

NAME

ffmpeg - ffmpeg video converter

SYNOPSIS

ffmpeg [*global_options*] {[*input_file_options*] -i *input_url*} ... {[*output_file_options*] *output_url*} ...

DESCRIPTION

ffmpeg is a very fast video and audio converter that can also grab from a live audio/video source. It can also convert between arbitrary sample rates and resize video on the fly with a high quality polyphase filter.

ffmpeg reads from an arbitrary number of input "files" (which can be regular files, pipes, network streams, grabbing devices, etc.), specified by the "-i" option, and writes to an arbitrary number of output "files", which are specified by a plain output url. Anything found on the command line which cannot be interpreted as an option is considered to be an output url.

Each input or output url can, in principle, contain any number of streams of different types (video/audio/subtitle/attachment/data). The allowed number and/or types of streams may be limited by the container format. Selecting which streams from which inputs will go into which output is either done automatically or with the "-map" option (see the Stream selection chapter).

To refer to input files in options, you must use their indices (0-based). E.g. the first input file is 0, the second is 1, etc. Similarly, streams within a file are referred to by their indices. E.g. "2:3" refers to the fourth stream in the third input file. Also see the Stream specifiers chapter.

As a general rule, options are applied to the next specified file. Therefore, order is important, and you can have the same option on the command line multiple times. Each occurrence is then applied to the next input or output file. Exceptions from this rule are the global options (e.g. verbosity level), which should be specified first.

Do not mix input and output files -- first specify all input files, then all output files. Also do not mix options which belong to different files. All options apply **ONLY** to the next input or output file and are reset between files.

- ⊕ To set the video bitrate of the output file to 64 kbit/s:

```
ffmpeg -i input.avi -b:v 64k -bufsize 64k output.avi
```

- ⊕ To force the frame rate of the output file to 24 fps:

```
ffmpeg -i input.avi -r 24 output.avi
```

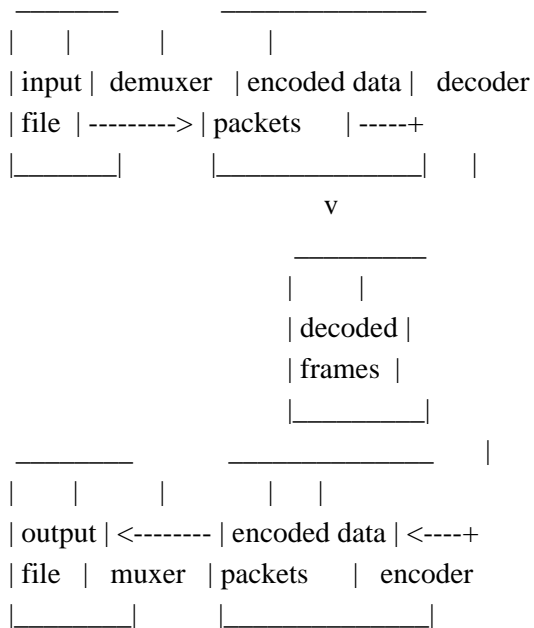
- ⊕ To force the frame rate of the input file (valid for raw formats only) to 1 fps and the frame rate of the output file to 24 fps:

```
ffmpeg -r 1 -i input.m2v -r 24 output.avi
```

The format option may be needed for raw input files.

DETAILED DESCRIPTION

The transcoding process in **ffmpeg** for each output can be described by the following diagram:



ffmpeg calls the libavformat library (containing demuxers) to read input files and get packets containing encoded data from them. When there are multiple input files, **ffmpeg** tries to keep them synchronized by tracking lowest timestamp on any active input stream.

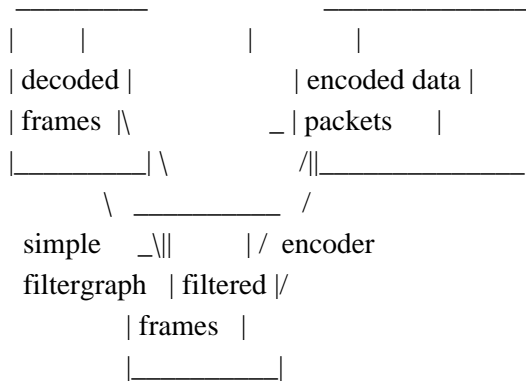
Encoded packets are then passed to the decoder (unless streamcopy is selected for the stream, see further for a description). The decoder produces uncompressed frames (raw video/PCM audio/...) which can be processed further by filtering (see next section). After filtering, the frames are passed to the encoder, which encodes them and outputs encoded packets. Finally those are passed to the muxer, which writes the encoded packets to the output file.

Filtering

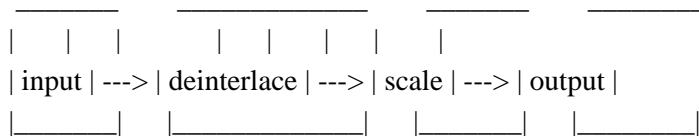
Before encoding, **ffmpeg** can process raw audio and video frames using filters from the libavfilter library. Several chained filters form a filter graph. **ffmpeg** distinguishes between two types of filtergraphs: simple and complex.

Simple filtergraphs

Simple filtergraphs are those that have exactly one input and output, both of the same type. In the above diagram they can be represented by simply inserting an additional step between decoding and encoding:



Simple filtergraphs are configured with the per-stream **-filter** option (with **-vf** and **-af** aliases for video and audio respectively). A simple filtergraph for video can look for example like this:



Note that some filters change frame properties but not frame contents. E.g. the "fps" filter in the example above changes number of frames, but does not touch the frame contents. Another example is the "setpts" filter, which only sets timestamps and otherwise passes the frames unchanged.

Complex filtergraphs

Complex filtergraphs are those which cannot be described as simply a linear processing chain applied to one stream. This is the case, for example, when the graph has more than one input and/or output, or when output stream type is different from input. They can be represented with the following diagram:



work on uncompressed data.

STREAM SELECTION

ffmpeg provides the "-map" option for manual control of stream selection in each output file. Users can skip "-map" and let ffmpeg perform automatic stream selection as described below. The "-vn / -an / -sn / -dn" options can be used to skip inclusion of video, audio, subtitle and data streams respectively, whether manually mapped or automatically selected, except for those streams which are outputs of complex filtergraphs.

Description

The sub-sections that follow describe the various rules that are involved in stream selection. The examples that follow next show how these rules are applied in practice.

While every effort is made to accurately reflect the behavior of the program, FFmpeg is under continuous development and the code may have changed since the time of this writing.

Automatic stream selection

In the absence of any map options for a particular output file, ffmpeg inspects the output format to check which type of streams can be included in it, viz. video, audio and/or subtitles. For each acceptable stream type, ffmpeg will pick one stream, when available, from among all the inputs.

It will select that stream based upon the following criteria:

- ⊕ for video, it is the stream with the highest resolution,
- ⊕ for audio, it is the stream with the most channels,
- ⊕ for subtitles, it is the first subtitle stream found but there's a caveat. The output format's default subtitle encoder can be either text-based or image-based, and only a subtitle stream of the same type will be chosen.

In the case where several streams of the same type rate equally, the stream with the lowest index is chosen.

Data or attachment streams are not automatically selected and can only be included using "-map".

Manual stream selection

When "-map" is used, only user-mapped streams are included in that output file, with one possible

exception for filtergraph outputs described below.

Complex filtergraphs

If there are any complex filtergraph output streams with unlabeled pads, they will be added to the first output file. This will lead to a fatal error if the stream type is not supported by the output format. In the absence of the map option, the inclusion of these streams leads to the automatic stream selection of their types being skipped. If map options are present, these filtergraph streams are included in addition to the mapped streams.

Complex filtergraph output streams with labeled pads must be mapped once and exactly once.

Stream handling

Stream handling is independent of stream selection, with an exception for subtitles described below. Stream handling is set via the "-codec" option addressed to streams within a specific *output* file. In particular, codec options are applied by ffmpeg after the stream selection process and thus do not influence the latter. If no "-codec" option is specified for a stream type, ffmpeg will select the default encoder registered by the output file muxer.

An exception exists for subtitles. If a subtitle encoder is specified for an output file, the first subtitle stream found of any type, text or image, will be included. ffmpeg does not validate if the specified encoder can convert the selected stream or if the converted stream is acceptable within the output format. This applies generally as well: when the user sets an encoder manually, the stream selection process cannot check if the encoded stream can be muxed into the output file. If it cannot, ffmpeg will abort and *all* output files will fail to be processed.

Examples

The following examples illustrate the behavior, quirks and limitations of ffmpeg's stream selection methods.

They assume the following three input files.

input file 'A.avi'

stream 0: video 640x360

stream 1: audio 2 channels

input file 'B.mp4'

stream 0: video 1920x1080

stream 1: audio 2 channels

```
stream 2: subtitles (text)
stream 3: audio 5.1 channels
stream 4: subtitles (text)

input file 'C.mkv'
stream 0: video 1280x720
stream 1: audio 2 channels
stream 2: subtitles (image)
```

Example: automatic stream selection

```
ffmpeg -i A.avi -i B.mp4 out1.mkv out2.wav -map 1:a -c:a copy out3.mov
```

There are three output files specified, and for the first two, no "-map" options are set, so ffmpeg will select streams for these two files automatically.

out1.mkv is a Matroska container file and accepts video, audio and subtitle streams, so ffmpeg will try to select one of each type. For video, it will select "stream 0" from *B.mp4*, which has the highest resolution among all the input video streams. For audio, it will select "stream 3" from *B.mp4*, since it has the greatest number of channels. For subtitles, it will select "stream 2" from *B.mp4*, which is the first subtitle stream from among *A.avi* and *B.mp4*.

out2.wav accepts only audio streams, so only "stream 3" from *B.mp4* is selected.

For *out3.mov*, since a "-map" option is set, no automatic stream selection will occur. The "-map 1:a" option will select all audio streams from the second input *B.mp4*. No other streams will be included in this output file.

For the first two outputs, all included streams will be transcoded. The encoders chosen will be the default ones registered by each output format, which may not match the codec of the selected input streams.

For the third output, codec option for audio streams has been set to "copy", so no decoding-filtering-encoding operations will occur, or *can* occur. Packets of selected streams shall be conveyed from the input file and muxed within the output file.

Example: automatic subtitles selection

```
ffmpeg -i C.mkv out1.mkv -c:s dvdsub -an out2.mkv
```

Although *out1.mkv* is a Matroska container file which accepts subtitle streams, only a video and audio stream shall be selected. The subtitle stream of *C.mkv* is image-based and the default subtitle encoder of the Matroska muxer is text-based, so a transcode operation for the subtitles is expected to fail and hence the stream isn't selected. However, in *out2.mkv*, a subtitle encoder is specified in the command and so, the subtitle stream is selected, in addition to the video stream. The presence of "-an" disables audio stream selection for *out2.mkv*.

Example: unlabeled filtergraph outputs

```
ffmpeg -i A.avi -i C.mkv -i B.mp4 -filter_complex "overlay" out1.mp4 out2.srt
```

A filtergraph is setup here using the "-filter_complex" option and consists of a single video filter. The "overlay" filter requires exactly two video inputs, but none are specified, so the first two available video streams are used, those of *A.avi* and *C.mkv*. The output pad of the filter has no label and so is sent to the first output file *out1.mp4*. Due to this, automatic selection of the video stream is skipped, which would have selected the stream in *B.mp4*. The audio stream with most channels viz. "stream 3" in *B.mp4*, is chosen automatically. No subtitle stream is chosen however, since the MP4 format has no default subtitle encoder registered, and the user hasn't specified a subtitle encoder.

The 2nd output file, *out2.srt*, only accepts text-based subtitle streams. So, even though the first subtitle stream available belongs to *C.mkv*, it is image-based and hence skipped. The selected stream, "stream 2" in *B.mp4*, is the first text-based subtitle stream.

Example: labeled filtergraph outputs

```
ffmpeg -i A.avi -i B.mp4 -i C.mkv -filter_complex "[1:v]hue=s=0[outv];overlay;aresample" \
    -map '[outv]' -an out1.mp4 \
    out2.mkv \
    -map '[outv]' -map 1:a:0 out3.mkv
```

The above command will fail, as the output pad labelled "[outv]" has been mapped twice. None of the output files shall be processed.

```
ffmpeg -i A.avi -i B.mp4 -i C.mkv -filter_complex "[1:v]hue=s=0[outv];overlay;aresample" \
    -an out1.mp4 \
    out2.mkv \
    -map 1:a:0 out3.mkv
```

This command above will also fail as the hue filter output has a label, "[outv]", and hasn't been mapped anywhere.

The command should be modified as follows,

```
ffmpeg -i A.avi -i B.mp4 -i C.mkv -filter_complex "[1:v]hue=s=0,split=2[outv1][outv2];overlay;aresample" \
    -map '[outv1]' -an out1.mp4 \
        out2.mkv \
    -map '[outv2]' -map 1:a:0 out3.mkv
```

The video stream from *B.mp4* is sent to the hue filter, whose output is cloned once using the split filter, and both outputs labelled. Then a copy each is mapped to the first and third output files.

The overlay filter, requiring two video inputs, uses the first two unused video streams. Those are the streams from *A.avi* and *C.mkv*. The overlay output isn't labelled, so it is sent to the first output file *out1.mp4*, regardless of the presence of the "-map" option.

The aresample filter is sent the first unused audio stream, that of *A.avi*. Since this filter output is also unlabelled, it too is mapped to the first output file. The presence of "-an" only suppresses automatic or manual stream selection of audio streams, not outputs sent from filtergraphs. Both these mapped streams shall be ordered before the mapped stream in *out1.mp4*.

The video, audio and subtitle streams mapped to "out2.mkv" are entirely determined by automatic stream selection.

out3.mkv consists of the cloned video output from the hue filter and the first audio stream from *B.mp4*.

OPTIONS

All the numerical options, if not specified otherwise, accept a string representing a number as input, which may be followed by one of the SI unit prefixes, for example: 'K', 'M', or 'G'.

If 'i' is appended to the SI unit prefix, the complete prefix will be interpreted as a unit prefix for binary multiples, which are based on powers of 1024 instead of powers of 1000. Appending 'B' to the SI unit prefix multiplies the value by 8. This allows using, for example: 'KB', 'MiB', 'G' and 'B' as number suffixes.

Options which do not take arguments are boolean options, and set the corresponding value to true. They can be set to false by prefixing the option name with "no". For example using "-nofoo" will set the boolean option with name "foo" to false.

Stream specifiers

Some options are applied per-stream, e.g. bitrate or codec. Stream specifiers are used to precisely specify which stream(s) a given option belongs to.

A stream specifier is a string generally appended to the option name and separated from it by a colon. E.g. "-codec:a:1 ac3" contains the "a:1" stream specifier, which matches the second audio stream. Therefore, it would select the ac3 codec for the second audio stream.

A stream specifier can match several streams, so that the option is applied to all of them. E.g. the stream specifier in "-b:a 128k" matches all audio streams.

An empty stream specifier matches all streams. For example, "-codec copy" or "-codec: copy" would copy all the streams without reencoding.

Possible forms of stream specifiers are:

stream_index

Matches the stream with this index. E.g. "-threads:1 4" would set the thread count for the second stream to 4. If *stream_index* is used as an additional stream specifier (see below), then it selects stream number *stream_index* from the matching streams. Stream numbering is based on the order of the streams as detected by libavformat except when a program ID is also specified. In this case it is based on the ordering of the streams in the program.

stream_type[:additional_stream_specifier]

stream_type is one of following: 'v' or 'V' for video, 'a' for audio, 's' for subtitle, 'd' for data, and 't' for attachments. 'v' matches all video streams, 'V' only matches video streams which are not attached pictures, video thumbnails or cover arts. If *additional_stream_specifier* is used, then it matches streams which both have this type and match the *additional_stream_specifier*. Otherwise, it matches all streams of the specified type.

p:*program_id[:additional_stream_specifier]*

Matches streams which are in the program with the id *program_id*. If *additional_stream_specifier* is used, then it matches streams which both are part of the program and match the *additional_stream_specifier*.

#stream_id or **i**:*stream_id*

Match the stream by stream id (e.g. PID in MPEG-TS container).

m:*key[:value]*

Matches streams with the metadata tag *key* having the specified value. If *value* is not given, matches streams that contain the given tag with any value.

u Matches streams with usable configuration, the codec must be defined and the essential information such as video dimension or audio sample rate must be present.

Note that in **ffmpeg**, matching by metadata will only work properly for input files.

Generic options

These options are shared amongst the ff* tools.

-L Show license.

-h, -?, -help, --help [*arg*]

Show help. An optional parameter may be specified to print help about a specific item. If no argument is specified, only basic (non advanced) tool options are shown.

Possible values of *arg* are:

long

Print advanced tool options in addition to the basic tool options.

full Print complete list of options, including shared and private options for encoders, decoders, demuxers, muxers, filters, etc.

decoder=*decoder_name*

Print detailed information about the decoder named *decoder_name*. Use the **-decoders** option to get a list of all decoders.

encoder=*encoder_name*

Print detailed information about the encoder named *encoder_name*. Use the **-encoders** option to get a list of all encoders.

demuxer=*demuxer_name*

Print detailed information about the demuxer named *demuxer_name*. Use the **-formats** option to get a list of all demuxers and muxers.

muxer=*muxer_name*

Print detailed information about the muxer named *muxer_name*. Use the **-formats** option to get a list of all muxers and demuxers.

filter=*filter_name*

Print detailed information about the filter named *filter_name*. Use the **-filters** option to get a list of all filters.

bsf=*bitstream_filter_name*

Print detailed information about the bitstream filter named *bitstream_filter_name*. Use the **-bsfs** option to get a list of all bitstream filters.

protocol=*protocol_name*

Print detailed information about the protocol named *protocol_name*. Use the **-protocols** option to get a list of all protocols.

-version

Show version.

-buildconf

Show the build configuration, one option per line.

-formats

Show available formats (including devices).

-demuxers

Show available demuxers.

-muxers

Show available muxers.

-devices

Show available devices.

-codecs

Show all codecs known to libavcodec.

Note that the term 'codec' is used throughout this documentation as a shortcut for what is more correctly called a media bitstream format.

-decoders

Show available decoders.

-encoders

Show all available encoders.

-bsfs

Show available bitstream filters.

-protocols

Show available protocols.

-filters

Show available libavfilter filters.

-pix_fmts

Show available pixel formats.

-sample_fmts

Show available sample formats.

-layouts

Show channel names and standard channel layouts.

-dispositions

Show stream dispositions.

-colors

Show recognized color names.

-sources *device[,opt1=val1[,opt2=val2]...]*

Show autodetected sources of the input device. Some devices may provide system-dependent source names that cannot be autodetected. The returned list cannot be assumed to be always complete.

```
ffmpeg -sources pulse,server=192.168.0.4
```

-sinks *device[,opt1=val1[,opt2=val2]...]*

Show autodetected sinks of the output device. Some devices may provide system-dependent sink names that cannot be autodetected. The returned list cannot be assumed to be always complete.

```
ffmpeg -sinks pulse,server=192.168.0.4
```

-loglevel [*flags+*]*loglevel* | **-v** [*flags+*]*loglevel*

Set logging level and flags used by the library.

The optional *flags* prefix can consist of the following values:

repeat

Indicates that repeated log output should not be compressed to the first line and the "Last message repeated n times" line will be omitted.

level

Indicates that log output should add a "[level]" prefix to each message line. This can be used as an alternative to log coloring, e.g. when dumping the log to file.

Flags can also be used alone by adding a '+'/'-' prefix to set/reset a single flag without affecting other *flags* or changing *loglevel*. When setting both *flags* and *loglevel*, a '+' separator is expected between the last *flags* value and before *loglevel*.

loglevel is a string or a number containing one of the following values:

quiet, -8

Show nothing at all; be silent.

panic, 0

Only show fatal errors which could lead the process to crash, such as an assertion failure. This is not currently used for anything.

fatal, 8

Only show fatal errors. These are errors after which the process absolutely cannot continue.

error, 16

Show all errors, including ones which can be recovered from.

warning, 24

Show all warnings and errors. Any message related to possibly incorrect or unexpected events will be shown.

info, 32

Show informative messages during processing. This is in addition to warnings and errors. This is the default value.

verbose, 40

Same as "info", except more verbose.

debug, 48

Show everything, including debugging information.

trace, 56

For example to enable repeated log output, add the "level" prefix, and set *loglevel* to "verbose":

```
ffmpeg -loglevel repeat+level+verbose -i input output
```

Another example that enables repeated log output without affecting current state of "level" prefix flag or *loglevel*:

```
ffmpeg [...] -loglevel +repeat
```

By default the program logs to stderr. If coloring is supported by the terminal, colors are used to mark errors and warnings. Log coloring can be disabled setting the environment variable **AV_LOG_FORCE_NOCOLOR**, or can be forced setting the environment variable **AV_LOG_FORCE_COLOR**.

-report

Dump full command line and log output to a file named "*program-YYYYMMDD-HHMMSS.log*" in the current directory. This file can be useful for bug reports. It also implies "-loglevel debug".

Setting the environment variable **FFREPORT** to any value has the same effect. If the value is a ':'-separated key=value sequence, these options will affect the report; option values must be escaped if they contain special characters or the options delimiter ':' (see the "Quoting and escaping" section in the ffmpeg-utils manual).

The following options are recognized:

file set the file name to use for the report; %p is expanded to the name of the program, %t is expanded to a timestamp, "%%" is expanded to a plain "%"

level

set the log verbosity level using a numerical value (see "-loglevel").

For example, to output a report to a file named *ffreport.log* using a log level of 32 (alias for log level "info"):

```
FFREPORT=file=ffreport.log:level=32 ffmpeg -i input output
```

Errors in parsing the environment variable are not fatal, and will not appear in the report.

-hide_banner

Suppress printing banner.

All Ffmpeg tools will normally show a copyright notice, build options and library versions. This option can be used to suppress printing this information.

-cpuflags flags (*global*)

Allows setting and clearing cpu flags. This option is intended for testing. Do not use it unless you know what you're doing.

```
ffmpeg -cpuflags -sse+mmx ...
```

```
ffmpeg -cpuflags mmx ...
```

```
ffmpeg -cpuflags 0 ...
```

Possible flags for this option are:

x86

mmx

mmxext

sse

sse2

sse2slow

sse3

sse3slow

ssse3

atom

sse4.1

sse4.2

avx

avx2

xop

fma3

fma4

3dnow

3dnowext

bmi1

bmi2

cmov

ARM

armv5te

armv6
armv6t2
vfp
vfpv3
neon
setend

AArch64

armv8
vfp
neon

PowerPC

altivec

Specific Processors

pentium2
pentium3
pentium4
k6
k62
athlon
athlonxp
k8

-cpucount *count* (*global*)

Override detection of CPU count. This option is intended for testing. Do not use it unless you know what you're doing.

```
ffmpeg -cpucount 2
```

-max_alloc *bytes*

Set the maximum size limit for allocating a block on the heap by ffmpeg's family of malloc functions. Exercise **extreme caution** when using this option. Don't use if you do not understand the full consequence of doing so. Default is INT_MAX.

AVOptions

These options are provided directly by the libavformat, libavdevice and libavcodec libraries. To see the list of available AVOptions, use the **-help** option. They are separated into two categories:

generic

These options can be set for any container, codec or device. Generic options are listed under AVFormatContext options for containers/devices and under AVCodecContext options for codecs.

private

These options are specific to the given container, device or codec. Private options are listed under their corresponding containers/devices/codecs.

For example to write an ID3v2.3 header instead of a default ID3v2.4 to an MP3 file, use the **id3v2_version** private option of the MP3 muxer:

```
ffmpeg -i input.flac -id3v2_version 3 out.mp3
```

All codec AVOptions are per-stream, and thus a stream specifier should be attached to them:

```
ffmpeg -i multichannel.mxf -map 0:v:0 -map 0:a:0 -map 0:a:0 -c:a:0 ac3 -b:a:0 640k -ac:a:1 2 -c:a:1 aac -b:2 128k
```

In the above example, a multichannel audio stream is mapped twice for output. The first instance is encoded with codec ac3 and bitrate 640k. The second instance is downmixed to 2 channels and encoded with codec aac. A bitrate of 128k is specified for it using absolute index of the output stream.

Note: the **-nooption** syntax cannot be used for boolean AVOptions, use **-option 0/-option 1**.

Note: the old undocumented way of specifying per-stream AVOptions by prepending v/a/s to the options name is now obsolete and will be removed soon.

Main options**-f *fmt*** (*input/output*)

Force input or output file format. The format is normally auto detected for input files and guessed from the file extension for output files, so this option is not needed in most cases.

-i *url* (*input*)

input file url

-y (*global*)

Overwrite output files without asking.

-n (*global*)

Do not overwrite output files, and exit immediately if a specified output file already exists.

-stream_loop *number* (*input*)

Set number of times input stream shall be looped. Loop 0 means no loop, loop -1 means infinite loop.

-recast_media (*global*)

Allow forcing a decoder of a different media type than the one detected or designated by the demuxer. Useful for decoding media data muxed as data streams.

-c[:stream_specifier] *codec* (*input/output,per-stream*)**-codec[:stream_specifier]** *codec* (*input/output,per-stream*)

Select an encoder (when used before an output file) or a decoder (when used before an input file) for one or more streams. *codec* is the name of a decoder/encoder or a special value "copy" (output only) to indicate that the stream is not to be re-encoded.

For example

```
ffmpeg -i INPUT -map 0 -c:v libx264 -c:a copy OUTPUT
```

encodes all video streams with libx264 and copies all audio streams.

For each stream, the last matching "c" option is applied, so

```
ffmpeg -i INPUT -map 0 -c copy -c:v:1 libx264 -c:a:137 libvorbis OUTPUT
```

will copy all the streams except the second video, which will be encoded with libx264, and the 138th audio, which will be encoded with libvorbis.

-t *duration* (*input/output*)

When used as an input option (before "-i"), limit the *duration* of data read from the input file.

When used as an output option (before an output url), stop writing the output after its duration reaches *duration*.

duration must be a time duration specification, see **the Time duration section in the ffmpeg-utils(1) manual**.

-to and -t are mutually exclusive and -t has priority.

-to *position* (*input/output*)

Stop writing the output or reading the input at *position*. *position* must be a time duration specification, see **the Time duration section in the ffmpeg-utils(1) manual**.

-to and -t are mutually exclusive and -t has priority.

-fs *limit_size (output)*

Set the file size limit, expressed in bytes. No further chunk of bytes is written after the limit is exceeded. The size of the output file is slightly more than the requested file size.

-ss *position (input/output)*

When used as an input option (before "-i"), seeks in this input file to *position*. Note that in most formats it is not possible to seek exactly, so **ffmpeg** will seek to the closest seek point before *position*. When transcoding and **-accurate_seek** is enabled (the default), this extra segment between the seek point and *position* will be decoded and discarded. When doing stream copy or when **-noaccurate_seek** is used, it will be preserved.

When used as an output option (before an output url), decodes but discards input until the timestamps reach *position*.

position must be a time duration specification, see **the Time duration section in the ffmpeg-utils(1) manual**.

-sseof *position (input)*

Like the "-ss" option but relative to the "end of file". That is negative values are earlier in the file, 0 is at EOF.

-isync *input_index (input)*

Assign an input as a sync source.

This will take the difference between the start times of the target and reference inputs and offset the timestamps of the target file by that difference. The source timestamps of the two inputs should derive from the same clock source for expected results. If "copyts" is set then "start_at_zero" must also be set. If either of the inputs has no starting timestamp then no sync adjustment is made.

Acceptable values are those that refer to a valid ffmpeg input index. If the sync reference is the target index itself or *-1*, then no adjustment is made to target timestamps. A sync reference may not itself be synced to any other input.

Default value is *-1*.

-itsoffset *offset (input)*

Set the input time offset.

offset must be a time duration specification, see **the Time duration section in the ffmpeg-utils(1)**

manual.

The offset is added to the timestamps of the input files. Specifying a positive offset means that the corresponding streams are delayed by the time duration specified in *offset*.

-itsscale *scale* (*input,per-stream*)

Rescale input timestamps. *scale* should be a floating point number.

-timestamp *date* (*output*)

Set the recording timestamp in the container.

date must be a date specification, see **the Date section in the ffmpeg-utils(1) manual**.

-metadata[:metadata_specifier] *key=value* (*output,per-metadata*)

Set a metadata key/value pair.

An optional *metadata_specifier* may be given to set metadata on streams, chapters or programs. See "-map_metadata" documentation for details.

This option overrides metadata set with "-map_metadata". It is also possible to delete metadata by using an empty value.

For example, for setting the title in the output file:

```
ffmpeg -i in.avi -metadata title="my title" out.flv
```

To set the language of the first audio stream:

```
ffmpeg -i INPUT -metadata:s:a:0 language=eng OUTPUT
```

-disposition[:stream_specifier] *value* (*output,per-stream*)

Sets the disposition for a stream.

By default, the disposition is copied from the input stream, unless the output stream this option applies to is fed by a complex filtergraph - in that case the disposition is unset by default.

value is a sequence of items separated by '+' or '-'. The first item may also be prefixed with '+' or '-', in which case this option modifies the default value. Otherwise (the first item is not prefixed) this options overrides the default value. A '+' prefix adds the given disposition, '-' removes it. It is also possible to clear the disposition by setting it to 0.

If no "-disposition" options were specified for an output file, ffmpeg will automatically set the 'default' disposition on the first stream of each type, when there are multiple streams of this type in the output file and no stream of that type is already marked as default.

The "-dispositions" option lists the known dispositions.

For example, to make the second audio stream the default stream:

```
ffmpeg -i in.mkv -c copy -disposition:a:1 default out.mkv
```

To make the second subtitle stream the default stream and remove the default disposition from the first subtitle stream:

```
ffmpeg -i in.mkv -c copy -disposition:s:0 0 -disposition:s:1 default out.mkv
```

To add an embedded cover/thumbnail:

```
ffmpeg -i in.mp4 -i IMAGE -map 0 -map 1 -c copy -c:v:1 png -disposition:v:1 attached_pic out.mp4
```

Not all muxers support embedded thumbnails, and those who do, only support a few formats, like JPEG or PNG.

-program [title=title:][program_num=program_num:]st=stream[:st=stream...] (output)

Creates a program with the specified *title*, *program_num* and adds the specified *stream*(s) to it.

-target type (output)

Specify target file type ("vcd", "svcd", "dvd", "dv", "dv50"). *type* may be prefixed with "pal-", "ntsc-" or "film-" to use the corresponding standard. All the format options (bitrate, codecs, buffer sizes) are then set automatically. You can just type:

```
ffmpeg -i myfile.avi -target vcd /tmp/vcd.mpg
```

Nevertheless you can specify additional options as long as you know they do not conflict with the standard, as in:

```
ffmpeg -i myfile.avi -target vcd -bf 2 /tmp/vcd.mpg
```

The parameters set for each target are as follows.

VCD

<pal>:

```
-f vcd -muxrate 1411200 -muxpreload 0.44 -packetize 2324
-s 352x288 -r 25
-codec:v mpeg1video -g 15 -b:v 1150k -maxrate:v 1150k -minrate:v 1150k -bufsize:v 327680
-ar 44100 -ac 2
-codec:a mp2 -b:a 224k
```

<ntsc>:

```
-f vcd -muxrate 1411200 -muxpreload 0.44 -packetize 2324
-s 352x240 -r 30000/1001
-codec:v mpeg1video -g 18 -b:v 1150k -maxrate:v 1150k -minrate:v 1150k -bufsize:v 327680
-ar 44100 -ac 2
-codec:a mp2 -b:a 224k
```

<film>:

```
-f vcd -muxrate 1411200 -muxpreload 0.44 -packetize 2324
-s 352x240 -r 24000/1001
-codec:v mpeg1video -g 18 -b:v 1150k -maxrate:v 1150k -minrate:v 1150k -bufsize:v 327680
-ar 44100 -ac 2
-codec:a mp2 -b:a 224k
```

SVCD

<pal>:

```
-f svcd -packetize 2324
-s 480x576 -pix_fmt yuv420p -r 25
-codec:v mpeg2video -g 15 -b:v 2040k -maxrate:v 2516k -minrate:v 0 -bufsize:v 1835008 -scan_offset 1
-ar 44100
-codec:a mp2 -b:a 224k
```

<ntsc>:

```
-f svcd -packetize 2324
-s 480x480 -pix_fmt yuv420p -r 30000/1001
-codec:v mpeg2video -g 18 -b:v 2040k -maxrate:v 2516k -minrate:v 0 -bufsize:v 1835008 -scan_offset 1
-ar 44100
-codec:a mp2 -b:a 224k
```

<film>:

```
-f svcd -packetize 2324
-s 480x480 -pix_fmt yuv420p -r 24000/1001
```

```
-codec:v mpeg2video -g 18 -b:v 2040k -maxrate:v 2516k -minrate:v 0 -bufsize:v 1835008 -scan_offset 1  
-ar 44100  
-codec:a mp2 -b:a 224k
```

DVD

```
<pal>:  
-f dvd -muxrate 10080k -packetize 2048  
-s 720x576 -pix_fmt yuv420p -r 25  
-codec:v mpeg2video -g 15 -b:v 6000k -maxrate:v 9000k -minrate:v 0 -bufsize:v 1835008  
-ar 48000  
-codec:a ac3 -b:a 448k
```

```
<ntsc>:  
-f dvd -muxrate 10080k -packetize 2048  
-s 720x480 -pix_fmt yuv420p -r 30000/1001  
-codec:v mpeg2video -g 18 -b:v 6000k -maxrate:v 9000k -minrate:v 0 -bufsize:v 1835008  
-ar 48000  
-codec:a ac3 -b:a 448k
```

```
<film>:  
-f dvd -muxrate 10080k -packetize 2048  
-s 720x480 -pix_fmt yuv420p -r 24000/1001  
-codec:v mpeg2video -g 18 -b:v 6000k -maxrate:v 9000k -minrate:v 0 -bufsize:v 1835008  
-ar 48000  
-codec:a ac3 -b:a 448k
```

DV

```
<pal>:  
-f dv  
-s 720x576 -pix_fmt yuv420p -r 25  
-ar 48000 -ac 2
```

```
<ntsc>:  
-f dv  
-s 720x480 -pix_fmt yuv411p -r 30000/1001  
-ar 48000 -ac 2
```

```
<film>:
```



```
-f dv
-s 720x480 -pix_fmt yuv411p -r 24000/1001
-ar 48000 -ac 2
```

The "dv50" target is identical to the "dv" target except that the pixel format set is "yuv422p" for all three standards.

Any user-set value for a parameter above will override the target preset value. In that case, the output may not comply with the target standard.

-dn (*input/output*)

As an input option, blocks all data streams of a file from being filtered or being automatically selected or mapped for any output. See "-discard" option to disable streams individually.

As an output option, disables data recording i.e. automatic selection or mapping of any data stream. For full manual control see the "-map" option.

-dframes *number* (*output*)

Set the number of data frames to output. This is an obsolete alias for "-frames:d", which you should use instead.

-frames[:*stream_specifier*] *framecount* (*output,per-stream*)

Stop writing to the stream after *framecount* frames.

-q[:*stream_specifier*] *q* (*output,per-stream*)

-qscale[:*stream_specifier*] *q* (*output,per-stream*)

Use fixed quality scale (VBR). The meaning of *q/qscale* is codec-dependent. If *qscale* is used without a *stream_specifier* then it applies only to the video stream, this is to maintain compatibility with previous behavior and as specifying the same codec specific value to 2 different codecs that is audio and video generally is not what is intended when no *stream_specifier* is used.

-filter[:*stream_specifier*] *filtergraph* (*output,per-stream*)

Create the filtergraph specified by *filtergraph* and use it to filter the stream.

filtergraph is a description of the filtergraph to apply to the stream, and must have a single input and a single output of the same type of the stream. In the filtergraph, the input is associated to the label "in", and the output to the label "out". See the ffmpeg-filters manual for more information about the filtergraph syntax.

See the **-filter_complex option** if you want to create filtergraphs with multiple inputs and/or outputs.

-filter_script*[:stream_specifier] filename (output,per-stream)*

This option is similar to **-filter**, the only difference is that its argument is the name of the file from which a filtergraph description is to be read.

-reinit_filter*[:stream_specifier] integer (input,per-stream)*

This boolean option determines if the filtergraph(s) to which this stream is fed gets reinitialized when input frame parameters change mid-stream. This option is enabled by default as most video and all audio filters cannot handle deviation in input frame properties. Upon reinitialization, existing filter state is lost, like e.g. the frame count "n" reference available in some filters. Any frames buffered at time of reinitialization are lost. The properties where a change triggers reinitialization are, for video, frame resolution or pixel format; for audio, sample format, sample rate, channel count or channel layout.

-filter_threads *nb_threads (global)*

Defines how many threads are used to process a filter pipeline. Each pipeline will produce a thread pool with this many threads available for parallel processing. The default is the number of available CPUs.

-pre*[:stream_specifier] preset_name (output,per-stream)*

Specify the preset for matching stream(s).

-stats *(global)*

Print encoding progress/statistics. It is on by default, to explicitly disable it you need to specify "-nostats".

-stats_period *time (global)*

Set period at which encoding progress/statistics are updated. Default is 0.5 seconds.

-progress *url (global)*

Send program-friendly progress information to *url*.

Progress information is written periodically and at the end of the encoding process. It is made of "*key=value*" lines. *key* consists of only alphanumeric characters. The last key of a sequence of progress information is always "progress".

The update period is set using "-stats_period".

-stdin

Enable interaction on standard input. On by default unless standard input is used as an input. To explicitly disable interaction you need to specify "-nostdin".

Disabling interaction on standard input is useful, for example, if ffmpeg is in the background process group. Roughly the same result can be achieved with "ffmpeg ... < /dev/null" but it requires a shell.

-debug_ts (*global*)

Print timestamp information. It is off by default. This option is mostly useful for testing and debugging purposes, and the output format may change from one version to another, so it should not be employed by portable scripts.

See also the option "-fdebug ts".

-attach *filename* (*output*)

Add an attachment to the output file. This is supported by a few formats like Matroska for e.g. fonts used in rendering subtitles. Attachments are implemented as a specific type of stream, so this option will add a new stream to the file. It is then possible to use per-stream options on this stream in the usual way. Attachment streams created with this option will be created after all the other streams (i.e. those created with "-map" or automatic mappings).

Note that for Matroska you also have to set the mimetype metadata tag:

```
ffmpeg -i INPUT -attach DejaVuSans.ttf -metadata:s:2 mimetype=application/x-truetype-font out.mkv
```

(assuming that the attachment stream will be third in the output file).

-dump_attachment[:*stream_specifier*]*filename* (*input,per-stream*)

Extract the matching attachment stream into a file named *filename*. If *filename* is empty, then the value of the "filename" metadata tag will be used.

E.g. to extract the first attachment to a file named 'out.ttf':

```
ffmpeg -dump_attachment:t:0 out.ttf -i INPUT
```

To extract all attachments to files determined by the "filename" tag:

```
ffmpeg -dump_attachment:t "" -i INPUT
```

Technical note -- attachments are implemented as codec extradata, so this option can actually be used to extract extradata from any stream, not just attachments.

Video Options

-vframes *number (output)*

Set the number of video frames to output. This is an obsolete alias for "-frames:v", which you should use instead.

-r*[:stream_specifier] fps (input/output,per-stream)*

Set frame rate (Hz value, fraction or abbreviation).

As an input option, ignore any timestamps stored in the file and instead generate timestamps assuming constant frame rate *fps*. This is not the same as the **-framerate** option used for some input formats like image2 or v4l2 (it used to be the same in older versions of FFmpeg). If in doubt use **-framerate** instead of the input option **-r**.

As an output option:

video encoding

Duplicate or drop frames right before encoding them to achieve constant output frame rate *fps*.

video streamcopy

Indicate to the muxer that *fps* is the stream frame rate. No data is dropped or duplicated in this case. This may produce invalid files if *fps* does not match the actual stream frame rate as determined by packet timestamps. See also the "setts" bitstream filter.

-fpsmax*[:stream_specifier] fps (output,per-stream)*

Set maximum frame rate (Hz value, fraction or abbreviation).

Clamps output frame rate when output framerate is auto-set and is higher than this value. Useful in batch processing or when input framerate is wrongly detected as very high. It cannot be set together with "-r". It is ignored during streamcopy.

-s*[:stream_specifier] size (input/output,per-stream)*

Set frame size.

As an input option, this is a shortcut for the **video_size** private option, recognized by some demuxers for which the frame size is either not stored in the file or is configurable -- e.g. raw video or video grabbers.

As an output option, this inserts the "scale" video filter to the *end* of the corresponding filtergraph. Please use the "scale" filter directly to insert it at the beginning or some other place.

The format is **wxh** (default - same as source).

-aspect[:stream_specifier] aspect (output,per-stream)

Set the video display aspect ratio specified by *aspect*.

aspect can be a floating point number string, or a string of the form *num:den*, where *num* and *den* are the numerator and denominator of the aspect ratio. For example "4:3", "16:9", "1.3333", and "1.7777" are valid argument values.

If used together with **-vcodec copy**, it will affect the aspect ratio stored at container level, but not the aspect ratio stored in encoded frames, if it exists.

-display_rotation[:stream_specifier] rotation (input,per-stream)

Set video rotation metadata.

rotation is a decimal number specifying the amount in degree by which the video should be rotated counter-clockwise before being displayed.

This option overrides the rotation/display transform metadata stored in the file, if any. When the video is being transcoded (rather than copied) and "-autorotate" is enabled, the video will be rotated at the filtering stage. Otherwise, the metadata will be written into the output file if the muxer supports it.

If the "-display_hflip" and/or "-display_vflip" options are given, they are applied after the rotation specified by this option.

-display_hflip[:stream_specifier] (input,per-stream)

Set whether on display the image should be horizontally flipped.

See the "-display_rotation" option for more details.

-display_vflip[:stream_specifier] (input,per-stream)

Set whether on display the image should be vertically flipped.

See the "-display_rotation" option for more details.

-vn (input/output)

As an input option, blocks all video streams of a file from being filtered or being automatically selected or mapped for any output. See "-discard" option to disable streams individually.

As an output option, disables video recording i.e. automatic selection or mapping of any video stream. For full manual control see the "-map" option.

-vcodec *codec (output)*

Set the video codec. This is an alias for "-codec:v".

-pass*[:stream_specifier] n (output,per-stream)*

Select the pass number (1 or 2). It is used to do two-pass video encoding. The statistics of the video are recorded in the first pass into a log file (see also the option -passlogfile), and in the second pass that log file is used to generate the video at the exact requested bitrate. On pass 1, you may just deactivate audio and set output to null, examples for Windows and Unix:

```
ffmpeg -i foo.mov -c:v libxvid -pass 1 -an -f rawvideo -y NUL
ffmpeg -i foo.mov -c:v libxvid -pass 1 -an -f rawvideo -y /dev/null
```

-passlogfile*[:stream_specifier] prefix (output,per-stream)*

Set two-pass log file name prefix to *prefix*, the default file name prefix is "ffmpeg2pass". The complete file name will be *PREFIX-N.log*, where N is a number specific to the output stream

-vf *filtergraph (output)*

Create the filtergraph specified by *filtergraph* and use it to filter the stream.

This is an alias for "-filter:v", see the **-filter option**.

-autorotate

Automatically rotate the video according to file metadata. Enabled by default, use **-noautorotate** to disable it.

-autoscale

Automatically scale the video according to the resolution of first frame. Enabled by default, use **-noautoscale** to disable it. When autoscale is disabled, all output frames of filter graph might not be in the same resolution and may be inadequate for some encoder/muxer. Therefore, it is not recommended to disable it unless you really know what you are doing. Disable autoscale at your own risk.

Advanced Video options

-pix_fmt*[:stream_specifier] format (input/output,per-stream)*

Set pixel format. Use "-pix_fmts" to show all the supported pixel formats. If the selected pixel format can not be selected, ffmpeg will print a warning and select the best pixel format supported by the encoder. If *pix_fmt* is prefixed by a "+", ffmpeg will exit with an error if the requested pixel format can not be selected, and automatic conversions inside filtergraphs are disabled. If *pix_fmt* is a single "+", ffmpeg selects the same pixel format as the input (or graph output) and automatic conversions are disabled.

-sws_flags *flags* (*input/output*)

Set SwScaler flags.

-rc_override[*:stream_specifier*] *override* (*output,per-stream*)

Rate control override for specific intervals, formatted as "int,int,int" list separated with slashes. Two first values are the beginning and end frame numbers, last one is quantizer to use if positive, or quality factor if negative.

-psnr

Calculate PSNR of compressed frames. This option is deprecated, pass the PSNR flag to the encoder instead, using "-flags +psnr".

-vstats

Dump video coding statistics to *vstats_HHMMSS.log*.

-vstats_file *file*

Dump video coding statistics to *file*.

-vstats_version *file*

Specifies which version of the vstats format to use. Default is 2.

version = 1 :

```
"frame= %5d q= %2.1f PSNR= %6.2f f_size= %6d s_size= %8.0fkB time= %0.3f br=
%7.1fkbits/s avg_br= %7.1fkbits/s"
```

version > 1:

```
"out= %2d st= %2d frame= %5d q= %2.1f PSNR= %6.2f f_size= %6d s_size= %8.0fkB time=
%0.3f br= %7.1fkbits/s avg_br= %7.1fkbits/s"
```

-top[*:stream_specifier*] *n* (*output,per-stream*)

top=1/bottom=0/auto=-1 field first

-vtag *fourcc/tag* (*output*)

Force video tag/fourcc. This is an alias for "-tag:v".

-qphist (*global*)

Show QP histogram

-vbsf *bitstream_filter*

Deprecated see -bsf

-force_key_frames[:*stream_specifier*] *time[,time...]* (*output,per-stream*)

-force_key_frames[:*stream_specifier*] **expr**:*expr* (*output,per-stream*)

-force_key_frames[:*stream_specifier*] **source** (*output,per-stream*)

-force_key_frames[:*stream_specifier*] **source_no_drop** (*output,per-stream*)

force_key_frames can take arguments of the following form:

time[,time...]

If the argument consists of timestamps, ffmpeg will round the specified times to the nearest output timestamp as per the encoder time base and force a keyframe at the first frame having timestamp equal or greater than the computed timestamp. Note that if the encoder time base is too coarse, then the keyframes may be forced on frames with timestamps lower than the specified time. The default encoder time base is the inverse of the output framerate but may be set otherwise via "-enc_time_base".

If one of the times is ""*chapters*"[*delta*]", it is expanded into the time of the beginning of all chapters in the file, shifted by *delta*, expressed as a time in seconds. This option can be useful to ensure that a seek point is present at a chapter mark or any other designated place in the output file.

For example, to insert a key frame at 5 minutes, plus key frames 0.1 second before the beginning of every chapter:

```
-force_key_frames 0:05:00,chapters-0.1
```

expr:*expr*

If the argument is prefixed with "expr:", the string *expr* is interpreted like an expression and is evaluated for each frame. A key frame is forced in case the evaluation is non-zero.

The expression in *expr* can contain the following constants:

n the number of current processed frame, starting from 0

n_forced

the number of forced frames

prev_forced_n

the number of the previous forced frame, it is "NAN" when no keyframe was forced yet

prev_forced_t

the time of the previous forced frame, it is "NAN" when no keyframe was forced yet

t the time of the current processed frame

For example to force a key frame every 5 seconds, you can specify:

```
-force_key_frames expr:gte(t,n_forced*5)
```

To force a key frame 5 seconds after the time of the last forced one, starting from second 13:

```
-force_key_frames expr:if(isnan(prev_forced_t),gte(t,13),gte(t,prev_forced_t+5))
```

source

If the argument is "source", ffmpeg will force a key frame if the current frame being encoded is marked as a key frame in its source.

source_no_drop

If the argument is "source_no_drop", ffmpeg will force a key frame if the current frame being encoded is marked as a key frame in its source. In cases where this particular source frame has to be dropped, enforce the next available frame to become a key frame instead.

Note that forcing too many keyframes is very harmful for the lookahead algorithms of certain encoders: using fixed-GOP options or similar would be more efficient.

-copyinkf[:stream_specifier] (*output,per-stream*)

When doing stream copy, copy also non-key frames found at the beginning.

-init_hw_device *type*[=*name*][:*device*[,*key=value*...]]

Initialise a new hardware device of type *type* called *name*, using the given device parameters. If no name is specified it will receive a default name of the form "*type*%d".

The meaning of *device* and the following arguments depends on the device type:

cuda

device is the number of the CUDA device.

The following options are recognized:

primary_ctx

If set to 1, uses the primary device context instead of creating a new one.

Examples:

-init_hw_device cuda:1

Choose the second device on the system.

-init_hw_device cuda:0,primary_ctx=1

Choose the first device and use the primary device context.

dxva2

device is the number of the Direct3D 9 display adapter.

d3d11va

device is the number of the Direct3D 11 display adapter.

vaapi

device is either an X11 display name or a DRM render node. If not specified, it will attempt to open the default X11 display (*\$DISPLAY*) and then the first DRM render node (*/dev/dri/renderD128*).

vdpa

device is an X11 display name. If not specified, it will attempt to open the default X11 display (*\$DISPLAY*).

qsv *device* selects a value in **MFx_IMPL_***. Allowed values are:

auto

sw

hw

auto_any

hw_any

hw2

hw3

hw4

If not specified, **auto_any** is used. (Note that it may be easier to achieve the desired result for QSV by creating the platform-appropriate subdevice (**dxva2** or **d3d11va** or **vaapi**) and then deriving a QSV device from that.)

Alternatively, **child_device_type** helps to choose platform-appropriate subdevice type. On Windows **d3d11va** is used as default subdevice type.

Examples:

```
-init_hw_device qsv:hw,child_device_type=d3d11va
```

Choose the GPU subdevice with type **d3d11va** and create QSV device with **MXF_IMPL_HARDWARE**.

```
-init_hw_device qsv:hw,child_device_type=dxva2
```

Choose the GPU subdevice with type **dxva2** and create QSV device with **MXF_IMPL_HARDWARE**.

opencl

device selects the platform and device as *platform_index.device_index*.

The set of devices can also be filtered using the key-value pairs to find only devices matching particular platform or device strings.

The strings usable as filters are:

- platform_profile**
- platform_version**
- platform_name**
- platform_vendor**
- platform_extensions**
- device_name**
- device_vendor**
- driver_version**
- device_version**
- device_profile**
- device_extensions**
- device_type**

The indices and filters must together uniquely select a device.

Examples:

-init_hw_device opencl:0.1

Choose the second device on the first platform.

-init_hw_device opencl:;device_name=Foo9000

Choose the device with a name containing the string *Foo9000*.

-init_hw_device opencl:1;device_type=gpu;device_extensions=cl_khr_fp16

Choose the GPU device on the second platform supporting the *cl_khr_fp16* extension.

vulkan

If *device* is an integer, it selects the device by its index in a system-dependent list of devices. If *device* is any other string, it selects the first device with a name containing that string as a substring.

The following options are recognized:

debug

If set to 1, enables the validation layer, if installed.

linear_images

If set to 1, images allocated by the hwcontext will be linear and locally mappable.

instance_extensions

A plus separated list of additional instance extensions to enable.

device_extensions

A plus separated list of additional device extensions to enable.

Examples:

-init_hw_device vulkan:1

Choose the second device on the system.

-init_hw_device vulkan:RADV

Choose the first device with a name containing the string *RADV*.

-init_hw_device

vulkan:0,instance_extensions=VK_KHR_wayland_surface+VK_KHR_xcb_surface

Choose the first device and enable the Wayland and XCB instance extensions.

-init_hw_device *type*[=*name*]@*source*

Initialise a new hardware device of type *type* called *name*, deriving it from the existing device with the name *source*.

-init_hw_device list

List all hardware device types supported in this build of ffmpeg.

-filter_hw_device *name*

Pass the hardware device called *name* to all filters in any filter graph. This can be used to set the device to upload to with the "hwupload" filter, or the device to map to with the "hwmap" filter. Other filters may also make use of this parameter when they require a hardware device. Note that this is typically only required when the input is not already in hardware frames - when it is, filters will derive the device they require from the context of the frames they receive as input.

This is a global setting, so all filters will receive the same device.

-hwaccel[:*stream_specifier*] *hwaccel* (*input,per-stream*)

Use hardware acceleration to decode the matching stream(s). The allowed values of *hwaccel* are:

none

Do not use any hardware acceleration (the default).

auto Automatically select the hardware acceleration method.

vdpa

Use VDPAAU (Video Decode and Presentation API for Unix) hardware acceleration.

dxva2

Use DXVA2 (DirectX Video Acceleration) hardware acceleration.

d3d11va

Use D3D11VA (DirectX Video Acceleration) hardware acceleration.

vaapi

Use VAAPI (Video Acceleration API) hardware acceleration.

qsv Use the Intel QuickSync Video acceleration for video transcoding.

Unlike most other values, this option does not enable accelerated decoding (that is used automatically whenever a qsv decoder is selected), but accelerated transcoding, without copying the frames into the system memory.

For it to work, both the decoder and the encoder must support QSV acceleration and no filters must be used.

This option has no effect if the selected hwaccel is not available or not supported by the chosen decoder.

Note that most acceleration methods are intended for playback and will not be faster than software decoding on modern CPUs. Additionally, **ffmpeg** will usually need to copy the decoded frames from the GPU memory into the system memory, resulting in further performance loss. This option is thus mainly useful for testing.

-hwaccel_device[:stream_specifier] hwaccel_device (*input,per-stream*)

Select a device to use for hardware acceleration.

This option only makes sense when the **-hwaccel** option is also specified. It can either refer to an existing device created with **-init_hw_device** by name, or it can create a new device as if **-init_hw_device type:hwaccel_device** were called immediately before.

-hwaccels

List all hardware acceleration components enabled in this build of ffmpeg. Actual runtime availability depends on the hardware and its suitable driver being installed.

-fix_sub_duration_heartbeat[:stream_specifier]

Set a specific output video stream as the heartbeat stream according to which to split and push through currently in-progress subtitle upon receipt of a random access packet.

This lowers the latency of subtitles for which the end packet or the following subtitle has not yet been received. As a drawback, this will most likely lead to duplication of subtitle events in order to cover the full duration, so when dealing with use cases where latency of when the subtitle event is passed on to output is not relevant this option should not be utilized.

Requires **-fix_sub_duration** to be set for the relevant input subtitle stream for this to have any effect, as well as for the input subtitle stream having to be directly mapped to the same output in which the heartbeat stream resides.

Audio Options

-aframes *number (output)*

Set the number of audio frames to output. This is an obsolete alias for "-frames:a", which you should use instead.

-ar*[:stream_specifier] freq (input/output,per-stream)*

Set the audio sampling frequency. For output streams it is set by default to the frequency of the corresponding input stream. For input streams this option only makes sense for audio grabbing devices and raw demuxers and is mapped to the corresponding demuxer options.

-aq *q (output)*

Set the audio quality (codec-specific, VBR). This is an alias for -q:a.

-ac*[:stream_specifier] channels (input/output,per-stream)*

Set the number of audio channels. For output streams it is set by default to the number of input audio channels. For input streams this option only makes sense for audio grabbing devices and raw demuxers and is mapped to the corresponding demuxer options.

-an *(input/output)*

As an input option, blocks all audio streams of a file from being filtered or being automatically selected or mapped for any output. See "-discard" option to disable streams individually.

As an output option, disables audio recording i.e. automatic selection or mapping of any audio stream. For full manual control see the "-map" option.

-acodec *codec (input/output)*

Set the audio codec. This is an alias for "-codec:a".

-sample_fmt*[:stream_specifier] sample_fmt (output,per-stream)*

Set the audio sample format. Use "-sample_fmts" to get a list of supported sample formats.

-af *filtergraph (output)*

Create the filtergraph specified by *filtergraph* and use it to filter the stream.

This is an alias for "-filter:a", see the **-filter option**.

Advanced Audio options

-atag *fourcc/tag (output)*

Force audio tag/fourcc. This is an alias for "-tag:a".

-absf *bitstream_filter*

Deprecated, see -bsf

-guess_layout_max *channels (input,per-stream)*

If some input channel layout is not known, try to guess only if it corresponds to at most the specified number of channels. For example, 2 tells to **ffmpeg** to recognize 1 channel as mono and 2 channels as stereo but not 6 channels as 5.1. The default is to always try to guess. Use 0 to disable all guessing.

Subtitle options**-scodec** *codec (input/output)*

Set the subtitle codec. This is an alias for "-codec:s".

-sn *(input/output)*

As an input option, blocks all subtitle streams of a file from being filtered or being automatically selected or mapped for any output. See "-discard" option to disable streams individually.

As an output option, disables subtitle recording i.e. automatic selection or mapping of any subtitle stream. For full manual control see the "-map" option.

-sbsf *bitstream_filter*

Deprecated, see -bsf

Advanced Subtitle options**-fix_sub_duration**

Fix subtitles durations. For each subtitle, wait for the next packet in the same stream and adjust the duration of the first to avoid overlap. This is necessary with some subtitles codecs, especially DVB subtitles, because the duration in the original packet is only a rough estimate and the end is actually marked by an empty subtitle frame. Failing to use this option when necessary can result in exaggerated durations or muxing failures due to non-monotonic timestamps.

Note that this option will delay the output of all data until the next subtitle packet is decoded: it may increase memory consumption and latency a lot.

-canvas_size *size*

Set the size of the canvas used to render subtitles.

Advanced options**-map** [-]*input_file_id[:stream_specifier][?] | [linklabel] (output)*

Create one or more streams in the output file. This option has two forms for specifying the data

source(s): the first selects one or more streams from some input file (specified with "-i"), the second takes an output from some complex filtergraph (specified with "-filter_complex" or "-filter_complex_script").

In the first form, an output stream is created for every stream from the input file with the index *input_file_id*. If *stream_specifier* is given, only those streams that match the specifier are used (see the **Stream specifiers** section for the *stream_specifier* syntax).

A "-" character before the stream identifier creates a "negative" mapping. It disables matching streams from already created mappings.

A trailing "?" after the stream index will allow the map to be optional: if the map matches no streams the map will be ignored instead of failing. Note the map will still fail if an invalid input file index is used; such as if the map refers to a non-existent input.

An alternative [*linklabel*] form will map outputs from complex filter graphs (see the **-filter_complex** option) to the output file. *linklabel* must correspond to a defined output link label in the graph.

This option may be specified multiple times, each adding more streams to the output file. Any given input stream may also be mapped any number of times as a source for different output streams, e.g. in order to use different encoding options and/or filters. The streams are created in the output in the same order in which the "-map" options are given on the commandline.

Using this option disables the default mappings for this output file.

Examples:

map everything

To map ALL streams from the first input file to output

```
ffmpeg -i INPUT -map 0 output
```

select specific stream

If you have two audio streams in the first input file, these streams are identified by *0:0* and *0:1*. You can use "-map" to select which streams to place in an output file. For example:

```
ffmpeg -i INPUT -map 0:1 out.wav
```

will map the second input stream in *INPUT* to the (single) output stream in *out.wav*.

create multiple streams

To select the stream with index 2 from input file *a.mov* (specified by the identifier *0:2*), and stream with index 6 from input *b.mov* (specified by the identifier *1:6*), and copy them to the output file *out.mov*:

```
ffmpeg -i a.mov -i b.mov -c copy -map 0:2 -map 1:6 out.mov
```

create multiple streams 2

To select all video and the third audio stream from an input file:

```
ffmpeg -i INPUT -map 0:v -map 0:a:2 OUTPUT
```

negative map

To map all the streams except the second audio, use negative mappings

```
ffmpeg -i INPUT -map 0 -map -0:a:1 OUTPUT
```

optional map

To map the video and audio streams from the first input, and using the trailing "?", ignore the audio mapping if no audio streams exist in the first input:

```
ffmpeg -i INPUT -map 0:v -map 0:a? OUTPUT
```

map by language

To pick the English audio stream:

```
ffmpeg -i INPUT -map 0:m:language:eng OUTPUT
```

-ignore_unknown

Ignore input streams with unknown type instead of failing if copying such streams is attempted.

-copy_unknown

Allow input streams with unknown type to be copied instead of failing if copying such streams is attempted.

-map_channel [*input_file_id.stream_specifier.channel_id*|-1][?][:*output_file_id.stream_specifier*]

This option is deprecated and will be removed. It can be replaced by the *pan* filter. In some cases it may be easier to use some combination of the *channelsplit*, *channelmap*, or *amerge* filters.

Map an audio channel from a given input to an output. If *output_file_id.stream_specifier* is not

set, the audio channel will be mapped on all the audio streams.

Using "-1" instead of *input_file_id.stream_specifier.channel_id* will map a muted channel.

A trailing "?" will allow the `map_channel` to be optional: if the `map_channel` matches no channel the `map_channel` will be ignored instead of failing.

For example, assuming *INPUT* is a stereo audio file, you can switch the two audio channels with the following command:

```
ffmpeg -i INPUT -map_channel 0.0.1 -map_channel 0.0.0 OUTPUT
```

If you want to mute the first channel and keep the second:

```
ffmpeg -i INPUT -map_channel -1 -map_channel 0.0.1 OUTPUT
```

The order of the "-map_channel" option specifies the order of the channels in the output stream. The output channel layout is guessed from the number of channels mapped (mono if one "-map_channel", stereo if two, etc.). Using "-ac" in combination of "-map_channel" makes the channel gain levels to be updated if input and output channel layouts don't match (for instance two "-map_channel" options and "-ac 6").

You can also extract each channel of an input to specific outputs; the following command extracts two channels of the *INPUT* audio stream (file 0, stream 0) to the respective *OUTPUT_CH0* and *OUTPUT_CH1* outputs:

```
ffmpeg -i INPUT -map_channel 0.0.0 OUTPUT_CH0 -map_channel 0.0.1 OUTPUT_CH1
```

The following example splits the channels of a stereo input into two separate streams, which are put into the same output file:

```
ffmpeg -i stereo.wav -map 0:0 -map 0:0 -map_channel 0.0.0:0.0 -map_channel 0.0.1:0.1 -y out.ogg
```

Note that currently each output stream can only contain channels from a single input stream; you can't for example use "-map_channel" to pick multiple input audio channels contained in different streams (from the same or different files) and merge them into a single output stream. It is therefore not currently possible, for example, to turn two separate mono streams into a single stereo stream. However splitting a stereo stream into two single channel mono streams is possible.

If you need this feature, a possible workaround is to use the *amerge* filter. For example, if you

need to merge a media (here *input.mkv*) with 2 mono audio streams into one single stereo channel audio stream (and keep the video stream), you can use the following command:

```
ffmpeg -i input.mkv -filter_complex "[0:1] [0:2] amerge" -c:a pcm_s16le -c:v copy output.mkv
```

To map the first two audio channels from the first input, and using the trailing "?", ignore the audio channel mapping if the first input is mono instead of stereo:

```
ffmpeg -i INPUT -map_channel 0.0.0 -map_channel 0.0.1? OUTPUT
```

-map_metadata[:*metadata_spec_out*] *infile*[:*metadata_spec_in*] (*output,per-metadata*)

Set metadata information of the next output file from *infile*. Note that those are file indices (zero-based), not filenames. Optional *metadata_spec_in/out* parameters specify, which metadata to copy. A metadata specifier can have the following forms:

g global metadata, i.e. metadata that applies to the whole file

*s[:*stream_spec*]*

per-stream metadata. *stream_spec* is a stream specifier as described in the **Stream specifiers** chapter. In an input metadata specifier, the first matching stream is copied from. In an output metadata specifier, all matching streams are copied to.

c:chapter_index

per-chapter metadata. *chapter_index* is the zero-based chapter index.

p:program_index

per-program metadata. *program_index* is the zero-based program index.

If metadata specifier is omitted, it defaults to global.

By default, global metadata is copied from the first input file, per-stream and per-chapter metadata is copied along with streams/chapters. These default mappings are disabled by creating any mapping of the relevant type. A negative file index can be used to create a dummy mapping that just disables automatic copying.

For example to copy metadata from the first stream of the input file to global metadata of the output file:

```
ffmpeg -i in.ogg -map_metadata 0:s:0 out.mp3
```

To do the reverse, i.e. copy global metadata to all audio streams:

```
ffmpeg -i in.mkv -map_metadata:s:a 0:g out.mkv
```

Note that simple 0 would work as well in this example, since global metadata is assumed by default.

-map_chapters *input_file_index (output)*

Copy chapters from input file with index *input_file_index* to the next output file. If no chapter mapping is specified, then chapters are copied from the first input file with at least one chapter. Use a negative file index to disable any chapter copying.

-benchmark (*global*)

Show benchmarking information at the end of an encode. Shows real, system and user time used and maximum memory consumption. Maximum memory consumption is not supported on all systems, it will usually display as 0 if not supported.

-benchmark_all (*global*)

Show benchmarking information during the encode. Shows real, system and user time used in various steps (audio/video encode/decode).

-timelimit *duration (global)*

Exit after ffmpeg has been running for *duration* seconds in CPU user time.

-dump (*global*)

Dump each input packet to stderr.

-hex (*global*)

When dumping packets, also dump the payload.

-readrate *speed (input)*

Limit input read speed.

Its value is a floating-point positive number which represents the maximum duration of media, in seconds, that should be ingested in one second of wallclock time. Default value is zero and represents no imposed limitation on speed of ingestion. Value 1 represents real-time speed and is equivalent to "-re".

Mainly used to simulate a capture device or live input stream (e.g. when reading from a file). Should not be used with a low value when input is an actual capture device or live stream as it

may cause packet loss.

It is useful for when flow speed of output packets is important, such as live streaming.

-re (*input*)

Read input at native frame rate. This is equivalent to setting "-readrate 1".

-vsync *parameter (global)*

-fps_mode[:*stream_specifier*] *parameter (output,per-stream)*

Set video sync method / framerate mode. vsync is applied to all output video streams but can be overridden for a stream by setting fps_mode. vsync is deprecated and will be removed in the future.

For compatibility reasons some of the values for vsync can be specified as numbers (shown in parentheses in the following table).

passthrough (0)

Each frame is passed with its timestamp from the demuxer to the muxer.

cfr (1)

Frames will be duplicated and dropped to achieve exactly the requested constant frame rate.

vfr (2)

Frames are passed through with their timestamp or dropped so as to prevent 2 frames from having the same timestamp.

drop

As passthrough but destroys all timestamps, making the muxer generate fresh timestamps based on frame-rate.

auto (-1)

Chooses between cfr and vfr depending on muxer capabilities. This is the default method.

Note that the timestamps may be further modified by the muxer, after this. For example, in the case that the format option **avoid_negative_ts** is enabled.

With -map you can select from which stream the timestamps should be taken. You can leave either video or audio unchanged and sync the remaining stream(s) to the unchanged one.

-frame_drop_threshold *parameter*

Frame drop threshold, which specifies how much behind video frames can be before they are dropped. In frame rate units, so 1.0 is one frame. The default is -1.1. One possible usecase is to avoid framedrops in case of noisy timestamps or to increase frame drop precision in case of exact timestamps.

-adrift_threshold *time*

Set the minimum difference between timestamps and audio data (in seconds) to trigger adding/dropping samples to make it match the timestamps. This option effectively is a threshold to select between hard (add/drop) and soft (squeeze/stretch) compensation. "-async" must be set to a positive value.

-apad *parameters (output,per-stream)*

Pad the output audio stream(s). This is the same as applying "-af apad". Argument is a string of filter parameters composed the same as with the "apad" filter. "-shortest" must be set for this output for the option to take effect.

-copyts

Do not process input timestamps, but keep their values without trying to sanitize them. In particular, do not remove the initial start time offset value.

Note that, depending on the **vsync** option or on specific muxer processing (e.g. in case the format option **avoid_negative_ts** is enabled) the output timestamps may mismatch with the input timestamps even when this option is selected.

-start_at_zero

When used with **copyts**, shift input timestamps so they start at zero.

This means that using e.g. "-ss 50" will make output timestamps start at 50 seconds, regardless of what timestamp the input file started at.

-copytb *mode*

Specify how to set the encoder timebase when stream copying. *mode* is an integer numeric value, and can assume one of the following values:

- 1 Use the demuxer timebase.

The time base is copied to the output encoder from the corresponding input demuxer. This is sometimes required to avoid non monotonically increasing timestamps when copying video streams with variable frame rate.

- 0** Use the decoder timebase.

The time base is copied to the output encoder from the corresponding input decoder.

- 1** Try to make the choice automatically, in order to generate a sane output.

Default value is -1.

-enc_time_base[:*stream_specifier*] *timebase* (*output,per-stream*)

Set the encoder timebase. *timebase* is a floating point number, and can assume one of the following values:

- 0** Assign a default value according to the media type.

For video - use 1/framerate, for audio - use 1/samplerate.

- 1** Use the input stream timebase when possible.

If an input stream is not available, the default timebase will be used.

- >0** Use the provided number as the timebase.

This field can be provided as a ratio of two integers (e.g. 1:24, 1:48000) or as a floating point number (e.g. 0.04166, 2.0833e-5)

Default value is 0.

-bitexact (*input/output*)

Enable bitexact mode for (de)muxer and (de/en)coder

-shortest (*output*)

Finish encoding when the shortest output stream ends.

Note that this option may require buffering frames, which introduces extra latency. The maximum amount of this latency may be controlled with the "-shortest_buf_duration" option.

-shortest_buf_duration *duration* (*output*)

The "-shortest" option may require buffering potentially large amounts of data when at least one of the streams is "sparse" (i.e. has large gaps between frames X this is typically the case for subtitles).

This option controls the maximum duration of buffered frames in seconds. Larger values may allow the "-shortest" option to produce more accurate results, but increase memory use and latency.

The default value is 10 seconds.

-dts_delta_threshold

Timestamp discontinuity delta threshold.

-dts_error_threshold *seconds*

Timestamp error delta threshold. This threshold use to discard crazy/damaged timestamps and the default is 30 hours which is arbitrarily picked and quite conservative.

-muxdelay *seconds (output)*

Set the maximum demux-decode delay.

-muxpreload *seconds (output)*

Set the initial demux-decode delay.

-streamid *output-stream-index:new-value (output)*

Assign a new stream-id value to an output stream. This option should be specified prior to the output filename to which it applies. For the situation where multiple output files exist, a streamid may be reassigned to a different value.

For example, to set the stream 0 PID to 33 and the stream 1 PID to 36 for an output mpegts file:

```
ffmpeg -i inurl -streamid 0:33 -streamid 1:36 out.ts
```

-bsf[:stream_specifier] *bitstream_filters (output,per-stream)*

Set bitstream filters for matching streams. *bitstream_filters* is a comma-separated list of bitstream filters. Use the "-bsfs" option to get the list of bitstream filters.

```
ffmpeg -i h264.mp4 -c:v copy -bsf:v h264_mp4toannexb -an out.h264
```

```
ffmpeg -i file.mov -an -vn -bsf:s mov2textsub -c:s copy -f rawvideo sub.txt
```

-tag[:stream_specifier] *codec_tag (input/output,per-stream)*

Force a tag/fourcc for matching streams.

-timecode *hh:mm:ssSEPff*

Specify Timecode for writing. *SEP* is ':' for non drop timecode and ';' (or '.' for drop).

```
ffmpeg -i input.mpg -timecode 01:02:03.04 -r 30000/1001 -s ntsc output.mpg
```

-filter_complex *filtergraph (global)*

Define a complex filtergraph, i.e. one with arbitrary number of inputs and/or outputs. For simple graphs -- those with one input and one output of the same type -- see the **-filter** options. *filtergraph* is a description of the filtergraph, as described in the "Filtergraph syntax" section of the ffmpeg-filters manual.

Input link labels must refer to input streams using the "[file_index:stream_specifier]" syntax (i.e. the same as **-map** uses). If *stream_specifier* matches multiple streams, the first one will be used. An unlabeled input will be connected to the first unused input stream of the matching type.

Output link labels are referred to with **-map**. Unlabeled outputs are added to the first output file.

Note that with this option it is possible to use only lavfi sources without normal input files.

For example, to overlay an image over video

```
ffmpeg -i video.mkv -i image.png -filter_complex '[0:v][1:v]overlay[out]' -map
[out] out.mkv
```

Here "[0:v]" refers to the first video stream in the first input file, which is linked to the first (main) input of the overlay filter. Similarly the first video stream in the second input is linked to the second (overlay) input of overlay.

Assuming there is only one video stream in each input file, we can omit input labels, so the above is equivalent to

```
ffmpeg -i video.mkv -i image.png -filter_complex 'overlay[out]' -map
[out] out.mkv
```

Furthermore we can omit the output label and the single output from the filter graph will be added to the output file automatically, so we can simply write

```
ffmpeg -i video.mkv -i image.png -filter_complex 'overlay' out.mkv
```

As a special exception, you can use a bitmap subtitle stream as input: it will be converted into a

video with the same size as the largest video in the file, or 720x576 if no video is present. Note that this is an experimental and temporary solution. It will be removed once libavfilter has proper support for subtitles.

For example, to hardcode subtitles on top of a DVB-T recording stored in MPEG-TS format, delaying the subtitles by 1 second:

```
ffmpeg -i input.ts -filter_complex \
    '[#0x2ef] setpts=PTS+1/TB [sub] ; [#0x2d0] [sub] overlay' \
    -sn -map '#0x2dc' output.mkv
```

(0x2d0, 0x2dc and 0x2ef are the MPEG-TS PIDs of respectively the video, audio and subtitles streams; 0:0, 0:3 and 0:7 would have worked too)

To generate 5 seconds of pure red video using lavfi "color" source:

```
ffmpeg -filter_complex 'color=c=red' -t 5 out.mkv
```

-filter_complex_threads *nb_threads* (*global*)

Defines how many threads are used to process a filter_complex graph. Similar to filter_threads but used for "-filter_complex" graphs only. The default is the number of available CPUs.

-lavfi *filtergraph* (*global*)

Define a complex filtergraph, i.e. one with arbitrary number of inputs and/or outputs. Equivalent to **-filter_complex**.

-filter_complex_script *filename* (*global*)

This option is similar to **-filter_complex**, the only difference is that its argument is the name of the file from which a complex filtergraph description is to be read.

-accurate_seek (*input*)

This option enables or disables accurate seeking in input files with the **-ss** option. It is enabled by default, so seeking is accurate when transcoding. Use **-noaccurate_seek** to disable it, which may be useful e.g. when copying some streams and transcoding the others.

-seek_timestamp (*input*)

This option enables or disables seeking by timestamp in input files with the **-ss** option. It is disabled by default. If enabled, the argument to the **-ss** option is considered an actual timestamp, and is not offset by the start time of the file. This matters only for files which do not start from timestamp 0, such as transport streams.

-thread_queue_size *size (input/output)*

For input, this option sets the maximum number of queued packets when reading from the file or device. With low latency / high rate live streams, packets may be discarded if they are not read in a timely manner; setting this value can force ffmpeg to use a separate input thread and read packets as soon as they arrive. By default ffmpeg only does this if multiple inputs are specified.

For output, this option specified the maximum number of packets that may be queued to each muxing thread.

-sdp_file *file (global)*

Print sdp information for an output stream to *file*. This allows dumping sdp information when at least one output isn't an rtp stream. (Requires at least one of the output formats to be rtp).

-discard *(input)*

Allows discarding specific streams or frames from streams. Any input stream can be fully discarded, using value "all" whereas selective discarding of frames from a stream occurs at the demuxer and is not supported by all demuxers.

none

Discard no frame.

default

Default, which discards no frames.

noref

Discard all non-reference frames.

bidir

Discard all bidirectional frames.

nokey

Discard all frames excepts keyframes.

all Discard all frames.

-abort_on *flags (global)*

Stop and abort on various conditions. The following flags are available:

empty_output

No packets were passed to the muxer, the output is empty.

empty_output_stream

No packets were passed to the muxer in some of the output streams.

-max_error_rate (*global*)

Set fraction of decoding frame failures across all inputs which when crossed ffmpeg will return exit code 69. Crossing this threshold does not terminate processing. Range is a floating-point number between 0 to 1. Default is 2/3.

-xerror (*global*)

Stop and exit on error

-max_muxing_queue_size *packets* (*output,per-stream*)

When transcoding audio and/or video streams, ffmpeg will not begin writing into the output until it has one packet for each such stream. While waiting for that to happen, packets for other streams are buffered. This option sets the size of this buffer, in packets, for the matching output stream.

The default value of this option should be high enough for most uses, so only touch this option if you are sure that you need it.

-muxing_queue_data_threshold *bytes* (*output,per-stream*)

This is a minimum threshold until which the muxing queue size is not taken into account. Defaults to 50 megabytes per stream, and is based on the overall size of packets passed to the muxer.

-auto_conversion_filters (*global*)

Enable automatically inserting format conversion filters in all filter graphs, including those defined by **-vf**, **-af**, **-filter_complex** and **-lavfi**. If filter format negotiation requires a conversion, the initialization of the filters will fail. Conversions can still be performed by inserting the relevant conversion filter (scale, aresample) in the graph. On by default, to explicitly disable it you need to specify "-noauto_conversion_filters".

-bits_per_raw_sample[:*stream_specifier*] *value* (*output,per-stream*)

Declare the number of bits per raw sample in the given output stream to be *value*. Note that this option sets the information provided to the encoder/muxer, it does not change the stream to conform to this value. Setting values that do not match the stream properties may result in encoding failures or invalid output files.

-stats_enc_pre[:*stream_specifier*] *path* (*output,per-stream*)**-stats_enc_post**[:*stream_specifier*] *path* (*output,per-stream*)**-stats_mux_pre**[:*stream_specifier*] *path* (*output,per-stream*)

Write per-frame encoding information about the matching streams into the file given by *path*.

-stats_enc_pre writes information about raw video or audio frames right before they are sent for encoding, while **-stats_enc_post** writes information about encoded packets as they are received from the encoder. **-stats_mux_pre** writes information about packets just as they are about to be sent to the muxer. Every frame or packet produces one line in the specified file. The format of this line is controlled by **-stats_enc_pre_fmt** / **-stats_enc_post_fmt** / **-stats_mux_pre_fmt**.

When stats for multiple streams are written into a single file, the lines corresponding to different streams will be interleaved. The precise order of this interleaving is not specified and not guaranteed to remain stable between different invocations of the program, even with the same options.

-stats_enc_pre_fmt[:stream_specifier] *format_spec* (*output,per-stream*)

-stats_enc_post_fmt[:stream_specifier] *format_spec* (*output,per-stream*)

-stats_mux_pre_fmt[:stream_specifier] *format_spec* (*output,per-stream*)

Specify the format for the lines written with **-stats_enc_pre** / **-stats_enc_post** / **-stats_mux_pre**.

format_spec is a string that may contain directives of the form *{fmt}*. *format_spec* is backslash-escaped --- use `\{`, `\}`, and `\\` to write a literal `{`, `}`, or `\`, respectively, into the output.

The directives given with *fmt* may be one of the following:

fidx Index of the output file.

sidx Index of the output stream in the file.

n Frame number. Pre-encoding: number of frames sent to the encoder so far. Post-encoding: number of packets received from the encoder so far. Muxing: number of packets submitted to the muxer for this stream so far.

ni Input frame number. Index of the input frame (i.e. output by a decoder) that corresponds to this output frame or packet. -1 if unavailable.

tb Encoder timebase, as a rational number *num/den*. Note that this may be different from the timebase used by the muxer.

tbi Timebase for *ptsi*, as a rational number *num/den*. Available when *ptsi* is available, *0/1* otherwise.

pts Presentation timestamp of the frame or packet, as an integer. Should be multiplied by the timebase to compute presentation time.

- pts** Presentation timestamp of the input frame (see *ni*), as an integer. Should be multiplied by *tbi* to compute presentation time. Printed as $(2^{63} - 1 = 9223372036854775807)$ when not available.
- t** Presentation time of the frame or packet, as a decimal number. Equal to *pts* multiplied by *tb*.
- ti** Presentation time of the input frame (see *ni*), as a decimal number. Equal to *pts* multiplied by *tbi*. Printed as *inf* when not available.
- dts** Decoding timestamp of the packet, as an integer. Should be multiplied by the timebase to compute presentation time. Post-encoding only.
- dt** Decoding time of the frame or packet, as a decimal number. Equal to *dts* multiplied by *tb*.
- sn** Number of audio samples sent to the encoder so far. Audio and pre-encoding only.
- samp**
Number of audio samples in the frame. Audio and pre-encoding only.
- size** Size of the encoded packet in bytes. Post-encoding only.
- br** Current bitrate in bits per second. Post-encoding only.
- abr** Average bitrate for the whole stream so far, in bits per second, -1 if it cannot be determined at this point. Post-encoding only.

The default format strings are:

pre-encoding

{fidx} {sidx} {n} {t}

post-encoding

{fidx} {sidx} {n} {t}

In the future, new items may be added to the end of the default formatting strings. Users who depend on the format staying exactly the same, should prescribe it manually.

Note that stats for different streams written into the same file may have different formats.

Preset files

A preset file contains a sequence of *option=value* pairs, one for each line, specifying a sequence of options which would be awkward to specify on the command line. Lines starting with the hash ('#') character are ignored and are used to provide comments. Check the *presets* directory in the FFmpeg source tree for examples.

There are two types of preset files: *ffpreset* and *avpreset* files.

ffpreset files

ffpreset files are specified with the "vpre", "apre", "spre", and "fpre" options. The "fpre" option takes the filename of the preset instead of a preset name as input and can be used for any kind of codec. For the "vpre", "apre", and "spre" options, the options specified in a preset file are applied to the currently selected codec of the same type as the preset option.

The argument passed to the "vpre", "apre", and "spre" preset options identifies the preset file to use according to the following rules:

First *ffmpeg* searches for a file named *arg.ffpreset* in the directories *\$FFMPEG_DATADIR* (if set), and *\$HOME/.ffmpeg*, and in the *datadir* defined at configuration time (usually *PREFIX/share/ffmpeg*) or in a *ffpresets* folder along the executable on win32, in that order. For example, if the argument is "libvpx-1080p", it will search for the file *libvpx-1080p.ffpreset*.

If no such file is found, then *ffmpeg* will search for a file named *codec_name-arg.ffpreset* in the above-mentioned directories, where *codec_name* is the name of the codec to which the preset file options will be applied. For example, if you select the video codec with "-vcodec libvpx" and use "-vpre 1080p", then it will search for the file *libvpx-1080p.ffpreset*.

avpreset files

avpreset files are specified with the "pre" option. They work similar to *ffpreset* files, but they only allow encoder-specific options. Therefore, an *option=value* pair specifying an encoder cannot be used.

When the "pre" option is specified, *ffmpeg* will look for files with the suffix *.avpreset* in the directories *\$AVCONV_DATADIR* (if set), and *\$HOME/.avconv*, and in the *datadir* defined at configuration time (usually *PREFIX/share/ffmpeg*), in that order.

First *ffmpeg* searches for a file named *codec_name-arg.avpreset* in the above-mentioned directories, where *codec_name* is the name of the codec to which the preset file options will be applied. For example, if you select the video codec with "-vcodec libvpx" and use "-pre 1080p", then it will search for the file *libvpx-1080p.avpreset*.

If no such file is found, then ffmpeg will search for a file named *arg.avpreset* in the same directories.

EXAMPLES

Video and Audio grabbing

If you specify the input format and device then ffmpeg can grab video and audio directly.

```
ffmpeg -f oss -i /dev/dsp -f video4linux2 -i /dev/video0 /tmp/out.mpg
```

Or with an ALSA audio source (mono input, card id 1) instead of OSS:

```
ffmpeg -f alsa -ac 1 -i hw:1 -f video4linux2 -i /dev/video0 /tmp/out.mpg
```

Note that you must activate the right video source and channel before launching ffmpeg with any TV viewer such as <http://linux.bytesex.org/xawtv/> by Gerd Knorr. You also have to set the audio recording levels correctly with a standard mixer.

X11 grabbing

Grab the X11 display with ffmpeg via

```
ffmpeg -f x11grab -video_size cif -framerate 25 -i :0.0 /tmp/out.mpg
```

0.0 is display.screen number of your X11 server, same as the DISPLAY environment variable.

```
ffmpeg -f x11grab -video_size cif -framerate 25 -i :0.0+10,20 /tmp/out.mpg
```

0.0 is display.screen number of your X11 server, same as the DISPLAY environment variable. 10 is the x-offset and 20 the y-offset for the grabbing.

Video and Audio file format conversion

Any supported file format and protocol can serve as input to ffmpeg:

Examples:

- ⊕ You can use YUV files as input:

```
ffmpeg -i /tmp/test%d.Y /tmp/out.mpg
```

It will use the files:

```
/tmp/test0.Y, /tmp/test0.U, /tmp/test0.V,
```

/tmp/test1.Y, /tmp/test1.U, /tmp/test1.V, etc...

The Y files use twice the resolution of the U and V files. They are raw files, without header. They can be generated by all decent video decoders. You must specify the size of the image with the `-s` option if ffmpeg cannot guess it.

- ⊕ You can input from a raw YUV420P file:

```
ffmpeg -i /tmp/test.yuv /tmp/out.avi
```

test.yuv is a file containing raw YUV planar data. Each frame is composed of the Y plane followed by the U and V planes at half vertical and horizontal resolution.

- ⊕ You can output to a raw YUV420P file:

```
ffmpeg -i mydivx.avi hugefile.yuv
```

- ⊕ You can set several input files and output files:

```
ffmpeg -i /tmp/a.wav -s 640x480 -i /tmp/a.yuv /tmp/a.mpg
```

Converts the audio file a.wav and the raw YUV video file a.yuv to MPEG file a.mpg.

- ⊕ You can also do audio and video conversions at the same time:

```
ffmpeg -i /tmp/a.wav -ar 22050 /tmp/a.mp2
```

Converts a.wav to MPEG audio at 22050 Hz sample rate.

- ⊕ You can encode to several formats at the same time and define a mapping from input stream to output streams:

```
ffmpeg -i /tmp/a.wav -map 0:a -b:a 64k /tmp/a.mp2 -map 0:a -b:a 128k /tmp/b.mp2
```

Converts a.wav to a.mp2 at 64 kbits and to b.mp2 at 128 kbits. '-map file:index' specifies which input stream is used for each output stream, in the order of the definition of output streams.

- ⊕ You can transcode decrypted VOBs:

```
ffmpeg -i snatch_1.vob -f avi -c:v mpeg4 -b:v 800k -g 300 -bf 2 -c:a libmp3lame -b:a 128k snatch.avi
```

This is a typical DVD ripping example; the input is a VOB file, the output an AVI file with MPEG-4 video and MP3 audio. Note that in this command we use B-frames so the MPEG-4 stream is DivX5 compatible, and GOP size is 300 which means one intra frame every 10 seconds for 29.97fps input video. Furthermore, the audio stream is MP3-encoded so you need to enable LAME support by passing "--enable-libmp3lame" to configure. The mapping is particularly useful for DVD transcoding to get the desired audio language.

NOTE: To see the supported input formats, use "ffmpeg -demuxers".

- ⊕ You can extract images from a video, or create a video from many images:

For extracting images from a video:

```
ffmpeg -i foo.avi -r 1 -s WxH -f image2 foo-%03d.jpeg
```

This will extract one video frame per second from the video and will output them in files named *foo-001.jpeg*, *foo-002.jpeg*, etc. Images will be rescaled to fit the new WxH values.

If you want to extract just a limited number of frames, you can use the above command in combination with the "-frames:v" or "-t" option, or in combination with -ss to start extracting from a certain point in time.

For creating a video from many images:

```
ffmpeg -f image2 -framerate 12 -i foo-%03d.jpeg -s WxH foo.avi
```

The syntax "foo-%03d.jpeg" specifies to use a decimal number composed of three digits padded with zeroes to express the sequence number. It is the same syntax supported by the C printf function, but only formats accepting a normal integer are suitable.

When importing an image sequence, -i also supports expanding shell-like wildcard patterns (globbing) internally, by selecting the image2-specific "-pattern_type glob" option.

For example, for creating a video from filenames matching the glob pattern "foo-*.jpeg":

```
ffmpeg -f image2 -pattern_type glob -framerate 12 -i 'foo-*.jpeg' -s WxH foo.avi
```

- ⊕ You can put many streams of the same type in the output:

```
ffmpeg -i test1.avi -i test2.avi -map 1:1 -map 1:0 -map 0:1 -map 0:0 -c copy -y test12.nut
```

The resulting output file *test12.nut* will contain the first four streams from the input files in reverse order.

- ⊕ To force CBR video output:

```
ffmpeg -i myfile.avi -b 4000k -minrate 4000k -maxrate 4000k -bufsize 1835k out.m2v
```

- ⊕ The four options *lmin*, *lmax*, *mblmin* and *mblmax* use 'lambda' units, but you may use the *QP2LAMBDA* constant to easily convert from 'q' units:

```
ffmpeg -i src.ext -lmax 21*QP2LAMBDA dst.ext
```

SEE ALSO

ffmpeg-all(1), **ffplay(1)**, **ffprobe(1)**, **ffmpeg-utils(1)**, **ffmpeg-scaler(1)**, **ffmpeg-resampler(1)**, **ffmpeg-codecs(1)**, **ffmpeg-bitstream-filters(1)**, **ffmpeg-formats(1)**, **ffmpeg-devices(1)**, **ffmpeg-protocols(1)**, **ffmpeg-filters(1)**

AUTHORS

The FFmpeg developers.

For details about the authorship, see the Git history of the project (<https://git.ffmpeg.org/ffmpeg>), e.g. by typing the command **git log** in the FFmpeg source directory, or browsing the online repository at <https://git.ffmpeg.org/ffmpeg>.

Maintainers for the specific components are listed in the file *MAINTAINERS* in the source code tree.