JAF [Just A Filter] COLLECTION

MANUAL V1.3

JAF COLLECTION - MODEL ONE (Free Version)

The JAF Collection (where MODEL ONE is derived from) is an exciting new set of digital modeled classic analog synthesizer filters (VCFs). The emphasis of this collection of audio effects is professional analog sounding and optimized behavior with touchscreen usage, features that not many currently available apps do offer.

The collection is exclusively available on the mobile iOS (version 11 and higher). The apps and the included Audio Units are "universal", meaning they work on iPhones and iPads the same way and also have the same appearance and functionality.

Note: These apps are so-called "Audio Units" (aka plugins), an Apple technology, which usually require hosting audio applications to operate. Although, they come with their own minimal host applications, which include for instance MIDI connectivity or mic input and also have inbuilt Bluetooth MIDI support, to use the filters out of the box, like external hardware devices, without requiring an extra host application.

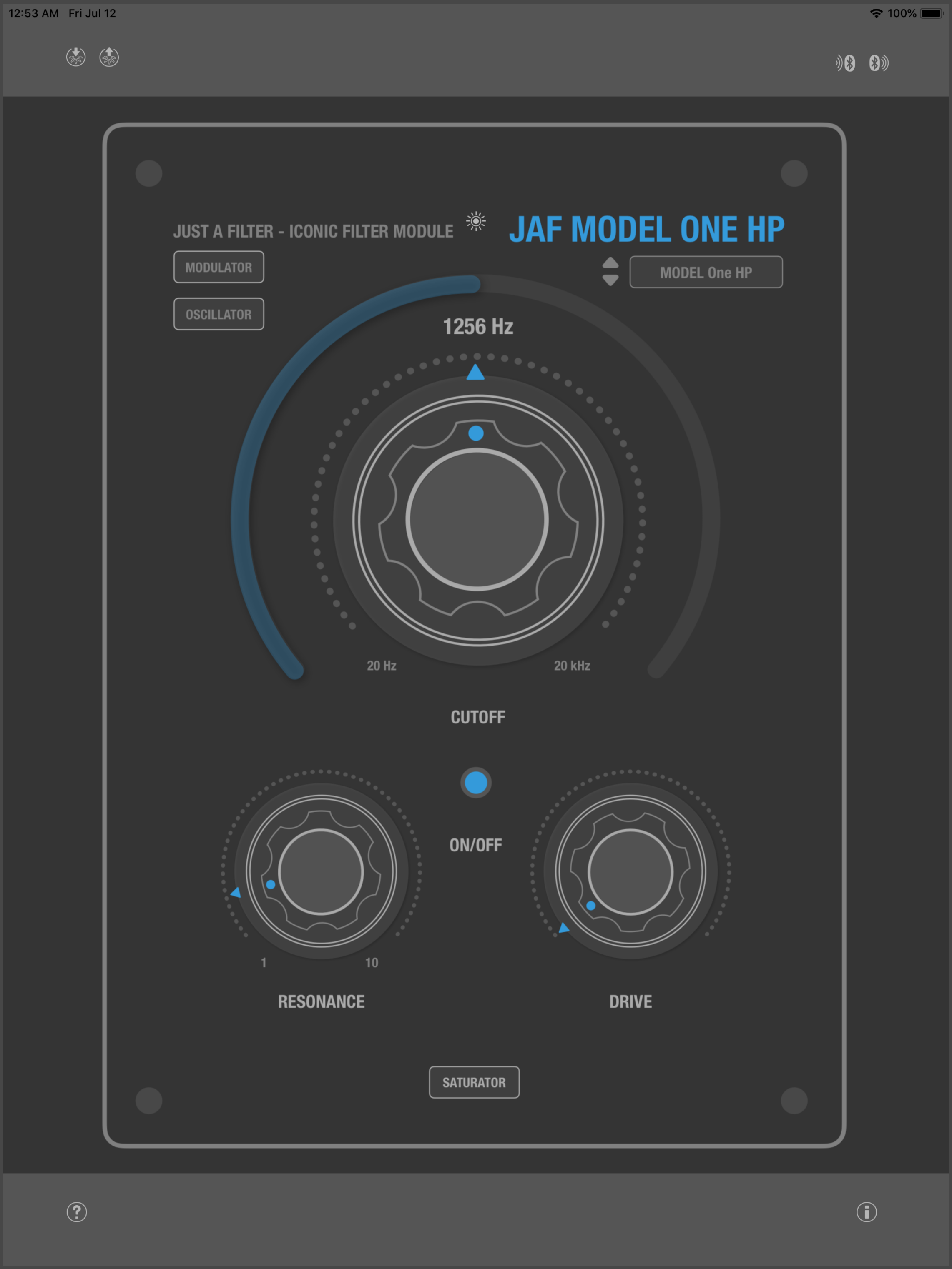
The MODEL ONE Filter can also act as a synthesizer module due to the fact, that there is an optional unison oscillator section (VCOs) built in for testing and for live performances. The analog modeled synthesizer has no advanced envelope control or similar such modulators. It just runs the oscillators continuously and can be controlled by external MIDI messages.
The complete MIDI assignment is documented inside the MIDI Implementation Chart for the JAF Collection at the end of this document.

JAF MODEL ONE (LP/HP FILTER MODULE)

The MODEL ONE is a selected, fine-tuned general purpose (transistor ladder model) filter with the ability of strong self oscillation. The cutoff slope is approximately 24dB. It can be switched to low pass or high pass processing mode.

This audio unit is released for free, as a showcase for the quality and features of our unique filter collection of over 30 different filter models. The main app for the JAF Collection allows to switch between different classic filter models, including a fairly complete set of well known legendary synthesizer filters of the golden synthesizer age.

The Model One, like all our filter models, is a true stereo processing effect, which includes a special audio-rate (high frequency) modulation oscillator, that can modulate the main parameters of the filter to its extremes. The modulation oscillator is not related to the inbuilt unison oscillators, which is a complete separate, different unit.



All our models have also an inbuilt additional saturation unit for the extra bit of analog drive. This saturation unit and the oscillator section can be used without the filter module, because the filter unit can be switched to pass the incoming audio signals thru.

The available vintage warmer tube/tape saturation type for instance is an excellent choice for breathing life into any obvious digital audio material. It can be applied to entire mixes as to individual tracks and voices.

The hosting app for the Model One has special MIDI connectivity, thus parameters can be automated with realtime MIDI Controllers (See MIDI implementation chart). The MIDI features on the audio unit depends on the fact, whether the host application will send MIDI messages to the plugin in any way with the audio processing block or not.

The audio unit can be loaded into all supporting audio unit hosts, but there are many different ones. At current state we cannot guarantee the seamless operation with all of these host applications. Please try the free MODEL ONE, before you eventually decide to purchase the commercial version JAF Collection.

If successfully loaded into a host, all parameter automation and preset handling will be handled by these host applications.

TOUCH SCREEN USAGE

The touch screen usage was overworked with the first maintenance update (version 1.1). Use 2 fingers tor common pinch&zoom and 3 fingers for panning the view. A double tap with 2 fingers will reset and center the view to default screen dimensions.

All the controls will react to single finger touches and movements. Multiple fingers are reserved for zooming. To change a value, one-finger-tap the control while sliding the fingers to any direction. Left and up usually will increase the value, right and down will decrease a value. The current zoom actually affects the parameter scaling, that means, the smaller the zoom, the faster a parameter will change.

There is also a permanent little contrast button overlay on the top of the screen (subject of future changes). You can adjust the background color (contrast) to your needs. Please note, that not all host applications behave the same way. Some allow to resize the screen or windows. This, however, may conflict with the internal zooming and panning of our plugins in some way.

UI updates: The touchscreen and user interface features got quite an evolutionary progress with the last updates. The contrast and the “pinch2zoom&move” overlay controls now can be hidden by a long press gesture. The freeze button is completely new and allows to lock the screen now. An animated hint will inform about these actions, so that the users do not get into more confusion with these changes.

These features were discussed very controversial, some users even have seen “bugs” in such features, which merely proves, that iOS users are too much trained to common static standards and confused by innovations of that kind.

THE MIDI BLUETOOTH CONNECTIVITY

We initially released the app with bluetooth MIDI support. JAF Collection and Model One was able to seamlessly connect to external bluetooth devices by establishing a connection as receiver and as a transmitter. So far so good. But the app store connect staff continuously rejected every of our apps with these features. The overcome this massive trouble and finally being able to release our apps to the consumers, we decided to remove the bluetooth support temporarily. This however, may have been changed at the time of writing this manual.

The bluetooth features were re-implemented with version 1.1.

MODEL ONE PARAMETERS

A. The Filter Module

Filter On / Off Switch:

In the center of the user interface, near the cutoff knob, sits the on/off switch for the filter in form of a small symbolized lighting diode. If the filter is switched off, the diode lights off and the incoming audio signal will be bypassed (not filtered). The oscillators and the saturation units may be used without the filter module by switching the processing mode to “Pass Thru”. But these modules are initially also switched off by default. You switch them on, by hitting the lighting diodes of these module(s) or using the MIDI controllers of a connected external controller or app. If a module is switched off, it will be bypassed by the audio processor completely.

Filter Cutoff:

The biggest knob in the middle of the screen adjusts the filter cutoff frequency, the most important parameter of every filter module. Depending on the current filter mode (high pass or low pass), the frequencies around the knobs indicated current frequency position will be filtered out of the audio signal, making a sound usually more “dark or light”.

Filter Resonance:

The more resonance (sometimes called “peak”) is applied, the more frequencies around the current cutoff frequency are emphasized. This gives a filter its characteristic sound.
Most of our filter models are able to produce high resonances up to the point of so-called “self oscillation”. If self oscillation occurs, a more or less intensive sine-like wave may become audible. It is caused by the filters internal feedback and a completely natural behavior of analog filter devices. Self oscillation is useful for generating drum sounds and special effects for instance. It is not an unwanted artifact, but an often looked feature by many synth and filter enthusiasts.

Filter Drive:

Digital filters, sometimes more or less exact mathematical models, usually behave not very natural (in a musical sense) in many aspects. If the volume boosts, especially with higher resonance, the sound actually would distort or (alternatively) reduce badly. Therefore our filters have an analog modeled amplification unit built in (sometimes this is called none linear processing), that produces natural sounding analog saturation rather than digital distortion or hard clipping, if the filters will be overdriven. The output of the filter (and sometimes the feedbacks) is saturated by these physical modeling algorithms to give the filter its own characteristics, a rich and natural warm sound, like one would expect from a analog filter emulation.
But the often used term “non linear processing” has many much more deep aspects, which alter analog filter design emulations. It is not just a simple saturation effect, that could be applied to any average digital filter model afterwards.

Filter Type:

With the sliding control under the model label, you can switch the filter between the 2 different main processing modi: high pass or low pass. The emphasized color and the model name will change, if you switch between these modi, to indicate that you have selected the right model.

The nature of non linear processing makes it actually quite difficult to produce high pass versions of all the available filters. These cannot simply be inverted to give a perfect corresponding inverse filter version for some certain reasons. But all of our low pass filter models consequently have their corresponding high pass versions. This is specially useful in our upcoming JAF Dual Pro Collection, where serial and parallel dual filter connections can be constructed and produce variable band pass and band stop filter effects this way.

B. The Integrated Modulator

Modulator On / Off:

The integrated modulator is an additional high frequency device (in fact a real audio oscillator without the sound output). You can switch it on or off. Unlike usually implemented low frequency oscillators, the parameters of the filter can be modulated up to audio rate frequency, producing interesting and aggressive special effects and if applied subtile, give the filters additionally individual character and special simulated high frequency responses, which are not possible with usual filters.

Note: Often digital filters are realized in a way, that makes their sound somewhat discontinuous or gritty because of block dependent coefficient calculations for performance reasons. Such filter models are not able to be modulated in realtime with any pleasing result. They just behave badly digital and are useful for static filtering and (possibly) analysis merely.

Modulator Destination:

You can select a destination parameter for the modulator with this slider button. There are some special destinations too, which are not directly related to filter processing, i.e. pan, amp and ring modulation of the input audio signal. The latest addition is a Bode (alike) frequency shifter, which uses the available parameters to modify audio in an interesting and unique way.

Modulator Wave:

Several different optimized waveforms can be selected for the high frequency modulator. The shape of these waveforms can be faded between two relative states in real time. These waveforms are all bi-directional, meaning the values will add or subtract cyclic modulation values from the destination value, depending on the currently selected modulation depth, which can for instance invert the wave shapes completely in direction. These two parameters must be seen in correlation.

Modulator Shape:

The modulator waveforms are, as described above, continuously adjustable between a sine shape and an alternatively selected wave form. This way a mixture or smoothing (or filtering) effect of the raw waveforms can be simulated and applied. The noise generator behaves somewhat differently. The noise generator can be adjusted in frequency and rawness.

Modulator Depth:

Adjusts the depth of the modulation in positive or negative direction (bi-directional). This means subtracting or adding modulation values from the current destination amount can be freely adjusted. So the oscillator waveform for instance will become inverted, if depth is adjusted more to the left and vice versa. If the filter cutoff knob is at maximum, then the modulation direction should be adjusted in a way that actually something can be subtracted from that value.

Modulator Frequency:

Adjusts the speed (frequency) of the modulation thru the modulating oscillator. This frequency is oscillating up to audio frequency range and may produce some sort of aliasing with very high amounts.

Modulator Linked:

This is a special behavior, if switched on, the frequency of the modulator will be linked to the cutoff frequency of the filter. The modulation frequency knob then adjusts the speed ratio. Sometimes this could be adorable, when simulating some special none linear behavior of a filter model, like certain resonance, cutoff or gain fluttering, that changes with the cutoff value in a constant relation.

C. The Integrated Oscillator

Oscillator On / Off:

The oscillator section is a completely additional and very special feature and can be switched on or off and also process independently of the filter. The tones, produced by the oscillators will be added to the current input audio signal. The oscillators always will be generated prior the filters and the saturation stage.

Oscillator Volume:

The loudness of the oscillators can be adjusted and also automated via MIDI with this parameter.

Oscillator Pitch:

This parameter adjusts the base frequency of the oscillators either freely or quantized to MIDI note pitches. If you send MIDI Messages via an external keyboard, the parameter will move and snap to the desired chromatic tuned pitch. Oscillator tuned should be on, to produce a chromatic pitch scaling. This way the filters can act as a synthesizer device for heavily modulated melodic effect voices.

Oscillator Wave:

The oscillators can be switched between several classic synthesizer waveforms and an extra variable noise frequency generator. The noise waveform generator is stereophonic but does not react to unison control (would be quite useless in the sonic result).

Oscillator Tuned:

If this option is switched on, the tones produced by the oscillator, will snap relatively to a MIDI note frequency table, making playing instant melodies more intuitive. Otherwise the frequency will be adjustable freely, i.e. for step-less, untuned sweepings.

Unison Voices:

You can set the number of oscillators, playing at same time. Up to a maximum of 8 oscillators can sound together. This is called oscillator unison, which produces interesting and rich new frequency spectra. Unison does not apply to the noise waveform for some mentioned reason. The filters will take the sum of the audio input plus the additional oscillators.

Unison Detune:

If you use multiple oscillators with the same pitch, merely the volume would raise and some phasing may occur due to oscillator instability. The detuning produces slightly differently micro tunings on each of the oscillators, so that an interesting ensemble and pitch phasing effect around the base frequency occurs. The more detuning, the more dramatically the sonic effect of the unison oscillator ensemble will become.

Unison Spread:

The 8 unison oscillators are equally distributed into the stereo field by using this parameter. Higher values separate the oscillators more and more individually to the left and right audio channels, widening the stereo field. If the parameter is zero. the oscillators will produce a pseudo monophonic signal at the center of the stereo field and phasing effects take clearly into account.

D. The Integrated Saturator

Saturator On / Off:

The saturation unit is a great addition to the analog modeled filters, because it can produce an additional special “grit” to a (possibly somewhat muddy) filter sound. The saturation unit can be switched off completely or used as a separate audio effect for processing any kind audio material.

Saturation Amount:

The more saturation is applied, the more new harmonics are rendered into the audio material. Saturation like overdrive and distortion always adds additional harmonics to the sound, not present in the original audio. This stage processes after the filter unit independently and is not related to the internal filter drive parameter in any way. Sometimes high values can produce light distortion alike effects, especially on basses or even hiss noise with high frequencies. It is recommended to be used with care, if applied to entire mixes and generally the effect should be reduced as much as possible so that it becomes a subtile subsonic additive. It is actually not thought nor is it even able to produce distortions like a guitar sound processor for instance would do.

Saturation Flavor:

This parameter is rather of a subtile nature. It adjusts the “color” of the saturation unit between a virtual tape and a virtual tube model or even a mixture between, which gives slightly different subsonic result. The effect is most distinguishable on lower and medium frequencies. A tube saturation is usually more harsh to the human ears. Audio saturation is generally a matter of tastes than a science. We provided a small bonus taste with that.

MIDI Implementation Chart

Parameter	Parameter Address	MIDI Controller	Value Range
Filter On/Off	1	0x66 102)	0 - 1
Filter Cutoff Frequency	2	0x4A (74)	0 - 127
Filter Resonance/Peak	3	0x47 (71)	0 - 127
Filter Drive	4	0x0B (11)	0 - 127
Filter Model/Type	5	0x6A (106)	0 - 32
Modulator ON/Off	6	0x67 (103)	0 - 1
Modulator Destinatio	7	0x6B (107)	0 - 8
Modulator Waveform	8	0x6C (108)	0 - 8
Modulator Shape	9	0x14 (20)	0 - 8
Modulator Depth	10	0x15 (21)	0 - 127 (63)
Modulator Frequency	11	0x16 (22)	0 - 127
Modulator Frequency Link	12	0x6D (109)	0 - 1
Oscillator On/Off	13	0x68 (104)	0 - 1
Oscillator Volume	14	(0x03) (3)	0 - 127
Oscillator Pitch	15	MIDI Note Number	0 - 127
Oscillator Wave	16	0x6E (110)	0 - 8
Oscillator Tuned	17	0x6F (111)	0 - 1
Unison Voices,	18	0x70 (112)	0 - 8
Unison Detune	19	0x17 (23)	0 - 127 (63)
Unison Spread	20	0x18 (24)	0 - 127
Saturator On/Off	21	0x69 (105)	0 - 1
Saturation Value	22	0x19 (25)	0 - 127
Saturation Flavor	23	0x1A (26)	0 - 127 (63)
Amplifier Volume	24	0x07 (7)	0 - 127
Amplifier Pan	25	0x0A (11)	0 - 127 (63)
Reset	-	0x79 (121)	0 - 1

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