FilterDeMystifier (FDM) Documentation

June 24, 2021, V 0.2.5, work in progress

> WARNING:
>
> If you switch off both security measures, this plug-in may produce very loud output signals (unstable filter) and poses a threat to your ears and gear. It can be fun – if you know what you're doing. Reduce the output level in your DAW before trying anything stupid.

1 Introduction

The name of the plug-in is just a funny name as a reminder of the early days when all plug-ins had an 'er' at the end (e.g. Loudness Maximizer, DeNoiser). Since filters are nothing but math and more math there are no mysteries here. However, never forget that we have all of these fabulous EQs and synthesizers in our DAWs because we are standing on the shoulders of giants, who started digital signal processing 50+ years ago. A special thanks goes to the unknown developer of the pzx.dll plug-in. This was our inspiration.

This plug-in shows how digital filters work on the inside. It is fun to play and see how a few poles and zeros determine the whole range of digital filters from cut to peak. From the famous RBJ cookbook to modern near analogue designs.

Have fun!

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2 Installation

The downloaded file is a standard zip file. It contains a readme.txt, a plug-in directory (.vst3) or an AU file (.component) and this manual. The first step is to copy the directory to the right place for VST3 / AU files.

2.1 Windows

For Windows™ the standard VST3-directory is:

```
c:\Program Files\Common Files\VST3
```

2.2 Mac VST

If you use the VST3 plugin, copy the .vst3 directory to

`/Users/yourUSERNAME/Library/Audio/Plug-Ins/VST3`

2.3 Mac AudioUnit

If you use the AudioUnit plugin, copy the .component file to

`/Users/yourUSERNAME/Library/Audio/Plug-Ins/Components`

2.4 Linux

For Linux copy the full directory to:

`/home/YourUSERNAME/.vst3/`

2.5 Preset (De)-Installation

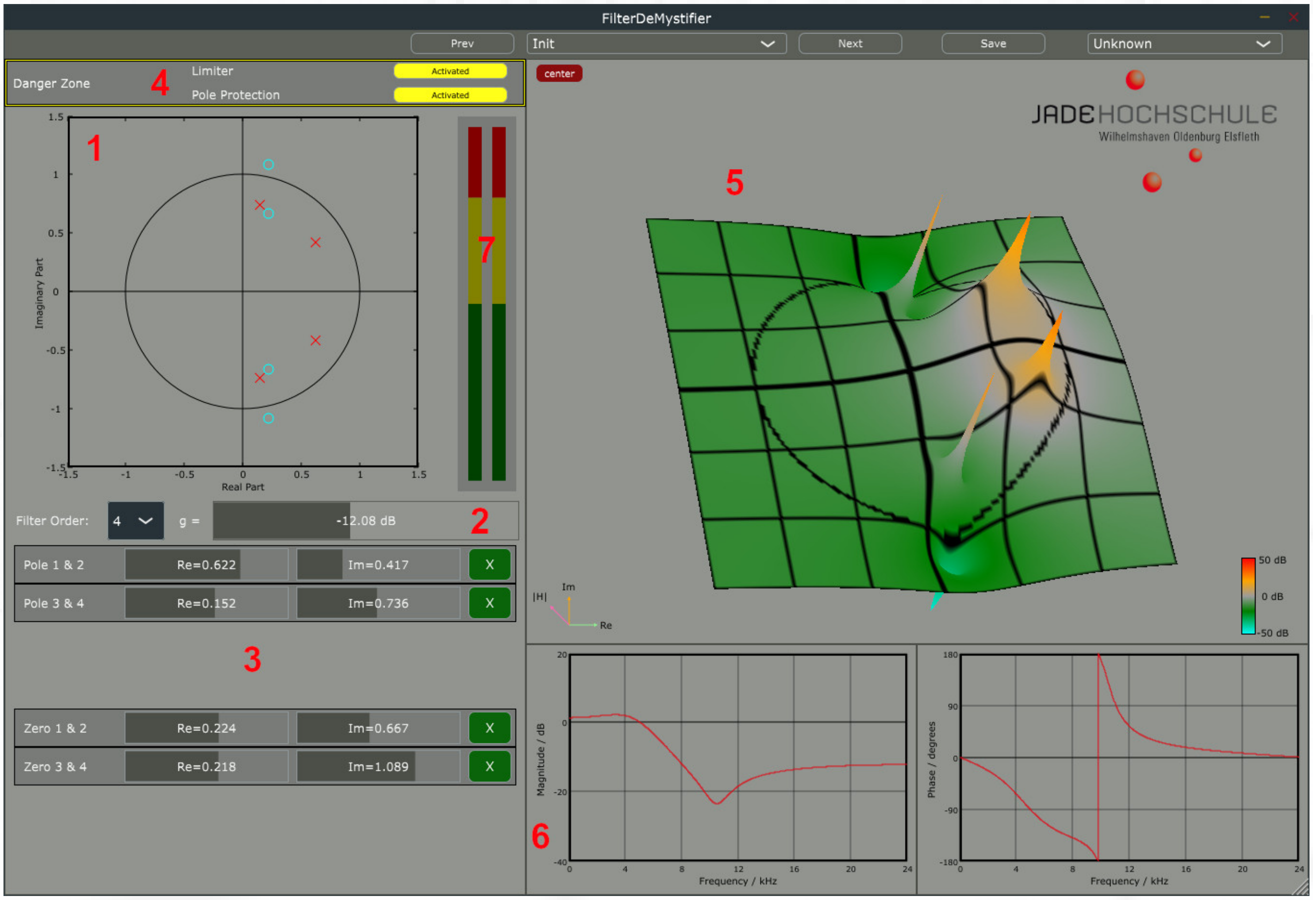
After the first launch of the plug-in you will find all presets at the following location (necessary to remove them after you delete the plug-in). These are plain xml files.

```
Windows:
C:\Users\yourUSERNAME\AppData\Roaming\Jade Hochschule\FilterDeMystifier

Mac:
/Users/yourUSERNAME/Library/Audio/Presets/Jade Hochschule/FilterDeMystifier

Linux:
/home/YourUSERNAME/.config/Jade Hochschule/FilterDeMystifier
```

3 How to use



FDM consists of three sections: The input parameter part, the output displays, and a special section for security settings.

3.1 Input parameter (1,2,3)

The input section controls the behaviour of the four second order filter sections and the gain.

3.1.1 PZ Diagram

The pole-zero diagram with the unit circle is the main user input. Drag a pole or a zero around. By default, the poles are restricted to remain within the boundaries of the unit circle to prevent unstable filter configurations. If you create an odd order filter, the single pole and single zero can only be dragged along the real axis. Only poles and zeros in the upper half can be used, the lower half is merely mirroring the upper half (this is necessary to have real-valued filter coefficients, and in audio everything is real.)

3.1.2 Gain

The gain slider allows for gain corrections, if the output is too loud or too soft.

3.1.3 PZ List

The slider for the real and imaginary parts of the poles and zeros is an equivalent user input to the PZ diagram. If your are interested in input coefficient values, it can be easier to use these sliders or numerical input.

The filter order combo box allows to change the order of the filter.

3.2 Security controls (4)

The "Danger Zone" features two buttons to control output signals that are far too loud for normal usage. The first one applies an internal brick-wall limiter to the output of the filter sections. This limiter guarantees output signals below *0dBFS*.

The second button toggles the restriction for every pole radius to be less than one. Deactivating it will allow for unstable filters. Usually there would be an audible click followed by silence. In this implementation the filters are internally restricted by a non-linear clipping function – meaning you will hear a loud and very distorted signal. Since this is very unpleasant, you should only use this setting with the limiter in place.

3.3 Output displays (5,6,7)

Three output displays help you to understand the behaviour of our filter for given poles and zeros.

3.3.1 z-Plane Graph

The most important one is the 3D z-plane magnitude graph. It shows the filter magnitude on the z-plane. All poles increase the magnitude and all zeros reduce the magnitude. This is visible as spikes and dips in the z-plane. The graph can be rotated by mouse moves and enlarged by using the mouse wheel.

3.3.2 Bode Diagram

The Bode diagram is the classical display of the transfer function of a filter. The left graph shows the magnitude, and the right graph the phase response with respect to the frequency. Typically, what you hear corresponds to what you see in the magnitude because our ears are less sensitive to phase changes.

If you look carefully you can see that the magnitude is exactly the height of the circle in the z-plane graph starting on the right side (zero degrees), if you rotate the display like the pole/zero input diagram and "walk" in counter-clockwise (mathematically positive) direction around the circle.

3.3.3 Level meter

The level meter is simply there to check how loud your output will be, if you listen to it. It shows RMS and peak values.

3.4 One simple example

Start with a second order filter (filter order = 2), put the pole close to the real axis (x axis) and close to the unit circle. The input signal will be amplified substantially, since you have put a pole close to the unit circle. Now move the zero close to the pole and the effect of the pole is counter-acted. If you move the zero onto the unit circle, the corresponding frequency is cancelled out completely (notch filter). If you move the zero outside the unit circle, look at the phase plot. The phase performs a jump and the final filter is not minimum-phase anymore (minimum-phase can be a desirable feature, should you want to invert the filter).

4 The presets

All presets are designed for a sampling rate of $f_s = 48000$ Hz. The parameters in this plug-in are all positions on the z-plane. This leads to normalised filter coefficients, and a different sampling rate will change the filter sound.

4.1 Standard cut filter

Several design methods exist for typical cut filter (lowpass, highpass, bandpass and bandstop). The four best known are:

- Butterworth: This design is maximally flat at 0 Hz. It is often used for loudspeaker filter-banks in its squared version, known as Linkwitz-Riley filter.
- Chebbycheff1: This filter has ripples in the passband, but a steeper response for the stopband compared to the Butterworth design
- Chebbycheff2: This version of the Chebbycheff filter has the ripple in the stopband, but is smooth in the passband.
- Elliptic or Cauer: The last filter design has ripples in both bands. It can be used to have a very fast change between pass- and stopband.

4.2 Audio Filter

For more or less all mixing jobs you need equalisers, and the typical audio filters are peak and shelving filters.

The peak filter is given by its centre frequency, the desired gain and the bandwidth (often called Q-factor). The approximation here is not the result of a typical peak design (RBJ cookbook) but behaves identically. Especially close to half of the sampling rate, the filter will be asymmetrical and the overall form does not correspond to the analogue original. This can be improved by using the design from Orfanidis [2] or by using analogue prototypes and a least-squares approach [5].

4.3 Vocals

Typical vocal filters are all-pole filters. Therefore, all zeros are in the middle $z = 0$. The poles are determined by the formant frequencies of the different vocals. If you want to determine the pole location, look at a formant list of your choice (e.g. https://www.classes.cs.uchicago.edu/

archive/1999/spring/CS295/Computing_Resources/Csound/CsManual3.48b1.HTML/Appendices/table3.html
) and compute the pole position with a radius close to one.

$$z_1 = 0.99 \exp\left(j\ 2\pi \frac{f_{\text{Formant}}}{f_s}\right) \tag{1}$$

The presets include two examples for an /a/, an /i/, and an /e/.

4.4 Synthesizer filter (without the non-linear part)

One of the most interesting kinds of filters are the ones for synthesizers. The approximation here is not very precise or close to the real filters, since all the non-linearities and the special behaviour for changing parameters are missing. However, the linear transfer function is close to the analogue counterparts.

4.5 Some signal processing specialities

All the filters so far were IIR filters. However, you can build linear-phase FIR filters as well. The poles must all be in the middle and therefore have no effect at all. If you now mirror a zero at the unit circle (one inside, one outside) you get a straight line in the phase response. It is not possible to have a linear-phase transfer function with stable IIR filters. Try it out! As long as your poles are inside the unit circle, you will have a curvy phase response.

Another special filter is a so-called all-pass filter. It does not change the magnitude at all, but changes the phase. This is a very versatile tool, if combined in other filter structures. For example you can use all-pass filters for phasers, filterbanks or decoupled filter equalisers. To build an all-pass filter you have to mirror all poles (remember for stable filters, all poles are inside the unit circle) at the unit circle and put zeros on these mirrored positions.

5 Boring background: The signal processing stuff

If you exclude non-linearities and automation/change of parameters over time, all standard filters are linear time-invariant systems. Thus, all of the wonderful LTI signal processing theory applies.

5.1 Time-domain difference equation

A higher-order IIR filter (Infinite Impulse Response) can (but should not) be implemented directly by using the difference equation

$$\begin{aligned} y(n) &= b_0 x(n) + b_1 x(n-1) + b_2 x(n-2) + \cdots \\ &\quad -a_1 y(n-1) - a_2 y(n-2) - \cdots \end{aligned} \tag{2}$$

where b_n denote the transversal coefficients and a_m the recursive coefficients.

Usually, you would implement this equation as second order sections (SOS) (see this plug-in for reference) and perhaps not in the direct form 1 – we use direct form 1 but alternatives could be e.g. a canonical form like direct form 2 (perhaps not), or ladder structures.

5.2 z-plane

By using the z-transform

$$\mathcal{Z}\{\cdot\} = \sum_{k=-\infty}^{\infty} (\cdot) z^{-k} \tag{3}$$

you can analyse the difference equation to get the system function

$$H(z) = \frac{b_0 + b_1 z^{-1} + b_2 z^{-2} + \cdots + b_{N-1} z^{-(N-1)}}{1 + a_1 z^{-1} + a_2 z^{-2} + \cdots + a_{M-1} z^{-(M-1)}} \tag{4}$$

or for the SOS structure (N = M)

$$\begin{aligned} H(z) &= \frac{b_0 + b_1 z^{-1} + b_2 z^{-2} + \cdots + b_{N-1} z^{-(N-1)}}{1 + a_1 z^{-1} + a_2 z^{-2} + \cdots + a_{N-1} z^{-(N-1)}} && (5) \\ &= g \, \frac{1 + b_{1_1} z^{-1} + b_{2_1} z^{-2}}{1 + a_{1_1} z^{-1} + a_{2_1} z^{-2}} \cdot \frac{1 + b_{1_2} z^{-1} + b_{2_2} z^{-2}}{1 + a_{1_2} z^{-1} + a_{2_2} z^{-2}} \cdot \\ &\quad \cdots \frac{1 + b_{1_K} z^{-1} + b_{2_K} z^{-2}}{1 + a_{1_K} z^{-1} + a_{2_K} z^{-2}} && (6) \end{aligned}$$

where g is the overall gain and K the number of SOS (usually $N/2$ for even order filters).

5.3 Poles and Zeros

By solving the polynomials of the system function you can compute the poles and zeros (necessary step for constructing the SOS).

$$\begin{aligned} H(z) &= g \, \frac{(z - n_0)(z - n_1) \cdots (z - n_{N-1})}{(z - p_0)(z - p_1) \cdots (z - p_{M-1})} && (7) \\ &= g \, \frac{\prod_{i=0}^{N-1} (z - n_i)}{\prod_{i=0}^{M-1} (z - p_i)} && (8) \end{aligned}$$

The positions of the poles and zeros determine the overall behaviour of the filter. For example, for a stable filter all poles have to be inside the unit circle (i.e. their distance from the centre must be less than one). If all zeros and poles are located inside the unit circle the system is minimum phase. A linear-phase filter has all poles at the centre of the z-plane, and the zeros are mirrored at the unit circle. For a stable all-pass filter the poles are mirrored by zeros at the unit circle.

6 Further topics

If you want to dive into dsp and filter design this is just the beginning! Think of quantisation (more details in the book of Zölzer [6]), non-linear and zero-delay filtering (book of Vadim Zavalishin) or time-variant behaviour (click-free morphing of parameters and filter designs, e.g. decoupled structures [4])

Here is a short list of interesting websites, books and papers for further reading (and since it cannot include everything, please forgive us this very subjective list, especially if you do not find your excellent paper, book, or website here.).

6.1 Source code

This plug-in is open source. you can find the source code here: (https://github.com/ArnoSchiller/FilterDeMystifier)

We are aware that the quality of the code could be improved significantly. However, if you are familiar with JUCE, you will have no problem to understand what's going on. Please report issues at the GitHub repository. Please keep in mind, this started as a student project for course work.

6.2 Web links

https://webaudio.github.io/Audio-EQ-Cookbook/audio-eq-cookbook.html
https://www.musicdsp.org/
https://www.willpirkle.com/

References

[1] Robert Bristow-Johnson. The equivalence of various methods of computing biquad coefficients for audio parametric equalizers. In *Audio Engineering Society Convention 97*. Audio Engineering Society, 1994.

[2] Sophoncles J Orfanidis. Digital parametric equalizer design with prescribed nyquist-frequency gain. *Journal of the Audio Engineering Society*, 45(6):444–455, 1997.

[3] Will C Pirkle. *Designing Audio Effect Plugins in C++: For AAX, AU, and VST3 with DSP Theory*. Routledge, 2019.

[4] Phillip Regalia and Sanjit Mitra. Tunable digital frequency response equalization filters. *IEEE transactions on acoustics, speech, and signal processing*, 35(1):118–120, 1987.

[5] Thorsten Schmidt and Joerg Bitzer. Digital equalization filter: New solution to the frequency response near nyquist and evaluation by listening tests. In *Audio Engineering Society Convention 128*. Audio Engineering Society, 2010.

[6] Udo Zölzer. *Digital audio signal processing*, volume 9. Wiley Online Library, 2008.

7 Legal stuff