AAC-LC Encoder

AAC-LC Encoder

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 48 kHz. It operates on a frame of 1024 samples. The bit-rate can vary in the range from 8 to 160 kbps/channel (depending on the sampling rate). Low Complexity version of AAC provides good compromise between the codec complexity and the audio quality.

This document describes the test plan for the MPEG-2/4 AAC-LC Encoder developed by Ittiam Systems Pvt Ltd.

Features

Supported

- MPEG2 and MPEG4 AAC LC(Low complexity)
- Up to two channels
- Sample rates supported

  8 kHz to 48 kHz
- Bit-rates supported 8-160 kbps/channel
- 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 for encoder
- Efficient scratch memory with reduced stack requirements.
- Optimized for low footprint & processing power.
- XDMI API

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 C64x+

CPU Load (MCPS)		Program Memory (Kbytes)	Data Memory (Kbytes)			
Peak	Average		Table	Scratch	Stack	Persistent
33.41	22.26	94	27.6	22.6	2.1	19.2

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

MCPS/MIPS indicate the CPU usage for processing AAC only for 44.1 kHz at 128 kbps for QL2 level with TNS enabled for two channel file.

MIPS or MCPS measurement on 0 wait-state memory access

Kane Computing Ltd
7 Theatre Court, London Road,
Northwich, Cheshire, CW9 5HB, UK.
Tel: +44(0)1606 351006
Fax: +44(0)1606 351007/8
Email: sales@kanecomputing.com
Web: www.kanecomputing.co.uk

Details of C64x+ Resources required

CPU Loading

Description	Simulator		Hardware	
	Peak MCPS	Avg MCPS	Peak MCPS	Avg MCPS
AAC only at 128 kbps, 44.1kHz	33.41	22.26	46.86	34.18

Memory Usage (KB)

Program	Tables	Static	Scratch	Stack	Input	Output
94	27.6	19.2	22.6	2.1	8	1.5

Note:

- I/O Buffers
  - Input Buffer Size : 8 kbytes
  - Output Buffer Size 1.5 kbytes
- Performance generated on *CCS* 3.2.39.5 *with C64x+ Cycle Accurate Simulator with 0 wait state memory access*
- Hardware Configuration performance generated on a DM6446 processor with all data and program memory sections placed in the external memory, with cache configuration of 32 KB L1 P Cache, 16 KB L1 D Cache & 64 KB L2 Cache, and cache thrashed after decoding each frame.
- MCPS numbers on the hardware will vary with the I-Cache and D-Cache size and with the memory configuration/placement
- 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