ACE

Any Cable Everywhere

Contents

A.C.E.

Does the world really need another simple synthesizer? We believe it does - but there's a lot more to ACE than meets the eye...

The software synthesizer ACE (Any Cable Everywhere) delivers top quality sound at a highly competitive price. The selection of modules and clear layout make ACE the ideal instrument for newcomers delving into the fascinating world of modular synthesis. The number of ways to connect modules together is practically infinite, and you will soon discover how much more fun it is to make your own sounds in ACE than in a non-modular synthesizer.

Concept and Features

Modular, ultrasonic

Most digital synths handle audio signals and modulation signals differently. Audio signals are normally evaluated at a sample rate between 44100 and 96000 Hertz, while modulation signals have to make do with 1000 Hz or even less (often called the "control rate" of the synthesizer).

ACE is very different in this respect. While the oscillators feature more than 500 times oversampling, all signals - including modulation - run at least twice as fast as the host application's sample rate. And this is only the lowest of ACE's quality settings!

ACE does not differentiate between audio signals and modulation/control signals at all. Any Cable Everywhere - you can plug any of the 24 signal outputs into any of the 30+ signal inputs and expect it to work just like a vintage modular hardware synth.

The sky is NOT the limit - all modulation can work beyond the limits of human hearing. For instance, the LFOs (Low Frequency Oscillators) can be set above 20 kHz, and still modulate e.g. the pulse width of an oscillator. This gives you a sonic freedom previously reserved for expensive analogue hardware. Either or both LFOs can be used as audio oscillators in their own right (e.g. for crystal-clear FM sounds). Conversely, the VCOs can be used as alternative LFOs, i.e. they can be set as low as 0.00 (zero) Hz. Note: DC components are quickly removed from the VCO outputs, so their shapes when used as LFOs might not be as you would expect.

Analogue modeling

To make this work, not only do the modules resemble their analogue counterparts, but their components do as well. Wherever necessary, the non-linear characteristics of analogue circuitry has been programmed directly into the code. For instance, the filter algorithm is entirely built around a very precise mathematical model of an analogue filter - as are the basic components of the oscillators and envelope generators. This technique is known as Analogue Modeling.

Only the LFOs, mixer, ramp and control-signal "logarithmizer" are not analogue models. You will soon see why: unlike its analogue ancestors, ACE is not susceptible to instabilities, and all oscillators can be synchronized to the song tempo. For instance, even perfectly host-synchronized beating between two oscillators.

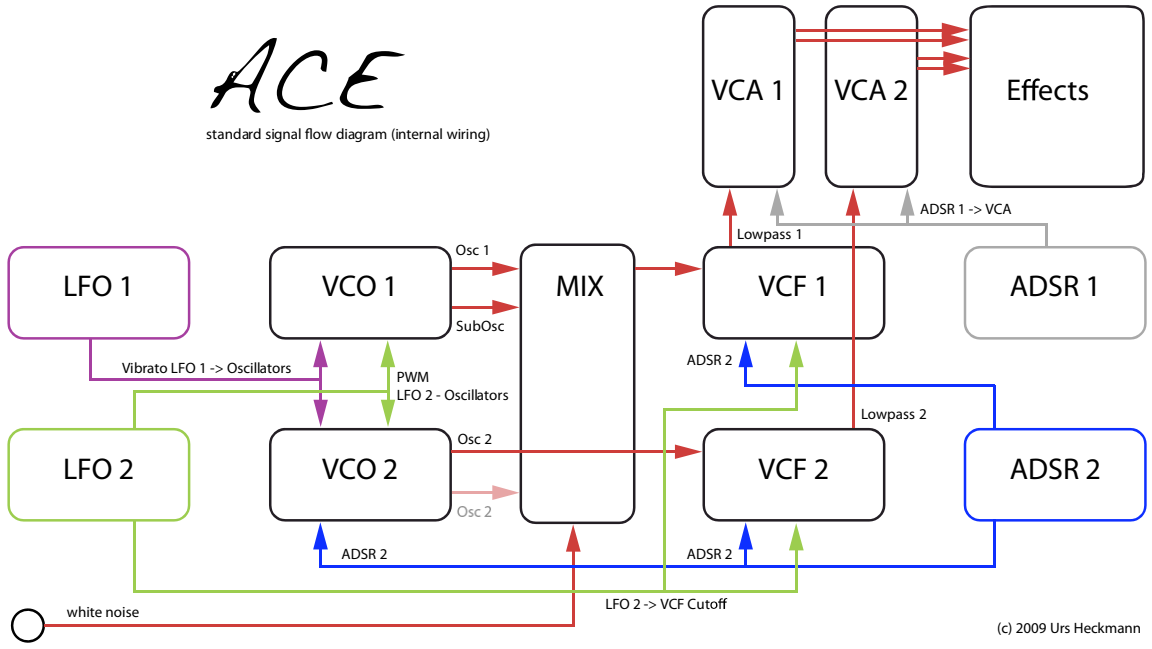
Non-linear distortion in the filters (which, by the way, are self-oscillating), the extremely fast envelopes and modulation pathways as well as certain other details

(e.g. "Glide2") open up a surprising palette of sound-sculpting techniques that are not available in other software synthesizers.

It is true that ACE places high demands on CPU power, and some of the typical features of recent digital synthesizers (e.g. "supersaw" or audio sample import) are nowhere to be found, but ACE rewards you with audio quality previously unheard of in the world of software synths. So can it really sound "analogue"? We'll let you be the judge of that.

Layout and Signal Flow

At first glance, the architecture is similar to a simple synthesizer of the late '70s and early '80s – which was also adopted by the first generation of "virtual analogue" hardware synthesizers of the '90s. Whenever you open a fresh instance of ACE, a default patch is loaded, without any cables. The pre-patched signal flow is as follows:



Oscillator 1, the sub-oscillator, oscillator 2 and the noise generator are merged in the mixer, and from there the signal is sent to filter 1. This filter is routed to output amplifier 1. At the same time i.e. in parallel, oscillator 2 is sent through filter 2 to the other amplifier.

LFO 1 is used for pitch modulation by default i.e. vibrato for both oscillators, and the amount is controlled by the modulation wheel (MIDI CC#1). LFO 2 is routed to the pulse width modulation inputs of the main oscillators, as well as to both filter cutoff frequencies.

ADSR 1 is used as envelope generator for both VCAs. ADSR 2 is set as modulation source for oscillator pitches, both filter cutoff frequencies and the amplitude of LFO 2.

Always remember that these are only the default connections - they can be replaced with any signal you like by simply connecting cables. Also note that you can have several cables connected to a single output, but only a single cable connected to an input (existing connections to inputs are replaced).

If you really feel the need to compare ACE with a classic modular synth (or three), think of it as a pimped-up ARP 2600 using modules from a Roland SH-7 with (almost) the patching flexibility of an EMS VCS3 / Synthi A - but polyphonic. Just like the ARP 2600, ACE is pre-patched so that it will work "out of the box", but all default connections can be overridden using patch cords. Roland's SH-7 includes - like ACE - a second filter and an extra envelope generator. However, many of the modules in ACE have been designed to take on a number of very different or unusual tasks. For instance, you can use the ramp generator as an LFO, a multiple as ring modulator, or LFO 1 as a waveshaper.

Operation Settings

Note: ACE does not include global settings i.e. all values are saved and recalled with each patch.

mode determines the polyphony and basic interpretation of MIDI notes:

- **poly** - polyphonic (several notes can be played at the same time)
- **retrigger** - monophonic, notes are always triggered, even when they overlap
- **legato** - monophonic, notes are only triggered after first releasing all current notes
- **duophonic** - VCO 1, LFO 1 and VCF 1 respond to the lowest note while VCO 2, LFO 2 and VCF 2 respond to the highest note.

voices (mainly for use in poly mode) sets the maximum number of notes that can be played before voice-stealing occurs. The **few** and **medium** settings can reduce

CPU load, especially sounds with relatively long envelope release times. Note that stacking (see below) uses multiple voices and therefore reduces polyphony.

The **quality** switch is also used for reducing CPU load. Depending on modulation speeds or the amount of filter distortion, quality can be reduced to **standard** or even **draft** without any audible compromises. Of course this can be decided according to how the patch is going to be used - it will sometimes be fine at the bass end, but less than optimal further up the keyboard.

pb up and **pb down** determine how many semitones the pitchbend wheel will bend notes.

drift determines whether individual voices are slightly detuned against each other for a fuller, more lively sound.

transpose adjusts the overall pitch, with a range of +/- 2 octaves.

tune adjusts the overall pitch, with a range of +/- 50 cents i.e. half a semitone.

output adjusts the overall volume.

Stack

The **stack** parameter determines the number of voices per note. Up to 8 voices can be stacked for a very powerful **unison** effect, just like a few classic polyphonic synthesizers e.g Oberheim OBXa. Depending on the mode and stack value, however, ACE can still be played polyphonically. Remember that this is not a "supersaw oscillator", it is true unison: the entire voice (all oscillators, filters etc..) is multiplied. Of course this kind of thing takes a lot of CPU power, but we think it is worth it. For instance, multiple filter distortion on one note is much more "alive" than a single filter could ever deliver. In the **tweak** page, up to 8 voices can be detuned within a range of +/- 24 semitones.

Glide

Glide (also called portamento) means slurring the pitch of notes from one to the next.

The **glide** knob controls either the glide **time** or the glide **rate**, depending on the state of the **glide mode** switch. In the **time** setting, gliding an interval of a semitone or e.g. 6 octaves takes exactly the same amount of time, whereas in the **rate** setting, it depends on how far apart the notes are.

ACE has a few unusual but interesting additions to conventional glide:

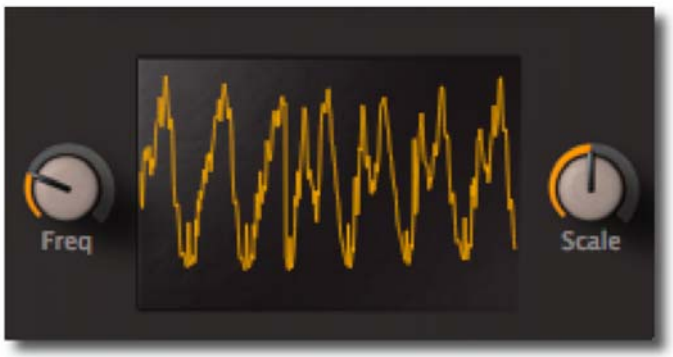
glide2 - this is an offset (i.e. relative to the value of **glide**) applied to LFO 2, VCO 2 and VCF 2. Careful use of this parameter can really bring otherwise static sounds to life.

range - In classic polysynths, polyphonic portamento was seldom used except for special effects. In ACE, the range parameter can be used to shift the initial position (where the glide starts) closer to the target note. So the glide can start e.g. "already half way there" for a more subtle effect. Tip: set the range to very low values for natural intonation effects.

Further operation settings are available in the **tweak** page.

Oscilloscope

The oscilloscope at the top of the ACE window displays a mono sum of the output (before the effects). This is very useful for e.g. tweaking waveforms, checking the effects of modulation or filtering, envelope shapes etc..



Especially in a synthesizer that allows modulation at audio rates, the oscilloscope is an invaluable aid to programming sounds. Note that the display is synchronized to played notes, as well as to zero-crossings (negative to positive) and whenever a longer scan is complete.

The **freq** knob adjusts horizontal resolution, **scale** adjusts vertical resolution.

Pitch Controls

All four oscillators (**LFO1**, **LFO2**, **VCO1** and **VCO2**) have the same set of pitch parameters. There are three knobs for this purpose: coarse, fine and modulation, plus associated mode switches (which also serve as labels for these knobs).

The **coarse** pitch knob (top left) of each oscillator has a range of 0.00 to 24.00, and the modes are as follows:

- **semi** – up to 24 semitones above the current pitch.
- **partial** – up to 24 overtones. A value of 1 is one octave up, 3 is two octaves up, 7 is three octaves up and 15 is four octaves up.
- **subhrm** – up to 24 subharmonics (see "Trautonium" in the Internet).
- **hertz** – 0 to 24 Hz. At 0.00, the oscillator is silent – note that DC components are removed.
- **sync** – the oscillator is synchronized to the song tempo. A value of 1.0 is a semibreve (whole note), 4.0 means a quarter note (crotchet) etc..

The **fine** pitch knob has a range of -50.00 to +50.00 (i.e. it is bipolar), and the modes are as follows:

- **cents** – detunes the oscillator by +/- 50 cents.
- **5 Hz** – detunes the oscillator by +/- 5 Hertz
- **beats** – detunes the oscillator in sync with song tempo. A value of +4.00 in this mode means that the oscillator completes one extra cycle for each quarter note (crotchet).
- **mtply** – oscillator frequency is either multiplied (0.00 – 50.00 times), or divided from 1/1 (at -1.00) to 1/50th (at -50.00).

The **modulation** knob determines the polarity and amount of pitch modulation from whichever source is connected, and the modes are as follows:

- **cents** – +/- 50 cent i.e. half a semitone
- **5 semi** – +/- 5 semitones
- **50 semi** – +/- 50 semitones

As you can see, control over pitches in ACE is very flexible. All oscillators – whether VCO or LFO – can be tuned very comfortably over an extremely wide range (from 0 Hz to 20 kHz). This means that the labels **LFO** and **VCO** only describe the default functionality.

LFO 1

By default, the first LFO delivers a pure sine wave. The pitch controls are described above.

The **reset** switch determines if and when the LFO is reset (to a definable phase - see below):

- **free** - runs continuously (and is thus monophonic).
- **gate** - polyphonic, resets whenever notes are played.

Note that if LFO 1 is also in **sync** mode, it is additionally reset in sync with the song

The **phase** knob usually determines phase position. This value can be modulated by any signal. For instance, connecting another oscillator gives you classic FM sounds (which illustrates that those synths were actually based on phase modulation - "FM" is a misnomer). Of course this modulation source could be LFO 1 itself ("FM feedback"), which results in a much brighter waveform similar to a sawtooth.

The amplitude parameter determines the level of LFO 1, and this value can also be modulated. The default source for this is the modulation wheel (MIDI CC#01) , so LFO 1 is useful for quick access to vibrato effects etc..

LFO 1 - Sample & Hold Mode

If any modulation source is connected to the **s&h** input, LFO 1 is put into **sample and hold** mode. LFO 1 samples the input with its own rate as "clock speed". Connect noise here for vintage random effects.

In s&h mode, the **phase** knob becomes a lag processor, smoothing out the jumps between successive output values. Note: at very high LFO 1 (i.e. clock) rates, the phase parameter acts as the cutoff of a lowpass filter, but going in the "wrong"

direction. So whenever you find that the LFO 1 signal seems to have disappeared, simply set its phase to zero.

Using LFO 1 in Unconventional Ways

Here are a few practical examples:

random waveforms. Connect white noise ("white") to the s&h input and use LFO 1 to modulate e.g. the frequency of an oscillator or filter.

sample rate reduction effects. Connect e.g. an oscillator to the **s&h** input. Set the coarse mode to **semi** and the fine mode to **mtply**. Turn the fine knob to a multiplication factor of about 20 or 30 – the LFO 1 output signal adopts the basic pitch of the oscillator, and delivers a rough version of same. Turn the oscilloscope frequency down to minimum to see the stepped waveform. Remember to set LFO 1 **phase** to zero or thereabouts, otherwise you won't hear anything!

waveshaping. ACE doesn't have a dedicated distortion or waveshaper module, but they aren't really necessary – the filters can already introduce plenty of distortion, especially when connected in series. However, LFO 1 can be used as an extra waveshaper...

Start with the default patch, and drag a cable from LFO 1 to one of the the main outputs. Set LFO 1 pitch modes to **semi** and **mtply**, and the multiplication factor to 0.00. Switch the reset mode to **gate** – LFO1 doesn't oscillate by itself now. Set the phase to 0.00 so that it completely resets to 0°.

Now patch the signal you want to process (e.g. VCO 1) into LFO 1's phase modulation input. Turn up the amount – you now have a "sine waveshaper"! Change the phase slightly to make the waveshaping asymmetrical. By the way, you can use this method to "bend" any signal, even an envelope! Simply plug an envelope generator into the phase modulation input instead.

LFO 2

LFO 2 is different from LFO 1. It doesn't offer phase modulation or sample & hold, but is more like a classic LFO with the following fixed waveforms:

- **sine** – sine wave
- **tri** – triangle wave
- **saw** – sawtooth wave
- **square** – square wave

Although LFO 2 is not as flexible as LFO 1, it offers a wider range of traditional modulation shapes. The other advantage of having these waves is that LFO 2 is the easiest one to use as a third audio oscillator alongside the two VCOs.

VCO - the Oscillators

The two VCOs ("voltage controlled oscillators") are the main sound-generation modules in ACE. Apart from the frequency control parameters mentioned above, both VCOs have a pair of blendable waveforms - sawtooth and pulse (**saw**/**pulse**). VCO 1 can also be switched to the mellower peak and triangle (**peak**/**tri**) waves.

Both oscillators have controllable pulse width for the pulse waves, ranging from 0% to 100%. Of course pulse width can be modulated (PWM) by any signal you like, including another oscillator. The default PWM source is LFO 2.

VCO 1 includes a sub-oscillator, which has a separate output socket. The **sub osc** has 3 modes: a square wave pitched either 1 or 2 octaves below the main oscillator, or a 75% pulse wave 2 octaves down.

VCO 2 has three fixed-source modulation controls, all of which depend on VCO 1:

ring - ring modulation. Crossfades between pure VCO 2 and VCO 2 ring modulated with VCO 1. Depending on the waveform and interval between the two oscillators, ring modulation can create metallic or nasal sounds, or even rhythms when VCO 1 is set to sync or hertz mode.

sync - turn this knob to maximum for the "hard-sync" found on many other synthesizers. The phase of VCO 2 is not only reset when it completes its own cycle (as always), but also whenever VCO 1 completes a cycle. The pitch of VCO 2 is typically set much higher than VCO 1, and is modulated by an envelope and/or LFO. The result is a very overtone-rich sound that retains the fundamental pitch of VCO 1.

Turn the sync knob down for a special kind of "soft-sync": The phase of VCO 2 is still reset by VCO 1, but not 100% (.e. not to 0°). The phase of VCO 2 moves by a certain proportion of its current value, e.g. 50%. This lets you create pure-interval overtones - try experimenting with the position of the sync knob and the interval between the two oscillators for some interesting overtones.

Connecting a cable to the modulation socket next to the sync knob effectively replaces a constant (+5V) default modulator. Tip: try modulating sync "hardness" from keyboard velocity or an envelope.

cross - cross-modulation. In ACE, this is analogue frequency modulation (unlike digital FM) in which VCO 1 modulates VCO 2. You should also think of this parameter as having a constant default modulator (+5V) which can be replaced by any signal you like.

Mixer

In the middle of the ACE window is a small but rather useful mixer which serves as the default link (and level control module) between the sound generation and sound processing - as you can see, MIX is the default input to VCF 1.

The upper **osc balance** knob controls the relative levels between VCO 1 and VCO 2. Below this are knobs for **sub osc** and **noise** levels.

The bottom knob is a general purpose input used for feeding any signal through the mixer. For instance, you could connect pink noise or a pitched LFO, or even experiment with feeding filters back into themselves via the mixer.

VCF - the Filters

ACE has two almost identical filters. Apart from the standard parameters **gain**, **cutoff** and **resonance**, the filters in ACE have two cutoff modulation inputs with amount knobs (the default sources are ADSR 2 and LFO 2), a resonance modulation input and a key-follow amount knob.

The cutoff value is measured in semitones from 0.00 to 150.00, a range of 12 ½ octaves. Accordingly, the modulation range is +/- 150 semitones.

The **keyfol** (key follow) parameter causes cutoff to follow the MIDI note. When set to maximum, the cutoff frequency follows notes 100%.

VCF 2 "cutoff" has 3 modes:

- **cutoff** – VCF 2 cutoff is independent of VCF 1
- **offset** – VCF 2 cutoff follows VCF 1 (including any modulation), but shifted by the offset value (negative or positive). This means that VCF 2 cutoff can be modulated by up to four sources (without using a multiple) – two from VCF 1 and two of its own.
- **spread** – similar to offset except that positive values are subtracted from VCF 1 and negative values are added to VCF 1. One obvious use is to set up the same modulation for both filters, but in opposite directions. Because adjusting the spread in VCF 2 also affects VCF 1 cutoff, you may have to keep adjusting VCF 1 cutoff.

res (resonance) range is 0.00 to 100.00, although self-oscillation can already start around 50.00. The actual amount of resonance depends on the level of the input signal (see **gain** below), so a generous range is necessary here. Resonance can be modulated by connecting any source to the socket to the left of the "res" label. Again, it is useful to think of the default modulator as being +**5V** here.

Each filter has two outputs. The upper ones offer four lowpass modes:

- **LP1** – 6dB/octave (1-pole lowpass)
- **LP2** – 12dB/octave (2-pole lowpass)
- **LP3** – 18dB/octave (3-pole lowpass)
- **LP4** – 24dB/octave (4-pole lowpass)

The lower outputs offer three other useful filter modes:

- **HP** – HighPass
- **BP** – BandPass
- **BR** – BandReject

Gain & Self-Oscillation

The filters in ACE have several properties generally considered unique to "real" analogue filters. For instance, they can easily be overdriven without sounding harsh. Unlike classic hardware filters, fairly heavy overdrive doesn't necessarily mean that you can't keep the resonance high - ACE has plenty of headroom here.

Especially around the threshold of self-oscillation where resonance seems to fight with the oscillators for control over pitch, there are surprising opportunities for sound design. Depending on the input signal and its level, it often sounds as if the input is actually modulating cutoff. Experimentation is the name of the game here!

The underlying cascade filter architecture allows the creation of different filter types at the same time, just like a classic multimode filter. In ACE, however, all types are capable of resonance and even self-oscillation. Note: It doesn't always have to be a 4-pole lowpass - depending on how the sound will be used in a track, the 1, 2 or 3-pole models are often more suitable.

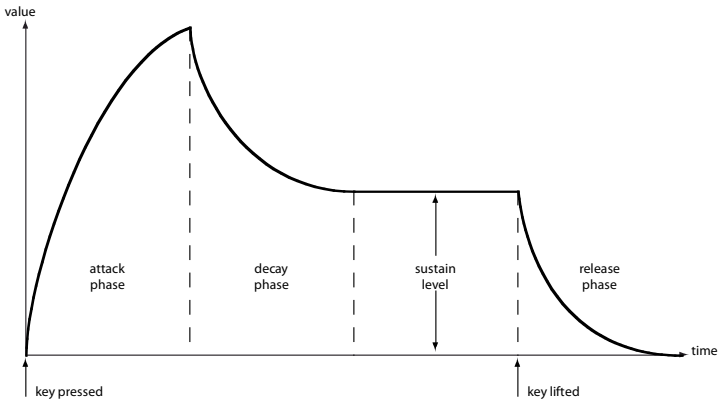
By the way: if a single filter still sounds too tame for your evil purpose, you can always patch the filters in series i.e. one after the other. This is a great way to make very bold, biting sounds similar to certain hardware filter units.

ADSR

What would a synthesizer be without envelopes to control the ebb and flow of levels? ACE features two classic ADSR generators. Like most classic synths, the main parameters are **A**ttack, **D**ecay, **S**ustain and **R**elease.

However, the envelopes in ACE have a few extra features. Firstly, the bipolar **fall/rise** knob causes the otherwise flat "sustain" to fall or rise at a definable speed.

Envelope levels can be scaled via MIDI velocity (**vel**), as well as via a source selected by right-clicking on the lower righthand knob ("..." means none i.e. undefined).



The lower lefthand knob is also user-definable, and lets you modulate the envelope times (attack, decay and release). For instance, selecting KeyFollow and setting a negative value here will make higher notes shorter (simulating the characteristics of plucked or struck instruments).

snap is a switch that causes the envelope curves to be more extreme, more "snappy" when the phases are relatively short.

ACE envelopes don't need an "invert" switch because the modulation destinations already have bipolar level controls.

Ramp Generator

If you find that two envelopes and two LFOs aren't quite sufficient for a complex patch, you should take a look at the ramp generator, which can fill either of these roles. The "ramp" in ACE is not a simple decay. It is basically an AHD (Attack Hold Decay) envelope, but with an "off time" similar to the "trapezoid" found in classic EMS synthesizers.

up - attack time
hold - time at maximum
down - release time
rest - time before repeat.

Unlike the normal envelopes, the ramp generator always holds as long as the **hold** setting. When **rest** is set to maximum, it becomes a one-shot envelope i.e it doesn't repeat. The ramp generator comes into its own whenever very slow modulation (each phase can last up to 20 seconds) or a perfectly linear envelope is required.

Mapping Generator

Mapping generators are thoroughly digital, and therefore alien to analogue synthesizers. The mapping generator is the only "digital" module implemented in ACE. Paradoxically, it is great for adding some of the lesser-known (but important) characteristics of analogue synthesis to the sound e.g. per-note tuning irregularities, non-linear modulation curves etc..

The output of the mapping generator is a socket at the bottom of the **synth** page called **mapper**. However, the controls are found in a different page - click on the **tweak** button to see them.

The mapping generator is basically a list of editable values that can be used for various modulation purposes. For instance, you can assign a separate value to every MIDI note (0 to 127) so that each one sounds different (but consistently so!) or emulate a classic "round-robin" architecture, or set up oriental tuning etc. etc..

The **number of steps** can be changed via the map's context menu (right-click on it): 2-12, 16, 24, 32, 48, 64, 96 or 128.

Map modes

- **alternate** – successive notes increment the position (play a few keys and watch it move from left to right). The default map is a list of 128 quasi-random values.
- **key** – selects a position according to which notes are played. If the map contains 128 values, these correspond directly to MIDI notes 0 to 127 (+1). If the numbers of steps is lower than 128, the list is repeated. For instance, setting 12 steps will let you tune each note (C, C#...) in all octaves at the same time.
- **map smooth** and **map quantized** – both these modes take a selectable source (including wheels and envelopes) to "scan" through the map. For instance, to transform a simple envelope into a complex one with several mounds and dips, or make abrupt timbral changes via velocity etc.. In **map smooth** mode, values are interpolated (intermediate values are created) for softer transitions. In **map quantized** mode the values are not interpolated, so this is often the better choice for e.g. sample & hold type effects or other sharp transitions.

Edit functions

Apart from the number of steps, the context menu (right-click) also contains several useful functions to make working with the mapping generator more comfortable:

copy/paste – copies the current map / replaces the current map with a previously copied one

randomize – creates a random variation from the current values

soften – smooths out any abrupt transitions in the map. Use multiple times in succession to soften more.

normalize – maximizes the range of values (-100 to +100)

straighten – draws a straight line between the first and last values

reset – sets all values back to 0

quantize 4, 6, 8, 12, 16, 24 – quantizes all values to the specified number of levels. Tip: the **12** and **24** settings are particularly useful for setting up mini-sequences – connect the mapper output to a pitch input, amount 12 or 24 semitones and use the ramp generator as mapping source.

Multiples

The "multiples" in analogue modular systems are simple mix/split units - often just a few sockets wired together. But because most modular systems have a very limited number of inputs and outputs per module, multiples are very important. Without them it would be impossible (or at least a waste of other modules) to modulate more than one parameter at a time from an envelope generator, or plug more than one or two audio sources directly into a filter.

ACE output sockets can accommodate any number of cables, and one popular modulation target in particular - filter cutoff - has two inputs as well as key follow. So classic multiples don't play such an important role in ACE - it simply doesn't need as many. Instead, the concept of the multiple in ACE has been extended, making them very versatile modules indeed...

Multiple as mixer

In this example, four signals are connected to each of the inputs. The lefthand knob is around 50% while the righthand knob is at maximum - the sum of the signals in inputs 1 and 2 is lower than the sum of the signals in 3 and 4.

You can mix up to 4 signals, but these are arranged in pairs with a common level control for each pair.

Multiple as ring modulator (RM)

In this example, the signal connected to input 1 is ring modulated (i.e. multiplied) with the signal in the **mod** (modulation) input. If another cable was connected to input 2, the sum of both inputs would be ring modulated with the mod signal.

Whenever a cable is connected to the **mod** input, the lefthand knob crossfades from the "dry" sum of inputs 1 and 2 to the ring modulated signal. In the above image, the lefthand knob is at maximum, meaning that the output (yellow cable)

is the ring modulated signal only. The value of the righthand knob is irrelevant here, as inputs 3 and 4 are not in use.

Because ring modulation is actually multiplication, the multiples can be used to scale control signals from another source. For instance, if you connect an LFO to input 1 and velocity to the mod socket, you will get **LFO level x velocity value** from the output – the harder you play a note, the more LFO signal will appear at the multiple's output.

Multiple as amplitude modulator (AM)

Another classic effect is **amplitude modulation**. This is like ring modulation except that, as well as the side bands, the output also contains the modulated original signal. While ring modulation could be written as **y = a x mod**, amplitude modulation is normally **y = a x (1 + mod)**. However, amplitude modulation in ACE's multiples is defined as **y = a x (1 – mod)**. There's a very good reason for this departure from the norm, as you will see shortly.

Amplitude modulation is achieved by using **inputs 3 and/or 4** in conjunction with the mod input. Similar to ring modulation, the righthand knob crossfades from only the original(s) to only the amplitude modulated signal.

Again (like in ring modulation), the level of a signal can be controlled via another, but in this case control is inverted – the "minus" symbol in **y = a x (1 – mod)**. If we took the previous example but used input 3 instead, the harder you played a note, the **less** LFO signal would appear at the output. So...

Multiple as balance controller

The RM and AM features can be used at the same time. In this example, the mod signal (green) controls the mix between inputs 1 and 3. If, for instance, an envelope was connected to the mod input, the envelope would crossfade smoothly between inputs 1 and 3. Please note that some signals are bipolar (e.g. oscillators) while others are not (e.g. envelopes). If you use a bipolar signal to crossfade between two other signals, you can get unexpected results due to the natures of the algorithms – see above. In such cases, you might have to e.g. bridge inputs 1 and 2 (to double the level) then set the lefthand knob to 50.00.

VCA and Effects

At the end of the synthesis chain there is always an amplifier unit (otherwise you wouldn't hear anything!). In analogue synthesizers this is usually called a VCA (Voltage Controlled Amplifier). VCAs often have their own dedicated envelopes to control transient volumes. As stereo is the de facto standard for software synthesizers, ACE has two amplifiers with associated pan controls.

The default inputs to the VCAs are filters 1 and 2 so that whenever you open a fresh instance of ACE, all you have to do to get a wide sound is to pan the two amplifiers apart and detune one (or both) of the oscillators. Such a feature would be unthinkable in older analogue synthesizers but, despite its simplicity, this tweak can have a dramatic effect.

The knobs in the VCA section are self-explanatory: volume and pan position. Above the input socket is a switch for selecting which envelope will be used - ADSR1, ADSR2 or Gate. The latter is a instantaneous on/off, and is especially useful if you want to free up an extra envelope (e.g. for classic sync sounds with separate oscillator and filter sweeps).

Why does ACE only let you use envelopes to control the final volume? In the "real" modular systems, you could use any signal (or none) to modulate VCAs, and even leave the system droning or bubbling away for hours without playing a note. Most of these old synthesizers were basically monophonic - the later polyphonic synthesizers always had envelope-controlled VCAs at the end of the chain (otherwise you would soon build up a cacophony of notes). ACE attempts to span both worlds, but you have to actively trigger and hold a note somehow. Not too much to ask, really!

After the VCA section, signals are sent to three effects to provide them with more "space" and/or presence:

Chorus

Traditionally, chorus is just a very short delay, with delay time periodically shortened and lengthened via a dedicated LFO. The pitch of the delayed signal rises and falls accordingly, similar to the Doppler Effect you hear when a fast car (or the classic example: an ambulance) passes close by.

By mixing the delayed signal with the original (dry) signal, the result is an interesting comb-filter effect similar to the movement you get from slightly detuned oscillators. The sound appears to be fuller. As the delays are very short (under 50 milliseconds), they blend with the original i.e. they aren't perceived as individual echos.

Chorus can be made more complex by using more than one delay line with differing modulation depths and LFO phases. Most chorus effects today are dual, they have two delay lines fully panned to the left and right channels.

The chorus unit in ACE has 4 different models with 4 or 8 voices. The first three are different varieties of chorus, while the last one is the classic phasing effect:

- **Chorus 1** is a 4-voice chorus with triangular LFO. Triangle modulation makes the detuning effect fairly constant and therefore more subtle than the traditional sine modulation...
- **Chorus 2** is also 4-voice, but uses a sine LFO for more dramatic movement than Chorus 1.
- **Chorus 3** is an 8-voice chorus that is particularly suitable for warm ensemble effects - but without the high noise floor usually associated with its analogue counterpart.
- **Phaser** is a classic phaser with a more subtle comb-filter effect than the chorus models. The phaser model includes a variable feedback instead of the mix parameter - higher feedback values result in a very dramatic resonant or metallic (due to atonal phase shifting) effect. Tip: Set depth to minimum for strong tonal coloration only.

The standard chorus parameters - **rate**, **depth** and **center** - affect LFO speed, LFO amount and phasing tone respectively. For the chorus models, the **mix** knob

controls the amount of delayed signal from 0 to 50%, while in the phaser model it controls the amount of feedback (at 0, you will hear phasing with zero feedback).

One special feature of ACE's chorus is that the low bass content of the signal bypasses the effect, which helps preserve the body of the sound - adding chorus in other synthesizers often means losing "oomph".

Delay

Delay is another traditional effect often used in for synthetic sounds. Unlike chorus, the delay times are long enough for repeats to be perceived as individual echos.

The first delay units used magnetic tape while the following generation was made of "bucket brigades". A large number of capacitors each provided a short delay, which added up to a single, long delay. However, both techniques had major drawbacks, the most serious of which were high noise level and lack of synchronization capability. Of course these artifacts can have their own special charm, which is why several more recent digital units try to emulate them.

The delay in ACE is a simple low-noise "digital" model with stable synchronized timing. In the '80s, when the price of memory dropped considerably, digital delays quickly displaced analogue bucket brigades - they were cheaper to manufacture, they were more precise and the sound quality was deemed to be better (to be honest here, people in the '80s also thought that digital synths sounded better than analogue).

The delay in ACE has two "taps" which are normally panned apart to widen the stereo effect. The **spread** knob controls panning: at 100 the taps are panned 100% to the left and right channels, at 0.00 both taps are in the center (mono), and at -100 the left and right taps are swapped.

feedback specifies the amount of delayed signal fed back into the delay input, which ultimately affects the number of echos. As the delay in ACE is synchronized to the clock of the host application, it is easy to set up precise rhythmic effects, and feedback can accentuate this.

Delay modes:

- **delay off**
- **8th + 8th**
- **8th groove**
- **8th dotted**
- **4th + 4th**
- **4th groove**
- **4th dotted**
- **slap**

damp reduces the high frequency content of successive echos, emulating real spaces: high frequencies are more readily absorbed (by carpets, trees etc.) than low frequencies.

EQ

The ACE equalizer is not a classic EQ, but offers enough treble and bass boost for most purposes.

In an attempt to achieve a bigger sound (often to cover for deficiencies in the basic sound), many digital synthesizers include a kind of "loudness contour" parameter. In contrast, ACE's basic sound is principally the same as analogue synthesizers: its filters don't output thin bass or irritating treble. When cutoff is set to maximum, however, ACE can deliver much higher frequencies than classic analogue synths - without having to use EQ.

As some analogue filters (notably classic Moog models) are famous for bass sounds, ACE can boost sub-bass frequencies several decibels. Be careful with the amount of sub-bass - otherwise you might need to reduce it in the mix (as is sometimes necessary when using "real" analogue synthesizers).

Modern mixes often demand ultra-crisp highs from synthesizers. Analogue synthesizers don't deliver these frequencies, but VA (virtual analogue) synths, with their purely digital filters, can. The treble control in ACE compensates for any possible losses due to the analogue-modeled oscillators and lowpass filters. ACE can sound as crisp as you like.

Other Signal Sources

At the bottom of the panel is a row of useful signal sources:

pink - pink noise (good for classic wind and wave effects)

white - white noise (brighter, great for percussion sounds)

+5V - constant "voltage". Can be used e.g. to create DC (direct current) offsets or modulate parameters beyond their normal ranges. Tip: to invert any signal, connect it to the mod input (!) of a multiple, then connect +5v to input 3 or 4.

breath - breath control (CC#02) output

m-wheel - modulation wheel (CC#01) output

p-wheel - pitch bender output. Tip: pitch bending can be set to +/-0 so that the bender can be used for other purposes without affecting pitch.

pressure - aftertouch output, either polyphonic (poly-pressure) or monophonic (channel pressure). ACE automatically recognizes which kind it is receiving. Channel pressure affects all notes equally, whereas poly-pressure is per voice (like e.g. the classic Yamaha CS80).

velocity - MIDI note velocity output

keyfollow - MIDI note number output. Below the note E3 (MIDI note 64) it is a negative value, above E3 it is positive.

mapper - mapping generator output

User Interface

If you are already familiar with software synthesizers, you will find the first steps in ACE relatively easy. Some of the controls are a little different, with additional features that may not be so obvious:

Knobs

ACE has two distinct types of knob: Unipolar and Bipolar. Unipolar knobs usually have a range of 0.00 to 100.00. Bipolar knobs usually have a range from -100.00 to +100.00, with zero in the centre (12 o'clock).

coarse adjustment: Click (and hold) on it with the lefthand mouse button, then drag up and down.

fine adjustment: Like coarse adjustment, but hold down a SHIFT key on your computer (for steps of 0.01).

mouse wheel: Owners of wheel mice can simply hover over a knob (i.e. without clicking) and move the wheel for coarse adjustments.

default reset: Double-clicking on a knob will revert to a sensible default value, usually 0.00.

MIDI learn: Clicking with the righthand mouse button (or left-click while holding down a Ctrl key) opens a context menu in which you can select "MIDI Learn". Then simply move the desired knob (or slider or whatever you have on your hardware) to link this to the knob. To remove the link, select "MIDI Unlearn" from the context menu.

Menu Switches

ACE has many small rectangular buttons (some of which double up as labels) indicating the status of various parameters. The values can be selected either by clicking on the switches, selecting a value from the context menu (via right-click) or by using the wheel of a "wheel mouse". Most of these switches can be remote-controlled by selecting "MIDI learn" from the context menu (see above).

Cables

Also known as "patch cords", these were what connected the modules of early synthesizers together, and they are still used in modern analogue modular systems. In ACE, all the outputs are dark grey, whereas the inputs are light grey. The latter often have associated knobs which are used to set the input levels. For instance, below the "phase" knob of LFO 1 is a bipolar level knob, and below this is the input for phase modulation. To connect sockets, simply drag and drop.

Note that two outputs or two inputs usually can't be connected together. However, inputs can be **daisy-chained** i.e. you can drag a cable from an empty input to one that is already in use - the output signal is sent to all inputs in the chain. The advantage: daisy-chained patches appear less cluttered.

To change inputs: click on the current input, drag the cable to another input, then release the mouse button.

To change outputs: right-click on the output and drag it to another output (you should see a straight line). Note that you can move several cables from one output at the same time.

Cable colour: this is more or less random so that overlapping cables can be easily differentiated. Colour-coding according to module was tried in prototypes of ACE, but for various reasons this was actually found to be disadvantageous.

Change the colour of a cable by clicking on the "input" end. Take care not to double-click...

Remove cables by double-clicking on the "input" end.

Patch Management

ACE comes with a collection of factory preset patches, stored as separate **.h2p** files at the following location:

MacOS X: MacHD/Library/Audio/Presets/u-he/ACE/

Windows: ...\Vstplugins\ACE.data\Presets\ACE\

There are several ways of loading a preset:

In the synth page

The main display above the oscilloscope not only shows the name of the current patch - clicking on the display will open a list containing a list of all patches in the current folder, which can then be selected.

To the left and right of the main display are arrows with which can be used to step through the patches. If you reach the beginning/end of the current folder, the last/first patch in the next folder will be selected.

In the patch page

At the top left of ACE are three buttons labelled **synth**, **tweak** and **patch**. Click on the **patch** button to open the library. The window on the left is used for selecting the desired folder. Some of the folders may be currently "collapsed" to save space and keep the list reasonably short. These have a small bright square next to the folder name - to open a collapsed folder, click on the square.

The central window of the patch manager is a list of all patches in the currently selected folder. Click on a name to load the patch. Tip: As soon as a patch has been selected in this way, this window has the "focus" and patches can be selected using the up-down cursor keys on your computer keyboard. Unfortunately, this doesn't work in all host programs - those which don't pass keystrokes on to the plugin.

Via MIDI

Programs can also be selected via MIDI "Bank" and "Program" messages, but the patches must reside in a special folder...

In the root patch folder is a folder called "MIDI Programs". You can copy or move your favourite patches into this folder, which are then selectable via standard MIDI Program Change messages. You can create new folders inside "MIDI Programs", and these are activated via MIDI Bank messages. Bank 0 accesses the patches in "MIDI Programs" itself, Bank 1 accesses the patches in the first sub-folder etc..

Note that ACE has to load all MIDI Programs (in a compressed format) into memory before Program Change will work. This means that the host application needs to be closed and opened again to update the patches. This would become tedious if you were constantly changing the contents of your MIDI Programs folder, so it is recommended that you collect a palette of sounds you want to be switched via MIDI before reloading the host application. After this, they will of course appear immediately.

Other Patch Manager Functions

Drag & Drop: Patches can be moved from one folder to another via Drag & Drop. The target folder is highlighted.

Favorites and **Junk**: Right-clicking on a patch in the patch manager opens a context menu where you can mark that patch as **favorite** or **junk**. Favorites have a small star next to the name, and junk patches are normally invisible. Or course you can make all "junk" patches visible again from the context menu, in which case they are indicated by a brown STOP sign.

Multiple selection: Several patches can be selected (for moving or marking) at once by holding down a SHIFT key.

Right-clicking on the left window also opens a small context menu containing **Create New Folder** and **Refresh Folder List**. The latter may be necessary after you have changed the contents of ACE's patch folder using OS functions (Finder, Windows Explorer).

The righthand window contains information written by the author of the current patch.

Undo/Redo

ACE has unlimited Undo and Redo functions. This applies only to the patch you are currently working on - if you switch to another patch or close ACE's window, all changes will be lost.

Enjoy!