FFMPEG ENCODING AND EDITING COURSE

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ABOUT ME

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GOALS

You should learn:

- Basic concepts
- Installing ffmpeg and tools
- Encoding videos
- Appling filters
- Analyzing videos

REQUIREMENTS

- These slides
- ffmpeg, ffprobe and ffplay installed
- Some sample videos

RESOURCES

If you need sample videos for testing, see overview from VQEG
(Video Quality Experts Group):

https://www.its.bldrdoc.gov/vqeg/video-datasets-and-organizations.aspx

INTRODUCTION TO FFMPEG

ABOUT THE PROJECT

- Free, open-source software for multimedia editing, conversion, …
- Started in 2000
- Continuous development until now

TOOLS

FFmpeg contains:

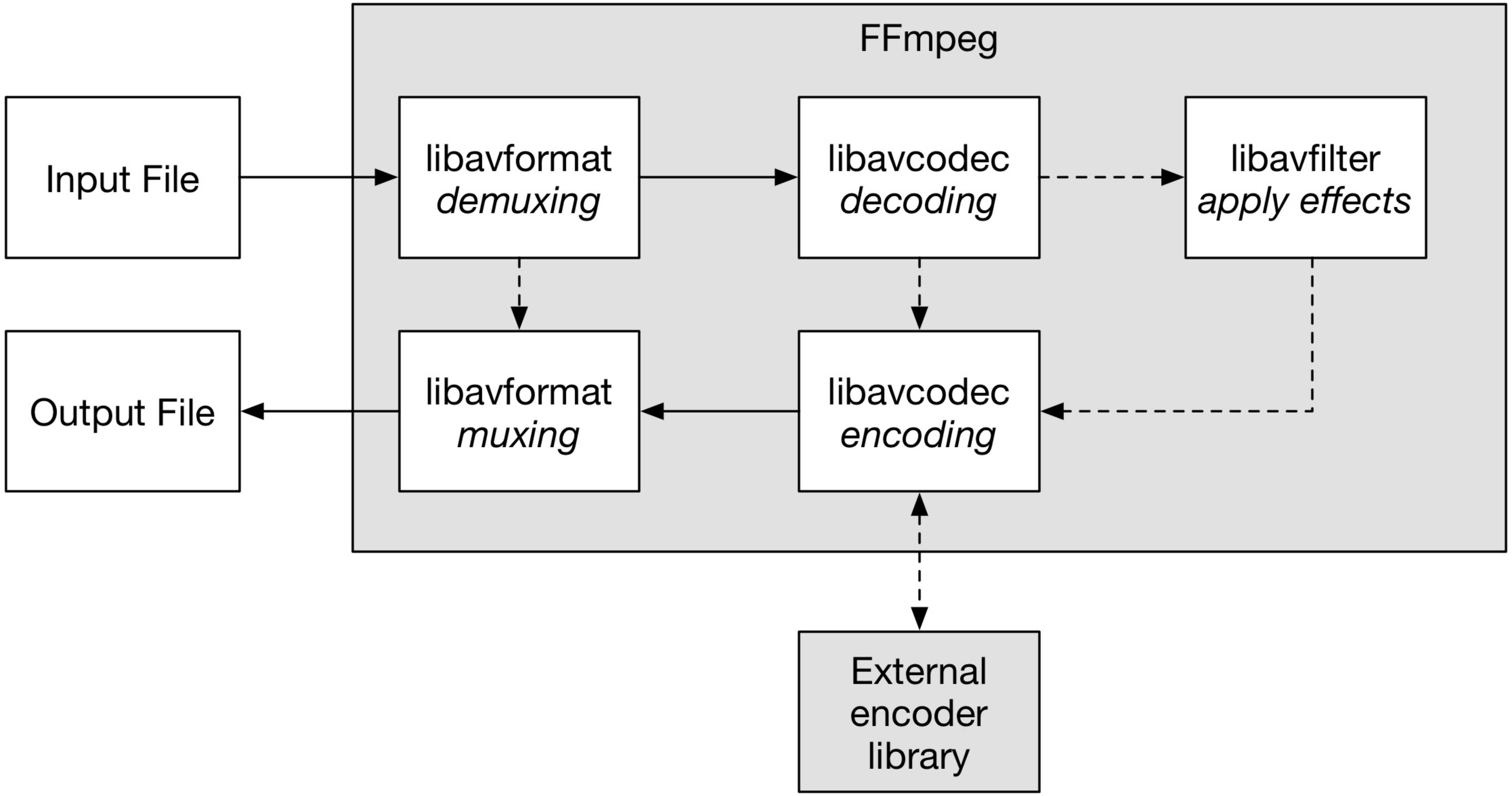
- Command-line tools: ffmpeg, ffprobe, ffserver, ffplay
- Libraries: libavformat, libavcodec, libavfilter, …

ABOUT THE LIBRARIES (LIBAV*)

- libavformat: Reads and writes container formats (AVI, MKV, MP4, …)
- libavcodec: Reads and writes codecs (H.264, H.265, VP9, …)
- libavfilter: Various filters for video and audio
- ... and many more

ARCHITECTURE

Simplfied overall architecture:



INSTALLATION / COMPILATION

Installation Method	Pro	Con
Building from source	Offers all options, tools, codecs	Takes time, hard to update
Downloading static build	Easy and fast	Does not offer all codecs, manual update
Installing from package manager (e.g., apt-get)	Easy and fast	Does not always offer latest version or all codecs

Get source code and static builds from: http://ffmpeg.org/download.html

GETTING HELP

Places to get help:

- Documentation: https://ffmpeg.org/ffmpeg-all.html
- Wiki: http://trac.ffmpeg.org/wiki
- IRC: #ffmpeg
- Mailing list: https://lists.ffmpeg.org/mailman/listinfo/ffmpeg-user/
- Stack Overflow: https://stackoverflow.com/ and use #ffmpeg
- Super User: http://superuser.com/ and use #ffmpeg

GENERAL VIDEO ENCODING CONCEPTS

CONTAINER FORMATS

Containers *contain* media data. Typical examples:

- MP4: MPEG-4 Part 14 container for H.264, H.264, AAC audio, …
- MKV: Versatile container for any media format
- AVI: Legacy container

See supported containers with:

```
$ ffmpeg -formats
File formats:
 D. = Demuxing supported
 .E = Muxing supported
 --
 D  3dostr          3DO STR
  E 3g2             3GP2 (3GPP2 file format)
  E 3gp             3GP (3GPP file format)
 D  4xm             4X Technologies
  E a64             a64 - video for Commodore 64
 D  aa              Audible AA format files
 ...
```

CODECS

- CODEC = *Coder* / *Decoder*
- Specification on how to code and decode video, audio, …
- Usually **not** a specification on how to encode / compress data

See supported codecs with:

```
$ ffmpeg -codecs
Codecs:
 D..... = Decoding supported
 .E.... = Encoding supported
 ..V... = Video codec
 ..A... = Audio codec
 ..S... = Subtitle codec
 ...I.. = Intra frame-only codec
 ....L. = Lossy compression
 .....S = Lossless compression
 -------
 D.VI.. 012v                 Uncompressed 4:2:2 10-bit
 D.V.L. 4xm                  4X Movie
 D.VI.S 8bps                 QuickTime 8BPS video
 .EVIL. a64_multi            Multicolor charset for Commodore 64 (encoders: a64multi )
 .EVIL. a64_multi5           Multicolor charset for Commodore 64, extended with 5th color (colram) (encoders
 D.V..S aasc                 Autodesk RLE
 D.VIL. aic                  Apple Intermediate Codec
```

MOST IMPORTANT (LOSSY) CODECS

Currently mostly used, standardized by ITU/ISO:

- H.262 / MPEG-2 Part H: Broadcasting, TV, used for backwards compatibility
- H.264 / MPEG-4 Part 10: The de-facto standard for video encoding today
- H.265 / MPEG-H: Successor of H.264, up to 50% better quality
- MP3 / MPEG-2 Audio Layer III: Used to be the de-facto audio coding standard
- AAC / ISO/IEC 14496-3:2009: Advanced Audio Coding standard

Competitors that are royalty-free:

- VP8: Free, open-source codec from Google
- VP9: Successor to VP8, almost as good as H.265
- AV1: Currently in development as a successor to VP9

MOST IMPORTANT LOSSLESS CODECS

Lossless codecs are useful for archival, editing, ...

- Raw YUV, HuffYUV, FFV1
- Raw PCM, FLAC, ALAC

Also, "visually lossless" codecs exist:

- Apple ProRes, Avid DNxHD, high-quality H.264/H.265, ...

ENCODERS

- Encoders are the *actual* software that outputs a codec-compliant bitstream
- Encoders can vary in quality and performance

Examples:

- libx264: most popular free and open-source H.264 encoder
- NVENC: NVIDIA GPU-based H.264 encoder
- libx265: free and open-source HEVC encoder
- libvpx: VP8 and VP9 encoder from Google
- libfdk-aac: AAC encoder
- aac: native FFmpeg AAC encoder
- …

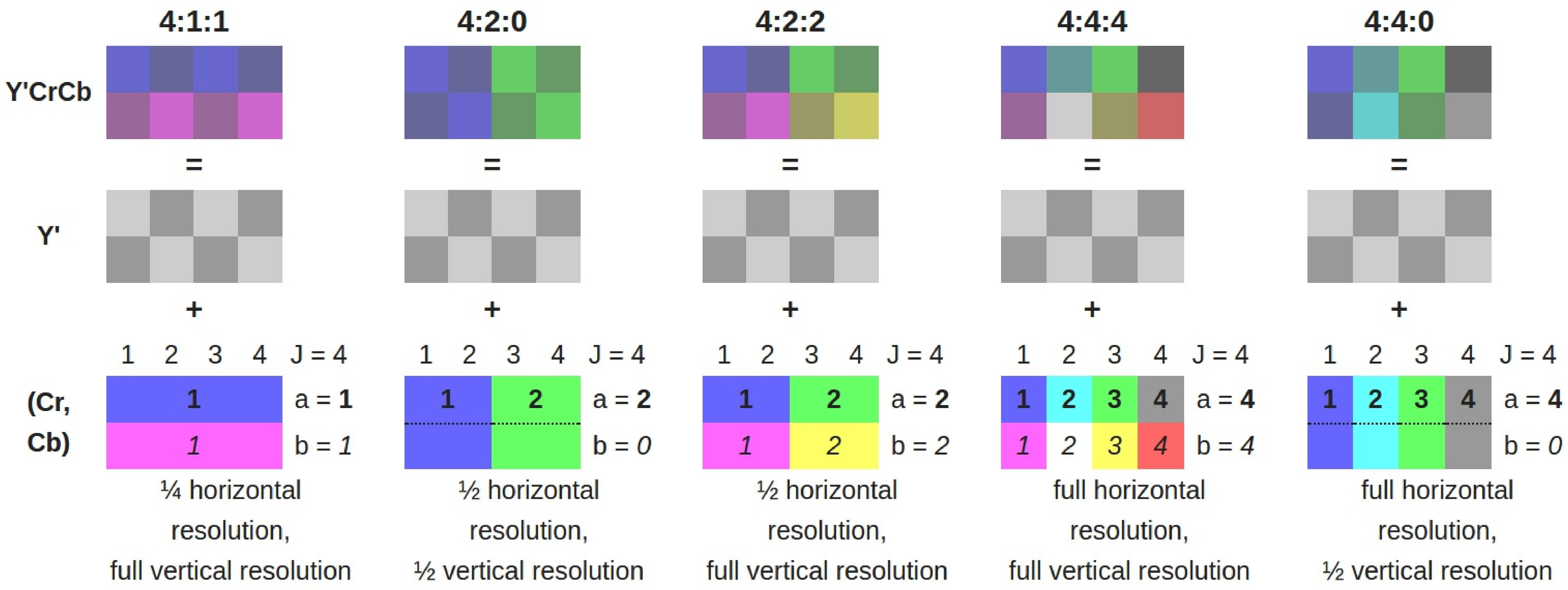
→ lots of competition

ENCODERS SUPPORTED IN FFMPEG

```
$ ffmpeg -encoders
Encoders:
 V..... = Video
 A..... = Audio
 S..... = Subtitle
 .F.... = Frame-level multithreading
 ..S... = Slice-level multithreading
 ...X.. = Codec is experimental
 ....B. = Supports draw_horiz_band
 .....D = Supports direct rendering method 1
 ------
 V..... a64multi             Multicolor charset for Commodore 64 (codec a64_multi)
 V..... a64multi5             Multicolor charset for Commodore 64, extended with 5th color (colram) (codec a64_m
 V..... alias_pix            Alias/Wavefront PIX image
...
```

PIXEL FORMATS

- Representation of raw pixels in video streams
- Specifies order of luma/color components and chroma subsampling



Supported pixel formats:

```
ffmpeg -pix_fmts
```

EXERCISE

Tasks:

- Download a PNG image from the Web
- Run the following command:

```
ffmpeg -loop 1 -i <image> -t 5 output.mp4
```

- What codec and encoder are used for the output file?
- What pixel format is auto-selected by ffmpeg?
- Why do you think this is done?

ENCODING WITH THE FFMPEG COMMAND LINE TOOL

GENERAL SYNTAX

```
ffmpeg <global-options> <input-options> -i <input> <output-options> <output>
```

- Global options for log output, file overwriting, ...
- Input options for reading files
- Output options for:
  - conversion (codec, quality, ...)
  - filtering
  - stream mapping

TRANSCODING AND TRANSMUXING

Transcoding from one codec to another:

```
ffmpeg -i <input> -c:v libx264 output.mp4
```

Transmuxing from one container/format to another – without re-encoding:

```
ffmpeg -i input.mp4 -c copy output.mkv
```

ffmpeg will take one video, audio, and subtitle stream from the input and map it to the output.

Explanation:

- `-c` sets the encoder (see `ffmpeg -encoders`)
- `-c copy` only copies bitstream
- `-c:v` sets only video encoders
- `-c:a` sets only audio encoders
- `-an` and `-vn` would disable audio or video streams

TRANSCODING BACKGROUND

From http://ffmpeg.org/ffmpeg-all.html:

> *ffmpeg […] **read[s] input files and get packets** containing encoded data from them. When there are multiple input files, ffmpeg tries to keep them synchronized […].*
>
> *Encoded packets are then **passed to the decoder**. […] The **decoder produces uncompressed frames** […] which can be processed further by **filtering** […]. After filtering, the frames are **passed to the encoder**, which encodes them and outputs encoded packets. Finally those are **passed to the muxer**, which writes the encoded packets to the output file.*

SEEKING AND CUTTING

Cut a video from timestamp <start> for <duration>, or until <end>:

```
ffmpeg -ss <start> -i <input> -t <duration> -c copy <output>
ffmpeg -ss <start> -i <input> -to <end> -c copy <output>
```

Examples:

```
ffmpeg -ss 00:01:50 -i <input> -t 10.5 -c copy <output>
ffmpeg -ss 2.5 -i <input> -to 10 -c copy <output>
```

Notes:

- When re-encoding, seeking is always accurate
- When copying bitstreams (-c copy), ffmpeg may copy frames that are not shown but necessary
- Also see: http://trac.ffmpeg.org/wiki/Seeking

SETTING QUALITY

- The output quality depends on encoder defaults
- Do not just encode without setting any quality level

Possible options (just examples):

- `-b:v` or `-b:a` to set bitrate (e.g., `-b:v 1000K`, `-b:v 8M`)
- `-q:v` or `-q:a` to set fixed-quality parameter (e.g., `-q:a 2` for native AAC encoder)

Examples of encoder-specific options:

- `-crf` to set Constant Rate Factor for libx264/libx265
- `-vbr` to set constant quality for FDK-AAC encoder
- Many many more; see `ffmpeg -h encoder=libx264` for examples

EXAMPLE: TRANSCODING TO H.264

Constant quality (CRF) encoding:

```
ffmpeg -i <input> -c:v libx264 -crf 23 -c:a aac -b:a 128k output.mkv
```

CRF between 18 and 28 looks "good", lower is better.

Two-pass encoding:

```
ffmpeg -y -i <input> -c:v libx264 -b:v 8M -pass 1 -c:a aac -b:a 128k -f mp4 /dev/null
ffmpeg -i <input> -c:v libx264 -b:v 8M -pass 2 -c:a aac -b:a 128k output.mp4
```

(Windows: Use NUL instead of /dev/null)

See https://trac.ffmpeg.org/wiki/Encode/H.264

RATE CONTROL

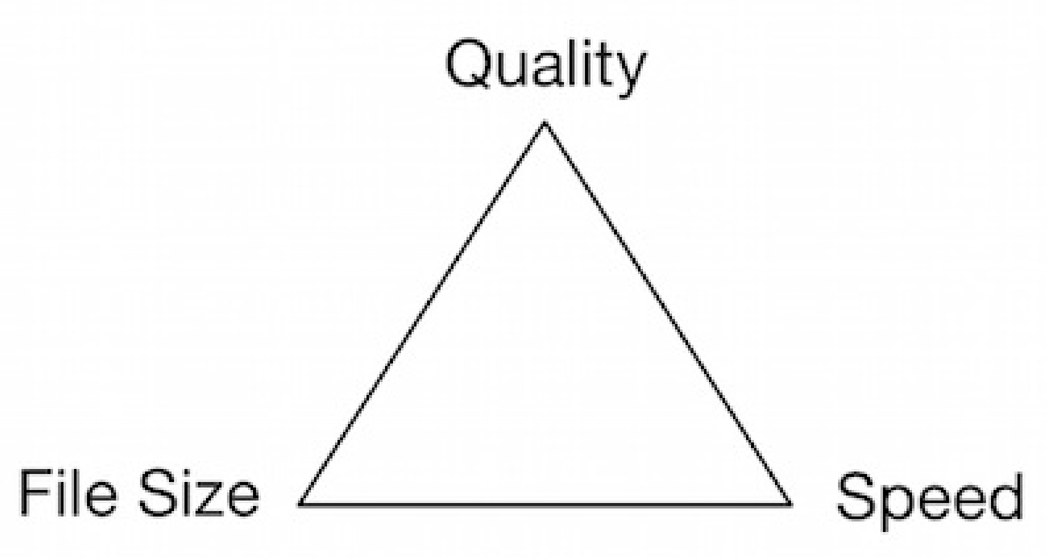
Different kinds of rate control:

- Constant Bitrate (CBR)
- Variable Bitrate (VBR)
    - Average bitrate (ABR)
    - Constant quantization parameter (CQP)
    - Constant quality, based on psychovisual properties, e.g. CRF in x264/x265/libvpx-vp9

Which rate control to use for which case? More info: https://slhck.info/video/2017/03/01/rate-control.html

SPEED VS. QUALITY VS. FILE SIZE

(Lossy) encoding is always a trade-off between:



For example:

- You can have fast, high-quality encoding, but the file will be large
- You can have high-quality, smaller file size, but the encoding will take longer
- You can have small files with fast encoding, but the quality will be bad

SPEED/QUALITY PRESETS IN X264

Choose the encoding speed for libx264 with the preset option:

```
ffmpeg -i <input> -c:v libx264 -crf 23 -preset ultrafast -an output.mkv
ffmpeg -i <input> -c:v libx264 -crf 23 -preset medium -an output.mkv
ffmpeg -i <input> -c:v libx264 -crf 23 -preset veryslow -an output.mkv
```

All presets: ultrafast, superfast, veryfast, faster, fast, medium, slow, slower, veryslow

Example results (all have the same quality!):

Preset	Encoding Time	File Size
ultrafast	4.85s	15M
medium	24.13s	5.2M
veryslow	112.23s	4.9M

CHANGING FRAMERATE

Simple way to change the framerate by dropping or duplicating frames:

```
ffmpeg -i <input> -r 24 <output>
```

More complex ways involve filtering, see fps, mpdecimate, minterpolate filters.

STREAM MAPPING

Each file and its streams have a unique ID, starting with 0.

Examples:

- 0:0 is the first stream of the first input file
- 0:1 is the second stream of the first input file
- 2:a:0 is the first audio stream of the third input file
- …

You can map input streams to output, e.g. to add audio to a video:

```
ffmpeg -i input.mp4 -i input.m4a -c copy -map 0:v:0 -map 1:a:0 output.mp4
```

See: http://trac.ffmpeg.org/wiki/Map

SIMPLE FILTERING

ffmpeg has lots of video, audio, subtitle filters:

```
ffmpeg -i <input> -filter:v "<filter1>,<filter2>,<filter3>" <output>
```

A <filter> has a name and several options, and some pre-defined variables:

```
-filter:v <name>=<option1>=<value1>:<option2>=<value2>
```

Notes:

- You can use `-filter:a` for audio filters.
- Filters can be chained by separating them with a ,
- See all filters with `ffmpeg -filters`
- Check http://trac.ffmpeg.org/wiki/FilteringGuide and http://ffmpeg.org/ffmpeg-filters.html

SCALING

Scale to 320×240:

```
ffmpeg -i <input> -vf "scale=w=320:h=240" <output>
```

Scale to a height of 240 and keep aspect ratio divisible by 2:

```
ffmpeg -i <input> -vf scale=w=-2:h=240 <output>
```

Scale to 1280×720 or smaller if needed:

```
ffmpeg -i <input> -vf "scale=1280:720:force_original_aspect_ratio=decrease" <output>
```

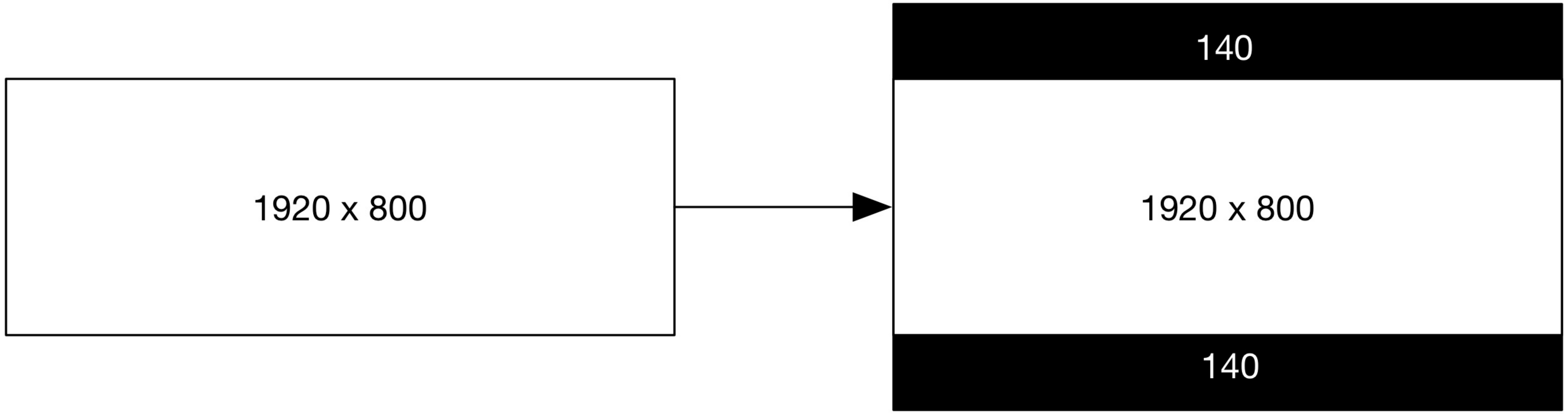
More tips:

- http://trac.ffmpeg.org/wiki/Scaling%20(resizing)%20with%20ffmpeg
- https://superuser.com/questions/547296/

PADDING

Add black borders to a file, e.g. 1920×800 input to 1920×1080:

```
ffmpeg -i <input> -vf "pad=1920:1080:(ow-iw)/2:(oh-ih)/2" <output>
```



Note that:

- You can use mathematical expressions
- ow and oh are output width and height
- iw and ih are input width and height

FADING

Simple fade-in and fade-out at a specific time for a specific duration.

```
ffmpeg -i <input> -filter:v \
  "fade=t=in:st=0:d=5,fade=t=out:st=30:d=5" \
  <output>
```

Notes:

- `t` sets the fade type (in or out)
- `d` sets the duration
- `st` sets the start time in seconds or `HH:MM:SS.msec`
- `ffmpeg` can't "search from the back"; you have to find the total duration yourself (e.g. with `ffprobe`)

COMPLEX FILTERING

Complex filters have more than one in- and/or output:

```
ffmpeg -i <input1> -i <input2> -filter_complex \
    "[0:v:0][1:v:0]overlay[outv]" \
    -map "[outv]" <output>
```

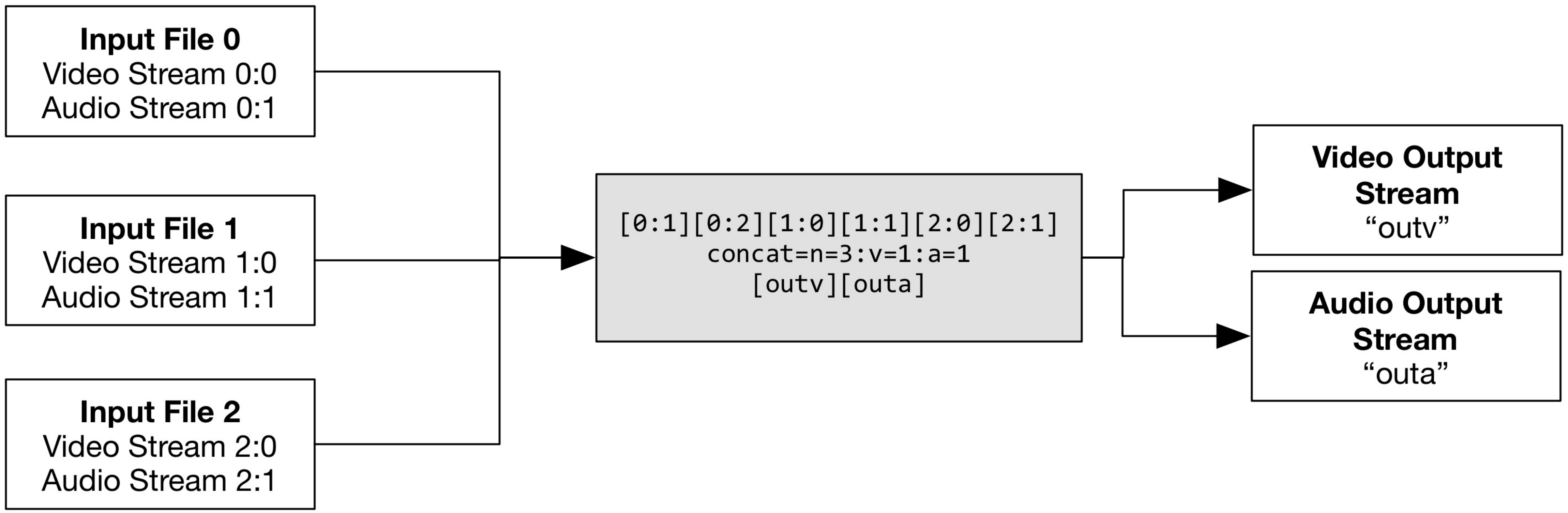
Steps:

- Specify inputs to filterchain (e.g. [0:v:0][1:v:0])
- Specify filters in the chain (e.g. overlay)
- Specify output labels of chain (e.g. [outv])
- Map output labels to final output file
- You can have multiple filterchains with ;

See: http://ffmpeg.org/ffmpeg-all.html#Filtergraph-syntax-1

CONCATENATING STREAMS

Decode three video/audio streams and append to one another:




```
ffmpeg -i <input1> -i <input2> -i <input3> -filter_complex \
    "[0:1][0:2][1:0][1:1][2:0][2:1]concat=n=3:v=1:a=1[outv][outa]" \
    -map "[outv]" -map "[outa]" <output>
```

See: http://trac.ffmpeg.org/wiki/Concatenate (also for other methods)

EXERCISE

Take an original 1920×1080 video file and downscale it to a height of 240 pixels. Keep the same aspect ratio. Keep the original audio stream without re-encoding.

Tasks/Questions:

- How do you make ffmpeg find out the output width?
- What is the final output width? What should the output width have been according to math?
- Why are these values sometimes different?
- Take the downscaled version of the video and append it to the original video by concatenating, and encode the final file with a lossless codec of your choice
    - *Hint: You have to first upscale it again*
    - Bonus points if you can do this in one command

TIMELINE EDITING

Enable filters only at a specific point in time.

Example:

- Show a watermark in the top left corner
- Between seconds 1 and 2 only

```
ffmpeg -i <video> -i <watermark> -filter_complex \
    "[0:v][1:v]overlay=10:10:enable='between(t,1,2)'[outv]" \
    -map "[outv]" <output>
```

See: http://ffmpeg.org/ffmpeg-all.html#Timeline-editing

CALCULATING SIMPLE QUALITY METRICS

PSNR (Peak Signal To Noise Ratio):

```
$ ffmpeg -i <degraded> -i <reference> -filter_complex psnr -f null /dev/null
[Parsed_psnr_0 @ 0x7fdb187045c0] PSNR y:33.437789 u:39.814416 v:39.319141 average:34.698320 min:
```

SSIM (Structural Similarity):

```
$ ffmpeg -i <degraded> -i <reference> -filter_complex ssim -f null /dev/null
[Parsed_ssim_0 @ 0x7fbf0500b660] SSIM Y:0.925477 (11.277116) U:0.948906 (12.916325) V:0.946795 (1
```

Notes:

- Optionally add: `2>&1 | grep SSIM` to filter only relevant output
- Windows users use `NUL` instead of `/dev/null`
- PSNR is an unreliable and inaccurate quality metric, SSIM is better but not perfect
- Try to use "proper" video quality metrics instead, e.g. VQM or VMAF

EXERCISE PT. 1

Tasks:

- Use two-pass encoding to transcode the sample video to H.264 with x264
- Use the following bitrates: 1M, 2M, 4M, 6M, 8M
- Encode the video with all existing speed presets for libx264 and all chosen bitrates.

Hints:

- This may take some time overall depending on your CPU speed
- It's easier to write a simple Batch or Bash script than type all the commands

EXERCISE PT. 2

Questions:

1. How long does the encoding take for a given speed preset, on average?
2. For every target bitrate, draw a curve that shows the time taken (y-axis) vs. preset used (x-axis). (*Bonus points if you overlay the curves on top of each other in one plot, e.g. through different colors.*)
3. Do you see a difference in quality between the encoded clips?
4. Calculate a quality measure for the encoded videos with the different presets.
5. For every target bitrate, draw a curve that shows the quality (y-axis) vs. preset used (x-axis). (*Bonus points if you overlay the curves on top of each other in one plot, e.g. through different colors.*)

Hints:

- You can use the built-in `ssim` filter as a rough measure for quality
- On Linux you can use the `time` command, on Windows this is a little harder (do a web search)
- This can be done with Excel, but other tools like Python or R are useful as well

GETTING MEDIA INFORMATION WITH FFPROBE

GENERAL CONCEPTS

```
ffprobe <input>
    [-select_streams <selection>]
    [-show_streamsl-show_formatl-show_framesl-show_packets]
    [-show_entries <entries>]
    [-of <output-format>]
```

Explanation:

- select_streams for specificing only video or audio, for example
- show_ for selecting which information to show
- show_entries for selecting fewer entries to show
- of to set output format

See:

- https://ffmpeg.org/ffprobe.html
- http://trac.ffmpeg.org/wiki/FFprobeTips

PRACTICAL FFPROBE EXAMPLES PT. 1

Show all available streams:

```
ffprobe <input> -show_streams
```

Show info on video stream:

```
ffprobe <input> -select_streams v -show_format
```

Show presentation timestamp and frame type of every frame, in CSV format (p=0 disables CSV section headers)

```
ffprobe <input> -show_frames -show_entries frame=pkt_pts_time,pict_type -of csv=p=0
```

PRACTICAL FFPROBE EXAMPLES PT. 2

Change output to JSON format for parsing:

```
ffprobe <input> -select_streams v -show_packets -of json
```

Get the number of streams in a file (nk=1 disables keys):

```
ffprobe <input> -show_format -show_entries format=nb_streams -of compact=nk=1:p=0
```

Get the duration in seconds or HH:MM:SS.ms:

```
ffprobe <input> -show_format -show_entries format=duration -of compact=nk=1:p=0
ffprobe -sexagesimal <input> -show_format -show_entries format=duration -of compact=nk=1:p=0
```

Get bitrate of audio stream in Bit/s:

```
ffprobe <input> -select_streams a -show_entries stream=bit_rate -of compact=nk=1:p=0
```

INSPECTING VIDEO CODECS

DEBUGGING MOTION VECTORS

Simple way to visualize motion in FFmpeg with MPEG codecs (H.264, H.265, …):

DEBUGGING MACROBLOCK TYPES

Visualize macroblock splits in FFmpeg with MPEG codecs (H.264, H.265, …):

VIDEO STREAM ANALYZERS

Different software for analyzing bitstreams graphically:

- https://arewecompressedyet.com/analyzer/

SUMMARY

SUMMARY

You should have learned how to:

- Understand FFmpeg libraries, codecs, containers, encoders, …
- Encode video and audio
- Apply basic filters
- Read stream information and metadata
- Find help if you get stuck