Compressor

VCarrara

Definition

Dynamic range compression or simply compression is the reduction of the dynamic range of an audio signal to avoid unintentional fluctuation in the dynamics (volume, for instance). According with Wikipedia[1], “Compression is an audio signal processing operation that reduces the volume of loud sounds or amplifies quiet sounds, thus reducing or compressing an audio signal's dynamic range. Compression is commonly used in sound recording and reproduction, broadcasting, live sound reinforcement and in some instrument amplifiers.”

Depending on where the controlled gain acts on the input signal, the compressor can be of downward or upward type, as shown in Figure 1. Wikipedia also clarifies that

“Downward compression reduces the volume of loud sounds above a certain threshold. The quiet sounds below the threshold remain unaffected. This is the most common type of compressor. A limiter can be thought of as an extreme form of downward compression as it compresses the sounds over the threshold especially hard.

Upward compression increases the volume of quiet sounds below a certain threshold. The louder sounds above the threshold remain unaffected.”

The opposite of compression is named expansion (sometimes also called expander or even “expandor”). Expansion increases the dynamic range of the audio signal and also comes in two types, downward and upward, as can be seen in Figure 2.

“Downward expansion makes the quiet sounds below the threshold even quieter. A noise gate can be thought of as an extreme form of downward expansion as the noise gate make the quiet sounds (for instance: noise) quieter or even silent, depending on the floor setting.

Upward expansion makes the louder sounds above the threshold even louder.”

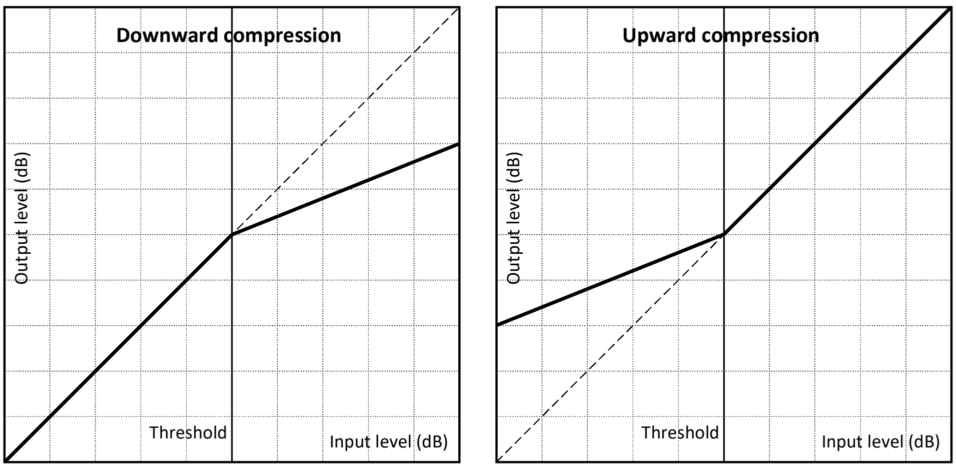

Fig. 1 – Compression of the type Downward (left) and Upward (right)

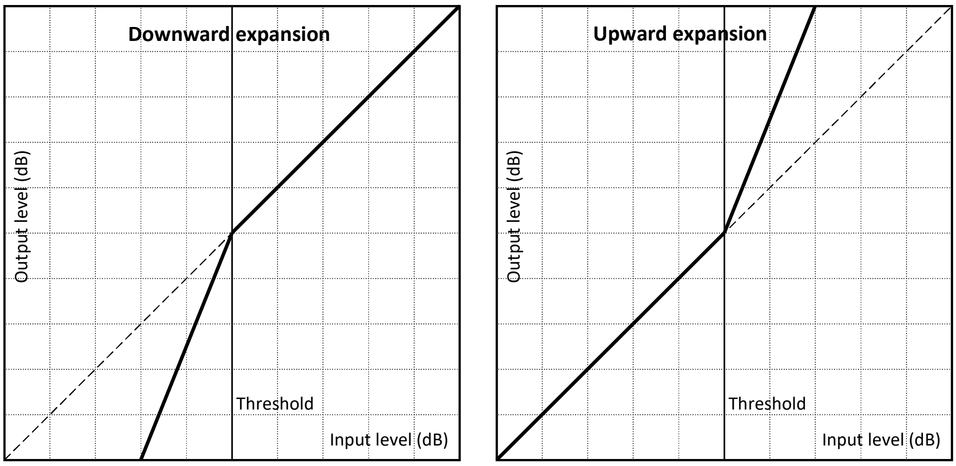

Fig. 2 – Expansion of the type Downward (left) and Upward (right)

More powerful compressors can be achieved by including threshold for several paths in the expansion, compression, and limiter[2]. Compressor can also be used to increase sustain.

Compressor

The adopted compressor model was presented by Giannoulis and others[3] with the formulation of a downward compression, and reproduced here.

Given the input at sampled time n, $x[n]$, the best approach to compute the input level is achieved with the level detector in the linear domain but including the threshold level. In this case the input level is computed by

$$x_L[n] = |x[n]| - 10^{T/20}$$

in which T is the selected threshold in dB, as seen in Figures 1 and 2.

Next step consists in smoothing the input level. Good results in both compression quality and fast algorithm were obtained with the decoupled smooth peak detector[3] which responds quickly when the input level rises, but decreases slowly when the input level fades:

$$y_1[n] = \max\left(x_L[n], \alpha_R\, y_1[n-1] + (1-\alpha_R)\, x_L[n]\right)$$

So the smoothed output level becomes

$$y_L[n] = \alpha_A\, y_L[n-1] + (1-\alpha_A)\, y_1[n]$$

where α_A and α_R are the low-pass filter attack and release parameters, respectively, obtained from

$$\alpha_A = e^{-\frac{1}{\tau_A f_s}}$$

and

$$\alpha_R = e^{-\frac{1}{\tau_R f_s}}$$

in which τ_A is the attack time and τ_R is the release time.

Giannoulis and others[3] suggests two different gain profiles for the compressor: the soft and the hard knee, as depicted in Figure 3. Soft knee gives a smooth transition between the gain ranges, while the gain in the hard knee changes abruptly on the threshold. Due to real time requirements, it was chosen the hard knee, with leads to the compressor gain given by

$$x_G[n] = 20\log_{10}\left(y_L[n] + 10^{T/20}\right),$$

in dB units. By placing the Level Detector before the Gain Computer, as suggested by Giannoulis and others[3], the output level can now be computed with

$$y_G = \begin{cases} x_G, & \text{for } x_G \le T \\ \dfrac{x_G - T}{R} + T, & \text{for } x_G > T \end{cases},$$

in which R is the compression rate (normally around 20 dB).

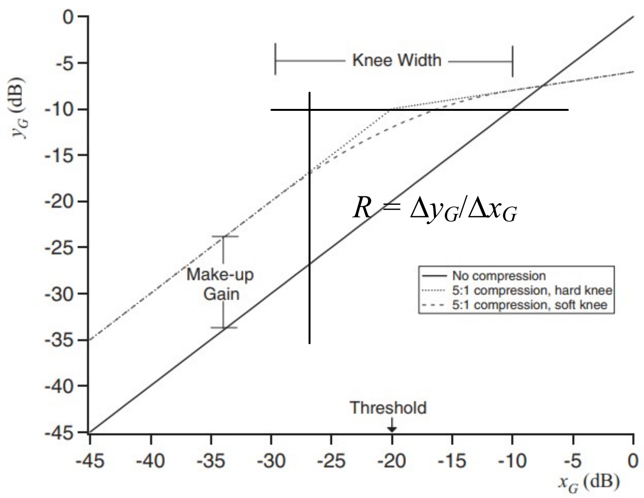

Fig. 3 – Compression curve with hard and soft knees. (Source: [3])

The compressor gain can be obtained in dB units by

$$c_{dB}[n] = y_G[n] - x_G[n],$$

and converting it to linear scale, it gives

$$c[n] = 10^{\frac{c_{dB}[n]}{20}}.$$

The output level (dB) is simply given by

$$y_{dB}[n] = x_{dB}[n] + c_{dB}[n] + M$$

where M (dB) is the Make-up gain that increases the output level of quiet sounds, and therefore the compressor output yields

$$y[n] = M\, c[n]\, x[n] \,.$$

Some procedures were taken to increase algorithm speed, like adopting look-up tables to perform logarithm and exponential functions, beside variables scaling to allow integer math operators.

References

[1] Wikipedia. Dynamic Range Compression. Available at: <https://en.wikipedia.org/wiki/Dynamic_range_compression>, 2023.

[2] McNally, G. W. Dinamic Range Control of Digital Audio Signals. Journal of the Audio Engineering Society, Vol. 32, No. 5, May 1984.

[3] Giannoulis, D.; Massberg, M.; Reiss, J. Digital Dynamic Range Compressor Design – A Tutorial and Analysis. Journal Audio Engineering Society, Vol. 60, No. 6, June 2012.