1 Student presentation (PA1)

•

2 Preview of remaining semester

• Command-line sound editing on Wed, 12/7
• Don’t miss class on Mon, 12/12, you would most likely regret it. 😊
• 5.1 surround sound workshop on Wed, 12/14

3 Why master?

![Diagram]

- Often performed by specialized engineer not involved in mixing
- Idea: Unbiased & very expert pair of ears optimizes end result
- Unites focus on very small details with concern for ‘big picture’

3.1 Improving the overall sound

- E.g., increase impact, punch, brilliance, etc.
- Primarily defines mastering as an art form (discussed in depth later)
- But mastering also includes other, more mundane aspects.
3.2 Assembling a coherent album

- Provide coherence between songs on an album (Eargle\textsuperscript{2003b} p. 333)
- Track listing: Defining the album sequence (Katz\textsuperscript{2014g} RNZ\textsuperscript{2015})
- Deciding on length of silence between tracks (Katz\textsuperscript{2007c})
  - Considerations: genre, listening environment, playback device
  - Listen & stop with eyes closed to find ‘correct’ position
- Separate problem: Where to put track marker in break between 2 tracks?
- Balancing the perceived loudness of all tracks on an album

3.3 Preparation for end-user distribution

- Preparation of files & tapes (cf., Katz\textsuperscript{2014e})
- Delivery of media to client & pressing plant
- Always listen through your final master in full before you sign it off!
- Anecdote: Gerd Kühr – \textit{Revue instrumental et électronique} (Kühr\textsuperscript{2007})
  - Automation envelope mishap in final chord at 31’46”
  - Detected during final listening session (3 am)
  - Another 31’46” of rendering, another 31’46” of proof-listening

4 History of mastering (record cutting)

Dates back to creation of metal masters used to press vinyl records

4.1 Process

- Dorsey\textsuperscript{2013}: Good general overview
- Eargle\textsuperscript{1996}: More technical detail
- \textit{Lathe} (turntable with cutting head) cuts \textit{lacquer} (aluminum disc topped with acetate plastic layer)
- Cutting head receives l/r signals and \textit{mechanically} m/s-encodes them\textsuperscript{1}
  - Lateral cutting head movement corresponds to sum signal $M = L + R$
  - Vertical movement corresponds to difference signal $S = L - R$
- One can listen to the sounds cut onto a lacquer, but:
  - Softness of acetate plastic layer optimized for cutting, not playback
  - Repeated playback will wear out HF content of groove
- Instead, multi-step process yields vinyl disc for end user (cf., figure\textsuperscript{2})

\textsuperscript{1} The animations by Sourisseau\textsuperscript{1997} illustrate this process very comprehensively.
1. Silver layer deposited on lacquer to make it conductive
2. Electroplated to create metal master (negative)
3. From that make a metal mother (positive)
4. From that make a stamp (negative)
5. From that press actual vinyl records

4.2 Bass requires space

- Record grooves do not represent $A(t)$ as in DAW waveform!
  - Reason: Stylus velocity (not elongation) constant across freq. range
  - Implication: Greater stylus excursion per waveform period $T$ for LF
  - So low frequencies occupy more groove width than high ones!\(^2\)
- Interesting: Requires aesthetic compromises inherent to medium:
  - Record with more bass has to be quieter or shorter
  - Wider stereo image $\rightarrow$ needle more likely pops out of groove
- Particularly relevant in electronic dance music (EDM):
  - Genre-specific emphasis on bass
  - Groove stability crucial (subwoofer meets Technics SL-1210 turntable)
  - But still need to fit all the music onto the vinyl!

5 Mastering engineer’s toolkit

Typically increased focus on very high quality hardware & software!

- Excellent pair of studio monitors
- But also alternative (and often cheap) ‘real-world’ playback systems
- A/D and D/A converters
• Sample rate converters
• Eqs
• Dynamics processors (more focus on limiters & expanders)
• Signal meters
• Noise reduction
• Stereo enhancers, exciters, tube emulators, etc.

6 Processing chain

Dedicated plugins, e.g.: Izotope’s Ozone suite (cf., Ariza 2012a, pp. 261 f.)
But not unusual to use analog devices plus high-quality ADC/DAC!
Selection and order of processors really depends on circumstances!
Figure 3: Example of a typical mastering chain

- Includes some guidelines re. order (e.g., excite early, requantize last)
- See Katz (2014a, p. 131) & Ariza (2012a, pp. 257 f.) for details

Rules of 👍:
- Less is more! Applies to number of processors & intensity of use
- Link L & R channels of stereo input to ensure identical FX parameters

6.1 Exciter

- Put early in the signal chain (Ariza 2012a, p. 257)
- Originally analog hardware. Pioneer: Aphex Aural Exciter (1975)
- Today often dsp-based (might model tube saturation etc.)
- No coherent definition, but usually some combination of:
  - Dynamic eqing depending on input signal
  - Subtle overdrive to add favorable harmonic distortion
  - Synthesis of higher harmonics (or sub-bass)
  - Phase manipulation
- (Dis)advantages (Ariza 2012a, p. 262):
  - In right doses can add warmth and presence to dry and cold mixes
  - Excessive use might make mix too bright or edgy
6.2 EQ

- Goal: Improve spectral balance (but inevitably affects level balance!)
- Linear-phase eqs more relevant than elsewhere (cf., Katz 2017)
- Katz (2014b) discusses mastering-specific eqing techniques
- Recommendations by Ariza (2012a, p. 261):
  - *Less is more!* ±3 dB may be sufficient
  - Confirm by ear over long listening periods & by A/B comparisons
- Common applications (Ariza 2012a, p. 261):
  - Focusing middle range: Use parametric eq to boost or cut
  - Controlling bass: Boost 80 Hz to 120 Hz; cut < 60 Hz
  - Boosting ‘air’ or ‘sparkle’: Seductive, but danger of long-term fatigue
  - Dc offset removal: high-pass at < 20 Hz
- Eq → compressor is perhaps the more common order in mastering:
  - Preferable with *multiband* compressors (Ariza 2012a, p. 257)
  - But compressor might undo eq effect if some emphasized frequency range causes it to overreact (Katz 2014a, p. 131)
  - In that case, consider reverse order (compressor → eq)
- Senior (2011, p. 180): General advice on order of eq & compression

6.3 Stereo compressor

- An essential mastering tool (Eargle 2003b, p. 333)
  - Closes the circle: We start & end with compression
  - But more subtly applied in mastering than in stem preparation
- May increase punch… or flatten sound and take life out of it
- Rules of 👍 (Ariza 2012a, p. 259):
  - Low ratios
  - Low thresholds
  - Long attacks, short releases
- Music mastering compression recipe (Katz 2014c, pp. 84, 93)
  - Attack time: 30 ms to 300 ms (average: 100 ms)
  - Release time: 50 ms to 500 ms (average: 150 ms to 250 ms)
  - Ratio: ⅓ to ⅓. Threshold: −20 dB to −10 dB
  - More subtle: Ratio: ⅓ to ⅓. Threshold: −40 dB to −30 dB
  - “Delicate painting”: Ratio: ⅓. Threshold: −3 dB

Figure 4. Stereo compressor
• Multiband compressors frequently used (Ariza 2012a, pp. 260 f.):
  – Blessing and curse: Allows to change mix balance after the fact
  – Potential tendency to overcompress (use with care!)
• Remember to link stereo channels (cf., figure 4)!

6.4 Stereo enhancer (m/s processing)
• Note: Not shown in signal chain of figure 3
• Goal: Improve spatial balance
• We previously discussed stereo enhancing mono signals (‘fake’ stereo)
• In mastering, however, we typically have a signal that is already stereo
• Tricky: How to adjust width of a stereo signal after the mix?
• Answer: m/s-based processing (cf., figure 5). Build your own in a DAW:
  1. Encode l/r stereo mix to m/s
  2. Adjust s/m ratio (e.g., increase to widen image)
  3. Optional: Compress, eq, reverberate m/s signal (Katz 2014a, pp. 135 ff.)
  4. Decode to l/r again
• Works best with coincident stereo recordings (why?)

6.5 Peak limiter
• Put late in the signal chain (last step before resampling & requantizing)
• Ariza (2012a, p. 258) provides suggestions for parameter settings

--- Dac output

− − − − − − −

Figure 5. M/s-based stereo enhancer (cf., Senior 2011, pp. 262 ff. Katz 2007a, pp. 210 ff.)

Figure 6. Inter-sample peaks in d/a conversion
• Rule of ⌈: Don’t limit all the way up to 0 dBFS!
  – Reason: *Inter-sample peaks* (cf., figure 6)
  – Clipping behavior can vary among different DACs
  – Limit to, say, −0.2 dBFS (Ariza 2012a, p. 258)

• Peak limiting frequently (mis)used as a means of maximizing loudness
  – Evidence shows: Louder media (radio stations, CDs) are preferred
  – But no predictable relation to perceived loudness
  – Resulted in *loudness war* (Katz 2014b)

• However, excessive limiting can make a mix sound very flat!
  – Katz (2007b, pp. 213 f.) suggests guidelines for compromises
  – Consider *loudness normalization* as an alternative

### 6.6 Loudness normalization (*Lufs*, *EBU R128*, CalM Act)

• Holistic, modern approach towards loudness management

• Let’s use it to end the *loudness war* (Katz 2014b)!

• Legislation for broadcasting exists in US (CalM Act, ATSC A/85) & EU

• Recommendations: Itu-r bs.1770-4 (Iru 2015); EBU R128 (Camerer 2011)

• New measure: *Lufs* (“loudness unit with regards to full scale”)
  – Models perceived loudness (cf., Hollerweger and Holzmann 2012)
  – Nice: Predict how loud production will be compared to others

• Requires dedicated loudness meters (Katz 2014i)
  – Robin Gareus’ excellent lv2 meter plugins (not just *Lufs*)
  – Debian/Ubuntu: sudo apt-get install x42-plugins
  – *Hindenburg* audio editor comes with integrated *Lufs* metering

• Requires calibrated monitoring (Katz 2014j)

• Which *Lufs* value should I use?
  – EBU R128 recommends −23 *Lufs* for broadcasting
  – Higher targets may be required for other applications
    (web audio, portable music players, etc.)
  – No definite standards yet (for guidelines, see Hollerweger 2013)
6.7 Resampling

- Different interpolation algorithms (e.g., sinc, zoh, linear)
- Significant differences in sound quality!
- Smith (2017) describes theory & implementation in depth
- High-quality floss converter: sndfile-resample
  Debian/Ubuntu: sudo apt-get install samplerate-programs

6.8 Requantizing (with dither & noise shaping)

- Requantizing… fancy term for ‘changing the bit depth’
- May be required in order to match target medium (e.g., cd: 16 bit)
- Should always happen last!
- Remember to dither & noise-shape when moving to a lower bit depth!
- Floss converter: sndfile-convert
  Debian/Ubuntu: sudo apt-get install sndfile-programs

7 Mastering workflows

- Idealized goal: Mastering happens after mixing stage is completed
  “When the mix is done, it should be done!” (Katz [2014] p. 133)
- Motivations:
  – Breaking down complex process into manageable entities
  – Allowing everyone to focus on what they are best at
  – Retaining sanity (don’t tweak the mix forever)

7.1 Mastering from stereo mix

- Traditional approach
- Mastering engineer receives only stereo mix from mixing engineer
- No possibility to ‘re-open’ the mix (can also be a blessing)
- Re-submission of mix requested only in case of significant problems

7.2 Mastering from stems

- Becoming more common, due to ease of rendering in daw environments
- Mastering engineer receives from mixing studio
  1. Stereo mix
  2. Set of stems which yields same mix when played together\(^3\)

\(^3\) Usually instrumental stems are submitted, i.e., one stem per instrument (rather than per microphone). Note that even a center-panned mono signal (e.g., bass) might be deliberately provided as a stereo stem (in this case with identical l/r signals), to avoid ambiguities with regards to the intended panning. The idea is that if all stems play at 0 dB, the result should perfectly match the submitted stereo mix.
• Mastering process focuses on stereo mix as usual
• But possibility to ‘re-open’ mix if need be (Izhaki 2011b p. 53)
• See also Katz (2014a pp. 133 f.)

7.3 Surround mastering
• Special requirements for mastering in 5.1 surround
• Workflows & best practices still under much development
• Consult Katz (2007d, 2014f) to learn more about state of the art

8 Daw workflow suggestion

1. Mix from 16 or 24 bit sources without fx on master output track
2. Render stereo mix from Daw in 24 bit without dither
3. Create a new 24 bit Daw session for mastering
   (a) Add stereo mix to new input track
   (b) Add mastering processing chain to input track with stereo mix
   (c) Duplicate clean input track (for quick A/B comparisons)
4. Render to final master. Depending on bit depth of target medium:
   • Render to 24 bit without dither & noisehaping, or:
   • Render to 16 bit with dither & noisehaping

References & further reading


