



VIRUS | TUTORIAL
PROGRAMMING ANALOG SYNTHESIZERS
A TUTORIAL BY HOWARD SCARR



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1: Introduction

About This Tutorial

Although primarily aimed at Virus C/KC/ Indigo II users with intermediate programming skills, many of the matters discussed could also apply to any type of modern subtractive synthesizer. Owners of earlier Virus models or the Rack XL will find plenty of information here that can be applied directly. Throughout the tutorial, readers are encouraged to plumb the depths of the Virus by way of hands-on analysis as well as several (hopefully enlightening) little experiments.

This tutorial does not describe each and every parameter (that's what the Virus manual is for), nor does it give precise instructions on how to sound exactly like Kraftwerk, Joe Zawinul or Shpongle. The input and output features are not discussed at all, even the powerful multi mode hardly gets a mention - and for good reason: Analysing individual patches will teach you a lot more than ploughing through reams of reductionist documentation. As long as you follow the analyses, walkthroughs and experiments diligently, you should notice a dramatic improvement in your Virus programming abilities. That is the sole purpose of this tutorial.

Chapter 2 provides some light information about classic analogue synthesizers which, apart from being a fascinating subject in its own right, should make the Virus heritage clear - after all, it was designed to carry on this tradition using modern technology. I hope you enjoy reading this chapter as much as I enjoyed writing it, and that it will whet your appetite for more!

Chapter 11 is a selection of useful but not-so-obvious techniques, several of which have already been alluded to in the previous chapters. If you come across any unfamiliar words you can look them up in Appendix B (the glossary). Appendix C is a list of the cryptic parts of Virus parameters to help you decipher anything you see in the display. The Internet is probably the best source of information for all those "what is pulse width modulation?" type questions.

Many thanks to Ben Crosland (BC, the 'C' of CS) for patches and advice, to Matt Picone (M@), Manuel Schleiss (DM), Stephan Vandeborn (SV), Christoph Kemper (CK) and Claudio Cordone for patches. To Zack Steinkamp and Robert Tygers for additional information. I am especially grateful to Marc Schlaile for his continuing confidence and support.

This tutorial is dedicated to the memory of Daft Eric Rigby



Setting Up

Please read this section carefully to avoid losing any data in your Virus!

You will find a bank of sounds containing all the patches mentioned in this document on the latest Virus CD as well as on the Access website. The file “tutorial.zip” contains the tutorial soundbank, this document in PDF format and a Readme file. The soundbank is available as a Sounddiver Library (.LIB) file and as a standard MIDI (.MID) file suitable for any sequencer. There is a Virus-specific version of Sounddiver on the CD which comes with every new Virus.

The recommended method of accessing the tutorial patches is to overwrite the first half of Bank A using Sounddiver - having saved what was already there of course!

...you can transmit individual patches from Sounddiver to your Virus whenever you encounter them in the tutorial. Find a patch in Bank A or B you are willing to overwrite. Open tutorial.lib in Sounddiver, make sure that Options / AutoAudition is selected, then click on the required sound (this will be sent to the Virus' edit buffer). Save each patch to the same location every time by pressing the STORE button: This is very important for the analyses/breakdowns because you would otherwise have trouble seeing the original values!

Using the Tutorial Soundbank

If you really don't want to overwrite bank A

Saving your original Bank A

Make sure that the Virus MIDI Out is connected to your computer's MIDI In.

A: as a Sounddiver library:

In Sounddiver, select the Singles Bank A bar in the Device window (the whole of Bank A is selected), then go to Edit / Request in the Sounddiver menu - this will ensure that Sounddiver has an up-to-date image of your Virus' Bank A. Once the Request task has been completed, go to Entry / Build Library / Selected Entries in the Sounddiver Menu. Save under the name “OriginalBankA.lib” via File / Save As...

B: as a MIDI file:

Make sure your sequencer program will record Sysex data, set the MIDI DUMP TX parameter on the Virus to SingleBankA, start recording then press STORE. Save/export the file as "OriginalBankA.mid", then use the VERIFY function (see the Virus manual) to make sure that everything worked properly. You may have to play back at a slower rate than you recorded at.

Loading the tutorial sounds

Make sure that your computer's MIDI Out is connected to the Virus MIDI In.

A: Sounddiver:

Set MIDI DUMP RX to ForceToBankA, open tutorial.lib in Sounddiver, select the entire library (Edit / Select All) and then Edit / Transmit. A "Paste Entry" pop-up appears. Check that the destination is A000, then click on the "All" button.

B: Sequencer:

Set MIDI DUMP RX to ForceToBankA, load/import tutorial.mid into your sequencer, make sure the MIDI port associated with the track is correct and that the sequencer is able to send Sysex. Press Play. If you encounter a problem, reduce the BPM in your sequencer and try again.

Restoring your original Bank A after completing the tutorial

Make sure that your computer's MIDI Out is connected to the Virus MIDI In.

A: Sounddiver:

Set MIDI DUMP RX to ForceToBankA, open OriginalBankA.lib in Sounddiver, select the entire library (Edit / Select All) then select Edit / Transmit. A "Paste Entry" pop-up appears. Check that the destination is A000, then click on the "All" button.

B: Sequencer:

Set MIDI DUMP RX to ForceToBankA, load/import OriginalBankA.mid into your sequencer, make sure the MIDI port associated with the track is correct and that the sequencer is able to send Sysex. Press Play. If you encounter a problem, reduce the BPM in your sequencer and try again.

Recommended SYSTEM settings

KnobMode should be set to 'Jump' (by default this is 'iRel'). LedMode should be either 'Lfo' or 'Auto'. The global Arpeggiator and Delay switches should both be 'On'. For synchronous editing between Sounddiver and the Virus panel, both LoPage and HiPage should be set to 'Sysex'.

To audition and/or play sounds you will need to connect a MIDI keyboard to your Virus. Either via computer (running Sounddiver or any MIDI sequencer) or directly if you have loaded all the tutorial patches at once. As many of the patches mentioned make use of keyboard velocity, aftertouch, modulation wheel and pitchbend, a fully-featured keyboard is recommended. The Indigo II and other small keyboards may need transposing up or down an octave for certain patches.

Syntax - Within the text

Upper case words within the main body of text usually refer to knobs or buttons. Otherwise common words which are used differently here (e.g. Time, Source, Off) as well as section names (e.g. Effects, Filters) and a few other Virus-specific terms (e.g. Env mode, Digital saturation) have been capitalised as they appear in the menus. The keyboard left-hand controls are always referred to as pitchbend and modulation wheels. Depending on the manufacturer and model, these controls may of course be levers, joysticks or touch-sensitive strips etc.

Syntax - Within edit lists

Each line starts with the section name, abbreviated to 3 letters:

Amp = AMPLIFIER
Arp = ARPEGGIATOR
Del = DELAY/REVERB
Eff = EFFECTS
Flt = FILTERS
Gen = GENERAL (all the buttons to the left and right of the display)
Mix = MIX
Mod = LFOS/MOD
Osc = OSCILLATORS

SELECT buttons do not appear in the lists - the selectable elements are referred to directly. Forward slashes separate physical controls from menus as well as menu headings from parameters (if there is more than one). Slashes and repeated words have been left out wherever the meaning is obvious.

Compound names such as SUB-OSC, FILT-GAIN, LFO1 or ASSIGN3 are either hyphenated or concatenated (SUB-OSC, FILT-GAIN, LFO1, ASSIGN3). Knobs with multiple functions e.g. TYPE / MIX or WAVE SEL / PW are indicated by their current function only. The parameter adjusted via INTENSITY is not specified further.

2: True Analogue Synthesizers

The First Synthesizer?

In 1955, the RCA (Radio Corporation of America) Electronic Music Synthesizer, a huge device made of vacuum tubes, summed sine waves at various harmonic intervals to form complex sounds - the birth of additive synthesis. It had all the modules found on modern synths e.g. envelope generators, portamento etc., although the terminology was a little different in those days. It even had filters, but apparently these were used for static formants only. Two Pianola mechanisms served as input devices, and the (non-realtime!) output was sent directly to a disk cutting lathe. This was clearly not a viable commercial venture, and many years would pass until the synthesizers we know and love became available...

The Best Synthesizer?

The Virus is a virtual analogue synthesizer (VA). All the sound-generation and processing modules you would find in any top-of-the-range analogue synthesizer have been simulated in digital signal processors (DSP).

This may seem like a step backwards after the digital revolution sparked off by the DX7, but the advantages are all too obvious. Familiar parameters in familiarly named modules (oscillators, envelopes, filters) let you programme your own sounds without having to learn an entirely new set of esoteric words and concepts every time you buy a synthesizer. Besides, subtractive synthesis has proved itself over many years to be the simplest way to create a wide range of complex sounds. Apart from FM and sampling, a few other promising approaches have been either too complicated for the average user (e.g. additive synthesis) or not as fruitful sound-wise (e.g. Casio's PD) to survive in what has become a highly competitive market.

Not surprisingly then, some of the easiest sounds to make on the Virus could be filed under the heading "analogue classics" - the kind of sounds first made popular by Moog, ARP, EMS, Sequential Circuits and Oberheim, and soon

RCA Electronic Music Synthesizer (1955)



Access Virus C (2002)



afterwards by generally more affordable products from their Japanese competitors: Yamaha, Roland and Korg. Below is a selection of analogue synthesizers from the 70s and 80s, followed by tips on how to create some of the sounds that made these instruments justly famous. Of course there were many other models, some of them equally good, but I have concentrated on the most innovative ones (including a few of my own favourites) for the sake of brevity.

Some Classic Analogue Synthesizers

Moog Minimoog (1971)



With an initial price tag of around \$1200, the Minimoog Model D was the first commercially successful keyboard synthesizer. Until the arrival of the Minimoog, most systems were far too cumbersome to take to a gig - and setting up different sounds on them would have been too time-consuming anyway. Synthesizers were the preserve of rock's aristocracy until well into the 1970s, and they remained a niche product until the mid 80s with the arrival of relatively cheap digital technology.

Despite its reputation, the Minimoog is a very simple device by today's standards: Three oscillators, noise generator, two ADS envelope generators (release time is the same as decay), a 4-pole filter, a 44-note keyboard and two control wheels all in a neat wooden case. No dedicated LFOs, no polyphony, no custom routing, no keyboard velocity, no aftertouch - and no programmability!

However, after falling out of favour when affordable polyphonic synthesizers arrived, the humble Minimoog is now considered to be a serious contender for the best-sounding synthesizer of all time (this is often attributed to the filter). Over the years the Minimoog has been used during countless recording sessions in all genres.

Comparable models: ARP Odyssey, Sequential Circuits Pro One, OSC Oscar and the new Macbeth M3X.

The ARP 2600 is a monophonic 3-oscillator synthesizer designed as an affordable alternative to Moog modular systems. The patching system was innovative - inserting cords into break-jack sockets on the front panel overrides preset connections between modules. This method was later adopted by Korg for their MS series (see below). Several versions appeared during its lifetime until production finally ceased in 1979.

ARP 2600 (1972)



The filter in version 2 is very similar to Moog's design. Moog found out, and a bitter legal battle between the two companies ensued. ARP eventually lost the case and were forced to change their filters to a 2-pole design which sounded relatively thin (unfortunately, this is the type used in the most common models). Later, ARP designed a much better filter which did not infringe on any Moog patents, and sealed it in a block of epoxy resin to ensure that neither Moog (nor anyone else) could copy the circuitry too easily.

Comparable models: Korg MS-20, EMS VCS3.

The Odyssey was ARP's answer to the Minimoog, and was essentially a repackaged 2600 - without the patching capability but with a built-in keyboard. The Odyssey had the same filter as the 2600, so several models also appeared over the years. With full-spec ADSR envelopes, sample-and-hold, highpass filter and more complex modulation possibilities, the Odyssey is considerably more flexible than the Minimoog, and many synthesizer players (including myself) consider the sound quality of the "good filter" models to be superior.

ARP Odyssey (1972)



Comparable models: Minimoog, Sequential Circuits Pro One, OSC Oscar, Macbeth M3X, Waldorf Pulse.

The VCS3 was the first truly portable synthesizer, predating the Minimoog. It still has a cult following, which is probably why VCS3s are still being built to this day - over 30 years with no improvements whatsoever! Manufactured in Britain by EMS, the VCS3 is a 3-oscillator modular synth with a pin-matrix patching system. Though notoriously prone to crosstalk and dirty contacts, the pin matrix alone turns what is (with all due respect) a box of noisy and unstable circuitry into a truly fantastic instrument worthy of being put on display in the London Science Museum (yes, it is still there). Setting up complex patches using a pin-matrix is considerably faster than using patch cords, and connect-

EMS VCS3 (1969)



ing multiple sources to single destinations or vice-versa is a doddle. The advantages of matrices over jack sockets are so great I'm surprised no other manufacturer has chosen to implement them until quite recently.

Synthi AKS (1971)



The Synthi AKS (my first synthesizer) is basically a VCS3 repackaged in a neat plastic suitcase together with a revolutionary velocity-sensitive touch keyboard cum realtime sequencer - the KS.

You can get an idea of the sheer diversity of EMS sounds from 70s albums by European artists such as Pink Floyd, Gong, Brian Eno or the late Delia Derbyshire (co-creator of the truly scary "Dr. Who" theme which has even been covered by Orbital). The Synthi A is being put to good effect (literally) these days by Sonic Youth among others.

Comparable models: EMS VCS3, ARP 2600, Analogue Solutions Vostok (new pin-matrix synth).

Korg MS-20 (1978)



The MS-20 was obviously Korg's answer to the ARP 2600. They not only copied the 2600's general appearance, but also the "hard-wired but patchable anyway" idea. Although it had fewer features and the plastic case is not to everyone's taste, the MS-20 still had a great deal to offer. It was also considerably cheaper than the ARP, possibly even contributing to the latter being discontinued. Scoring over the ARP with an extra highpass filter, integrated keyboard and a pitch to voltage converter, the MS-20 sold very well and helped establish Korg as one of the big three Japanese companies soon destined to oust their western competition.

Many well-known musicians who started out in the 80s quote the MS-20 (or the smaller MS-10) as their first synthesizer, and it was without doubt a good instrument to learn on. Apparently the MS-20 is still being used by the likes of Aphex Twin (for its wild noises) and Air (for its retro-appeal).

Comparable models: ARP 2600, EMS VCS3, EMS Synthi A.

The Prophet 5 could safely be dubbed the mother of all modern polyphonic synthesizers. It may not have been the very first of its kind, but unlike earlier models such as the mighty Yamaha CS80 (1977), it was fully programmable i.e. all relevant knob and switch values could be saved and recalled later at the press of a button. There were basically two different versions depending upon the ICs used. Early models had SSM synthesizer chips (also used later by Korg, Siel and others) while later models had the same Curtis (CEM) chips used by most other manufacturers. These analogue integrated circuits were designed specifically for the synthesizer market - which shows that what had previously been esoteric niche instruments were rapidly becoming big business.

Reminiscent of the Minimoog with its polished wooden case and black panel, the 5-voice polyphonic Prophet was a very desirable machine at the time. Although highly collectable today, the sound quality and versatility is not quite as wonderful as it is often cracked up to be (in my humble opinion). Prophets can be heard on any number of early 80s albums by e.g. Peter Gabriel, Tears For Fears or Japan.

Comparable models: Oberheim OB series, Elka Synthex, Roland Jupiter 8, Memorymoog, Native Instruments Pro-52 (extended software emulation of the original).

Conceived as the bass-playing companion to the Roland TR606 drum machine, the now (in)famous TB303 was a terrible flop during its brief production lifespan. However, this humble monophonic toy has since become instrumental in spawning and sustaining entire genres of electronic music e.g. (Acid)-House, Techno and Trance. Until recently it was a “must have” for anyone requiring that unique overdriven, squelchy resonant sound. For lack of any alternative, second-hand prices soon shot up to over \$1000.

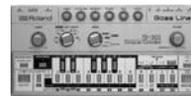
A major feature of the TB303 was its built-in step sequencer with programmable pitch, accent (like a drum machine) and per-note portamento. Performance was thus reduced to twisting knobs on the panel, especially those parameters affecting filter frequency and resonance, so you didn't even need to take piano lessons. TB303s can be heard on too many Techno tracks to mention.

Comparable models: Propellerhead Rebirth (software), Novation Bass Station and several other hardware and software clones.

Sequential Circuits Prophet 5 (1978)



Roland TB303 (1982)



Roland Alpha Juno 2 (1986)



the 6-voice polyphonic Alpha Juno 2 is a step up from the Alpha Juno 1, with a velocity sensitive keyboard, aftertouch and a memory cartridge slot added to the original specifications. Roland dubbed the large rotary encoder used to edit and browse through the parameters an “Alpha Dial”, and such encoders have since become quite common on a variety of instruments from other manufacturers, notably Waldorf. The oscillators can be stacked up in unison and heavily detuned against each other, forming the basis of the popular “Hoover” sounds. Like the TB303, second-hand prices have recently soared. Hoovers are often featured heavily on Dance/Eurobeat albums.

Comparable models: Roland Alpha Juno 1, Roland JX-8P.

Oberheim OB-8 (1983)



The stylish eight-voice polyphonic OB-8 was the last model in their OB series of synthesizers (OB-X, OB-Xa and OB-SX), and is treasured for its smooth, warm filters. Released just before the MIDI protocol was finalised, MIDI was soon available as a retrofit (and later as standard). The dedicated controls and logical layout make programming the OB-8 very easy, although more advanced features can be accessed via a “page 2” mode.

Some specifications: 2 oscillators, 2 or 4-pole lowpass filter, 120 single patches, 24 performances (for split/double sounds), 8-note arpeggiator, sound storage to external cassette recorder.

Comparable models: all other OB-series synths, Elka Synthex, Roland Jupiter 8, Matten+Wiechers Banana, Alesis Andromeda A6.

Oberheim Matrix 1000 (1988)



The Oberheim Matrix 1000 is arguably the most underrated true analogue synthesizer ever, possibly due to the fact that patches cannot be edited directly on the panel. Successful enough to be re-issued in 1994, production finally ceased in 1999. It should be of special interest to Virus owners that 1994 was also the year when Access brought out their very first product - a Matrix 1000 hardware programmer!

With the same 6-voice architecture as the Matrix 6, including quite a comprehensive modulation matrix, the Matrix 1000 is an excellent source of rich brass and string sounds, complex pads and some meaty basses. The lowpass filter is extremely smooth and the architecture is very flexible. Unfortunately, any rapid pitch modulations such as vibrato or portamento can cause the sound to become very grainy (like a VA without parameter smoothing).

If you want to create your own sounds instead of simply recalling one of the 800 presets or 200 user sounds, you will need an editor program for your computer e.g. OB 6000 (for Windows). At the time of writing you can pick up a second-hand Matrix 1000 for well under \$300, so if you don't own an analogue synthesizer yet and are on a very tight budget, this is definitely one to look out for!

Comparable models: Oberheim Matrix 6R, Cheetah MS6.





3: Programming Classic Analogue Sounds

An Egg-Laying Woolmilk-Sow?

Despite what some analogue purists may say, the Virus (whether A, B or C) is quite capable of reproducing just about any sound originally made on true analogue synthesizers - from the simple Minimoog to the polyphonic flagships of the 1980s. Admittedly you have to apply a few tricks to coax (for instance) an Oberheim-type filter sound out of the Virus' DSP, but in my opinion the Virus filter is still the most versatile, most 'analogue' sounding digital filter around. Besides, if made of discrete analogue components, a synthesizer with specifications similar to a Virus C would not only weigh a ton but would also cost more than... well, more than I could afford.

This chapter introduces several methods of recreating the sounds which made synthesizers highly desirable objects in the first place, and also some landmark sounds which helped generate fresh interest in all things analogue from the mid '90s to the present day.



Classic analogue lead sounds must be played well, otherwise they can sound rather bland - or even downright boring!

Analogue Leads

Superficially, Minimoog architecture differs from a subset of the Virus in only two ways: portamento is linear and the keyboard has low-note priority. Although the Virus cannot emulate these two directly (e.g. for live work), they are not all-important, and it is possible to make a pretty good clone of just about any Minimoog sound. Differences between the raw waveforms in the Minimoog and Virus can be worked around by mixing waveforms within each

The Minimoog in your Virus - Part I

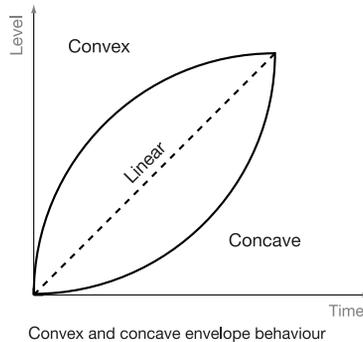
oscillator, as well as by meticulous use of Sync, saturation and other parameters. Getting the filter right is less of a problem than e.g. emulating an Oberheim filter.

PATCH: AT-MINI HS

“Objection, your Honour! The Virus envelope attack is absolutely linear, whereas the Minimoog attack is slightly convex and has a short plateau. The Decays are different as well”. No problem: Load the sound AT-Mini HS and look into the ASSIGN 2 parameters in the mod matrix. What’s this? The filter envelope is controlling its own attack and decay times?

This is a prime example of ‘recursive modulation’, and here’s how it works for an envelope: Attack time is being continuously lengthened by the current attack level so it becomes convex. Strangely, decay time is continuously lengthened by a *negative* Amount of modulation from the current decay level, and decay becomes more concave.

To explain this as simply as possible: decay level is a downward-moving source, and a minus times a minus is a plus.

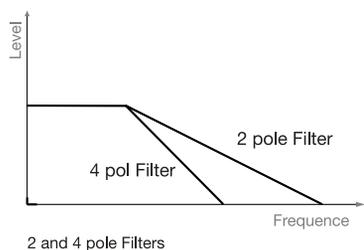


Recursive behaviour in general is difficult to visualise and possibly even harder to explain, but as long as you remember that positive self-modulation of envelope times always causes convexity and negative self-modulation always causes concavity, you don’t really need to think in any other terms.

 ***Solo sounds of the 70s often have a characteristic “pluck” caused by concave Decay. For this type of sound, use a spare slot in the modulation matrix to make at least the filter envelope Decay more concave than it normally is.***

I assigned Aftertouch to vibrato because the mod wheel is already being used to close the filter in this patch (particularly expressive with Roland-style integrated left hand controllers instead of a pair of wheels). If you look in the ASSIGN 3 parameters, you will see that channel pressure (Aftertouch) is not only controlling LFO 3 depth, but also increasing LFO 3 rate as well as opening the filter a bit. Although the Minimoog is not touch-sensitive, I often use this

method to make a sound more responsive to live playing, and I think its use is also valid here. If you are wary of RSI (repetitive strain injury) or your keyboard is not pressure-sensitive, you could set ASSIGN 1 source to Off and ASSIGN 3 source to ModWheel instead.



If you have already taken a look at the filter parameters, you might be wondering why they are set to Ser4 (i.e. both 2-pole, 12 dB/octave each) although the Mini-moog has a single 4-pole filter. There is a simple reason for this: A pair of 2-pole filters in series can sound very similar to a single 4-pole, but the default setting (Ser4) gives you more options to refine a sound, especially when trying to emulate

particular filter characteristics (see “Filter Balancing Acts” in chapter 11, page 17).

This is a more cutting lead sound as used by both Stevies (Wonder and Win-wood). Originally a clone of the solo sound heard at the end of Stevie Wonder’s “I’m New” (Conversation Peace, 1995), it eventually turned into something which can wail like a blues harp if you practice enough. SmoothMode has been turned off to make pitchbending much less sluggish - one of the very few good reasons to do this! There is no real need for a more concave decay here because the sound stays bright, but ASSIGN 2 has been used to make the Attack convex. I know it makes only a subtle difference, but...

PATCH: SYNTHAR HS

The secret is to be obsessive about it...

The EDIT light in the filter section is on, which means that Saturation is being used. Open the filter’s EDIT menu and check the Saturation curve setting. OSC VOL is at +63, so the signal is being completely rectified. Try turning OSC VOL down to 0 (not -64) and notice how much difference this makes!

If you have an oscilloscope or an equivalent program in your computer (e.g. the freeware “Wave Tools” for Windows), it is well worth looking at what rectification does to a wave. It can be used for morphing up an octave - see Flutoon HS for a good example of this trick.

PATCH: C.TOWNS HS

You will find a sound called C.Towns HS in the tutorial bank, a clone of the melody synth in “Kate” from the Full Circle soundtrack (Colin Towns). Actually, I cloned an entire section of this piece just to show how analogue the Virus can sound - KateClone.mp3 might become available on the Access web site: www.access-music.de. Also included is a very good retro-lead sound called Moog-ST< > (I have taken the liberty of changing the source for vibrato to channel pressure).

You might also like check out some more of the clones in Bank D e.g. Cpt Kork HS, Froese HS (“Detroit Snackbar Dreamer” from Stuntman), Manfman HS (Manfred Mann), MoonweedHS (a typical Tim Blake filter sweep sound), Oddigy HS (Moog Prodigy) or ShineOn HS (courtesy of Pink Floyd). Note that these patches are much less aggressive than modern lead sounds. Some others to look out for in the Virus C presets are: FunkLd-3SM, LuckyMan J and a few of the RP sounds in bank H. TheramosM@ is wonderfully dated, and should only be played while watching Forbidden Planet!

Analogue Basses

The Minimoog in your Virus - Part II

These days, the Minimoog seems even more famous for its bass sounds than its leads, although most analogue synthesizers are capable of good basses. With a little equalisation of course. THE analogue bass sound per se doesn’t exist, there are simply too many usable variations out there. However, there is a seemingly harmless patch in your Virus which, with a few simple tweaks, can be turned into just about any old (sic) analogue bass sound you like: Club-BassSV. Of course there are several other good patches lurking in the presets, but I have often found this one to be the best starting point because of its simplicity.

PATCH: CLUBBASSSV

This patch already sounds meaty. One obvious reason is that the Analog Boost intensity (look in the Effects section) is at maximum, but I think there are some equally important factors involved:

Punch. This aptly-named parameter is set very high. Try turning it down and notice how the sound loses its aggressive attack.

Sub-oscillator. Although an octave lower, the sub-oscillator is mixed in at a level which still allows the main oscillators to be perceived as delivering the fundamental.

The short filter envelope decay, low CUTOFF and rather high ENV AMOUNT interact to cause the low frequencies to stand out shortly after a note is played.

It is simple. The oscillators are phase-locked (because SYNC is switched on), and there are no added effects such as chorus or reverb mushing up the sound.

Right, let's see how quickly ClubbassSV can be turned into something a lot more squelchy. Before you start tweaking parameters I should mention that it is best to play a few notes after each step so that you don't fall into the trap of doing things parrot-fashion.

PATCH: REZBASS HS

Step up a patch and then down again using the VALUE buttons to get rid of any changes you might have made. Turn SUB OSC up to maximum and RESONANCE to 45. Even with this amount of resonance, the sound gets too boomy when Cutoff (the frequency where all that resonance is) reaches the fundamental. This can be remedied by turning CUTOFF up just a little - to 38. The higher notes sound muffled? Setting Keyflw (in the filter EDIT menu) to around 25 cures this. Now let's make the filter decay more concave using ASSIGN 4: Source = FiltEnv, Dest = FiltDecay, Amount = -20. Adjust filter DECAY to 60 to compensate.

The same tweaks as a list: Walkthrough from ClubbassSV to RezBass

Mix SUB-OSC = 127
Filt RESONANCE = 45
Filt CUTOFF = 38
Filt EDIT / FILTERS / Keyflw = +25 +25
Mod ASSIGN4 / Source = FiltEnv
Mod ASSIGN4 / Dest = FiltDecay
Mod ASSIGN4 / Amount = -20
Filt DECAY = 60



Just about everything in the Virus is interactive. Adjust one parameter and you often need to adjust another to compensate. Make a point of getting to know how and why parameters interact.

After saving this patch (if you like it) you might also like to try the following. These three little walkthroughs demonstrate how just a few tweaks can change a patch drastically. As always, play a few notes after each and every step:

Walkthrough from RezBass to WiryBass

Eff DIST TYPE = Shaper
Flt DECAY = 0

Walkthrough from WiryBass to WahBass

Flt FILT1 = BS
Eff DIST TYPE = Rectifier
Eff INTENSITY = 127
Flt ATTACK = 55
Flt DECAY = 50

Walkthrough from WahBass to AcrylBass

Eff DIST TYPE = Hard
Flt CUTOFF = 50
Flt ATTACK = 20
Flt DECAY = 64
Eff EDIT / EQUALIZER-MID / Gain = +8
Eff PHA MIX = 64

There are approximately 5,742 other bass sounds waiting to be made just as quickly. By the way, did you know that 63.4 percent of all statistics, facts and figures are simply made up on the spot?

Go back to the original ClubBassSV for a subtle but interesting effect you might be able to make good use of in future: First, temporarily turn SUB OSC down to 0 and Cutoff up to 50. Now try slowly turning up DETUNE while listening carefully to the sound. You should be able to hear different harmonics being picked out as you turn the knob. At 43 you get quite a strong fifth, and between 60 and 64 you get different mixtures of an octave higher. If you set DETUNE to 64 (there is hardly any fundamental now), then go into the modulation matrix and assign Random to OSC 2Detune with Amount = +63, you will get different harmonics each time you play a note. I guess this “life” is one reason why you bought the Virus instead of yet another ROMpler.



Subtle changes in the attack each time you play a note can make a sound less tiring to the ear, more “alive”

Check out some more old-school bass sounds in the Virus C presets e.g. Back280sSV, Da Funk BC, Goomby HS, TimeStep HS, MiniBS-5SM, Smooth-BsBC and several of the RP sounds in Bank H.

Analogue Brass and Strings

To our sampler-pampered ears, the so-called brass and string sounds of the 70s and 80s bear little resemblance to brass bands or orchestral string sections. However, the ability of a particular brand of polyphonic synthesizer to deliver buttery (though synthetic-sounding) brass and strings was, and still is, considered a sign of quality.

This is an example of the kind of “brass” sound which sold Oberheims by the score. It is basically just a mix of sawtooth and pulse oscillators sent through a lowpass filter.

Note that the level of oscillator 3 is not 127 but 64 - the same level as oscillators 1 and 2 when OSC BAL is in the centre! To get some extra sheen, I set the Unison parameter to Twin but otherwise I have left it raw i.e. without additional delay effects. Oberheims sound to me as if they are already in a medium-sized wooden room! Turn up MIX and SEND in the EFFECTS section if you like - I have already set up some default values for Phasing and Delay.



Make a habit of checking which particular selection LEDs are lit immediately before saving a sound. If you want immediate access to e.g. LFO 2, oscillator 3, filter 2 and the Phaser every time you recall a patch, you should use the SELECT buttons to activate these. Then save.

To my ears, Oberheims often sound as if envelope decays are initially linear or even slightly convex. Take a look in the modulation matrix: ASSIGN 2 connects the modulation wheel to LFO 3 (for vibrato). In ASSIGN 3, FiltEnv is modulating filter envelope Attack and Decay times (as well as amplifier Release time) so that attack becomes convex and the other segments become (practically) linear - remember that positive recursive modulation causes envelope segments to become more convex or less concave. For the cream on top, I have modu-



The OB-8 in your Virus

PATCH: OBIFANF HS

lated EQ mid-frequency from both the filter envelope and Aftertouch in an attempt to emulate the totally un-fizzy Oberheim sound more closely - it worked!



For classic analogue brass sounds, it is a good idea to spend most of your time and effort on the filter and its envelope.

PATCH: STTHEORYBC

This is a typical analogue strings sound with just the right brilliance and length of decay for my taste. As usual for analogue strings, the two main oscillators are fairly narrow pulse waves with vibrato in opposite directions (see LFO 1 destinations). Extra movement comes from LFO 2 (panning) and LFO 3 (vibrato) but there is no added Chorus or Phasing. Other factors which make this patch so expressive are the velocity-sensitive Attack time (see ASSIGN 2) and the slightly concave (!) Attack (see ASSIGN 1).

Let's see how important the Amplifier envelope and EQ settings are for the sound quality (I don't really care for the word "timbre" myself):

Experiment from StTheoryBC: Env and EQ

Amp ATTACK = 0
Amp RELEASE = 0
(I would still call this a string-sound...)
Eff EDIT / EQUALIZER-LOW / Gain = +0
Eff EDIT / EQUALIZER-MID / Gain = +0
Eff EDIT / EQUALIZER-HIGH / Gain = +0

By removing the EQ, we have transformed StTheory into a cheap brass sound.

EQ is what makes the Virus C brilliant instead of just great!

PATCH: STR ABEJA

The tutorial bank includes a patch called STR ABEJA. This one also captures the spirit of '80s flagship synthesizers.

Analogue Effects

The EMS VCS3 and other modular synths are capable of so many weird and wonderful effects that it is absolutely impossible to categorise them. On second thoughts, according to the late Douglas Adams, anything is possible in an infinite universe, so in order of historical occurrence and/or patch-complexity we have...

1. Early sci-fi effects
2. Filtered noise effects
3. Psychedelic bubbles and swoops
4. Ambience

Apart from Theremin sounds, the cheesiest effect used in old Sci-Fi films was of the “chaotic bleeps” variety. The standard method is very simple - an LFO set to Sample and Hold (not available on the VCS3!) modulates pitch. Go to Old S+H HS and look into the LFO 1 parameters. WAVE is set to S+H and modulates both oscillators in opposite directions. Filter 1 takes out the highest frequencies to simulate the reduced bandwidth of old tape recordings (I made this sound pre-Virus C, so I couldn't use EQs). Try this sound only if you like very strong cheese...

Another characteristic of old Sci-Fi noises is that they were almost invariably sent through short delays (using tape recorders) and/or spring reverbs. A much cheaper alternative to plate reverbs, you will find these in guitar amplifiers, Hammond organs and the VCS3.

Go to SpacePat HS and give it a listen. Now turn down SEND to defeat the reverb and hold down a key on your keyboard. The rising pitch you hear is not an oscillator, but Filter 1 set to a high resonance. When pitch reaches maximum you can hear a deep vibrato (Oscillator 2 only, as OSC BAL is set to maximum). This is what excites the reverb, giving the sound its retro-appeal. The vibrato here is caused by FM (tweak FM AMOUNT to check this): the pitch of the modulator i.e. oscillator 1 is very low, it is even sub-audio. More about the FM-vibrato trick in the section “Psychedelic Bubbles and Swoops” below.

Step up and down a patch to defeat any changes, and look in the Effects section to find out how the delay/spring reverb combination is simulated. SEND is very high, DELAY/REVERB Mode is set to Rev+Feedb2, of type Ambience with a medium Predelay and a good dollop of Feedback. Real spring reverbs reflect

The VCS3 / ARP 2600 in your Virus

Early Sci-Fi Effects



PATCH: SPACEPAT HS

a sound back and forth along the length of the spring, and so Rev+Feedb2 is perfect for the job - the feedback parameter can be used to simulate these reflections. I originally set Predelay to 71 for quite a convincing spring reverb (try this), but later decided that increasing the value made it sound more like an old delay with a very short reverb, which I particularly liked. It is well worth experimenting with reverb by sending a simple impulse through the Rev+Feedb modes and tweaking *all* the parameters.

Now turn OSC BAL down to minimum (-64), bringing in oscillator 1 instead of oscillator 2. Play several low notes at random on your keyboard for a kind of rising “bone soup” effect caused by spikes from oscillator 1 exciting the resonance in the Filter.



Getting good variations on a sound at the turn of a knob is very useful, so try to set up optimum values for unused modules/parameters. Filter 2, oscillator 3, the sub-oscillator, DIST/PHA/CHO are often left out, so these are particularly good candidates.

PATCH: TANKYOU HS

I have included this as a further example of using spikes to excite resonance in filters. A similar method is particularly good for xylophone-type sounds - see chapter 9 “EasyXylo”, page 81.

Filtered Noise Effects



Noise effects were often used by Robert Margouloff and Malcolm Cecil of TONTO (who were also sound designers for Stevie Wonder), the eclectic Bernie Krause (of Beaver and Krause), Tangerine Dream, Hawkwind, Pink Floyd etc. during the 1970s. Many of these sounds were attempts at emulating wind and waves.

This is my Virus simulation of howling wind, “the force 10 gale in your Virus”. Both LFOs are set to S+G (a smooth random wave): LFO 1 modulates the oscillator frequencies in opposite directions, LFO 2 does the same for the filters as well as modulating the overall pitch and even its own rate (another example of recursive modulation - look in the LFOS/MOD section to check all this).

The filters are in parallel and one of them is set to high-pass to prevent there being too much rumble. The idea while making this sound was to introduce enough random factors to make the results as bleak as a Grimsby suburb on a cold and grey November night...

PATCH: MCCLLOUD HS

...you are lost. You have just missed the last bus.

This patch was originally supposed to sound exactly like breaking waves, but I got carried away (often a source of new sounds) by the wonderfully variable Virus phaser, which became heavily featured in the final version. Using only a single Stage makes the phaser sound much more like an old analogue stomp-box such as the Electro-Harmonix "Small Stone" than any modern multi-stage unit.

PATCH: SUNDER HS

A bit of info for all fellow anoraks: although made in the USA, the very popular Electro-Harmonix devices were actually designed by a Brit, the very same person who had designed the EMS synths and later all those Akai samplers. David Cockerell (who still works for Akai) probably deserves to be mentioned in the same breath as synthesizer pioneers Max Mathews, Don Buchla, Bob Moog or John Chowning. While I'm naming trailblazers, it's worth looking up "Hugh LeCain" in the Internet!

The method used to mute the oscillators in Sunder HS is interesting - they are both set to an infinitely narrow Pulse (i.e. PW at 127). Although this is redundant here (you could turn OSC VOL down to -64 instead), you will learn a good reason to know about this later on in the tutorial.

The type of effects peculiar to just a few psychedelic rock bands - Gong, Hawkwind and Ozric Tentacles, originated in the mind and hands of a certain Tim Blake (aka "High T Moonweed", who played EMS synthesisers with both Gong and Hawkwind at different times). Self-oscillating filter swoops and bubbles, oriental drones and arpeggios are sent through a stereo tape delay to create a wide and truly psychedelic wash of sound. Although relatively obscure these days, some interest in these sound effects has started to reappear recently in the Trance scene.

Psychedelic Bubbles and Swoops

Although much fuss is still being made about whether the digital filters of synthesizer XYZ can self-oscillate, this feature is redundant...

Unless you really want to morph a sound in and out of resonant chaos you can use an oscillator instead. Would you like to experiment with a slightly fragile sine wave typical of the EMS self-oscillating filter? Start with patch C127 and set both SHAPE and OSC BAL to minimum. Take Amplifier ATTACK up to 42. Change LFO 1 SHAPE to WAVE = S+G and its RATE to maximum. Select the OSC 1 destination for LFO 1 and change the value to +5.

Experiment from C127: Fragile EMS filter

```
Osc OSC1 SHAPE = 0
Mix OSC-BAL = -64
Amp ATTACK = 42
Mod LFO 1 SHAPE = WAVE S+G
Mod LFO 1 RATE = 127
Mod LFO 1 OSC 1 = +5
```

PATCH: FLYINGT CS

Bubbles: this is quite a descriptive term for the type of sounds made by rapidly modulating the cutoff frequency of a self-oscillating filter with an LFO (or by EMS's self-repeating envelope), then sweeping cutoff frequency as well as LFO speed. Load FlyingT CS, hold down a note on your keyboard and play around with the left-hand controls for a while. The sine wave is from oscillator 2 (the Virus filters do not self-oscillate). You might think the bubbling is caused by an LFO, but it is audio-oscillator FM. I chose this method because the Virus LFOs are incapable of audio frequencies, and I like the sound of sweeping the modulator frequency over a very wide range i.e. in and out of audio (which starts at around 30Hz). Try it.

To see how this patch works, first press EDIT in the Oscillators section. Oscillator 1 KeyFollow is set to zero and Oscillator 2 FmMode is PosTri. Oscillator 1 SEMITONE is quite low (-38), and together with Transpose at -1 this makes its nominal frequency sub-audio. There is a medium FM AMOUNT (56) for the right size of bubbles - try different values here.



Swoops are simpler than bubbles - just leave out the rapid modulation and concentrate on controlling pitch and volume. Early electronic musicians such as Pierre Henry and Todd Dockstader often used sine wave generators, but it was Tim Blake who turned this simple effect into something a bit more psychedelic than a toot on a swanee whistle!

The trick was this: hook up your VCS3 or Synthi A to a ping-pong delay, patch the filter into one of the outputs, grab the filter frequency and resonance (Q) knobs and move your elbows very dramatically while pulling faces as if you've

just had an overdose of Viagra. Twisting the filter Q between not-quite-self-oscillating and self-oscillating gives you volume control while filter frequency gives you pitch control. If you also patch in an LFO (or the EMS self-repeating envelope) controlling filter frequency, you can easily morph this sound into bubbles (see above).

What do you mean, you don't have a VCS3 or Synthi A handy? OK, you can do it on a Virus C instead: Go to Swoopie HS and turn SOFT KNOB 2 (an unfortunate choice of name!) all the way down. Use the AUDITION function so you don't have to play the keyboard (press EDIT and SYNC for two seconds). Now tweak away on the two SOFT KNOBS. If you happen to have three hands, you should also try some FM AMOUNT.

PATCH: SWOOPIE HS

How it works: The SOFT KNOB destinations are MIDI Controllers 13 and 14, and these are also the sources used for Transpose and OscMainVol in the modulation matrix. I took this roundabout route because Transpose as a direct SOFT KNOB destination gave me quantised pitch, not nice and smooth as I had expected. Also, OscMainVol is not included in the list of SOFT KNOB destinations (in my current OS) because it already has a dedicated knob. Although this patch is simple, it took a while and a bit of outside help until I got it working properly. In the immortal words of King Missile:

If at first you don't succeed, fail and fail again. Or ask a Virus expert, which is what I did (thanks Ben).

As far as I know, the father of ambient sounds in western popular music was one Bernie Krause, who teamed up with organist Paul Beaver during the late 60s. In 1967 Beaver and Krause released "The Nonesuch Guide to Electronic Music", still considered a standard reference in the annals of electronic music history.

Ambience



Bernie Krause is a highly interesting character who was not only instrumental in promoting the use of synthesizers in the broadcast media, but also put forward quite a plausible theory on how music-making began all those millennia ago. The following paragraph is a short excerpt from Bernie Krause's article "THE NICHE HYPOTHESIS: How Animals Taught Us To Dance and Sing". I think it is worth quoting here, although the good Doctor Krause says this section is a bit out of date...

Experienced musical composers know that in order to achieve an unimpeded resonance the sound of each instrument must have its own unique voice and place in the spectrum of events being orchestrated. All too little attention has

been paid to the possibility that insects, birds and mammals in any given environment have been finding their aural niche since the beginning of time... A complex vital beauty emerges that the best of sonic artists in Western culture have yet to achieve. Like the recent acknowledgement that medicine owes much to rainforest flora, it is my hunch that the development our sound arts owes at least as much to the "noise" of our natural environments.

Whether this really applies to Cro-Magnon bone-bashing or not, this article certainly got me thinking a bit further than usual. I came up with the following, very obvious answer to a big question that had been bugging me for many years: Because of the huge advantage of listening intently and recognising patterns in all the animal noises around us (otherwise we get eaten and/or fail to catch our prey), early humans evolved to take great pleasure in this activity - it was another survival factor like eating, sex and physical exercise.

OK, that applies more or less to all animals with ears, but we humans were the only species brainy enough to develop highly structured music over a few generations (memes require intelligence). So the sonic pleasure of the whole tribe is maximised and they stick together through thick and thin. I suppose we now have "death by chocolate" type foods, porn and hooliganism for similar reasons - there are a few downsides to being an intelligent species of pleasure-seekers!

If you are interested in Bernie Krause's lifework you should check out "Wild Soundscapes: Discovering the Voice of the Natural World" (Wilderness Press, Berkeley, California, 2002), or "Into a Wild Sanctuary: A Life in Music and Natural Sound" (Heyday Books, Berkeley, California, 1998).

To make complex ambient sounds in the very early days of synthesizers you needed a lot of 'patchability' to get away from the Sci-Fi clichés. When the affordable VCS3 arrived it became easier to create weird and wonderful sounds. In Europe, EMS synths were very popular with radio and television people such as the BBC Radiophonic Workshop (e.g. Dr. Who) or Edward Williams (who composed music for the classic TV-series "Life On Earth"). One of the patches in the presets reminded me of sounds made at the Beeb in the late 1960s, so it is well worthy of analysis in this context - Nowhere BC...

 *To analyse a complex synthesizer programme, defeat individual elements top-down to isolate the factors responsible for the basic characteristics of the sound. You can usually start with the effects.*

The most obvious attribute of this sound is its heavy reverb. As the reverb parameter values are fairly standard (press EDIT and have a look), turn SEND fully anticlockwise and let's move on. There is still a noticeable shatter effect, so the next job is to find out where this is coming from. Simply turning amplifier RELEASE down immediately proves that this is not due to any of the Virus effects.

We could now check the RING MOD level as this is often the basis of metallic sounds (including the shatter). Bingo - it is set to 119. Turning this down to zero leaves us with what seems to be a random frequency sine wave with some coloration from the effects section (chorus or phaser or both). To check which of these is the source of the coloration, press SELECT in the Effects section until the Phaser LED lights up. Turning the TYPE/MIX knob shows that this is set to 66, but turning it right down doesn't seem to make much difference.

Press SELECT once to access the Chorus (CHO) parameters. MIX (Dir/Eff) is high, and turning this down to zero shows that we have finally found the source of the coloration. You won't be able to see any parameters in the Chorus edit menu as long as Dir/Eff is Off, so turn the MIX knob back up. As Rate (the speed of the Chorus' own LFO) is set to zero, this makes the effect static and any modulation depth is redundant here.

Why Virus Oscillators Sound so "Analogue"

Each time you play a note, the pitch (and tone!) of the oscillators vary slightly. This typical attribute of real analogue oscillators can be defeated by changing the Phaselnit parameter:

Experiment from C127: Analogue-feel

Gen EDIT / PUNCH Intensity = 0
 (Play middle C repeatedly while listening to the slight variations in pitch and tone, then...)
 Osc EDIT / Phaselnit = 1

You should notice quite a difference once your ears have become attuned to the sound!



4: Programming Modern Analogue Sounds

Tee-Bees

These sounds (as well as Hoovers - see below) have become so widely (over)used during the last decade that they merit a separate section in this tutorial. Looking through the presets in the Virus C, I noticed there are two fundamentally different types of TeeBee sounds there: those which use the arpeggiator and synchronised LFOs as an integral part of the sound, and those which don't.

FuzzlineDM is a great example of the former type. As I prefer it this sound with less resonance than the original, I have taken the liberty of turning this down a bit (apologies, Manuel). The most obvious characteristic is the regular filter modulation. Looking in the modulation matrix (which is often the first thing I do when analysing a sound) shows that Cutoff is being modulated by LFO 1, so we should take a look at the LFO 1 parameters in that section's EDIT menu first. LFO 1 is clocked at one cycle per bar and that's why the 'squelch' is so regular.

Interestingly, the waveform is one of the additional waves (wave 9 - press SHAPE to check this) and Contour is set to -9. To find out why, we could do a little experiment. As always, be fully aware of everything you do here, and why:

Go to the -START- patch (C127) and press the SHAPE button three times to select WAVE. Change it to wave 9. Press the destination SELECT button (the one below the EDIT button in the LFOS/MOD section) and take the modulation level up to maximum by holding the Value '+' button, then the Value '-' button. Release '+' before you release '-'. Once you get the hang of it, you won't go straight for the Value knob every time you need minimum or maximum values.

The TB303 in your Virus



PATCH: FUZZLINEDM



Whenever you need to set minimum or maximum values (including patch numbers) you can use the above method instead of turning the VALUE knob or holding a VALUE button down. See Appendix A for a list of all such shortcuts, page 107.

If you now press a note on your keyboard you should hear a simple ‘scratch’ type sound. Press ARP ON and then go into the LFO edit menu. Set the same values as in FuzzlineDM i.e. LFO 1 Contour to -9, Clock to 1/1. If you like, set EnvMode to On. Because it is clocked the LFO would be started at the same phase in its cycle anyway, but setting EnvMode on also makes it unipolar, and that’s why the pitch goes up.

Hold a note on your keyboard and listen to how the pitch moves - it mirrors the way the filter opens and closes in FuzzlineDM. Controlling pitch demonstrates the LFO shape much better - which is the object of this experiment. Changing the value of LFO 1 Contour to more negative values clearly shows that you can zoom right into the wave. More about the Contour parameter below and in chapter 11 under “Non-Standard LFO shapes”, page 91. See Pstyro3 HS for a much more complex version of this idea.

Experiment from C127: Zooming into waves

```
Mod LFO1 SHAPE = WAVE 9
Mod LFO1 / OSC1 = +63
Arp ARP-ON (on)
Mod EDIT / LFO 1 / Contour = -9
(then tweak this value further)
```

Back to FuzzlineDM. So where does the ‘fuzz’ come from? Apart from the police station there are two obvious candidates - either the filter saturation stage or distortion from the Effects section. As the LED above the EDIT button in the Filter section is lit, we should look there first. The saturation curve is set to Hard and OSC VOL is at maximum, so there we have it.

Next question: Why does the arpeggiator only give you the last note i.e. it does not cycle through all the notes you hold down? This is because the arpeggiator is in Chord mode and the patch is monophonic. You might think that ASSIGN 3 is redundant because the destination (OscMainVol i.e. OSC VOL) is already at maximum. However, if you happen to turn OSC VOL down...

Representing the non-arpeggiated TeeBee sounds I chose UniVoV BC because I think it has the typical nasal quality of an overdriven TB303 (caused by filter 2 in bandpass mode). Turning FILTER BALANCE all the way up makes it even more nasal. This type of sound is best played by a sequencer with most of the notes overlapping - you will see why when you try!

PATCH: UNIVOV BC

Hoovers

Until last year I was blissfully unaware of what people meant when they talked about 'Hoover' sounds, so by default I filed the word under 'passing fad'. However, curiosity soon got the better of me and I started collecting example MP3s I found on the Internet. I still don't fully understand the appeal of these noisy and out-of-tune sounds, but at least I now know why the name is synonymous with a brand of vacuum cleaner.

The Alpha Juno 2 in your Virus



💡 *Try not to dismiss any particular types of sound (or even entire music-genres) out of hand like I tend to. They might just grow on you...*

Apparently, the original Hoover sound was a preset called "What The ..." in the Roland Alpha Juno 2. Strangely, I have yet to find a sample of this patch on the Internet.

I programmed J.Edgar HS to be the lowest common denominator of all the Hoovers I had found, so it is fairly standard fare. However, it is certainly dusty enough and has oodles of suction when the mod wheel is pushed up. All that dust comes from sawtooth waves stacked up and heavily detuned in Unison mode without using the filter envelope. This seems to be the simple secret behind Hoover sounds.

PATCH: J.EDGAR HS

Press the main EDIT button (the one above the BPM LED) and scroll through the menu until you see UNISON, which is set to 6 (with maximum Detune). Try turning Unison mode to Off to see how important Unison is for this sound. The only used slot in the modulation matrix is ASSIGN 1 - the mod wheel is controlling DistInt (distortion intensity). The distortion TYPE set in the effects section is LowPass, and the negative Amount (-32) in ASSIGN 1 causes the mod wheel to open the filter from its nominal value of 70. Select DIST and turn the INTENSITY knob to check or change this.

PATCH: DYSON++ BC

Now this is the sort of Hoover you wouldn't want to meet down a dark alley! Filter1 is set to Bandpass and its cutoff is controlled via the mod wheel or by SoftKnob2 (Scream). Instead of using sawtooth, both main oscillators are set to Pulse with quite a narrow width. This aggressive source is made even tougher by using Hard saturation (look in the Filters menu). Also check out what the mod wheel is doing (look in both ASSIGN 3 and ASSIGN 4). The pulse width of both oscillators is being rapidly modulated by LFO 3, and there is quite a lot of chorus for maximum movement.

What's really special about Dyson++ is the Dive parameter controlled by SOFT KNOB 1. Press the main EDIT button (the one above the BPM LED) and check that it has been assigned to Contr3 i.e. MIDI continuous controller number 3. This controller is also the source in ASSIGN 2, modulating both LFO 1>OSC 1 and LFO>OSC 2. The amount (26) is enough to send the pitch of the oscillators up 2 octaves - turn SOFT KNOB 1 all the way up and check this.

So how does the pitch stay up for a while before finally 'diving'? The trick is in the shape of LFO 1 - a sawtooth in Env mode with plenty of positive contour. Here's a little experiment. Go to C127 (START) and set up the following parameters:

Experiment from C127: Diving lesson

```
Flt CUTOFF = 90
Amp SUSTAIN = 127
Mod LFO 1 SHAPE = [Sawtooth]
Mod EDIT / LFO 1 / EnvMode = on
Mod LFO1 / OSC1 = 44
```

Play your keyboard. This is probably what you had expected - a simple linear drop in pitch. Now try this:

```
Mod EDIT / LFO 1 / Contour = +50
```

If you experiment a while with different Contour values, you will notice that Contour affects the concavity/convexity of the downward slope in a sawtooth wave. Now try out all the other LFO waveforms at various rates. Look in the manual to find out what the Contour (both positive and negative) parameter does to each of these waveforms. Or see "Non-Standard LFO shapes" in chapter 11, page 91.

This is a lush polyphonic Hoover sound highly suitable for Dance to Trance rhythmic chords. Play around with it for a while, then take a look in the modulation matrix: The source of ASSIGN 1 is Data and the destination is EqMidGain. Press SINGLE to exit the MOD menu. As the text above SOFT KNOB 1 reads “EqMid”, you can safely assume that it has been defined to send MIDI continuous controller (CC) number 6 i.e. Data Entry.

You’re right, let’s not assume anything! Press the main EDIT button and scroll through the menu with the PART buttons until you see SOFTKNOB 1 in the display. Press the right-hand PARAMETER button once to check that our assumption was correct. Although SOFT KNOB 1 could have controlled EqMidGain directly, it has been set to an unused MIDI CC (i.e. Data). In the modulation matrix, Data is the source which modulates EqMidGain. This indirect method has been used to restrict the range (check this - the Amount is only +11).

Above SOFT KNOB 2 it says ‘Filter’, and turning this up makes the sound darker (try this). If you have moved the mod wheel, you will have noticed that this has exactly the same effect. Now guess what values the SOFT KNOB 2 and ASSIGN 2 parameters have!





5: Digital Synthesizers

Digital synthesis did not begin with the famous Yamaha DX7, although there is a direct lineage from the first ever digital synthesizer - an IBM 704 vacuum-tube computer: In 1957 an engineer at Bell Labs called Max Matthews wrote a program called MUSIC 1 for this computer. Several years later, while collaborating on a later version of the MUSIC program at Stanford University, John Chowning started experimenting with audio frequency 'vibrato' and thus FM synthesis was born. Mathews suggested to Chowning that he should try to emulate real-life instruments to show that FM could be a commercially viable alternative to subtractive synthesis - the rest is history!

The First Digital Synthesizers

The New England Digital Synclavier was way ahead of its time. Originally conceived in the mid to late 70's, there are basically three models. Released in 1980, the Synclavier II (pictured above) was by far the most common. It specialised in additive synthesis and 12-operator FM - four years before the DX7. A 64-voice system with on-board sequencer, arpeggiator, sample RAM expandable to 768MB and 16-track digital recording, it was many times more expensive than modern samplers (well over \$20,000). Before putting his stamp on the New Music (modern orchestral) scene shortly before his untimely death, Frank Zappa made extensive use of his Synclavier system. Other famous Synclavier owners are George Duke, Pat Metheny, Sting, Michael Jackson, Pink Floyd, Kraftwerk and Genesis.

The PPG Wave was certainly a breath of fresh air when it first appeared. With traditional analogue envelopes and filters (unlike an earlier product, the Wave Computer 360), sweepable wavetables and user-samples instead of analogue oscillators, these revolutionary instruments caused other manufacturers to think very hard about developing their own hybrid synths. Built towards the end of the age of "knobbage", it was relatively easy to edit sounds on the Wave - which is more than can be said of the DX7 (see below) and its immediate suc-

**NED Synclavier II
(1980)**



**PPG Wave 2.2
(1982)**



cessors. The deep blue colour was a novelty at the time, but the tendency towards strong or even garish colours is still being held high by PPG's heirs to wavetable technology, Waldorf.

OSC Oscar (1983)



The Oscar was designed by one Chris Huggett, who later joined forces with David Cockerell (of EMS and Electro-Harmonix fame) on the Akai sampler team for ten years. Mr. Huggett now works for Novation.

Another eccentric UK product like the EMS instruments, this almost-duophonic synthesizer's quirky appearance belied its powers. The thick rubber end-pieces and section dividers certainly didn't appeal to prospective synthesizer buyers at the time. Needless to say the Oscar flopped, only to achieve cult-status quite recently with Dance/Trance producers - alongside the TB303 and Alpha Juno.

The Oscar's great strength lies in its ability to deliver truly *different* sounds. With its dual multi-mode filter, step sequencer and additive synthesis (the levels of 24 harmonics are set by holding down keys on the keyboard), it was unique. Now often dubbed the last great monophonic synthesizer (although my candidate for this honour would be the Yamaha VL7 virtual acoustic synth), the only thing analogue about it is the filter section - everything else in the Oscar is digital.

Yamaha DX7 (1984)



The instrument which brought affordable digital synthesis to the masses and thus changed the face (literally) of synthesizers for almost two decades was the Yamaha DX7. It had a radically different method of synthesis called FM (frequency modulation). The potential of FM was discovered around 1967 by John Chowning of Stanford University, who licensed the idea to Yamaha. The basic idea behind FM is this: if you modulate the frequency of a sine wave with another (also within the audio range), you can create rich harmonics which would be almost impossible to achieve using traditional additive or subtractive synthesis. The timbre depends on the frequency ratio between the carrier and modulator as well as on the current level of modulation, so the results of even 2-oscillator FM can be difficult to predict. If you then take six of these 'operators' arranged in complex 'algorithms' then add a pressure-sensitive keyboard and optional breath controller, you end up with enough synthesizer power to outsell all the competition.

And outsell they certainly did! The sheer expressiveness of the DX7 took the synthesizer world by storm and very soon it became the biggest selling synthesizer. Shortly after its introduction, many heavyweight companies, including

Moog, were forced to throw in the towel. Others such as Roland reduced the knobbage on their new instruments to such an extent that the job of programming a post-DX7 synthesizer was more like repainting your house through the letterbox than synth heaven. However, this didn't really matter to most buyers who recognised this new breed of synthesizers for what they were - more flexible alternatives to organs and pianos. Luckily, times have changed and there is now plenty of room in the market for knobby instruments like the Virus.

Comparable models: Many Yamaha FM synths, Native Instruments FM-7 (an extended software emulation of the original DX7).





6: Programming Classic Digital Sounds

Digital synthesizers in general are seldom called ‘classic’, but when a software company (or two) actually recreate and market the PPG Wave and DX7 as plug-ins for your computer, this label is certainly justified.

FM Pianos

This is a typical DX7 II ‘Rhodes’ including the inimitable (?) ‘ting’. The chosen FM mode is very important for this patch - it is not the default PosTri (which would give you pitch changes), but Tri.

Press the EDIT button in the OSCILLATORS section and play around with the OSCILLATOR 2 FmMode parameter for a while...then restore the original by stepping up and down a patch. And don’t forget to remember what you have learned...

The basic sound (without the ‘ting’) is just two sine wave oscillators with FM (try turning FM AMOUNT up) and some ring modulation. Play your keyboard (which is hopefully velocity-sensitive) as dynamically as possible. Three parameters are being modulated by Velocity in ASSIGN 3: LFO 1AssAmt, FmEnvAmt, FltDecay. LFO 1AssAmt means the amount of whatever the destination of LFO 1 ASSIGN is.

The LFO 1 ASSIGN destination is OSC 3 Volume and LFO 1 is in Env mode (press EDIT in the LFOS/MOD section to check). What do we know already? That LFO 1 is an additional envelope used for the short ‘ting’ sound, and that its amplitude is controlled by Velocity. The nominal volume of Oscillator 3 is 0 (press EDIT in the OSCILLATORS section to check this) so that it will decay right down to zero when LFO 1 has finished its cycle. LFO 1 Contour is set to -64 i.e. extremely concave.

The DX7 in your Virus

PATCH: DX ROAD HS



Try to reserve an LFO or even two for use as additional envelope(s), especially if you are making percussive sounds.

Time to experiment with the “ting”: To isolate oscillator 3, turn OSC BAL down to minimum, select oscillator 1 and turn SHAPE up to maximum. Play around with LFO 1 RATE for a while, then try out different LFO shapes and Contour values. Hmm - should be able to get a good triangle (the percussion instrument, not the waveform) out of this one!

The oscillator 3 SEMITONE value is only +21, which is an octave and a major sixth. However, wave 14 has very prominent 1st and 2nd harmonics (that’s why it is so useful for organ sounds), so the ‘ting’ sounds high enough while adding some medium range frequencies. If you’re good at hearing intervals, play a low C - the major third you can hear is the 2nd harmonic (i.e. the “fifth of the sixth”).

Is there anything more to say about this patch? Well, LFO 3 gives oscillator 1 a slow subtle vibrato which also affects the FM sound, making it seem as if Cut-off is being modulated slightly. The DX7 doesn’t have any filters, so I didn’t use them either (Filter 1 is LP, CUTOFF is at maximum and Keyfollow is 0).

If you’re feeling clever after all this, you could try your hand at some other DX-7 type sounds, for instance basses. Tip: turn off Osc 3 and play around with OSC 2 SEMITONE first. You should make sure that Unison is off while experimenting.

Wavetable Effects

The PPG Wave in your Virus?

The additional waves in the Virus do not have smooth transitions when you sweep through them - they are primarily intended to be a source of static waveforms. However, there are two modulation destinations (OSC 1WavSel and OSC 2WavSel) which let you modulate the current position within the wavetable. One good use of this feature is for organ key click (see “Hammond Organ” in chapter 7, page 55), but a more obvious one is for that chaotic ‘sparkle’ first heard on the PPG wave and later on the Korg Wavestation.

Play with this sound for a while then look into the ASSIGN 1 parameters. LFO 1 is sweeping through the table with a range of 19 waves although the Amount in ASSIGN 1 is only +9. An experiment:

PATCH: SPARKLE BC

Experiment from C127: Modulating wave numbers

```
Osc OSC 1 SHAPE = 0
Osc OSC 1 WAVE-SEL = 16
Amp SUSTAIN = 127
Fit CUTOFF = 100
Mod LFO SHAPE = [square]
Mod LFO 1 / ASSIGN / Dest = OSC 1WavSel
Mod LFO 1 / ASSIGN / Amount = 0
```

Now manually move WAVE SEL up to 17 and down through 16 to 15, getting to know the sound of all three waves (I chose these because they are obviously different). Set WAVE SEL to 16 again and LFO 1 ASSIGN Amount to +1. You should now hear only waves 15 and 17. Setting LFO 1 ASSIGN Amount to +2 gives you waves 14 and 18 (check this manually with zero Amount). What have we learned? That with a bipolar LFO source you will sweep through ASSIGN Amount * 2 + 1 waves. Nice to know this kind of thing whenever you are using LFOs to modulate tables e.g. the arpeggios (see 2 Gtrs CS in the next chapter and “Modulating Tables” in chapter 11, page 91).





7: Electromechanical Instruments

Rhodes Piano

This was the instrument of choice for countless Jazz and Soul players such as George Duke, Herbie Hancock, Stevie Wonder, Joe Sample, Joe Zawinul etc.

Harold Rhodes founded the Rhodes Piano Corporation in the 1940s and sold it to Fender in the late fifties. Although the name “Fender Rhodes” was retained until the mid 1970s, Leo Fender had already sold the company to CBS in 1965.



A good emulation of an early, punchy Stage Piano (the ‘portable’ one with metal legs) with oodles of 1st harmonic but not much bell tone. I have taken the liberty of changing a few values: I turned off oscillator 3 because it reduced polyphony without making much difference to the sound. The LFO 2 destination in the original was SHAPE, but as SHAPE is already at maximum for both oscillators, and LFO 2 in Env mode is a positive-only (unipolar) source, I sacrificed this for the sake of some panning. Turn up the mod wheel or SOFT KNOB 1.

PATCH: EPSTAGE?M@

The punch comes from a relatively high ENV AMOUNT in combination with a very short filter DECAY. The extra decay is dependant on the filter Time parameter (go into the filter EDIT menu and try changing the value). Both oscillators are fairly narrow pulse waves (tweak the PW knob to check this) which gives the filters plenty of high frequencies to work with.

Pulse width is modulated by LFO 1 in Env mode with a negative amount - the oscillator sound is initially duller, then ‘decays’ up to the narrower nominal PW value(s). As the filter envelope is doing the opposite (causing the filter to close at about the same rate), the sound quality changes over time in a complex but subtle manner.

I already mentioned the strong 1st harmonic (an octave above the fundamental). This is a neat by-product of SYNC: Turn OSC BAL to maximum, select OSC 2 and move the SEMITONE knob. From +5 to +9 you get a strong 1st har-

monic. Around +15 you should be able to hear a good 2nd harmonic (an octave and a fifth above the fundamental). You might want to take CUTOFF up a little to hear the harmonics better.

Step up and down a patch to defeat any changes you have made. Turn up SOFT KNOB 1. The panning effect is not round, it is more like a soft square although the selected LFO 2 SHAPE is a sinewave. Select LFO 2 and press EDIT. LFO 2 Contour is set to +45, which transforms the sine wave into a rounded-off square. Try maximising Contour to see just how square you can get! By the way: Contour values below 0 make sine waves more triangular. Check that.

Select LFO 2 and then press the lower SELECT button to see the LFO 2 Panorama value. I set this to +1 for the sole purpose of lighting up the LED so that I don't get the impression that LFO 2 is free! This nominal value is increased by ASSIGN 2 in the modulation matrix (take a peek there). The ASSIGN 1 source is Contr3, which is also the destination of SOFT KNOB 2. This controls LFO 2Rate (i.e. panning speed) within a fairly narrow range. Although the Amount here is only +10, it is quite enough to make panning a lot faster when you turn up SOFT KNOB 2.

As a challenge, see if you can use oscillator 3 to add some 'bell tone' - you might have to turn off SYNC and adjust OSC 2 SEMITONE accordingly. LFO 1 is already in Env mode, so you could use this to make the bell tone shorter than the other oscillators. Or you could sacrifice LFO 2 for this purpose instead. Refer to the FM Piano section (above) for a few tips which could point you in the right direction.

Wurlitzer Piano



After bringing out several wooden-cased models for the home market, the Wurlitzer EP200 electric piano was the first 'Wurly' built especially for the stage. It uses flat metal tines without resonators, so it doesn't have as much sustain as the Rhodes. Pros: The EP200 is much more portable and has a lighter action. Wurlitzer pianos became popular with Rock bands whereas Jazz groups tended towards the Rhodes. I bought a Wurly around 1977 to put my ARP Axse on. Aahh Jazzrock (sigh)...

Most synthetic Wurlitzer piano sounds I've heard are based on pulse waves. This one is different (it uses wave 13), and in my opinion more realistic and very dynamic. Surprisingly, if you turn CUTOFF up to maximum, EPWhirlyM@ sounds like a cheap and nasty digital synth e.g. from a certain Japanese watch and digital metronome manufacturer beginning with 'S' - so the filter settings are very important here!

PATCH: EPWHIRLYM@

Yet another good Matt Picone piano, EPZeply M@ takes me right back to the early 70s. Based on a very simple Wurlitzer-type sound, what makes this one so special is the chorus-vibrato effect similar to a Leslie cabinet. Select CHO in the EFFECTS section and look at the chorus settings by pressing EDIT. Rate is very fast (112), the nominal delay is very short (4) and there is quite a lot of negative Feedback (-34). Chorus MIX is at maximum, This is unusual - the normal setting for maximum chorus is 64 i.e. half of the signal is passed untreated, causing complex phasing effects. With Chorus MIX at maximum you get a lot more vibrato (see chapter 12, page 99).

PATCH: EPZEPLY M@

At the risk of digressing from the Wurlitzer theme, let's apply the same effect to E-Grand HS and turn it into the typical Leslie-piano sound from Pink Floyd's "Echoes" (Meddle, 1971). Everyone should be familiar with this track - it has accompanied countless nature documentaries:

PATCH: PHLOID HS

Walkthrough from E-Grand HS to Phloid HS

Amp RELEASE = 60
Filt CUTOFF = 50
Eff CHO EDIT / CHORUS / Rate = 112
Eff CHO EDIT / CHORUS / Delay = 4
Eff CHO EDIT / CHORUS / Feedback = -34
Eff MIX = 127
Eff INTENSITY = 127
Del EDIT / Color = -42
Del SEND = 80
Del REV-TIME = 100

Sit in the lotus position and put your hands together like the "Delay Lama". Suddenly play a staccato high B with your index finger, put your hands back together immediately and wait for 6 seconds. Repeat for about 20 minutes...

Hohner Clavinet



The Clavinet D6 was the most successful (and certainly the funkiest) keyboard ever produced by Hohner, a German company famous for their harmonicas. Unlike their earlier keyboard products such as the Pianet, the Clavinet is velocity-sensitive and has a very brilliant tone (the D6 could really 'cut it' on stage). Although it doesn't sound quite as silvery as a harpsichord, it also has a distinctive click when notes are released. Suggested listening: of course there's "Superstition" by Stevie Wonder, but let's not forget about all those Reggae songs by Peter Tosh or Bob Marley!

PATCH: CLAVKEY CS

This is about the best clone Ben Crosland and I could manage after sending several versions back and forth. ClavKey is too complicated to analyse in depth so I'm just going to concentrate on two aspects here: Digital saturation and release-click. Get funky widdit for a while - and try out the wah-wah effect!

Filter saturation is responsible for a great deal of the sound - turn OSC VOL down to 0 (not -64) to check this. Time for a little experiment to show how little signal you actually need to get clipping from Digital saturation: Go to C127 and turn the following up to maximum: SHAPE, WAVE SEL/PW (to mute oscillator 1) and OSC VOL (for maximum saturation). Select OSC 2 and turn both SHAPE and WAVE SEL/PW down to minimum (for a sine wave). Play a note while fading oscillator 2 in and out via OSC BAL. Now press EDIT in the filter section and change the SATURATION curve to Digital. Again, play a note and fade in oscillator 2.

Experiment from C127: Digital clipping

```
Osc OSC 1 SHAPE = 127
Osc OSC 1 PW = 127
Mix OSC-VOL = +63
Osc OSC 2 SHAPE = 0
Osc OSC 2 WAVE-SEL = Sin
(play with OSC BAL for a while)
Fit EDIT / SATURATION Curve = Digital
```

Back to ClavKey. You should be able to hear a slight click when you release a key. This is because the filter closes before the amplifier does - turn Amplifier RELEASE up to about 25 to hear this better. See Harpsie HS in chapter 8 (page 63) for more about this effect.

Hammond Organ

Weighing in at rather less than a Hammond B3 plus Leslie cabinet, the Virus is nevertheless capable of some very convincing organ sounds

This started out as an attempt to clone the B3 (or was it a C3?) in “Sweet Child in Time” (Deep Purple in Rock, 1970), but I added some optional 8’ which you can turn down with SOFT KNOB 1. The method I used to get a convincing key-click is interesting: Press the lower SELECT button in the LFOS/MOD section. The ASSIGN destination for LFO 1 (in Env mode) is OSC 1WavSel and the Amount is +32. LFO 1 is a square wave and RATE is at maximum.



This means that, for a very short time, oscillator 1 is not a sine wave but wave 64. Why isn't it wave 32, seeing as the Amount is only +32? Modulating tables (wave or arpeggio number) is tricky - see below and in chapter 11 (page 91).

PATCH: LORDORG HS

The percussion is oscillator 2 - turn down SOFT KNOB 1 to hear this more clearly. OSC 2 SEMITONE is +19, which is the 2nd harmonic. The filter envelope is being used in the modulation matrix (see ASSIGN 1) to affect OSC BAL, which is nominally -64 i.e. oscillator 1 only. If you like you could experiment with shorter or longer percussion by tweaking Filter DECAY.

Restore the original patch by stepping up and down with the VALUE buttons. the 8’ drawbar is the Ring Modulator signal only - a good opportunity for another experiment:

Experiment from C127: Ring Mod as 4th oscillator

```
Mix OSC-BAL = 0
Osc OSC 1 SHAPE = 0
Osc OSC 2 SHAPE = 0
Osc OSC 2 SEMITONE = +19
Osc OSC 2 DETUNE = 0
(play the keyboard for a while)
Mix OSC-VOL = -64
Mix RING-MOD = 64
```

What have we learned from this? That the ring modulator can be used as a fourth oscillator if the pitch of the sources (oscillators 1 and 2) are favourable - which they are in this case. Now try different SEMITONE values for oscillator 2.

The rotary (Leslie) effect is a combination of Phaser, Chorus and panning plus a little filter and volume modulation. Select ASSIGN 2 in the modulation matrix. Destination 1 is RingMod, so all this does is pull the 8' drawbar out a bit. Destination 2 is Phaser Freq. I didn't modulate PhaserRate (which would have been more obvious) here because I just wanted the Phaser to add a bit of psychedelic 'whoosh', but only when the speed changes. This is something I had noticed in the sound of real Leslies - I love it when they ramp up and down!

The ASSIGN 3 source is ModWheel and destination 1 is ChorusRate, fine. Destination 2 is ChorusDpth with a negative Amount (-15) because otherwise it would sound either too wobbly when fast or too subtle when slow. LFO 2 is mainly responsible for panning, also a major aspect of rotary effects. So now you know: it is possible to cram a Leslie cabinet into your Virus next to the Hammond - although it's a fairly tight fit.

Clean Electric Guitar

This patch sounds as if it uses some of the additional waves. However, if you tweak WAVE SEL/PW for both main oscillators, you will see that this just isn't the case - they are both set to pure sawtooth. All the 'thinness' comes from the ring modulator (OSC VOL is at minimum) with a little help from FM. Another little experiment:

PATCH: FENDER!M@

Experiment from C127: Thinness

Amp DECAY = 64
(listen to the sound for a while)
Mix OSC-VOL = -64
Mix RING-MOD = 127
Osc SYNC (on)
Osc EDIT / OSCILLATOR 2 / FmMode = Tri
Osc FM-AMOUNT = 75

Note that the ring modulator gets half of its input from oscillator 2, whatever the value of OSC BAL is (-64 here).

PATCH: 2 GUITARS CS

A great automatic rhythm guitarist patch from Ben Crosland (with a few tweaks from yours truly). Play sustained chords for a while using pitch bender, modulation wheel and (especially) the sustain pedal.

Open the ARP menu. The arpeggiator mode is Chord (which of course plays all notes simultaneously). The arpeggio pattern number is 7, but this can be changed by stepping on your sustain pedal. Looking in ASSIGN 3, you will see that HoldPedal is modulating ArpPattern with Amount +4, so the other pattern should be number 11. For some reason it isn't, it's actually pattern number 14. Check this by changing the Arpeggiator Pattern parameter directly. If you want to find out what is happening here, see chapter 11, page 91, under "Modulating Tables".

Distorted Guitar

Similar to a power-chord, this uses Hard saturation in the Filter section as well as Bit Reduction in the Effects section. If you push up the modulation wheel on your keyboard, it becomes less harsh without sounding as if you are simply closing a lowpass filter: It is EQUALIZER MID i.e. a single bandpass filter over the entire sound. Look in ASSIGN 1.

PATCH: WILDTHNGHS



As well as being good for static formants, Equaliser mid frequency is a very useful modulation destination. Try programming a harsh sound and then use EQ to add a wah-wah effect on top.

I have included this patch in the tutorial bank as an example of how far you can go trying to clone one of the most complex sounds (sound-genres?) in existence - lead guitar. HevyHevyM@ is far from being a perfect clone - if you manage to improve on the original, I would be very happy to hear your results!

PATCH: HEVYHEVYM@



8: Acoustic Instruments

Attempting to clone acoustic instruments on a subtractive synth is an academic exercise - you might as well use a quality sampler or, better still, the real thing. I suppose the same also applies to electromechanical instruments.

You're probably asking yourself what this chapter (and the previous one) is good for? Well, seeing as the sounds of acoustic instruments are generally well-known although relatively complex, it makes a great deal of sense to use them for sound-design target practice. Besides, once you have a good clone you can easily turn it into a dynamic 'fantasy' sound without losing complexity.

Plucked Strings

This is one of my personal favourites because it is simple enough to serve as a template for countless related sounds. I not only based NylSolo HS on this patch, but also all my pianos. Play low notes to see why.

PATCH: NYLON BC

Let's head straight for the modulation matrix: ASSIGN 1 is used for vibrato (ModWheel controls LFO 3OscAmt), ASSIGN 2 shortens amplifier attack as well as opening Filter 2 a little when notes are played harder, and ASSIGN 3 balances out the volume across the keyboard.

Checking OSC BAL shows us that only oscillator 1 is being used - one reason why Nylon BC sounds like single strings. The oscillator certainly doesn't sound like a sawtooth or pulse wave, so tweak the WAVE SEL knob to find out. Yes, it's one of the extra waves (5). Temporarily turn SHAPE up to 64 (pure sawtooth) so that we can hear the effect of the filters better.



To test what the filters are really doing, defeat any effects and use a harmonically rich sound source (either sawtooth or noise).

Restore Nylon BC by stepping up and down a patch. The soft ‘pluck’ and some of the dynamics are due to meticulous work on the envelope settings: Amplifier ATTACK is nominally quite slow (40), but we already know that this parameter is being modulated by Velocity (see ASSIGN 2). Velocity is also being used to modulate Filt1EnvAmt and Filt2EnvAmt (press the main EDIT button and check the Velocity parameters) so the sound gets brighter when you play harder. Both DECAYS are quite short - the sound only disappears slowly because of the Time parameters - look in the Filter EDIT menu for these. Now let’s see whether this sound can be turned into something like a mandolin.

The following is a complete walkthrough from Nylon BC to a fairly complex mandolin sound, pulling out all the stops. Although there is a complete list at the end of this section, you should follow the text first.

PATCH: MANDOID HS

Restore Nylon BC and turn the reverb right down so that we can hear everything clearly. Mandolins have two strings for each note, so turn OSC BAL up to 0. Oscillator 2 is obviously a few octaves lower than it should be - my guess is that Ben Crosland (Mr. BC) originally wanted to use it for a short acoustic guitar ‘thump’ by modulating OSC BAL with LFO 1, but later decided it sounded better without. Select OSC 2 and turn SEMITONE up to -12 so it has the same pitch as oscillator 1. The wave is pure square (check this), which seems to complement wave 5 quite well. OK, it could do with just a little more bite, so turn PW (pulse width) up to 10.

Take DETUNE up to 60 (double strings are always slightly out of tune). Steel strings are brighter than nylon, so turn CUTOFF up - 80 sounds about right. Mandolins decay even faster than nylon guitars, so use ASSIGN 4 to make the secondary decay (i.e. the Time parameter) a tad more concave: Source = AmpEnv, Dest = AmpSusTime, Amount = -5.

If you now play G, D, A and E (consecutive fifths like open mandolin strings) in very quick succession, you can tell that we are already getting somewhere! However, there are still a few important factors missing: oscillator 2 should be plucked slightly later than oscillator 1, and we haven’t tried simulating the resonance in a small wooden box with a hole in the middle yet!

How to ‘delay’ one oscillator only...

First problem: How on earth can we get oscillator 2 to come in slightly later? We could “cheat” and add a short delay, but that would sound too static, too obviously tacked on. Besides, we might need some reverb later. After thinking about this for a minute I came up with the following simple solution: A pulse wave LFO in Env mode could modulate OSC BAL so that it starts at -64 then, a split second later, returns immediately to 0.

When you have understood the last sentence, set the following parameters: LFO 1 is already in Env mode, so we'll use that. Set LFO 1 SHAPE to the square wave. Keep pressing the lower SELECT button until the bottom LED (ASSIGN) lights up. Set Dest to OscBalance and Amount to -64. Play a key. Hah, it works! The ‘delay’ is too long, so turn LFO 1 RATE up slowly until you can hear very short ‘flams’ - 108 is about right. The short square wave from LFO 1 could also be used to add a bit of squeak to the attack: Select the destination RESO 1+2 and take the Amount up to +33.

It would be nice to randomly detune one of the strings for even more realism. Select ASSIGN 5 and set Source = Random, Dest = OSC 2Detune and Amount +14.

We've only got ASSIGN 6 left, but why not make good use of it? As mandolins stopped at high frets sound very short, this final slot could be used to shorten the decay as you move further up the keyboard: Select ASSIGN 6 and set Source = Keyflw, Dest = FltSusTime and Amount = -7.

Next on the agenda - formants. The Phaser can be (mis)used to give acoustic instrument clones some extra realism (see chapter 12 under “Phaser”, page 99): Press the EFFECTS EDIT button, set phaser frequency to 64, Depth to 0, Stages to 1. Exit the menu by pressing SINGLE. Turn the MIX up to 50 and INTENSITY (Feedback) to -40. Couldn't it do with a bit more volume and some EQ? Press the main EDIT button and take COMMON / PatchVolume up to 110. Press Effects EDIT and change EQUALIZER / MID Gain to +5 and Frequency to 5012.

Et voilà!

Walkthrough from Nylon BC to Mandoid HS

```
Del SEND = 0
Mix OSC-BAL = 0
Osc OSC2 SEMITONE = -12
Osc OSC2 PW = 10
Osc DETUNE = 60
Flt CUTOFF = 80
Mod ASSIGN4 / Source = AmpEnv
Mod ASSIGN4 / Dest = AmpSusTime
Mod ASSIGN4 / Amount = -5
Mod LFO1 SHAPE = [square]
Mod LFO1 ASSIGN / Dest = OscBalance
Mod LFO1 ASSIGN / Amount = -64
Mod LFO1 RATE = 108
Mod LFO1 RESO1+2 Amount = +3
Mod ASSIGN5 / Source = Random
Mod ASSIGN5 / Dest = Osc2Detune
Mod ASSIGN5 / Amount = +14
Mod ASSIGN6 / Source = Keyflw
Mod ASSIGN6 / Dest = FltSusTime
Mod ASSIGN6 / Amount = -7
Eff PHA EDIT / Frequency = 64
Eff PHA EDIT / Depth = 0
Eff PHA EDIT / Stages = 1
Eff MIX = 50
Eff INTENSITY = -40
Gen EDIT / COMMON / PatchVolume = 110
Eff EDIT / EQUALIZER MID / Gain = +5
Eff EDIT / EQUALIZER MID / Freq (Hz) = 5012
```

To make this patch sound like a real mandolin, you have to play it like one. Start off by strumming G, D, A, E i.e. the open strings, then try your hand at some traditional Irish jigs and reels.

PATCH: K:GEET-R

This is a very acoustic-sounding plucked string patch which has been around for as long as I can remember owning a Virus. It demonstrates the classic method of getting a click when releasing notes: Go into the Filter EDIT menu and scroll through the parameters until you see FILTER 1 / EnvPolarity. This is set to Neg, inverting the effect that ENV AMOUNT has on filter 1 frequency. ATTACK becomes a decay and, most importantly, RELEASE opens the filter instead of closing it. Filter RELEASE has been set to zero (thus opening the filter immediately) whereas Amplifier RELEASE is +6 - long enough to hear the release-click. More about release click when we get to the next sound.

In the back of your mind you might have the idea that the filter envelope itself is inverted, although there are two EnvPolarity parameters and only one filter envelope. A little experiment makes this clear:

Experiment from C127: Filter envelope polarity

```
Fit EDIT / FILTER 1 EnvPolarity = Neg
Fit ATTACK = 64
Mod ASSIGN1 / Source = FiltEnv
Mod ASSIGN1 / Dest = Osc1Pitch
Mod ASSIGN1 / Amount = +20
```

If you now play a note, the pitch rises and falls as if you hadn't changed polarity. Therefore, only the way that the filter envelope is interpreted by the filter(s) is inverted, not the filter envelope in general.

I had already made several unsuccessful attempts before hitting on the obvious method of getting a good silvery harpsichord tone - not using the filter at all (well, hardly at all) but still using the filter envelope for release-click. The initial pluck is LFO1 in Env mode modulating oscillator 1 pitch. Although OSC BAL is at +63 you can still hear oscillator 1 because LFO2 is modulating oscillator balance (look in LFO2s dedicated ASSIGN slot).

PATCH: HARPSIE HS

Filter RELEASE is at minimum, and the filter envelope is being used (see ASSIGN 1) to suppress the nominally high amount of FM (see FM AMOUNT). An experiment:

Experiment from C127: Release-click

```
Fit EDIT / FILTER 1 / EnvPolarity = Neg
Fit ENV-AMOUNT = 127
Fit RELEASE = 0
Fit SUSTAIN = 127
Amp RELEASE = 20
```

The following experiment is similar but a bit more involved. It throws some light on the strange behaviour of release velocity (VeloOff).

Experiment from C127: Release velocity

```
Mix OSC-BAL = +63  
Osc OSC2 SHAPE = 127  
Filt CUTOFF = 85  
Filt ENV-AMOUNT = 50  
Amp RELEASE = 32  
Mod ASSIGN1 / SOURCE = VeloOff  
Mod ASSIGN1 / Dest = Osc2Pitch  
Mod ASSIGN1 / Amount = +10  
Gen TRANSPOSE = -1
```

Play your keyboard. The initial pitch is velocity-dependant as if VeloOff also includes VeloOn, something I didn't expect. You can easily prove that it's not the release velocity of the previous note affecting normal pitch - just play extremely softly and release at different speeds. Of course this method can be used for any destination parameter you like, but if you now switch SYNC on and maybe turn up oscillator 2 SEMITONE a bit...

```
Osc SYNC = [on]  
Osc OSC2 SEMITONE = +7
```

PATCH: SITARS HS

Another plucked string sound from my Virus A days, with a relatively simple structure. The basic sitar sound is a combination of two additional waves (59 and 72) sent through a bandpass filter with a little ENV AMOUNT and plenty of Amplifier RELEASE. The most interesting thing about this patch is the use of the sustain pedal (HoldPed) for oriental-style autobend: Look into the ASSIGN 2 parameters - the Source is HoldPed and the two destinations are Lfo1>Osc1 and LFO1>Osc2 i.e. the pitch of both oscillators.

Restore the original patch. LFO1 is a triangle wave in Env mode (of course) with very negative Contour (-60), a value which almost transforms the triangle into a sawtooth. Almost but not quite - there is still a short upward slope. Play the keyboard and tap the sustain pedal every now and again. I didn't use Transpose as LFO 1 destination because the range is still too high, even with a minimum Amount. Try this:

Experiment from Sitars HS: Modulation ranges

Mod LFO1 ASSIGN / Dest = Transpose
Mod ASSIGN2 / Dest1 = Lfo1AssAmt
Mod ASSIGN2 / Amount1 = +1
Mod ASSIGN2 / Dst2 = Off

Bowed Strings

Where do I start? The raw sound is two extra waves (64 and 54) and the sub oscillator to take away the chill. As you can see, both LFOs are in Env mode: LFO1 makes notes slur slightly, LFO2 adds a burst of FM to the attack (simulating the scratch of a bow). Now look in the mod matrix and try to work out for yourself what is what...

PATCH: CHAMBER HS

Please don't read any further until you have researched the mod matrix!

It's fairly straightforward, despite the full complement of lit-up matrix LEDs: the mod wheel takes out the LFO3 vibrato, Velocity and Keyfollow affect the amp envelope and levels, Aftertouch (ChanPres) causes the volume to swell and adds more vibrato. Did you recognise all those destination parameters?

What actually makes this patch realistic is not so much the techniques used, but the amount of time I spent tweaking each and every parameter into submission. I would listen intently to every little change and compare it with the unedited version until I was absolutely sure that the new version was better. M@ has a better method: Reserve successive memory locations as a "history list" (e.g. A120 - A127) and save to these locations only. Compare all the versions so far, choose the best version, tweak it a bit and save to the location of the worst version. Repeat ad infinitum until you have a 'killer' sound, then save it somewhere else.



Reserve e.g. A120 to A127 as your general sound-designing "history list". This lets you compare several versions, with little risk of losing the best one(s)

Flutes

PATCH: EASYFLUTE

A pure flute-like sound is easy to program on just about any subtractive synth - just take a triangle wave and add some delayed tremolo/vibrato/filter modulation. Use a little distortion if possible. Here's a typical example for the Virus. As always, you should play a note or two after each step:

Walkthrough from C127 to EasyFlute

```
Osc SHAPE = 0
Osc WAVE-SEL = Tri
Filt CUTOFF = 90
Amp ATTACK = 48
Amp SUSTAIN = 127
Gen TRANSPOSE = +1
Filt EDIT / SATURATION Curve = Soft
Filt EDIT / FILTERS / Keyflw = +30 +30
Filt CUTOFF = 66
Mod LFO3 / OSC1+2 = 35
Mod LFO3 EDIT / Fadeln = 42
```

So that was the basic sound - you should be able to do something very similar within minutes on any subtractive synth. Why was Cutoff adjusted after changing the Saturation and KeyFollow parameters? Because Saturation adds high frequencies and the breakpoint for keyboard scaling (KeyflwBase) is set to quite a low note (C1).



You will come across parameter interdependency again and again when you edit patches - that's what makes even relatively simple synthesizers like the ARP Odyssey so powerful!

PATCH: FLUTES HS

This is a more refined version, but its ancestry is clear - it is based on the patch you have just programmed. Turn SEND down to kill the reverb. Especially the short percussion an octave below the fundamental makes this patch more realistic than EasyFlute. Look in ASSIGN3: LFO2 in Env mode is modulating the volume of the sub-oscillator, which is also set to triangle wave. The Amount of modulation here is being affected by a Random (look in the ASSIGN 2 parameters) so that the percussion level is different every time you play a note. I have also tried to simulate breath noise by simply turning NOISE up a bit and taking out all the low frequencies (NOISE Color is at maximum - look in the Oscillators edit menu).

This is a monophonic solo flute sound which features a clarinet-like (2nd harmonic) overblow effect when you push the modulation wheel. Using the pitch bender 'morphs' down an octave as well as making it sound purer. How does this work? The secret is in the Rectifier - time for another little experiment. Go back to EasyFlute and change the Saturation curve to Rectifier. Then play around with OSC VOL or assign ModWheel to OscMainVol (with negative Amount) in the modulation matrix:

PATCH: FLUTOON HS

Experiment from EasyFlute: Morphing octaves

```
Fit EDIT / SATURATION Curve = Rectifier
Mix OSC-VOL = +63
Mod ASSIGN1 / Source = ModWheel
Mod ASSIGN1 / Dest = OscMainVol
Mod ASSIGN1 / Amount = -32
```

Note that this only works properly with relatively pure sources such as sine or triangle (try the others). However, you can get some quite interesting wavetable-type effects by using one of the cleaner additional waves (e.g. wave 14, the 'organ wave').

Flutoon HS brought in the 2nd harmonic by simply cross-fading from oscillator 1 to oscillator 2. However, there is even a way to morph a single oscillator to the 2nd harmonic. Load ShaperMorf and tweak SOFT KNOB 1 to its limits. OK, this doesn't sound very spectacular (I wish there was an audio-specific version of that word!), but the method is interesting and may be useful in future, especially when the filters are in split mode. How it works? A triangle wave is sent through the Shaper (look in the filters EDIT menu), and I've adjusted OSC VOL and ASSIGN 1 Amount so that the SOFT KNOB 1 range starts at the fundamental and finishes at the 2nd harmonic.

PATCH: SHAPERMORF

Saxophones

Saxophones have a characteristic vibrato which affects tone as much as pitch, so I used the filter to simulate this in Saxpet HS:

PATCH: SAXPET HS

Meant to be a cross between a soprano sax and a trumpet (honest guy! See “Naming Patches” in chapter 11). The basic waveform is a narrow pulse, which is usually best for saxophone sounds, plus a lot of rectification. This is what the modulation matrix contents look like - saving you the trouble of looking in there yourself:

List: Saxpet HS mod-matrix assignments

Mod ASSIGN1 / Source = LFO3
Mod ASSIGN1 / Dest = Cutoff
Mod ASSIGN1 / Amount = +1 (to light the LED)

Mod ASSIGN2 / Source = VeloOn
Mod ASSIGN2 / Dest1 = AmpAttack
Mod ASSIGN2 / Amount1 = -20

Mod ASSIGN3 / Source = ModWheel
Mod ASSIGN3 / Dest1 = Lfo3OscAmt (pitch of osc 1)
Mod ASSIGN3 / Amount1 = +34
Mod ASSIGN3 / Dest2 = Cutoff
Mod ASSIGN3 / Amount2 = -7
Mod ASSIGN3 / Dest3 = Ass1Amt (the interesting bit)
Mod ASSIGN3 / Amount3 = -9

ASSIGN 3 has ModWheel as Source, and Dest1 is the normal vibrato because, although you can't see it (no OSC3 LED is lit), LFO3 is routed to oscillator 1 pitch. Maybe I should have set a minimum LFO3 OSC1 Amount just to light up the LED. Too late - it's already in the presets.

ASSIGN 3 destination 3 modulates the Amount of ASSIGN1, so cutoff follows the vibrato whenever the modulation wheel is pushed up. Note that Cutoff moves in the opposite direction from Pitch.



Modulating the Amounts of other matrix slots is a very powerful feature which lets you do things you probably thought were impossible.

PATCH: BARYON HS

The fluffy attack is the interesting part of this patch. If you play high notes, the attack is rather reminiscent of some DX7 saxophone patches. Press the lower SELECT button in the LFOS/MOD section. The nominal value of OSC BAL is -64, but LFO1 in Env mode is causing oscillator balance to start at -40 (+24 -64 = -40, gedditt?) and then 'decay' very quickly back to -64.

Now turn OSC BAL right up and listen to the FM grunge which is being used for the attack. Check the FM AMOUNT. Did you know that the FM signal is always routed together with oscillator 2, whatever the FM mode setting?

Double-Reed Instruments

Narrow double-reed instruments such as the oboe tend to sound as if they have been sent through a bandpass filter (see filter 1 MODE) i.e. they generally don't have as much 'belly' as single reed instruments. Otherwise there are strong similarities - I used only a little less width of pulse than I did for Baryon HS. I think I should have released this patch with reverb already turned up, as it would have sounded more typically orchestral.

PATCH: OBOE HS

LFO1 in Env mode (again) is causing the pitch of oscillator 1 to start a bit lower, a subtle and rapid autobend. This seems to be normal intonation for double reed instruments - especially bagpipes can sound terribly flat when there is not quite enough air pressure. Because of these similarities between the oboe and the bagpipes, it's time for another of those walkthroughs:

Walkthrough from Oboe HS to McNasty HS

```
Mod LFO3 / OSC1+2 = 0
Osc SYNC [off]
Mix OSC-BAL = -25
Osc OSC2 SHAPE = 127
Osc OSC2 PW = 95
Osc EDIT / OSCILLATOR-2 Keyfollow = 0
Osc OSC2 SEMITONE = -5 (the drone is a 4th down)
```

Hold down a middle C with your left hand and play white notes only.

```
Del SEND = 42
Amp RELEASE = 42
Fit CUTOFF = 77 (nice/nasty and harsh!)
Osc OSC2 DETUNE = 32
Mod ASSIGN4 / Source = Random
Mod ASSIGN4 / Dest = Osc2Detune
Mod ASSIGN4 / Amount = +63
```

That should impress all the bonnie wee lassies in the front row! BagginsM@ (go to patch G07) is another set of bagpipes, and has the distinction of being the silliest Virus patch I have come across to date! There are some great modulation tricks in that one.

Horns

PATCH: TRUMPY HS

This one has the typical Mexican trumpet vibrato when you press harder on your (aftertouch-sensitive) keyboard. Over the years I have often seen sawtooth waves used for trumpet sounds, but I think a narrow pulse sounds much more realistic. If you really want a warmer sound, turn up SOFT KNOB 1 (check which parameter this has been assigned to). Two other factors which make this sound more realistic:

1. Both the amplifier and filter envelope have a rapid 'warble' at the beginning of each note. Attack and decay are very short, sustain is zero and there is a very quick positive envelope 'Time' to take the volume and cutoff back to maximum - look in the filters EDIT menu and check out those (unusual) Time values.
2. Rectification with OSC VOL = +28. At this level, the rectifier doesn't have a strong effect in itself, but LFO 3 (also used for vibrato - see ASSIGN 3) is being routed to OscMainVol in ASSIGN 1, and this makes quite a difference. It causes an interesting timbral 'tremolo' in parallel with the normal vibrato. The Amount of modulation is +1 to light up the LED, but this value is being altered by aftertouch (in ASSIGN 3 destination 2). OK, OK - I'll list the mod matrix again:

List: Trumpy HS mod-matrix assignments

Mod ASSIGN1 / Source = LFO3
Mod ASSIGN1 / Dest = OscMainVol
Mod ASSIGN1 / Amount = +1 (to light the LED)

Mod ASSIGN2 / Source = VeloOn
Mod ASSIGN2 / Dest1 = Cutoff2
Mod ASSIGN2 / Amount1 = +20

Mod ASSIGN3 / Source = ChanPres
Mod ASSIGN3 / Dest1 = Lfo3OscAmt
Mod ASSIGN3 / Amount1 = +30
Mod ASSIGN3 / Dest2 = Ass1Amt (to LFO3)
Mod ASSIGN3 / Amount2 = -20
Mod ASSIGN3 / Dest3 = Cutoff2
Mod ASSIGN3 / Amount3 = +5

Mod ASSIGN4 / Source = ModWheel (from SOFT KNOB 1)
Mod ASSIGN4 / Dest = OscBalance
Mod ASSIGN4 / Amount = -64

Did you notice the similarity between this one and Saxpet HS? I had to start somewhere and I didn't have any trumpet sounds already, so I chose a saxophone and made it successively more like trumpet-like. This took about half an hour. Again, you have to play this like a real trumpet if you want to impress the señoritas...

Bells

The classic method of creating bell-like sounds was to send two sine waves through a ring modulator and tweak the frequencies to taste. Bells are also a simple exercise on the DX7 (assuming that your paintbrush fits through the letterbox!), using FM of course. So let's see which one of these Ben used in Bell-Boy BC:

Not only is RING MOD level set to maximum, but FM AMOUNT is quite high too. Turn down RING MOD to see how much of the bell-effect is coming from FM. The sound is now rather dull, but if you also turn FM AMOUNT right down, you can only hear oscillator 2. Try turning only RING MOD back up - now it sounds more like an electronic chime (the store is closing in 5 minutes). Oscillator 2 FmMode is PosTri (positive triangle) - this is the setting which gives you the purest FM signal.

PATCH: BELLBOY BC

Now restore the original sound and play around with both oscillator 2 SEMI-TONE and FM AMOUNT to see what other types of bell you can create. There are some very metallic sounds when the carrier is higher than the modulator (remember that oscillator 1 is always the modulator, and oscillator 2 is the carrier). Finally, some non-advice: the results of FM and ring modulation can be very complex and unpredictable, so trial and error can actually be the best method here.

Double Bass

PATCH: DOUBLEB HS

I think this is my first Virus C patch that uses up all the slots in the mod matrix, so here it is in full:

List: DoubleB HS mod-matrix assignments

```
Mod ASSIGN1 / Source = VeloOn
Mod ASSIGN1 / Dest = AmpAttack (more velocity = faster)
Mod ASSIGN1 / Amount = -30
-----
```

```
Mod ASSIGN2 / Source = ChanPres
Mod ASSIGN2 / Dst1 = Lfo3OscAmt (vibrato)
Mod ASSIGN2 / Amount1 = +25
Mod ASSIGN2 / Dst2 = Lfo3Rate
Mod ASSIGN2 / Amount2 = +2
-----
```

```
Mod ASSIGN3 / Source = Keyflw (high notes have...)
Mod ASSIGN3 / Dst1 = Lfo2Rate (...faster purr)
Mod ASSIGN3 / Amount1 = +24
Mod ASSIGN3 / Dst2 = Lfo2AssAmt (less purr)
Mod ASSIGN3 / Amount2 = -20
Mod ASSIGN3 / Dst3 = AmpDecay (shorter decay)
Mod ASSIGN3 / Amount3 = -32
-----
```

```
Mod ASSIGN4 / Source = Random
Mod ASSIGN4 / Dest = Lfo1>Osc1 (random slur)
Mod ASSIGN4 / Amount = +10
-----
```

```
Mod ASSIGN5 / Source = ChanPres
Mod ASSIGN5 / Dest = Ass4Amt (pressure affects slur)
Mod ASSIGN5 / Amount = +3
-----
```

```
Mod ASSIGN6 / Source = Lfo2
Mod ASSIGN6 / Dest = OscBalance (extra purr)
Mod ASSIGN6 / Amount = -20
```

Might as well list the LFO ASSIGNS as well because these are an integral part of the modulation 'spaghetti':

List: DoubleB HS LFO assignments

Mod LFO1 / ASSIGN Dest = NoiseVol (dirty attack)

Mod LFO1 / ASSIGN Amount = +42

Mod LFO1 / OSC1 = -1 (Slur)

Mod LFO2 / ASSIGN Dest = OscMainVol

Mod LFO2 / ASSIGN Amount = +40 (Purr)

Mod LFO3 OSC1+2 = 1 (Vibrato)

"Purr": If you listen to a jazz tune with a double-bass or fretless bass solo in it, you should be able to hear the sound evolving after it is plucked. I have tried to simulate this effect with a combination of saturation (see ASSIGN3 and LFO2 ASSIGN) and oscillator balance (see ASSIGN 6), both of which use LFO2 as their 'envelope'.

"Slur": Because double basses are fretless, it is almost impossible to play all notes 100% in tune without having to adjust the pitch slightly after playing a note. I tried to simulate this effect by using LFO1 in Env mode to slur notes towards the correct pitch. To make this at all realistic the initial pitch should therefore be slightly different every time - see ASSIGNS 4 and 5. If you can't work out how all these modulations interact right now, just play away - you can always come back to this section later!

Vocal Sounds

I count four completely different types of Virus patches which could be filed under ‘vocal sounds’: speaking voices, singing voices (which both make use of bandpass filters), choirs and Vocoder sounds. Let’s start with some fun-stuff - a speaking voice.

PATCH: PNTSTCHRM@ “Peanuts Teacher” is a clone of Charlie Brown’s completely unintelligible school teacher, and it had me in stitches once I had worked out what the cryptic name actually stood for! There are 3 random sources used here - both LFOs are set to complex waves (60 and 58) and Random is used as a Source in the modulation matrix (ASSIGN 1). These are all used to modulate Cutoff, pitch, resonance and filter gain - there is so much ‘spaghetti’ modulation here (all used to make the sound more human) that I hope you will forgive me for leaving you to unravel this on your own.

PATCH: TUVAWHEELHS This is a single voice similar to Mongolian throat singing (or “Delay Lama”, a freeware VSTi plug-in highly recommended for its graphics alone). Two resonating bandpass filters are used in parallel (look in the Filters EDIT menu), and the two cutoff frequencies are independently controlled by pitchbend and modulation wheel (look in ASSIGNS 2 and 3).

LFO 1 (S+G wave) is being used to randomise vibrato speed (LFO3Rate) slightly so that the voice doesn’t sound too synthetic. LFO3 is also being used so that a stab of Aftertouch can give the voice a vocal ‘kick’ (see ASSIGN 1). Pressing hard for longer than a split second just gives you some unpleasantly deep vibrato. Eee-Aaah-Ohhh...

PATCH: CHOIR2 HS Incapable of packing a 50-piece Gospel choir into my Virus B, I refined an old Virus A classical choir sound instead, and this was included in the Virus presets. The oscillators are extra waves 41 (OSC 3) and 42 (OSC 1+2). I admit these are unlikely candidates, but it’s hard to judge the potential of raw waves! Setting FILTER BALANCE to -15 allows some high frequencies to pass (because Filter 1 is BS). Its cutoff frequency is being modulated by LFO 2 for some subtle phasing. Try all the following steps to hear the basic raw sound:

Breakdown: Choir2 HS

Del SEND = 0 (no delay)
Eff CHO MIX = 0 (no chorus)
Eff DIST INTENSITY = 0 (no extra highpass)
Eff PHA MIX = 0 (no static coloration)
Amp ATTACK = 0
Amp RELEASE = 0
(now defeat all the LFO modulations and RANDOM)
Mod LFO1 / OSC1+2 = +0 +0 (no opposing vibrato)
Mod LFO2 / FILTER1 = +0 (no filter-phasing)
Mod LFO3 / OSC1+2 = 0 (no rapid random pitch)
Mod ASSIGN1 / Source = Off (no random cutoff)
Fit RESONANCE = 0 (might as well defeat this)

Although very static, it still sounds vaguely vocal, and this is only because of the choice of waves in conjunction with the CUTOFF 2 value. Having said this, I usually find selecting the best waves for the job to be a matter of “suck it and see”.

This only took about 10 minutes, starting from Choir2. I really like that phased string machine effect (and the sub octave), so I decided to include this one as a bonus. The EQ settings are very important here: The ear-splitting resonance I found so annoying in Choir2 has been taken out with some quite radical MID cut!

PATCH: CHOIR3 HS

Another early Virus A sound, but not improved in the slightest. I tried to clone the Vocoder sound from the Beastie Boys’ “Intergalactic Planetary”, and I remember this patch being pretty close to the original. If you don’t have much experience with the vocoder, I suggest you read the corresponding section in the Virus manual again.

PATCH: PLANETARHS

List: PlanetarHS vocoder settings

Fit CUTOFF (Center) = 42
Fit RESONANCE (Q-Mod) = 50
Fit ENV-AMOUNT (Spread) = 127
Fit FILTER-BALANCE (VocBal) = +0
Fit CUTOFF2 (Offset) = -8
Fit ATTACK (VocAttk) = 20
Fit DECAY (VocRel) = 50
Fit SUSTAIN (SpecBal) = 64
Fit RELEASE (VocBand) = 32

Send any source of pre-recorded speech to the external inputs and play a note or two on your keyboard. The sound you can hear is a sawtooth plus a little noise so that sibilants become more prominent. You should now try tweaking each and every vocoder parameter, noting the parameter names you see in the display and the effect each one has. I find VocAttk (ATTACK knob) and VocRel (DECAY knob) particularly important for smooth results.

Now let's turn this into a polyphonic sound. Restore the original PlanetarHS and change the following:

Walkthrough from PlanetarHS to PolyVoder

```
Gen EDIT / COMMON / KeyMode = Poly
Gen EDIT / COMMON / Portamento = 0
Gen TRANSPOSE = -1
(play chords)
```

If you have some drum loops or a drum machine handy, try sending some of these to the Virus inputs instead of speech. See “Can't I modulate Vocoder parameters?” in chapter 13, then try modulating these parameters with clocked LFOs and the filter envelope. Set EDIT / CLOCK Tempo to the same BPM as your drum loop beforehand.

Another thing you might like to try is using the Envelope Follower at the same time! Route the filter envelope (FiltEnv) to whatever you like (e.g. Cutoff 2 or Transpose) in the modulation matrix. I'm sure there's a lot of potential in this, although I haven't even tried it myself yet...

9: Drums and Percussion

The Phaselnit Parameter

Before getting into the nitty-gritty of percussion sounds, there is an important common factor worth more than a passing mention: Normally, all Virus oscillators are free-running, which means they won't start at the same point in the waveform each time you play a note. To avoid different-sounding clicks every time you play a note, the oscillators should always be restarted at the same position within the wave. You can do this by setting the Phaselnit parameter (in the Oscillators EDIT menu) to any value except Off.

If you don't feel like following some rather academic experiments right now, skip this section and come back to it later... but you would miss out on a free beer!

A little experiment illustrates what 'Phase' means here.

Experiment from C127: TrigPhase

Fit CUTOFF = 100
Mod LFO1 / OSC1 = +50
(play some notes)
Mod EDIT / LFO1 / TrigPhase = 1 (0°)

All notes now start at the bottom of the wave. I actually expected them to start at the middle (going up), but I suppose it makes more sense the way it is: when LFOs are in Env mode (unipolar!), I wouldn't want them to start anywhere else by default. Let's try some different phases now:

Mod EDIT / LFO1 / TrigPhase = 32 (45°)
(middle of the wave going up)
Mod EDIT / LFO1 / TrigPhase = 64 (90°)
(top of the wave)
Mod EDIT / LFO1 / TrigPhase = 96 (135°)
(middle of the wave going down)
Mod EDIT / LFO1 / TrigPhase = 127 (180°)
(bottom of the wave)

Those 'degrees' are the standard units of phase - I'll come back to that shortly. Now let's see where oscillator 1 starts when we set Phaselnit to '1'. The best way to do this is to use FM with a very low modulator (i.e. oscillator 1):

Experiment from C127: Phaselnit

```
Osc OSC1 SHAPE = 0
Osc OSC1 SEMITONE = -48
Mix OSC-BAL = +63
Osc OSC2 SHAPE = 0
Gen EDIT / PUNCH Intensity = 0
Osc FM-AMOUNT = 127
Osc EDIT / OSCILLATORS Phaselnit = 1
Osc EDIT / OSCILLATOR 1 Keyfollow = 0
Gen TRANPOSE = -2
(osc1 is still too fast to hear the phase properly,
so it needs to be taken even further down...)
Mod EDIT / LFO1 / EnvMode = On
Mod LFO1 SHAPE = [square]
Mod LFO1 RATE = 0
Mod LFO1 / OSC1 = -64
```

The modulator is now slow enough to hear that it always starts at the top of the wave. Try different Phaselnit values (but not 'Off'). Hmm - Phaselnit makes no difference whatsoever to where oscillator 1 (the modulator) starts. However, the click (you are tuned in to Oscillator 2 FM!) sounds different. There is minimum click when Phaselnit is exactly 1, 64 or 127. This suggests that the range is 360° (a complete cycle) and that these values correspond to 0°, 180° and 360° respectively (i.e. wherever the wave crosses zero). Eureka! My scientist heart skips a beat and I'm back in "Smug Mode" again.

One practical use of this discovery is that if you want the hardest click (in addition to PUNCH) for a sine or triangle wave, set Phaselnit to exactly 32 (90°, top of the wave) or 96 (270°, bottom of the wave). Must try that myself, actually...

Walkthrough from C127 to EasyXylo

Mix OSC-BAL = +63
Osc OSC2 SHAPE = 0
Amp DECAY = 32
Amp RELEASE = 32
Fit FILTER-BALANCE = -64
Fit RESONANCE = 127
Fit EDIT / Keyflw = +63 +63
Fit CUTOFF = 47
Gen EDIT / PUNCH Intensity = 127
Osc EDIT / OSCILLATORS Phaselnit = 32

To cut an even longer story shorter, Phaselnit works for oscillators like Trig-Phase does for LFOs, but with double the range (360° as opposed to 180°). Also, instead of setting all oscillators to the same phase, oscillator 1 always starts at 90° while oscillator 2 is shifted forwards and oscillator 3 is shifted backwards.

You can pick up your free beer at the next Musikmesse in Frankfurt - just go to the Access stand on Friday midday and tell me what the phase angle of oscillator 3 at Phaselnit = 32 is!

Some Drum Kit Sounds

I don't think it is worth spending hours on end trying to simulate acoustic drums on the Virus, although I'm sure it's quite possible. Being a virtual analogue synthesizer, the Virus is predestined for all those analogue drum-machine sounds which became so much more popular the second time around. You can program many of these sounds within minutes - after a little practise.

I chose BassDrumCK because it uses the standard method: modulate the frequency of a very low sine wave from any source of a linear downward slope (in this case LFO1 in Env mode). Add some Light saturation - especially important after the pitch reaches it lowest point. Any other 'major' parameters? Yes - as usual for percussion sounds, Phaselnit has been used to make the attack consistent. Finally, oscillator KeyFollow has been set to zero so you don't need to

PATCH: BASSDRUMCK

worry about playing any particular note. For a more distorted bass drum just turn up OSC VOL. Of course you could lengthen or shorten the sound, pitch it higher or add some noise or...

PATCH: SNARE3 BC

In recent years, “snare” have become more of a sound-genre (like “strings”) than any single recognisable sound. Here we only have room to dismantle one of them.



The best way to find out how something works is to take it apart, remembering each step. Then try to reassemble it in the reverse order...

Turn down SEND to defeat the reverb. Turn down NOISE so we can hear what else is going on. Interestingly, there is still quite a lot of noise in the sound. This is not due to any NoiseVol modulation (look in all the ASSIGNS). From FM and/or ring modulation? Turn down FM AMOUNT and RING MOD. Remarkably, the sound is still very noisy although taking FM out of the equation did make a difference. Right, Saturation is the next candidate, seeing as its LED is lit. Press the Filters EDIT button and scroll through the menu using the PART buttons until you see SATURATION. There we have it - bit reduction! Press both Value buttons at the same time to defeat this.

We are now down to the ‘bedrock’. Three oscillators, all set to wave 54 (oscillator 3 is a Slave). Turn oscillator 3 off and take OSC BAL right down to isolate oscillator 1. If you turn CUTOFF down you will see why Ben chose to use a highpass filter! To hear the raw wave better, you could also turn up Amplifier SUSTAIN. The only other important parameter I found was Analog Boost. Check out those unusual values, and then change them to find out how much difference this makes to the sound.

Now put Ben’s snare back together before he finds it in pieces. Save your reconstructed version somewhere else, then compare with the original. Bravo - you have just created a completely new patch! Add your initials to the name.

PATCH: FLATHAT HS
(CONVEX ENVELOPE)

There are some realistic BC cymbals already in (the original) bank A, but I chose to make a quick ‘beat box’ type hi-hat (or is it a snare or shaker?) to show you something with a one-shot convex envelope. Play any note for any length of time, there’s no difference. The sound stays loud before dropping sharply, which implies that decay/release is convex. Look into the ASSIGN 2 parameters to see why: the Source is AmpEnv and the two destinations are

AmpDecay and AmpRelease. The Amounts here are both +25, which turns the decay values (zero) into something longer than just a click. I think you will find “one-shot convex envelopes” very useful in future.

Restore FlatHat HS if you have changed anything. The sound itself is just white noise sent through a highpass filter, with some added treble (EQ) and static formants (see chapter 12 under “Phaser”).



The pitchbend wheel is a very useful source because it is the only bipolar control on most keyboards. Don't take the name too seriously - you can always bend something else.

The Pitchbend wheel has been put to good use in this patch (look in ASSIGN 3). Note that the BendUp and BendDown parameters (in EDIT / COMMON) only apply to the oscillators, the range for other uses is not affected.

I hereby donate a short sleighbell-type sound to the tutorial bank. See if you can lengthen Sleigh HS (in a natural way) by making the amplifier envelope convex, then use velocity (VeloOn) to modulate this. Tweak a few more parameters to taste (e.g. RESONANCE) and when you are finished, save it. Call it “Sleighter” and replace my initials with your own.

PATCH: SLEIGH HS

The same idea as an old one of mine (TomColl HS), but I have to admit that Matt's electronic tom patch is much better. It is very dynamic and sounds more ‘organic’ somehow. Certainly worth picking apart - turn down all the following (some of these are off already, but we are not supposed to know that yet):

PATCH: ETOM2002M@

Breakdown: Etom2002M@ (possible noise sources)

Del SEND = 0 (that was quite a nice reverb)
Mix RING-MOD = 0 (wasn't used anyway)
Mix NOISE = 0 (makes quite a difference)
Osc FM-AMOUNT = 0 (wasn't used anyway)
Eff DIST TYPE = Off (hmm - difficult to say...)

The drop in pitch is caused by LFO1 (in Env mode) modulating both main oscillators. Note that when Env mode is on, you never have to set the TrigPhase parameter to 1. The filter's ‘envelope’ is a combination of normal ENV

AMOUNT (for the initial quick decay) and LFO2 in Env mode (for the main, slow decay). The Filter envelope Time value (Filters EDIT / FILTERENVELOPE / Time) is redundant. Try setting it to 0.

 *While analysing patches you will come across redundant parameters all the time. These are often remnants of previous versions, and can give clues about how the sound originally developed. I have even added a few little 'clues' on purpose so that I can tell who has based their sounds on mine! :-)*

Now go to LFO1 destination OSC1+2 (both LEDs flash) and defeat the pitch drop by pressing both VALUE buttons at the same time. The pitch is now too low to hear the tone properly (both oscillators' SEMITONE values are very low), so take TRANSPOSE up to 0. What we hear now is a 'dirty' attack followed by a constant tone. The keyboard range is smaller than usual (Keyfollow is only +10 for both main oscillators). The big question is: where does that dirty attack come from? We have already defeated FM and RING MOD, haven't we?

 *Even if some parameters appear to be unused, they can still be modulated*

Not quite. Look in ASSIGN3 - the Source is Keyflw, the destination is FmEnvAmount and Amount1 is -37. Set Amount to 0. Have we got rid of FM once and for all now? Nope - 'FmEnvAmount' as destination is a clue. Look in the oscillators EDIT menu and scroll through until you reach the OSCILLATOR 2 parameters. The filter envelope is adding a burst of FM (FitEnv>Fm = +37). Press both VALUE buttons to zero this.

If you scroll one parameter backwards, you should see what appears to be pitch modulation (FitEnv>Ptch = +26). Because SYNC is on, this only results in a bit of sync-sweep - you can listen to the oscillator 2 pitch dropping by switching SYNC off. Now that you have finally hit bedrock with this sound, you could think about building a new house on it. Applause!

Some Percussion Sounds

This is meant to sound like a typical analogue clap, and uses both Delay and Chorus (as a very short delay!) to make the sound more dense. Turn up EqMid (soft knob 2), play it for a while and then defeat the effects by turning down SEND and MIX. The 'rattle' comes from LFO1 (a rapid sawtooth) modulating oscillator 2 frequency. Turn SEND and MIX back up (SEND = 90, MIX = 64). What about those two short delays?

PATCH: CLAPS2 HS

Breakdown: Claps2 HS (delays)

```
Eff CHO MIX = 64
Eff CHO INTENSITY = 0
Eff CHO EDIT / Delay = 54
Eff CHO EDIT / Feedback = -37 (try -60!)
-----
Del EDIT / Mode = Delay
Del SEND = 90
Del DELAY = 21.8
Del FEEDBACK = 7
```

This patch is one of the few percussion sounds where Phaselnit can safely be left off so that the 'clicks' are different for every hit - one reason why I find Congoid HS more useful than a sample. Play with it for a while, then isolate oscillator 1 (and make the tone a little clearer) by changing the following values:

PATCH: CONGOID HS

Breakdown: Congoid HS (choice of wave)

```
Mix OSC-BAL = -64
Del SEND = 0
Eff PHA MIX = 0
Amp ATTACK = 15
Amp DECAY = 64
(there are still 2 tones, so...)
Mix RING-MOD = 0
```

This last parameter turned an almost-conga into a sine wave, which shows how important oscillator 2 (which contributes to the RM signal) is.

```
Mix OSC-BAL = +63 (still some grunge in there...)
Osc FM-AMOUNT = 0 (that got rid of it)
Ft CUTOFF = 127
```

Would you even consider starting with THAT sound to make congas/bongos? I'm pretty sure I wouldn't either - I must have arrived at wave 17 (tweak WAVE SEL to check this) in the middle of developing this patch. Especially after I had turned CUTOFF down.



It is easy to underestimate the potential of the extra waves. If a wave seems unusable, filter it

Experiment from C127: Bad Wave?

Osc SHAPE = 0
Osc WAVE-SEL = 17 (eeek!)
Filt CUTOFF = 0
Filt ENV-AMOUNT = 100 (ahaaah!)
Gen TRANSPOSE = -1

PATCH: VR-78 2 RP

A very brash agogo/cowbell type sound which can cut through any mix. Many notes sound completely different - a few of them have more 'rattle' than pitch (e.g. the highest F on a 5-octave keyboard). Take out the reverb and play away. The only real 'trick' to this sound (simply FM from two sine waves) is that one of the Keyfollow parameters has been set to zero so that the carrier and modulator pitches don't follow each other.

PATCH: SLAPPY1 HS

An automatic percussion sound in the same vein as Voodoo HS or Kitchen HS (both of these are in bank D). The mod wheel adds a pitched bell-like element by increasing amplifier sustain. Of course the LFOs are synchronized to the clock, which is usually the case with such 'automata'. Now it's your turn: Write down a list of all modulation sources (including LFOs), destinations and amounts, then work out what each one is actually doing to the sound. Also try a 'breakdown' by selectively defeating parts of the sound.

10: Think of a Sound...

After all that 'cloning' nonsense you are now proficient enough to create some unheard-of fantasy instruments without resorting to the RANDOM button. Its all down to your expertise and imagination from now on. If you ever ask yourself "How did he/she make THAT sound", you just analyse the patch, OK?

 *Ever had an AMAZING SOUND in your mind's ear? Write down a list of all its properties first, otherwise it will be lost forever. Do not attempt to program it directly - that never works!*

I asked Ben what he was thinking about when he made this sound. He just noticed a certain "squeakiness" while tweaking one of his other patches.

PATCH: SQUEAKERBC

 *Recognise the single most interesting property of a sound, then accentuate this to the max.*

If you are still interested in learning how to capture e.g. the fleeming zoomocity of a well-played Patagarian Squigglephone, then read on...

Actually the 4th in a trilogy of comic sawing sounds, this one sounds very vocal, especially when the mod wheel and pitchbender are both up. Note that downward pitchbending has no effect (why?). You probably noticed that the LFOs are in sync (1/8 and 1/1). The trick behind the basic sound was to find the right combination of two very interdependent parameters: the selected LFO2 wave and its zoom-factor (Contour). I had to scroll back and forth between these two parameters many times before I found something that worked. My advice, which I will take myself next time? Use Sounddiver for jobs like this!

PATCH: PSTYRO3 HS

I have included this one in the tutorial bank because, although the concept is similar to Pstyro3 HS, it is easier to analyse. OK, the real reason I included Handsaw is because I think it's funny...

PATCH: HANDSAW HS

PATCH: DR.WHAT?HS

This is a TeeBee with additional percussion that you can isolate by turning up either the mod wheel or SOFT KNOB 2. Question: How can you get two different sounds from a single patch? Answer: By putting the filters in Split mode!



11: Special Techniques

Non-Standard LFO Shapes

The Contour parameter works differently for each waveform. The effects of Contour (negative, positive) are:

Sine: triangle, squareness

Triangle: falling saw, rising saw

Sawtooth: concave ramp, convex ramp

Square: narrow top, narrow bottom

S+H: unused

S+G: unused

Waves 3 to 64: zoom, unused

Alternatively (or even additionally), LFO shapes can be manipulated via recursive modulation. 'Recursive' means something like "affecting itself". Modulate either LFOxRate or LFOxCont from LFOx:

Experiment from C127: Brown Alert

```
Amp SUSTAIN = 127
Mod LFO1 / OSC1 = -60
Mod EDIT / LFO1 / TrigPhase = 32
Mod EDIT / LFO1 / Contour = -40
Mod LFO1 RATE = 75
Mod LFO1 ASSIGN / Dest = Lfo1Rate
Mod LFO1 ASSIGN / Amount = +63
Mod LFO1 RATE = 100 (compensation)
Del SEND = 32
```

SELF-DESTRUCT SEQUENCE INITIATED - DISINTEGRATION WILL OCCUR IN FIFTY-SEVEN SECONDS. PLEASE RELAX.

Here's a really weird one. Instead of modulating oscillator frequency directly, the parameter LFO1>Osc1 is modulated in LFO1's own ASSIGN slot. The result doesn't resemble a sawtooth at all:

LFO Contour

Recursive modulation 1

Recursive modulation 2

Experiment from C127: Purple Alert

```
Amp SUSTAIN = 127
Gen TRANSPOSE = +1
Mod LFO1 SHAPE = [Sawtooth]
Mod LFO1 RATE = 70
Mod EDIT / LFO1 / TrigPhase = 1
Mod LFO1 / ASSIGN / Dest = Lfo1>Osc1 (!)
Mod LFO1 / ASSIGN / Amount = -64
```

Non-Standard Envelope Shapes

Recursive modulation

I often use this technique for manipulating the shapes of envelope segments (especially useful for the filter envelope). See AT-Mini HS in chapter 3, page 25.

Adding LFO in Env mode

Here's a much simpler yet more flexible method of modifying an envelope curve, but it will cost you an LFO: Set the destination of an LFO (in Env mode) to whatever you are using that particular envelope for, e.g. Cutoff. Then adjust SHAPE, RATE, Contour and TrigInIt to taste - there are literally grillions of possibilities (see above).

Filter Balancing Acts

Would Sir like a 3-pole filter perhaps, Sir?

FILTER BALANCE doesn't actually morph between the two filters, but it can give you similar results. If, for instance, you want a simple 18dB/octave low-pass filter, set filter Routing to Ser4, both MODEs to LP and FILTER BALANCE to -32 or +32. Modulate Cutoff (1) only - the filters will be modulated in parallel and to the same degree if Cutoff2 = +0 and CutoffLink = On.

Attack Consistency

Phaselnit

This is a very important parameter for drum and percussion sounds, especially bass drums. Setting Phaselnit to any value except Off ensures that the 'click' always sounds the same (under normal conditions). You will get minimum click from oscillators 2 or 3 at Phaselnit values of 1 or 64 and maximum at 32 or 96 (this varies slightly depending on oscillator frequency). See the beginning of chapter 9, page 81.

Detune

Detuned sounds which sound fine in the centre of the keyboard often beat too quickly when you play high notes and/or too slowly when you play low notes. You can minimise this effect on the Virus by applying some negative KeyFlw to Osc2Detune.

The minimum amount of pitch modulation in the mod matrix (e.g. using Osc1Pitch or Transpose) is two semitones. For more subtle pitch modulation, you can modulate Osc2Detune or Osc3Detune. Unfortunately, Oscillator 1 doesn't have a detune parameter, so we are forced to resort to some very clever (though I say it myself) indirect recursive jiggery-pokery:

Experiment from C127: Recursive Osc1 Detune

```
Amp SUSTAIN = 127
Mix OSC-BAL = +0
Osc DETUNE = 0 (just for now)
Mod LFO1 SHAPE = [square]
(RATE doesn't matter)
Mod LFO1 / ASSIGN / Dest = Lfo1>Osc1
Mod LFO1 / ASSIGN / Amount = ? (detune)
```

The detune can be modulated using Lfo1AssAmt as destination. OK, now I'll admit I don't have the foggiest notion of how this trick works! Whatever, it could come in very handy for adjusting the overall pitch of highly detuned sounds without resorting to Multi mode. Or for subtle pitchbend via Aftertouch:

Walkthrough from Nylon BC to NylonP CS

```
Mod EDIT / LFO1 EnvMode = Off
Mod LFO1 SHAPE = [square]
Mod LFO1 / ASSIGN / Dest = Lfo1>Osc1
Mod ASSIGN4 / Source = ChanPres
Mod ASSIGN4 / Dest = Lfo1AssAmt
Mod ASSIGN4 / Amount = +6 (adjust to taste)
```

Detune consistency

Detuning oscillator 1

Filter Phasing

Bandstop filters were the only source of phasing in the Virus A, and I still find this trick useful on the C every now and again. Leave the BS filter's resonance at minimum for maximum phasing.

Experiment from C127: Filter phasing

```
Amp SUSTAIN = 127
Mix OSC-VOL = -64
Mix NOISE = 127
Flt FILT2 = BS
Flt EDIT / FILTER2 CutoffLink = Off
(so that 'neutral' cutoff is dead centre)
Mod LFO2 FILTER2 = +50
```

Uses of Indirect Modulation

Modulating ASSIGN Amounts

Think of this as "Modulating the amount that another source modulates its own destination". This sounds strange at first, but the uses are too numerous to mention. That is why there are also quite a few fixed-source destinations available:

FmAmtVel, FmEnvAmt, Lfo1>FltGn, Lfo1>Osc1, Lfo1>Osc2, Lfo1>PlsWd, Lfo1>Reso, Lfo1AssAmt, Lfo2>Cut1, Lfo2>Cut2, Lfo2>Fm, Lfo2>Pan, Lfo2>Shape, Lfo2AssAmt, Lfo3OscAmt, Osc1Keyflw, Osc1ShpVel, Osc2EnvAmt, Osc2Keyflw, Osc2ShpVel, PanVel, PhaserDept, PlsWhdVel, Reso1Vel, Reso2Vel.

For instance, if you modulate FMAmtVel, you are modulating the amount that velocity modulates the amount of FM. If you modulate Lfo1>Osc1, you are modulating the amount that LFO1 modulates oscillator 1 frequency. OK, just one more to confuse you: if you have LFO1 as source in ASSIGN1 and the destination is Ass2Amt1, whose source is Keyfollow and whose destination is Lfo2>Cut1, then...

...LFO1 is modulating the amount that Keyfollow is modulating the amount that LFO2 is modulating Filter 1 cutoff frequency

Not so easy after all, is it?

Destination Soft knob...

Parameters which do not appear in the list of destinations for the SOFT KNOBs can still be controlled by them. Set the SOFT KNOB destination to an unused MIDI CC (e.g. Breath or Contr3 or Data) and set this CC as Source in the mod matrix.

Even if the parameter you want to control is a direct SOFT KNOB destination, using the indirect route (via MIDI CC - see above) is still very useful because the range can be set by adjusting the Amount parameter in the modulation matrix.

The Transpose destination reacts differently, depending on whether it is controlled directly from a SOFT KNOB or indirectly via the mod matrix. See Swoopie HS in chapter 3 under “Analogue Effects”, page 25. There may be other parameters which behave in a similar manner - I haven't looked into this yet.

Soft Knobs: Missing destinations

Soft Knobs: Custom ranges

Soft Knobs: Different behaviour

Filter Offset

The standard value of KeyflwBase (in the Filters EDIT menu) is C1 i.e. two octaves below middle C. If you want C3 to be unaffected by the Keyfollow value, set KeyflwBase to C3 instead. KeyflwBase is also useful for adjusting the Cutoff when the CUTOFF knob is at minimum or maximum (for live tweaking purposes):

Experiment from C127: Minimum CUTOFF

Fit CUTOFF = 0
Fit ENV-AMOUNT = 127
Fit EDIT / Keyflw = +63 +63
Arp EDIT / Hold = On
Arp ARP-ON [On]
(CUTOFF knob doesn't close the filters enough...)
Fit EDIT / KeyflwBase = G8
(...now it does)

Vocoding External Signals

The following is a quote from the Virus users' mailing list (see www.amp-fea.org). See also "Can't I modulate Vocoder parameters?" in chapter 13, page 103:

<Matt>

when using the vocoder, is it possible for me to use the left input as a modulator and the right as a carrier?

<Zack>

you can definitely do this ... i just set this up with my virus A (os 2.8)...in the edit screen (maybe it's effects for those of you with fancy new machines), set:

INPUT Mode = Static

INPUT Select = In L

Vocoder Mode = In R

Modulating Tables

so the INPUT stuff gets inserted where the Oscs usually go in the soundchain, so they are the carrier. since Vocoder Mode is "In R", that becomes the modulator. to test this, i made a file in soundforge with a drumloop in the right channel and a synth sound in the left. my computer is hooked into one of the stereo strips on my Mackie 1202. the "Alt 3+4" out on the back of the mixer is connected to the virus input. when you mute a channel on the mackie, the output for that channel is sent to this "Alt 3+4" output. so i muted the channel strip the computer is connected to so that the sound was routed to the virus input. smart guy, greg mackie.

i did have to play with the vocoder settings to get sound out of it.

You normally don't notice that minimum oscillator pitch modulation (Amount = +1) is usually a whole-tone (instead of a semitone or less). However, when you start modulating tables e.g. arpeggio pattern or wave number, the problems become obvious:

Experiment from C127: Pattern modulation

```
Fit ENV-AMOUNT = 64
Fit CUTOFF = 50
Mod ASSIGN1 / Source = HoldPedal
Mod ASSIGN1 / Dest = ArpPattern
Mod ASSIGN1 / Amount = +1
Arp ARP-ON [on]
```

OK, this works fine - stepping on the sustain pedal changes to pattern 2 (check this by going into the ARP edit menu and changing Pattern to 2). Now do the following:

```
Arp EDIT / Pattern = 1
Mod ASSIGN1 / Amount = 2
```

When you step on the pedal, the pattern changes to number 4 instead of 3!
Again, check this by going into the ARP edit menu and changing Pattern to 4.
Three more tries, and we should be able to work out a formula for future reference:

```
Arp EDIT / Pattern = 1
Mod ASSIGN1 / Amount = 3
(then...)
Arp EDIT / Pattern = 1
Mod ASSIGN1 / Amount = 4
(then...)
Arp EDIT / Pattern = 2
Mod ASSIGN1 / Amount = 3
```

This gives us pattern 6, 8 and 7 respectively when stepping on the pedal. So the formula for unipolar sources (such as HoldPed) modulating tables seems to be:

New Number = Original Number + (Amount * 2) - 1

If you want a bit of a challenge, work out the formula for bipolar sources. Please let me know your answers: mail to hscarr@web.de. I will send a bunch of my latest patches for the most precise, interesting or humorous replies!

Naming Patches

A very important tip for would-be professional sound designers!

Having failed dismally at cloning a grand piano, I renamed my best shot 'E-Grand HS' instead of 'Grand HS' so that everyone would think I actually meant it to sound much worse than a concert Steinway. I had already named a less sophisticated version "DancePno" for similar reasons. Don't go the other extreme and call your very best lead guitar sound 'Guitar4' - call it 'Slashed' or 'Jimdrix'...



12: Getting More out of the Effects

Distortion

Now that the Virus C series has a parametric EQ built in, the distortion types 'LowPass' and 'HiPass' in the Effects section are practically redundant. They have been retained for compatibility with sounds made on the Virus B.

Minimum INTENSITY is often enough to add just a little 'grunge', though you may only notice this when you play chords.

Why those two filter algorithms?

Grunge

Phaser

For a very 'retro' sounding phaser, use only two stages and turn the Spread down - the effect becomes much warmer.

You don't always have to use extreme amounts of Feedback for strong resonant effects. MIX at 64 gives you maximum effect when there is zero Feedback. Once you have more than a little feedback (positive or negative), turning MIX way beyond the usual 64 may be the better option.

Maximum Effect

The phaser is very useful for realistic formants i.e. the complex static coloration you get from hollow-bodied instruments. Go to BerimTamHS (in bank D) and set SOFT KNOB 1 to different positions - you can't do that with the EQ!

Formants

Chorus

Flanging is simply Chorus with feedback. I find that turning the Delay parameter down from its default value (127) to around 24 helps keep the sound from getting too washy. You may have to turn INTENSITY up to compensate. Positive feedback seems to be the better choice for strong flanging effects.

Flanging

Chorus-Vibrato

If you have used up all your LFOs, there is still a way you can get a sort of stereo vibrato using the Chorus. Note that this also colours the sound quite a lot:

Experiment from C127: Chorus vibrato

```
Amp SUSTAIN = 127
Flt CUTOFF = 100
Eff CHO MIX = 127
Eff CHO INTENSITY = 64
Eff CHO / EDIT / Rate = 108 (telephone?)
Eff CHO / EDIT / Delay = 10 (vibratoid)
```

Formants

Like the Phaser, the Chorus can also be useful for static coloration. See MelloVI HS in bank D for an example. I had already used the Phaser for something else - phasing!

EQ

An Integral Part of the Sound

Implementing a parametric EQ in the Virus C series was a simple but brilliantly effective way to make the Virus more versatile than it already was. You could say "I don't need this because I already have quality parametric EQs in my mixer", but I think you would be missing the point(s): Different EQ settings can be stored with each and every patch, and the three mid-parameters are available as modulation destinations...

Modulating Mid Frequency

While trying to make Obifan HS sound closer to a real Oberheim (my Matrix 1000 actually), I tried shifting EQ mid frequency in parallel with the filter. This worked like a dream - the result is a very analogue-sounding 'Polysynth' patch with as much butter in it as a triple portion of Viennese ice-cream. Yum!

Delay/Reverb

Analogue delay

The normal method of simulating analogue delay is to apply a small amount of rapid S+G modulation to the delay time. This delivers relatively clean results while adding plenty of random 'wobble':

Experiment from C127: Analogue delay 1

```
Fit CUTOFF = 0
Fit ENV-AMOUNT = 127
Fit DECAY = 70
Del SEND = 64
Del EDIT / DELAY/REVERB Mode = Delay
Del EDIT / DELAY / Clock = 3/16
Del EDIT / DELAY / Color = +20
Del EDIT / DELAY / Rate = 118
Del EDIT / DELAY / Depth = 1
Del EDIT / DELAY / Shape = S+G
Del EDIT / REVERB / Feedback = 80
```

For a dirtier “Born Slippy” kind of delay, try using Rev+Feedb1 instead:

Experiment from C127: Analogue delay 2

```
Fit CUTOFF = 0
Fit ENV-AMOUNT = 127
Fit DECAY = 70
Del SEND = 64
Del EDIT / DELAY/REVERB Mode = Rev+Feedb1
Del EDIT / REVERB / Type = Ambience
Del REV-TIME = 0
Del DAMPING = 64
Del EDIT / REVERB / Color = -20
Del EDIT / REVERB / Clock = 3/16
Del EDIT / REVERB / Feedback = 70
```

Very short delays are at the heart of comb filters and the standard “physical modelling” method of simulating plucked strings. I have included a patch (FX-Fun1 HS) which is basically just a very short burst of noise.

With SEND turned up, this is transformed into an interesting showcase for short delay effects. The mod wheel adds vibrato by modulating the Depth parameter (look in ASSIGN3). See also TankYou HS.

To emulate the shattering reverb of old organs and amplifiers, use the mode Rev+Feedb2 with zero Color. Predelay should be somewhere between 70 and 120. Also, REV TIME should not swamp the ‘shatter’ - anything over 64 is probably too long. Try turning FEEDBACK up instead. See SpacePat HS.

Very Short Delays

PATCH: FX-FUN1 HS

Spring Reverb

Gated Reverb

Because RevbDecay is a modulation destination, reverb is very easy to 'gate'. Have a look inside GateRim HS (in bank D) - a triggered square-wave LFO modulates RevbDecay.



13: Possible Sources of Confusion

If you go to www.access-music.de and click on SUPPORT, you will find quite a comprehensive FAQ list there. This chapter is meant as a supplement to the FAQ, and only covers questions which may arise due to minor inconsistencies in the Virus user interface.

What's up with LFO3 SELECT?

You may have been a little confused by the LFO3 destination select function - even I was when I first got my Virus C. It seemed to me that every destination from OSC 1 to SYNC PHASE adopted the last Amount I set. What I didn't realise was that LFO3 destination selection is fundamentally different: It selects ONE of the destinations i.e. you can't define multiple destinations for LFO3 here. Of course there is only one Amount, and this will not change when you select a different destination!

When am I in a menu, and how do I get out again?

You are always in some kind of menu. A menu could be defined as any view in the display which can be exited by selecting another menu. According to this circular definition we have the following:

- Arpeggiator EDIT
- Effects EDIT (with Distortion, Phaser or Chorus)
- Delay/Reverb EDIT
- Lfos/Mod EDIT
- Lfos/Mod SELECT (the lower button)
- Lfos/Mod SHAPE "WAVE"
- General EDIT
- SYSTEM
- MULTI EDIT (via the same button as SYSTEM)
- STORE (exited only via STORE, SINGLE, MULTI or UNDO/Compare)
- Compare (sub-menu of STORE entered by pressing UNDO)
- Oscillators EDIT
- MULTI
- SINGLE
- MULTISINGLE
- Filters EDIT (with filter 1, 2 or both)

The way to exit any other menus is to press SINGLE (or MULTI if you are currently working in multi mode).

What is the difference between PatchVol and ChannelVol?

PatchVol: The destination used for modulating total volume in Single mode - look in the main EDIT menu under COMMON.

ChannelVol: If ChanVolRx is enabled, volume is controlled via MIDI CC #7 (volume pedal), so this is the one you should use if you want to mix Multi mode arrangements via sequencer or MIDI fader box. Initial values of ChannelVol can be stored in Multis only.

Is FILT GAIN the same as OscMainVol?

Not quite. FILT GAIN (an LFO1 preset destination) is simply the master level at the filter inputs. OscMainVol (a generally available destination) is just like automating the OSC VOL knob. The upshot of this is that FILT GAIN can affect noise and ring modulator levels but not saturation, whereas OscMainVol can affect saturation but not the noise or ring modulator levels. Alles klar?

Can't I modulate Vocoder parameters?

You just have to modulate the corresponding filter parameter. There is a table showing how filter and vocoder parameters relate to each other in the Tips and Tricks section of the manual. There is a typo there: Filter Env Release is listed twice - the first one should read "Filter Env Decay".





Appendix A – Virus C Shortcuts

Sect.	Combination	Result
Flt	FILT1 + FILT2	Select both filters
Gen	TRANPOSE + TRANPOSE	PANIC function
Gen	MULTI+SINGLE	Enter MULTISINGLE mode
Gen	SINGLE + PART or SINGLE + PARAMETER or SINGLE + VALUE knob	Scroll through categories
Gen	SINGLE + VALUE button	Jump to the next / previous patch in the current category
Gen	MULTI + VALUE knob	Scroll through patches in the current bank
Gen	VALUE + VALUE	Reset to default value
Gen	VALUE held	Scroll through values
Gen	VALUE held + VALUE Release second button to stop	Faster scroll through values
Gen	VALUE held + VALUE then release first button	Jump to minimum / maximum value
Gen	PART+PART	Demo function. Not available for 8-bank OS versions.
Mod	EDIT + SELECT or SELECT + EDIT	Step upwards through the LFO destinations or MOD slots. SELECT here means the lower button.
Mod	SHAPE + Power On	Reset global parameters
Osc	EDIT + SYNC	AUDITION. Pressing for 2 seconds holds the note indefinitely.



Appendix B – Glossary

Several of the terms in this list also have common, everyday meanings. Most of the ‘definitions’ below only apply to synthesizer technology.

Acronym

Abbreviation using the first letter of each word.

Adaptive Parameter Smoothing

Rapid interpolation of parameter changes in order to eliminate zipper noise.

Additive synthesis

Method of creating sound by summing relatively simple waveforms (usually many sine waves) to form more complex sounds.

See *subtractive synthesis*

ADS (Attack Decay Sustain)

Minimoog-style envelope in which the decay and release segments are ganged together.

See *ADSR, attack, decay, sustain, release, envelope*

ADSR (Attack Decay Sustain Release)

The de facto standard envelope form.

See *attack, decay, sustain, release, envelope*

Aftertouch

Older term for either Channel Pressure or Key Pressure. Usually the former, as Key Pressure is rarely implemented for its original purpose.

Algorithm

Word (mis)used by Yamaha for the different arrangements of modulators and carriers in their FM synthesizers.

AM

Acronym for Amplitude Modulation. Usually modulating volume from an audio source (as in AM radio).

See *FM, amplitude, modulation*

Amplitude

The level of a signal.

Analogue Synthesis

A more common but less descriptive term for subtractive synthesis.

Arpeggiator

Originally a simple clocked processor that takes chords played on a keyboard and plays them back sequentially. The Virus arpeggiator also features repeated chords, preset rhythmic patterns, indefinite hold etc..

Attack

In an *ADSR*: The initial “A” stage where the level rises from zero to maximum.

See *decay, sustain, release*

Autobend

Name originally given to a feature of early preset synthesizers where each note sweeps rapidly up to its nominal pitch. Sometimes used for any form of automatic pitch bending.

See *pitchbend*

Bandpass filter (BP)

Passes a range of frequencies while rejecting all others (i.e. above and below the cutoff point). Particularly useful for vocal sounds.

See *bandstop, highpass, lowpass, comb*

Bandstop filter (BS)

Also called notch filter or band reject filter. Suppresses a range of frequencies while passing those above and below the cutoff point.

See *bandpass, highpass, lowpass, comb*

Bandwidth

For audio: Frequency range.

Bipolar

Describes a modulation source which can have negative as well as positive values.

See *unipolar*

BPM

Acronym for Beats Per Minute.

Buchla, Donald

Father of the voltage controlled, modular synthesizer. Robert Moog added a keyboard to this concept and became famous.

Carlos, Wendy (nee Walter)

Electronic music pioneer of the 1960s. “Switched On Bach”, made by multitracking sounds from a Moog modular system, was the album which introduced the general public to synthesizers.

Carrier

Term used in FM to mean the oscillator being modulated by another audio-frequency signal (the modulator). In FM radio this has a fixed frequency.

Channel pressure

Single modulation source from pressure applied to the keyboard after notes have been played.

See *key pressure*, *aftertouch*

Chorus

Effect used to make sounds more complex by using short delays, the lengths of which are modulated by an LFO.

Comb filter

Causes cancellation/accentuation at regular frequency intervals.

Concave

In envelopes (e.g. *ADSR*): the segment is valley-shaped.
See *convex, linear, non-linear*

Control Voltage

The source used in analogue synthesizers to modulate parameters. A quasi-standard for oscillator or filter frequency is 1 Volt per octave. Voltage control has been supplanted by MIDI.

Controller

Either a MIDI CC (continuous controller) or a physical device such as joystick, modulation wheel or pitchbend wheel.

Convex

In envelopes (e.g. *ADSR*): the segment is hill-shaped.
See *concave, linear*

CPU

Acronym for Central Processing Unit. The component in a computer that sequentially carries out program instructions.

Curtis Chips (CEM)

Integrated circuits designed to replace entire modules in analogue synthesizers. Curtis is the brand used in the Oberheim Matrix 1000 as well as many other 80s synths.
See *SSM chips*

Cutoff

Threshold frequency of a filter above and/or below which frequencies are attenuated or boosted. Resonance accentuates frequencies around the cutoff point.

CV

Acronym for Control Voltage.

Default

The standard (preset) value.

Decay

In an ADSR: The second “D” stage, where the level falls from maximum to the sustain level.

See *ADSR, attack, sustain, release*

DSP

Acronym for Digital Signal Processor (or Processing). DSPs are the heart of all modern synthesizers.

Duophonic

A maximum of two notes can be played at the same time e.g. Aulos (Greek double flute), Yamaha CS40m, EMS DK2, OSC Oscar.

See *monophonic, polyphonic*

Encoder

Non-stop rotary control (as opposed to a standard potentiometer which physically stops at minimum and maximum).

Envelope (generator)

Synthesizer module which creates a contour over time, originally to emulate the way acoustic sounds appear and fade away.

See *ADSR*

Feedback

Routing part of the output signal back into the input e.g. for echoes or resonance.

Fifth

Note interval of 7 semitones. In diatonic scales this is only five notes (inclusive).

Filter

Module in all subtractive synthesizers used to pass only certain frequencies while attenuating others.

See *highpass, lowpass, bandpass, bandstop, comb*

FM

Acronym for Frequency Modulation (e.g. FM radio, FM synthesis).

See *AM, frequency, modulation*

Frequency

The rate at which something vibrates. Measured in Hertz (Hz), the maximum range for human hearing is from about 20 Hz to 20kHz (20,000 Hertz).

Fundamental

The strongest pitch reference in a sound. Usually the lowest frequency component.

See *harmonic*

Harmonic

Component of a sound whose frequency is an integer multiple of the fundamental. Any other frequencies in the sound are 'enharmonic'.

Highpass filter (HP)

Rejects frequencies below the cutoff point.

See *bandpass, bandstop, lowpass, comb, cutoff*

IC

Acronym for Integrated Circuit.

Integrated circuit

Electronic circuit on a single chip.

Joystick

Two-dimensional controller in the form of a short stick. Used to modulate two parameters at the same time.

Key Follow

Modulation source proportional to MIDI note number.

Key pressure

In MIDI specification: independent, multiple modulation sources from pressure applied to several notes after they have been played. Rarely implemented for its original purpose, this is used in the Virus to supplement System Exclusive.

Also called poly-pressure.

See *channel pressure*, *aftertouch*

Legato

Leaving no space between consecutive notes.

LFO

Acronym for Low Frequency Oscillator. Used as a modulation source.

Linear

In a straight line. Attack in the Virus is linear, all other segments (Decay, SustainTime, Release) are slightly concave.

Lowpass filter (LP)

Rejects frequencies above the cutoff point.

See *bandpass*, *bandstop*, *highpass*, *comb*

Mainframe

The central part of a very large computer system.

MB

Acronym for MegaBytes. Unit of computer memory equal to $2^{20} = 1,048,576$ Bytes. Sometimes quoted as being exactly a million Bytes.

Meme

The cultural equivalent of “gene”. Also subject to a kind of evolution.

MIDI

Acronym for Musical Instrument Digital Interface. International standard for communication between musical instruments and computers.

MIDI CC (continuous controller)

Standard MIDI control numbers. Modulation wheel = CC#01, main volume = CC#07, sustain pedal = CC64 etc. Most of the 128 CCs have not been specified.

Modular

Composed of modules i.e. separate units which are meant to be connected to each other.

Modulation

Any varying of a parameter (destination) according to the level of another parameter (source).

Modulation matrix

That part of a synthesizer where general modulation routing can be defined. See *pin matrix*

Modulation wheel

Multi-purpose left-hand control first seen on the Minimoog. Often used for vibrato.

Modulator

Most often used in FM for the audio source which modulates an audio-frequency destination (carrier) - in FM radio this is the broadcast signal you hear. Can also mean any modulation source.

Monophonic

Only one note can be played at a time e.g. saxophone, Minimoog. See *duophonic, polyphonic*

Multimode filter

Filter which can be switched into different modes e.g. lowpass, highpass, bandpass, bandstop, comb.

Nominal value

In this tutorial: The value before being affected by modulation.

Non-linear

Not in a straight line. For envelopes, this means that segments are either concave or convex.

Notch filter

Alternative term for bandstop filter.

Octave

12-semitone interval.

Offset

Shift of any value by a defined constant. Also called 'bias'.

Operator

Term used in FM for oscillator i.e. either carrier or modulator.

Parameter

Setting, variable value.

Patch

Originally the arrangement of connections between modules in a modular synthesizer. Now synonymous with 'sound', 'program' etc. in any synthesizer with user-memory.

Peak

Two related meanings: 1) the upper portion of an oscillating signal 2) a transient maximum within a signal.

See *trough*

PD

Acronym for Phase Distortion.

Phase

Position within a wave. Measured in degrees, whereby 360° is a full cycle.

Phase distortion

Synthesis method similar to FM used by Casio in their CZ range of synthesizers. Has not been commercially pursued since then.

Phaser

Signal processor which uses very short delay lines (sometimes filters) to cause frequency cancellation/accentuation effects.

Physical Modelling

Any method of synthesis that digitally emulates the physical characteristics of real instruments (either acoustic or analogue).

Pin matrix

A two-dimensional array of sockets into which pins are inserted. Flexible patching system used by only a few modular synthesizers (e.g. EMS VCS3).
See *modulation matrix*

Pitch

Frequency. 'Pitch' is used specifically for musical notes, whereas 'frequency' is the more general term.

Pitchbend wheel

Left hand controller giving the player immediate control of pitch. Usually centre-sprung.

Pitch to voltage converter

Device used to transform relatively pure tones (e.g. voice or guitar) into control voltages suitable for driving (non-MIDI) analogue synthesizers.

Polyphonic

When several notes can be played at the same time.
See *duophonic*, *monophonic*

Polyphonic pressure

See *key pressure*

Portamento

Gliding smoothly from one note to another.

Pulse

Oscillator waveform which alternates between two levels.

See *pulse width*

Pulse width

Ratio of times between upper and lower levels in a pulse wave. Often given as percentage.

PW

Acronym for Pulse Width.

Q

“Quality”. The slope of a filter band.

Rectification

Signal processing method where one half (upper or lower) of a signal is inverted (full wave rectification) or zeroed (half wave rectification). The Virus C offers variable full wave rectification with subsequent DC-offset.

Recursive modulation

When a modulation destination affects the source itself.

Release

In an *ADSR*: The final “R” stage where level falls to zero.

See *attack, decay, sustain*

Resonance

In filters: Boost around the cutoff frequency originally achieved via feedback.

Retrofit

Fitted afterwards e.g. MIDI capabilities for the (pre-MIDI) Memorymoog.

Rigby, Daft Eric

Eccentric Welsh cellist and self-styled “iconoclassical decomposer”. Rigby’s better-known works include “Disconcerto for String Quartet and Prepared Chainsaw in A minor” (1927) and “65 Little Pieces” (1938). Smashing stuff!

RAM

Acronym for Random Access Memory. Volatile, can be written to.

ROM

Acronym for Read-Only Memory. Permanent, cannot be written to.

RM

Acronym for Ring Modulator.

ROMpler

Instrument similar to a sampler, but with preset audio samples in ROM.

S+G

Acronym for Sample and Glide. Like Sample and Hold but smooth i.e. not stepped.

S+H

Acronym for Sample and Hold.

Sample and Hold

Clocked processor which regularly “freezes” momentary input levels (so the output is stepped). S+H in the Virus is random source because the input is always noise.

Sampler

Instrument capable of recording sounds into RAM and playing these back via MIDI note-on messages.

Saturation

True filter ‘saturation’ is the distortion/clipping caused by overloading the input of a filter. Access has greatly expanded on the idea, but kept the term.

Saw / Sawtooth

Oscillator waveform in the shape of a sawtooth. Very rich in harmonics.

Sine

The simplest, purest oscillator waveform. No harmonics.

Square

Special form of pulse whereby peaks and troughs have the same length. Available on the Virus when SHAPE = 127 and PW = 0.

SSM chips

Integrated circuits designed to replace entire modules in analogue synthesizers. SSM is a brand name. Used in the version 2 Sequential Circuits Prophet 5.

See *Curtis chips*

Staccato

Separate, short notes.

See *legato*

Sub-oscillator

Slave oscillator always pitched one octave below oscillator 1 in the Virus.

Subtractive synthesis

Also called analogue synthesis. The method of shaping sound by filtering out certain frequencies from relatively complex waveforms.

See *additive synthesis*

Sustain

In an ADSR: The third "S" stage, where the level remains constant until a note is released. In the Virus, this stage can also fall to minimum or rise to maximum depending upon the Sustain Time parameter (now in the filter menu).

See *attack, decay, release, envelope*

Sync

Synchronisation. Oscillator sync means that one oscillator resets the phase

another.

Sysex

Short for System Exclusive. Device-specific MIDI data used for transferring patches, editing parameters via computer etc.

Timbre

Tonal quality (in general).

Tweak

Colloquial term for adjusting knobs or menu parameters on a synthesizer.

Unipolar

Describes a modulation source which can only have positive values.
See *bipolar*

Unison

Several instances of the same note at the same time.

VA

Acronym for Virtual Analogue.

Velocity

MIDI data proportional to the speed at which a key is struck.

Vibrato

Regular, fairly rapid pitch modulation, 'wobble'. Common use of LFOs, normally triangle or sine.

Virtual Acoustic

Physical modelling of acoustic instruments.

Virtual Analogue

Either DSP hardware or CPU-based software synthesizers which can emulate traditional analogue i.e. subtractive synthesizers.



Voltage control

The method used in analogue synthesizers to modulate parameters.
See *Donald Buchla, control voltage*

Wavetable

- 1) Digital oscillator comprised of several waveforms in RAM.
- 2) Samples in a computer sound-card e.g. Soundblaster.

Zipper noise

Grainy artefacts caused by quantised modulators. Eliminated in the Virus by adaptive parameter smoothing.





Appendix C – Virus C Cryptology

Virus C Series Abbreviations

Abbreviation	as in...	Meaning
A	AboostInt	Analog
Amp	AmpAttack	Amplifier
Amt	Ass1Amt	Amount
Arp	ArpClock	Arpeggiator
Arpegg	ArpeggSend	Arpeggio
Ass	Ass1Amt	Assign
Attk	FltAttk	Attack
Auto	Auto	Automatic
Bal	FltBal	Balance
Bnd	PitchBnd	Bend
Bst	AnalogBstTune	Boost
Buff	SingleBuff	Buffer
Chan	ChanPres	Channel
Chg	MultiPrgChg	Change
Cont	Lfo1Cont	Contour
Contr	Contr3	MIDI Controller (CC)
Contr	Contr	Control
Cut	Lfo2>Cut1	Cutoff
Dec	FltDec	Decay
Dept	PhaserDept	Depth
Dest	Dest	Destination
Dir	Dir/Eff	Direct
Dis	Dis	Disable
Displ	KnobDispl	Display

Virus C Series Abbreviations

Dist	DistInt	Distortion
Dly	ChorusDly	Delay
Dpth	ChorusDpth	Depth
Dst	Dst1	Destination
Eff	Dir/Eff	Effect
Ena	Ena	Enable
Env	AmpEnv	Envelope
Fdbk	PhaserFdbk	Feedback
Feed	DelayFeed	Feedback
Feedbck	RevFeedbck	Feedback
Filt	Filt1EnvAmount	Filter
Flt	FltAttack	Filter
Flw	KeyFlw	Follow
Fm	FmEnvAmt	FM
Freq	EqMidFreq	Frequency
Frequenc	PhaserFrequenc	Frequency
Gn	Lfo1>FltGn	Gain
Hi	HiPage	High
i	iSnap	Intelligent
Id	DeviceId	ID
Init	OscInItPhase	Initial
Int	AboostInt	Intensity
Int	Int+MIDI	Internal
Intens	Intens	Intensity
Keyflw	Flt1Keyflw	Key follow
Len	ArpNoteLen	Length
Lo	LoPage	Low
Mem	MemProtect	Memory
Mod	ModWheel	Modulation
Mod	RingModMix	Modulator
Neg	Neg	Negative

Virus C Series Abbreviations

Osc	SubOscVol	Oscillator
Pan	Lfo2>Pan	Panorama
Par	Par4	Parallel
Para	ParaDepth	Parameter
Ped	HoldPed	Pedal
Phs	OscInItPhs	Phase
Pls	Lfo1>PlsWd	Pulse
Porta	PortaSw	Portamento
Pos	PosTri	Positive
Predely	RevPredely	Predelay
Prg	MultiPrgChg	Program
Prog	ProgChange	Program
Prs	PolyPrs	Pressure
Ptch	FiltEnv>Ptch	Pitch
Puls	Lfo1>PulsWidth	Pulse
QFact	EqMidQFact	Q-Factor
Reduc	RateReduc	Reduce
Rel	iRel	Relative
Rel	FiltRel	Release
Reso	Filt1Reso	Resonance
Rev	RevDecayTime	Reverb
Revb	RevbDecay	Reverb
RX	MIDI DUMP RX	Receive
S+G	S+G	Sample and Glide
S+H	S+H	Sample and Hold
Sec	SecBalance	Secondary Output
Sel	Osc1WavSel	Select
Ser	Ser6	Serial
Semi	Osc3Semi	Semitone
Semiton	Semiton	Semitone
Ser	Ser6	Serial

Virus C Series Abbreviations

Shp	Osc1ShpVel	Shape
Sin	Sin	Sine wave
Sost	SostPed	Sostenuto
Sprd	PhaserSprd	Spread
Sus	FltSusTime	Sustain
Sust	FltSust	Sustain
Sw	PortaSw	Switch
SysEx	SysEx	System Exclusive
Tri	PosTri	Triangle wave
Trig	TrigPhase	Trigger
TX	MIDI DUMP TX	Transmit
Uni	UniDetune	Unison
Vel	Flt1EnvVel	Velocity
Velo	VeloOn	Velocity
Vol	ChannelVol	Volume
Wav	Osc1WavSel	Waveform
Wd	Lfo1>PlsWd	Width
Wdh	Osc1PlsWdh	Width

Appendix D - Reference to patches

2 Guitars CS	60	K:Geet-R	66
AT-Mini HS	26	LordOrg HS	59
Baryon HS	72	Mandoid HS	64
BassDrumCK	83	McCloud HS	35
BellBoy BC	75	Nowhere BC	39
C.Towns HS	28	Nylon BC	63
Chamber HS	69	ObiFanf HS	31
Choir2 HS	78	Oboe HS	73
Choir3 HS	79	Phloid HS	57
Claps2 HS	87	PlanetarHS	79
ClavKey CS	58	PntsTchrM@	78
ClubBassSV	28	Pstyro3 HS	89
CommerseSV	45	RezBass HS	29
Congoid HS	87	Saxpet HS	72
DoubleB HS	76	ShaperMorf	71
Dr.What?HS	90	Sitars HS	68
DX Road HS	51	Slappy1 HS	88
Dyson++ BC	44	Sleigh HS	85
EasyFlute	70	Snare3 BC	84
EPStage?M@	55	SpacePat HS	33
EPWhirlyM@	57	Sparkle BC	53
EPZeply M@	57	SqueakerBC	89
ETom2002M@	85	STR ABEJA	32
Fender!M@	60	StTheoryBC	32
FlatHat HS (convex envelope)	84	Sunder HS	35
Flutes HS	70	Swoopie HS	37
Flutoon HS	71	Synthar HS	27
FlyingT CS	36	TankYou HS	34
FuzzlineDM	41	Trumpy HS	74
FX-Fun1 HS	101	TuvaWeelHS	78
Handsaw HS	89	UniVoV BC	43
Harpisie HS	67	VR-78 2 RP	88
HevyHevyM@	61	WildThngHS	61
J.Edgar HS	43		

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