16019509 / 15015680

Audio Process Design and Implementation – Report

Purpose

The purpose of this plug-in is to serve as a simplified multiband compressor which can be used in both creative mixing and mastering processes. By offering a low and a high-pass band in which unique compression can be applied, each setting can be closely tailored to the user's needs, whilst also adding richness and presence to the sound in full stereo output.

Parameters

Firstly, the user can select between Peak or RMS compression using a drop-down menu at the top of the interface, this simply adjusts the envelope follower which is used as the input signal for the compressor.
The multiband compressor consists of two bands, a low and high; the cut-off frequency for both of which is controlled by a singular dial on the interface (*F_c* in Figure 1). The user can therefore decide how much frequency information is compressed; a range from 20 Hz to 5000 Hz is assigned, which was chosen after tests which found that this range yielded the best sounding results.

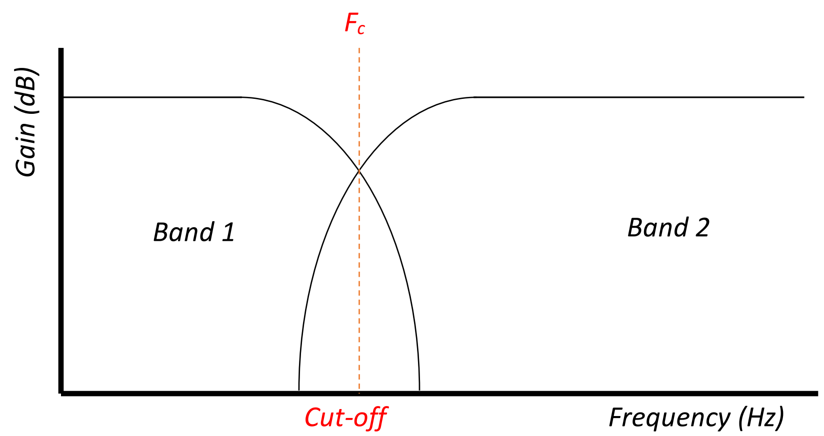

Figure 1.

For each of the frequency bands, individual settings for both threshold and ratio can be set using the appropriate dials. The threshold sets the signal amplitude level at which compression is engaged; the ratio is the amount of attenuation which is then applied when the signal surpasses this threshold. The ranges for these parameters were carefully selected to allow a rich, compressed sound to be achieved quickly and easily even when the parameters are set to extreme levels.
Attack and Release dials are also an essential part of dynamics processing. The attack setting dictates how long it takes for the compression to take full effect once the threshold is exceeded; and the release time is the time taken for the compressed signal to return to no compression.

An example of compressor settings can be seen in Figure 2.

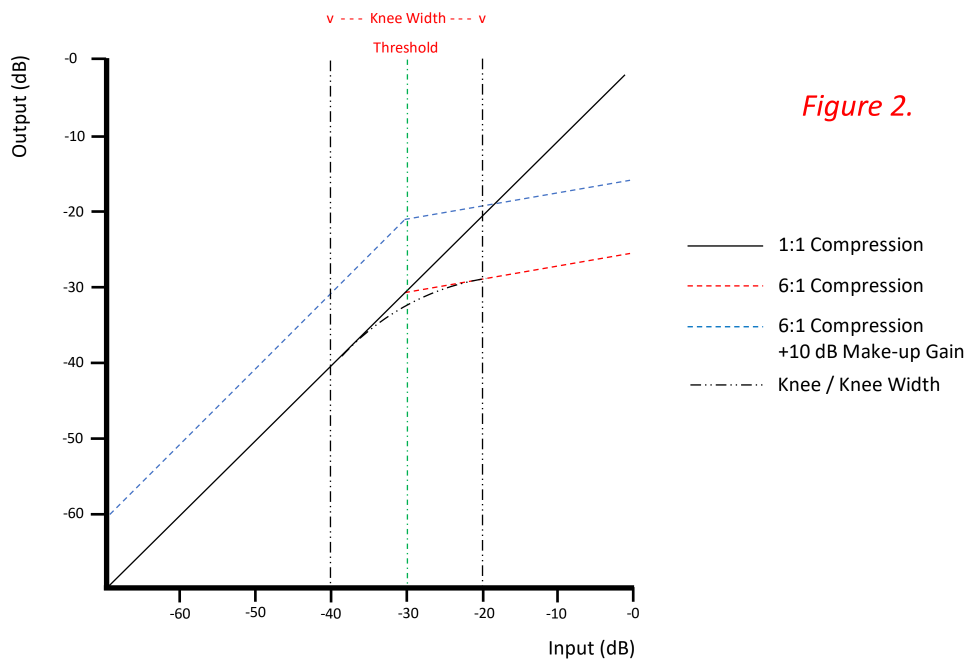

Figure 2.

The *Knee* feature of the compressor refers to how the signal transitions between the non-compressed and compressed states. The knee width can be controlled by the user with a dial at the bottom of the interface.
Individual make-up gain parameters allow the user to increase or decrease the level after compression is applied. With a range of +0.0 dB to +30 dB, this is the stage where the user can amplify the compressors effects and have volume control over specific frequency ranges within a given sound.
Finally, a Mono button allows the user to switch between having Stereo or Mono output.
The interface and its parameters can be seen below in Figure 3.

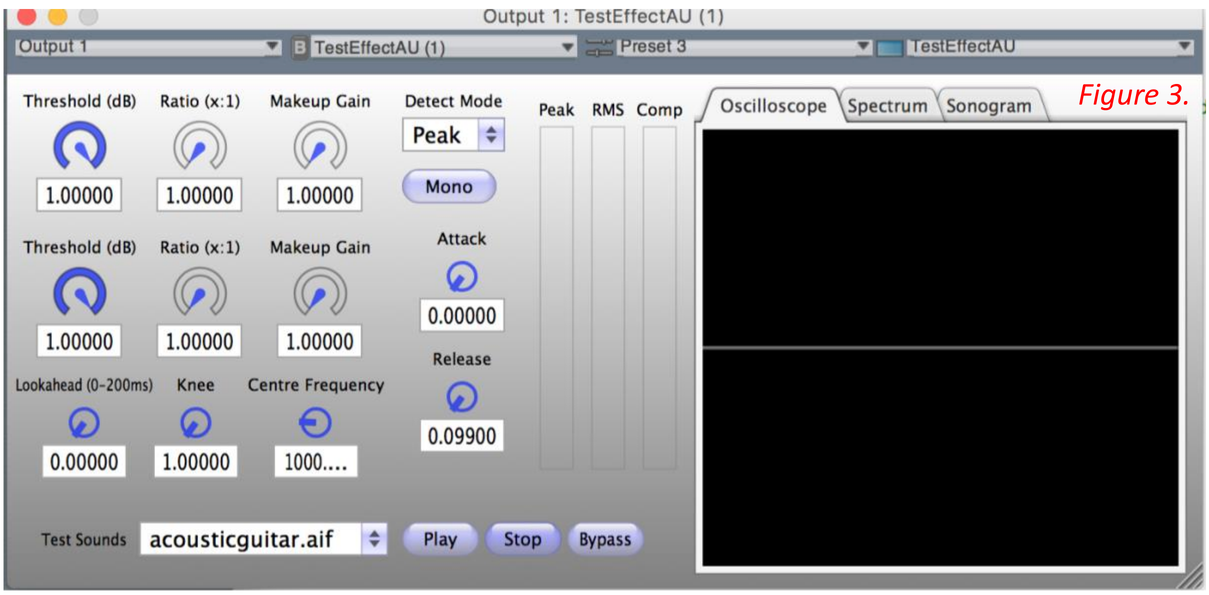

Figure 3.

Processing Technique

A block diagram can be seen in Figure 4 which shows the signal routing of the compressor from input to output. Firstly, both the left and right signals are split into two channels (totalling four channels) which are then sent to low and high-pass filters. These individual frequency bands are then sent into their own compressors which run in parallel to one another. Finally, the signals are then sent to a gain parameter to make up for any attenuation which may have occurred during compression, before the two signals are then summed and sent to the output.

The left and right signals were kept separate throughout the program to allow for complete stereo compatibility. Additional delay lines on both the low and high-pass channels are fed to the compressor which accommodates a *Look-ahead*, granting the user the ability to compensate for any delay in the compression caused by the peak/RMS detector.

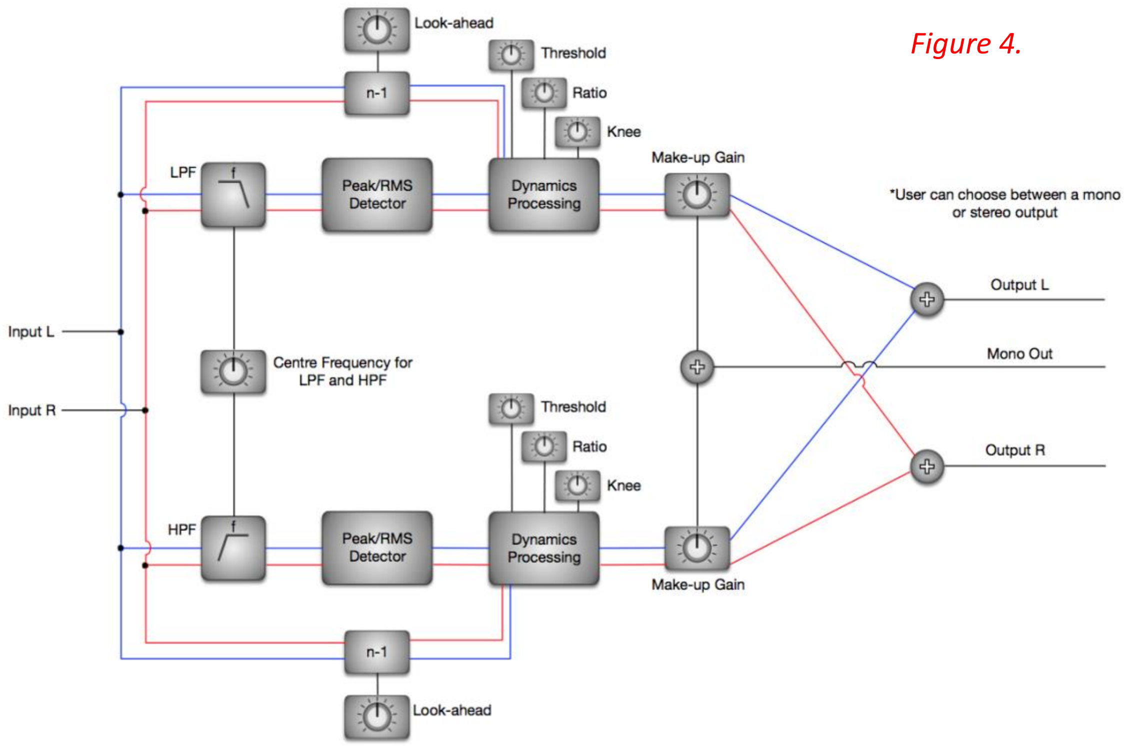

Figure 4.

Two main digital signal processors are used in the compressor: filtering, and amplitude manipulation in the form of dynamic range processing.

A filter chain was programmed initially, whereby the user could define a centre frequency which controlled the cut-off for both a low and high-pass filter. Before the two bands could be recombined, one of them had to be inverted in order to avoid phasing issues. Pirkle writes, 'Remember that filters alter the phase response as well, so recombining them in parallel offers a special problem with how the phases recombine. For the signals to recombine properly, you must invert one of the filter outputs' (2013, p. 475).

A peak meter was then coded so that the peak levels were stored and could be used as the input signal for dynamics processing.

A simple compressor was then programmed, with fixed values for threshold, ratio, attack and release. The filtering and dynamic range processing programs were then combined, and

the resulting program was an amalgamation of *Practical 5* and *Practical 7* from the APDI sessions.
To build on this further, parameter control was integrated into the plug-in so that the user can control the threshold, ratio, attack and release times, as well as the make-up gain. Additionally, a knee parameter allows for adjustment between hard and soft knee compression, implemented using second order interpolation (Pirkle, 2013. USC, n.d.).
As the entire signal chain is programmed to accommodate stereo output, a simple *if* statement in the program allows the user to decide between either output depending on their preference within the plug-in.
It is important that the signal routing was programmed in such a way that processes happened in a specific order. Filtering has to occur before compression otherwise the user would not be able to decide which spectral components were compressed, meaning the plug-in would not serve its intended purpose.

Modifications were made in the user interface controls within the code to allow for greater precision (down to 0.00001, seen in Figure 5) when setting the release times. This is due the use of first order filters within the peak detector, which required very small values to produce an audible compression envelope.

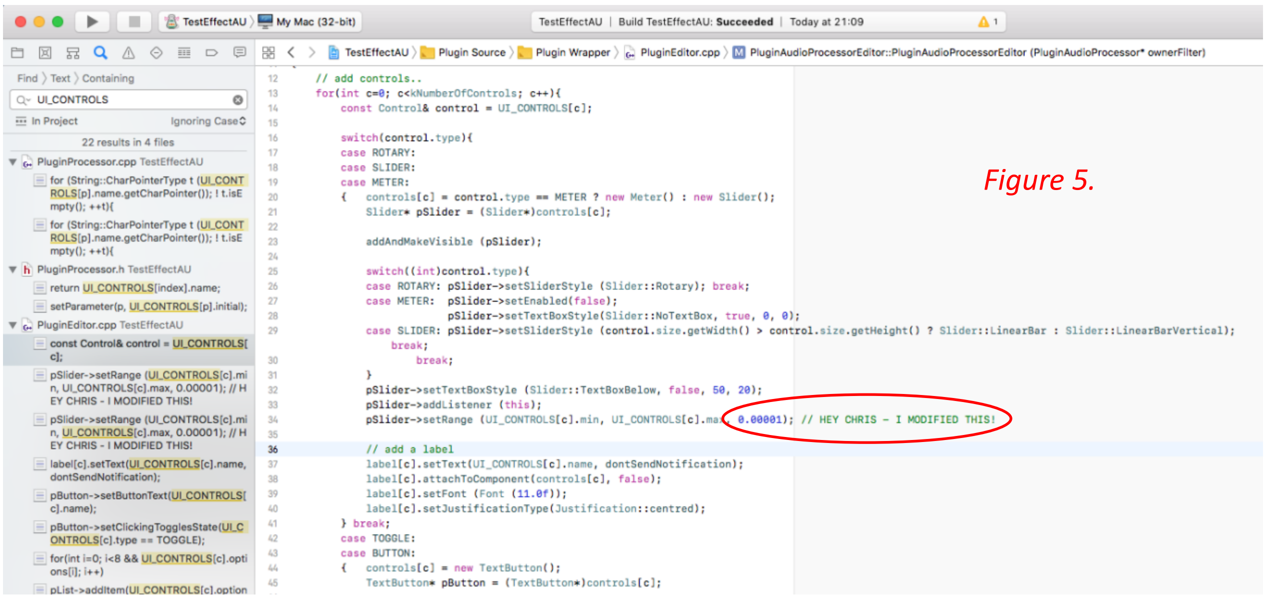

Figure 5.

Audio Analysis

Figure 6.

Figure 6 shows a side-by-side comparison of a drum break which was processed using the compressor. Label 1 shows how the low-frequency information (in this case the kick drums) has been attenuated, whilst Label 2 shows that the hi-hats and other high-frequency information have been amplified, showing the effects of the multiband compression. Label 3 shows how the snare, a broadband frequency sound, remains mostly unaltered.

Figure 7.

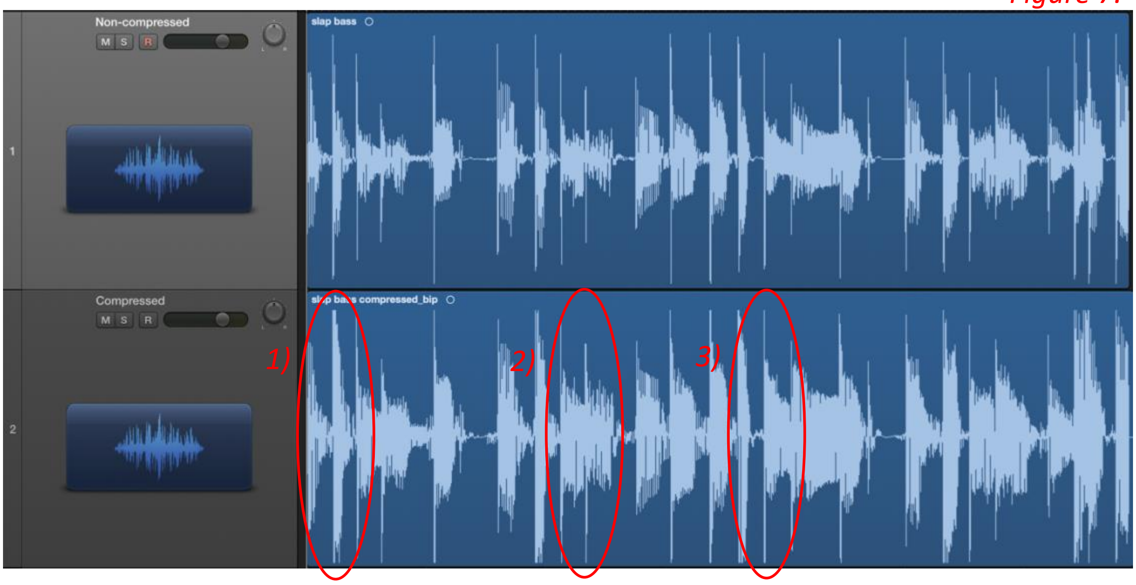

Figure 7 shows a side-by-side comparison of a slap-bass guitar loop which was processed using the multiband compressor. Label 1 shows how some of the transients are being limited, a result of adjusting the attack/look-ahead times within the compressor. Label 2 demonstrates how the low-frequencies in the sustain sections of the bass being played have been amplified. The high frequencies however have mostly retained their dynamic range, as displayed in Label 3.

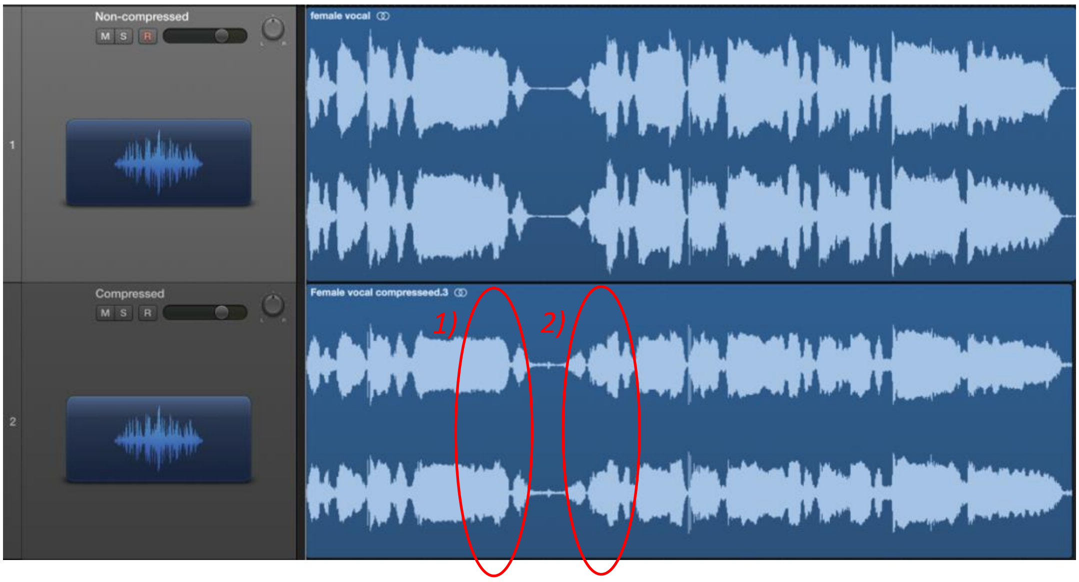

Figure 8 shows a side-by-side comparison of a vocal sample both before and after multiband compression. Firstly, the entire signal has been noticeably attenuated as a result of both low and high frequency compression. However, Label 1 and 2 both show areas within the waveform where quieter signals have been amplified, showing the results of a raised noise floor due to dynamic processing.

Word Count: 1,091

References:

Drum Break (n.d.) *Music Radar: DnB Break Samples*. [Sample] Music Radar

Female Vocal (n.d.) *Bright Lights Vocal Sample Pack.* [Sample] Splice

Pirkle, W. (2013) Designing Audio Effect Plug-ins in C++. United Kingdom: Taylor and Francis.

Slap Bass (n.d.) *Anna Sentina – Pop Bass and Piano* [Sample] Splice

USC. (n.d.) Lagrange's Interpolation Formula. [Online]. pp. 1-2. [Accessed 10 December 2018].