



VIRUS | POWERCORE
ACCESS VIRUS VIRTUAL ANALOG SYNTHESIZER
USER REFERENCE MANUAL IN ENGLISH



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Welcome

Thank you for purchasing the Access Virus PowerCore (VPC). You now have amongst your sonic arsenal a truly amazing synthesizer, fully equipped to infect your music with some serious attitude! Rest assured that no compromises have been made here - Access have equipped the Virus PowerCore with a sound engine that is identical in every way to that of the now legendary Virus series. The oscillator section features 3 main oscillators, each capable of generating one of 66 waveforms, plus a sub-oscillator for those extra-deep basses. Add to this a noise generator, oscillator sync, FM and ring modulation capabilities and you have a tone-palette that is truly vast! What's more, up to 16 voices can be achieved on each DSP, meaning those of you lucky enough to own four PowerCores and the full license for your Virus PowerCore can infect their tracks with up to 256 simultaneous voices of Virus phattitude!

The filters in the Virus PowerCore are second to none and with the flexible routing options and a wide variety of saturation modes, all manner of filter characteristics can be achieved. Anything from the smooth, warm tones associated with traditional analog synthesizers to the most gut-wrenching, teeth-rattling, speaker-shredding sonic terror is possible!

The Virus PowerCore also sports a powerful effects section, featuring effects such as chorus, phaser, distortion, delay and reverb as well as input ring modulation and a vocoder. Furthermore, with its flexible audio inputs, the Virus becomes an incredibly powerful effects processor. Any audio signal can be routed through both the filters and effects, so none of your audio tracks will be immune to infection!

One feature which you are bound to appreciate immediately is Access' acclaimed Adaptive Parameter Smoothing technology. Unlike many other soft-synths, where every parameter change is accompanied by obvious 'stepping' through the values, every adjustment you make will happen seamlessly and in realtime, meaning that live tweaking is now a truly joyful experience.

Loading and saving patches has never been easier, thanks to the advanced patch management system in the Virus PowerCore. As soon as the Virus plug-in is enabled, it scans the PowerCore folder on your hard-drive and loads all of the patches it finds there into RAM. From there it organises them into hierarchical menus and categories, so you

can get straight to the patches you want. All patches used in the current session are stored in the Recently Used folder, and over time, your most commonly-used patches will be automatically stored to the Favourites folder for instant recall.

NOTE: *The Virus PowerCore is 100% compatible with the majority of patches available for the Virus series, of which there are thousands available for free download at www.access-music.de*

Installation

Installing the VPC On Your Computer

> INSTALLING ON A PC/WIN32

Run the installer program 'Virus PowerCore Installer.exe' included in the download.

The installer will copy both the Virus Powercore.dll and the Virus Input.dll into the Vst-PlugIns folder of your sequencer/host software. Virus Powercore.dll is the main synthesizer plugin, and Virus Input.dll is the side-chain effect plugin. (If the installer selects the wrong host software, just locate the .dll files and copy/paste them into the VstPlugins\POWERCORE folder of your preferred host software.)

A folder will be created in the location:

C:\Documents and Settings*(your username)*\My Documents\Access Music\Virus Powercore.

Into this folder you should copy any .mid files you have which contain Virus patch libraries. Check the download section of www.access-music.de for a large selection of freely downloadable patches. The Virus PowerCore is compatible with the Virus hardware synthesizers, so any existing Virus patches are suitable.

> INSTALLING ON A MAC

Run the installer program 'Virus PowerCore Installer' included in the download.

The installer will copy the Virus Powercore and Virus Powercore Input plugin to

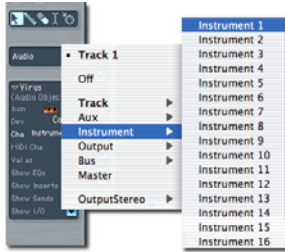
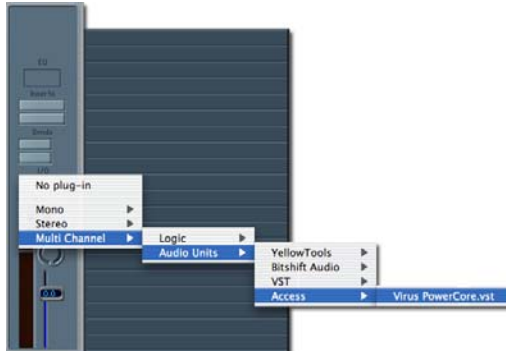
/Library/Audio/Plug-Ins/VST/POWERCORE.


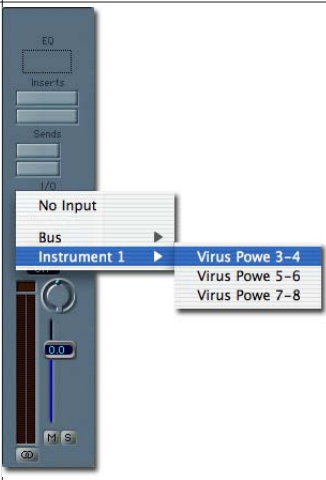

A folder will be created in the location ~/Library/Application Support/Access Music/Virus Powercore. Into this folder you should copy any .mid files you have which contain Virus patch libraries. Check the download section of www.access-music.de for a large selection of freely downloadable patches. The Virus PowerCore is compatible with the Virus hardware synthesizers, so any existing Virus patches are suitable.

> AUDIO UNITS

To use the Virus PowerCore as an Audio Unit, you need to run the TCAU Audio Unit wrapper after you have run the Virus installer. Installing the PowerCore driver also installs the TCAU wrapper although it's recommended to check the TC website for updates frequently.

Instancing the VPC in Logic

<p>STEP1: Create a new Audio Instrument in the arrange window</p>	 A screenshot of the Logic Pro software interface. The 'Audio' menu is open, and the 'Instrument' submenu is also open, showing a list of instrument options from 'Instrument 1' to 'Instrument 16'. The 'Instrument' option is highlighted in blue.
<p>STEP2: Now select the Virus PowerCore by opening the instrument popup menu within the audio object. For maximum flexibility choose "Multi Channel" which enables you to route the four parts of the Virus to individual outputs and thus treat them individually with additional effects.</p>	 A screenshot of the Virus PowerCore instrument's popup menu. The 'Multi Channel' option is selected and highlighted in blue. A submenu is open for 'Multi Channel', showing 'Logic' and 'Audio Units'. The 'Audio Units' submenu is also open, showing 'YellowTools', 'Bitshift Audio', 'VST', and 'Access'. The 'Access' option is highlighted in blue, and a further submenu is open showing 'Virus PowerCore.vst'.

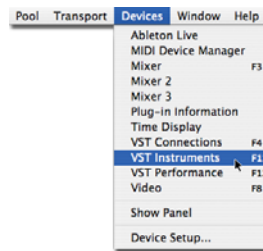
<p>STEP3: To gain access to the Virus' individual outs, create an Aux Object. Please note that you need to create an Aux object for every stereo output you wish to gain access to.</p>	
<p>STEP4: Now select the Virus' individual outs from the Aux channel's Input options</p>	
<p>STEP5: Now open up the environment, select the Virus instrument and change the MIDI in channel ("Cha") from 1 to All.</p>	

STEP6: The Virus is up to 4 part multi timbal for each DSP used. The four multiparts receive MIDI data on the corresponding channels 1-4. To easily access all four parts, create four MIDI instruments within the environment, assign the MIDI channels 1,2,3 and 4 to the instruments and cable them into the Virus Instrument.



Instancing the VPC in Cubase SX2

STEP1: Select "VST Instruments" from the "Devices" menu to show the "VST Instruments" window.



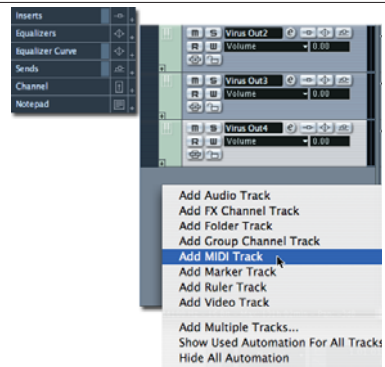
STEP2: In the "VST Instruments window", find an empty VST Instrument slot. Empty VST Instrument slots are labelled "no instrument". Click on the label "no instrument" to open the VST Instrument Selection menu and select "Virus PowerCore" in the "Access Music" sub-menu to create an instance of the Virus PowerCore. After a couple of seconds, the Virus PowerCore is initialized and its editor window is shown.



STEP3: In the Arrange Window, the Virus is represented by a number of tracks that are created automatically by Cubase SX. The folder track "VST Instruments" shows all currently running VST Instruments; in this case it only shows the folder track "Virus Powercore" which contains a number of tracks to automate various aspects of the Virus Powercore. The topmost track labelled "Virus Powercore" allows you to automate the parameters of the Virus itself while the other four tracks labelled "Virus Out1" to "Virus Out4" automate the VST Mixer parameters of the four stereo outputs of the Virus PowerCore. If you are confused by these tracks, just click the little "-" symbol to the left of the "VST Instruments" track to collapse the whole tree. You can ignore these tracks for now but you should read the Cubase SX documentation later to learn what to do with these tracks if you are unsure.



STEP4: Create a MIDI Track to play the Virus PowerCore via MIDI and to record MIDI messages that are played back by Cubase SX. To do that, select "MIDI" from the sub-menu "Add Track" in the "Project" menu. Alternatively, you can open the Track context menu in the track list (Mac: Ctrl+Click, PC: right mouse button click) as shown below.



STEP5: Select "Virus PowerCore" from the "out" menu in the track inspector to route the MIDI output of the track to the Virus PowerCore.



STEP6: Select MIDI channel "1" from the "chn" menu in the track inspector to play Part 1 of the Virus PowerCore.
 Now you can play the Virus Powercore via MIDI and record on the created MIDI track.
 You can create further MIDI tracks and assign them to any of the four Parts of the Virus by selecting the respective MIDI channel from the "chn" menu.



Patch Management

Total Recall

The Virus PowerCore has a sophisticated Patch Management system, designed to make browsing and sourcing patches easier than ever before. As soon as you instance the Virus PowerCore in your sequencer, it will automatically scan the Virus PowerCore folder on your computer's hard-drive for any .mid files containing banks of Virus patches and store them in RAM for instant recall. Patches are then sorted alphabetically within each bank, and organised into Categories, to make browsing a breeze.

> THE POWERCORE FOLDER

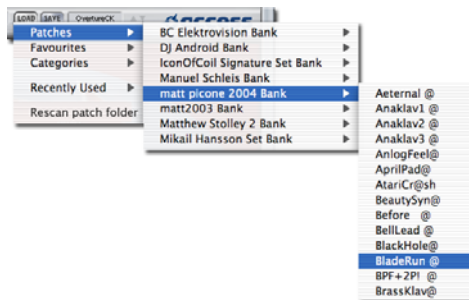
This is where you store all your Virus PowerCore patches. If you are running Windows it is located in "My Documents\Access Music\PowerCore". If you are running Mac OS it is located in ~/Library/Application Support/Access Music/PowerCore.

NOTE: If you have a large number of banks you may wish to create sub-folders within the PowerCore folder - these will also be scanned automatically, and appear within the heirarchical menu structure.

Loading Patches

To load a patch click on the LOAD tab at the top right of the screen. The following drop-down menu will appear:

- Position the mouse pointer on 'Patches' and the next level will appear listing all the available banks.
- Position the mouse pointer on the bank you wish to browse, and a list of all the patches contained in this bank will appear.



- Select the patch you wish to audition with a single mouse click.

NOTE: If the selected bank contains too many entries to display all at once, you will see a little arrow at the bottom of the list. Click on this arrow to scroll through the rest of the entries.

> CATEGORIES

The Virus PowerCore will scan all the patches it finds and organise them according to their category settings. This means if you need a bass sound, you need only look in the 'Bass' category if you wish. The patches are displayed alphabetically with name of the bank from which they were sourced.

NOTE: If the selected category contains too many entries to display all at once, you will see a little arrow at the bottom of the list. Click on this arrow to scroll through the rest of the entries.



> RECENTLY USED

Patches which have been used in the current session are stored in the Recently Used menu, so you needn't go searching through all the banks again to find that great lead sound you had half an hour ago. Banks are displayed according to the date of the session in which they were created ie:

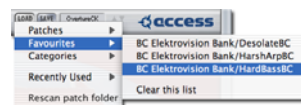


NOTE: Note that patches that are accessed via the browse buttons will not be added to the Recently Used menu automatically. However, if they are being used in the song when it is saved, they will then be added

> **Clear this list:** Use this to clear all entries from the Recently Used list.

> FAVOURITES

The Virus PowerCore will automatically compile your most commonly used patches in the Favourites folder. It works by assigning 'points' to the patches you have used in your projects, for instance:



Patch is used in a project (host has saved the settings to a song) = 5 points

Patch was saved (using the plug-in's Save menu) = 2 points

Up to 100 favourites can be stored internally, with the top twenty visible to the user. As soon as one patch drops out off the list (by scoring less points than No.100) the value of all remaining patches will be reduced so that the lowest scoring patch has only one point. This ensures that 'new' patches stand a chance of being included in the list.

> **Clear this list:** Use this to clear all entries from the Favourites list.

> BROWSING PATCHES

At the top right hand corner of the Virus PowerCore window, you will see the name of the currently selected patch with two arrows (up/down) alongside.



By clicking on the browsing arrows, you can easily audition every patch in the currently selected bank. You may also use this method to browse the currently selected Category menu.

NOTE: *The browsing arrows will only be available once you have selected a patch from one of the menus.*

> RESCAN PATCH FOLDER

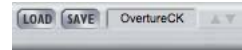
Click on 'Rescan patch folder' if you have added any new .mid files containing Virus patches to the PowerCore folder since instancing the Virus PowerCore. The new banks will now be displayed.

Saving Patches

If you make any changes to a patch, or indeed create a completely new patch, you may wish to save it for future use.

> NAMING A PATCH

It is always advisable to choose an appropriate name for a new patch, so you can locate it more easily in the future.

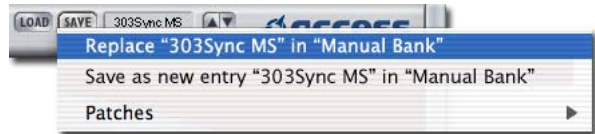


To change the current patch name, click on the name pane at the top of the window.

Type in the new name. Press Return on your computer keyboard to enter the new name.

> SAVE

To save the current patch, click on the Save tab.



The Save menu will appear (see screenshot).

You can now choose from three options:

- > **Replace “(old patchname) xxx” in “xxx Bank”:** The original version of the current patch will be overwritten by the new version.
- > **Save as new entry “(new patchname) xxx” in “xxx Bank”:** The newly edited patch will be appended to the currently selected bank.
- > **Patches:** Alternatively you can select ‘Patches’ which will give you the option of appending the newly edited patch to any of the banks in the PowerCore folder ie

How To Install Your Own Patches

Firstly, you will need to download some new soundsets. There are several available for free at www.access-music.de, between them covering a wide variety of musical tastes.

Once you have downloaded some soundsets, locate the .mid versions and copy/paste them to the Virus Powercore folder. On a Windows PC, this will be located in: C:\Documents and Settings*(your username)*\My Documents\Access Music\Virus Powercore. On a Mac, it will be located in: ~/Library/Application Support/Access Music/Virus Powercore.

NOTE: *It is important to realise that only sounds saved in .mid format will be recognised by the Virus PowerCore. Only SMF format 0 is supported*

Next time the you instance the Virus PowerCore, you will find the new soundbanks listed in the Patches menu.

You may wish to organise the banks into folders for easier browsing. If you do not wish a bank to be loaded every time you instance the Virus PowerCore, simply delete it from the Virus Powercore folder, or cut/paste it into another location.

Using Multiple Parts

Parts and DSPs

The TC PowerCore has four DSP processors which it uses to process the various plugins including Virus PowerCore. The Base License package allows one of these DSPs to be used for the Virus, which can be split up into four 'Parts' - each with it's own voice and individual output routing.

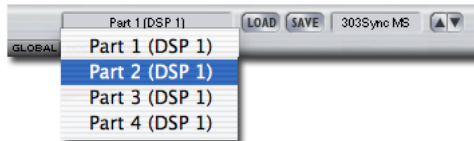
With the Base License, you can only use one Virus simultaneously, if you find that you really need to use more than one instance at a time, you should consider purchasing the Extended License.

The Part/DSP box at the top of the plugin window displays the current part and the DSP it is using i.e.

To choose a different part, click on the box and select the desired part from the drop-down list.

Each part corresponds to it's equivalent MIDI channel i.e. Part 1 = MIDI channel 1, Part 2 = MIDI channel 2. Please refer to your sequencer manual for instructions on how to address different MIDI channels on the same audio instrument.

With the Extended License package, then you can utilise as many DSPs as there are in your PowerCore system. Each time you instance the Virus PowerCore on a new audio instrument, another DSP is used.



> GLOBAL EFFECTS

Most of the effects in the Virus PowerCore are specific to each part, but the delay/reverb effect is global i.e. all parts using the same DSP use the one delay/reverb effect. Please refer to the Delay/Reverb section of the FX1 Page section of the manual for more specific instructions. If you wish to use more than one delay/reverb effect simultaneously, you must instance the Virus PowerCore on another audio instrument (Extended License only).

Adjusting Parameters

The majority of Virus PowerCore controls can be adjusted by either dragging a knob, clicking on a button or typing in a specific value.

The majority of parameters have a value range of 0 - 127. The value increases as you rotate the knob clockwise.

There are some knobs that are bipolar. In this case the value range is -64 to 0 to +63. Centre position (12 o'clock) is zero. Rotate the knob in an anti-clockwise direction to specify values down to -64. Rotate clockwise to specify values up to +63.

> ADJUSTING KNOBS

Drag the knob either vertically or horizontally. (Some host software may also allow you to use the mouse-wheel to adjust knobs.)

> TYPING IN VALUES

The majority of parameters also have a text box, which displays the current value. This can be edited via the computer keyboard:

> TYPING A VALUE INTO A TEXT BOX

Click on the text box and then type in the desired value. If the parameter is bipolar and you wish to specify a negative value, you must type a minus sign in front of the numbers.

To input the value without leaving the selected text box, press Enter on your numeric keypad. To input the value and exit keyboard editing mode, press Return (Mac) or Return on the Alpha keypad (Windows PC).

NOTE: Many parameters in the Virus PowerCore are inter-dependent, meaning that adjusting certain parameters may or may not have an audible effect, depending on the value of another. For instance if Filter Balance is set to -64, only filter1 is audible; consequently adjustments to filter 2 parameters will be inaudible.

Using Remote Controllers

If you prefer you can choose to use an external controller to adjust the parameters of the Virus PowerCore. Ideally this will be a hardware Virus, as all of the controllers will be automatically mapped to their equivalent in the Virus PowerCore, but you could also use a third-party remote controller or hardware synthesizer.

> USING A VIRUS SYNTHESIZER AS A REMOTE CONTROL DEVICE

Make sure that the MIDI OUT of the Virus is connected to the MIDI IN of your PC MIDI interface. On the external Virus, set SYSTEM>MIDI>Panel to Int+Midi or Midi. This tells the Virus to send controller data to the MIDI OUT port. Now set SYSTEM>MIDI CONTROL>LoPage to 'Contr' and MIDI CONTROL> HiPage to PolyPrs.

Check that when you move a knob on the Virus or indeed adjust any parameter in the edit menus that the equivalent parameter in the Virus PowerCore is updated accordingly.

NOTE: Please note that in certain sequencer hosts some parameters, such as Channel Volume and Panorama will be intercepted by the audio instrument channel. In this case, you will not be able to adjust these parameters by remote.

> USING A THIRD-PARTY DEVICE AS A REMOTE CONTROL

Please refer to the instruction manual of the third-party device for instructions on how to configure it to send the appropriate control data. For a list of the controller numbers associated with the various parameters in the Virus PowerCore, please refer to the Appendix.

> USING YOUR SEQUENCER TO AUTOMATE THE VIRUS POWERCORE

Most sequencers allow you to automate the parameters of VST instruments. For instructions on how to do this in your chosen sequencer, please refer to the instruction manual that was included with your software. For a list of the controller numbers associated with the various parameters in the Virus PowerCore, please refer to the Appendix.

Getting Started

An Introduction to Viral Synthesis

For the benefit of synthesis newbies, we will first discuss some of the basic concepts to help you get an idea of what this is all about. The Virus PowerCore is based on '*subtractive*' synthesis, which means that the sound begins life in the *oscillator* section as harmonically rich as it's going to get, then certain elements are removed via the *filter* section and what's left is sculpted into the desired shape by the *amplifier* section. Of course the Virus PowerCore is a lot more complex than this! What follows is an overview of the main stages that constitute any Virus sound or 'patch'.

Oscillators

This is where life begins for a Virus PowerCore sound. An oscillator generates a basic *waveform*, which, if left untreated, will sound as a continuous tone for as long as a key is held.

Different waveforms contain different amounts of *harmonics* - it's these that give the waveform its timbre, or tonal character. For instance:

- > **Saw:** Saw waveforms contain both odd and even harmonics, resulting in a very bright and harsh tone.
- > **Pulse:** Pulse waveforms (also known as a square wave) contains only the odd harmonics, which results in a bright, but somewhat 'hollow' tone.
- > **Sine:** Sine waveforms contain no harmonics at all, only the *fundamental*, which is what we refer to when we talk about the pitch of a note. Consequently, a sine wave has a very pure timbre.

The 3 main oscillators in the Virus PowerCore are capable of generating all these, including 63 additional *spectral waves* of all sorts of shapes and timbres.



The Virus also has a *sub-oscillator*, which is used to add extra bass to a sound by tracking the pitch of *oscillator 1*, only 1 octave lower.

In addition to the 3 *oscillators* and *sub-oscillator*, there is also a *noise generator*. This generates 'white' noise, which sounds rather like the noise a television makes when you unplug the aerial. This is useful for creating snare drum sounds, sound FX such as wind or surf, or simply for adding warmth to the sound.

As if this wasn't enough, the Virus PowerCore is also capable of *ring modulation* (ring mod) and *FM* (frequency modulation), both of which can be used to add extra harmonics and overtones, useful for creating more complex timbres like bells and electric pianos.

Mixer

This is where the outputs of the oscillators, noise, and ring modulator are balanced before the resulting mixture is sent to the filter section.



Filter

This is an extremely important stage of any subtractive synthesizer, and the Virus PowerCore is no exception to this with its *dual filters*, each of which can be assigned to one of 4 different types.

A filter is used to remove certain components (frequencies) of a sound, whilst allowing others to pass freely. For instance, a *lowpass filter* will remove all the frequencies above a user-definable 'cutoff' frequency, whilst all those *below* will *pass* - geddit? The cutoff can be adjusted in realtime, which results in a 'filter sweep' - the life-blood of many famous synth sounds.



Amplifier

Once the signal has been suitably filtered, it then passes to the *amplifier* to be shaped by the *amp envelope*. The way that the volume of a sound changes over time is crucial to our perception of it, and so just like the filter, this is a very important stage of any synth sound. Believe it or not, on certain old electronic organs, the only difference between the piano and trombone presets was the shape of their amp envelopes!! (Bear in mind of course that it took a certain degree of imagination to hear either as a piano or a trombone!)



A typical 4-stage amplitude envelope forces the sound to follow a specific volume curve through the following stages - *Attack*, *Decay*, *Sustain*, and *Release*, with *Attack* determining the time it takes for the note to reach full volume once a key is pressed, *Decay* determining the rate at which it falls to the level at which the note will *Sustain*, and *Release* being the time it takes the sound to decay to silence once the key is released. The Virus PowerCore actually contains a 5-stage envelope, due to the addition of a *Sustain Time* parameter. This is used to create an additional crescendo (volume swell) or decrescendo once the note has reached the sustain level.

Note that the filter section also has a dedicated envelope for dynamically controlling the cut-off in a similar fashion.

Effects

Once the sound has been successfully mangled by the filters and amplifier, it then gets sent to the *Effects* section. Here you can choose to colour the sound with several different (simultaneous) effect types, including *Analog Boost*, *Distortion*, *Chorus*, *Phaser*, and *Delay/Reverb*.

Modulation Matrix

Last, but by no means last as it happens, is the *mod matrix*. This is a very powerful tool that you can use to dynamically control several parameters at once via a number of different *control sources*. The available control sources include *LFO's*, (see below) *envelopes*, *MIDI note velocity*, *modwheel* and *pitch bend* as well as several others.

NOTE: LFO is an abbreviation of Low Frequency Oscillator. Whereas the Virus PowerCore's main oscillators generate frequencies high enough to be in the audio spectrum, the LFOs generate much slower oscillations, which are very useful for modulating any of the parameters in the both the synthesis section and the effects section. There are 3 LFOs in the Virus PowerCore.

The *mod matrix* itself is used to route the outputs of 3 control 'sources' to six 'destination' parameters.

There are also 10 pre-defined parameters which can be controlled by *velocity* (how hard you press the key on a touch-sensitive keyboard).

Added together, this means a total of 29 parameters can be controlled simultaneously by up to 7 modulation sources, which allows for a great deal of movement, interest and expression within a single sound.

So hopefully you now have a general overview of the way the Virus PowerCore works, it's time to look in more detail at how to achieve all this sonic naughtiness...

Viewing Pages



To make things easier, the controls of the Virus PowerCore have been organised into seven different pages. To view a particular page, click on the corresponding name in the upper left of the Indigo pane.

> EASY

The Easy Page contains controls for the most commonly edited parameters, i.e. filter cutoff, resonance, amp attack, FX Send etc.

> OSC

The Osc Page contains the controls for the oscillators section, including ring mod, FM and noise. The controls for the mixer section are also located on this page.

> FILTERS/ENV

The Filters/Env Page contains the controls for the filter section and both of the envelope generators.

> **LFO**

The LFO Page contains the controls for the 3 low frequency oscillators (LFOs).

> **FX-1**

The FX-1 Page contains the controls for Analog Boost, Input RingMod, Phaser, Patch Distortion, Chorus and Delay/Reverb effects.

> **FX-2/GLOBAL**

The FX-2 Page contains the controls for the Vocoder, Input, Envelope Follower, and several global settings such as Master Tune, Pitch Wheel, Unison Mode, etc.

> **MODMATRIX**

The Mod Matrix Page contains the controls for assigning all of the modulation sources and destinations and their respective amounts.

Easy Page



Here you will find a selection of controls for the most commonly edited parameters in the Virus PowerCore. If you just want to tweak an existing patch without getting into it too deeply, then this is the page for you.

Parameters of the Easy Page

> CUTOFF

This adjusts the cutoff frequency of both filters. The result will vary depending on the currently selected filter modes.

> RESONANCE

This adjusts the amount of resonance (Q) for both filters. This determines how much those frequencies near the cutoff frequencies will be emphasised, so the result will depend largely on the cutoff frequency of both filters. Those who want to make their PowerCore squeal like a pig need look no further.

> CHORUS MIX

Adjusts the balance between the dry signal and the chorus effect.

> PHASER MIX

Adjusts the balance between the dry signal and the phaser effect.

> SEND

Adjusts the balance between the dry signal and the delay/reverb effect.

> OSC VOLUME

Adjusts the volume of the combined outputs of the 3 oscillators and the sub-oscillator. Turning the knob past 12 o'clock position will increase the level of filter 1 saturation (if selected). This controller has no effect on the volume of the noise generator or the ring modulator, both of which have independent volume controls (Osc Page).

> SUB VOLUME

An independent volume control for the sub-oscillator. This is a very simple oscillator, which always tracks the pitch of oscillator 1, but 1 octave lower. Use this to add more bass to a sound.

> RING MODULATOR

Adjusts the volume of the ring modulator (see Osc Page).

> **AMP ATTACK**

Adjusts the time it takes for the sound to reach maximum volume. At zero, the sound has immediate impact, at higher values it will fade in gradually.

> **AMP DECAY**

This adjusts the time it takes for the sound to fall from maximum volume to the sustain level (see Filters/Env Page).

The Osc Page



The Osc Page is the birthplace of any Virus PowerCore patch. Here you will find the controls for adjusting the 3 main oscillators, the sub-oscillator, the noise generator, ring modulator and FM.

Osc 1

This is where you will find the controls for adjusting oscillator 1.

> SHAPE

This defines the shape of the waveform generated by osc 1. When set to 0, the waveform will be whatever is selected with the Wave Select knob. Moving the knob in a clockwise direction will result in a gradual transition to a pure saw wave at value 64 (default). As you continue to increase the value above 64, a square wave will be gradually mixed in with the saw until, at a value of 127, a pure square wave is achieved.



> WAVE SELECT

The Wave Select knob allows you to choose from one of 64 waveshapes, including sine, triangle and 62 additional *spectral* waves. This parameter has no effect unless the value of osc 1 SHAPE is between of 0 – 63. If you just want the pure waveform, set osc 1 Shape to 0.

If you would like to see a graphical representation of these waveforms, see the Appendix.

> PW

This determines the width of the pulses in a square wave, or the *pulse width*. At a setting of 0, the wave is completely square – the pulse has 50% of the wave cycle. As you increase the value, the pulses become thinner until, at a value of 127, they have 0% of the wave cycle, resulting in silence.

> SEMITONE

This determines the pitch for osc 1, and is adjustable in semitone intervals. The default setting is 0, at which middle C will sound like middle C, but you have the option of transposing osc 1 anywhere within a 4-octave range above or below this point.

> KEY FOLLOW

This determines how MIDI note number affects the pitch of osc 1. At the default setting of +32, playing an ascending chromatic scale (play every key in sequence, including the black keys) will cause the pitch of osc 1 to rise in semitone intervals. At a setting of +63, playing the

same scale will cause the pitch to rise in whole tone intervals. A setting of 0 results in every key sounding at the same pitch, and minus values will make osc 1 *descend* in pitch as you play *up* the keyboard (useful for left-handed players ;-)

Osc 2

This is where you will find the controls for adjusting oscillator 2. Many parameters are identical in function to their counterparts in osc1.



> SHAPE

This defines the shape of the waveform generated by osc 2. When set to 0, the waveform will be whatever is selected with the Wave Select knob. Moving the knob in a clockwise direction will result in a gradual transition to a pure saw wave at value 64 (default). As you continue to increase the value above 64, a square wave will be gradually mixed in with the saw until, at a value of 127, a pure square wave is achieved.

> WAVE SELECT

The Wave Select knob allows you to choose from one of 64 waveshapes, including sine, triangle and 62 additional *spectral* waves. This parameter has no effect unless the value of osc 2 Shape is between of 0 – 63. If you just want the pure waveform, set osc 1 Shape to 0.

> PW

This determines the width of the pulses in a square wave, or the *pulse width*. At a setting of 0, the wave is completely square – the pulse has 50% of the wave cycle. As you increase the value, the pulses become thinner until, at a value of 127, they have 0% of the wave cycle, resulting in silence.

> SEMITONE

This determines the pitch for osc 2, and is adjustable in semitone intervals. The default setting is 0, at which middle C will sound like middle C, but you have the option of transposing osc 2 anywhere within a 4-octave range above or below this point.

> **DETUNE**

This determines the detuning of osc 2 relative to oscillators 1 and 3. The subtle tuning variations between all the individual instruments in a string ensemble is a major factor in the 'warmth' that is associated with that sound – the same is true of many famous synth patches, which use detuned oscillators to achieve a similar effect.

NOTE: *Detune is affected in a subtle way by keyfollow – the higher up the keyboard you play, the less osc 2 is detuned. This achieves a more musical result than a linear detune.*

> **KEY FOLLOW**

This determines how MIDI note number affects the pitch of osc 2. At the default setting of +32, playing an ascending chromatic scale (play every key in sequence, including the black keys) will cause the pitch of osc 2 to rise in semitone intervals. At a setting of +64, playing the same scale will cause the pitch to rise in whole tone intervals. A setting of 0 results in every key sounding at the same pitch, and minus values will make osc 2 *descend* in pitch as you play *up* the keyboard!

> **SYNC**

When switched to On, the wave cycle of osc 2 is forced to synchronise to that of osc 1. Depending on the pitch of osc 2 the result can be a cold, hard sound with a lot of overtones. Try sweeping the pitch (Semitone) of osc 2 for a dramatic demonstration of this effect.

NOTE: *Tip: Try using LFO3 to modulate Sync Phase (see LFO Page).*

> **FM AMOUNT**

This determines to what degree osc 2 is frequency-modulated by osc 1. The results will vary considerably, depending on the interval between osc 1 and 2, but this will often create complex timbres, which may sound quite atonal. Try using Sync as well for even more timbral possibilities - the effect of FM can be more subtle when used in conjunction with oscillator sync.

> **FM MODE**

This selects the waveform with which the frequency will be modulated:

> **PosTri:** Modulates osc 2 with a positive triangle wave according to the pitch of osc 1.

- > **Tri:** Modulates osc 2 with a triangle wave according to the pitch of osc 1.
- > **Wave:** Modulates osc2 with the waveform currently assigned to osc 1, according to the pitch of osc 1.
- > **Noise:** Modulates the frequency of osc 2 with the noise generator.
- > **Input:** Modulates the frequency of osc 2 with the audio input signal.

NOTE: Note that depending on your host software, you will most likely have to use the Virus PowerCore Side-chain module to use the Input mode.

> **ENV FM**

This determines to what degree FM Amount is modulated by the filter envelope. As this is a bipolar parameter, you can set this to positive or negative values – try setting FM Amount to +63, and experiment with different values for Env FM, filter Attack and filter Decay (see Filters/Env page).

> **ENV OSC 2**

This determines to what degree osc 2 Pitch is modulated by the filter envelope. Experiment with both negative and positive values, and different values of filter attack and filter decay for a wide range of pitch curves.

> **PHASE INIT**

This determines the point in the wave cycle at which the oscillators will start when a key is pressed. At the default value of 0, the oscillators are free-running, so the start of each note can occur at any point along the wave cycle, resulting in the subtle variations between each note; an effect traditionally associated with analog synthesizers. At values between 1 and 127, osc 1 is forced to a phase angle of 0, whilst the start point of osc 2 is shifted further along the wave cycle, the higher the value. The end result is a consistent attack which can prove very useful in the creation of drum and percussion patches.

Osc 3

This is where you will find the controls relating to oscillator 3.



> OSC MODE

This allows you to choose from a selection of 66 waveshapes, including saw, pulse, sine and triangle, including 62 additional spectral waves (the same selection available to osc 1 and 2). There is also an additional 'Slave' mode, in which osc 3 takes it's characteristics from osc 2. No osc 3 parameters will have any effect in this mode, which was designed as a simple way to add more depth to the sound. *Since most sounds only require 2 oscillators, osc 3 is optional, hence the default setting is 'Off'. It also takes more calculating power, and as such will result in a reduction in polyphony when activated.*

> SEMITONE*

This adjusts the pitch of osc 3 within a range of 4 octaves (48 semitones) above or below middle C. If osc 1 and 2 are both set to Semitone = 0, try setting osc 3 Semitone to +7 (perfect 5th) – a technique used in many classic lead sounds.

> DETUNE*

This offsets the pitch of osc 3 relative to osc 1 and 2. This works in the opposite direction to osc 2 Detune, so the higher the value, the lower the pitch.

Use this to add more warmth and subtle movement to the sound.

NOTE: Like Osc2 Detune, this parameter is affected by keyfollow (MIDI note number) for a more musical result.

NOTE: *This parameter has no effect if Osc Mode is set to Off.

Sub/Noise

The sub-oscillator beefs up the sound by adding a square or triangle wave an octave below the pitch of osc 1.

The noise generator adds ‘white’ noise to the sound. It is un-pitched, and is therefore used for percussion sounds, FX such as wind or surf, or for adding a little warmth to pitched sounds.



> SUB SHAPE

This toggles the sub-oscillator waveform between square and triangle.

> NOISE COLOR

Changes the characteristics of the noise generator:

> **Middle (0)** : Neutral (*‘white’ noise, all the frequencies are evenly distributed*).

> **Negative (-64)** : Lowpass (*‘pink’ noise, less top and more bass*).

> **Positive (+63)** : Hipass (*bright and thin*).

Velocity Mod

Several parameters in the Virus PowerCore can be controlled by *velocity*. In practice this means that these parameters can be set to respond to how hard you press the keys. Think of a traditional piano – as you hit the keys progressively harder, the notes become not only louder, but also brighter (more overtones) and even very slightly sharper in pitch. If you want to program sounds that respond in a really musical way to the nuances of your keyboard technique, then it’s definitely worthwhile spending some time on this section. (There again, if your keyboard technique is comparable to that of a one-fingered chimpanzee, perhaps you should move quickly on!)



Many of the following destinations are *bipolar* parameters, in which case, positive values result in a positive offset as velocity increases, whereas negative values result in a negative offset as velocity increases.

NOTE: Tip 1: Most of the arpeggiator patterns are programmed with velocity data – so be sure to experiment with different velocity mod settings in your arp patches!

NOTE: Tip 2: Make sure the target destinations have somewhere to go – for example, there is no point applying a negative offset to a parameter that is already set to zero!!

NOTE: Tip 3: You will need to make sure your controller keyboard is velocity sensitive to benefit fully from this functionality!

> **OSC 1**

This determines to what degree Osc 1 Shape is affected by velocity. At positive values, a positive offset is applied to Osc 1 Shape as you play harder. For example, if Osc 1 Shape is set to 0, and Osc 1 Vel Mod is set to +64, as you play progressively harder, Osc 1 Shape will move towards saw, and at full velocity (127) will reach pulse. At negative values, a negative offset is applied to Osc 1 Shape as you play harder; therefore Osc 1 Shape needs to be set to something considerably higher than 0 for you to notice any effect here.

> **OSC 2**

This determines to what degree Osc2 Shape is affected by velocity. (See Osc 1 for more details).

> **PW**

This determines to what degree the pulse width of oscillators 1 and 2 are affected by velocity. At least one of these oscillators need to be set to Shape = +65 or higher before you will notice any difference here.

> **FM**

This determines to what degree Osc 2 frequency modulation is affected by velocity. Positive values result in an increase in FM depth as you play harder, whereas negative values result in a decrease in FM depth as you play harder.

> **VOL**

This determines to what degree the patch volume is affected by velocity. Set this to a positive value for the traditional response, i.e. the harder you play, the louder the note. Negative values have the opposite effect, i.e. the harder you play, the *quieter* the note!

> PAN

This determines to what degree panorama is affected by velocity. Positive values mean that as you play harder the sound will move towards the right speaker, whereas negative values mean that as you play harder, the sound will move towards the left speaker.

> PUNCH INTENSITY

This determines the volume of the click at the beginning of each note, which can help to add bite and punch to percussive sounds. This will only be audible if Amp Attack (Easy Page, Filters/Env Page) is set to a very low value.

Unison Mode

In Unison mode, the Virus PowerCore will trigger between 2 and 16 instances of the same voice for every key played, depending on the chosen value. The voices are then detuned and spread across the stereo field, resulting in a much fuller sound than is otherwise possible.



Be aware that Unison mode will reduce the available polyphony by a factor equal to the number of unison voices!! For instance, a sound that uses Unison Mode = Twin, will have a maximum polyphony of 8 voices.

> MODE

Use this to switch Unison Mode On or Off and specify the number of voices to be played in unison.

> DETUNE*

This determines to what degree the unison voices are detuned against each other. Higher values will result in a warmer, thicker sound.

> PAN SPREAD*

This determines the stereo separation between each voice. At 0, all voices are panned centre, at 127 the voices are spread evenly across the entire stereo field.

NOTE: If the filters are in Split Mode, then Pan Spread will also affect the stereo separation of the signals from filter 1 and 2.

> LFO PHASE*

This determines the phase offset between the LFOs in each voice. At extreme settings, the LFOs of Voice 1 will modulate in opposite phase to the LFOs of Voice 2, resulting in a much busier sound.

NOTE: *This parameter will have no effect if Unison Mode is Off.

Mixer

Here you will find the controls for balancing the different sound sources within the oscillator section.



> OSC 1/2 BAL

This adjusts the balance between osc1 and osc 2. At -64 you will hear only osc 1, and at +63 only osc 2. At +0 (default) both oscillators are equally balanced.

> OSC 3 VOL

This adjusts the volume of oscillator 3. At a setting of 0, osc 3 will be inaudible (but still active, so if you don't intend to hear it at all, better switch it off and free up some polyphony!). To balance it evenly with osc 1 and 2, try setting Osc 3 Vol to 64.

NOTE: This parameter will have no effect if Osc 3 > Osc Mode is set to Off.

> SUB VOL

This adjusts the volume of the sub-oscillator, which is used to make the sound fuller and deeper by adding an extra tone at a pitch of one octave below osc 1.

> **OSC VOL**

This adjusts the *overall* volume of oscillators 1, 2, 3 and the sub-oscillator in relation to the ring modulator and noise generator. At a setting of -64, only the ring modulator and noise are audible. At +0, the oscillators are at maximum volume, and from +0 upwards only the level of filter saturation is affected. If no saturation curve has been selected (see Filter/Env Page), then values between +0 and +63 will have no effect.

If the Virus PowerCore is in Input mode, then Osc Vol can be used to adjust the level of the input signal.

> **NOISE VOL**

This adjusts the volume of the white noise generator. Noise is often used to create snare drum or other percussive sounds, but can also be used to create wind and surf fx or add extra warmth to the oscillators.

> **PAN**

This adjusts the position of the sound in the stereo field. Virus PowerCore must be enabled as a stereo insert for this to have any effect.

> **PATCH VOL**

This adjusts the overall volume of the patch (sound).

> **RING VOL**

This adjusts the volume of the ring modulator, which multiplies the output of osc 1 and 2 to create additional harmonic overtones. The result is highly dependent on both the pitch and waveform of each oscillator. For bell-like tones, try setting both oscillators to a sine wave and experiment with the semitone settings of each.

> **FX SEND**

This determines how much of the dry signal is sent to the delay/reverb effect. At 0, the signal is dry. At 127, only the output of the delay/reverb is audible.

The Filter/Env Page



The filter is arguably the most important section of any subtractive synthesizer – it is here that you determine the overall character of the sound. Filters work by removing certain frequencies from the signal, leaving only those that are required. The exact way in which this is done is different for every synthesizer, which is one reason why they all have such a distinctive character.

The Virus PowerCore is endowed with two filters with a flexible routing system and several modes of saturation, making it possible to achieve all manner of filtering effects.

The controls in this section determine the way in which the signal is routed through the filters (Series, Parallel or Split) and the kind of saturation (distortion/drive) applied to the output of filter 1.

> SATURATION

Saturation is the term used to describe the distortion effect applied to the output of filter 1. Some types just add overtones, whilst others can be used to completely warp the sound beyond all recognition. Saturation is always post-filter 1 and pre-filter 2, so you can use filter 2 to remove unwanted overtones if you like. Choose between the following types:

Off, Light, Soft, Middle, Hard, Digital, Shaper, Rectifier, BitReduce, RateReduce, Rate+Flw, LowPass*, Low+Flw*, HighPass* and High+Flw*.

NOTE: *Flw* means that the amount of saturation applied will be adjusted according to the MIDI note number (Keyfollow).

NOTE: *These last four options aren't really saturation effects at all, instead they are additional 1-pole filters.

> OSC VOL

Between values of -64 and 0, this adjusts the overall level of the 3 oscillators and the sub-oscillator. From 0 to +63, this increases the level of saturation applied to filter 1. If Saturation is set to Off, then values above 0 will have no effect.

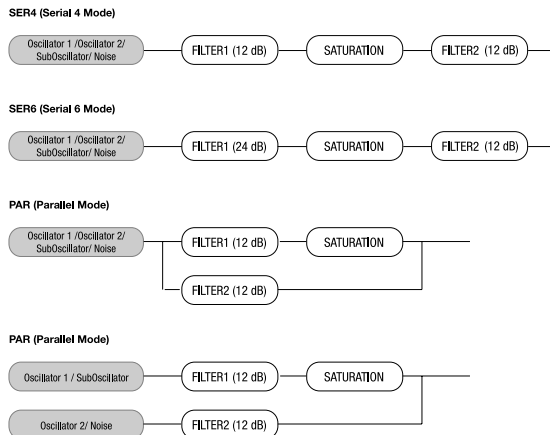
> FILTER ROUTING

Determines the way in which the output of the mixer section is routed through the filters.

NOTE: *The more poles a filter has, the more severe the filtering effect.*

- > **Serial 4:** The output of the mixer section is routed through filter 1 (+saturation) into filter 2. Both filters are 2 pole, with a 12dB/Oct slope. This results in a 4 pole filter with a 24 dB slope.
- > **Serial 6:** The output of the mixer section is routed through filter 1 (+saturation) into filter 2. Filter 1 is 4 pole, 24 dB/Oct, and filter 2 is 2 pole, 12 dB/Oct. This results in a 6 pole filter with a 36 dB slope.
- > **Parallel 4:** The output of the mixer section is routed through filter 1 (+saturation) and filter 2 simultaneously. Both filters are 2 pole with a 12 dB/Oct slope. In this configuration, the saturation applied to filter 1 is not affected by filter 2.

> **Split** : Like Parallel 4 mode, the filters run in parallel, with 2 poles each (12 dB/Oct), but the difference here is that the filters each receive different elements of the oscillator section. Oscillator 1 and the sub-oscillator are routed through filter 1 (+saturation), whilst oscillators 2, 3 and noise are routed through filter 2. Filter 1 is panned hard left, and filter 2 is panned hard right. The depth of panning can be controlled via Unison Pan Spread (Osc Page).



> FILTER BALANCE

This balances the output levels of the two filters. The default setting is 12 o'clock (0).

In Serial 4 and Serial 6 modes, this means that the output of filter1 is routed through filter2, and then to the amplifier section. If filter balance is set to a negative value, then some of the signal from filter1 (+ saturation) bypasses filter2 and the output of filter2 is also reduced. With filter balance set to -64 only the output of filter1 (+ saturation) is audible. When positive values are used, some of the mix signal bypasses filter1 and is sent directly into filter2, whilst the output of filter1 is also reduced. With filter balance set to +63, filter1 is bypassed completely and only the output of filter2 is audible.

In Parallel and Split modes, Filter Balance simply balances the outputs of filter1 and filter2.

Filter 1

> MODE

Select from the following filter types:

> **Low Pass:** The most commonly used filter type. Removes frequencies higher than the cutoff point.

> **High Pass:** Removes frequencies lower than the cutoff point.

> **Band Pass:** Allows only a narrow band of frequencies through - Resonance determines the width of the band in this case.

> **Band Stop:** The opposite of Band Pass, this filter removes a narrow band of frequencies. Often referred to as a 'notch' filter, as it effectively cuts a notch in the tonal spectrum. Resonance determines the width of the notch.

> CUTOFF

This determines the frequency above or below which filter 1 will come into effect. For instance in a Low Pass filter, frequencies below this point are allowed to pass, and those above are removed according to the slope of the filter.

> LINK (CUTOFF)

Synchronises the cutoff frequencies of both filters. When switched to ON, filter 2 cutoff tracks filter 1 cutoff. The offset between the two determined by the Filter 2 Cutoff controller.

> RESONANCE

In Low Pass or High Pass modes, this determines the emphasis of those frequencies near to the cutoff point. As you increase the level it results in a more nasal or 'honking' tone. In Band Pass and Band Stop modes, increasing the resonance reduces the width of the band.

> ENV AMT

This determines to what degree the cutoff frequency of filter 1 is affected by the filter envelope.



> **KEYFOLLOW**

This determines to what extent the cutoff frequency of filter 2 is affected by MIDI note number. Positive values will cause the cutoff frequency to increase as you play up the keyboard, negative values will cause it to increase as you play down the keyboard.

> **RESO VEL**

This determines to what extent the resonance of filter 1 is affected by MIDI velocity. Positive values mean as you play harder, the resonance *increases* in intensity. Negative values mean that as you play harder, the resonance *decreases* in intensity.

> **ENV VEL**

This determines to what extent the Filter 1 Env Amt is affected by MIDI velocity. Positive values result in velocities of less than 127 applying a negative offset to the current value of Filter 1 Env Amt. And if you understand that, then you're doing very well indeed!

I think a practical example is in order:

Set Env Vel to +63, and Filter 1 Env Amt to 127. At this setting, playing as gently as possible will result in the filter envelope having no effect on the cutoff frequency of filter 1. As you play harder, the filter envelope has an increasingly noticeable effect until, at maximum velocity, the effect is equivalent to Filter 1 Env Amt = 127 again.

If Env Vel is set to -64, and filter 1 Env Amt is set to 127, then playing as gently as possible will result in the filter envelope having maximum effect on the cutoff frequency of filter 1. As you play harder, the filter envelope has less and less effect until, at minimum velocity, the effect is equivalent to Filter 1 Env Amt = 0.

NOTE: *Positive values are the most common application of this parameter, as it simulates the way in which the tone of an acoustic instrument becomes brighter the louder it is played. Many synth sounds benefit from this effect as well.*

> **ENV POLARITY**

Determines which way the filter envelope will sweep the cutoff frequency of filter 1. Normal polarity means that attack sweeps upwards and decay sweeps downwards. Inverse polarity effectively turns the filter envelope upside-down i.e. attack sweeps downwards and decay sweeps upwards.

> **LINK (ENVELOPE AMOUNT, RESONANCE, KEY FOLLOW)**

When enabled, Env Amt, Resonance and Key Follow become linked with their counterparts in filter 2, so that both filters can be adjusted simultaneously.

> **KEY BASE**

This determines the MIDI note number from which the keyfollow parameter will start to have an effect. To set the key base, either type in the MIDI note number in the text window, or click on the button and press a key on your controller keyboard or the onscreen keyboard.

Filter 2

> **MODE**

Selects the filter type:

> **Low Pass:** The most commonly used filter type. Removes frequencies higher than the cut-off point.

> **High Pass:** Removes frequencies lower than the cutoff point.

> **Band Pass:** Allows only a narrow band of frequencies through - Resonance determines the width of the band in this case.

> **Band Stop:** The opposite of Band Pass, this filter removes a narrow band of frequencies. Often referred to as a 'notch' filter, as it effectively cuts a notch in the tonal spectrum. Resonance determines the width of the notch.

> **CUTOFF**

This determines the frequency above or below which filter 2 will come into effect. For instance in a lowpass filter, frequencies below this point are allowed to pass, and those above are attenuated according to the slope of the filter.

> **RESONANCE**

In Low Pass or High Pass modes, this determines the emphasis of those frequencies near to the cutoff point. As you increase the level it results in a more nasal or 'honking' tone. In Band Pass and Band Stop modes, increasing the resonance reduces the width of the band.

> **ENV AMT**

This determines to what degree filter 2 cutoff is affected by the filter envelope.

> **KEY FOLLOW**

This determines to what extent the filter 2 cutoff is affected by MIDI note number. Positive values will cause the cutoff frequency to increase as you play up the keyboard, negative values will cause it to increase as you play down the keyboard.

> **RESO VEL**

This determines to what extent the resonance of filter 2 is affected by MIDI velocity. Positive values mean as you play harder, the resonance *increases* in intensity. Negative values mean that as you play harder, the resonance *decreases* in intensity.

> **ENV VEL**

This determines to what extent Filter 2 Env Amt is affected by MIDI velocity. Positive values result in velocities of less than 127 applying a negative offset to the current value of Filter 2 Env Amt. And if you understand that, then you're doing very well indeed!

I think a practical example is in order:

Set Env Vel to +63, and Filter 2 Env Amt to 127. At this setting, playing as gently as possible will result in the filter envelope having no effect on the cutoff frequency of filter 2. As you play harder, the filter envelope has an increasingly noticeable effect until, at maximum velocity, the effect is equivalent to Filter 2 Env Amt = 127 again.

If EnvVel is set to -64, and Filter 2 Env Amt is set to 127, then playing as gently as possible will result in the filter envelope having maximum effect on the cutoff frequency of filter 2. As you play harder, the filter envelope has less and less effect until, at minimum velocity, the effect is equivalent to Filter 2 Env Amt = 0.

Positive values are the most common application of this parameter, as it simulates the way in which the tone of an acoustic instrument becomes brighter the louder it is played. Many synth sounds benefit from this effect as well.

> **ENV POLARITY**

Determines which way the filter envelope will sweep the cutoff frequency of filter 2. Normal polarity means that attack sweeps upwards and decay sweeps downwards. Inverse polarity effectively turns the filter envelope upside-down i.e. attack sweeps downwards and decay sweeps upwards.

Filter Envelope

The filter envelope is used to automatically control the cutoff frequency over a certain time period. In order for these parameters to have any effect on the sound, Filter 1 Env Amt and/or Filter 2 Env Amt must be set to a positive value.



> **ATTACK**

This determines how long it takes for the envelope to reach its peak.

> **DECAY**

This determines how long it takes for the envelope to decay from its peak to the sustain level.

> **SUSTAIN**

This determines the level at which the envelope will sustain.

> **SUSTAIN TIME**

This determines how long the envelope will remain at the sustain level. At 0, the sustain level will be maintained as long as the key is held. At positive values, the envelope will rise back up to its peak level (the higher the value, the quicker the rise). At negative values, the envelope will sweep down to its minimum level (the lower the value, the quicker the fall).

> **RELEASE**

This determines how long the envelope will take to fall to its minimum level after the key is released. The higher the value, the longer it will take.

Amplifier Envelope

The amp envelope affects patch volume in the same way that the filter envelope affects frequency cutoff. Use this to automatically control the volume of a sound over time.



> ATTACK

This determines how long it takes for the sound to reach maximum volume.

> DECAY

This determines how long it takes for the sound to decay from maximum volume to the sustain volume.

> SUSTAIN

This determines the volume at which the sound will sustain as long as the key is held.

> SUSTAIN TIME

At +0, the sound will remain at the sustain level so long as the key is held. At positive values, the volume will rise to maximum (the higher the value, the quicker the rise). At negative values, the volume will fall to silence (the lower the value, the quicker the fall).

> RELEASE

This determines how long it takes for the sound to fade to silence once the key is released. Higher values result in a slower fade.

The LFO Page



LFO is an abbreviation of **Low Frequency Oscillator**. Basically, these generate waveforms that oscillate so slowly that they are only useful as a modulation source for other parameters, such as pitch, cutoff, panorama etc. The Virus PowerCore has 3 LFOs.

LFO 1

The controls in this section are used to affect the characteristics of LFO 1, and determine which parameters it will modulate.

> RATE

This determines how fast LFO 1 will oscillate. The higher the value, the faster it will oscillate.

> CLOCK

This overrides Rate and forces LFO 1 to sync its wave cycle to the clock control. The fractions are relative to a whole note, or semi-breve, which in turn are equivalent to a whole bar/measure in 4/4 time. So, by choosing a clock rate of 1/4, LFO 1 will sync to a quarter note (crotchet) cycle, whereas a setting of 4/1 will cause it to sync to cycle of 4 bars.

> SHAPE

This determines the waveform of LFO 1 - different waves can result in very different modulation effects. Choose between the following options:



>**Sine**: A smooth up/down cyclic wave.



>**Triangle**: Similar to sine wave, but with sharper peaks and a linear curve.



>**Sawtooth**: This will cause the sound to rise/fall, then jump back suddenly to rise/fall again.



>**Square**: Use this to jump back and forth between two values.



>**Sample and Hold (Random)**: This isn't a waveform as such – instead LFO1 will generate random values, causing the affected parameter(s) to jump around unpredictably.



>**Sample and Glide**: Like Sample and Hold, this generates random values between which the LFO will interpolate smoothly, causing the affected parameter(s) to slide up and down unpredictably.





>**Oscillator Waveform** : Selecting this button will open up a window from which you can choose any of the 62 spectral waves.

> **CONTOUR**

This determines the contour or slope of the currently selected waveform. It basically has the effect of 'squashing' the waveform from one side or the other, depending on whether you enter a positive or negative value. For instance by using extreme values you can turn a triangle wave into a rising or falling 'ramp' wave. You can also use this parameter to adjust the pulse width of the pulse (square) waveform.

***NOTE:** If the current waveform is one of the spectral waveforms, then negative values will 'zoom' into the wave. Positive values will have no effect on the spectral waveforms.*

> **ENV MODE**

When enabled, the LFO will only cycle through the waveform once. This effectively turns the LFO into an additional envelope which is shaped according to the currently selected waveform.

> **KEY FOLLOW**

This determines to what extent LFO rate is affected by MIDI note number. When set to Off, the LFO will oscillate at the same frequency regardless of which key is played. When set to 127, the frequency will increase the higher you play up the keyboard. Lesser values result in a more subtle increase.

> **KEY TRIG**

When set to Off, the LFO is free-running. When set to any other value, the LFO is forced to start its cycle at the same point as soon as a key is played. Changing the value moves the start point along the wave cycle.

> **LFO MODE**

When set to Mono, LFO 1 is applied to each voice (note) in sync. When set to Poly, each voice is modulated independently.

***NOTE:** Note: there will only be a difference between these two modes as long as **Key Trig** is set to **Off** and **LFO 1** is **not in clock mode**.*

> **OSC 1**

Determines the amount of modulation applied to the pitch of osc1.

> **OSC 2**

Determines the amount of modulation applied to the pitch of osc 2.

> **LINK (OSC1 & OSC2)**

When enabled, adjusts the amounts of LFO 1-> Osc 1 and LFO 1->Osc 2 simultaneously.

> **PW 1+2**

Determines the amount of modulation applied to the pulse widths of oscillators 1 and 2. One or both of these oscillators must be set to a square wave for this parameter to have an effect.

> **RESO 1+2**

Determines the amount of modulation applied to the resonance of filters 1 and 2.

> **ASSIGN AMT**

Determines the amount of modulation applied to the parameter selected in the Assign Destination window.

> **ASSIGN DESTINATION**

Select virtually any parameter in the Virus PowerCore to be modulated by LFO 1.

> **FILTER GAIN**

Determines the amount of modulation applied to the input gain to the filter section. This is often used to create tremolo effects (try a sine or triangle wave).

LFO 2

The controls in this section are used to affect the characteristics of LFO 2, and determine which parameters it will modulate. Many of the parameters are identical in function to their counterparts in LFO 1.

> RATE

This determines how fast LFO 2 will oscillate. The higher the value, the faster it will oscillate.

> CLOCK

This overrides Rate and forces LFO 2 to sync its wave cycle to the clock control. The fractions are relative to a whole note, or semi-breve, which in turn are equivalent to a whole bar/measure in 4/4 time. So, by choosing a clock rate of 1/4, LFO 2 will sync to a quarter note (crotchet) cycle, whereas a setting of 4/1 will cause it to sync to cycle of 4 bars.

> SHAPE

This determines the waveform of LFO 2 - different waves can result in very different modulation effects. Choose between the following options:



>**Sine**: A smooth up/down cyclic wave.



>**Triangle**: Similar to sine wave, but with sharper peaks and a linear curve.



>**Sawtooth** : This will cause the sound to rise/fall, then jump back suddenly to rise/fall again.



>**Square**: Use this to jump back and forth between two values.



>**Sample and Hold (Random)**: This isn't a waveform as such – instead LFO1 will generate random values, causing the affected parameter(s) to jump around unpredictably.



>**Sample and Glide** : Like Sample and Hold, this generates random values between which the LFO will interpolate smoothly, causing the affected parameter(s) to slide up and down unpredictably.





>**Oscillator Waveform** : Selecting this button will open up a window from which you can choose any of the 62 spectral waves.

Clicking on this box will open up a window from which you can choose any of the 62 spectral waves.

> **CONTOUR**

This determines the contour or slope of the currently selected waveform. It basically has the effect of 'squashing' the waveform from one side or the other, depending on whether you enter a positive or negative value. For instance by using extreme values you can turn a triangle wave into a rising or falling 'ramp' wave. You can also use this parameter to adjust the pulse width of the pulse (square) waveform.

***NOTE:** If the current waveform is one of the spectral waveforms, then negative values will 'zoom' into the wave. Positive values will have no effect on the spectral waveforms.*

> **LFO MODE**

When set to Mono, LFO 2 is applied to each voice (note) in sync. When set to Poly, each voice is modulated independently.

Note: there will only be a difference between these two modes as long as **Key Trig** is set to **Off** and LFO 2 is **not in Clock Mode**.

> **KEY FOLLOW**

This determines to what extent LFO 2 Rate is affected by MIDI note number. When set to Off, LFO 2 will oscillate at the same frequency regardless of which key is played. When set to 127, the frequency will increase the higher you play up the keyboard. Lesser values result in a more subtle increase.

> **FILTER 1**

Determines the amount of modulation applied to the cutoff frequency of filter .

> **FILTER 2**

Determines the amount of modulation applied to the cutoff frequency of filter 2.

> **LINK (FILTER 1& FILTER 2)**

When enabled, adjusts the cutoff frequencies of filter 1 and filter 2 simultaneously.

> **SHAPE 1+2**

Determines the amount of modulation applied to osc 1 Shape and osc 2 Shape.

> **PANORAMA**

Determines the amount of modulation applied to the stereo pan position.

> **ASSIGN AMOUNT**

Determines the amount of modulation applied to the parameter selected in the Assign Destination window.

> **ASSIGN DESTINATION**

Select virtually any parameter in the Virus PowerCore to be modulated by LFO 2.

> **FM AMT**

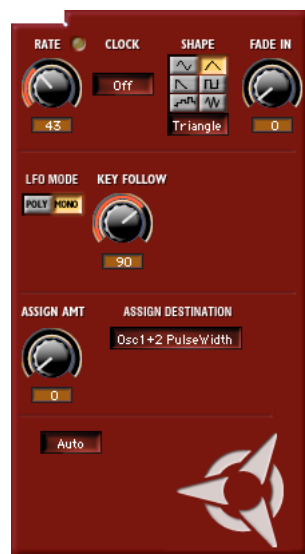
Determines the amount of modulation applied to osc 2 FM amount.

LFO 3

The controls in this section are used to affect the characteristics of LFO 3, and determine which parameter(s) it will modulate. Many parameters are identical in function to their equivalents in LFOs 1 and 2.

NOTE: Note: the functionality of LFO 3 is not the same as that of LFO 1 and 2. This LFO is designed for creating vibrato effects, and as such only has one destination slot.

NOTE: Tip: If you want LFO 3 to modulate other parameters, you can choose it as a Source in the Mod Matrix (See ModMatrix page).



> RATE

This determines how fast LFO 3 will oscillate. The higher the value, the faster it will oscillate.

> CLOCK

This overrides Rate and forces LFO 3 to sync its wave cycle to the clock control. The fractions are relative to a whole note, or semibreve, which in turn are equivalent to a whole bar/measure in 4/4 time. So, by choosing a clock rate of 1/4, LFO 3 will sync to a quarter note (crotchet) cycle, whereas a setting of 4/1 will cause it to sync to cycle of 4 bars.

> SHAPE

This determines the waveform of LFO 3 - different waves can result in very different modulation effects. Choose between the following options:



>**Sine:** A smooth up/down cyclic wave.



>**Triangle:** Similar to sine wave, but with sharper peaks and a linear curve.



>**Sawtooth :** This will cause the sound to rise/fall, then jump back suddenly to rise/fall again.



>**Square:** Use this to jump back and forth between two values.



>**Sample and Hold (Random)**: This isn't a waveform as such – instead LFO1 will generate random values, causing the affected parameter(s) to jump around unpredictably.



>**Sample and Glide** : Like Sample and Hold, this generates random values between which the LFO will interpolate smoothly, causing the affected parameter(s) to slide up and down unpredictably.



>**Oscillator Waveform** : Selecting this button will open up a window from which you can choose any of the 62 spectral waves.

> **FADE IN**

If set to 0, LFO 3 will be fully effective as soon as a note is played. At values of 1 – 127, LFO 3 takes a certain amount of time to develop in intensity. Use this to simulate how a string or woodwind player gradually introduces vibrato on longer notes.

> **LFO MODE**

When set to Mono, LFO 3 is applied to each voice (note) in sync. When set to Poly, each voice is modulated independently.

Note: there will only be a difference between these two modes as long as **Key Trig** is set to **Off** and LFO 3 is **not in Clock Mode**.

> **KEY FOLLOW**

This determines to what extent LFO Rate is affected by MIDI note number. When set to Off, the LFO will oscillate at the same frequency regardless of which key is played. When set to 127, the frequency will increase the higher you play up the keyboard. Lesser values result in a more subtle increase.

> **ASSIGN AMOUNT**

Determines the amount of modulation applied to the parameter selected in the Assign Destination window.

> **ASSIGN DESTINATION**

Choose the parameter(s) to be modulated by LFO 3. Choose between:

> **Osc1 Pitch**: LFO 3 will modulate the pitch of oscillator 1.

- > **Osc1+2 Pitch:** LFO 3 will modulate the pitch of oscillator 2.
- > **Osc2 Pitch:** LFO 3 will modulate the pitch of osc1 and oscillator 2.
- > **Osc1 PulseWidth*:** LFO 3 will modulate the pulse width of oscillator 1.
- > **Osc1+2 PulseWidth*:** LFO 3 will modulate the pulse width of oscillator 1 and oscillator 2.
- > **Osc2 PulseWidth*:** LFO 3 will modulate the pulse width of oscillator 2.
- > **Osc2 SyncPhase:** LFO 3 will modulate the phase position of oscillator2 when it is synced to oscillator1 via the sync function (Osc Page).

* This will only have an audible effect if the relevant oscillators are set to generate a pulse wave (Osc Page).

The FX-1 Page



Here you will find the controls for the Analog Boost, Input Ring Modulator, Phaser, Patch Distortion, Chorus and Delay/Reverb effects as well as the output selection for the current part.

Analog Boost

This effect imbues your sound with the kind of bottom-end warmth more commonly associated with analogue synthesizers, by emphasizing certain frequencies.



> INTENSITY

This determines the degree of emphasis applied to the sound.

> TUNE

This determines the central frequency around which the emphasis will occur.

Ring Mod

With this effect you can ring-modulate an audio input signal with the oscillators of the Virus PowerCore. This means that the signals from the inputs and the oscillators are multiplied. The end result will vary greatly depending on the settings of the oscillators – try choosing a sine wave for both oscillator 1 and 2, and experiment with different pitches for each.



> MIX

This determines the balance between the oscillators, ring mod and audio input signals. At 0, the effect is Off, and only the output of the oscillators can be heard. As the value increases from 1 to 64 the level of the ring mod is increased until at 64, only the ring modulation is audible. As the value increases from 65 towards 127, the ring mod signal is dialed out of the mix, being gradually replaced with the audio input signal (see “Using the Audio Inputs” on page 83). At 127, only the audio input signal is audible.

Phaser

Use the phaser to achieve the sweeping, swooshing effect made popular in the 60s and 70s by many psychedelic and prog-rock artists.



The phaser will also turn a mono sound into stereo.

> **MIX**

Determines the balance between the dry signal and the phaser effect. At 0, the phaser is Off. At 127, the signal is 100% wet.

> **RATE**

Determines how fast the phaser will sweep through the frequency spectrum.

> **DEPTH**

Determines the frequency range through which the phaser will sweep. The higher the value, the greater the intensity of the phasing effect.

> **FREQUENCY**

Determines the center frequency about which the phaser will sweep.

> **STAGES**

Determines the number of filter stages used to create the phasing effect. The higher the number, the more complex the effect becomes.

> **SPREAD**

Determines the bandwidth of the phaser's filters.

> **FEEDBACK**

Determines the amount of signal to be fed back into the phaser. Experiment with both positive and negative values for different types of extreme phasing.

Patch Distortion

Use Patch Distortion to add saturation or clipping effects to the patch as a whole. Although the list is very similar to that of filter saturation (see Filter Page), the effect is quite different, as all the voices (notes) go through the same instance of the distortion effect. This means that the intensity of the distortion is enhanced as you play more notes simultaneously, whereas filter saturation is applied to each voice individually.



> INTENSITY

Determines the level of distortion. In the case of LowPass and HighPass (1-pole filters), this determines the cutoff frequency.

> CURVE

Choose from Off, Light, Soft, Middle, Hard, Digital, Shaper, Bit Reducer, Rate Reducer, Low-Pass and HighPass.

Chorus

Use the renowned Virus chorus effect to add depth and movement to the sound. The chorus works by adding a delayed copy to the original signal and modulating the delay time with an LFO to create a detuned effect. Like the phaser, this will also convert a mono sound into stereo.



> MIX

Determines the balance between the original signal and the chorus signal. At 0 the effect is Off, at 64, the balance is equal, and at 127 only the output of the chorus is audible. It usually makes sense to set Mix at a value somewhere between 0 and 64, as both signals are required to achieve the chorusing effect.

> RATE

Determines the rate of the chorus LFO.

> **DEPTH**

Determines the depth of modulation applied to the chorus delay time. Use this to increase the intensity of the chorusing effect.

> **DELAY TIME**

Determines the time-delay between the original signal and the output of the chorus effect. Longer times result in a warmer sound, whilst very short times result in a flanging effect.

NOTE: Tip:: Set Chorus Delay Time to 0 and increase the Chorus Feedback amount to boost the levels of quieter sounds.

> **FEEDBACK**

Determines the amount of signal to be fed back into the chorus. At extreme values this can create an intense, howling effect which can easily overload your equipment and your ears, so approach with caution!

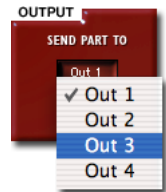
> **SHAPE**

Determines the wave shape of the chorus LFO. Triangle is the default, but you might want to experiment with the other options for more exotic effects.

Output

The Virus PowerCore has 4 pairs of stereo output channels. Here you can choose to which output channel the currently selected part is sent.

NOTE: Remember the signal being sent to any particular output channel is pre-delay/reverb. The delay/reverb signal is always sent to output 1.



Delay/Reverb

Use the controls in this section to determine the characteristics of the delay/reverb effect. Delay is used to create complex echo effects and reverb is used to simulate the effect of being in an acoustic space, such as a room or hall.

> USE GLOBAL FX FROM

The Virus PowerCore has just one global delay or reverb per DSP, therefore only one part can control the parameters of the Delay/Reverb at any given time.

Click on this box to select which Part shall control the Delay/Reverb effect.

All currently active Parts sharing the same DSP will use the delay/reverb settings from the Part selected here. The delay/reverb controls will be greyed out in the other three Parts.

NOTE: Remember that although there is only one Global Delay/Reverb per DSP, each Part still has it's own individual Send parameter.

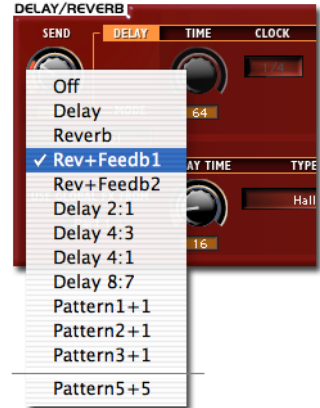
> MODE

Select the type of effect to be used. Options include Delay, Reverb, Reverb+Feedback and several 'Groove' delays which can be used for complex rhythmic effects

> **Off:** When switched to Off, the Delay/Reverb effect is deactivated.

> **Delay:** Use this for a simple echo effect where each repeat is evenly spaced.

> **Reverb:** Use this to create the effect of an acoustic space, such as a room or a hall.



- > **Reverb+Feedback 1:** This variation of the reverb algorithm feeds the pre-delay signal back into itself to create rhythmic repetitions of the reverb signal.
- > **Reverb+Feedback 2:** This behaves in a similar fashion to Reverb+Feedback 1, with the only difference being that the Reverb is audible immediately after the dry signal.

NOTE: Tip: When using either of the Reverb+Feedback algorithms, try using Ambience or Small Room reverb type, as both of these offer reasonably long pre-delay times. Set a short reverb time, a long pre-delay and set reverb feedback to a high value. These settings should give you a good indication of the purpose of these algorithms.

- > **Delay X:Y:** These are 'ping-pong' delays, which means that the delay times for the left and right channels are different, making the sound appear to bounce back and forth between the speakers, like a ping-pong ball. The different ratios refer to the relationship between the left and right signals, i.e. 2:1 means that the left delay time is twice the length of the right delay time.
- > **Pattern X+Y:** These 'Pattern' delays are also ping-pong delays, but instead of referring to the delay time or clock parameters, they refer instead to the global tempo. The two numbers represent how many sixteenth note (semiquaver) increments the left and right channels will be delayed by. Note that neither Delay Time nor Delay Clock will have any effect in this mode.

NOTE: Tip: Try using these 'Groove' delays in conjunction with the arpeggiator to create interesting rhythmic patterns.

> SEND

Determines the amount of the dry signal to be sent to the delay/reverb effect. This behaves exactly like a post-fader effect bus on a mixing console. As the value nears 127, the dry signal is faded out and only the delay/reverb signal is audible.

Delay



> TIME

Determines the length of the delay time in milliseconds, to a maximum of 693.6ms.

> CLOCK

This slaves the delay time to the global tempo or incoming MIDI Clock signal. Choose a note value that represents the desired delay time i.e. a Clock value of 1/4 will result in quarter note (crotchet) repeats.

NOTE: Note: The maximum delay time is 693.6 ms, so if the current clock setting requires a delay time longer than this, it will automatically be adjusted to half the value i.e. 1/2 will sound as 1/4.

> FEEDBACK

Determines the amount of signal to be fed back into the delay. The higher the value, the more repetitions you will hear.

> COLORATION

Determines the character of the delay by applying a filter to the delayed signal. At +0 the character is unchanged. Negative values apply a lowpass filter, resulting in a duller, warmer sound. Positive values apply a highpass filter, resulting in a much thinner, brighter sound.

> RATE

Determines the rate of modulation applied to the delay time by the delay LFO.

> DEPTH

Determines the amount of modulation applied to the delay time. Higher values result in strange effects that can be made even more interesting if Rate is set to a high value.

> SHAPE

Determines the wave shape for the delay LFO. The default setting is Triangle, but you might want to experiment with the other shapes for more extreme effects.

NOTE: Using different shapes for the delay LFO can result in audible artefacts.

Reverb

Use these controls to affect the various characteristics of the reverb effect.



> DECAY TIME

Determines the length of the decay or 'tail' of the reverberation.

> TYPE

Select from one of 4 types of reverb: Ambience, Small Room, Large Room or Hall.

NOTE: Ambience is useful for creating a sense of space around a sound, but isn't the most realistic of the available algorithms. The advantage is that, due to the lower calculation required, a much longer pre-delay is possible than with the other types.

> DAMPING

Determines the degree of high frequency damping. As a sound bounces around a room, the high frequencies get gradually attenuated, especially by soft furnishings, carpets etc. As the value increases, the more quickly the high frequencies are damped in the reverb tail.

> COLORATION

Determines the character of the reverb tail. This behaves exactly like Delay Coloration, by applying a lowpass filter (negative control range) or highpass filter (positive control range) to the reverb signal. To create natural sounding reverb effects, you should only use negative amounts, which will result in a warmer, darker sound. If you wish to experiment with thin reverbs where the lower frequencies are attenuated instead, then try positive values.

> PRE-DELAY

Determines the time delay between the dry signal and the start of the room reflections. Longer values can be useful for creating the illusion of a wider space.

The maximum pre-delay times available depend on the reverb type, ranging from 150ms (Hall) to 500ms (Ambience).

> **CLOCK**

Selects a clock-synchronised pre-delay time. This is particularly useful for the Reverb+Feedback algorithms, as the repeats can be synchronised in the same way as in the delay. (See Delay Clock for further explanation.)

> **FEEDBACK**

This parameter is only functional in the Reverb+Feedback modes. It determines the amount of signal to be fed back into the reverb. Higher values result in more repeats of the early reflections. To emphasise this effect, try using short reverb times and a long pre-delay.

The FX2/Global Page



On this page you will find the controls for the Vocoder, Envelope Follower, External Audio Input, Arpeggiator as well as Global Tuning, Key Mode, Pitch Wheel etc.

Vocoder

> ABOUT THE VOCODER

The Vocoder is used to take the harmonic and dynamic properties of one signal (the modulator) and impose them on another (the carrier). The most widely known application of this is to make ‘talking’ synth sounds by using a voice as the modulator and an oscillator waveform as the carrier. Those of you old enough to remember Sparky’s Magic Piano will know this effect well, but the most famous proponents of the vocoder are probably Kraftwerk, who have made extensive use of this effect. Another popular and highly effective technique is to use the Vocoder to process drum loops.



So how does it work? Well, the vocoder applies a series of bandpass filters and envelope followers to the modulator signal and uses the resulting signal to modulate the equivalent frequencies in the carrier signal. Widely different results can be achieved depending on the number of bands used, the sensitivity of the envelope followers and the frequency content of each signal.

NOTE: The vocoder is the most complex of the Virus PowerCore’s effects, and takes a lot of DSP calculating power; therefore the filter section becomes disabled when you use it, and polyphony is reduced. If you need to filter the output of the vocoder, then open up another instance of the Virus PowerCore, and use the audio inputs to process the output of the first (Extended License only).

NOTE: In order to make use of the vocoder, you may need to instance the Virus PowerCore as a side-chain effect. See the chapter “Using the Audio Inputs” for more information.

> MODE

Use this to select the source of the carrier signal.

- > **Off:** The vocoder is disabled.
- > **Osc:** The output of the oscillators, including noise and ring mod, provides the signal for the carrier.
- > **Osc Hold:** The same as Osc mode, but with key latch.
- > **Noise:** The output of the noise generator is used for the carrier signal.
- > **Input:** The audio inputs provide the signal for the carrier. This means that the same signal is used for both carrier and modulator, which can result in some very interesting effects.

- > **CENTER FREQ**
Determines the central frequency of the bandpass filters in the modulator and carrier banks.

- > **MOD OFFSET**
Use this to offset the center frequency of the modulator bank relative to the carrier bank.

- > **MOD Q**
Determines the width of the gap between the bandpass filters in the modulator bank. This is unlikely to have more than a very subtle effect on the sound.

- > **MOD SPREAD**
Determines the spread of the bandpass filters in the modulator bank. Lower values mean a narrower spectrum of the modulator signal is applied to the carrier signal. A value of +63 means that the whole frequency spectrum of the modulator signal is applied to the carrier signal.

- > **CARR Q**
Determines the resonance or quality of the filters bands in the carrier bank. Lower values will result in a fairly neutral tone, whereas higher values result in a more artificial sound.

- > **LINK**
With Link enabled, Carr Q and Mod Q are adjusted simultaneously, as are Mod Spread and Carr Spread.

> **CARR SPREAD**

Determines the spread of the bandpass filters in the carrier bank. Low values result in a narrow frequency spectrum, whilst a value of +63 results in the entire frequency spectrum being covered in the carrier signal. The value chosen for this parameter can have a dramatic affect on the intelligibility of speech.

> **SPECTRAL BAL**

Determines the spectral balance between the higher and lower frequencies of the vocoder signal. This parameter has a lot of influence on the overall character of the vocoder.

> **BANDS**

Determines the number of bandpass filters used by the vocoder. It is possible to choose between 1 and 32 filter bands. Less filter bands result in a more typical vocoder sound, whereas more filter bands increase the intelligibility of speech. The more filter bands there are, the more calculating power is required, so polyphony might vary depending on the number of active filter bands.

> **ATTACK**

Determines the response time of the vocoder's envelope followers. With longer attack times, the response of the vocoder will become slower, reducing the intelligibility of speech.

> **RELEASE**

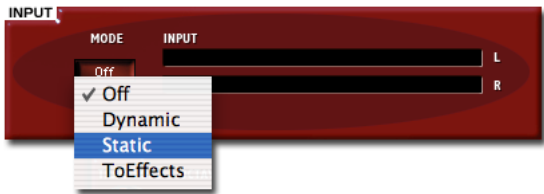
Determines the release time of the vocoder's envelope followers. With longer release times, the definition of the signal peaks becomes blurred, reducing intelligibility.

> **SOURCE BAL**

Determines the balance between the source signals and the vocoder signal. At +0 (default) only the output of the vocoder is heard. Negative values add a certain amount of the carrier signal to the vocoder signal, whereas positive values add a certain amount of the modulator signal. At either extreme, you will hear only the carrier or modulator signals respectively.

Using the Audio Inputs

The two audio input channels allow you to pass an external audio signal through the filters, amplifier and effects sections of the Virus PowerCore. A typical application of this technique is to run a drum-loop through a lowpass filter, but with the Virus PowerCore's powerful modulation capabilities, the possibilities for sonic manipulation go a lot further than this!



> USING THE SIDE-CHAIN PLUGIN

Not all VST host software will allow you to route audio data into a VST instrument, so for this reason we have included the Virus Input plugin which sets up a side-chain between the audio track and the Virus PowerCore plugin. Latency is compensated as soon as the relevant information is provided by the host software.



To use this, you first need to instance the Virus PowerCore as an audio instrument, so that the input plugin has something to connect to. Now, in the audio track you wish to process, select 'Virus Input' from the PowerCore folder in the VST menu.

Now click on the Destination window, and choose the Virus instance from the menu. The output of the audio track will now be routed into the Virus plugin.

NOTE: *If your host software provides side-chaining capabilities, or allows audio to be routed to a VSTi then you should use these methods instead, as the side-chain plugin may not behave correctly under such hosts. Note that you may have to instance the Virus PowerCore as a multi-channel instrument in order to access the side-chain functionality.*

> MODE

Enables the external audio inputs.

> **Off:** the audio inputs are disabled.

> **Dynamic:** In Dynamic mode, the audio signal is passed through the amp envelope as well as the filters (inc. saturation) and effects. This means that no signal is heard until you press a key, and the volume curve of the signal will follow the shape of the amp envelope.

- > **Static:** In Static mode, the audio signal is passed through the filters and effects sections, but bypasses the amp envelope. This means that the signal can be heard continuously, regardless of any keys played.
- > **To Effects:** Use this mode if you wish to bypass the filter section and send the input signal straight to the effects section of the Virus PowerCore. This mode does not reduce polyphony

NOTE: When in Dynamic or Static modes, the polyphony will be reduced by two voices.

NOTE: In both modes, the level of the external audio signal can be attenuated by the **Osc Vol** control (Global Page).

Envelope Follower

The envelope follower replaces the filter envelope with an envelope that is dynamically controlled by an external audio signal. In other words, the external audio signal is continuously monitored, and the peaks and troughs in the signal are translated into data which can be used to control the levels of other parameters. A typical application of this would be to use a drum-loop to modulate the filter cutoff frequency of a pad sound.



NOTE: See the chapter “Using the Audio Inputs” for instructions on how to route an audio signal into the Virus PowerCore.

> MODE

Enables the envelope follower. If you are using multiple instances of the Virus PowerCore on the same DSP, only one can use the envelope follower.

By default, the envelope follower modulates the cutoff frequency of filter 1 and filter 2, but because it replaces the filter envelope, you will need to adjust Env Amt (Filter page) of one or both filters to hear any effect. If you need the envelope follower to modulate another parameter, then use the Mod Matrix (see Mod Matrix page) and choose ‘Filter Envelope’ as the modulation source.

> **LEVEL**

Determines the sustain level of the envelope follower. This has a significant influence on the intensity of the effect, and it is recommended that you start with a value of 127 and reduce as necessary.

> **ATTACK**

Determines the envelope follower's response time to the incoming signal peaks. It is recommended that you start with a value of 0 and only increase Attack if you require a softer response.

> **RELEASE**

Determines the rate at which the envelope follower responds to a reduction in the input signal strength. Lower values result in a more dynamic, grittier sound, whereas higher values result in a smoother sound which still contains a rhythmic element.

Global

The following parameters are global i.e. they affect each part/channel of the current instance in the same way.

> **TUNE**

Determines the tuning of all parts within the current instance. Default is 0 (A=440Hz).

> **TEMPO**

Determines the tempo of the Virus PowerCore's internal clock, which is used to control the rate of all the clock-syncable parameters, such as the arpeggiator, delay and lfo's. If MIDI Clock data is being received from the host software, this controller will be greyed out and unavailable for adjustment.

> **SYNC**

Use this button to determine whether the Virus PowerCore synchronises its tempo to the clock signal generated by the host software, or whether it uses its own internal clock.



NOTE: The state of this button is saved with the project data by your host software, i.e. it is not saved within the patch.

> **Extern:** The Virus will automatically synchronise to the clock signal generated by the host application. In this mode, the Tempo control will be greyed out.

> **Intern:** The Virus ignores the clock signal generated by the host software and uses the tempo as set within the patch. Only the tempo setting of the patch in part 1 is used, as it is not possible to have multiple tempi within the same DSP.

Pitch

The following controls determine the overall pitch and tuning offset for the current patch.



> TRANSPOSE

Determines the pitch offset in semitone increments for the current patch. You can transpose a patch down to 64 semitones lower, or up to 63 semitones higher. All oscillators keep their relative intervals between each other.

> OCTAVE

Use the Octave buttons to transpose the pitch in octave increments, up to 4 octaves higher or lower.

> DETUNE

Determines the overall detuning of the patch. This is a subtle effect which is similar to Osc 2/3 Detune, but applied to all oscillators simultaneously. Like Osc2/3 Detune, this is affected by MIDI note number i.e. there is a subtle keyfollow applied to the intensity of the detuning.

NOTE: Use Detune to increase the overall warmth when playing more than one sound from different parts within the same instance of the Virus PowerCore.

Pitch Wheel

The pitch wheel (also known as a pitch bender) is a controller found on pretty much all keyboard controllers. It usually takes the form of a sprung wheel that is moved either up or down, but is sometimes in the form of a lever, which is moved to the left or right. The most common application of this controller is to affect subtle changes in pitch on some notes, in order to make a performance more expressive.



> CURVE

Determines the rate of pitch bend as you move towards either extreme. Choose the curve which best suits your playing style and the musical context of the patch.

- > **Linear:** This affects a consistent change in pitch as you move the controller towards either extreme.
- > **Exponential:** This affects a gradual change in pitch as you first move the controller, but as you move towards either extreme, the change becomes much more pronounced.

> BEND UP

Determines the degree of pitch bend at the upper extreme of the pitch controller.

The value is set in semitone intervals.

> BEND DOWN

Determines the degree of pitch bend at the lower extreme of the pitch controller.

The value is set in semitone intervals.

> LINK (BEND UP & BEND DOWN)

When enabled, the Bend Up and Bend Down values are linked so as to be symmetrical i.e. setting Bend Up to +3 will automatically set Bend Down to -3.

Patch Param



> CTRL SMOOTH

Many digital synths, including many 'soft' synths in particular, cannot affect parameter changes without producing audible artefacts or 'zipper noise'. This is because the resolution of the controllers is not high enough to affect smooth changes in realtime. The Virus PowerCore compensates for this however, by using Adaptive Parameter Smoothing or 'Control Smoothing'. This means that instead of jumping between each step at the resolution of the controller (128 steps maximum), the Virus PowerCore takes advantage of the higher internal resolution of the PowerCore's DSP to interpolate between the values. The result is smooth, discreet changes with no zipper noise. Some patches may require more instantaneous jumps than Ctrl Smooth can achieve though, so the following options are made available:

- > **Off:** No smoothing is applied to the current patch.
- > **On:** Smoothing is applied to all relevant parameters.
- > **Auto:** Smoothing is applied to parameters whose value is changed slowly, but sudden jumps are not smoothed.
- > **Note:** Smoothing is applied within a note's duration, but is disabled at Note Off i.e. parameter changes affected by MIDI note number or Note On will not be smoothed.

> HOLD PEDAL

Enables or disables the response to a sustain pedal, which keyboard players often use to hold notes even after the key is released.

> NOTE STEAL

If the Virus PowerCore is asked to play more notes than the available polyphony allows, then 'note-stealing' occurs. This means that all currently playing voices are analysed for their current state, and the Virus PowerCore intelligently silences one voice to make room for the new one. When more than one part within the same instance is being used, then the note-stealing algorithm has to decide which plug-in loses a voice in favour of another. This is where High and Low Priority come in:

- > **Low Priority :** When an instance is set to Low Priority, preference is deferred to any other instances which are set to High Priority.

- > **High Priority:** When set to High Priority, the Virus Powercore will try to ignore this patch and look to other, Low Priority patches when stealing notes.

Keyboard

The following parameters define how the Virus PowerCore will respond to a keyboard controller.



> LOW KEY

Determines the lowest MIDI note number to which the current patch will respond. Any notes played below this key will be ignored. There are two ways to set the Low Key, either:

- 1) Click the mouse pointer on the Low Key box and type the MIDI note number, or:
- 2) Click the Low Key button and press a key on either your MIDI keyboard controller or the on-screen keyboard.

> HIGH KEY

Determines the highest MIDI note number to which the current patch will respond. Any notes played above this key will be ignored. There are two ways to set the High Key, either:

- 1) Click the mouse pointer on the High Key box and type the MIDI note number, or:
- 2) Click the High Key button and press a key on either your MIDI keyboard controller or the on-screen keyboard.

> MODE

Determines whether a patch is polyphonic or monophonic, and the behaviour of Portamento.

- > **Poly:** The Virus PowerCore will play as many simultaneous voices (notes) as possible, up to a maximum of 16. Portamento is always active and envelopes are re-triggered with every new note.
- > **Mono 1:** Only one voice is allowed to be played at any time. Portamento is always active, and envelopes are re-triggered with each new note.

- > **Mono 2:** Only one voice is allowed to be played at any time. Portamento is only activated between notes that are played using a legato technique (legato technique = the first key is only released once the next is played). Envelopes are re-triggered with each new note.
- > **Mono 3:** Only one voice is allowed to be played at any time. Portamento is always active. Envelopes are only re-triggered when played in a non-legato manner (detached).
- > **Mono 4:** Only one voice is allowed to be played at any time. Portamento is only activated between notes that are played using a legato technique. Envelopes are only re-triggered when played in a non-legato manner.
- > **Hold:** The Virus PowerCore will play as many simultaneous voices as possible, up to a maximum of 16. Note Off is ignored, meaning all voices will continue to sound indefinitely, even after the keys are released. As soon as you press a key, any held voices will be replaced by the new voice(s).

> PORTAMENTO

Determines the rate at which the pitch of one note will glide towards the next. A value of 0 means that Portamento is off. The higher the value, the longer it takes for the pitch of the first note to glide towards the next.

Arpeggiator

The arpeggiator is a powerful creative tool that takes any currently played notes and plays them in a cyclic, rhythmic pattern. If more than one note is being played i.e. a chord, then the arpeggiator will break up the chord into its individual notes, and cycle through them with each consecutive step in the pattern. Think of the bit just before the chorus of 'Rio' by Duran Duran, and you'll understand ;-)



> MODE

This determines the order in which the arpeggiator will cycle through the notes of a chord.

- > **Off:** The arpeggiator is off.
- > **Up:** The arpeggiator will cycle through the notes of a chord in ascending order.

- > **Down:** The arpeggiator will cycle through the notes of a chord in descending order.
- > **Up & Down:** The arpeggiator will cycle through the notes of a chord in ascending then descending order.
- > **As Played:** The arpeggiator will cycle through the notes of a chord in the order in which they were played.
- > **Random:** The arpeggiator will play the notes of a chord in random order.
- > **Chord:** The arpeggiator will play all the notes of a chord repeatedly in the chosen rhythm pattern.

> **PATTERN**

Select from one of 40 different rhythmic patterns for the arpeggiator.

***NOTE:** Please note that a few of the patterns may sound fairly similar at first. This is because they differ in rhythmic accentuation as opposed to timing. To make the difference more apparent, try adjusting some of the Velocity Mod parameters (see Osc Page).*

> **NOTE LENGTH**

Determines an offset in the length of each note in the current arpeggiator pattern. Positive values lengthen each note in the pattern, whilst negative values shorten each note.

> **SWING**

Similar to the quantization presets available in many of today's sequencers, this delays the up-beats by up to 75%. This parameter only affects patterns 2 – 64. Select pattern 2, and set Swing to Off. When you hold down a key, you should hear a continuous 1/8th note rhythm. As you increase Swing towards 75%, you will notice every second 1/8th note move towards the next down beat.

> **OCT RANGE**

Determines the transposition range of the arpeggiator in octaves. With Oct Range set to 1, the arpeggiator only cycles through the currently held notes. When set to 2, the arpeggiator cycles through the held notes and the equivalent notes one octave higher. The maximum range is 4 octaves.

> **CLOCK**

Determines the rate of arpeggiation relative to the MIDI beat clock. Choose a value between 1/64 and 1/1. Patterns 2 – 64 are designed to be used with a clock value of 1/8, but can of course be used with any of the available values.

> **HOLD**

When enabled, the arpeggiator will continue to arpeggiate the most recently played notes even after the keys have been released.

The ModMatrix Page



Here is where you determine which parameters are to be modulated by which control sources. Up to 6 parameters can be modulated by up to 3 control sources, allowing for some very complex movement within a sound. Practically any parameter in the Virus Powercore can be modulated by any one of 27 control sources! The most common application of the ModMatrix is to assign the modwheel to add vibrato, or to open the filters. A little imagination can go a long way though – see Howard Scarr’s excellent tutorial “Programming Analog Synths” for some ideas in this area.

Assign 1

> SOURCE

Select the control source for Mod Slot 1.

> AMOUNT

Determines the degree of modulation.

> DESTINATION

Select the parameter to be modulated by Assign Source 1.



Assign 2

> SOURCE

Select the control source for Mod Slot 2.

> AMOUNT 1

Determines the degree of modulation for Destination 1.

> DESTINATION 1

Select the first parameter to be modulated by Assign Source 2.

> AMOUNT 2

Determines the degree of modulation for Destination 2.

> DESTINATION 2

Select the second parameter to be modulated by Assign Source 2.



Assign 3

> SOURCE

Select the control source for Mod Slot 3.

> AMOUNT 1

Determines the degree of modulation for Destination 1.

> DESTINATION 1

Select the first parameter to be modulated by Assign Source 3.

> AMOUNT 2

Determines the degree of modulation for Destination 2.

> DESTINATION 2

Select the second parameter to be modulated by Assign Source 3.

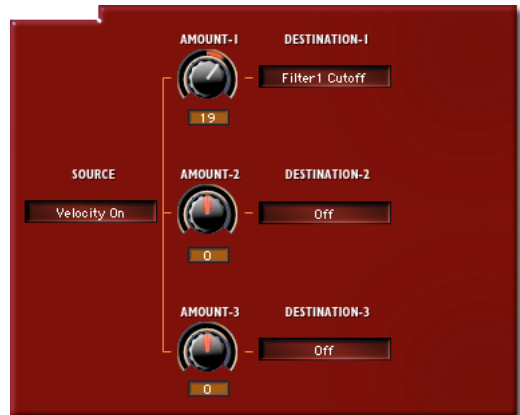
> AMOUNT 3

Determines the degree of modulation for Destination 3.

> DESTINATION 3

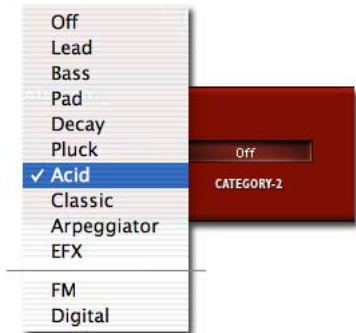
Select the third parameter to be modulated by Assign Source 3.

NOTE: For a full list of available modulation sources and destinations, please refer to the Appendix.



Categories

Here you can select up to two categories with which to label your patch. If you choose to use two different categories, then the patch will appear in both lists in the categories option in the Load menu. Click on the relevant pane to select the appropriate category for your patch.



Customer support

For PowerCore and installation related questions, please contact

<http://www.tcelectronic.com/>

For Virus PowerCore synthesis related questions, please contact

<http://www.access-music.de/?go=support>

Additional patches can be downloaded here:

<http://www.access-music.de>

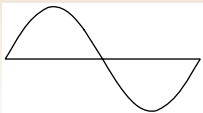
The latest PowerCore driver version and TCAU TC Audio Unit wrapper can be downloaded here:

<http://www.tcelectronics.com>

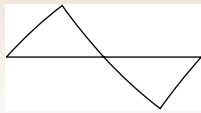
Appendix

Oscillator and LFO waveforms

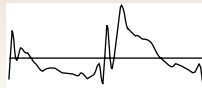
Waveform Sine



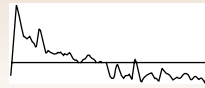
Waveform Triangle



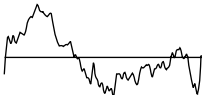
Waveform 3



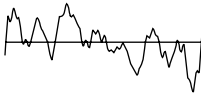
Waveform 4



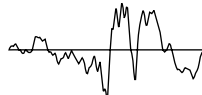
Waveform 5



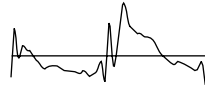
Waveform 6



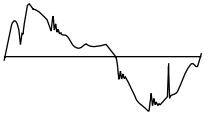
Waveform 7



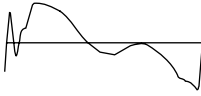
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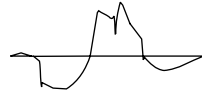
Waveform 9



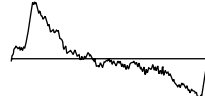
Waveform 10



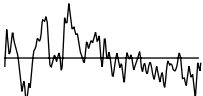
Waveform 11



Waveform 12



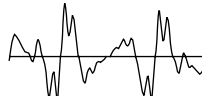
Waveform 13



Waveform 14



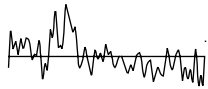
Waveform 15



Waveform 16



Waveform 17



Waveform 18



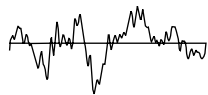
Waveform 19



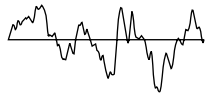
Waveform 20



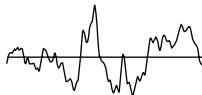
Waveform 21



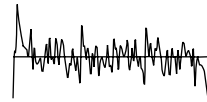
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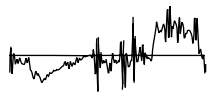
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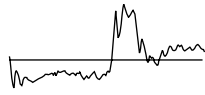
Waveform 24



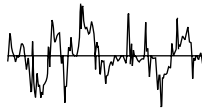
Waveform 25



Waveform 26



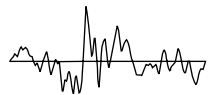
Waveform 27



Waveform 28



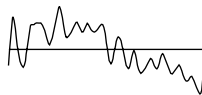
Waveform 29



Waveform 30



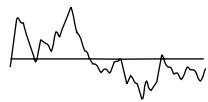
Waveform 31



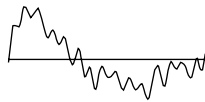
Waveform 32



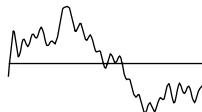
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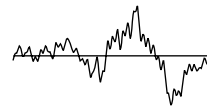
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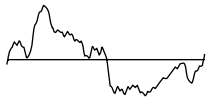
Waveform 35



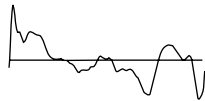
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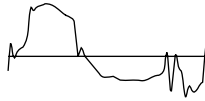
Waveform 37



Waveform 38



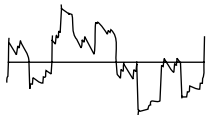
Waveform 39



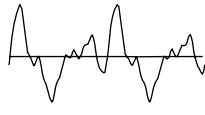
Waveform 40



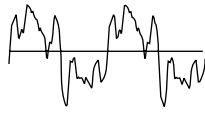
Waveform 41



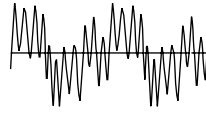
Waveform 42



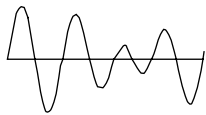
Waveform 43



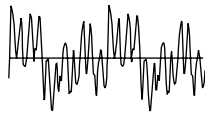
Waveform 44



Waveform 45



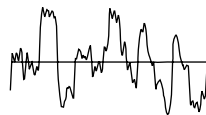
Waveform 46



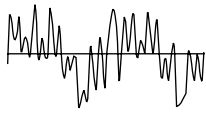
Waveform 47



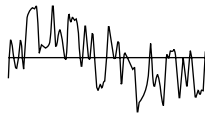
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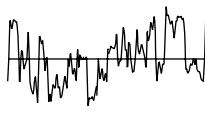
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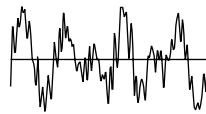
Waveform 50



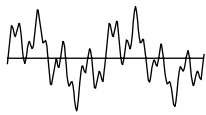
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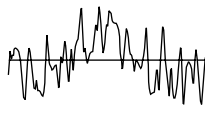
Waveform 52



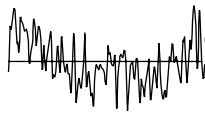
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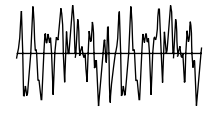
Waveform 54



Waveform 55



Waveform 56



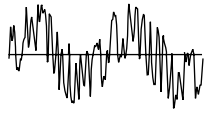
Waveform 57



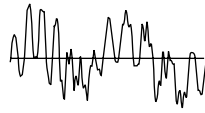
Waveform 58



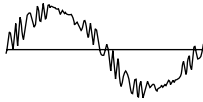
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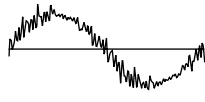
Waveform 60



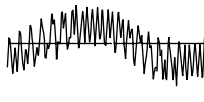
Waveform 61



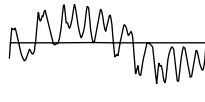
Waveform 62



Waveform 63



Waveform 64



Appendix

System Exclusive Data

The Parameters of the VIRUS are organized in three so-called pages A, B and C. Each page contains 128 parameters, addressed by numbers from 0 to 127. Each parameter is represented by one byte with a maximum value of 127. All parameters are individually accessible by SysEx-Parameterchange.

The pages A and B represent a Single-Program, thus a Single-Program contains 256 Bytes.

Page A (LowPage) contains Single parameters that are usefull for MIDI automation. The parameters of this page are additionally controllable by MIDI Control Change.

The remaining Single parameters in Page B (HiPage) are additionally controllable by MIDI Polyphonic Pressure (!).

Page C contains Multi parameters and Global parameters. These parameters are sent and received only by SysEx Parameter Change.

In the following, all bytes are shown in hexadecimal representation.

CONTROL CHANGE MESSAGE (ONLY PAGE A)

Byte	Meaning	Remark
Bc	Status byte	c=MIDI channel
nn	Parameter Number 0..127	see parameter list Page A)
vv	Parameter Value 0..127	see parameter list Page A)

The Control Change messages are defined as Performance Controller (e.g. Modulation Wheel or Hold Pedal) or Sound Parameters (e.g. Cutoff or Patch Volume). The Performance Controllers are not stored with a Single-Sound. If more than one Multi Part is set to the same MIDI channel, all Parts on this MIDI channel receive the same Performance Controllers. The Sound Parameters are stored with a Single Sound. If more than one Multi Part is set to the same MIDI channel, the Sound Parameter is received only by the Multi Part with the lowest part number.

Example: B0,21,40 Set oscillator balance (21 hex = 33 dec) on MIDI channel 1 to the middle position (40 hex =64 dec).

POLYPHONIC PRESSURE MESSAGE (ONLY PAGE B)

Byte	Meaning	Remark
Ac	Status byte	c=MIDI channel
nn	Parameter Number 0..127	see parameter list Page B
vv	Parameter Value 0..127	see parameter list Page B

Example: A2,07,25 Control LFO3 Rate on MIDI channel 3 (!).

Parameters Description

No.	Class	Name	Range	Value	Text
PAGE A					
A 1	p	Modulation Wheel			
A 2	p	Breath Controller			
A 3	p	Contr 3			
A 4	p	Foot Controller			
A 5	a	Portamento Time	0..127		
A 6	p	Data Slider			
A 7	p	Channel Volume	0..127		
A 8	p	Balance			
A 9	p	Contr 9			
A 10	a	Panorama	0..127	-64..0..+63:	Left..Center..Right
A 11	p	Expression	0..127		
A 12	p	Contr 12			
A 13	p	Contr 13			
A 14	p	Contr 14			
A 15	p	Contr 15			
A 16	p	Contr 16			
A 17	a	Osc1 Shape	0..127	-64..0..+63:	Wave..Saw..Pulse
A 18	a	Osc1 Pulsewidth	0..127		
A 19	a	Osc1 Wave Select	0..64		Sine, Triangle, Wave 3..64
A 20	a	Osc1 Semitone	0..127	-64..+63	
A 21	a	Osc1 Keyfollow	0..127	-64..+63,	Default: 32
A 22	a	Osc2 Shape	0..127	-64..0..+63:	Wave..Saw..Pulse
A 23	a	Osc2 Pulsewidth	0..127		
A 24	a	Osc2 Wave Select	0..64		Sine, Triangle, Wave 3..64
A 25	a	Osc2 Semitone	0..127	-64..+63	
A 26	a	Osc2 Detune	0..127		
A 27	a	Osc2 FM Amount	0..127		
A 28	a	Osc2 Sync	0..1		0:Off 1:On
A 29	a	Osc2 Filt Env Amt	0..127	-64..+63	
A 30	a	FM Filt Env Amt	0..127	-64..+63	

No.	Class	Name	Range	Value	Text
A 31	a	Osc2 Keyfollow	0..127	-64..+63:	Default: 32
A 32	p	Bank Select	0..3		Bank A..D
A 33	a	Osc Balance	0..127		-64..+63:
A 34	a	Suboscillator Volume	0..127		
A 35	a	Suboscillator Shape	0..1		0:Square 1:Triangle
A 36	a	Osc Mainvolume	0..127		
A 37	a	Noise Volume	0..127		
A 38	a	Ringmodulator Volume	0..127		
A 39	a,Vb	Noise Color	0..127	-64..0..+63	
A 40	a	Cutoff	0..127		
A 41	a	Cutoff2	0..127	-64..+63	
A 42	a	Filter1 Resonance	0..127		
A 43	a	Filter2 Resonance	0..127		
A 44	a	Filter1 Env Amt	0..127		
A 45	a	Filter2 Env Amt	0..127		
A 46	a	Filter1 Keyfollow	0..127	-64..+63	
A 47	a	Filter2 Keyfollow	0..127	-64..+63	
A 48	a	Filter Balance	0..127	-64..+63	
A 49	a	Saturation Curve	0..6		0:Off 1:Light 2:Soft 3:Middle 4:Hard 5:Digital ..
A 51	a	Filter1 Mode	0..3		0:LP 1:HP 2:BP 3:BS
A 52	a	Filter2 Mode	0..3		0:LP 1:HP 2:BP 3:BS
A 53	a	Filter Routing	0..3		0:Ser4 1:Ser6 2:Par4 3:Split
A 54	a	Filter Env Attack	0..127		
A 55	a	Filter Env Decay	0..127		
A 56	a	Filter Env Sustain	0..127		
A 57	a	Filter Env Sustain Time	0..127	-64..+63:	Fall..Infinite..Rise
A 58	a	Filter Env Release	0..127		
A 59	a	Amp Env Attack	0..127		
A 60	a	Amp Env Decay	0..127		
A 61	a	Amp Env Sustain	0..127		
A 62	a	Amp Env Sustain Time	0..127	-64..+63:	Fall..Infinite..Rise
A 63	a	Amp Env Release	0..127		

No.	Class	Name	Range	Value	Text
A 64	p	Hold Pedal			
A 65	p	Portamento Pedal			
A 66	p	Sostenuto Pedal			
A 67	a	Lfo1 Rate	0..127		
A 68	a	Lfo1 Shape	0..5		0:Sine 1:Tri 2:Saw 3:Square 4:S&H 5:S&G ..
A 69	a	Lfo1 Env Mode	0..1		0:Off 1:On
A 70	a	Lfo1 Mode	0..1		0:Poly 1:Mono
A 71	a	Lfo1 Symmetry	0..127	-64..+63	
A 72	a	Lfo1 Keyfollow	0..127		
A 73	a	Lfo1 Keytrigger	0..127		0:Off, 1..127:Keytrigger Phase
A 74	a	Osc1 Lfo1 Amount	0..127	-64..+63	
A 75	a	Osc2 Lfo1 Amount	0..127	-64..+63	
A 76	a	PW Lfo1 Amount	0..127	-64..+63	
A 77	a	Reso Lfo1 Amount	0..127	-64..+63	
A 78	a	FiltGain Lfo1 Amount	0..127	-64..+63	
A 79	a	Lfo2 Rate	0..127		
A 80	a	Lfo2 Shape	0..5		0:Sine 1:Tri 2:Saw 3:Square 4:S&H 5:S&G ..
A 81	a	Lfo2 Env Mode	0..1		0:Off 1:On
A 82	a	Lfo2 Mode	0..1		0:Poly 1:Mono
A 83	a	Lfo2 Symmetry	0..127	-64..+63	
A 84	a	Lfo2 Keyfollow	0..127		
A 85	a	Lfo2 Keytrigger	0..127		0:Off, 1..127:Keytrigger Phase
A 86	a	OscShape Lfo2 Amount	0..127	-64..+63	
A 87	a	FmAmount Lfo2 Amount	0..127	-64..+63	
A 88	a	Cutoff1 Lfo2 Amount	0..127	-64..+63	
A 89	a	Cutoff2 Lfo2 Amount	0..127	-64..+63	
A 90	a	Panorama Lfo2 Amount	0..127	-64..+63	
A 91	a	Patch Volume	0..127		
A 93	a	Transpose	0..127	-64..+63	
A 94	a	Key Mode	0..4		0:Poly 1..4: Mono1-4
A 97	a	Unison Mode	0..15		0:Off 1:Twin 2..15
A 98	a	Unison Detune	0..127		

No.	Class	Name	Range	Value	Text
A 99	a	Unison Panorama Spread	0..127		
A100	a	Unison Lfo Phase	0..127	-64..+63	
A101	a	Input Mode	0..2		0:Off 1:Dynamic 2:Static 3:ToEffects
A102	a	Input Select	0..8		0:In1L 1:In1L+R 2:In1R ..
A105	a	Chorus Mix	0..127		
A106	a	Chorus Rate	0..127		
A107	a	Chorus Depth	0..127		
A108	a	Chorus Delay	0..127		
A109	a	Chorus Feedback	0..127	-64..+63	
A110	a	Chorus Lfo Shape	0..5		0:Sine 1:Tri 2:Saw 3:Square 4:S&H 5:S&G ..
A112	a	Delay/Reverb Mode	0..1		0:Off 1:Delay 2:Reverb 3:Rev+Feedb1
A113	a,ms	Effect Send	0..127		
A114	a,ms,np	Delay Time	0..127		
A115	a,ms,np	Delay Feedback	0..127		
A116	a,ms,np	Delay Rate	0..127		
		Reverb Decay Time	0..127		
A117	a,ms,np	Delay Depth	0..127		
		Reverb Room Size	0..3		0:Ambience 1:SmallRoom 2:LargeRoom 3:Hall
A118	a,ms,np	Delay Lfo Shape	0..5		0:Sine 1:Tri 2:Saw 3:Square 4:S&H 5:S&G ..
		Reverb Damping	0..127		
A119	a,ms,np	Delay Color	0..127	-64..+63	
A123	p	All Notes Off			

No.	Class	Name	Range	Value	Text
PAGE B					
B 1	b	Arp Mode	0..6		0:Off 1:Up 2:Down 3:Up&Down 4:AsPlayed 5:Random 6:Chord
B 2	b	Arp Pattern Select	0..64		
B 3	b	Arp Octave Range	0..3		
B 4	b	Arp Hold Enable	0..1		0:Off 1:On
B 5	b	Arp Note Length	0..127	-64..+63c	
B 6	b	Arp Swing	0..127	50%..75%	
B 7	b	Lfo3 Rate	0..127		
B 8	b	Lfo3 Shape	0..5		0:Sine 1:Tri 2:Saw 3:Square 4:S&H 5:S&G ..
B 9	b	Lfo3 Mode	0..1		0:Poly 1:Mono
B 10	b	Lfo3 Keyfollow	0..127		
B 11	b	Lfo3 Destination	0..5		0:Osc1 1:Osc1+2 2:Osc2 3:PW1 4:PW1+2 5:PW2
B 12	b	Osc Lfo3 Amount	0..127		
B 13	b	Lfo3 Fade-In Time	0..127		
B 16	b	Clock Tempo	0..127	63..190 BPM	
B 17	b	Arp Clock	1..17	1/64..1/1	
B 18	b	Lfo1 Clock	0..19		Off, 1/64..4/1
B 19	b	Lfo2 Clock	0..19		Off, 1/64..4/1
B 20	b,ms,np	Delay Clock	0..16		Off, 1/64..3/4
B 21	b	Lfo3 Clock	0..19		Off, 1/64..4/1
B 25	b	Control Smooth Mode	0..3		0:Off, 1:On, 2:Auto, 3:Note
B 26	b	Bender Range Up	0..127	-64..+63	
B 27	b	Bender Range Down	0..127	-64..+63	
B 28	b	Bender Scale	0..1		0:Linear 1:Exponential
B 30	b	Filter1 Env Polarity	0..1		0:Negative 1:Positive
B 31	b	Filter2 Env Polarity	0..1		0:Negative 1:Positive
B 32	b	Filter2 Cutoff Link	0..1		0:Off 1:On
B 33	b	Filter Keytrack Base	0..127		C-1..G9
B 34	b,Vb	Osc FM Mode	0..12		0:Pos-Tri 1:Tri 2:Wave 3:Noise 4:Ln L 5:Ln L+R ..

No.	Class	Name	Range	Value	Text
B 35	b	Osc Init Phase	0..127		0:Off 1..127
B 36	b	Punch Intensity	0..127		
B 38	b,Vb	Input Follower Mode	0..9		0:Off 1:In L 2:In L+R ...
B 39	b	Vocoder Mode	0..12		0:Off 1:OSC 2:OscHold 3:Noise 4:In L 5:In L+R ..
B 41	b,Vb	Osc3 Mode	0..67		0:Off 1:Osc2Slave 2:Saw 3:Pulse 4:Sine 5 Triangle ..
B 42	b,Vb	Osc3 Volume	0..127		
B 43	b,Vb	Osc3 Semitone	0..127	-64..+63	
B 44	b,Vb	Osc3 Detune	0..127		
B 47	b	Osc1 Shape Velocity	0..127	-64..+63	
B 48	b	Osc2 Shape Velocity	0..127	-64..+63	
B 49	b	PulseWidth Velocity	0..127	-64..+63	
B 50	b	Fm Amount Velocity	0..127	-64..+63	
B 54	b	Filter1 EnvAmt Velocity	0..127	-64..+63	
B 55	b	Filter1 EnvAmt Velocity	0..127	-64..+63	
B 56	b	Resonance1 Velocity	0..127	-64..+63	
B 57	b	Resonance2 Velocity	0..127	-64..+63	
B 60	b	Amp Velocity	0..127	-64..+63	
B 61	b	Panorama Velocity	0..127	-64..+63	
B 62	b	Soft Knob1 Single			see Soft Knob List
B 63	b	Soft Knob2 Single			see Soft Knob List
B 64	b	Assign1 Source			see Assign Sources List
B 65	b	Assign1 Destination			see Assign Destinations List
B 66	b	Assign1 Amount	0..127	-64..+63	
B 67	b	Assign2 Source			see Assign Sources List
B 68	b	Assign2 Destination1			see Assign Destinations List
B 69	b	Assign2 Amount1	0..127	-64..+63	
B 70	b	Assign2 Destination2			see Assign Destinations List
B 71	b	Assign2 Amount2	0..127	-64..+63	
B 72	b	Assign3 Source			see Assign Sources List
B 73	b	Assign3 Destination1			see Assign Destinations List
B 74	b	Assign3 Amount1	0..127	-64..+63	
B 75	b	Assign3 Destination2			see Assign Destinations List

No.	Class	Name	Range	Value	Text
B 76	b	Assign3 Amount2	0..127	-64..+63	
B 77	b	Assign3 Destination3			see Assign Destinations List
B 78	b	Assign3 Amount3	0..127	-64..+63	
B 79	b	LFO1 Assign Dest			see Assign Destinations List
B 80	b	LFO1 Assign Amount	0..127	-64..+63	
B 81	b	LFO2 Assign Dest			see Assign Destinations List
B 82	b	LFO2 Assign Amount	0..127	-64..+63	
B 84	b,Vb	Phaser Mode	0..6		0:Off, 1..6 Phaser Stages
B 85	b,Vb	Phaser Mix	0..127		
B 86	b,Vb	Phaser Rate	0..127		
B 87	b,Vb	Phaser Depth	0..127		
B 88	b,Vb	Phaser Frequency	0..127		
B 89	b,Vb	Phaser Feedback	0..127	-64..+63	
B 90	b,Vb	Phaser Spread	0..127		
B 97	b,Vb	Analog Boost Intensity	0..127		
B 98	b,Vb	Analog Boost Tune	0..127		
B 99	b,Vb	Input Ringmodulator	0..127		0:Off 1..127: Direct..Ringmodulator..Input
B100	b,Vb	Distortion Curve	0..6		0:Off 1:Light 2:Soft 3:Middle 4:Hard 5:Digital ..
B101	b,Vb	Distortion Intensity	0..127		
B123	b,Vb				Category1
B124	b,Vb				Category2

All bytes are shown in decimal representation.

Classes

P: PERFORMANCE CONTROLLER

Accessible by Control message. Performance Controllers are not stored with a Single-Sound. If more than one Multi Part is set to the same MIDI channel, all Parts on this MIDI channel receive the same Performance Controllers.

A: SOUND PARAMETER OF BANK A

Accessible by Control message, SysEx-Parameterchange and Single-Dump. The Sound Parameters are stored with a Single Sound. When received as Control Message, the Sound Parameter is received only by the Multi Part with the lowest part number, if more than one Multi Part is set to the same MIDI channel. When received as SysEx-Parameterchange or Single-Dump, the part is addressed by the part number irrespective of the actual MIDI channel setting.

B: SOUND PARAMETER OF BANK B

Accessible by MIDI Polyphonic Pressure, SysEx-Parameterchange and Single-Dump. The Sound Parameters are stored with a Single Sound. When received as Polyphonic Pressure, the Sound Parameter is received only by the Multi Part with the lowest part number, if more than one Multi Part is set to the same MIDI channel. When received as SysEx-Parameterchange or Single-Dump, the part is addressed by the part number irrespective of the actual MIDI channel setting.

MS: MULTI/SINGLE PARAMETER

When in Single Mode, the parameter is received and stored with the Single Sound. When in Multi Mode, the parameter is received and stored with the Multi Patch. In Multi Mode the Single Sound settings are ignored while the corresponding Multi Patch settings are active.

NP: NON-PART-SENSITIVE SOUND PARAMETER

When in Multi Mode, the parameter affects all Multi Parts.

BPC: BANK/PROGRAM-CHANGE PARAMETER BANK SELECT

selects the Single bank accessed by a subsequent Program Change, similar to the regular Bank Select. Bank Change directly changes the Single program to the requested bank, without changing the program number. Program Change directly changes the Single program to the requested program number, without changing the bank number; similar to the regular Program Change. Part number \$40 will address the Single buffer in Single Mode.

On non-part-sensitive parameters the part number is ignored, but must still be sent as any value.

Remarks for editor/librarian programs Not all 256 bytes of a Single or Multi Dump are defined as a parameter. Some of them are defined for internal use or reserved for future applications. In a bulk dump these byte should not be changed, they should be sent to the Virus on the same value as they were received in the dump.

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- Resonance
- Chorus Mix
- Phaser Mix
- Send
- Osc Volume
- Sub Volume
- Ring Modulator
- Amp Attack
- Amp Decay

Osc 1 38

- Shape
- Wave Select
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- Semitone
- Key follow

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- Shape
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- PW
- Semitone
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- Semitone*
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- Sub Shape
- Noise Color
 - Middle (0)
 - Negative (-64)
 - Positive (+63)

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- Osc 1
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- Mode
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 - Band Pass
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- Reso Vel
- Env Vel
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- Link (Envelope Amount, Resonance, Key Follow)
- Key Base

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- Mode
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 - High Pass
 - Band Pass
 - Band Stop
- Cutoff
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- Env AMT
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- Reso Vel
- Env Vel
- Env Polarity

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- Decay
- Sustain
- Sustain Time
- Release

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- Attack
- Decay
- Sustain
- Sustain Time
- Release

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 - Triangle
 - Sawtooth
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 - Sample and Glide
 - Oscillator Waveform
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- Key Follow
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- LFO Mode
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- Shape
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 - Triangle
 - Sawtooth
 - Square
 - Sample and Hold (Random)
 - Sample and Glide
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- Shape
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- Polyphonic Pressure message (only Page B)

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- b: Sound Parameter of Bank B
- ms: Multi/Single Parameter
- np: Non-part-sensitive Sound Parameter
- bpc: Bank/Program-Change Parameter
Bank Select

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