Important Changes (Since Ver. 2.0)

In this version we have enabled Audio Unit V3 Support.
You can load multiple instances of apeFilter in most of the popular AudioUnit Hosts as: AUM, Cubasis, Studiomux etc… You will see two kinds of ports: 'Effect' and 'Music Effect', the only difference is that 'Music Effect' supports MIDI (see ApeFilter User Guide for more information). Please note that some of the Audio Hosts do not support both kinds, in that case you will see only the 'Effect Node', without MIDI.

⇆   Locks the FilterGraph X axes.

⇅   Locks the FilterGraph Y axes.

Important note, in the Standalone mode these settings are not saved in the Presets and they will not affect the FilterGraph when you are playing via MIDI. Conversely when you load apeFilter as an Audio Unit v3 these will be saved in the Presets, particulary for the 'Music Effect' instances, will affect the FilterGraph MIDI input.

Audio Unit Extension support

Since iOS 9 Apple introduced the Audio Unit V3 format which enables multiple instances of the same plug-in running at the same time.
You can load with compliant hosts like Apple's own GarageBand or other audio editing and mixing apps, compatible with the host-automations, host-save/load presets and host-save/load state in the session. The Audiounit Extension (i.e. AUv3) can share data only with the main App. The extension have two Manager called QuickSamples and QuickPresets for facilitate the audio/presets sharing. When you copy samples or preset banks, from/to, you will make a copy in a shared folder which is accessible from both.

WARNING:
The memory limit imposed for AU Extensions on iOS is 300 MB for 32-bit devices and 360 MB on 64-bit devices.
These are performance considerations as you can expect from iOS devices having limited shared resources.

We do not recommend loading more than 12 instances.

Sharing procedure:

1) From the AUv3 plugin open QuickSamples or QuickPresets Manager

2) Tap the 'Download icon' ⤓ this will launch the main app

3) From the main app, if you are importing an Audio Sample, tap one of them and chose 'Share with AUv3'. If you are importing a Preset Bank, select it and tap the 'Share icon' ⬆

4) Now you needs to coming back to the Host where you are running the AUv3 plugin, double tap on the iDevice Home Button and select the app.

5) To complete the retrieve process you need to tap the 'Sync icon' ↻

Tips & Tricks

- Double tap on the **Knobs** to show the options. In the '**Edit**' field you can enter the value from the keyboard. Tap '**Random**' to generate a random value, '**Default**' to reset to the factory settings and configure the knobs dragging behavior. Step fine resolution sets how many points of movement result in a one degree rotation in the knob's position. Only used in the horizontal/vertical slider modes. You can set Knobs behavior from Settings.

- Long Press Tap on the **Toggles** to open Control Manager Assignations.

- When the user presses the Home button, the app goes to the background process and will store the whole status of parameters to reload when next opening the app. See Settings for more explanations.

- Pinch In/Out on the Filter Bandwidth for changing **Q** value.

- Tap to select a Filter, Double Tap to add or remove a Filter.

- apeFilter can play audio in the background and/or mix its own audio with other iOS apps. apeFilter also supports Apple IAA (i.e. Inter-App Audio for iOs 7 or higher) and Audiobus, so you can send and receive audio through applications.

- Swipe gesture (right to left) on a preset or bank to rename/duplicate/delete it. See Presets for more details.

- You can load Presets (current selected bank) by remotely sending a MIDI Program Change message. See MIDI section, below for further details. You

can select the MIDI channel where you can receive PC messages. A PC value of zero will select the first preset in the list, a value of 1 for the second etc… The range is from 0 to 127.

- Send to *apeFilter* MIDI Note messages, to controls the Filtergraph **Center Freq**./**Cutoff** (Note Number) and **Gain** (Velocity).

- When you are connected in Audiobus or Inter-App Audio as FX, Input Mode will be selected automatically. Every time you will disconnect from AB or IAA, you will turn back in Sampler. A warning message will displayed be in order to prevent dangerous Speaker audio feedback. The same behaviour will occur if you have not attached, in your iPad/iPhone/iPod Touch, an external device as headset or Mic. etc… You can disable this message through the app Settings, see later for more explanations.

- We suggest you to disable the screen rotation, otherwise the accelerometer doesn't work properly.

- **Init** will reset to default values, a reset message will be displayed.

- **Create Random Sound** values for all of the parameters including the FX parameters. The result can be very interesting and wayward.

Enable/Disable Accelerometer X/Y and LFO, see the Control Manager for more details.

Captures Output stereo signal (post Effects) and creates an audio file *wav* format 44.1 kHz 16 bit in the Documents Folder. Files created by *apeFilter* are saved with an id-name followed by a number, for example: **Output Rec 1.wav, Output Rec 2.wav etc.**
Each new file will be numbered with the next consecutive number.

SYNC

The goal of SYNC is to setup a global Metronome (MASTER BPM). This will affect the LFO (Low Frequency Oscillator) and/or the Sampler (or Sequencer etc…) when enabled to receive SYNC.

You can sets the BPM (Beats Per Minute) by scrolling up/down (or left/right) the label value (♩ 120 ←○→) or tap out the tempo.

Single Tap or Long Tap the minus/plus buttons to increase or decrease by 0.25. By default 120 BPM corresponds to 1 Hz frequency for the LFOs.

Ableton Link description
Ableton Link is a new technology that synchronizes the beat, phase and tempo of Ableton Live and Link-enabled iOS apps over a wireless network. It lets you play devices together with the freedom of a live band. Anyone can start and stop their part while others keep playing, and anyone can adjust the tempo and the rest will follow. You can use Link to play with several copies of Ableton Live, with Live and iOS apps, or with iOS apps together – even without Live in your setup.

MIDI Clock description
The clock signal is broadcasted via MIDI to ensure that several MIDI-enabled devices such as a synthesizer or music sequencer stay in synchronization.
The iOS implementation is designed to provides precise, jitter-free MIDI clock transmission and robust, error-tolerant MIDI clock reception.

Quantization
When you are connected to an Ableton Live Session, the Play/Stop/Rewind transport will wait until the next sync quantum boundary before starting.

The 'Sync' button waits for the next beat (Tempo Forte) from the MIDI Clock or Ableton Link and resets all phases for LFOs, Sequencer etc.

Send MIDI Clock
Will create a virtual MIDI port for the MIDI Clock Output.
In Play/Stop mode the MIDI Start/Stop Clock messages
are sent according to Transport Play/Stop (Sampler, Sequencer etc…). In 'Always' they will not be sent every time, but only at start.

Please note when you are connected to Ableton Link, the incoming MIDI Clock is disabled while you can send MIDI Clock out, converting the Ableton Link to MIDI Clock.

Receive Sync
1) Local-Host
2) MIDI Clock signal
3) Ableton Link.

When the Ableton Link its enabled the Local-Host and MIDI Clock will be disabled and viceversa.

In Local-Host when the app is connected through Inter-App Audio, will receive the Sync from the HOST (if available, i.e. tempo/start/stop) and the current BPM is overridden.

When MIDI Clock is activated, you cannot change the BPM (Slave) since is controlled by the MIDI Clock (Master). Through the 'CLOCK' button you can select a MIDI Clock Source.

Sampler

You can feed *apeFilter* with a stereo sampler or stereo signal from microphone/line or Audiobus/Inter-App Audio when connected.

Touch and drag the waveform to scrub the audio file. The speed of the scrubbing is relative to the position of the finger inside the waveform: higher positions produce faster scrubbing

Tap the symbol to set the loop points. When in "loop edit mode" you can:
- tap and hold inside the selection for the fine regulation of the loop
- double tap inside the selection to select again the whole file

When a loop is set, deselect the (tap again) to zoom-in into the loop selection.

Reset scanning to the start position.

Waveform Scrubbing Mode, when selected the pointer jumps to the **Absolute** position of the finger that touches the waveform. If unselected (**Relative)** it does not jump but the scrubbing occurs from Sampler position toward finger touch.

Load an audio sample, see Samples Manager for mode details.

Include/exclude the Sampler state when Presets are loaded. The Sampler state is composed by File Path, Loop Points (Zoom), Play/Pause and selected Input Source. Please note even if it is disabled, the Sampler still saves its current state in the presets.

⊖⊖ Switches the Sampler Play function between Ratio and Time Signature mode. When in Ratio mode (blue icon) you can set the play ratio: 1 means play the sound file at the original speed, 0.5 means play at half speed, -0.5 means play at half speed and in reverse and so on. When in Time Signature mode (orange icon) you specify the number of Measures and Quarter per Measure the sound file contains, and the tempo is linked to the general Metronome Tempo; for instance if we specify 2 measures of 4 quarters each, and the general Tempo is 120 BPM, the sound file will be executed in 4 seconds.

← **BPM** The button appears only when the Sampler is in Time Signature mode (see above). If you tap the <-BPM button you overwrite the general Metronome Tempo with a tempo calculated using the sound file length and the Measures/Quartes setting. For instance, if the sound file length is 2 seconds, and it has 2 measures of 4 quartes each, the general Tempo is set to 240 bpm. This function is not available when the General Metronome is in SYNC mode (MIDI Clock or Ableton Link).

N.B. You can disable the Crossfade and change Waveform style from the Settings. See the Settings for more details.

Input - Sampler Switch

You can choose between the internal sample player and a live input via Microphone, Audiobus or Inter-App-Audio.

IMPORTANT NOTE:
When you are connected in Audiobus or Inter-App Audio as FX, Mic-AB-IAA will be selected automatically. Thus, every time you will disconnect from AB or IAA, you will turn back in Sampler. A warning message will be displayed in order to prevent dangerous Speaker audio feedback. The same behaviour will occur if you have not attached, in your iPad/iPhone/iPod Touch, an external device as headset or Mic. etc.
You can disable this message through the app Settings, see later for more explanations.

Input Gain, Dry/Wet, Pan and Output Gain

"Input Gain" is the Sample/Input volume, Dry/Wet mixes the input sound with the apeFilter sound, "Pan" puts it in the stereo continuum and "Output Gain" is the output volume.

FilterGraph

Use this graphical interface to generate filter coefficients for the **biquad/cascade** filters.

The Filtergraph recognise this gestures:
Tap, Double Tap, Pinch and Horizontal/Vertical drag.

⊖

Lock/Unlock the filters band for multiple operations, if enabled, **Center Freq**., **Gain** and **Q** will work for all filters, if unlocked will work for only selected filter band.
You can select a filter band by tapping over filter region. A band rectangle is highlighted, indicating that tapping and dragging will map x-axis for **Center Freq**.,
and y-axis for **Gain.** You can Pinch in/out to adjust the filter bandwidth (Q). Alternatively you can adjust filter bandwidth left/right dragging ←○→ on the symbol lower the band.

When is locked you will change all filters parameters by dragging on a region. Every filter will keep own state and will be scaled according to general average.
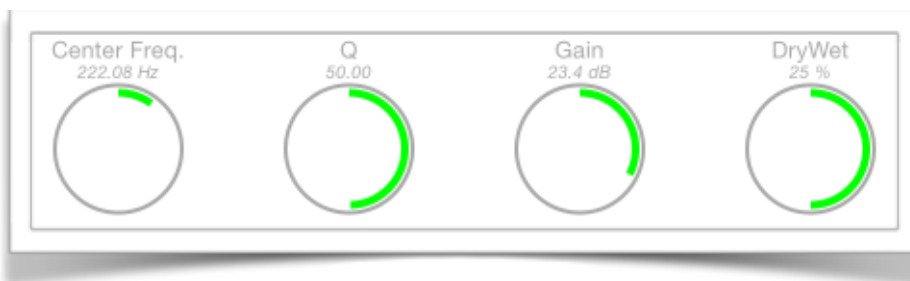
Tap for select a filter region, Double tap to create new band or remove an existing filter. Pinch a filter region to change **Q**.

[—] Reset the **Gain** to 0 dB for all filters, no **Center Freq**./**Cutoff** and **Q** will affected.

Alternatively at Filtergraph, you can adjust the filters parameters through the Knobs in the main User Interface.

The frequency response of the filter is based on three parameters: **Center Freq.** (Default for Peak Filters or **Cutoff** for other Filters Type), **Gain**, and **Q** (resonance).

These parameters are bound at Filtergraph, you can see the values when you change the selected filter in the Filtergraph. For the locked mode, will shown the values of the first filter (also called SENTINEL, i.e. the first of serie).

⇆   Locks the FilterGraph X axes.

⇅   Locks the FilterGraph Y axes.

Important note, in the Standalone mode these settings are not saved in the Presets and they will not affect the FilterGraph when you are playing via MIDI. Conversely when you load apeFilter as an Audio Unit v3 these will be saved in the Presets, particulary for the 'Music Effect' instances, will affect the FilterGraph MIDI input.

Spectral Analyzer

The Spectral frequency analysis FFT (i.e. Fast Fourier Transform) is of 8192 samples, so you have 4096 analysis channels in the range 0 ÷ 22.5 kHz. The frequency gap of the FFT bin is 5.3833 Hz (i.e. Nyquist / (FFT/2)).

Spectral Grid Compilers

Through the Filter Type menu, you can select the filter kind for the synthesis. The default Filter is **Peak**, this will enable all features of apeFilter, as well the Spectral Grid Compilers features. By selecting **Peak** you can also setup more than one filter in the Filtergraph, while for all the others, only one is permitted. The Lowpass, Hipass, Bandpass and Notch filters, will disable the **Gain** parameter in the main UI, while the Lowshelf and Hishelf will disable **Q**.

The **Center Freq**. parameter (Default name for Peak Filters) will change in **Cutoff** for all others Filters.

As well as you can setup ad hoc, your equalisation curve for every filter bandwidth (in Peak mode), the Compilers increase the complexity and coherence between the filters bandwidth space. You can create your pitch space by the way of four different criteria of construction by determining a pitches grid.

For all Spectral Compilers, the frequency range is depending by two parameters: the fundamental frequency (i.e. **Center Freq.** knob on the main UI) and the **Numbers of Filters**.

When a filter band exceed the Nyquist[1] frequency (i.e. SR/2) the filter band generation is inhibited.

Fibonacci[2]: the *ratio* of two consecutive Fibonacci numbers converges on the Golden Mean Ratio (approximately 1:1.618) as its limit.
Fibonacci compiler uses this sequence of numbers for compile the grid pitches.

The only parameter is the **Serie Stretch** for the purpose to create a certain Fibonacci pitch grid.
The first member of the series is 2 and **Serie Stretch** expands or compress the resulting grid according to the formula:

expr = (1.0 + pow(STRETCH, 0.5)) / 2.0;
Hz_ = Hz * ((((pow(expr,i+2) - pow(1.0-expr,i+2)) / pow(STRETCH, 0.5)));

where:
i = i-th value of serie
2 = offset of first serie value.

To complete the operation and generate the pitch grid you must press the **compile** button on the right of every Compilers.
If the **Auto Compile** is enabled, by changing the parameter value, (for every compilers methods) the grid pitch it will be automatically compiled.

Harmonic COMPILER creates a bands grid according to the natural series of integer numbers (1,2,3...etc), starting from a fundamental frequency (i.e. **Center Freq**. on the main UI)
The HARMONIC Inharmonicity is for spectral expansion/compression. You can stretched or compressed harmonic spectra putting the parameter at a non zero value;

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[1] The **Nyquist frequency**, named after the Swedish-American engineer Harry Nyquist or the Nyquist–Shannon sampling theorem, is half the sampling frequency of a discrete signal processing system. It is sometimes known as the folding frequency of a sampling system. The sampling theorem shows that aliasing can be avoided if the Nyquist frequency is greater than the bandwidth, or maximum component frequency, of the signal being sampled. (Wikipedia)

[2] **Leonardo Pisano Bigollo** (c. 1170 – c. 1250)also known as **Leonardo of Pisa**, **Leonardo Pisano**, **Leonardo Bonacci**, **Leonardo Fibonacci**, or, most commonly, simply **Fibonacci**, was an Italian mathematician, considered by some "the most talented western mathematician of the Middle Ages. (Wikipedia)

Stretched spectrum is obtained from values greater than zero, while for compressed spectrum you have to select values less than zero.

The stretched or compressed spectra are created using the following formula:

$$F_n = F_0 * n^{(1+exp)}$$

where:

Fn = n-th frequency
F0 = fundamental frequency
n= index of Fn frequency
exp = factor of expansion/compression (inharm)
The inharm factor can vary from -0.2 +0.2 When inharm = 0 the generated frequency grid is pure harmonic.

Geometric COMPILER create "geometric pitch space" based on a *Ratio number*. For example if you specify 1.5 (i.e. 3/2), you will create a pitch space grid based on the natural interval of 5th. Since you can choose any relationship, you can virtually have infinite pitch spaces.
The **Ratio** parameter defines a coefficient of multiplication, this serves to build up the whole pitch grid while the *fundamental frequency* (**Center Freq**. on the main UI) is the starting point from which grid take origin. The Ratio of 1.6 for the geometric spectrum (also represented by the ratio 8/5) in the below example, creates a geometric pitch grid based on the interval 1.6 starting from the fundamental frequency (**Center Freq**.) of 55 Hz. The grid will consist of a total of 13 bands (according to selected **Number of Filters**) starting from 55 Hz up to 15.481,1279 Hz as sketched in the next table:

#of_FREQS	Hz	operation
13	15539.276	$55*1.6^{12}$
12	9675.702	$55*1.6^{11}$
11	6047.313	$55*1.6^{10}$
10	3779.571	$55*1.6^{9}$
9	2362.232	$55*1.6^{8}$
8	1476.395	$55*1.6^{7}$
7	922.746	$55*1.6^{6}$
6	576.716	$55*1.6^{5}$
5	360.488	$55*1.6^{4}$
4	225.280	$55*1.6^{3}$
3	140.800	$55*1.6^{2}$
2	88.000	$55*1.6^{1}$
1	55.000	$55*1.6^{0}$

You can obtain the same result using the next recursive formula:

$$fi = f(i-1) * iterval\_ratio$$

where:
fi is the i-th frequency (i.e. f2 = f1 * 1.6 = 55.00 * 1.6 = 88.00).

Scalar COMPILER method is very easy and it linearly distributes all bands between Fundamental Frequency (i.e. **Center Freq**.) and **Max Frequency** parameter.

Remove all Filters will set the Number of Filters to one but also will clean the filters coefficients and restore all defaults values for **Gain** = 0 dB and **Q** = 50.

Remove upper Niquist Filters will set the **Number of Filters** parameter to the actual number of filters contained in the FIltergraph range and discarding all the Filters who fall out of the Niquist range.

Presets

Preset Manager will collect all the **Banks** *(i.e. .apefilter suffix)* and **Snapshots.** A *Snapshot* is a simple "photo" of the parameters in the UI (i.e. User Interface). Both the Matrix and Keyboard/Sequencer are included but they can also be excluded, see the Settings for more details.

So, once you find an interesting sound you can take a snapshot (i.e. save this new preset) by touching the 'plus' icon and this new preset will be added to the selected Bank. The top four elements of the Snapshots list are assigned for 'HV Snapshot Pad', you can then sort or delete the Snapshots in the List by tapping the 'List Icon'. You can even share the selected Bank and you can restore the factory Built-In Banks from the 'Cloud' icon.

Backup

Factory preset-banks (Excluded from iCloud back up)

Library preset-banks (if available as In-App Purchasing) (Excluded from iCloud back up)

User preset-banks (Included in iCloud back up)

If you delete, overwrite, rename or save new presets in factory/library banks, the icon will change as User preset-banks , being backed up.

Snapshot banks are located in Documents folder, you can access files via iTunes Sharing, see 'Samples'.

You can add new files by copying in the app's documents (through iTunes or from another app) and refresh the list from there.

Load a preset, randomly from the banks list.

Delete all Presets for a selected bank. A warning message will be displayed.

The Snapshots bank file is a kind of Property List format (*.apeFilter*), you can edit as text and modify the values just rename in *.plist*

Swipe (right to left) on a preset or bank to rename/duplicate/delete it.

The purpose of the unique **Snapshot Pad** is to obtain intermediate values between four snapshots (corners). You can change snapshots assigned in the corner through the Snapshots list (see above).

The Snapshots Pad performs complex interpolations between four snapshots on a two-dimensional space. By scrubbing your finger on the Snapshots Pad a temporary new snapshot will be created by mixing the four snapshots depending on distance. Thus you can modify hundreds of parameters with a single touch, this is also called *Hyper Vectorial Synthesis.*
Snapshots Pad can be resized on fullscreen for detailed controls (left bottom icon).

N.B. Snapshot Pad have no effect on Switches and Sampler.

Samples

Samples manager can read/write in the Documents folder, you can also share files through AudioCopy, Document Picker (iCloud, Dropbox etc…) and AudioShare. Samples manager can also read the Built-in files, but you cannot rename or delete them. Touch on a file in the list to show the actions popup. Here it is possible to **Load** a file in one of two Samplers (i.e. Sampler Ch.1 and Sampler Ch.2), Open In... contains all system actions over the file, according to file type. For example you can send files as an e-mail, open file with a listed app etc…

 You can rename files in the Documents Folder, if you omit the file extension then Samples Manager will fill the original audio file extension for you. N.B. Samples manager shows only audio files supported, the other files in the Documents folder will not be seen.

 You can pre-listen to the samples before you load them.

 Starts audio capture from the microphone immediately and saves new stereo audio file in wav format 44.1 kHz, 16 bit in the Documents Folder. Files created are composed by id-name followed by a number, for example:
Input Rec 4.wav, Input Rec 5.wav etc…
Every new file will be numbered with the next consecutive number.

N.B. You can record a mono or stereo file in wav format 16 bit 44.1 kHz according to the iDevice channels. Even if the apeFilter is connected to Audiobus or IAA, the Grab Audio will record always from iPad microphone/line

⬇ You can import files from AudioCopy, Music Library, Document Picker (iCloud, Dropbox etc…) and AudioShare.

Access to the iPod library, this allows the application to read user's songs, audio books, and audio podcasts. The file can be converted in *.wav* (PCM 16 bit 44.1 kHz) or *.caf* formats and stored in the local app Documents Folder. If the file exist at path then progressive number will be attached to the file name.

The document picker feature lets users select documents from outside your app's sandbox. These include documents stored in iCloud Drive and documents provided by a third-party extension. Users can open these documents directly, editing them in place. This access simplifies sharing documents between apps and easily enables more complex workflows.

Use iTunes to share files between your computer and your device
1) Open iTunes on your Mac or PC.
2) Connect your iPhone, iPad, or iPod touch to your computer using the USB cable that came with your device.
3) Click your device in iTunes.
4) In the left sidebar, click File Sharing.
5) Select an app to see which files are available for sharing in that app on your device.

Control Manager

MIDI

Accelerometer X

Accelerometer Y

LFO

After selecting **MIDI**, *apeFilter* enters into "learn mode" i.e. it "hears" incoming MIDI Control Change messages. It's then possible to send messages with MIDI hardware, Virtual MIDI and Network MIDI. Flashing screen means the association between the parameter and the MIDI message has been successful and displays the corresponding control Change and Channel. Alternatively you can enter direct CC and MIDI Channel number through number keyboard. You will also receive an additional setting: 'Widget Range' that allows you to resize the range of action on the parameter. This is

because you can use all 128 steps MIDI to control precisely particular part of the parameters.

Accelerometer X and **Y** are used to control the parameters through the horizontal or vertical inclination of the device.

LFO, Low Frequency Modulation is a table look-up oscillator with seven built-in shapes. The frequency of LFO is expressed in Hz and BPM (Beats per Minute). You can get the period time by dividing 1/freq. The LFO Frequency value is converted in BPM (Beats Per Minute), you can adjust this value through the minus/plus buttons, these will replace current value with the integer BPM (previous/next) or you can TAP to set your own Tempo.

φ   Will reset to zero the phase of current LFO.

Φ   Will reset to zero the phase of ALL LFO actives.

🔗 Enabling the Link symbol, you will disable the 'Low Frequency Oscillator in Hz' Slider, so the value of the current LFO is sampled from the global 'MASTER BPM'. You can adjust the BPM value from SYNC view. When the MIDI Clock or Ableton Link is enabled (see MIDI Manager and SYNC) the BPM value will affect all LFO enabled for SYNC mode. The goal of the 'SYNC Subdivision' Slider is to rescale the 'MASTER BPM' from SYNC according to the relative symbol (abbreviate musical note time values):

Normal : 1/1, 1/2, 1/4, 1/8, 1/16, 1/32, 1/64
Triplets : 1/1T, 1/2T, 1/4T, 1/8T, 1/16T, 1/32T, 1/64T
Dotted : 1/1D, 1/2D, 1/4D, 1/8D, 1/16D, 1/32D, 1/64D

Where 'D' is a **dotted** note and 'T' is **triplet**.
With 1/4 (Default), the BPM follows the 'SYNC BPM' or 'MIDI Clock' BPM when enabled, i.e. 120 BPM corresponds to 1 Hz.

-     1/4 - each step is a quarter note (or BPM), e.g. 60 BPM is 0.5 Hz
-     1/4D - each step is a dotted quarter note (or 1.5 x BPM),
-     1/4T - each step is a triplet quarter note (or 0.5 / 3) etc.

♩ Reset default value (1/4 = 120 BPM)

MIDI

You can enable/disable the incoming MIDI signal from all the physical, virtual and network ports available. The MIDI Clock is managed by SYNC, see the SYNC section for more details.

 If the Core MIDI Network is connected a wifi symbol will appear in the Network port cell.

Channel sets the input channel for the selected port, the *apeFilter* DK (Keyboard) will receive the Note On/Off messages in OMNI (from all 16 channels) or, one of the 16 channels.

Configure Bluetooth LE MIDI
With the release of iOS 8 and OS X Yosemite, sending and receiving MIDI data is supported using Bluetooth Low Energy connections on any iOS device or Mac that has native Bluetooth Low Energy support. All established connections are secure which means that pairing is enforced and connections cannot be made to your devices without your explicit consent. After a connection is established, it simply appears as an ordinary MIDI device that any MIDI application can communicate with.

Enable NRPN
NRPN stands for "Non-Registered Parameter Number" and is part of the MIDI specification for the control of electronic musical instruments. NRPNs allow for manufacturer-specific or instrument-specific MIDI controllers that are not part of the basic MIDI standard.
Unlike other MIDI controllers (such as velocity, modulation, volume, etc.), NRPNs require more than one piece of controller data to be sent. First, controller 99 - NRPN Most Significant Byte (MSB) - followed by 98 - NRPN Least Significant Byte (LSB) sent as a pair specify the parameter to be changed. Controller 6 then sets the value of the parameter in question. Controller 38 may optionally then be sent as a fine adjustment to the value set by controller 6.
This fine adjustment is part of the conventional MIDI controller specification, where any of the first 32 controls can be optionally paired with a control offset 32 higher. This is the rare 14-bit Continuous Controller feature of the MIDI specification, and NRPNs simply take advantage of that existing option in the same way offering 16,384 possible values instead of only 128.
(http://en.wikipedia.org/wiki/NRPN)

When you enable NRPN, you should not employ CC 99 and 98 as the normal CC in order to avoid an undesired overlap. An NRPN message must be sent with this order: CC 99 98 6 38 where the first couple is the CC number and the second couple is the data. You can MIDI learn as for the normal CC, and NRPN works for all knobs and parameters. Although an NRPN CC number can be greater than 127, you will not be able to learn CC beyond 127.

Program Change for Presets Selection
You can select Presets (current selected bank) by remotely sending a MIDI Program Change. You can select the MIDI channel where you can receive PC messages. A PC value of zero will select the first preset in the list, a value of 1 for the second etc… The range is from 0 to 127.

Effects Chain

From left to right we have: *Stereo Echo, ClassicVerb and Dynamics s Processor*. The first effect of the chain is **Stereo Echo** which receives signal from the **apeFilter**. If enabled, the processed signal will pass to the next effect or bypass if disabled. You will see enabled effects on the highlighted toolbar icons, you can switch between popovers effects while scrubbing on the Filtergraph.
The amazing ClassicVerb is a classic reverberation based on an high-quality algorithm is also provided. It is possible to control, independently, the early-reflections and the reverberation tail.
Special thanks to **Carmine Cella.**
www.sonic-pad.com

Dynamics Processor

After the *apeFilter*, you can change the output sound using a conventional compressor and expander effect.

This is used to either reduce or expand the dynamic range of an audio signal. The Dynamics Processor controls set high (compression) and low (expansion) threshold volume levels. When the incoming audio is between these levels, it is left unchanged. Audio that rises above the upper threshold

is cut in volume to bring it back down to that level (compression). Audio that falls below the lower threshold is cut in volume to make quiet sounds even quieter (downward expansion).

Additional controls adjust the attack and release time for the plugin's volume envelopes, the expansion ratio, the compression headroom, and a master gain to boost or cut volume on the Dynamics Processor's output.

Threshold: when the signal is above the threshold it is compressed
Headroom: upper limit for the compressor, a limiter prevents the signal to go above this threshold
Attack and Release: activation and de-activation times for the compression when it is above the threshold
Exp Threshold: when the signal is below the Exp Threshold it is backward expanded
Exp Ratio: expansion ratio

Settings

Buffer Size set the Audio Device latency. When you run *apeFilter* from **Audiobus**, or **Inter-App** this value will be taken from **Audiobus** or **Inter-App Host**. Warning! When an app runs first and keeps running in the background, the Sampling Rate and Buffer Size will impose for the whole iOs while running.

Save/Restore the Last App Setup, when you exit from the application, save the status (all *apeFilter* and Fx Parameters) and reload when next opening the app.

Filter Graph Scrub Relative/Absolute (Jump)
If **Relative**, it does not jump but still maintains its position and you can move the pointer in relation to the relative position. If **Absolute** it does jump to the value of the finger that touches the filter band.

Filtergraph Frequency Logarithmic Scale
Valid for only x-axis, Frequency scale.

Enable MIDI Note Velocity for Filter Graph Gain
If enabled, the MIDI note key velocity will controls the Gain parameter. As you know, MIDI range is 0 ÷ 127 while Gain range is -36 ÷ +36 dB.
Therefore, a MIDI dynamic of zero it means -36 dB, 64 (mf) 0 dB and 127 (fff) +36 dB.

Others Settings
The switch in this section are self-explanatory. As for all warning messages; include/exclude Control Manager, Assignations and Knob Color in the presets etc…

Snapshot Pad Resolution Update sets the time interval for the interpolations, from about 15 to 350 milliseconds. Short times for higher resolution but higher consumption of CPU.