NAME

ffmpeg−resampler − FFmpeg Resampler

DESCRIPTION

The FFmpeg resampler provides a high-level interface to the libswresample library audio resampling utilities. In particular it allows one to perform audio resampling, audio channel layout rematrixing, and convert audio format and packing layout.

RESAMPLER OPTIONS

The audio resampler supports the following named options.

Options may be set by specifying −*option value* in the FFmpeg tools, *option*=*value* for the aresample filter, by setting the value explicitly in the `SwrContext` options or using the *libavutil/opt.h* API for programmatic use.

ich, in_channel_count
Set the number of input channels. Default value is 0. Setting this value is not mandatory if the corresponding channel layout **in_channel_layout** is set.

och, out_channel_count
Set the number of output channels. Default value is 0. Setting this value is not mandatory if the corresponding channel layout **out_channel_layout** is set.

uch, used_channel_count
Set the number of used input channels. Default value is 0. This option is only used for special remapping.

isr, in_sample_rate
Set the input sample rate. Default value is 0.

osr, out_sample_rate
Set the output sample rate. Default value is 0.

isf, in_sample_fmt
Specify the input sample format. It is set by default to `none`.

osf, out_sample_fmt
Specify the output sample format. It is set by default to `none`.

tsf, internal_sample_fmt
Set the internal sample format. Default value is `none`. This will automatically be chosen when it is not explicitly set.

icl, in_channel_layout
ocl, out_channel_layout
Set the input/output channel layout.

See **the Channel Layout section in the ffmpeg−utils (1) manual** for the required syntax.

clev, center_mix_level
Set the center mix level. It is a value expressed in deciBel, and must be in the interval [−32,32].

slev, surround_mix_level
Set the surround mix level. It is a value expressed in deciBel, and must be in the interval [−32,32].

lfe_mix_level
Set LFE mix into non LFE level. It is used when there is a LFE input but no LFE output. It is a value expressed in deciBel, and must be in the interval [−32,32].

rmvol, rematrix_volume
Set rematrix volume. Default value is 1.0.

rematrix_maxval
Set maximum output value for rematrixing. This can be used to prevent clipping vs. preventing volume reduction. A value of 1.0 prevents clipping.

flags, swr_flags
Set flags used by the converter. Default value is 0.

It supports the following individual flags:

res force resampling, this flag forces resampling to be used even when the input and output sample rates match.

dither_scale
Set the dither scale. Default value is 1.

dither_method
Set dither method. Default value is 0.

Supported values:

rectangular
select rectangular dither

triangular
select triangular dither

triangular_hp
select triangular dither with high pass

lipshitz
select Lipshitz noise shaping dither.

shibata
select Shibata noise shaping dither.

low_shibata
select low Shibata noise shaping dither.

high_shibata
select high Shibata noise shaping dither.

f_weighted
select f−weighted noise shaping dither

modified_e_weighted
select modified-e-weighted noise shaping dither

improved_e_weighted
select improved-e-weighted noise shaping dither

resampler
Set resampling engine. Default value is swr.

Supported values:

swr
select the native SW Resampler; filter options precision and cheby are not applicable in this case.

soxr
select the SoX Resampler (where available); compensation, and filter options filter_size, phase_shift, exact_rational, filter_type & kaiser_beta, are not applicable in this case.

filter_size
For swr only, set resampling filter size, default value is 32.

phase_shift
For swr only, set resampling phase shift, default value is 10, and must be in the interval [0,30].

linear_interp
Use linear interpolation when enabled (the default). Disable it if you want to preserve speed instead of quality when exact_rational fails.

exact_rational

For swr only, when enabled, try to use exact phase_count based on input and output sample rate. However, if it is larger than `1 << phase_shift`, the phase_count will be `1 << phase_shift` as fallback. Default is enabled.

cutoff

Set cutoff frequency (swr: 6dB point; soxr: 0dB point) ratio; must be a float value between 0 and 1. Default value is 0.97 with swr, and 0.91 with soxr (which, with a sample-rate of 44100, preserves the entire audio band to 20kHz).

precision

For soxr only, the precision in bits to which the resampled signal will be calculated. The default value of 20 (which, with suitable dithering, is appropriate for a destination bit-depth of 16) gives SoX's 'High Quality'; a value of 28 gives SoX's 'Very High Quality'.

cheby

For soxr only, selects passband rolloff none (Chebyshev) & higher-precision approximation for 'irrational' ratios. Default value is 0.

async

For swr only, simple 1 parameter audio sync to timestamps using stretching, squeezing, filling and trimming. Setting this to 1 will enable filling and trimming, larger values represent the maximum amount in samples that the data may be stretched or squeezed for each second. Default value is 0, thus no compensation is applied to make the samples match the audio timestamps.

first_pts

For swr only, assume the first pts should be this value. The time unit is 1 / sample rate. This allows for padding/trimming at the start of stream. By default, no assumption is made about the first frame's expected pts, so no padding or trimming is done. For example, this could be set to 0 to pad the beginning with silence if an audio stream starts after the video stream or to trim any samples with a negative pts due to encoder delay.

min_comp

For swr only, set the minimum difference between timestamps and audio data (in seconds) to trigger stretching/squeezing/filling or trimming of the data to make it match the timestamps. The default is that stretching/squeezing/filling and trimming is disabled (**min_comp** = `FLT_MAX`).

min_hard_comp

For swr only, 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 (trim/fill) and soft (squeeze/stretch) compensation. Note that all compensation is by default disabled through **min_comp**. The default is 0.1.

comp_duration

For swr only, set duration (in seconds) over which data is stretched/squeezed to make it match the timestamps. Must be a non-negative double float value, default value is 1.0.

max_soft_comp

For swr only, set maximum factor by which data is stretched/squeezed to make it match the timestamps. Must be a non-negative double float value, default value is 0.

matrix_encoding

Select matrixed stereo encoding.

It accepts the following values:

none

select none

dolby

select Dolby

dplii
select Dolby Pro Logic II

Default value is none.

filter_type
For swr only, select resampling filter type. This only affects resampling operations.

It accepts the following values:

cubic
select cubic

blackman_nuttall
select Blackman Nuttall windowed sinc

kaiser
select Kaiser windowed sinc

kaiser_beta
For swr only, set Kaiser window beta value. Must be a double float value in the interval [2,16], default value is 9.

output_sample_bits
For swr only, set number of used output sample bits for dithering. Must be an integer in the interval [0,64], default value is 0, which means it's not used.

SEE ALSO

ffmpeg (1), **ffplay** (1), **ffprobe** (1), **libswresample** (3)

AUTHORS

The FFmpeg developers.

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

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