
Processing Hardware and Software

In this chapter, I will detail all the tools that can be used during a sound recording or during the mixing of a vocal part, whether it is a voice-over, dubbing or singing.

There are a multitude of devices that can be used to shape, modify and improve the sound of the voice. In Chapters 2 and 3 of Volume 2, we will see how to make the best use of the possibilities of each of them.

1.1. The materials

In a recording studio, there are many electronic systems that are part of the audio chain that is necessary for mixing. A few years ago, they were all exclusively present in the form of electronic racks; today, they can also be found in the form of software plugins¹, as can the mixing console, which can also be entirely virtual, in the form of a DAW (Digital Audio Workstation²).

1.1.1. *The compressor*

As its name indicates, the compressor generates *compression*, one of the most popular and effective treatments used by sound engineers. Its principle is simple; it

For a color version of all the figures in this chapter, see: www.iste.co.uk/reveillac.recording2.zip.

1 Appendix 3 of this book provides a non-exhaustive list of some of these plugins.

2 The most popular DAWs are Avid Protocols, Apple Logic Pro X, Ableton Live, Steinberg Cubase, Image-Line FL Studio, Presonus Studio One, Reason 11, Bitwig Studio, Cockos Reaper, BandLab Cakewalk, etc.

brings the level of a sound signal to a predefined value without the operator having to intervene to adapt it to the desired value according to its volume variations.

Compression modifies the quality and the perception that we can have of a sound signal; this is due to its essential action on the *attack* and *release*.

It is used in practically all areas of the music production chain, whether it be during sound recording, mixing or even broadcasting.

Before the advent of compressors, the sound engineer or technician had to vary the volume in relation to the different fluctuations of the incoming sound signal.

It was in the 1950s that the first compressors appeared. They were first developed for radio, although the first-level limiters were created in the 1930s, a little after the development of the first vacuum tubes.

It was not until 1960 that recording studios used them regularly.



Figure 1.1. *The Urei 1176 LN studio compressor*

A compressor acts according to several parameters: *threshold*, *ratio*, *attack*, *release* and *make-up*. We can also add some optional features like *knee* adjustment, the *limiter*, the *side chain*, etc.

The threshold is expressed in decibels (dB). Below this threshold, compression is not active and the sound message is not transformed.

When the threshold is exceeded, dynamic compression is activated and the other parameters become active.

1.1.1.1. *The ratio*

This expresses the proportion of compression that the sound message will undergo. The greater it is, the greater the compression.

Let x be the number of decibels above the threshold; each time x is exceeded by n dB, the compressor will only let through y dB (the part corresponding to what is below the threshold).

$x:y$ represents the ratio or compression rate.

The following equation expresses the output level of a compressor:

$$n_o = \frac{y(n_i - s)}{x} + s$$

with:

n_o : output level in dB

n_i : input level in dB

s : threshold in dB

x : numerator of the ratio

y : denominator of the ratio

Let us take an example:

With a threshold of 15 dB, a ratio of 2:1 and an incoming signal of 17 dB, we have:

$$n_o = \frac{1(17 - 15)}{2} + 15$$

$$n_o = 16$$

The compressor will reduce the signal to 16 dB at the output.

1.1.1.2. *The attack*

This is expressed in milliseconds (ms) and represents the time it takes for the compressor to kick in. By playing on the attack, the *timbre* of the sound signal can be greatly modified.

If the duration of the attack is too long, it is difficult for the compression to establish itself and this is detrimental to the quality of the signal. In fact, a well-dosed length allows the *transients* to pass and preserves the first information of a sound wave, that is, the beginning of the amplitude of the wave.

Some sounds, such as the voice, have present and pronounced transients, while others, such as stringed instruments, have softer transients.

If the duration is too short, the signal will tend to flatten out and the percussive aspects of the signal will be lost.

1.1.1.3. *The release*

Like the attack, this is also expressed in milliseconds (ms). This represents the time it takes for the compressor to become inactive when the threshold is no longer reached or exceeded. Again, the release has a great influence on the timbre of the sound message.

On many compressors, the release can be set to an automatic mode. In this case, the compressor intelligently adapts to the nature of the input signal.

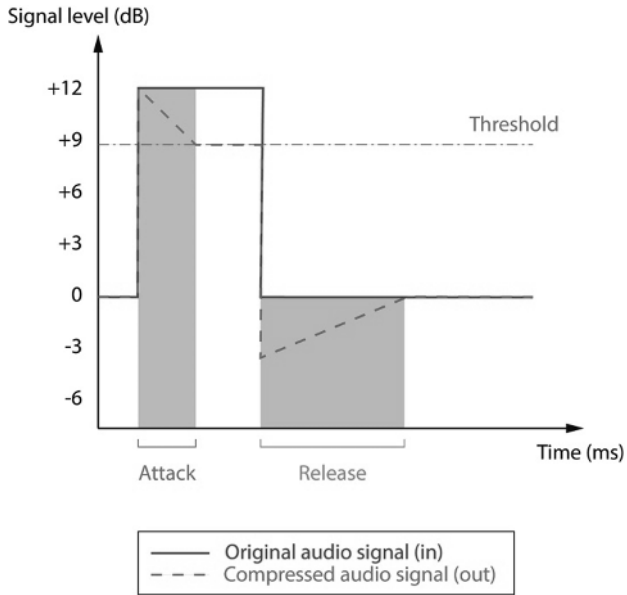


Figure 1.2. *Attack and release during compression*

1.1.1.4. *Gain compensation*

This parameter is present to compensate for the loss of signal gain due to compression.

1.1.1.5. Optional features

The *knee* setting determines whether the compressor kicks in gradually, known as the *soft knee* mode, or abruptly, the *hard knee* mode.

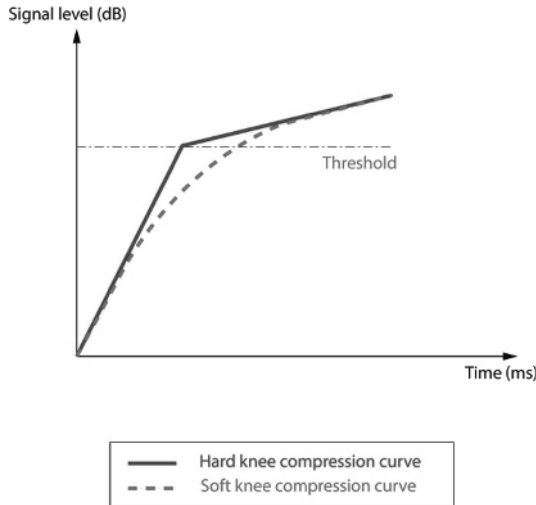


Figure 1.3. The shape of the compression curve as a function of the knee setting

A compressor can also act as a limiter. Many devices are called compressor-limiters. The limiter blocks the signal below the threshold, which is the case when the ratio reaches values of 10:1. It is often used in special cases, such as staying below a critical level during a recording.

The *side chain* is added to the compressor to control the triggering of the processing via an external signal. It is often only available on recent and high-end compressors, and other (internal) treatments included in this chain are also available, such as, for example, the activation of a low-pass filter to suppress *pumping*³ or switching to an insert such as an equalizer (see section 1.1.2) or a de-esser (see section 1.1.5).

The mixer is used to determine the proportion of the original audio signal, at the input, and the compressed audio signal, at the output.

³ During compression, if the release is too fast, we will hear the action of the compressor, which raises the sound level after having reduced it on the peaks. This phenomenon is called “pumping”.

A *low cut comp* filter may exist on some compressors. It eliminates low frequencies in the side-chain control signal.

A *rotation point compressor* establishes a rotation at the point where the audio signal meets the threshold, the purpose of which is to amplify the signal below the level and compress it above.

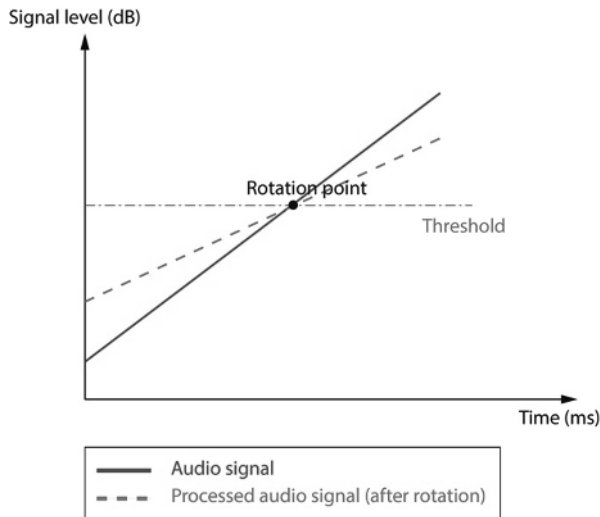


Figure 1.4. *The influence of the rotation point*

1.1.1.6. Some compressors

In Table 1.1, you will find a non-exhaustive list of some studio compressors.

The technology embedded in these devices varies; we can find optical compressors, compressors with field effect transistors (FET), tube compressors, Voltage Controlled Amplifier (VCA) compressors, OTA (Operational Transconductance Amplifier) compressors and many others.

They all have advantages and disadvantages with very different performances in terms of sound rendering. Only by listening to them can you make your choice, knowing that using them to process voices requires sensitivity. Most studios have several compressors, allowing them to provide the necessary coloration or transparency wanted by the sound engineer during mixing.

Some of them integrate other features (*de-esser, noise gate, expander, etc.*).

Manufacturer	Model	Remarks
Alesis	3632	Compressor/limiter/expander/gate – 2 channels
Aphex	CX-500	Compressor/noise gate – 1 channel
ART	PRO VLA II	Optical* tube compressor – 2 channels
Avalon	AD 2044	Optical compressor* – 2 channels
Behringer	MDX2200	Compressor/limiter/gate – 2 channels
Chandler Limited	RS124	Tube compressor – 1 channel
Chandler Limited	Germanium compressor	Compressor – 1 channel
DBX	160VU	Compressor – 1 channel
DBX	266XL	Compressor/limiter/gate – 2 channels
DBX	166XS	Compressor/limiter/gate – 2 channels
Drawner	1968 MKII	Tube compressor – 2 channels
Manley	Variable MU Mastering	Compressor/limiter – lamp – 2 channels
Fairchild	670	Compressor/limiter – 2 channels
NEVE	33609J/D	Compressor/limiter – 2 channels – vintage
Rupert Neve Designs	Portico 5043	Compressor/limiter – 2 channels
Samson	S-Com plus	Compressor/limiter/expander/gate/de-esser – 2 channels
SSL	G Comp	Stereo compressor – Rack Mount API 500
Tornado Music Systems	E-Series stereo bus compressor	Compressor – 2 channels – VCA
Universal Audio	LA2A	Optical compressor – 1 channel
Urei JBL	1176LN	Compressor – 1 channel
Vertigo Sound	VSC-2	Compressor – 2 channels – 4 VCA**
Vintech Audio	609CA	Compressor/limiter – 2 channels
Warm Audio	WA-2A	Tube compressor – 1 channel
Weiss Engineering	Gambit DS1	Dynamic compressor/limiter – broadband, multiband and parallel – AES/EBU only

* Optical compressors have a light source that varies in intensity according to the amount of signal that passes through it, and a photocell that reacts to the brightness of this source. Compressors of this type are generally quite colored or even very neutral. They bring an “organic” side to the sound signal.

** The VCA (Voltage Controlled Amplifier) is a transistorized chip that tracks the incoming signal level to determine the amount of reduction to apply. VCA compressors are associated with fast and accurate compression.

Table 1.1. Some studio compressors

1.1.2. *The equalizer*

The equalizer will correct the timbre of a sound by accentuating or minimizing one or more frequency bands present in an audio signal.

Equalization has, over time, become an indispensable process, allowing the sound engineer to obtain the right frequency balance for a signal and to define its place in the sound space by making it stand out correctly.

The frequency range (tessitura) of the human voice stretches from approximately 100 Hz to 8 kHz for men and 250 Hz to 8 kHz for women, that is, the octaves C2–C8⁴.

It is important not to forget that tessitura is not the only element to be taken into account; every voice has a sound spectrum that is rich in harmonics that must be taken into consideration.

The equalizers are divided into several types: the graphic equalizer, the parametric equalizer and the semi-parametric equalizer.

1.1.2.1. *The graphic equalizer*

The name of this equalizer is derived from its ergonomics, which propose a series of linear potentiometers (sliders), delimiting frequency bands, aligned and positioned in the increasing order of the audible sound spectrum, from left to right.

The advantage is that it is easy to visually evaluate the corrections made to the incoming signal. The gain/attenuation level for each band can vary by a maximum of +/-18 dB.



Figure 1.5. A DBX 1231 2 x 31 band graphic equalizer

⁴ The octaves are distributed from C0 to C10, that is, 16.3 Hz to 19,912 Hz.

Over the years and with the evolution of technology, the number of bands on these equalizers has increased from 5 bands to 31 bands (1/3 octave), or even up to 64.

1.1.2.2. *The parametric equalizer*

This type of equalizer provides the choice of the type of filter to be applied, its cutoff frequency or center frequency, the bandwidth and the gain to be applied.

This equalizer is more precise than a graphic equalizer but requires more finesse in its use.



Figure 1.6. *A Tube-Tech PE1C parametric equalizer*

This type of equalizer only works on one frequency band; several are needed to make more selective corrections. Most manufacturers offer two- or four-band parametric equalizers.

There are parametric equalizers called passive equalizers on which you can only decrease the gain of a frequency. An additional active circuit restores the output gain level. The advantage of this type of equalizer is that there is little signal alteration caused by the electronics, which is only linked to the slight losses induced by the passive, non-powered components.

1.1.2.3. *The semi-parametric equalizer*

Equalizers of this type combine some of the characteristics of the two previous types.



Figure 1.7. *The Aphex EQF 500 semi-parametric equalizer*

Generally, they are made up of three (high, medium and low pitch) or four frequency bands on which a central cut-off frequency can be defined while being able to choose the gain.

1.1.2.4. *Some equalizers*

Manufacturer	Model	Remarks
Apex Audio	DBQ Zero	Graph – 2 x 30 bands – +/-15 dB
Avalon	AD 2055	Parametric – 3 bands – 2 channels
Boss	GE-215	Graphic – 2 x 15 bands – +/-15 dB
BSS Audio	FCS-966 Opal	Graphic – 2 x 30 bands – +/-15 dB
DBX	2231	Graphic – 2 x 31 bands – +/-15 dB
D.W. Fearn	VT-4	Parametric – 1 channel – electronic tubes
Empirical Labs	Lil FreQ EL-Q	Parametric – 4 bands – 2 Baxandall ⁵
GML	Model 8200	Parametric – 5 bands – 2 channels
Gyraf Audio	Gyratec XIV	Parametric – 5 bands – passive – electronic tubes
Manley	Massive Passive EQ	Parametric – passive – electronic tubes – 4 bands – 2 channels
Moog Music	10 Band Graphic EQ	Chart – 10 bands
Moog	3 Band Parametric EQ	Parametric – 3 bands

⁵ Bass-to-treble tone correction around a center frequency of about 1 kHz.

Manufacturer	Model	Remarks
Orban	672A	Parametric – 8 bands
Pultec	MEQ5-SS	Parametric – electronic tubes – 2 boost bands and 1 attenuation band – 300–5,000 Hz
Rupert Neve Designs	Portico 5033	Parametric – 3 bands + 2 shelf bands ⁶
Samson technologies	E62i	Graphic – 2 x 31 bands – +/-12 dB
Urei/JBL	527A	Graphic – 27 bands – +/-10 dB
Summit	EQP-200B	Parametric – 2 bands + 1 band low tray – electronic tubes
Tube-Tech	EQ 1AM	Parametric – 3 bands – +/-20 dB – electronic tubes
Tube-Tech	PE1C	Parametric – 2 bands – passive electronic tubes
Weiss	EQ1	Parametric – digital – 7 bands – 2 channels

Table 1.2. *Some studio equalizers*

1.1.3. Reverberation and delay

Because this effect is present in certain natural environments and can also be created artificially, it has given rise to much controversy. Purists will say that using artificial reverberation is simply detestable. Many others will tell you that it is a creative tool of great relevance, completely customizable, which brings the musician and the sound engineer infinite possibilities of redefining and reshaping the ambience of the sound palette.

The heading of this section also mentions the term delay. What is the difference between the two?

The delay reproduces, among other things, the phenomenon that everyone knows under the name of echo. That is, a repetition of the sound that weakens in time. The analogy in text form would be this one:

– You shout **ECHO**

⁶ A *plateau* is a filter that starts at one end of the audible frequency range (or covered by the equalizer) and continues to a selected frequency. There are two shelves, low and high, for each of the two ends.

- and a few tenths of a second later, the place answers you:
- ECHO ... ECHO ... ECHO ... echo... ech... ec...

This repetition is related to the fact that the sound waves produced by your cry are reflected far away and return to you. The intensity, when they return, is lessened because their collision with the material that reflects them absorbs part of their energy. Once they return to you, the process starts again and continues for several round trips until the signal intensity is completely exhausted.

The fact that you can distinguish between the different repetitions is specific to echo or delay, unlike reverb where the distinction is not possible.

To be precise, delay is different from echo because its number of repetitions can be infinite. However, in common parlance, professionals mostly use the terms echo and delay without differentiation.

1.1.3.1. *Principle of reverberation*

A sound wave when it travels in the air undergoes reflections on the obstacles, walls, floor, ceiling and various objects that it meets. These reflections follow one another and make the sound wave travel longer and longer paths while reducing its energy, as it is absorbed by the materials it hits.

When a person is placed in a reverberant environment, they will hear the same sound signal several times with slight temporal differences, which will stretch it, increasing its duration compared to that of its emission. This phenomenon is called reverberation, and the duration of the prolongation of the sound signal is the *reverberation time*.

Reverberation is often perceived as a trail (in large venues such as churches and cathedrals) and/or as a reinforcement of the sound signal (small rooms with low absorption) compared to a sound emission in the open air.

When the space is small, the temporal shifts are weak so the sound signal is not repeated and we hear it as one. If the dimensions of the space where the emission takes place become significant (valleys) or have concave forms (tunnels), the sound signal takes a much longer time to reach the ears of the listener, who can then distinguish between each sound arrival (repetition). In this case, the reverberation becomes an *echo*.

In order to measure reverberation time, acousticians have defined the theoretical reverberation time of a place as the time after which the sound energy is reduced to 1 millionth of its initial value, that is, an attenuation of 60 dB.

In architecture, it is important to be able to know the reverberation time before construction. The Sabine equation⁷ theoretically calculates the reverberation time

$$T = \frac{0.163 \times V}{A}$$

with:

0.163: constant expressed in s/m

T : reverberation time in s

V : volume of the room in m^3

A : absorption area in m^2 of opening (sum of the product of each of the surfaces S_n in m^2 by their Sabine absorption coefficient α_n)

$$A = (\alpha_1 \times S_1) + (\alpha_2 \times S_2) + \dots$$

You will find in Table 1.3 some Sabine absorption coefficients depending on the frequency of the emitted sound waves.

Material	125 Hz	250 Hz	500 Hz	1 kHz	2 kHz	5 kHz
Wood	0.09	0.11	0.1	0.08	0.08	0.1
Cement plaster	0.01	0.01	0.02	0.02	0.02	0.03
Wallpaper	0.01	0.02	0.04	0.1	0.2	0.3
Carpet	0.12	0.2	0.25	0.45	0.4	0.35
Rock wool: thickness 4 cm	0.3	0.7	0.88	0.85	0.65	0.6
Raw plaster	0.04	0.03	0.03	0.04	0.05	0.08
Painted plaster	0.01	0.01	0.02	0.03	0.04	0.05
Raw concrete	0.01	0.01	0.01	0.02	0.05	0.07
5 mm plywood + 5 cm of air	0.47	0.34	0.3	0.11	0.08	0.08
Tile	0.01	0.015	0.02	0.025	0.03	0.04
Ordinary glazing	0.3	0.22	0.17	0.14	0.1	0.02

Table 1.3. Sabine absorption coefficients according to materials and frequencies

⁷ Wallace Clement Sabine, 1868–1919, American acoustic engineer and Harvard graduate.

The more α_n tends to 1, the more the wall absorbs all the sound energy. When nothing is reflected, the material is perfectly absorbent. The more α_n tends to 0, the more the material is reverberant.

Sabine also showed (Sabine's laws):

– The curve of sound establishment and its curve of attenuation have a practically exponential appearance and, moreover, they are complementary. The increase in sound energy during a given time is equal to its decrease during the same time.

– The effect of an absorbing material is independent of its position.

– The decay time of a sound is approximately the same at all points of the considered sound space.

– Reverberation is independent of the position of the sound source in the space considered.

However, it is necessary to weigh these laws carefully, as they can vary in certain cases, such as in the presence of furniture, people or concave or convex walls in the place considered.

1.1.3.2. *Types of reverberation*

Within recording studios, we can find several types of reverberations which all have their specificities and their sound rendering: the analog reverbs (spring, plate, etc.), the digital reverbs with algorithms or convolutions, and the atypical reverbs (magnetic tape, magnetic plate, etc.).

Analog reverbs, which date back to the 1950s, are based on mechanical principles with transducers that excite a metal plate or springs, while digital reverbs, known as algorithmic – which are more recent, the first having been presented to the public in 1976 – use a digitized audio signal processed by specialized processors linked to memory storage⁸.

These have been further refined with the implementation of the principle of signal convolution in 1999, which models the impulse response of a room.

In parallel to these evolutions, other more atypical models appeared to be built around systems mixing electronics and mechanics associated with magnetic tapes or trays.

⁸ For more information, see the References section at the end of the book.



Figure 1.8. *The first EMT 250 digital reverberator, dating from 1976*



Figure 1.9. *The first convolution reverb, Sony DRE-S777*



Figure 1.10. *A magnetic tape reverb, Roland Space Echo RE-201*

1.1.3.3. *The parameters of the reverberation*

A reverb has a set of parameters that allow you to refine the effect you want to obtain:

– The *reverberation time* or RT60 is the time during which the sound signal is perceived after the disappearance of the original signal. This measure is defined as the time necessary for the signal to decrease by 60 dB (hence the acronym RT60 – *Reverb Time 60 dB*) compared to the original level (direct sound).

– The *pre-delay* is the time it takes for the audio signal to produce its first reflections, called *primary reflections*. In reality, the size of the room and the various obstacles define this parameter. The closer they are, the shorter this time is. For this parameter, the higher its value, the larger the simulated location will be. We can say that this parameter aerates the signal by separating the *dry* sound from the *wet* sound of the reverberation.

– The *early reflections* or the initial level that forms the initial reflections by gathering the first responses that appear when the audio signal meets the reflecting obstacles. If the room or place is huge (valley, tunnel, etc.), these reflections are echoes but generally in classical places (empty room, church, cathedral, etc.), they are perceived as reverberations. As a parameter, it has a direct relationship with the *decay* level of the reverberation. When the first reflections are very weak, you simulate a displacement of the listener that varies from the back to the center of the space considered.

– The fall (*decay* or *decay time*), the fading or the attenuation. It occurs within the *secondary reflections*, that is, the reflections of reflections and leads to a phenomenon of diffusion of sound waves. As the number of reflections increases, the intensity decreases until it disappears completely. This phenomenon is due to the absorption of the waves by the sound space. The fall is a fundamental parameter of the reverberation.

In addition to the classic parameters, there are a multitude of settings specific to certain reverb technologies:

– The *crossover frequency*, which defines the switch point between high and low frequencies, used by other parameters.

– The *damping* which characterizes the reflections according to the material on which they take place, a hard surface which will have a great dynamic and will favor the high frequencies or a soft surface which will absorb them and provide a warmer sound. The modification of this component gives the reverberation a texture ranging from artificial to natural.

– The *density* which increases or decreases the space between the first reflections of the reverberation and the following ones. The nature of the sound signal is of great importance for this component; generally, the drier it is (very short attack), the greater the density should be and vice versa. The density can be seen as the thickness of the reverberation, in that it is quite close to the diffusion.

– The *diffusion*, which affects the overall effect and whose influence is essential in the rendering of the reverberation. When this component increases, the primary reflections come closer together, which increases the mass of the sound signal. The echoes are less and less perceptible individually. The nature of the sound message is the determining factor.

– The frequency *attenuation* (*frequency level*, *frequency decay* or *frequency attenuation*) is a parameter that is often broken down to work on two or more frequency bands (bass, midrange, treble, etc.). In practice, the attenuation occurs first of all on the high frequencies, but nothing prevents you from doing the opposite, in which case the sound becomes more unreal. This parameter is proportional to the time elapsed. It is an important parameter for the globality of the effect.

– The *gate* defines the signal level below which the reverb is *cut off*.

– *Gate decay* determines how long the reverb will be off when the signal falls below the *gate* value.

– *Mix* is a parameter that determines the proportions of the mix between the initial audio signal (*dry*) and the reverb (*wet*).

– The type of algorithm (*preset*) which defines the impulse response. It is often associated with a simulated real acoustic, or even a virtual one that is non-existent in reality. This type is often present in the form of presets (pre-selections) within digital reverberations.

– The *room size* (or *size*) which determines the reverberation time in accordance with the dimensions of the virtual room where it is simulated. The longer it takes for the sound waves to travel before they are reflected against an obstacle (wall), the larger the room.

– The *spread* widens or narrows the stereo image of the *reverb tail*⁹ from a monophonic spread at its minimum value to a very wide stereo field at its maximum.

– The *width* defines the width of the stereo image in order to reduce or enlarge it.

⁹ We call the tail of reverberation or “*diffuse reverberation*” the reverberation that appears when the multiplications of the reflections are such that they merge in time, giving them a homogeneous character that gives them a diffuse aspect of great density.

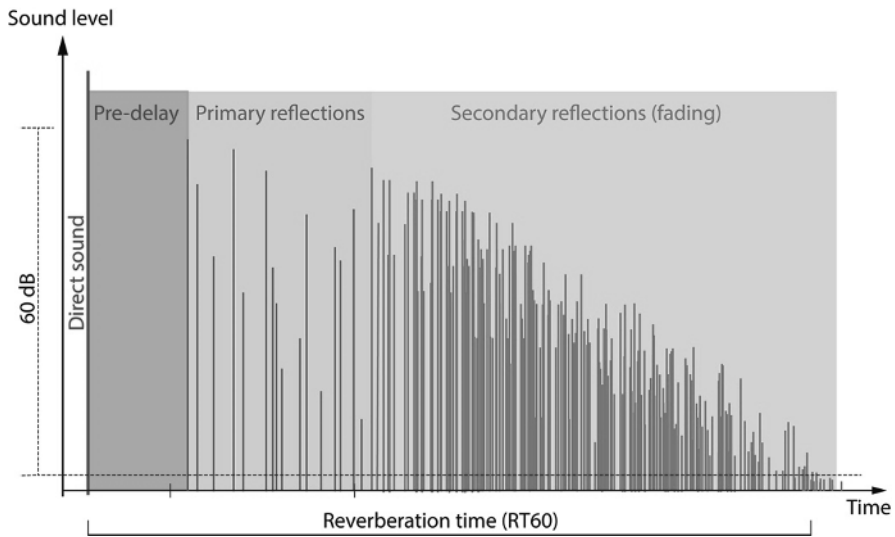


Figure 1.11. *The main parameters of the reverberation*

1.1.3.4. Some reverberations

Most of the reverbs in Table 1.4 have been or are still being marketed as new or used, but they are still very much in vogue and often provide an outstanding sound.

Despite their age, they are often very expensive. Many of them have remote controls, a strong point when working in a studio.

The list below is far from being exhaustive, even if rack reverbs are becoming rare and are being replaced by software equivalents which often have nothing to do with grain and coloration.

Manufacturer	Model	Main features
AKG	ADR68K	Digital – 16 bits – 32 kHz + remote control (year 1986)
AMS Neve	RMX16 500	Digital – 32 bits – 48 kHz – USB-C (year 2020)
Binson	Echorec 2	Echo chamber with a magnetic plate (year 1960)

Manufacturer	Model	Main features
Dynacord	Echocord Super 76	Tape echo chamber and spring reverb (year 1976)
Lexicon	PCM-42	Digital (year 1982)
Lexicon	MPX-1	Digital – 24 bits – 44.1 kHz. Digital multi-effects (year 1997)
Lexicon	MPX-550	Digital – digital multi-effects (year 2012)
Lexicon	PCM-92	Digital – Reverb + Effects – 24 bits – 96 kHz (year 2009)
Lexicon	PCM-96	Digital – 24 bits – 96 kHz + effects (year 2009)
Lexicon	480L	Digital – 18 bits – 48 kHz + remote control (year 1988)
Lexicon	224	Digital + remote control (year 1978)
Roland	R-880	Digital 16 bits – 48 kHz + remote control (year 1989)
Roland	Space Echo RE-201	Magnetic tape echo chamber (year 1973)
Sony	DRE 2000	Digital + remote control (year 1981)
TC Electronic	M3000	Digital – 24 bits – 48 kHz (year 2002)
TC Electronic	System 6000 MKII	Digital – 24 bits – 96 kHz+ remote control (year 2010)
Vermona	DSR-3	Spring reverb – 2 band equalizer included – stereo (year 2013)
Yamaha	Pro R3	Digital – 20 bits – 44.1 kHz + effects (year 1995)

Table 1.4. *Some reverbs, delays and echo chambers for studios*

1.1.4. *The de-esser*

This is a tool that reduces sibilance, whistling or hissing sounds (“ch”, “s”, etc.) that are naturally present in a voice but can become annoying following a sound treatment using compression, delay or reverberation.



Figure 1.12. The SPL 9629 studio de-esser

1.1.4.1. Principle

The principle of the de-esser is simple: it filters out the sibilance and compresses it, leaving the rest of the audio signal unprocessed.

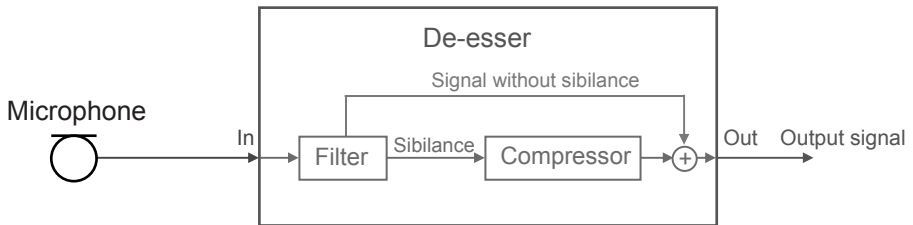


Figure 1.13. Schematic diagram of a de-esser

By default, a de-esser is almost exclusively used as an insert on a track. Its settings are simple and are often limited to a *threshold* (which can be automatic), a *frequency range* or a *male/female* preset and an *attenuation range*.

There is also a frequency detector, an *attack* and *release* control and various filters.

Some compressors and some equalizers have a de-esser function.

1.1.4.2. Some de-essers

Table 1.5 shows a few de-essers, some of which have other features.

Manufacturer	Designation or model	Remarks
Behringer	Pro-XL MDX2600	Expander/gate/compressor/peak limiter/enhancer – 2 channels
BSS Audio	DPR 422 Opal	Compressor/de-esser – 2 channels
Dane	#31 Optical De-esser	1 channel
DBX	520 De-esser	De-esser – rack module model 500 – 1 channel
DBX	263X	1 channel
Drawmer	MX50 - Dual de-esser	2 channel de-esser
Empirical Labs DerrEsser	EL-DS	1 channel
SPL	De-esser 9629	One of the most used in studios – 2 channels
Valley people	Micro FX	Expander/gate – 4 channels

Table 1.5. Some studio de-essers

It should be noted that before the appearance of de-essers, in a way that is still possible today, we can create an excellent de-esser equivalent with a compressor and a filter and possibly an amplifier. To do this, we use side chaining on the compressor, as shown in Figure 1.14.

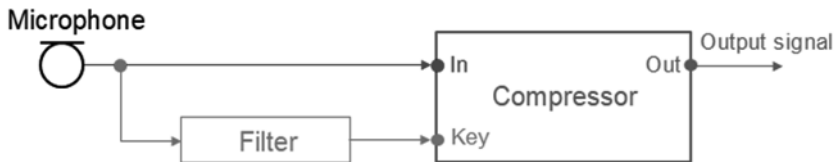


Figure 1.14. Equivalent of the de-esser with a filter and a compressor. Very often, the filter is associated with an amplifier which increases the presence of sibilance for better compression

1.1.5. The expander

The dynamics *expander* is often integrated into compressors.

It is a device whose role is to extend the dynamics of an audio signal. It works in opposition to the audio compressor.

1.1.5.1. Parameters and principle

As for a compressor, for an expander we find the *threshold*. Above a certain value, the expander remains inactive, but below this value, it comes into action according to a certain expansion ratio. The values are 1:2, 1:4 etc. up to 1:10.

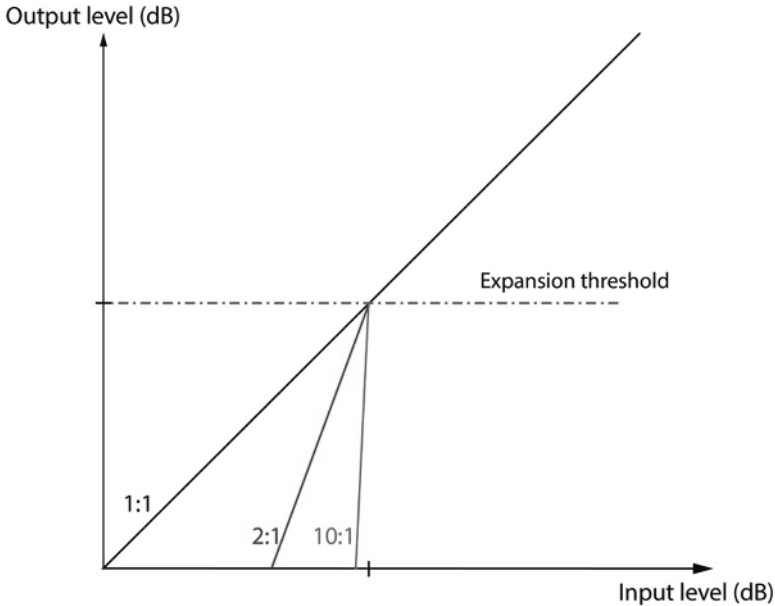


Figure 1.15. *The principle of the expander*

There are also temporal parameters such as *attack*, *release*, *knee* and a *hold* function whose purpose is to maintain the action when the signal has passed the threshold and before the release has elapsed.



Figure 1.16. *The DBX 1074 gate expander*

There is also a *side-chain* mode, as on many other dynamics tools.

1.1.5.2. Some expanders

The expanders in Table 1.6 usually have other features, including a *gate* function that blocks the audio signal when the level falls below a certain threshold.

When the expansion ratio of an expander is greater than or equal to 1:10, it can be considered to behave like a gate.

Manufacturer	Model	Remarks
Behringer	Multigate Pro XR4400	Expander/gate – 4 channels
Behringer	Multicom Pro-XL MDX4600	Expander/gate/compressor/limiter – 4 channels
DBX	1074 QuadGate	Expander/gate – 4 channels
Drawmer	DS101	Expander/door – Model 500 Rack Module
Samson Technologies	S-Com plus	Compressor/gate/expander/de-esser/limiter

Table 1.6. Some studio expanders

1.1.6. The exciter or enhancer

The exciter, which is also called an *aural exciter*, *harmonic exciter* or *enhancer*, depending on the different manufacturers, is a device that modifies the audio signal by bringing a dynamic equalization, generally on the high frequencies of the sound spectrum. Its purpose is also to increase the harmonics in the sound message in order to make it more contrasting and therefore more intelligible.

In French, the equivalent term used is “*embellisseur sonore*”.

Its use is difficult and requires sensitivity so as not to fall into caricature, especially with voices.



Figure 1.17. The aural exciter model 602 from ApheX

Its appearance, in 1975, with the famous “Aural Exciter” from Aphex, gave rise to a real craze in many studios which snatched it up and used it on a huge number of productions. However, its use did not last; as its qualitative contribution was very debatable, the professionals quickly realized that a good equalization could easily replace it.

Brand	Model	Remarks
BBE	Sonic Stomp	-
TC Electronic	Bodyrez	-
Boss	EH-2	-
Aphex	1401	Transistor preamp
Aphex	Xciter	-
Behringer	SE200 Spectrum Enhancer	-
Aphex	The Exciter	2 channels – XLR and jack inputs/outputs
Aphex	204 Aural Exciter	2 channels – XLR and jack inputs/outputs
Aphex	103A Aural Exciter	2 channels – jack inputs/outputs
Aphex	204 Aural Exciter	2 channels – XLR and jack inputs/outputs
BBE	Sonic Maximiser 882i	2 channels – XLR and jack inputs/outputs – diode input meter and peak indicator
Behringer	Ultrafex II EX3100	2 channels – XLR and jack inputs/outputs – surround
Behringer	Ultrafex Pro EX3200	2 channels – XLR and jack inputs/outputs – surround
Rocktron	RX20	Designed for the guitar
Rocktron	RX1	Designed for the guitar
SPL	Vitalizer Tube	2 channels – XLR and jack inputs/outputs – high-end exciter
SPL	Spectralizer	99 presets – AES/EBU – S/P-DIF – RS422 – MIDI In/thru
SPL	Vitalizer SX2 pro	2 channels – XLR and jack inputs/outputs
SPL	Vitalizer Mk2-T	2 channels – XLR and jack inputs/outputs

Table 1.7. *Some studio enhancers*

1.2. The software

All the materials that have been presented above, compressors, equalizers, reverbs, de-essers, expanders and exciters, also exist in software version in the form of plugins recognized by all DAWs.

The market is very dynamic, and these new tools are very numerous, some disappearing while others are entering or evolving into new versions.

Compared to hardware devices, they often offer more functionality and are generally more open. Many adopt the design of their elders by copying their front panel, even adding, for some, additional controls.

Their cost is much lower than the originals, and for a few hundred euros, you can own tools that in the form of electronic racks would be worth tens of thousands.

Though I recognize that plugins can be very useful, they only work with software that runs on a computer and that's where the problem lies. Where is the purely physical aspect linked to the manipulation of control buttons and switches, which makes the correction(s) you want to make to the sound sequence tangible? Everything remains virtual, frozen in the digital, without soul, a complex combination of sequences of bits, an uninterrupted string of 0s and 1s.

To summarize, and this is only for me and the many sound engineers who still work with this equipment, software tools are a necessary evil but will never replace hardware tools.

In Table 1.8, you will find a list of plugins classified by functionality; the order is the same as in the previous sections. Most of them exist for Microsoft Windows and macOS operating systems, in the form of VST, AAX, RTAS or AU¹⁰.

1.3. Conclusion

As you will have understood after reading this chapter, there are many possible treatments and it is difficult to make a choice. At the very least, a compressor, an equalizer and a reverb are three essential devices for satisfactorily reworking a vocal recording, as Chapter 3 of this volume shows.

There are other hardware and software programs that can be classified as special effects that can also add value to your work, which will be briefly discussed in Chapter 4 of this volume.

¹⁰ See Appendix 5 at the end of the book.

Function	Publisher	Designation
Compressor	Audiocaption	Compressor AC-1
	Eventide	Omnipressor
	IK Multimedia	Classic T-RackS Compressor
	MeldaProduction	MCompressor
	PSP Audioware	MasterComp
	Softube	Valley People Dyna-mite
	URS	1975 Classic
	Waves	H-Comp
	Waves	PuigChild 670
	Waves	API 2500
	Waves	SSL G-Master Buss Compressor
	Waves	VComp
	Waves	CLA-3A
	Waves	C1 Compressor
	Waves	Renaissance Compressor
Equalizer	Abbey Road	TG Mastering Pack
	Bomb Factory	Pultec Bundle
	FabFilter	Pro-Q2
	IK Multimedia	T-Racks Classic Equalizer
	MeldaProduction	MEqualizerLinearPhase
	PSP Audioware	Retro Q
	RJProjects	Aqualizer
	Rob Papan	RP-EQ
	Softube	Tube-tech ME1B
	Softube	Tube-Tech PE1C
	Sonnox	Oxford EQ
	SPL	Free Ranger
	TC Electronic	Assimilator
	Universal Audio	Pultec Passive EQ Collection
	Universal Audio	NEVE 1073 EQ
	Voxengo	HarmoniEQ
	Voxengo	GlissEQ 3
	Waves	Linear Phase
	Waves	Q10
	Waves	H-EQ
Waves	RS56 Passive EQ	
Waves	V-EQ4	

Function	Publisher	Designation
Algorithmic reverb	Waves	H-Reverb
	UVI	Sparkverb
	Lexicon	MPX PCM LXP
	Eventide	UltraReverb TVerb
	Universal Audio	EMT 140
	Valhalla DSP	ValhallaRoom ValhallaPlate
	Sinusweb	FreeverbToo
	Kresearch	KR-Reverb FSQ
	Rhythm Lab	Mo'Verb
Convolution reverberation	TC Electronic	ME30 ClassicVerb VSS3 DVR2 powercore
	Acustica Audio	Nebula HS Reverb
	Audio Ease	Altiverb 6 XL
	Audio Ease	Altiverb 7
	Audio Ease	Altiverb 7 XL
	Native instrument	Reflektor
	Thomas Resch	Tconvolution
	VSL	Vienna MIR
	VSL	Vienna MIR Pro
De-esser	Waves	IR360
	Digitalfishphones	Spitfish
	Eiosis	E2deesser
	FabFilter	Pro-DS
	Nomad Factory	BT Deesser DS-2S
	Sonnox	Oxford SuprEsSer
	Waves	DeEsser
Expander	Waves	Renaissance DeEsser
	Audio Damage	AD022 BigSeq-2
	FabFilter	Pro-G
	Kjaerhus Audio	GAG-1
	MeldaProduction	MStereoSpread
	Nomad Factory	AS Gate Expander
Waves	Primary Source Expander	

Function	Publisher	Designation
Exciter	BBE	Sonic Sweet
	Crysonic	Spectralive NXT-3
	MeldaProduction	MStereoProcessor
	Nomad Factory	AS-Exciter
	SPL	Vitalizer MK2-T
	Voxengo	Warmifier
	Waves	Renaissance Bass
	Waves	Maserati B72
Waves	Vitamin	

Table 1.8. *Software plugins for digital audio workstations*