ACE

any cable everywhere

user guide for ACE version 1.3

Urs Heckmann 2010, Howard Scarr 2012 – 2014

TOC GUI VCO LFO VCF ENV MLT OPS EFX TWK

Introduction

Installation

Go to the ACE webpage, grab the appropriate installer for your system, double-click on the downloaded file and follow further instructions. For more information, please refer to the ReadMe file included with the installer. Note: The only demo restriction is a crackling sound that appears at irregular intervals.

online resources

For u-he news, downloads, support etc., go to the u-he website
For a lively discussion about u-he products (including ACE), go to the u-he forum
For friendship and informal news, visit to our facebook page
For video tutorials and much more, go to our youtube channel
For thousands of u-he presets (commercial and free), go to PatchLib

uninstall

To uninstall, delete the plugin itself, then all associated files from the following directories (the precise locations will depend on paths chosen during installation):

Windows Patches	*...\VstPlugins\ACE.data\Presets\ACE*
Windows Preferences	*...\VstPlugins\ACE.data\Support* (*.txt files)
Mac Patches	*MacHD/Library/Audio/Presets/u-he/ACE/*
Mac Preferences	*~/Library/Application Support/u-he/com.u-he.ACE...* (*.* files)

ACE Concept and Features

modular ultrasound

Most digital synths handle audio signals and modulation signals separately. Audio is usually evaluated at a rate between 44100 and 96000 Hertz, while modulation signals update at 1000 Hz or slower (often called the "control rate" of the synthesizer).

ACE is very different in this respect. While the oscillators have 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 just the lowest of ACE's quality settings!

ACE does not differentiate between audio signals and modulation/control signals. You can plug any of the 24 signal outputs into any of the 30+ signal inputs and expect it to work just like vintage modular hardware.

So any modulation can function beyond the limits of human hearing. For instance, the **LFO**s (low frequency oscillators) can be sent above 20 kHz and still modulate e.g. the pulse width of another oscillator. This gives you a sonic freedom previously reserved for expensive analogue hardware. Both LFOs can be used as audio oscillators e.g. for **FM** (frequency modulation) sounds. Conversely, the **VCO**s (voltage controlled oscillators) can be used as alternative LFOs. Note: Any DC (direct current) is removed from VCO outputs, so when used as LFOs their shapes may not be precisely as you would expect.

analogue modeling

Wherever necessary, the non-linear characteristics of analogue circuitry has been programmed directly into the code. For instance, the filter algorithm is built around a very precise mathematical model of a hardware analogue filter – as are the basic components of the oscillators and envelope generators.

Only the LFOs, mixer, ramp generator and control-signal conversions are not analogue models. You will soon hear why: unlike its analogue ancestors, ACE is not susceptible to instabilities, and all oscillators can be synchronized to the song tempo. In ACE, even perfect host-synchronized beating between two oscillators is possible.

Non-linear distortion in the self-oscillating filters, extremely fast envelopes and modulation channels (as well as other unique details such as "Glide2" and “Tap Map”) open up a myriad of sound-sculpting techniques unavailable in other software synths.

If you really want to compare ACE to 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 these default connections can be overridden by plugging in patch cables. Many of the modules were designed to carry out a number of seemingly unrelated tasks. For instance the ramp generator can be used as an LFO, the multiples as ring or amplitude modulators, LFO1 as a waveshaper, or the filters as slew limiters.

3D rendered ACE prototype

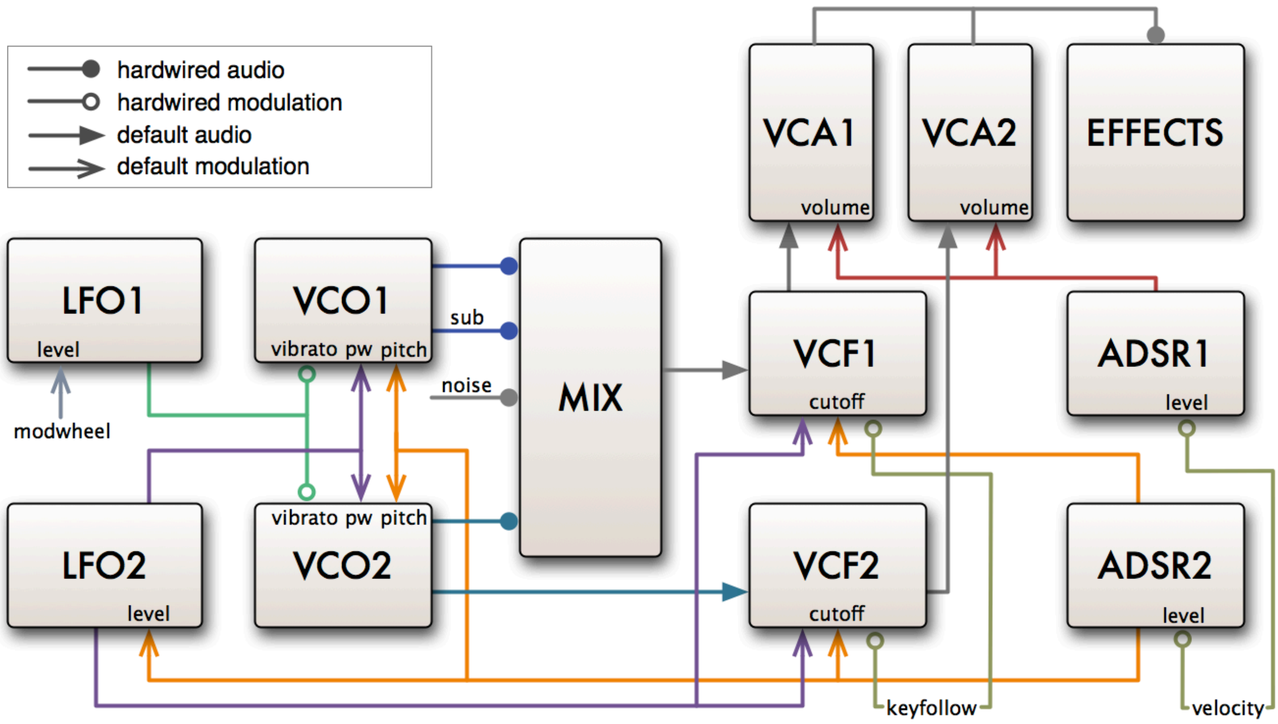
User Interface

You should feel very comfortable with ACE – it was designed to pack a lot of functionality into a compact but clear user interface:

Default Signal Flow

Like e.g. the ARP 2600 but unlike most other real modular systems, you don't need to plug any cables in before you can get a humble squeak out of ACE. That's because the modules are already connected in the typical configuration of a fixed architecture synth by default. Of course the real fun begins when you dip into your infinite supply of cables and start overriding those defaults, connecting modules together any way you like...

ACE signal flow when no patch cables are connected



VCO1 and sub-oscillator, VCO2 and white noise are MIXed and routed to VCF1, which is in turn routed to VCA1 (to right of the oscilloscope, unlabelled). VCO2 is also sent through VCF2 to VCA2.

LFO1 is hardwired as the source of vibrato for both VCOs. The output level of LFO1 and therefore vibrato depth is controlled by the modulation wheel (MIDI CC#1). LFO2 (violet) modulates both VCO pulse widths and both VCF cutoffs.

ADSR1 (red) is used as the envelope generator for both VCAs. ADSR2 (orange) modulates both VCO frequencies, both VCF cutoffs and the output level of LFO2.

Patches

ACE programs are called *patches*, a term borrowed by the modular synth community from the telephone world (calls used to be connected via jack plugs/sockets).

load

To browse through the available patches in ACE, click on the *Patch* button at the top left of ACE's window. You will see a set of panels like this:

ACE's patch window, with the two context menus open (via right-click)

Folders are listed in the left pane, patches in the centre, and patch information in the right. After having clicked on a patch, you should be able to use the up/down cursor keys on your computer to step through the others.

You can also step through patches without opening the browser – click on the arrows to the left and right of the central data display. To select from a list of all patches in the current folder only, click on the data display.

save

In the patch browser (see *load* above), make sure that the folder where you want to put your sounds is currently selected. Click on the *Save* button to the left of the data display. A dialog window opens in which you can name your sound and enter any details you would like to add (patch description, playing tips etc.). Confirm by clicking on *Apply*.

Whenever you need to create a new folder or refresh the list (e.g. if folders or patches have been added from Explorer / Finder), right-click in the left pane of the browser. Note: Simply clicking on a folder should also refresh the list.

favourite or junk?

Right-clicking on a patch in the browser will open a context menu in which you can classify patches as *Favourite* or *Junk*. Junked files disappear, but can be made visible again by selecting *show Junk* from the same context menu.

reveal in...

As the functionality of ACE'S browser is limited, the context menus let you open your operating system's file system and highlight the current folder or file: Right-click and select *Reveal in Finder / Explorer.*

the MIDI Programs folder

Local also contains a special folder called *MIDI Programs*, which is initially empty. If you put a bunch of patches (up to 128) in there, they are **all** loaded into a cache (for performance reasons) when the first instance of ACE starts. Important: changes will only take affect after restarting the host software – *MIDI Program* patches cannot be added, removed or renamed on the fly.

Individual patches are selected via MIDI program change messages. As they are accessed in alphabetical order, it is a good idea to put a number at the beginning of each name e.g. '000 rest-of-name' to '127 rest-of-name' or similar.

Banks: The MIDI Programs folder can contain up to 127 sub-folders (of 128 patches each), and these are switchable via MIDI bank select messages 1-127. The MIDI bank select message is CC#0 (ACE only interprets the MSB) – send this value, then a Program Change message.

GUI Elements

knobs

ACE has two distinct types of knob: *unipolar* and *bipolar*. Unipolar knobs only allow positive values, usually within a range of 0.00 to 100.00. Bipolar knobs also allow negative values, usually within a range of -100.00 to +100.00 with zero in the central position.

coarse control: Click+hold with the lefthand mouse button, then drag up or down.

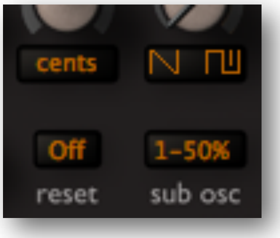
fine control: for steps of 0.01, hold down either SHIFT key before moving the knob.

mouse wheel: If your mouse has a scroll wheel, you can hover over a knob and roll the wheel for coarse adjustment. Fine control via SHIFT.

default reset: Double-clicking a knob reverts to a sensible default value, usually 0.00.

MidiLearn: Clicking with the righthand mouse button (or left-click while holding down a Ctrl key) opens a context menu in which you can select *MidiLearn*. Moving any knob/slider on your hardware controller will then create the link. To remove links, right-click and select *MidiUnlearn*. Note: accidental MIDI Learn is the reason for many erroneous bug reports – if knob values appear to magically reset themselves, try *MidiUnlearn* first!

switches / orange labels



All orange text elements are switches, and many of them do double duty as knob labels. Values can be incremented via left-click, right-click opens the list, mouse wheel movement scrolls through values.

Most of the switches in ACE can be remote-controlled by selecting *MidiLearn* from the context menu (see above).

sockets and cables

In most hardware modular synthesizers, standard jack sockets and leads are used to connect modules together. ACE's virtual cables always connect **outputs** to **inputs**:

OUTPUT sockets are brown-grey

INPUT sockets are silver-grey

To create a connection in ACE, drag and drop between sockets. Outputs will happily accommodate several patch cables, while inputs will only accept one.

Most of the modulation inputs have controls for setting the **modulation amount**. For instance below LFO1's *Phase* knob is an input socket and control for *phase modulation*.

Although you normally can't connect two inputs together, they can be **daisy-chained** by dragging a cable from an unused input to one that is already being used – the source signal is then sent to all inputs in the chain. The main advantage is that daisy-chained patches appear less cluttered. Note that editing daisy-chains can be tricky, and removing parts of the chain can leave orphan input-to-input connections.

to change inputs, drag+drop from the current input to another input.

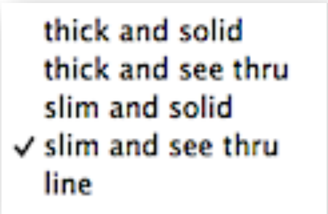
to select any source from a list, right-click on any input.

to change outputs, right-click on the output and drag it to a different output. A straight line will appear. Several cables connected to one output can only be moved together.

to remove cables, double-click (or drag away) the input end.

to change colour, click on the input end. Take care not to double-click, as this would remove the cable. Colours are initially selected more or less at random so that overlapping cables can be differentiated easily. Module-specific colour-coding was tried in early ACE prototypes, but found to have too many disadvantages.

to change cable appearance, right-click on any input socket and select one of the following options:



Oscilloscope

The oscilloscope displays a mono sum of the output (pre-effects). It is used for e.g. finely adjusting waveforms, for checking the effects of audio-rate modulation or filtering on the waveform, for viewing envelope shapes. Or simply for its entertainment value!

For many decades, oscilloscopes have proved invaluable while programming sounds, especially if the synth allows audio-rate modulation – like all classic modular systems... and ACE.

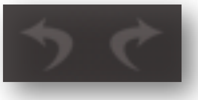
The oscilloscope in ACE is synchronized to MIDI notes as well as to zero-crossings (negative to positive transitions). The display is also updated whenever a longer scan is completed.

Because synchronization is fully automatic, the oscilloscope only requires two controls: **Freq** adjusts the horizontal resolution and **Scale** adjusts the vertical resolution.

eco, fast, glow, fire, wind

Right-clicking in the oscilloscope window lets you switch the drawing mode: *glow, fire* and *wind* add different fade-out effects at the cost of some extra CPU. These modes are also a bit more sluggish than *eco* or *fast*. Tip: If you need to keep CPU-usage down to a minimum, stick to *eco* (economy) mode.

Undo / Redo



To the right of the data display is a pair of buttons for ACE’s undo and redo functions, with a practically unlimited number of steps.

Note: Undo/Redo only works while editing the same patch – if you switch patches or close ACE's window, all edits will be lost.

GUI Size

cute	512 x 288
tiny	682 x 384
small	768 x 432
normal	**1024 x 576**
large	1536 x 864
huge	2048 x 1152
cinematic	2560 x 1140

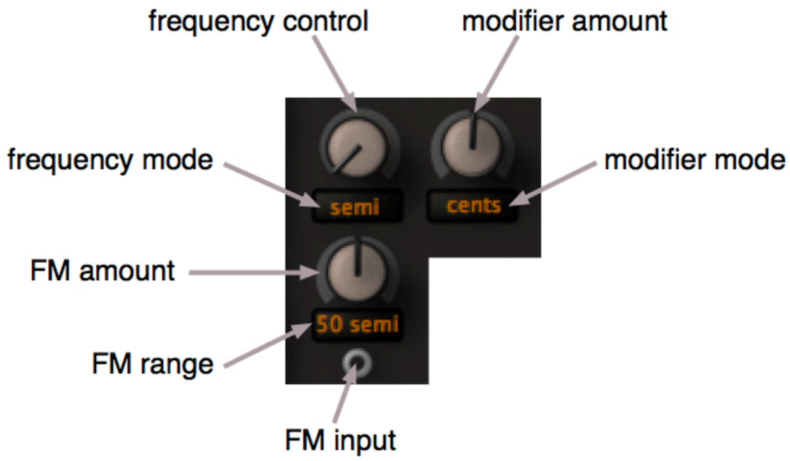
The size of ACE’s user interface can be changed to suit your personal preference / eye-sight / monitor size. To open the GUI size menu, right-click **anywhere** in the background of ACE’s window.

Temporarily switching to *huge* or even *cinematic* can be helpful if you need the best possible precision while editing the *mapping generator* (see the Tweak page). Although the *cute* setting is a bit of a joke, especially on large monitors, it still works!

Modules

VCOs (common parameters)

The two VCOs are ACE's main sound-generation modules, with a frequency range of 0 Hz to 20 kHz. All oscillators (VCOs and LFOs) include the same set of frequency parameters: Three knobs with associated switches that also serve as labels:



frequency

The frequency control has a range of 0.00 to 24.00. The frequency modes are:

semi	maximum 24 semitones above the current pitch.
partial	the first 24 overtones – octaves are at 1.00, 3.00, 7.00, 15.00
subhrm	the first 24 subharmonics – look up "Trautonium" on the web
hertz	0Hz to 24Hz. 0.00Hz is no signal because DC components are removed
sync	sync to song tempo, divider – 1.0 is a whole note, 4.0 a quarter note etc.

modifier

The modifier control range is -50.00 to +50.00 (bipolar). The modifier modes are:

cents	detunes the oscillator by +/- 50 cents
5 Hz	detunes the oscillator by +/- 5 Hertz
beats	detune in sync with song tempo. +4.00 = one extra cycle per quarter note
mtply	frequency is multiplied (from 0 to 50) or divided (from 1/1 at -1.00 to 1/50th)

FM amount

The amount of frequency modulation from the FM input. Available FM ranges are:

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

reset

The oscillators in analogue synths run *continuously*, they never stop. In digital synths, oscillators are not computed until a note is played. This means they either start at a random phase (most similar to analogue) or at the same phase every time (for consistent attack). The VCOs in ACE have a switch offering both options:

off – random phase
on – fixed phase

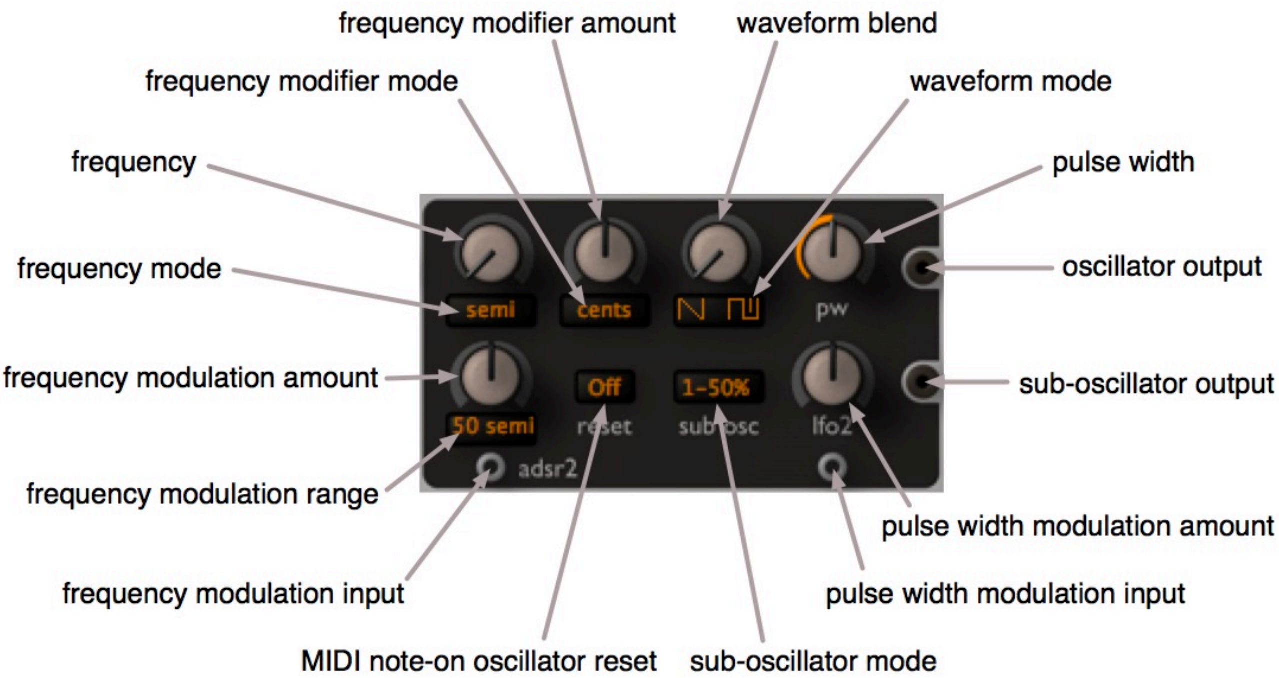
waveform blend

Crossfades between the sawtooth and pulse waves (or peak and triangle in VCO1).

pulse width (pw)

The width of the pulse wave is variable from 0% to 100%. Any signal can be used for pulse width modulation (PWM), even an audio oscillator. The default source is LFO2.

VCO1



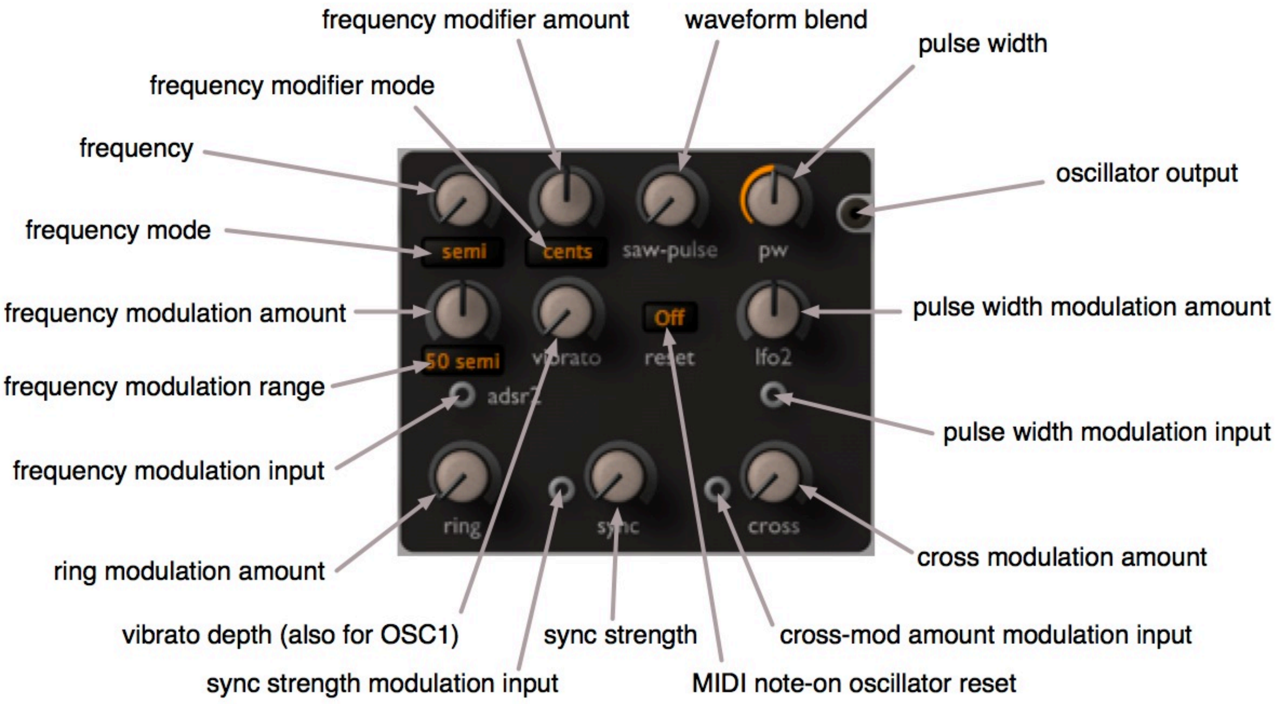
waveform mode

As mentioned above, both VCOs can blend between sawtooth and pulse waves. VCO1 also lets you switch to the mellower **peak / triangle** (click on the waveform icon).

sub-oscillator

VCO1 includes a sub-oscillator with its own separate output socket. Click on the **sub-osc** switch to select a 50% pulse wave (perfect square) pitched 1 or 2 octaves below the main oscillator, or a 75% pulse wave 2 octaves below.

VCO2



vibrato

Frequency modulation **for both VCOs** from LFO1, hardwired. Although it also affects VCO1, the position of this knob (i.e. in the VCO2 panel) was a necessary compromise.

ring

Ring modulation. This knob cross-fades between pure VCO2 and VCO2 ring modulated with VCO1. Depending on the waveform and interval between the two oscillators, ring modulation can create metallic sounds, nasal sounds – even rhythms when VCO1 is set to e.g. *sync* mode.

sync

Turn this knob to maximum for the standard ‘hard sync’ found on most other synths. The phase of VCO2 is not only reset when it completes its own cycle (as always), but also whenever VCO1 completes a cycle. The pitch of VCO2 is normally set higher than VCO1, and VCO2 is often modulated by an envelope or LFO to sweep the effect. Hard sync can deliver sounds that are very rich in harmonics without losing the fundamental pitch (of VCO1).

Turn the sync knob down for a special kind of ‘soft sync’: Again, the phase of VCO2 is reset by VCO1 – but not to 0°. The phase of VCO2 moves by a certain proportion of its current value, e.g. 50%, which lets you create pure-interval overtones. Experiment with the sync knob and the interval between the two oscillators – you should quickly discover some very interesting overtones and quasi-chaotic effects!

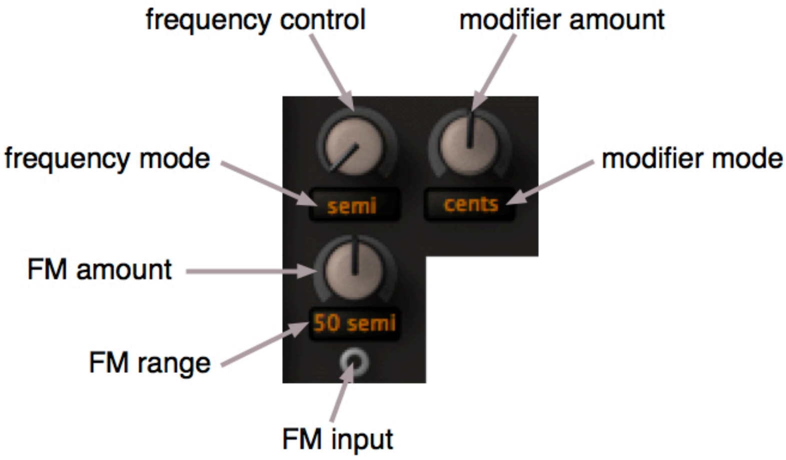
Connecting a cable to the sync modulation socket effectively replaces a +5V default modulator. Tip: Try patching velocity or an envelope into the sync input.

cross

Short for “cross modulation”. In ACE this means analogue FM (frequency modulation), with VCO1 modulating VCO2. Connecting a cable to the cross modulation socket effectively replaces a +5V default modulator.

LFOs (common parameters)

The label **LFO** only describes the default function of these modules, as they are actually full range (0Hz–20kHz) i.e. from static to inaudibly high! All oscillators have the same set of frequency parameters – three knobs with associated switches that also serve as labels:



frequency

The frequency control has a range of 0.00 to 24.00. The frequency modes are:

semi	maximum 24 semitones above the current pitch.
partial	the first 24 overtones – octaves are at 1.00, 3.00, 7.00, 15.00
subhrm	the first 24 subharmonics – look up "Trautonium" on the web
hertz	0Hz to 24Hz. 0.00Hz is no signal because DC components are removed
sync	sync to song tempo, divider – 1.0 is a whole note, 4.0 a quarter note etc.

modifier

The modifier control range is -50.00 to +50.00 (bipolar). The modifier modes are:

cents	detunes the oscillator by +/- 50 cents
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beats	detune in sync with song tempo. +4.00 = one extra cycle per quarter note
mtply	frequency is multiplied (from 0 to 50) or divided (from 1/1 at -1.00 to 1/50th)

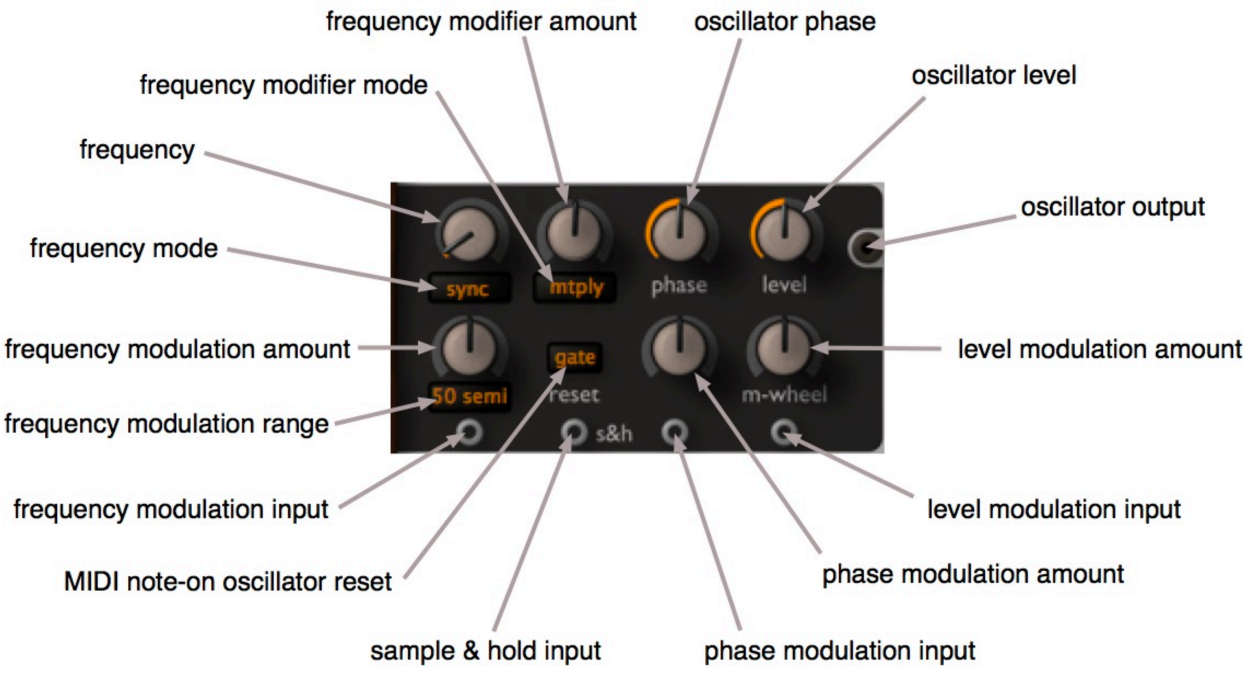
FM amount

The amount of frequency modulation from the FM input. Available FM ranges are:

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

LFO1

LFO1 normally generates a pure sine wave. The LFO1 specialities are vibrato, phase modulation i.e. classic FM, sample & hold... and even waveshaping.



phase

The **phase** knob adjusts the phase position i.e. where in its cycle the waveform will start whenever the LFO is *reset* (see below). LFO1 has a **phase modulation** input – connecting another oscillator here gives you classic FM sounds (all so-called "FM" synths were actually using phase modulation, and should have been called "PM" synths). Of course the phase modulation source can be LFO1 itself, which skews the sine wave towards something very similar to a sawtooth.

Note: The phase knob has a very different role in sample & hold mode (see below).

level

LFO1 output level. Both LFOs have amplitude modulation inputs, and the default source for LFO1 is the modulation wheel (*m-wheel*) for e.g. quick “vibrato via mod wheel”.

reset

This switch determines whether the LFO phase is reset by MIDI note-on events. Note: If the frequency mode is *sync*, LFOs are also reset in sync with the host program.

free not reset, runs continuously (“monophonic”)
gate per-voice reset whenever a note is played (“polyphonic”)

sample & hold

If anything is connected to the *s&h* input, LFO1 switches into **sample & hold mode**, and samples the input at its own “clock speed". For vintage random effects, connect noise here.

In s&h mode the *phase* knob becomes a lag processor, smoothing out jumps between successive values. At very high LFO1 rates, the phase parameter acts like the cutoff control of a lowpass filter (but in the opposite direction). If you find LFO1 strangely silent in s&h mode, set its *phase* closer to zero.

a few LFO1 tricks

random modulation: Connect white noise ("white") to the s&h input and use LFO1 to modulate e.g. VCO frequency or VCF cutoff.

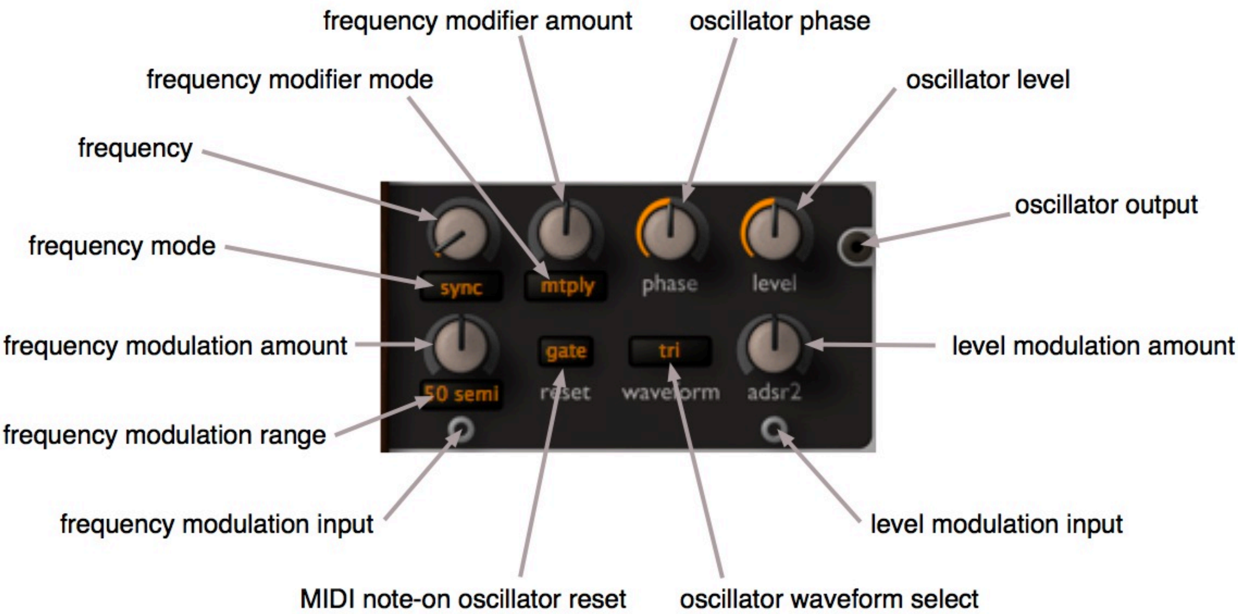
sample rate reduction effects: Start with the default patch and drag a cable from LFO1 output directly to one of the VCAs. Connect an audio signal (e.g. a VCO) to LFO1’s s&h input. Set the frequency mode to *semi* and the modifier to *mtply*. Set LFO1 phase to around zero or you won't hear anything. Now try different multiplication factors (modifier values) between 1.50 and around 30. LFO1 adopts the pitch of the sampled oscillator and delivers a “rougher” version of same. To see the steps in the waveform, turn the oscilloscope frequency way down.

waveshaping: Although the VCFs can deliver plenty of distortion, especially when connected in series, you can also use LFO1 as an extra waveshaper – try this: Start with the default patch and drag a cable from LFO1 output directly to one of the VCAs. Set *semi* and *mtply* modes, and a multiplication factor of 0.00. Switch the *reset* to *gate* (LFO1 doesn't oscillate now) and set the phase to 0.00 (LFO1 always resets to 0° now).

Then patch the signal you want to process (e.g. VCO1) into LFO1's phase modulation input. Turn up the amount – there’s your *sine waveshaper*! Change the phase to make the effect asymmetrical. By the way, the sampled signal doesn’t have to be a VCO – you can use this method to alter the shape of any signal: try waveshaping an envelope.

LFO2

Instead of phase modulation and s&h, LFO2 offers a much wider variety of waveforms than LFO1... which makes LFO2 easy to use as a third audio oscillator.



phase

The phase knob adjusts the phase position i.e. where in its cycle the waveform will start whenever the LFO is *reset* (see below).

level

LFO2 output level. Both LFOs have level/amplitude modulation (AM) inputs and associated amount controls, and the default source for LFO2 is ADSR2.

reset

This switch determines whether the LFO phase is reset by MIDI note-on events. Note: If the frequency mode is *sync*, LFOs are also reset in sync with the host program.

free	not reset, runs continuously (monophonic)
gate	per-voice reset whenever a note is played (polyphonic)

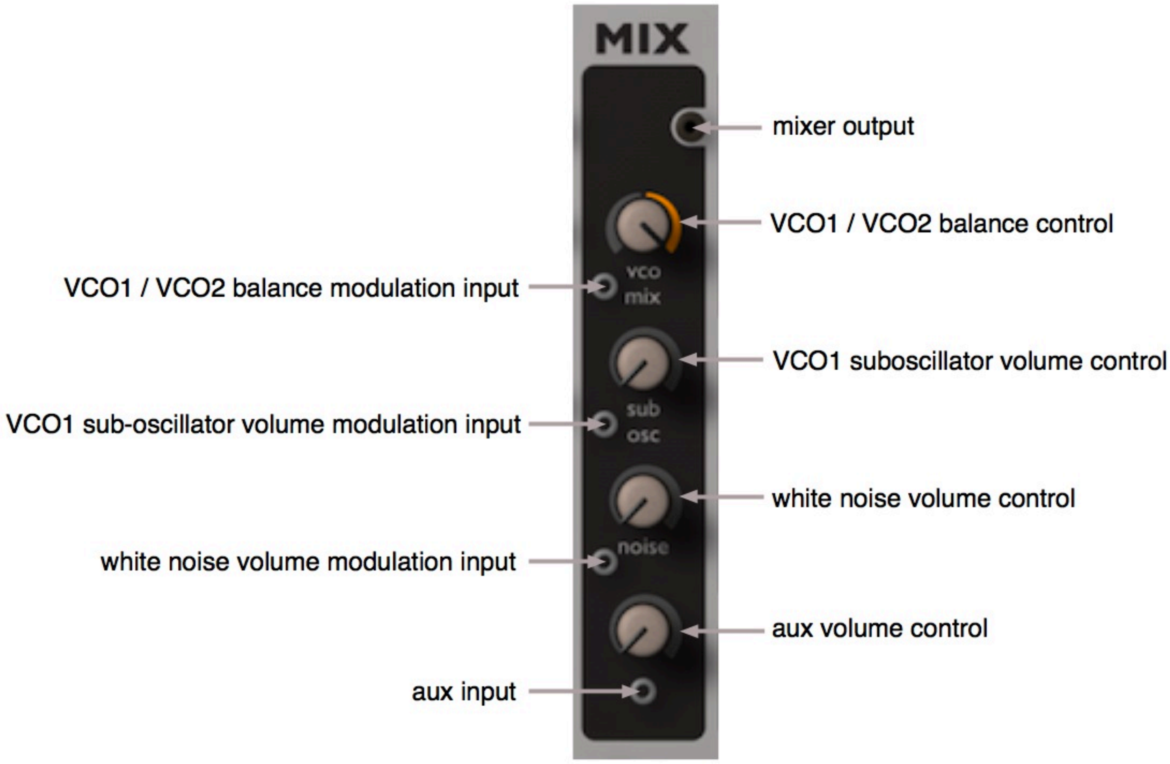
waveform

The first four LFO2 waveforms are standards, but the last in the list is very special:

sine	sine wave, pure
tri	triangle wave, pure
saw	sawtooth wave, bright
square	square wave, hollow
tap map	mapping generator, interpreted as a waveform! For details of the mapping generator, go here. Note that setting the LFO2 waveform to *tap map* doesn't prevent the mapping generator from being used as a modulation source (its other role) at the same time.

MIX

In the middle of the window is a mixer that serves as the default link between the main sound generators and the sound processing in ACE. The MIX module also includes amplitude modulation inputs. Note that its output is sent to VCF1 by default.



vco mix

The upper knob controls the relative levels between VCO1 and VCO2. The central position (0.00) is a 50-50 mix of both VCOs.

Of course the balance modulation input will also accept audio rate signals: remember *any cable everywhere!* Tip: For bipolar modulation sources (LFO, VCO) set the mix knob to the centre, for unipolar sources (e.g. mod wheel, ramp) set it to maximum.

sub osc

VCO1's sub-oscillator.

noise

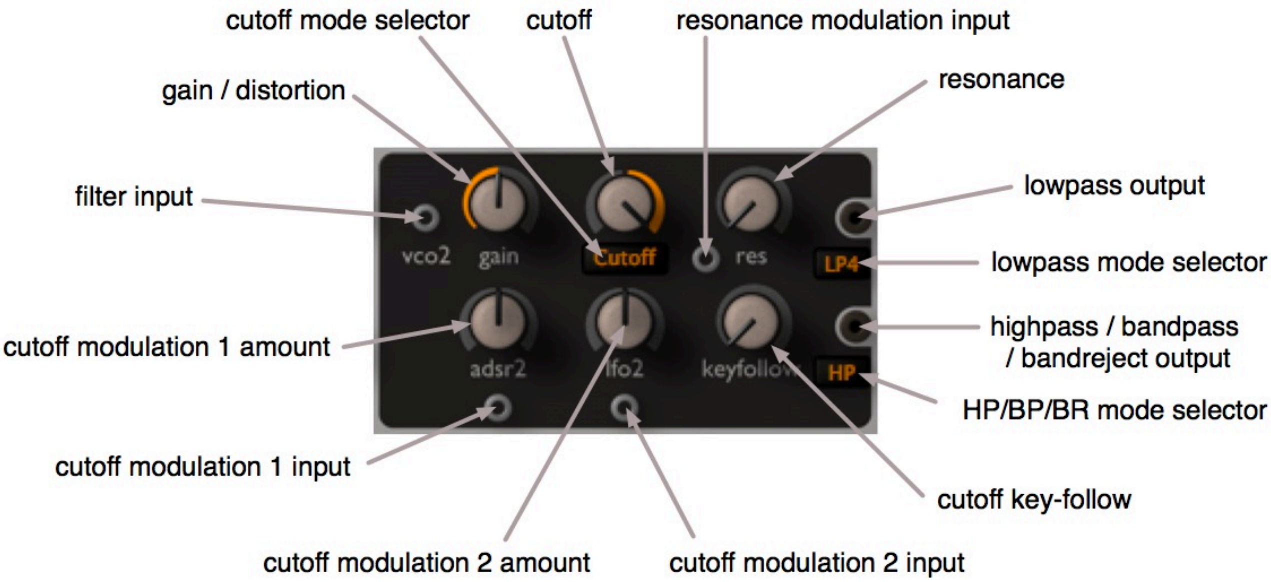
White noise. Tip: A small amount of white noise mixed into a pad patch can give the filters and chorus more frequencies to work with, making the sound fuller.

aux

The unlabelled knob is the level control for the auxiliary input at the bottom of the mixer. Connect anything you like here: pink noise, a pitched LFO – or even VCF1 for instant *filter feedback*.

VCF

ACE's two filters are almost (but not quite) identical. The screenshot here shows VCF2, with its cutoff mode selector and bipolar cutoff knob...



The filters in ACE have several properties normally associated with analogue hardware only. For instance, they can easily be overdriven without sounding harsh. Unlike classic hardware filters, strong overdrive in ACE won't necessarily kill the resonance. Just turn it up – there's plenty of headroom there.

Especially around the self-oscillation threshold, where the resonance appears to struggle with the oscillators for control over pitch, there are surprising opportunities for organic/chaotic sound design. Depending on the input signal and its gain, it can even sound as if the input is actually *modulating* cutoff. Experimentation is the name of the game here!

The underlying cascade filter architecture gives you different filter types in parallel, just like hardware multimode filters. In ACE however, *all* types are capable of resonance and even self-oscillation.

Tip: if a single filter still sounds too tame for your evil purposes, you could try patching the filters in series i.e. one after the other, and increasing the gain of the second filter. This is a great way to make very bold, biting sounds similar to hardware filter units.

gain

VCF input level (negative values) and overdrive (positive values).

Tip: for typical screaming distortion (TB303 etc.), use another VCF in series. Set it to LP1 mode, with maximum cutoff and very high gain.

cutoff

VCF1: Cutoff frequency is measured in semitones from 0.00 to 150.00 (12 octaves) and the modulation range is +/-150 semitones. Note: the input / knob at the bottom left of the VCF panel also modulates cutoff, *not* gain.

VCF2: Instead of a simple positive-only cutoff, VCF2 has 3 modes and bipolar control:

cutoff like VCF1, but bipolar

offset VCF2 cutoff follows VCF1 – *including any modulation* – but shifted negative or positive. This means that VCF2 cutoff can be modulated directly by up to *four* sources: two within the VCF2 panel and two adopted from VCF1.

spread like offset, but also affects VCF1 cutoff *in the opposite direction*.

keyfollow

Keyfollow causes cutoff to follow the MIDI note (pivoting around E3 – the only note that always remains unaffected by keyfollow). If keyfollow is set to maximum, cutoff follows MIDI notes 100%, like the VCOs.

resonance

The **res** range is 0.00 to 100.00. Although self-oscillation can start around 50.00, the actual amount of resonance depends on the level of the input signal (see **gain** above), so a generous range was necessary here. Resonance can be modulated by connecting a signal to the socket to the left of the *res* label (effectively replacing a +5V default).

outputs

Each filter has two parallel outputs. The upper one offers four grades of lowpass...

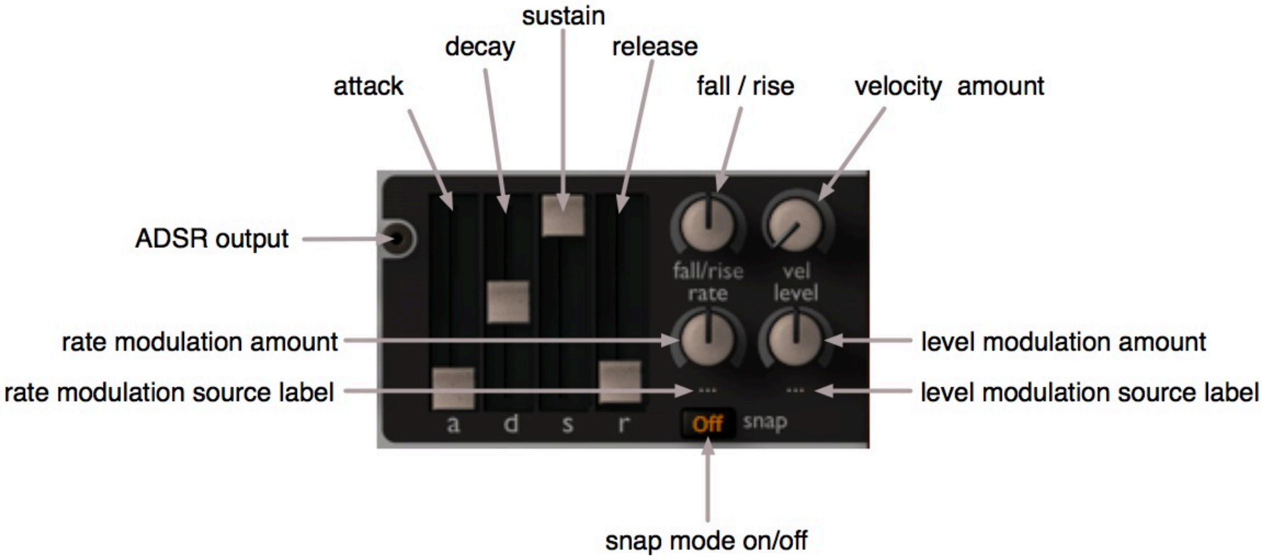
LP1 6dB/octave (1-pole lowpass)
LP2 12dB/octave (2-pole lowpass)
LP3 18dB/octave (3-pole lowpass)
LP4 24dB/octave (4-pole lowpass)

...and the lower one has three other types:

HP high pass
BP band pass
BR band reject (notch)

ADSR

What would a synthesizer be without envelopes to control the ebb and flow of levels? ACE has two identical envelope generators:



a, d, s, r

Like the vast majority of synthesizers, the main envelope parameters are **A**ttack time, **D**ecay time, **S**ustain level and **R**elease time. But ACE also has a few extras...

fall/rise

Firstly, the bipolar **fall/rise** knob causes the normally flat sustain to fall or rise at a defined rate. There's a parameter in the Tweak page called fall/rise range that limits how far towards zero / maximum the sustain level will fall / rise.

rate modulation (...)

The lower lefthand knob is user-definable (hence the '...' default label). This parameter lets you modulate the envelope rates (attack, decay and release). Right-click on the knob to select a modulation source. For instance, selecting KeyFollow and setting a negative value here will make higher notes shorter, simulating the characteristics of plucked or struck instruments.

velocity amount (vel)

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 yet i.e. undefined).

level modulation (...)

The lower righthand knob is user-definable (hence the '...' default label). Lets you modulate the overall ADSR level. Right-click on the knob to select a modulation source.

snap

This switch makes the decay and release more extreme, more "snappy" if the envelope stages are relatively short.

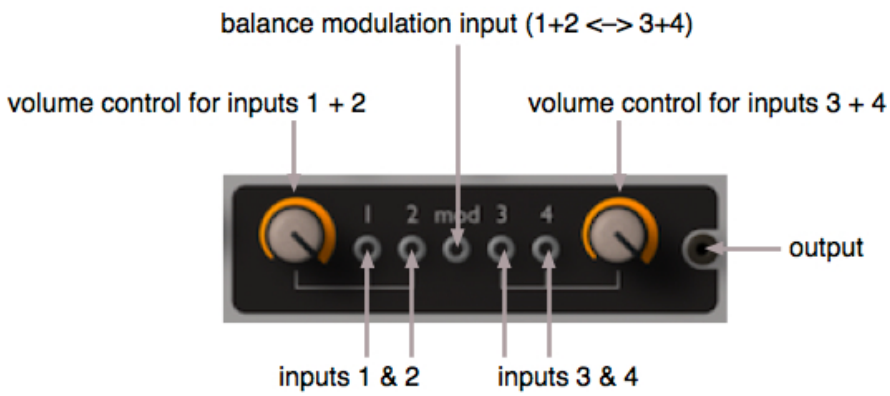
Ramp Generator

If you find that two envelopes and two LFOs aren't quite enough for a complex patch, you could take a look at the ramp generator – it can fill either of these roles quite well. The ramp is not a simple decay, it is a linear attack-hold-decay (AHD) envelope with an off time – so its closest relative is the *trapezoid* of classic EMS synthesizers *Synthi A* and *VCS3*.

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

Unlike standard envelopes, the ramp generator will stay at maximum for the period set by *hold*. If *rest* is set to maximum, the ramp is a one-shot envelope i.e it does not repeat. Note: the Tweak page includes a parameter called ramp clock that sets the ramp segments to either seconds or one of two different sync values.

Multiples



The *multiples* you find in most analogue modular systems are simple mix/split devices, often just four sockets bridged together. As most modular synthesizers have a very limited number of inputs and outputs per module, multiples are important – without them it would be impossible to modulate more than one parameter at a time from e.g. an envelope generator, or plug more than one or two audio sources into a filter.

Because ACE's output sockets can accommodate any number of cables, the humble multiple underwent a serious redesign, finally emerging as something **much** more useful!

simple mixing

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, arranged in pairs with a common level control for each pair.

ring modulation (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.

amplitude modulation (AM)

Another classic effect is amplitude modulation (AM). This is like ring modulation except that, as well as the side bands, the output also contains the modulated original signal.

While RM could be written as $y = a \times mod$, AM is normally $y = a \times (1 + mod)$. However, AM in ACE's multiples is defined as $y = a \times (1 - mod)$. 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 $y = a \times (1 - mod)$. 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.

balance processing

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 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 may get unexpected results due to the natures of the algorithms. In such cases, you might have to e.g. bridge inputs 1 and 2 to double the level and set the lefthand knob to 50.00.

signal inversion

To invert a signal, connect it to the *mod* input of a multiple and +5v to input 3 or 4.

VCA

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 synths 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 an instantaneous on/off, and is 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 you having to play a note. Most of these old synthesizers were monophonic – all the early polyphonic synths had envelope-controlled VCAs at the end of the signal chain. ACE tries to span both worlds, but you **do** have to play a MIDI note somehow... not too much to ask, really!

Other Signal Sources

At the bottom of the panel is a row of signal sources that don't require any knobs:

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.
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 you can use the bender for other purposes without affecting pitch.
pressure	aftertouch output, either polyphonic (poly-pressure) or monophonic (channel pressure). ACE automagically recognizes which type it is receiving. Channel pressure affects all notes equally, whereas poly-pressure is per voice.
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

Operation Settings

The top left panel contains controls that are not specific to individual modules. ACE does not include global settings i.e. all values are saved and recalled with each patch.

output

ACE's master volume control, after the effects.

Polyphony and Quality

mode

Determines the polyphony and how MIDI notes are interpreted:

poly polyphonic
mono monophonic, retrigger
legato monophonic, no retrigger
duo duophonic

voices

Mainly relevant for poly mode (see above), this parameter sets the maximum number of notes that can be played before voice-stealing occurs.

few 4 voices
medium 8 voices
many 16 voices

quality

draft, standard, good, accurate

The quality switch is mostly for reducing CPU load, an important consideration in ACE. Tip: start with *good* and compare the sound with other quality settings. Depending on modulation rates, filter distortion and/or whether the sound of high notes is important, *standard* or *draft* can be used without compromising the result.

multicore

Switching this on causes ACE to distribute voices across available CPU cores, which will usually allow more voices to be played without overloading the CPU. This mode appears to work well on processors such as Intel i5 and i7, but please note that performance can even be reduced if your CPU is older. Also, active *multicore* might interact with multi-threaded hosts in an unpredictable fashion (not yet observed).

stack

The number of voices played in unison.

Up to 8 voices can be stacked for a very powerful unison effect like a few classic polyphonic synths e.g the Oberheim OBXa. However, ACE can still be played polyphonically. This is not a "supersaw", it is true unison i.e. the entire voice is multiplied.

Of course this feature eats a lot of CPU power, but we think it is worth it. For instance, multiple filter distortion on one note is more lively than a single filter could possibly be.

Using the stacked voice tuning knobs in the Tweak page, the 8 voices can be detuned within a range of +/- 24 semitones.

Pitch Settings

pb up / down

Separate pitchbend ranges, from 0 to +/- 24

drift

If *on*, voices are slightly detuned against each other for a fuller, more lively sound.

transpose / tune

Transpose adjusts the overall pitch over a +/- 2 octave range.
Tune also adjusts the overall pitch, but the range is only +/- 50 cents (half a semitone).

glide controls

Glide or ‘portamento’ is a smooth pitch transition between consecutive notes. In ACE it also affects the ‘Key Follow’ modulator.

glide controls either the *time* or the *rate*, depending on the state of the *glide mode* switch (see below)

glide2 offset relative to the **glide** value, applied to LFO2, VCO2 and VCF2 only. Careful use of this parameter can really bring 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. This means that the glide can start "already half way there" for a more subtle effect. Tip: set the range to very low values for natural intonation effects.

glide mode *time:* However far apart notes are, glide will take exactly the same amount of time. *rate:* Glide is proportionally slower when notes are further apart.

Effects Section

The upper-right panel controls ACE's three post-VCA effects: Chorus, Delay and Tone...

Chorus

Traditionally, chorus is a simple very fast delay periodically shortened and lengthened by a dedicated LFO. The pitch of the delayed signal rises and falls like the *Doppler Effect* you hear when a fast car (or the classic example: an ambulance) passes by.

Mixing the delayed signal with the original dry signal results in a warm comb-filter effect similar to slightly detuned oscillators. As the delays are under 50 milliseconds, they blend well with the dry signal i.e. they aren't perceived as individual echos.

Chorus can be made richer by using more than one delay line with different modulation depths and LFO phases. Most of today's chorus units are stereo, using two delay lines fully panned apart. The one in ACE has four different models – 3 varieties of chorus (4 or 8 voices) plus a classic phaser:

mode

Chorus 1 is a 4-voice chorus with triangle LFO. Triangle modulation keeps the detuning effect fairly constant and therefore more subtle than Chorus 2....

Chorus 2 is also 4-voice, but has a sine LFO for more dramatic movement.

Chorus 3 is an 8-voice chorus for lush ensemble effects – of course without the high noise floor typical of the original hardware units.

Phaser is a classic phaser with a more subtle comb-filter effect than the chorus models. The phaser 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 the depth to minimum for strong tonal coloration but no movement.

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 usually means losing a lot of "oomph".

rate

Speed of the effect’s own modulation LFOs

mix

In all *chorus* modes, this knob controls the amount of delayed signal (0 to 50%), in *phaser* mode it controls the amount of resonance

center

Nominal delay time before modulation, affects the overall **tone** of the effect

depth

LFO modulation depth

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 next (solid state) generation was made of *bucket brigades* – a large number of capacitors each provided a short delay, which were arranged in series to produce a single long delay. Both techniques had major drawbacks, the most serious of which were noise and lack of synchronization capability. However, these units do have their own special charm, which is why digital emulations of tape and bucket brigade delays are still available, as hardware or plugin effects.

In the '80s, when the price of memory dropped considerably, digital delays quickly displaced analogue – they were cheaper to manufacture, they were more precise and the sound quality was deemed better. However, most people in the 1980s were convinced that the early digital synths sounded much better than analogue... how times change!

ACE’s delay is a simple low-noise digital type with two taps and synchronized timing...

times

Click on the button to select delay times/patterns:

off, *8th + 8th, 8th groove, 8th dotted,*
4th + 4th, 4th groove, 4th dotted, slap

mix

Dry/wet mix for the delay.

spread

The **spread** knob controls stereo width: at 100 the taps are panned 100% to the left and right channels, at 0.00 both taps are in the centre (mono), and at -100 the left and right taps are swapped.

feedback

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

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.

Tone controls

ACE doesn't have a classic EQ, but the pair of tone controls offer enough high and low boost for most purposes. In an attempt to achieve a bigger sound (often to make up for deficiencies in other areas), many digital synthesizers include a kind of "loudness contour". In contrast, ACE's basic sound is principally the same as analogue synthesizers: its filters do not deliver irritating treble or lifeless bass...

bass

As some analogue filters (notably classic Moog models) are famous for bass sounds, ACE lets you boost sub-bass frequencies by several decibels.

treble

Modern mixes often demand ultra-crisp highs from synthesizers. Analogue synths 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.

Effects On/Off

In the top righthand corner of the effects panel is a button that **globally** switches all effects on or off, so you can browse through the "raw" patches without effects. Tip: Remember to switch it back on afterwards!

Tweak Page

The **Tweak Page** is where you will find the thoroughly digital **mapping generator** plus anything else that doesn't fit on the main *synth* panel or is of secondary importance...

Mapping Generator

Mapping generators are alien to analogue synthesizers, and the mapping generator is the only "digital" type module implemented in ACE. Paradoxically, it is great for adding some of the important characteristics of analogue synthesizers – per-note tuning irregularities, non-linear modulation curves etc..

The mapping generator is a list of 128 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 consistently different, you can emulate a classic *round-robin* architecture or pan stacked voices apart etc..

The mapping generator actually has two outputs: Firstly, the socket at the bottom of the *synth* page labelled **mapper**. Secondly, the LFO2 output when in tap map mode.

map modes

Typical uses for ACE's mapping generator: quasi-random, sequencer, modulation-shaper

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 hills and valleys, or make abrupt timbral changes via velocity etc.. In *map smooth* mode, the values are interpolated for softer transitions. In *map quantize* mode the values are not interpolated, so this is usually the better choice for e.g. sequencer-type effects or sharp transitions.

Note: A **mapping source** is only used in the *map smooth* and *map quantize* modes. It is ignored in *alternate* and *key* modes...

alternate – successive notes increment the position (play a few keys and watch the highlighted bar step from left to right). The default map is a list of 128 quasi-random values, but even a two-value map can be useful.

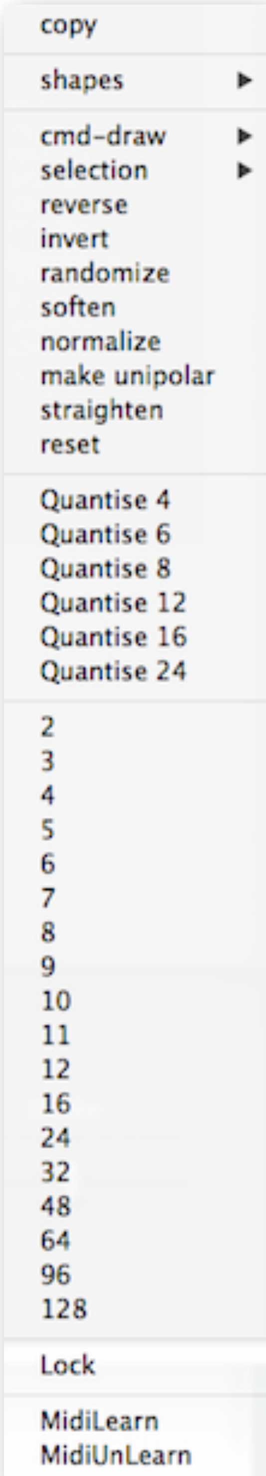
Example: To pan stacked voices apart, connect the mapping generator to VCA pan modulation, set *stack* to 2 and the number of mapping generator steps to 2, set the map values to maximum and minimum and the mode to *Alternate*.

key – selects a position within the map according to which notes are being played. If the map contains 128 values, these correspond directly to MIDI notes 0 to 127. If the numbers of steps is less 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.

Drawing and Selection

To edit the map, simply draw in the window. For straight lines, hold down **ctrl** (Win) or **alt** (Mac) while drawing. To make a selection, hold down the **SHIFT** key: the functions (see below) are restricted to the selection. To deselect, either click in the background i.e. away from the selection, or choose 'deselect' from the *selection* sub-menu – which brings us to the context menu...

Context menu



Right-clicking on the Mapping Generator's edit window gives you access to various map editing tools. This newly expanded version is provisional – selecting tools will be much easier in future. For the time being, remember the following: holding down the **SHIFT** key lets you select a subset of the values in the window, and the **ALT** (Win) or **CMD** (Mac) key lets you apply the functions.

Experiment with all these functions:

copy / paste – copies the current map to the clipboard, or replaces the current map with a previously copied one

shapes – ready-made *ramp*, *triangle*, *sine*, *cosine*, *root, quadric*

alt-draw or **cmd-draw** – sets the drawing mode to *erase* (zero), *scale* (multiply), *shift* (2D move) or *warp* (2D bend).

selection – *deselect*, *invert*, *shift left*, *shift right*, *every 2nd*, *every 3rd*, or *every 4th* (if nothing is selected, only the 'every' options will appear).

reverse – flips the current window or selection horizontally

invert – flips the current window or selection vertically

randomize – a random variation based on current values

soften – removes abrupt transitions

normalize – scales all values in the current window or selection so that the lowest and highest are at the bottom and top.

straighten – draws a straight line between the first and last values in the current window or selection

reset – sets all values in the current window or selection to zero

quantize 4, 6, 8, 12, 16, 24 – quantizes all values to the specified number of levels. Tip: the 12 and 24 settings are useful for setting up little sequences: connect the mapper output to a pitch input, set the amount to 12 or 24 semitones then use the *ramp generator* (with minimum *rest*) as mapping source. Simpler still: use LFO2 with the *tap map* waveform!

2 to 12, 16, 24, 32, 48, 64, 96 or 128 – sets the number of visible steps in the mapping generator. Note that the original data is retained when the number of steps is reduced.

Stacked Voice Tuning

This block of knobs is used for tuning the individual voices within a stack. The total range for each voice is +/- 24 semitones, so as well as setting up mild detuning you can even create massive one-finger chords. For fine tuning, hold down the SHIFT key on your computer keyboard.

Important: Stacking voices will significantly increase the CPU-hit per played note.

Circuit Bending

slop

Slop has become the insider term of choice for tuning instability (this is generally attributed to Dave Smith of *Sequential Circuits* and *Dave Smith Instruments*). The slop parameter in ACE adds slow random detuning. Note: When slop is at maximum, the oscillators will sound obviously out of tune, while low values can be extremely subtle.

crosstalk

Once considered an even less desirable feature of analogue synths than tuning instability, even crosstalk has its own special charm in this digital age. Quoting from Wikipedia:

Crosstalk is any phenomenon by which a signal transmitted on one circuit or channel of a transmission system creates an undesired effect in another circuit or channel.

osc cap failure

Finally, let's make sure the capacitors in your emulated analogue synth sound like they will need replacing very soon. No joke, that's what this parameter emulates... try it!

filter reset

none.............quasi ‘free-running’, transients may have a little analogue randomness.
full ....reduces randomness by flushing the filter at the beginning of each note. Note: Presets with self-oscillating filters might fade in more slowly.
full+click ....same, but with an extra transient. Use for ‘self-oscillation’ presets.

Envelope Tweaks

This area is still fairly simple in the current version of ACE – future revisions might include a few extra parameters – there is certainly enough room!

fall/rise range

These two knobs affect the level after fall/rise for each of the ADSRs. Normally this would be either maximum (positive fall/rise) or zero (negative fall/rise). The range knob lets you set a percentage of the difference (from the nominal sustain level) instead of always 100%.

singing envs

Switching **singing envs** on causes the envelope of a new voice to start at the current level of the stolen voice’s envelope instead of at zero, emulating the typical behaviour of classic analogue envelopes more closely.

ramp clock

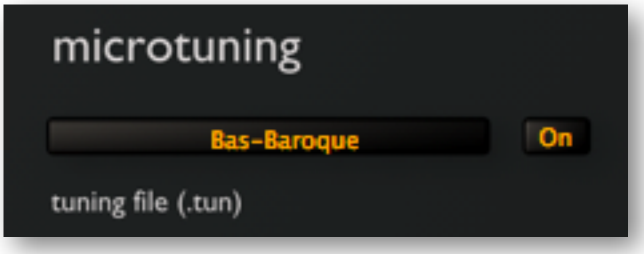
This switch sets the maximum of the ramp generator stages to either absolute time or host-synchronized values:

0-20 sec absolute time, maximum = 20 seconds per stage
1/4 sync synchronized, maximum = one quarter note per stage
4/4 sync synchronized, maximum = one whole note per stage

Note: The ramp generator scaling is absolutely linear, so setting e.g. an attack time of 25 will divide the maximum by four (i.e. it will be 5 seconds, 1/16 or 1/4 long)

Microtuning

ACE supports standard .TUN microtuning tables. Hundreds or even thousands of tuning tables are available online, and most of them are free.



Put .tun files into the following folder on your hard drive:

Win: *...\VST Plugins\u-he\Ace.data\Tunefiles*
Mac: *~/Library/Application Support/u-he/Tunefiles*

*O*r similar, depending on paths chosen during installation.

The End