

DSEQ

Manual



TBProAudio 2020

1 Introduction

Welcome to **DSEQ**, a dynamic spectral equalizer.

Dynamic equalizers are widely used in the mixing and mastering arsenal. They usually work in the time domain and have limited frequency selectivity due to the nature of their filters. High frequency selectivity can be achieved in the frequency domain. This means that disturbing singular frequencies are dimmed without touching the frequencies on the left and right too much. Such characteristics are essential for de-essing, softening digital highs, taming annoying resonances or de-masking frequency regions.

DSEQ works fully in the frequency domain, providing very high frequency selectivity thanks to self-adjusting dynamic equalizers. **DSEQ** does its job with highest transparency even pushed hard. This avoids distortion, phasing and artefacts even with very critical sound material. And **DSEQ** offers several quality modes taming even singular frequencies.

On top **DSEQ** provides a very easy to use interface. With just a few parameters (threshold, selectivity and attack/release) a broad range of applications can be covered. Even the initial parameter set could make your mix more transparent. Just pick the threshold parameter and lower it step by step. From there you can start focusing on specific frequency regions by using the pre-filters.

DSEQ can be used in all mix and master situations:

- de-essing vocals
- taming resonances in e.g. drum, guitar, vocal recordings
- removing digital harshness
- balancing the mix
- de-masking frequency regions
- support of pink noise mixing/mastering

2 How does DSEQ work

DSEQ analyses the input or side chain signal in the frequency domain and triggers then the dynamic equalizers depending on the **threshold**. This is done for each singular frequency that is available during the Fast Fourier Transformation (FFT).

The frequency **selectivity** determines whether and to what extent neighbouring frequencies are also affected. **Attack** and **Release** control how quickly the dynamic equalizers react.

The **slope** parameter lets you turn the spectrum of the input or side chain signal before going into the dynamic equalizers. With a positive **slope** value the dynamic equalizers with higher frequencies trigger earlier, with lower frequencies later. Just set the **slope** value to +3dB if you work with **pink noise mixing/mastering** method.

DSEQ works normally in the entire frequency range. You can limit the effect to certain frequency regions (e.g. highs for de-essing). Therefore, the signal for the dynamic equalizers can be processed using **pre-filters**. **DSEQ** offers various filter types such as cut, peak and shelf filters. The filter **sensitivity** determines how strongly the signal is recognized by the dynamic equalizers.

DSEQ can either work with a fixed threshold values over the spectrum or with a **custom threshold curve** calculated from any audio file.

DSEQ needs some time to analyse the signal and trigger the dynamic equalizers. **DSEQ** therefore creates a notable latency that is normally compensated by any modern DAW when PDC (plugin delay compensation) is activated.

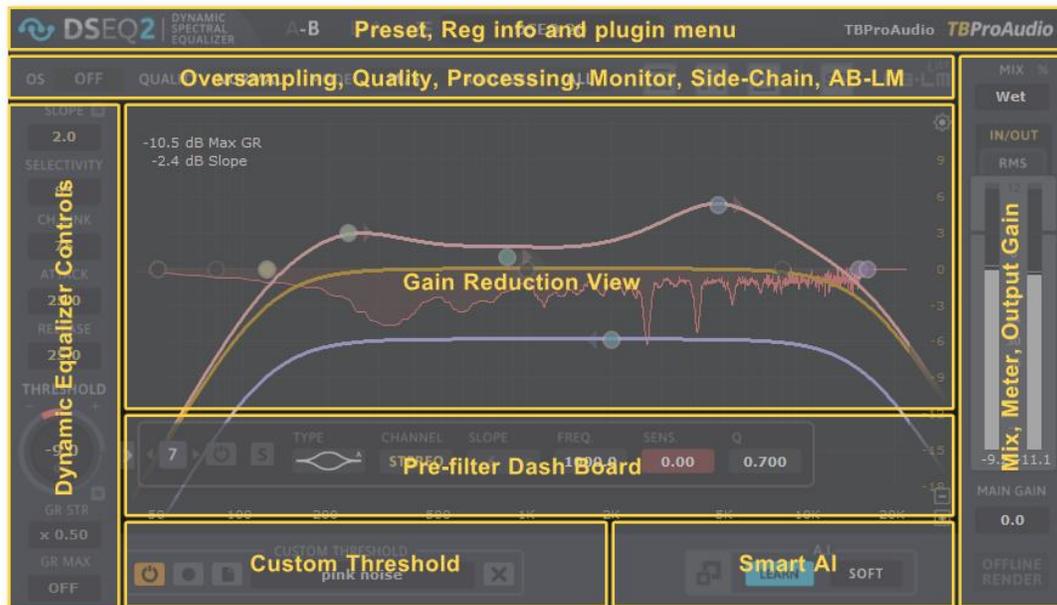
DSEQ V2.0 introduces the smart AI function and a corresponding GUI page to setup **DSEQ** automatically depending on the input material.

3 Features

DSEQ offers following features:

- **NEW**: smart AI function,
- **NEW**: smart AI GUI page
- smooth linear phase dynamic filter
- full frequency range processing, 10Hz – 22kHz
- **NEW**: 7 different quality modes: eco eco, eco, normal, high and ultra, ultra 2/3
- left/right and mid/side processing
- global slope for detector signal (e.g. pink noise mixing/mastering)
- custom threshold curve
- limit of gain reduction
- compression control
- automatic optimal slope parameter calculation
- **NEW**: automatic optimal threshold parameter calculation
- 12 independent pre-filter bands, analogue/digital style
- side-chain support
- parameter A/B
- multiple signal monitoring modes (pre-filter, delta, side chain)
- multiple spectrum analyzing modes (stereo, left, right, mid, side)
- perceptual loudness matching (powered by AB-LM Lite)
- extended input/output meter, Peak/RMS/EBU/VU
- narrow-band sweeping mode
- **NEW**: filter band listen mode
- sample-exact bypass toggle for all processing modes
- sample rates min. up to 192 kHz
- "real" over sampling, up to 4x
- offline render modes
- EQ cheat sheets
- continuous calculation of the spectrum slope
- precise parameter input
- easy to use GUI
- **NEW**: GUI resize up to 300%
- **NEW**: many presets including smart AI setups
- preset management including custom setups
- full DAW automation support
- 64-bit internal processing
- very efficient CPU usage design

4 Overview



- The upper section offers plug-in settings such as over-sampling, quality modes and processing mode.
- In the middle is the main display for EQ / spectrum, with which you can monitor the delta spectrum (input - output) and adjust the filter response curve.
- Below the spectrum display is the filter dashboard that controls the pre-filter parameter.
- The controls of the dynamic equalizer are on the left.
- Measuring devices and AB-LM for the perception of the volume are on the right side.
- Below are the controls for the custom threshold and the smart AI functions

Interactive spectrum/filter curve display



- The filter button activates the filter and can be dragged with the mouse. Right mouse click opens the filter popup menu.
- The pre-filter dash board shows current filter settings. The filter number is set by the “Nr.” control.
- The analyzer popup menu hosts settings of the spectrum analyzer.
- The top left area shows the current maximum gain reduction.

5 Minimum System Requirements

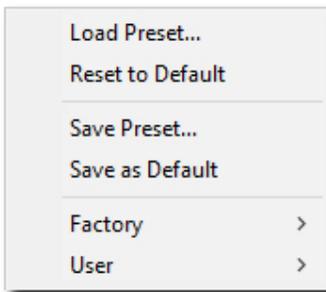
- Windows XP SP3 or newer
- Mac OS X 10.9 or newer
- Win: 32/64 Bit VST, 32/64 Bit VST3, 32 Bit RTAS, 32/64 Bit AAX
- OS X: 32/64 Bit VST, 32/64 Bit VST3, 32/64 Bit AU, 32 Bit RTAS, 32/64 Bit AAX
- Tested with: Cockos Reaper, Steinberg Cubase/Nuendo/Wavelab 6/7/8/9/10, FL Studio 12.x/20.x, PT10/12/2019, Reason 9.5/10, Studio One, Ableton Live, Logic Pro X, Cakewalk, Samplitude, Adobe Audition, Bitwig Studio

6 Plugin Controls

Mouse usage:

Click and drag horizontally changes parameter value.
Shift click and drag slows down parameter change.
Ctrl click and drag accelerates down parameter change.
Alt click rests to default value.
Double click resets parameter.
Right click opens value edit box, finish with enter.

6.1 Preset Menu



Preset menu loads and saves user presets. Presets stored under %localappdata%/DSEQ (Windows) or /Users/xxx/Library/Application Support/ DSEQ (Mac OSX) are imported as user presets. Previous and next button step through all presets. “Reset to default” sets the current plugins settings to the default settings. “Save as Default” sets the current settings as start-up/default settings.

DSEQ comes with several factory presets demonstrating the capabilities of the plugin. With version 1.5.2 DSEQ stores the current custom threshold with the user preset.

6.2 Parameter AB

This control lets you on the fly AB different parameter settings. Additionally the current set could be copied to B or A.

With version 1.5.2 DSEQ includes also custom threshold.

So select “A”, load a user preset with custom threshold, switch to “B”, load another user preset with custom threshold. Now you can toggle A/B on the fly. Please make sure that both settings have the same quality mode to avoid audio drops.

6.3 Over-sampling

Over-sampling reduces the magnitude/phase distortions near Nyquist frequency. Both pre-filters and dynamic equalizers are over-sampled. Please to note that over-sampling impacts mainly the smoothing of the gain reduction curve (gain reduction of all dynamic equalizers together). Depending on the sound material it could make the impact of DSEQ smoother.

6.4 Quality Mode

Quality mode determines the size of the FFT and therefore the number of frequencies to be analysed. This control also impacts the plugin latency.

6.5 Channel Mode

Channel mode selects either left/right or mid/side processing mode. Usually left/right is the choice. In special cases independent processing of the mid and side channel is useful (see M/S processing)

6.6 Analyzer

Analyzer mode selects the main analyzer view mode:

All: shows the summed left/right spectrum.

Left: left spectrum.

Right: right spectrum.

Mid: M/S mid spectrum.

Side: M/S side spectrum.

6.7 Bypass

Bypass enables click-free plugin bypassing.

6.8 Pre-filter monitoring

Pre-filter monitoring enables the listening of the pre-filter signal (either normal signal or side chain signal).

6.9 Delta monitoring

Delta monitoring enables the listening of the difference signal of input minus output. This makes the impact of pre-filter and dynamic equalizers much more audible.

6.10 Filter band listen

The filter band listen mode enables the listening of the output signal in a specific filter band with all DSEQ processing. Filter frequency and Q are determined by the selected pre-filter. Just click on one of the pre-filter buttons (no need to be active) and move button around. In combination with the delta monitoring mode, you can precisely verify the effect of DSEQ on the signal.



6.11 Side chain

Side chain toggles usage of side chain (for VST2 CH 3+4) off/on

6.12 Main Gain

Main gain adjusts the output-gain of **DSEQ**, in dB.

6.13 Mix

Mix selects the mix of processed and non-processed (wet and dry) signals.

6.14 Dynamic Equalizer



6.14.1 Slope

Slope turns the spectrum of the input signal by the value in dB/Oct around 1kHz. Positive values bring the highs more in focus, negative values the lows. If you are doing pink noise mixing/mastering, just set slope to +3dB. This brings the highs more and the lows less in the focus of the dynamic equalizers.

6.14.2 Calculate slope parameter

This control sets automatically the optimal slope parameter based on main input signal and optionally custom threshold.

6.14.3 Selectivity

Selectivity determines how much the trigger of a singular dynamic equalizer affects the neighbours. Higher values limit the effect to more singular dynamic equalizer, but could also create more distortion.

6.14.4 Channel Link

Channel link determines how strong left/mid and right/side channel are linked together. Lower values give more separate processing. For stereo master a value 75% is useful.

6.14.5 Attack

Attack controls how quickly the gain reduction of the dynamic equalizers starts. As **DSEQ** could use more than 1000 dynamic equalizers (depending on processing quality) the control of each dynamic equalizer parameter is not suitable. Therefore **DSEQ** controls all of them combined. Lower values give shorter attack times, higher values longer attack times.

6.14.6 Release

Release controls how quickly the gain reduction of the dynamic equalizers releases. As **DSEQ** could use more than 1000 dynamic equalizers (depending on processing quality) the control of each dynamic equalizer parameter is not suitable. Therefore **DSEQ** controls all of them combined. Lower values give shorter release times, higher values longer release times.

6.14.7 Threshold

Threshold sets the signal level when the dynamic equalizers are triggered. Use lower values for more gain reduction. The current gain reduction is displayed in the spectrum view. You can use the slope parameter to increase or decrease the threshold additionally by x dB/Oct.

6.14.8 Calculate threshold parameter

This control sets automatically the optimal threshold parameter based on main input signal.

6.14.9 GR strength

GR strength boosts or lowers the gain reduction additionally. Usually the factor x1 is sufficient for most cases. In surgical cases (e.g. eliminating outstanding frequencies) higher factors can be applied.

GR strength works similar to the ratio parameter of a compressor. The formula is:

Strength = 1 - 1 / Ratio or

Ratio = 1 / (1 - Strength).

6.14.10 Max GR

Max GR limits the maximum gain reduction to the set value.

6.15 Main Meter

A small LED on top lids up if level goes over 0.0 dBFS. Numbers below the meters display the maximum values.

6.15.1 Channel Mode

Channel mode selects which channel combination displayed by the meters: In/out, In Left/Right, Out Left/Right, In Mid/Side, Out Mid/Side and off.

6.15.2 Meter Mode

Meter mode selects various measurement modes of the meters: Peak (dBFS), RMS (AES 17, dBFS), EBU ML (LUFS), EBU SL (LUFS) and VU (dBVU).

6.16 Pre-Filter Parameter



Pre-filter control the signal fed into the dynamic equalizer. You can take parts of the spectrum out (not processed by the dynamic equalizers) or boost (more processed by the dynamic equalizers).

The display of the pre-filter dash board can be toggle with the small icon just below.

6.16.1 Nr.

Controls display of current pre-filter parameter set. Click on it to open the number menu, use the mouse wheel or the small arrows left and right to change current set.

6.16.2 Active

Active enables/disables the current pre-filter.

6.16.3 Solo

Solo makes all other pre-filters inactive.

6.16.4 Type

Type selects the filter type of the current pre-filter:

Peak: peak filter

Low Cut: high pass filter

High Cut: low pass filter

High Shelf: shelf filter boosting high frequencies

Low Shelf: shelf filter boosting low frequencies

Tilt Shelf: combination of low shelf/high shelf in one filter

Flat Tilt: flat tilt filter

Band Pass: passes certain frequencies of the spectrum

Notch: cuts certain frequencies out of the spectrum

Gain: simple “gain filter”. To be used to lower e.g. side signal while processing mid/side.

Digital Peak: peak filter

Digital Low Cut: high pass filter

Digital High Cut: low pass filter

Digital High Shelf: shelf filter boosting high frequencies

Digital Low Shelf: shelf filter boosting low frequencies

Digital Tilt Shelf: combination of low/high shelf in one filter

Digital Band Pass: passes certain frequencies of the spectrum

6.16.5 Channel

Channel controls which part of the signal (all/left/right/mid/side) is affected by the filter

6.16.6 Slope

Slope sets the steepness of the filter (if applicable), 6dB to 96dB/octave.

6.16.7 Frequency

Frequency sets the frequency of the filter band, 50Hz to 20kHz

6.16.8 Sensitivity

Sensitivity sets the filter sensitivity (if applicable), -48dB to 30dB.

6.16.9 Q-Factor

Q sets the bandwidth of the filter band (if applicable), widening or narrowing the filter response.

6.16.10 Filter button

Click and drag the filter button to set frequency, gain or Q of the filter. Additional shift key slows down the mouse movement.

Double click on the button activates/de-activates the filter. Additional ctrl key resets filter settings. Right mouse click (or shift + ctrl key + left mouse click) opens the filter popup menu:

Title: activates/de-activates the filter

Type: selects current filter type

Channel: selects current stereo placement

Slope: selects filter steepness

Solo: solos filter

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Reset: resets filter to default values

Mouse usage:

Mouse drag: changes filter **frequency** and **sensitivity**

Shift key + mouse drag: slows down mouse movement

Ctrl key + mouse drag: changes filter **frequency** only

Ctrl + shift key + mouse drag: slows down mouse movement

Alt key + mouse drag: changes filter **sensitivity** only

Alt + shift key + mouse drag: slows down mouse movement

Mouse wheel: changes filter **Q** (where applicable)

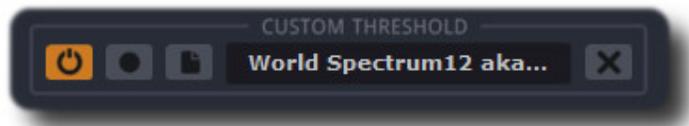
Shift key + mouse wheel: slows down change of **Q** (where applicable)

Control key + mouse wheel: changes filter **channel**

Control + shift key + mouse wheel: changes filter **type**

Alt key + mouse wheel: changes filter **slope** (where applicable)

6.17 Custom threshold



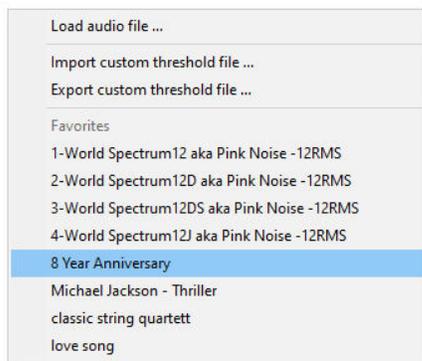
The dynamic EQs can either work with fixed threshold values or custom values.

DSEQ calculates the custom threshold values from any audio file or even the recorded input signal.

On/Off: toggles custom or fixed threshold. GR display reflects current setting: blue (fixed threshold) and red (custom threshold).

Record: analyses the current input signal (either main or side chain) and calculates the threshold curve. Toggle to start or stop. During recording all **DSEQ** processing is bypassed. Record clear all current data.

File menu:



The menu lets you load and analyse audio files (Mono/stereo **WAV** and **AIFF**) and use as custom threshold. Custom threshold data can be imported and exported (ext. cth). The export is done to the **DSEQ** data folder. The menu lists the first 20 data files from there as favourites for quick import. You can also drag&drop wav/aiff/cth files from DAW internal browser or from Windows Explorer or Mac Finder on this control.

Clear: clears current custom threshold curve

Notes:

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- With version V1.5.2 you can A/B custom thresholds, either load with the custom threshold menu or with a user preset.
- Each custom threshold curve has its own internal slope, so it is useful to set the parameter slope to 0.0 dB at the beginning.
- After the recording or importing of audio files the custom threshold curve is normalized during the analysis. So it is useful to start with a high threshold parameter value like +30.0 and decreased it afterwards.
- Usually the threshold values of the spectrum border regions (below 50 Hz and above 16 kHz) are very low. So it is useful to set a low- and high-cut pre-filter additionally. Please refer to the preset “General – custom threshold”.

6.18 Smart AI



DSEQ V2 makes it even easier to tame digital harshness thanks to its smart AI function. Set up DSEQ as normal, selected desired AI learn mode, press play to start the DAW and then the learn button. DSEQ calculates the optimal slope and threshold parameter within seconds.

Open smart AI GUI: the icon opens the simplified smart AI GUI.

Learn: starts DSEQ smart AI analysing. Place the play cursor at the desired part of the track (most probably the loudest part), and then press play. It takes ca. 10 seconds to analyse the input signal.

Learn mode: sets the different smart AI parameters depending on the modes soft, normal and strong.

6.19 Smart AI GUI



The smart AI GUI simplifies the working with DSEQ. It presents the most important controls in an easy to use GUI. The smart AU GUI works optimal with the provided AI presets from the preset menu.

Controls from top left to right:

- close smart AI GUI and return to expert GUI
- show/hide smart AI instructions
- AB-LM
- plugin bypass
- delta monitoring

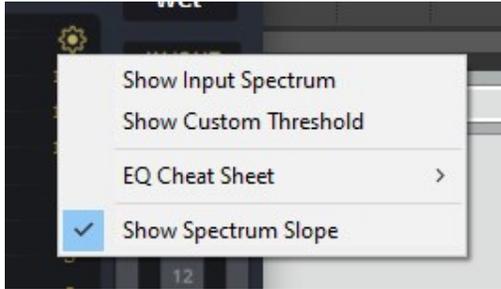
On middle left side there is the smart AI learn button. Just press to start the learning process. On the right side the smart AI gain reduction meter. It shows the current gain reduction applied to the signal. The meter should be between 25% and 75%. Otherwise select another part for the analysis.

You can switch seamlessly between smart AI and expert GUI. All settings made in expert mode stay when switching to smart AI GUI.

How to use smart AI GUI:

- Select any of the AI presets from preset menu drop down.
- Set the DAW play cursor at the desired part of the track (usually the loudest part) and then press play.
- Press the learn button and wait for a few seconds until DSEQ finished the detection process.
- DSEQ will automatically adjust the best setting for taming digital harshness.

6.20 Analyzer popup menu



Show Input Spectrum: toggles display of input spectrum

Show Custom Threshold: toggles display of custom threshold curve

EQ Cheat Sheet: toggles display of various overlaid cheat sheets.

Show Spectrum Slope: toggles display of the estimated spectrum slope

6.21 Narrow-band Sweeping Mode (detector signal)



The narrow-band sweeping mode is enabled by left mouse click + ctrl key in the free area of the spectrum view. The frequency is changed by moving the mouse left/right, the Q of the filter is changed by moving the mouse up/down.

7 Info menu



Click on the DSEQ 2 logo to access the info menu.

7.1.1 About

Info about the plugin

7.1.2 Open Manual

Opens online manual

7.1.3 Check version

Opens online version info.

7.1.4 Enable tooltips

Toggles GUI hints.

7.1.5 GUI Scale

Scales GUI from 50-300%

7.1.6 Link quality mode to sample rate

If over-sampling is set to 2x, quality mode n+1 is used, e.g. if quality mode is set to “Normal” internally quality mode “High” is used.

If over-sampling is set to 4x, quality mode n+2 is used, e.g. if quality mode is set to “Normal” internally quality mode “Ultra” is used.

Please note that the quality mode is always finally limited to “Ultra 3”.

7.1.7 Offline render quality

Select quality mode for offline render (aka bounce, mix-down, audio export). If item “1:1” is selected no change from online to offline render is made.

7.1.8 Offline render over-sampling mode

Select quality mode for offline render (aka bounce, mix-down, audio export). If item “1:1” is selected no change from online to offline render is made.

7.1.9 Note to offline render modes

Please note that not all DAWs either report offline render status properly or handle PDC messages in offline render mode properly. So either the offline render modes of **DSEQ** are not activated or the tracks with **DSEQ** are out of sync due to missing PDC.

TBProAudio tested following DAWs and plugin formats successfully:

- Reaper: VST2x32, VST2x64, VST3x32, VST3x64
- Wavelab: VST3x64
- Cubase: VST3x64
- Studio One: VST3x64, AUx64
- Ableton: VST2x64, AUx64
- FL Studio: VST3x32, VST3x64
- ProTools: AAXx64

So if your DAW/plugin format is not listed just leave the setting to “1:1”.

8 AB-LM Lite

AB-LM Lite is a **perceptual A/B loudness matching** algorithm to help you avoid the pitfall of 'louder is better', so you can evaluate the impact of the plugin on your incoming and outgoing audio signal at equal loudness.

When you engage AB-LM Lite, the algorithm analyzes the incoming signal and compares it to the outgoing signal via its RMS values and applies an automatic gain adjustment so you can focus on the plugin settings and what they are actually doing to your audio, without being distracted by loudness differences.

Please to note that AB-LM Lite limits the gain adjustment to +/- 12dB to avoid gain overshoots.



AB-LM Lite lets you automatically adjust the loudness either quickly (mode fast) or gently (mode slow).

AB-LM Lite is an automatic, simplified algorithm to help your workflow, based on our more **advanced dedicated AB-LM** plugin, which allows perceptual loudness matching of third party plugins and even complete plugin chains. AB-LM also offers more advanced options, including advanced measurement modes and snapshots. For the more advanced features of AB-LM, please see https://www.tb-software.com/TBProAudio/ab_lm.html.

9 Demo mode versus Registered mode

In demo mode (without registering) the plugin mutes audio every 90 seconds for a short period. This could be circumvented by clicking on the "TBProAudio" logo within 90 seconds.

You can register and activate here: <http://www.tb-software.com/tbproaudio>.

10 Registration/Activation

This plugin must be registered and activated in order to be fully functional. You can purchase the license key here: <http://www.tb-software.com/tbproaudio>.

After purchase, you will receive the license key packed in a zip file by email. Unzip the license file to a suitable location on your PC (e.g. desktop). The license file is called "product name.license". Add the plug-in to any track in your DAW, click on the text "Demo: Click here to activate it" and select the license file. Remove the plug-in in your DAW and add it again. The GUI now says "Registered to your name".

11 Conclusion

So finally if you have any questions or suggestions just let us know. And have fun with our tools.

Your team from TBProAudio :-)