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Introduction to Dolby Digital Plus, an Enhancement to the Dolby Digital Coding System

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ABSTRACT

An extension to the existing Dolby Digital (AC-3) multichannel audio coding standard is described and its new capabilities explored. This coding system is designed with extensive compatibility with the existing standard by retaining most of its metadata and data-framing structure to preserve and extend functionality in existing applications. New features include simultaneous support for multiple program streams, carriage of multichannel signals beyond 5.1 channels, and fine-grained control and support for data rates up to 6.144 Mbps. New coding “tools” including spectral extension, enhanced channel coupling, transient pre-noise processing, and improved filterbank/quantization enhance the performance of earlier AC-3 technology.

1. INTRODUCTION

The existing multichannel coding standard, AC-3, has found widespread use in popular audio and audio/video applications. This fact, coupled with interest in a wider range of supported data rates, channel formats, and reproduction circumstances, led to the requirement for an AC-3 compatible coding system with additional functionality. The new system is called Dolby Digital Plus or Enhanced AC-3 (E-AC-3).

The added functionalities are shown to be useful in applying this new system in a number of emerging

applications. Some of these applications are examined here.

The existing AC-3 coding standard is reviewed, and the additions, modifications, and extensions employed in this advanced system are described. The benefits of building on the existing coding standard are also discussed.

1.1. Increased Flexibility

This new coding system is based on the existing AC-3 standard by preserving the present metadata carriage, underlying filterbank, and basic framing structure. The operating range has been increased by allowing data rates spanning 32 kbps – 6.144 Mbps. In addition, the data rate control has a finer resolution, as little as 0.333 bits per second (bps) at a sample rate of 32 kHz and a six-block transform frame. The data-rate control resolution is proportional to sample rate and inversely proportional to frame size.

This enhancement of AC-3 preserves the AC-3 frame structure of six 256-coefficient transforms, while also allowing for shorter frames composed of one, two, and three 256-coefficient transform blocks. This new feature enables the transport of audio at data rates greater than 640 kbps in formats limiting the amount of data per frame, such as the DVD.

The flexibility of the channel format has been greatly expanded to allow for a significantly larger number of channels than 5.1 (five main channels, plus the low frequency effects channel, denoted “.1”). This is valuable for the support of 6.1 or 7.1 channels being deployed today, and allows for the carriage of at least 13.1 channels to support home delivery of future multichannel audio sound tracks that might stem from digital cinema applications.

The increased channel capacity is obtained by associating the main audio program bitstream with up to eight additional substreams, all of which are multiplexed into one E-AC-3 bitstream. This allows the main audio program to convey the existing 5.1-channel format of AC-3, while the additional channel capacity comes from the dependent bitstreams. This means that a 5.1-channel version and the various conventional downmixes to mono are always available and that matrix subtraction-induced coding artifacts are eliminated by the use of a channel substitution process.

Multiple program support is also available through the ability to carry seven more independent audio streams, each with possible associated substreams, to increase the channel carriage of each program beyond 5.1 channels.

1.2. New Coding Tools for Greater Efficiency

Coding efficiency has also been increased to allow for the beneficial use of lower data rates. This is accomplished using an improved filterbank, improved quantization, enhanced channel coupling, spectral extension, and a technique called transient pre-noise processing.

The improved filterbank is a result of the addition of a second stage DCT after the existing AC-3 filterbank when audio with stationary characteristics is present. This converts the six 256-coefficient transform blocks into a single 1536-coefficient hybrid transform block with increased frequency resolution.

This increased frequency resolution is combined with 6-dimensional Vector Quantization (VQ) and Gain Adaptive Quantization (GAQ) to improve the coding efficiency for “hard to code” signals such as pitch pipe and harpsichord. VQ is used to efficiently code frequency bands requiring lower accuracies, while GAQ provides greater efficiency when higher accuracy quantization is required.

Improved coding efficiency is also obtained through the use of channel coupling with phase preservation. This method expands on the AC-3 method of employing a high frequency mono composite channel which reconstitutes the high-frequency portion of each channel on decoding. The addition of phase information and encoder-controlled processing of spectral amplitude information sent in the bitstream improves the fidelity of this process so that the mono composite channel can be extended to lower frequencies than was previously possible. This decreases the effective bandwidth encoded, and thus increases the coding efficiency.

Another powerful tool added is spectral extension. This method builds on the channel coupling concept by replacing the upper frequency transform coefficients with lower frequency spectral segments translated up in frequency. The spectral characteristics of the translated segments are matched to the original through spectral modulation of the transform coefficients.

An additional technique to improve audio quality at low data rates is the use of transient pre-noise processing. This is a post-decoding process that reduces pre-noise error via time scaling synthesis techniques which reduce pre-noise duration and therefore audibility of transient artifacts common to bit-rate-reduction audio coders. The

parameters used in the post-decoding time scaling synthesis processing, which employs auditory scene analysis, are provided by metadata calculated by the encoder and transmitted in the E-AC-3 bitstream.

2. OVERVIEW OF AC-3

E-AC-3 is best understood by comparison with AC-3. E-AC-3 is essentially AC-3 with increased flexibility, expanded metadata functionality, and improved coding tools. Given the close relationship, a brief description of standard AC-3 is appropriate here. Beyond this overview, further information about AC-3 can be found in the ATSC standard [1] or written descriptions by Fielder et al. [2] and Davidson [3].

AC-3 is a multichannel coding system that uses either a 256-coefficient filterbank or a pair of 128-coefficient filterbanks, based on the MDCT, to perform the frequency division process and allows for carriage of one to five channels of audio, plus a low frequency effects channel. In addition, the AC-3 encoder obtains better spectral power estimation through the power summation of modified discrete sine transform (MDST) spectral values, along with those of the modified discrete cosine transform (MDCT). See Princen and Bradley [4] for further information on the MDCT and MDST.

The choice of these particular transform lengths was based on balancing the coding efficiency benefits of a high frequency-resolution filterbank with the memory costs to implement it. Shorter length transforms are used during transient signal conditions to better optimize the time-frequency tradeoff for these types of signals.

The selection of the transform length is performed by a transient detector in the AC-3 encoder and a parameter is transmitted in the bitstream to inform the decoder of the transform length. This detector first performs a high-pass filter on the input signal with a 4th order Chebychev filter with an 8 kHz bandwidth. A 256-sample frame of the filtered signal is divided into four equal time segments for analysis. The time segments are examined for an increase in energy from one time segment to the next. Segments are also examined at different time scales. If a significant increase in signal energy is detected, a transient is indicated. More detail on transient detection can be found in the AC-3 ATSC standard [1].

Quantization of the transform coefficients is performed by linear quantization of the mantissas of an exponent-mantissa representation.

The exponents are the coarse quantization of transform coefficient levels in steps of 6.02 dB. The first exponent is coded as a single 4-bit value with a range of 0 to -90.30 dB, and each subsequent exponent is coded as a difference of 0 dB, ± 6.02 dB, or ± 12.04 dB from the previous exponent. The differences are grouped into triplets and coded as a 7-bit word. Exponents are associated with one (D15 state), two (D25 state), or four (D45 state) frequency-adjacent mantissas and are shared in time over as many as six transform blocks. The encoder picks the best balance of sharing over time and groups of one, two, or four mantissas.

Mantissas are exponent-scaled versions of the transform coefficients. These are linearly quantized, based on the assumption that the exponent scaling process more fully loads the quantizer levels, resulting in a probability distribution of quantizer values better approximating a uniform distribution. As a result, the linear quantization performs well.

The bit-allocation process drives the setting of mantissa quantization accuracy. Coarsely quantized mantissas are coded with a symmetric level quantizer of 3, 5, 7, 11 or 15 levels. The fine resolution mantissas are quantized with an asymmetrical quantizer and coded using a standard 2's complement word. The three-level quantized mantissas are grouped into triplets and coded as 5-bit words, the 5-level quantized mantissas are grouped into triplets and coded as 7-bit words, and the 11-level quantized mantissas are grouped into pairs and coded as 7-bit words. The rest of the quantized mantissas are singly coded as binary words of length 3 – 16 bits.

The bit-allocation control in AC-3 is a hybrid of a backward and forward adaptive process since the perceptual model and bit-allocation process use parametric variables to modify the action of its basically backwards adaptive decoder bit-allocation algorithm, see Davidson et al. [5] for more details.

The exponents are grouped into masking bands approximating one half an auditory Critical Bandwidth, as defined by Zwicker [6]. These exponents within a band are summed together on a power basis to result in a masking-band-level spectrum, which is combined with a hearing threshold spectrum to derive a masking

threshold spectrum. The difference between these two spectra indicates the required accuracy of quantization for the mantissas associated with the exponents within each masking band. This difference is scaled by a variable called the *snroffset*, which adjusts the data rate given to the mantissas so that the overall data rate of the encoded signal matches a given target data rate. The use of the *snroffset* parameter to adjust the mantissa accuracy results in the best match to the masking and hearing threshold characteristics of the listener for a given data rate. The bit-allocation process remains essentially unchanged in the E-AC-3 system.

A distinguishing feature of AC-3 is that it extensively employs multichannel coding methods. It processes multiple channels as a single ensemble to improve coding efficiency and is designed for single point to multi-point consumer applications. Above a designated frequency some or all channels are summed together on a spectral basis in bands that are multiples of 12 transform coefficients in size.

These frequency coefficients for the coupled channel are transmitted in place of the individual channel spectra, together with additional side information. The side information consists of a unique set of coupling coefficients for each channel.

In the decoder, frequency coordinates for each output channel are computed by multiplying the coupled channel frequency coordinates by the coupling coordinates for that channel. Coupling coordinates are computed in the encoder in a manner which preserves the short-time energy envelope of the original signals, thereby preserving spatial cues used by the listener.

The coupling coordinates for each channel and coupling band are coded in an exponent-mantissa form. The exponents are 4-bit words, representing steps of 6.02 dB that span the range of 0 to -90.30 dB, while the mantissas are positive value, 4-bit linear words. Additionally, each channel's spectral levels are scaled by a 2-bit master exponent that scales each channel's spectral levels by 0, -18.06, -36.12, or -54.19 dB.

AC-3 makes extensive use of channel downmixing and dynamic level compression to provide a high degree of adaptability to consumer playback conditions. The use of dynamic level compression in the decoder allows for a decoder adjustment of the amount of dynamic level compression employed and the possibility to avoid its use in the highest fidelity applications.

Dynamic level compression and static level control take place at the decoder via bitstream metadata. The overall gain is set via a dialog normalization word sent at 1536-sample intervals and the dynamic level compression words supplied at both 1536 and 256 sample intervals. This remains unchanged in E-AC-3 when the frame length is six transform blocks in length.

AC-3 includes provisions for sending metadata, i.e., multiple auxiliary data streams, language identifiers, copyright protection, time stamps, and control information.

The AC-3 algorithm is capable of being used at sample rates of 32, 44.1 and 48 kHz. Its data-rate options are 32, 40, 48, 56, 64, 80, 96, 112, 128, 160, 192, 224, 256, 320, 384, 448, 512, 576, and 640 kbps.

AC-3 supports channel configurations of 3/2, 3/1, 3/0, 2/2, 2/1, 2/0, 1/0, in format X/Y, where X represents the number of forward channels and Y the number of rear channels. Additionally, a low frequency effects channel can be carried, and a dual mono configuration is supported.

3. NEW APPLICATIONS

A number of important new applications for which E-AC-3 is well suited are emerging in the audio-video marketplace today. These are: lower data-rate carriage of audio and its conversion to the AC-3 coding standard for playback on today's installed base of audio/video entertainment equipment; interactive multimedia capabilities that allow the combination of streamed content with a main audio program; the reproduction of greater than 5.1 channels for the support of playback of existing and future cinema content; and the efficient transcoding of AC-3 program content to lower data-rate E-AC-3 bitstreams and conversion back to AC-3 for playback on the installed base of AC-3 decoders. These applications will be discussed in the following section.

3.1. Compatible Lower Data-Rate Carriage

There are a growing number of applications that require lower data rates, but also require compatibility to the existing broadcast-reception and audio/video decoding infrastructure.

The E-AC-3 system is an excellent solution for these applications because of its inherent lower tandem coding losses with AC-3 and its greater coding

efficiency provided by the new coding tools. Because of the very large installed base of AC-3 decoders, the E-AC-3 system has been designed to permit a very low loss conversion to standard AC-3 over a digital audio interconnect such as S/PDIF and decoding by a standard AC-3 decoder.

The conversion stage is a special form of transcoder that minimizes quality degradations resulting from tandem coding losses. This is feasible with the use of the same filterbank, transform block alignment, bit-allocation process, and basic framing structure.

The reduction of the tandem losses results from the use of a frame-synchronous partial E-AC-3 decoder and partial AC-3 encoder that are designed to minimize changes in the original quantization for mantissas. The minimization of changes in the quantization of the transform coefficients is essential for the reduction of tandem coding losses. See Sections 6 and 7 for more details. A 2-channel PCM output is produced by performing the rest of the decoding, downmixing, and dynamic level control.

Two applications that are likely to benefit from lower data rates and compatibility with existing home theater systems are lower data-rate cable television audio and fallback audio under impaired terrestrial digital television reception conditions.

3.1.1. Cable/Satellite Television Set Top Box

The cable/satellite television application is very similar to the conventional AC-3 reception paradigm except that the need for greater channel capacity requires the transmission of audio programming at lower data rates than is typical for AC-3 applications. Traditionally, AC-3 has been employed at 128 – 192 kbps for stereo and 384 – 448 kbps for 5.1-channel applications. The use of the new coding tools in E-AC-3 allows for the practical use of lower data rates while permitting efficient conversion into a conventional AC-3 bitstream at 640 kbps for compatibility with existing home theaters. Figure 1 shows the configuration of this converter/decoder.

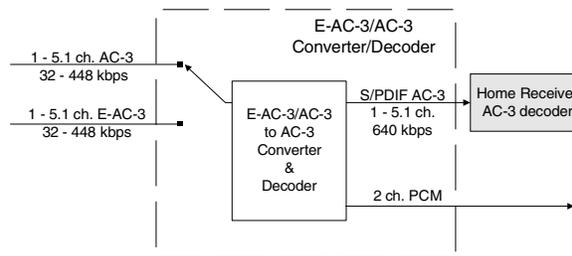


Figure 1: Cable/Satellite set-top box converter/decoder (existing AC-3 devices in gray).

The device shown in figure 1 accepts both the E-AC-3 and AC-3 bitstreams but always outputs standard AC-3 as the consumer switches channels within the network. The greater efficiency of the E-AC-3 system allows for a greater number of programs within the cable system while preserving full functionality for legacy receiver hardware, shown with a gray interior.

The conversion of the E-AC-3 bitstream is done via the process described in Section 6. In addition, the conversion of the input to output AC-3 bitstream is done by padding out the output AC-3 with the appropriate number of bits to result in a 640 kbps output.

This application requires that the E-AC-3/AC-3 converter decodes and creates a valid AC-3 bitstream in S/PDIF form as it switches between programming with both AC-3 and E-AC-3 bitstreams. The channel format of the input bitstream is preserved but the output data rate is always 640 kbps. This application involves the use of E-AC-3 data rates as low as 96 kbps and 192 kbps, for stereo and 5.1-channel configurations, respectively.

3.1.2. Terrestrial Broadcast Fallback Audio

The fallback audio application is similar to the cable set top box application except that it requires more seamless switching between standard AC-3 and the lower data-rate E-AC-3 bitstreams carried on the more robust terrestrial broadcast channel. This channel is carried using an improved data carriage method that uses enhanced vestigial sideband modulation (E-VSB), for further details see Bretl et al. [7] and Gaddam [8]. This paradigm is shown in figure 2.

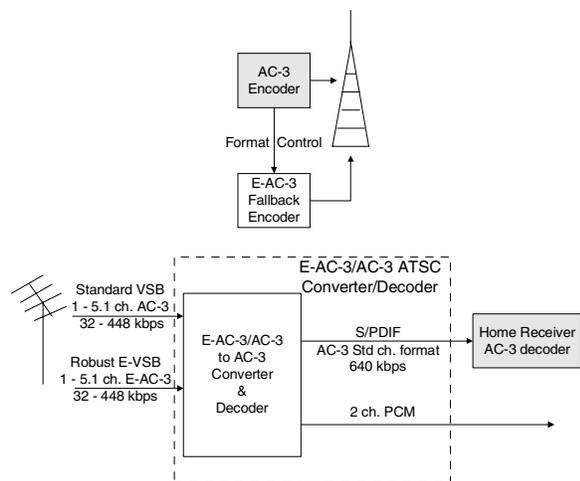


Figure 2: ATSC terrestrial television fallback audio converter/decoder (existing AC-3 devices in gray).

Examination of this figure shows that the E-AC-3/AC-3 converter can switch from the AC-3 to the lower data-rate E-AC-3 bitstream when reception conditions demand it. Both bitstreams can contain 1 – 5.1-channel programs at data rates between 32 – 448 kbps. The E-AC-3 bitstream has similar data rates to those of the previous application and is dynamically selected when the standard VSB channel is corrupted or unavailable due to poor reception conditions.

Although each bitstream may represent a different combination of channel configurations and data rates, the converter/decoder always supplies an AC-3 bitstream in the original AC-3 channel configuration at a 640 kbps data rate. This prevents any possible problems in the home receiver since the data rate and channel format remain constant as the reception conditions vary. The input channel configuration of the fallback channel is appropriately upmixed to the standard channel format.

3.2. Interactive Multimedia Playback

Another new application involves the addition of interactive content support to traditional audio/video programs. This requires the development of an E-AC-3 mixer-converter-decoder that provides the ability to mix an additional E-AC-3 program bitstream into an existing multimedia program bitstream, such as that coming from a DVD.

A system of this type is very useful in adding special effects or director's commentary on an already released DVD. The special effects or commentary track can originate from the Internet and be used to supply "late release" additional programming or be present as another feature on the DVD. The mixing process should allow for suitable attenuation, compression, and mixing of the sources while preserving the metadata functionality of the resultant AC-3 bitstream. The appropriate system is shown in figure 3.

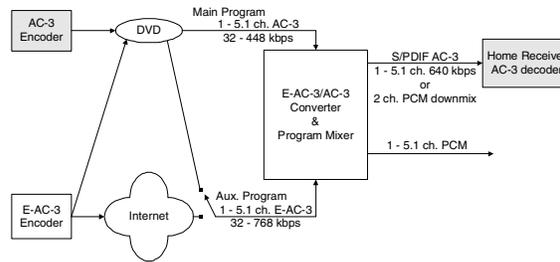


Figure 3: Interactive DVD mixer-converter-decoder (existing AC-3 devices in gray).

This E-AC-3 mixer-converter-decoder takes advantage of the additional metadata present in E-AC-3 to allow for mixing of the main and auxiliary program bitstreams with the appropriate mixing levels and dynamic level compression. The E-AC-3 auxiliary bitstream acts as the controlling element for the mixing process so main programs embodied in both AC-3 and E-AC-3 bitstreams can be utilized. As before, the conversion to the output AC-3 stream is done in such a way to provide artifact-free operation for the final AC-3 decoder.

3.3. Advanced Multimedia Carriage

The E-AC-3 system allows for carriage of more channels than the 5.1 channels transported by AC-3. This is accomplished through use of multiple bitstreams embedded in the E-AC-3 framing structure, as will be described later.

The channels are arranged as a combination of the 5.1-channel downmix and additional channels are substituted to construct the larger number-of-channels configuration. For instance, a 7.1-channel configuration is composed of the three front channels of the 5.1-channel downmix and four additional rear channels that substitute for the original two rear channels.

Figure 4 shows an E-AC-3 converter/decoder configuration that accepts either a conventional AC-3 bitstream of 1 – 5.1 channels, or an E-AC-3 bitstream of up to 13.1 channels composed of the main bitstream and two dependent substreams. The system shown produces an AC-3 compatible 5.1-channel downmix, a 5.1-channel downmix in PCM form, a pass through of the original bitstream, and up to 13.1-channel PCM.

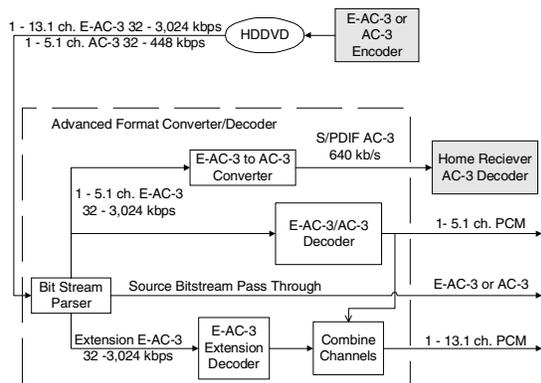


Figure 4: Advanced channel configuration converter/decoder (existing AC-3 devices in gray).

Examination of this figure shows that the combination of a bitstream parser, E-AC-3 to AC-3 converter, E-AC-3 extended parser, E-AC-3 extended bitstream decoder, E-AC-3/AC-3 decoder, and PCM channel combiner provides a wide variety of channel downmixes and extended channel reproduction without matrix induced artifacts.

The channel substitution method is chosen to avoid the unmasking of coding errors that can occur when a downmix is upconverted to the original channel configuration by matrixing that involves subtraction of signal components. This effect is important to avoid as it is a major problem for the MPEG-2 Layer II and III when extended to 5.1-channel operation. For more details see Grill [9].

3.4. AC-3 to E-AC-3 Transcoder

The increasing number of channels and services supplied by cable television networks has produced the need for audio and video distribution at even lower data rates than before. One such broadcasting-distribution paradigm is shown in figure 5.

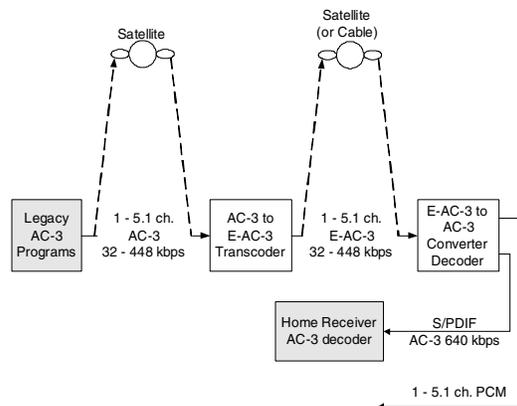


Figure 5: Distribution of legacy AC-3 content over an original and a lower data-rate link (existing AC-3 devices or content in gray).

Figure 5 shows the circumstance where AC-3 content is distributed over two network links, one allowing the AC-3 content to be sent as originally encoded and another link requiring transmission at a lower data rate. The second link requires transcoding to a lower data rate and then conversion back to AC-3 for 5.1-channel content.

This presents a very difficult situation because there is the potential for two additional generations of tandem coding loss, above and beyond the original source material's quality limitations. Each additional generation loss reduces the effective data-rate savings. In addition, metadata included in the original AC-3 bitstream is lost if transcoded to an unrelated coder. The use of E-AC-3 is particularly desirable as it provides greater coding efficiency without sacrificing the related metadata, while introducing less quality loss from tandem coding effects. The above factors result in acceptable quality for broadcast situations at E-AC-3 data rates of 300 kbps and below for 5.1-channel applications.

4. NEW CODING TOOLS

The improved functionality and flexibility is a result of the addition of a number of new coding tools and enhancements to the bitstream format of E-AC-3. These augmentations are described in the following sections. The improved filterbank, quantization, coupling, and transient pre-noise processing will be explained.

4.1. High Resolution Transform and Quantization Tools

AC-3 employs a relatively short transform and simple scalar quantization to perceptually code audio material (see Section 2 for more details). This approach met audio coding requirements in the 1990s without adding undue computational complexity; however, with today's requirement for higher coding gain in E-AC-3, improved spectral resolution, quantization, and coding are necessary.

This section describes the adaptive hybrid transform and the new quantization tools available in E-AC-3. The combination of higher resolution bit allocation, vector quantization, and gain adaptive quantization, along with the improved spectral resolution provided by the hybrid transform, allows for better compression of pseudo-stationary material.

4.1.1. Adaptive Hybrid Transform

The adaptive hybrid transform (AHT) is composed of two linear transforms connected in cascade. The first transform is an MDCT which produces 256 transform coefficients, and employs a Kaiser Bessel derived window (KBD) with an alpha factor of 5.0, as defined by Fielder et al. [2]. This provides a level of interoperability with AC-3 without the need to return to the time domain. However, the 256 transform coefficient MDCT used in AC-3, while appropriate for non-stationary audio signals, provides limited spectral resolution and hence limited coding gain for stationary audio signals.

In order to alleviate this condition, the AHT system includes a secondary block transform in the form of a non-windowed, non-overlapped Type II discrete cosine transform (DCT), as defined by Rao and Yip [10]. This transform provides improved coding efficiency.

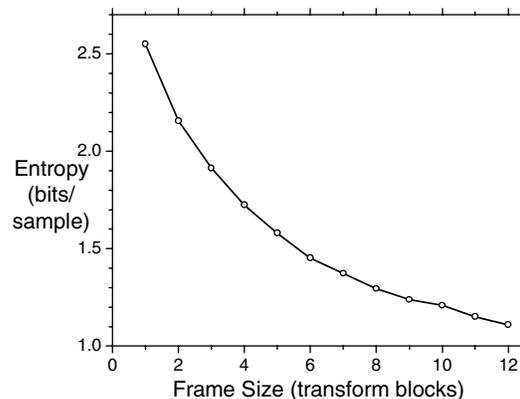


Figure 6: Entropy versus DCT block length for a stationary signal (pitch pipe) as used in AHT.

Figure 6 demonstrates the entropy, as defined by Cover and Thomas [11], for one audio signal (pitch pipe) as a function of DCT block length. The figure indicates a notable reduction in entropy as the length of the DCT is increased. Since the DCT is non-windowed and non-overlapped, the length of the DCT can in principle be varied in time and frequency depending on signal conditions. The arbitrary adaptation of AHT in time and frequency is not supported in E-AC-3; however, the bitstream syntax provides for adaptively switching the DCT in or out (AHT on or off) depending on signal conditions.

Computation of AHT is given in the following equations. Equation 1 is the computation of the MDCT where the window used ($w(n)$) is the KBD.

$$X(k, m) = \sum_{n=0}^{N-1} x\left(n + m \frac{N}{2}\right) \cdot w(n) \cdot \cos\left(\frac{2\pi\left(k + \frac{1}{2}\right) \cdot \left(n + \frac{N}{4} + \frac{1}{2}\right)}{N}\right) \quad (1)$$

where

$$k = 0, 1, \dots, \frac{N}{2} - 1;$$

N = the length in samples of the KBD window;

n = the sample index;

and

m = the block index.

Equation 2 gives the computation of the DCT type 2 over successive blocks in each bin, hence creating a higher resolution time to frequency mapping.

$$X^{AHT}(kM+l) = \frac{\sqrt{2}}{M} C_l \sum_{m=0}^{M-1} X(k,m) \cdot \cos\left(\frac{(2m+1)l\pi}{2M}\right) \quad (2)$$

where

M = the number of blocks over which the DCT is applied;

$$l = 0, 1, \dots, M-1.$$

and

$$C_l = \begin{cases} 1/\sqrt{2} & l = 0 \\ 1 & l \neq 0 \end{cases}$$

The application of AHT in E-AC-3 is limited to the case in which the exponent strategy is shared over all six blocks in a frame. Referring to Equation 2, $M = 6$.

4.1.2. High Precision Bit Allocation and Vector Quantization

In AC-3, the bit-allocation level, defined by the bit-allocation pointer (*bap*) variable, is computed as the difference between the mask level and the power spectral density estimate, in approximately 6 dB increments. As a result, some spectral coefficients will receive up to one bit higher allocation than is strictly necessary as determined by the psychoacoustic masking model. To reduce this error, the new quantization techniques applied with AHT use a high efficiency *bap* parameter called *hebp* and vector quantization (VQ), as explained by Gersho and Gray [12], which allow for 1.5 dB increments in noise allocation at the lowest allocation levels. The vector quantizer uses six dimensions and table lengths ranging from 4 to 512, depending on the *hebp* level, which is detailed in Table 1. Table 1 also shows the average SNR in dB achieved by the VQ tables.

<i>hebp</i> value	VQ Table Length	Average SNR of VQ Table
0	0	N/A
1	4	1.32 dB
2	8	2.29 dB
3	16	3.79 dB
4	32	4.92 dB
5	128	6.94 dB
6	256	8.20 dB
7	512	9.43 dB
8 – 20	GAQ	N/A

Table 1 VQ table length for each *hebp* level.

4.1.3. Gain Adaptive Quantization

Gain-adaptive quantization (GAQ) is a method for quantizing transform coefficient mantissas using variable-length codewords when the desired mantissa accuracy is greater than that covered by the VQ process. This technique was developed for use in the Dolby E audio distribution codec, see Truman et al. [13] and Fielder and Davidson [14] for more detail.

GAQ is based on conditionally amplifying one or more of the smaller and typically more frequently occurring transform coefficient mantissas in a single block, and representing these with a shorter uniform length code. Larger mantissas always bypass the amplifier, and are transmitted using longer uniform length codes. Even though larger mantissas require longer codes, an overall savings in bit-rate often results since small mantissas occur much more frequently in typical audio signals.

In E-AC-3, a gain word called G_k is selected by the encoder for each GAQ-coded DCT block, of length six. The gain is packed together with the quantized mantissas and transmitted as side information. The gain can assume values of 1, 2, or 4. With gain-adaptive quantization, the encoder is able to adapt to changing local signal statistics from frame to frame, and/or from channel to channel. Since a coding mode using uniform-length output symbols is included as a special case, gain-adaptive quantization cannot cause a noticeable coding loss compared to the codes employed in AC-3.

For a DCT block having a predominately uniform amplitude distribution, a gain of 1 will be selected. This results in uniform-length mantissa codewords. As the amplitude distribution (pdf) becomes increasingly skewed toward small values, gains of 2 and then 4 become advantageous. Since the AHT DCT is typically

enabled only for audio frames containing stationary signals, the DCT typically produces transform coefficient mantissas that are significantly skewed toward small values.

In the decoder, the individual gain words are unpacked first, followed by a bitstream parsing operation (using the gains) to reconstruct the individual transform coefficient mantissas. To compensate for amplification applied in the encoder, the decoder applies a corresponding attenuation of 1, 0.5, or 0.25 to the small mantissas. The large mantissas always bypass the attenuator.

Four different gain transmission modes are available for use in the encoder. The different modes employ switched 0, 1 or 1.67-bit gains. For each independent, coupling, and LFE channel in which AHT is in use, a 2-bit parameter called *gaqmod* is transmitted once per frame to the decoder. The bitstream parameters, values, and active *hebp* range are shown for each mode in Table 2. If *gaqmod* = 0x0, GAQ is not in use and no gains are present in the bitstream. This case emulates the uniform-length quantization scheme used in AC-3. If *gaqmod* = 0x1, a 1-bit gain value is present for each block of DCT coefficients having an *hebp* value between 8 and 11, inclusive. Coefficients with *hebp* higher than 11 are decoded using the same quantizer as *gaqmod* = 0x0. If *gaqmod* = 0x2 or 0x3, gain values are present for each block of DCT coefficients having an *hebp* value between 8 and 16, inclusive. Coefficients with *hebp* higher than 16 are decoded using the same quantizer as *gaqmod* = 0x0. The difference between the two last modes lies in the gain word length, as shown in table 2.

<i>gaqmod</i>	GAQ Mode for the frame	Active <i>hebp</i> Range
0x0	GAQ not in use	None
0x1	1-bit gains ($G_k = 1$ or 2)	$8 \leq hebp \leq 11$
0x2	1-bit gains ($G_k = 1$ or 4)	$8 \leq hebp \leq 16$
0x3	1.67 bit gains ($G_k = 1, 2, \text{ or } 4$)	$8 \leq hebp \leq 16$

Table 2 Gain adaptive quantization modes.

For the case of *gaqmod* = 0x1 and 0x2, the gains are coded as 1-bit values. For the case of *gaqmod* = 0x3, the

gains are composite-coded in triplets (three 3-state gains packed into 5-bit words).

4.2. Spectral Extension

Spectral extension is a process which synthesizes high frequency transform coefficients. In E-AC-3, spectral extension is implemented in a way similar to coupling, as described in Section 2. Spectral extension achieves more data-rate reduction than coupling because it does not require the transmission of a mono composite channel. Only metadata related to high frequency synthesis is transmitted. Additionally, it can be used for mono signals, whereas channel coupling requires at least two channels.

In E-AC-3 spectral extension may be used in conjunction with coupling or enhanced coupling. In this mode of operation, the coupling region resides between the baseband and the spectral extension regions.

4.2.1. System Description

At the core of spectral extension is its banding structure. Similar to coupling, energy ratios and region borders are calculated on a banded basis to reduce data rate. Spectral extension bands that approximately match Critical Bands are used for optimal tradeoff between data rate and sonic quality.

A sample spectrum is shown in figure 7. It is divided into two regions: a baseband region and an extension region. The extension region is further divided into bands according to the banding structure. A single energy ratio is calculated for each band. Also, the degree of noise-like character of the extension region is measured, and from that measurement a single noise blending parameter for the entire region is derived. After calculating these parameters, the original extension region transform coefficients are discarded.

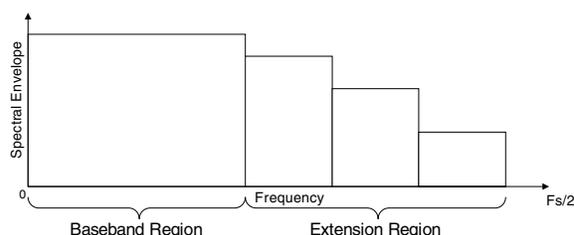


Figure 7: Original spectrum.

Figure 8 illustrates the translation process. The translation process involves linear copying of transform coefficients from the baseband into the extension region.

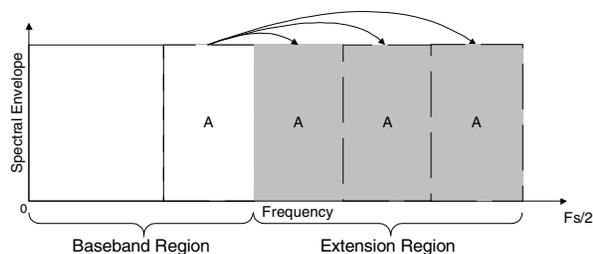


Figure 8: Translation.

Through inspection of figure 8, note that the translated components are taken from a section of the baseband called the copy region (CR). Typically, the copy region is smaller than the extension region, and thus it is necessary to translate the coefficients in the copy region multiple times to completely fill the extension region. This process is referred to as 'wrapping.' In figure 8, the copy region has been wrapped two times. Wrapping causes a discontinuity in the translated spectrum, as it places two transform coefficients from separate portions of the spectrum adjacent to one another. Because a single scale factor is used for each band, if wrapping occurs within a band, the scale factor for the band cannot reduce the discontinuity. However, if wrapping occurs only at band borders, the scale factors for the adjacent bands can reduce the discontinuity. The translation function therefore ensures that wrapping always occurs between bands.

Simple linear translation from baseband coefficients may not result in a natural sounding signal. One reason for this is that most signals become more noise-like as frequency increases. Therefore, the translated transform coefficients are blended with a noise spectrum according to a noise-blending function. Because most signals become more noise-like as frequency increases, the noise blending function is weighted toward higher frequencies. The noise blending function is linear with a slope that depends on the bandwidth of the original signal and an intercept that depends on the noise blending parameter.

Once the noise blending function has been determined, a noise spectrum is generated whose banded energies are equivalent to the banded energies of the translated

bands. The noise spectrum is then scaled by the noise-blending function. One such noise spectrum is shown in figure 9.

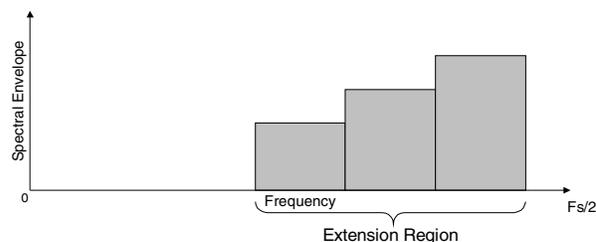


Figure 9: Noise spectrum, multiplied by blending function.

In preparation for blending, the translated spectrum is multiplied by the inverse of the noise blending function, as shown in figure 10.

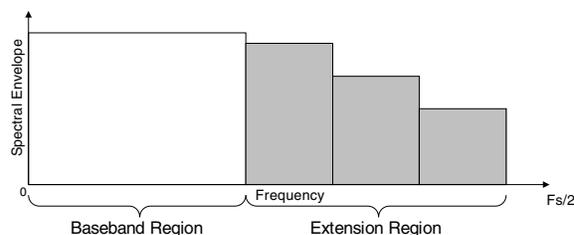


Figure 10: Translated spectrum, multiplied by inverse of blending function.

The translated spectrum and noise spectrum are then blended together, as shown in figure 11. Note that the energy of each blended band matches the energy of the translated bands in figure 8.

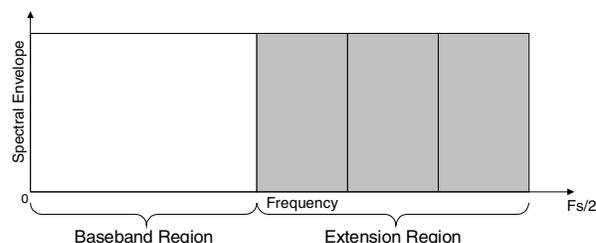


Figure 11: Blended spectrum.

Finally, the blended, translated spectrum is scaled by the energy ratios to match the banded envelope of the original spectrum, shown in figure 12.

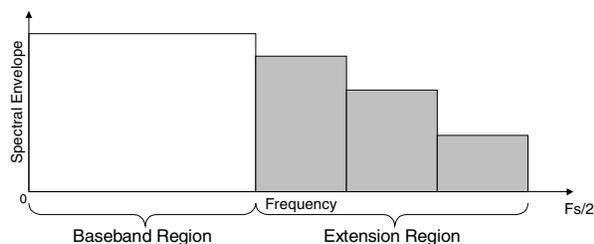


Figure 12: Final blended spectrum, multiplied by energy ratios.

4.2.2. Encoder Operation

This section describes E-AC-3 spectral extension encoder operations. The encoder analyzes the extension region to calculate the noise-blending parameter, sets up the banding structure, and simulates decoder translation to calculate energy ratios. Spectral extension encoding is illustrated as a block diagram in figure 13.

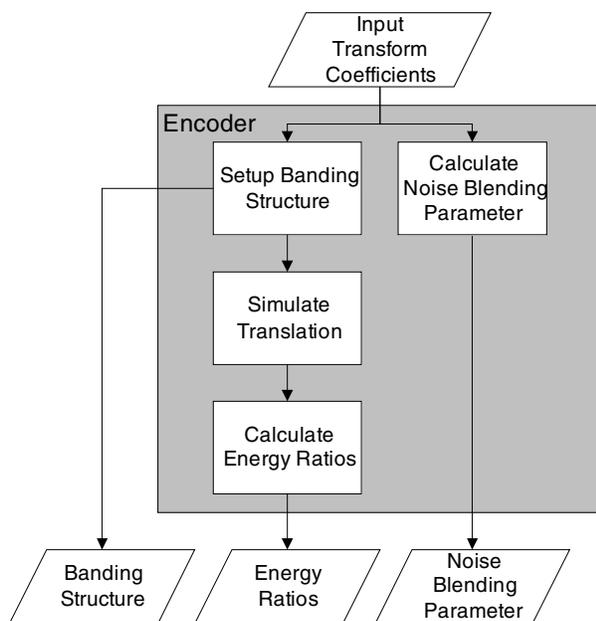


Figure 13: Spectral extension encoding.

To facilitate proper noise blending in the decoder, a noise-blending parameter is calculated and transmitted. For best operation, the original extension region spectral

envelope is analyzed for two terms: the rate of change in angle precession, and the amount of change in magnitude over time. If either term is inconsistent, the original spectrum is likely to be noisy, and more noise should be blended after translation. If the terms are consistent, it is likely to contain important harmonic components, and less noise should be blended. The noise-blending parameter is quantized to 5 bits for transmission in the bitstream.

Next, a banding structure is created. The banding structure consists of subbands; each subband contains twelve transform coefficients, spanning a range of about 1.1 kHz at 48 kHz sample rate. These subbands can be grouped into bands; the banding structure indicates how subbands are grouped together. The default banding structure for E-AC-3 approximates auditory Critical Bands. To further reduce data rate, the banding structure is only transmitted if it differs from the default banding structure.

Once the banding structure has been determined, the energy ratios are calculated. The first step in this process is to calculate the banded energy of the MDCT and MDST coefficients in the extension region.

$$E(m) = \sum_{k=m_1}^{m_2} (X_{MDCT}^2(k) + X_{MDST}^2(k)) \quad (3)$$

Where:

E = the energy of band m ;

$X_{MDCT}(k)$ = coefficient k of the MDCT;

$X_{MDST}(k)$ = coefficient k of the MDST;

m_1 = the first transform coefficient in band m ;

m_2 = the last transform coefficient in band m .

Note that the spectral energies could be transmitted to the decoder. However, this would require the decoder to calculate energy ratios. This operation is computationally complex, and frequently, resources on decoder implementations are limited. In order to minimize decoder complexity, the encoder calculates energy ratios and transmits them to the decoder.

To calculate accurate energy ratios, the encoder must create the decoder copy region transform coefficients as accurately as possible. Therefore, it is desirable to account for coding artifacts that may exist in the

decoded transform coefficients. Decoder operations, such as coupling, enhanced coupling and AHT, may be simulated to improve energy ratio accuracy.

Once the copy region has been created, decoder translation from copy region to extension region is simulated. The simulation must exactly match the actual translation, with respect to wrapping and band sizes.

After the translation has been simulated, the banded energies of the synthesized extension region transform coefficients are calculated.

$$ES(m) = \sum_{k=m_1}^{m_2} Y_{MDCT}^2(k) \quad (4)$$

Where:

ES = the energy of the synthesized band m ;

Y_{MDST} = coefficient k of the synthesized MDCT.

Finally, once the banded energies of the original and synthesized extension regions have been calculated, the banded energy ratios of the extension region are generated.

$$ER(m) = 2^{-5} \times \sqrt{\frac{E(m)}{2 * ES(m)}} \quad (5)$$

Where:

ER = the energy ratio of band m .

Note that the range of E is double that of ES . This is because ES is derived from MDCT coefficients only, and E is derived from MDCT and MDST coefficients. Therefore ES is doubled in this operation so that it has the same range as E . Also note that in order to keep energy ratios between zero and one while simultaneously allowing for some gain, they are normalized by 2^{-5} . The normalization is undone by the decoder and allows the energy ratios to indicate up to ~30 dB of gain.

Energy ratios are formatted similarly to coupling scale factors for transmission in the bitstream. A 2-bit, 18.06 dB step-size master exponent is derived for the energy ratios of each channel. The master exponent provides 0 – 54.19 dB of additional dynamic range to

the energy ratios. Energy ratios are renormalized by the master exponent for each channel. Finally, each resultant energy ratio is transmitted as a 6-bit exponent-mantissa set.

The energy ratio formatting scheme provides an approximate range of energy ratios from +28.94 dB to -126.43 dB.

Spectral extension metadata requires an average of 3 kbps per channel to convey a spectrum from approximately 8 to 15 kHz.

4.2.3. Decoder Operation

This section describes E-AC-3 spectral extension decoder operations. The decoder synthesizes high frequency transform coefficients from baseband transform coefficients and metadata sent by the encoder. Spectral extension decoding is illustrated as a block diagram in figure 14.

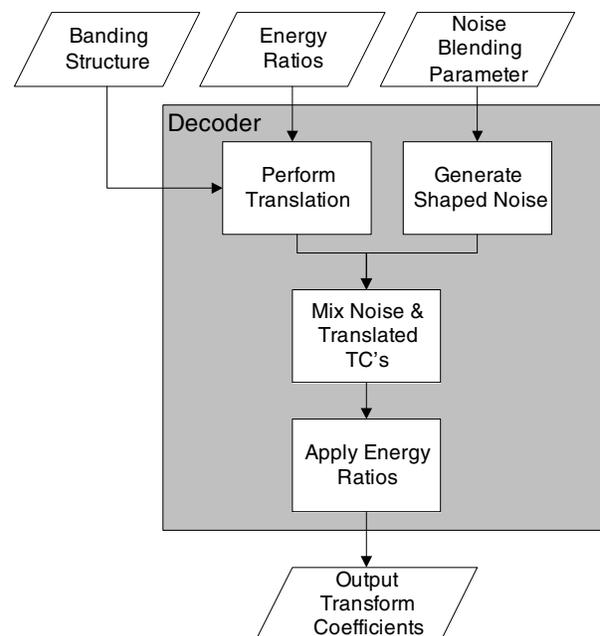


Figure 14: Spectral extension decoding.

The first step of the spectral extension process is translation. Translation consists of sequentially copying transform coefficients from the baseband copy region into the extension region. If the end of the baseband is reached before the end of the extension region, the translation function wraps around and continues

copying from the beginning of the copy region. As mentioned earlier, the decoder must not wrap within a band. Translation continues in this manner until the extension region is completely synthesized.

Next, a noise spectrum is generated to blend with the translated transform coefficients. The noise spectrum is created using a zero-mean, unity-variance pseudo-random noise generator.

Then, noise-blending factors are derived for each band from the noise blending parameter.

$$BF(m) = \frac{\left(m_1 + \frac{m_2 - m_1}{2}\right)}{BW} - NB \quad (6)$$

Where:

BF = the noise blending factor of band m ;

BW = the highest transform coefficient of the extension region;

NB = the noise blending parameter.

Then, mixing coefficients C_N and C_Y are derived from the noise blending factor for each band.

$$C_N(m) = \sqrt{\frac{ES(m)}{m_2 - m_1}} \times \sqrt{BF(m)} \quad (7)$$

$$C_Y(m) = \sqrt{1 - BF(m)}$$

Where:

C_N = the noise mixing coefficient for band m ;

C_Y = the translated spectrum mixing coefficient for band m ;

ES = the energy of the translated transform coefficients in band m .

The mixing coefficients are calculated such that the average energy of each translated band is unaffected by the mixing operation.

Since calculation of the mixing parameters requires square root operations, and the parameter BF has a finite set of values, a table lookup is used to reduce computational complexity.

The translated transform coefficients are then blended with the shaped noise spectrum.

$$Y_B(k) = C_Y(m) \times Y(k) + C_N(m) \times N(k) \quad (8)$$

Where:

Y_B = the blended transform coefficient k ;

Y = the translated transform coefficient k ;

N = the noise component for transform coefficient k .

Once the blended spectrum is complete, each band is multiplied by the energy ratio for that band.

$$Z(k) = 2^5 \times ER(m) \times Y_B(k) \quad (9)$$

Where:

Z = the real portion of transform coefficient k ;

ER = the energy ratio for band m ;

Y_B = the blended transform coefficient k .

Note that the energy ratio is multiplied by 2^5 to compensate for normalization performed by the encoder.

4.3. Enhanced Coupling

Channel coupling in AC-3 is a bit-savings technique that allows high frequency sounds to be combined and conveyed by a single composite encoded channel, rather than multiple independent channels. Side information is conveyed to the decoder and is used to reconstruct the high frequency amplitude envelope of each individual channel on a frequency subband by subband basis, where a subband is a group of six, twelve, a multiple of twelve, frequency-adjacent transform coefficients.

The use of this type of channel-composite coding is attractive since it reduces the effective bandwidth of audio coded, and therefore the data rate at a given quality. In a 5.1-channel application this means that five channels are coded up to a specified bandwidth and then only one composite channel is used to represent them at higher frequencies above the “coupling frequency.”

Unfortunately, this coding advantage comes with the potential for the audio quality to suffer from the

composite coding process if the frequency where this process starts is too low. Optimal use involves the correct tradeoff between reducing the effective bandwidth to encode and controlling the artifacts caused by the composite-channel process.

Enhanced channel coupling is a spatial coding technique that elaborates on conventional channel coupling, principally by adding phase compensation, a de-correlation mechanism, variable time constants, and more compact amplitude representation. The intent is to reduce coupling cancellation artifacts in the encode process by adjusting inter-channel phase before downmixing, to improve dimensionality of the reproduced signal by restoring the phase angles, and inter-channel correlations in the decoded output. This allows the process to be used at lower frequencies than conventional channel coupling while not significantly increasing coupling artifacts, thus improving coder efficiency.

Additional benefits include preservation of phase information required by 2:N matrix systems, such as Dolby Pro Logic and Dolby Pro Logic II. In the case of a 2-channel fallback service, inclusion of this technique makes it practical to optionally apply additional processing (e.g. matrix decoding) to reconstruct a multichannel signal from the 2-channel fallback service, thus significantly improving the experience for the consumer.

4.3.1. Encoder Enhanced Coupling Process

The enhanced coupling encoder is responsible for producing the mono composite signal in a form compatible with the rest of the E-AC-3 coder, and for producing the side chain information that will be used by the decoder to restore the signals. As such, it carries a much greater processing load than the decoder.

The enhanced coupling encoder normally operates on signal blocks of 512 samples, with 50% overlap, but requires the addition of the MDST in addition to the MDCT used in AC-3 to allow for angle adjustment. The minimization of the side chain data rate is accomplished by collecting the groups of up to six 512-sample blocks that comprise an E-AC-3 frame. These are then divided up into enhanced coupling frequency subbands.

The constraints imposed by this coarseness in time (up to frame length intervals) and frequency (subband bandwidths) must be properly managed for best sound

quality and coding efficiency, particularly for complex dynamic signals. This accounts for many of the specific processing steps, particularly the decorrelation mechanisms and smoothing of time constants.

Figure 15 shows a block diagram of the encoding operations for this process.

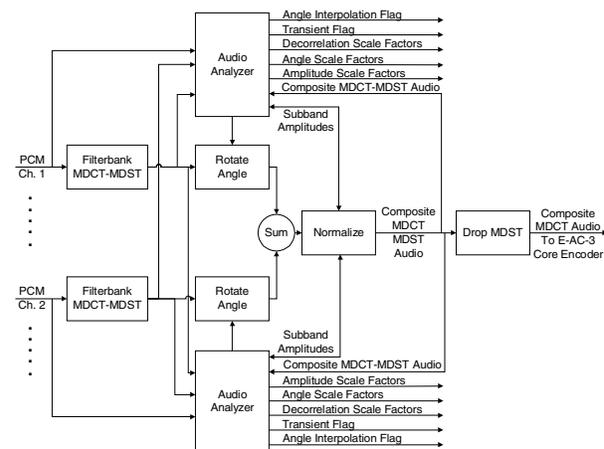


Figure 15: Encoder enhanced coupling process (two out of up to five channels shown for clarity).

As shown in figure 15, each input PCM channel sample block is applied to both a filterbank and a transient detector in the audio analyzer block. The filterbank uses a standard AC-3 KBD analysis window, followed by both an MDCT and MDST of 256-coefficient length. The use of a complex-valued pair, composed of the MDCT as the real-number part and the MDST as the imaginary-number part, allows phase rotations to be accurately applied prior to downmixing.

The transient detector in the audio analyzer block is similar to that employed in a standard AC-3 encoder to control the MDCT transform block length (see Section 2), except set with higher sensitivity to allow it to respond to weaker transients. The encoder uses the presence of a transient to send a transient flag in the bitstream to select the type of decorrelator used in the decoder upmixing process, and to momentarily shorten smoothing time constants in the encoder that produce the amplitude and angle scale factors.

The complex-valued MDCT-MDST pairs are analyzed by first converting each coefficient to a magnitude/angle representation. The overall amplitude of each subband is

derived by smoothing and summing transform coefficient energy values across frequency and across the blocks within the frame. These amplitudes are subsequently used to normalize the amplitude of the composite sum on a subband basis, and to calculate amplitude scale factor side chain information.

The angle value of each transform coefficient pair is used to calculate an inter-channel angle by subtracting from it the corresponding angle of a reference channel, typically the left front channel. Frame-wide inter-channel angles are then calculated on a subband basis by smoothing (low-pass filtering) and performing an amplitude-weighted average of the transform coefficient inter-channel angles. These averaged inter-channel angles are then used to phase compensate the transform coefficients. The avoidance of cancellation is achieved by applying the inverse value of each angle to a phase rotator. This phase rotator is simply a complex multiplication of each MDCT-MDST coefficient pair in the subband by the appropriate phase rotation factor.

Phase compensated transform coefficients from each channel are then summed on a transform-coefficient basis to derive the raw complex-valued mono composite signal. The amplitude of each subband of the mono composite is then normalized so that its power is equal to the sum of the powers of the input channels. This step avoids the incorrect emphasis of signals common to more than one input channel, and guarantees that resulting side chain amplitude scale factors do not exceed a value of 1.0.

The normalized complex-valued composite signal is fed back to each channel's audio analyzer block, so that amplitude side chain scale factors can be calculated on a subband basis, as the ratio of input amplitude to corresponding mono composite amplitude.

The final step in the signal path of the encoder is to discard the MDST coefficients, while retaining the MDCT coefficients for use in the remaining encoding process.

The audio analyzer block for each channel uses the calculated signal information to derive side-chain values for the decoder, typically at a frame rate. For each subband, the following values are calculated:

- An amplitude scale factor, as described above. This is a 5-bit value with a granularity of 1.5 dB/step.
- A phase scale factor representing averaged inter-channel phase angle. This is a 6-bit value linearly covering the range π to $-\pi$.
- A 3-bit decorrelation factor covering a range of 0 to 1.0.

Note that the decorrelation value is basically the normalized standard deviation of the transform coefficient inter-channel angles across subbands and blocks within the frame. A high value, at or near 1.0, indicates negligible consistency of interchannel angles, invoking a high degree of decorrelation by the decoder, and conversely for a low value at or near 0.

A 1-bit transient flag is also sent, but unlike the other side-chain information, this is a value that applies to all subbands within the coupled frequency region for a given channel. Separate transient flags are conveyed for each block in the frame to provide for higher temporal resolution. The transient flag is used by the decoder to mediate the choice of decorrelation mechanism. The transient flag controls the choice between a decorrelator optimized for continuous signals or for transients.

Similarly, a 1-bit interpolation flag is sent to control whether transmitted subband angle values are to be applied equally to all transform bins, or linearly interpolated across frequency, based on the similarity to the original angle distribution.

4.3.2. Decoder Enhanced Coupling Process

The enhanced coupling process in the decoder is designed for minimum computation load, with most of the computationally demanding signal analysis performed by the encoder. The decoder unpacks the enhanced coupling audio and side chain information, and then uses both to derive the individual output channels.

Figure 16 is a block diagram of the decoding upmix operation. It shows how the decorrelators, amplitude modulators, and phase modulators are applied to the composite mono signals to generate MDCT coefficients for the coupled band associated with the enhanced coupling frequency region for each output channel.

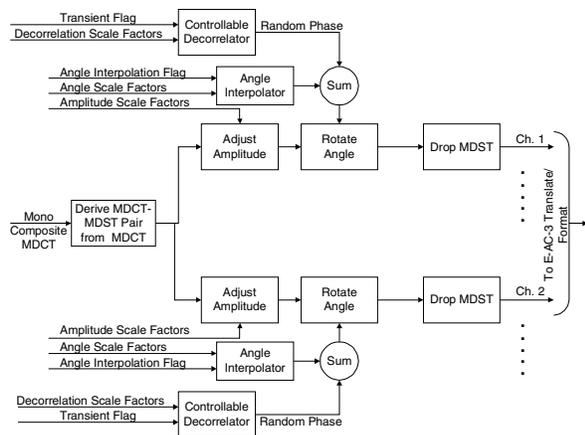


Figure 16: Enhanced coupling process in the decoder (two out of up to five channels shown for clarity).

Examination of this figure shows that the MDCT values from the composite channel are first converted to MDCT-MDST complex-valued coefficient pairs by derivation of the MDST values, treating them as imaginary-valued, and then pairing them with the MDCT coefficients. The derivation of the MDST coefficients is performed by transformation to mono composite PCM values using an IMDCT, standard KBD windowing, and an overlap process similar to that used by the AC-3 decoder. The MDST coefficients are derived by standard KBD windowing of the PCM values, use of the 50% overlap, and employing the MDST.

Next, amplitude scaling of the groups of MDCT-MDST pairs grouped in subbands for each channel is applied that molds the coarse spectral characteristics of each channel to those of the original. Following that, the transform-pair subbands are modulated by an angle rotation process similar to that used in the encoder, albeit of opposite sign. Again, it is this step that requires that the MDCT and MDST coefficients be arranged in complex-valued pairs. The transmitted interpolation flag controls whether identical or interpolated phase angles are applied across the transform coefficients of each subband.

The angle scale value applied to each MDCT-MDST coefficient pair is a sum of two values: the corresponding dequantized subband angle conveyed as side chain information by the encoder, possibly modified by the interpolation factor, and a decorrelating angle value derived by the decoder, based on the

decorrelation scale factor, the transient flag, and an associated random number sequence.

In the absence of a transient, a separate random decorrelating angle value is used for each MDCT-MDST coefficient pair, but maintained constant as a function of time. Additionally, the angle values are scaled by the decorrelation scale factor, in order to minimally disturb continuous signal and impart an individual time envelope to each output channel.

In the presence of a transient, a new subband-wide phase-offset value is chosen for each block and scaled by the decorrelation scale factor. The use of single random phase offset value per subband ensures the temporal integrity of each transient is maintained.

The MDCT coefficients from the enhanced coupling process are combined with the discrete low frequency ones to form a complete spectral representation for each output channel. This allows both an inverse transformation to PCM or translation and repackaging into a standard AC-3 bitstream.

4.4. Transient Pre-Noise Processing

E-AC-3 has been designed to operate and provide better audio quality at lower data rates than the original AC-3 system. At lower data rates the audio quality of coded audio can be negatively impacted, especially for difficult to code, transient material. This impact on audio quality is primarily due to the limited number of data bits available to accurately code these types of signals. Coding artifacts of transients are exhibited as a reduction in the definition of the transient signal as well as the “transient pre-noise” artifact which smears audible noise throughout the encoding window due to coding quantization errors. The new transient pre-noise processing coding tool uses time scaling synthesis to reduce or completely remove the transient pre-noise artifacts introduced at the lower audio coding data rates provided by E-AC-3.

4.4.1. Algorithmic Description – Time Scaling Synthesis

The new transient pre-noise coding tool reduces transient pre-noise artifacts that are introduced when audio containing transients is encoded using the E-AC-3 audio coding system. The relationship between the length and location of pre-noise with respect to the transient and E-AC-3 audio coding windows is shown below in figure 17. As shown in the example in figure

17, the audio coding block 3 has switched into the standard AC-3 block switching mode.

In AC-3 style block switching the standard 512-point KBD transform window is transformed into two, 256-point, half windows, which reduces the size of the audio window and therefore reduces the length of the transient pre-noise. While block switching reduces the length of the transient pre-noise at very low data rates, transient pre-noise artifacts can still negatively impact audio quality and so by using the transient pre-noise coding tool along with block switching the quality of E-AC-3 coded audio can be substantially improved.

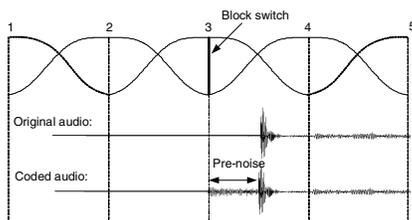


Figure 17: Relationship of transient pre-noise to the location of the transient and the coding windows.

Figure 18 shows how the transient pre-noise coding tool reduces or removes transient pre-noise by replacing the appropriate audio segment with audio that is synthesized using the audio located prior to the transient pre-noise. The audio is processed using time scaling synthesis so that its duration is increased such that it is of sufficient length to replace the audio containing the transient pre-noise.

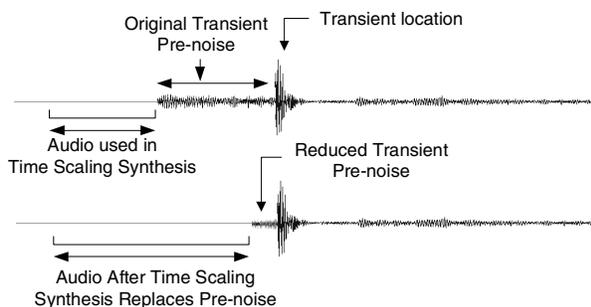


Figure 18: Application of the transient pre-noise coding tool using time scaling synthesis.

The E-AC-3 transient pre-noise processing tool incorporates a multi-channel, time-domain based time scaling synthesis technique that uses auditory scene analysis (ASA) to provide very high quality results, see

Crockett [15] for more details. The audio synthesis buffer is analyzed using ASA and maximum similarity processing and then time scaled such that its duration is increased enough to replace the audio which contains the transient pre-noise. The synthesized audio of increased length is used to replace the transient pre-noise and is cross-faded into the existing transient pre-noise just prior to the location of the transient to ensure a smooth transition from the synthesized audio into the originally coded audio data. For additional details of the E-AC-3 transient pre-noise processing audio coding tool, see Crockett [16].

By using transient pre-noise processing, the length of the transient pre-noise can be dramatically reduced or removed, even for the case when standard block-switching is disabled. It is well known that the coding efficiency of bit-rate-reduction audio coders increases with transform size and that coding efficiency becomes significantly more important as the data rate is reduced. In many cases, at low data rates (i.e., ≤ 96 kbps for stereo content), the use of block switching and increased transient performance is weighed against the improved audio quality provided by not block switching and exclusively using longer windows and transform lengths. With the addition of the transient pre-noise processing tool in E-AC-3, it is now possible to provide both improved coding efficiency and good transient performance at very low data rates by disabling block switching and using the transient pre-noise processing coding tool.

4.4.2. Algorithmic Description – Application in E-AC-3

Time scaling synthesis analysis and processing for the transient pre-noise processing tool is performed on time-domain data. Analysis is performed using PCM audio data prior to E-AC-3 encoding and the metadata information generated is transmitted along with the encoded E-AC-3 bitstream. The transmitted transient pre-noise metadata is used to perform time-domain processing on the decoded audio to reduce or remove the transient pre-noise introduced by low bit-rate audio coding at low data rates. Figure 19 provides a flow diagram of the processing steps that are performed.

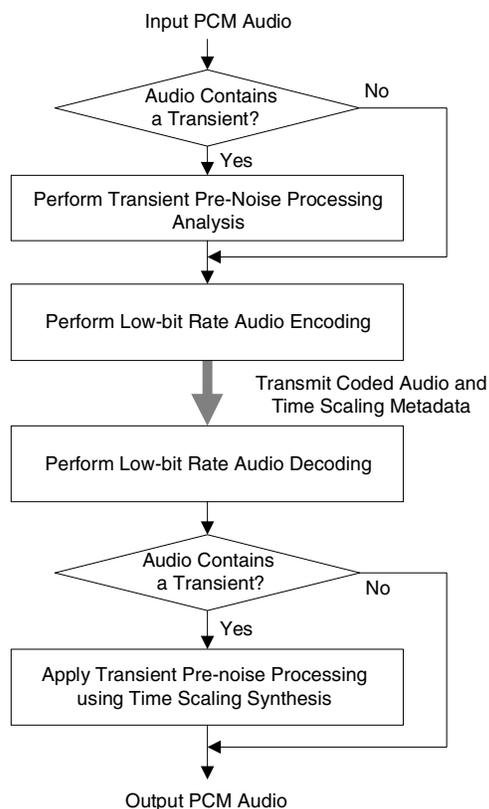


Figure 19: Overview of the application of the transient pre-noise processing tool in E-AC-3.

As shown in figure 19, the E-AC-3 encoder performs time scaling synthesis analysis and determines the optimal time scaling parameters, based on the audio content, for each detected transient. The time scaling parameters are transmitted as additional metadata, along with the encoded audio data. Experiments performed using standard 5.1-channel transient test material show that the additional transient pre-noise processing metadata requires on average only 200 – 500 bps additional overhead data rate. Analysis is performed in the E-AC-3 encoder to offload processing from the decoder and to provide a low-latency implementation of the transient pre-noise processing tool in decoders.

5. DATA FORMAT EXTENSIONS

Current and future applications require the ability to transmit more channels and more programs in a single data stream, with finer control over a wider range of allowable data rates than the AC-3 coding system

provides. To meet these demands, the E-AC-3 coding system was also designed to support programs of more than 5.1 channels through the use of channel extensions, simultaneous transmission of multiple independent programs through the use of program extensions, and finer control over a wider range of data rates through increased control over the bitstream framing structure.

5.1. Increased Channel and Program Carriage

The E-AC-3 system supports the carriage of programs of more than 5.1 channels by enabling multiple frames, or substreams, to be time-multiplexed into a single data stream. A data stream carrying a program of more than 5.1 channels is constructed from an independent substream containing a 5.1-channel downmix of the program followed by up to eight dependent substreams containing a combination of replacement and supplemental channels.

The replacement and supplemental channels carried by the dependent substreams are coded discretely rather than derived from matrix subtraction, and are thus not subject to coding artifacts induced by matrix subtraction. In a program of more than 5.1-channels, the substream containing the 5.1-channel downmix is independent because it is sufficient to completely reproduce the program without data from any other substreams.

The substreams containing the replacement and supplemental channel data are dependent because they must be combined with data carried by other substreams to completely reproduce the program. For example, an independent program consisting of 7.1 channels (left, center, right, left-surround, right-surround, left-back, right-back, and low-frequency-effects [LFE]) could be created and reproduced as follows:

1. Create a 5.1-channel downmix (left, center, right, left-surround, right-surround, and LFE) of the 7.1-channel program
2. Encode the 5.1-channel downmix as an independent substream
3. Encode the left-surround, right-surround, left-back and right-back channels of the 7.1-channel program as a dependent substream
4. Decode the independent substream
5. Decode the dependent substream
6. Replace the left-surround and right-surround channels of the independent substream with the

left-surround and right-surround channels from the dependent substream

The use of time-multiplexed substreams by the E-AC-3 system to carry programs of more than 5.1 channels yields two classes of decoders: simple decoders that reproduce a program for configurations of 5.1 or fewer speakers and advanced decoders that reproduce the complete program. Simple decoders ignore all dependent substreams and decode only the 5.1-channel downmix carried by the independent substream. Advanced decoders decode the independent substream and all dependent substreams, and then perform channel replacement to reproduce the complete program. Figure 20 shows two frames of an E-AC-3 stream carrying 7.1 channels of discrete audio data.

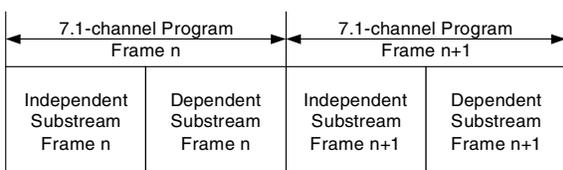


Figure 20: E-AC-3 data stream carrying a 7.1-channel program.

The E-AC-3 system supports carriage of up to eight independent programs by time-multiplexing up to eight independent substreams into a single data stream. For example, it may be desirable to simultaneously transmit both an English language sound track and a Spanish language sound track in a single data stream. Using the E-AC-3 system, the English track could be encoded as independent substream 0 and the Spanish track could be encoded as independent substream 1. A simple decoder would always decode independent substream 0 and ignore all other substreams. A more advanced decoder could allow the user to select either the English or the Spanish track, decode the substream corresponding to the selected sound track, and ignore the other substream.

Figure 21 shows two frames of an E-AC-3 stream carrying two independent programs.

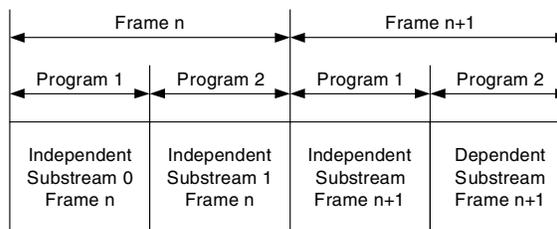


Figure 21: E-AC-3 data stream carrying two independent programs.

It should be noted that the channel and program extension capabilities of the E-AC-3 system can operate together to enable the carriage of up to eight independent programs each containing more than 5.1 channels.

5.2. Flexible Data Rate and Framing Structure

The E-AC-3 system enables more control over the data rate of the bitstream than the standard AC-3 system in order to provide greater flexibility required by new applications. The expansion of available data rates is achieved through control of two coding parameters: the number of data words per frame, and the number of blocks per frame. The data rate in bits per second for a compressed data stream can be given by the formula:

$$\text{data rate} = \frac{n \cdot F_s \cdot \left(16 \frac{\text{bits}}{\text{word}}\right)}{m \cdot \left(256 \frac{\text{samples}}{\text{block}}\right)} \quad (10)$$

where

F_s = sample rate;

m = number of blocks per frame;

n = 16-bit words specified per frame;

This data rate is directly proportional to the number of data words per frame, and inversely proportional to the number of blocks per frame. The E-AC-3 system allows the number of words per frame to vary between 1 and 2048, and the number of blocks per frame to be 1, 2, 3, or 6. This increased flexibility allows for both a greater maximum data rate and a finer resolution between data rates than the AC-3 system.

The maximum data rate at which the E-AC-3 coding system can operate occurs when the number of data words per frame is 2048 and the number of blocks per frame is one. Table 3 shows the maximum possible data rate as a function of the sample rate and the number of blocks per frame. The number of compressed data words for all entries in the table is 2048.

		Blocks per Frame (m)			
		1	2	3	6
Sample Rate (kHz)	48	6,144	3,072	2,048	1,024
	44.1	5,644	2,822	1,881	940
	32	4,096	2,048	1,365	682

Table 3 Maximum E-AC-3 data rate (kbps).

Note that for all combinations of sample rate and number of blocks per frame, the maximum E-AC-3 data rate is greater than the maximum AC-3 data rate of 640 kbps.

For each combination of sample rate and number of blocks per frame, 2048 data-rate settings are possible. The number of usable data rates is significantly expanded over the AC-3 system.

With the addition of many more possible data rates, the step size between successive data-rate settings for the E-AC-3 system is smaller for every combination of sample rate and number of blocks per frame than the AC-3 system. Table 4 shows the step size between successive data-rate settings as a function of the sample rate and the number of blocks per frame:

		Blocks per Frame (m)			
		1	2	3	6
Sample Rate (kHz)	48	3.000	1.500	1.000	0.500
	44.1	2.756	1.378	0.918	0.459
	32	2.000	1.000	0.666	0.333

Table 4 E-AC-3 data-rate adjustment resolution (kbps).

Although the data rate is dependent on the PCM sample rate, it is not practical to vary the data rate this way. As a result, the desired data rates are achieved by adjusting

the number of compressed words and number of blocks per frame parameters rather than a single data-rate parameter.

5.3. Improvements to the Bitstream Syntax

In many cases, the AC-3 bitstream syntax transmits bitstream parameters that can easily be inferred from other bitstream parameters. In other cases, bitstream parameters have several possible values, of which only one is typically used in practice. The removal of such bitstream inefficiencies is extremely important in bit-rate-reduction applications where small bitstream inefficiencies can have an audible impact on sound quality. On the other hand, the removal of these inefficiencies becomes less important in moderate to high data-rate applications where the impact of the inefficiencies on audio quality is negligible.

Careful consideration was taken during the development of the E-AC-3 bitstream syntax to allow an encoder to sacrifice flexibility for bitstream efficiency for low data-rate applications. Likewise, for moderate to high data-rate applications, the encoder can sacrifice bitstream efficiency to regain the flexibility built into the original AC-3 coding system.

The E-AC-3 bitstream syntax was also designed to support mixing of two independent bitstreams, whether contained in a single program stream through the use of program extensions or in separate data streams. Several bitstream parameters were defined that provide information about how to mix two streams together.

The syntax includes separate scaling parameters that indicate the amount of gain or attenuation to apply to the streams prior to mixing. The syntax allows the selection of one of four models for performing dynamic range control on signals before and after they are mixed together. The simplest model is fixed and requires no additional bitstream data, while the most sophisticated model is parametric and can require up to 269 bits per frame. The syntax includes a parameter that indicates the ideal transform length to use for each audio block when re-encoding the mixed bit stream into an AC-3 bit stream.

Lastly, the syntax includes parameters that specify directional cues that enable a mono signal to be rendered into a 2-dimensional sound field. This enables the addition of directionality to a mono signal when mixed with a multichannel signal.

All of these parameters are conditionally present in the bitstream, and can be removed when they are not necessary or when bitstream efficiency is desired.

6. E-AC-3 TO AC-3 CONVERSION

The E-AC-3 system provides an effective method to convert an E-AC-3 bitstream to an AC-3 bitstream to ensure compatibility with the large installed base of AC-3 decoders. The conversion process is designed to minimize loss in audio quality while keeping the complexity at a level suitable for low-cost consumer devices.

Unlike standard transcoding processes for converting between incompatible formats, the E-AC-3 to AC-3 conversion process does not require a full decode to PCM samples followed by a full encode to AC-3. Because E-AC-3 and AC-3 use the same filterbank, bit-allocation process and framing structure, the conversion device needs only to perform a partial decode followed by a partial encode.

The amount of decoding and encoding required during the conversion process depends on how the E-AC-3 frame is encoded. In the worst case, when the audio data is encoded using new features of E-AC-3, the audio is decoded to a frequency domain representation that is common to both AC-3 and E-AC-3. No time-domain decoding is necessary due to identical filterbanks for both formats. In the best case, if the audio data is encoded without using any E-AC-3 features, then the encoded audio data can be repackaged into an AC-3 frame without any decoding or encoding of audio data. In this case, no requantization is necessary due to the identical bit-allocation process shared by both formats.

The partial decode must also take into account interdependencies between the AC-3 coding tools and the E-AC-3 coding tools. For example, if an E-AC-3 frame is encoded using both standard AC-3 coupling and spectral extension, the partial decode must decode the coupling channel data into independent channel data prior to performing the spectral extension decoding process. Once transform coefficients have been synthesized using the spectral extension process, it is not possible to employ channel coupling in the converted AC-3 frame without performing a complete re-encode of the coupling channel, which is prohibitively complex and requires data not available to the conversion device. Therefore, when spectral extension is enabled in the E-AC-3 frame, all channels

in the converted AC-3 frame are coded independently. Because converted AC-3 frames are encoded at 640 kbps, the decreased efficiency due to coding channels independently is not significant.

Once the E-AC-3 frame data has been partially decoded, the audio data is encoded into an AC-3 frame. The partial encode process involves the same steps as an AC-3 encoder except that it begins from frequency domain data, and is thus not required to perform any of the PCM calculations or time to frequency transformations performed by the encoder. There are two main challenges in the partial encode process. The first is keeping the computational load low enough for implementation in consumer devices. The second is performing a high quality encode without having access to all the metadata normally available to a full encoder.

The most computationally expensive part of the encode process is the calculation of the *snroffset* value. The calculation is an iterative process that determines the optimal bit-allocation for the desired data rate. In order to make the partial encode feasible for consumer devices, the calculation is off-loaded from the converter to the E-AC-3 encoder. The E-AC-3 encoder calculates the AC-3 *snroffset* and packs the result into the E-AC-3 bitstream. The E-AC-3 converter extracts the *snroffset* value when performing bit allocation for the AC-3 bitstream.

There are two consequences to having the encoder perform the *snroffset* calculation. The first is that the encoder must know ahead of time what the desired data rate is for the converted AC-3 bitstream. To accommodate this, E-AC-3 specifies that all E-AC-3 bitstreams are converted to the data rate of 640 kbps. This is the maximum AC-3 data rate and thus minimizes tandem coding losses associated with the conversion process.

The second consequence is that care must be taken by the encoder to make sure that the *snroffset* value it calculates is identical to the *snroffset* value that the converter would have calculated if it performed the calculation itself. This means that the data from which the *snroffset* is derived in the encoder must be identical to the data available to the converter. Any differences between the encoder calculation and the converter's hypothetical calculation may result in an over-allocation of bits and failure to complete the conversion.

Another challenge for the partial encode is in performing a high quality encode without access to the original PCM data. One calculation that has a significant impact on the quality of the encoder is the exponent strategy calculation. The exponent strategy determines how exponents are shared across time and frequency and is based on how much the exponents change from one block to the next within a frame. In a full encoder, this calculation is done using exponents derived from the MDCT-MDST complex-valued pairs to accurately measure changes in the power spectrum. However, the partial encoder only has access to exponents derived from the MDCT, which provide a relatively poor approximation of the power spectrum and result in an inefficient exponent strategy. For information on the AC-3 exponent strategy process, see Davidson [3].

Alternatively, the partial encoder could derive the AC-3 exponent strategy based on the channel exponent strategy and coupling exponent strategy associated with the E-AC-3 bitstream. This technique yields an improved exponent strategy, but is still less optimal than that of a full encoder. Given this, the exponent strategy calculation is also off-loaded to the full E-AC-3 encoder. The E-AC-3 encoder calculates the AC-3 exponent strategy using complex exponents. This value is transmitted in the E-AC-3 bitstream and extracted during the conversion process. This technique not only provides the highest audio quality, but also further reduces the computational load of the converter. Figure 22 shows how the encoder and converter work together to perform the E-AC-3 to AC-3 conversion.

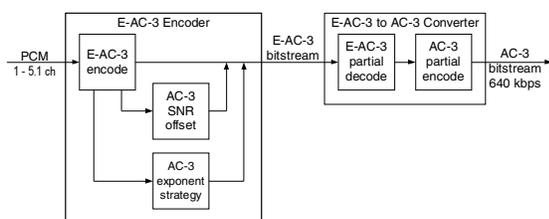


Figure 22: Block diagram of E-AC-3 to AC-3 conversion.

An additional complication arises in the conversion process when the E-AC-3 bitstream consists of frames that contain less than six audio blocks. Since an AC-3 frame always consists of six audio blocks, an E-AC-3 bitstream consisting of 2-block frames would require three E-AC-3 frames to produce one AC-3 frame. In this case, both the encoder and converter must accumulate

multiple E-AC-3 frames before performing the conversion calculations.

Since the encoder is performing calculations for the conversion process, the encoder must indicate which six blocks of audio data it is using for its calculations. The encoder does this by setting a flag in the E-AC-3 bitstream called *convsync* which indicates that the frame contains the first block of a 6-block set of frames. The converter can then synchronize to the 6-block set of frames by detecting the *convsync* flag.

Another issue that arises with such bitstreams is how metadata present in multiple E-AC-3 frames is resolved into metadata for a single AC-3 frame. This applies to the bitstream information (BSI) field present in every E-AC-3 and AC-3 frame. Any differences between BSI parameters must be resolved. For example, if each E-AC-3 frame has a different compression (*compr*) value, the converter must choose one value to represent the entire AC-3 frame. In this case, the converter will resolve to the value that provides the greatest compression in order to guarantee against clipping during downmixing. In general, the converter will resolve differences by choosing the most conservative value.

7. TRANSCODING & CONVERSION ISSUES

This section examines how the similarity between E-AC-3 and AC-3 can be exploited to optimize tandem coding performance. This optimization is particularly important because of the need to work well within the existing broadcast and multimedia infrastructures, which include a wide prevalence of AC-3 coding devices.

Tandem coding losses occur when the coding errors from each coding system in a series configuration combine with each other to result in larger errors than produced by each coding system alone. This happens when the quantization levels in one coder do not map to the same levels in the succeeding ones, and therefore produce new errors in addition to the old ones.

The use of different filterbanks, presence of time delays between coding stages, presence of changes in signal amplitude, employment of different coding tools that modify the data being quantized, and the use of different perceptual models all contribute to tandem coding losses. Unfortunately, the desire to lower the data rate from standard AC-3 applications involves some or all of

the above changes that can incur tandem coding losses. It is important to minimize these to achieve the best possible audio quality.

Coding system design under the tandem coding paradigm of today's broadcast and multimedia environment requires a balance between maintaining a match to the source AC-3 encoder and destination AC-3 decoder, plus the need to use new coding tools to increase the basic efficiency and lower the data rate. A higher overall quality/data rate favors a high similarity between the AC-3 and E-AC-3 systems, while a lower quality/data rate favors the use of coding tools that increase E-AC-3 efficiency but result in greater tandem coding losses.

The overall quality of a broadcast or multimedia application is driven by basic coding system distortions and tandem losses, combined with the audible distortions arising from the use of the coding tools, such as bandwidth extension, enhanced channel coupling, and transient pre-noise processing. An advantage of the aggressive use of bandwidth extension and the enhanced coupling tool is that the effective bandwidth of the core audio coder is reduced, thus lowering the basic coding error at a given data rate. This is offset by the disadvantage of a greater chance of coding-tool-induced distortions occurring. The use of transient pre-noise processing also requires the selection of an optimal balance between the reduction of pre-noise effects versus sound-segment-substitution artifacts.

The transcoding and conversion applications discussed earlier require finding the best balance between basic coding errors, tandem-coding losses, and new coding tool advantages/disadvantages. This must be done in context of the complete tandem configuration and not just the single E-AC-3 coding stage.

7.1. Basic Coding Efficiency and Tandem Coding Losses

This section will focus on the determination of basic quantization and tandem coding losses. The PEAQ objective measure of subjective quality, standardized by the ITU [17], is used to provide a rapid estimation of subjective sound quality so that basic trends in coding efficiency and similarity effects of E-AC-3 to AC-3 are quantitatively examined. Carefully used, this algorithm gives reliable results; see Thiede et al. [18].

Because current embodiments of the PEAQ algorithm require a reference and coded signal of only one or two channels, the following tests were done using 2-channel audio coders serving as a model for the actual 1 – 5.1 channel configurations. The selection of equivalent 2-channel E-AC-3 coder configurations to model the actual application is done by matching the *snroffset* variable value. Since the coding errors scale proportionally to this variable and both the AC-3/E-AC-3 coding systems use the same perceptual model, identical *snroffset* values indicate a close match in basic coding errors.

Although bandwidth extension and channel coupling may be utilized in many applications, the PEAQ system is not designed to assess them accurately. As a result the model coders do not include them. Additionally, the outputs from the model coders and the reference signal are selected to have a 20 kHz audio bandwidth so that bandwidth effects are eliminated. The compensation for the effect of these configuration choices is built in the *snroffset* matching process.

The following experiments were performed using a 6.8 second segment of a pitch pipe note with a fundamental frequency of approximately 290 Hz. The silence periods were edited out and the amplitude adjusted so the peak amplitude was set to -4.60 dB relative to digital full scale. This test signal is used because it is challenging to code efficiently with good sound quality and is stationary in its spectral characteristics. The results from these tests using the pitch pipe are used to represent the performance over a wide class of input signals.

7.1.1. Tandem Effects in the E-AC-3 Converter

The first situation examined is the use of the E-AC-3 to AC-3 converter when audio content encoded in the E-AC-3 format needs to be converted to AC-3 encoded bitstreams at 640 kbps for final decoding in a home receiver. The simulation of a 5.1-channel AC-3 operation at 640 kbps is performed with the use of a 2-channel E-AC-3 to AC-3 converter with an output data rate of 317 kbps.

The performance of E-AC-3 alone is compared to the E-AC-3 to AC-3 at 640 kbps tandem model with and without the AHT coding tool. The data rates are all referenced to the model coder data rate of 221 kbps simulating 5.1-channel AC-3 operating at 384 kbps, as used in digital television.

Note that actual coding systems employed in the applications mentioned earlier have lower data rates than used by the coder models used in this test. This is true because of the beneficial effect of coding tools employed and configuration adjustments.

Figure 23 shows this comparison using the noise-to-mask ratio (NMR) parameter of the PEAQ standard. This is appropriate because NMR is directly proportional to the *snroffset* parameter which controls the basic coding errors in E-AC-3 and AC-3. Because the masking model is identical for all the coding configurations tested, a measurement of NMR produces accurate assessment of basic coding errors. Lower coding errors are indicated by larger negative values of NMR.

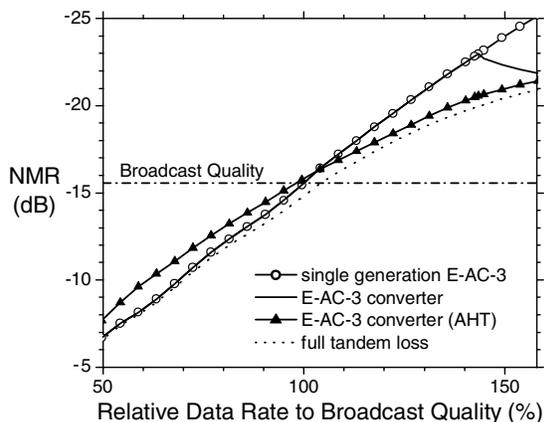


Figure 23: NMR as a function of coding system configuration and data rate for E-AC-3 to AC-3 conversion.

This figure shows the NMR values for the following configurations:

- Single generation of E-AC-3.
- E-AC-3 to AC-3 converter with final decoding to the AC-3 bitstream.
- E-AC-3 to AC-3 converter including the AHT tool and final decoding to the AC-3 bitstream.
- Model of a two coder tandem configuration where a full tandem coding loss occurs.

The full tandem loss condition is estimated by power summation of the coding errors. The data rate is expressed as a percentage of the broadcast-quality test metric of 221 kbps.

Examination of figure 23 shows that the E-AC-3 to AC-3 converter performs equally well as a single generation of E-AC-3 until its data rate exceeds that of the final AC-3 bitstream. This occurs because the converter carries the original E-AC-3 quantization state within the AC-3 bitstream. When the E-AC-3 data rate exceeds that of the final AC-3, requantization of the audio causes tandem coding losses to occur. In addition, the final AC-3 stage limits the ultimate sound quality.

Also evident is that the AHT coding tool shows better performance at lower data rates because of its improved coding efficiency, but suffers from tandem coding losses caused by the changed quantization of the new transform/VQ-GAQ compared to the final AC-3 process. Note that the converter with AHT performance roughly parallels, but is better than that of the non-AHT coding system combination with full tandem losses. This allows for the practical application of the AHT tool since it results in improved performance at or below broadcast-quality.

In summary, the E-AC-3 to AC-3 converter is shown to perform noticeably better, from a basic coding error point of view, than a conventional transcoding process. The AHT tool is also shown to be increasingly appropriate as the data rate target falls below 384 kbps in the actual application.

7.1.2. Tandem Effects in the AC-3 to E-AC-3 Transcoder

This discussion investigates the performance of another device, the AC-3 to E-AC-3 transcoder, which is used to distribute 5.1-channel audio content that has already been encoded in the AC-3 format at 384 kbps. The intermediate E-AC-3 bitstream is created by transcoding to E-AC-3 at a lower data rate. This is converted back to an AC-3 bitstream at 640 kbps and finally decoded to PCM at the home receiver.

As before, the analysis of basic coding errors is performed by the use of model coders that simulate the action of the actual coders but work in a 2-channel mode without coupling or spectral extension. The actual AC-3 to E-AC-3 transcoder is modeled by the combination of an E-AC-3 decoder and encoder with

synchronized frame alignment. This is used because it is easy to implement and creates only a small degradation in quality when modeling the actual AC-3 source and AC-3 to E-AC-3 transcoder. These degradations occur because small arithmetic errors in the frame-aligned encoder and decoder result in small quantization differences not present in the actual transcoder.

The source bitstreams are modeled by the output from a 2-channel E-AC-3 encoder set to have the same *snroffset* value as the AC-3 encoder at 384 kbps in a 5.1-channel configuration. This bitstream is transcoded by the previously mentioned decoder-encoder pair modeling the transcoder where its output data rate is adjusted over the range of 110 – 300 kbps. The AHT coding tool is turned on or off to investigate the effect of its employment on overall quality. Next, the E-AC-3 to AC-3 converter is used to generate an AC-3 bitstream that is finally decoded to PCM for PEAQ analysis. The source encoder data rate is adjusted to be 221 kbps, while the final AC-3 bitstream is modeled by an E-AC-3 bitstream at a 317 kbps data rate. Data rates are shown relative to 221 kbps.

The following configurations/models were modeled and results in the NMR values shown in figure 24:

- single generation of E-AC-3
- AC-3 source, AC-3 to E-AC-3 transcoding, E-AC-3 to AC-3 conversion, decoding to PCM (AHT off)
- AC-3 source, AC-3 to E-AC-3 transcoding, E-AC-3 to AC-3 conversion, decoding to PCM (AHT on)
- AC-3 source, transcoding to an intermediate bitstream, transcoding back to AC-3, decoding to PCM where a full tandem coding loss occurs at each transcoding operation

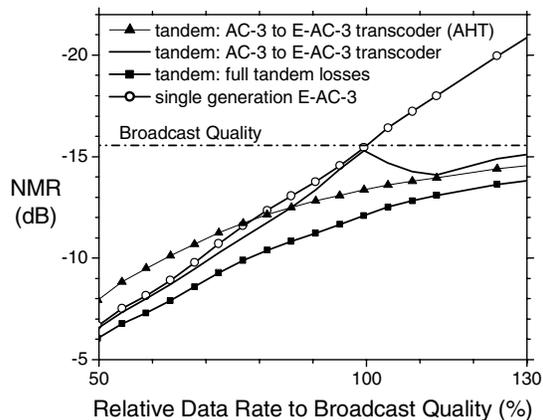


Figure 24: NMR as a function of coding system configuration and data rate for transcoding and conversion.

Examination of this figure indicates that transcoding from AC-3 to E-AC-3 and conversion back to AC-3 has only a small degradation compared to a single generation of E-AC-3 coding when the E-AC-3 uses the same transform as AC-3 and the intermediate data rate is equal or lower than the source bitstreams.

The use of the AHT tool produces a different result. At lower data rates the improved efficiency due to the AHT system results in improved performance, but this benefit quickly degrades as the tandem coding loss becomes more significant.

The importance of tandem coding errors is shown by the difference between the single generation performance of E-AC-3 and the transcoder/converter configuration versus what is predicted if the same efficiency coding systems were used but full tandem losses occurred. Figure 25 shows the difference between the predicted performance of the transcoder/converter combination compared to the full-tandem-loss transcoder.

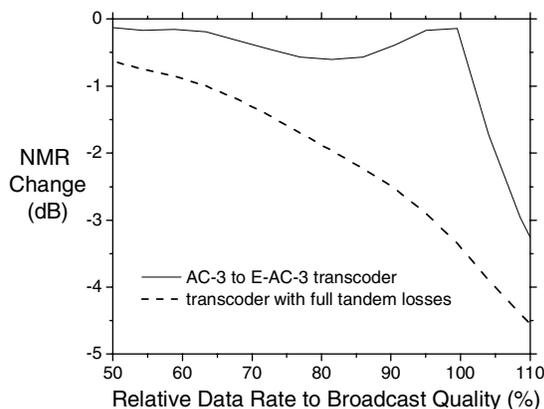


Figure 25: NMR differences between proposed transcoder/converter and general two stage transcoder.

This figure shows the substantial benefit of inter-coding system compatibility. At data rates below that of the source, the compatible coding system performs much better. The AC-3 compatible configuration experiences minor losses of less than 0.60 dB up to the reference data rate. In fact, the degradation decreases to as little as 0.12 dB at the reference data rate. Note that the actual transcoder performs better than this, not being subject to arithmetic-accuracy induced tandem losses.

The situation with full tandem losses has substantially worse performance of a 3.40 dB loss at the reference data rate, implying that transcoding to an incompatible coding system engenders a severe quality penalty.

Since the quality of the distribution process with any transcoding or conversion must lie below that of the first element in the chain, the perceived audio quality associated with the different options and lower data rates must be estimated. This is accomplished with the use of the objective difference grade (ODG) variable from the PEAQ standard. This variable directly indicates the sound quality as the difference in the subjective impairment scale, as defined by the ITU [19], between a transparent system and the one being measured. An ODG value of greater than -1.0 indicates better than “perceptible but not annoying” performance, down to a “slightly annoying” performance level, for an ODG value of -2.0 . This more direct relationship to overall sound quality for the various configurations is shown in figure 26.

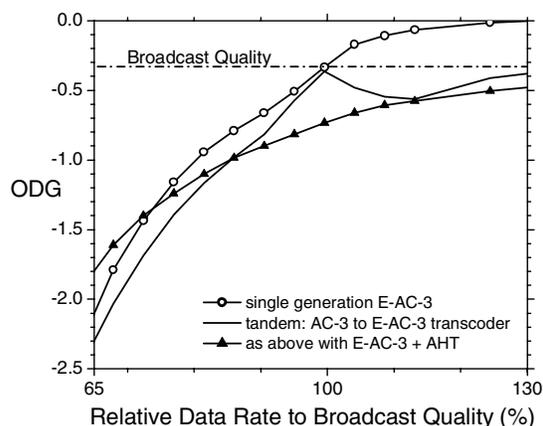


Figure 26: ODG relationships between the proposed transcoder/converter and general two stage transcoder.

This figure demonstrates that the relationship between single generation E-AC-3 and the two transcoder/converter options is very similar to that of the NMR parameter, with the additional information that the lower data rates quickly result in rather poor sound quality. The ODG value falls below -1.0 when the data rate falls below 75% of nominal. Since the AHT version of the transcoder/converter only performs better below that point, a properly designed AC-3 to E-AC-3 transcoder should only employ AHT when the need for lower data rates outweighs the need for broadcast quality.

The rapid drop in quality with data rate implies that only a very modest increase of basic coding errors is possible if a desirable result is obtained. Because of this limited ODG/NMR margin and the strong effect of tandem coding losses, transcoding to other coding systems is at a very strong disadvantage. In addition, the successful operation of E-AC-3 at lower data rates must be arrived at by the use of coding tools of enhanced coupling and bandwidth extension, not via increases in basic coding errors.

8. CONCLUSION

A new multichannel audio coding system called enhanced AC-3 has been described and its advanced features and characteristics explored. This system is shown to be based on and very compatible with the existing coding standard AC-3. This compatibility has been shown to be very beneficial for use within the

existing broadcast and audio/video media infrastructure because it reduces tandem coding effects while also providing coding efficiency improvements due to the use of new coding tools. The new system is very attractive for use in cable/satellite television set top boxes, terrestrial broadcast fallback audio decoder/converters, and broadcast distribution transcoding devices.

An overview of AC-3 was given and the new E-AC-3 coding tools of enhanced coupling, bandwidth extension, adaptive-hybrid transform, vector quantization, gain-adaptive quantization, and transient pre-noise processing were examined. These were shown to be a significant means for improving coding efficiency over the existing AC-3 coding standard.

The bitstream enhancements found in E-AC-3 for increasing the flexibility in number of channels carried, data rate range, and mixing/conversion metadata were also described. This increased functionality in E-AC-3 over AC-3 was shown to be useful in adapting E-AC-3 to a number of important new applications.

Finally, several applications that benefited from the basic compatibility between AC-3 and E-AC-3 were examined in more detail. In particular, the development of an E-AC-3 to AC-3 converter and AC-3 to E-AC-3 transcoder were examined. The analysis of basic quantization errors from the bit-rate-reduction process and tandem coding effects demonstrated a big advantage due to the compatibility between E-AC-3 and AC-3. This was shown to be quite significant for applications requiring eventual conversion/transcoding to AC-3 in home receivers and even more important when AC-3 source material is distributed over broadcast systems and delivered to the consumer.

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