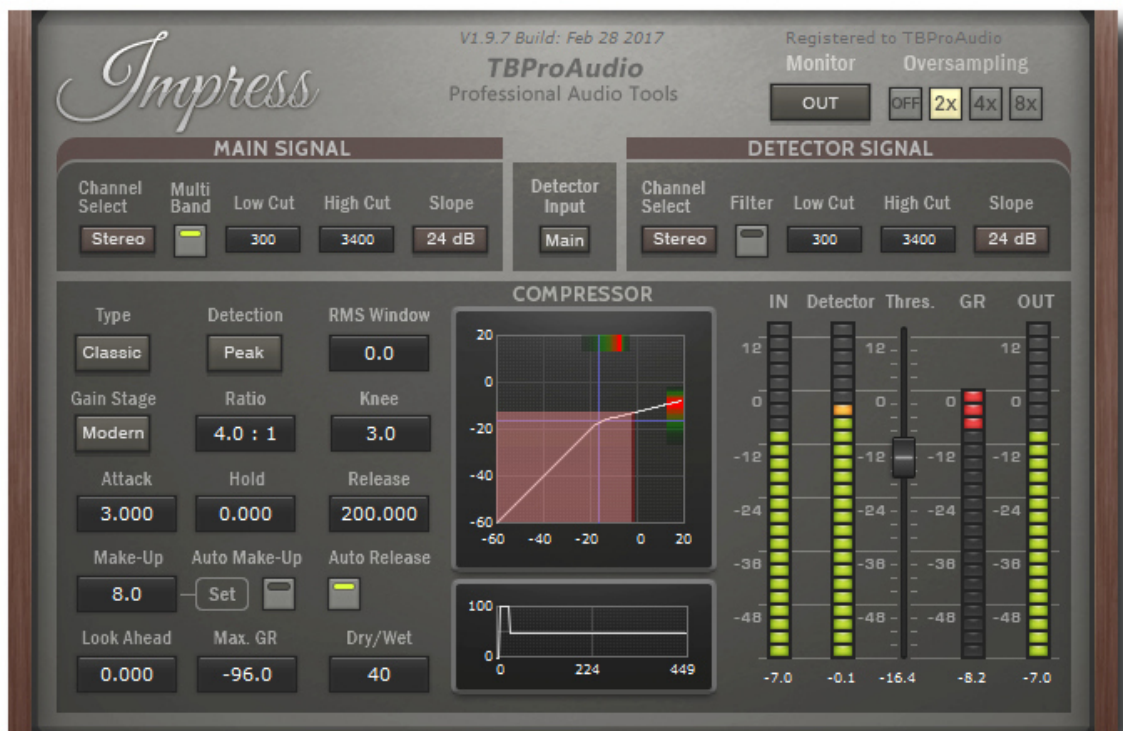


Impress

Manual



by TBProAudio 2017

1 Introduction

Welcome to Impress, a wideband stereo compressor including side-chain and various pre-filter options. The goal of the design was to cover a broad range of compression applications in today's mixing and mastering situations with strong focus on *low aliasing*.

So, Impress could be used to shape subtle a snare sound, glue different tracks on instrument bus and even do complex tasks like deessing or voice over's.

All Impress parameters are easy to read and control. Additionally Impress gives visual feedback to make musical decisions right.

2 Features

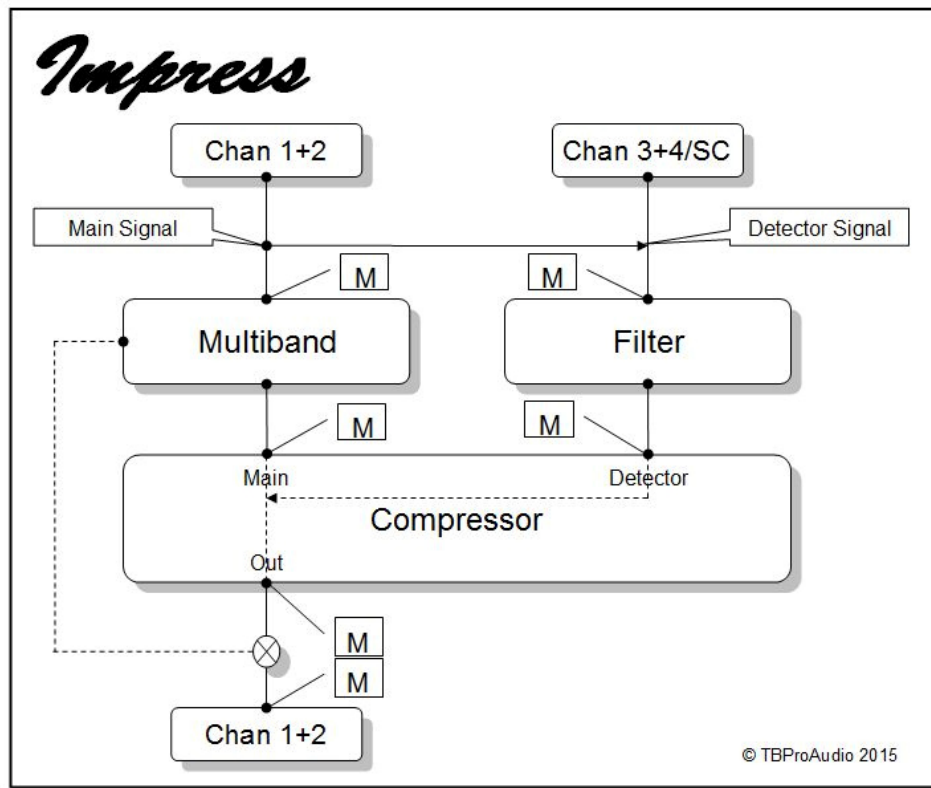
Impress offers following features:

- zero latency click-free 64-bit internal processing
- low aliasing compressor design
- "real" oversampling, up to 8x
- stereo/left/right/mid/side processing
- external side-chain
- main signal multiband filter, up to 192 dB/Oct (*Linkwitz-Riley* design)
- detector signal pre filter, up to 192 dB/Oct
- 2 compressor modes: classic and clean (low aliasing)
- 3 gain stage modes: modern, vintage and vintage clean
- 2 level detection modes: peak, RMS and RMS with BS.1770 pre-filter
- short/long attack/ release times
- Look Ahead
- low aliasing soft knee
- Gain reduction limit
- dry/wet
- peak/RMS measurement
- flexible signal monitoring

- Easy to use GUI
- large and accurate live meters
- gain transfer graph
- time response graph

3 Design

Diagram below shows the basic structure of Impress:



Compressor: The compressor part has two inputs (Main and Detector) and one output (Out): the main signal is the audio stream to be processed by the compressor, the detector signal is the signal for the level detection. In most applications both signals are the same, for side-chain effects separate signals are used.

Beside the classic "to go" design Impress offers a clean, *low aliasing* design with two envelope follower. The gain stage could run in "vintage" mode with a thermal noise emulation. The "knee" function is optimized for *low aliasing* operation.

Multiband Filter: The main signal could run through a multiband filter stage. The signal is split into three frequency bands, the mid band feeds the compressor, the output of the compressor is mixed with the low and high band again. The multiband filter is based on the *Linkwitz-Riley* design (up to 192 dB/Oct steepness), which prevents any phase shifts.

Filter: The detector signal could also run through a filter stage .

Detector Input: The detector signal path could use either channel 1+2 or channel 3+4/side-chain.

Monitoring: Signals through Impress can be individually monitored in order to control the effect of each signal stage (M).

4 Minimum System Requirements

- Windows XP SP3 or newer
- Mac OS X 10.5 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, FL Studio 12.x, PT10/12
- All sample rates

5 How to use

Impress is designed to reset all internal measurements if significant parameter changes are made. This keeps all displays updated.

Impress uses various graphical elements to control the parameters of the plugin:

Value button:



- Mouse click and drag up/down changes value of the parameter.
- Mouse double click opens value input box.
- Mouse shift click resets value.
- Mouse-wheel increments/decrements value

Button:



- Mouse click enables/disables function, small red LED means "on/enabled"

Popup menu button:



- Mouse click opens popup menu.

Slider:



Mouse click on slider handle and drag up/down changes value. Hold ctrl key for small adjustments.

Mouse click on slider outside handle sets handle and value directly.

Mouse shift click resets slider.

Mouse double click on value below the slider opens value input box

Mouse-wheel increments/decrements value

6 Plugin Controls

6.1 Main

6.1.1 Monitor

This control selects plugin audio output (please refer to 3, monitoring). Following signals could be routed to output:

- original main signal
- main signal, filtered by multiband filter
- output of compressor, not mixed with low/high band of multiband filter
- original detector signal
- detector signal, filtered
- final output signal of Impress plugin

6.1.2 Oversampling

Main and detector signal could be oversampled up to 8 times. As Impress uses highest quality up/down-sampling algorithms for *low aliasing* operation significantly more CPU resources are used.



6.2 Main Signal Stage

Before the main signal goes into the compressor, the full stereo signal or parts of it (e.g. left or mid signal) could be selected. after the compressor stage the unprocessed part (e.g. right or side signal) is added.

Then the signal could be sent through 3 multi band filter, only the mid band is then treated by the compressor. After the compressor stage the low and high band are mixed to the compressed signal. As the design of the multiband filter is based on *Linkwitz-Riley* the absence of phase shifts is guaranteed. Placing several instances of Impress in series gives a flexible multiband compressor.



ChanSel (Channel Selector): stereo (default), left, right, mid+side, mid, side

MB (Multiband): off, on (red light)

LC (LowCut): sets the lower limit of mid band, 1 - 22049Hz

HC (HighCut): sets the higher limit of mid band, 1 - 22049Hz

Slope: sets filter steepness, 24dB/Oct - 192dB/Oct, in 24dB/Oct steps

6.3 Detector Selector

This control selects which plugin input is routed to the detector signal stage.

Channel 1+2 of the plugin are always routed to the main signal stage.

Channel 1+2 could also be routed to detector signal stage (normal operation). Or channel 3+4/Side-chain could be used as detector signal. In this mode the level detection of the compressor is controlled by a different signal than the main signal (side-chain effects)



Det Input (Detector input): Main (channel 1+2), SC (channel 3+4/side-chain)

6.4 Detector Signal Stage

Before the signal from channel 1+2 or 3+4/side-chain is routed to the detector input of the compressor parts of the signal can be selected. Additionally the signal can be filtered in order to let the level detector focus on a specific part for the spectrum.



ChanSel (Channel Selector): stereo (default), left, right, mid+side, mid, side

Flt (Filter): off, on (red light)

LC (LowCut): sets the lower limit of the filter, 1 - 22049Hz

HC (HighCut): sets the higher limit of the filter, 1 - 22049Hz

Slope: sets filter steepness, 24dB/Oct - 192dB/Oct, in 24dB/Oct steps

6.5 Compressor

The compressor module is a soft knee variable-ratio processor with maximum compression becomes a limiter.

6.5.1 Compressor Type

Classic: Classic digital compressor design with well known gain follower

Clean: Low aliasing design, gain follower and double level envelope follower

6.5.2 Level Detector

Peak: Classic peak level detector

RMS: Signal level is calculated from RMS value

6.5.3 RMS Window

Selectable RMS window size in ms, 0 - 3000ms, in 0,1ms steps

6.5.4 Threshold

Calibrated in dBFS. If detector level is below threshold level, compression starts to act. Click and drag handle to adjust the threshold level, additionally press ctrl key to fine tune. Shift click resets to default value. Double click on the threshold value below opens input window. -60dB - +12dB in 0.1dB steps

6.5.5 GainStage Type

Modern: Classic digital gain stage design

Vintage: Analog gain stage with thermal noise emulation

Vintage C: Analog gain stage, without thermal noise emulation

6.5.6 Ratio

This controls the compression ratio in the region above threshold, 1 - inf (limiter), in 0.1 steps

6.5.7 Knee

Adjustable *low aliasing* soft knee, knee starts -x dB below threshold and ends +x dB above, 0 - 24dB, in 1dB steps

6.5.8 Attack

Calibrated in ms, adjusts attack time of the compressor from 0 - 1000 ms in 0.001ms steps
Envelope is shown in the response graph on the right side

6.5.9 Hold

Calibrated in ms, adjusts hold time of the compressor from 0 - 2000 ms in 0.001ms steps
Parameter is only used in compressor type "classic".

6.5.10 Release

Calibrated in ms, adjusts release time of the compressor from 0 - 5.000 ms in, in 0.001ms steps.

If clean compressor mode is selected release time also effects hold times of double level envelope follower

Current Envelope is shown in the response graph on the right side

6.5.10.1 Auto Release

Release time is calculated program dependant, 125dB/s

6.5.11 Makeup Gain

Adjust the output level of the compressor in dB, -24dB - +24dB, in 0.1dB steps

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6.5.11.1 Auto Makeup Gain (AMG)

If switched on makeup gain is calculated ratio dependant

6.5.11.2 Set Makeup Gain

Sets makeup gain so that output level reaches 0dBFS. Hold shift key to reach -0.1dBFS

6.5.12 Dry/Wet

This control mixes the dry signal with the wet (compressed) signal, 0 (dry) - 100 (wet)

6.5.13 Look Ahead

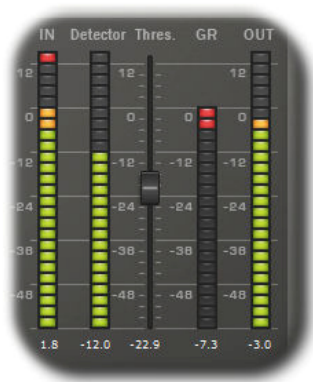
Main/Detector signal is "pre-delayed" in order to let compression start "earlier", 0 ms - 3000 ms, in 0.001ms steps

6.5.14 Max Gr

Limit maximum gain reduction. This is very useful for vocal compression, just to avoid over compression.

-96 - 0dB, in 0.1dB steps

6.6 Meter



Meters for main signal (Inp), detector signal (Det), output (Out) and gain reduction (GR).

LED signal meter range is from -60dBFS to +12dBFS, in 3dB steps.

LED gain reduction meter is from -60dB to 0dB, in 3dB steps.

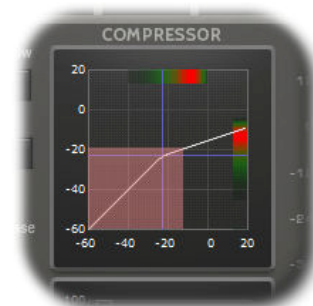
Value below meters show maximum value.

If signal clips (above 0dBFS) top LED lights red.

Click on the meter resets max. values and heat-maps in the scope display.

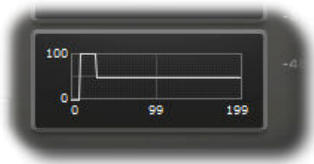
6.7 Scope

6.7.1 Gain Transfer Graph



This display show the current gain transfer curve (white line), current threshold value (blue line), maximum main signal level (red area) and current main signal level (light red area). Additionally heat-maps of main signal (top) and output signal (right) are shown

6.7.2 Response Graph



This display shows current response graph of the compressor (in ms).

7 Demo mode versus Registered mode

In demo mode (without registering) the plugin stops working after 90 secs. By clicking on one of the meters it continues for next 90 secs.

You can register at <http://www.tb-software.com/tbproaudio>.

8 Conclusion

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

Your team from TBProAudio :-)