Name

rdconvert — Convert an audio file to a different format

Synopsis

`rdconvert` [*OPTIONS*] {*src-file*}

Description

rdconvert(1) can be used to convert audio files between different formats.

Options

`--destination-bit-rate=`*bit-rate*
: Use a bit rate of *bit-rate* bits per second. This option is ignored for PCM and FLAC formats, and is mutually exclusive with the `--destination-quality` option. The default value is **0**.

`--destination-channels=`*chans*
: Use *chans* channels. Supported values are **1** and **2**. The default value is **2**.

`--destination-file=`*filename*
: Write the converted data to *filename*. If not specified, the data will be written to the name of the input file with the default extension of the destination format appended.

`--destination-format=`*format*
: Write the converted data to the specified format. *format* can be one of the following:

  0 PCM16 WAV

  2 MPEG Layer 2 (Raw)

  3 MPEG Layer 3 (Raw)

  4 Free Lossless Audio Codec (FLAC)

  5 OggVorbis

  6 MPEG Layer 2 (BWF WAV Container)

  7 PCM24 WAV

`--destination-quality=`*qual*
: Use a variable bitrate with a quality of *chans*. Supported values are **-1** through **10**. This parameter is used only with a format of **5** (OggVorbis). The default value is **0**.

`--destination-sample-rate=`*rate*
: Use a sample rate of *rate* samples per second. Not all sample rates are supported for all formats; see the relevant MPEG specifications for details. The default value is **48000**.

`--end-point=`*msec*
: Stop converting the audio data at the point *msec* mS from the start of the source file. A value of **-1** means to continue conversion to the end of the source file, which is the default.

`--normalization-level=`*lvl*
: Peak-normalize the audio to *lvl* dBFS. A value of **0** disables normalization, which is the default.

`--speed-ratio=`*`ratio`*
: Alter the tempo of the audio by *`ratio`*. A value of **`1.0`** specifies no tempo alteration, which is the default.

`--start-point=`*`msec`*
: Start converting the audio data at the point *`msec`* mS into the source file. The default value is **`0`**.

See Also

rdexport(1) , rdimport(1) , rdmarkerset(8)