the following types:

- AIFF
- AIFC (however , no compressed data)
- AU (Sun Audio)
- Parallax JMOVIE file

If the input file does not match the specified sampling rate, then it will be converted on-the-fly .

EXAMPLE

Given anAIFC audio file, named `file.aiff`, containing 16-bit, stereo, 44.1kHz audio, this can be compressed to an MPEG audio file named `file.ampeg` as shown below . Note that since not attributes were specified the attributes of the AIFF file (bits per sample, rate channels) will be the same as the input.

```
cvtaudio file.smpeg file.aiff
```

SEE ALSO

player , recorder , vmpegcomp, smpegcomp, playaudio, cvtaudio

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BUGS

The option flags will soon change to a `name=value` model.

The resampling method used to convert sampling rates is fast, but low quality .

Because the audio library used to read the input file can read any file type, it is easy (too easy) to specify MPEG audio as your input. It doesn' t compress it any more, it just decompresses it and recompresses it. This will just make the quality get worse. Y ou should use the original material.

NAME

ampegcomp - NVR' s MPEG audio compressor/encoder

SYNOPSIS

```
ampegcomp options
```

OPTIONS

-in *filename* Specifies the input file. Default is standard input (stdin).

-out *filename* Specifies the output file. Default is standard output (stdout).

-layer *n* Specifies the MPEG layer to be used for encoding. Currently of the three layers, only layers 1 and 2 are supported. Default is Layer 2.

-mode *name* Where name is one of *stereo* , *dual_channel* , *joint_stereo* . Specifies the 1) the mode (mono or otherwise) used to capture the raw audio data contained in the input file, and 2) the mode to encode the data (i.e., when data is captured in two channels you'll need to specify if the data should be treated as two disjoint channels ("dual channel") or in stereo. If the latter , you have the choice of encoding the data in regular "stereo" or "joint stereo" mode. Joint stereo encoding exploits stereophonic irrelevance or stereophonic redundancy . Default mode is mono.

-ext_mode *n* Where *n* is 0, 1 2 or 3. When mode is joint_stereo, this option indicates which subbands are encoded as intensity_stereo:
0: subbands 4-31 in intensity_stereo
1: subbands 8-31 in intensity_stereo
2: subbands 12-31 in intensity_stereo
3: subbands 16-31 in intensity_stereo Default is 0.

-sampling *s* Where *s* is one of 32000, 44100, or 48000. Specifies the sampling frequency . Default is 32000 kHz. V alid MPEG audio sampling frequencies 48000, 44100, and 32000 Hz.

-bitrate *b* Specifies the target bitrate in kilobits. V alid bitrates depend on the layer selected. V alid bit rates are
Layer 1: 32, 64, 96, 128, 160, 192, 224, 256, 288, 320, 352, 384, 4416, 448
Layer 2: 32, 48, 56, 64, 80, 96, 1 12, 128, 160, 192, 224, 256, 320, 384
The default for Layer 1 is 448 kb/sec.; the default for Layer 2 is 384 kb/sec.

-psy *m* Where *m* is 1 or 2. Specifies the psychoacoustic model to use. Default is 2.

-emphasis *e* Where e is *5* or *c*. Specifies the type of de-emphasis to use: *5* for 50/15, *c* for ccitt j.17. The default is none.

-copyright Indicates that the input material is NOT copyrighted.

+copyright Indicates that the input material is IS copyrighted.

-original Indicates that the input material is NOT the original.

+original Indicates that the input material is IS the original.

DESCRIPTION

audiocomp is an MPEG audio encoder/compressor that takes a variety of audio files and writes out an MPEG audio data stream to the specified output file. Input files may be any of