DSEQ3

Manual

TBProAudio 2024

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 witch **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.

DSEQ V3.0 introduces the gain reduction spectrogram and free GUI scaling and resizing.

DSEQ V3.5 introduces **natural phase** dynamic filter and adaptive slope/threshold.
The natural phase dynamic filter creates less pre-ringing.
The **adaptive slope** and **threshold** combined with the internal loudness matching (AB-LM Lite Slow) let DSEQ work in "auto pilot" mode with no or small user interaction. It gives an instant improvement of resonances or digital harshness. Just select one of the presets "Adaptive – xxx" and start right away.

DSEQ V3.6 introduces **smart silence processing** to save CPU with multiple instances.

DSEQ V3.7 introduces **undo/redo** und **fixed GUI scaling**.

**DSEQ V3.8** introduces rewritten GR display with adjusted feature set.

DSEQ V3.9 introduces side-chain gain and display of output spectrum.

3 Features

DSEQ offers following features:

- smart AI function,
- smart AI GUI page
- smooth linear and natural phase dynamic filter
- full frequency range processing, 10Hz – 22kHz
- 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/ threshold parameter calculation
- adaptive slope and threshold
- 12 independent pre-filter bands, analogue/digital style
- side-chain support including gain control
- parameter A/B
- parameter undo/redo
- multiple signal monitoring modes (pre-filter, delta, side chain)
- multiple spectrum analyzing modes (stereo, left, right, mid, side, SC)
- perceptual loudness matching (powered by AB-LM Lite) and gain "freezing"
- extended input/output meter, Peak/RMS/EBU/VU
- 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
- gain reduction spectrogram
- easy to use GUI
- free GUI scaling and resizing up to 300%
- fixed GUI scaling up to 300%
- many presets including smart AI setups
- preset management including custom setups
- full DAW automation support
- touch control
- 64-bit internal processing
- smart silence processing
- very efficient CPU usage design

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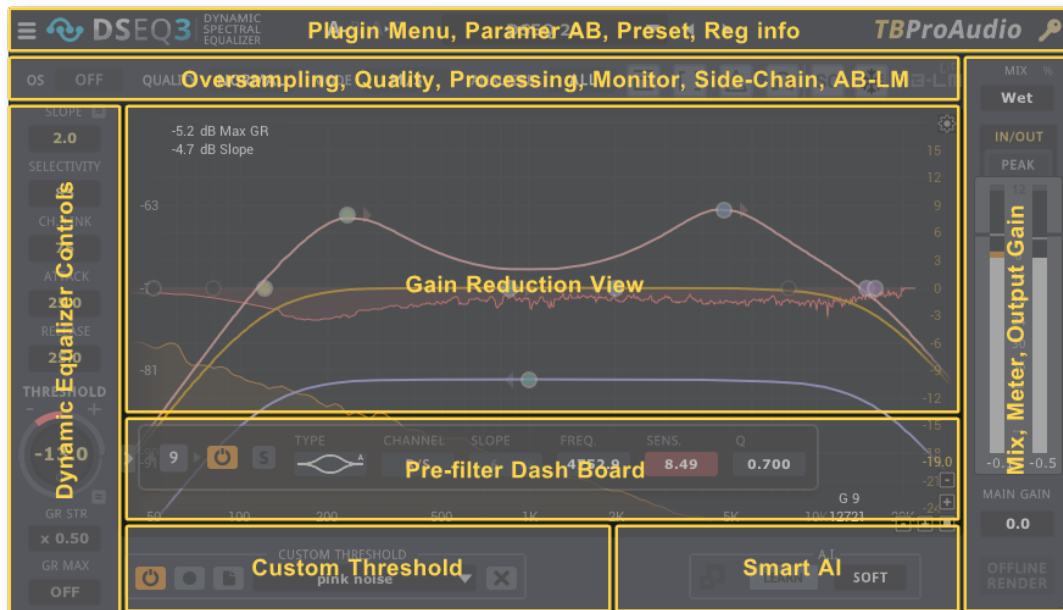
- The upper section offers plug-in settings such as over-sampling, quality modes and processing mode.
- In the middle is the main display for EQ / spectrum, with which you can monitor the delta spectrum (input - output) and adjust the filter response curve.
- Below the spectrum display is the filter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

1. Mouse double click activates the filter and is dragged with the mouse. Right mouse click opens the filter popup menu. See more under the section pre-filter board.
2. The lines between the filter buttons represent the filter response curves of the different channels (stereo, left/right/mid/side).
3. The analyzer popup menu hosts the settings of the spectrum analyzer.
4. The left scale shows the filter/GR magnitude. Change zoom level with mouse wheel or mouse click and drag left/right
5. The right scale shows the spectrum magnitude. Change zoom level with mouse wheel or mouse click and drag left/right. Change offset with mouse click and drag up/down
6. The pre-filter dash board shows current filter settings. The filter number is set by the "Nr." control.
7. The top left area shows the current maximum gain reduction and signal slope.
8. The bottom scale shows the frequency scale. Change zoom level with mouse wheel or mouse click and drag up/down. Change offset with mouse click and drag left/right.

5 Minimum System Requirements

- Windows 7, OpenGL 2 GFX card
- Mac OS X 10.11, Metal GFX card
- SSE2 CPU
- Win: 32/64 Bit VST, 32/64 Bit VST3, 32/64 Bit AAX
- OS X: 64 Bit VST, 64 Bit VST3, 64 Bit AU, 64 Bit AAX
- Tested with: Cockos Reaper, Steinberg Cubase/Nuendo/Wavelab 6+, FL Studio 12+, PT2018+, Reason 9.5+, Studio One, Ableton Live
- For latest information please visit www.tbproaudio.de

6 Plugin Controls

Mouse usage:

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

6.1 Main plugin area

6.1.1 Plugin menu

About: shows plugin version and further information
Open manual: opens plugin manual
Change log: opens web site with latest version info
Check for updates: new version check
Activate plugin: shows plugin activation screen
Open data folder: opens local data folder
Tooltips: enables display parameter tooltips
Reset GUI size: resets GUI size to default
Reset GUI scale: resets GUI scale to default
Set Min GUI scale: sets GUI scale to minimum size
Fixed GUI scale: sets GUI to defined scales
Set GUI scale globally: propagate current GUI scale to all DSEQ instances

6.1.1.1 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".

6.1.1.2 Set smart silence processing globally

Propagates smart silence processing setting to all DSEQ instances (including SSP custom threshold). This functions only if all DSEQ instances are in the same process space. Please consult the DAW manual for more information.

6.1.2 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
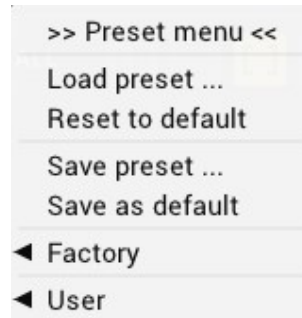
preset with custom threshold. Now you can toggle A/B on the fly. Please make sure that both settings have the same quality mode to avoid audio drops.

6.1.3 Parameter undo/redo

Undos and redos parameter changes.

6.1.4 Preset menu

>> Preset menu <<

Load preset ...
Reset to default

Save preset ...
Save as default

◄ Factory
◄ User

Preset menu loads and saves user presets. Presets stored under %localappdata%/DSEQ3 (Windows) or /Users/xxx/Library/Application Support/ DSEQ3 (Mac OSX) are imported as user presets. Previous and next button step trough 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 together with the user preset.

6.1.5 Previous and next preset

Set previous and next preset.

6.1.6 TBProAudio logo

Click to prevent audio mute every 90 seconds (demo mode only)

6.1.7 Key symbol

If gray colored, click to select license key and activate the plugin. If golden colored plugin is activated.

6.1.8 Check plugin update

Click to check for updates.

6.2 Top plugin area

6.2.1 Over-sampling (Online/offline render)

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.
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.

6.2.2 Quality mode (Online/offline render)

Quality mode determines the size of the FFT and therefore the number of frequencies to be analysed. This control also impacts the plugin latency.
Select quality mode of offline rendering (aka bounce, mix-down or audio export).
If item "1:1" is selected no change from online to offline render is made.

6.2.3 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)

6.2.4 Analyzer mode

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.

6.2.5 Linear/natural phase switch

Toggle linear (LP) and natural (NP) phase dynamic filter. Please to note that the natural phase filter introduces more phasing compared to the linear phase filter. The natural phase filter creates less pre-ringing.

6.2.6 Bypass

Bypass enables click-free plugin bypassing.

6.2.7 Pre-filer monitoring

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

6.2.8 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.

6.2.9 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.

6.2.10 Side chain

Side chain toggles usage of side chain (VST2 CH 3+4) off/on. The side chain signal triggers the dynamic filters. Click on the small top arrow open the gain control window.

6.2.11 Spectrum analyser freeze

Stop updating of the GR/spectrum analyzer.

6.2.12 ABLM

See chapter ABLM.

### 6.2.13 Some notes 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".

6.3 Left plugin area
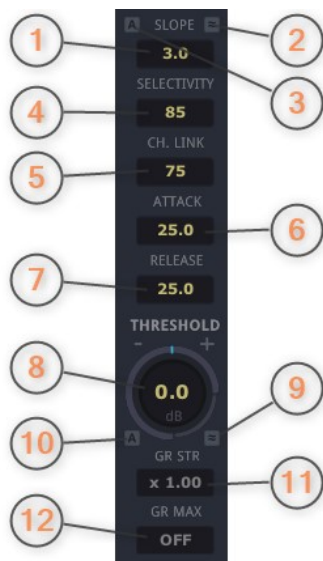
6.3.1 Slope

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.

6.3.2 Calculate slope parameter

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

6.3.3 Adaptive slope

Enable to let DSEQ continuously calculate the optimal slope. Use the slope parameter to increase or decrease the calculated slope additively.

6.3.4 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.

6.3.5 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.

6.3.6 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.

6.3.7 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.

6.3.8 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.

6.3.9 Calculate threshold parameter

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

6.3.10   Adaptive threshold

Enable to let DSEQ continuously calculate the optimal threshold. Use the threshold parameter to increase or decrease the calculated threshold additively.

6.3.11   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.
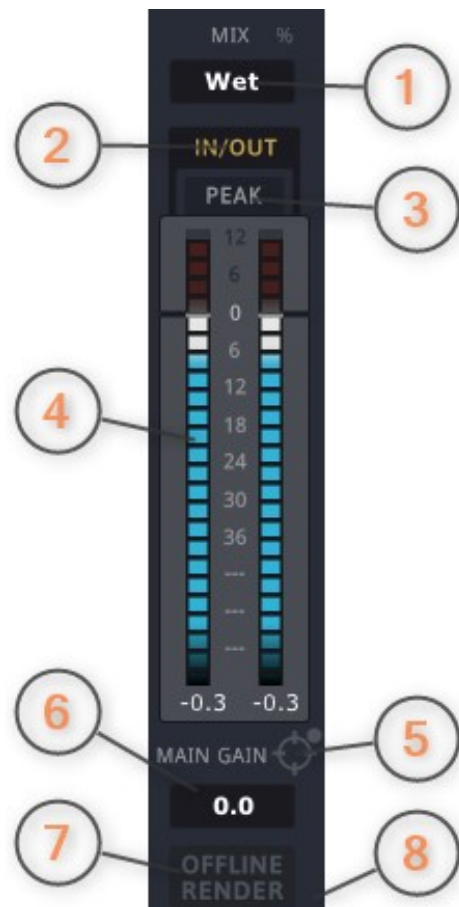GR strength works similar to the ratio parameter of a compressor.
The formula is: Strength = 1 – 1 / Ratio or Ratio = 1 / (1 - Strength).

6.3.12 Max GR

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

6.4 Right plugin area

6.4.1 Mix

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

6.4.2 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.

6.4.3 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).

6.4.4 Main Meter

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

6.4.5 AB-LM Lite gain freeze

Freezes AB-LM Lite gain and switches AB-LM Lite to off. Small white LED indicates if freeze gain is active. Please refer to chapter 7 for more information.

6.4.6 Main Gain

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

6.4.7 Offline render

Lights if DAW offline render mode is active

6.4.8 GUI resizing and scaling

Click and drag to resize the GUI. Additional ctrl key scales the GUI

6.5 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.

6.5.1 Hide pre-filter board

Click to show/hide the pre-filter board.

6.5.2 Nr.

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

6.5.3 Show/hide pre-filter response curve.

Shows/hides pre-filter response curve.

6.5.4 Active

Active enables/disables the current pre-filter.

6.5.5 Solo

Solo makes all other pre-filters inactive.

6.5.6 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

6.5.7 Channel

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

6.5.8 Slope

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

6.5.9 Frequency

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

6.5.10   Sensitivity

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

6.5.11   Q-Factor

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

6.5.12   Copy parameter

Copy the current filter parameter set to target, given by number below.

6.5.13   Target pre-filter number

Target pre-filter number, used by parameter set copy.

6.5.14   Filter button (spectrum view)

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-actives 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
Reset: resets filter to default values

Mouse usage:

Mouse drag: changes filter **frequency** and **sensitivity**
Shift key + mouse drag: slows down mouse movement
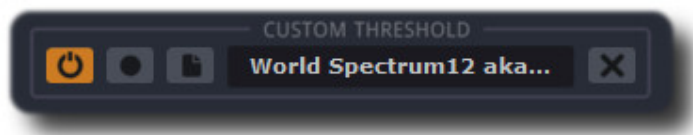Ctrl key + mouse click: resets filter to default values

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

6.6 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 recoding all **DSEQ** processing is bypassed. Record clear all current data.
File menu:
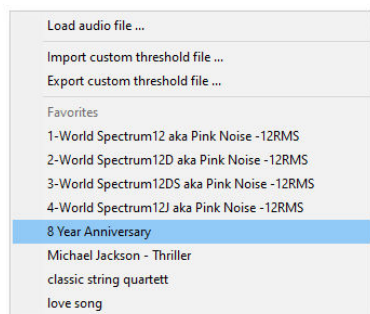
The menu lets you load and analyse audio files (Mono/stereo **WAV** and **AIFF)** and use as custom threshold. Custom threshold data can be imported and exported (ext. cth). The export is done to the **DSEQ** data folder. The menu lists the first 20 data files from there as favourites for quick import.
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:
- 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".

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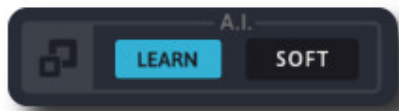
DSEQ makes it even easier to tame digital harshness thanks to its smart AI function. Set up DSEQ as normal, selected desired AI learn mode, press play to start the DAW and then the learn button. DSEQ calculates the optimal slope and threshold parameter within seconds.

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.

6.8 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 AI 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
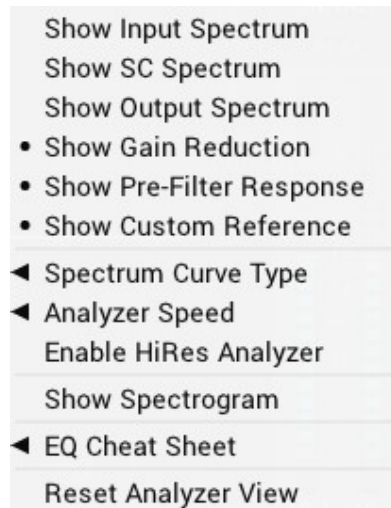- AB-LM
- plugin bypass
- delta monitoring

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.

6.9 Analyzer popup menu

Show Input Spectrum
Show SC Spectrum
Show Output Spectrum
• Show Gain Reduction
• Show Pre-Filter Response
• Show Custom Reference
◀ Spectrum Curve Type
◀ Analyzer Speed
Enable HiRes Analyzer
Show Spectrogram
◀ EQ Cheat Sheet
Reset Analyzer View

Show Input Spectrum: toggles display of input spectrum
Show SC Spectrum: toggles display of side chain spectrum
Show Output Spectrum: toggles display of output spectrum
Show Gain Reduction: toggles display of GR view
Show Pre-Filter Response: toggles display of filter response
Show Custom Threshold: toggles display of custom threshold curve
Spectrum Curve Type: customizes display of spectrum/GR curve
Analyzer speed: customizes analyzer speed
Enable HiRes Analyzer: toggles high resolution analyzer
EQ Cheat Sheet: toggles display of various overlaid cheat sheets.
Reset analyzer view: resets scale and offset of spectrum, GR and EQ response view

6.9.1 Spectrogram

Enable the spectrogram view.
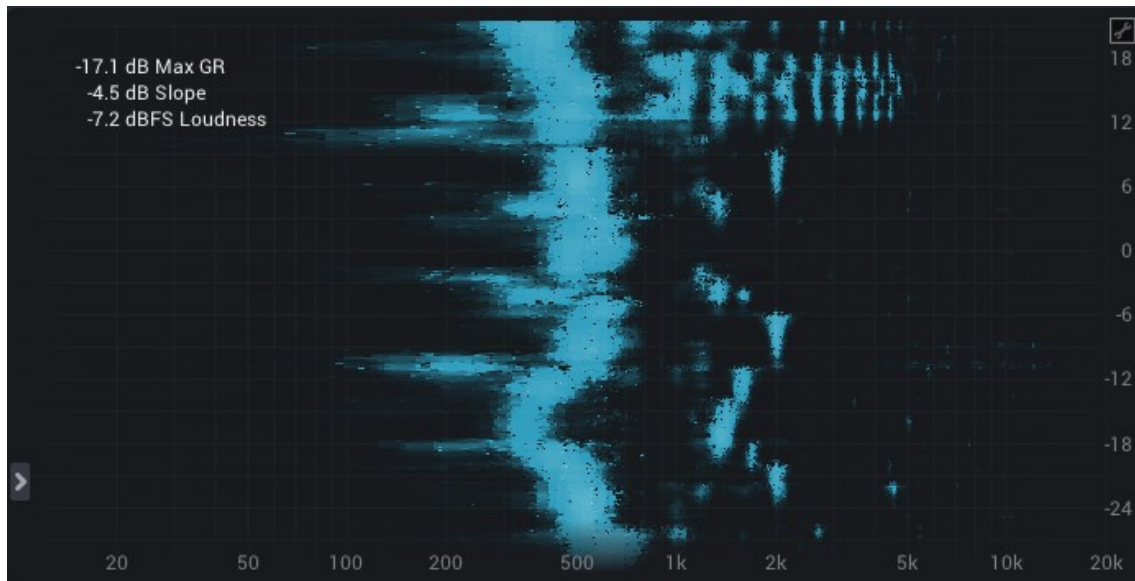
Brighter areas indicate higher gain reduction, darker areas less or no gain reduction.

7 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 minimize the loudness either quickly (mode fast) or gently (mode slow).

The "target" symbol top right of the fader sections freezes the last AB-LM Lite gain and switches AB-LM Lite off. This function prints the last AB-LM Lite gain settings to the gain structure of DSEQ.

The small white LED indicates if AB-LM Lite freeze gain is active. Click on the "target" symbol sets freeze gain to 0.0 dB.

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 www.tbproaudio.de.

8 Smart silence processing (SSP)

Smart silence processing reduces CPU if silence is processed.

1) Click to set SSP threshold to either -186/-138/-90dBFS. This corresponds to SNR of 32/24/16Bit. Select custom threshold for user defined SSP threshold. The SSP icon indicates the processing state.
2) Click to set SSP threshold based on current loudness.

9 Demo mode versus Registered Mode

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

10 Plugin activation

The plugins needs to be registered/activated to remove demo restrictions. Please go to www.tb-proaudio.de to purchase the activation key. After purchase you will receive an email from TBProAudio with either the (zipped) activation key file or the activation key in text form. Go to the plugin menu->Activate plugin. Please follow the steps described here: https://www.tbproaudio.de/support/productactivation. After successful activation the key symbol shown in the GUI appears in golden colour.

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 :-)