
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

<code>--destination-bit-rate=<i>bit-rate</i></code>	Use a bit rate of <i>bit-rate</i> bits per second. This option is ignored for PCM and FLAC formats, and is mutually exclusive with the <code>--destination-quality</code> option. The default value is 0 .
<code>--destination-channels=<i>chans</i></code>	Use <i>chans</i> channels. Supported values are 1 and 2 . The default value is 2 .
<code>--destination-file=<i>filename</i></code>	Write the converted data to <i>filename</i> . If not specified, the data will be written to the name of the input file with the default extension of the destination format appended.
<code>--destination-format=<i>format</i></code>	Write the converted data to the specified format. <i>format</i> 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
<code>--destination-quality=<i>qual</i></code>	Use a variable bitrate with a quality of <i>chans</i> . Supported values are -1 through 10 . This parameter is used only with a format of 5 (OggVorbis). The default value is 0 .
<code>--destination-sample-rate=<i>rate</i></code>	Use a sample rate of <i>rate</i> samples per second. Not all sample rates are supported for all formats; see the relevant MPEG specifications for details. The default value is 48000 .
<code>--end-point=<i>msec</i></code>	Stop converting the audio data at the point <i>msec</i> 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.
<code>--normalization-level=<i>lvl</i></code>	Peak-normalize the audio to <i>lvl</i> dBFS. A value of 0 disables normalization, which is the default.

<code>--speed-ratio=<i>ratio</i></code>	Alter the tempo of the audio by <i>ratio</i> . A value of 1.0 specifies no tempo alteration, which is the default.
<code>--start-point=<i>msec</i></code>	Start converting the audio data at the point <i>msec</i> mS into the source file. The default value is 0 .

See Also

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