AACENC 1.0 Documentation

April 19, 2001

Introduction – What is AACENC

AACENC is audio compression technology demo based on MPEG-2 and MPEG-4 international multimedia standards (ISO/IEC 13818-7 and ISO/IEC 14496-3) implementation, developed by PsyTEL Research and providing very high audio quality. The audio quality provided is indistinguishable from CD-Audio to most users at compression ratios 11:1 to 14:1

PsyTEL AACENC uses most sophisticated technologies to simulate human hearing (perceptual model) so it is able to pick parts of signal that are not audible to humans. This phenomenon is so fantastic that we can leave only 1/10 of original material and it will still sound exactly as original to average listener. Our technology is much more optimized than example technology provided as source code with ISO/IEC standard. Lot of "know-how" was engaged in order to tune every part of the technology for maximum quality.

AACENC command line tool can be used to store and organize music in very high quality audio format suitable for PC-Multimedia, or to evaluate our implementation of MPEG-4 AAC and maybe integrate it into your custom devices (Broadcasting Equipment, Internet Audio, Music Storage, etc…)

This package includes both consumer and professional quality optimized AAC encoder.

Compression ratios and sampling rates supported

PsyTEL AACENC 1.0 supports bit-rates from 32 kilobits per second to 512 kilobits per second. Compression ratio can be fixed (constant through time) or variable – where encoder decides whether additional bits must be used in order to provide transparent audio quality. Current consumer version supports 1 (mono) to 6 (multichannel) wave files.

For typical recordings from CD, transparent (indistinguishable) quality can be achieved at compression ratio of 11:1 – or – 128 kilobits per second.

CD-Audio = 1411 kilobits per second
"CD Quality" Ratio = 11:1

1411 / 11 = 128 kilobits per second for compressed audio

Bit rates that PsyTEL AACENC supports are:

Bit Rate	Bandwidth	Quality
16 kBits/s	6 kHz	SW Radio
32 kBits/s	8 kHz	Better than AM Radio
64 kBits/s	11 kHz	Better than FM Radio
128 kBits/s	18 kHz	CD Quality (average)
192 kBits/s	24 kHz	DAT Quality
256 kBits/s	24-48 kHz	Archival/Studio quality

Some Command Line Examples

To encode file with default AAC CBR 128 quality:

```
aacenc -if myfile.wav
```

This is the same as if you typed:

```
aacenc –br 128 -if myfile.wav –of myfile.aac
```

For encoding in *professional* CBR mode type:

```
aacenc –production –br 128 -if myfile.wav
```

To encode file with default VBR quality (constant quality – variable bitrate)

```
aacenc –vbrhi -if myfile.wav
```

To encode file with **recommended VBR** preset (it is always recommended to use presets):

`aacenc –normal -if myfile.wav`

Available VBR presets are: -tape, -radio, -internet, -streaming, -normal, -extreme, -archive and -ultra

Note: current version of AACEnc does not have *resampling* tool implemented. For maximum performance over a given bitrate it might be required to use resampling to other sampling rate (for example: 32 kHz for 78 kbits/s stereo). External resampling tool must be used in these conditions.

Command Line Reference

Switch	**Description**
-br <n>	Bitrate in kBits per second. N should be between 8 and 512 – Default "CD Quality" for Stereo 44.1 kHz sampled .wav files is 128 kBits/s (-br 128) and it is set by default.
-vbrhi	Recommended manual Variable Bitrate Mode. In this mode encoder will allocate more or less bits than average in order to maintain constant audio quality. Can be used in combination with –br switch, where –br selects *base* bitrate. Base bitrate is set to 64 kBits/s * number of channels, but it can be changed if additional transcoding is expected.
-tape, -radio, -internet, -streaming, -normal, -extreme, -archive and -ultra	Automatic VBR presets, it is **recommended** to use these profiles for highest quality encodings. Normal profile is in most cases enough even for higest demanding users. Archive mode is designed for music-archiving purposes while Ultra mode is designed for multiple encoding-decoding cycles. Syntax for using these profiles is very easy: aacenc -profile_name -if <input_file.wav>
-vr	Lower quality manual Variable Bitrate Mode. Recommended for files prepared for internet streaming of FM broadcasts.
-c <n>	Lowpass filter (cut-off) in Hertz (Hz). You may need to change default cut-off

	presets in order to achieve »warmer« or »smoother« sound. In general, it is not recommended to change this value because it may degrade sound quality. By default, cut-off at 18 kHz is used at 128 kbits/s encoding, and 22.05 for 192 kbits/s encodings. For professional quality encoding at 128 kbits/s (44.1 kHz, stereo) it is desirable to set the lowpass filter to 16 kHz (-c 15995)
-if <filename>	Input file name, for example –if myfile.wav If file contains spaces it could be used with –if "my long file.wav" switch. By default output file name is set to be same as input, but with .aac extension istead of .wav
-of <filename>	Output file name. Uses same convention as –if switch. May be ommited, because encoder will automatically set best output file name.
-lc	Tells encoder to use MPEG-2 AAC Low Complexity profile instead of MPEG-4 AAC Main LTP profile. Use this switch if you plan to transfer .aac files to some portable device supporting MPEG-2 AAC LC only, or to play it on slower machines. LC mode is designed for compatibility with large amount of software and hardware decoders.
-qual <n>	Encoder quality level (from 1 to 9). By default, highest setting (9) is used, but in order to increase encoding speed lower value could be used. Note that this switch affects sound quality.
-adif	Use ADIF instead of default ADTS header. For compatibility with some decoding software.
-nh	Tells encoder not to use headers at all. Provided for compatibility with some decoding software and very low bitrate encoding. At low bitrates it might increase encoding quality since ISO ADTS header carries 56-58 bits of information.
-production	Highest-Quality CBR encoding switch. However, encoding speed will be much lower than normal mode. This switch is used for "professional quality" encoding instead of "consumer quality".
-low_ath	Tells encoder to use highest sensitivity threshold of audibility. This option is automatically used on -extreme and higher VBR presets. It is not recommended to use this switch at bitrates lower than 192 kbits/s
-ihsc	Improved Human Speech Coding mode, automatically enabled for –archive and –extreme presets. Not recommended for CBR or low bitrate VBR modes.
-artist, -title, -genre, -year	Tagging options (for music taging and identification)

Contact Informaton

Web site: http://www.psytel-research.co.yu
AAC Technology Info: aac-info@psytel-research.co.yu

Project Manager: Ivan Dimkovic (dim@psytel-research.co.yu)
Telephone: ++381 64 11 40 600