Preparing the weekly release as one audio file

A complete, cross-platform solution to record, convert and stream audio and video.

https://ffmpeg.org/

FFmpeg

FFmpeg is a command line tool for working with audio and video formats. You can use it to do a whole range of things, it comes with a lot of detailed settings and options.

This is an example of how you can use FFmpeg to convert a `mp4` video file into an `avi` file.

```
$ ffmpeg -i input.mp4 output.avi
```

You can use FFmpeg on the soupboat, where it is installed with `sudo apt install ffmpeg`; or you can install it on your own computer, in which case you can find installation instructions here:

- Mac: https://formulae.brew.sh/formula/ffmpeg
- Windows: https://community.chocolatey.org/packages/ffmpeg
- More options: https://trac.ffmpeg.org/wiki/CompilationGuide

When you are planning to FFmpeg often and specially to work with bigger files, it is recommended to install it on your own computer.

From wav to mp3

Many of the recording devices save audio as wav files. This is a high quality audio file format, which is too heavy to use on the web and most browsers will not play it in a webpage. \ So we will convert the audio files into a compressed audio format, that is supported by all browsers: `mp3`.

To convert a `wav` file into a `mp3` file, you can use the following FFmpeg command:

```
In [… ! ffmpeg -hide_banner -i samples/raphael.wav samples/raphael.mp3
```

```
Guessed Channel Layout for Input Stream #0.0 : stereo
Input #0, wav, from 'samples/raphael.wav':
  Duration: 00:00:16.39, bitrate: 1411 kb/s
    Stream #0:0: Audio: pcm_s16le ([1][0][0][0] /
0x0001), 44100 Hz, stereo, s16, 1411 kb/s
Stream mapping:
  Stream #0:0 -> #0:0 (pcm_s16le (native) -> mp3
(libmp3lame))
Press [q] to stop, [?] for help
Output #0, mp3, to 'samples/raphael.mp3':
  Metadata:
    TSSE            : Lavf58.20.100
    Stream #0:0: Audio: mp3 (libmp3lame), 44100 Hz,
stereo, s16p
    Metadata:
      encoder         : Lavc58.35.100 libmp3lame
size=     257kB time=00:00:16.40 bitrate= 128.3kbits/s
speed=18.2x
video:0kB audio:257kB subtitle:0kB other streams:0kB
global headers:0kB muxing overhead: 0.096236%
```

You can specify more options to convert your file with a specific codec or quality, set the number of channels and select a bit rate. \ See the command below, and this page for more information: https://trac.ffmpeg.org/wiki/Encode/MP3

In…
```
! ffmpeg -i samples/raphael.wav -acodec libmp3lame -aq 2 -ac 2 -ar 48000 -hide_banner samples/raphael.mp3
```

- `-acodec` : name of the audio codec
- `-aq` : quality scale (1-10)
- `-ac` : number of audio channels
- `-ar` : frequency in hz
- `-ab` : audio bitrate

See: https://ffmpeg.org/ffmpeg.html#Audio-Options

From aiff to mp3

In […
```
! ffmpeg -hide_banner -i samples/dialed.aiff samples/dialed.mp3
```

Hide "WARNING: library configuration mismatch"

Add `-hide_banner` to your `ffmpeg` commands, to avoid getting the `[warning]` messages in your output.

In [ ]:
```
! ffmpeg -hide_banner
```

Generating playlist.txt

To start making a single audio file of all the recordings, we will first make a `playlist.txt` file.

To do this, we will use the concatenate function of FFmpeg. The following examples come from this page: https://trac.ffmpeg.org/wiki/Concatenate

We will only search for all the `mp3` files in the `./samples/` folder:

```
In [2…  ! for f in ./samples/*.mp3; do echo "file '$f'" >> playlist.txt; done
```

See if it worked:

```
In [29]:  ! cat playlist.txt
```

```
file './samples/dialed.mp3'
file './samples/owl.mp3'
file './samples/raphael.mp3'
```

Remove the file `playlist.txt` if you want to overwrite it and make a whole new one with:

```
In [26]:  ! rm playlist.txt
```

Concatenating release.mp3

Now we will use the playlist.txt file to generate a single audio file, using FFmpeg's `concat`, which is the command for *concatenating*.

```
In [… ! ffmpeg -hide_banner -f concat -safe 0 -i playlist.txt -c copy release.mp3
```

```
Input #0, concat, from 'playlist.txt':
  Duration: N/A, start: -0.023021, bitrate: 128 kb/s
    Stream #0:0: Audio: mp3, 48000 Hz, stereo, fltp,
128 kb/s
    Metadata:
      encoder         : Lavc58.35
Output #0, mp3, to 'release.mp3':
  Metadata:
    TSSE            : Lavf58.20.100
    Stream #0:0: Audio: mp3, 48000 Hz, stereo, fltp,
128 kb/s
    Metadata:
      encoder         : Lavc58.35
Stream mapping:
  Stream #0:0 -> #0:0 (copy)
Press [q] to stop, [?] for help
size=    1630kB time=00:01:56.68 bitrate= 114.4kbits/s
speed= 725x
video:0kB audio:1629kB subtitle:0kB other streams:0kB
global headers:0kB muxing overhead: 0.025712%
```

Remove the file `release.mp3` before you make a new one with:

```
In [52]: ! rm release.mp3
```

Check the format, encoder, bitrate, and channels of your audio file

```
In [54]: ! ffprobe -hide_banner samples/raphael.mp3
```

```
Input #0, mp3, from 'samples/raphael.mp3':
  Metadata:
    encoder         : Lavf58.20.100
  Duration: 00:00:16.44, start: 0.023021, bitrate: 129
kb/s
    Stream #0:0: Audio: mp3, 48000 Hz, stereo, fltp,
129 kb/s
    Metadata:
      encoder         : Lavc58.35
```

In [49]: `! ffprobe -hide_banner samples/owl.mp3`

```
Input #0, mp3, from 'samples/owl.mp3':
  Metadata:
    comment         : dUBITS=$02021600
                    : dSCENE=1
                    : dTAKE=8
                    : dTAPE=20160224
                    : dFRAMERATE=24
                    : dTRK1=LEFT
                    :
    encoded_by      : iZotope RX 4
    originator_reference:
USIZT0G88181ZVXYL092953326532331
    date            : 2016-03-05
    coding_history  :
A=PCM,F=48000,W=24,M=1,T=ZaxConvert
                    :
    time_reference  : 3186656781
    encoder         : Lavf58.20.100
  Duration: 00:01:28.25, start: 0.023021, bitrate: 109
kb/s
    Stream #0:0: Audio: mp3, 48000 Hz, stereo, fltp,
109 kb/s
    Metadata:
      encoder         : Lavc58.35
```

In [50]: `! ffprobe -hide_banner samples/dialed.mp3`

```
Input #0, mp3, from 'samples/dialed.mp3':
  Metadata:
    encoder         : Lavf58.20.100
  Duration: 00:00:12.02, start: 0.023021, bitrate: 128
kb/s
    Stream #0:0: Audio: mp3, 48000 Hz, stereo, fltp,
128 kb/s
    Metadata:
      encoder         : Lavc58.35
```