# VECTOR QUANTIZATION OF SCALE FACTORS IN ADVANCED AUDIO CODER (AAC)

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# ABSTRACT

This paper describes some experiments to reduce the load of side information in the MPEG AAC scheme using vector quantization (VQ) methods. The VQ replaces the existing differential and entropy coding of the scale factors. Various types of VQ are considered, such as subvector/product VQ,, multi-stage VQ and tree-structured VQ which provide some advantages in the context of MPEG-4 AAC characteristics such as scaleability etc. However, the VQ being a lossy compression scheme, psycho-acoustic sensitivity of the losses is very important. These are studied using objective measures such as NMR and listening tests to make proper choices for the VQ design.

#### **1. INTRODUCTION**

MPEG-2 Advanced Audio Coder (AAC) is the newest and most effective audio coder in the MPEG family of audio/video compression schemes. In the present on-going efforts towards MPEG-4, where scaleability is an important goal to achieve, once again the AAC scheme appears to have a good promise. However, there are several new efforts underway, within the framework of AAC, to achieve improved quality at lower bitrates. One of the challenging tasks is to realize transparent quality (CD reference) at or lower than 64 Kb/s/channel. While this goal is already within reach for many types of music material, critical items such as castanets, pitch-pipe, etc., are just short of the threshold. Interestingly, some speech items also pose difficulties. Towards lower bitrates, such as 16 Kb/s or lower, the performance reference is no longer the CD quality, but a correspondingly lower bandwidth signal. Even for telephone quality speech, transparent compression has not been achieved at < 2 b/samp rate. Whereas CD quality audio has been achieved at about 1.5 b/samp by AAC. The challenge is to achieve this higher compression for low bandwidth signals including telephone speech.

On the other hand, speech coding techniques are evolving towards wide-band speech and music, providing selectively better performance than AAC at lower bitrates. Naturally, these methods perform very well for speech because of the underlying speech models used and not so well for music because of not exploiting psycho-acoustic properties as effectively as the AAC. Reducing the bitrate of AAC, thus, has great potential for new applications requiring scalability of bitrate and performance such as internet mobile usage.

#### 2. OVERVIEW OF AAC

Fig. 1 shows a block diagram of AAC encoder. The decoder is essentially the inverse operations of the encoder. The details of the AAC standard and its earlier evolution are available in the literature [1,2]. However, the functionality and effectiveness of AAC can be intuitively explained as follows. The basic approach is that of an adaptive transform coder utilizing detailed psycho-acoustic models to conceal the quantization noise; however, there are several special features at each stage compared to a basic transform coder. A high resolution, switched lapped transform is used, viz., MDCT [3], which effectively avoids the blocking effects of a transform coder. The adaptive switching provides for trading between coding gain and pre-echos that occur in transient portions of the signal. Temporal noise shaping (TNS) [4] aids this process by applying an adaptive predictor, in a novel way, in the transform domain. This is followed by a backward-adaptive multi-channel predictor [5], for each transform coefficient. This provides for exploiting signal redundancy longer than the transform width. However, this is not the same as adaptive long-term prediction (LTP of speech coders) which can exploit (spectral) periodicity within a block. This latter scheme is also under consideration for MPEG-4 AAC.

The last, but most important feature of AAC, is that of quantizing transform domain information. Unlike standard transform coders, an *adaptive noise-allocation* strategy is used instead of adaptive bit-allocation [6,7]. To optimize quantization, several multi-dimensional entropy coders are used, which are adaptively switched to achieve least bit rate. The usual bit allocation algorithms assume a fixed bitrate, which is not suited to entropy coding. This is circumvented by developing an iterative *noise-allocation* scheme in which the transform coefficients are normalized and then quantized using entropy codes; the normalization



Fig.1: AAC Encoder

factor (scale factor) itself determines the amount of quantization noise due to each coefficient. The iterative scheme tries to meet or exceed the masking threshold. In principle, this scheme can provide a fixed rate bit-stream. However, a small variation in the bitrate is permitted and using a bit reservoir, some bits from a previous stationary segment can be passed on to the demanding transient segments. The uniqueness of the scale-factor approach is that it reduces the granularity of the standard bit-allocation methods as well as accrue some advantage due to *successive* quantization of transform coefficients.

### **3. SCF QUANTIZATION**

The exponential normalization, called non-uniform quantization in AAC, of transform coefficients is given by [2]

$$X_{n}(k) = s \cdot round \left[ \left( \frac{|X(k)|}{(\sqrt[4]{2})^{\gamma} m} \right)^{0.75} - 0.0946 \right] ..(1)$$

s = sign[X(k)], X(k) are the where spectral coefficients before normalization, and  $\gamma_m, 1 \le m \le M$  are the scale factors; index m is the critical-band index to which frequency k belongs. SCFs are constrained to be integers and the exponent base is chosen to provide a 1.5 dB step-size for  $X_n(k)$ . The above equation determines the lossy quantization; the subsequent entropy coding of  $X_n(k)$  and  $\gamma_m$  are lossless. Multi-dimensional entropy codes are used for X(k) and differential scale factors, i.e.,  $\gamma_m' = \gamma_m - \gamma_{m-1}$ , are 1-D entropy coded. The bits used for X(k) is the main information in the encoded bit stream, whereas that of  $\gamma_m$ ' and a few other bits from TNS, multi-channel predictor, etc., constitute the total side information. The SCF information is thus crucial for the reconstruction of the signal at the decoder.

The application range of AAC being 128 Kb/s (CD quality) down to 8 Kb/s (telephone quality), Table-I gives a comparison of SCF information (SI) to main information (MI) at different bit rates. It can be seen that SI takes a larger portion of the total bits, upto 20%, particularly towards lower bit rate, where the coder performance is lacking. Also, the MI is already using multi-dimensional entropy coding, exploiting any correlation between successive normalized spectral values. But, the normalizing factors themselves, i.e.,  $\gamma_m$ , are coded using only a first order predictor and a 1-D entropy code. The SCFs correspond to the overall spectral envelope which has a lot more redundancy that should be exploited for quantization.

Table-I: Bit distribution: avg-bits/frm for SI & SI+MI

Bitrate	All frames		Long Blocks		Short Blocks	
64Kb/s/ch	82	1351	80	1337	147	1879
40Kb/s/ch	109	1261	107	1253	219	1586
16Kb/s/ch	156	979	150	966	240	1169

### 4. VECTOR QUANTIZATION OF SCF

The SCF,  $\gamma_m$ ,  $1 \le m \le M$ , is an integer valued vector, where M varies between long-blocks and short-blocks, because of the switched transform window. Also, transform size in samples is independent of signal sampling frequency, resulting in higher frequency resolution at lower bitrates (=> lower sampling rate) and slightly different vector lengths to adjust to the critical-band frequencies.

Vector quantization is the most effective bit reduction scheme, which can exploit simultaneously the linear dependencies between components as well as the vector pdf shape. Thus, VQ will subsume the functions of both differential coding and entropy coding, currently used for SCF. However, VQ is lossy and complexity constrained.



Fig.2: SCF Vector Quantization schemes

The much higher complexity of VQ encoding and low complexity of VO decoding is well suited to the AAC philosophy and hence, is not a problem. However, the lossy nature of VQ requires that we map the losses to the psychoacoustic domain and determine the nature and levels of acceptability. The VQ encoding complexity can be controlled to within limits by using slightly sub-optimal constrained VQ schemes, such as (i) sub-vector/product VQ, (ii) multi-stage VQ and (iii) tree-structured VQ [8]. These schemes also have certain advantages to specific applications of AAC requiring scaleability, etc. Fig.2 briefly indicates the three VQ structures. The distance measure in VQ is an important design consideration. However, since the perceptual sensitivity of the SCF vector is yet unknown, a simple Euclidian distance measure is used in all the experiments.

# VQ in DCT domain:

Another important issue concerned with complexity is the vector dimension. In principle, larger dimensions provide better compression at the cost of higher complexity; they also require larger amount of training data, hence tedious design effort. To reduce dimensionality, the linear dependency should be reduced. This can be achieved through decorrelation using an orthogonal transform. The transform closest to the optimum KLT (Karhunen-Loeve transform) is the DCT which can result in close to diagonal covariance matrix in the transform domain. The AAC already uses the DCT kernel in realizing the MDCT, and hence it would be easy to implement the of the scale factor vector; DCT/IDCT i.e.,  $\{\Gamma_m\} = DCT\{\gamma_m\}$ . However, the decoder has to bear slightly higher complexity because of the introduction of IDCT.

Fig.3 shows the variances of the individual scale factors, at different bitrates of AAC encoding; i.e.,

 $\sigma_m^2 = \mathbf{\mathcal{E}} (\gamma_m - \overline{\gamma_m})^2$  and  $\sigma_k^2 = \mathbf{\mathcal{E}} (\Gamma_k - \overline{\Gamma_k})^2$ . It can be seen that, both for long blocks and short blocks, the variance fluctuates much between band-index and bitrates. This would require a large dimension vector for good

compression. However, in the DCT domain, large variance is concentrated towards low indices and much lower variance at high indices, for all bitrates. This will permit a sub-vector/product VQ to be more effective in the DCT domain than the SCF domain. However, complexity permitting, a full-vector multi-stage VQ can give comparable performance in both the DCT and SCF domains.

#### 5. PSYCHO-ACOUSTIC SENSITIVITY

The sensitivity of reconstructed signal due to the lossy VQ of  $\gamma_m$  or  $\Gamma_m$  is studied using simulated random errors as well as some types of VQ itself. For simulation purpose, these errors can be viewed as occurring either (i) only at encoder, or (ii) only at decoder or (iii) both at encoder and decoder. These cases correspond to different operational conditions of the audio coder: (i) corresponds to suboptimality of the noise-allocation process, (ii) corresponds to possible errors in the bit-stream and (iii) is simultaneous (i) and (ii). Case-(i) is simulated by running the outer loop of the noise-allocation process, just one more time using  $\hat{\gamma}_m = \gamma_m + q_m$ , whereas case-(ii) can be directly simulated. The error vector  $q_m$  is Gaussian distributed with variance corresponding to the VQ residual and it can be added either in the SCF domain or in the DCT domain,  $\Gamma_m$ . The DCT domain quantization can provide some other advantages justifying the additional IDCT operation at the decoder.

The sensitivity of the reconstructed signal is determined through informal randomized-pair comparison test, using the no error decoded signal as reference. Also, an objective measure of subjective quality, NMR (noise-to-mask ratio), is computed using the new EQ system [9]. (NMR is computed wrt original) The most critical music items, viz., castanets (tk2) and German male speech (tk3), are considered for initial evaluation. It is found that deterioration of ~1 dB *Total-NMR* becomes perceptually noticeable by an expert listener in case-(i) type of errors. This corresponds to Gaussian errors of  $\sigma = 1.5$ . The different VQ schemes shown in Table-II are: (3) 8 subvector mean-removed product code for  $\gamma_m$ , (4) 4 subvectors of non-uniform dimensions in  $\Gamma_m$ , (5) 8 subvectors of uniform dimension in  $\gamma_m$ . The quantization bits required for these VQs range from 70 to 40, respectively, for coding a long block frame; a short-block will use proportionately smaller bits. As can be seen from the NMRs in the table, most items are hardly distinguishable, except G-rnd2 ( $\sigma = 2$ ). Thus, using the best VQ, one can achieve a saving of about 50% bits from the SI.

For case-(ii) errors, the measure is not distinguishability, but tolerability. Table-III gives the NMR values at three bitrates for the same critical audio items. These items were judged as deteriorated by more than 2 points on a perceptual scale. However, at lower bitrates, the degree of deterioration is less as also shown by the NMR differences between no-error and error. This is found to be so in case-(i) errors also. By comparing with Table-I, one can see that the bitrate saving due to VQ would be more significant and *lesser* loss in quality at lower bitrates.

**Conclusion:** Vector quantization of SCF information is very promising within the AAC framework, particularly for lower than 64 Kb/s bitrate. The saved bits from SI would be useful for coding the main spectral information, which can improve the quality.

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Table-II: *Total-NMR* for 64 Kb/s coded items tk2 & tk3, respectively, for case-(i) error.

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1	G-RND 1.5	-13.9	-10
2	G-RND 2	-13.6	-9.7
3	VQ 4x8	-14.6	-10.3
4	DCT/VQ 8x4	-14.6	-10.3
5	zmn/VQ 5x8	-14.6	-10.3
	No error	-15	-10.5

**Table-III:** *Total-NMR* at lower bitrates for tk2 & tk3, respectively, for no-error and case-(ii) error

respectively, for no error and ease (ii) error.								
64 KB/S; 48 KB/S	-15	-10.5	-10.5	-4.5				
40 Kb/s; 32 KS/s	-10.9	-7.8	-11.4	-5.9				
16 Kb/s; 16 KS/s	-4.2	-2.8	-5.3	-3.4				
G-RND 1.5 (64Kb)	-15	-9.8	-10.5	-7.6				



Fig:3: Standard deviation of SCFs and their DCT coefficients (+); 64 Kb/s (-), 40 Kb/s (--), 16 Kb/s (-.)