A NEW CRITERION AND ASSOCIATED BIT ALLOCATION METHOD FOR CURRENT AUDIO CODING STANDARDS

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ABSTRACT

This paper presents a new noise-shaping criterion. Based on the new criterion, we derive an efficient bit allocation method. The bit allocation method is applicable to the current audio standards like MPEG1 Layer 3 and MPEG4 AAC. The bit allocation method has gained a speed up for more than ten and has resulted in better quality over the traditional two nested loop method presented in ISO draft. The experiments illustrated the correction of the objective measurement criterion and the new allocation has shown the deterministic method instead of the iteration method to achieve the high allocation efficiency and best quality.

1. INTRODUCTION

The bit allocation aims to assign suitable parameters to the encoder to achieve the best audio quality under the restricted bit number. Hence control over the quality and the bit number are two fundamental requirements for the bit allocation. The complexity of the task depends on the difficulties to have the quality and bit control. For MPEG Layers 1 and 2 [1], both the quality and the bit requirement are controlled by a uniform quantizer. Hence the bit allocation is just to apportion the total number of bits available for the quantization of the subband signals to minimize the audibility of the quantization noise. For coders such as MPEG Layer 3 [1], MPEG-2 AAC [2], and MPEG4 T/F coding [3], controls over the quality and the bit rate are difficult. This is mainly due to the fact that they both use a non-uniform quantizer whose quantization noise is varied with respect to the input values. In other words, it fails to control the quality by assigning quantizer parameters according to the perceptually allowable noise. In addition, the bit-rate control issue can be examined from the variable length coding used in MPEG Layer 3 and MPEG-2 AAC. The variable length coding assigns variable bit-length to different values, which means that the bits consumed should be obtained from the quantization results, and cannot be from the quantizer parameters alone. Thus, the bit allocation is one of the main tasks leading to the high complexity of the encoder. This paper presents a new bit allocation method to ease the complexity. We take MPEG Layer 3 as the detail derivation and experiment example.

The above two difficulties lead to the problem in evaluating the quantization parameters. A two-nested loop iterative method referred to as the OCF (optimum coding in the frequency domain) has been proposed in [4] to solve the problem. Te OCF method evaluates the quantization parameters through two iteration loops: the rate-controlling loop and the quality-controlling loop. The rate-controlling loop adjusts in iteration the parameter values to fit to the limited bits the consumed bits which are obtained by performing quantization and Huffman coding for spectral lines. The quality-controlling loop adjusts in iteration the parameter values to fit to a perceptual criterion the quantization noise that needs to be evaluated by performing the inverse quantization. The method can be examined from the complexity and the induced audio quality. The complexity of the method for a frame with F spectral lines is $O(F \cdot R \cdot \eta + F \cdot Q \cdot \gamma)$, where Q and R are respectively the numbers of quality-controlling iterations and rate-controlling iterations while the η and γ are the computation complexity to handle a spectral line in the rate-controlling loop and the quality-controlling loop, respectively. The rate-controlling loop complexity η will be from the quantization and the Huffman coding of a spectral line while the quality-controlling loop complexity γ from the dequantization and noise evaluation. Both the complexity η and γ are high. Also, the numbers of iterations Q and R depend on the initial values of quantization parameters and the adjust manners. The second problem is on the quality of the coded audio. The method of assigning bits to quantization bands in the quality-controlling loop decides the quality. There have been two approaches in the assigning manners. One method is to assign in each iteration only the band with the worst noise-to-masking ratio. The method leads to a large number of iterations in the quality-controlling loop, which means the immense complexity. Another method assigns in iteration to all the bands with noise-to-masking ratio higher than one until all available bits are consumed. This method has a much lower complexity than the first method. However, the quality of the method is the concerns. The first method can shape the noise so that the masking threshold will be in parallel to the noise threshold, which has been a widely accepted criterion [5]. The second method that has been in the draft provided by ISO can be referred to as the approximate method on the first method. Since that there are two separate rules controlling the quality and bits consumed in two loops, there may lead to infinite loops, generally referred to as "dead-lock problem". A general method to handle the deadlock problem is to set the maximum number of iterations; however, the quality may be sacrificed to meet the bit rate constraint. This paper presents a new bit-allocation method that has merits in both complexity and audio quality.

2. BANDWIDTH-PROPORTIONAL NOISE SHAPING

Consider the minimum over segmental noise-to-masking ratio.

$$R(i) = \arg \underset{R(i)}{Min} \sum_i \left\{ \left(\frac{\sigma^2_{N(i)}}{\sigma^2_{M(i)}} \right) \right\}, \tag{1}$$

where $\sigma^2_{N(i)}$ and $\sigma^2_{M(i)}$ is the noise energy and the masking energy associated with the perceptual band *i*. *R(i)* is the bit rate to minimize the segmental NMR. In an *R(i)* bits/sample PCM coder, the quantization error variance is given by

$$N(i) = \rho 2^{-2R(i)} \sigma^2_{x(i)} \tag{2}$$

So, the minimization should be constrained by the total bit rate; that is

$$\sum_i \{R(i)B(i)\} = R \cdot \tag{3}$$

where *B(i)* is the effective bandwidth of the critical band *i*. According to the method of Lagrange multipliers, According to the method of Lagrange multipliers, the solution must satisfy

$$\frac{\partial}{\partial R(j)} \left\{ \left(\sum_i \{R(i)B(i)\} - R \right) + \lambda \sum_i \left\{ \left(\frac{\rho 2^{-2R(i)} \sigma^2_{x(i)}}{\sigma^2_{M(i)}} \right) \right\} \right\} = 0, \tag{4}$$

for all *j*. Then

$$\lambda = \frac{B(j)}{(2\log 2)\left(\frac{\rho 2^{-2(R(j))} \sigma^2_{x(j)}}{\sigma^2_{M(j)}} \right)} = \frac{B(j)}{2\log 2 \left(\frac{\sigma^2_{N(j)}}{\sigma^2_{M(j)}} \right)}, \tag{5}$$

for all *j*. So, *R(j)* should be allocated such that the noise-to-masking ratio is proportional to the *B(j)*. That is

$$\sigma^2_{N(j)} = \kappa \sigma^2_{M(j)} B(j), \text{ for all } j. \tag{6}$$

where κ is a constant. The noise shape is different from the traditional criterion that noise is shaped parallel to the masking threshold.

The noise level for the quantization bands is selected in consideration of the masking threshold and critical bandwidth in the quantization band. In other words, the $\sigma^2_{N(q)}$ instead of the $\sigma^2_{N(j)}$ is to be found to minimize the segmental NMR

$$\sigma^2_{N(q)} = \kappa \sigma^2_{M(q)} B(q) \tag{7}$$

where *q* is the index of the quantization band. The problem is equivalent to finding *B(q)* to approximate best the energy defined to minimize the segmental NMR; that is

$$\hat{B}(q) = \arg \underset{B(q)}{Min} \sum_{j \in q} \left| \sigma^2_{N(q)} - \sigma^2_{N(j)} \right| \tag{8}$$

Assume that the masking energies of the critical bands in the quantization bands are uniform, the selection after calculation is

$$\hat{B}(q) = \underset{j \in q}{Average}\,(B(j)) \tag{9}$$

To avoid the bits allocated to the bands with masking level higher than the noise level, the criteria to minimize the segmental NMR is modified so that the bands with negative NMR should be rounded to 0. That is, the quantization noise for each band should have a lower bound. On the other hand, the noise higher than the masking threshold leads to a phenomenon that the associated band will be rounded to zero, referred to as the zero bands. The zero bands are quite perceptually noticeable. So, the quantization levels should also be restricted to be no larger than the signal energy.

To summarize, the criterion is to allocate bits to bands proportional the bandwidth when the quantization noise is higher than masking threshold.

3. NEW BIT ALLOCATION METHOD

Based on the bit allocation criterion mentioned in last Section, this paper proposes a method to evaluate the quantization parameters directly from the masking threshold. The resultant evaluation method can guarantee the quality and will be free from the iteration used in the analysis-by-synthesis method. We adopt MPEG Layer 3 as the example to derive the parameter estimator.

3.1. Noise Predictor for Nonuniform Quantization

Nonuniform quantization can lead to a quantization noise with level proportional to the signal energy, which has been adopted in current audio standards like MPEG Layer 3, MPEG AAC (Advanced Audio Coding), and MPEG4 T/F coding. However, the difficulty in evaluating the quantization noise from the parameters leads to the analysis-by-synthesis method. From MPEG layer 3 standard [1], the simplified formula for the non-uniform quantizer of layer 3 is

$$is_i = \text{int}\left(\frac{xr_i^{\frac{3}{4}}}{\Delta_q} \right), \tag{10}$$

where quantization step size

$$\Delta_q = 2^{\frac{3}{4}(gain_{gr} - scale_q)}. \tag{11}$$

From the MPEG standard, the non-uniform quantizer is expressed as

$$is_i = \text{int}\left(xr_i 2^{scale_q - gain_{gr}} - 0.0946 \right)^{\frac{3}{4}}, \tag{12}$$

where $scale_q = 1/2(1 + scalefac\_scale)(scalefac_q + preflag_{gr} \cdot pretab_q)$ for each quantization band *q*; $scalefac\_scale$ is 0 or 1, $scalefac_q$ is in the range of 0~15 depending on bands, and the pre-amplified value $preflag_{gr} \cdot pretab_q$; global gain $gain_{gr} = (global\_gain_{gr} - 210)/4$ for each granule of MPEG layer 3 frame. By ignoring 0.0946, (12) can be derived as

$$
\begin{aligned}
is_i &= \operatorname{int}\left(xr_i \, 2^{scale_q - gain_{gr}} \right)^{\frac{3}{4}} \\
&= \operatorname{int}\left(xr_i^{\frac{3}{4}} \, 2^{\frac{3}{4}(scale_q - gain_{gr})} \right) \\
&= \operatorname{int}\left(\frac{xr_i^{\frac{3}{4}}}{\Delta_q} \right)
\end{aligned} \qquad . \qquad (13)
$$

The input signal xr_i and reconstructed signal $\tilde{xr}_i$ have the following two formulae:

$$xr_i = \left((is_i + \varepsilon_i)\Delta_q\right)^{\frac{4}{3}} \text{ , and } \tilde{xr}_i = \left(is_i\Delta_q\right)^{\frac{4}{3}} .$$

The quantization error of the non-uniform quantizer e_i will be equal to the difference of input signal xr_i and reconstructed signal $\tilde{xr}_i$:

$$
\begin{aligned}
e_i &= xr_i - \tilde{xr}_i = \left((is_i + \varepsilon_i)\Delta_q\right)^{\frac{4}{3}} - \left(is_i\Delta_q\right)^{\frac{4}{3}} \\
&= (1 + is_i^{-1}\varepsilon_i)^{\frac{4}{3}} is_i^{\frac{4}{3}} \Delta_q^{\frac{4}{3}} - \left(is_i\Delta_q\right)^{\frac{4}{3}}
\end{aligned} \qquad (14)
$$

Let $f(\varepsilon_i) = (1 + is_i^{-1}\varepsilon_i)^{\frac{4}{3}}$. By Tyler expansion with the first order approximation of $f(\varepsilon) \approx 1 + f'(\varepsilon)\varepsilon$, this leads to

$$e_i = f(\varepsilon_i) is_i^{\frac{4}{3}} \Delta_q^{\frac{4}{3}} - \left(is_i\Delta_q\right)^{\frac{4}{3}} \approx \tfrac{4}{3} is_i^{\frac{1}{3}} \varepsilon_i \Delta_q^{\frac{4}{3}} . \quad (15)$$

Assume that the quantized signals is_i and the quantized error of the uniform quantizer ε_i are independent; the expectation of the quantization error of the non-uniform quantizer e_i is as follows:

$$E[e_i^2] \approx \tfrac{16}{9} \Delta_q^{\frac{8}{3}} E[IS_i^{\frac{2}{3}} \varepsilon^2] \approx \tfrac{16}{9} \Delta_q^{\frac{8}{3}} E[IS_i^{\frac{2}{3}}] E[\varepsilon_i^2] \quad (16)$$

If the spectrum of the quantization bands is uniform, the noise of lines can be the average energy of quantization band; that is

$$E(e_i^2) = E(e_q^2) \quad (17)$$

Since $E[\varepsilon_i^2] = \frac{1}{12}$, (17) becomes

$$E[e_i^2] \approx \tfrac{4}{27} \Delta_q^{\frac{8}{3}} E[(\frac{XR_i^{\frac{3}{4}}}{\Delta_q})^{\frac{2}{3}}] = \tfrac{4}{27} \Delta_q^{2} E[|XR_i|^{\frac{1}{2}}] \quad (18)$$

3.2. Quantization Parameters Evaluation

If the spectrum of the quantization bands is uniform, we can let the noise of lines be the average energy of quantization band.

$$E(e_i^2) = E(e_q^2) \quad (19)$$

Substituting (7) into (19) yields

$$E[e_q^2] = \kappa \sigma_{M(q)}^2 B(q) \quad (20)$$

Define $T_q = \sigma_{M(q)}^2 B(q)$. So,

$$E[e_q^2] = \kappa \cdot T_q^2 \approx \tfrac{4}{27} \Delta_q^2 E[|XR_q|^{0.5}] ,$$

That is

$$\Delta_q^2 \approx \tfrac{27}{4} \kappa \cdot T_q^2 / E[|XR_q|^{0.5}] \quad (21)$$

Combining (11) and (21) yields

$$\Delta_q^2 = 2^{\frac{3}{2}(gain_{gr} - scale_q)} \approx \tfrac{27}{4} \kappa \cdot T_q^2 / E[|XR_q|^{0.5}] \quad (22)$$

The difference of global gain and scalefactor is

$$gain_{gr} - scale_q = \tfrac{2}{3}(\log_2 \frac{27}{4} + \log_2 \kappa + \log_2 T_q^2 - \log_2 E[|XR_q|^{0.5}]) \quad (23)$$

Since scalefactor $scale_q$ is in the range of 0~16 and the minimum scale for these quantization bands must be zero. So,

$$
\begin{aligned}
gain_{gr} &= \underset{q}{Max}\{gain_{gr} - scale_q\} \\
&= \underset{q}{Max}\{\tfrac{2}{3}(\log_2 6.25 + \log_2 \kappa + \log_2 T_q^2 - \log_2 E[|XR_q|])^{0.5}\}
\end{aligned} \quad (24)
$$

and the scale factors for all sub-bands are obtained. It can be seen that the global gain varies with the bit rate related constant κ, and the scale factor varies for each sub-band according to the masking threshold and the input signals.

3.3. Bounds on Scale factors

As mentioned before, the bits should be allocated under non-negative NMR and the constraint of zero bands. For the non-negative NMR issues, the noise level is set to be the masking threshold; that is $T_q = \sigma_{M(q)}^2$ and $\kappa = 1$. This yields to the upper bound of the U $scale_q$ relative to the global scale.

$$gain_{gr} - Uscale_q = \tfrac{2}{3}(\log_2 \frac{27}{4} + \log_2 \sigma_{M(q)}^2 - \log_2 E[|XR_q|^{0.5}]) \quad (25)$$

That is,

$$scale_q \le Uscale_q = gain_{gr} - \tfrac{2}{3}(\log_2 \frac{27}{4} + \log_2 \sigma_{M(q)}^2 - \log_2 E[|XR_q|^{0.5}]) \quad (26)$$

The $gain_{gr}$ will be adjusted according to the available bits. The lower bounds can be derived under the constraint of the zero bands. The zero bands occur when the noise is greater than the signal energy; that is

$$\Delta_q^2 = (2^{\frac{3}{4}(gain_{gr} - Dscale_q)})^2 < \{E[|XR_q|^{0.5}]\}^{\frac{3}{4}} \quad (27)$$

Thus, the lower bound on the scale will be

$$scale_q \geq Dscale_q = gain_{gr} - \frac{1}{2}\log_2 E[\left|XR_q\right|^{0.5}] \tag{28}$$

4. EXPERIMENTS

Table 1 illustrates the average iteration number with different testing materials for the present invention and the MPEG bit allocation process respectively, where Q is the quality-controlling iterations and R is the rate-controlling iterations. As shown in Table 1, the new allocation method has removed the iterations required for the quality-controlling iteration and have reduced the rate controlling iterations by a factor more than three.

Table 2 illustrates the objective score of the invented method compared to the bit allocation method in Lame [6]. Here the invention adopts PEAQ (perceptual evaluation of audio quality) system which is the recommendation system by ITU-R Task Group 10/4. Here the invention adopts PEAQ (perceptual evaluation of audio quality) system which is the recommendation system by ITU-R Task Group 10/4. The objective difference grade (ODG) is the output variable from the objective measurement method. The ODG values should ideally range from 0 to -4, where 0 corresponds to an imperceptible impairment and -4 to an impairment judged as very annoying. All test tracks are chosen that they are long enough to stress the codec under test. These test signals are used in formal subjective or objective test by EBU [6]. As shown in Table 2, the quality from the method of the new allocation algorithm is even better than Lame which is claimed to be the best mp3 encoder.

The configuration adopted in this invention for PEAQ is the basic version. The basic version uses the FFT-based ear model. It uses the following model output variables: $BandwidthRef_B$, $BandwidthTest_B$, $Total\ NMR_B$, $WinModDiff1_B$, ADB_B, EHS_B, $AvgModDiff1_B$, $AvgModDiff2_B$, $RmsNoiseLoud_B$, $MFPD_B$ and $RelDistFrames_B$. These 11 model output variables are mapped to a single quality index using an artificial neural network with three nodes in the hidden layer. Table 3 provides a list with a subset of test signals that were used during the objective and subjective test.

Tracks	1	2	3	4	5	6	7	8
R in ISO	38.33	39.52	29.66	40.77	37.79	39.39	38.98	38.07
Q in ISO	11.39	13.76	8.97	11.32	11.10	11.71	12.68	11.63
R in this paper	12.20	12.58	11.98	11.33	12.99	12.90	11.61	11.30
Q in this paper	0	0	0	0	0	0	0	0

Table 1 *Iteration numbers for each granule in MP3.*

5. CONCLUSION

This paper have presented the bandwidth-proportional allocation criterion for the noise shaping in audio compression. Based on the new criterion, we derive an efficient bit allocation method. The bit allocation method is applicable to the current audio standards like MPEG1 Layer 3 and MPEG4 AAC. The bit allocation method has gained a speed up for more than ten and has resulted in better quality over the traditional two nested loop method presented in ISO drafts and that used in Lame. The method has been illustrated to improve the computing speeds by a factor more than ten and has a quality even better than Lame.

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	1	2	3	4	5	6	7	8
ISO	-2.11	-2.32	-2.06	-1.48	-2.42	-2.22	-1.28	-1.67
Lame 3.88	-1.52	-1.01	-1.2	-0.62	-1.71	-1.55	-0.54	-0.94
New Method	-1.42	-0.95	-1.08	-0.6	-1.37	-1.33	-0.47	-0.83

Table 2 *Objective test on the new method and Lames. The configuration on the encoder has adopted the stereo mode coding and long window only.*

No.	Items	File name	Remarks
1	Bass	Bass	(d)
2	Glockenspiel	Gspi	(a) & (b) & (e)
3	Harpsichord	Harp	(a) & (b) & (d)
4	Horn	Horn	(d)
5	Quarter	Quar	(d)
6	Soprano	Sopr	(d)
7	Trumpet	Trpt	(b)
8	Violoncello	Vioo	(e)

Remarks:
(a) Transients: pre-echo sensitive, smearing of noise in temporal domain.
(b) Tonal structure: noise sensitive, roughness.
(c) Natural speech (critical combination of tonal parts and attacks): distortion sensitive, smearing of attacks.
(d) Complex sound: stresses the Device Under Test.
(e) High bandwidth: stresses the Device Under Test, loss of high frequencies, program-modulated high frequency noise.

Table 3 *Test tracks*

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