Fraunhofer
IIS

Advanced Audio Coding Encoder Library

MPEG-2 and MPEG-4,
AAC Low-Complexity (AAC-LC),
High-Efficiency AAC v2 (HE-AAC v2),
AAC Low-Delay (AAC-LD),
AAC Enhanced Low-Delay (AAC-ELD v2)
encoder

Confidential

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Chapter 1

Introduction

1.1 Scope

This document describes the high-level interface and usage of the ISO/MPEG-2/4 AAC Encoder library developed by the Fraunhofer Institute for Integrated Circuits (IIS).

The library implements encoding on the basis of the MPEG-2 and MPEG-4 AAC Low-Complexity standard, and depending on the library's configuration, MPEG-4 High-Efficiency AAC v2 and/or AAC-ELD standard.

All references to SBR (Spectral Band Replication) are only applicable to HE-AAC or AAC-ELD versions of the library. All references to PS (Parametric Stereo) are only applicable to HE-AAC v2 versions of the library.

1.2 Encoder Basics

This document can only give a rough overview about the ISO/MPEG-2 and ISO/MPEG-4 AAC audio coding standard. To understand all the terms in this document, you are encouraged to read the following documents.

- ISO/IEC 13818-7 (MPEG-2 AAC), which defines the syntax of MPEG-2 AAC audio bitstreams.
- ISO/IEC 14496-3 (MPEG-4 AAC, subparts 1 and 4), which defines the syntax of MPEG-4 AAC audio bitstreams.
- Lutzky, Schuller, Gayer, Krämer, Wabnik, ”A guideline to audio codec delay”, 116th AES Convention, May 8, 2004

MPEG Advanced Audio Coding is based on a time-to-frequency mapping of the signal. The signal is partitioned into overlapping portions and transformed into frequency domain. The spectral components are then quantized and coded.
An MPEG-2 or MPEG-4 AAC audio bitstream is composed of frames. Contrary to MPEG-1/2 Layer-3 (mp3), the length of individual frames is not restricted to a fixed number of bytes, but can take on any length between 1 and 768 bytes.

Chapter 2

Library Usage

2.1 API Files

All API header files are located in the folder /include of the release package. All header files are provided for usage in C/C++ programs. The AAC encoder library API functions are located in aacenc_lib.h.

In binary releases the encoder core resides in statically linkable libraries called for example libAACenc.a/libFDK.a (LINUX) or FDK_fastaaclib.lib (MS Visual C++) for the plain AAC-LC core encoder and libSBRenc.a (LINUX) or FDK_sbrEncLib.lib (MS Visual C++) for the SBR (Spectral Band Replication) and PS (Parametric Stereo) modules.

2.2 Calling Sequence

For encoding of ISO/MPEG-2/4 AAC bitstreams the following sequence is mandatory. Input read and output write functions as well as the corresponding open and close functions are left out, since they may be implemented differently according to the user's specific requirements. The example implementation uses file-based input/output.

1. Call aacEncOpen() to allocate encoder instance with required configuration.

```
HANDLE_AACENCODER hAacEncoder = NULL;
if ( (ErrorStatus = aacEncOpen(&hAacEncoder,0,0)) != AACENC_OK ) {
```

2. Call aacEncoder_SetParam() for each parameter to be set. AOT, samplingrate, channelMode, bitrate and transport type are mandatory.

```
ErrorStatus = aacEncoder_SetParam(hAacEncoder, parameter, value);
```

3. Call aacEncEncode() with NULL parameters to initialize encoder instance with present parameter set.

```
ErrorStatus = aacEncEncode(hAacEncoder, NULL, NULL, NULL, NULL);
```

4. Call aacEncInfo() to retrieve a configuration data block to be transmitted out of band. This is required when using RFC3640 or RFC3016 like transport.

```
AACENC_InfoStruct encInfo;
aacEncInfo(hAacEncoder, &encInfo);
```

5. Encode input audio data in loop.

```
do
{
```

Feed input buffer with new audio data and provide input/output arguments to aacEncEncode().

```
ErrorStatus = aacEncEncode(hAacEncoder,
                           &inBufDesc,
                           &outBufDesc,
                           &inargs,
                           &outargs);
```

Write output data to file or audio device.

```
} while (ErrorStatus==AACENC_OK);
```

6. Call aacEncClose() and destroy encoder instance.

```
aacEncClose(&hAacEncoder);
```

2.3 Encoder Instance Allocation

The assignment of the aacEncOpen() function is very flexible and can be used in the following way.

- If the amount of memory consumption is not an issue, the encoder instance can be allocated for the maximum number of possible audio channels (for example 6 or 8) with the full functional range supported by the library. This is the default open procedure for the AAC encoder if memory consumption does not need to be minimized.

```
aacEncOpen(&hAacEncoder,0,0)
```

- If the required MPEG-4 AOTs do not call for the full functional range of the library, encoder modules can be allocated selectively.

```
-------------------------------------------------------
 AAC | SBR |  PS | MD |          FLAGS          | value
-----+-----+-----+----+-------------------------+-------
  X  |  -  |  -  |  - | (0x01)                  |  0x01
  X  |  X  |  -  |  - | (0x01|0x02)             |  0x03
  X  |  X  |  X  |  - | (0x01|0x02|0x04)        |  0x07
  X  |  -  |  -  |  X | (0x01           |0x10)  |  0x11
  X  |  X  |  -  |  X | (0x01|0x02      |0x10)  |  0x13
  X  |  X  |  X  |  X | (0x01|0x02|0x04|0x10)  |  0x17
-------------------------------------------------------
 - AAC: Allocate AAC Core Encoder module.
 - SBR: Allocate Spectral Band Replication module.
 - PS: Allocate Parametric Stereo module.
 - MD: Allocate Meta Data module within AAC encoder.
```

```
aacEncOpen(&hAacEncoder,value,0)
```

- Specifying the maximum number of channels to be supported in the encoder instance can be done as follows.
  - For example allocate an encoder instance which supports 2 channels for all supported AOTs. The library itself may be capable of encoding up to 6 or 8 channels but in this example only 2 channel encoding is required and thus only buffers for 2 channels are allocated to save data memory.

```
aacEncOpen(&hAacEncoder,0,2)
```

– Additionally the maximum number of supported channels in the SBR module can be denoted separately.
In this example the encoder instance provides a maximum of 6 channels out of which up to 2 channels support SBR. This encoder instance can produce for example 5.1 channel AAC-LC streams or stereo HE-AAC (v2) streams. HE-AAC 5.1 multi channel is not possible since only 2 out of 6 channels support SBR, which saves data memory.

```
aacEncOpen(&hAacEncoder,0,6|(2<<8))
```

2.4 Input/Output Arguments

2.4.1 Provide Buffer Descriptors

In the present encoder API, the input and output buffers are described with buffer descriptors. This mechanism allows a flexible handling of input and output buffers without impact to the actual encoding call. Optional buffers are necessary e.g. for ancillary data, meta data input or additional output buffers describing superframing data in DAB+ or DRM+.
At least one input buffer for audio input data and one output buffer for bitstream data must be allocated. The input buffer size can be a user defined multiple of the number of input channels. PCM input data will be copied from the user defined PCM buffer to an internal input buffer and so input data can be less than one AAC audio frame. The output buffer size should be 6144 bits per channel excluding the LFE channel. If the output data does not fit into the provided buffer, an AACENC_ERROR will be returned by aacEncEncode().

```
static INT_PCM          inputBuffer[8*2048];
static UCHAR            ancillaryBuffer[50];
static AACENC_MetaData  metaDataSetup;
static UCHAR            outputBuffer[8192];
```

All input and output buffer must be clustered in input and output buffer arrays.

```
static void* inBuffer[]        = { inputBuffer, ancillaryBuffer, &metaDataSetup };
static INT   inBufferIds[]     = { IN_AUDIO_DATA, IN_ANCILLRY_DATA,
      IN_METADATA_SETUP };
static INT   inBufferSize[]    = { sizeof(inputBuffer), sizeof(ancillaryBuffer),
                                   sizeof(metaDataSetup) };
static INT   inBufferElSize[]  = { sizeof(INT_PCM), sizeof(UCHAR),
                                   sizeof(AACENC_MetaData) };

static void* outBuffer[]       = { outputBuffer };
static INT   outBufferIds[]    = { OUT_BITSTREAM_DATA };
static INT   outBufferSize[]   = { sizeof(outputBuffer) };
static INT   outBufferElSize[] = { sizeof(UCHAR) };
```

Allocate buffer descriptors

```
AACENC_BufDesc inBufDesc;
AACENC_BufDesc outBufDesc;
```

Initialize input buffer descriptor

```
inBufDesc.numBufs            = sizeof(inBuffer)/sizeof(void*);
inBufDesc.bufs               = (void**)&inBuffer;
inBufDesc.bufferIdentifiers = inBufferIds;
inBufDesc.bufSizes           = inBufferSize;
inBufDesc.bufElSizes         = inBufferElSize;
```

Initialize output buffer descriptor

```
outBufDesc.numBufs           = sizeof(outBuffer)/sizeof(void*);
outBufDesc.bufs              = (void**)&outBuffer;
outBufDesc.bufferIdentifiers = outBufferIds;
outBufDesc.bufSizes          = outBufferSize;
outBufDesc.bufElSizes        = outBufferElSize;
```

2.4.2 Provide Input/Output Argument Lists

The input and output arguments of an aacEncEncode() call are described in argument structures.

```
AACENC_InArgs     inargs;
AACENC_OutArgs    outargs;
```

2.5 Feed Input Buffer

The input buffer should be handled as a modulo buffer. New audio data in the form of pulse-code-modulated samples (PCM) must be read from external and be fed to the input buffer depending on its fill level. The required sample bitrate (represented by the data type INT_PCM which is 16, 24 or 32 bits wide) is fixed and depends on library configuration (usually 16 bit).

```
inargs.numInSamples += WAV_InputRead ( wavIn,
                                       &inputBuffer[inargs.numInSamples],
                                        FDKmin(encInfo.inputChannels*encInfo.
      frameLength,
                                               sizeof(inputBuffer) /
                                               sizeof(INT_PCM)-inargs.
      numInSamples),
                                        SAMPLE_BITS
                                      );
```

After the encoder's internal buffer is fed with incoming audio samples, and aacEncEncode() processed the new input data, update/move remaining samples in input buffer, simulating a modulo buffer:

```
if (outargs.numInSamples>0) {
    FDKmemmove( inputBuffer,
                &inputBuffer[outargs.numInSamples],
                sizeof(INT_PCM)*(inargs.numInSamples-outargs.
      numInSamples) );
    inargs.numInSamples -= outargs.numInSamples;
}
```

2.6 Output Bitstream Data

If any AAC bitstream data is available, write it to output file or device. This can be done once the following condition is true:

```
if (outargs.numOutBytes>0) {

}
```

If you use file I/O then for example call mpegFileWrite_Write() from the library libMpegFileWrite

```
mpegFileWrite_Write(hMpegFile, outputBuffer, outargs.numOutBytes,
      aacEncoder_GetParam(hAacEncoder, AACENC_GRANULE_LENGTH));
```

2.7 Meta Data Configuration

If the present library is configured with Metadata support, it is possible to insert meta data side info into the generated audio bitstream while encoding.

To work with meta data the encoder instance has to be allocated with meta data support. The meta data mode must be be configured with the AACENC_METADATA_MODE parameter and aacEncoder_SetParam() function.

```
aacEncoder_SetParam(hAacEncoder, AACENC_METADATA_MODE, 0-3);
```

This configuration indicates how to embed meta data into bitstrem. Either no insertion, MPEG or ETSI style. The meta data itself must be specified within the meta data setup structure AACENC_MetaData.

Changing one of the AACENC_MetaData setup parameters can be achieved from outside the library within IN_METADATA_SETUP input buffer. There is no need to supply meta data setup structure every frame. If there is no new meta setup data available, the encoder uses the previous setup or the default configuration in initial state.

In general the audio compressor and limiter within the encoder library can be configured with the AACENC_METADATA_DRC_PROFILE parameter AACENC_MetaData::drc_profile and and AACENC_MetaData::comp_profile.

2.8 Encoder Reconfiguration

The encoder library allows reconfiguration of the encoder instance with new settings continuously between encoding frames. Each parameter to be changed must be set with a single aacEncoder_SetParam() call. The internal status of each parameter can be retrieved with an aacEncoder_GetParam() call.
There is no stand-alone reconfiguration function available. When parameters were modified from outside the library, an internal control mechanism triggers the necessary reconfiguration process which will be applied at the beginning of the following aacEncEncode() call. This state can be observed from external via the AACENC_INIT_STATUS and aacEncoder_GetParam() function. The reconfiguration process can also be applied immediately when all parameters of an aacEncEncode() call are NULL with a valid encoder handle.

The internal reconfiguration process can be controlled from extern with the following access.

```
aacEncoder_SetParam(hAacEncoder, AACENC_CONTROL_STATE,
      AACENC_CTRLFLAGS);
```

2.9 Encoder Parametrization

All parameteres listed in AACENC_PARAM can be modified within an encoder instance.

2.9.1 Mandatory Encoder Parameters

The following parameters must be specified when the encoder instance is initialized.

```
aacEncoder_SetParam(hAacEncoder, AACENC_AOT, value);
aacEncoder_SetParam(hAacEncoder, AACENC_BITRATE, value);
aacEncoder_SetParam(hAacEncoder, AACENC_SAMPLERATE, value);
aacEncoder_SetParam(hAacEncoder, AACENC_CHANNELMODE, value);
```

Beyond that is an internal auto mode which preinitializes the AACENC_BITRATE parameter if the parameter was not set from extern. The bitrate depends on the number of effective channels and sampling rate and is determined as follows.

```
AAC-LC (AOT_AAC_LC): 1.5 bits per sample
HE-AAC (AOT_SBR): 0.625 bits per sample (dualrate sbr)
HE-AAC (AOT_SBR): 1.125 bits per sample (downsampled sbr)
HE-AAC v2 (AOT_PS): 0.5 bits per sample
```

2.9.2 Channel Mode Configuration

The input audio data is described with the AACENC_CHANNELMODE parameter in the aacEncoder_SetParam() call. It is not possible to use the encoder instance with a 'number of input channels' argument. Instead, the channelMode must be set as follows.

```
aacEncoder_SetParam(hAacEncoder, AACENC_CHANNELMODE, value);
```

The parameter is specified in ::CHANNEL_MODE and can be mapped from the number of input channels in the following way.

```
CHANNEL_MODE chMode = MODE_INVALID;

switch (nChannels) {
  case 1:  chMode = MODE_1;          break;
  case 2:  chMode = MODE_2;          break;
  case 3:  chMode = MODE_1_2;        break;
  case 4:  chMode = MODE_1_2_1;      break;
  case 5:  chMode = MODE_1_2_2;      break;
  case 6:  chMode = MODE_1_2_2_1;    break;
  case 7:  chMode = MODE_6_1;        break;
  case 8:  chMode = MODE_7_1_BACK;   break;
  default:
    chMode = MODE_INVALID;
}
return chMode;
```

2.9.3 Bitreservoir Configuration

In AAC, the default bitreservoir configuration depends on the chosen bitrate per frame and the number of effective channels. The size can be determined as below.

$$bitreservoir = nEffChannels \quad 6144 \quad (bitrate \quad framelength/samplerate)$$

Due to audio quality concerns it is not recommended to change the bitreservoir size to a lower value than the default setting! However, for minimizing the delay for streaming applications or for achieving a constant size of the bitstream packages in each frame, it may be necessaray to change the bitreservoir size. This can be done with the AACENC_PEAK_BITRATE parameter.

```
aacEncoder_SetParam(hAacEncoder, AACENC_PEAK_BITRATE, value);
```

By setting AACENC_BITRATEMODE to fixed framing, the bitreservoir is disabled. A disabled bitreservoir results in a constant size for each bitstream package. Please note that especially at lower bitrates a disabled bitreservoir can downgrade the audio quality considerably! The default bitreservoir configuration can be achieved as follows.

```
aacEncoder_SetParam(hAacEncoder, AACENC_BITRESERVOIR, -1);
```

To achieve acceptable audio quality with a reduced bitreservoir size setting at least 1000 bits per audio channel is recommended. For a multichannel audio file with 5.1 channels the bitreservoir reduced to 5000 bits results in acceptable audio quality.

2.9.4 Variable Bitrate Mode

The encoder provides various Variable Bitrate Modes that differ in audio quality and average overall bitrate. The given values are averages over time, different encoder settings and strongly depend on the type of audio signal. The VBR configurations can be adjusted via AACENC_BITRATEMODE encoder parameter.

VBR_MODE	Approx. Bitrate in kbps/channel	
	AAC-LC	AAC-LD/AC_ELD
VBR_1	32 - 48	32 - 56

```
    VBR_2 |    40 -  56   |      40 -  64
    VBR_3 |    48 -  64   |      48 -  72
    VBR_4 |    64 -  80   |      64 -  88
    VBR_5 |    96 - 120   |     112 - 144
-----------------------------------------
```

The bitrate ranges apply for individual audio channels. In case of multichannel configurations the average bitrate might be estimated by multiplying with the number of effective channels. This corresponds to all audio input channels exclusively the low frequency channel. At configurations which are making use of downmix modules the AAC core channels respectively downmix channels shall be considered. For AACENC_AOT which are using SBR, the average bitrate can be estimated by using the ratio of 0.5 for dualrate SBR and 0.75 for downsampled SBR configurations.

2.9.5 Audio Quality Considerations

The default encoder configuration is suggested to be used. Encoder tools such as TNS and PNS are activated by default and are internally controlled (see Encoder Tools).

There is an additional quality parameter called AACENC_AFTERBURNER. In the default configuration this quality switch is deactivated because it would cause a workload increase which might be significant. If workload is not an issue in the application we recommended to activate this feature.

```
aacEncoder_SetParam(hAacEncoder, AACENC_AFTERBURNER, 0/1);
```

2.9.6 ELD Auto Configuration Mode

For ELD configuration a so called auto configurator is available which configures SBR and the SBR ratio by itself. The configurator is used when the encoder parameter AACENC_SBR_MODE and AACENC_SBR_RATIO are not set explicitly.

Based on sampling rate and chosen bitrate a reasonable SBR configuration will be used.

```
-------------------------------------------------------------------
 Sampling Rate |   Total Bitrate | No. of | SBR |       SBR Ratio
     [kHz]     |      [bit/s]    |  Chan  |     |
               |                 |        |     |
---------------+-----------------+--------+-----+-----------------
     ]min, 16[ |    min -    max |      1 | off |             ---
---------------+-----------------+--------+-----+-----------------
          [16] |    min -  27999 |      1 |  on | downsampled SBR
               |  28000 -    max |      1 | off |             ---
---------------+-----------------+--------+-----+-----------------
     ]16 - 24] |    min -  39999 |      1 |  on | downsampled SBR
               |  40000 -    max |      1 | off |             ---
---------------+-----------------+--------+-----+-----------------
     ]24 - 32] |    min -  27999 |      1 |  on |    dualrate SBR
               |  28000 -  55999 |      1 |  on | downsampled SBR
               |  56000 -    max |      1 | off |             ---
---------------+-----------------+--------+-----+-----------------
   ]32 - 44.1] |    min -  63999 |      1 |  on |    dualrate SBR
               |  64000 -    max |      1 | off |             ---
---------------+-----------------+--------+-----+-----------------
   ]44.1 - 48] |    min -  63999 |      1 |  on |    dualrate SBR
               |  64000 -  max   |      1 | off |             ---
               |                 |        |     |
---------------+-----------------+--------+-----+-----------------
     ]min, 16[ |    min -    max |      2 | off |             ---
---------------+-----------------+--------+-----+-----------------
          [16] |    min -  31999 |      2 |  on | downsampled SBR
               |  32000 -  63999 |      2 |  on | downsampled SBR
               |  64000 -    max |      2 | off |             ---
---------------+-----------------+--------+-----+-----------------
```

]16 - 24]	min - 47999	2	on	downsampled SBR
	48000 - 79999	2	on	downsampled SBR
	80000 - max	2	off	---
]24 - 32]	min - 31999	2	on	dualrate SBR
	32000 - 67999	2	on	dualrate SBR
	68000 - 95999	2	on	downsampled SBR
	96000 - max	2	off	---
]32 - 44.1]	min - 43999	2	on	dualrate SBR
	44000 - 127999	2	on	dualrate SBR
	128000 - max	2	off	---
]44.1 - 48]	min - 43999	2	on	dualrate SBR
	44000 - 127999	2	on	dualrate SBR
	128000 - max	2	off	---

2.9.7 Reduced Delay (Downscaled) Mode

The downscaled mode of AAC-ELD reduces the algorithmic delay of AAC-ELD by virtually increasing the sampling rate. When using the downscaled mode, the bitrate should be increased for keeping the same audio quality level. For common signals, the bitrate should be increased by 25% for a downscale factor of 2.

Currently, downscaling factors 2 and 4 are supported. To enable the downscaled mode in the encoder, the framelength parameter AACENC_GRANULE_LENGTH must be set accordingly to 256 or 240 for a downscale factor of 2 or 128 or 120 for a downscale factor of 4. The default values of 512 or 480 mean that no downscaling is applied.

```
aacEncoder_SetParam(hAacEncoder, AACENC_GRANULE_LENGTH, 256);
aacEncoder_SetParam(hAacEncoder, AACENC_GRANULE_LENGTH, 128);
```

Downscaled bitstreams are fully backwards compatible. However, the legacy decoder needs to support high sample rate, e.g. 96kHz. The signaled sampling rate is multiplied by the downscale factor. Although not required, downscaling should be applied when decoding downscaled bitstreams. It reduces CPU workload and the output will have the same sampling rate as the input. In an ideal configuration both encoder and decoder should run with the same downscale factor.

The following table shows approximate filter bank delays in ms for common sampling rates(sr) at framesize(fs), and downscale factor(dsf), based on this formula:

$$1000 \quad fs=(dsf \quad sr)$$

	512/2	512/4	480/2	480/4
22050	17.41	8.71	16.33	8.16
32000	12.00	6.00	11.25	5.62
44100	8.71	4.35	8.16	4.08
48000	8.00	4.00	7.50	3.75

2.10 Audio Channel Configuration

The MPEG standard refers often to the so-called Channel Configuration. This Channel Configuration is used for a fixed Channel Mapping. The configurations 1-7 and 11,12,14 are predefined in MPEG standard and used for implicit signalling within the encoded bitstream. For user defined Configurations the Channel Configuration is set to 0 and the Channel Mapping must be explecitly described with an appropriate Program Config Element. The present Encoder implementation

does not allow the user to configure this Channel Configuration from extern. The Encoder implementation supports fixed Channel Modes which are mapped to Channel Configuration as follow.

ChannelMode	ChCfg	Height	front_El	side_El	back_El	lfe_El
MODE_1	1	NORM	SCE			
MODE_2	2	NORM	CPE			
MODE_1_2	3	NORM	SCE, CPE			
MODE_1_2_1	4	NORM	SCE, CPE		SCE	
MODE_1_2_2	5	NORM	SCE, CPE		CPE	
MODE_1_2_2_1	6	NORM	SCE, CPE		CPE	LFE
MODE_1_2_2_2_1	7	NORM	SCE, CPE, CPE		CPE	LFE
MODE_6_1	11	NORM	SCE, CPE		CPE, SCE	LFE
MODE_7_1_BACK	12	NORM	SCE, CPE		CPE, CPE	LFE
MODE_7_1_TOP_FRONT	14	NORM	SCE, CPE		CPE	LFE
		TOP	CPE			
MODE_7_1_REAR_SURROUND	0	NORM	SCE, CPE		CPE, CPE	LFE
MODE_7_1_FRONT_CENTER	0	NORM	SCE, CPE, CPE		CPE	LFE

- NORM: Normal Height Layer. - TOP: Top Height Layer. - BTM: Bottom Height Layer.
- SCE: Single Channel Element. - CPE: Channel Pair. - LFE: Low Frequency Element.

The Table describes all fixed Channel Elements for each Channel Mode which are assigned to a speaker arrangement. The arrangement includes front, side, back and lfe Audio Channel Elements in the normal height layer, possibly followed by front, side, and back elements in the top and bottom layer (Channel Configuration 14).
This mapping of Audio Channel Elements is defined in MPEG standard for Channel Config 1-7 and 11,12,14.
In case of Channel Config 0 or writing matrix mixdown coefficients, the encoder enables the writing of Program Config Element itself as described in encPCE. The configuration used in Program Config Element refers to the denoted Table.
Beside the Channel Element assignment the Channel Modes are resposible for audio input data channel mapping. The Channel Mapping of the audio data depends on the selected AACENC_CHANNELORDER which can be MPEG or WAV like order.
Following table describes the complete channel mapping for both Channel Order configurations.

ChannelMode	MPEG-Channelorder								WAV-Channelorder							
MODE_1	0								0							
MODE_2	0	1							0	1						
MODE_1_2	0	1	2						2	0	1					
MODE_1_2_1	0	1	2	3					2	0	1	3				
MODE_1_2_2	0	1	2	3	4				2	0	1	3	4			
MODE_1_2_2_1	0	1	2	3	4	5			2	0	1	4	5	3		
MODE_1_2_2_2_1	0	1	2	3	4	5	6	7	2	6	7	0	1	4	5	3
MODE_6_1	0	1	2	3	4	5	6		2	0	1	4	5	6	3	
MODE_7_1_BACK	0	1	2	3	4	5	6	7	2	0	1	6	7	4	5	3
MODE_7_1_TOP_FRONT	0	1	2	3	4	5	6	7	2	0	1	4	5	3	6	7
MODE_7_1_REAR_SURROUND	0	1	2	3	4	5	6	7	2	0	1	6	7	4	5	3
MODE_7_1_FRONT_CENTER	0	1	2	3	4	5	6	7	2	6	7	0	1	4	5	3

The denoted mapping is important for correct audio channel assignment when using MPEG or WAV ordering. The incoming audio channels are distributed MPEG like starting at the front channels and ending at the back channels. The distribution is used as described in Table concering Channel Config and fix channel elements. Please see the following example for clarification.

Example: MODE_1_2_2_1 - WAV-Channelorder 5.1

Input Channel	Coder Channel

```
 2 (front center)    | 0 (SCE channel)
 0 (left center)     | 1 (1st of 1st CPE)
 1 (right center)    | 2 (2nd of 1st CPE)
 4 (left surround)   | 3 (1st of 2nd CPE)
 5 (right surround)  | 4 (2nd of 2nd CPE)
 3 (LFE)             | 5 (LFE)
-----------------------------------------
```

2.11 Supported Bitrates

The FDK AAC Encoder provides a wide range of supported bitrates. The minimum and maximum allowed bitrate depends on the Audio Object Type. For AAC-LC the minimum bitrate is the bitrate that is required to write the most basic and minimal valid bitstream. It consists of the bitstream format header information and other static/mandatory information within the AAC payload. The maximum AAC framesize allowed by the MPEG-4 standard determines the maximum allowed bitrate for AAC-LC. For HE-AAC and HE-AAC v2 a library internal look-up table is used.

A good working point in terms of audio quality, sampling rate and bitrate, is at 1 to 1.5 bits/audio sample for AAC-LC, 0.625 bits/audio sample for dualrate HE-AAC, 1.125 bits/audio sample for downsampled HE-AAC and 0.5 bits/audio sample for HE-AAC v2. For example for one channel with a sampling frequency of 48 kHz, the range from 48 kbit/s to 72 kbit/s achieves reasonable audio quality for AAC-LC.

For HE-AAC and HE-AAC v2 the lowest possible audio input sampling frequency is 16 kHz because then the AAC-LC core encoder operates in dual rate mode at its lowest possible sampling frequency, which is 8 kHz. HE-AAC v2 requires stereo input audio data.

Please note that in HE-AAC or HE-AAC v2 mode the encoder supports much higher bitrates than are appropriate for HE-AAC or HE-AAC v2. For example, at a bitrate of more than 64 kbit/s for a stereo audio signal at 44.1 kHz it usually makes sense to use AAC-LC, which will produce better audio quality at that bitrate than HE-AAC or HE-AAC v2.

2.12 Recommended Sampling Rate and Bitrate Combinations

The following table provides an overview of recommended encoder configuration parameters which we determined by virtue of numerous listening tests.

2.12.1 AAC-LC, HE-AAC, HE-AACv2 in Dualrate SBR mode.

Audio Object Type	Bit Rate Range [bit/s]	Supported Sampling Rates [kHz]	Preferred Sampl. Rate [kHz]	No. of Chan.
AAC LC + SBR + PS	8000 - 11999	22.05, 24.00	24.00	2
AAC LC + SBR + PS	12000 - 17999	32.00	32.00	2
AAC LC + SBR + PS	18000 - 39999	32.00, 44.10, 48.00	44.10	2
AAC LC + SBR + PS	40000 - 64000	32.00, 44.10, 48.00	48.00	2
AAC LC + SBR	8000 - 11999	22.05, 24.00	24.00	1
AAC LC + SBR	12000 - 17999	32.00	32.00	1
AAC LC + SBR	18000 - 39999	32.00, 44.10, 48.00	44.10	1
AAC LC + SBR	40000 - 64000	32.00, 44.10, 48.00	48.00	1
AAC LC + SBR	16000 - 27999	32.00, 44.10, 48.00	32.00	2
AAC LC + SBR	28000 - 63999	32.00, 44.10, 48.00	44.10	2
AAC LC + SBR	64000 - 128000	32.00, 44.10, 48.00	48.00	2
AAC LC + SBR	64000 - 69999	32.00, 44.10, 48.00	32.00	5, 5.1

AAC LC + SBR	70000 - 239999	32.00, 44.10, 48.00	44.10	5, 5.1
AAC LC + SBR	240000 - 319999	32.00, 44.10, 48.00	48.00	5, 5.1
AAC LC	8000 - 15999	11.025, 12.00, 16.00	12.00	1
AAC LC	16000 - 23999	16.00	16.00	1
AAC LC	24000 - 31999	16.00, 22.05, 24.00	24.00	1
AAC LC	32000 - 55999	32.00	32.00	1
AAC LC	56000 - 160000	32.00, 44.10, 48.00	44.10	1
AAC LC	160001 - 288000	48.00	48.00	1
AAC LC	16000 - 23999	11.025, 12.00, 16.00	12.00	2
AAC LC	24000 - 31999	16.00	16.00	2
AAC LC	32000 - 39999	16.00, 22.05, 24.00	22.05	2
AAC LC	40000 - 95999	32.00	32.00	2
AAC LC	96000 - 111999	32.00, 44.10, 48.00	32.00	2
AAC LC	112000 - 320001	32.00, 44.10, 48.00	44.10	2
AAC LC	320002 - 576000	48.00	48.00	2
AAC LC	160000 - 239999	32.00	32.00	5, 5.1
AAC LC	240000 - 279999	32.00, 44.10, 48.00	32.00	5, 5.1
AAC LC	280000 - 800000	32.00, 44.10, 48.00	44.10	5, 5.1

2.12.2 AAC-LD, AAC-ELD, AAC-ELD with SBR in Dualrate SBR mode.

Unlike to HE-AAC configuration the SBR is not covered by ELD audio object type and needs to be enabled explicitly. Use AACENC_SBR_MODE to configure SBR and its samplingrate ratio with AACENC_SBR_RATIO parameter.

Audio Object Type	Bit Rate Range [bit/s]	Supported Sampling Rates [kHz]	Preferred Sampl. Rate [kHz]	No. of Chan.
ELD + SBR	18000 - 24999	32.00 - 44.10	32.00	1
ELD + SBR	25000 - 31999	32.00 - 48.00	32.00	1
ELD + SBR	32000 - 64000	32.00 - 48.00	48.00	1
ELD + SBR	32000 - 51999	32.00 - 48.00	44.10	2
ELD + SBR	52000 - 128000	32.00 - 48.00	48.00	2
ELD + SBR	78000 - 160000	32.00 - 48.00	48.00	3
ELD + SBR	104000 - 212000	32.00 - 48.00	48.00	4
ELD + SBR	130000 - 246000	32.00 - 48.00	48.00	5, 5.1
LD, ELD	16000 - 19999	16.00 - 24.00	16.00	1
LD, ELD	20000 - 39999	16.00 - 32.00	24.00	1
LD, ELD	40000 - 49999	22.05 - 32.00	32.00	1
LD, ELD	50000 - 61999	24.00 - 44.10	32.00	1
LD, ELD	62000 - 84999	32.00 - 48.00	44.10	1
LD, ELD	85000 - 192000	44.10 - 48.00	48.00	1
LD, ELD	64000 - 75999	24.00 - 32.00	32.00	2
LD, ELD	76000 - 97999	24.00 - 44.10	32.00	2
LD, ELD	98000 - 135999	32.00 - 48.00	44.10	2
LD, ELD	136000 - 384000	44.10 - 48.00	48.00	2
LD, ELD	96000 - 113999	24.00 - 32.00	32.00	3
LD, ELD	114000 - 146999	24.00 - 44.10	32.00	3
LD, ELD	147000 - 203999	32.00 - 48.00	44.10	3

LD, ELD	204000 - 576000	44.10 - 48.00	48.00	3
LD, ELD	128000 - 151999	24.00 - 32.00	32.00	4
LD, ELD	152000 - 195999	24.00 - 44.10	32.00	4
LD, ELD	196000 - 271999	32.00 - 48.00	44.10	4
LD, ELD	272000 - 768000	44.10 - 48.00	48.00	4
LD, ELD	160000 - 189999	24.00 - 32.00	32.00	5, 5.1
LD, ELD	190000 - 244999	24.00 - 44.10	32.00	5, 5.1
LD, ELD	245000 - 339999	32.00 - 48.00	44.10	5, 5.1
LD, ELD	340000 - 960000	44.10 - 48.00	48.00	5, 5.1

2.12.3 AAC-ELD with SBR in Downsampled SBR mode.

Audio Object Type	Bit Rate Range [bit/s]	Supported Sampling Rates [kHz]	Preferred Sampl. Rate [kHz]	No. of Chan.
ELD + SBR (downsampled SBR)	18000 - 24999	16.00 - 22.05	22.05	1
	25000 - 31999	16.00 - 24.00	24.00	1
	32000 - 47999	22.05 - 32.00	32.00	1
	48000 - 64000	22.05 - 48.00	32.00	1
ELD + SBR (downsampled SBR)	32000 - 51999	16.00 - 24.00	24.00	2
	52000 - 59999	22.05 - 24.00	24.00	2
	60000 - 95999	22.05 - 32.00	32.00	2
	96000 - 128000	22.05 - 48.00	32.00	2
ELD + SBR (downsampled SBR)	78000 - 99999	22.05 - 24.00	24.00	3
	100000 - 143999	22.05 - 32.00	32.00	3
	144000 - 159999	22.05 - 48.00	32.00	3
	160000 - 192000	32.00 - 48.00	32.00	3
ELD + SBR (downsampled SBR)	104000 - 149999	22.05 - 24.00	24.00	4
	150000 - 191999	22.05 - 32.00	32.00	4
	192000 - 211999	22.05 - 48.00	32.00	4
	212000 - 256000	32.00 - 48.00	32.00	4
ELD + SBR (downsampled SBR)	130000 - 171999	22.05 - 24.00	24.00	5, 5.1
	172000 - 239999	22.05 - 32.00	32.00	5, 5.1
	240000 - 320000	32.00 - 48.00	32.00	5, 5.1

2.12.4 AAC-ELD v2, AAC-ELD v2 with SBR.

The ELD v2 212 configuration must be configured explicitly with AACENC_CHANNELMODE parameter according MODE_212 value. SBR can be configured separately through AACENC_SBR_MODE and AACENC_SBR_RATIO parameter. Following configurations shall apply to both framelengths 480 and 512. For ELD v2 configuration without SBR and framelength 480 the supported sampling rate is restricted to the range from 16 kHz up to 24 kHz.

Audio Object Type	Bit Rate Range [bit/s]	Supported Sampling Rates [kHz]	Preferred Sampl. Rate	No. of Chan.

			[kHz]	
ELD-212 (without SBR)	16000 - 19999	16.00 - 24.00	16.00	2
	20000 - 39999	16.00 - 32.00	24.00	2
	40000 - 49999	22.05 - 32.00	32.00	2
	50000 - 61999	24.00 - 44.10	32.00	2
	62000 - 84999	32.00 - 48.00	44.10	2
	85000 - 192000	44.10 - 48.00	48.00	2
ELD-212 + SBR (dualrate SBR)	18000 - 20999	32.00	32.00	2
	21000 - 25999	32.00 - 44.10	32.00	2
	26000 - 31999	32.00 - 48.00	44.10	2
	32000 - 64000	32.00 - 48.00	48.00	2
ELD-212 + SBR (downsampled SBR)	18000 - 19999	16.00 - 22.05	22.05	2
	20000 - 24999	16.00 - 24.00	22.05	2
	25000 - 31999	16.00 - 24.00	24.00	2
	32000 - 64000	24.00 - 24.00	24.00	2

Chapter 3

Encoder Behaviour

3.1 Bandwidth

The FDK AAC encoder usually does not use the full frequency range of the input signal, but restricts the bandwidth according to certain library-internal settings. They can be changed in the table ”bandWidthTable” in the file bandwidth.cpp (if available).

The encoder API provides the AACENC_BANDWIDTH parameter to adjust the bandwidth explicitly.

```
aacEncoder_SetParam(hAacEncoder, AACENC_BANDWIDTH, value);
```

However it is not recommended to change these settings, because they are based on numerous listening tests and careful tweaks to ensure the best overall encoding quality. Also, the maximum bandwidth that can be set manually by the user is 20kHz or fs/2, whichever value is smaller.

Theoretically a signal of for example 48 kHz can contain frequencies up to 24 kHz, but to use this full range in an audio encoder usually does not make sense. Usually the encoder has a very limited amount of bits to spend (typically 128 kbit/s for stereo 48 kHz content) and to allow full range bandwidth would waste a lot of these bits for frequencies the human ear is hardly able to perceive anyway, if at all. Hence it is wise to use the available bits for the really important frequency range and just skip the rest. At lower bitrates (e. g. <= 80 kbit/s for stereo 48 kHz content) the encoder will choose an even smaller bandwidth, because an encoded signal with smaller bandwidth and hence less artifacts sounds better than a signal with higher bandwidth but then more coding artefacts across all frequencies. These artefacts would occur if small bitrates and high bandwidths are chosen because the available bits are just not enough to encode all frequencies well.

Unfortunately some people evaluate encoding quality based on possible bandwidth as well, but it is a double-edged sword considering the trade-off described above.

Another aspect is workload consumption. The higher the allowed bandwidth, the more frequency lines have to be processed, which in turn increases the workload.

3.2 Frame Sizes & Bit Reservoir

For AAC there is a difference between constant bit rate and constant frame length due to the so-called bit reservoir technique, which allows the encoder to use less bits in an AAC frame for those audio signal sections which are easy to encode, and then spend them at a later point in time for more complex audio sections. The extent to which this ”bit exchange” is done is limited to allow for reliable and relatively low delay real time streaming. Therefore, for AAC-ELD, the bitreservoir is limited. It varies between 500 and 4000 bits/frame, depending on the bitrate/channel.

- For a bitrate of 12kbps/channel and below, the AAC-ELD bitreservoir is 500 bits/frame.
- For a bitrate of 70kbps/channel and above, the AAC-ELD bitreservoir is 4000 bits/frame.
- Between 12kbps/channel and 70kbps/channel, the AAC-ELD bitrervoir is increased linearly.
- For AAC-LC, the bitrate is only limited by the maximum AAC frame length. It is, regardless of the available bit reservoir, defined as 6144 bits per channel.

Over a longer period in time the bitrate will be constant in the AAC constant bitrate mode, e.g. for ISDN transmission. This means that in AAC each bitstream frame will in general have a different length in bytes but over time it will reach the target bitrate.

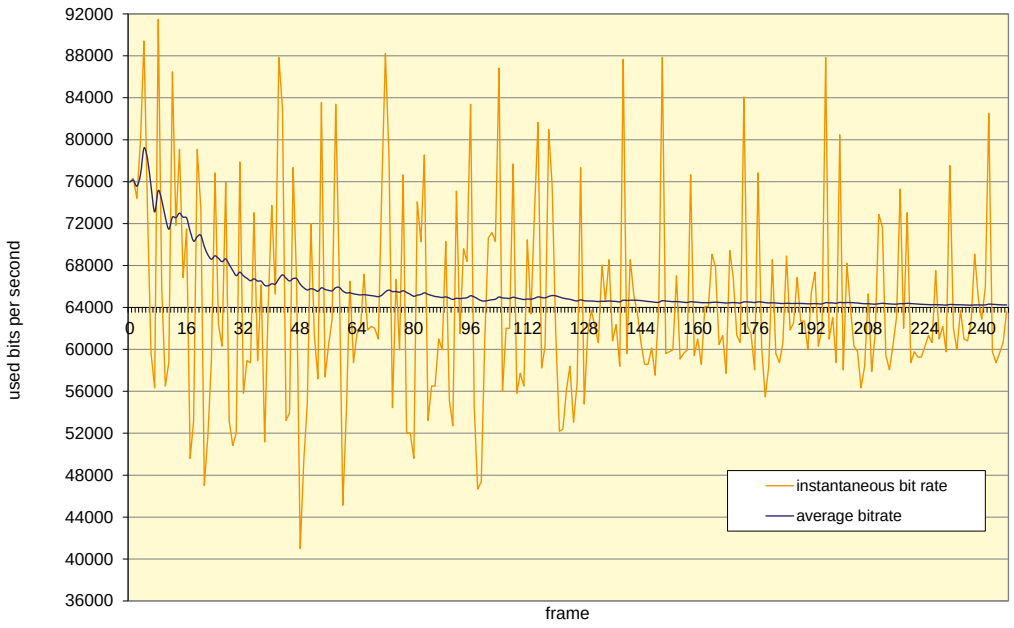


Figure 3.1: Average bitrate over time

The figure shows by means of a stereo file encoded with a constant bitrate of 64kbps, that the average bitrate converges quickly to the set bitrate. After 64 frames, the average bitrate already is 65.3 kbps. After 230 frames, the average bitrate is below 64.3 kbps.

One could also make an MPEG compliant AAC encoder which always produces constant length packages for each AAC frame, but the audio quality would be considerably worse since the bit reservoir technique would have to be switched off completely. A higher bit rate would have to be used to get the same audio quality as with an enabled bit reservoir.

For mp3 by the way, the same bit reservoir technique exists, but there each bit stream frame has a constant length for a given bit rate (ignoring the padding byte). In mp3 there is a so-called ”back pointer” which tells the decoder which bits belong to the current mp3 frame - and in general

some or many bits have been transmitted in an earlier mp3 frame. Basically this leads to the same ”bit exchange between mp3 frames” as in AAC but with virtually constant length frames.

This variable frame length at ”constant bit rate” is not something special in this Fraunhofer IIS AAC encoder. AAC has been designed in that way.

3.2.1 Estimating Average Frame Sizes

A HE-AAC v1 or v2 audio frame contains 2048 PCM samples per channel (there is also one mode with 1920 samples per channel but this is only for special purposes such as DAB+ digital radio).

The number of HE-AAC frames N_FRAMES per second at 44.1 kHz is:

$$N\_FRAMES = 44100/2048 = 21.5332$$

At a bit rate of 8 kbps the average number of bits per frame $N\_BITS\_PER\_FRAME$ is:

$$N\_BITS\_PER\_FRAME = 8000/21.5332 = 371.52$$

which is about 46.44 bytes per encoded frame.

At a bit rate of 32 kbps, which is quite high for single channel HE-AAC v1, it is:

$$N\_BITS\_PER\_FRAME = 32000/21.5332 = 1486$$

which is about 185.76 bytes per encoded frame.

These bits/frame figures are average figures where each AAC frame generally has a different size in bytes. To calculate the same for AAC-LC just use 1024 instead of 2048 PCM samples per frame and channel. For AAC-LD/ELD it is either 480 or 512 PCM samples per frame and channel.

3.3 Encoder Tools

The AAC encoder supports TNS, PNS, MS, Intensity and activates these tools depending on the audio signal and the encoder configuration (i.e. bitrate or AOT). It is not required to configure these tools manually.

PNS improves encoding quality only for certain bitrates. Therefore it makes sense to activate PNS only for these bitrates and save the processing power required for PNS (about 10 % of the encoder) when using other bitrates. This is done automatically inside the encoder library. PNS is disabled inside the encoder library if an MPEG-2 AOT is choosen since PNS is an MPEG-4 AAC feature.

If SBR is activated, the encoder automatically deactivates PNS internally. If TNS is disabled but PNS is allowed, the encoder deactivates PNS calculation internally.

Chapter 4

Class Index

4.1 Class List

Here are the classes, structs, unions and interfaces with brief descriptions:

AACENC_BufDesc 25
AACENC_InArgs 26
AACENC_InfoStruct 26
AACENC_MetaData 28
AACENC_OutArgs 31

Chapter 5

File Index

5.1 File List

Here is a list of all files with brief descriptions:

aacenc_lib.h
FDK AAC Encoder library interface header file 33

Chapter 6

Class Documentation

6.1 AACENC_BufDesc Struct Reference

`#include <aacenc_lib.h>`

Public Attributes

- INT numBufs
- void bufs
- INT bufferIdentifiers
- INT bufSizes
- INT bufElSizes

6.1.1 Detailed Description

Describes the input and output buffers for an aacEncEncode() call.

6.1.2 Member Data Documentation

numBufs

`INT AACENC_BufDesc::numBufs`

Number of buffers.

bufs

`void** AACENC_BufDesc::bufs`

Pointer to vector containing buffer addresses.

bufferIdentifiers

`INT* AACENC_BufDesc::bufferIdentifiers`

Identifier of each buffer element. See AACENC_BufferIdentifier.

bufSizes

`INT* AACENC_BufDesc::bufSizes`

Size of each buffer in 8-bit bytes.

bufElSizes

`INT* AACENC_BufDesc::bufElSizes`

Size of each buffer element in bytes.

The documentation for this struct was generated from the following file:

- aacenc_lib.h

6.2 AACENC_InArgs Struct Reference

`#include <aacenc_lib.h>`

Public Attributes

- INT numInSamples
- INT numAncBytes

6.2.1 Detailed Description

Defines the input arguments for an aacEncEncode() call.

6.2.2 Member Data Documentation

numInSamples

`INT AACENC_InArgs::numInSamples`

Number of valid input audio samples (multiple of input channels).

numAncBytes

`INT AACENC_InArgs::numAncBytes`

Number of ancillary data bytes to be encoded.

The documentation for this struct was generated from the following file:

- aacenc_lib.h

6.3 AACENC_InfoStruct Struct Reference

`#include <aacenc_lib.h>`

Public Attributes

- UINT maxOutBufBytes
- UINT maxAncBytes
- UINT inBufFillLevel
- UINT inputChannels
- UINT frameLength
- UINT nDelay
- UINT nDelayCore
- UCHAR confBuf [64]
- UINT confSize

6.3.1 Detailed Description

Provides some info about the encoder configuration.

6.3.2 Member Data Documentation

maxOutBufBytes

`UINT AACENC_InfoStruct::maxOutBufBytes`

Maximum number of encoder bitstream bytes within one frame. Size depends on maximum number of supported channels in encoder instance. For superframing (as used for example in DAB+), size has to be a multiple accordingly.

maxAncBytes

`UINT AACENC_InfoStruct::maxAncBytes`

Maximum number of ancillary data bytes which can be inserted into bitstream within one frame.

inBufFillLevel

`UINT AACENC_InfoStruct::inBufFillLevel`

Internal input buffer fill level in samples per channel. This parameter will automatically be cleared if samplingrate or channel(Mode/Order) changes.

inputChannels

`UINT AACENC_InfoStruct::inputChannels`

Number of input channels expected in encoding process.

frameLength

`UINT AACENC_InfoStruct::frameLength`

Amount of input audio samples consumed each frame per channel, depending on audio object type configuration.

nDelay

`UINT AACENC_InfoStruct::nDelay`

Codec delay in PCM samples/channel. Depends on framelength and AOT. Does not include framing delay for filling up encoder PCM input buffer.

nDelayCore

`UINT AACENC_InfoStruct::nDelayCore`

Codec delay in PCM samples/channel, w/o delay caused by the decoder SBR module. This delay is needed to correctly write edit lists for gapless playback. The decoder may not know how much delay is introdcued by SBR, since it may not know if SBR is active at all (implicit signaling), therefore the deocder must take into account any delay caused by the SBR module.

confBuf

`UCHAR AACENC_InfoStruct::confBuf[64]`

Configuration buffer in binary format as an AudioSpecificConfig or StreamMuxConfig according to the selected transport type.

confSize

`UINT AACENC_InfoStruct::confSize`

Number of valid bytes in confBuf.

The documentation for this struct was generated from the following file:

- aacenc_lib.h

6.4 AACENC_MetaData Struct Reference

`#include <aacenc_lib.h>`

Public Attributes

- AACENC_METADATA_DRC_PROFILE drc_profile
- AACENC_METADATA_DRC_PROFILE comp_profile
- INT drc_TargetRefLevel
- INT comp_TargetRefLevel
- INT prog_ref_level_present
- INT prog_ref_level
- UCHAR PCE_mixdown_idx_present
- UCHAR ETSI_DmxLvl_present
- SCHAR centerMixLevel
- SCHAR surroundMixLevel
- UCHAR dolbySurroundMode
- UCHAR drcPresentationMode
- struct *f*
  - UCHAR extAncDataEnable
  - UCHAR extDownmixLevelEnable
  - UCHAR extDownmixLevel_A
  - UCHAR extDownmixLevel_B
  - UCHAR dmxGainEnable

INT dmxGain5
INT dmxGain2
UCHAR lfeDmxEnable
UCHAR lfeDmxLevel
g ExtMetaData

6.4.1 Detailed Description

Meta Data setup structure.

6.4.2 Member Data Documentation

drc_profile

AACENC_METADATA_DRC_PROFILE AACENC_MetaData::drc_profile
MPEG DRC compression profile. See AACENC_METADATA_DRC_PROFILE.

comp_profile

AACENC_METADATA_DRC_PROFILE AACENC_MetaData::comp_profile
ETSI heavy compression profile. See AACENC_METADATA_DRC_PROFILE.

drc_TargetRefLevel

INT AACENC_MetaData::drc_TargetRefLevel
Used to define expected level to: Scaled with 16 bit. x 2^16.

comp_TargetRefLevel

INT AACENC_MetaData::comp_TargetRefLevel
Adjust limiter to avoid overload. Scaled with 16 bit. x 2^16.

prog_ref_level_present

INT AACENC_MetaData::prog_ref_level_present
Flag, if prog_ref_level is present

prog_ref_level

INT AACENC_MetaData::prog_ref_level
Programme Reference Level = Dialogue Level: -31.75dB .. 0 dB ; stepsize: 0.25dB Scaled with 16 bit. x 2^16.

PCE_mixdown_idx_present

UCHAR AACENC_MetaData::PCE_mixdown_idx_present
Flag, if dmx-idx should be written in programme config element

ETSI_DmxLvl_present

```
UCHAR AACENC_MetaData::ETSI_DmxLvl_present
```

Flag, if dmx-lvl should be written in ETSI-ancData

centerMixLevel

```
SCHAR AACENC_MetaData::centerMixLevel
```

Center downmix level (0...7, according to table)

surroundMixLevel

```
SCHAR AACENC_MetaData::surroundMixLevel
```

Surround downmix level (0...7, according to table)

dolbySurroundMode

```
UCHAR AACENC_MetaData::dolbySurroundMode
```

Indication for Dolby Surround Encoding Mode.

- 0: Dolby Surround mode not indicated
- 1: 2-ch audio part is not Dolby surround encoded
- 2: 2-ch audio part is Dolby surround encoded

drcPresentationMode

```
UCHAR AACENC_MetaData::drcPresentationMode
```

Indicatin for DRC Presentation Mode.

- 0: Presentation mode not inticated
- 1: Presentation mode 1
- 2: Presentation mode 2

extAncDataEnable

```
UCHAR AACENC_MetaData::extAncDataEnable
```

extDownmixLevelEnable

```
UCHAR AACENC_MetaData::extDownmixLevelEnable
```

extDownmixLevel_A

```
UCHAR AACENC_MetaData::extDownmixLevel_A
```

extDownmixLevel_B

```
UCHAR AACENC_MetaData::extDownmixLevel_B
```

dmxGainEnable

```
UCHAR AACENC_MetaData::dmxGainEnable
```

dmxGain5

```
INT AACENC_MetaData::dmxGain5
```

dmxGain2

```
INT AACENC_MetaData::dmxGain2
```

lfeDmxEnable

```
UCHAR AACENC_MetaData::lfeDmxEnable
```

lfeDmxLevel

```
UCHAR AACENC_MetaData::lfeDmxLevel
```

ExtMetaData

```
struct { ... } AACENC_MetaData::ExtMetaData
```

The documentation for this struct was generated from the following file:

- aacenc_lib.h

6.5 AACENC_OutArgs Struct Reference

```
#include <aacenc_lib.h>
```

Public Attributes

- INT numOutBytes
- INT numInSamples
- INT numAncBytes
- INT bitResState

6.5.1 Detailed Description

Defines the output arguments for an aacEncEncode() call.

6.5.2 Member Data Documentation

numOutBytes

`INT AACENC_OutArgs::numOutBytes`

Number of valid bitstream bytes generated during aacEncEncode().

numInSamples

`INT AACENC_OutArgs::numInSamples`

Number of input audio samples consumed by the encoder.

numAncBytes

`INT AACENC_OutArgs::numAncBytes`

Number of ancillary data bytes consumed by the encoder.

bitResState

`INT AACENC_OutArgs::bitResState`

State of the bit reservoir in bits.

The documentation for this struct was generated from the following file:

- aacenc_lib.h

Chapter 7

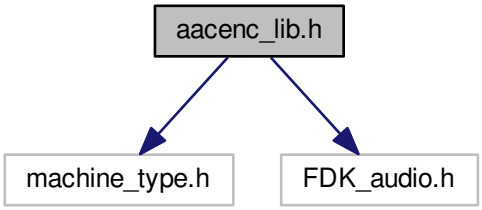
File Documentation

7.1 aacenc_lib.h File Reference

FDK AAC Encoder library interface header file.

```
   #include "machine_type.h"
#include "FDK_audio.h"
```

Include dependency graph for aacenc_lib.h:



Classes

- struct AACENC_InfoStruct
- struct AACENC_BufDesc
- struct AACENC_InArgs
- struct AACENC_OutArgs
- struct AACENC_MetaData

Typedefs

- typedef struct AACENCODER HANDLE_AACENCODER

Enumerations

- enum AACENC_ERROR *f*
  AACENC_OK = 0x0000,

AACENC_INVALID_HANDLE = 0x0020,
AACENC_MEMORY_ERROR = 0x0021,
AACENC_UNSUPPORTED_PARAMETER = 0x0022,
AACENC_INVALID_CONFIG = 0x0023,
AACENC_INIT_ERROR = 0x0040,
AACENC_INIT_AAC_ERROR = 0x0041,
AACENC_INIT_SBR_ERROR = 0x0042,
AACENC_INIT_TP_ERROR = 0x0043,
AACENC_INIT_META_ERROR = 0x0044,
AACENC_INIT_MPS_ERROR = 0x0045,
AACENC_ENCODE_ERROR = 0x0060,
AACENC_ENCODE_EOF = 0x0080 *g*

- enum AACENC_BufferIdentifier *f*
  IN_AUDIO_DATA = 0,
  IN_ANCILLRY_DATA = 1,
  IN_METADATA_SETUP = 2,
  OUT_BITSTREAM_DATA = 3,
  OUT_AU_SIZES = 4 *g*
- enum AACENC_METADATA_DRC_PROFILE *f*
  AACENC_METADATA_DRC_NONE = 0,
  AACENC_METADATA_DRC_FILMSTANDARD = 1,
  AACENC_METADATA_DRC_FILMLIGHT = 2,
  AACENC_METADATA_DRC_MUSICSTANDARD = 3,
  AACENC_METADATA_DRC_MUSICLIGHT = 4,
  AACENC_METADATA_DRC_SPEECH = 5,
  AACENC_METADATA_DRC_NOT_PRESENT = 256 *g*
- enum AACENC_CTRLFLAGS *f*
  AACENC_INIT_NONE = 0x0000,
  AACENC_INIT_CONFIG = 0x0001,
  AACENC_INIT_STATES = 0x0002,
  AACENC_INIT_TRANSPORT = 0x1000,
  AACENC_RESET_INBUFFER = 0x2000,
  AACENC_INIT_ALL = 0xFFFF *g*
- enum AACENC_PARAM *f*
  AACENC_AOT = 0x0100,
  AACENC_BITRATE = 0x0101,
  AACENC_BITRATEMODE = 0x0102,
  AACENC_SAMPLERATE = 0x0103,
  AACENC_SBR_MODE = 0x0104,
  AACENC_GRANULE_LENGTH = 0x0105,
  AACENC_CHANNELMODE = 0x0106,
  AACENC_CHANNELORDER = 0x0107,
  AACENC_SBR_RATIO = 0x0108,
  AACENC_AFTERBURNER = 0x0200,
  AACENC_BANDWIDTH = 0x0203,
  AACENC_PEAK_BITRATE = 0x0207,
  AACENC_TRANSMUX = 0x0300,
  AACENC_HEADER_PERIOD = 0x0301,
  AACENC_SIGNALING_MODE = 0x0302,
  AACENC_TPSUBFRAMES = 0x0303,
  AACENC_AUDIOMUXVER = 0x0304,
  AACENC_PROTECTION = 0x0306,
  AACENC_ANCILLARY_BITRATE = 0x0500,
  AACENC_METADATA_MODE = 0x0600,

AACENC_CONTROL_STATE = 0xFF00,
AACENC_NONE = 0xFFFF *g*

AAC encoder setting parameters.

Functions

- AACENC_ERROR aacEncOpen (HANDLE_AACENCODER phAacEncoder, const UINT encModules, const UINT maxChannels)

  Open an instance of the encoder.
- AACENC_ERROR aacEncClose (HANDLE_AACENCODER phAacEncoder)

  Close the encoder instance.
- AACENC_ERROR aacEncEncode (const HANDLE_AACENCODER hAacEncoder, const AACENC_BufDesc inBufDesc, const AACENC_BufDesc outBufDesc, const AACENC_InArgs inargs, AACENC_OutArgs outargs)

  Encode audio data.
- AACENC_ERROR aacEncInfo (const HANDLE_AACENCODER hAacEncoder, AACENC_InfoStruct pInfo)

  Acquire info about present encoder instance.
- AACENC_ERROR aacEncoder_SetParam (const HANDLE_AACENCODER hAacEncoder, const AACENC_PARAM param, const UINT value)

  Set one single AAC encoder parameter.
- UINT aacEncoder_GetParam (const HANDLE_AACENCODER hAacEncoder, const AACENC_PARAM param)

  Get one single AAC encoder parameter.
- AACENC_ERROR aacEncGetLibInfo (LIB_INFO info)

  Get information about encoder library build.

7.1.1 Detailed Description

FDK AAC Encoder library interface header file.

7.1.2 Typedef Documentation

HANDLE_AACENCODER

```
typedef struct AACENCODER* HANDLE_AACENCODER
```

AAC encoder handle.

7.1.3 Enumeration Type Documentation

AACENC_ERROR

```
enum AACENC_ERROR
```

AAC encoder error codes.

Enumerator

AACENC_OK	No error happened. All ne.
AACENC_INVALID_HANDLE	Handle passed to function call was invalid.
AACENC_MEMORY_ERROR	Memory allocation failed.
AACENC_UNSUPPORTED_PARAMETER	Parameter not available.
AACENC_INVALID_CONFIG	Con guration not provided.
AACENC_INIT_ERROR	General initialization error.
AACENC_INIT_AAC_ERROR	AAC library initialization error.
AACENC_INIT_SBR_ERROR	SBR library initialization error.
AACENC_INIT_TP_ERROR	Transport library initialization error.
AACENC_INIT_META_ERROR	Meta data library initialization error.
AACENC_INIT_MPS_ERROR	MPS library initialization error.
AACENC_ENCODE_ERROR	The encoding process was interrupted by an unexpected error.
AACENC_ENCODE_EOF	End of le reached.

AACENC_BufferIdenti er

enum AACENC_BufferIdentifier

AAC encoder buffer descriptors identi er. This identi er are used within buffer descriptors AACENC_BufDesc::bufferIdenti ers.

Enumerator

IN_AUDIO_DATA	Audio input buffer, interleaved INT_PCM samples.
IN_ANCILLRY_DATA	Ancillary data to be embedded into bitstream.
IN_METADATA_SETUP	Setup structure for embedding meta data.
OUT_BITSTREAM_DATA	Buffer holds bitstream output data.
OUT_AU_SIZES	Buffer contains sizes of each access unit. This information is necessary for superframing.

AACENC_METADATA_DRC_PROFILE

enum AACENC_METADATA_DRC_PROFILE

Meta Data Compression Pro les.

Enumerator

AACENC_METADATA_DRC_NONE	None.
AACENC_METADATA_DRC_FILMSTANDARD	Film standard.
AACENC_METADATA_DRC_FILMLIGHT	Film light.

A[redacted]

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BAND[redacted]

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AND[redacted]

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