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

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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.

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 as follows.

```
 if (outargs.numOutBytes>0) { FDKfwrite(outputBuffer,
outargs.numOutBytes, 1, pOutFile);
}
```

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 preinitizializes 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 Peak Bitrate 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.

$$bitreservoir = nEffChannels * 6144 - (bitrate * framelength/samplerate)$$

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 limit the maximum bits per frame size. This can be done with the AACENC_PEAK_BITRATE parameter.

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

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:

$$1000 * fs/(dsf * sr)$$

```
--------------------------------------
      | 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   |            Supported  | Preferred  | No.
of |         [bit/s]  |       Sampling Rates  |     Sampl.  |  Chan. |
|                 [kHz]  |      Rate  | |                   |
|     [kHz]  |
-------------------+-------------------+-----------------------+------------+-------
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  |            Supported  | Preferred  | No.
of |             [bit/s]  |        Sampling Rates  |     Sampl.  |  Chan. |
|                 [kHz]  |       Rate  | |                 |
|     [kHz]  |
-------------------+------------------+-----------------------+------------+-------
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  |          Supported  | Preferred  | No.
of |          [bit/s]  |       Sampling Rates  |     Sampl.  |  Chan. |
|                  [kHz]  |       Rate  | |                   |
|     [kHz]  |
-------------------+------------------+-----------------------+------------+-------
ELD + SBR          |  18000 - 24999   |        16.00 - 22.05  |     22.05  | 1
(downsampled SBR)  |  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          |  32000 - 51999   |        16.00 - 24.00  |     24.00  | 2
(downsampled SBR)  |  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          |  78000 -  99999  |        22.05 - 24.00  |     24.00  | 3
(downsampled SBR)  | 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          | 104000 - 149999  |        22.05 - 24.00  |     24.00  | 4
(downsampled SBR)  | 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          | 130000 - 171999  |        22.05 - 24.00  |     24.00  |
5, 5.1 (downsampled SBR)  | 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  |          Supported  | Preferred  | No.
of |          [bit/s]  |       Sampling Rates  |     Sampl.  |  Chan. |
|                  [kHz]  |       Rate  | |                   |
|     [kHz]  |
-------------------+------------------+-----------------------+------------+-------
ELD-212            |  16000 -  19999  |        16.00 - 24.00  |     16.00  | 2
(without SBR)      |  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     |  18000 -  20999  |                32.00  |     32.00  | 2
(dualrate SBR)    |  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     |  18000 -  19999  |        16.00 - 22.05  |     22.05  | 2
(downsampled SBR) |  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.

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.

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:

Chapter 5

File Index

5.1 File List

Here is a list of all files with brief descriptions:

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.

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 {
  UCHAR extAncDataEnable
  UCHAR extDownmixLevelEnable
  UCHAR extDownmixLevel_A
  UCHAR extDownmixLevel_B
  UCHAR dmxGainEnable

INT dmxGain5
INT dmxGain2
UCHAR lfeDmxEnable
UCHAR lfeDmxLevel
} 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

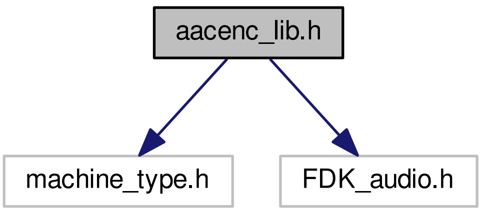
File Documentation

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 {
  AACENC_OK = 0x0000,

AACENC_INVALID_HANDLE,
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,
AACENC_INIT_MPS_ERROR = 0x0045,
AACENC_ENCODE_ERROR = 0x0060,
AACENC_ENCODE_EOF = 0x0080 }

- enum AACENC_BufferIdentifier {
IN_AUDIO_DATA = 0,
IN_ANCILLRY_DATA = 1,
IN_METADATA_SETUP = 2,
OUT_BITSTREAM_DATA = 3,
OUT_AU_SIZES }
- enum AACENC_METADATA_DRC_PROFILE {
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 }
- enum AACENC_CTRLFLAGS {
AACENC_INIT_NONE = 0x0000,
AACENC_INIT_CONFIG,
AACENC_INIT_STATES = 0x0002,
AACENC_INIT_TRANSPORT,
AACENC_RESET_INBUFFER,
AACENC_INIT_ALL = 0xFFFF }
- enum AACENC_PARAM {
AACENC_AOT,
AACENC_BITRATE = 0x0101,
AACENC_BITRATEMODE = 0x0102,
AACENC_SAMPLERATE = 0x0103,
AACENC_SBR_MODE = 0x0104,
AACENC_GRANULE_LENGTH,
AACENC_CHANNELMODE = 0x0106,
AACENC_CHANNELORDER,
AACENC_SBR_RATIO,
AACENC_AFTERBURNER,
AACENC_BANDWIDTH = 0x0203,
AACENC_PEAK_BITRATE,
AACENC_TRANSMUX = 0x0300,
AACENC_HEADER_PERIOD,
AACENC_SIGNALING_MODE,
AACENC_TPSUBFRAMES,
AACENC_AUDIOMUXVER,
AACENC_PROTECTION = 0x0306,
AACENC_ANCILLARY_BITRATE,
AACENC_METADATA_MODE = 0x0600,

AACENC_CONTROL_STATE,
AACENC_NONE = 0xFFFF }

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 fine.
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	Configuration 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 file reached.

AACENC_BufferIdentifier

`enum AACENC_BufferIdentifier`

AAC encoder buffer descriptors identifier. This identifier are used within buffer descriptors AACENC_BufDesc::bufferIdentifiers.

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 Profiles.

Enumerator

AACENC_METADATA_DRC_NONE	None.
AA-CENC_METADATA_DRC_FILMSTANDARD	Film standard.
AACENC_METADATA_DRC_FILMLIGHT	Film light.

Enumerator

AACENC_METADATA_DRC_MUSICSTANDARD	Music standard.
AACENC_METADATA_DRC_MUSICLIGHT	Music light.
AACENC_METADATA_DRC_SPEECH	Speech.
AACENC_METADATA_DRC_NOT_PRESENT	Disable writing gain factor (used for comp_profile only).

AACENC_CTRLFLAGS

`enum AACENC_CTRLFLAGS`

AAC encoder control flags.

In interaction with the AACENC_CONTROL_STATE parameter it is possible to get information about the internal initialization process. It is also possible to overwrite the internal state from extern when necessary.

Enumerator

AACENC_INIT_NONE	Do not trigger initialization.
AACENC_INIT_CONFIG	Initialize all encoder modules configuration.
AACENC_INIT_STATES	Reset all encoder modules history buffer.
AACENC_INIT_TRANSPORT	Initialize transport lib with new parameters.
AACENC_RESET_INBUFFER	Reset fill level of internal input buffer.
AACENC_INIT_ALL	Initialize all.

AACENC_PARAM

`enum AACENC_PARAM`

AAC encoder setting parameters.

Use aacEncoder_SetParam() function to configure, or use aacEncoder_GetParam() function to read the internal status of the following parameters.

Enumerator

AACENC_AOT	Audio object type. See ::AUDIO_OBJECT_TYPE in FDK_audio.h. • 2: MPEG-4 AAC Low Complexity. • 5: MPEG-4 AAC Low Complexity with Spectral Band Replication (HE-AAC). • 29: MPEG-4 AAC Low Complexity with Spectral Band Replication and Parametric Stereo (HE-AAC v2). This configuration can be used only with stereo input audio data. • 23: MPEG-4 AAC Low-Delay. • 39: MPEG-4 AAC Enhanced Low-Delay. Since there is no ::AUDIO_OBJECT_TYPE for ELD in combination with SBR defined, enable SBR explicitely by AACENC_SBR_MODE parameter. The ELD v2 212 configuration can be configured by AACENC_CHANNELMODE parameter. • 129: MPEG-2 AAC Low Complexity. • 132: MPEG-2 AAC Low Complexity with Spectral Band Replication (HE-AAC). Please note that the virtual MPEG-2 AOT's basically disables non-existing Perceptual Noise Substitution tool in AAC encoder and controls the MPEG_ID flag in adts header. The virtual MPEG-2 AOT doesn't prohibit specific transport formats.
AACENC_BITRATE	Total encoder bitrate. This parameter is mandatory and interacts with AACENC_BITRATEMODE. • CBR: Bitrate in bits/second. • VBR: Variable bitrate. Bitrate argument will be ignored. See Supported Bitrates for details.

Enumerator

AACENC_BITRATEMODE	Bitrate mode. Configuration can be different kind of bitrate configurations: • 0: Constant bitrate, use bitrate according to AACENC_BITRATE. (default) Within none LD/ELD ::AUDIO_OBJECT_TYPE, the CBR mode makes use of full allowed bitreservoir. In contrast, at Low-Delay ::AUDIO_OBJECT_TYPE the bitreservoir is kept very small. • 1: Variable bitrate mode, very low bitrate. • 2: Variable bitrate mode, low bitrate. • 3: Variable bitrate mode, medium bitrate. • 4: Variable bitrate mode, high bitrate. • 5: Variable bitrate mode, very high bitrate.
AACENC_SAMPLERATE	Audio input data sampling rate. Encoder supports following sampling rates: 8000, 11025, 12000, 16000, 22050, 24000, 32000, 44100, 48000, 64000, 88200, 96000
AACENC_SBR_MODE	Configure SBR independently of the chosen Audio Object Type ::AUDIO_OBJECT_TYPE. This parameter is for ELD audio object type only. • -1: Use ELD SBR auto configurator (default). • 0: Disable Spectral Band Replication. • 1: Enable Spectral Band Replication.
AACENC_GRANULE_LENGTH	Core encoder (AAC) audio frame length in samples: • 1024: Default configuration. • 512: Default length in LD/ELD configuration. • 480: Length in LD/ELD configuration. • 256: Length for ELD reduced delay mode (x2). • 240: Length for ELD reduced delay mode (x2). • 128: Length for ELD reduced delay mode (x4). • 120: Length for ELD reduced delay mode (x4).
AACENC_CHANNELMODE	Set explicit channel mode. Channel mode must match with number of input channels. • 1-7, 11,12,14 and 33,34: MPEG channel modes supported, see ::CHANNEL_MODE in FDK_audio.h.

Enumerator

AACENC_CHANNELORDER	Input audio data channel ordering scheme: • 0: MPEG channel ordering (e. g. 5.1: C, L, R, SL, SR, LFE). (default) • 1: WAVE file format channel ordering (e. g. 5.1: L, R, C, LFE, SL, SR).
AACENC_SBR_RATIO	Controls activation of downsampled SBR. With downsampled SBR, the delay will be shorter. On the other hand, for achieving the same quality level, downsampled SBR needs more bits than dual-rate SBR. With downsampled SBR, the AAC encoder will work at the same sampling rate as the SBR encoder (single rate). Downsampled SBR is supported for AAC-ELD and HE-AACv1. • 1: Downsampled SBR (default for ELD). • 2: Dual-rate SBR (default for HE-AAC).
AACENC_AFTERBURNER	This parameter controls the use of the afterburner feature. The afterburner is a type of analysis by synthesis algorithm which increases the audio quality but also the required processing power. It is recommended to always activate this if additional memory consumption and processing power consumption is not a problem. If increased MHz and memory consumption are an issue then the MHz and memory cost of this optional module need to be evaluated against the improvement in audio quality on a case by case basis. • 0: Disable afterburner (default). • 1: Enable afterburner.
AACENC_BANDWIDTH	Core encoder audio bandwidth: • 0: Determine audio bandwidth internally (default, see chapter Bandwidth). • 1 to fs/2: Audio bandwidth in Hertz. Limited to 20kHz max. Not usable if SBR is active. This setting is for experts only, better do not touch this value to avoid degraded audio quality.

Enumerator

AACENC_PEAK_BITRATE	Peak bitrate configuration parameter to adjust maximum bits per audio frame. Bitrate is in bits/second. The peak bitrate will internally be limited to the chosen bitrate AACENC_BITRATE as lower limit and the number_of_effective_channels∗6144 bit as upper limit. Setting the peak bitrate equal to AACENC_BITRATE does not necessarily mean that the audio frames will be of constant size. Since the peak bitate is in bits/second, the frame sizes can vary by one byte in one or the other direction over various frames. However, it is not recommended to reduce the peak pitrate to AACENC_BITRATE - it would disable the bitreservoir, which would affect the audio quality by a large amount.
AACENC_TRANSMUX	Transport type to be used. See ::TRANSPORT_TYPE in FDK_audio.h. Following types can be configured in encoder library: • 0: raw access units • 1: ADIF bitstream format • 2: ADTS bitstream format • 6: Audio Mux Elements (LATM) with muxConfigPresent = 1 • 7: Audio Mux Elements (LATM) with muxConfigPresent = 0, out of band StreamMuxConfig • 10: Audio Sync Stream (LOAS)
AACENC_HEADER_PERIOD	Frame count period for sending in-band configuration buffers within LATM/LOAS transport layer. Additionally this parameter configures the PCE repetition period in raw_data_block(). See encPCE. • 0xFF: auto-mode default 10 for TT_MP4_ADTS, TT_MP4_LOAS and TT_MP4_LATM_MCP1, otherwise 0. • n: Frame count period.

Enumerator

<table>
<tr><td>AACENC_SIGNALING_MODE</td><td>Signaling mode of the extension AOT:

0: Implicit backward compatible signaling (default for non-MPEG-4 based AOT's and for the transport formats ADIF and ADTS)

A stream that uses implicit signaling can be decoded by every AAC decoder, even AAC-LC-only decoders
An AAC-LC-only decoder will only decode the low-frequency part of the stream, resulting in a band-limited output
This method works with all transport formats
This method does not work with downsampled SBR

1: Explicit backward compatible signaling

A stream that uses explicit backward compatible signaling can be decoded by every AAC decoder, even AAC-LC-only decoders
An AAC-LC-only decoder will only decode the low-frequency part of the stream, resulting in a band-limited output
A decoder not capable of decoding PS will only decode the AAC-LC+SBR part. If the stream contained PS, the result will be a a decoded mono downmix
This method does not work with ADIF or ADTS. For LOAS/LATM, it only works with AudioMuxVersion==1
This method does work with downsampled SBR

2: Explicit hierarchical signaling (default for MPEG-4 based AOT's and for all transport formats excluding ADIF and ADTS)

A stream that uses explicit hierarchical signaling can be decoded only by HE-AAC decoders
An AAC-LC-only decoder will not decode a stream that uses explicit hierarchical signaling
A decoder not capable of decoding PS will not decode the stream at all if it contained PS
This method does not work with ADIF or ADTS. It works with LOAS/LATM and the MPEG-4 File format
This method does work with downsampled SBR

For making sure that the listener always experiences the best audio quality, explicit hierarchical signaling should be used. This makes sure that only a full HE-AAC-capable decoder will decode those streams. The audio is played at full bandwidth. For best backwards compatibility, it is recommended to encode with implicit SBR signaling. A decoder capable of AAC-LC only will</td></tr>
<tr><td></td><td></td></tr>
</table>

Enumerator

AACENC_TPSUBFRAMES	Number of sub frames in a transport frame for LOAS/LATM or ADTS (default 1). • ADTS: Maximum number of sub frames restricted to 4. • LOAS/LATM: Maximum number of sub frames restricted to 2.
AACENC_AUDIOMUXVER	AudioMuxVersion to be used for LATM. (AudioMuxVersionA, currently not implemented): • 0: Default, no transmission of tara Buffer fullness, no ASC length and including actual latm Buffer fullnes. • 1: Transmission of tara Buffer fullness, ASC length and actual latm Buffer fullness. • 2: Transmission of tara Buffer fullness, ASC length and maximum level of latm Buffer fullness.
AACENC_PROTECTION	Configure protection in transport layer: • 0: No protection. (default) • 1: CRC active for ADTS transport format.
AACENC_ANCILLARY_BITRATE	Constant ancillary data bitrate in bits/second. • 0: Either no ancillary data or insert exact number of bytes, denoted via input parameter, numAncBytes in AACENC_InArgs. • else: Insert ancillary data with specified bitrate.
AACENC_METADATA_MODE	Configure Meta Data. See AACENC_MetaData for further details: • 0: Do not embed any metadata. • 1: Embed dynamic_range_info metadata. • 2: Embed dynamic_range_info and ancillary_data metadata. • 3: Embed ancillary_data metadata.
AACENC_CONTROL_STATE	There is an automatic process which internally reconfigures the encoder instance when a configuration parameter changed or an error occured. This paramerter allows overwriting or getting the control status of this process. See AACENC_CTRLFLAGS.
AACENC_NONE	

7.1.4 Function Documentation

aacEncOpen()

```
AACENC_ERROR aacEncOpen (
            HANDLE_AACENCODER * phAacEncoder,
            const UINT encModules,
            const UINT maxChannels )
```

Open an instance of the encoder.

Allocate memory for an encoder instance with a functional range denoted by the function parameters. Preinitialize encoder instance with default configuration.

Parameters

<table>
<tr><td>phAacEncoder</td><td>A pointer to an encoder handle. Initialized on return.</td></tr>
<tr><td>encModules</td><td>Specify encoder modules to be supported in this encoder instance: 0x0: Allocate memory for all available encoder modules. else: Select memory allocation regarding encoder modules. Following flags are possible and can be combined. 0x01: AAC module. 0x02: SBR module. 0x04: PS module. 0x08: MPS module. 0x10: Metadata module. example: (0x01|0x02|0x04|0x08|0x10) allocates all modules and is equivalent to default configuration denotet by 0x0.</td></tr>
<tr><td>maxChannels</td><td>Number of channels to be allocated. This parameter can be used in different ways: 0: Allocate maximum number of AAC and SBR channels as supported by the library. nChannels: Use same maximum number of channels for allocating memory in AAC and SBR module. nChannels | (nSbrCh<<8): Number of SBR channels can be different to AAC channels to save data memory.</td></tr>
</table>

Returns

- AACENC_OK, on succes.
- AACENC_INVALID_HANDLE, AACENC_MEMORY_ERROR, AACENC_INVALID_CONFIG, on failure.

aacEncClose()

```
AACENC_ERROR aacEncClose (
            HANDLE_AACENCODER * phAacEncoder )
```

Close the encoder instance.

Deallocate encoder instance and free whole memory.

Parameters

phAacEncoder	Pointer to the encoder handle to be deallocated.

Returns

- AACENC_OK, on success.
- AACENC_INVALID_HANDLE, on failure.

aacEncEncode()

```
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.

This function is mainly for encoding audio data. In addition the function can be used for an encoder (re)configuration process.

- PCM input data will be retrieved from external input buffer until the fill level allows encoding a single frame. This functionality allows an external buffer with reduced size in comparison to the AAC or HE-AAC audio frame length.
- If the value of the input samples argument is zero, just internal reinitialization will be applied if it is requested.
- At the end of a file the flushing process can be triggerd via setting the value of the input samples argument to -1. The encoder delay lines are fully flushed when the encoder returns no valid bitstream data AACENC_OutArgs::numOutBytes. Furthermore the end of file is signaled by the return value AACENC_ENCODE_EOF.
- If an error occured in the previous frame or any of the encoder parameters changed, an internal reinitialization process will be applied before encoding the incoming audio samples.
- The function can also be used for an independent reconfiguration process without encoding. The first parameter has to be a valid encoder handle and all other parameters can be set to NULL.
- If the size of the external bitbuffer in outBufDesc is not sufficient for writing the whole bitstream, an internal error will be the return value and a reconfiguration will be triggered.

Parameters

hAacEncoder	A valid AAC encoder handle.

Parameters

inBufDesc	Input buffer descriptor, see AACENC_BufDesc: • At least one input buffer with audio data is expected. • Optionally a second input buffer with ancillary data can be fed.
outBufDesc	Output buffer descriptor, see AACENC_BufDesc: • Provide one output buffer for the encoded bitstream.
inargs	Input arguments, see AACENC_InArgs.
outargs	Output arguments, AACENC_OutArgs.

Returns

- AACENC_OK, on success.
- AACENC_INVALID_HANDLE, AACENC_ENCODE_ERROR, on failure in encoding process.
- AACENC_INVALID_CONFIG, AACENC_INIT_ERROR, AACENC_INIT_AAC_ERROR, AACENC_INIT_SBR_ERROR, AACENC_INIT_TP_ERROR, AACENC_INIT_META_ERROR, AACENC_INIT_MPS_ERROR, on failure in encoder initialization.
- AACENC_UNSUPPORTED_PARAMETER, on incorrect input or output buffer descriptor initialization.
- AACENC_ENCODE_EOF, when flushing fully concluded.

aacEncInfo()

```
AACENC_ERROR aacEncInfo (
            const HANDLE_AACENCODER hAacEncoder,
            AACENC_InfoStruct * pInfo )
```

Acquire info about present encoder instance.

This function retrieves information of the encoder configuration. In addition to informative internal states, a configuration data block of the current encoder settings will be returned. The format is either Audio Specific Config in case of Raw Packets transport format or StreamMuxConfig in case of LOAS/LATM transport format. The configuration data block is binary coded as specified in ISO/IEC 14496-3 (MPEG-4 audio), to be used directly for MPEG-4 File Format or RFC3016 or RFC3640 applications.

Parameters

hAacEncoder	A valid AAC encoder handle.
pInfo	Pointer to AACENC_InfoStruct. Filled on return.

Returns

- AACENC_OK, on succes.
- AACENC_INIT_ERROR, on failure.

aacEncoder_SetParam()

```
AACENC_ERROR aacEncoder_SetParam (
            const HANDLE_AACENCODER hAacEncoder,
            const AACENC_PARAM param,
            const UINT value )
```

Set one single AAC encoder parameter.

This function allows configuration of all encoder parameters specified in AACENC_PARAM. Each parameter must be set with a separate function call. An internal validation of the configuration value range will be done and an internal reconfiguration will be signaled. The actual configuration adoption is part of the subsequent aacEncEncode() call.

Parameters

hAacEncoder	A valid AAC encoder handle.
param	Parameter to be set. See AACENC_PARAM.
value	Parameter value. See parameter description in AACENC_PARAM.

Returns

- AACENC_OK, on success.
- AACENC_INVALID_HANDLE, AACENC_UNSUPPORTED_PARAMETER, AACENC_INVALID_CONFIG, on failure.

aacEncoder_GetParam()

```
UINT aacEncoder_GetParam (
            const HANDLE_AACENCODER hAacEncoder,
            const AACENC_PARAM param )
```

Get one single AAC encoder parameter.

This function is the complement to aacEncoder_SetParam(). After encoder reinitialization with user defined settings, the internal status can be obtained of each parameter, specified with AACENC_PARAM.

Parameters

hAacEncoder	A valid AAC encoder handle.
param	Parameter to be returned. See AACENC_PARAM.

Returns

Internal configuration value of specifed parameter AACENC_PARAM.

aacEncGetLibInfo()

```
AACENC_ERROR aacEncGetLibInfo (
            LIB_INFO * info )
```

Get information about encoder library build.

Fill a given LIB_INFO structure with library version information.

Parameters

info	Pointer to an allocated LIB_INFO struct.

Returns

- AACENC_OK, on success.
- AACENC_INVALID_HANDLE, AACENC_INIT_ERROR, on failure.

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