AAC-LC Encoder

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MPEG-2/4 AAC-LC (Advanced Audio Coding - Low Complexity version) is a popular audio coding technique recommended by MPEG committee. The codec handles audio signals sampled in the range of 8 kHz to 96 kHz. It operates on a frame of 1024 samples. The bit-rate can vary in the range from 8 to 576 kbps/channel (depending on the sampling rate). Low Complexity version of AAC provides good compromise between the codec complexity and the audio quality.

Features Supported

- MPEG2 and MPEG4 AAC LC(Low complexity)
- Up to two channels
- Sample rates supported
  - 8,11.025,12,16,22.05,24,32, 44.1, 48, 88.2 and 96kHz
- Bit-rates per channel 8 – 576 kbps
  - Max Bit-rate per channel is 6 times sampling frequency.
- Channels: Mono/Stereo
- Bit-streams ADIF (Audio Data Interchange Format), ADTS (Audio Data Transport Stream)
- Average bit-rate
- Tools: MS (Mid-side Stereo), TNS (Temporal Noise shaping).
- Efficient psycho-acoustic model.
- C Callable interface/TI XDM API for encoder
- Efficient scratch memory with reduced stack requirements.
- Optimized for low footprint & processing power.

Not Supported

- More than two channels of audio
- IS (Intensity Stereo) Coding
- PNS (Perceptual Noise Substitution)
- MP4 packetization

Encoder Validation

AAC-LC Encoder is an informative standard. There is no standard measure or tool for evaluating the quality /fidelity of the encoder. The encoders produce complex artifacts, which is dependent on the source material. Taking these into consideration, the test bench for the audio coders includes the following types of test.

Features

- Bit Stream Compliance: Tests to ensure that the generated bit-stream is in conformance with the specification.
- Objective Quality Evaluation: Audio Quality test based on the ITU BS.1387 standard for objective audio quality evaluation.
- Subjective Quality Evaluation: Listening tests to evaluate the quality.
- Artifact Listening Tests: Listening tests to ensure that the encoder does not produce the artifacts.

Resource requirements on ARM9E

MCPS	Program Memory (Kbytes)	Data Memory (Kbytes)			
Peak		Tables	Scratch	Stack	Persistent
42.28	75.3	26.5	23.55	2.5	19.4

Note

Input/ Output buffers details are given in the next page.

MCPS indicate the CPU usage for processing AAC only for 44.1 kHz at 128 kbps with TNS disabled for two channel file.

MCPS measurement on 0 wait-state memory access

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Details of ARM9E Resources required

CPU Loading

Description	Simulator		Hardware	
	Average MCPS	Peak MCPS	Average MCPS	Peak MCPS
AAC only at 128 kbps, 44.1kHz, TNS enabled	37.30	52.58	57.8	77.8
AAC only at 128 kbps, 44.1kHz, TNS disabled	31.62	42.28	50.7	67.1

Memory Usage (kB)

Program	Tables	Static	Scratch	Stack	Input	Output
75.3	26.5	19.4	23.55	2.5	4	1.5

Memory Breakup (kB)

Tables						Static		Scratch
9.57	6.0	1.52	2.59	0.75	6.16	19.2	0.4	23.55

Note:

- Performance numbers on Simulator generated with ARM RVDS Tools version 2.1 *with 0-wait state memory access and without cache.*
- Hardware performance generated on a ARM9E processor with 16kB of I Cache and 8kB of D-Cache
- Hardware performance generated under Linux2.6 , using the ARM-GCC 3.4.3 Compiler
- MCPS numbers on the hardware will vary with the I-Cache and D-Cache size and with the memory configuration/place
- I/O Buffers mentioned above are for single input and output buffers
- MCPS/MIPS indicate the CPU usage for processing 2 channel music streams (AAC, 27.wav (SQAM file), 44.1 kHz, 128 kbps).
- Program memory doesn't include the code size of the test bench and standard library functions
- Data memory should be aligned to desired byte-boundary to meet the performance/functionality requirement