Marco Massimi

MITB
Mastering in the box

Concepts and applications for audio mastering

including exercises for Wavelab Pro 10

Marco Massimi

This is a demo version of

MITB MASTERING IN THE BOX

Concepts and applications for audio mastering
Theory and practice on Wavelab Pro 10

© Contemponet 2020

full version at:
www.contemponet.com

INDEX

Chapter 8
DISTORTION AND SATURATION • 311

FOREWORD
by Steve Lyon

I can't recall over my various years of being a recording engineer and producer how many times I've faced the same question, what is Mastering, and is it required? You would think the answers would be quite obvious and easy, however, as we all know that not really how the conversation ends.

Over the years, the changes are well documented and evident as to our listing habits, from good old Vinyl to streaming with new software in particular ready to assault our ears. The technical requirements have required new solutions to challenges still being fought between the various "platforms" and different mediums. However, the basic outcome across all these are the same, and we just want the music to sound good.

That might seem a little simple in its analysis, but it's so true. You don't need to be a technical wizard to enjoy music or to have an opinion on the Genre, it's content, or it's sound but to those of us battling to get music heard, develop artists and create something new, reliance on experience, our own or especially on others who work in other avenue's within music has always been essential.

I have great memories of Mastering in various studios around Europe, especially in the UK and NYC. It would take too long to recall them here, but I absorbed as such as possible from the wildly eccentric engineers to the fanatically technical, appreciating that these personalities had it all within their grasp to make sure the tracks I had brought in the door on tape or digital formats would sound great.

Those possibilities to learn and trust from those experiences are sadly limited now, but the desire to find out more about certain issues is still there, especially the mysteries of mastering. Digital requirements are evolving, Vinyl Mastering is making a reassurance, all good, but where can I find a reference to a vast array of questions and endless curiosity, technical or otherwise? Well, you have it in hand laid out in diagrams and formulas for the technical, discussions, and advice for the curious and a wealth of ideas and insight into what is now generally called "in the box."

Personal taste and feelings are as important if not better regarded than rules and tech jargon, but wouldn't it be nice to be assured that you can access all of these in one place.
Happy reading :)

Steve Lyon
Producer, Mixing Engineer (Depeche Mode, The Cure)

INTRODUCTION

This book deals with digital sound processing for audio mastering, entirely inside a computer, **ITB**, **In the Box**. ITB mode has been chosen for various reasons. The computing power of today's computers allows a high level of accuracy and calculation. These computers are now within anyone's reach; such a text may help the reader make things that only a few mastering studios, provided with expensive equipment, could have done before. Nowadays, a home studio can produce good masters. If we consider the parameters and take the precautions to avoid degrading the numeric signal, our master can compete, quality speaking, with any commercial product.

Do not be astonished if you find a copy of this book in a mastering studio, however. The purpose of this book is to analyze and experiment with the various elements of this post-production stage. The reader has the tools to analyze the sonic content and make the right choices for the signal processing within a **DAW Digital Audio Workstation**.

PREREQUISITES

This book alternates between theory and practice on a computer, which is supposed to be studied in close connection. Users with different levels of knowledge can read this text. The following is the minimum level required for those who start studying MITB: a basic proficiency with a computer (how to save, copy and erase a file); being comfortable with terms such as preamplifier, amplifier, mixer, speakers. Each theoretical section needs to be studied in conjunction with its corresponding practical section, including each computer exercise. You can study this tome either by yourself or under the guidance of a teacher.

LEARNING TIMES

Learning times are different for each person. Use the following as a reference in the two modes: self-learning and learning led by an expert teacher.

Self-learning (200-hour independent study)

Chapter	Subject	Total hours
1	Production and background history	6
2	Types of processing	8
3	Metering	10
4	Signal analysis	9
5	Signal filtering	20
6	Spatial processing	10
7	Dynamics processing	25
8	Distortion	10
9	Noise reduction	6
10	M/S mastering	14
11	Stem mastering	12
12	Surround mastering	15
13	Dither	9
14	Authoring	9
15	Red Book	14
16	Streaming	13

Learning with a lecturer
(40-hour course + 80-hour independent study)

Chapter	Subject	Lessons	Feedback	Study	Total hours
1	Production and background history	1	1	2	4
2	Types of processing	1	1	2	4
3	Metering	2	1	3	6
4	Signal analysis	2	1	3	6
5	Signal filtering	3	2	5	10
6	Spatial processing	3	2	4	9
7	Dynamics processing	6	2	8	16
8	Distortion	2	1	3	6
9	Noise reduction	1	1	2	4
10	M/S mastering	3	2	4	9
11	Stem mastering	2	1	3	6
12	Surround mastering	3	2	4	9
13	Dither	1	1	2	4
14	Authoring	3	2	4	9
15	Red Book	3	2	4	9
16	Streaming	3	2	4	9

AUDIO EXAMPLES AND INTERACTIVE EXAMPLES

This path includes interactive examples downloadable from the Virtual Sound website. By using such examples, you can experience the sound and sound processing. The studies are interrelated with how sound is perceived and processed. Making perception and knowledge interact is at the core of education, and the reader is provided with further online, upgraded, and widen material.

WAVELAB Pro 10

Steinberg WAVELAB 10 Pro or higher, one of the most advanced software for audio signal analysis and digital processing, is the reference software. The latest version includes the necessary tools for real-time and off-line processing, encoding codecs simulation, and premaster production. It is possible to download a free and complete 30-day trial copy at **www.steinberg.net**. The same theories and concepts we deal with throughout this book can be applied to other software or external hardware equipment within a mixer or within the mixer. Both a software and hardware version of a compressor or a filter have the same parameters.

TEACHING APPROACH

There are not many books on mastering (especially ITB mastering). For this reason, besides interactive examples, for each chapter, there is a learning agenda, listening and analysis tasks, tests, a glossary, and background history. The system, made up of this book and an online section, is multi-platform, and the theory section is built to allow practicing on other software, only using the same educational approach. Each chapter begins with a theory section that may include interactive examples. The book includes exercises on the software Wavelab. This book is not a Wavelab handbook, however. Please refer to the Wavelab user's manual for any non-listed software function.

Moreover, it is possible to download PDF documents, including further exercises from the Virtual Sound website. An icon on the page side indicates that exercises are available for that specific chapter.

Some subjects may include an insight section that can be identified thanks to a dark edge around the specific section.

To conclude, bear in mind that the subjects in this book are specific and relative to the analysis, concepts, and practice for audio mastering. To master notions, and basics to understand better and use this book, please read:

Electronic Music and Sound Design vol. 1 e 2 – Contemponet Editions

PRACTICAL INDICATIONS

This book contains interactive examples, presets, audio files, which are essential for learning. The material needed in MITB is available for download from the Virtual Sound website

Interactive Examples

While studying, before the practice, it is vital to use interactive examples. Using them helps to practice on each subject.

Example files
Use example files, and master section presets within the software Steinberg Wavelab 10 Pro (or later versions), which can be downloaded at **www.steinberg.net**.

Interchanging theory and practice
In each chapter, there is a theory section before the practice section. The terms in **bold** are listed in the glossary at the end of each chapter, accompanied by synthetic descriptions. The <u>underlined</u> terms correspond to the software menu items. The path to download the support material is indicated as follows: e.g., **\MITB\Examples\Chapter 08\07F.wav.** Each **** represents a folder.

Bibliography and sitography
The author includes only an essential bibliography and reference texts, relative to the texts quoted in the book. A more comprehensive bibliography and sitography can be found on the website.

Before you start
To start working with this book, download the Interactive Examples from the Virtual Sound website. Reference is always made to the examples in the book.

Comments and reports
Comments and reports are welcome. If any, please send an e-mail to the author at: mitb.massimi@gmail.com

Audio examples
Marco Massimi, Vincenzo Mario Cristi, Anuhel, Gianluigi Farina, Lingala (Azeglio Izzizzari, Massimo Izzizzari, Rocco Zifarelli, Francesco Carlesi, Filiberto Palermini, Marco Massimi, Giovanni Imparato)

Graphics and original design
Marco Massimi

ACKNOWLEDGMENT

LEGEND OF SYMBOLS

• **EXERCISES AND INTERACTIVE EXAMPLES**

• **SUPPLEMENTARY EXERCISES AVAILABLE ONLINE**

PDF

• **UNIT ASSIGNMENTS**

• **TEST**

1

PRODUCTION, BACKGROUND HISTORY AND DEFINITION OF MASTERING

LEARNING AGENDA

PREREQUISITES FOR THIS CHAPTER
- Basic knowledge of computer tools
  (Basic operations, managing folders, audio interface, etc.)

TARGETS
What you will learn
- Learning history of recording and used supports
- Learning each step of music production
- Learning needs and reasons why mastering exists

Skills
- Being able to identify the loudest song by listening

CONTENTS
- of the music production process
- Tracking modes
- Mixing modes
- Production evolution process
- Loudness war

LEARNING TIMES – Ch.1
Self-learning
For a 200-hour independent study session: about 6 hours

Courses
For a 40-hour classroom course + a 80-hour independent study session:
- about 1-hour lecture + 1-hour feedback
- about 2-hour independent study

EXERCISES
- Interactive examples

TESTING
- Listening and analysis tasks
- Short-answer questions

SUPPORTING MATERIALS
- Glossary

1.1 MUSIC PRODUCTION CHAIN

The music we listen to, come from different sound sources. These may be analog, like a vinyl disc or a magnetic tape; digital, like a Compact Disc or a linear-format or compressed file. These can be radio or TV broadcast programs, or the score of a movie, or web-streamed material, or a smartphone or a tablet. As we listen, we do not think about the production processes done to achieve the final product. It is right that we care about the artistic, emotional, and sonic content of the piece. Artistic value is crucial and generates emotions and sensations that can deeply involve the listener. Those emotions and sounds are the results of a multi-step production chain, capable of affecting the quality of the final product.
The quality of audio production is born before the recording process. The artist conceives it, it comes from the performer's and the arranger's skills, and it is made up of hardly-measurable-and-analyzable processes.

However, even the most inspired artist, the most excellent performer, and the most sophisticated composer may not be appreciated and considered in the right and the proper way if the technical quality of the product is weak. We must take the main elements of an audio production process into account.

Let us consider the audio production process into three main stages:

TRACKING, MIXING, MASTERING.

Tracking
During the tracking stage, we transfer the sonic material coming from microphones or signal generators on physical support. Such support can be the magnetic tape of a reel-to-reel multitrack tape recorder, see fig.1.1.
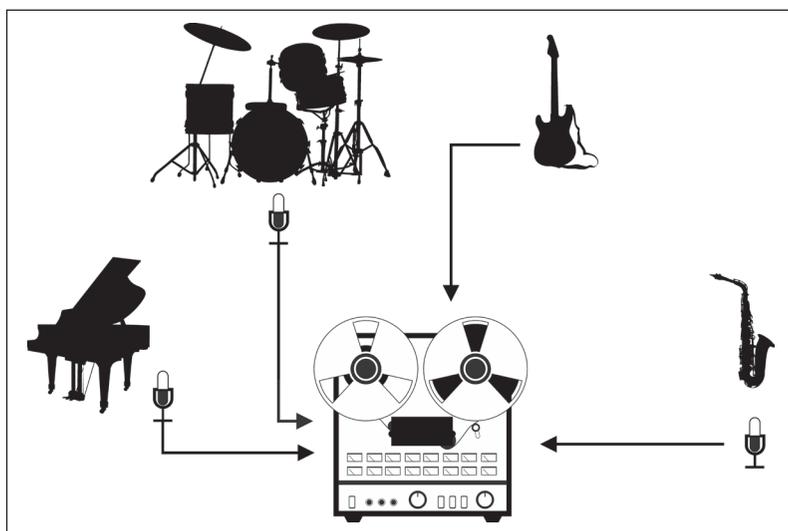
fig.1.1 Tracking on magnetic tape

Currently, the most used support is a computer hard disk, which is part of a system made up of both a specific audio software and an audio interface. Such a system is called a **DAW (Digital Audio Workstation)**. Alternatively, a dedicated hardware audio workstation can as well be used. First, analog signals are converted into digital signals thanks to an audio interface, see fig.1.2.
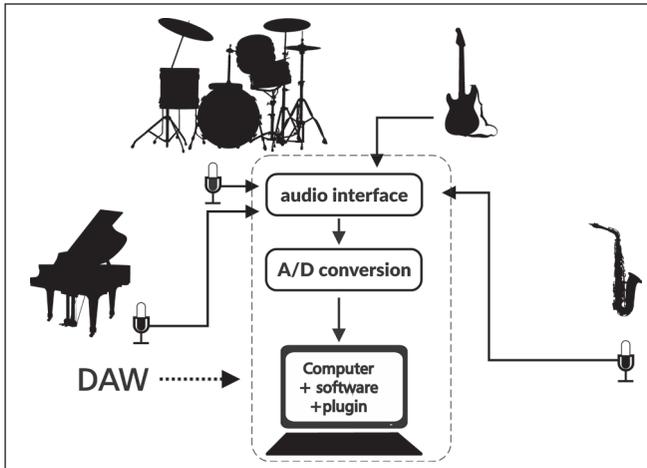
fig.1.2 Tracking on a DAW

Tracking is the recording stage done in a studio or a room of a house designated as a home recording studio. During the recording stage, a high number of audio tracks, each corresponding to a sound source, can be generated. These sound sources can be:
- Acoustic sources, captured with a microphone
- Electromagnetic sources, coming from pickups and transducers for electric or electrified guitars and strings.
- Electronic sources such as synthesizers
- Virtual sources like generators, processors and software-based sound samplers

Thanks to modern DAWs, the number of recordable tracks is virtually infinite. The number of audio tracks that can be recorded on a single analog multitrack recorder goes up to a maximum of 24. Only a few recording studios have more than one recorder to bring the number of the tracks up to 48 or, more rarely, 72 or even 96.

Mixing
During the mixing stage, the tracks are processed. These kinds of processing may be spectral by using filters, equalizers, and spectral-type noise reduction systems. They may be dynamic processing, by using level controls and dynamics processors such as compressors, expanders, limiters, gates, and dynamic-type noise reduction systems. They may be spatial processing by using reverbs, delays, and panoramic potentiometers or joysticks. As you mix, you edit tracks. There are three modes to accomplish a mix.

OTB mode (out of the box)
Signal coming from the magnetic tape is eventually mastered with outboard analog processors and printed on vinyl or Compact Disc. Each signal coming from a multitrack tape recorder is sent to the inputs of an analog mixer, which is, in turn, connected to the external outboard. Tracks and effects are mixed in an analog mixer. The mixer output is sent to stereo or multitrack output, which is sent to a master recorder, or a DAW to produce the final mix.

By doing so, the signal path stays electric until the end of the production (e.g., various sound sources, linked to an analog mixer). These signals are acquired thanks to a multitrack tape recorder on a magnetic tape. Each track coming out of the recorder is mixed within an analog mixer. The mixer output is then recorded on a magnetic master recorder. The signal coming from the magnetic tape is eventually mastered with outboard analog processors and printed on vinyl or Compact Disc. From a mathematical point of view, each conversion from analog to digital and vice versa introduces sampling and quantization errors, which damages the signal quality. This kind of technology allows using a single **A/D (analog-to-digital)** conversion (or even no conversion if the production is on vinyl) and minimizes signal degradation, see fig. 1.3.
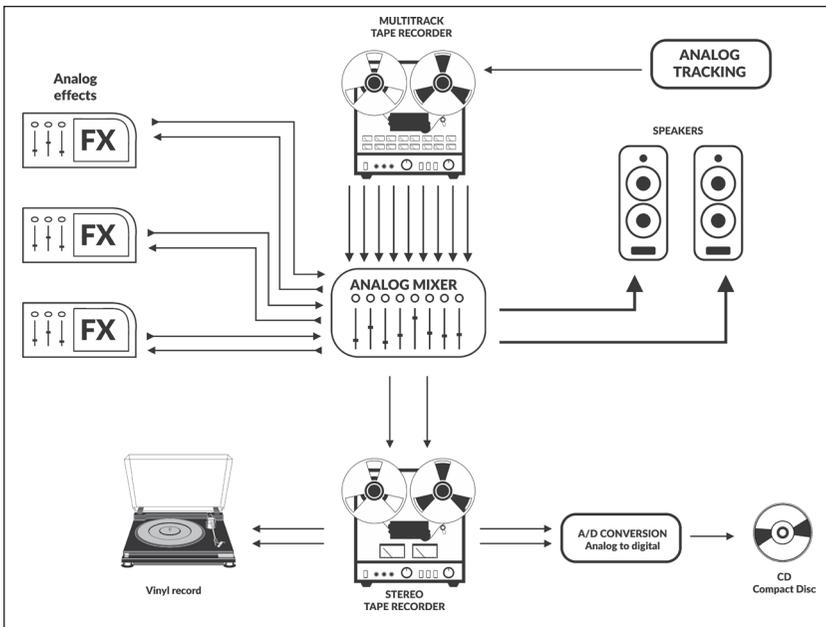
fig.1.3 OTB Mixing

ITB mode (in the box)
The signal is sampled in a DAW, then is processed within the DAW and the audio production software. We may use software plugins that emulate external signal processors and eventually produce the final mix file. Even with this technique, there is just a single A/D conversion by converting the electric signal coming from the various sound sources into numbers, thanks to an audio interface. There is a D/A conversion (digital-to-analog) used for the playback system. This one does not affect the quality of the final product, which is generated in a digital format with no need to be furthermore converted. In such a case, the mixing and mastering stage is done entirely in the digital domain. ITB includes the other kinds of processing, entirely made into the numeric domain, e.g., when we use digital mixers and digital hardware effects. There is no conversion after the tracking process. Using the **floating-point** arithmetic for processing allows achieving a computing accuracy and a nearly infinite sonic resolution, without regretting the old-fashioned analog/electric signal path. We are dealing with this subject from a mathematical point of view, ignoring the questions about whether digital is better than analog and vice versa, see fig. 1.4.
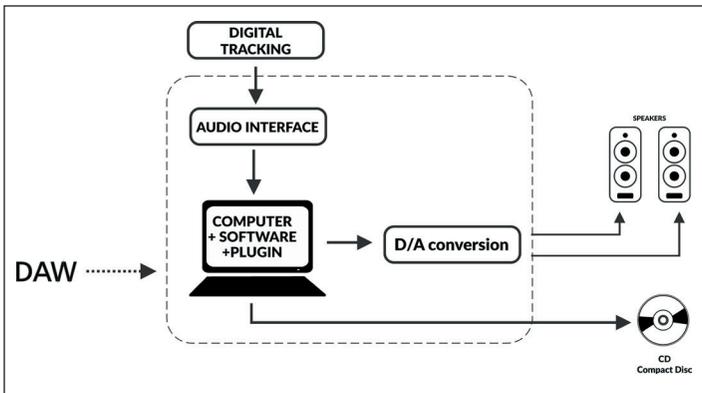
fig. 1.4 ITB Mixing

Hybrid mode
A lot of producers and recording studios go for hybrid mode. Tracks are recorded on a DAW and then sent to an analog mixer and other outboard processors. The output of the mixer and processors are sent back to the DAW to record the final mix. Consider the previous examples, with such a signal path, conversions increase in number, and so make sample and quantization errors.

As you can see in fig. 1.5, we are supposed to:

- Connect the sound sources to the analog mixer and the mixer's output to the DAW for tracking purposes: **first conversion, A/D**
- Send the signal, which has been previously recorded into the DAW, to the analog mixer for mixing purposes: second conversion, D/A
- Send the mixer's and outboards outputs to the DAW to create the final mix: **third conversion, A/D**
- Consider the A/D and D/A conversions needed to connect digital hardware effects, such as delay and reverb units
- Deliver the mix file to an analog mastering studio: **fourth conversion, D/A**
- Eventually print the CD: **fifth conversion, A/D**
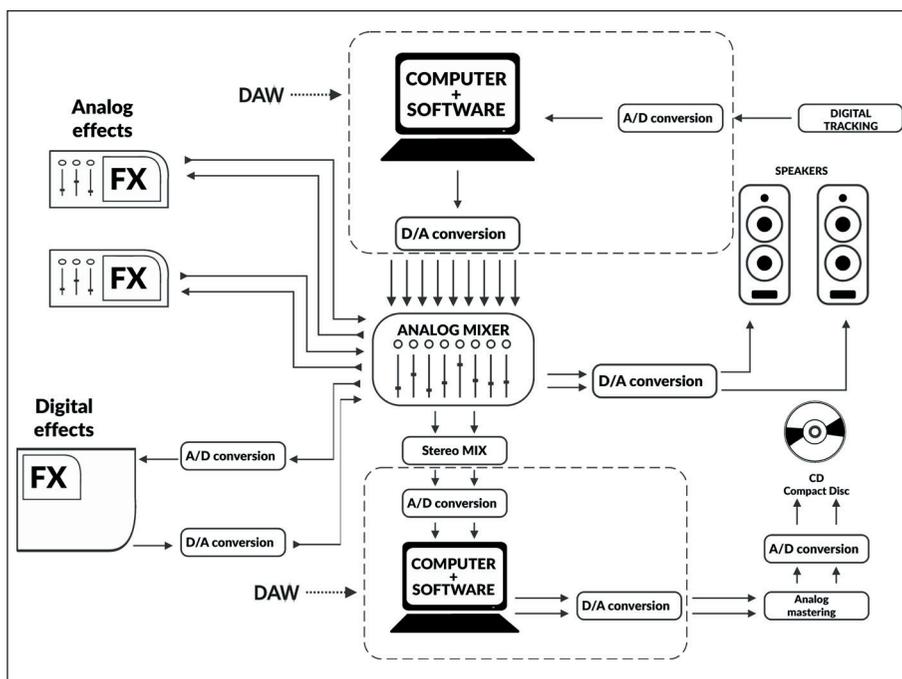
fig. 1.5 Hybrid mixing

From a mathematical point of view, five conversions may highly compromise signal quality.
Sampling a signal compromises its integrity. Whichever the sample rate and bit depth, the values are approximated during this process. The term *sample* means an *example*; indeed, an *example* of the original signal. Do not be surprised if some ITB productions have such quality and sound, which is comparable or even better than sought-after and expensive *hybrid* productions.

To conclude, we may claim that *respect* for signal integrity must be considered. If you are interested in digital production, try and keep the signal in a numeric form as much as possible; if you want an analog production, keep it in its electric form.

By the end of this process, the tracks are mixed into variable-number outputs. Two outputs, if the final product is stereo, six outputs, or more if the final product is a surround-format file like in Movies and Videogames. Production is finalized, and the final mix is complete, although still not ready for commercial distribution.

Mastering
It is the last element in the production chain. It is a post-production stage, and it includes various processes accomplished to bring the mix into technical and sonic standard compliance, as requested in the music business. In such a case, spectral, dynamic, numeric, and spatial processes are accomplished, but consider that manipulating sound in this stage affects the whole mix.
During the mixing stage, we may be highly creative and experiment with extreme variations of each signal processor parameter for every single track. Mastering is not supposed to be like that. Once the mix is done, each sound and instrument can no longer be separated to be processed individually. Each choice made from now on affects the whole song in regards to its level, timbre, ambiance, and signal content.

(...)

other sections in this chapter:

1.2 Direct to disc on wax and Bakelite discs
1.3 Vinyl 1
1.4 Transfer
1.5 Mastering
1.6 Why master a song?
> **Listening consistency**
> **Exercises**
> **Playback system adjustment**
> **Fixing errors and editing**
> **Create a tracklist**
> **Inserting PQ codes and metadata**
1.7 Playback system

Test
Glossary

2
TYPES OF PROCESSING

LEARNING AGENDA

PREREQUISITES FOR THIS CHAPTER
- Contents of chapter 1

TARGETS
What you will learn
- Learning various types of processing that can be applied on an audio signal
Skills
- Being able to identify the type of processing by listening to the signal

CONTENTS
- Graphs and domain
- Basic concepts on dynamic range
- Amplitude
- Frequency
- Timbre
- Envelope

LEARNING TIMES - CH. 2
Self-learning
For a 200-hour independent study session:
- About 8 hours
courses
For a 40-hour classroom course + a 80-hour independent study session:
- About 1-hour lecture + 1-hour feedback
- About 2-hour independent study

ACTIVITY
- Interactive examples

TESTING
- Listening and analysis tasks
- Short answer questions

SUPPORTING MATERIALS
- Glossary

2.1 SPECTRAL PROCESSING

When we talk about the spectrum of a signal, we mean its frequency content, and it is often depicted on a graph within the frequency domain, see fig. 2.1. The domain of a function is the value on the x-axis. The spectrum of a signal is depicted as a graph. The amplitude on the y-axis and frequency on the x-axis.
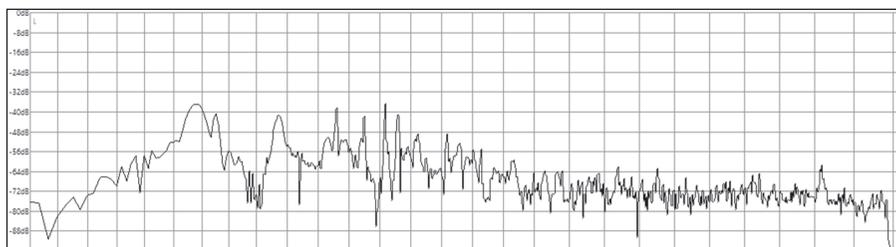
fig. 2.1 Frequency domain graph

The sound processing within the frequency domain affects the sound character and timbre. It can make it darker, brighter, harsh. It can fix the flaws of a non-linear signal (see DC offset, chapter 5). In mastering, spectral processing achieves a homogeneous album sound and fix timbral flaws resulting from an excessive high/low-frequency content.
The processing uses various kinds of filters: high-pass, low-pass, notch, shelving; also, a set of filters is used, like parametric, semi-parametric, graphic equalizers (see chapter 5). Each processing affects the whole song. It is better to use *subtractive* equalization rather than *additive*. If the timbre of the song seems dark, we are prone to amplify the high-frequency content, whereas it is better to analyze the spectrum to identify the excessive low-frequency content and lower it. It allows us to minimize the phase variations around the cutoff frequency without amplifying the filter noise. In this stage, we must try to reduce what is too much.

(...)

other sections in this chapter:

2.2 Dynamics processing
2.3 Time-based dynamics processing
2.4 Spatial processing
2.5 Noise reduction
      Exercises

Test
Glossary

3
METERING

LEARNING AGENDA

PREREQUISITES FOR THIS CHAPTER
- Contents of chapters 1-2

TARGETS
What you will learn
- Learning different types of db, unit of measurements, reference levels, operative levels
- Learning different types of meters
- Learning measurement scales

Skills
- Being able to use and set up the standard meter in wavelab
- Being able to use and set up the r128 meter in wavelab

CONTENTS
- Decibel
- Ppm and qppm meters
- Integration times
- Vu meter
- The dbfs scale
- Ebu r128 measurement system

LEARNING TIMES - Ch. 3
Self-learning
For a 200-hour independent study session:
- about. 10 hours

Courses
For a 40-hour classroom course + a 80-hour independent study session:
- about 2-hour lectures + 1-hour feedback
- about 3-hour independent study

EXERCISES
- Interactive examples

TESTING
- Analysis and listening tasks
- Short-answer questions

SUPPORTING MATERIALS
- Glossary

Preface

When sorting a problem out, we may apply various workarounds, computing modes and find practical solutions. We cannot overlook some crucial stages that are key to solving a problem like analysis and data acquirement. This section of the book covers tools and units of measurements used to accomplish an accurate and complete analysis of the audio material. Thanks to the analysis, we can get those data. We can evaluate what kind and how much processing to make. The mathematical formulas we may encounter throughout the reading may discourage. Such formulas are easy to understand, and they are thoroughly explained. It is asked for is a little bit of a sacrifice, to understand the concepts and procedures of the audio mastering. In order to do this, we need specific measuring tools, i.e., **meters** capable of displaying instant-by-instant, or average, or integrated values. The meters used in production differ from those used in mastering. During the tracking and mixing stage is essential to measure the input signal level swiftly and accurately, as to avoid distortions of any kind in the signal.

In a mastering studio, although instantaneous levels are regularly taken care of, we accurately measure the average value of the signal too, which corresponds to loudness, i.e., the perceived volume.

The most used unit of measurement to measure pressure, power, and intensity values is the **decibel**, **dB**. It is a tenth of a Bel, a no longer used unit, which can be called a unit of *representation* rather than a unit of measurement. Values in dB are relative to various units of measurement and physical quantities. Such quantities and units may be: pascal, volt, watt, depending on the type of dB. dB can represent data in a logarithmic way.

A **logarithm** is a **number (power) to which you raise another number (base) in order to get a result**.

The base-10 logarithm of 100 is 2.

The mathematical expression is $\log_{10} 100 = 2$.

As to obtain 100, we must raise 10 to the power of 2. $100 = 10^2$

When we talk about dB, the logarithm base is ten, and by **log**, we mean **log$_{10}$**. Thus, the progressive increase in values is not linear, and that is the reason why we use dB. It allows us to depict significant numeric variations of pressure, voltage, power, with few digits, see fig. 3.1. We express dB with the ratio between a quantity to measure and a reference one
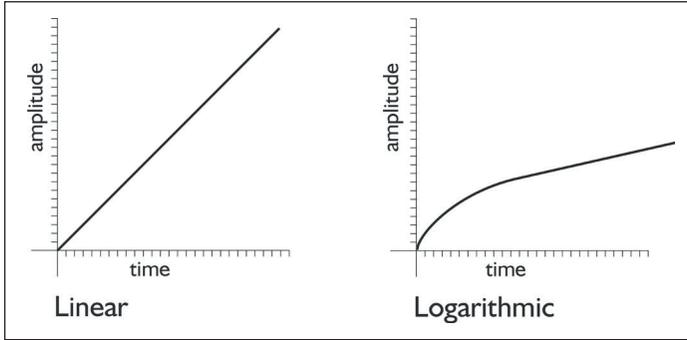
fig. 3.1 Logarithmic scale and linear scale

In the previous example, two curves that describe the amplitude in the time domain. The values on the two axes may vary. They can represent the output and input level of a circuit, the level of both the left channel and right channel of a stereo program, or the amplitude in the frequency domain.

3.1 dBSPL SOUND PRESSURE LEVEL

The **dBSPL** measures the sound pressure level. It is a quantity that expresses the sound pressure level compared to the threshold of hearing. **SPL meter** is the tool used to measure sound pressure, see fig. 3.2.

fig. 3.2 SPL meter

It contains a microphone and a circuit that measures the electrical values in order to apply various values **weighting curves**, see fig. 3.3. These curves filter the signal both in the low and the high-frequency content. The most used curve is A-type.
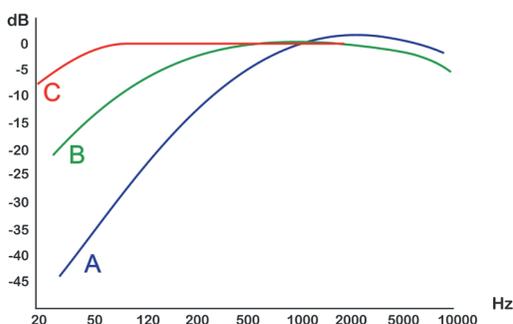
fig. 3.3 Measurement weightings

The dBSPL measures the maximum pressure value that the microphone or the human ear can tolerate. It measures the sound pressure generated by a loudspeaker at a specific power at a certain distance (sensitivity). Each doubling in pressure level corresponds to an increase by 6dBSPL[5]. The double of 40dBSPL is 46, the double of 46 is 52, and so on. dB cannot represent absolute values if not associated with a unit of measurement and a reference quantity. If we read 6dB, we can claim that there is a value that is twice as another value, but we cannot tell the actual value.

---

[5] SPL values measure the pressure on a surface and refer to the human threshold of hearing. When we measure energy, the absolute intensity, we are supposed to use SIL, Sound Intensity Level. SIL values double every 3dB. Lots of SPL meteres can measure both SPL and SIL. In order to measure the sound intensity produced by a loudspeaker we use the SIL, which corresponds to 10^{-12} watts on a square meter, 10^{-12} W/m^2.

IN DEPTH

The unit of measurement for dBSPL is the **pascal (Pa)**, which is the applied force on a surface. 1 pascal is a force of 1 Newton applied on a 1 square meter surface:

$$Pa = \frac{N}{m^2}$$

The reference quantity of the dBSPL is **20µPa**, and it represents the human threshold of hearing, i.e., the weakest sound one can hear in good hearing condition. The dBSPL is 20 times the base-10 logarithm of the ratio between the pressure to measure and the reference one:

$$dBSPL = 20 \log_{10} \frac{px}{pr}$$

px being the pressure to measure and *pr* being the reference pressure of 20µPa.

We use logarithm because we must represent non-linear values. The multiplier of 20, i.e., 10 x 2, is used because dB is a tenth of a Bel and since these are squares resulting from the surfaces, we may write the formula as follows:

$$dBSPL = 10 \log_{10} \frac{px^2}{pr^2}$$

though we can simplify as

$$dBSPL = 20 \log_{10} \frac{px}{pr}$$

If we measure a pressure of 40µPa, twice as the reference pressure, the expression is

$$dBSPL = 20 \log_{10} \frac{40}{20} = 20 \log_{10} 2 = 6dBSPL$$

To measure the pressure of 80µPa in dBSPL, four times as the reference pressure, the expression is

$$dBSPL = 20 \log_{10} \frac{80}{20} = 20 \log_{10} 4 = 12dBSPL$$

Consider that the reference pressure is of 20µPa, the minimum pressure variation one can hear.

The values to measure in dBSPL are higher than that one, and the results obtained by this logarithmic calculation are positive on a scale that generally goes from 0 up to 194dBSPL, see fig. 3.4.
Below a simplified table listing the sound pressure values in dBSPL

Source	Pressure	dBSPL
Physical limitation	101.325 Pa	194dB
Instant injuries	50.000 Pa	185dB
Jet 30 meters away	630 Pa	150dB
Threshold of pain	100 Pa	130dB
Disco	2 Pa	100dB
Traffic 10 meters away	0,6 Pa	90dB
Office	0,02 Pa	60dB
Steady-tone human voice 1 meter away	0,002 Pa	40dB
Quiet room	0,0006 Pa	30dB
Breathing	0,00006 Pa	10dB
Threshold of hearing	0,00002 Pa	0dB

fig. 3.4 Pressure values in dBSPL

(...)

other sections in this chapter:

4
SIGNAL ANALYSIS

LEARNING AGENDA

PREREQUISITES FOR THIS CHAPTER
- Contents of chapters 1-3

TARGETS
What you wil learn
- Learning various types of supports for material acquirement
- Learning the basic concepts of sampling
- Learning the analysis tools in wavelab

Skills
- organizing material to process withing a daw
- Analyzing material developing critical and comprehension skills of the various measurement parameters

CONTENTS
- File and folders structure
- Sample rate
- Bit depth
- Floating-point arithmetic
- Dc offset
- Level, volume, dynamic range
- Spectral analysis

LEARNING TIMES - Ch. 4
Self-learning
For a 200-hour independent study session:
- About 9 hours

Courses
For a 40-hour classroom course + a 80-hour independent study session
- About 2-hour lectures + 1-hour feedback
- About 3-hour independent study

EXERCISES
- Interactive examples

TESTING
- Short-answer questions

SUPPORTING MATERIALS
- Glossary

4.1 ACQUIRING FROM ANALOG SUPPORT

An ITB mastering studio can be provided with analog support material to acquire on the hard disk. It may be a magnetic tape or vinyl. In this stage, it is essential to use at least a 24-bit resolution to acquire material. Later, we will deal with this subject more in-depth (see section 4.4).

In the Wavelab recording window, there are many options for automatic indexing or automatic start and stop. These functions are useful as we acquire more tracks from the same support. Vinyl or magnetic tape is a classic example. In order to automatically assign a marker at the beginning of each song, we can use the specific function, see fig. 4.1.
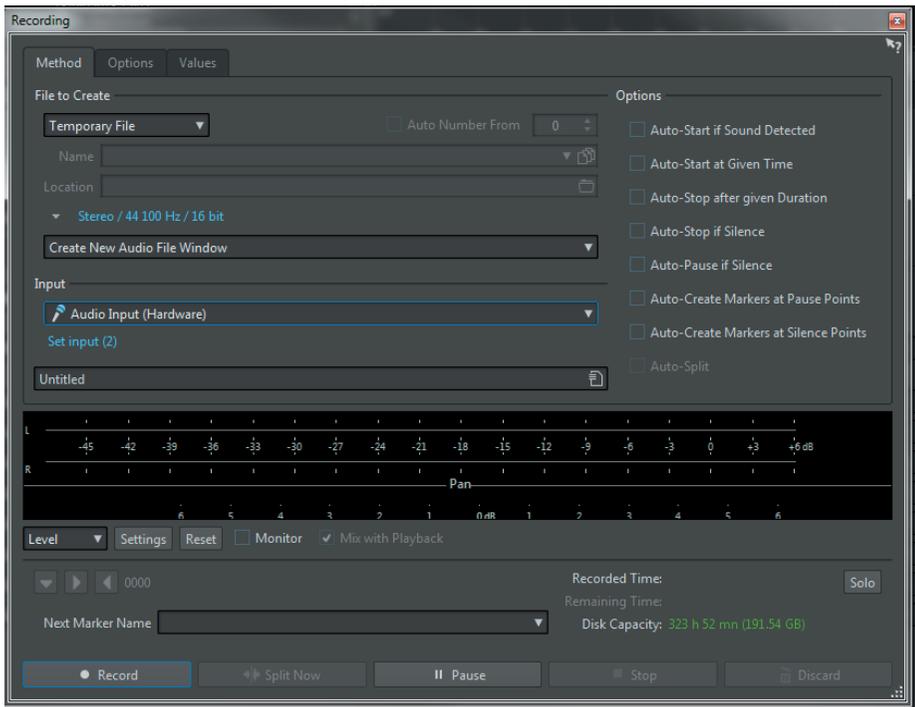
fig. 4.1 Recording in Wavelab

In the Values settings window, we can set a *pre-record* time, thanks to the Record Previous Samples function. If we set up 2 seconds, the recording includes the two seconds before pushing the Record key. This function is handy when acquiring vinyl in that it is difficult to identify the actual beginning of a song, see fig. 4.2.
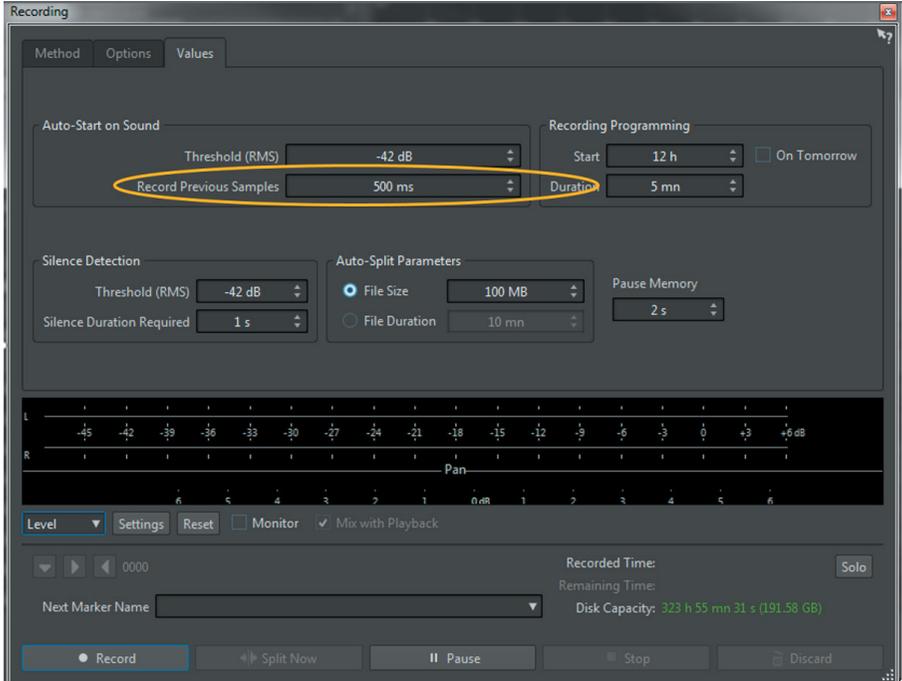
fig. 4.2 Pre-record option

EXERCISES

- For this exercise, we need an analog audio source. This can be a turntable plugged into an amplifier with a phono input (see section 14.3), a multimedia player, a television receiver, or a CD player, provided that they have an analog audio output.

- Press the Record key on the transport bar, see fig. 4.3.
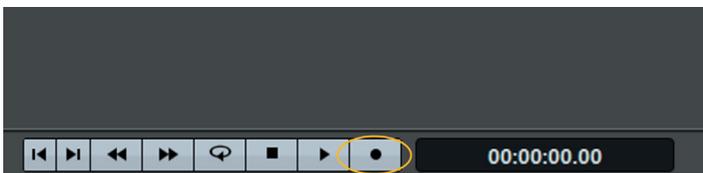
fig. 4.3 Record key

- The recording menu pops up, see fig. 4.1.

- The recording window meter is customizable in the same way the main meter is. It can display peaks, RMS values and can be used as a K-meter too. By pressing Settings, we can enter the relative customizing window, see fig. 4.4.
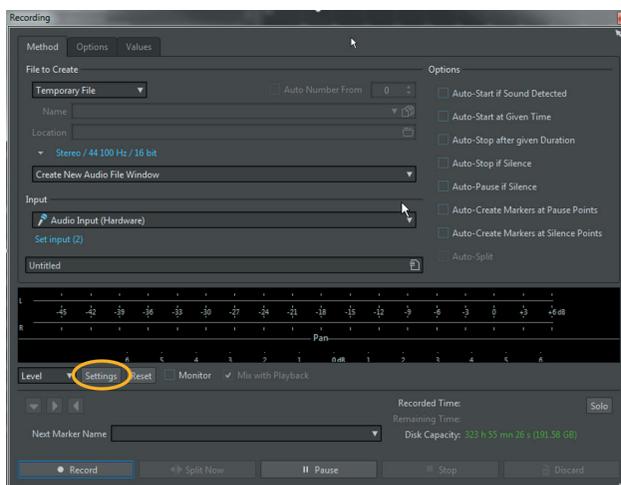
fig. 4.4 Recording meter settings

- In the recording window, we can choose the file format. As we will see later, it is better to set a sample rate of 44.1kHz or multiples for CD production and a sample rate of 48kHz or multiples for video production. Bit depth during recording must be at least 24 bits. Recording an analog source at 32 bits, from a 24-bit source, means wasting the hard disk storage. The bits from 25 up to 32 indeed have a value of zero.

- On the bottom right, disk storage relative to the used sample rate and bit depth is displayed.

- In the File to Create section, it is possible to choose the type of document. Temporary, to create a temporary file that can be named and saved later, or Named File, to create a final file with the name setup in the relative window.

- By enabling the Monitor option, it is possible to listen to the signal as we record.

- Plug the analog device output into the audio interface input and start recording.

- Adjust the audio interface gain so that the recording meter reads a maximum value of -18dBFS. In this stage, the external PPM meter is the reference. Accurate measurement of the signal instantaneous values is possible, avoiding distortion.

(...)

other sections in this chapter:

5

SIGNAL FILTERING, DC OFFSET REMOVAL AND TIMBRAL CHANGES

LEARNING AGENDA

PREREQUISITES FOR THIS CHAPTER
- Contents of chapters 1-4

TARGETS
What you will learn
- Learning types of filters
- Learning the basic concepts of filtering
- Learning various types of equalizers

Skills
- Using each individual filter and equalizer in spectral processing intended for mastering
- Analyzing material and develop critical skills for targeting frequencies to process
- Understanding the structure and functionalities of the master section

CONTENTS
- Types of filters
- Types of equalizers
- DC offset
- Types of equalization

LEARNING TIMES - Ch. 5
Self-learning
For a 200-hour independent study session:
- About 20 hours

Courses
For a 40-hour classroom course + a 80-hour independent study session:
- About 3-hour lectures + 1-hour feedback
- About 5-hour independent study

EXERCISES
- Interactive examples

TESTING
- Short-answer questions

SUPPORTING MATERIALS
- Glossary

5.1 DC OFFSET CAUSES

As stated above in the previous chapter, the presence of DC offset in a signal can create either acoustic or psychoacoustic or electronic issues. In order to remove the DC offset, we can use both dynamics and spectral processing. Let us identify its causes and type.

DC offset can have multiple causes, such as:

- Asymmetrical power supply circuits. The reference voltage of 0V is not 0; there is not an equal potential difference between positive and negative voltages.
- A/D converters with poor bit resolution. The measurement circuit of the input values does not provide a perfectly symmetrical detection between the positive half-wave and the negative one, since the available values are powers of 2, hence even numbers.
- Inefficient plugins algorithms. Internal calculations made through poorly efficient and precise programming.
- AM synthesis. When using amplitude modulation synthesis, a DC offset voltage is introduced. Generally, this voltage is not part of the generated sound. However, errors may occur when programming synthesis algorithms, eventually introducing DC offset in the audio signal.

5.2 REMOVAL THROUGH AN INVERTED VOLTAGE

The Wavelab off-line analysis tool allows measuring the amount of DC offset in the audio signal. The resulting value represents the average level of DC offset within the whole song. Often, we take for granted that the DC offset level is constant within the file from start to end. If this is the case, we can remove the DC offset by applying a voltage, which is the inverse of the detected one. If the detected value is +1% (i.e., +0.007V), we can sum the signal to a voltage of -0.007V, bringing its value back to 0.

In Wavelab, this kind of processing is made on the numeric values thanks to the function Remove DC offset, see fig. 5.1.
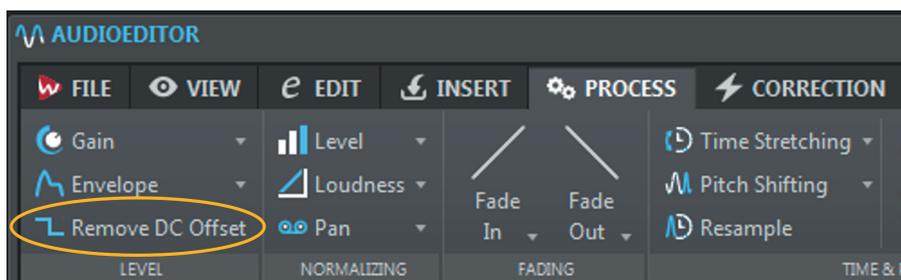
fig.5.1 DC offset removal, by applying an inverted voltage within Wavelab

5.3 REMOVAL THROUGH FILTERING

Regardless of its cause, the DC offset may be inconstant within the whole signal. When mixing a song, the DC offset may change in amplitude within the tracks. If the volume of these tracks is automated, i.e., their levels are changed during the mix, DC offset level changes over time. The off-line analysis measures the DC offset average level. If we apply an inverted voltage, based on the average detected value, we fix DC offset in some parts of the song. Furthermore, other DC offset (opposite in value) occurs in those parts where there is no DC offset. In such a case, although DC offset average value is 0, there are some parts of the file where this is still present. Fig 5.2 shows the original file on the left and the *fixed* one on the right. By doing so, we created other DC offset (opposite in value), rather than completely removing it.
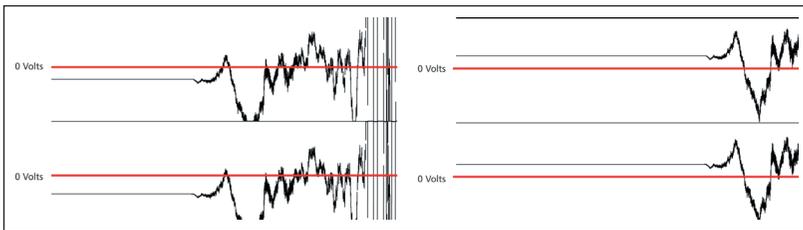
fig. 5.2 Error in DC offset removal through inverted voltage

In order to remove a variable-level DC offset, it is not advisable to apply dynamics processing. We may think of an alternative solution that allows filtering out the direct current within the signal. From a spectral point of view, the direct current has a frequency of 0Hz. Its voltage is constant and does not oscillate between positive and negative values. That being said, since the useful audio spectrum goes from 20 up to 20.000Hz, we can use a high-pass filter with a cutoff frequency of 20Hz to remove DC offset, regardless of its level, see fig. 5.3.
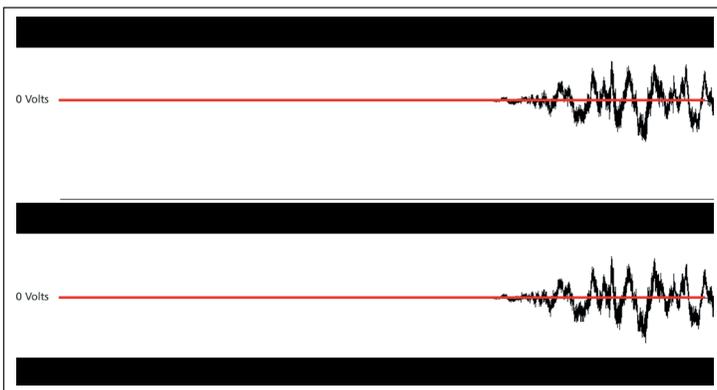
fig. 5.3 DC offset removal through filtering

EXERCISES

- Load up the example **05A**. Loop it, enabling the corresponding function in the transport bar, see fig. 5.4.

fig. 5.4 Loop playback

· ·

INTERACTIVE EXAMPLE 05A • 0 % DC OFFSET

· ·

- Make a global analysis of the file, see fig. 3.33 and look at the Extra section, see fig. 5.5.
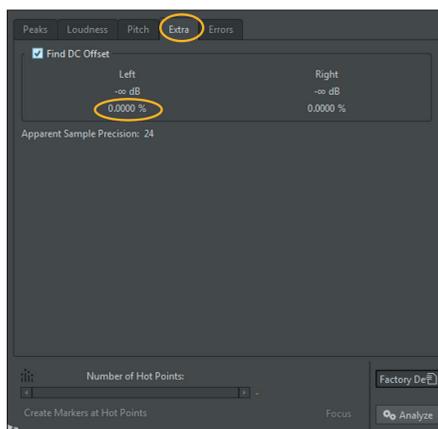
fig. 5.5 No DC offset detected

- Notice the value of 0,0000%. In this file, there is no DC offset and the reference zero of the waveform matches the digital zero. The symbol -∞ indicates the DC offset value, i.e., infinitely low. That does not mean that the example **05A** has an infinite dynamic range, but that, since there is no DC offset, we can have a high dynamic range, i.e., the maximum available in the system.

- Although a DC offset value of 0 is ideal, it is not always possible to retain such a low level of deviation from the zero. An acceptable DC offset value must be proportional to the dynamic range of the final medium. In the case of an audio CD, it has to be at least 96dB.

- Load up and analyze the example **05B**.

· ·

INTERACTIVE EXAMPLE 05B • 0,0016 % DC OFFSET

· ·

- The DC offset is 0,0016%. Such a value indicates a maximum dynamic range of 95.99dB, which is suitable for the audio CD medium. A value next to the second zero of the decimal number is fine, see fig. 5.6.
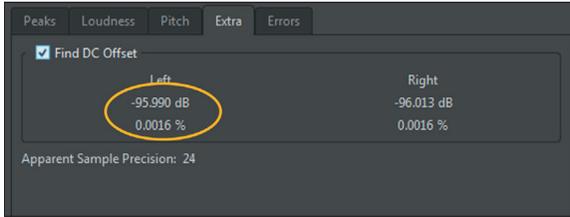
fig. 5.6 0,0016 % DC offset

- Load up and analyze the example **05C**.

· ·

INTERACTIVE EXAMPLE 05C • 0,15 % DC OFFSET

· ·

- The DC offset level is 0,16 %; the available dynamic range is about 56dB, see fig. 5.7.
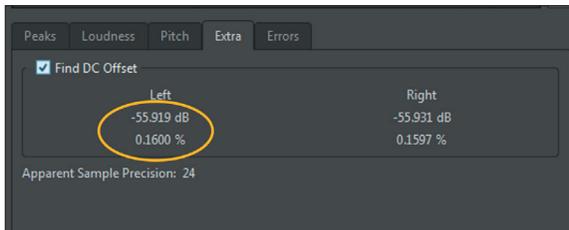
fig. 5.7 0,15% DC offset

- With such a DC offset value, the dynamic range of the final product cannot exceed 56dB, regardless of the quality of the hardware.
- We must process the signal to remove the direct current.
- Do this on a signal with a high level of DC offset.
- Load up the example **05D**.

INTERACTIVE EXAMPLE 05D • 50 % DC OFFSET

· ·

- Look at the whole waveform, see fig. 5.8.
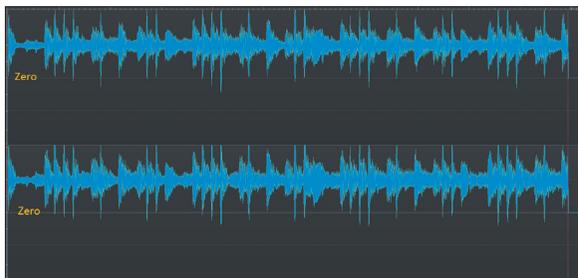
fig. 5.8 Positive 50% DC offset

- The signal shifts to the top compared to the digital zero. It is practically impossible to receive a file to master like this. If we exaggerate these values, we can understand how to fix these issues, however.

- Look at the values within the global analysis windows. The lowest value of the signal cannot go lower than -6dB. The remaining part of the signal is direct current, see fig. 5.9.
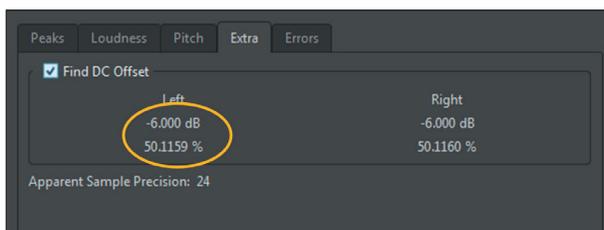
fig. 5.9 50% DC offset and dynamic range

- By clicking **G** and **H** on the keyboard, we can horizontally zoom in and out the waveform. Compare the two waveforms in **05A** and **05D** and notice the wave shifting due to DC offset.

- Reduce the playback level to the minimum. The high level of DC offset can be stressful for the loudspeakers.

- Listen to the example and look at the standard meter. Notice a high RMS level, mainly because of the direct current, rather than the audio signal. Compare it to the example **05A** and examine the difference.

- Look at the Phase meter. The pattern shifts to the top because of the positive DC offset, see fig. 5.10.
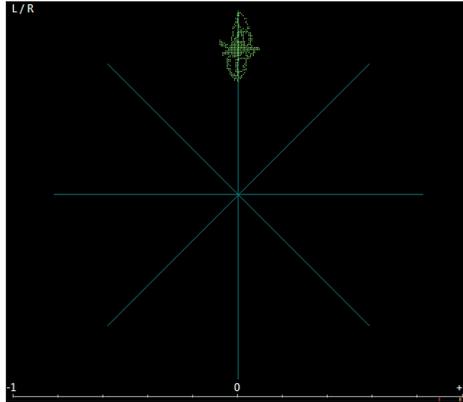
fig. 5.10 DC offset on the Phase meter

- Besides stressing the loudspeakers, a high DC offset value compromises the plugin calculations, eventually producing a poor-quality audio signal.
- Run an off-line FFT analysis and look at the low-end of the spectrum. Notice the presence of a *red wall*. This wall graphically represents the direct current, DC, detected in the signal, see fig. 5.11.
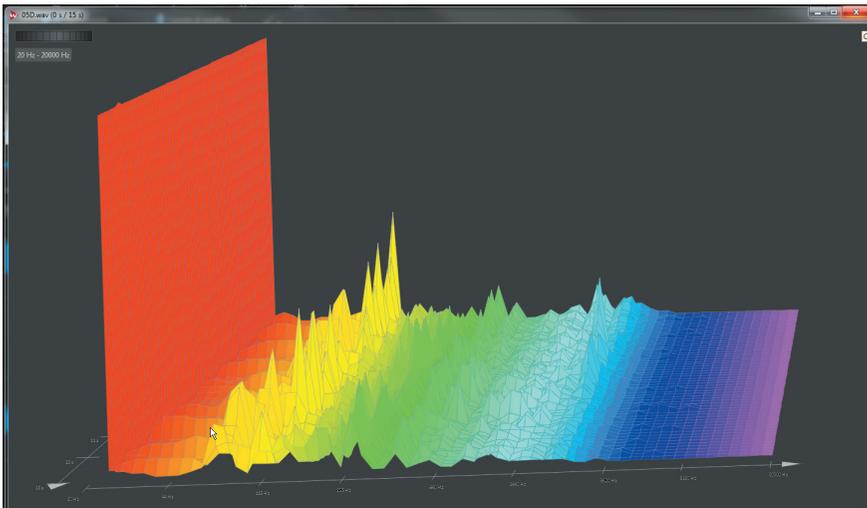
fig. 5.11 DC offset FFT analysis

(...)

other sections in this chapter:

6
SPATIAL PROCESSING

LEARNING AGENDA

PREREQUISITES FOR THIS CHAPTER
- Contents of chapters 1-5

TARGETS
What you will learn
- Learning concepts of correlation and mono-compatibility
- Learning concepts of signal sum and difference
- Learning tools to analyze the spatial content

Skylls
- Using specific plugins meant to fix and adjust the balance
- Using m/s encoding and decoding
- Using reverbs and equalizers for spatial processing

CONTENTS
- Spatial content analysis
- Phase meter advanced use
- Matrix for m/s encoding and decoding
- Types of mastering reverbs

LEARNING TIMES – CH. 6
Self-learning
For a 200-hour independent study session:
- About 20 hours

Courses
For a 40-hour classroom course +a 80-hour independent study session:
- About 3-hour lectures + 2-hour feedback
- About 4-hour independent study

EXERCISES
- Interactive examples

TESTING
- Short-answer questions

SUPPORTING MATERIALS
- Glossary

6.1 ANALYZING BALANCE AND PHASE CORRELATION

Balance

The Wavelab Pan meter allows detecting differences in level and volume between the channels. Continuous intensity variations between the channels usually occur, but if these are constant and long-lasting, they represent a stereo image imbalance that needs to be fixed. Fig. 6.1 shows in light blue a constant RMS imbalance on the right channel. The value on the lower bar of the pan meter is the one that most indicates a problem of level between the channels. The upper line indicates instantaneous variations, typical of stereo content.
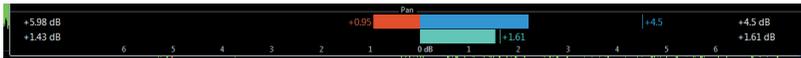
fig. 6.1 Unbalanced signal to the right

To correct the imbalance, we can either attenuate the right channel or amplify the left one or even both. The ideal tool for this kind of processing is Wavelab Stereo Tools. This plugin allows performing the necessary operations on the signal level and phase, see fig. 6.2.
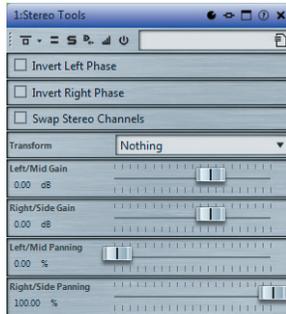
fig. 6.2 Stereo Tools

EXERCISES

- Load up a song

- Look at the pan meter and check if there are constant intensity variations on one of the channels in the lower bar. If the brown or light blue bar permanently remains in the same position, it is necessary to modify the level of one of the two channels. The ideal situation for a correct stereo balance is one in which neither the brown bar nor the light blue appears. Be careful, however. **In the case of a mono signal**, we have the **same levels** on both channels. Not even in that case, the bars appear, but this does not represent an ideal condition in a stereo song.

- The Pan meter, like all Wavelab measuring tools, has a high level of accuracy, capable of detecting small variations of thousandths of dB and of displaying them at the edges of the meter. In older recordings, level differences between channels can be considerable. That does not represent a mix or mastering mistake. At that time, such accurate measuring tools did not exist.

- Load up the master section preset saved in section 5.15. Insert the Stereo Tools plugin in the second slot of the master section, see fig. 6.3.
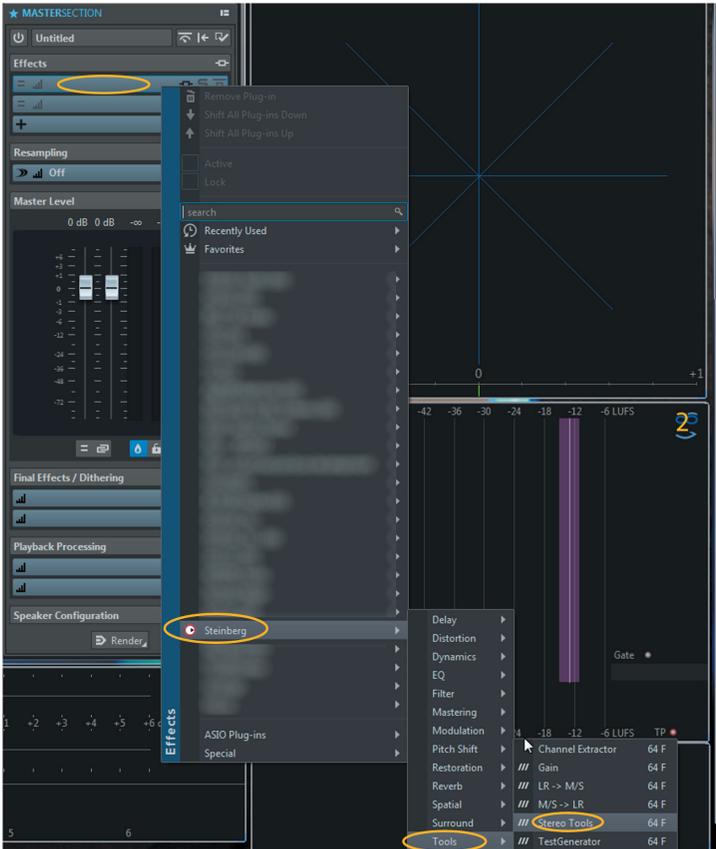
fig. 6.3 Selecting the Stereo Tools plugin

- Save this master section preset too. It is an additional starting point for future processing

- Look at the pan meter and reduce the level of the channel with the highest RMS by using the gain controls. Leave the pan controls untouched, see fig. 6.2. Try to achieve minimal variations in the amplitude of the brown and light blue bars. In this process, it is critical to use a subtractive approach. If we amplify the weakest channel, we may generate distortion or inter-sample peaks higher than 0dBTP

Phase

Phase meter detects correlation issues. Wavelab Phase meter is an advanced measuring tool. It includes both the vector representation of the values of the left and right channels and a phase correlation meter as a horizontal bar with numerical values. This last indication gives a reliable value about signal mono-compatibility. The bar displays, between two red dashes, the minimum, and maximum phase shift between the two channels and with a green dash the instantaneous value, see fig. 6.4.
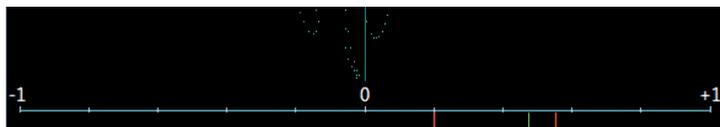
fig. 6.4 Phase meter

The value +1 indicates two identical signals on the left and right, L and R, channels, thus a mono program whose signals are entirely in phase. The phase meter shows a vertical straight line.

Values between +1 and 0 indicate a phase shift between 0 and 90 degrees. 90 degrees is the maximum phase shift that allowed to be mono-compatible. The ambiance processors, reverbs and delay, produce a signal with a phase shift of 90 degrees between the right and left channels.

Values between 0 and -1 indicate a shift from 90 to 180 degrees. Phase cancellation and comb filtering may occur during mono playback. It can result from the common mix practice of creating a stereo image by delaying one of the two channels of a mono program by few milliseconds or even samples. The resulting signals are then sent to L and R and perceived as a stereo signal. The spectral and dynamic content of the two signals is the same, listening to them in mono produces audible comb filtering effects.

The value -1 means that the two channels are identical, a mono program whose signals are entirely out of phase. The phase meter shows a horizontal straight line. By switching to *mono* playback, the two summed signals cancel each other out. Speakers do not reproduce any sound

Check mono-compatibility by using the relative feature in the master section, or a monitor controller in the studio. We can customize the shortcuts in Wavelab. We can configure the F11 and F12 keys to switch between stereo and mono playback easily, see fig. 6.5.

fig. 6.5 Mono playback

Compare every time mono and stereo content. This trains the ear to tell phase cancellation and comb filtering phenomena.

(...)

other sections in this chapter:

6.2   Stereo image width
         Exercises
6.3   M/S spatial processing
         In depth
         M/S stereo image processing
         Exercises
6.4   Mastering reverbs
         Exercises
6.5   Spatial equalization
         Exercises

Test
Glossary

7
DYNAMICS PROCESSING

LEARNING AGENDA

PREREQUISITES FOR THIS CHAPTER
- CONTENTS OF CHAPTERS 1-6

TARGETS
WHAT YOU WILL LEARN
- LEARNING DYNAMICS PROCESSING CHARACTERISTICS AND ITS FEATURES
- LEARNING HOW TO ANALYZE AMPLITUDE AND ENVELOPE
- LEARNING VARIOUS KINDS OF DYNAMICS PROCESSING: ACTIVE, PASSIVE
- LEARNING KINDS OF DYNAMICS PROCESSORS: COMPRESSORS, EXPANDERS, LIMITERS, GATES, TRANSIENT DESIGNERS
- LEARNING HOW TO USE MULTIBAND DYNAMICS PROCESSORS AND DYNAMIC EQUALIZERS

SKILLS
- BEING ABLE TO ANALYZE AND IDENTIFY WHICH KIND OF PROCESSING TO USE IN MASTERING
- USING SOFTWARE DOWNWARD AND UPWARD DYNAMICS PROCESSORS
- USING TRANSIENT DESIGNERS
- USING DYNAMICS PROCESSORS TO REACH THE DESIDERD LOUDNESS VALUE, WITHOUT DESTROYING SIGNAL QUALITY

CONTENTS
- DYNAMIC RANGE ANALYSIS
- DOWNWARD AND UPWARD PROCESSORS
- MULTIBAND PROCESSORS
- SPECIFIC APPLICATIONS FOR DYNAMICS PROCESSING

LEARNING TIMES – Ch. 7
SELF-LEARNING
FOR A 200-HOUR INDEPENDENT STUDY SESSION:
- ABOUT 20 HOURS

COURSES
FOR A 40-HOUR CLASSROOM COURSE + A 80-OURS INDEPENDENT STUDY SESSION:
- ABOUT 6-HOUR LECTURES + 2-HOUR FEEDBACK
- ABOUT 8-HOUR INDEPENDENT STUDY

EXERCISES
- INTERACTIVE EXAMPLES

TESTING
- SHORT-ANSWER QUESTIONS

SUPPORTING MATERIALS
- GLOSSARY

Preface and basic concepts

One of the most delicate and complex operations during the mastering process is dynamic processing. Often, it simply means increasing the perceived volume, completely neglecting the ratio between each moment of the performance. Though it is necessary, dynamics processing must be respectful to the dynamic balance of the song, still ensuring the best listening experience possible on most of the playback systems. Tools for this purpose are called the dynamics processors. They work by modifying the input and output levels. Such an operation is accomplished by varying the gain of a signal amplifier. If the gain is negative, the signal is attenuated. If the gain is positive, the signal is amplified. The function of a dynamics processor is graphically represented on a Cartesian axes system, whose vertical and horizontal lines represent the output and input levels, respectively, see fig. 7.1.
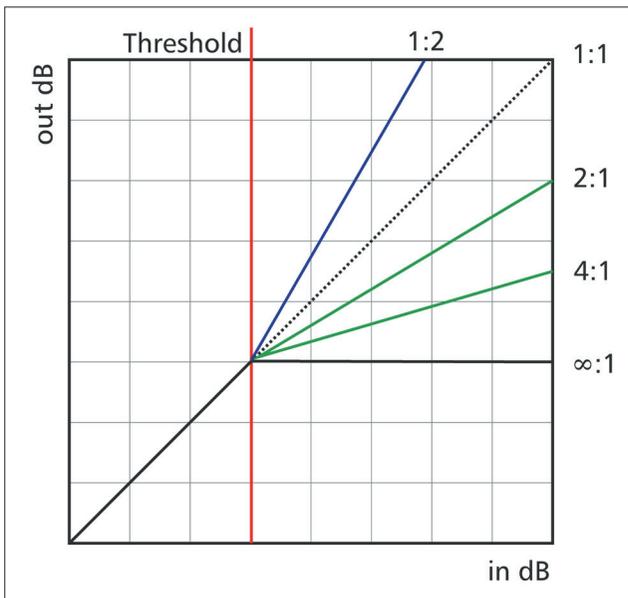
fig. 7.1 Function graph of a dynamics processor

A dynamics processor modifies the difference between the highest and lowest level of a signal, i.e., the dynamic range. Dynamic range and volume are inversely proportional to each other. It means that the signal with the lowest dynamic range sounds louder, see fig. 4.75.

One can make different kinds of processing in order to modify the dynamic range of a signal; the main ones are:

Compression
Dynamic range compression leads to **reducing the level difference between the highest and the lowest value of a signal**. Just like any other dynamics processing, it is used in many areas such as mixing, mastering, broadcasting, noise reduction. In mastering, it is used to control the perceived volume, the RMS level of the sound. Fig. 7.2 shows a signal and its Highest Peak, Lowest Peak and Dynamic Range. A higher peak than the average of the other peaks (High Peak Average) is highlighted in red. The average volume (RMS) is highlighted in yellow. It corresponds to the surface covered by the waveform. The bigger this area, the higher the energy of the wave and its RMS value.
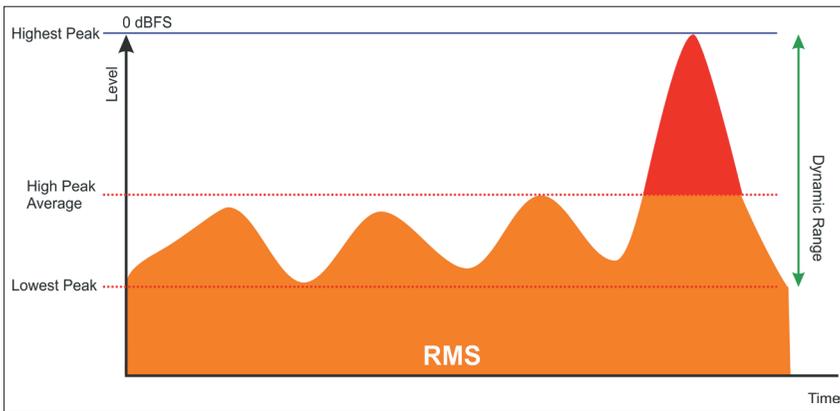
fig.7.2 Uncompressed signal

By reducing the gain (Gain Reduction) in the red peak area, a dynamic range reduction is achieved, see fig. 7.3.
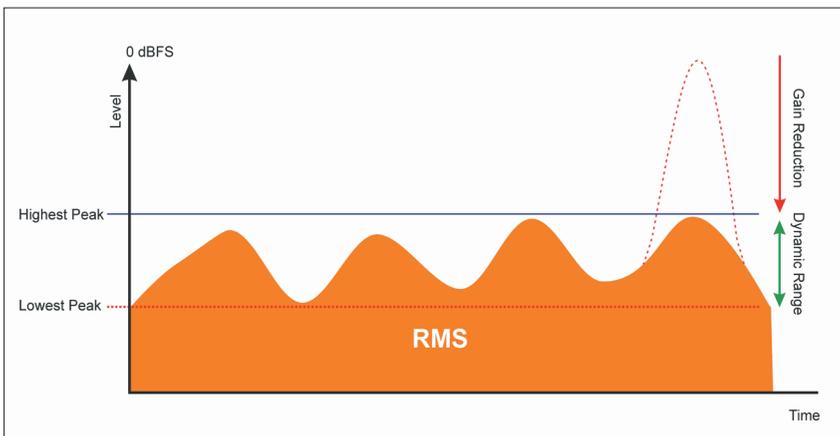
fig. 7.3 Compressed signal

The original signal had a maximum level equal to the red peak. After compression, it is possible to increase the gain until that value is reached again, thus increasing in RMS value, the yellow area, see fig. 7.4.
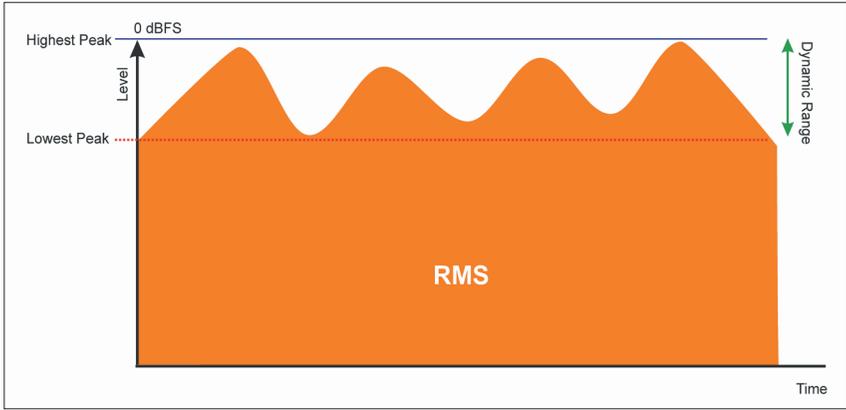
fig. 7.4 Compressed and amplified signal

• •

EXERCISES

- Load up the example **07A**.

• •

INTERACTIVE EXAMPLE 07A – UNCOMPRESSED SIGNAL

• •

- Again, the level of the playback system must stay the same during the following listening tasks.
- Listen to the song, look at the meters, and take note of the peak and RMS levels. We can use the off-line global analysis to collect more accurate data. Try to remember the perceived loudness. The example **07A** is taken from a mix made with no compression and can be compared to the signal in fig. 7.2.
- Load up the example **07C**.

• •

INTERACTIVE EXAMPLE 07C – COMPRESSED SIGNAL

• •

- Do not modify the level of the playback system.
- Listen to the song, look at the meters, and take note of the peak and RMS levels. We can use the off-line global analysis to collect more accurate data. Compare the perceived loudness to that of the example **07A**. Open the two examples at the same time and switch from one to the other one while listening. The example **07C** is a compressed mix similar to the signal in fig. 7.4. The loudness increase resulting from the dynamics processing is apparent.

Expansion

Dynamic range expansion leads to **increasing the level difference between the highest and lowest value of a signal**. It is widespread in audio restoration for noise reduction, as well as in any processing intended for controlling the ratio between ambiance sound and direct sound. Dynamic range and volume are inversely proportional to each other. Dynamics expansion leads to a decrease in perceived volume. Fig. 7.5 shows an expansion process to reduce the noise floor. Notice the dynamic range of the signal before (Original) and after expansion (Expanded). The level of the noise floor (Noise) is highlighted in gray. The example refers to a case in which noise is masked (see section 9.1).
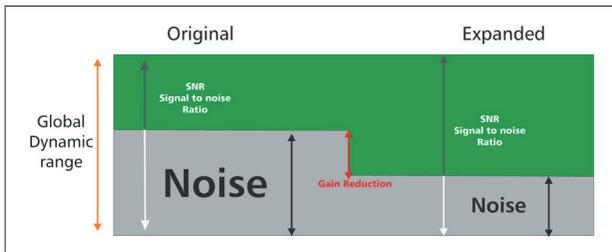
fig. 7.5 Expanded signal

. . . .

EXERCISES

- Load up the example **07D**.

. . . .

INTERACTIVE EXAMPLE 07D – SIGNAL WITH NOISE

. . . .

- The level of the playback system must stay the same during the following listening tasks
- Listen to the song. Notice the noise floor throughout the whole song. This noise is audible at the beginning and end of the song when there is silence
- Adjust the volume of the playback system to hear the noise.
- Use the Loop function, shortcut key ⌐/⌐, to select, and listen to the area of the waveform that includes the noise, in a loop, see fig. 7.6.
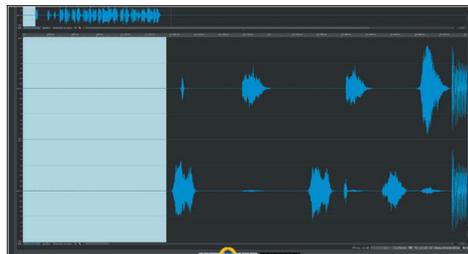
fig. 7.6 Loop function

(...)

other sections in this chapter:

8
DISTORTION AND SATURATION

LEARNING AGENDA

PREREQUISITES FOR THIS CHAPTER
- CONTENTS OF CHAPTERS 1-7

TARGETS
WHAT YOU WILL LEARN
- LEARNING SATURATION AND DISTORTION FEATURES
- LEARNING KINDS OF HARMONIC DISTORTION
- LEARNING TRANSFER FUNCTIONS FEATURES

SKILLS
- BEING ABLE TO ANALYZE AND IDENTIFY WHICH KIND OF DISTORTION OR SATURATION TO USE IN MASTERING
- USING DISTORTION AND SATURATION PLUGINS
- USING EVEN OR ODD HARMONICS
- USING DISTORTION TO ACHIEVE THE DESIRED LOUDNESS VALUE, WITHOUT DESTROYING SIGNAL QUALITY

CONTENTS
- ANALYZING SIGNAL TRANSFER FUNCTIONS
- PROCESSORS AND THEIR FEATURES
- MULTIBAND PROCESSORS

LEARNING TIMES - Ch. 8
SELF-LEARNING
FOR A 200-HOUR INDEPENDENT STUDY SESSION:
- ABOUT 10 HOURS

COURSES
FOR A 40-HOUR CLASSROOM COURSE + A 80-HOUR INDEPENDENT STUDY SESSION:
- ABOUT 2-HOUR LECTURES + 1-HOUR FEEDBACK
- ABOUT 3-HOUR INDEPENDENT STUDY

EXERCISES
- INTERACTIVE EXAMPLES

TESTING
- LISTENING AND ANALYSIS TESTS
- SHORT-ANSWER QUESTIONS

SUPPORTING MATERIALS
- GLOSSARY

8.1 DISTORTION

Dynamics processing is not the only way to modify the loudness. Distortion can help to increase the RMS value of a signal as well. As the name suggests, distortion modifies the shape and the original content of a signal, see fig. 8.1.
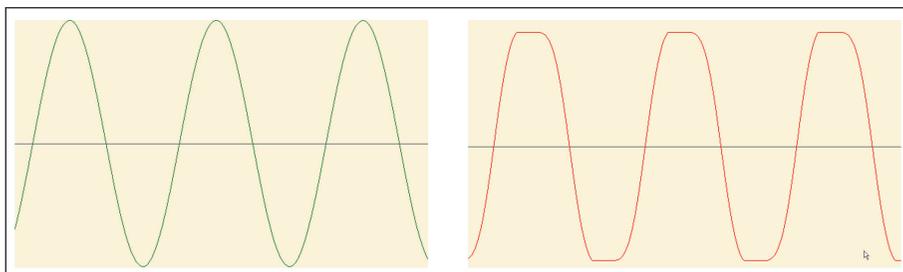
fig. 8.1 Original signal in green, the distorted signal in red

Distortion can be achieved, or caused, by many factors. It can depend on a voltage or current overload of an electric or digital circuit. It can be a peculiar transfer function of some devices like vacuum tubes, magnetic tape, transistors, transformers. Often, distortion indicates a defect in circuitry, a wrong ratio between the output and input level, a circuit overload. So, it seems like it is everything but the desired element in a signal. However, distortion, along with other factors, is what makes so many analog devices, especially the old ones, sound the way they do. In a signal, the distortion level must be low, but it is because of the linearity and accuracy of digital devices that, sometimes, one wants to emulate *old-style* devices. The purpose is, besides increasing the RMS value, obtaining the typical sound of a particular period or epoque, when the digital processing still was not considered as a tool for sound processing. Distortion leads to an increase in RMS, and that is one of the reasons why it is used in mastering. Much analog equipment is so appreciated because they can add something more to the signal, i.e., what we often call *warmth*. There are many physical and electronic processes behind this process, but distortion has a crucial role. Many plugins are developed to digitally emulate the behavior of different kinds of analog circuits, as to recreate that specific sound.

8.2 SATURATION

Modifying the waveform of the signal means distorting it. However, there are different stages and modes to generate it. We can alter the amplitude, frequency, and harmonic content of a signal. Such alterations may depend on using various circuits or electronic components but, before dealing with real distortion, we have to consider that there is an altered state of the signal, which is called **Saturation**. Saturation can occur in circuits close to overloading or with a non-linear response, and it generally changes the way the signal is sent to the output or the recording media. It represents its transfer function. Several devices can generate saturation, like reel-to-reel tape recorders, see fig. 8.2.

fig. 8.2 Reel-to-reel tape recorder

The way the signal is magnetically transferred from the recording heads to the magnetic tape is not linear. It seems that the tape is *deaf* to the low-level signal. The response then becomes linear until, beyond a certain level, the tape becomes non-linear again.

(...)

other sections in this chapter:

9
NOISE REDUCTION

LEARNING AGENDA

PREREQUISITES FOR THIS CHAPTER
- Contents of chapters 1-8

TARGETS
What you will learn
- Learning all kind of noise
- Learning different methods to reduce noise
- Learning tools for spectral editing

Skills
- Being able to tell the difference between masked and unmasked noise
- Being able to tell the difference between impulse and continuous noise
- Using dynamics processing to reduce noise
- Using spectral processing to reduce noise

CONTENTS
- Analysis and difference between noise, hum, buzz
- Processors for dynamic noise reduction
- Processors for spectral noise reduction

LEARNING TIMES - Ch. 9
Self-learning
For a 200-hour independent study session:
- About 6 hours

Courses
For 40-hour classroom course + 80-hour independent study:
- About 1-hour lecture + 1-hour feedback
- About 2-hour independent study

EXERCISES
- Interactive examples

TESTING
- Short-answer questions

SUPPORTING MATERIALS
- Glossary

9.1 DYNAMIC NOISE REDUCTION

As stated above in section 2.5, a signal may contain different kinds of noise.

These can be:

- **Impulse-like**: The clicks occurring when playing back vinyl, noise resulting from oxide degradation in a magnetic tape, impulses resulting from clock interruption in a digital system, or microphone handling.

- **Constant**: Rustling of magnetic tape, the background noise of the electrical circuits, ambiance sounds in noisy environments, unfiltered mains voltage.

- **Constant frequency**: When the frequency content is constant. 50 or 60 Hz frequency plus harmonics (in which case we are dealing with **Hum**), resulting from domestic appliances, engines, AC transformers.

- **Wideband**: Noise made up of several frequencies or even all frequencies of the spectrum, as for White and Pink noise, or noises with a complex spectrum.

- **Masked**: Noise audible during silence or musical rests.

- **Unmasked**: Noise audible, even when there is sound.

There are various kinds of tools and methods to remove or reduce such noises. Before using them, we must analyze the signal content.

If during signal analysis, the noise seems to be constant and masked, we can apply dynamic noise reduction to reduce it. Noise is audible when the original signal stops playing. Use a downward expander and set its threshold to a value just above the noise floor. Fig. 9.1 shows masked noise in blue, signal in green, and audible noise in red.
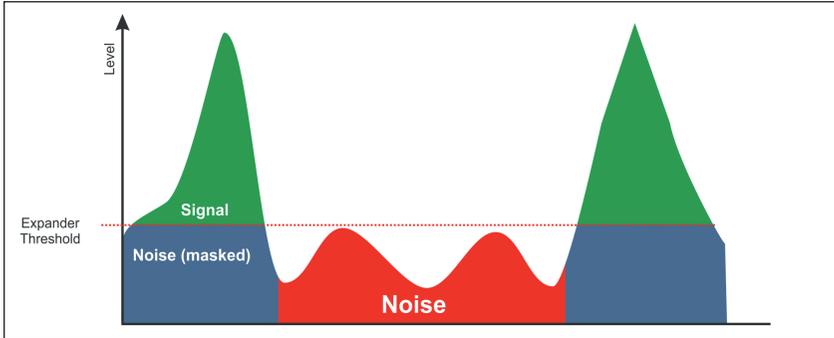
fig. 9.1 Masked constant noise

If we use fast attack and release times, we can reduce the noise as soon as it goes below the threshold level. Fig 9.2 shows the Wavelab expander set up to reduce the masked noise.

fig. 9.2 Expander for noise reduction

If noise is constant, masked, and falls within specific frequency ranges, it is possible to use the Wavelab multiband expander to reduce noise in the affected frequency range, see fig. 9.3.

fig. 9.3 Multiband expander for top-end noise reduction

• •

For the exercises concerning broadband and multiband dynamic noise reduction, refer to the exercises in sections 7.7 and 7.11

• •

When noise is like an impulse, analysis parameters need to include, besides noise level, its envelope too. An expander cannot analyze the attack time of a signal. Processors to reduce impulse-like noise are sort of hybrids between Transient Designers and Expanders. Wavelab features a plugin suite entirely dedicated to audio restoration. RestoreRig includes three modules. One of them is dedicated to impulse-like noises reduction. Fig. 9.4 shows the DE CLICKER.

fig. 9.4 DE CLICKER module in RestoreRig

This processor features three modes to detect impulse-like noise:

- **POP**: for noises with a duration between 2 ms and 10 ms
- **CLICK**: for noises with a duration between 0,1 ms and 3 ms
- **CRACKLE**: for noises with a duration shorter than 0,4 ms

The various sensitivity curves allow applying impulse-noise reduction gradually. Firstly, reduce impulses that last longer, through POP control. Then, use the CLICK control, and finally, if small transients persist, use the CRACKLE module. Each module can be enabled separately, to allow for a benchmarking between the various detecting algorithms.

(...)

other sections in this chapter:

10
M/S MASTERING

LEARNING AGENDA

PREREQUISITES FOR THIS CHAPTER
- CONTENTS OF CHAPTERS 1-9

TARGETS
WHAT YOU WILL LEARN
- LEARNING TO RECOGNIZE THE CHARACTERISTICS OF SIGNALS SUM AND DIFFERENCE PROCESSES
- LEARNING SERIAL AND PARALLEL ROUTING OF THE M/S MATRIX
- LEARNING THE MAIN CHARACTERISTICS OF THE MASTERING PROCESS FOR VINYL

SKILLS
- BEING ABLE TO IDENTIFY WHAT KIND OF AUDIO MATERIAL NEEDS M/S PROCESSING
- BEING ABLE TO CREATE M/S MATRIX WITHIN THE EDITOR AND THE AUDIO MONTAGE
- USING PLUGINS IN L-R AND M/S MODE
- USING THE DIFFERENT OPTIONS AVAILABLE IN THE MASTER SECTION FOR ADVANCED ROUTING

CONTENTS
- ANALYZING SPATIALIZATION AND ORIGINAL CONTENT
- SERIAL PROCESSING
- PARALLEL PROCESSING
- TECHNOLOGY FOR VINYL REPRODUCTION

LEARNING TIMES - Ch. 10
SELF-LEARNING
FOR A 200-HOUR INDEPENDENT STUDY SESSION:
- ABOUT 14 HOURS

COURSES
FOR A 40-HOUR CLASSROOM COURSE + A 80-HOUR INDEPENDENT STUDY SESSION:
- ABOUT 3-HOUR LECTURES + 2-HOUR FEEDBACK
- ABOUT 4-HOUR INDEPENDENT STUDY

EXERCISES
- INTERACTIVE EXAMPLES

TESTING
- SHORT-ANSWER QUESTIONS

SUPPORT MATERIALS
- GLOSSARY

10.1 SERIAL M/S PROCESSING

So far, we have been processing the left and right channels of a stereo program. It allows us to equalize, compress, expand the **L** and **R** channels. The biggest problem with **L-R** processing is the ambiance. Dynamics processing tends to modify the ratio between the middle and the side components, often leading to unwanted results and pumping effect on the ambiance sound[27]. As stated in section 6.3, if we encode an **L-R** stereo signal in **M/S**, we can extract **M** and **S** that correspond to the Mono, or **Middle** and the Stereo, or **Side** component of the sound, respectively. We can process the central channel dynamically, leaving the ambiance sound unaltered. After dynamics processing has been performed, the level of the ambiance sound is lower than the central channel, if we compare the result to the original signal. Thus, we need to compensate for such variation either by increasing the level of the **S** signal or by decreasing that of the **M** signal, until we get the same stereo width of the original signal. After this process, we use an **M/S** decoding matrix to get the **L** and **R** channels back, see fig. 10.1.
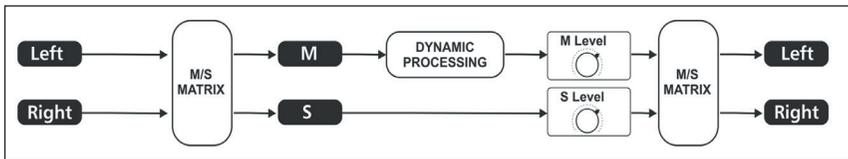
fig. 10.1 M/S dynamics processing

Even when mastering in the box in **M/S** mode, we must rely on floating-point processing. It is the only way to create the necessary headroom to make up for variations between **M** and **S** without losing bit resolution and dynamic range. In the same way, using the phase meter properly is critical in order to understand how the stereo image is modified, see fig. 10.2.
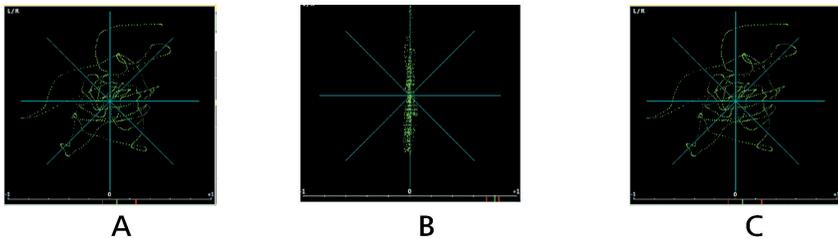
fig. 10.2 Adjusting the stereo width

---

[27] Although most of the ambience sound can be on the S signal, the M channel includes information about it too. Yet, since reflections resulting from spatialization are generally different from each other, rarely they are identical on both L and R, on the M signal.

Figure **A** shows the stereo image before applying dynamics processing on **M**.
Figure **B** shows the stereo image after applying dynamics processing on **M**.
Figure **C** shows the stereo image after applying dynamics processing on **M** and compensating for the level of **S**.

As explained in section 6.3, there are various ways to create an **M/S** matrix. In Wavelab, we can process the **M** and **S** channels individually both in the master section and in the audio montage. The software creates the necessary encoding and decoding matrix automatically. In the master section, we can choose the type of processing for each effect slot, see fig. 10.3.
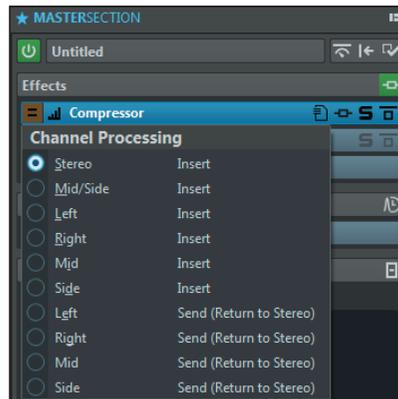
fig. 10.3 **M/S** processing within the master section

By **Middle**, we mean the **M** channel, by **Side**, we mean the **S** channel.
In the audio montage, we can use **M/S** processing both on the clip and on the track and the whole montage, see fig. 10.4.
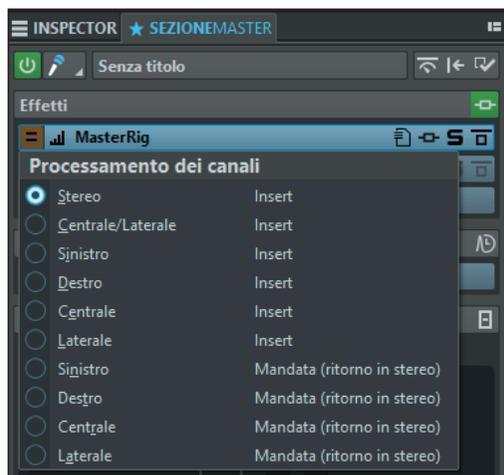
fig. 10.4 **M/S** processing in the audio montage

Wavelab allows the processing of the signal in many ways. We can use the **M/S** technique in the audio montage and then process the signal in **L-R** mode within the master section. All can be done in real-time, listening to each parameter variation simultaneously, thanks to the automatic **M/S** matrix function. Fig. 10.5 shows a high-pass filter on the **S** channel in **M/S** mode within the audio montage and a compressor in the master section in **L-R** mode.

fig. 10.5 **M/S** processing in the audio montage and **L-R** processing in the master section

(...)

other sections in this chapter:

10.2 Mastering for vinyl
         Exercises
10.3 Parallel M/S processing
         Exercises

Test
Glossary

11
STEM MASTERING

LEARNING AGENDA

PREREQUISITES FOR THIS CHAPTER
- Contents of chapters 1-10

TARGETS
What you will learn
- Learning the characteristics of multitrack mastering
- Learning the types and formats of stem files
- Learning how to use serial and parallel processing on a multitrack project

Skills
- Being able to acquire, arrange and import stems
- Being able to create and manage multitrack projects through the audio montage
- Understanding which is the proper processing for each track
- Using serial, parallel processing, and matrix on each stem

CONTENTS
- Analyzing material and creating a montage
- Tracks effect menu
- Montage effect menu
- Rendering a multitrack montage

LEARNING TIMES - Ch. 11
Self-learning
For a 200-hour independent study session:
- About 12 hours

Courses
For a 40-hour classroom course + a 80-hour independent study session:
- About 2-hour lectures + 1-hour feedback
- About 3-hour independent study

EXERCISES
- Interactive examples

TESTING
- Short-answer questions

SUPPORT MATERIALS
- Glossary

11.1 MIXING VS MASTERING

The mastering process is generally performed on a full mix provided by the recording studio. Though the producer, or the mastering studio, may ask for a different kind of processing called **Stem Mastering**. The idea is that separate processing for each sub-mix, i.e., the stems, including drums, vocals, effects, and synths, can give better results than those typically get with a simple stereo mix processing. Generally, stems are stereo tracks that are equal in duration and can then be inserted in a DAW project or, as in the case of Wavelab, in an audio montage, see fig. 11.1.
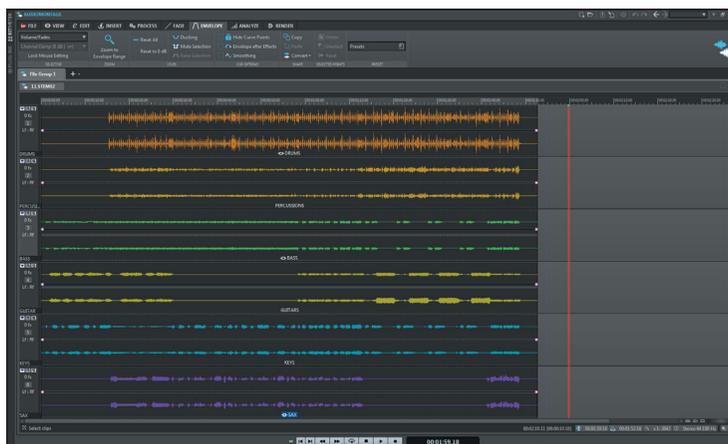
fig. 11.1 Stem mastering in an audio montage

The chance to use specific processing for each track of a stem mastering allows optimizing the plugins and set them up according to the specific content of each track. Nonetheless, this way, the mastering engineer is responsible for the mix, the balance, and the spatial positioning of every element of the song. These details are generally decided during the mixing process in the recording studio. It is strongly recommended to carefully evaluate the choice to perform or not perform a stem mastering and go for this solution only when needed.

11.2 ACQUIRING AND ARRANGING MATERIAL

As stated in section 4.2, the first operation to do is copying the delivered material in a secure location. Never apply any process neither on original files nor on backup copies. After pasting a second copy of the stems in a folder, analyze the content of the files. Usually, the level of the stems matches the actual peak level of the section in the mix. It means that if we import the stems on separate tracks of the audio montage and leave the level of each track at 0dB, we get the same peak level of the stereo mix on the master bus. Another parameter to check is the bit depth of each stem. If the bit depth is lower than a 32-bit floating-point, it is strongly recommended to change it to 32-bit float before inserting the stems within the montage.

Converting the file to a higher bit depth does not increase its resolution, but it allows the software to save intermediate files at a high resolution.

(...)

other sections in this chapter:

11.3 Setting up a stem project
         Exercises

Test
Glossary

12
SURROUND MASTERING

LEARNING AGENDA

PREREQUISITES FOR THIS CHAPTER
- CONTENTS OF CHAPTERS 1-11

TARGETS
WHAT YOU WILL LEARN
- LEARNING FEATURES OF THE MULTICHANNEL OUTPUT BUS
- LEARNING TO ANALYZE THE SURROUND IMAGE
- LEARNING TO USE MULTICHANNEL DYNAMICS PROCESSING
- LEARNING TYPES AND FEATURES OF MULTICHANNEL PLUGINS

SKILLS
- BEING ABLE TO CONFIGURE AND MANAGE THE ROUTING OF THE MULTICHANNEL OUTPUT BUS
- BEING ABLE TO USE THE AUDIO MONTAGE FOR SURROUND PROCESSING
- BEING ABLE TO USE THE TOOLS TO ANALYZE THE SURROUND IMAGE
- BEING ABLE TO MAKE COMPLEX PROCESSING OPERATING MULTICHANNEL OUTPUT ROUTING

CONTENTS
- SURROUND MASTER SECTION
- SURROUND METER
- MULTICHANNEL PLUGINS
- MULTICHANNEL RENDERING

LEARNING TIMES - Ch. 12
SELF-LEARNING
For a 200-HOUR INDEPENDENT STUDY SESSION:
- ABOUT 15 HOURS

COURSES
For a 40-HOUR CLASSROOM COURSE + A 80-HOUR INDEPENDENT STUDY SESSION:
- ABOUT 3-HOUR LECTURES + 2-HOUR FEEDBACK
- ABOUT 4-HOUR INDEPENDENT STUDY

EXERCISES
- INTERACTIVE EXAMPLES

TESTING
- SHORT-ANSWER QUESTIONS

SUPPORT MATERIALS
- GLOSSARY

12.1 MULTICHANNEL BUS

Just like Stem Mastering, surround mastering implies using a certain number of tracks inside an audio montage. The number of tracks depends on the standard. 5.1 surround needs six tracks: Left, Center, Right, Left surround, Right surround, LFE. The number of outputs of the master section is different. In surround mastering, the master section has as many channels as the standard we choose for the soundtrack. If the standard is 5.1, the master section has six output channels, see fig. 12.1.
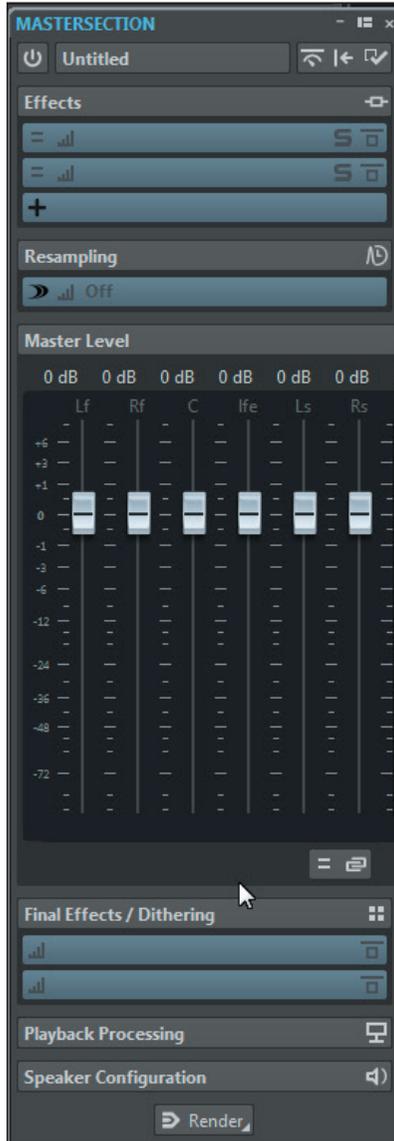
fig. 12.1 5.1 master section

The Wavelab meter is configured similarly, displaying the six channels of the mix. Pan meters display the front L-R and the surround L-R sections, see fig. 12.2.
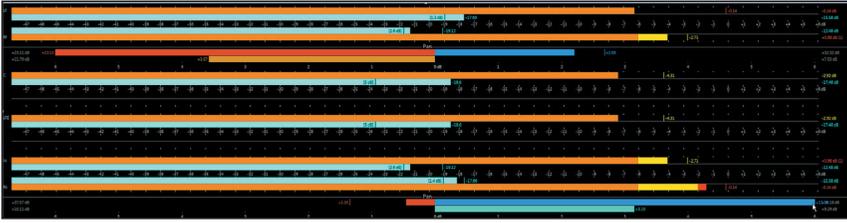
fig. 12.2 5.1 Multichannel meter

The audio montage contains as many tracks as the audio channels. There are six tracks, see fig. 12.3.
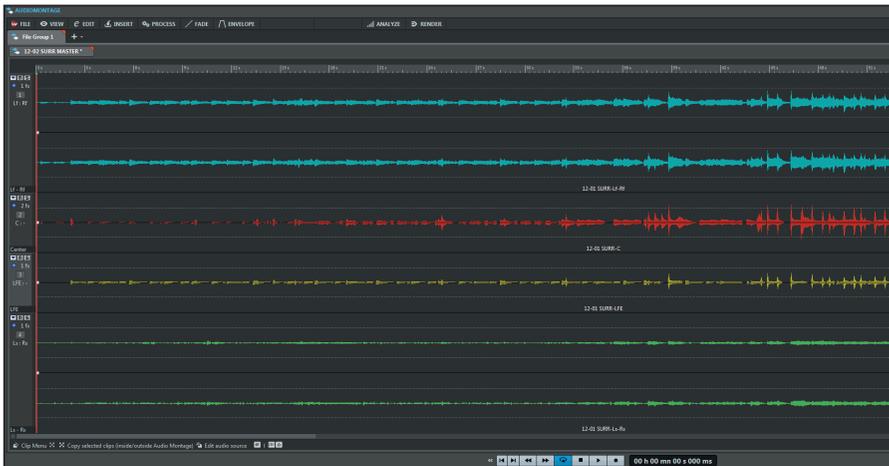
fig. 12.3 5.1 audio montage

Each track is assigned to the corresponding audio output of the control room. Fig. 12.4 shows the assignment window for the outputs of the track. In this specific case, the track LFE is assigned to the output LFE.
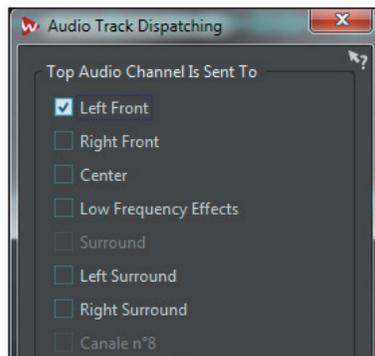
fig. 12.4 Assigning the outputs of the tracks within the audio montage

Usually, in surround mastering, we want to insert one channel of the final mix in each track. This way, as shown above, the output of the track matches the mix bus. The tracks of the audio montage may correspond to sections of the mix or specific groups of sounds, both mono, stereo, and multichannel. We can select the outputs of the busses needed for each track, see fig. 12.5.
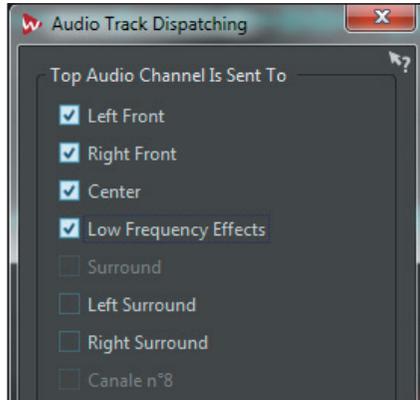
fig. 12.5 Assigning multiple outputs of the tracks within the audio montage

When a track is assigned to more outputs at the same time, and some of these are surround-busses, we can open the **surround panner**, right-clicking on the track panel, see fig. 12.6.
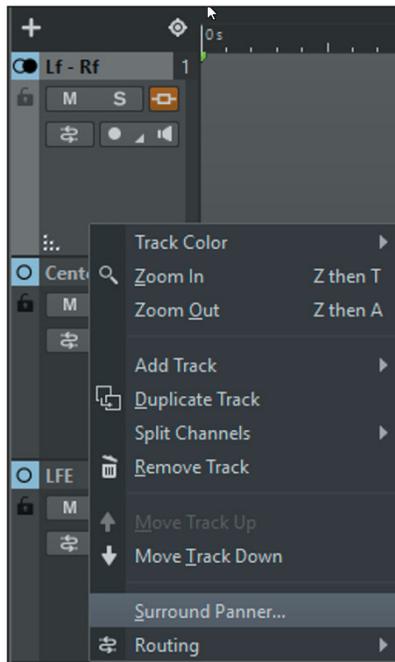
fig. 12.6 Surround panner control of the tracks within the montage

For fast adjustments, use the mouse within the area of the panner to move the green dot, i.e., the sound source. For more accurate adjustments, right-click the panner as to get a more accurate view, see fig. 12.7.
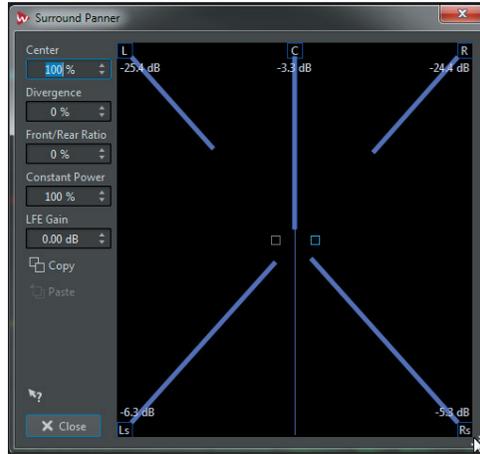
fig. 12.7 Accurate surround panner

Surround panning automation is possible, using the envelope curves. Fig. 12.8 shows the automation for Front-to-Rear movement.
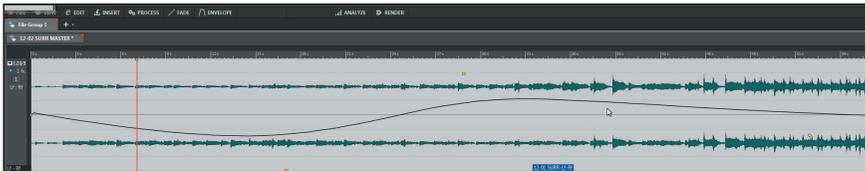
fig. 12.8 Front-to-Rear automation

It is possible to store different control room configurations in order to use various speaker systems in various configurations. Stereo Near field, Stereo Main, Surround 5.1, Quad, Stereo 2.1, see fig. 12.9.
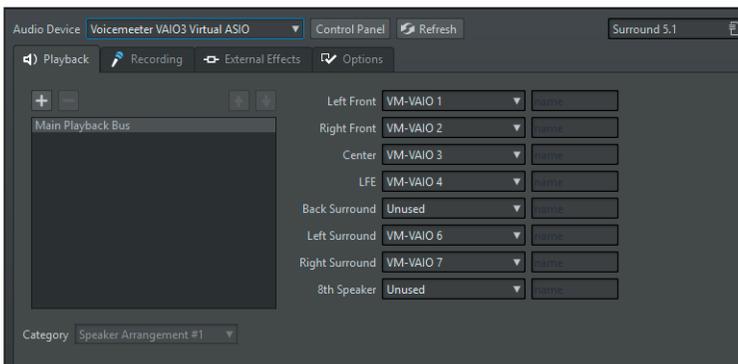
fig. 12.9 Control room configuration menu

(...)

other sections in this chapter:

13
DITHER

LEARNING AGENDA

PREREQUISITES FOR THIS CHAPTER
- CONTENTS OF CHAPTERS 1-12

TARGETS
WHAT YOU WILL LEARN
- LEARNING TO REDUCE BIT RESOLUTION
- LEARNING TO RECOGNIZE THE DISTORTION RESULTING FROM BIT REDUCTION
- LEARNING TO RECOGNIZE THE HARMONIC CONTENT RESULTING FROM BIT REDUCTION
- LEARNING THE TYPES OF NOISE USED IN DITHER
- LEARNING TO FILTER THE NOISE

SKILLS
- BEING ABLE TO CHOOSE WHICH NOISE TO USE
- BEING ABLE TO USE THE DITHER PLUGIN
- BEING ABLE TO VIEW DITHER THROUGH REAL-TIME AND OFF-LINE ANALYSIS
- BEING ABLE TO USE DITHER OF VARIOUS COMPLEXITY AND SPECTRAL SHAPES

CONTENTS
- BIT METER
- DITHER IN MASTER SECTION
- TYPES OF DITHER
- TYPES OF NOISE SHAPING

LEARNING TIMES - Ch. 13
SELF-LEARNING
FOR A 200-HOUR INDEPENDENT STUDY SESSION:
- ABOUT 9 HOURS

COURSES
FOR A 40-HOUR CLASSROOM COURSE + A 80-HOUR INDEPENDENT STUDY SESSION:
- ABOUT 1-HOUR LECTURES + 1-HOUR FEEDBACK
- ABOUT 2-HOUR INDEPENDENT STUDY

EXERCISES
- INTERACTIVE EXAMPLES

TESTING
- SHORT-ANSWER QUESTIONS

SUPPORT MATERIALS
- GLOSSARY

13.1 BIT DEPTH REDUCTION

During ITB mastering, a high bit resolution, the highest possible indeed, is used. Using floating-point arithmetic increases calculation accuracy both in 32-bit float mode and 64-bit float mode. At the end of the production process, the number of bits must be reduced. For an Audio CD, the number of bits is 16. Thus, we need to *truncate* the file resolution by removing the bits after the sixteenth. In the digital domain, bits represent the amplitude of the signal. The theoretical dynamic resolution of a 16-bit signal is 96,32dB, i.e., 6,02dB for every bit. Considering that the maximum level of a digital signal is 0dBFS, the lowest level of a signal, that can be represented by a 16-bit converter, is -96,32dBFS. Fig. 13.1 shows the way bits the signal amplitude. **MSB, Most Significant Bits**, are used for maximum-amplitude values. **LSB, Less Significant Bits**, are used for minimum-amplitude values.
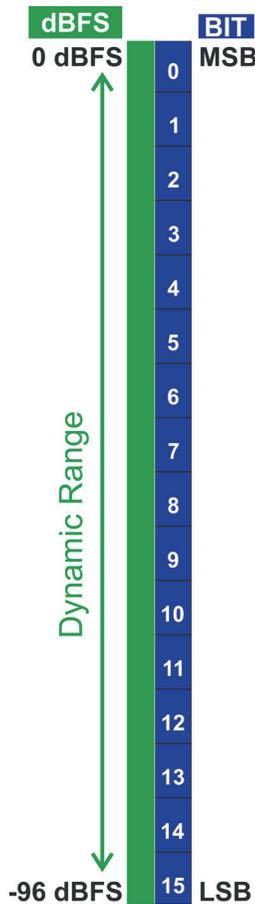
fig. 13.1 Comparison between bits and dynamic range

When transferring a 24-bit file on an Audio CD, we need to truncate the last eight less significant bits, as to preserve the maximum level at 0dBFS, see fig. 13.2.
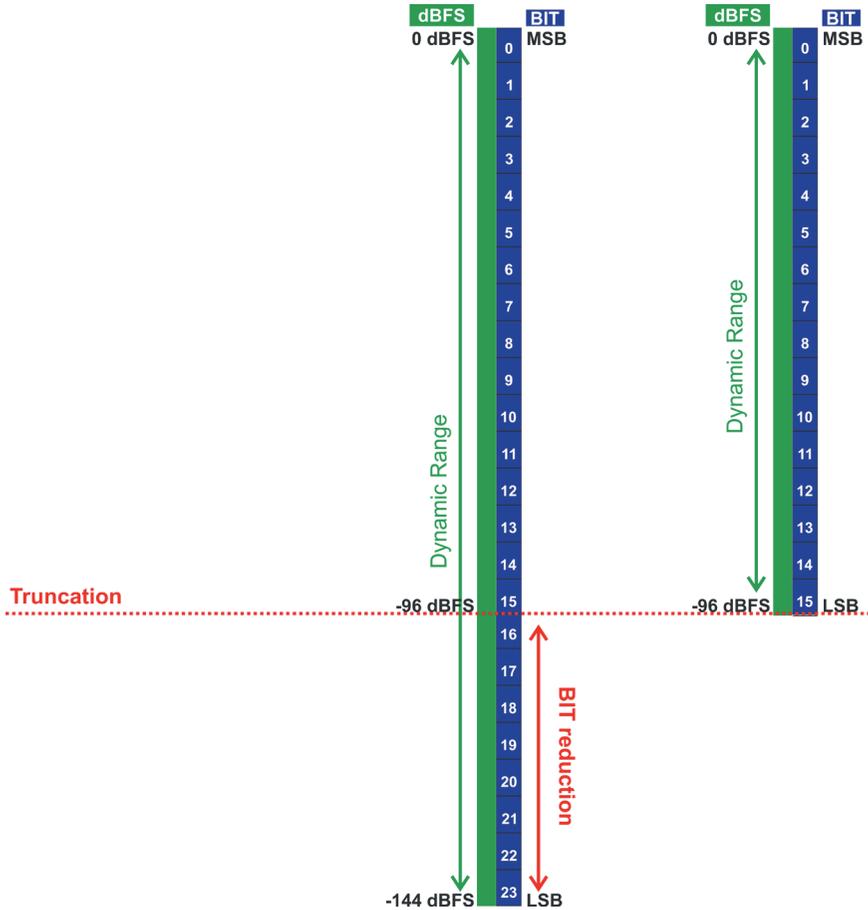
fig. 13.2 Truncating from 24 down to 16-bits

The dynamic range of a 24-bit signal is 144,48dB. Every signal lower than -96,32dB and the information down to -144,48dB, is lost. Generally, there is not that much sound information in the dynamic range between -96 and -144dBFS, yet few problems come out. This is the step when, as we stated at the beginning, we are forced to make a mistake. If the acquisition and processing parameters have appropriately been set up during the production process, the error is minimum. Approximating a numerical value calculated with extreme accuracy is better than approximating an inaccurately calculated value. Once again, the importance of using the floating-point calculation during the ITB production stages is reiterated.

(...)

other sections in this chapter:

14
AUTHORING

LEARNING AGENDA

PREREQUISITES FOR THIS CHAPTER
- CONTENTS OF CHAPTERS 1-13

TARGETS
WHAT YOU WILL LEARN
- LEARNING MECHANICAL FEATURES OF A VINYL
- LEARNING THE CONCEPT OF ANGULAR VELOCITY
- LEARNING THE FEATURES OF THE EQUALIZATION USED DURING TRANSFER AND VINYL REPRODUCTION
- LEARNING THE PHYSICAL FEATURES OF A CD
- LEARNING THE CONCEPT OF LINEAR VELOCITY
- LEARNING DIGITAL PARAMETERS USED BOTH IN AUDIO AND VIDEO
- LEARNING THE SPDIF PROTOCOL

SKILLS
- BEING ABLE TO ANALYZE AND RECOGNIZE DIFFERENT TYPES OF MEDIA SUPPORT
- BEING ABLE TO SET THE RIGHT SAMPLE RATE DEPENDING ON THE TYPE OF PROJECT
- BEING ABLE TO CONNECT DEVICES THROUGH THE SPDIF STANDARD

CONTENTS
- RIAA EQUALIZATION
- CD DATA ENCODING
- TOC AND ITS AREAS
- ELECTRICAL AND OPTICAL SPDIF

LEARNING TIMES - Ch. 14
SELF-LEARNING
FOR A 200-HOUR INDEPENDENT STUDY SESSION:
- ABOUT 9 HOURS

COURSES
FOR A 40-HOUR CLASSROOM COURSE + A 80-HOUR INDEPENDENT STUDY SESSION:
- ABOUT 3-HOUR LECTURES + 2-HOUR FEEDBACK
- ABOUT 4-HOUR INDEPENDENT STUDY

EXERCISES
- INTERACTIVE EXAMPLES

TESTING
- SHORT-ANSWER QUESTIONS

SUPPORT MATERIALS
- GLOSSARY

Preface

In the last phase of the mastering, the material is organized and encoded on a specific storage medium. During this stage, known as **Authoring**, a tracklist is compiled, and the auxiliary information is added. Although this process is typically associated with CDs, DVDs, and other digital media, sound material needs to be adapted to the type of medium used for playback, even for analog media. It is the case of productions on magnetic tape or, as explained in the subsequent section, of vinyl records

14.1 VINYL

Vinyl is a technological challenge for physics and mechanics. Such support is subject to different types of forces: centripetal, centrifugal, friction. The system provides for physical contact between the reading head and the groove. The vinyl production chain includes many steps such as engraving, chemical baths, metallic coating, washes, mold, each of which can influence the quality of the final product. A mastering engineer must be familiar with the features of this medium to avoid errors that may easily compromise the integrity of the transfer process from the master onto the vinyl.

14.2 CONSTANT ANGULAR VELOCITY, CAV

The rotational speed of vinyl can be either **45rpm (revolutions per minute)** or **33rpm** (33 revolutions and $1/3$ to be precise) on discs that can have a diameter of either 7" (17,78 cm) or 12" (30,48 cm). 45rpm discs complete a full rotation 45 times every minute. 45 revolutions per minute are completed both in the center of the disc and at the edge, see fig. 14.1.
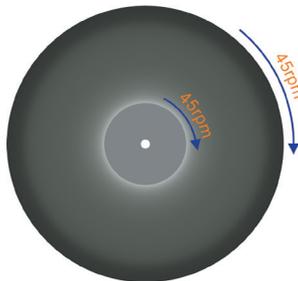

fig. 14.1 Constant angular velocity

This means that vinyl has a **Constant angular velocity**.
The disc is a circumference, and the angle to complete it all is 360°. Thus, the angular velocity of a 45rpm vinyl is 360° x 45/min.

If we consider the linear velocity, the one we can measure using the speedometer of a car, this becomes a whole other matter. The diameter of the disc at the edge is higher than that in the center. Provided the same angle has been completed, the head runs across much more space at the edge than in the center, see fig. 14.2.
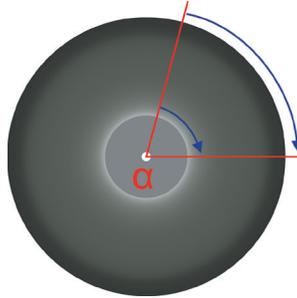
fig. 14.2 Variable linear velocity

The linear velocity of the head is higher at the disc edge than at the center. Considering that frequency response and dynamic range are directly proportional to the linear playback velocity, it means that sound has better quality at the outer edge of the disc. Such quality decreases as the head gets closer to the center. Thus, during authoring for vinyl, when creating the tracklist, one must take the timbral and dynamic content of each track into account. Placing a song with a high dynamic range, full of high frequencies at the end of one side affects sound quality. This is the reason why the essential tracks of an album are the first or second place of the tracklist, in an area of the disc where the linear velocity is high. The songs at the end of each side, i.e., those closer to the center of the disc, are generally less complex, with few instruments and reduced low-frequency content.

(...)

other sections in this chapter:

14.3 RIAA curve
14.4 CD
14.5 Why 44.100Hz?
14.6 Why 48kHz for video?
14.7 Constant linear velocity
14.8 SPDIF

Test
Glossary

15
RED BOOK

LEARNING AGENDA

PREREQUISITES FOR THIS CHAPTER
- CONTENTS OF CHAPTERS 1-14

TARGETS
WHAT YOU WILL LEARN
- LEARNING THE STRUCTURE OF A COMPACT DISC
- LEARNING THE RED BOOK PROTOCOL
- LEARNING THE CD-DA ENCODING
- LEARNING THE DATA AREAS OF A CD
- LEARNING THE SCMS PROTOCOL
- LEARNING THE ISRC UPC EAN CODES
- LEARNING THE DDP SUPPORT

SKILLS
- BEING ABLE TO MASTER A PMCD
- BEING ABLE TO CREATE A DDP
- BEING ABLE TO CREATE GHOST TRACKS AND BONUS TRACKS
- BEING ABLE TO MANAGE AN AUTHORING PROJECT WITHIN THE AUDIO MONTAGE

CONTENTS
- RED BOOK
- CODES AND SUB-CODES
- TRACKS AND SUB-INDICES E SUB-INDEX
- METADATA

LEARNING TIMES - Ch. 15
SELF-LEARNING
FOR A 200-HOUR INDEPENDENT STUDY SESSION:
- ABOUT 14 HOURS

COURSES
FOR A 40-HOUR CLASSROOM COURSE + A 80-HOUR INDEPENDENT STUDY SESSION:
- ABOUT 3-HOUR LECTURES + 2-HOUR FEEDBACK
- ABOUT 4-HOUR INDEPENDENT STUDY

EXERCISES
- INTERACTIVE EXAMPLES

TESTING
- SHORT-ANSWER QUESTIONS

SUPPORT MATERIALS
- GLOSSARY

Preface

In 1980 Sony and Philips agreed-upon physical, mechanical, electrical, and digital encoding parameters of the **CD-DA**, **Compact Disc Digital audio**. These specifications were then added in the **Rainbow Books**, a series of publications about Compact Disc standards and formats. The name derives from the color encoding used to label each standard. The **Red Book** sets audio CD parameters; the **Yellow Book** sets those of a **CD-ROM**, (**Read-Only Memory**, Data), the **Orange Book** those of a **CD-R**, **CD-Recordable**, the **Blue Book** those of a **CD-Extra** and an **Enhanced-CD**, etc… The red book was then added to the IEC international standards in 1987, with protocol ID 60908. The red book describes the digital parameters about the **Reed-Solomon** error correction system and the **EFM Eight to Fourteen Modulation** encoding. The EFM encoding divides each information in an 8-bit block. Each byte made up of 8-bits is then turned into a 14-bit encoded block (word), by adding auxiliary data that allow for a proper transition in pits and lands encoding on the CD surface. The red book includes information about **Sub-codes** from **P** to **W**, which are used for time positioning of each track, ancillary functions, barcodes metadata, **ISRC**, **CD-TEXT**, and copy protection system. The IEC 60908 standard was later updated with the addition of new features. Currently, it is possible to add images, videos, and MIDI data on an audio CD. These functions must be supported by the CD player or the software player. The red book sets the standard of a CD-DA at 74 minutes. It is said that this value was requested by Sony's vice president, Norio Ohga, in 1979, to allow Beethoven's ninth symphony, performed at the Bayreuth Festival in 1951 and directed by Wilhelm Furtwängler, to be published on a single CD. Today, it is possible to record up to 100 minutes of audio on a CD, even if this is not compliant with the specifications of the red book. Furthermore, playback of CDs more extended than 74 minutes is not guaranteed on all types of CD players.

15.1 CD STRUCTURE

Considering the encoding, i.e., the way data are organized on the support, the CD is divided into three main areas: Lead-in, Program area, and Lead-out, see fig. 15.1.
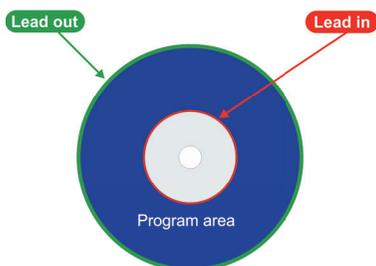

fig. 15.1 Information area on an audio CD

- The **Lead-In** section includes about 9 MB. It is located between 48 and 50 mm away from the center and includes the **TOC**, **table of contents**, the index of the tracks. The lead-in contains information on synchronization and laser focus
- The **Program area** section includes audio information and the error correction code, encoded in EFM
- The **Lead-out** section, which is 116 to 120 mm in diameter, does not include audio data, but a small part of the code, made up of zeros, that identifies the end of a CD.

The lead-in, program area, and lead-out make up the **Information Area**, **Session**, see fig.15.2.
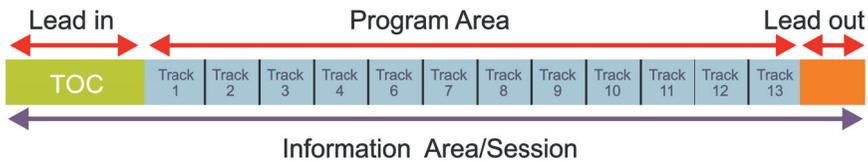
fig. 15.2 Session structure

The information area is located within a **session**. The CD-DA can contain a single session, while the CD-ROM can contain several sessions, recorded at different times. As explained later, with the introduction of the Blue Book, it is possible to insert both an audio and a data session on the same CD, thus allowing the audio session to be read both by CD players and computers, while allowing data to be read-only by computers.

(...)

other sections in this chapter:

16
MASTERING FOR STREAMING SERVICES AND COMPRESSED FILE FORMATS

LEARNING AGENDA

PREREQUISITES FOR THIS CHAPTER
- Contents of chapters 1-15

TARGETS
What you will learn
- Learning features of various audio compression codecs
- Learning target levels required by streaming platforms
- Learning which types of metadata can be used

Skills
- Being able to export files in various formats
- Being able to analyze compressed rendering

CONTENTS
- Codecs
- Loudness in compressed files
- Metadata in compressed files

LEARNING TIMES - Ch. 16
Self-learning
For a 200-hour independent study session:
- About 13 hours

Courses
For a 40-hour classroom course + a 80-hour independent study session:
- About 3-hour lectures + 2-hour feedback
- About 4-hour independent study

EXERCISES
- Interactive examples

TESTING
- Short-answer questions

SUPPORT MATERIALS
- Glossary

Preface

The processes we have been dealing with so far involved using linear PCM audio. That makes sense for professional production, where the final product quality is related to the resolution of the original material. Most of the music market is based on compressed audio formats playback and marketing. **AAC**, **Mp3**, **WMA**, **Ogg**. This kind of algorithm is called **Lossy**, **i.e.**, **with data loss**. As the name suggests, rather than compression, it is more of a data *resizing* that can be achieved by removing part of the information, which cannot be restored anymore. This way, we can reduce, depending on the level of compression, the amount of data up to a hundred times compared to the original file. The resulting sound quality is inversely proportional to the level of compression and the resulting bitrate. The audio of a CD-DA has a bit rate of **1411,2kbps, kilobits per second**.

A *medium-quality* Mp3 file has a bit rate of 128 kbps, and a *high-quality* one has a bit rate of 320kbps. A high-quality AAC file can go up to 320 kbps, but iTunes uses 256kbps. The purpose of lossy compression is using less storage space on storage media and servers, as well as using fewer data during streaming on the web. Multimedia providers use a compressed lossy audio format. iTunes, YouTube, Soundcloud, Spotify, to name a few. There are **Lossless**, **i.e.**, **with no data loss** compression algorithms. These reduce the size of the file without removing information. A lossless file, once decompressed, generate the same information on the original file. The most known lossless algorithm is the **ZIP**. There are specific lossless algorithms used for audio, like **FLAC**, **ALAC**, **WavPack**. Unfortunately, lossless codecs do not provide for high compression ratios; they are rarely used by multimedia services providers, which in turn prefer using the lossy codecs for speed and storage space. The lossy compression of a digital signal, just like all the digital processes, affects the timbre, the dynamic range, the volume, produces distortion, and is everything but *transparent*. The quality of an uncompressed signal is undoubtedly higher than a compressed one. We must put up with the fact that the music market requires specific kinds of supports and formats. We must optimize the conversion process of the mastered linear material in a compressed format.

16.1 MP3

MPEG-1 Layer 3, better known as MP3, is a lossy compression algorithm that features bitrates from 32 up to 320kbps. Developed in 1989 and updated in 1992, it even became a standard under the name **ISO/IEC 11172-3**. Its highest-quality version has a lower bitrate than a CD WAV PCM by 4,5 times. The compression algorithm supports files with a sample rate of 32, 44.1, and 48kHz, at 16 or 24-bits. It uses ID3v1 and ID3v2 metadata. Although a 320kbps MP3 can be a good compromise in terms of quality, for many multimedia services providers, a 128kbps file can be already considered *good…* Much information on the original file is removed with data compression of 11:1. This occurs both in the spectrum and dynamic range. Especially for what concerns the inter-sample peaks and clipping, a loud file can be acceptable in WAV format and distorted when converted to MP3. For this reason, we must never exceed an I value of -16LUFS, an **LRA** around 8LU, and a maximum **dBTP** value of -1, -2dBTP. Wavelab Playback Processing section features an Encoder Checker plugin, see fig. 16.1.
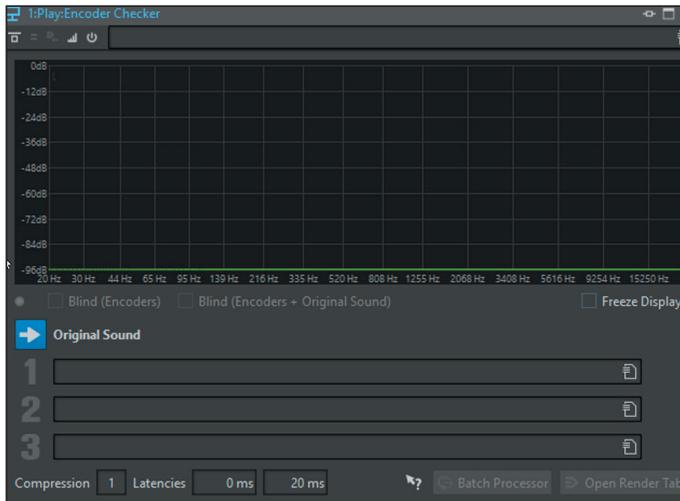
fig. 16.1 Encoder Checker

The plugin allows for real-time listening of the results of 3 different conversions. There is a *Blind* mode, which randomly switches between the various codecs and the original file. This way, while listening, the opinion is not influenced by knowing the kind of codec.

(...)

other sections in this chapter:

16.2 AAC, Advanced Audio Coding
      Exercises
16.3 Ogg
      Exercises
16.4 Flac, Free Lossless Audio Codec
      Exercises
16.5 iTunes
16.6 Soundcloud
16.7 Spotify
16.8 YouTube

Test
Glossary

Marco Massimi

MITB Mastering in the box

Concepts and applications for audio mastering

Topics

Production chain – Background history and meaning of mastering – Types of processing - Metering – Analyzing material to master - Filtering, DC offset removal and timbral changes – Spatial processing – Dynamics processing – Distortion and saturation – Noise reduction - M/S mastering - Stem mastering – Surround mastering - Dither - Authoring - Red book - Mastering for streaming and compressed formats

"The possibilities to learn and trust from those experiences are sadly limited now, but the desire to find out more about certain issues is still there, especially the mysteries of mastering. Digital requirements are evolving, Vinyl Mastering is making a reassurance, all good, but where can I find a reference to a vast array of questions and endless curiosity, technical or otherwise? Well, you have it in hand laid out in diagrams and formulas for the technical, discussions, and advice for the curious and a wealth of ideas and insight into what is now generally called "in the box." Personal taste and feelings are as important if not better regarded than rules and tech jargon, but wouldn't it be nice to be assured that you can access all of these in one place. Happy reading."
from the Foreword by **Steve Lyon** (Producer, Mixing Engineer)

This book deals with digital sound processing for audio mastering, entirely inside a computer, **ITB**, **In the Box**. We chose to deal with the ITB mode mainly because, nowadays, thanks to the computing power of our computers, it is possible to achieve a high level of accuracy and calculation. Considering that these computers are now within anyone's reach, such a text may help the reader make things that only few mastering studios, provided with expensive equipment, could have done before.

MARCO MASSIMI sound engineer, mastering engineer. Lecturer in Electroacoustics at the "Licinio Refice" Music Conservatory of Frosinone, lecturer in Post-Production and Mastering at the "Tor Vergata" University of Rome. He holds masterclasses and press conferences in Italian and European universities. He worked as a sound engineer in RAI Radiotelevisione Italiana for: Gino Vannelli, Barry White, Michael Bolton, Sting, Elvis Costello, Mother Earth, Al di Meola, Maria Carey. As a freelance sound engineer, he worked in Italy and abroad with: Gloria Gaynor, Harbie Hancock, Peter Erskine, Tom Harrel, Enrico Rava, Gary Allen, Dave Holland, Bill Frisell, Gonzalo Rubalcaba, Scott Henderson, Hiram Bullock, Mike Stern. As a mastering engineer, he worked for Ennio Morricone, Céline Dion, and many more. He studied Sound Design for Electronic Music at Berklee College of music, Boston USA. His works as a Sound Engineer and Mastering Engineer have been published by Sony, Venus, BMG, RAI trade. He played bass and double bass for Ennio Morricone, Solomon Burke, Michele Piovani, Alicia Keys, Patty Austin, Michael Bolton, Gabrielle, Quincy Jones.