Fraunhofer
IIS

Advanced Audio Coding Decoder 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),
MPEG-D Extended High Efficiency AAC (xHE-AAC) / USAC,
decoder

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Revision 3.0.0 , February 26, 2018

Contents

Chapter 1

Introduction

1.1 Scope

This document describes the high-level application interface and usage of the ISO/MPEG-2/4 AAC Decoder library developed by the Fraunhofer Institute for Integrated Circuits (IIS). Depending on the library configuration, decoding of AAC-LC (Low-Complexity), HE-AAC (High-Efficiency AAC v1 and v2), AAC-LD (Low-Delay) and AAC-ELD (Enhanced Low-Delay) is implemented.

All references to SBR (Spectral Band Replication) are only applicable to HE-AAC and AAC-ELD configurations of the FDK library. All references to PS (Parametric Stereo) are only applicable to HE-AAC v2 decoder configuration of the library.

1.2 Decoder Basics

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

- ISO/IEC 13818-7 (MPEG-2 AAC) Standard, defines the syntax of MPEG-2 AAC audio bitstreams.
- ISO/IEC 14496-3 (MPEG-4 AAC, subpart 1 and 4) Standard, defines the syntax of MPEG-4 AAC audio bitstreams.
- ISO/IEC 23003-3 (MPEG-D USAC), defines MPEG-D USAC unified speech and audio codec.
- Lutzky, Schuller, Gayer, Krämer, Wabnik, ”A guideline to audio codec delay”, 116th AES Convention, May 8, 2004

In short, MPEG Advanced Audio Coding is based on a time-to-frequency mapping of the signal. The signal is partitioned into overlapping time portions and transformed into frequency domain. The spectral components are then quantized and coded using a highly efficient coding scheme.
Encoded MPEG-2 and MPEG-4 AAC audio bitstreams are 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 any length between 1 and 768 bytes.

In addition to the above mentioned frequency domain coding mode, MPEG-D USAC also employs a time domain Algebraic Code-Excited Linear Prediction (ACELP) speech coder core. This operating mode is selected by the encoder in order to achieve the optimum audio quality for different content type. Several enhancements allow achieving higher quality at lower bit rates compared to MPEG-4 HE-AAC.

Chapter 2

Library Usage

2.1 API Description

All API header files are located in the folder /include of the release package. The contents of each file is described in detail in this document. All header files are provided for usage in specific C/C++ programs. The main AAC decoder library API functions are located in aacdecoder_lib.h header file.

In binary releases the decoder core resides in statically linkable libraries, for example libAACdec.a.

2.2 Calling Sequence

The following sequence is necessary for proper decoding of ISO/MPEG-2/4 AAC, HE-AAC v2, or MPEG-D USAC bitstreams. In the following description, input stream read and output write function details are left out, since they may be implemented in a variety of configurations depending on the user's specific requirements. The example implementation uses file-based input/output, and in such case one may call mpegFileRead_Open() to open an input file and to allocate memory for the required structures, and the corresponding mpegFileRead_Close() to close opened files and to de-allocate associated structures. mpegFileRead_Open() will attempt to detect the bitstream format and in case of MPEG-4 file format or Raw Packets file format (a proprietary Fraunhofer IIS file format suitable only for testing) it will read the Audio Specific Config data (ASC). An unsuccessful attempt to recognize the bitstream format requires the user to provide this information manually. For any other bitstream formats that are usually applicable in streaming applications, the decoder itself will try to synchronize and parse the given bitstream fragment using the FDK transport library. Hence, for streaming applications (without file access) this step is not necessary.

1. Call aacDecoder_Open() to open and retrieve a handle to a new AAC decoder instance.

```
aacDecoderInfo = aacDecoder_Open(transportType, nrOfLayers);
```

2. If out-of-band config data (Audio Specific Config (ASC) or Stream Mux Config (SMC)) is available, call aacDecoder_ConfigRaw() to pass this data to the decoder before beginning the decoding process. If this data is not available in advance, the decoder will configure itself while decoding, during the aacDecoder_DecodeFrame() function call.

3. Begin decoding loop.

```
do {
```

4. Read data from bitstream file or stream buffer in to the driver program working memory (a client-supplied input buffer ”inBuffer” in framework). This buffer will be used to load AAC bitstream data to the decoder. Only when all data in this buffer has been processed will the decoder signal an empty buffer. For file-based input, you may invoke mpegFileRead_Read() to acquire new bitstream data.

5. Call aacDecoder_Fill() to fill the decoder's internal bitstream input buffer with the client-supplied bitstream input buffer. Note, if the data loaded in to the internal buffer is not sufficient to decode a frame, aacDecoder_DecodeFrame() will return AAC_DEC_NOT_ENOUGH_BITS until a sufficient amount of data is loaded in to the internal buffer. For streaming formats (ADTS, LOAS), it is acceptable to load more than one frame to the decoder. However, for RAW file format (Fraunhofer IIS proprietary format), only one frame may be loaded to the decoder per aacDecoder_DecodeFrame() call. For least amount of communication delay, fill and decode should be performed on a frame by frame basis.

```
ErrorStatus = aacDecoder_Fill(aacDecoderInfo, inBuffer, bytesRead, bytesValid);
```

6. Call aacDecoder_DecodeFrame(). This function decodes one frame and writes decoded PCM audio data to a client-supplied buffer. It is the client's responsibility to allocate a buffer which is large enough to hold the decoded output data.

```
ErrorStatus = aacDecoder_DecodeFrame(aacDecoderInfo, TimeData, OUT_BUF_SIZE, flags);
```

   If the bitstream configuration (number of channels, sample rate, frame size) is not known a priori, you may call aacDecoder_GetStreamInfo() to retrieve a structure that contains this information. You may use this data to initialize an audio output device. In the example program, if the number of channels or the sample rate has changed since program start or the previously decoded frame, the audio output device is then re-initialized. If WAVE file output is chosen, a new WAVE file for each new stream configuration is be created.

```
p_si = aacDecoder_GetStreamInfo(aacDecoderInfo);
```

7. Repeat steps 5 to 7 until no data is available to decode any more, or in case of error.

```
} while (bytesRead[0] > 0 || doFlush || doBsFlush || forceContinue);
```

8. Call aacDecoder_Close() to de-allocate all AAC decoder and transport layer structures.

```
aacDecoder_Close(aacDecoderInfo);
```

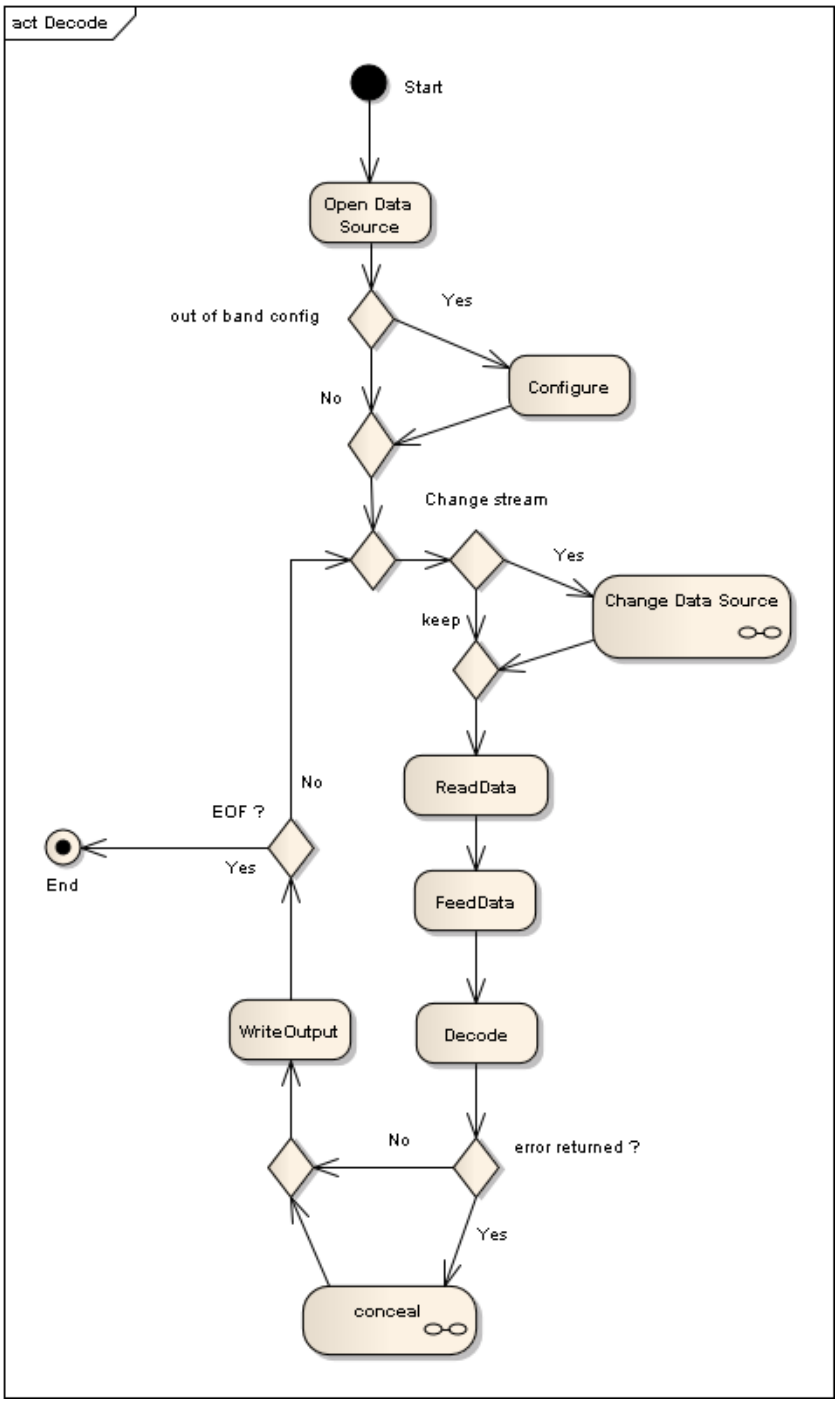


Figure 2.1: Decode calling sequence

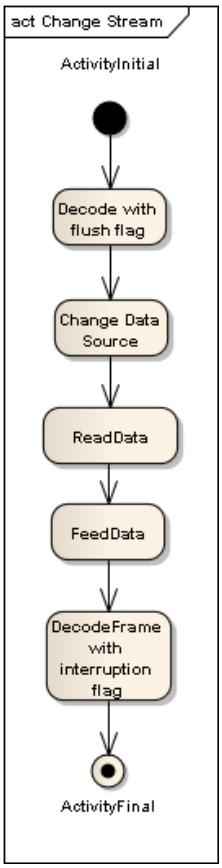


Figure 2.2: Change data source sequence

width 5cm

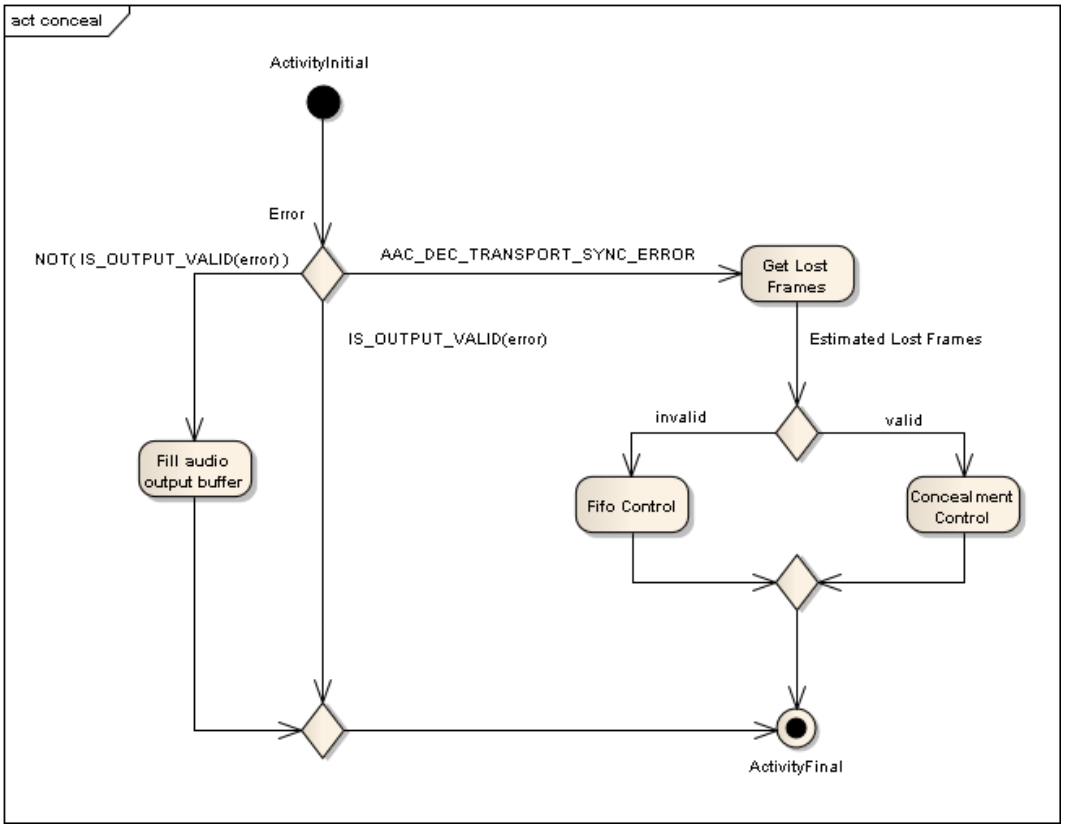


Figure 2.3: Error concealment sequence

2.2.1 Error Concealment Sequence

There are different strategies to handle bit stream errors. Depending on the system properties the product designer might choose to take different actions in case a bit error occurs. In many cases the decoder might be able to do reasonable error concealment without the need of any additional actions from the system. But in some cases its not even possible to know how many decoded PCM output samples are required to fill the gap due to the data error, then the software surrounding the decoder must deal with the situation. The most simple way would be to just stop audio playback and resume once enough bit stream data and/or buffered output samples are available. More sophisticated designs might also be able to deal with sender/receiver clock drifts or data drop outs by using a closed loop control of FIFO fulness levels. The chosen strategy depends on the final product requirements.

The error concealment sequence diagram illustrates the general execution paths for error handling.

The macro IS_OUTPUT_VALID(err) can be used to identify if the audio output buffer contains valid audio either from error free bit stream data or successful error concealment. In case the result is false, the decoder output buffer does not contain meaningful audio samples and should not be passed to any output as it is. Most likely in case that a continuous audio output PCM stream is required, the output buffer must be filled with audio data from the calling framework. This might be e.g. an appropriate number of samples all zero.

If error code AAC_DEC_TRANSPORT_SYNC_ERROR is returned by the decoder, under some particular conditions it is possible to estimate lost frames due to the bit stream error. In that case the bit stream is required to have a constant bitrate, and compatible transport type. Audio samples for the lost frames can be obtained by calling aacDecoder_DecodeFrame() with flag

AACDEC_CONCEAL set n-times where n is the count of lost frames. Please note that the decoder has to have encountered valid configuration data at least once to be able to generate concealed data, because at the minimum the sampling rate, frame size and amount of audio channels needs to be known.

If it is not possible to get an estimation of lost frames then a constant fullness of the audio output buffer can be achieved by implementing different FIFO control techniques e.g. just stop taking of samples from the buffer to avoid underflow or stop filling new data to the buffer to avoid overflow. But this techniques are out of scope of this document.

For a detailed description of a specific error code please refer also to AAC_DECODER_ERROR.

2.3 Buffer System

There are three main buffers in an AAC decoder application. One external input buffer to hold bitstream data from file I/O or elsewhere, one decoder-internal input buffer, and one to hold the decoded output PCM sample data. In resource limited applications, the output buffer may be reused as an external input buffer prior to the subsequence aacDecoder_Fill() function call.

The external input buffer is set in the example program and its size is defined by ::IN_BUF_SIZE. You may freely choose different buffer sizes. To feed the data to the decoder-internal input buffer, use the function aacDecoder_Fill(). This function returns important information regarding the number of bytes in the external input buffer that have not yet been copied into the internal input buffer (variable bytesValid). Once the external buffer has been fully copied, it can be completely re-filled again. In case you wish to refill the buffer while there are unprocessed bytes (bytesValid is unequal 0), you should preserve the unconsumed data. However, we recommend to refill the buffer only when bytesValid returns 0.

The bytesValid parameter is an input and output parameter to the FDK decoder. As an input, it signals how many valid bytes are available in the external buffer. After consumption of the external buffer using aacDecoder_Fill() function, the bytesValid parameter indicates if any of the bytes in the external buffer were not consumed.

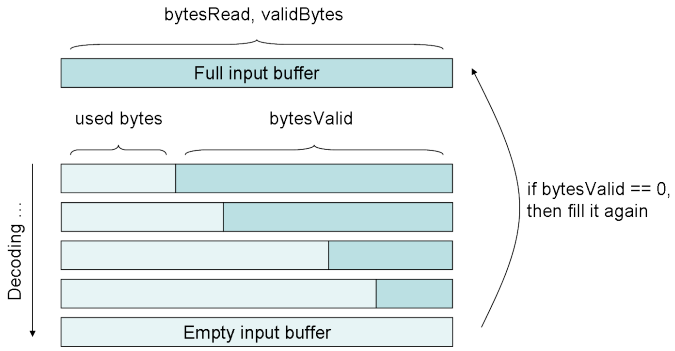


Figure 2.4: Life cycle of the external input buffer

Chapter 3

Decoder audio output

3.1 Obtaining channel mapping information

The decoded audio output format is indicated by a set of variables of the CStreamInfo structure. While the struct members sampleRate, frameSize and numChannels might be self explanatory, pChannelType and pChannelIndices require some further explanation.

These two arrays indicate the configuration of channel data within the output buffer. Both arrays have CStreamInfo::numChannels number of cells. Each cell of pChannelType indicates the channel type, which is described in the enum ::AUDIO_CHANNEL_TYPE (defined in FDK_audio.h). The cells of pChannelIndices indicate the sub index among the channels starting with 0 among channels of the same audio channel type.

The indexing scheme is structured as defined in MPEG-2/4 Standards. Indices start from the front direction (a center channel if available, will always be index 0) and increment, starting with the left side, pairwise (e.g. L, R) and from front to back (Front L, Front R, Surround L, Surround R). For detailed explanation, please refer to ISO/IEC 13818-7:2005(E), chapter 8.5.3.2.

In case a Program Config is included in the audio configuration, the channel mapping described within it will be adopted.

In case of MPEG-D Surround the channel mapping will follow the same criteria described in ISO/IEC 13818-7:2005(E), but adding corresponding top channels (if available) to the channel types in order to avoid ambiguity. The examples below explain these aspects in detail.

3.2 Changing the audio output format

For MPEG-4 audio the channel order can be changed at runtime through the parameter AAC_PCM_OUTPUT_CHANN
See the description of those parameters and the decoder library function aacDecoder_SetParam()
for more detail.

3.3 Channel mapping examples

The following examples illustrate the location of individual audio samples in the audio buffer that is passed to aacDecoder_DecodeFrame() and the expected data in the CStreamInfo structure which can be obtained by calling aacDecoder_GetStreamInfo().

3.3.1 Stereo

In case of AAC_PCM_OUTPUT_CHANNEL_MAPPING set to 1, a AAC-LC bit stream which has channelConfiguration = 2 in its audio specific config would lead to the following values in CStream-

Info:
CStreamInfo::numChannels = 2
CStreamInfo::pChannelType = { ::ACT_FRONT, ::ACT_FRONT }
CStreamInfo::pChannelIndices = { 0, 1 }
The output buffer will be formatted as follows:

```
<left sample 0>  <left sample 1>  <left sample 2>  ... <left sample N>
<right sample 0> <right sample 1> <right sample 2> ... <right sample N>
```

Where N equals to CStreamInfo::frameSize .

3.3.2 Surround 5.1

In case of AAC_PCM_OUTPUT_CHANNEL_MAPPING set to 1, a AAC-LC bit stream which has channelConfiguration = 6 in its audio specific config, would lead to the following values in CStreamInfo:
CStreamInfo::numChannels = 6
CStreamInfo::pChannelType = { ::ACT_FRONT, ::ACT_FRONT, ::ACT_FRONT, ::ACT_LFE, ::ACT_BACK, ::ACT_BACK }
CStreamInfo::pChannelIndices = { 1, 2, 0, 0, 0, 1 }
Since AAC_PCM_OUTPUT_CHANNEL_MAPPING is 1, WAV file channel ordering will be used. For a 5.1 channel scheme, thus the channels would be: front left, front right, center, LFE, surround left, surround right. Thus the third channel is the center channel, receiving the index 0. The other front channels are front left, front right being placed as first and second channels with indices 1 and 2 correspondingly. There is only one LFE, placed as the fourth channel and index 0. Finally both surround channels get the type definition ACT_BACK, and the indices 0 and 1.
The output buffer will be formatted as follows:

```
<front left sample 0> <front right sample 0>
<center sample 0> <LFE sample 0>
<surround left sample 0> <surround right sample 0>

<front left sample 1> <front right sample 1>
<center sample 1> <LFE sample 1>
<surround left sample 1> <surround right sample 1>

...

<front left sample N> <front right sample N>
<center sample N> <LFE sample N>
<surround left sample N> <surround right sample N>
```

Where N equals to CStreamInfo::frameSize .

3.3.3 ARIB coding mode 2/1

In case of AAC_PCM_OUTPUT_CHANNEL_MAPPING set to 1, in case of a ARIB bit stream using coding mode 2/1 as described in ARIB STD-B32 Part 2 Version 2.1-E1, page 61, would lead to the following values in CStreamInfo:
CStreamInfo::numChannels = 3
CStreamInfo::pChannelType = { ::ACT_FRONT, ::ACT_FRONT, ::ACT_BACK }
CStreamInfo::pChannelIndices = { 0, 1, 0 }
The audio channels will be placed as follows in the audio output buffer:

```
<front left sample 0> <front right sample 0>  <mid surround sample 0>

<front left sample 1> <front right sample 1> <mid surround sample 1>
```

```
...
<front left sample N> <front right sample N> <mid surround sample N>
Where N equals to CStreamInfo::frameSize .
```

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 CStreamInfo Struct Reference

This structure gives information about the currently decoded audio data. All fields are read-only.

`#include <aacdecoder_lib.h>`

Public Attributes

- INT sampleRate
- INT frameSize
- INT numChannels
- AUDIO_CHANNEL_TYPE * pChannelType
- UCHAR * pChannelIndices
- INT aacSampleRate
- INT profile
- AUDIO_OBJECT_TYPE aot
- INT channelConfig
- INT bitRate
- INT aacSamplesPerFrame
- INT aacNumChannels
- AUDIO_OBJECT_TYPE extAot
- INT extSamplingRate
- UINT outputDelay
- UINT flags
- SCHAR epConfig
- INT numLostAccessUnits
- UINT numTotalBytes
- UINT numBadBytes
- UINT numTotalAccessUnits
- UINT numBadAccessUnits
- SCHAR drcProgRefLev
- SCHAR drcPresMode

6.1.1 Detailed Description

This structure gives information about the currently decoded audio data. All fields are read-only.

6.1.2 Member Data Documentation

sampleRate

`INT CStreamInfo::sampleRate`

The sample rate in Hz of the decoded PCM audio signal.

frameSize

`INT CStreamInfo::frameSize`

The frame size of the decoded PCM audio signal.
Typically this is:
1024 or 960 for AAC-LC
2048 or 1920 for HE-AAC (v2)
512 or 480 for AAC-LD and AAC-ELD
768, 1024, 2048 or 4096 for USAC

numChannels

`INT CStreamInfo::numChannels`

The number of output audio channels before the rendering module, i.e. the original channel configuration.

pChannelType

`AUDIO_CHANNEL_TYPE* CStreamInfo::pChannelType`

Audio channel type of each output audio channel.

pChannelIndices

`UCHAR* CStreamInfo::pChannelIndices`

Audio channel index for each output audio channel. See ISO/IEC 13818-7:2005(E), 8.5.3.2 Explicit channel mapping using a program_config_element()

aacSampleRate

`INT CStreamInfo::aacSampleRate`

Sampling rate in Hz without SBR (from configuration info) divided by a (ELD) downscale factor if present.

profile

`INT CStreamInfo::profile`

MPEG-2 profile (from file header) (-1: not applicable (e. g. MPEG-4)).

aot

`AUDIO_OBJECT_TYPE CStreamInfo::aot`

Audio Object Type (from ASC): is set to the appropriate value for MPEG-2 bitstreams (e. g. 2 for AAC-LC).

channelConfig

`INT CStreamInfo::channelConfig`

Channel configuration (0: PCE defined, 1: mono, 2: stereo, ...

bitRate

`INT CStreamInfo::bitRate`

Instantaneous bit rate.

aacSamplesPerFrame

`INT CStreamInfo::aacSamplesPerFrame`

Samples per frame for the AAC core (from ASC) divided by a (ELD) downscale factor if present.
Typically this is (with a downscale factor of 1):
1024 or 960 for AAC-LC
512 or 480 for AAC-LD and AAC-ELD

aacNumChannels

`INT CStreamInfo::aacNumChannels`

The number of audio channels after AAC core processing (before PS or MPS processing). CAUTION: This are not the final number of output channels!

extAot

`AUDIO_OBJECT_TYPE CStreamInfo::extAot`

Extension Audio Object Type (from ASC)

extSamplingRate

`INT CStreamInfo::extSamplingRate`

Extension sampling rate in Hz (from ASC) divided by a (ELD) downscale factor if present.

outputDelay

`UINT CStreamInfo::outputDelay`

The number of samples the output is additionally delayed by.the decoder.

flags

`UINT CStreamInfo::flags`

Copy of internal flags. Only to be written by the decoder, and only to be read externally.

epConfig

`SCHAR CStreamInfo::epConfig`

epConfig level (from ASC): only level 0 supported, -1 means no ER (e. g. AOT=2, MPEG-2 AAC, etc.)

numLostAccessUnits

`INT CStreamInfo::numLostAccessUnits`

This integer will reflect the estimated amount of lost access units in case aacDecoder_DecodeFrame() returns AAC_DEC_TRANSPORT_SYNC_ERROR. It will be < 0 if the estimation failed.

numTotalBytes

`UINT CStreamInfo::numTotalBytes`

This is the number of total bytes that have passed through the decoder.

numBadBytes

`UINT CStreamInfo::numBadBytes`

This is the number of total bytes that were considered with errors from numTotalBytes.

numTotalAccessUnits

`UINT CStreamInfo::numTotalAccessUnits`

This is the number of total access units that have passed through the decoder.

numBadAccessUnits

`UINT CStreamInfo::numBadAccessUnits`

This is the number of total access units that were considered with errors from numTotalBytes.

drcProgRefLev

`SCHAR CStreamInfo::drcProgRefLev`

DRC program reference level. Defines the reference level below full-scale. It is quantized in steps of 0.25dB. The valid values range from 0 (0 dBFS) to 127 (-31.75 dBFS). It is used to reflect the average loudness of the audio in LKFS according to ITU-R BS 1770. If no level has been found in the bitstream the value is -1.

drcPresMode

`SCHAR CStreamInfo::drcPresMode`

DRC presentation mode. According to ETSI TS 101 154, this field indicates whether light (MPEG-4 Dynamic Range Control tool) or heavy compression (DVB heavy compression) dynamic range control shall take priority on the outputs. For details, see ETSI TS 101 154, table C.33. Possible values are:
-1: No corresponding metadata found in the bitstream
0: DRC presentation mode not indicated
1: DRC presentation mode 1
2: DRC presentation mode 2
3: Reserved

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

- aacdecoder_lib.h

Chapter 7

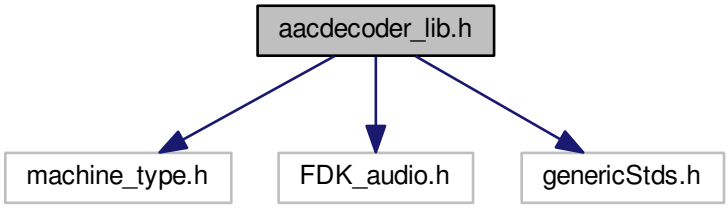
File Documentation

7.1 aacdecoder_lib.h File Reference

FDK AAC decoder library interface header file.

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

Include dependency graph for aacdecoder_lib.h:



Classes

- struct CStreamInfo

  This structure gives information about the currently decoded audio data. All fields are read-only.

Macros

- #define IS_INIT_ERROR(err) ((((err)>=aac_dec_init_error_start) && ((err)<=aac_dec_init_error_end)) ? 1 : 0)
- #define IS_DECODE_ERROR(err) ((((err)>=aac_dec_decode_error_start) && ((err)<=aac_dec_decode_error_er ? 1 : 0)
- #define IS_OUTPUT_VALID(err) (((err) == AAC_DEC_OK) || IS_DECODE_ERROR(err))
- #define AACDEC_CONCEAL 1
- #define AACDEC_FLUSH 2

- #define AACDEC_INTR 4
- #define AACDEC_CLRHIST 8

Typedefs

- typedef struct AAC_DECODER_INSTANCE * HANDLE_AACDECODER

Enumerations

- enum AAC_DECODER_ERROR {
  AAC_DEC_OK = 0x0000,
  AAC_DEC_OUT_OF_MEMORY = 0x0002,
  AAC_DEC_UNKNOWN = 0x0005,
  aac_dec_sync_error_start = 0x1000,
  AAC_DEC_TRANSPORT_SYNC_ERROR = 0x1001,
  AAC_DEC_NOT_ENOUGH_BITS = 0x1002,
  aac_dec_sync_error_end = 0x1FFF,
  aac_dec_init_error_start = 0x2000,
  AAC_DEC_INVALID_HANDLE = 0x2001,
  AAC_DEC_UNSUPPORTED_AOT = 0x2002,
  AAC_DEC_UNSUPPORTED_FORMAT = 0x2003,
  AAC_DEC_UNSUPPORTED_ER_FORMAT = 0x2004,
  AAC_DEC_UNSUPPORTED_EPCONFIG = 0x2005,
  AAC_DEC_UNSUPPORTED_MULTILAYER = 0x2006,
  AAC_DEC_UNSUPPORTED_CHANNELCONFIG = 0x2007,
  AAC_DEC_UNSUPPORTED_SAMPLINGRATE = 0x2008,
  AAC_DEC_INVALID_SBR_CONFIG = 0x2009,
  AAC_DEC_SET_PARAM_FAIL = 0x200A,
  AAC_DEC_NEED_TO_RESTART = 0x200B,
  AAC_DEC_OUTPUT_BUFFER_TOO_SMALL = 0x200C,
  aac_dec_init_error_end = 0x2FFF,
  aac_dec_decode_error_start = 0x4000,
  AAC_DEC_TRANSPORT_ERROR = 0x4001,
  AAC_DEC_PARSE_ERROR = 0x4002,
  AAC_DEC_UNSUPPORTED_EXTENSION_PAYLOAD = 0x4003,
  AAC_DEC_DECODE_FRAME_ERROR = 0x4004,
  AAC_DEC_CRC_ERROR = 0x4005,
  AAC_DEC_INVALID_CODE_BOOK = 0x4006,
  AAC_DEC_UNSUPPORTED_PREDICTION = 0x4007,
  AAC_DEC_UNSUPPORTED_CCE = 0x4008,
  AAC_DEC_UNSUPPORTED_LFE = 0x4009,
  AAC_DEC_UNSUPPORTED_GAIN_CONTROL_DATA = 0x400A,
  AAC_DEC_UNSUPPORTED_SBA = 0x400B,
  AAC_DEC_TNS_READ_ERROR = 0x400C,
  AAC_DEC_RVLC_ERROR = 0x400D,
  aac_dec_decode_error_end = 0x4FFF,
  aac_dec_anc_data_error_start = 0x8000,
  AAC_DEC_ANC_DATA_ERROR = 0x8001,
  AAC_DEC_TOO_SMALL_ANC_BUFFER = 0x8002,
  AAC_DEC_TOO_MANY_ANC_ELEMENTS = 0x8003,
  aac_dec_anc_data_error_end = 0x8FFF }

  AAC decoder error codes.

- enum AAC_MD_PROFILE {
  AAC_MD_PROFILE_MPEG_STANDARD = 0,
  AAC_MD_PROFILE_MPEG_LEGACY = 1,
  AAC_MD_PROFILE_MPEG_LEGACY_PRIO = 2,
  AAC_MD_PROFILE_ARIB_JAPAN = 3 }

  The available metadata profiles which are mostly related to downmixing. The values define the arguments for the use with parameter AAC_METADATA_PROFILE.
- enum AAC_DRC_DEFAULT_PRESENTATION_MODE_OPTIONS {
  AAC_DRC_PARAMETER_HANDLING_DISABLED = -1,
  AAC_DRC_PARAMETER_HANDLING_ENABLED = 0,
  AAC_DRC_PRESENTATION_MODE_1_DEFAULT = 1,
  AAC_DRC_PRESENTATION_MODE_2_DEFAULT = 2 }

  Options for handling of DRC parameters, if presentation mode is not indicated in bitstream.
- enum AACDEC_PARAM {
  AAC_PCM_DUAL_CHANNEL_OUTPUT_MODE = 0x0002,
  AAC_PCM_OUTPUT_CHANNEL_MAPPING = 0x0003,
  AAC_PCM_LIMITER_ENABLE = 0x0004,
  AAC_PCM_LIMITER_ATTACK_TIME = 0x0005,
  AAC_PCM_LIMITER_RELEAS_TIME = 0x0006,
  AAC_PCM_MIN_OUTPUT_CHANNELS = 0x0011,
  AAC_PCM_MAX_OUTPUT_CHANNELS = 0x0012,
  AAC_METADATA_PROFILE = 0x0020,
  AAC_METADATA_EXPIRY_TIME = 0x0021,
  AAC_CONCEAL_METHOD = 0x0100,
  AAC_DRC_BOOST_FACTOR = 0x0200,
  AAC_DRC_ATTENUATION_FACTOR = 0x0201,
  AAC_DRC_REFERENCE_LEVEL = 0x0202,
  AAC_DRC_HEAVY_COMPRESSION = 0x0203,
  AAC_DRC_DEFAULT_PRESENTATION_MODE = 0x0204,
  AAC_DRC_ENC_TARGET_LEVEL = 0x0205,
  AAC_QMF_LOWPOWER = 0x0300,
  AAC_TPDEC_CLEAR_BUFFER = 0x0603,
  AAC_UNIDRC_SET_EFFECT = 0x0903 }

  AAC decoder setting parameters.

Functions

- LINKSPEC_H AAC_DECODER_ERROR aacDecoder_AncDataInit (HANDLE_AACDECODER self, UCHAR *buffer, int size)

  Initialize ancillary data buffer.
- LINKSPEC_H AAC_DECODER_ERROR aacDecoder_AncDataGet (HANDLE_AACDECODER self, int index, UCHAR **ptr, int *size)

  Get one ancillary data element.
- LINKSPEC_H AAC_DECODER_ERROR aacDecoder_SetParam (const HANDLE_AACDECODER self, const AACDEC_PARAM param, const INT value)

  Set one single decoder parameter.
- LINKSPEC_H AAC_DECODER_ERROR aacDecoder_GetFreeBytes (const HANDLE_AACDECODER self, UINT *pFreeBytes)

  Get free bytes inside decoder internal buffer.
- LINKSPEC_H HANDLE_AACDECODER aacDecoder_Open (TRANSPORT_TYPE transportFmt, UINT nrOfLayers)

Open an AAC decoder instance.

- LINKSPEC_H AAC_DECODER_ERROR aacDecoder_ConfigRaw (HANDLE_AACDECODER self, UCHAR *conf[ ], const UINT length[ ])

  Explicitly configure the decoder by passing a raw AudioSpecificConfig (ASC) or a StreamMuxConfig (SMC), contained in a binary buffer. This is required for MPEG-4 and Raw Packets file format bitstreams as well as for LATM bitstreams with no in-band SMC. If the transport format is LATM with or without LOAS, configuration is assumed to be an SMC, for all other file formats an ASC.

- LINKSPEC_H AAC_DECODER_ERROR aacDecoder_LoudnessBox (HANDLE_AACDECODER self, UCHAR *buffer, UINT length)

  Submit raw payload of ISO base media file format loudness box ('ludt').

- LINKSPEC_H AAC_DECODER_ERROR aacDecoder_Fill (HANDLE_AACDECODER self, UCHAR *pBuffer[ ], const UINT bufferSize[ ], UINT *bytesValid)

  Fill AAC decoder's internal input buffer with bitstream data from the external input buffer. The function only copies such data as long as the decoder-internal input buffer is not full. So it grabs whatever it can from pBuffer and returns information (bytesValid) so that at a subsequent call of aacDecoder_Fill(), the right position in pBuffer can be determined to grab the next data.

- LINKSPEC_H AAC_DECODER_ERROR aacDecoder_DecodeFrame (HANDLE_AACDECODER self, INT_PCM *pTimeData, const INT timeDataSize, const UINT flags)

  Decode one audio frame.

- LINKSPEC_H void aacDecoder_Close (HANDLE_AACDECODER self)

  De-allocate all resources of an AAC decoder instance.

- LINKSPEC_H CStreamInfo * aacDecoder_GetStreamInfo (HANDLE_AACDECODER self)

  Get CStreamInfo handle from decoder.

- LINKSPEC_H INT aacDecoder_GetLibInfo (LIB_INFO *info)

  Get decoder library info.

7.1.1 Detailed Description

FDK AAC decoder library interface header file.

7.1.2 Macro Definition Documentation

IS_INIT_ERROR

```
#define IS_INIT_ERROR(
              err ) ( (((err)>=aac_dec_init_error_start) && ((err)<=aac_dec_init_error_end))
?  1 :  0)
```

Macro to identify initialization errors. Output buffer is invalid.

IS_DECODE_ERROR

```
#define IS_DECODE_ERROR(
              err ) ( (((err)>=aac_dec_decode_error_start) && ((err)<=aac_dec_decode_error_end))
?  1 :  0)
```

Macro to identify decode errors. Output buffer is valid but concealed.

IS_OUTPUT_VALID

```
#define IS_OUTPUT_VALID(
              err ) ( ((err) == AAC_DEC_OK) || IS_DECODE_ERROR(err) )
```

Macro to identify if the audio output buffer contains valid samples after calling aacDecoder_DecodeFrame(). Output buffer is valid but can be concealed.

AACDEC_CONCEAL

```
#define AACDEC_CONCEAL 1
```

Flag for aacDecoder_DecodeFrame(): Trigger the built-in error concealment module to generate a substitute signal for one lost frame. New input data will not be considered.

AACDEC_FLUSH

```
#define AACDEC_FLUSH 2
```

Flag for aacDecoder_DecodeFrame(): Flush all filterbanks to get all delayed audio without having new input data. Thus new input data will not be considered.

AACDEC_INTR

```
#define AACDEC_INTR 4
```

Flag for aacDecoder_DecodeFrame(): Signal an input bit stream data discontinuity. Resync any internals as necessary.

AACDEC_CLRHIST

```
#define AACDEC_CLRHIST 8
```

Flag for aacDecoder_DecodeFrame(): Clear all signal delay lines and history buffers. CAUTION: This can cause discontinuities in the output signal.

7.1.3 Typedef Documentation

HANDLE_AACDECODER

```
typedef struct AAC_DECODER_INSTANCE* HANDLE_AACDECODER
```

Pointer to a AAC decoder instance.

7.1.4 Enumeration Type Documentation

AAC_DECODER_ERROR

```
enum AAC_DECODER_ERROR
```

AAC decoder error codes.

Enumerator

AAC_DEC_OK	No error occurred. Output buffer is valid and error free.

Enumerator

AAC_DEC_OUT_OF_MEMORY	Heap returned NULL pointer. Output buffer is invalid.
AAC_DEC_UNKNOWN	Error condition is of unknown reason, or from a another module. Output buffer is invalid.
aac_dec_sync_error_start	
AAC_DEC_TRANSPORT_SYNC_ERROR	The transport decoder had synchronization problems. Do not exit decoding. Just feed new bitstream data.
AAC_DEC_NOT_ENOUGH_BITS	The input buffer ran out of bits.
aac_dec_sync_error_end	
aac_dec_init_error_start	
AAC_DEC_INVALID_HANDLE	The handle passed to the function call was invalid (NULL).
AAC_DEC_UNSUPPORTED_AOT	The AOT found in the configuration is not supported.
AAC_DEC_UNSUPPORTED_FORMAT	The bitstream format is not supported.
AAC_DEC_UNSUPPORTED_ER_FORMAT	The error resilience tool format is not supported.
AAC_DEC_UNSUPPORTED_EPCONFIG	The error protection format is not supported.
AAC_DEC_UNSUPPORTED_MULTILAYER	More than one layer for AAC scalable is not supported.
AAC_DEC_UNSUPPORTED_CHANNELCONFIG	The channel configuration (either number or arrangement) is not supported.
AAC_DEC_UNSUPPORTED_SAMPLINGRATE	The sample rate specified in the configuration is not supported.
AAC_DEC_INVALID_SBR_CONFIG	The SBR configuration is not supported.
AAC_DEC_SET_PARAM_FAIL	The parameter could not be set. Either the value was out of range or the parameter does not exist.
AAC_DEC_NEED_TO_RESTART	The decoder needs to be restarted, since the required configuration change cannot be performed.
AAC_DEC_OUTPUT_BUFFER_TOO_SMALL	The provided output buffer is too small.
aac_dec_init_error_end	
aac_dec_decode_error_start	
AAC_DEC_TRANSPORT_ERROR	The transport decoder encountered an unexpected error.
AAC_DEC_PARSE_ERROR	Error while parsing the bitstream. Most probably it is corrupted, or the system crashed.
AAC_DEC_UNSUPPORTED_EXTENSION_PAYLOAD	Error while parsing the extension payload of the bitstream. The extension payload type found is not supported.

Enumerator

AAC_DEC_DECODE_FRAME_ERROR	The parsed bitstream value is out of range. Most probably the bitstream is corrupt, or the system crashed.
AAC_DEC_CRC_ERROR	The embedded CRC did not match.
AAC_DEC_INVALID_CODE_BOOK	An invalid codebook was signaled. Most probably the bitstream is corrupt, or the system crashed.
AAC_DEC_UNSUPPORTED_PREDICTION	Predictor found, but not supported in the AAC Low Complexity profile. Most probably the bitstream is corrupt, or has a wrong format.
AAC_DEC_UNSUPPORTED_CCE	A CCE element was found which is not supported. Most probably the bitstream is corrupt, or has a wrong format.
AAC_DEC_UNSUPPORTED_LFE	A LFE element was found which is not supported. Most probably the bitstream is corrupt, or has a wrong format.
AAC_DEC_UNSUPPORTED_GAIN_CONTROL_DATA	Gain control data found but not supported. Most probably the bitstream is corrupt, or has a wrong format.
AAC_DEC_UNSUPPORTED_SBA	SBA found, but currently not supported in the BSAC profile.
AAC_DEC_TNS_READ_ERROR	Error while reading TNS data. Most probably the bitstream is corrupt or the system crashed.
AAC_DEC_RVLC_ERROR	Error while decoding error resilient data.
aac_dec_decode_error_end	
aac_dec_anc_data_error_start	
AAC_DEC_ANC_DATA_ERROR	Non severe error concerning the ancillary data handling.
AAC_DEC_TOO_SMALL_ANC_BUFFER	The registered ancillary data buffer is too small to receive the parsed data.
AAC_DEC_TOO_MANY_ANC_ELEMENTS	More than the allowed number of ancillary data elements should be written to buffer.
aac_dec_anc_data_error_end	

AAC_MD_PROFILE

`enum AAC_MD_PROFILE`

The available metadata profiles which are mostly related to downmixing. The values define the arguments for the use with parameter AAC_METADATA_PROFILE.

Enumerator

AAC_MD_PROFILE_MPEG_STANDARD	The standard profile creates a mixdown signal based on the advanced downmix metadata (from a DSE). The equations and default values are defined in ISO/IEC 14496:3 Ammendment 4. Any other (legacy) downmix metadata will be ignored. No other parameter will be modified.
AAC_MD_PROFILE_MPEG_LEGACY	This profile behaves identical to the standard profile if advanced downmix metadata (from a DSE) is available. If not, the matrix_mixdown information embedded in the program configuration element (PCE) will be applied. If neither is the case, the module creates a mixdown using the default coefficients as defined in ISO/IEC 14496:3 AMD 4. The profile can be used to support legacy digital TV (e.g. DVB) streams.
AAC_MD_PROFILE_MPEG_LEGACY_PRIO	Similar to the AAC_MD_PROFILE_MPEG_LEGACY profile but if both the advanced (ISO/IEC 14496:3 AMD 4) and the legacy (PCE) MPEG downmix metadata are available the latter will be applied.
AAC_MD_PROFILE_ARIB_JAPAN	Downmix creation as described in ABNT NBR 15602-2. But if advanced downmix metadata (ISO/IEC 14496:3 AMD 4) is available it will be preferred because of the higher resolutions. In addition the metadata expiry time will be set to the value defined in the ARIB standard (see AAC_METADATA_EXPIRY_TIME).

AAC_DRC_DEFAULT_PRESENTATION_MODE_OPTIONS

`enum AAC_DRC_DEFAULT_PRESENTATION_MODE_OPTIONS`

Options for handling of DRC parameters, if presentation mode is not indicated in bitstream.

Enumerator

AAC_DRC_PARAMETER_HANDLING_DISABLED	DRC parameter handling disabled, all parameters are applied as requested.
AAC_DRC_PARAMETER_HANDLING_ENABLED	Apply changes to requested DRC parameters to prevent clipping.
AAC_DRC_PRESENTATION_MODE_1_DEFAULT	Use DRC presentation mode 1 as default (e.g. for Nordig)
AAC_DRC_PRESENTATION_MODE_2_DEFAULT	Use DRC presentation mode 2 as default (e.g. for DTG DBook)

AACDEC_PARAM

enum AACDEC_PARAM

AAC decoder setting parameters.

Enumerator

AAC_PCM_DUAL_CHANNEL_OUTPUT_MODE	Defines how the decoder processes two channel signals: 0: Leave both signals as they are (default). 1: Create a dual mono output signal from channel 1. 2: Create a dual mono output signal from channel 2. 3: Create a dual mono output signal by mixing both channels (L' = R' = 0.5∗Ch1 + 0.5∗Ch2).
AAC_PCM_OUTPUT_CHANNEL_MAPPING	Output buffer channel ordering. 0: MPEG PCE style order, 1: WAV file channel order (default).
AAC_PCM_LIMITER_ENABLE	Enable signal level limiting. -1: Auto-config. Enable limiter for all non-lowdelay configurations by default. 0: Disable limiter in general. 1: Enable limiter always. It is recommended to call the decoder with a AACDEC_CLRHIST flag to reset all states when the limiter switch is changed explicitly.
AAC_PCM_LIMITER_ATTACK_TIME	Signal level limiting attack time in ms. Default configuration is 15 ms. Adjustable range from 1 ms to 15 ms.
AAC_PCM_LIMITER_RELEAS_TIME	Signal level limiting release time in ms. Default configuration is 50 ms. Adjustable time must be larger than 0 ms.

Enumerator

AAC_PCM_MIN_OUTPUT_CHANNELS	Minimum number of PCM output channels. If higher than the number of encoded audio channels, a simple channel extension is applied (see note 4 for exceptions). -1, 0: Disable channel extension feature. The decoder output contains the same number of channels as the encoded bitstream. 1: This value is currently needed only together with the mix-down feature. See AAC_PCM_MAX_OUTPUT_CHANNELS and note 2 below. 2: Encoded mono signals will be duplicated to achieve a 2/0/0.0 channel output configuration. 6: The decoder tries to reorder encoded signals with less than six channels to achieve a 3/0/2.1 channel output signal. Missing channels will be filled with a zero signal. If reordering is not possible the empty channels will simply be appended. Only available if instance is configured to support multichannel output. 8: The decoder tries to reorder encoded signals with less than eight channels to achieve a 3/0/4.1 channel output signal. Missing channels will be filled with a zero signal. If reordering is not possible the empty channels will simply be appended. Only available if instance is configured to support multichannel output. NOTE: 1. The channel signaling (CStreamInfo::pChannelType and CStreamInfo::pChannelIndices) will not be modified. Added empty channels will be signaled with channel type AUDIO_CHANNEL_TYPE::ACT_NONE. 2. If the parameter value is greater than that of AAC_PCM_MAX_OUTPUT_CHANNELS both will be set to the same value. 3. This parameter does not affect MPEG Surround processing. 4. This parameter will be ignored if the number of encoded audio channels is greater than 8.

Enumerator

AAC_PCM_MAX_OUTPUT_CHANNELS	Maximum number of PCM output channels. If lower than the number of encoded audio channels, downmixing is applied accordingly (see note 5 for exceptions). If dedicated metadata is available in the stream it will be used to achieve better mixing results. -1, 0: Disable downmixing feature. The decoder output contains the same number of channels as the encoded bitstream. 1: All encoded audio configurations with more than one channel will be mixed down to one mono output signal. 2: The decoder performs a stereo mix-down if the number encoded audio channels is greater than two. 6: If the number of encoded audio channels is greater than six the decoder performs a mix-down to meet the target output configuration of 3/0/2.1 channels. Only available if instance is configured to support multichannel output. 8: This value is currently needed only together with the channel extension feature. See AAC_PCM_MIN_OUTPUT_CHANNELS and note 2 below. Only available if instance is configured to support multichannel output. NOTE: 1. Down-mixing of any seven or eight channel configuration not defined in ISO/IEC 14496-3 PDAM 4 is not supported by this software version. 2. If the parameter value is greater than zero but smaller than AAC_PCM_MIN_OUTPUT_CHANNELS both will be set to same value. 3. The operating mode of the MPEG Surround module will be set accordingly. 4. Setting this parameter with any value will disable the binaural processing of the MPEG Surround module 5. This parameter will be ignored if the number of encoded audio channels is greater than 8.

Enumerator

AAC_METADATA_PROFILE	See AAC_MD_PROFILE for all available values.
AAC_METADATA_EXPIRY_TIME	Defines the time in ms after which all the bitstream associated meta-data (DRC, downmix coefficients, ...) will be reset to default if no update has been received. Negative values disable the feature.
AAC_CONCEAL_METHOD	Error concealment: Processing method. 0: Spectral muting. 1: Noise substitution (see ::CONCEAL_NOISE). 2: Energy interpolation (adds additional signal delay of one frame, see ::CONCEAL_INTER. only some AOTs are supported).
AAC_DRC_BOOST_FACTOR	Dynamic Range Control: Scaling factor for boosting gain values. Defines how the boosting DRC factors (conveyed in the bitstream) will be applied to the decoded signal. The valid values range from 0 (don't apply boost factors) to 127 (fully apply boost factors). Default value is 0.
AAC_DRC_ATTENUATION_FACTOR	Dynamic Range Control: Scaling factor for attenuating gain values. Same as AAC_DRC_BOOST_FACTOR but for attenuating DRC factors.
AAC_DRC_REFERENCE_LEVEL	Dynamic Range Control (DRC): Target reference level. Defines the level below full-scale (quantized in steps of 0.25dB) to which the output audio signal will be normalized to by the DRC module. The parameter controls loudness normalization for both MPEG-4 DRC and MPEG-D DRC. The valid values range from 40 (-10 dBFS) to 127 (-31.75 dBFS). Any value smaller than 0 switches off loudness normalization and MPEG-4 DRC. By default, loudness normalization and MPEG-4 DRC is switched off.
AAC_DRC_HEAVY_COMPRESSION	Dynamic Range Control: En-/Disable DVB specific heavy compression (aka RF mode). If set to 1, the decoder will apply the compression values from the DVB specific ancillary data field. At the same time the MPEG-4 Dynamic Range Control tool will be disabled. By default, heavy compression is disabled.

Enumerator

AAC_DRC_DEFAULT_PRESENTATION_MODE	Dynamic Range Control: Default presentation mode (DRC parameter handling). Defines the handling of the DRC parameters boost factor, attenuation factor and heavy compression, if no presentation mode is indicated in the bitstream. For options, see AAC_DRC_DEFAULT_PRESENTATION_MODE_OPTIONS. Default: AAC_DRC_PARAMETER_HANDLING_DISABLED
AAC_DRC_ENC_TARGET_LEVEL	Dynamic Range Control: Encoder target level for light (i.e. not heavy) compression. If known, this declares the target reference level that was assumed at the encoder for calculation of limiting gains. The valid values range from 0 (full-scale) to 127 (31.75 dB below full-scale). This parameter is used only with AAC_DRC_PARAMETER_HANDLING_ENABLED and ignored otherwise. Default: 127 (worst-case assumption).
AAC_QMF_LOWPOWER	Quadrature Mirror Filter (QMF) Bank processing mode. -1: Use internal default. Implies MPEG Surround partially complex accordingly. 0: Use complex QMF data mode. 1: Use real (low power) QMF data mode.
AAC_TPDEC_CLEAR_BUFFER	Clear internal bit stream buffer of transport layers. The decoder will start decoding at new data passed after this event and any previous data is discarded.
AAC_UNIDRC_SET_EFFECT	MPEG-D DRC: Request a DRC effect type for selection of a DRC set. Supported indices are: -1: DRC off. Completely disables MPEG-D DRC. 0: None (default). Disables MPEG-D DRC, but automatically enables DRC if necessary to prevent clipping. 1: Late night 2: Noisy environment 3: Limited playback range 4: Low playback level 5: Dialog enhancement 6: General compression. Used for generally enabling MPEG-D DRC without particular request.

7.1.5 Function Documentation

aacDecoder_AncDataInit()

```
LINKSPEC_H AAC_DECODER_ERROR aacDecoder_AncDataInit (
              HANDLE_AACDECODER self,
              UCHAR * buffer,
              int size )
```

Initialize ancillary data buffer.

Parameters

self	AAC decoder handle.
buffer	Pointer to (external) ancillary data buffer.
size	Size of the buffer pointed to by buffer.

Returns

Error code.

aacDecoder_AncDataGet()

```
LINKSPEC_H AAC_DECODER_ERROR aacDecoder_AncDataGet (
              HANDLE_AACDECODER self,
              int index,
              UCHAR ** ptr,
              int * size )
```

Get one ancillary data element.

Parameters

self	AAC decoder handle.
index	Index of the ancillary data element to get.
ptr	Pointer to a buffer receiving a pointer to the requested ancillary data element.
size	Pointer to a buffer receiving the length of the requested ancillary data element.

Returns

Error code.

aacDecoder_SetParam()

```
LINKSPEC_H AAC_DECODER_ERROR aacDecoder_SetParam (
              const HANDLE_AACDECODER self,
              const AACDEC_PARAM param,
              const INT value )
```

Set one single decoder parameter.

Parameters

self	AAC decoder handle.
param	Parameter to be set.
value	Parameter value.

Returns

Error code.

aacDecoder_GetFreeBytes()

```
LINKSPEC_H AAC_DECODER_ERROR aacDecoder_GetFreeBytes (
            const HANDLE_AACDECODER self,
            UINT * pFreeBytes )
```

Get free bytes inside decoder internal buffer.

Parameters

self	Handle of AAC decoder instance.
pFreeBytes	Pointer to variable receiving amount of free bytes inside decoder internal buffer.

Returns

Error code.

aacDecoder_Open()

```
LINKSPEC_H HANDLE_AACDECODER aacDecoder_Open (
            TRANSPORT_TYPE transportFmt,
            UINT nrOfLayers )
```

Open an AAC decoder instance.

Parameters

transportFmt	The transport type to be used.
nrOfLayers	Number of transport layers.

Returns

AAC decoder handle.

aacDecoder_ConfigRaw()

```
LINKSPEC_H AAC_DECODER_ERROR aacDecoder_ConfigRaw (
            HANDLE_AACDECODER self,
```

```
            UCHAR * conf[],
            const UINT length[] )
```

Explicitly configure the decoder by passing a raw AudioSpecificConfig (ASC) or a StreamMuxConfig (SMC), contained in a binary buffer. This is required for MPEG-4 and Raw Packets file format bitstreams as well as for LATM bitstreams with no in-band SMC. If the transport format is LATM with or without LOAS, configuration is assumed to be an SMC, for all other file formats an ASC.

Parameters

self	AAC decoder handle.
conf	Pointer to an unsigned char buffer containing the binary configuration buffer (either ASC or SMC).
length	Length of the configuration buffer in bytes.

Returns

Error code.

aacDecoder_LoudnessBox()

```
LINKSPEC_H AAC_DECODER_ERROR aacDecoder_LoudnessBox (
            HANDLE_AACDECODER self,
            UCHAR * buffer,
            UINT length )
```

Submit raw payload of ISO base media file format loudness box ('ludt').

Parameters

self	AAC decoder handle.
buffer	Pointer to an unsigned char buffer containing the binary loudness box payload (without size and type).
length	Length of the payload in bytes.

Returns

Error code.

aacDecoder_Fill()

```
LINKSPEC_H AAC_DECODER_ERROR aacDecoder_Fill (
            HANDLE_AACDECODER self,
            UCHAR * pBuffer[],
            const UINT bufferSize[],
            UINT * bytesValid )
```

Fill AAC decoder's internal input buffer with bitstream data from the external input buffer. The function only copies such data as long as the decoder-internal input buffer is not full. So it grabs whatever it can from pBuffer and returns information (bytesValid) so that at a subsequent call of aacDecoder_Fill(), the right position in pBuffer can be determined to grab the next data.

Parameters

self	AAC decoder handle.
pBuffer	Pointer to external input buffer.
bufferSize	Size of external input buffer. This argument is required because decoder-internally we need the information to calculate the offset to pBuffer, where the next available data is, which is then fed into the decoder-internal buffer (as much as possible). Our example framework implementation fills the buffer at pBuffer again, once it contains no available valid bytes anymore (meaning bytesValid equal 0).
bytesValid	Number of bitstream bytes in the external bitstream buffer that have not yet been copied into the decoder's internal bitstream buffer by calling this function. The value is updated according to the amount of newly copied bytes.

Returns

Error code.

aacDecoder_DecodeFrame()

```
LINKSPEC_H AAC_DECODER_ERROR aacDecoder_DecodeFrame (
              HANDLE_AACDECODER self,
              INT_PCM * pTimeData,
              const INT timeDataSize,
              const UINT flags )
```

Decode one audio frame.

Parameters

self	AAC decoder handle.
pTimeData	Pointer to external output buffer where the decoded PCM samples will be stored into.
timeDataSize	Size of external output buffer.
flags	Bit field with flags for the decoder: (flags & AACDEC_CONCEAL) == 1: Do concealment. (flags & AACDEC_FLUSH) == 2: Discard input data. Flush filter banks (output delayed audio). (flags & AACDEC_INTR) == 4: Input data is discontinuous. Resynchronize any internals as necessary.

Returns

Error code.

aacDecoder_Close()

```
LINKSPEC_H void aacDecoder_Close (
              HANDLE_AACDECODER self )
```

De-allocate all resources of an AAC decoder instance.

Parameters

self	AAC decoder handle.

Returns

void.

aacDecoder_GetStreamInfo()

```
LINKSPEC_H CStreamInfo* aacDecoder_GetStreamInfo (
            HANDLE_AACDECODER self )
```

Get CStreamInfo handle from decoder.

Parameters

self	AAC decoder handle.

Returns

Reference to requested CStreamInfo.

aacDecoder_GetLibInfo()

```
LINKSPEC_H INT aacDecoder_GetLibInfo (
            LIB_INFO * info )
```

Get decoder library info.

Parameters

info	Pointer to an allocated LIB_INFO structure.

Returns

0 on success.

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