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## 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 <b>0</b> .
<code>--destination-channels=<i>chans</i></code>	Use <i>chans</i> channels. Supported values are <b>1</b> and <b>2</b> . The default value is <b>2</b> .
<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:  <b>0</b> PCM16 WAV  <b>2</b> MPEG Layer 2 (Raw)  <b>3</b> MPEG Layer 3 (Raw)  <b>4</b> Free Lossless Audio Codec (FLAC)  <b>5</b> OggVorbis  <b>6</b> MPEG Layer 2 (BWF WAV Container)  <b>7</b> PCM24 WAV
<code>--destination-quality=<i>qual</i></code>	Use a variable bitrate with a quality of <i>chans</i> . Supported values are <b>-1</b> through <b>10</b> . This parameter is used only with a format of <b>5</b> (OggVorbis). The default value is <b>0</b> .
<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 <b>48000</b> .
<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 <b>-1</b> 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 <b>0</b> 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 <b>1.0</b> 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 <b>0</b> .

## See Also

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