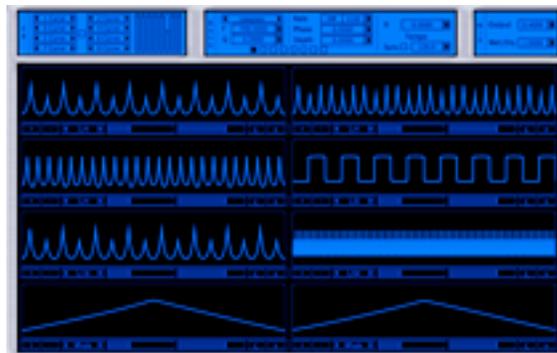


Music Unfolding Collective 8 Audio Units User Guide



Requirements:

OS X 10.6 or higher is required. F8-V2, Mod8, and Res8 require an Audio Unit (AU)-capable host. A host that supports Cocoa user interfaces (UI) for AU's is needed. The sequence pattern data can only be set in a Cocoa UI. F8-V2, Mod8, and Res8 will work in hosts that present generic UI's, but the sequence data can't be set in these hosts. GarageBand supports Cocoa UIs. Logic also supports Cocoa UIs, though 7.2 or higher is required to fully access F8-V2, Mod8, and Res8. Please check the operation in your host of choice for suitability before purchasing.

Installation:

F8-V2, Mod8, and Res8 can be installed in any of the standard Audio Unit component locations. These are commonly /Library/Audio/Plug-Ins/Components and ~/Library/Audio/Plug-Ins/Components (where "~" stands for the user's home directory). Drag the F8V2.component, Mod8.component, and Res8.component into the Components directory and relaunch your AU host software.

Description:

F8-V2, Mod8, and Res8 are similar in the way they operate. The sequence configuration is the same in all three. The differences are in the effects that each of the AU's provides. F8-V2 consists of 8 filters: Low Pass, Band Pass, or High Pass with LFO-controlled frequency. Res8 is similar to F8-V2, but the filters are different in character and do not have an LFO. The filters in Res8 include our driven, multistage non-linear filter and formant filters. Mod8 has a range of effects including modulation effects like vibrato and ring modulation. It also has echo, reverb, sub-octave, and pitch shifting.

Note that some of the settings can be very processor intensive. Res8 in particular can add up to a whole lot of non-linear filters running at once. This can take some processing time.

We'll describe F8-V2 first and then note some of the differences with Res8 and Mod8.

F8-V2 works as either a mono-to-mono or stereo-to-stereo effect.

All of the settings of numerical parameters are made in the editable text fields. Next to each text field, there is a little rectangular button. This button behaves as a slider. If you click and drag (the button turns yellow to indicate it is active), the text fields will change. The control behavior is linear. Dragging up or to the right (or both) increases the parameter. Dragging down or to the left (or both) decreases the parameter. If you click and drag while holding down the control key, the control goes into "fine" mode. If you have a two button mouse, right dragging also invokes a fine control mode. This depends a bit on the parameter being set, but it is usually about 10 times finer than the standard drag mode for the sliders. If you have a scroll wheel on your mouse, then scrolling the mouse wheel while hovering the mouse over the slider button moves the parameter value.

F8-V2 has eight independent filters. Each filter has its own LFO. The filters can be set to be either "Lowpass," "Bandpass," or "Highpass" independently. Each filter has a resonance and cut-off frequency parameter. The LFO applies to the cut-off frequency of the filter. The rate, phase, and depth can be set independently for each filter. This allows for some pretty complex effects. Note that the resonance of the filters (Q) can be driven into self-oscillation. It can make things loud (and fun). The "X" parameter applies to all 8 filters equally (it's not set on a filter-by-filter basis). If the value is greater than zero, the filters have a faster roll off. "X" ranges from 0.0 to 1.0. As "X" is increased, the resonance of the filters gets sharper.

There is a checkbox control for selecting whether or not the tempo of the F8-V2 syncs to the hosts tempo. If your host doesn't support tempo, the tempo of the F8-V2 can still be set using the BPM control.

F8-V2 works by copying the input audio into eight independent streams. The filters are applied to these eight streams, one filter per stream. Then the eight streams are mixed back into one by the corresponding sequence section for each filter. Essentially, the

sequence section can be viewed as a time varying mixer. There's a mode selector for each of the 8 streams in the top left panel. You can choose between "Off," "Level" and "Curve." Off turns that individual stream off. When the mode selector is set to "Level," the volume of the corresponding stream is controlled solely by the setting of mini mixer slider at the right end of this same panel. If the mode selector is set to "Curve," you can set the amplitude curve in the corresponding sequence editor. The mini mixer sliders apply to this mode as well.

Clicking on the check button between the two groups of mode selectors enables "stereo split." In stereo split mode, the audio streams are split into left/right stereo channels. The split is indicated in the left/right groupings of mode selectors and matching sequence editor panels.

The Mix section has overall output level control and a wet/dry control. The "Wet/Dry" parameter has two modes. If the value is greater than zero, it works as a straight forward wet/dry mix. If the value is negative, the mix works as $D - w E$ where D is the dry signal, w is the parameter value, and E is the wet signal. Basically, this is similar to the mix from a phase shifter. It gives a bit of a different character to the sound.

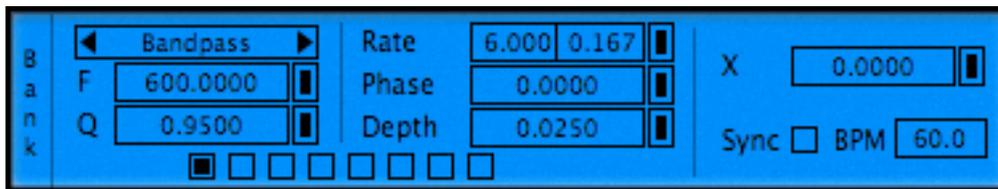
Configuration of the Settings:

The F8-V2 AU has three control groupings—sequences, filter bank and mix. The mix section contains the global parameters for the AU.



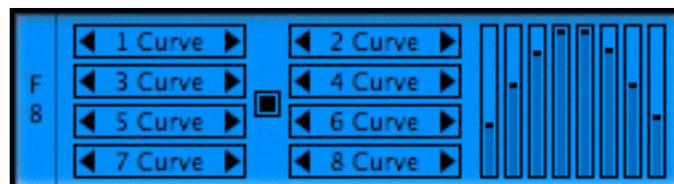
"Output" sets the overall output level. The range is 0 to 1. "Wet/Dry" sets the mix of effect-to-raw input signal. To get pure effect output, set the value to 1.0. To get the raw signal (no effect) set the value to 0.0. The wet/dry level can be used to add texture to a sound. It is also very effective with phase shifter like sounds. If the value is greater than zero, it works as a straight forward wet/dry mix. If the value is negative, the mix works as $D - w E$ where D is the dry signal, w is the parameter value, and E is the wet signal.

Next, we'll look at the Filter/LFO section.

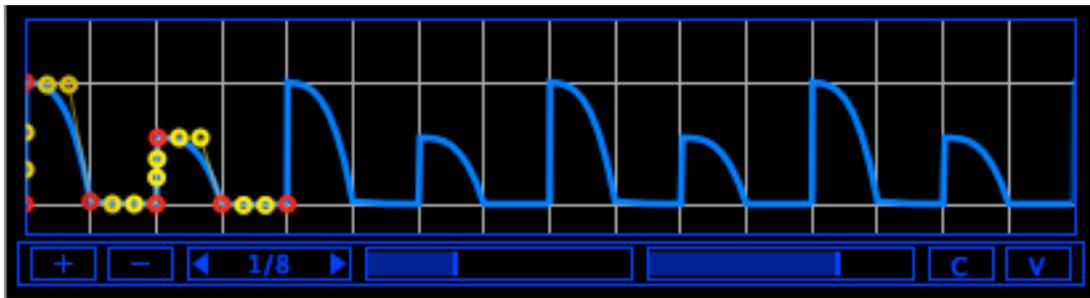


Across the bottom is the selector to choose the filter you are editing. The buttons go in order with the farthest left corresponding to filter 1. At the top-left is a popup menu to choose the filter's type. Below this is the filter's cut-off frequency and "Q" corresponds to the filter's resonance. The LFO section is on the right. The rate is displayed in both beat duration and frequency. A change in tempo will alter the frequency while leaving the beat duration constant. The phase sets the starting point for the LFO. The parameter's range is 0.0 to 1.0. It is a circular value in that 1.0 is in phase with 0.0. You can view the values as the fraction of shift for the sinusoidal waveform. For example, 0.25 is a shift of 25% of the distance through the LFO's sweep. The "Depth" parameter corresponds to the amplitude of the sweep in cut-off frequency. The range is 0 to 1. A setting of zero is equivalent to turning the LFO off. You need to play with this parameter to get a feel for it. It has guards that won't allow it to get outside of stable ranges. So, just play with it and see how it sounds. Different filter types in different situations (input audio, etc.) have an effect on how this parameter sounds.

Next, we'll look at the sequence section.



There are 8 selectors for choosing the sequence type. These allow you to choose between "Off", "Line", and "Curve." "Off" turns the sequence off and it will not sound in the final mix. "Line" selects a constant value for the volume. This volume is set by the corresponding slider in the mini-mixer. "Curve" enables the application of the amplitude modulation that is built in the sequence editors. After application of the amplitude modulation curve, the mini-mixer setting adjusts the overall volume of the corresponding stream. The checkbox between the two curve selector groups enables "stereo-split" mode. In this mode, the streams are split into two groups of four streams each. The split is stereo left/right as the curve selectors are arranged.



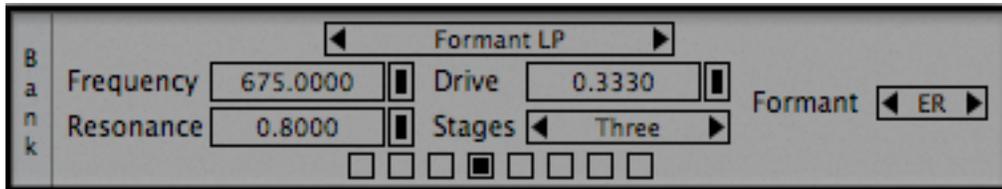
The amplitude sequences are set in the editors in the bottom screen of the AU. There are eight sequence editors. They are arranged to match the order of the curve type selectors. Hovering your mouse over each segment curve will reveal the editing environment. After your mouse leaves the area of the curve panel, the editing environment will disappear. If hovering doesn't reveal the editing environment, click on the curve panel and this should reveal it. The amplitude curves are drawn using Bezier curves. There can be between two and twelve individual curve segments. There are two horizontal white grid lines on the screen. These lines mark the zero and one amplitudes. The middle slider, in the controls section of the segment editor can be used to adjust the vertical zoom level. The slider on the right sets the horizontal zoom level. The vertical lines mark the beat divisions. The drop down menu/selector allows you to choose the beat division that is in effect. Note that selecting a new beat division rescales the current pattern to match this new beat. The "+" button adds new curve segments to the end of the sequence. The "-" removes the last segment.

Now we'll investigate drawing the curves. Note that the final end point is always restricted to match the initial. This ensures that the loop is smooth. The Bezier curves are from a restricted class of curves. Looking at the curve above, you'll see that there are two groups of points. They are marked yellow and red. The red points are always on the curve. The yellow points are used to influence the shape of the curve. A little experimentation will give you the idea of how this works pretty quickly. If you have used graphics programs based on Bezier curves, then drawing the amplitude curves will be easy. The one thing to note is the restriction on the Bezier segments that is different from the way a Bezier curve usually works. The control points cannot be moved horizontally. They are tied to the horizontal spacing of the red "curve" points. This restriction enables fast computation in the audio rendering.

The red points are not allowed to cross over each other. Time only moves forward. The editor will block you from doing anything illegal. Double clicking on a red point will snap the point to the nearest active beat division. Using this feature is important for keeping the segment timed to exactly where you want. It will help to avoid the timing from drifting. It is particularly important to make sure the final point is at the correct time for the duration of loop you want. This will ensure that the entire segment stays in sync.

The "C" button is a copy button. The "V" is a paste. You can copy individual sequences both between streams in the same AU and between F8-V2, Mod8, and Res8. This can be very helpful in building sequence sets and in moving a set of sequence patterns from

one AU to another. For example, to copy a sequence from F8-V2 to Res8, click on the "C" button in the panel of the sequence that you want to copy in F8-V2. Then bring up Res8 and select the preset you want to alter. Then click the "V" button in the panel into which want to copy the sequence.



The effects in Res8 can be "Low Pass," "Band Pass," "High Pass" or "Formant Filters." The drop-down menu selects the effect type. For LP, BP, and HP filters, the "Frequency" and "Resonance" set the characteristics of the filter. Setting the Resonance to 1.0 can induce self oscillation, which can be a good thing, depending on what you are looking for. It can also be a loud thing. There are two types of formant filter available. Both types of formants are defined by three filter frequencies. Selecting the active formant from the "Formant" menu sets the filter frequencies that correspond to a particular vowel sound. The vowel sounds are EE, OO, I, U, ER, UH, E, OW, AE, A and are ordered in roughly a low to high ordering.

The "Drive" and "Stages" controls apply to all types of filters. The drive controls the amount of non-linear distortion in the filter. The number of stages sets the roll-off of the filter.



The effects in Mod8 are "None," "Vibrato," "Vibrato Reverb," "Ring Mod," "Suboctave" and "Harmonic Generation." The "None" setting can be used to make a pure tremolo that is defined by the amplitude modulation curve. "Vibrato" is an LFO-modulated delay. "Rate" sets the rate of the LFO. "Phase" is the phase of the LFO. "Delay" is the length of the time delay, and "Depth" is the time sweep that the LFO causes in the delay time. "Vibrato Reverb" is similar to "Vibrato" except that where "Vibrato" is a pure delay (an echo), "Vibrato Reverb" is an allpass filter delay. It is the type of delay used to make reverb effects. "Q" sets the feedback, feed-forward parameter. A "Q" of 0.75 makes for a good reverb setting. "Q" ranges from -1 to 1. Unlike a normal reverb type setting, "Vibrato Reverb" can be modulated. This creates an interesting chorused-reverb like sound. "Ring mod" is a straight forward ring modulator. The frequency of the modulating tone is set using the "Rate" parameter. "Suboctave" generates a tone one octave below

the input note. And, finally, "Harmonic Generation" generates a note up to two octaves above the input note. The generated tone is set by the "Tone" parameter. The table below gives the values for the "Tone" parameter for given notes.

Hints:

Detuning the harmonic voices by a small amount, 0.005 to 0.01 provides a nice doubling effect.

To aid in the setting of harmonics:

Table of Harmonic Ratios

Tone Factor	Note Ratio	Example
0	1:1	C
0.125	9:8	D
0.25	5:4	E
0.3333	4:3	F
0.5	3:2	G
0.6667	5:3	A
0.875	15:8	B
1	2:1	C
1.5	5:2	G2
2	3:1	C2

For any given note ratio, the formula to determine the Tone Factor for a ratio N:M is
Tone Factor = $N/M - 1.0$.