VIRUS C SERIES OS 6.5
ADDENDUM IN ENGLISH

access

VIRUS C SERIES OS6.5 ADDENDUM

Content

New features OS6.5

THE ANALOG FILTER ALGORITHMUS

With the Virus analogue filter you have a choice of 1, 2, 3 or 4 poles - impossible in its analogue counterpart! The number of poles can be selected via the parameter "Filter1 Mode", which you will find in the filter edit menu (press the EDIT button). Filter 2 is not affected – it always remains the original 2-pole Virus-filter.

```
 1  FILTER  1
Mode Analog 2P
```

When the analogue filter is activated, the usual Saturation curves are disabled because, like its true analogue forebears, the Virus analogue filter is already capable of saturation. The level of this saturation is controlled via the upper half of OSC VOL – also impossible in its analogue counterpart.

Analogue filter saturation is not simple clipping before or after the filter, as is sometimes assumed. On the contrary, we have made an implied non-linearity in the entire virtual filter circuit controllable. The fewer poles you select, the more the resulting characteristic overtones are allowed to pass through.

The analogue filter is perfectly integrated into the existing Virus filter model and can be operated seamlessly together with the second filter. The classic ladder filter had/has 4 poles and therefore a Q-factor of 24 dB/Oct. There are two ways to achieve this in the Virus. Try either of the following:

METHOD 1 Starting with the -Init- program (C126), press filter EDIT and switch Filter 1 Mode over to "Analog 2P". Filter 2 is in series, contributing the other two poles. With Cutoff 2 in the centre and Cutoff Link switched on, filter 2 follows filter 1 – they appear as a single 4-pole ladder filter.

METHOD 2 Starting with the -Init- program (C126) again, take filter 2 out of the equation, either by turning Cutoff2 all the way up or Filter Balance all the way down. Now press the EDIT button and select the Filter1 Mode "Analog 4P".

While the analogue filter is active, there is no difference between Serial 6 and Serial 4 modes.

SOUNDDIVER IMPLEMENTATION

The latest version of SoundDiver Virus, which can be downloaded from the Access webiste, supports the new Analog Filter algorithm. After choosing one of the new filter algorithms SoundDiver will offer you the option to initialise corresponding filter parameters. In case you plan to check the new filter with an old sound we recommend you to not choose this option

since it will alter the filter settings. In case you are working on a new sound initialisation is a good starting point for your own filter settings.

ABOUT POWER CONSUMPTION

Due to it's precise simulation of the original hardware it was modelled after, the Analog Filter Algorithmus needs more computing power than the conventional Virus filter. This can result in a lower voice count than the 32 voices. As a thumb rule each voice with Analog Filter activated will count as two voices which is similar to the Oscillator Twin Mode. Voices on other channels which make use of the conventional Virus filter don't need more calculating power.

TECHNICAL INFORMATION AND HISTORY

The new ladder filter in the Virus is modelled after the legendary Moog filter.

Real analogue circuits often have a life of their own, especially when operated close to or beyond their natural limits. The phenomena caused by interaction between a large number of electronic components can become highly complex and uncontrollable.

Digital technology is usually 100% controllable, and this has several advantages. For instance, an audio algorithm can be developed from scratch without having to worry about the peculiarities and limitations of analogue electronic components.

On the other hand, character and "life" is very difficult to achieve in the digital domain. The real challenge lies in transforming an analogue circuit into a truly accurate digital model – including all the physical effects that can affect the sound. However, once a good model has been developed, the advantages of digital technology comes into its own. You can implement features that would be much too expensive or unreliable if realized using analogue circuitry!

The character of an analogue filter is especially noticeable at high resonance and self-oscillation:

· Self-oscillation intermodulates with the oscillator signal at the filter input. The tonal quality of this intermodulation depends upon the amount of resonance. If two or more oscillators are slightly detuned against each other, the intermodulation gives the resonance a pleasantly rough but rich quality. Slight self-oscillation appears unstable and full of "life".

· Even while the filter is self-oscillating, the resonance control still has an effect on the sound. It ranges from a delicate sinusoidal whistle to a stable waveform rich in overtones.

· Self-oscillation is available over the entire cut-off range - it is not reduced at higher frequencies.

· Self-oscillation even continues when there is no signal at the filter input. It starts quite smoothly, rising and falling according to the amount of resonance.

· The self-oscillation signal is always at a musically sensible level compared with the oscillator signal.

· The usual drop in total volume at higher resonance is moderate - the filter does not lose power.

New features OS6

PURE TUNING

For centuries we have been playing musical instruments that are out of tune...

Ever since the emergence of polyphonic instruments, including all keyboards as well as fretted instruments such as the lute and guitar, tuning has always been a compromise. Several attempts from the 16th-18th centuries to standardise the temperament (tuning) of church organs and virginals helped a little, but they were battling against the mighty laws of physics – see the "Theory" section below.

First suggested in 1636, our modern "Equal Temperament" was only in common use from the late 18th century onwards because it was considered much too much of a compromise at the time. Despite it's one advantage (the freedom to play in any key), Equal Temperament simply dilutes the fundamental problem, spreading it across all the notes in the octave.

Experienced singers and string players use "just" intonation – they adapt to any keys and modulations (key changes) because they have infinitely variable control over pitch. Within certain limits, the pitch of wind instruments can also be varied by adjusting embouchure (lip position/tension). A group of musicians instinctively approaches a common overtone structure, minimizing the "friction" between all the voices in a chord. This results in the wonderfully rich but compact sound of symphony orchestras or gospel choirs.

Unfortunately, realtime intonation was not a feasible proposition for makers of keyboard instrument. Finding a usable method of performing fine adjustments to each and every note seemed physically impossible, especially when playing polyphonically.

These days, digital musical instruments can automate this process. The Pure Tuning (aka. Hermode) algorithm analyses chords and immediately adjusts the pitch of each note so that the prominent harmonics line up. Especially for normal synthesizer sounds, the difference between Equal Temperament and Pure Tuning may appear to be rather subtle at first (though this difference can be accentuated – see below):

Select the factory preset C126 (-Init-) on your Virus. Turn the Detune value down to 7 to make the phasing between the two oscillators nice and slow. Play a few simple major chords, or just a C-major triad if you prefer, and vary the PureTuning Intensity parameter.

At 127, PureTuning is fully on: The chord is well intonated (like a choir) and does not exhibit any beating between notes. This may seem a little unusual at first, but it sounds perfectly "straight" and correct after you have familiarized yourself with the effect. Setting PureTuning back to zero ("Equal") switches it off again: The chord is beating and, after comparing the two extremes, this original setting sounds equally unusual. The chord now appears to be tuned rather oddly, not quite pure enough, slightly spoiled compared with the maximum setting.

Astonishingly, this is the very same Equal Temperament we have been hearing all our lives – the big compromise!

Let's accentuate the effect now: Change the Distortion Intensity parameter (the one in the Effects section, not Filter saturation) to "Hard". Distortion in the Effects section is applied to the chord as a whole, blending the sound in the same way a guitar amplifier would. Medium range major chords now sound nice and grungy, but when played higher up the keyboard, this grunge becomes rather annoying. Turning up PureTuning causes this unpleasant grunge to disappear entirely because all the notes are perfectly in tune i.e. they have a simple frequency ratio.

The root cause of the unpleasant grunge is easy to find: Turn Pure-Scale down to zero and play middle C together with its fifth (G): Sounds clean enough. This is a typical "power chord" as played by guitarrists with their ampflifiers at maximum distortion settings. Now add the E to form a major C chord: There's that dirt again. The major third (E) is obviously way out of tune. Now turn PureTuning slowly up to maximum; you should be able to hear the major third being continuously adjusted downwards until it reaches purity.

When playing very high chords with distortion you can hear the lowest common denominator of chords: A subharmonic – a low note which is not actually being played. It is the same note as the root of the chord, but two octaves lower.

If the sound appears too sterile at maximum PureTuning, this is likely to be because there is always minimum beating between notes. Turn PureTuning down to the central position: "Natural". Although tuning is also nominally 100% pure at this setting, the actual pitch of notes is once again subject to the fluctuations, the natural anomalies responsible for a lot of the "life" in acoustic instruments (as well as true analogue synthesizers).

Even if you only play octaves, the "Natural" setting often causes beating between the notes! This effect is therefore independant of the chord structure, and can/should be accentuated via oscillator Detune (or even Unison mode) to beef up the sound. That's how the monumental sound of a symphony orchestra arises: the richness of the sound is solely dependant upon the number of musicians (oscillators), not upon harmonic complexity.

PureTuning works perfectly for major triads and dominant seventh chords. Due to the physics involved (see below), minor chords are more of a problem – they don't sound quite as pure. However, PureTuning is also very effective here because the subharmonics are managed well.

Equal-temperament was a radical break from all the other "tempered" methods, where polyphonic instruments could only be played in a few diatonic modes. Note that Johann Sebastian Bach's "The Well-Tempered Clavier" is thought to have been based on a variation of "Werckmeister 3" tuning. Much more suitable for playing in any key, but not quite the same as Equal Temperament!

PureTuning combines classical intonation with the universality of Equal Temperament.

Let Pure Tuning inspire you!

THEORY

Interestingly, the feeling of harmony in a major chord is a phenomenon based upon physics, not on psycho-acoustics alone. The notes in a perfectly tuned C-major triad (C, E, G) have exact integer frequency ratios: From E to C is 5:4 (or 1:1.25) and from G to C is 3:2 (1: 1.5). The

same applies to any major chord. It is fairly common knowledge that the frequency ratio of an octave is 2:1 – the upper note is exactly double the frequency of the lower one. Assuming a C is exactly 1000 Hz, the E would be 1250 Hz and the G would be 1500 Hz. The next C would be 2000 Hz. Very simple, very nice...

Reality isn't quite that simple. We expect more from our (chromatic) keyboard instruments. We demand that the frequency ratio of all semitone steps should be the same, that no particular key has "priority" over any other. Chords in the key of B or C# should sound just as good as they do in C. The truth is that they actually sound just as bad!

Rule Number One: An octave interval should be an octave, it should be exactly double/half the frequency of the reference note. If you take this rule as the basis and work out a constant ratio you can apply to all 12 semitones, you arrive at the 12th root of 2, i.e. 1.059463. Test this by multiplying this value with itself 12 times, or take a pocket calculator with the "power" (^) function and tap in 1.059463 ^ 12. The result is exactly 2 (or 1.9999 recurring). A perfect octave. Sounds complicated, but it seems to work – so where's the problem?

Let's get back to our C-major chord and work out the frequency ratios using this "magic" number. The interval C-E is four semitones, which means we should multiply the magic number with itself four times (1.059463 ^ 4). Feel free to try it – your Virus is doing this kind of thing all the time :-). The result is 1.2599. Fairly close to 1.25 (see above), but not near enough: It is detuned all of 14 cents (14% of a semitone). A singer with perfect intonation would sing the E exactly 14 cents lower than this!

The interval from C to G is seven semitones: 1.059463 to the power of 7 is 1.4983. Not bad, but not quite the 1.5 me might have expected. This detuning is equivalent to only 2 cents, but still...

So that's the big disadvantage of the tempered scale, the standard for all modern (western) keyboard instruments. Why can't the notes simply be tuned "correctly" instead? Just intonation depends upon the current situation, upon the context: An A-minor triad (A, C, E) also includes the notes E and C like in our C-major chord, but their functions are now Fifth and minor Third (instead of major Third and Root). So the frequency ratio should certainly not be 5:4 (1.25) for them to be perfectly in tune!

Is it actually possible to hear the disadvantages of Equal Temperament? Not immediately – we have become too accustomed to it throughout our lives. However, we can certainly react to its more unpleasant side-effects: Electric guitarists avoid playing full chords when using very distorted sounds. Instead, they often limit their playing to the so-called "power chords" which leave out the Third (so they are neither major nor minor). Attempting to add the Third to a power chord only results in a very rough and muddy sound.

SHORTCUTS

Shortcuts are SHIFT functions to directly access parameters which are not represented on the panel. Any of the seven local EDIT buttons can be used to enable the shift function by holding it down while tweaking the corresponding knob. The table below shows all combinations.

Shift Function	Original function	Remark
Part Volume	Master Volume Knob	Only in Multi or Multi/Single mode
Keyfollow 1/2	[Filter] ENV AMOUNT knob	Not available when vocoder is active
Filter envelope time	[Filter Envelope] sustain Knob	
Amplifier evelope time	[Amplifier Envelope] sustain knob	
Noise color	[Mixer] Noise knob	
Oscillator 3 volume	[Mixer] Sub OSC knob	Not available when OSC3 is inactive
Master clock tempo	Soft knob 1	
Mid EQ Gain	[FX] Delay/Reverb send knob	
Mid EQ Frequency	[FX] Delay/Reverb delay knob	
Mid EQ Q-factor	[FX] Delay/Reverb feedback knob	
FM Mode	FM Amount knob	
Saturation type	OSC Volume knob	

ARRANGEMENT DUMP

To trigger an arrangement dump without stepping through the SYSTEm menu, press STORE and SINGLE simmultaneously.

SINGLE DUMP

To trigger a Single dump in Single Mode, Press STORE and SINGLE simmultaneously.

With OS 5.0 build 256 we introduce keyboard shortcuts which will assist you to control even more functions directly from the front panel.

GLOBAL EQ BYPASS

Press [EFX-Edit] or [EFX-Select] plus [Undo] to temporarily enable and disable all equalizers.

ARPEGGIATOR HOLD

Press [ARP Edit] and [ARP On] to toggle the Arpeggiator hold parameter.

JUMP TO NEXT SOUND OF THE SAME CATEGORY

Press [Single-Mode] plus [Part-] or [Part+] to jump to the next or previous sound of the selected category.

NEW ARPEGGIATOR PATTERNS

The 24 new styles can be found from arpeggiator pattern number 41 onwards. They have been created with especially dance musicians in mind. There is a small number of sequences which sound very similar in the first place. They work great in sync - try to use two of those arpeggiator pattern on two different patches which running in parallel. You might also notice that some patterns contain very short notes. Those "dead notes" will spice up your music with additional groove elements.

New features OS 5.5

STORE TO FLASH

This function allows you to write your favourite patches into the patch ROM of the Virus. The actual process is similar to that used to update the operating system. For technical reasons, only complete banks can be written.

The Store To Flash function can be found here: SYSTEM>STORE TO FLASH

```
 1  STORE  TO  FLASH
               A > C  ◀
```

A>C, A>D, A>E, A>F, A>G, A>H and B>C, B>D, B>E, B>F, B>G, B>H indicate which RAM bank will be written into which segment of the Virus patch ROM. A>C means, for instance, that the RAM bank A will be written to ROM bank C. Hit the [STORE] button and confirm the safety alert message to write the chosen bank.

```
  New  ROM-Bank?
 [VAL+]→execute!
```

WARNING: DESPITE THE FACT THAT THE MANUFACTURER OF THE FLASH ROMS GUARANTEE THOUSANDS OF WRITE CYCLES, WE RECOMMEND YOU USE THIS FUNCTION WISELY. WE TOOK EVERY TECHNICALLY POSSIBLE PRECAUTION TO ENSURE THAT EVEN A POWER FAILURE DURING THE PROCESS OF WRITING WON'T AFFECT THE OPERATING SYSTEM OF THE VIRUS. IN THE UNLIKELY CASE OF PERMANENT DAMAGE TO THE FLASH ROM, THE FLASH ROM HAS TO BE EXCHANGED BY AN AUTHORISED ACCESS DEALER OR SERIVICE FACILITY. THIS IS NOT COVERED BY THE STANDARD WARRANTY.

One of the most effective ways to put your patches into the right order is to use SoundDiver Virus.

But before you start, save any precious sounds in banks A and B first! To do this, you can use SoundDiver: Request either bank by clicking the long button 'Singles Bank A' in the Device window, and then choose **Entry>Build Library>Selected Entries**. Save the library and repeat for bank B if necessary.
Alternatively, go to SYSTEM>MIDI DUMP TX>SingleBank (A or B), set your sequencer into Record mode, and hit the STORE button.

To prepare your custom banks in SoundDiver, open a library and the Device window. Shrink the windows so that both can fit on the same screen, with the Library window on the left. Now you can drag and drop any of the patches onto a specific patch location in banks A or B. When the dialog opens, click on 'Store'. The patch will then be transmitted to the equivalent location in the Virus' RAM. If you like, you can drag and drop multiple patches in one go, or if

you wish to do an entire library of 128 patches at once, choose 'Select All' from the Edit menu and make sure that you position the first patch over location 000 in the relevant bank.

Once you have compiled banks A and B to your liking, use Store To Flash to write them to your least favourite ROM banks :-)

All that's left to do, is to load your old sounds back into banks A and B, should you wish to do so. Either play the sequence containing the Midi dump, or open the library you saved earlier, Select All, and then drag and drop to the relevant bank in the Device window.

SYNC XTREME

Is Access new clock engine technology. Sync Xtreme locks much faster to an incoming MIDI clock and all parameters including the delays follow tempo changes dynamically. Sync Xtreme is based on LTB (Linear Time Base), a new technology products like Steinberg's Midex-8 multi port interface also benefit from.

COPY DELAY/REVERB FX

Copies the delay or reverb settings from a single patch into the current multi patch. The function is located in the Delay/Reverb edit menu.

```
Copy SingleDelay
  → multi [STORE]
```

To perform a copy, switch to single mode and recall the patch which contains the delay/reverb setting you wish to use within the multi patch. Next switch back to the multimode, select the copy function and press [STORE] to proceed.

The same procedure works in multi/single mode.

New features OS5.0

MULTI MODE PART PANORAMA

Each of the 16 multimode slots has a dedicated panorama setting. This setting controls the position of the sound in the stereo field. It works as overlay parameter to the single patch panorama setting. Once you enable PartPan in the Multi Edit menu, the panorama setting of the patch is being disabled.

OFF   The multi part is using the panorama setting of the single patch.

-63 .. 64   The single patch's panorama setting is disabled and this value is being used instead.

```
 1  A1  Proto-EPSV
PartPan       +1◀
```

■ *Active Multimode panorama settings disable dynamic panorama changes by e.g. the modualtion matrix.*

KEYBOARD VELOCITY CURVE

Keyboard Velocity curve adjusts the velocity response of the Virus keyboard to your personal preferences.

Negative values flatten the velocity response, positive values create a more sensible reaction. Linear (value=0) is how the Virus keyboard was set to before.

The parameter is located in the Global menu.

```
 1  KEYB  VELOCITY
Curve        +10◀
```