

# 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

**Fraunhofer Institut fuer Integrierte Schaltungen IIS,  
Fraunhofer Institute for Integrated Circuits IIS**  
<http://www.iis.fraunhofer.de/amm>

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# Contents

<b>1</b>	<b>Introduction</b>	<b>1</b>
1.1	Scope . . . . .	1
1.2	Decoder Basics . . . . .	1
<b>2</b>	<b>Library Usage</b>	<b>3</b>
2.1	API Description . . . . .	3
2.2	Calling Sequence . . . . .	3
2.2.1	Error Concealment Sequence . . . . .	7
2.3	Buffer System . . . . .	8
<b>3</b>	<b>Decoder audio output</b>	<b>9</b>
3.1	Obtaining channel mapping information . . . . .	9
3.2	Changing the audio output format . . . . .	9
3.3	Channel mapping examples . . . . .	9
3.3.1	Stereo . . . . .	9
3.3.2	Surround 5.1 . . . . .	10
3.3.3	ARIB coding mode 2/1 . . . . .	10
<b>4</b>	<b>Class Index</b>	<b>13</b>
4.1	Class List . . . . .	13
<b>5</b>	<b>File Index</b>	<b>15</b>
5.1	File List . . . . .	15
<b>6</b>	<b>Class Documentation</b>	<b>17</b>
6.1	CStreamInfo Struct Reference . . . . .	17
6.1.1	Detailed Description . . . . .	17
6.1.2	Member Data Documentation . . . . .	18
	sampleRate . . . . .	18
	frameSize . . . . .	18
	numChannels . . . . .	18
	pChannelType . . . . .	18
	pChannelIndices . . . . .	18
	aacSampleRate . . . . .	18
	profile . . . . .	18
	aot . . . . .	18
	channelConfig . . . . .	18
	bitRate . . . . .	18
	aacSamplesPerFrame . . . . .	19
	aacNumChannels . . . . .	19
	extAot . . . . .	19
	extSamplingRate . . . . .	19
	outputDelay . . . . .	19

flags	19
epConfig	19
numLostAccessUnits	19
numTotalBytes	19
numBadBytes	19
numTotalAccessUnits	19
numBadAccessUnits	19
drcProgRefLev	20
drcPresMode	20
<b>7 File Documentation</b>	<b>21</b>
7.1 aacdecoder_lib.h File Reference	21
7.1.1 Detailed Description	24
7.1.2 Macro Definition Documentation	24
IS_INIT_ERROR	24
IS_DECODE_ERROR	24
IS_OUTPUT_VALID	24
AACDEC_CONCEAL	25
AACDEC_FLUSH	25
AACDEC_INTR	25
AACDEC_CLRHIST	25
7.1.3 Typedef Documentation	25
HANDLE_AACDECODER	25
7.1.4 Enumeration Type Documentation	25
AAC_DECODER_ERROR	25
AAC_MD_PROFILE	27
AAC_DRC_DEFAULT_PRESENTATION_MODE_OPTIONS	27
AACDEC_PARAM	27
7.1.5 Function Documentation	30
aacDecoder_AncDataInit	30
aacDecoder_AncDataGet	31
aacDecoder_SetParam	31
aacDecoder_GetFreeBytes	31
aacDecoder_Open	32
aacDecoder_ConfigRaw	33
aacDecoder_RawISOBMFFData	33
aacDecoder_Fill	33
aacDecoder_DecodeFrame	34
aacDecoder_Close	34
aacDecoder_GetStreamInfo	34
aacDecoder_GetLibInfo	35
<b>Index</b>	<b>36</b>

# 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);
```



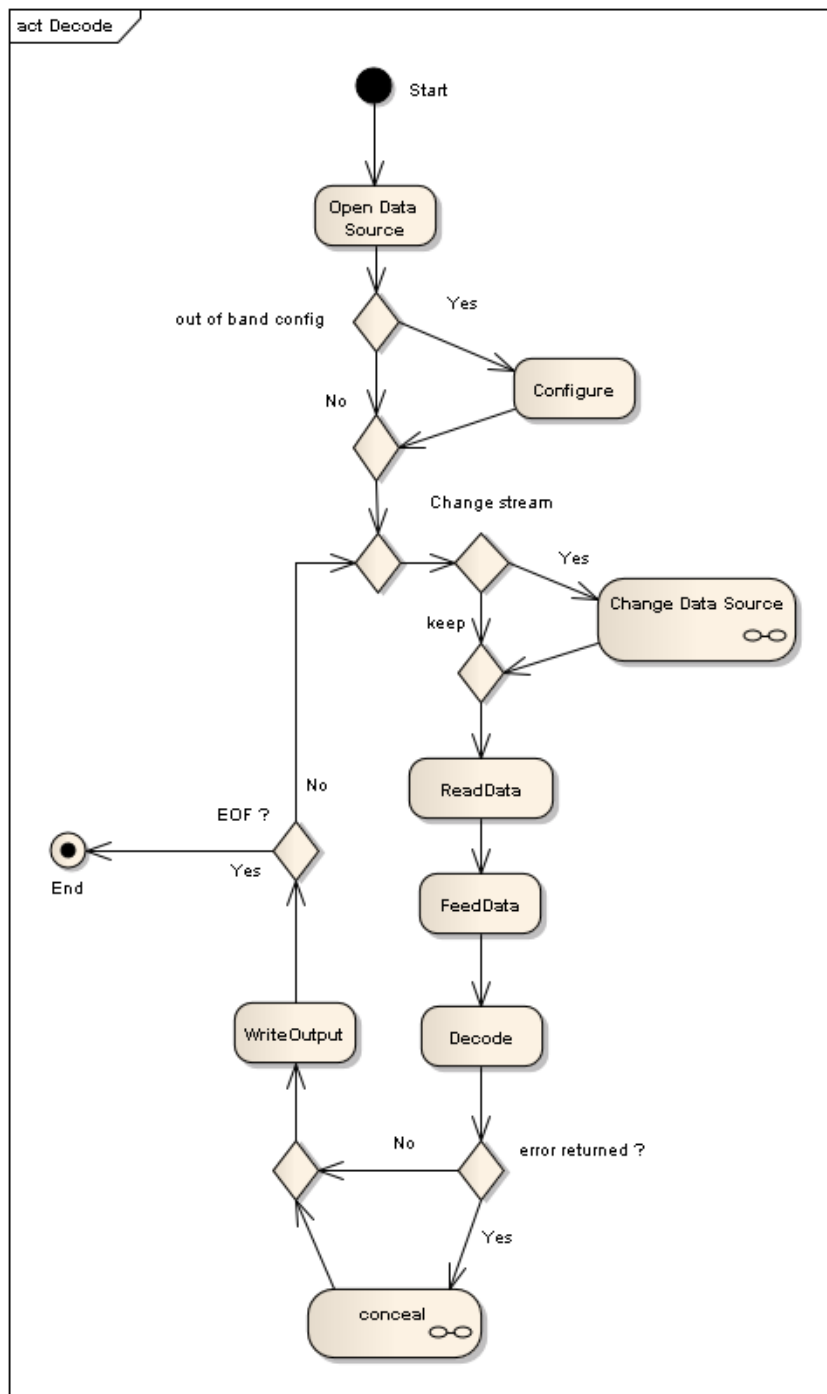


Figure 2.1: Decode calling sequence

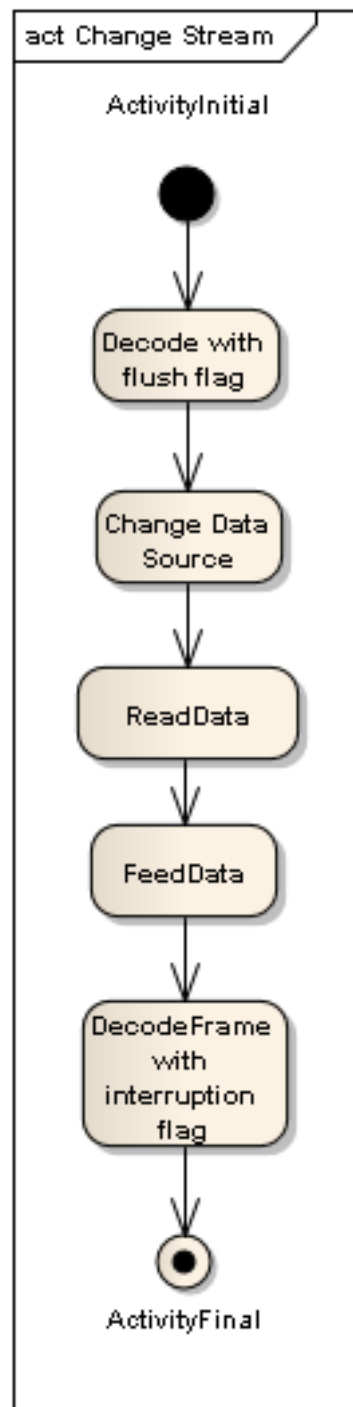


Figure 2.2: Change data source sequence

width 5cm

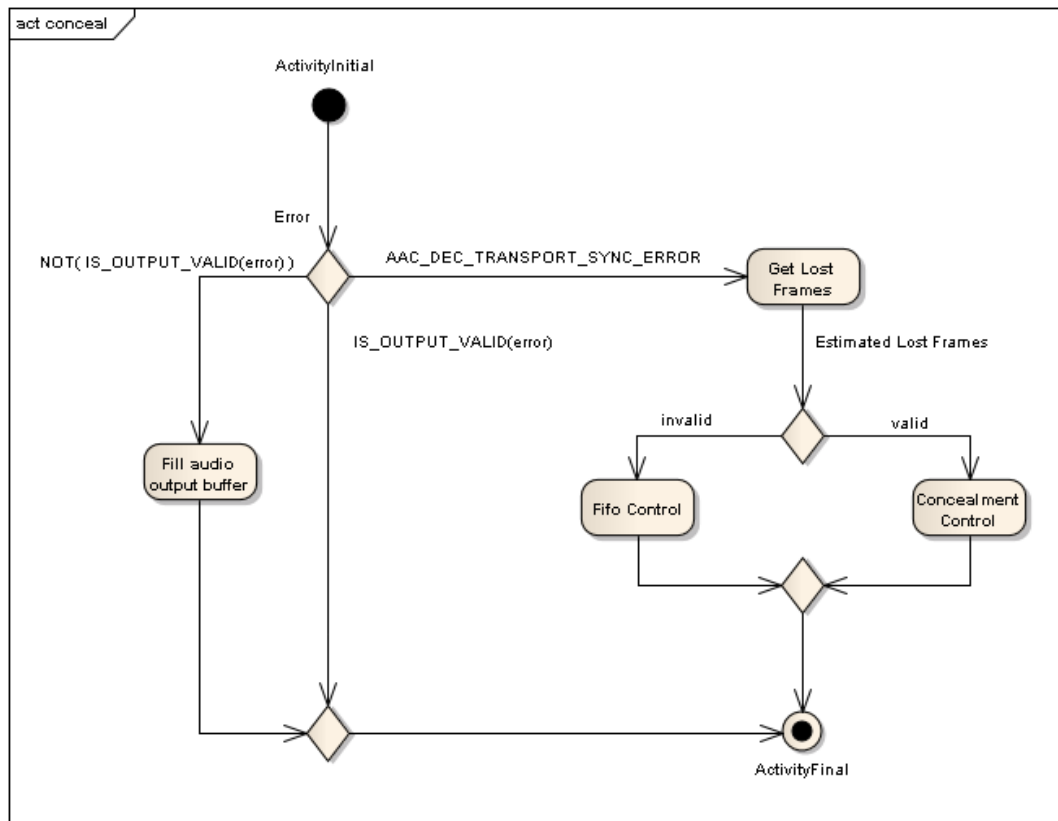


Figure 2.3: Error concealment sequence

### 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.

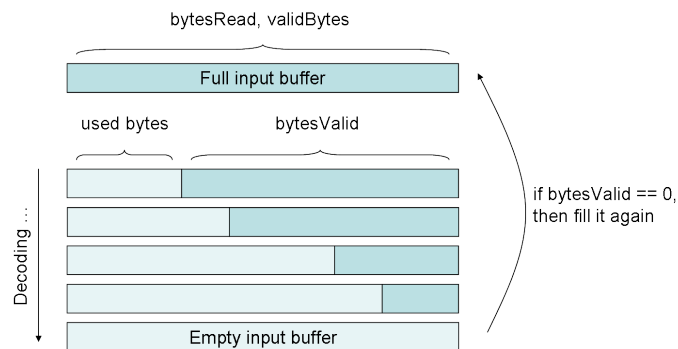


Figure 2.4: Life cycle of the external input buffer

## 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\\_CHANNEL\\_MAPPING](#). 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:

[CStreamInfo](#)

This structure gives information about the currently decoded audio data. All fields are read-only . . . . . 17



# Chapter 5

## File Index

### 5.1 File List

Here is a list of all files with brief descriptions:

[aacdecoder\\_lib.h](#)

FDK AAC decoder library interface header file . . . . . 21



# 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](#)
- INT64 [numTotalBytes](#)
- INT64 [numBadBytes](#)
- INT64 [numTotalAccessUnits](#)
- INT64 [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

### INT CStreamInfo::sampleRate

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

### 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

### INT CStreamInfo::numChannels

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

### AUDIO\_CHANNEL\_TYPE\* CStreamInfo::pChannelType

Audio channel type of each output audio channel.

### 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()

### INT CStreamInfo::aacSampleRate

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

### INT CStreamInfo::profile

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

### 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).

### INT CStreamInfo::channelConfig

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

### INT CStreamInfo::bitRate

Instantaneous bit rate.

**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

**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!

**AUDIO\_OBJECT\_TYPE CStreamInfo::extAot**

Extension Audio Object Type (from ASC)

**INT CStreamInfo::extSamplingRate**

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

**UINT CStreamInfo::outputDelay**

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

**UINT CStreamInfo::flags**

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

**SCHAR CStreamInfo::epConfig**

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

**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.

**INT64 CStreamInfo::numTotalBytes**

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

**INT64 CStreamInfo::numBadBytes**

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

**INT64 CStreamInfo::numTotalAccessUnits**

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

**INT64 CStreamInfo::numBadAccessUnits**

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

**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

1. If no level has been found in the bitstream the value is -1.

**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

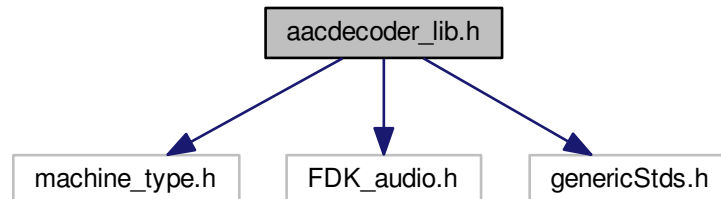
## File Documentation

### 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)
- #define [IS\\_DECODE\\_ERROR](#)(err)
- #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](#),  
[AAC\\_DEC\\_OUT\\_OF\\_MEMORY](#),  
[AAC\\_DEC\\_UNKNOWN](#),  
[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](#),  
[AAC\\_DEC\\_UNSUPPORTED\\_AOT](#),  
[AAC\\_DEC\\_UNSUPPORTED\\_FORMAT](#),  
[AAC\\_DEC\\_UNSUPPORTED\\_ER\\_FORMAT](#),  
[AAC\\_DEC\\_UNSUPPORTED\\_EPCONFIG](#),  
[AAC\\_DEC\\_UNSUPPORTED\\_MULTILAYER](#),  
[AAC\\_DEC\\_UNSUPPORTED\\_CHANNELCONFIG](#),  
[AAC\\_DEC\\_UNSUPPORTED\\_SAMPLINGRATE](#) = 0x2008,  
[AAC\\_DEC\\_INVALID\\_SBR\\_CONFIG](#),  
[AAC\\_DEC\\_SET\\_PARAM\\_FAIL](#) = 0x200A,  
[AAC\\_DEC\\_NEED\\_TO\\_RESTART](#) = 0x200B,  
[AAC\\_DEC\\_OUTPUT\\_BUFFER\\_TOO\\_SMALL](#),  
[aac\\_dec\\_init\\_error\\_end](#) = 0x2FFF,  
[aac\\_dec\\_decode\\_error\\_start](#) = 0x4000,  
[AAC\\_DEC\\_TRANSPORT\\_ERROR](#),  
[AAC\\_DEC\\_PARSE\\_ERROR](#) = 0x4002,  
[AAC\\_DEC\\_UNSUPPORTED\\_EXTENSION\\_PAYLOAD](#),  
[AAC\\_DEC\\_DECODE\\_FRAME\\_ERROR](#) = 0x4004,  
[AAC\\_DEC\\_CRC\\_ERROR](#) = 0x4005,  
[AAC\\_DEC\\_INVALID\\_CODE\\_BOOK](#) = 0x4006,  
[AAC\\_DEC\\_UNSUPPORTED\\_PREDICTION](#),  
[AAC\\_DEC\\_UNSUPPORTED\\_CCE](#) = 0x4008,  
[AAC\\_DEC\\_UNSUPPORTED\\_LFE](#) = 0x4009,  
[AAC\\_DEC\\_UNSUPPORTED\\_GAIN\\_CONTROL\\_DATA](#),  
[AAC\\_DEC\\_UNSUPPORTED\\_SBA](#),  
[AAC\\_DEC\\_TNS\\_READ\\_ERROR](#) = 0x400C,  
[AAC\\_DEC\\_RVLC\\_ERROR](#),  
[aac\\_dec\\_decode\\_error\\_end](#) = 0x4FFF,  
[aac\\_dec\\_anc\\_data\\_error\\_start](#) = 0x8000,  
[AAC\\_DEC\\_ANC\\_DATA\\_ERROR](#),  
[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](#),  
[AAC\\_MD\\_PROFILE\\_MPEG\\_LEGACY](#),  
[AAC\\_MD\\_PROFILE\\_MPEG\\_LEGACY\\_PRIO](#),

[AAC\\_MD\\_PROFILE\\_ARIB\\_JAPAN](#) }

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](#),  
[AAC\\_DRC\\_PRESENTATION\\_MODE\\_1\\_DEFAULT](#),  
[AAC\\_DRC\\_PRESENTATION\\_MODE\\_2\\_DEFAULT](#) }

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

- enum [AACDEC\\_PARAM](#) {  
[AAC\\_PCM\\_DUAL\\_CHANNEL\\_OUTPUT\\_MODE](#),  
[AAC\\_PCM\\_OUTPUT\\_CHANNEL\\_MAPPING](#),  
[AAC\\_PCM\\_LIMITER\\_ENABLE](#),  
[AAC\\_PCM\\_LIMITER\\_ATTACK\\_TIME](#) = 0x0005,  
[AAC\\_PCM\\_LIMITER\\_RELEASE\\_TIME](#) = 0x0006,  
[AAC\\_PCM\\_MIN\\_OUTPUT\\_CHANNELS](#),  
[AAC\\_PCM\\_MAX\\_OUTPUT\\_CHANNELS](#),  
[AAC\\_METADATA\\_PROFILE](#),  
[AAC\\_METADATA\\_EXPIRY\\_TIME](#) = 0x0021,  
[AAC\\_CONCEAL\\_METHOD](#) = 0x0100,  
[AAC\\_DRC\\_BOOST\\_FACTOR](#),  
[AAC\\_DRC\\_ATTENUATION\\_FACTOR](#),  
[AAC\\_DRC\\_REFERENCE\\_LEVEL](#),  
[AAC\\_DRC\\_HEAVY\\_COMPRESSION](#),  
[AAC\\_DRC\\_DEFAULT\\_PRESENTATION\\_MODE](#),  
[AAC\\_DRC\\_ENC\\_TARGET\\_LEVEL](#),  
[AAC\\_UNIDRC\\_ALBUM\\_MODE](#),  
[AAC\\_QMF\\_LOWPOWER](#) = 0x0300,  
[AAC\\_TPDEC\\_CLEAR\\_BUFFER](#),  
[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_RawISOBMFFData` ([HANDLE\\_AACDECODER](#) self, UCHAR \*buffer, UINT length)
 

*Submit raw ISO base media file format boxes to decoder for parsing (only some box types are recognized).*
- 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

**#define IS\_INIT\_ERROR( err )**

**Value:**

```
((err) >= aac_dec_init_error_start) && ((err) <=
aac_dec_init_error_end) \
? 1
: 0)
```

Macro to identify initialization errors. Output buffer is invalid.

**#define IS\_DECODE\_ERROR( err )**

**Value:**

```
((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.

**#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.

**#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.

**#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.

**#define AACDEC\_INTR 4**

Flag for `aacDecoder.DecodeFrame()`: Signal an input bit stream data \ discontinuity. Resync any internals as necessary.

**#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****typedef struct AAC\_DECODER\_INSTANCE\* HANDLE\_AACDECODER**

Pointer to a AAC decoder instance.

**7.1.4 Enumeration Type Documentation****enum AAC\_DECODER\_ERROR**

AAC decoder error codes.

Enumerator

**AAC\_DEC\_OK** No error occurred. Output buffer is valid and error free.

**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.

**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\_CODEBOOK** 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**

**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 Amendment 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](#)).

**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)

**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\_RELEASE\_TIME** Signal level limiting release time in ms. Default configuration is 50 ms. Adjustable time must be larger than 0 ms.

**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.

**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.

**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.

**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\_UNIDRC\_ALBUM\_MODE** MPEG-D DRC: Enable album mode. 0: Disabled (default), 1: Enabled.

Disabled album mode leads to application of gain sequences for fading in and out, if provided in the bitstream.

Enabled album mode makes use of dedicated album loudness information, if provided in the bitstream.

**AAC\_QMF\_LOWPPOWER** 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

**LINKSPEC.H AAC\_DECODER\_ERROR** `aacDecoder_AncDataInit ( HANDLE_AACDECODER self, UCHAR * buffer, int size )`

Initialize ancillary data buffer.

## Parameters

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

## Returns

Error code.

**LINKSPEC.H AAC\_DECODER\_ERROR aacDecoder.AncDataGet ( HANDLE\_AACDECODER *self*, int *index*, UCHAR \*\* *ptr*, int \* *size* )**

Get one ancillary data element.

## Parameters

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

## Returns

Error code.

**LINKSPEC.H AAC\_DECODER\_ERROR aacDecoder.SetParam ( const HANDLE\_AACDECODER *self*, const AACDEC\_PARAM *param*, const INT *value* )**

Set one single decoder parameter.

## Parameters

<i>self</i>	AAC decoder handle.
<i>param</i>	Parameter to be set.
<i>value</i>	Parameter value.

## Returns

Error code.

**LINKSPEC.H AAC\_DECODER\_ERROR aacDecoder.GetFreeBytes ( const HANDLE\_AACDECODER *self*, UINT \* *pFreeBytes* )**

Get free bytes inside decoder internal buffer.

## Parameters

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

## Returns

Error code.

**LINKSPEC.H HANDLE\_AACDECODER aacDecoder\_Open ( TRANSPORT\_TYPE  
*transportFmt*, UINT *nrOfLayers* )**

Open an AAC decoder instance.

## Parameters

<i>transportFmt</i>	The transport type to be used.
<i>nrOfLayers</i>	Number of transport layers.

## Returns

AAC decoder handle.

**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 StreamMux-Config (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

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

## Returns

Error code.

**LINKSPEC.H AAC\_DECODER\_ERROR aacDecoder\_RawISOBMFFData ( HANDLE\_AACDECODER self, UCHAR \* buffer, UINT length )**

Submit raw ISO base media file format boxes to decoder for parsing (only some box types are recognized).

## Parameters

<i>self</i>	AAC decoder handle.
<i>buffer</i>	Pointer to an unsigned char buffer containing the binary box data (including size and type, can be a sequence of multiple boxes).
<i>length</i>	Length of the data in bytes.

## Returns

Error code.

**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

<i>self</i>	AAC decoder handle.
<i>pBuffer</i>	Pointer to external input buffer.
<i>bufferSize</i>	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).
<i>bytesValid</i>	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.

**LINKSPEC.H AAC\_DECODER\_ERROR aacDecoder\_DecodeFrame ( HANDLE\_AACDECODER *self*, INT\_PCM \* *pTimeData*, const INT *timeDataSize*, const UINT *flags* )**

Decode one audio frame.

## Parameters

<i>self</i>	AAC decoder handle.
<i>pTimeData</i>	Pointer to external output buffer where the decoded PCM samples will be stored into.
<i>timeDataSize</i>	Size of external output buffer.
<i>flags</i>	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. (flags & AACDEC_CLRHIST) == 8: Clear all signal delay lines and history buffers.

## Returns

Error code.

**LINKSPEC.H void aacDecoder\_Close ( HANDLE\_AACDECODER *self* )**

De-allocate all resources of an AAC decoder instance.

## Parameters

<i>self</i>	AAC decoder handle.
-------------	---------------------

## Returns

void.

**LINKSPEC.H CStreamInfo\* aacDecoder\_GetStreamInfo ( HANDLE\_AACDECODER *self* )**

Get [CStreamInfo](#) handle from decoder.

**Parameters**

<i>self</i>	AAC decoder handle.
-------------	---------------------

**Returns**

Reference to requested [CStreamInfo](#).

**LINKSPEC.H INT aacDecoder\_GetLibInfo ( LIB\_INFO \* *info* )**

Get decoder library info.

**Parameters**

<i>info</i>	Pointer to an allocated LIB_INFO structure.
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**Returns**

0 on success.

# Index

- AAC\_CONCEAL\_METHOD  
aacdecoder.lib.h, 29
- AAC\_DEC\_ANC\_DATA\_ERROR  
aacdecoder.lib.h, 26
- AAC\_DEC\_CRC\_ERROR  
aacdecoder.lib.h, 26
- AAC\_DEC\_DECODE\_FRAME\_ERROR  
aacdecoder.lib.h, 26
- AAC\_DEC\_INVALID\_CODE\_BOOK  
aacdecoder.lib.h, 26
- AAC\_DEC\_INVALID\_HANDLE  
aacdecoder.lib.h, 25
- AAC\_DEC\_INVALID\_SBR\_CONFIG  
aacdecoder.lib.h, 26
- AAC\_DEC\_NEED\_TO\_RESTART  
aacdecoder.lib.h, 26
- AAC\_DEC\_NOT\_ENOUGH\_BITS  
aacdecoder.lib.h, 25
- AAC\_DEC\_OK  
aacdecoder.lib.h, 25
- AAC\_DEC\_OUT\_OF\_MEMORY  
aacdecoder.lib.h, 25
- AAC\_DEC\_OUTPUT\_BUFFER\_TOO\_SMALL  
aacdecoder.lib.h, 26
- AAC\_DEC\_PARSE\_ERROR  
aacdecoder.lib.h, 26
- AAC\_DEC\_RVLC\_ERROR  
aacdecoder.lib.h, 26
- AAC\_DEC\_SET\_PARAM\_FAIL  
aacdecoder.lib.h, 26
- AAC\_DEC\_TNS\_READ\_ERROR  
aacdecoder.lib.h, 26
- AAC\_DEC\_TOO\_MANY\_ANC\_ELEMENTS  
aacdecoder.lib.h, 26
- AAC\_DEC\_TOO\_SMALL\_ANC\_BUFFER  
aacdecoder.lib.h, 26
- AAC\_DEC\_TRANSPORT\_ERROR  
aacdecoder.lib.h, 26
- AAC\_DEC\_TRANSPORT\_SYNC\_ERROR  
aacdecoder.lib.h, 25
- AAC\_DEC\_UNKNOWN  
aacdecoder.lib.h, 25
- AAC\_DEC\_UNSUPPORTED\_AOT  
aacdecoder.lib.h, 25
- AAC\_DEC\_UNSUPPORTED\_CCE  
aacdecoder.lib.h, 26
- AAC\_DEC\_UNSUPPORTED\_CHANNELCONFIG  
aacdecoder.lib.h, 26
- AAC\_DEC\_UNSUPPORTED\_EPCONFIG  
aacdecoder.lib.h, 25
- AAC\_DEC\_UNSUPPORTED\_ER\_FORMAT  
aacdecoder.lib.h, 25
- AAC\_DEC\_UNSUPPORTED\_EXTENSION\_PAYLOAD  
aacdecoder.lib.h, 26
- AAC\_DEC\_UNSUPPORTED\_FORMAT  
aacdecoder.lib.h, 25
- AAC\_DEC\_UNSUPPORTED\_GAIN\_CONTROL\_DATA  
aacdecoder.lib.h, 26
- AAC\_DEC\_UNSUPPORTED\_LFE  
aacdecoder.lib.h, 26
- AAC\_DEC\_UNSUPPORTED\_MULTILAYER  
aacdecoder.lib.h, 25
- AAC\_DEC\_UNSUPPORTED\_PREDICTION  
aacdecoder.lib.h, 26
- AAC\_DEC\_UNSUPPORTED\_SAMPLINGRATE  
aacdecoder.lib.h, 26
- AAC\_DEC\_UNSUPPORTED\_SBA  
aacdecoder.lib.h, 26
- AAC\_DRC\_ATTENUATION\_FACTOR  
aacdecoder.lib.h, 29
- AAC\_DRC\_BOOST\_FACTOR  
aacdecoder.lib.h, 29
- AAC\_DRC\_DEFAULT\_PRESENTATION\_MODE  
aacdecoder.lib.h, 29
- AAC\_DRC\_ENC\_TARGET\_LEVEL  
aacdecoder.lib.h, 30
- AAC\_DRC\_HEAVY\_COMPRESSION  
aacdecoder.lib.h, 29
- AAC\_DRC\_PARAMETER\_HANDLING\_DISABLED  
aacdecoder.lib.h, 27
- AAC\_DRC\_PARAMETER\_HANDLING\_ENABLED  
aacdecoder.lib.h, 27
- AAC\_DRC\_PRESENTATION\_MODE\_1\_DEFAULT  
aacdecoder.lib.h, 27
- AAC\_DRC\_PRESENTATION\_MODE\_2\_DEFAULT  
aacdecoder.lib.h, 27
- AAC\_DRC\_REFERENCE\_LEVEL  
aacdecoder.lib.h, 29



- AAC\_MD\_PROFILE\_ARIB\_JAPAN  
   aacdecoder.lib.h, 27  
 AAC\_MD\_PROFILE\_MPEG\_LEGACY  
   aacdecoder.lib.h, 27  
 AAC\_MD\_PROFILE\_MPEG\_LEGACY\_PRIO  
   aacdecoder.lib.h, 27  
 AAC\_MD\_PROFILE\_MPEG\_STANDARD  
   aacdecoder.lib.h, 27  
 AAC\_METADATA\_EXPIRY\_TIME  
   aacdecoder.lib.h, 29  
 AAC\_METADATA\_PROFILE  
   aacdecoder.lib.h, 29  
 AAC\_PCM\_DUAL\_CHANNEL\_OUTPUT\_MODE  
   aacdecoder.lib.h, 27  
 AAC\_PCM\_LIMITER\_ATTACK\_TIME  
   aacdecoder.lib.h, 28  
 AAC\_PCM\_LIMITER\_ENABLE  
   aacdecoder.lib.h, 28  
 AAC\_PCM\_LIMITER\_RELEASE\_TIME  
   aacdecoder.lib.h, 28  
 AAC\_PCM\_MAX\_OUTPUT\_CHANNELS  
   aacdecoder.lib.h, 28  
 AAC\_PCM\_MIN\_OUTPUT\_CHANNELS  
   aacdecoder.lib.h, 28  
 AAC\_PCM\_OUTPUT\_CHANNEL\_MAPPING  
   aacdecoder.lib.h, 27  
 AAC\_QMF\_LOWPOWER  
   aacdecoder.lib.h, 30  
 AAC\_TPDEC\_CLEAR\_BUFFER  
   aacdecoder.lib.h, 30  
 AAC\_UNIDRC\_ALBUM\_MODE  
   aacdecoder.lib.h, 30  
 AAC\_UNIDRC\_SET\_EFFECT  
   aacdecoder.lib.h, 30  
 aac\_dec\_anc\_data\_error\_end  
   aacdecoder.lib.h, 26  
 aac\_dec\_anc\_data\_error\_start  
   aacdecoder.lib.h, 26  
 aac\_dec\_decode\_error\_end  
   aacdecoder.lib.h, 26  
 aac\_dec\_decode\_error\_start  
   aacdecoder.lib.h, 26  
 aac\_dec\_init\_error\_end  
   aacdecoder.lib.h, 26  
 aac\_dec\_init\_error\_start  
   aacdecoder.lib.h, 25  
 aac\_dec\_sync\_error\_end  
   aacdecoder.lib.h, 25  
 aac\_dec\_sync\_error\_start  
   aacdecoder.lib.h, 25  
 AAC\_DECODER\_ERROR  
   aacdecoder.lib.h, 25  
 AAC\_DRC\_DEFAULT\_PRESENTATION\_MODE\_OPTIONS  
   aacdecoder.lib.h, 27  
 AAC\_MD\_PROFILE  
   aacdecoder.lib.h, 26  
 AACDEC\_CLRHIST  
   aacdecoder.lib.h, 25  
 AACDEC\_CONCEAL  
   aacdecoder.lib.h, 24  
 AACDEC\_FLUSH  
   aacdecoder.lib.h, 25  
 AACDEC\_INTR  
   aacdecoder.lib.h, 25  
 AACDEC\_PARAM  
   aacdecoder.lib.h, 27  
 aacdecoder.lib.h  
   AAC\_CONCEAL\_METHOD, 29  
   AAC\_DEC\_ANC\_DATA\_ERROR, 26  
   AAC\_DEC\_CRC\_ERROR, 26  
   AAC\_DEC\_DECODE\_FRAME\_ERROR, 26  
   AAC\_DEC\_INVALID\_CODE\_BOOK, 26  
   AAC\_DEC\_INVALID\_HANDLE, 25  
   AAC\_DEC\_INVALID\_SBR\_CONFIG, 26  
   AAC\_DEC\_NEED\_TO\_RESTART, 26  
   AAC\_DEC\_NOT\_ENOUGH\_BITS, 25  
   AAC\_DEC\_OK, 25  
   AAC\_DEC\_OUT\_OF\_MEMORY, 25  
   AAC\_DEC\_OUTPUT\_BUFFER\_TOO\_SMALL,  
     26  
   AAC\_DEC\_PARSE\_ERROR, 26  
   AAC\_DEC\_RVLC\_ERROR, 26  
   AAC\_DEC\_SET\_PARAM\_FAIL, 26  
   AAC\_DEC\_TNS\_READ\_ERROR, 26  
   AAC\_DEC\_TOO\_MANY\_ANC\_ELEMENTS,  
     26  
   AAC\_DEC\_TOO\_SMALL\_ANC\_BUFFER, 26  
   AAC\_DEC\_TRANSPORT\_ERROR, 26  
   AAC\_DEC\_TRANSPORT\_SYNC\_ERROR, 25  
   AAC\_DEC\_UNKNOWN, 25  
   AAC\_DEC\_UNSUPPORTED\_AOT, 25  
   AAC\_DEC\_UNSUPPORTED\_CCE, 26  
   AAC\_DEC\_UNSUPPORTED\_CHANNELCONFIG,  
     26  
   AAC\_DEC\_UNSUPPORTED\_EPCONFIG, 25  
   AAC\_DEC\_UNSUPPORTED\_ER\_FORMAT,  
     25  
   AAC\_DEC\_UNSUPPORTED\_EXTENSION\_PAYLOAD,  
     26  
   AAC\_DEC\_UNSUPPORTED\_FORMAT, 25  
   AAC\_DEC\_UNSUPPORTED\_GAIN\_CONTROL\_DATA,  
     26  
   AAC\_DEC\_UNSUPPORTED\_LFE, 26  
   AAC\_DEC\_UNSUPPORTED\_MULTILAYER,  
     25  
   AAC\_DEC\_UNSUPPORTED\_PREDICTION,  
     26

- AAC\_DEC\_UNSUPPORTED\_SAMPLINGRATE, 26
- AAC\_DEC\_UNSUPPORTED\_SBA, 26
- AAC\_DRC\_ATTENUATION\_FACTOR, 29
- AAC\_DRC\_BOOST\_FACTOR, 29
- AAC\_DRC\_DEFAULT\_PRESENTATION\_MODE, 29
- AAC\_DRC\_ENC\_TARGET\_LEVEL, 30
- AAC\_DRC\_HEAVY\_COMPRESSION, 29
- AAC\_DRC\_PARAMETER\_HANDLING\_DISABLED, 27
- AAC\_DRC\_PARAMETER\_HANDLING\_ENABLED, 27
- AAC\_DRC\_PRESENTATION\_MODE\_1\_DEFAULT, 27
- AAC\_DRC\_PRESENTATION\_MODE\_2\_DEFAULT, 27
- AAC\_DRC\_REFERENCE\_LEVEL, 29
- AAC\_MD\_PROFILE\_ARIB\_JAPAN, 27
- AAC\_MD\_PROFILE\_MPEG\_LEGACY, 27
- AAC\_MD\_PROFILE\_MPEG\_LEGACY\_P10, 27
- AAC\_MD\_PROFILE\_MPEG\_STANDARD, 27
- AAC\_METADATA\_EXPIRY\_TIME, 29
- AAC\_METADATA\_PROFILE, 29
- AAC\_PCM\_DUAL\_CHANNEL\_OUTPUT\_MODE, 27
- AAC\_PCM\_LIMITER\_ATTACK\_TIME, 28
- AAC\_PCM\_LIMITER\_ENABLE, 28
- AAC\_PCM\_LIMITER\_RELEASE\_TIME, 28
- AAC\_PCM\_MAX\_OUTPUT\_CHANNELS, 28
- AAC\_PCM\_MIN\_OUTPUT\_CHANNELS, 28
- AAC\_PCM\_OUTPUT\_CHANNEL\_MAPPING, 27
- AAC\_QMF\_LOWPPOWER, 30
- AAC\_TPDEC\_CLEAR\_BUFFER, 30
- AAC\_UNIDRC\_ALBUM\_MODE, 30
- AAC\_UNIDRC\_SET\_EFFECT, 30
- aac\_dec\_anc\_data\_error\_end, 26
- aac\_dec\_anc\_data\_error\_start, 26
- aac\_dec\_decode\_error\_end, 26
- aac\_dec\_decode\_error\_start, 26
- aac\_dec\_init\_error\_end, 26
- aac\_dec\_init\_error\_start, 25
- aac\_dec\_sync\_error\_end, 25
- aac\_dec\_sync\_error\_start, 25
- aacDecoder\_AncDataGet
  - aacdecoder.lib.h, 31
- aacDecoder\_AncDataInit
  - aacdecoder.lib.h, 30
- aacDecoder\_Close
  - aacdecoder.lib.h, 34
- aacDecoder\_ConfigRaw
  - aacdecoder.lib.h, 33
- aacDecoder\_DecodeFrame
  - aacdecoder.lib.h, 34
- aacDecoder\_Fill
  - aacdecoder.lib.h, 33
- aacDecoder\_GetFreeBytes
  - aacdecoder.lib.h, 31
- aacDecoder\_GetLibInfo
  - aacdecoder.lib.h, 35
- aacDecoder\_GetStreamInfo
  - aacdecoder.lib.h, 34
- AAC\_DECODER\_ERROR, 25
- AAC\_DRC\_DEFAULT\_PRESENTATION\_MODE\_OPTIONS, 27
- AAC\_MD\_PROFILE, 26
- AACDEC\_CLRHIST, 25
- AACDEC\_CONCEAL, 24
- AACDEC\_FLUSH, 25
- AACDEC\_INTR, 25
- AACDEC\_PARAM, 27
- aacDecoder\_AncDataGet, 31
- aacDecoder\_AncDataInit, 30
- aacDecoder\_Close, 34
- aacDecoder\_ConfigRaw, 33
- aacDecoder\_DecodeFrame, 34
- aacDecoder\_Fill, 33
- aacDecoder\_GetFreeBytes, 31
- aacDecoder\_GetLibInfo, 35
- aacDecoder\_GetStreamInfo, 34
- aacDecoder\_Open, 31
- aacDecoder\_RawISOBMFFData, 33
- aacDecoder\_SetParam, 31
- HANDLE\_AACDECODER, 25
- IS\_DECODE\_ERROR, 24
- IS\_INIT\_ERROR, 24
- IS\_OUTPUT\_VALID, 24
- aacDecoder\_Open
  - aacdecoder.lib.h, 31
- aacDecoder\_RawISOBMFFData
  - aacdecoder.lib.h, 33
- aacDecoder\_SetParam
  - aacdecoder.lib.h, 31
- aacNumChannels
  - CStreamInfo, 19
- aacSampleRate
  - CStreamInfo, 18
- aacSamplesPerFrame
  - CStreamInfo, 18
- aot
  - CStreamInfo, 18
- bitRate
  - CStreamInfo, 18

- channelConfig
    - CStreamInfo, 18
  - CStreamInfo, 17
    - aacNumChannels, 19
    - aacSampleRate, 18
    - aacSamplesPerFrame, 18
    - aot, 18
    - bitRate, 18
    - channelConfig, 18
    - drcPresMode, 20
    - drcProgRefLev, 19
    - epConfig, 19
    - extAot, 19
    - extSamplingRate, 19
    - flags, 19
    - frameSize, 18
    - numBadAccessUnits, 19
    - numBadBytes, 19
    - numChannels, 18
    - numLostAccessUnits, 19
    - numTotalAccessUnits, 19
    - numTotalBytes, 19
    - outputDelay, 19
    - pChannelIndices, 18
    - pChannelType, 18
    - profile, 18
    - sampleRate, 18
  - drcPresMode
    - CStreamInfo, 20
  - drcProgRefLev
    - CStreamInfo, 19
  - epConfig
    - CStreamInfo, 19
  - extAot
    - CStreamInfo, 19
  - extSamplingRate
    - CStreamInfo, 19
  - flags
    - CStreamInfo, 19
  - frameSize
    - CStreamInfo, 18
  - HANDLE\_AACDECODER
    - aacdecoder\_lib.h, 25
  - IS\_DECODE\_ERROR
    - aacdecoder\_lib.h, 24
  - IS\_INIT\_ERROR
    - aacdecoder\_lib.h, 24
  - IS\_OUTPUT\_VALID
    - aacdecoder\_lib.h, 24
  - numBadAccessUnits
    - CStreamInfo, 19
  - numBadBytes
    - CStreamInfo, 19
  - numChannels
    - CStreamInfo, 18
  - numLostAccessUnits
    - CStreamInfo, 19
  - numTotalAccessUnits
    - CStreamInfo, 19
  - numTotalBytes
    - CStreamInfo, 19
  - outputDelay
    - CStreamInfo, 19
  - pChannelIndices
    - CStreamInfo, 18
  - pChannelType
    - CStreamInfo, 18
  - profile
    - CStreamInfo, 18
  - sampleRate
    - CStreamInfo, 18
-