NAME

twolame - an optimised MPEG Audio Layer 2 (MP2) encoder

SYNOPSIS

twolame [options] <infile> [outfile]

DESCRIPTION

TwoLAME is an optimised MPEG Audio Layer 2 (MP2) encoder based on tooLAME by Mike Cheng, which in turn is based upon the ISO dist10 code and portions of LAME. Encoding is performed by the libtwolame library backend.

OPTIONS

Input File

twolame uses libsndfile for reading the input sound file, so the input file can be in any format supported by libsndfile. To read raw PCM audio from STDIN, then use - as the input filename.

Output File

If no output filename is specified, then suffix of the input filename is automatically changed to .mp2. To write the encoded audio to STDOUT then use - as the output filename.

Input Options

-r, --raw-input

Specifies that input is raw signed PCM audio. If audio is stereo, than audio samples are interleaved between the two channels.

-x, --byte-swap

Force byte-swapping of the input. Endian detection is performed automatically by libsndfile, so this option shouldn’t normally be needed.

-s, --samplerate <int>

If inputting raw PCM sound, you must specify the sample rate of the audio in Hz. Valid sample rates: 16000, 22050, 24000, 32000, 44100, 48000Hz. Default sample rate is 44100Hz.

--samplesize <int>

Specifies the sample size (in bits) of the raw PCM audio. Valid sample sizes: 8, 16, 24, 32. Default sample size is 16-bit.

-N, --channels <int>

If inputting raw PCM sound, you must specify the number of channels in the input audio. Default number of channels is 2.

-g, --swap-channels

Swap the Left and Right channels of a stereo input file.

--scale <float>

Scale the input audio prior to encoding. All of the input audio is multiplied by specified value. Value between 0 and 1 will reduce the audio gain, and a value above 1 will increase the gain of the audio.

--scale-l <float>

Same as --scale, but only affects the left channel.

--scale-r <float>

Same as --scale, but only affects the right channel.

Output Options

-m, --mode <char>

Choose the mode of the resulting audio. Default is auto.

- auto - choose mode automatically based on the input
- stereo
- dual channel
- joint stereo
- mono

-a, --downmix

If the input file is stereo then, downmix the left and right input channels into a single mono channel.

-b, --bitrate <int>

Sets the total bitrate (in kbps) for the output file. The default bitrate depends on the number of input channels and samplerate.

Sample Rate	Mono	Stereo
48000	96	192
44100	96	192
32000	80	160
24000	48	96
22050	48	96
16000	32	64

-P, --psyc-mode <int>
Choose the psycho-acoustic model to use (-1 to 4). Model number -1 is turns off psycho-acoustic modelling and uses fixed default values instead. Please see the file *psycho* for a full description of each of the models available. Default model is 3.

-v, --vbr
Enable VBR mode. See *vbr* documentation file for details. Default VBR level is 5.0.

-V, --vbr-level <float>
Enable VBR mode and set quality level. The higher the number the better the quality. Maximum range is -50 to 50 but useful range is -10 to 10. See *vbr* documentation file for details.

-l, --ath <float>
Set the ATH level. Default level is 0.0.

-q, --quick <int>
Enable quick mode. Only re-calculate psycho-acoustic model every specified number of frames.

-S, --single-frame
Enables single frame mode: only a single frame of MPEG audio is output and then the program terminates.

Miscellaneous Options

-c, --copyright
Turn on Copyright flag in output bitstream.

-o, --non-original
Turn off Original flag in output bitstream.

--original

Turn on Original flag in output bitstream.

-p, --protect

Enable CRC error protection in output bitstream. An extra 16-bit checksum is added to frames.

-d, --padding

Turn on padding in output bitstream.

-R, --reserve <int>

Reserve specified number of bits in the each from of the output bitstream.

-e, --deemphasis <char>

Set the de-emphasis type (n/c/5). Default is none.

-E, --energy

Turn on energy level extensions.

Verbosity Options

-t, --talkativity <int>

Set the amount of information to be displayed on stderr (0 to 10). Default is 2.

--quiet

Don't send any messages to stderr, unless there is an error. (Same as --talkativity=0)

--brief

Only display a minimal number of messages while encoding. This setting is quieter than the default talkativity setting. (Same as --talkativity=1)

--verbose

Display an increased number of messages on stderr. This setting is useful to diagnose problems. (Same as --talkativity=4)

RETURN CODES

If encoding completes successfully, then twolame will return 0. However if encoding is not successful, then it will return one of the following codes.

0

(No encoding performed)

- (Error opening input file)
- (Error opening output file)
- (Error allocating memory)
- (Error in chosen encoding parameters)
- (Error reading input audio)
- (Error occurred while encoding)
- (Error writing output audio)

EXAMPLES

This will encode sound.wav to sound.mp2 using the default constant bitrate of 192 kbps and using the default psycho-acoustic model (model 3):

```
twolame sound.wav
```

Constant bitrate of 160kbps and joint stereo encoding, saved to file sound_160.mp2:

```
twolame -b 160 -m j sound.aiff sound_160.mp2
```

Encode sound.wav to newfile.mp2 using psycho-acoustic model 2 and encoding with variable bitrate:

```
twolame -P 2 -v sound.wav newfile.mp2
```

Same as example above, except that the negative value of the "-V" argument means that the lower bitrates will be favoured over the higher ones:

```
twolame -P 2 -V -5 sound.wav newfile.mp2
```

Resample audio file using sox and pipe straight through twolame:

sox sound_11025.aiff -t raw -r 16000 | twolame -r -s 16000 - - > out.mp2

AUTHORS

The twolame frontend was (re)written by Nicholas J Humfrey. The libtwolame library is based on toolame by Mike Cheng. For a full list of authors, please see the AUTHORS file.

RESOURCES

TwoLAME web site: **http://www.twolame.org/**

SEE ALSO

lame(1), mpg123(1), madplay(1), sox(1)

COPYING

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