WAVES SERIES

Recording and Voice Processing 2

Working in the Studio

Jean-Michel Réveillac

iSTE

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Preface

If you want to know if this book is for you, how it is constructed and organized, what it contains and what conventions will be used, you’ve come to the right place, this is the place to start.

Target audience

This book is intended for all those who, amateur or professional, are interested in sound recording, recording and mixing in the field of singing and voice or musicians, performers, commentators and composers.

The work presented in some sections requires minimum knowledge in the field of acoustics and digital audio.

You must have a good knowledge of your computer’s operating system (paths, folders and directories, files, names, extensions, copying, moving, etc.) and know how to handle a DAW (Digital Audio Workstation), such as Avid Pro Tools, Apple Logic Pro X, Ableton Live, Steinberg Cubase, FL Studio, MOTU Digital performer, Cockos Reaper, etc., or a digital integrated studio, such as Tascam DP-03SD, Tascam DP32, Roland VS-1680, Akai DPS16HD, Yamaha AW4416, etc.

Structure and contents of the book

This work is composed of two volumes:

1) history and generalities;

2) studio work.

Volume 1 presents a preface, specifying the contents and the writing conventions used, then an introduction followed by four chapters, a conclusion and five appendices:

– recording history;

– voice;

– microphones;

– acoustic environment.

The conclusion summarizes the main topics discussed and introduces the concepts that will be addressed in the second volume.

Appendices 1–5 provide some additional information. You will find in this order:

– sound unit;

– audio connectivity;

– audio processing plugins;

– tube and JFET mic amplifiers;

– microphone pairs.

Volume 2 presents a preface and an introduction identical to those of Volume 1 followed by four chapters, a conclusion and five appendices:

– processing hardware and software;

– configuration and audio channel;

– voice recording;

– special effects.

Appendices 1–4 are taken from Volume 1 to complement the previous chapters by including:

– sound unit;

– audio connectivity;

– audio processing plugins;

– microphone pairs.

Appendix 5 of Volume 2 provides details on the types of software plugins available from different vendors and operating systems.

The conclusion sheds light on the whole book and gives a brief overview of the future evolution of voice recording.

Each volume can be read separately. While there are concepts that are dependent on another chapter, references to the relevant sections are given. However, the first two chapters of Volume 1, devoted to the history of recording and to the human voice, provide a contextual basis for the understanding of several notions that you will find in the following chapters.

If you're a novice on the subject, I strongly advise you to read them first, to discover the basics of the subject of this book.

For the others, I hope that you will discover new notions that will enrich your knowledge.

At the end of each volume, you will find a reference list and a list of Internet links.

A glossary is also present; it will explain some acronyms and terminologies very specific to sound recording, recording and mixing.

Conventions

This book uses the following typographical conventions:

Italics, which are reserved for important terms used for the first time in the text which may be present in the glossary at the end of the book, mathematical terms, comments, equations, expressions or variables.

Remarks are indicated by the presence of the keyword: NOTE. They complete the explanations already provided.

The figures and tables all have a legend that is often useful for understanding.

Vocabulary and definition

As with all techniques, voice recording has its own vocabulary. Certain words, acronyms, abbreviations, initialisms and proper names are not always familiar and will be included in the glossary.

Acknowledgments

I would especially like to thank the ISTE Ltd team and my editor Chantal Ménascé, who trusted me.

Finally, I would like to thank my wife, Vanna, and my daughters Océane and Léa who supported me throughout the writing of this book.

August 2021

Introduction

As you may have discovered in Volume 1 of this book, voice recording and voice processing have been around for several decades. During the 20th century, the techniques and means of capture rapidly evolved from an environment that was analog and electromechanical, then electronic, to digital technologies which currently allow an almost systematic dematerialization of the transcription and archiving chain.

The computer and the appearance of digital audio processing stations, mixing editors and processing plugins have become indispensable, imposing new standards.

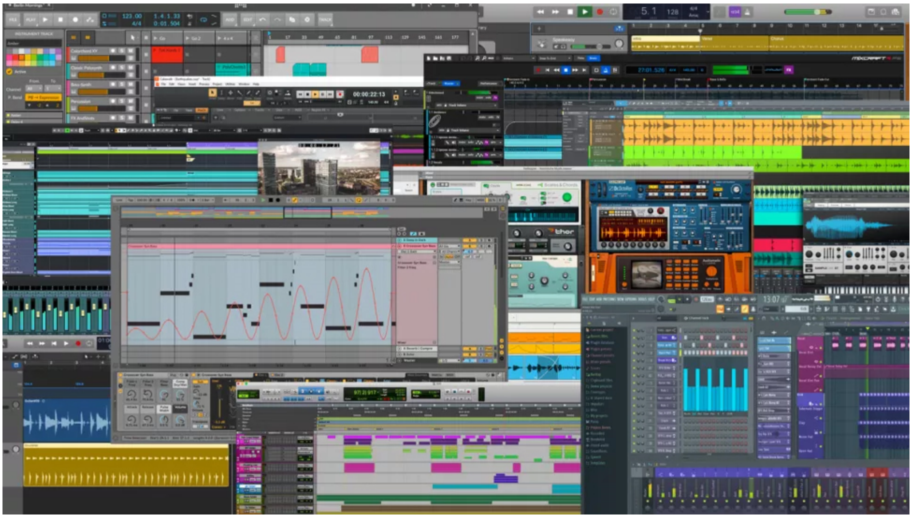

Figure I.1. *Some digital audio workstations (DAWs) – source: www.musicradar.com. For a color version of this figure, see www.iste.co.uk/reveillac/recording2.zip*

The predominance of the Internet and networks has shaken up the entire music industry, from sound recording to distribution to the final customer. The listener has changed their relationship with the broadcasting medium, seeing the disc, the tape, the walkman and soon the CD or even the MP3 disappear, having only a subscription to a streaming platform (Spotify, Deezer, Apple Music, YouTube Music, Amazon Music, etc.).

There are some nostalgists who still defend the vinyl record, which is making a comeback, but it is far from being the most widespread medium. The worldwide invasion of smartphones and the constant evolution of the speed and quality of networks has quickly changed habits, making obsolete many technologies that were considered innovative only a few years ago.

Figure I.2. *The first portable tape player (Sony TPS-L2 – 1979) and the MP3 player (Apple iPod – 2002)*

In this second volume, more oriented towards recording, voice mixing and studio work, I will start by describing the hardware and software tools that are effective in recording and include them in the audio chain that will support them.

To continue, I will present several examples of recording studio configurations, from the most modest to the most advanced.

In this context, I will approach sound recording, without neglecting the comfort of the performer and the management of their immediate environment, within which will be placed the various essential materials, respecting the rules necessary for a correct recording.

Figure I.3. *An example of a home studio – source: thomann.de. For a color version of this figure, see www.iste.co.uk/reveillac/recording2.zip*

To go further, I will study the constraints related to the recording of a group, a choir or a voice-over.

Once the recording has been made, it will be time to process it by implementing different processes during the mixing process that will ensure a correct sound rendering while respecting the style and coloring that you wish to impose.

To conclude, I will examine the sound effects or possible corrections that can be made to finalize your project.

I hope that the four chapters that make up this second volume will enrich your knowledge of vocal recording and enable you to make quality recordings that respect your personal feelings and style.

1

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 plugins[1], as can the mixing console, which can also be entirely virtual, in the form of a DAW (Digital Audio Workstation[2]).

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

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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.

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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.

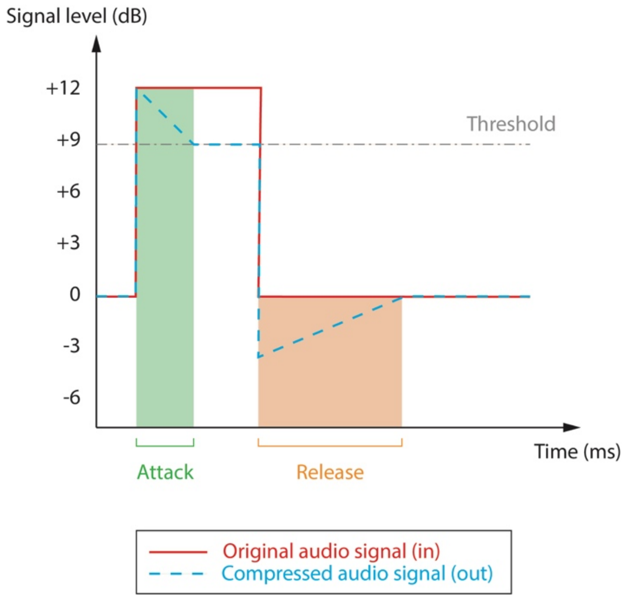

Figure 1.2. *Attack and release during compression*

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.

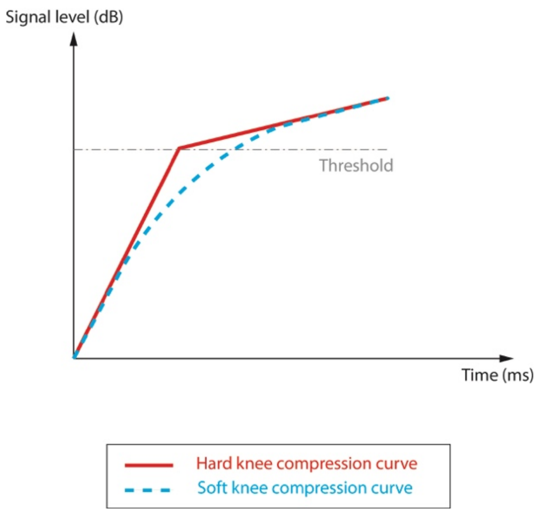

Figure 1.3. *The shape of the compression curve as a function of the knee setting*

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.

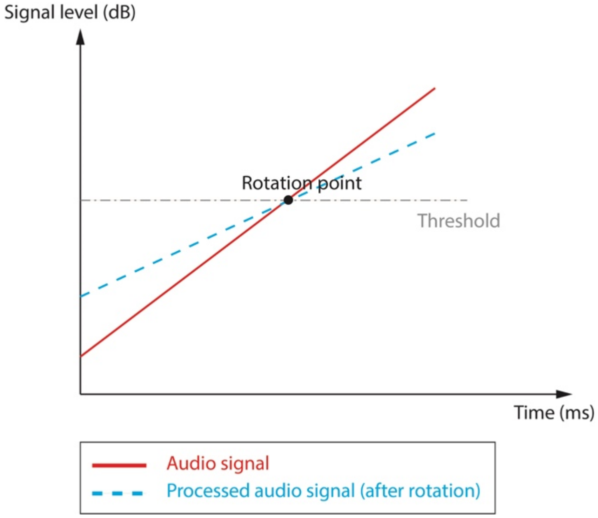

Figure 1.4. *The influence of the rotation point*

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.).