Friedemann Tischmeyer

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Internal Mixing

How to create a professional mix on your computer – a systematic approach

- *Create punchy, powerful, clear, and three-dimensional mixes on Mac and PC-based digital audio workstations*
- *How to work best with external DSP power, such as PowerCore, UAD-1, and SSL Duende*
- *Numerous exercises and extras on the DVD; For example: How to use compressors*
- *New chapter about analog summing units including a 650 MB OMF file to compare on your own*

SECOND REVISED EDITION

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Tischmeyer Publishing

Second, revised edition 2008

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Cover Design:	aim Werbeagentur Hamburg, Michael Prahl
Layout:	Mott Jordan www.mottjordan.com
Graphics:	Friedemann Tischmeyer and Gregoire Vanoli
Copyeditor:	Leina Gonzalez Baird
Proofreaders:	Omid Bürgin, Namin Nooman
Production:	Media Print GmbH, printed in Germany
DVD-Production:	optimal media production GmbH, manufactured in Germany

www.tischmeyer-publishing.de
www.proworkshops.de

ISBN: 978-3-9811217-1-1

Foreword

Dear Reader,

WELCOME ABOARD! We are about to embark on a voyage through the entire mixing process. This book will give you easy-to-understand suggestions on how to systematically carry out a successful mix.

Very often people ask if computer-based production can provide the same punch and especially the same degree of space that a production made with classical tools and the best outboard equipment. Step by step, I will explain the necessary tools and techniques for professional mixing with computers. We will look at how mixing was done with tape-based recording technology, and will examine the supposed advantages and disadvantages of these production methods while applying this to the world of digital production.

On the way, we will discover a treasure of inspiration for improving our current working methods.

Beginners, DJs, audio engineers, musicians, producers, and audio engineering students will all be able to use the ideas and suggestions in this book for their productions.

Here's a suggestion for all those whose highest priority is the best possible conservation of their ideas and songs: In order to make progress in mixing, please plan on finishing your projects within a predetermined period of time. This is the only way that you will be able to look back and see how your work methods and your hearing have developed. On the other hand, if you let yourself be a slave to the possibilities of total recall incorporated in today's digital audio workstations (DAW) by never really finishing your projects, then it will be much more difficult to see any progress in your mixing techniques.

Concerning my Background and the Creation of this Book

I AM A TRAINED guitar and bass player and have learned everything I know about audio engineering on my own. As both a musician and as an engineer, I have been very lucky to have worked with a variety of experienced engineers; I have used these experiences and opportunities to further increase my knowledge. Later I had a large 48-track studio with a Studer 2" tape recorder and a Trident analog mixing desk/console, along with a large quantity of outboard equipment. During this time, I gained quite a bit of experience working with tape – quite a bit different from the predominately individual or mini-team working methods that are characteristic of today's production methods using home computers. Before the digital revolution could wipe out my studio, I was still able to sell off my equipment to concentrate fully on the new computer-based techniques in a smaller studio, spending more time working on my own projects. I constantly tried to achieve results of a quality equal to that attained using the familiar analog techniques. In the beginning, it was not easy to attain the same quality with digital workstations as analog technology. Therefore, I began to work with manufacturers and developers of software-based workstations and plug-ins, and always went to the limits of what was technically feasible.

Like my mastering book "Audio Mastering with PC Workstations," this book has "organically" grown from the many workshops I have given in my studio and at various educational institutions. Workshops given to small groups are ideal for the mutual exchange of information and experiences. They helped me to fine-tune pedagogical concepts and to tailor to the participants' needs – to your needs. This was the foundation of both the book and the DVD series I created based on the books. Theoretical issues, which were inappropriate for the DVDs, along with all information about quickly evolving matters such as plug-in descriptions, are reserved for the book, which will be revised in a cycle of approximately two to three years. On the other hand, the tutorial DVDs focus on practical working methods illustrated by numerous audio examples. The book's accompanying DVD-ROM contains audio examples, exercises, and demo versions of plug-ins from many different manufacturers.

I would like to thank the owners of copyrighted material who have allowed me to reproduce parts of their work for the audio examples. There are also single-track excerpts that are suitable for exercises concerning compressors and EQs.

Please do not use any excerpts as samples for productions as their use is not allowed without express permission from the copyright owners.

The knowledge contained in my books therefore stems from pure practical user experience, from one user to another. Wherever appropriate, it is supported by extensive research and studies.

Enjoy yourself and have fun applying the following techniques to your work!

Friedemann Tischmeyer

Working with this Book

NEARLY ALL OF THE INFORMATION and suggestions in this book are cross-platform and are therefore equally valid for both PC and Mac users; exceptions will be indicated. The same is the case for mixing strategies with analog mixing consoles and computer-based systems. Nevertheless, the focus of the book is clearly based on computer-based mixing.

For the sake of clarity, screenshots are taken from only one sequencer program. Because I personally work with Steinberg's Nuendo and 99% of the features that concern us also are included in Cubase, I have used screenshots from Nuendo.

To avoid unnecessary complication, I will not discuss other sequencers. Every professional sequencer or hard disk recording application has the necessary basic functions for our work and it will not be difficult for you to apply the workflows that we describe to your own software environment. My choice is not a recommendation concerning the very similar qualities of the various sequencers or DAWs available today.

In this book we will concentrate almost entirely on systematically organized techniques for interpreting and answering artistic requirements. The right sound and how to obtain it is our main task, provided you already have a clear idea of the sound aesthetic and the artistic flow you want.

The system involving the complex, intuitive, and creative process of music mixing can be divided into two large categories: absolute processes and relative processes.

Absolute processes refer to recurring steps or rules, which should be carried out in a specific manner and order. For example it makes sense to begin with automation after finishing the static mix.

Relative processes also follow rules, but are not dependent on being done in a strict order. Working with EQs, while independent of the mixing process order, still follows the same rules. Relative process steps can therefore occur at different points in the mixing process or even more often than once. It will be self-evident which steps are relative and which are absolute processes. With the information in this book, you will be able to correctly navigate yourself through the entire mixing process.

After exploring the individual areas in depth, the section “Workflow Overview” represents the mixing process as a timeline in order to give you an overview of the “absolute” process workflow.

Contents

EXTRACT

Chapter 1: The Three Phases of Classical Production – A Retrospective

USE THE ADVANTAGES OF BOTH analog **and** digital work methods to achieve better results.This chapter adapts the experiences of classical mixing production in a tape-based studio to the predominating work methods in today's modern computer-based studios. It helps to take a brief look into the past to see what has changed and if some of the traditional methods can be applied today for improving quality.

Traditionally, music production can be divided into three basic work phases that overlap when working with digital audio workstations (DAWs): Recording – Mixing – Mastering.

Phase 1 – The Recording Session

The producer's job is to make sure that a large number of tracks are filled with the most musically relevant content possible. Ideally, producers should not have any personal relationships with the composer, songwriter, singer, musicians, arranger, and engineer, so that they can remain neutral with regards to the production. They need to have a strong imagination in order to record individual tracks, which – with the exception of a few details – are ready for the mix.

Good sound is created in front of the mic, not behind it!

When working with analog mixers and analog multitrack tape recorders, well-organized track planning, as well as recording at consistent levels, was necessary in order to be able to jump quickly from one song to another during an overdub session. This way, a separate production, or headphone, or rough mix could easily be made by simply adjusting channel faders, panning and headphone send knobs.

This technically driven need for highly disciplined organization is diametrically opposed to the methods of non-destructive computer-based workstations. The advantages of a practically unlimited number of tracks, loop recording, and total recall of all parameters are all advantages of modern DAWs, as long as we are consistent in our use of all these possibilities. For example, event-based level adjustment (which is possible in most audio sequencers) is a quick and helpful way of correcting recording level inconsistencies. Despite all of its advantages, loop recording often leads to mediocre results: "there must be a good take in there somewhere..." The freedom of non-destructive loop recording can become a forced labor marathon for sound editors. If you've already worked professionally with tape media or can mentally imagine the process of tape-based recording, then you can apply a number of valuable tips to your work with DAWs.

Improve your Decision-making Capabilities

The classic method of working trained the ability to decide whether a take was good or bad. Because of a limited number of tracks and the resulting necessity of recording over previous takes, decisions had to constantly be made regarding the quality of each take. To avoid unnecessary searching and editing before things got out of control, high standards should also be set for nondestructive editing. For example, while recording a lead vocal track, the workflow could be as follows: Open a recording track (for listening to the input signal, recording, and punching), a keeper track A (listening) for the "first choice" and a muted "spare parts" track B. During loop recording, categorize running the takes in writing as A, B, and C = trash. Another option is to limit the number of loops to a relatively small number to be judged mentally as you go.

Improved efficiency with better decision-making

Right after the recording process, the takes are divided between the keeper track (A), the "spare parts" track (B), and the trash (C), so that the recording track is empty for the next take. The keeper track is for listening and should have exactly the same effect settings (insert and send) as the recording track. Other than small details like manual tuning, sibilant and breather editing, etc., the A track should sound good after the session. This makes things much more enjoyable for the musician, since he or she can have a very good idea of the results and can go home with a good feeling about the day's work. Instead of being overwhelmed by the idea of editing a mountain of disorderly takes, you can open the project at any time – even weeks later – and still have immediate access to the tracks that may need editing. During the editing process, the spare parts track (B) serves as a reserve for replacing details that you might not have heard

during recording (pronunciation, consonant endings and sibilants, unwanted accents, or noises).

Without systematic pre-sorting, you can be fairly sure that after 60 takes of a loop recording, on the next day you will not remember whether or not THE take was the 47th, since after the 30th you were probably already somewhat dazed. In any case, without pre-sorting, your work is laborious, time-consuming, and certainly not much fun.

But by trusting your decision-making abilities, when you finish editing the track in question, you can make space on your hard disk by completely emptying the spare parts track (B). Only when you decide to close off theoretical possibilities can you move forward unhindered.

Building Confidence in Rhythmic Hearing

Back in the days when tape machines were still common, you had to use your ears to know if a track grooved or not after overdubbing (for example an e-bass recorded onto drums). Moving tracks was very time-consuming and only in extreme cases could a MIDI-synchronized sampler help with timing problems.

Nowadays, a graphic interface and therefore the eye, is allowed to judge whether or not a track grooves well or is rhythmically tight. To allow the ears to do the work they were meant to do, reduce the visual waveform display to a minimum or eliminate it completely so that you depend 100% on your hearing. After all, later we want to ***listen*** to the music, not watch it! Once the ears have decided that a sound comes too early or too late, you can switch the waveform display back on with a keyboard shortcut and go back to the graphic editing mode. Most DAWs let you change screen layouts with keyboard shortcuts that can be used for this way of working. Another option is to simply close your eyes while listening for rhythmic precision and to trust your ears.

A Few Good Tracks are Better than Many Mediocre Ones!

Another advantage of the limited number of tracks on tape machines was that bad content could not be compensated by an overly inflated number of tracks. If 20 tracks cannot transport emotion and do not create a particular sound, then it is highly unlikely that the solution will be found in more tracks.

Clearly there are many reasons why working in analog studios led to more focused, disciplined work that involved spontaneous and quick decision-making, along with a great deal of imagination. The convenience – inherent in DAWs – resulting from the lack of clearly-defined production phases has both advantages and disadvantages. Using Cubase, Logic, Pro Tools, DP, or Sonar becomes a blessing only if we work more systematically.

The "analog" working methods had the advantage that in the second phase – the mixdown – a good basic sound had already been supplied; the mixing engineer had a number of fundamental musical aspects already laid out for him. One point is as valid today as it was in the days of analog recording: the better the recording and the "front-end" – mics, pre-amps, compressors, and other equipment used – the more easily a signal integrates into the mix later on. The popular saying "we'll fix it in the mix" should not be synonymous with musically or technically inferior recording, even when we are tempted by the endless correction possibilities of modern DAWs.

Phase 2 - The Mixdown

In the past, a 2-inch tape was frequently brought to another studio or sent to a mixing engineer. The mixing studio was usually outfitted with high-end outboard equipment and mixer automation. Phase 2 began with all tracks being laid down next to each other without panning and all faders set according to taste, in order to get a first impression. Masking tape was used to label the tracks and a long tape roll from each song was hung on walls and doors until the production was finished. Today, the equivalent process would typically be done by importing an OMF (Open Media Framework) file, where individual .wav or .aif files are brought into the arrangement without EQs, insert effects, or level and panning information.

The advantages of division of labor here are important: because the sound "grows" throughout the entire computer-based music production process, by the time the mixdown begins, a production is often already halfway finished.

That is not necessarily an advantage!

The last fine-tuning can turn into a rocky path because of awkwardness or a lack of systematic working methods. It is important to keep questions in mind such as: what is the reason for this send effect? Which strategy is used for panning? What is my mixing concept?

Being deeply involved in the production can hamper a fresh and systematic approach to the mixing process. Very few people would go to the trouble of taking out all plug-ins and setting all levels, pans, and EQs to zero and start everything from the beginning. A further disadvantage in being both the recording and mix engineer has to do with something we will mention later: the mix concept. Often, when creating a mix concept, the mute button is a very important tool. The producer, who has been involved since the very start, has a difficult time muting tracks that were created as the result of hard recording and editing work. This is why I recommend creating teams with friends and colleagues, and occasionally changing the roles of mixing engineer and producer with that of the client; taking on the role of accepting or rejecting final results while delegating the actual mixing process. Doing this will give you new ideas and teach you a great deal.

The mix is done when the mixing engineer creates a stereo master. In earlier days, this involved a DAT recorder fed by an Apogee converter with onboard limiting. Often, an analog limiter – like the Urei 1178 LN – was used to cut out peaks in order to make the best use out of the 16 bits of the DAT recorder. Now that we have the luxury of 24 or 32-bit mixdowns (bounce/export/render/apply), you do not have to dynamically limit the master with a bad native converter. The 32-bit floating-point files provide enough headroom to eliminate a limiter completely, since even overs can be handled cleanly. The 24-bit files can be protected from overs using a brick wall limiter, because the limiter only kicks in when a peak is detected. Generally, dynamic processing and other level processing steps should be left to the mastering engineer, who takes over in phase 3.

How to Achieve a Quality Mix

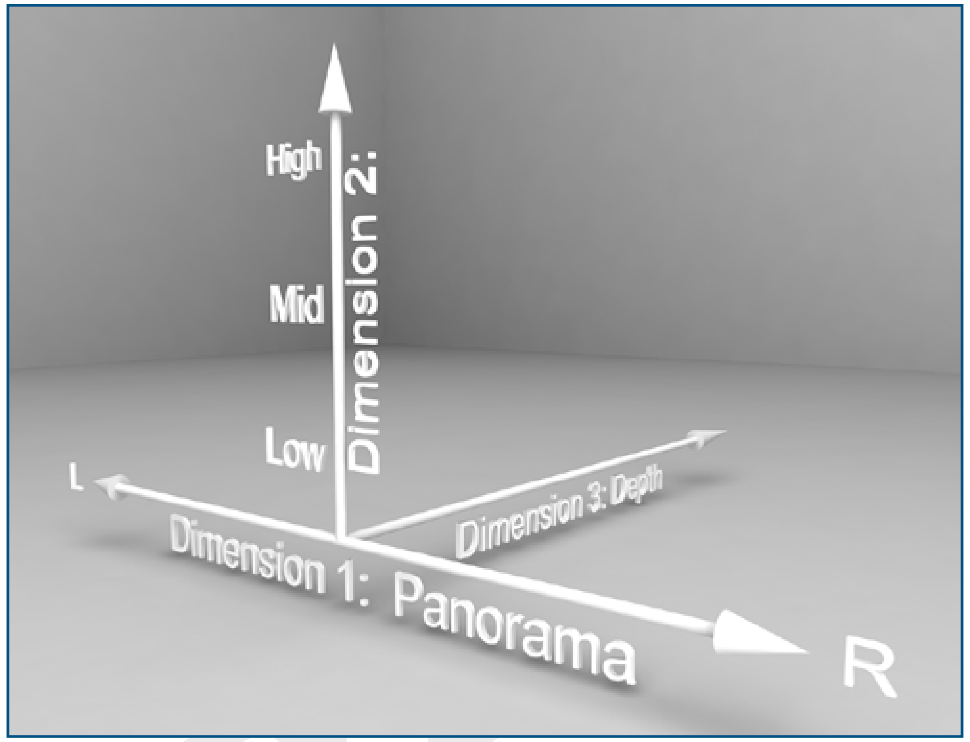
An additional **artistic aspect** is dramatic form and can be especially achieved by muting or using special effects.

During the recording stage of the production, many tracks are filled with content that is to be later sorted during the mixing process. Nothing is more boring than a song where all instruments are audible from beginning to end. By intelligently muting individual tracks a song can become interesting. For example, if the vocals are good enough you can even create a cappella passages with this process.

Mixing Goals

The goal of a good mix is a warm, clear, deep, and punchy sound, where all events

The secret – or fundamental skill – necessary to obtain a good mix lies in intelligently distributing all events in the three spatial dimensions: width, height, and depth!



The Three Dimensions

The Three Dimensions:	Aspects	Sub-Aspects
1. Width (L/R Panorama):	Panning	Basis Widening
2. Height (Frequency Distribution):	EQing/ Level	Compression
3. Depth (Front-Back Space)	Reverb & Delay	EQing Reverb & Delay

are clearly defined, or correspond to the genre and sound aesthetic. Critically examine every event – other than a quiet background pad for giving warmth – that does not have a clearly defined place in the mix to see if it would be better to do without it. Less is often better!

What conclusions can we make from analog techniques for working with DAWs?

- When you mix your own song and you are not happy with the results, make a copy of your project, remove all insert and send effects, and put all panning to the center. Start right from the beginning with a clear mixing strategy (see following chapters).

- Ask a like-minded friend or colleague to help by switching roles (client/engineer).
- Give yourself only a limited amount of time. In the past, it was often only possible to have one night to "sleep on" mix decisions – maybe listening on a different set of monitors – with the sole possibility of coming in the next morning, before the next production day begins, in order to make a few minor changes. Afterwards the patch bay and mixer settings would be irreversibly changed. Do not be led to collect a pile of unfinished projects just because of the existence of the total recall functions in modern DAWs. It is much better to decide to close mixing projects which can be saved and wiped from the hard disk, so that you can realize that in a year's time you have gotten better! Then, if required you can mix another version. A mix should be 90% done after four hours of work. The rest is fine-tuning and takes the largest amount of time (1 to 2 days), but of course this can vary largely and depends on the person.

The mixing process is a continuous chain of decisions!

- The mixing process consists of a continuous chain of decisions. The ability to quickly make decisions is crucial for maintaining an efficient work pace.

Delivering the Mix Master to the Mastering Studio

As long as you are working on a **native basis** – in other words, with the computer and without the aid of an external analog mixing desk – the mix master should be delivered on CD-ROM as an undithered 32-bit floating point .wav or .aif file. If your DAW does not provide for the 32-bit floating point format, use the 24-bit integer format. Since the 24-bit integer format cannot handle overs (level values exceeding 0 dB), it is essential that you use good brickwall limiting.

To keep the BLER (Block Error Rate) values of the medium (CD-R or DVD-R) low enough, you should use a low speed to write your data-CDs or DVDs. In the past, the writing speed for best BLER values were indicated on the blank media, but today this is seldom the case.

The BLER value is a statistical error value of digital media. The Red Book specification stipulates the highest acceptable value.

For comparative listening, you can create a second file, dithered to 16 bits in the last position of the master insert of the virtual DAW mixer.

If you are working with a **digital mixer**, it is a good idea to create the greatest possible bit depth – in this case 24 bits – when creating the mix master. Because the S/P-DIF and AES/EBU digital transfer formats are limited to 24-bit integer bit depth, it is impossible to process files on a 32-bit basis with external equipment. The 16-bit DAT recorders are to be avoided nowadays; the DAT format is no longer suited as a mastering medium. In addition, very few project studios have high-resolution tape machines. For mixing down using an external digital mixer, it is best to copy back over to the DAW. Here it is important to make sure that the synchronization (wordclock/houseclock) settings are correct.

For good sound in the digital studio using a houseclock, the rule is: **converters clock themselves internally, are clock masters, and drive the clock distributor**. (If you own several different converters, then the most important converter – that the mic and line inputs are fed through – should be internally clocked.)

If they are in the chain, devices like the TC Finalizer should be set to bypass or used only as "technical" limiters for eliminating occasional peaks. This does not contradict the high quality of the Finalizer, but avoids unnecessary compression, which can be difficult to correct in the mastering process.

Be careful! When you are using S/P-DIF with your digital mixer, make sure that the processing chain supports 24-bit word depth. Some audio interfaces – especially low-cost models – are limited to 16 bits. Such devices should be replaced with 24-bit audio interfaces if they are to be used for mixdowns. WaveLab provides a bit-depth metering function that shows the actual allowed bit depth.

If you are working with an **analog mixer**, you can make a ½-inch master parallel to the digital master. These formats are still accepted in some mastering studios in North America. In Europe large mastering studios also support such formats. Beware of the country-specific measuring standards of analog machines (USA: IEC/Germany: NAB) and be gentle with tape saturation. With a digital mix master, high-quality 24-bit converters should be used. These also serve as clock masters for the DAW being used for recording. 24-bit or 32-bit floating point files should be delivered on CD-ROMs. If reliable digital metering is available, I recommend keeping pop and radio music at an average loudness of not more than -14dB/RMS during loud passages. This ensures that the mastering engineer's job can remain enjoyable. Over-compressed masters are very difficult to work on and are difficult to shape. When mixdowns are louder than -14dB/RMS, it is difficult to correct mixing mistakes.

Please see Chapter 2: Using Metering for Monitoring the Three Dimensions.

Here is a summary of the most important points for delivering a mix master to a mastering studio:

- Highest possible bit depth; Cubase & Nuendo: 32-bit floating point files; otherwise use 24-bit files.
- In situations where the computer uses internal 32-bit floating point processing, but you must go to 24-bit files for exporting, use either a brickwall limiter without dithering or a simple limiter and dither down to 24 bits.
- Do not dither 32-bit files.
- Use CD-ROMs, DVD-ROMs (slow writing speeds), hard disks, or memory sticks as transport media.
- For CD productions, use a sampling rate of 44.1 kHz. If conversion is necessary, use a high-quality sample rate converter (SRC) with internal oversampling. If this is not an option, leave the conversion to the mastering studio. Sample rate conversion is a complex and CPU-intensive process, which can only be done with oversampling (multiplication of the sampling rate) in order to prevent rounding errors.
- Before mastering, avoid fades at all costs unless they are musically arranged into the song. If fades are run through dynamic processing during mastering, pumping and digital artifacts can result.
- Do not cut beginnings and endings. Many mastering plug-ins require short lead-ins so that the "predict" function can work. Without the lead-ins, artifacts might be created during the processing phase. When marking areas to be bounced or exported in the arrangement window of your DAW, it is smart to leave small lead-in times instead of selecting the area start point exactly at the beginning of the song.
- When using analog mixers, a lead-in can be used as a fingerprint for possible de-noising during mastering.
- Label track files according to their numerical order on the album. (For example, 01-32Bit my song.wav.)
- Give the mastering studio a processing wishlist along with reference tracks, if desired.