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Distorted References

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Two books published by Berklee Press ("a publishing activity of Berklee College of Music," www.BerkleePress.com), one written by David Franz (engineer and Berklee College graduate student) and the other by Daniel M. Thompson (Berklee College of Music Assistant Chair of Music Production and Engineering), purport to provide enough information to enable effective setup and use of a home or professional recording studio. I give each a failing grade and hope they are not used in the classroom.

Recording and Producing in the Home Studio: A Complete Guide (©2004, ISBN 0-87639-048-3, 236 pages, $25) is Franz's attempt. Franz introduces the book as "for artists, songwriters, bands and aspiring engineers and producers who want to better understand how to engineer and produce their own recordings." This book is of limited use to a musician who is starting to produce and create his own recordings, and who wants an overview of what topics he needs to learn about, plus how they interact. It reads as though written by a musician whose ego led him to write, but in spite of his engineering degree seems without the depth of knowledge, or the ability to teach that knowledge, that is necessary to fully inform the budding musician/producer/engineer.

The most useful and informative part of the book are sections 2-4 (chapters 4-14), which address practical operational techniques useful in producing, recording, and mixing. Franz explains the role of the producer versus that of the engineer, acknowledging that in many home studios, the musician does it all. He suggests microphone selection and techniques, channel assignments, various traditional mixing board functions, how to take advantage of MIDI, editing techniques, and mastering. These seem to be based on his preferences with conventional mixing boards. He doesn't make clear that there are other ways to approach each task that might be better suited to the way other musicians work or think. The novice musician could become confused if he uses this book as a tutorial and his equipment doesn't offer the same capabilities.

The book fails when it comes to science and engineering (true electrical engineering, not the art of operating a mixing board). Some examples:

- (page 10) "Voltage is a measure of power" is a misstatement. Voltage is only part of the equation. Changes in voltage only directly relate to changes in power when other factors are fixed.
- (page 10) "dBV is the unit of measure for power ratios based on voltage" is also incorrect. dBV is a decibel measure of voltage relative to 1.0V RMS.
- (pages 10–11) "Line level signals are either −10dBV or +4dBm (dBm is another unit of measure for power ratios based on milliwatts)" is at best partially correct. dBm is defined as relative to 1mW into a 600Ω load impedance. Without the 600Ω load impedance, dBm is meaningless (and erroneously used throughout the book). dBu is the proper unit, as it is referenced to 0.7746V RMS, which is the voltage that would yield 1mW with a 600Ω load. Other than some microphone preamps, I am aware of no equipment for audio recording, mixing, or reproducing that offers a 600Ω input impedance, thus making dBm an incorrect unit of measure. In addition, -10dBV and +4dBu are reference levels used for design purposes and to determine reasonable compatibility between an output and an input, as well as to track the gain structure of the devices. They are almost never the absolute levels of music signals, even averaged over time.
- (page 12) " . . . 0dBV, which is actually an industry standard of 0.775 volts." 0dBV equals 1.0V RMS. 0dBu equals 0.7746V RMS.
- (pages 13–14) "If you plug a low-powered, low-impedance device (like a microphone) into a high-impedance input, the lo-Z signal from the mic might meet too much resistance, and you often won't be able to get enough gain for a healthy recording level, hence the need

for preamps." The problem has nothing to do with feeding a low-impedance source into a high-impedance input, but is due to the low voltage output capability of the source versus the much higher voltage level required by the line-level input of the recording or mixing equipment. The microphone preamplifier's sole purpose is to raise the voltage level from the low output of the microphone to the level needed by the line-level input.

• (page 21) "If you want the input mic preamp signal to go only to the compressor and not back to the recording device directly, break the normal by patching the top jack signal to the compressor and then inserting a patch cable into the bottom jack input connected to nothing." ["Breaking the normal" is the author's phrase for interrupting the default path within the patch bay from the output jack back to the related input jack by inserting a plug into the patch panel input jack.] You should never leave any cable end unconnected. It could touch a source of voltage and cause damage. It will definitely act as an antenna and can add noise to the signal. At best, plug a low-impedance dummy load into the unused input (bottom jack) to prevent noise pickup.

• (page 22) "Sound modules (also synthesizers and samplers) translate MIDI data and turn it into audio." They do not. They respond to digital control commands in MIDI format and activate circuitry in the devices to create audio.

In the section discussing sampling rates and digital recording:

• (page 38) Figure 3.10 alleges to show "The waveform created by a cymbal ringing at 30kHz," yet shows a pure sine wave. Then figure 3.11 claims to show "the waveform sampled at a rate of 44.1kHz," yet the samples aren't at equal time intervals, and *averages* 110kHz sampling rate, with 11 samples shown over three waveform cycles.

• (page 38) Figure 3.12 alleges to show digital levels for a two-bit and a three-bit system. However, a two-bit system can represent only four levels, and the diagram shows five levels. A three-bit system can support eight levels, yet the diagram shows nine levels.

Under mastering:

• (page 226) Figure 14.3(b) claims to be "the mastered mix waveform. Notice the overall signal level is higher [compared to figure 14.3(a)], yet there is no clipping." From the figure, you really don't know if any clipping occurred. There appears to be no digital clipping of the waveform during the final mastering stage, assuming the window frame top and bottom lines represent clipping levels, but the fact that the overwhelming majority of the waveform is a solid gray rectangle with essentially flat top and bottom edges implies that the waveform was hard limited (clipped) prior to being mastered for the release medium, such as a CD. In other words, it might have been clipped and set to peak less than a dB below the CD's 16-bit clipping level. Much more waveform detail would need to be displayed to convince me it wasn't clipped somewhere in the processing.

Not all is lost:

• (page 35) Franz presents a very nice analogy of digital audio sampling to taking timed snapshots of the sky: "If you set up a camera to take a picture of the sky once every hour, you can follow the weather patterns in a very rough way. . . However, if you took pictures every second, you would see much more [time-related] detail. . ."

It seems telling that the only testimonials on the back cover of this book are for Franz's earlier book on the use of Pro Tools.

This isn't true of the other book, *Understanding Audio: Getting the Most Out of Your Project or Professional Recording Studio* (©2005, ISBN 0-634-00959-1, 348 pages, $25), for which Elliot Scheiner, Don Puluse (former Dean of Music Technology, Berklee College of Music), and Ken Pohlmann (Director of Music Engineering Technology, University of Miami and writer for *Sound & Vision*) contributed glowing recommendations. There is enough wrong with this book that they should be embarrassed. Some examples:

• (page 13) "Our hearing mechanism transduces the vibrations [of air] into electrical impulses that are sent back and forth between the brain and the inner ear." I'm not convinced that "transduces" is the right term for what the basilar membrane does. You usually think of transducers as converting a waveform in one type of energy into its analog in another type of energy, e.g., electrical waves to mechanical vibrations. The ear *transforms* the time-domain signal into multiple windowed frequency-domain signals of peculiar, overlapped-filter nature, in a highly nonlinear fashion. Also, the implication of the phrase "back and forth" is misleading. While there are electrical impulses sent from the brain to the ear, they are not those directly generated by the inner ear (cochlea) from sound, but are feedback control signals generally understood to control the transmission efficiency of the middle ear, thereby applying a limited form of dynamic compression. The signals from the cochlea are unidirectional.

• (page 17) Regarding the design concept of condenser microphones: "The capacitance (a form of electrical resistance) of the element is determined by the distance between the two plates and the voltage across them." The voltage across the two plates has no effect on the capacitance, which is only based on the area of the plates, their separation, and the dielectric between them.

• (page 22) "Often, an external mic preamp will be used instead for its superior sonic characteristics, particularly if it is tube-based or 'vintage'." The external tube-based microphone preamp is not necessarily a superior device, but certainly one that provides specific sonic character, usually in the form of harmonious distortion and soft clipping effects, plus higher noise than almost all properly designed solid state microphone preamps—transformer or transformerless.

• (page 65) Although the earlier portions of chapters 3 and 4 are written for multi-track recording, the "Master Section" is written for two-channel mastering. It should be generalized for 2- through 5.1-channel mastering, as these are the primary formats currently used. Missing is a section explaining the limited purpose behind the ".1" Low Frequency Effects (LFE) channel, explaining how it is different from a subwoofer speaker, and that it isn't needed for almost all music recordings since the five main recording and transmission channels are all full-bandwidth; the playback system's bass management should control which speakers get the bass, *not* the original recording; the recordist shouldn't design a high-quality recording based on the presumption of limited playback system capabilities.

• The diagram on page 102 shows an instrument string with the full length of the string as a half-wavelength first harmonic (same as the fundamental) vibration, and labels the second harmonic vibration as the first, and the fourth harmonic as the second, never showing the correct third harmonic.

• (page 114) "The sounds that we call musical are made up of sine waves that are exact multiples of the fundamental." The sentence is meaningless from either of two perspectives—Fourier analysis or musical. There are cosine and sine waves in traditional Fourier analysis, in order to account for relative phase of overtones; plus, sounds that we call UNmusical are constituted the same way. From a musical perspective, the tones that make up a perfect third or fifth, chords that are considered quite musical, are not related by integer multiplication.

• (page 134) Addressing loudspeaker impedances and efficiency: "Some acoustic power can be regained by acoustically coupling the cabinets, placing them side-by-side, or even better, one on top of the other." Practically speaking, this is only true for frequencies whose wavelengths are large relative to cabinet dimensions (usually the bass), which are long enough that multiple cabinets can be close enough to yield a coupling effect. In addition, placing speakers side-by-side will cause comb filtering effects in the midrange and high frequencies, which can seriously color the sound. Also, one can't "regain" acoustic power; for a given electrical power input, the acoustic output in the relevant frequency range can be increased due to coupling among two or more speaker drivers radiating the same signal.

• (page 155) In the section on damping factor, there is no mention of the speaker-wire-resistance and crossover-impedance effects on damping factor, or their impact on the frequency response heard from the speaker due to the nonlinear speaker load in series with a variable-but-linear speaker wire impedance. (The variant nonlinearity and resistance of expensive speaker wires is the biggest reason they cause the sound to change, not necessarily for the better.)

• (page 175) "Sometimes it is necessary to interface balanced with unbalanced gear. When connecting such gear directly, it is advisable to use a cable (easily modified) that facilitates the connection." First, it is preferable not to directly interconnect balanced with unbalanced gear, but to use an interface box such as made by Jensen Transformers. Second, if you are forced to do so, check out the white papers at www.JensenTransformers.com to design a cable that is least likely to result in a hum or noise problem. Remember also that connecting unbalanced directly to balanced equipment will have an effect on the levels transferred.

• (pages 209-213) In the section on the Fletcher-Munson equal loudness contours, the numbers given correlating measured loudness at a specific frequency in dBspl with phons do not match the results when plotting the numbers on Figure 10.2 (page 211), which shows the Fletcher-Munson curves, labeled in phons, along with intensity level in dBspl. They come closer to matching ISO R226, but are still not a match. They also don't match any curves I could find on the Internet, such as at http://hyperphysics.phy-astr.gsu.edu/hbase/sound/eqloud.html#c1 (The hyperphysics section of gsu.edu provides a lot of useful reference information).

• (page 253-254) "Crossover distortion can occur in push-pull or class AB-type amplifiers in which a separate power supply is used to generate each portion of a waveform—one for the positive and another for the negative." This is not true. The presence of two power supplies or only one is irrelevant. There are two complementary portions of the output circuitry, each of which primarily provides the current for one half (positive or negative portion) of the signal waveform. Crossover distortion arises from improper biasing of the gain devices used for each half of the signal swing, not from how they are powered.

• (page 255) On measuring frequency response: Footnote 28. "A faster method might involve pink noise as a source, in which all octave bands are represented equally, but measurements would have to be in octave bands and would therefore be less precise." This is patently untrue. Wideband pink noise is made up of all frequencies in the accepted audio spectrum of statistically distributed amplitudes versus frequency such that each octave has the same amount of power. The same is true for each third octave, or any other fraction thereof. Measurements can be taken of an overall level, in octave bands, in third- or even twelfth-octave bands, based on the test instrument used. Using pink noise and measuring twelfth-octave levels at low frequencies is becoming a standardized practice as part of a room equalization technique.

There are a few positives:

• (pages 233-235) The sections on sound masking effects are quite clearly informative.

• (page 236) The concept and diagram of "The Audio Window—Relating What We Hear to the Gear" is nicely introduced.

• (page 252) The caution regarding the need for comprehensive explanations of the conditions surrounding a distortion claim is good advice for interpreting the value of any specifications or measurements.

This book would benefit from real-world examples within each section of the text. The concepts presented and jargon used will likely hold little meaning for most novices, who would find this book much more useful after gaining some experience working on a mixing board.

Both of these books give the impression that the authors' association with Berklee College allowed them to use the Berklee Press as a vanity press with no effective editing or proofreading. If the books are required acquisitions for Berklee College students, they have my sympathy.

If these books were just two more publications with no ties to a music college, I wouldn't grade them so harshly. However, they bring with them the implied authority of Berklee College, so I hold them to a higher standard. I wish they had passed. aX