CS-5501 V2

CHANNEL STRIP

MANUAL

TBProAudio

2025

1   Introduction

Welcome to CS-5501, a channel strip plugin with gate/expander, EQs, compressor, saturation, limiter, de-esser, extensive side-chain and stereo placement options, flexible module routing, over sampling and AB_LM-Lite for perceptual loudness matching.

2 Features

CS-5501 offers following features:

- 2x 7 band EQ
- 2x noise gate/expander
- 2x compressor, VCA/FET/OPTP design
- 2x de-esser
- 2x limiter
- 2x saturation
- 2x thermal noise for analogue sound emulation
- 2x clip
- 2x pre-amp
- 2x transient shaper
- full dual mono processing
- serial and parallel processing
- HQ over sampling
- linkable input/output fader including phase inversion
- module parameter link
- copy/paste module settings (across all instances)
- extended metering, input/output/gain reduction, Peak, RMS, EBU and VU
- DAW synced oscilloscope
- spectrum analyzer
- peak level display
- perceptual loudness matching (powered by ABLM Lite) and gain "freezing"
- flexible ABLM Lite routing options
- module routing
- extensive side-chain support for many effect modules
- many comfort functions like soloing and signal monitoring
- mono mode on stereo channel
- parameter randomizer
- preset management
- module preset management
- undo/redo
- selectable colour theme
- large and easy to use GUI
- free GUI scaling
- 64-bit internal processing
- very efficient CPU usage design
- smart silence processing

3 Overview

CS-5501 is a multi functional channel strip providing all necessary tools to process all kind of audio signals like vocals and instruments.

CS-5501 comes with double EQ/compressor/limiter/gate/expander/de-esser/saturation/clip/ thermal noise emulation/pre-amp and transient shaper. All modules have stereo placement options and are freely routable. Some of the effect modules have external side chain options. Each module can work either in serial or parallel mode.

CS-5501 provides also IN/OUT/GR VU meter, IN/OUT fader, extensive monitoring functions and over sampling.

CS-5501 includes an oscilloscope, a spectrum analyzer and a peak level display.

On top CS-5501 adds the function called ABLM Lite providing perceptual loudness matching.

4 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

5 Plugin Controls

<u>Mouse usage:</u>

Click and drag horizontally changes parameter value
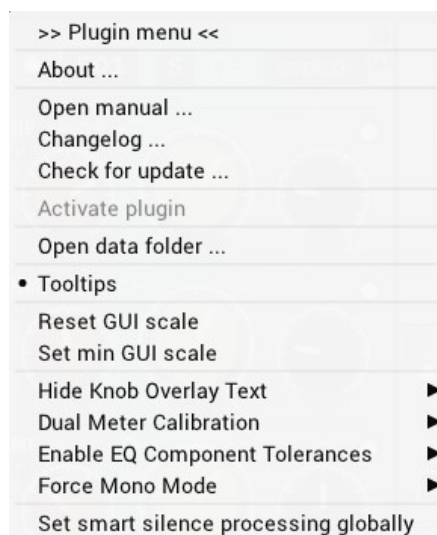Shift click and drag changes parameter value slower
Ctrl click resets to default value, double click on text opens text entry

Please note that any parameter change of an effect parameter also enables the unit itself (auto activation)

5.1 Top area

5.1.1   Plugin menu

About: Info about the plugin.
Open manual: open manual.
Changelog: open online change log information.
Check for update: contact TBProAudio server for plugin updates.
Activate plugin: open activation screen.
Open data folder: open plugin data folder
Tooltips: toggle GUI control hints
Reset GUI scale: resets GUI scale to default (100%).
Set min GUI scale: set GUI to minimum GUI scale (50%)
Hide Kob Overlay: prevent the display of the parameter value during adjustment, if you want to keep the GUI "old-school".
Dual Meter Calibration: enable individual calibrations of left and right VU meter.
Enable EQ Component Tolerances: enable emulation of components of EQ section (currently experimental).
Force Mode Mode: CS-5501 detects current channel configuration automatically. Some DAWs provide only stereo tracks even for mono audio material. In case you process mono audio on a stereo track enable to save resources.
Set smart silence processing globally: Propagates smart silence processing setting to all CS-5501 instances (including SSP custom threshold). This functions only if all CS-5501 instances are in the same process space. Please consult the DAW manual for more information.

5.1.2   Background themes
Changes background plane.

5.1.3   Undo/redo
Undo and redo of parameter changes.

5.2 Display section

The display section shows the VU meters, oscilloscope, spectrum analyzer or peak level display.

5.2.1   VU Meter

CS5501 provides a dual meter display which can be configured in multiple ways.

1) Current hold value
2) The black needle represents the current value
3) VU meter mode: PEAK (dB), RMS (dB), EBU ML (LU), EBU SL (LU) and VU (dBVU). RMS uses 600ms ballistics, VU 300ms. Click to select the desired meter mode.
4) The red needle represents the current hold value.
5) Max value, click to reset
6) Clip LED: lights for a few seconds if black needle goes over the zero value.
7) The meter configuration enables the display of IN, GR (gain reduction), OUT or IN/OUT values
8) The meter calibration controls the reference level of Left/Right IN/OUT meter, in dBFS.
9) Set the monitor mode either to in-place or centred.
10) The monitor control adjusts the main monitor mode: either stereo (normal), left, right, mid or side signal.
11) Smart silence processing, please refer to chapter 7)

5.2.2   Oscilloscope

1) The oscilloscope displays the signal amplitude over the time. The buffer of the oscilloscope stores one second maximum.

2) Shows configuration page

3) Time scale. Mouse wheel zooms in/out. Mouse drag moves zoomed time range. Mouse click, hold and move up zooms and moves time range.

4) Magnitude scale. Mouse wheel zooms in/out. Mouse click, hold and move right zooms magnitude.

5) Signal mode selector. Shows either summed stereo, left, right, mid or side channel

6) Signal display selector. Shows input (after input fader), side-chain and output (before output fader) signal.

7) Pauses update of the display

8) Time setting: either buffer length in ms if DAW sync is off or tempo synced if DAW sync is on:

LOC: DAW time selection/locator information is used

1/x note: loop buffer is set to specific note length

x bar: loop buffer is set to DAW bar length (tempo and time signature)

9) DAW sync off/on: if off buffer length is adjusted with time control

5.2.3   Spectrum analyzer

1) The spectrum analyzer displays the frequency spectrum of the audio signals.
2) Shows configuration page
3) Frequency scale. Mouse wheel zooms in/out. Mouse drag moves zoomed frequency range. Mouse click, hold and move up zooms and moves frequency range.
4) Magnitude scale. Mouse wheel zooms in/out. Mouse click, hold and move right zooms magnitude.
5) Signal mode selector. Shows either summed stereo, left, right, mid or side channel
6) Signal display selector. Shows input (after input fader), side-chain and output (before output fader) signal.
7) Changes spectrum tilt
8) Changes analyzer speed
9) Pauses update of the display
10) Shows maximum spectrum
11) Shows Spectrum as bars
12) Shows EQ response curve

5.2.4 Peal level display
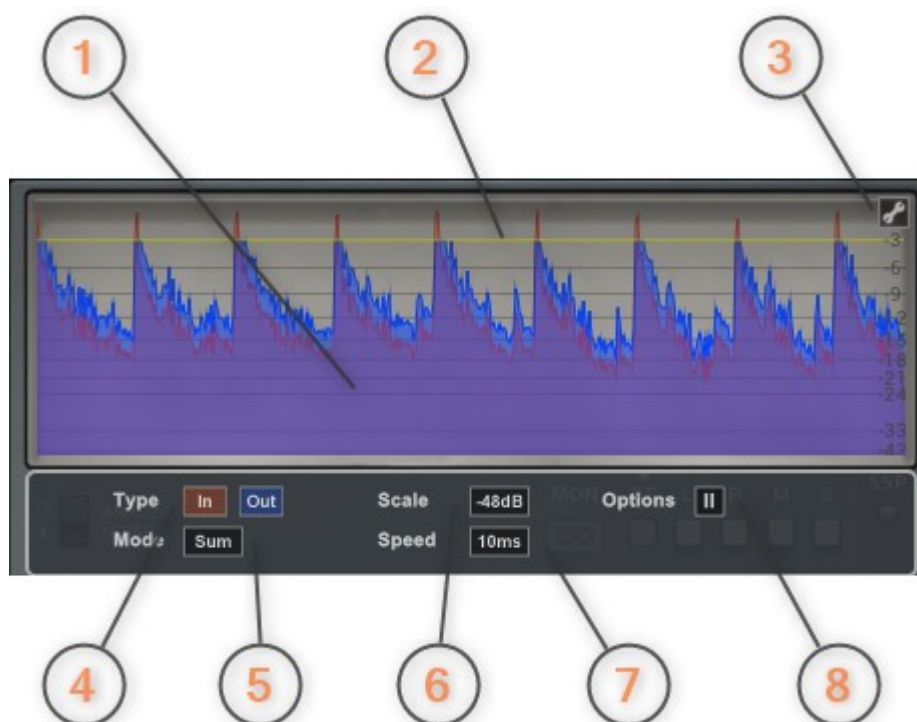
1) The peak level display shows the envelope of the audio signals over a period of time.
2) Yellow line shows the clip level
3) Configuration page
4) Input/output signal
5) Sum/left/right/mid or side signal
6) Display scale -12/-18/-24/-36/-48 dBFS
7) Update speed 10/20/50/100ms or infinite
8) Pause display update

5.3 Over sampling & Bypass

1) AB-LM Lite: control perceptual A/B loudness matching. Please refer to chapter 6.
2) Over-sampling: Incoming signal could be over-sampled up to 2 or 4 times. Over sampling reduces magnitude/phase distortions near Nyquist frequency and let the filter sound much more "analogue".
3) By-Pass: Global plugin by-pass.
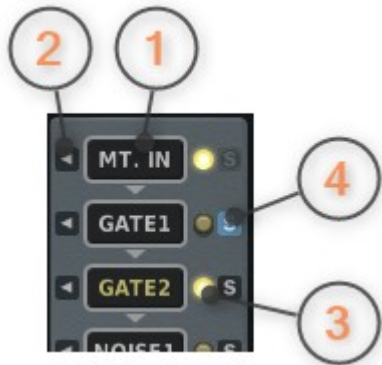
5.4 Fader section
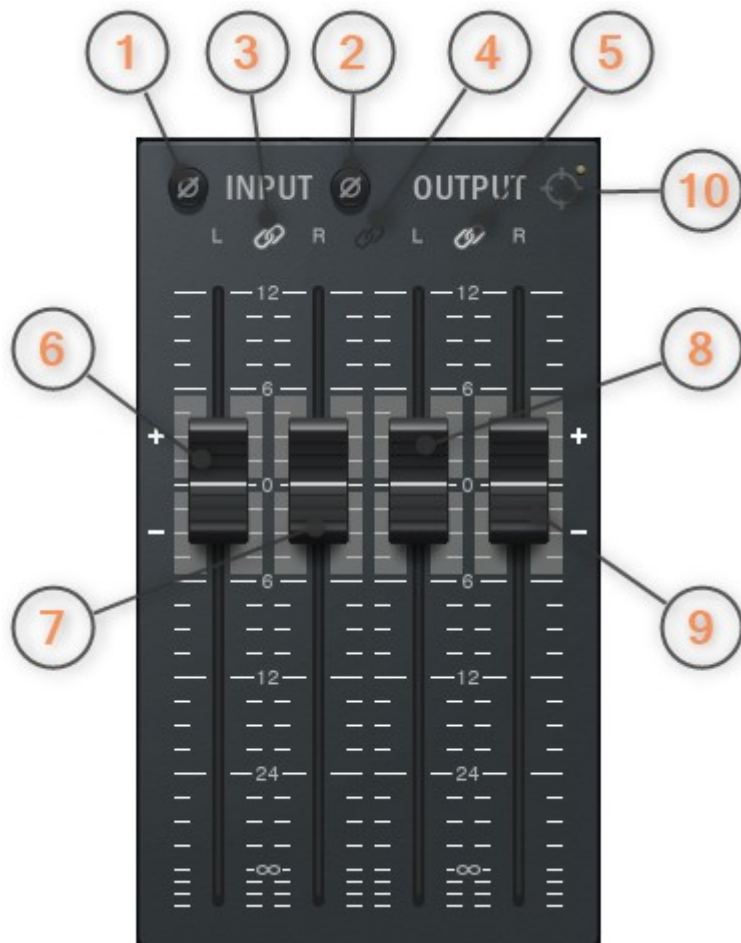
1/2) Inverts the phase of incoming left and right signal

3/4/5): Link input LR, output LR and input/output fader

6/7/8/9): Fader sets the input/output gain

10) Freezes ABLM Lite gain and switches ABLM Lite to off. Small yellow LED indicates if freeze gain is active. Please refer to chapter 6.

5.5 Routing section

This section controls the order of the individual effect modules. Click the slot where you want to change the effect and select the new effect from the drop-down menu.
Shift click shows the respective module. Click on the small left icon below the drop-down menu to show the corresponding module.
The yellow LED indicates the power status of the module. Click to activate/deactivated the effect, shift click to show the module.
1) Module menu
2) Show module
3) Activate module
4) Solo module

5.6 EQ section

1) Select either EQ1 or EQ2
2) Enable EQ
3) EQ menu: Copy/paste module settings (even across CS-5501 instances), load/save module settings and quick load of presets.
4) Solo the module. Shift mouse click resets solo states of all modules.
5) The stereo placement of the effect can be changed to stereo/left/right/mid/side.

Each module can work either in serial or parallel mode. In parallel mode module 1 and 2 are processed in parallel and then mixed depending on the mix knob. The routing position of both modules is determined by position of module 1.

1) Serial or parallel mode
2) Show/hide mix knob for parallel operation
3) Mix knob

CS-5501 provides 2 independent EQ modules with 7 filters each. The filters are based on analogue design and minimize amplitude cramping near Nyquist frequency. This lets the filter sound much softer than normal digital filters.
1) Activate filter
2) Filter band monitor
3) Filter mode: peak/shelf filter
4) Filter center frequency
5) Filter gain
6) Filter Q factor

1/4) Activate filter
2/5) Filter slope
3/6) Filter center frequency

5.7 Effect section
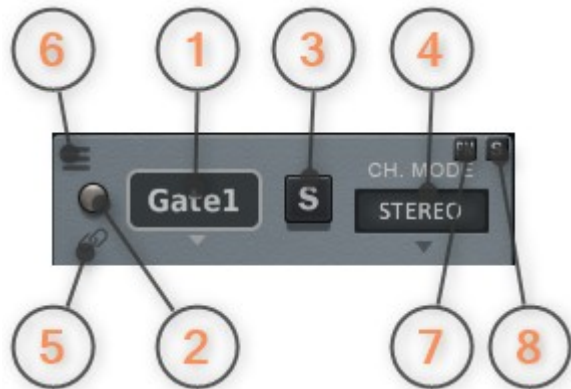
1) Select the effect module from drop-down menu
2) Activate effect
3) Solo effect
4) Stereo placement of the effect can be changed to stereo/left/right/mid/side
5) Links parameter of effect 1 and 2
6) Effect menu: Copy/paste module settings (even across CS-5501 instances), load/save module settings and quick load of presets.
7/8) Serial or parallel mode and mix knob (parallel mode)

1) Selects either internal or external side chain. VST2 uses channel 3+4 as ext. side chain. For VST3/AU/AAX/RTAS please consult your DAW manual.
2/5) Enables low/high cut filter
3/6) Center frequency
4/7) Filter slope
8) Link center frequency of low/high cut filter
9) Monitor side-chain signal

5.7.1   Compressor

CS-5501 provides 2 independent compressor modules. The compressor modifies the incoming signal depending on threshold and ratio.

The compressor design is based on TBProAudio's Impress and offers 3 different classical types:

VCA stands for "Voltage Controlled Amplifier" and its compression behavior is based on PEAK, with fast attack and release.

FET stands for "Field Effect Transistor". The 1176 is his most prominent representative and offers very short attack and release times.

OPTO stands for "Optical compressor" and its compression behavior is based on RMS, with slow attack and release.

FET 1 offers a compressor ratio much closer to the original CS3301 compressor.

1) Threshold
2) Ratio
3) Attack: lower values make the compressor more responsive.
4) Release: higher values make the compressor longer working.
5) Auto-Release: program dependent release time calculation
6) DAW tempo synced release times
7) Mix dry/wet signal (parallel compression)
8) Output gain
9) Compressor types
10) Gain reduction meter

5.7.2 Gate/expander

CS-5501 provides 2 independent gate/expander modules. Gate/Expander opens only if the signal reaches a certain threshold. In Gate mode, the signal is fully suppressed below the threshold and left through above the threshold. In Expander mode, the signal below the threshold is suppressed depending on the signal strength.

1) Gate/Expander mode
2) Threshold
3) Attack: lower values make the gate/expander more responsive.
4) Hold: lower values start release earlier.
5) Release: higher values make the gate/expander longer open.
6) Range: in gate mode min value -72dB is treated as range off.
7) Attenuation meter

5.7.3   De-esser

CS-5501 provides 2 independent de-esser modules. Deeser is a kind of dynamic filter. It compresses/expands the signal in a defined frequency range depending on the input/side-chain signal. The design is based on TBProAudio's dEQ6.

1) Threshold
2) Ratio
3) Attack
4) Release
5) Filter center frequency
6) Filter gain
7) Filter Q ratio
8) Compression range
9) Output gain
10) In wideband (WB) mode the de-esser reacts on events in the full frequency range, in narrowband mode (NB) only on events around de-esser center frequency.
11) Filter type: low-shelf/bell/high-shelf filter type.
12) Monitor mode: signal removed by the de-esser or signal focused by the de-esser.
13) Current gain reduction (blue) or gain boost (green).

5.7.4   Limiter

CS-5501 provides 2 independent limiter modules.
The limiter keeps the signal below a certain "ceiling" level. The design is based on
TBProAudio's LAxLimiter.
Please to note that the limiter introduces 128 smp latency, this is compensated by the DAW.

1) Gain
2) Ceiling: maximum output level
3) Release
4) Auto-release: program dependent release time calculation
5) Gain reduction meter

5.7.5   Saturation
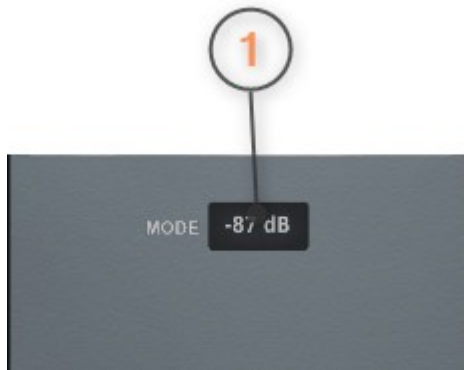
CS-5501 provides 2 independent saturation modules.
Saturation adds odd or even harmonics to the signal. The saturation effect emulates several tubes with odd and even harmonics.

1) Tube character: select either clean/warm/crisp/classic saturation type. 'classic' is from former CS-3301.
2) Stages: select number of tubes in a row.
3) Fluctuation: controls the amount of saturation fluctuation.
4) Odd: controls the amount of odd harmonics added to the signal.
5) Even: controls the amount of even harmonics added to the signal.
6) Mix: controls the amount of saturation added to the main signal

5.7.6 Thermal noise emulation

CS-5501 provides 2 independent modules to emulate thermal noise of analogue channel strips.

1) Mode: select from several noise profiles/floors: -87/-114dB and dither 16/24 bit.
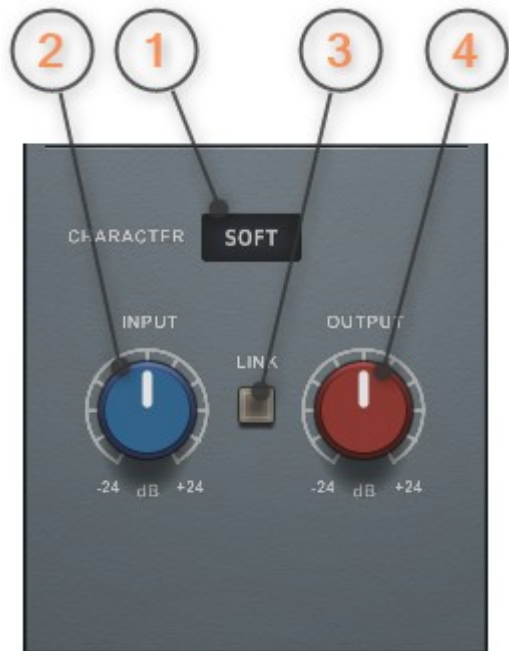
5.7.7 Clip

CS-5501 provides 2 independent modules to clip the signal.

1/2) Ceiling sets the clip level of the module. The LED indicates activity of the clip processor.
3) Mode selects various clip modes: tanh (default), hard, sinus and quintic clipping.

5.7.8   Pre-Amp

CS-5501 provides 2 independent pre-amp modules.

1) Character: 2 different tube mode, soft/hard
2) Input gain
3) Link input/output gain
4) Output gain

5.7.9 Transient Shaper

CS-5501 provides 2 independent transient shaper modules.

1) Reduces or enhances transients of the signal.
2) Reduces or enhances sustain of the signal.
3) Attack time of the transient processing.
4) Release time of sustain processing.
5) Adjust the processing according source signal.
6) Output gain.
7) Enables additional smoothing of the transient shaping.
8) Enables internal clipper.

6 ABLM Lite

ABLM 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 ABLM 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 ABLM Lite limits the gain adjustment to +/- 12dB to avoid gain overshoots.

ABLM Lite lets you automatically minimize the loudness either quickly (mode fast) or gently (mode slow).

The 'target' icon top right of the fader sections freezes the last ABLM Lite gain and switches ABLM Lite off. This function prints the last ABLM Lite gain settings to the gain structure of CS-5501.

The small yellow LED indicates if ABLM Lite freeze gain is active. Click on the 'target' icon sets freeze gain to 0.0 dB.

ABLM Lite is an automatic, simplified algorithm to help your workflow, based on our more advanced dedicated ABLM plugin, which allows perceptual loudness matching of third party plugins and even complete plugin chains. ABLM also offers more advanced options, including advanced measurement modes and snapshots.

For the more advanced features of ABLM, please see www.tbproaudio.de.

7 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.

8 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.

9 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.

10 Conclusion

Special thanks go to Crimson Merry who designed the GUI and Taviq Mastering who gave constant valuable input to improve CS-5501.

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

Your team from TBProAudio :-)