjustment:** -24 to +24
VocalTune applies both fixed and dynamic pitch shifting. PitchShift 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 KBControl). PitchShift intervals are in semitones.

**Parameter:** VocalTune Pitch Wheel Range
**Displayed As:** BendShift
**Default value:** 12
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BendShift 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 KBControl apply additional correction prior to the Bend Shift stage.
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 synthetic sound, and what you hear is a synthetic 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 singing into a microphone. This creates the distinctive robotic or ‘talky like’ sounds which

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 EKT 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

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Displayed As</th>
<th>Default value</th>
<th>Range of adjustment</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>VocalTune Gate Threshold</td>
<td>GateThr</td>
<td>-50</td>
<td>-96 to 0</td>
<td>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.</td>
</tr>
<tr>
<td>VocalTune Gate Release Time</td>
<td>GateRel</td>
<td>64</td>
<td>0 to 127</td>
<td>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.</td>
</tr>
<tr>
<td>Modulator Level</td>
<td>ModuliLv1</td>
<td>0</td>
<td>0 to 127</td>
<td>ModuliLv1 adjusts the level of the microphone (or other external input) that is mixed with the Vocoder output signal.</td>
</tr>
<tr>
<td>Vocoder On/Off</td>
<td>On/Off</td>
<td>Off</td>
<td>On or Off</td>
<td>Enables/disables the Vocoder function.</td>
</tr>
<tr>
<td>Vocoder Level</td>
<td>VocoderLv1</td>
<td>0</td>
<td>0 to 127</td>
<td>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. VocoderLv1 adjusts the level of the Vocoder output in this mix.</td>
</tr>
<tr>
<td>Carrier Level</td>
<td>CarrierLv1</td>
<td>0</td>
<td>0 to 127</td>
<td>CarrierLv1 adjusts the level of the Carrier signal (the currently selected synth Patch) in the Vocoder output mix.</td>
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</table>

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.

Parameter: Modulator Level  
Displayed As: ModuliLv1  
Default value: 0  
Range of adjustment: 0 to 127  
ModuliLv1 adjusts the level of the microphone (or other external input) that is mixed with the Vocoder output signal.

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: VocoderLv1  
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. VocoderLv1 adjusts the level of the Vocoder output in this mix.

Parameter: Carrier Level  
Displayed As: CarrierLv1  
Default value: 0  
Range of adjustment: 0 to 127  
CarrierLv1 adjusts the level of the Carrier signal (the currently selected synth Patch) in the Vocoder output mix.

### Submenu: Vocoder

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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.

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.

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.

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.

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.

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).

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 microphone signal 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.

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

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Vocoder Noise Gate Release Time
- **Displayed As:** GateRel
- **Default value:** 0
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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).

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.

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.

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.

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.

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.

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).

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.

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.

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.
### Waveform Table

<table>
<thead>
<tr>
<th>DISPLAY</th>
<th>FORM</th>
</tr>
</thead>
<tbody>
<tr>
<td>Sine</td>
<td>Sine</td>
</tr>
<tr>
<td>Triangle</td>
<td>Triangle</td>
</tr>
<tr>
<td>Sawtooth</td>
<td>Sawtooth</td>
</tr>
<tr>
<td>Saw9:1PW</td>
<td>Sawtooth Pulse Width 9:1</td>
</tr>
<tr>
<td>Saw8:2PW</td>
<td>Sawtooth Pulse Width 8:2</td>
</tr>
<tr>
<td>Saw7:3PW</td>
<td>Sawtooth Pulse Width 7:3</td>
</tr>
<tr>
<td>Saw6:4PW</td>
<td>Sawtooth Pulse Width 6:4</td>
</tr>
<tr>
<td>Saw5:5PW</td>
<td>Sawtooth Pulse Width 5:5</td>
</tr>
<tr>
<td>Saw4:6PW</td>
<td>Sawtooth Pulse Width 4:6</td>
</tr>
<tr>
<td>Saw3:7PW</td>
<td>Sawtooth Pulse Width 3:7</td>
</tr>
<tr>
<td>Saw2:8PW</td>
<td>Sawtooth Pulse Width 2:8</td>
</tr>
<tr>
<td>Saw1:9PW</td>
<td>Sawtooth Pulse Width 1:9</td>
</tr>
<tr>
<td>PWM</td>
<td>Pulse Width</td>
</tr>
<tr>
<td>Square</td>
<td>Square</td>
</tr>
<tr>
<td>BassCamp</td>
<td>Camp Bass</td>
</tr>
<tr>
<td>Bass_FM</td>
<td>Frequency Modulated Bass</td>
</tr>
<tr>
<td>EP_Dull</td>
<td>Dull Electric Piano</td>
</tr>
<tr>
<td>EP_Bell</td>
<td>Bell Electric Piano</td>
</tr>
<tr>
<td>Clav</td>
<td>Clavinova</td>
</tr>
<tr>
<td>DoubleReed</td>
<td>Double Reed</td>
</tr>
<tr>
<td>Retro</td>
<td>Retro</td>
</tr>
<tr>
<td>StrnMch1</td>
<td>String Machine 1</td>
</tr>
<tr>
<td>StrnMch2</td>
<td>String Machine 2</td>
</tr>
<tr>
<td>Organ_1</td>
<td>Organ 1</td>
</tr>
<tr>
<td>Organ_2</td>
<td>Organ 2</td>
</tr>
<tr>
<td>Evil10or9</td>
<td>Evil Organ</td>
</tr>
<tr>
<td>HiStuff</td>
<td>High Stuff</td>
</tr>
<tr>
<td>Bell_FM1</td>
<td>Frequency Modulated Bell 1</td>
</tr>
<tr>
<td>Bell_FM2</td>
<td>Frequency Modulated Bell 2</td>
</tr>
<tr>
<td>DigBell1</td>
<td>Digital Bell 1</td>
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<tr>
<td>DigBell2</td>
<td>Digital Bell 2</td>
</tr>
<tr>
<td>DigBell3</td>
<td>Digital Bell 3</td>
</tr>
<tr>
<td>DigBell4</td>
<td>Digital Bell 4</td>
</tr>
<tr>
<td>DigiPad</td>
<td>Digital Pad</td>
</tr>
<tr>
<td>Wtable 1</td>
<td>Wavetable 1</td>
</tr>
<tr>
<td>Wtable ....</td>
<td>Wavetable ...</td>
</tr>
<tr>
<td>Wtable36</td>
<td>Wavetable 36</td>
</tr>
<tr>
<td>AudioInL/M</td>
<td>Left Audio Input (or Gooseneck Microphone)</td>
</tr>
<tr>
<td>AudioInR</td>
<td>Right Audio Input</td>
</tr>
</tbody>
</table>

### Sync Values Table

<table>
<thead>
<tr>
<th>DISPLAY</th>
<th>DETAILS</th>
<th>CHORUS SYNC</th>
<th>LFO RATE SYNC</th>
<th>LFO DELAY SYNC</th>
<th>PAN SYNC</th>
<th>ARP SYNC</th>
<th>GATOR SYNC</th>
<th>FX DELAY SYNC</th>
</tr>
</thead>
<tbody>
<tr>
<td>32nd T</td>
<td>48 cycles per 1 bar</td>
<td>✔</td>
<td></td>
<td></td>
<td></td>
<td></td>
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<td>✔</td>
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<tr>
<td>16th T</td>
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<td>✔</td>
<td></td>
<td></td>
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<tr>
<td>16th</td>
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<td>✔</td>
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<td></td>
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<td>8th T</td>
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<td>✔</td>
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<td></td>
</tr>
<tr>
<td>16th D</td>
<td>8 cycles per 3 beats / 32 cycles per 3 bars</td>
<td>✔</td>
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<td></td>
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<td>8 cycles per 1 bar</td>
<td>✔</td>
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</tr>
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<td>4th T</td>
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<td></td>
</tr>
<tr>
<td>8th D</td>
<td>4 cycles per 3 beats / 16 cycles per 3 bars</td>
<td>✔</td>
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<td></td>
<td></td>
</tr>
<tr>
<td>4th</td>
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<td>✔</td>
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<td></td>
<td></td>
<td></td>
<td></td>
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</tr>
<tr>
<td>1 + 1/3</td>
<td>3 cycles per 1 bar</td>
<td>✔</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>4th D</td>
<td>2 cycles per 3 beats / 8 cycles per 3 bars</td>
<td>✔</td>
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<td></td>
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</tr>
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<td>2nd</td>
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</tr>
<tr>
<td>2 + 2/3</td>
<td>3 cycles per 2 bars</td>
<td>✔</td>
<td></td>
<td></td>
<td></td>
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<td></td>
<td></td>
</tr>
<tr>
<td>3 Beats</td>
<td>1 cycle per 3 beats / 4 cycles per 3 bars</td>
<td>✔</td>
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<td></td>
<td></td>
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<td></td>
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</tr>
<tr>
<td>4 Beats</td>
<td>1 cycle per 1 bar</td>
<td>✔</td>
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<td>5 + 1/3</td>
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<td>✔</td>
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</tr>
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<td>6 Beats</td>
<td>1 cycle per 6 beats / 2 cycles per 3 bars</td>
<td>✔</td>
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<td></td>
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<td>1 cycle per 1 bar</td>
<td>✔</td>
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<tr>
<td>10 + 2/3</td>
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<tr>
<td>12 Beats</td>
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<td>✔</td>
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<tr>
<td>16 Beats</td>
<td>1 cycle per 4 bars</td>
<td>✔</td>
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</tr>
<tr>
<td>18 Beats</td>
<td>1 cycle per 18 beats / 2 cycles per 9 bars</td>
<td>✔</td>
<td></td>
<td></td>
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<td>18 + 2/3</td>
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</tr>
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<td>20 Beats</td>
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<td>✔</td>
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<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>21 + 1/3</td>
<td>3 cycles per 16 bars</td>
<td>✔</td>
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<td></td>
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<td></td>
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</tr>
<tr>
<td>24 Beats</td>
<td>1 cycle per 6 bars</td>
<td>✔</td>
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</tr>
<tr>
<td>28 Beats</td>
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<td>✔</td>
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</tr>
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<td>30 Beats</td>
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<td>✔</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>32 Beats</td>
<td>1 cycle per 8 bars</td>
<td>✔</td>
<td></td>
<td></td>
<td></td>
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<td></td>
<td></td>
</tr>
<tr>
<td>36 Beats</td>
<td>1 cycle per 9 bars</td>
<td>✔</td>
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<tr>
<td>42 Beats</td>
<td>2 cycles per 21 bars</td>
<td>✔</td>
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</tr>
<tr>
<td>48 Beats</td>
<td>1 cycle per 12 bars</td>
<td>✔</td>
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</tr>
<tr>
<td>64 Beats</td>
<td>1 cycle per 16 bars</td>
<td>✔</td>
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</table>
### LFO Waveforms Table

<table>
<thead>
<tr>
<th>DISPLAY</th>
<th>WAVEFORM</th>
<th>EXTRA INFO</th>
</tr>
</thead>
<tbody>
<tr>
<td>Sine</td>
<td>Traditional LFO shapes</td>
<td></td>
</tr>
<tr>
<td>Triangle</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Sawtooth</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Square</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Rand S/H</td>
<td></td>
<td>Jumps to random values every cycle of the LFO</td>
</tr>
<tr>
<td>Time S/H</td>
<td></td>
<td>Jumps to min and max value each held for a random amount of time</td>
</tr>
<tr>
<td>PianoEnv</td>
<td></td>
<td>A curved sawtooth shape</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Sequence</th>
<th>DESCRIPTION</th>
</tr>
</thead>
<tbody>
<tr>
<td>Seq 1</td>
<td>These are sequences that jump to different values, holding each for a sixteenth of the LFO cycle rate.</td>
</tr>
<tr>
<td>Seq 2</td>
<td></td>
</tr>
<tr>
<td>Seq 3</td>
<td></td>
</tr>
<tr>
<td>Seq 4</td>
<td></td>
</tr>
<tr>
<td>Seq 5</td>
<td></td>
</tr>
<tr>
<td>Seq 6</td>
<td></td>
</tr>
<tr>
<td>Seq 7</td>
<td></td>
</tr>
<tr>
<td>Altern 1</td>
<td>These are sequences that jump between a minimum and a maximum value, each value held for a varying interval of time.</td>
</tr>
<tr>
<td>Altern 2</td>
<td></td>
</tr>
<tr>
<td>Altern 3</td>
<td></td>
</tr>
<tr>
<td>Altern 4</td>
<td></td>
</tr>
<tr>
<td>Altern 5</td>
<td></td>
</tr>
<tr>
<td>Altern 6</td>
<td></td>
</tr>
<tr>
<td>Altern 7</td>
<td></td>
</tr>
<tr>
<td>Altern 8</td>
<td></td>
</tr>
<tr>
<td>Chromat</td>
<td>These are &quot;melodic&quot; sequences of various kinds. When modulating oscillator pitch, to obtain chromatic results, set Modulation Depth to either ±30 or ±36.</td>
</tr>
<tr>
<td>Major</td>
<td></td>
</tr>
<tr>
<td>Major 7</td>
<td></td>
</tr>
<tr>
<td>Minor 7</td>
<td></td>
</tr>
<tr>
<td>MinArp 1</td>
<td></td>
</tr>
<tr>
<td>MinArp 2</td>
<td></td>
</tr>
<tr>
<td>Diminish</td>
<td></td>
</tr>
<tr>
<td>DecMinor</td>
<td></td>
</tr>
<tr>
<td>Minor3rd</td>
<td></td>
</tr>
<tr>
<td>Pedal</td>
<td></td>
</tr>
<tr>
<td>4ths</td>
<td></td>
</tr>
<tr>
<td>4ths x12</td>
<td></td>
</tr>
<tr>
<td>1625 Maj</td>
<td></td>
</tr>
<tr>
<td>1625 Min</td>
<td></td>
</tr>
<tr>
<td>2511</td>
<td></td>
</tr>
</tbody>
</table>

### Modulation Matrix Sources Table

<table>
<thead>
<tr>
<th>DISPLAY</th>
<th>SOURCE</th>
<th>COMMENTS</th>
</tr>
</thead>
<tbody>
<tr>
<td>Direct</td>
<td>No modulation source selected.</td>
<td></td>
</tr>
<tr>
<td>ModWheel</td>
<td>Mod Wheel</td>
<td>Mod Wheel is the controller.</td>
</tr>
<tr>
<td>AftTouch</td>
<td>Aftertouch</td>
<td>Modulation is proportional to the pressure applied to a key while it is held down. (Monophonic aftertouch).*</td>
</tr>
<tr>
<td>Express</td>
<td>Expression pedal</td>
<td>An external foot pedal provides the control.</td>
</tr>
<tr>
<td>Velocity</td>
<td>Key velocity</td>
<td>Modulation is proportional to hard key is played.</td>
</tr>
<tr>
<td>Keyboard</td>
<td>Key position</td>
<td>Modulation is proportional to key position.</td>
</tr>
<tr>
<td>Lfo1+</td>
<td>LFO 1</td>
<td>+ = LFO increases value of controlled parameter in a positive sense only.</td>
</tr>
<tr>
<td>Lfo1+/-</td>
<td>LFO 1</td>
<td>+/- = LFO increases and decreases value of controlled parameter equally.</td>
</tr>
<tr>
<td>Lfo2+</td>
<td>LFO 2</td>
<td>+ = LFO increases and decreases value of controlled parameter equally.</td>
</tr>
<tr>
<td>Lfo2+/-</td>
<td>LFO 2</td>
<td>+/- = LFO increases and decreases value of controlled parameter equally.</td>
</tr>
<tr>
<td>Lfo3+</td>
<td>LFO 3</td>
<td>+ = LFO increases and decreases value of controlled parameter equally.</td>
</tr>
<tr>
<td>Lfo3+/-</td>
<td>LFO 3</td>
<td>+/- = LFO increases and decreases value of controlled parameter equally.</td>
</tr>
<tr>
<td>Env1Rmp</td>
<td>Envelope 1 to 6</td>
<td>These are envelopes that are triggered by a keypress, and any/all may be used to vary parameters over time. Note that Env 1 and Env 2 are “hard-wired” to control Amplitude and Filter parameters, but are still available to control other parameters.</td>
</tr>
<tr>
<td>AudInEnv</td>
<td>Audio Input Envelope</td>
<td>Output of Envelope Follower in Mic/Audio Input signal path.</td>
</tr>
</tbody>
</table>

*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

<table>
<thead>
<tr>
<th>DISPLAY</th>
<th>DESTINATION</th>
<th>COMMENTS</th>
</tr>
</thead>
<tbody>
<tr>
<td>02Pitch</td>
<td>Per-oscillator pitch</td>
<td>Oscillator 2: Pitch Transpose</td>
</tr>
<tr>
<td>03Pitch</td>
<td>Per-oscillator pitch</td>
<td>Oscillator 3: Pitch Transpose</td>
</tr>
<tr>
<td>01Pitch</td>
<td>Per-oscillator pitch</td>
<td>Oscillator 1: Pitch Transpose</td>
</tr>
<tr>
<td>012Pitch</td>
<td>Global oscillator pitch</td>
<td>All Oscillators: Pitch Transpose</td>
</tr>
<tr>
<td>01Usync</td>
<td>Per-oscillator Variable Sync</td>
<td>Oscillator 1: Virtual Sync</td>
</tr>
<tr>
<td>02Usync</td>
<td>Oscillator 2: Virtual Sync</td>
<td>Oscillator 2: Virtual Sync</td>
</tr>
<tr>
<td>03Usync</td>
<td>Oscillator 3: Virtual Sync</td>
<td>Oscillator 3: Virtual Sync</td>
</tr>
<tr>
<td>01PW/Idx</td>
<td>Per-oscillator pulse width / Wave Table Index</td>
<td>Oscillator 1: Pulsewidth / Wavetable Index</td>
</tr>
<tr>
<td>02PW/Idx</td>
<td>Oscillator 2: Pulsewidth / Wavetable Index</td>
<td>Oscillator 2: Pulsewidth / Wavetable Index</td>
</tr>
<tr>
<td>03PW/Idx</td>
<td>Oscillator 3: Pulsewidth / Wavetable Index</td>
<td>Oscillator 3: Pulsewidth / Wavetable Index</td>
</tr>
<tr>
<td>01Hard</td>
<td>Per-oscillator hardness</td>
<td>Oscillator 1: Hardness</td>
</tr>
<tr>
<td>02Hard</td>
<td>Oscillator 2: Hardness</td>
<td>Oscillator 2: Hardness</td>
</tr>
<tr>
<td>03Hard</td>
<td>Oscillator 3: Hardness</td>
<td>Oscillator 3: Hardness</td>
</tr>
</tbody>
</table>

### Mixers:

- **01Level**: Mixer input levels | Mixer: Oscillator 1 Level
- **02Level**: Mixer: Oscillator 2 Level
- **03Level**: 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
- **L1Rate**: Per-LFO frequency | LFO 1: Rate
- **L2Rate**: LFO 2: Rate
- **L3Rate**: LFO 3: Rate

### LFOs:

- **PortTime**: Voice: Portamento Time
- **FXWetLvl**: FX: Wet Level
- **PstFXLvl**: Mixer: Post FX Level
- **PanPosn**: FX: Pan Position
- **UniDetune**: Voice: Unison Detune

### Envelopes:

- **Env1Dec**: Envelope Decay Time | Envelope 1 (Amp): Decay Time
- **Env2Dec**: Envelope 2 (Filter): Decay Time

### FX:

- **FX1Amnt**: FX 1: FX Amount
- **FX2Amnt**: FX 2: FX Amount
- **FX3Amnt**: FX 3: FX Amount
- **FX4Amnt**: FX 4: FX Amount
- **FX5Amnt**: FX 5: FX Amount
- **FXFeedback**: FX: FX Feedback
- **FXWetLvl**: FX: Wet Level
- **Ch1Rate**: Chorus parameters | Chorus 1: Rate
- **Ch1Depth**: Chorus 1: Depth
- **Ch1Delay**: Chorus 1: Delay
- **Ch1Fdbck**: Chorus 1: Feedback
- **Ch2Amnt**: Chorus 2: Rate
- **Ch2Depth**: Chorus 2: Depth
- **Ch2Delay**: Chorus 2: Delay
- **Ch2Fdbck**: Chorus 2: Feedback
- **Ch3Rate**: Chorus 3: Rate
- **Ch3Depth**: Chorus 3: Depth
- **Ch3Fdbck**: Chorus 3: Feedback
- **Ch4Rate**: Chorus 4: Rate
- **Ch4Depth**: Chorus 4: Depth
- **Ch4Delay**: Chorus 4: Delay
- **Ch4Fdbck**: Chorus 4: Feedback
- **Dlv1TTime**: Delay parameters | Delay 1: Delay Time
- **Dlv1Fdbck**: Delay 1: Feedback
- **Dlv2TTime**: Delay 2: Delay Time
- **Dlv2Fdbck**: Delay 2: Feedback
- **E0BasLvl**: EQ settings | EQ: Bass Level
- **E0BasFr**: EQ: Bass Frequency
- **E0MidLvl**: EQ: Mid Level
- **E0MidFr**: EQ: Mid Frequency
- **E0TrblLvl**: EQ: Treble Level
- **E0TrblFr**: EQ: Treble Frequency
- **PanPosn**: Pan Position | Pan: Pan Position
- **VocShift**: Vocoder Shift
- **VocSpread**: 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

<table>
<thead>
<tr>
<th>DISPLAY</th>
<th>AREA</th>
<th>DETAIL</th>
</tr>
</thead>
<tbody>
<tr>
<td>--------</td>
<td>------</td>
<td>--------</td>
</tr>
<tr>
<td>PortTime</td>
<td>Voice: Portamento Time</td>
<td></td>
</tr>
<tr>
<td>FXWetLvl</td>
<td>FX: Wet Level</td>
<td></td>
</tr>
<tr>
<td>PstFXLvl</td>
<td>Mixer: Post FX Level</td>
<td></td>
</tr>
<tr>
<td>PanPosn</td>
<td>FX: Pan Position</td>
<td></td>
</tr>
<tr>
<td>UniDetune</td>
<td>Voice: Unison Detune</td>
<td></td>
</tr>
</tbody>
</table>

### Oscillators:

- **01WTInt**: Oscillator 1 parameters | Oscillator 1: Wavetable Interpolation
- **01Pu/Idx**: Oscillator 1: Pulswidth / Wavetable Index
- **01Usync**: Oscillator 1: Virtual Sync
- **01Hard**: Oscillator 1: Hardness
- **01Dense**: Oscillator 1: Density
- **01DnsDtn**: Oscillator 1: Density Detune
- **01Semi**: Oscillator 1: Semitone Transpose
- **01Cents**: Oscillator 1: Cents Transpose
- **02WTInt**: Oscillator 2 parameters | Oscillator 2: Wavetable Interpolation
- **02Pu/Idx**: Oscillator 2: Pulswidth / Wavetable Index
- **02Usync**: Oscillator 2: Virtual Sync
- **02Hard**: Oscillator 2: Hardness
- **02Dense**: Oscillator 2: Density
- **02DnsDtn**: Oscillator 2: Density Detune
- **02Semi**: Oscillator 2: Semitone Transpose
- **02Cents**: Oscillator 2: Cents Transpose
### Tweak Parameters Table - Continued

| 03WTInt | Oscillator 3 parameters | Oscillator 3: Wavetable Interpolation |
| 03Pw/Idx | Oscillator 3: Pulsewidth / Wavetable Index |
| 03Usync | Oscillator 3: Virtual Sync |
| 03Hard | Oscillator 3: Hardness |
| 03Dense | Oscillator 3: Density |
| 03DnsDtn | Oscillator 3: Density Detune |
| 03Semi | Oscillator 3: Semitone Transpose |
| 03Cents | Oscillator 3: Cents Transpose |

| Mixer: |
| 01Level | Mixer: Oscillator 1 Level |
| 02Level | Mixer: Oscillator 2 Level |
| 03Level | Mixer: Oscillator 3 Level |
| RM1+3Lv1 | Mixer: Ring Mod 1+3 Level |
| RM2+3Lv1 | Mixer: Ring Mod 2+3 Level |
| NoiseLv1 | Mixer: Noise Level |

| Filters: |
| Fbalance | Filter Balance |
| F1Freq | Filter 1: Frequency |
| F1Res | Filter 1: Resonance |
| F1DAmtnt | Filter 1: Distortion Amount |
| F1Track | Filter 1: Keyboard Tracking |
| F2Freq | Filter 2: Frequency |
| F2Res | Filter 2: Resonance |
| F2DAmtnt | 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 |

### FX: |
| FX1Ammnt | FX1: FX Amount |
| FX2Ammnt | FX2: FX Amount |
| FX3Ammnt | FX3: FX Amount |
| FX4Ammnt | FX4: FX Amount |
| FX5Ammnt | FX5: FX Amount |
| FXFedbck | FX: FX Feedback |
| Dist1Lv1 | Distortion: Distortion 1 Level |
| Dist2Lv1 | Distortion: Distortion 1 Level |

### Delay parameters |
| Dlv1Time | Delay 1: Delay Time |
| Dlv1Sync | Delay 1: Delay Sync Time |
| Dlv1Slew | Delay 1: Slew Amount |
| Dlv2Time | Delay 2: Delay Time |
| Dlv2Sync | Delay 2: Delay Sync Time |
| Dlv2Slew | Delay 2: Slew Amount |

### Chorus parameters |
| Ch1Rate | 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 |

### Gator parameters |
| StSlew | Gator: Slew Amount |
| StDecay | Gator: Decay Time |
| StL/RDml | Gator: Left/Right Delay Time |

### Arpeggiator parameters |
| ArrGTm | Arpeggiator: Gate Time |
| ArrSwing | Arpeggiator: Swing |

### Modulation Depth: |
| M1Depth | Modulation Matrix: Slot 1 Depth |
| M...Depth | Modulation Matrix: Slot ... Depth |
| M20Depth | Modulation Matrix: Slot 20 Depth |
### Filter Table

<table>
<thead>
<tr>
<th>DISPLAYED AS</th>
<th>DESCRIPTION</th>
</tr>
</thead>
<tbody>
<tr>
<td>LP6NoRes</td>
<td>Lo-pass, 6 dB/oct, no resonance</td>
</tr>
<tr>
<td>LP12</td>
<td>Lo-pass, 12 dB/oct</td>
</tr>
<tr>
<td>LP18</td>
<td>Lo-pass, 18 dB/oct</td>
</tr>
<tr>
<td>LP24</td>
<td>Lo-pass, 24 dB/oct</td>
</tr>
<tr>
<td>BP6/6</td>
<td>Symmetric Band-pass, 6 dB/oct</td>
</tr>
<tr>
<td>BP12/12</td>
<td>Symmetric Band-pass, 12 dB/oct</td>
</tr>
<tr>
<td>BP6/12</td>
<td>Asymmetric Band-pass, 6 dB/oct (hi-pass), 12 dB/oct (lo-pass)</td>
</tr>
<tr>
<td>BP12/6</td>
<td>Asymmetric Band-pass, 12 dB/oct (hi-pass), 6 dB/oct (lo-pass)</td>
</tr>
<tr>
<td>BP6/18</td>
<td>Asymmetric Band-pass, 6 dB/oct (hi-pass), 18 dB/oct (lo-pass)</td>
</tr>
<tr>
<td>BP18/6</td>
<td>Asymmetric Band-pass, 18 dB/oct (hi-pass), 6 dB/oct (lo-pass)</td>
</tr>
<tr>
<td>HP6NoRes</td>
<td>Hi-pass, 6 dB/oct, no resonance</td>
</tr>
<tr>
<td>HP12</td>
<td>Hi-pass, 12 dB/oct</td>
</tr>
<tr>
<td>HP18</td>
<td>Hi-pass, 18 dB/oct</td>
</tr>
<tr>
<td>HP24</td>
<td>Hi-pass, 24 dB/oct</td>
</tr>
</tbody>
</table>

### Arp Mode Table

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

### Gator Modes Table

<table>
<thead>
<tr>
<th>DISPLAY</th>
<th>MODE</th>
<th>DESCRIPTION</th>
</tr>
</thead>
<tbody>
<tr>
<td>Mono16</td>
<td>16-note mono</td>
<td>16-note mono sequence: {A}</td>
</tr>
<tr>
<td>MonoAlt1</td>
<td>32-note mono</td>
<td>32-note mono sequence: {AB}</td>
</tr>
<tr>
<td>MonoAlt2</td>
<td>2 x 32-note mono</td>
<td>2 x 16-note sequences, each repeated: {AABB}</td>
</tr>
<tr>
<td>Stereo16</td>
<td>16-note stereo</td>
<td>2 x 16-note sequences simultaneously, {A} L, {B} R</td>
</tr>
<tr>
<td>SterAlt1</td>
<td>16-note stereo</td>
<td>2 x 16-note sequences simultaneously: {A} L, {B} R, {A} R, {B} L</td>
</tr>
<tr>
<td>SterAlt2</td>
<td>16-note stereo</td>
<td>As SterAlt1, but each sequence pair is repeated</td>
</tr>
</tbody>
</table>

### Effects Type Table

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

#### DISPLAY

- **A**: Display A
- **B**: Display B
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).