

Dolby AC3 Version3 Decoder on C64x+

User Guide



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Read This First

About This Manual

This document describes how to install and work with Texas Instruments' (TI) Dolby AC3 Version3 Decoder implementation on the C64x+ platform. It also provides a detailed Application Programming Interface (API) reference and information on the sample application that accompanies this component.

TI's codec implementations are based on the eXpressDSP Digital Media (XDM) standard. XDM is an extension of the eXpressDSP Algorithm Interface Standard (XDAIS).

Intended Audience

This document is intended for system engineers who want to integrate TI's codecs with other software to build a multimedia system based on the C64x+ platform.

This document assumes that you are fluent in the C language, have a good working knowledge of Digital Signal Processing (DSP), digital signal processors, and DSP applications. Good knowledge of eXpressDSP Algorithm Interface Standard (XDAIS) and eXpressDSP Digital Media (XDM) standard will be helpful.

How to Use This Manual

This document includes the following chapters:

- ❑ **Chapter 1 - Introduction**, provides a brief introduction to the XDAIS and XDM standards. It also provides an overview of the codec and lists its supported features.
- ❑ **Chapter 2 - Installation Overview**, describes how to install, build, and run the codec.
- ❑ **Chapter 3 - Sample Usage**, describes the sample usage of the codec.
- ❑ **Chapter 4 - API Reference**, describes the data structures and interface functions used in the codec.

Related Documentation From Texas Instruments

The following documents describe TI's DSP algorithm standards such as, XDAIS and XDM. To obtain a copy of any of these TI documents, visit the Texas Instruments website at www.ti.com.

- ❑ *TMS320 DSP Algorithm Standard Rules and Guidelines* (literature number SPRU352) defines a set of requirements for DSP algorithms that, if followed, allow system integrators to quickly assemble production-quality systems from one or more such algorithms.
- ❑ *TMS320 DSP Algorithm Standard API Reference* (literature number SPRU360) describes all the APIs that are defined by the TMS320 DSP Algorithm Interoperability Standard (also known as XDAIS) specification.
- ❑ *Technical Overview of eXpressDSP - Compliant Algorithms for DSP Software Producers* (literature number SPRA579) describes how to make algorithms compliant with the TMS320 DSP Algorithm Standard which is part of TI's eXpressDSP technology initiative.
- ❑ *Using the TMS320 DSP Algorithm Standard in a Static DSP System* (literature number SPRA577) describes how an eXpressDSP-compliant algorithm may be used effectively in a static system with limited memory.
- ❑ *DMA Guide for eXpressDSP-Compliant Algorithm Producers and Consumers* (literature number SPRA445) describes the DMA architecture specified by the TMS320 DSP Algorithm Standard (XDAIS). It also describes two sets of APIs used for accessing DMA resources: the IDMA2 abstract interface and the ACPY2 library.
- ❑ *eXpressDSP Digital Media (XDM) Standard API Reference* (literature number SPRUEC8)

The following documents describe TMS320 devices and related support tools:

- ❑ *Design and Implementation of an eXpressDSP-Compliant DMA Manager for C6X1X* (literature number SPRA789) describes a C6x1x-optimized (C6211, C6711) ACPY2 library implementation and DMA Resource Manager.
- ❑ *TMS320C64x+ Megamodule* (literature number SPRAA68) describes the enhancements made to the internal memory and describes the new features which have been added to support the internal memory architecture's performance and protection.
- ❑ *TMS320C64x+ DSP Megamodule Reference Guide* (literature number SPRU871) describes the C64x+ megamodule peripherals.
- ❑ *TMS320C64x to TMS320C64x+ CPU Migration Guide* (literature number SPRAA84) describes migration from the Texas Instruments TMS320C64x™ digital signal processor (DSP) to the TMS320C64x+™ DSP.
- ❑ *TMS320C6000 Optimizing Compiler v 6.0 Beta User's Guide* (literature number SPRU187N) explains how to use compiler tools such as compiler, assembly optimizer, standalone simulator, library-build utility, and C++ name demangler.
- ❑ *TMS320C64x/C64x+ DSP CPU and Instruction Set Reference Guide* (literature number SPRU732) describes the CPU architecture, pipeline, instruction set, and interrupts of the C64x and C64x+ DSPs.

- ❑ TMS320DM6446 Digital Media System-on-Chip (literature number SPRS283)
- ❑ *TMS320DM6446 Digital Media System-on-Chip Errata (Silicon Revision 1.0)* (literature number SPRZ241) describes the known exceptions to the functional specifications for the TMS320DM6446 Digital Media System-on-Chip (DMSoC).
- ❑ TMS320DM6443 Digital Media System-on-Chip (literature number SPRS282)
- ❑ *TMS320DM6443 Digital Media System-on-Chip Errata (Silicon Revision 1.0)* (literature number SPRZ240) describes the known exceptions to the functional specifications for the TMS320DM6443 Digital Media System-on-Chip (DMSoC).
- ❑ *TMS320DM644x DMSoC DSP Subsystem Reference Guide* (literature number SPRUE15) describes the digital signal processor (DSP) subsystem in the TMS320DM644x Digital Media System-on-Chip (DMSoC).
- ❑ *TMS320DM644x DMSoC ARM Subsystem Reference Guide* (literature number SPRUE14) describes the ARM subsystem in the TMS320DM644x Digital Media System on a Chip (DMSoC).

Related Documentation

You can use the following documents to supplement this user guide:

- ❑ *ATSC A/52, Digital Audio Compression (AC-3), United States Advanced Television Committee*
- ❑ *Dolby Digital Consumer Decoder (AC3) algorithm - 3 (7.1.0) of the Dolby Digital C Simulator Decoder Source Code*
- ❑ *DolbySIP5_0.pdf – Dolby Software Interface Protocol version 5.0*

Abbreviations

The following abbreviations are used in this document:

Table 1-1. List of Abbreviations

Abbreviation	Description
API	Application Programming Interface
AC3	Audio Coding 3
DRM	Digital Rights Management
EVM	Evaluation Module
PCM	Pulse Coded Modulation
ATSC	Advanced Television Systems Committee
HDTV	High Definition Television
DVD	Digital versatile Disc
XDAIS	eXpressDSP Algorithm Interface Standard
XDM	eXpressDSP Digital Media

Text Conventions

The following conventions are used in this document:

- Text inside back-quotes (“”) represents pseudo-code.
- Program source code, function and macro names, parameters, and command line commands are shown in a `mono-spaced` font.

Product Support

When contacting TI for support on this codec, please quote the product name (Dolby AC3 Version3 Decoder on C64x+) and version number. The version number of the codec is included in the Title of the Release Notes that accompanies this codec.

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Introduction

This chapter provides a brief introduction to XDAIS and XDM. It also provides an overview of TI's implementation of the Dolby AC3 Version3 Decoder on the C64x+ platform and its supported features.

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1.1 Overview of XDAIS and XDM

TI's multimedia codec implementations are based on the eXpressDSP Digital Media (XDM) standard. XDM is an extension of the eXpressDSP Algorithm Interface Standard (XDAIS).

1.1.1 XDAIS Overview

An eXpressDSP-compliant algorithm is a module that implements the abstract interface IALG. The IALG API takes the memory management function away from the algorithm and places it in the hosting framework. Thus, an interaction occurs between the algorithm and the framework. This interaction allows the client application to allocate memory for the algorithm and also share memory between algorithms. It also allows the memory to be moved around while an algorithm is operating in the system. In order to facilitate these functionalities, the IALG interface defines the following APIs:

- ❑ `algAlloc()`
- ❑ `algInit()`
- ❑ `algActivate()`
- ❑ `algDeactivate()`
- ❑ `algFree()`

The `algAlloc()` API allows the algorithm to communicate its memory requirements to the client application. The `algInit()` API allows the algorithm to initialize the memory allocated by the client application. The `algFree()` API allows the algorithm to communicate the memory to be freed when an instance is no longer required.

Once an algorithm instance object is created, it can be used to process data in real-time. The `algActivate()` API provides a notification to the algorithm instance that one or more algorithm processing methods is about to be run zero or more times in succession. After the processing methods have been run, the client application calls the `algDeactivate()` API prior to reusing any of the instance's scratch memory.

The IALG interface also defines three more optional APIs `algControl()`, `algNumAlloc()`, and `algMoved()`. For more details on these APIs, see *TMS320 DSP Algorithm Standard API Reference* (literature number SPRU360).

1.1.2 XDM Overview

In the multimedia application space, you have the choice of integrating any codec into your multimedia system. For example, if you are building a video decoder system, you can use any of the available video decoders (such as MPEG4, H.263, or H.264) in your system. To enable easy integration with the client application, it is important that all codecs with similar functionality use similar APIs. XDM was primarily defined as an extension to XDAIS to ensure uniformity across different classes of codecs

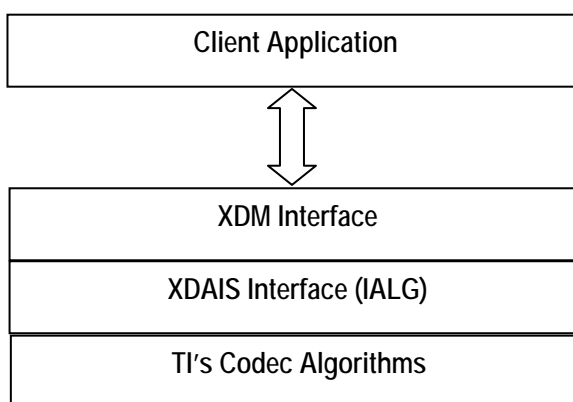
(for example audio, video, image, and speech). The XDM standard defines the following two APIs:

- ❑ `control()`
- ❑ `process()`

The `control()` API provides a standard way to control an algorithm instance and receive status information from the algorithm in real-time. The `control()` API replaces the `algControl()` API defined as part of the IALG interface. The `process()` API does the basic processing (encode/decode) of data.

Apart from defining standardized APIs for multimedia codecs, XDM also standardizes the generic parameters that the client application must pass to these APIs. The client application can define additional implementation specific parameters using extended data structures.

The following figure depicts the XDM interface to the client application.



As depicted in the figure, XDM is an extension to XDAIS and forms an interface between the client application and the codec component. XDM insulates the client application from component-level changes. Since TI's multimedia algorithms are XDM compliant, it provides you with the flexibility to use any TI algorithm without changing the client application code. For example, if you have developed a client application using an XDM-compliant MPEG4 video decoder, then you can easily replace MPEG4 with another XDM-compliant video decoder, say H.263, with minimal changes to the client application.

For more details, see eXpressDSP Digital Media (XDM) Standard API Reference (literature number SPRUEC8).

1.2 Overview of Dolby AC3 Decoder

The Dolby AC3 Decoder is an audio coding technology used to store and transmit high quality multi-channel sound.

This standard is used in movie and home theaters and has been chosen as the audio standard for next-generation systems such as DVD, HDTV, digital broadcasts, computer audio, and DVD ROMs for video games. The technology has been adopted by the ATSC (Advanced Television Systems Committee) as the audio service standard for HDTV in United States.

From this point onwards, all references to AC3 Decoder means Dolby AC3 Version3 Decoder only.

1.3 Supported Services and Features

This user guide accompanies TI's implementation of Dolby AC3 Decoder on the C64x+ platform.

This version of the codec has the following supported features of the standard:

- ❑ Supports 32, 44.1, and 48 kHz output sampling rates and 32-640 kbps input bit rates
- ❑ Supports all specified audio coding modes namely Dual Mono (1+1), 1/0, 2/0, 3/0, 2/1, 3/1, 2/2, 3/2
- ❑ Supports Dialog Normalization, Dynamic Range Control, and Peak Level Control features
- ❑ Supports dual-mono modes: Stereo, Ch1 Mono, Ch2 Mono, and Mixed Mono
- ❑ Compliant with Dolby Digital Development Kit version 3.0 (7.1.0)
- ❑ Implementation is Karaoke Aware and Karaoke capable
- ❑ Supports the Dolby Digital Extended Bit Stream syntax provided in Annex D of the ATSC specifications
- ❑ Supports output of 16-bit and 24-bit PCM samples. In case of 24-bit output PCM samples, the samples are written as 32-bit words where the 24-bit sample word is towards the MSB and LSB 8-bits are zero padded which can be ignored. In case of 16-bit PCM samples, the MSB 16-bit has the PCM data and the LSB 16-bits are zero-padded.
- ❑ Supports both block format data output and sample-interleaved data format
- ❑ Does not support Digital Rights Management (DRM)
- ❑ Does not support Up-Mix and Pro-Logic processing
- ❑ eXpressDSP compliant
- ❑ eXpressDSP Digital Media (XDM) compliant

Installation Overview

This chapter provides a brief description on the system requirements and instructions for installing the codec component. It also provides information on building and running the sample test application.

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2.1 System Requirements

This section describes the hardware and software requirements for the normal functioning of the codec component.

2.1.1 Hardware

This codec has been built and tested on the DM6437 EVM with XDS560 USB.

2.1.2 Software

The following are the software requirements for the normal functioning of the codec:

- ❑ **Development Environment:** This project is developed using Code Composer Studio version 3.2.37.12.
- ❑ **Code Generation Tools:** This project is compiled, assembled, archived, and linked using the code generation tools version 6.0.8.

2.2 Installing the Component

The codec component is released as a compressed archive. To install the codec, extract the contents of the zip file onto your local hard disk. The zip file extraction creates a directory called 100_A_AC3_D. Figure 2-1 shows the sub-directories created in this directory, under which another directory named DM6437_AC3 is created. Figure 2-1 shows the sub-directories created in DM6437_AC3 directory.

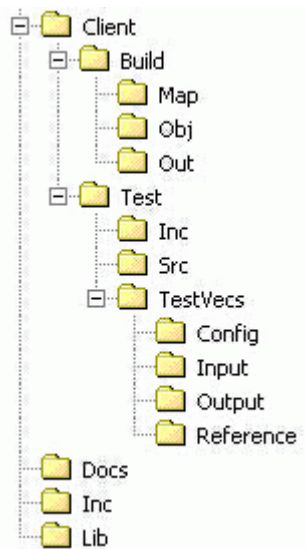


Figure 2-1. Component Directory Structure

Note:

If you are installing an evaluation version of this codec, the parent directory name will be 100E_A_AC3_D.

Table 2-1 provides a description of the sub-directories created in the DM6437_AC3 directory.

Table 2-1. Component Directories

Sub-Directory	Description
\Inc	Contains XDM related header files which allow interface to the codec library
\Lib	Contains the codec library files
\Docs	Contains user guide, datasheet, and release notes
\Client\Build	Contains the sample test application project (.pj) file
\Client\Build\Map	Contains the memory map generated on compilation of the code
\Client\Build\Obj	Contains the intermediate .asm and/or .obj file generated on compilation of the code
\Client\Build\Out	Contains the final application executable (.out) file generated by the sample test application
\Client\Test\Src	Contains application C files
\Client\Test\Inc	Contains header files needed for the application code
\Client\Test\TestVecs\Input	Contains input test vectors
\Client\Test\TestVecs\Output	Contains output generated by the codec
\Client\Test\TestVecs\Reference	Contains read-only reference output to be used for verifying against codec output
\Client\Test\TestVecs\Config	Contains configuration parameter files

2.3 Before Building the Sample Test Application

This codec is accompanied by a sample test application. To run the sample test application, you need DSP/BIOS. This version of the codec has been validated with DSP/BIOS version 5.31.

2.3.1 Installing DSP/BIOS

You can download DSP/BIOS from the TI external website:

https://www-a.ti.com/downloads/sds_support/targetcontent/bios/index.html

Install DSP/BIOS at the same location where you have installed Code Composer Studio. For example:

<install directory>\CCStudio_v3.2

The sample test application uses the following DSP/BIOS files:

- ❑ Header file, bcache.h available in the <install directory>\CCStudio_v3.2\<bios_directory>\packages\ti\bios\include directory.
- ❑ Library file, biosDM420.a64P available in the <install directory>\CCStudio_v3.2\<bios_directory>\packages\ti\bios\lib directory.

2.4 Building and Running the Sample Test Application

The sample test application that accompanies this codec component will run in TI's Code Composer Studio development environment. To build and run the sample test application in Code Composer Studio, follow these steps:

- 1) Verify that you have an installation of TI's Code Composer Studio version 3.2.37.12 and code generation tools version 6.0.8.
- 2) Verify that the codec object library, dolbyac3dec_ti.l64P exists in the \Lib sub-directory.
- 3) Open the test application project file, TestAppDecoder.pjt in Code Composer Studio. This file is available in the \Client\Build sub-directory.
- 4) Select **Project > Build** to build the sample test application. This creates an executable file, TestAppDecoder.out in the \Client\Build\Out sub-directory.
- 5) Select **File > Load**, browse to the \Client\Build\Out sub-directory, select the codec executable created in step 4, and load it into Code Composer Studio in preparation for execution.
- 6) Select **Debug > Run** to execute the sample test application.

The sample test application takes the input files stored in the \Client\Test\TestVecs\Input sub-directory, runs the codec, to generate

an output file that will be stored in the \Client\Test\TestVecs\Output sub-directory.

On successful completion, the application displays the following message for each frame “Decoder frame dump completed”

- 7) Compare the output file stored in \Client\Test\TestVecs\Output sub-directory with the reference file stored in the \Client\Test\TestVecs\Reference sub-directory to verify that the codec is functioning as expected.

2.5 Configuration Files

This codec is shipped along with a generic configuration file (Testvecs.cfg), which specifies the input and output files for the sample test application.

2.5.1 Generic Configuration File

The sample test application shipped along with the codec uses the configuration file, Testvecs.cfg for determining the input and output files for running the codec. The Testvecs.cfg file is available in the \Client\Test\TestVecs\Config sub-directory.

The format of the Testvecs.cfg file is:

```
X
Input
Output
Bit precision
Satellite Channel configuration
Sub woofer
Auxiliary channel
Source dual
Source Karaoke
Operation Mode
Dynamic RangeLo
Dynamic RangeHi
```

where:

- `x` may be set to:
 - 1 – for compliance checking, no output file is created
 - 0 – for writing output to output file
- `Input` is the input file name (use complete path)
- `Output` is the root of the output stereo file's name (use complete path). For example, if output file name is output_001, the three stereo PCM files will be output_001_lr.pcm, output_001_lsrs.pcm, and output_001_csw.pcm.
- `Bit precision`
 - 16 – 16-bit PCM output
 - 24 – 24-bit PCM output
- `Satellite Channel configuration` is the Satellite channel configuration value (See Table 4-3. Speaker Satellite Configuration

for details)

- ❑ `Sub Woofer` is the user request for subwoofer channel or LFE channel
 - 0 – LFE absent
 - 1 – LFE present
- ❑ `Auxiliary Channel` is the user request for auxiliary channel
 - 0 – Unknown
 - 1 – Stereo Lo Ro
 - 2 – Stereo Lt Rt
 - 3 – Stereo Mono
 - 4 – Stereo Dual
- ❑ `Source Dual` is the dual mono reproduction mode
 - 0 – Stereo (default)
 - 1 – Left mono
 - 2 – Right mono
 - 3 – Mixed mono
- ❑ `Source Karaoke`
 - -1 – Karaoke mode
 - 0 – no vocal
 - 1 – left vocal
 - 2 – right vocal
 - 3 – both vocals (default)
- ❑ `Operation Mode` indicates operational mode selected for decoding
 - 0 – Custom 0 mode
 - 1 – Custom 1 mode
 - 2 – Lineout mode(default)
 - 3 – RF mode
- ❑ `Dynamic RangeLo` indicates Dynamic range scale factor (low). Range: 0.0 to 1.0 in Q6 format i.e. 0x00 to 0x40. (The default value is 1.0)
- ❑ `Dynamic RangeHi` indicates Dynamic range scale factor (high). Range: 0.0 to 1.0 in Q6 format i.e. 0x00 to 0x40. (The default value is 1.0)

A sample `Testvecs.cfg` file is as shown:

```
1
..\..\Test\TestVecs\Input\default.ac3
..\..\Test\TestVecs\Reference\output_001
24
10
1
0
0
0
3
0
0.0
0.0
```

2.6 Standards Conformance and User-Defined Inputs

To check the conformance of the codec for the default input file shipped along with the codec, follow the steps as described in Section 2.4.

To check the conformance of the codec for other input files of your choice, follow these steps:

- ❑ Copy the input files to the \Client\Test\TestVecs\Inputs sub-directory.
- ❑ Copy the reference files to the \Client\Test\TestVecs\Reference sub-directory.
- ❑ Edit the configuration file, Testvecs.cfg available in the \Client\Test\TestVecs\Config sub-directory. For details on the format of the Testvecs.cfg file, see Section 2.5.1.
- ❑ Execute the sample test application. The application displays the following message for each frame “Decoder compliance test passed for the frame No x”.

You may use any standard file comparison utility to compare the codec output with the reference output and check for conformance.

2.7 Uninstalling the Component

To uninstall the component, delete the codec directory from your hard disk.

2.8 Evaluation Version

If you are using an evaluation version of this codec there will be an audible tone will be heard occasionally.

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Sample Usage

This chapter provides a detailed description of the sample test application that accompanies this codec component.

3.1 Overview of the Test Application

The test application exercises the IAUDDEC base class of the Decoder library. The main test application files are TestAppDecoder.c and TestAppDecoder.h. These files are available in the \Client\Test\Src and \Client\Test\Inc sub-directories respectively.

Figure 3-1 depicts the sequence of APIs exercised in the sample test application.

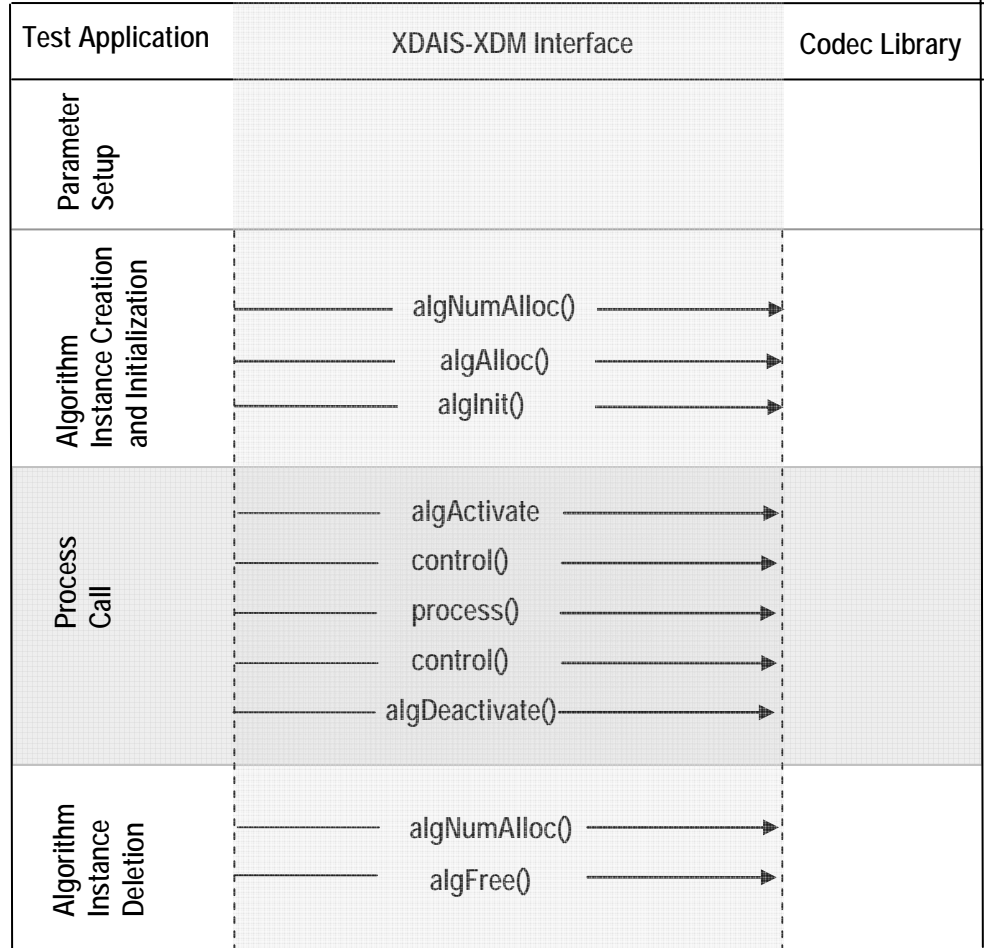


Figure 3-1. Test Application Sample Implementation

Note:

Audio codecs do not use `algActivate()` and `algDeactivate()` APIs.

The test application is divided into four logical blocks:

- ❑ Parameter setup
- ❑ Algorithm instance creation and initialization
- ❑ Process call
- ❑ Algorithm instance deletion

3.1.1 Parameter Setup

Each codec component requires various codec configuration parameters to be set at initialization. For example, a video codec requires parameters such as video height, video width, etc. The test application obtains the required parameters from the Decoder configuration files.

In this logical block, the test application does the following:

- 1) Opens the generic configuration file, `Testvecs.cfg` and reads the compliance checking parameter, input file name, output/reference file name, output samples bit precision, user satellite configuration request parameters, dual mono mode and karaoke mode parameters
- 2) Reads the input bit stream into the application input buffer.
- 3) Assigns the corresponding parameters as extended input arguments. The decoder assumes the default values (see Section 4.2) if these parameters are not provided.

After successful completion of the above steps, the test application does the algorithm instance creation and initialization.

3.1.2 Algorithm Instance Creation and Initialization

In this logical block, the test application accepts the various initialization parameters and returns an algorithm instance pointer. The following APIs are called in sequence:

- 1) `algNumAlloc()` - To query the algorithm about the number of memory records it requires.
- 2) `algAlloc()` - To query the algorithm about the memory requirement to be filled in the memory records.
- 3) `algInit()` - To initialize the algorithm with the memory structures provided by the application.

A sample implementation of the create function that calls `algNumAlloc()`, `algAlloc()`, and `algInit()` in sequence is provided in the `ALG_create()` function implemented in the `alg_create.c` file.

3.1.3 Process Call

After algorithm instance creation and initialization, the test application does the following:

- 1) Sets the dynamic parameters (if they change during run time) by calling the `control()` function with the `XDM_SETPARAMS` command.
- 2) Sets the input and output buffer descriptors required for the `process()` function call. The input and output buffer descriptors are obtained by calling the `control()` function with the `XDM_GETBUFINFO` command.
- 3) Calls the `process()` function to encode/decode a single frame of data. The behavior of the algorithm can be controlled using various dynamic parameters (see Section 4.2.1.5). The inputs to the process function are input and output buffer descriptors, pointer to the `IAUDDC_InArgs` and `IAUDDC_OutArgs` structures.

There could be any ordering of `control()` and `process()` functions. The following APIs are called in sequence:

- 1) `control()` (optional) - To query the algorithm on status or setting of dynamic parameters etc., using the six available control commands.
- 2) `process()` - To call the Decoder with appropriate input/output buffer and arguments information.
- 3) `control()` (optional) - To query the algorithm on status or setting of dynamic parameters etc., using the six available control commands.

The do-while loop encapsulates frame level `process()` call by updating the input buffer pointer to each frame every time before the next call. For each frame, frame level decoding is achieved through the `process()` call. The do-while loop breaks off whenever the input buffer exhausts. The do-while loop also protects the `process()` call from file operations by placing appropriate calls for cache operations as well. The test application does a cache invalidate for the valid input buffers before `process()` and a cache write back invalidate for output buffers after `process()`.

In the sample test application, after calling `process()`, the output data is either dumped to a file or compared with a reference file.

3.1.4 Algorithm Instance Deletion

Once decoding/encoding is complete, the test application must delete the current algorithm instance. The following APIs are called in sequence:

- 1) `algNumAlloc()` - To query the algorithm about the number of memory records it used.
- 2) `algFree()` - To query the algorithm to get the memory record information.

A sample implementation of the delete function that calls `algNumAlloc()` and `algFree()` in sequence is provided in the `ALG_delete()` function implemented in the `alg_create.c` file.

API Reference

This chapter provides a detailed description of the data structures and interfaces functions used in the codec component.

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4.1 Symbolic Constants and Enumerated Data Types	4-2
4.2 Data Structures	4-5
4.3 Interface Functions	4-5

4.1 Symbolic Constants and Enumerated Data Types

This section summarizes all the symbolic constants specified as either #define macros and/or enumerated C data types. Described alongside the macro or enumeration is the semantics or interpretation of the same in terms of what value it stands for and what it means.

Table 4-1. List of Enumerated Data Types

Group or Enumeration Class	Symbolic Constant Name	Description or Evaluation
IAUDIO_ChannelId	IAUDIO_MONO	Single channel
	IAUDIO_STEREO	Two channel
	IAUDIO_THREE_ZERO	Three channel
	IAUDIO_FIVE_ZERO	Five channel
	IAUDIO_FIVE_ONE	5.1 channel
	IAUDIO_SEVEN_ONE	7.1 channel. Not supported in this version of AC3 Decoder.
IAUDIO_PcmFormat	IAUDIO_BLOCK	Front Left, Front Right, Surround Left, Surround Right, Centre, and LFE channels data are output in block format in a single output buffer.
	IAUDIO_INTERLEAVED	Front Left, Front Right, Surround Left, Surround Right, Centre, and LFE channels data are output in interleaved format in a single output buffer.
XDM_DataFormat	XDM_BYTE	Big endian stream
	XDM_LE_16	16-bit little endian stream
	XDM_LE_32	32-bit little endian stream
XDM_CmdId	XDM_GETSTATUS	Query algorithm instance to fill <code>Status</code> structure
	XDM_SETPARAMS	Set run-time dynamic parameters via the <code>DynamicParams</code> structure.
	XDM_RESET	Reset the algorithm
	XDM_SETDEFAULT	Initialize all fields in <code>Params</code> structure to default values specified in the library.

Group or Enumeration Class	Symbolic Constant Name	Description or Evaluation
	XDM_FLUSH	Handle end of stream conditions. This command forces algorithm instance to output data without additional input. Not applicable for AC3 Decoder. Returns IALG_EOK.
	XDM_GETBUFINFO	Query algorithm instance regarding the properties of input and output buffers.
XDM_ErrorBit		The bit fields in the 32-bit error code are interpreted as shown.
	XDM_APPLIEDCONCEALMENT	Bit 9 <input type="checkbox"/> 1 - Applied concealment <input type="checkbox"/> 0 - Ignore Not supported for this version of AC3 Decoder.
	XDM_INSUFFICIENTDATA	Bit 10 <input type="checkbox"/> 1 - Insufficient input data <input type="checkbox"/> 0 - Ignore
	XDM_CORRUPTEDDATA	Bit 11 <input type="checkbox"/> 1 - Invalid data <input type="checkbox"/> 0 - Ignore
	XDM_CORRUPTEDHEADER	Bit 12 <input type="checkbox"/> 1 - Corrupted frame header <input type="checkbox"/> 0 - Ignore
	XDM_UNSUPPORTEDINPUT	Bit 13 <input type="checkbox"/> 1 - Unsupported feature/parameter in input <input type="checkbox"/> 0 - Ignore
	XDM_UNSUPPORTEDPARAM	Bit 14 <input type="checkbox"/> 1 - Unsupported input parameter or configuration <input type="checkbox"/> 0 - Ignore
	XDM_FATALERROR	Bit 15 <input type="checkbox"/> 1 - Fatal error (stop decoding) <input type="checkbox"/> 0 - Recoverable error

Note:

The remaining bits that are not mentioned in XDM_ErrorBit are interpreted as:

- Bit 16-32: Reserved
- Bit 8: Reserved
- Bit 0-7: Codec and implementation specific

The algorithm can set multiple bits to 1 depending on the error condition.

The AC3 Decoder specific error status messages are listed in Table 4-2. The value column indicates the decimal value of the last 8-bits reserved for codec specific error statuses.

Table 4-2. Decoder Error Status

Group or Enumeration Class	Symbolic Constant Name	Value	Description or Evaluation if set
AC3_ErrorBit	XDM_BADBITRATE	1	Unsupported bit rate
	XDM_BADSAMPLERATE	2	Unsupported sampling rate
	XDM_UNSUPPORTEDDBSID	3	Unsupported bit stream configuration ID
	XDM_CRCFAILED	4	CRC failed
	XDM_CPLSTRTGFAILED	5	Channel coupling strategy failed
	XDM_SETCHBWCODHIGH	6	Channel bandwidth code too high
	XDM_REMATRIXFAILED	7	Re-matrix failed
	XDM_EXPDECODEFAILED	8	Exponent decoding failed
	XDM_BITALLOCFAILED	9	Bit allocation failed
	XDM_SYNCNOTFOUND	10	AC3 synchronization not found
	XDM_INVALID_BUFFER_CONFIG	11	Invalid I/O buffer configuration
	XDM_INVALID_ARGUMENT	12	Invalid input argument.
	XDM_NULL_POINTER_DETECTED	13	Null pointer detected

Note:

The AC3 errors are categorized based on the status of the decoder based on whether,

- (i) the application can proceed decoding with additional input data, or
- (ii) the application can proceed decoding with the next frame or
- (iii) decoder should be configured or initialized by the application.

- ❑ Decoder can continue with next frame or with additional input data
 - XDM_SYNCNOTFOUND
 - XDM_BADBITRATE
 - XDM_BADSAMPLERATE
 - XDM_SETCHBWCODHIGH
- ❑ Decoder cannot decode the present frame
 - XDM_UNSUPPORTEDDBSID
 - XDM_CRCFAILED
 - XDM_CPLSTRGFAILED
 - XDM_EXPDECODEFAILED
 - XDM_BITALLOCFAILED
 - XDM_REMATRIXFAILED
- ❑ Decoder is not configured or initialised properly by the application
 - XDM_INVALID_BUFFER_CONFIG
 - XDM_INVALID_ARGUMENT
 - XDM_NULL_POINTER_DETECTED

4.2 Data Structures

This section describes the XDM defined data structures that are common across codec classes. These XDM data structures can be extended to define any implementation specific parameters for a codec component.

4.2.1 Common XDM Data Structures

This section includes the following common XDM data structures:

- ❑ XDM_BufDesc
- ❑ XDM_AlgBufInfo
- ❑ IAUDDEC_Fxns
- ❑ IAUDDEC_Params
- ❑ IAUDDEC_DynamicParams
- ❑ IAUDDEC_InArgs
- ❑ IAUDDEC_Status
- ❑ IAUDDEC_OutArgs

4.2.1.1 XDM_BufDesc

|| Description

This structure defines the buffer descriptor for input and output buffers.

|| Fields

Field	Datatype	Input/ Output	Description
**bufs	XDAS_Int8	Input	Pointer to the vector containing buffer addresses
numBufs	XDAS_Int32	Input	Number of buffers
*bufSizes	XDAS_Int32	Input	Size of each buffer in bytes

4.2.1.2 XDM_AlgBufInfo

|| Description

This structure defines the buffer information descriptor for input and output buffers. This structure is filled when you invoke the `control()` function with the `XDM_GETBUFINFO` command.

|| Fields

Field	Datatype	Input/ Output	Description
minNumInBufs	XDAS_Int32	Output	Number of input buffers
minNumOutBufs	XDAS_Int32	Output	Number of output buffers
minInBufSize[XDM_MAX_IO_BUFFERS]	XDAS_Int32	Output	Size in bytes required for each input buffer
minOutBufSize[XDM_MAX_IO_BUFFERS]	XDAS_Int32	Output	Size in bytes required for each output buffer

Note:

For AC3 Decoder, the buffer details are:

- ❑ Number of input buffer required is 1.
- ❑ Number of output buffer required is 1.
- ❑ The size of the input buffer should allow atleast one frame of encoded data to be present in the input buffer. If the size of input buffer is less than the minimum frame size of AC3 decoder (128 bytes), the decoder returns `XDM_INVALID_BUFFER_CONFIG` error. The input buffer size (in bytes) for worst case is 4096 bytes.

- ❑ The size of output buffer should be sufficient enough to hold the decoded output of a complete frame for all six channels. The output buffer size (in bytes) for worst case is 36864 bytes.
- ❑ The decoded output of all six channels are placed in channel order: Front Left, Front Right, Surround Left, Surround Right, Centre, and LFE. The channels are interleaved sample-wise or block-wise based on output format. See `IAUDIO_PcmFormat` enumeration for details.

4.2.1.3 `IAUDDDEC_Fxns`

|| Description

This structure contains pointers to all the XDAIS and XDM interface functions.

|| Fields

Field	Datatype	Input/ Output	Description
<code>ialg</code>	<code>IALG_Fxns</code>	Input	Structure containing pointers to all the XDAIS interface functions. For more details, see <i>TMS320 DSP Algorithm Standard API Reference</i> (literature number SPRU360).
<code>*process</code>	<code>XDAS_Int32</code>	Input	Pointer to the <code>process()</code> function
<code>*control</code>	<code>XDAS_Int32</code>	Input	Pointer to the <code>control()</code> function

4.2.1.4 `IAUDDDEC_Params`

|| Description

This structure defines the creation parameters for an algorithm instance object. Set this data structure to `NULL`, if you are unsure of the values to specify for these parameters.

|| Fields

Field	Datatype	Input/ Output	Description
<code>size</code>	<code>XDAS_Int32</code>	Input	Size of the basic or extended (if being used) data structure in bytes.
<code>maxSampleRate</code>	<code>XDAS_Int32</code>	Input	Maximum sampling frequency to be supported in Hertz (Hz). For example, if maximum sampling frequency is 44.1 kHz, set this field to 44100.
<code>maxBitrate</code>	<code>XDAS_Int32</code>	Input	Maximum bit rate to be supported in bits per second. For example, if maximum bit rate is 128 kbps, set this field to 128000.

Field	Datatype	Input/Output	Description
maxNoOfCh	XDAS_Int32	Input	Maximum channels to be supported. See <code>IAUDIO_ChannelId</code> enumeration for details.
dataEndianness	XDAS_Int32	Input	Endianness of input data. See <code>XDM_DataFormat</code> enumeration for details.

Note:

- ❑ AC3 decoder implementation supports both little endian (`XDM_LE_16`) and big endian (`XDM_BYTE`) format of input data. The decoder detects the endianness of input data from the bitstream and hence this field is ignored. However during initialization, the decoder validates this field and returns error if this field is set values other than `XDM_LE_16` or `XDM_BYTE`.
- ❑ AC3 decoder supports maximum 6 channels, that is, 5.1 channels. LFE channel is referred as “.1”
- ❑ Supported `maxBitrate` and `maxSampleRate` values are 640 kbps and 48 kHz respectively.
- ❑ The `maxBitrate`, `maxSampleRate`, and `maxNoOfCh` fields are used by the application only to poll the capabilities of the decoder. The fields are not considered by the decoder during the process of decoding the bitstream.

4.2.1.5 IAUDDEC_DynamicParams

|| Description

This structure defines the run time parameters for an algorithm instance object. Set this data structure to `NULL`, if you are unsure of the values to be specified for these parameters.

|| Fields

Field	Datatype	Input/Output	Description
size	XDAS_Int32	Input	❑ Size of the basic or extended (if being used) data structure in bytes
outputFormat	XDAS_Int32	Input	To set interleaved/block format for output. See <code>IAUDIO_PcmFormat</code> enumeration for details.

Note:

- ❑ AC3 decoder implementation supports both `IAUDIO_BLOCK` and `IAUDIO_INTERLEAVED` format for output PCM samples. The channel order followed is Front Left, Front Right, Surround Left, Surround Right, Centre, and LFE channels.

4.2.1.6 IAUDDEC_InArgs

|| Description

This structure defines the run time input arguments for an algorithm instance object.

|| Fields

Field	Datatype	Input/ Output	Description
size	XDAS_Int32	Input	❑ Size of the basic or extended (if being used) data structure in bytes.
numBytes	XDAS_Int32	Input	❑ Number of valid input data (in bytes) in input buffer. For example, if number of valid input data in input buffer is 128 bytes, set this field to 128.

4.2.1.7 IAUDDEC_Status

|| Description

This structure defines parameters that describe the status of the decoder.

|| Fields

Field	Datatype	Input/ Output	Description
size	XDAS_Int32	Input	Size of the basic or extended (if being used) data structure in bytes.
extendedError	XDAS_Int32	Output	Extended error enumeration for XDM compliant encoders and decoders. See XDM_ErrorBit data structure for details.
bitRate	XDAS_Int32	Output	Bit rate in bits per second. For example, if the value of this field is 128000, it indicates that bit rate is 128 kbps.
sampleRate	XDAS_Int32	Output	Sampling frequency in Hertz (Hz). For example, if the value of this field is 44100, it indicates that the sample rate is 44.1kHz.
numChannels	XDAS_Int32	Output	Number of channels. See IAUDIO_ChannelId enumeration for details.
numLFEChannels	XDAS_Int32	Output	Number of LFE (Low Frequency Effects) channels in the stream

Field	Datatype	Input/ Output	Description
outputFormat	XDAS_Int32	Output	The output PCM format. See <code>IAUDIO_PcmFormat</code> enumeration for details.
autoPosition	XDAS_Int32	Output	Flag to indicate support for random position decoding, which means that a stream can be decoded from any point: <ul style="list-style-type: none"> <input type="checkbox"/> 1 - Supports random position decoding <input type="checkbox"/> 0 - Does not support random position decoding
fastFwdLen	XDAS_Int32	Output	Recommended Fast Forward length in bytes in case of random position decoding.
frameLen	XDAS_Int32	Output	Number of samples decoded per decode call.
outputBitsPerSample	XDAS_Int32	Output	Number of output bits per output sample. For example, if the value of the field is 16, it indicates 16 output bits per PCM sample.
bufInfo	XDM_AlgBufInfo	Output	Input and output buffer information. See <code>XDM_AlgBufInfo</code> data structure for details.

4.2.1.8 `IAUDEEC_OutArgs`

|| Description

This structure defines the run time output arguments for the algorithm instance object.

|| Fields

Field	Datatype	Input/ Output	Description
size	XDAS_Int32	Input	<input type="checkbox"/> Size of the basic or extended (if being used) data structure in bytes.
extendedError	XDAS_Int32	Output	<input type="checkbox"/> Extended error enumeration for XDM compliant encoders and decoders. See <code>XDM_ErrorBit</code> data structure for details.
bytesConsumed	XDAS_Int32	Output	Bytes consumed during the process call

4.2.2 Dolby Digital AC3 Decoder Data Structures

This section includes the following AC3 Decoder specific extended data structures:

- ❑ IDOLBYAC3DEC_Params
- ❑ IDOLBYAC3DEC_DynamicParams
- ❑ IDOLBYAC3DEC_InArgs
- ❑ IDOLBYAC3DEC_Status
- ❑ IDOLBYAC3DEC_OutArgs

4.2.2.1 IDOLBYAC3DEC_Params

|| Description

This structure defines the creation parameters and any other implementation specific parameters for the AC3 Decoder instance object. The creation parameters are defined in the XDM data structure, IAUDDEC_Params.

|| Fields

Field	Datatype	Input/ Output	Description
auddec_params	IAUDDEC_Params	Input	See IAUDDEC_Params data structure for details.

4.2.2.2 IDOLBYAC3DEC_DynamicParams

|| Description

This structure defines the run time parameters and any other implementation specific parameters for the AC3 Decoder instance object. The run time parameters are defined in the XDM data structure, IAUDDEC_DynamicParams.

|| Fields

Field	Datatype	Input/ Output	Description
auddec_dynamicparams	IAUDDEC_DynamicParams	Input	See IAUDDEC_DynamicParams data structure for details.
operationMode	XDAS_Int32	Input	Indicates operational mode selected for decoding. <ul style="list-style-type: none"> ❑ 0—Custom 0 mode ❑ 1—Custom 1 mode ❑ 2—Lineout mode(default) ❑ 3—RF mode

Field	Datatype	Input/Output	Description
dynrngScaleHiQ6	XDAS_Int32	Input	Dynamic range scale factor (high). Range: 0.0 to 1.0 in Q6 format i.e. 0x00 to 0x40. (The default value is 1.0)
dynrngScaleLoQ6	XDAS_Int32	Input	Dynamic range scale factor (low) Range: 0.0 to 1.0 in Q6 format i.e. 0x00 to 0x40. (The default value is 1.0)
RFModeGainApply	XDAS_Int32	Input	Indicates whether 11 dB gain to be applied in the RF mode has been applied. <input type="checkbox"/> 0 – Apply 11 dB gain (default) <input type="checkbox"/> 1 – Do not apply 11 dB gain
dlgNormApply	XDAS_Int32	Input	Apply dialog normalization <input type="checkbox"/> 0 – Apply dialog normalisation (default) <input type="checkbox"/> 1 – Do not apply dialog normalisation
stereomod	XDAS_Int32	Input	Facilitate preferred stereo downmix mode <input type="checkbox"/> 0 – Auto or LtRt (default) <input type="checkbox"/> 1 – Perform LtRt downmix <input type="checkbox"/> 2 – Perform LoRo downmix <input type="checkbox"/> 3 – Auto or LoRo Note: Auto implies stereo downmix as indicated in the bitstream.
adconvtyp	XDAS_Int32	Input	A/D conversion Type; required in consumer players for playback. The default value set is 0.

4.2.2.3 IDOLBYAC3DEC_InArgs

|| Description

This structure defines the run time input arguments for the AC3 Decoder instance object.

|| Fields

Field	Datatype	Input/ Output	Description
auddec_inArgs	IAUDDEC_InArgs	Input	See IAUDDEC_InArgs data structure for details.
channelConfigurationRequest	PAF_ChannelConfiguration	Input	Request for desired output channel configuration. See Section 4.2.2.4 for more details on PAF_ChannelConfiguration.
sourceDual	XDAS_Int32	Input	Request for dual mono reproduction mode: <input type="checkbox"/> 0 – Stereo (default) <input type="checkbox"/> 1 – Left mono <input type="checkbox"/> 2 – Right mono <input type="checkbox"/> 3 – Mixed mono
sourceKaraoke	XDAS_Int32	Input	Request for karaoke mode: <input type="checkbox"/> -1 – Karaoke mode <input type="checkbox"/> 0 – No vocal <input type="checkbox"/> 1 – Left vocal <input type="checkbox"/> 2 – Right vocal <input type="checkbox"/> 3 – Both vocals (default)
outputBitsPerSample	XDAS_Int32	Input	Number of bits per output PCM sample: <input type="checkbox"/> 16 – 16 bits per PCM sample <input type="checkbox"/> 24 – 24 bits per PCM sample

4.2.2.4 PAF_ChannelConfiguration

|| Description

This structure defines the arguments for configuring speaker satellite for the AC3 Decoder instance object.

|| Fields

Field	Datatype	Input/ Output	Description
Sat	XDAS_Int32	Input	Speaker satellite configuration. <input type="checkbox"/> 0 – Unknown <input type="checkbox"/> 1 – None <input type="checkbox"/> 2 – Mono <input type="checkbox"/> 3 – Stereo or PHANTOM0

			<ul style="list-style-type: none"> <input type="checkbox"/> 4 – PHANTOM1 <input type="checkbox"/> 5 – PHANTOM2 <input type="checkbox"/> 6 – PHANTOM3 <input type="checkbox"/> 7 – PHANTOM4 <input type="checkbox"/> 8 – 3 Stereo or SURROUND0 <input type="checkbox"/> 9 – SURROUND1 <input type="checkbox"/> 10 – SURROUND2
			See Table 4-3. Speaker Satellite Configuration for details.
Sub	XDAS_Int32	Input	Subwoofer or LFE configuration <input type="checkbox"/> 0 – LFE absent <input type="checkbox"/> 1 – LFE present
Aux	XDAS_Int32	Input	Auxiliary channel configuration <input type="checkbox"/> 0 – Unknown <input type="checkbox"/> 1 – Stereo Lo Ro <input type="checkbox"/> 2 – Stereo Lt Rt <input type="checkbox"/> 3 – Stereo Mono <input type="checkbox"/> 4 – Stereo Dual

Note:

- If the application requested values for the `sat` and `aux` fields of `channelConfigurationRequest` are invalid/unsupported, the decoder defaults them to unknown. For `sub` field, the decoder defaults them to `LFE present`.
- The `Stereo`, `Mono`, and `Stereo Dual` values for `aux` field are used only as an indicator by the decoder and any requests by the application for these values will be defaulted as unknown.
- The decoder returns error for invalid values of `sourceKaraoke` and `sourceDual` fields of `IDOLBYAC3DEC InArgs`

Table 4-3 describes the field `sat` in `PAF_ChannelConfiguration` data structure and explains the speaker satellite configuration

Table 4-3. Speaker Satellite Configuration

Field 'sat'	Satellite Channel Configuration	Left	Right	LSur	RSur	Cntr
0	UNKNOWN					
1	NONE					
2	MONO					x
3	STEREO or PHANTOM0	x	x			
4	PHANTOM1	x	x	x		
5	PHANTOM2	x	x	x	x	
6	PHANTOM3	x	x	x	x	
7	PHANTOM4	x	x	x	x	
8	3STEREO or SURROUND0	x	x			x
9	SURROUND1	x	x	x		x
10	SURROUND2	x	x	x	x	x

4.2.2.5 IDOLBYAC3DEC_Status

|| Description

This structure defines parameters that describe the status of the AC3 Decoder and any other implementation specific parameters. The status parameters are defined in the XDM data structure, `IAUDDC_Status`.

|| Fields

Field	Datatype	Input/Output	Description
<code>auddec_status</code>	<code>IAUDDC_Status</code>	Output	See <code>IAUDDC_Status</code> data structure for details.
<code>isValid</code>	<code>XDAS_Int32</code>	Output	Flag indicating if the last decode call was successful: <ul style="list-style-type: none"> <input type="checkbox"/> 1 – Indicates last decode call was successful and the values of the fields in the <code>IAUDDC_Status</code> structure are valid. <input type="checkbox"/> 0 – Indicates last decode call was not successful and the values are not valid.
<code>bsi[10]</code>	<code>XDAS_Int16</code>	Output	Bit Stream information required for application. Refer <code>DolbySIP5_0.pdf</code> in related documentation for complete description of them.
<code>bsiParams</code>	<code>IDOLBYAC3DEC_BsiParams</code>	Output	Common bitstream information required for application for instant easy reference. See Section 4.2.2.7 for more details on <code>bsiParams</code>

4.2.2.6 IDOLBYAC3DEC_OutArgs

|| Description

This structure defines the run time output arguments for the AC3 Decoder instance object.

|| Fields

Field	Datatype	Input/ Output	Description
auddec_outArgs	IAUDEDEC_OutArgs	Output	See IAUDEDEC_OutArgs data structure for details.
channelConfigurat ionProgram	PAF_ChannelConf iguration	Output	Decoder transmits to the application, the output channel configuration encoded in the bit stream.
channelConfigurat ionOut	PAF_ChannelConf iguration	Output	Decoder transmits to the application, the final output channel configuration decoded.
bsi[10]	XDAS_Int16	Output	Bit Stream information required for application. Refer DolbySIP5_0.pdf in related documentation for complete description of them.
bsiParams	IDOLBYAC3DEC_Bs iParams	Output	Common bitstream information required for application for instant easy reference. See Section 4.2.2.7 for more details on bsiParams

4.2.2.7 IDOLBYAC3DEC_BsiParams

|| Description

This structure defines few important BSI elements, which are frequently required by the run time output arguments for the AC3 Decoder instance object.

|| Fields

Field	Datatype	Input/ Output	Description
sampleRate	XDAS_Int32	Output	Sampling frequency in Hz. For example, a value of 44100 indicates sampling frequency of 44.1 KHz
bitRate	XDAS_Int32	Output	Bit rate in bits per second. For example, a value of 128000 indicates a bit rate of 128 kbps.
sampleRateModifier	XDAS_Int32	Output	Sampling rate <input type="checkbox"/> 0 – Standard sample rate <input type="checkbox"/> 1 – Half sample rate <input type="checkbox"/> 2 – Quarter sample rate <input type="checkbox"/> 3 – Reserved
lfeChannelFlag	XDAS_Int32	Output	Flag to indicate presence of LFE channel <input type="checkbox"/> 0 – LFE absent <input type="checkbox"/> 1 – LFE present
acmod	XDAS_Int32	Output	Audio coding mode <input type="checkbox"/> 0 – 1+1 (Ch1, Ch2) <input type="checkbox"/> 1 – 1/0 (C) <input type="checkbox"/> 2 – 2/0 (L, R) <input type="checkbox"/> 3 – 3/0 (L, C, R) <input type="checkbox"/> 4 – 2/1 (L, R, S) <input type="checkbox"/> 5 – 3/1 (L, C, R, S) <input type="checkbox"/> 6 – 2/2 (L, R, SL, SR) <input type="checkbox"/> 7 – 3/2 (L, C, R, SL, SR)
frameSize	XDAS_Int32	Output	<input type="checkbox"/> Frame size in words
bsid	XDAS_Int32	Output	<input type="checkbox"/> Bit stream identification number

Field	Datatype	Input/Output	Description
bsmod	XDAS_Int32	Output	<input type="checkbox"/> Bit stream Mode <input type="checkbox"/> 0 – Main audio service <input type="checkbox"/> 1 – Main audio service minus dialog <input type="checkbox"/> 2 – Associated service: visually impaired <input type="checkbox"/> 3 – Associated service: hearing impaired <input type="checkbox"/> 4 – Associated service: dialog <input type="checkbox"/> 5 – Associated service: commentary <input type="checkbox"/> 6 – Associated service: emergency flash <input type="checkbox"/> 7 – Associated service: voice-over (audio coding mode 1+1 or 1/0 only) <input type="checkbox"/> 8 – Main audio service: karaoke (audio coding mode 2/0 or higher)
dsurmod	XDAS_Int32	Output	<input type="checkbox"/> Dolby surround mode <input type="checkbox"/> 0 – No indication <input type="checkbox"/> 1 – Not dolby surround encoded <input type="checkbox"/> 2 – Dolby surround encoded <input type="checkbox"/> 3 – Reserved
copyrightbit	XDAS_Int32	Output	<input type="checkbox"/> Copyright protection bit <input type="checkbox"/> 0 – Not copyright protected <input type="checkbox"/> 1 – Copyright protected
originalBitStream	XDAS_Int32	Output	<input type="checkbox"/> Original bit stream flag <input type="checkbox"/> 0 – Copy of original bitstream <input type="checkbox"/> 1 – Original bitstream

4.3 Interface Functions

This section describes the Application Programming Interfaces (APIs) used in the AC3 Decoder. The APIs are logically grouped into the following categories:

- Creation** – `algNumAlloc()`, `algAlloc()`
- Initialization** – `algInit()`
- Control** – `control()`
- Data processing** – `algActivate()`, `process()`, `algDeactivate()`
- Termination** – `algFree()`

You must call these APIs in the following sequence:

- 1) `algNumAlloc()`
- 2) `algAlloc()`
- 3) `algInit()`
- 4) `algActivate()`
- 5) `process()`
- 6) `algDeactivate()`

7) `algFree()`

`control()` can be called any time after calling the `algInit()` API.

`algNumAlloc()`, `algAlloc()`, `algInit()`, `algActivate()`, `algDeactivate()`, and `algFree()` are standard XDAIS APIs. This document includes only a brief description for the standard XDAIS APIs. For more details, see *TMS320 DSP Algorithm Standard API Reference* (literature number SPRU360).

Note:

Audio codecs do not use `algActivate()` and `algDeactivate()` APIs.

4.3.1 Creation APIs

Creation APIs are used to create an instance of the component. The term creation could mean allocating system resources, typically memory.

|| Name

`algNumAlloc()` – determine the number of buffers that an algorithm requires

|| Synopsis

```
XDAS_Int32 algNumAlloc(Void);
```

|| Arguments

Void

|| Return Value

```
XDAS_Int32; /* number of buffers required */
```

|| Description

`algNumAlloc()` returns the number of buffers that the `algAlloc()` method requires. This operation allows you to allocate sufficient space to call the `algAlloc()` method.

`algNumAlloc()` may be called at any time and can be called repeatedly without any side effects. It always returns the same result. The `algNumAlloc()` API is optional.

For more details, see *TMS320 DSP Algorithm Standard API Reference* (literature number SPRU360).

|| See Also

`algAlloc()`

|| Name

`algAlloc()` – determine the attributes of all buffers that an algorithm requires

|| Synopsis

```
XDAS_Int32 algAlloc(const IALG_Params *params, IALG_Fxns  
**parentFxns, IALG_MemRec memTab[]);
```

|| Arguments

```
IALG_Params *params; /* algorithm specific attributes */
```

```
IALG_Fxns **parentFxns; /* output parent algorithm  
functions */
```

```
IALG_MemRec memTab[]; /* output array of memory records */
```

|| Return Value

```
XDAS_Int32 /* number of buffers required */
```

|| Description

`algAlloc()` returns a table of memory records that describe the size, alignment, type, and memory space of all buffers required by an algorithm. If successful, this function returns a positive non-zero value indicating the number of records initialized.

The first argument to `algAlloc()` is a pointer to a structure that defines the creation parameters. This pointer may be `NULL`; however, in this case, `algAlloc()` must assume default creation parameters and must not fail.

The second argument to `algAlloc()` is an output parameter. `algAlloc()` may return a pointer to its parent's IALG functions. If an algorithm does not require a parent object to be created, this pointer must be set to `NULL`.

The third argument is a pointer to a memory space of size `nbufs * sizeof(IALG_MemRec)` where, `nbufs` is the number of buffers returned by `algNumAlloc()` and `IALG_MemRec` is the buffer-descriptor structure defined in `ialg.h`.

After calling this function, `memTab[]` is filled up with the memory requirements of an algorithm.

For more details, see *TMS320 DSP Algorithm Standard API Reference* (literature number SPRU360).

|| See Also

```
algNumAlloc(), algFree()
```

4.3.2 Initialization API

Initialization API is used to initialize an instance of the algorithm. The initialization parameters are defined in the `Params` structure (see Data Structures section for details).

|| Name

`algInit()` – initialize an algorithm instance

|| Synopsis

```
XDAS_Int32 algInit(IALG_Handle handle, IALG_MemRec
memTab[], IALG_Handle parent, IALG_Params *params);
```

|| Arguments

```
IALG_Handle handle; /* algorithm instance handle*/
IALG_memRec memTab[]; /* array of allocated buffers */
IALG_Handle parent; /* handle to the parent instance */
IALG_Params *params; /* algorithm initialization
parameters */
```

|| Return Value

```
IALG_EOK; /* status indicating success */
IALG_EFAIL; /* status indicating failure */
```

|| Description

`algInit()` performs all initialization necessary to complete the run-time creation of an algorithm instance object. After a successful return from `algInit()`, the instance object is ready to be used to process data.

The first argument to `algInit()` is a handle to an algorithm instance. This value is initialized to the base field of `memTab[0]`.

The second argument is a table of memory records that describe the base address, size, alignment, type, and memory space of all buffers allocated for an algorithm instance. The number of initialized records is identical to the number returned by a prior call to `algAlloc()`.

The third argument is a handle to the parent instance object. If there is no parent object, this parameter must be set to `NULL`.

The last argument is a pointer to a structure that defines the algorithm initialization parameters.

For more details, see *TMS320 DSP Algorithm Standard API Reference* (literature number SPRU360).

|| See Also

`algAlloc()`, `algMoved()`

4.3.3 Control API

Control API is used for controlling the functioning of the algorithm instance during run-time. This is done by changing the status of the controllable parameters of the algorithm during run-time. These controllable parameters are defined in the `Status` data structure (see Data Structures section for details).

|| Name

`control()` – change run-time parameters and query the status

|| Synopsis

```
XDAS_Int32 (*control) (IAUDDDEC_Handle handle, IAUDDDEC_Cmd  
id, IAUDDDEC_DynamicParams *params, IAUDDDEC_Status  
*status);
```

|| Arguments

```
IAUDDDEC_Handle handle; /* algorithm instance handle */  
IAUDDDEC_Cmd id; /* algorithm specific control commands*/  
IAUDDDEC_DynamicParams *params /* algorithm run time  
parameters */  
IAUDDDEC_Status *status /* algorithm instance status  
parameters */
```

|| Return Value

```
IALG_EOK; /* status indicating success */  
IALG_EFAIL; /* status indicating failure */
```

|| Description

This function changes the run-time parameters of an algorithm instance and queries the algorithm's status. `control()` must only be called after a successful call to `algInit()` and must never be called after a call to `algFree()`.

The first argument to `control()` is a handle to an algorithm instance.

The second argument is an algorithm specific control command. See `XDM_CmdId` enumeration for details.

The third and fourth arguments are pointers to the `IAUDDDEC_DynamicParams` and `IAUDDDEC_Status` data structures respectively.

Note:

If you are using extended data structures, the third and fourth arguments must be pointers to the extended `DynamicParams` and `Status` data structures respectively. Also, ensure that the `size` field is set to the size of the extended data structure. Depending on the value set for the `size` field, the algorithm uses either basic or extended parameters.

|| Preconditions

The following conditions must be true prior to calling this function; otherwise, its operation is undefined.

- ❑ `control()` can only be called after a successful return from `algInit()` and `algActivate()`.
- ❑ If algorithm uses DMA resources, `control()` can only be called after a successful return from `DMAN3_init()`.

|| Postconditions

- `handle` must be a valid handle for the algorithm's instance object.

The following conditions are true immediately after returning from this function.

- If the control operation is successful, the return value from this operation is equal to `IALG_EOK`; otherwise it is equal to either `IALG_EFAIL` or an algorithm specific return value.
- If the control command is not recognized, the return value from this operation is not equal to `IALG_EOK`.

|| Example

See test application file, `TestAppDecoder.c` available in the `\Client\Test\Src` sub-directory.

|| See Also

`algInit()`, `algActivate()`, `process()`

Note:

Audio codecs do not use `algActivate()`, `algDeactivate()`, and `DMAN3_init()` APIs.

4.3.4 Data Processing API

Data processing API is used for processing the input data.

|| Name

`process()` – basic encoding/decoding call

|| Synopsis

```
XDAS_Int32 (*process)(IAUDDDEC_Handle handle, XDM_BufDesc
*inBufs, XDM_BufDesc *outBufs, IAUDDDEC_InArgs *inargs,
IAUDDDEC_OutArgs *outargs);
```

|| Arguments

```
IAUDDDEC_Handle handle; /* algorithm instance handle */
XDM_BufDesc *inBufs; /* algorithm input buffer descriptor
*/
XDM_BufDesc *outBufs; /* algorithm output buffer descriptor
*/
IAUDDDEC_InArgs *inargs /* algorithm runtime input
arguments */
IAUDDDEC_OutArgs *outargs /* algorithm runtime output
arguments */
```

|| Return Value

```
IALG_EOK; /* status indicating success */
IALG_EFAIL; /* status indicating failure */
```

|| Description

This function does the basic encoding/decoding. The first argument to `process()` is a handle to an algorithm instance.

The second and third arguments are pointers to the input and output buffer descriptor data structures respectively (see `XDM_BufDesc` data structure for details).

The fourth argument is a pointer to the `IAUDDDEC_InArgs` data structure that defines the run time input arguments for an algorithm instance object.

The last argument is a pointer to the `IAUDDDEC_OutArgs` data structure that defines the run time output arguments for an algorithm instance object.

Note:

If you are using extended data structures, the fourth and fifth arguments must be pointers to the extended `InArgs` and `OutArgs` data structures respectively. Also, ensure that the `size` field is set to the size of the extended data structure. Depending on the value set for the `size` field, the algorithm uses either basic or extended parameters.

|| Preconditions

The following conditions must be true prior to calling this function; otherwise, its operation is undefined.

- ❑ `process()` can only be called after a successful return from `algInit()` and `algActivate()`.

- ❑ If algorithm uses DMA resources, `process()` can only be called after a successful return from `DMAN3_init()`.
- ❑ `handle` must be a valid handle for the algorithm's instance object.
- ❑ Buffer descriptor for input and output buffers must be valid.
- ❑ Input buffers must have valid input data.

|| Postconditions

The following conditions are true immediately after returning from this function.

- ❑ If the process operation is successful, the return value from this operation is equal to `IALG_EOK`; otherwise it is equal to either `IALG_EFAIL` or an algorithm specific return value.
- ❑ After successful return from `process()` function, `algDeactivate()` can be called.
- ❑ Example
- ❑ See test application file, `TestAppDecoder.c` available in the `\Client\Test\Src` sub-directory.
- ❑ See Also
- ❑ `algInit()`, `algDeactivate()`, `control()`

Note:

- ❑ Audio codecs do not use `algActivate()`, `algDeactivate()`, and `DMAN3_init()` APIs.

4.3.5 Termination API

Termination API is used to terminate the algorithm instance and free up the memory space that it uses.

|| Name

`algFree()` – determine the addresses of all memory buffers used by the algorithm

|| Synopsis

```
XDAS_Int32 algFree(IALG_Handle handle, IALG_MemRec  
memTab[]);
```

|| Arguments

```
IALG_Handle handle; /* handle to the algorithm instance */  
IALG_MemRec memTab[]; /* output array of memory records */
```

|| Return Value

```
XDAS_Int32; /* Number of buffers used by the algorithm */
```

|| Description

`algFree()` determines the addresses of all memory buffers used by the algorithm. The primary aim of doing so is to free up these memory regions after closing an instance of the algorithm.

The first argument to `algFree()` is a handle to the algorithm instance.

The second argument is a table of memory records that describe the base address, size, alignment, type, and memory space of all buffers previously allocated for the algorithm instance.

For more details, see *TMS320 DSP Algorithm Standard API Reference* (literature number SPRU360).

|| See Also

```
algAlloc()
```