1 Student presentation (PA1)

2 Announcement: I want you for schlepping

3 Quiz 1 (QZ1)

4 Digital audio basics

4.1 Audio file formats

<table>
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<th>Data compression</th>
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<td>Lossless (reversible)</td>
<td>FLAC</td>
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<tr>
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<td>.ogg</td>
</tr>
<tr>
<td></td>
<td>Opus</td>
<td>.opus</td>
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</table>

Rules of 👍:

- Uncompressed for editing and production (recommended: .wav)
- Lossless compressed for archiving large projects (recommended: .flac)
- Lossy compression only for end-user distribution (recommended: .ogg)
- Avoid re-encoding in lossy compressed formats!
### 4.2 Sample rate & bit depth

- Sample rate = temporal resolution (x axis; determines max. frequency)
- Bit depth = amplitude resolution (y axis; determines dynamic range)
- General recommendation for this class: 44.1 kHz, 24 bit
  - 24 bit probably more important than 96 kHz
  - Stick to one sample rate & bit depth for given project (avoid unnecessary conversions)
- Bit rate ... describes data throughput of compressed audio formats
  - Unit indicates product of bit depth (kbit) & sample rate (s\(^{-1}\) = Hz)
  - In reality not quite as simple (since data is compressed)
  - Modern codecs often use Variable bit rate (vbr) to minimize file size

![Figure 1. Sample rate and bit depth in digital audio](image-url)
5 Essential sound editing principles

5.1 Avoiding clicks through crossfades

- Rapid changes in amplitude results in audible clicks. Why?
- Likely to happen whenever you splice together two audio snippets!
- Use crossfades to prevent clicks
- Different crossfade shapes available. Rules of 
- Use constant-gain ($g_{in} + g_{out} = 1$) for strongly correlated materials
- Use constant-power ($g_{in}^2 + g_{out}^2 = 1$) for uncorrelated materials
- Crossfades can also be asymmetrical (fade-out time of first sound ≠ fade-in time of second)

5.2 Setting appropriate output levels & avoiding clipping

- Hard border of digital audio is 0 dBFS (“dB full-scale”)
  - Violating that border results in clipping of waveform
Audible result: Harmonic distortion (i.e., additional high-frequency components). Why distortion? Why harmonic?

• But don’t be tempted to record or render at low levels ‘to be safe’ and amplify later. Why?

• Rules of 👍 for recording, editing & mixing:
  – Always aim at getting good levels as early as possible in the process
  – Make levels ‘as high as possible, but not higher’ to optimize s/N ratio

• Rules of 👍 for mastering:
  – Don’t render all the way to 0 dBFS (inter-sample peaks; also depends on DAC)
  – Normalize to a loudness target (discussed below) suitable for target medium

• Recommendation for ed assignments: Render to −3 dB to −1 dB peak level

5.3 Normalization

• Motivations:
  – Exploit available headroom of low-level signals
  – Match perceived loudness of different audio files (album; playlist)
  – Win the loudness war (Katz 2014a)

• Peak or rms normalization is a two-step process:
  1. Find maximum (peak or rms) level in signal to be normalized
  2. Amplify (or attenuate) signal to a new pre-defined target maximum
• Example:
  – Maximum peak level is −7 dB
  – We want to normalize to −3 dB peak level
  – Hence entire signal has to be amplified by +4 dB

• Properties:
  – Normalization can be automated
  – Inherently non-realtime!
  – Only changes overall level (but neither spectrum nor dynamics!)

• Modern productions use loudness normalization (Katz [2014a,b,c])
  – Neither peak nor rms level say much about perceived loudness ☺
  – Hence nobody knew how their production’s loudness would compare to others
  – So everybody tried to make theirs as loud as possible → loudness war
  – Will be discussed in more depth in future lecture on mastering

• Lufs (“loudness unit with regards to full scale”)
  – Models perceived loudness for purpose of loudness normalization
  – Target recommendations available, e.g., -23 Lufs for TV broadcasts
  – Details: Itu-r bs.1770-4 (ITU [2015]); EBU r128 (Camerer [2011])

• When to normalize?
  – Peak/rms: When you have a very low-level audio file where you reach the top of the fader in the mix
  – Loudness normalization: Before rendering your mix (or during mastering)
  – But no need to normalize each and every individual audio snippet!
• How to normalize?
  – As a destructive offline process (e.g., sox in.wav out.wav norm -3, or Effect Normalize menu in Audacity)
  – Manually. Project renders to meager peak level of −13 dB? Raise master fader by 12 dB and re-render to −1 dB peak level!
  – Automatically. Many DAWs provide functions to automatically normalize individual audio snippets or master fader

5.4 Basic mixing: Panning & level balance

![Panpot position](image)

- Basic act of mixing: Setting level balance and stereo balance:
- Faders adjust level at which input signal is routed to stereo mix
- Panpot determines position of a mono phantom source in stereo image
  - Panning refers to positioning of sound between L & R speakers
  - Reciprocally adjusts gains of 2 amps for L & R speakers (amplitude panning, by contrast to time delay panning)
  - Implemented on analog mixers by means of a potentiometer, hence the term panpot
  - Different panpot laws determine precise gain functions (usually equal power panning)
- Panpot vs. balance control:
  - Panpot: Mono input signal (entire input signal will be heard regardless of panpot position)
  - Balance control: Stereo input signal (parts of input might become inaudible for hard L/R panning!)

Figure 6. Stereo panpot laws

![Figure 6. Stereo panpot laws](image)

Figure 7. Stereo loudspeaker setup

![Figure 7. Stereo loudspeaker setup](image)
6 Basic sound editing in Reaper

6.1 Licensing information
- $60 for personal use, educational institutions, small businesses
- Free 2-month trial

6.2 Download
- On Linux, download Windows installation file for installation via Wine
- Do I have a 32 or 64 bit os? Probably 64 bit, but:
  - macOS: [Apple](https://www.apple.com) > About This Mac (top left desktop corner)
  - Windows 8 and 10: [Win + X] and [System] > System type
  - Linux: Type arch in command line (i686 = 32 bit, x86_64 = 64 bit)

6.3 Installation
- macOS: Drag from .dmg to Applications
- Windows: Double-click .exe
- Linux:
  1. On 64-bit machines, enable multiarch, e.g.: 
     sudo dpkg --add-architecture i386 && sudo apt-get update
  2. Install Wine and Mono, e.g., via Ubuntu Software Center or
     sudo apt-get install wine winetricks mono-complete wine32

6.4 Overview of Reaper window
- Track control panel (left)
- Transport bar (below)
- Arrange area (right)
- Mixer control panel (bottom)

6.5 Recording sound
- Add new track(s): [Tracks] > Insert new track
- Arm track(s) for recording:
  1. Click small red circular button on respective track
  2. Track meter should now show input signal
- Start recording: Large red circular button (transport area) or [Ctrl] + [R]
- Stop recording: Click record button again (or stop button)
6.6 Importing audio files

- Insert Media file...
- Can also select multiple sound files and insert them as Separate Tracks

6.7 Playing back sound

- Start playback using button or (preferably) Space bar
- Hit Space again to stop playback and go back to starting point
- [Ctrl] + Space to stop playback and leave cursor in place
- R key (or use button in transport menu) to toggle looped playback
- Mute a channel by pressing M button in track control panel
- Solo a channel by pressing S button in track control panel

6.8 Mixing & panning

- Input channel faders (on track control panel and mixer)
- Panpot (on track control or mixer control panel)
- Left: master fader (mixed stereo output)

6.9 Arranging

- Drag & drop items on timeline or between tracks (left mouse button)
- Hold [Ctrl] to select multiple items with left mouse button
- Disabling grid lines:
  - [Alt] + G (Win)
  - [Cmd] + G (Mac)
  - Details: Displaying Grid Lines in Reaper User Guide (Francis 2017)

6.10 Cutting items

1. Select item(s) to cut
2. Position cursor on timeline and press
3. S key to cut

6.11 Shortening, lengthening & looping

1. Move cursor over bottom half of left or right edge of audio snippet
2. Hold left mouse button and move mouse left/right

Note: Extending item beyond its actual length will automatically loop it
6.12 Splicing, fades & crossfades

- **Fade in/out:**
  1. Move cursor over top half of left | right edge of audio item
  2. Left-click and drag mouse to right | left to set fade length

- **Crossfades:**
  - Reaper automatically creates crossfade between overlapping items
  - Double-click on crossfade area to edit in detail

6.13 Rendering a completed project to a new sound file

- Don’t submit .RPP (Reaper project) files – they contain no audio data!
- Instead, render completed project to new audio file via File >> Render…:
  - Select channels to render (usually Master mix)
  - Select render bounds (avoid mysterious periods of silence at end)
  - Decide on output file format, sample rate, bit depth, channel number
  - Watch output level meter
  - Adjust master fader, and re-render if necessary (to avoid clipping)

7 Preview ed1 assignment

References & further reading


On course reserve at the Lewis Music Library.

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