AC3 Decoder Spec

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1. AC3 Codec Introduction

The AC3 digital compression algorithm can encode for 1 to 5.1 channels of source audio from a PCM representation into a serial bit stream at data rates ranging from 32kbps to 640kbps. The 0.1 channel refers to a fractional bandwidth channel intended to convey only low frequency (subwoofer) signals.

A typical application of the algorithm is shown in Figure 1.1. In this example, a 5.1 channel audio program is converted from a PCM representation requiring more than 5Mbps (6 channels $\times$48 kHz $\times$18 bits = 5.184 Mbps) into a 384 kbps serial bit stream by the AC-3 encoder.

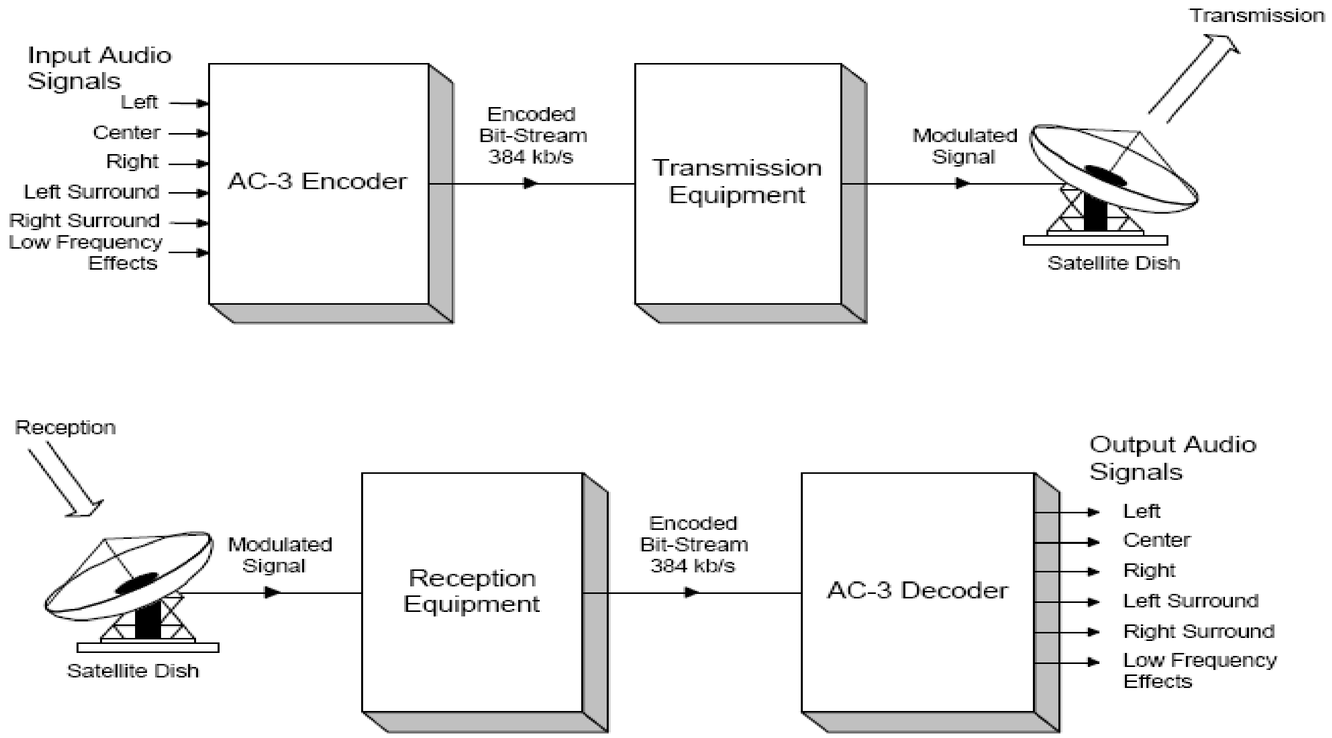

Figure 1.1. Example application of AC-3 to satellite audio transmission.

2. AC3 Decoder processing

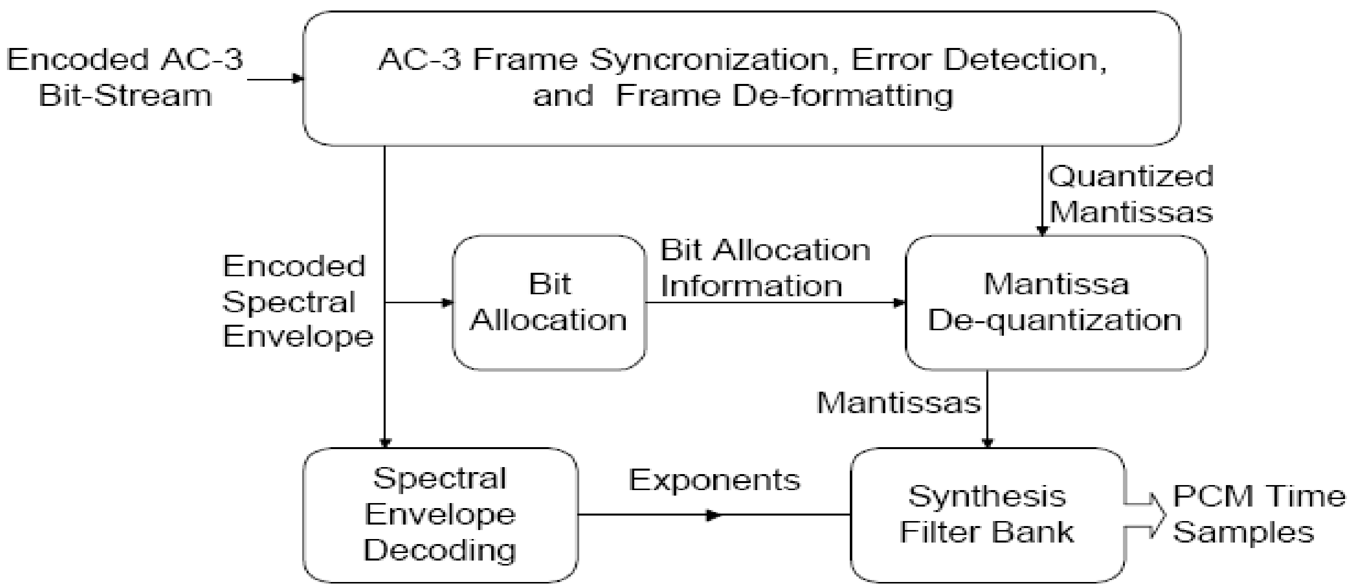

Figure 1.2. The AC-3 Decoder

3. AC3 Bit stream Syntax

1). Synchronization frame

An AC-3 serial coded audio bit stream is made up of a sequence of synchronization frames (see Figure 1.3.). Each synchronization frame contains 6 coded audio blocks (AB), each of which represents 256 new audio samples. A synchronization information (SI) header at the beginning of each frame contains information needed to acquire and maintain synchronization. A bit stream information (BSI) header follows SI, and contains parameters describing the coded audio service. The coded audio blocks may be followed by an auxiliary data (Aux) field. At the end of each frame is an error check field that includes a CRC word for error detection. An additional CRC word is located in the SI header, the use of which is optional.

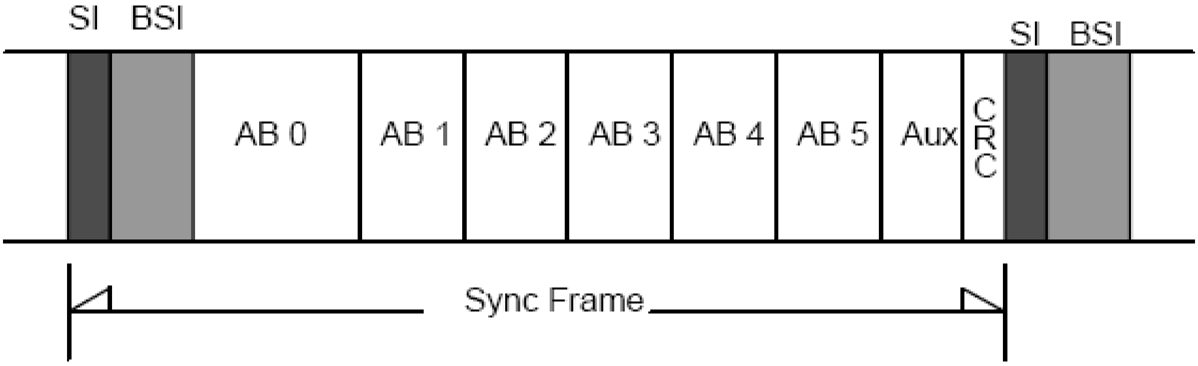

Figure 1.3. AC3 Synchronization Frame

2). SI --- synchronization Information

Syncword(16bits)

The syncword is always 0x0B77, or ‘0000 1011 0111 0111’. Transmission of the syncword, like other bit field elements, is left bit first.

Crc1(16bits)

This 16 bit-CRC applies to the first 5/8 of the frame. Transmission of the CRC, like other numerical values, is most significant bit first.

Fscod(2bits)

This is a 2-bit code indicating sample rate according to Table 1.1. If the reserved code is indicated, the decoder should not attempt to decode audio and should mute.

fscod	sampling rate, kHz
‘00’	48
‘01’	44.1
‘10’	32
‘11’	reserved

Table1.1 Sample Rate Code

Frmsizecod(6bits)

The frame size code is used along with the sample rate code to determine the number of (2-byte) words before the next syncword.

frmsizecod	nominal bit rate	fs = 32 kHz words/syncframe	fs = 44.1 kHz words/syncframe	fs = 48 kHz words/syncframe
'000000' (0)	32 kbps	96	69	64
'000001' (0)	32 kbps	96	70	64
'000010' (1)	40 kbps	120	87	80
'000011' (1)	40 kbps	120	88	80
'000100' (2)	48 kbps	144	104	96
'000101' (2)	48 kbps	144	105	96
'000110' (3)	56 kbps	168	121	112
'000111' (3)	56 kbps	168	122	112
'001000' (4)	64 kbps	192	139	128
'001001' (4)	64 kbps	192	140	128
'001010' (5)	80 kbps	240	174	160
'001011' (5)	80 kbps	240	175	160
'001100' (6)	96 kbps	288	208	192
'001101' (6)	96 kbps	288	209	192
'001110' (7)	112 kbps	336	243	224
'001111' (7)	112 kbps	336	244	224
'010000' (8)	128 kbps	384	278	256
'010001' (8)	128 kbps	384	279	256
'010010' (9)	160 kbps	480	348	320
'010011' (9)	160 kbps	480	349	320
'010100' (10)	192 kbps	576	417	384
'010101' (10)	192 kbps	576	418	384
'010110' (11)	224 kbps	672	487	448
'010111' (11)	224 kbps	672	488	448
'011000' (12)	256 kbps	768	557	512
'011001' (12)	256 kbps	768	558	512
'011010' (13)	320 kbps	960	696	640
'011011' (13)	320 kbps	960	697	640
'011100' (14)	384 kbps	1152	835	768
'011101' (14)	384 kbps	1152	836	768
'011110' (15)	448 kbps	1344	975	896
'011111' (15)	448 kbps	1344	976	896
'100000' (16)	512 kbps	1536	1114	1024
'100001' (16)	512 kbps	1536	1115	1024
'100010' (17)	576 kbps	1728	1253	1152
'100011' (17)	576 kbps	1728	1254	1152
'100100' (18)	640 kbps	1920	1393	1280
'100101' (18)	640 kbps	1920	1394	1280

3) Bsi --- Bit Stream Information
For more information, refer to the AC3 standards spec : ATSC_A52.pdf

4. Visualon AC3 decoder software release note

1) You can get the "Dolby Digital Decoder for Portable Solutions implementation Development Kit" --- CD1 and "Dolby Portable Solution implementation Development Kit" --- CD2, meanwhile you can get the Codec Certification pass report from Dolby.
Docs Directory: ../ trunk/Codec/Audio/AC3/DECODER/spec

2) APIs Description
Following SDK3.0 APIs

3) SetParam IDs Description

VO_PID_AC3_WORDSIZE:
Output PCM sample bits, Now only support 16-bit interger, but you can update these code, and support 17 ~ 32 bits
If(wordsize == 0)
  Support 16bits PCM output
Else if (wordsize == 1)
  Support 32bits PCM output
else
  Support 17~24bits PCM output

VO_PID_AC3_KCAPABLEMODE:
Karaoke capable reproduction mode
0 = no vocal
1 = left vocal
2 = right vocal
3 = both vocal (default) --- Karaoke capable mode

VO_PID_AC3_DRCMODE:
Compression mode out of range
Default: 2 --- line out

VO_PID_AC3_OUTLFEON:
Output subwoofer present flag
Default: 1 --- On

VO_PID_AC3_OUTPUTMODE:
Output channel configuration
Default: 2 --- L, R

VO_PID_AC3_NUMCHANS:
Output channel number
Default: 2 --- 2 channels

VO_PID_AC3_STEREOMODE:
Downmix Type, if you want to enable the ID, you have to set p_confparam->outputmode == 2 firstly
1 = Lt/Rt
2 = Lo/Ro

VO_PID_AC3_DUALMONOMOD:
Dual mono reproduction mode

VO_PID_AC3_USEVERBOSE:
Verbose message flag
It only bring into correspondence with the standerds algorithm, have not usefull for appilication

VO_PID_AC3_DYNX:
Dynamic range scale factor(high)
Default(0x7FFFFFFF)

VO_PID_AC3_DYNY:
Dynamic range scale factor(low)
Default (0x7FFFFFFF)

VO_PID_AC3_OUTPUTFLAG:
Output PCM flag
Set 1, enable dump PCM data, and Set0, disable dump PCM data

VO_PID_AC3_CHARI:
Channel routing information, Default: -0L -1R -2C -3l -4r -5s