

Quality Evaluation Of Watermarked Audio Tracks

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ABSTRACT

This paper presents an exhaustive evaluation of the quality of an audio watermarking algorithm.¹ The integration of the psychoacoustic model into the audio watermarking approach is demonstrated. The quality parameter relating the power of the watermark noise and the masking threshold is presented.

The evaluation method is detailed and the quality of the watermarked audio tracks is evaluated with regard to different settings of the quality parameter used to adjust the power of the embedded watermarks.

The subjective listener test compares the quality of the original audio track with the watermarked one. Different quality parameter settings were used in order to enable the adjustment between quality and maximum robustness according to the items to be watermarked and the target audience.

Keywords: Audio Watermarking, Copyright Protection, Multimedia Security, MPEG

1. INTRODUCTION

The digital information age and especially the WWW stands at a crossroad. Commercial distribution of data and information over the Internet will disappear if it is not possible to protect the intellectual property rights (IPR) in a sufficient manner. Especially the music industry is seeking for reliable solutions to problems associated with the protection and misuse of audio data.² Restriction of the access to the data is not manageable in applications where the user needs access to the data in order to enjoy it. New methods like digital watermarking techniques have been proposed to meet the needs by embedding a robust, perceptually and statistically undetectable link directly into the raw data.

Despite the fact that watermarking is a relatively new research field a lot of publications are already available. Most of the research work has been performed in the image and video field. In comparison there exist only a few audio watermarking methods³⁴⁵⁶⁷¹⁸.

The organization of the paper is as follows. Section 2 starts with a discussion about audio watermarking and the requirement of quality preservation of the watermarked audio track. The next section presents the watermarking algorithm. The encoder and decoder are discussed in the section 4 describing the watermarking system. Section 5 presents the quality evaluation of the watermarked audio tracks. This includes the setup of the testbed, statistical considerations and corresponding test results. The paper is completed by a discussion of the results and possible improvements.

2. AUDIO WATERMARKING AND QUALITY

The international federation of phonographic industry (IFPI) has specified the desired features of an optimal audio watermarking method. This requirements can be detailed and further subdivided in different categories describing the properties of the algorithm and the watermark

- Signal processing properties
- Security properties
- Application specific requirements

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According to the intended application of the watermarking algorithm different variations and corresponding design criteria are relevant for the development of an effective audio watermarking method. The most important requirement addresses the quality of the watermarked items. If the quality of the audio tracks cannot be preserved the user and especially the recording industry will not accept this new technology. Consequently the definition of robustness of the watermark depends on the quality of the watermarked audio track. A watermark is robust if it's removed from the audio track, but the quality is decreased at the same time. This emphasizes the priority in ranking among the requirements from quality (first) to robustness (second) and data capacity (third). Most of the work already done aims at the development of efficient embedding and reliable watermark detection procedures. We claim that assessing the quality of the watermarked audio track is the first step in a complete evaluation of any audio watermarking method.

3. ALGORITHM

This section provides a short overview of the underlying embedding and detection algorithm. *The original audio track is subdivided into individual frames each one carrying one bit. One frame in turn consists of N_B blocks. The number of samples for each frame is not fixed, but from a practical point of view a multiple of the N_S samples of one block. One block is the basis for the analysis of the audio stream according to the underlying psychoacoustic model. The audio watermarking method is based on hypothesis testing. It works in the frequency domain and does not require the original in order to detect the watermark.

Embedding the watermark

1. Use the secret key to map each bit $w \in \{0, 1\}$ of the watermark pseudo-randomly onto two different pattern $p^w = (0, 1, -1, 0, 1 - 1, \dots, 1, 0)$, with length $2N = N_B * N_S$. The sign of the elements p_j^w specify the alteration of the magnitudes of Fourier coefficients forming the set $f_j \in \mathcal{F}, j = 1, \dots, 2N$ for each frame. The same pattern are generated during the detection procedure to enable the hypothesis testing. The two intermixed subsets $\mathcal{A}_w = \{a_i\}_{i=1, \dots, M} = \{f_j | p_j^w = 1\}$ and $\mathcal{B}_w = \{b_i\}_{i=1, \dots, M} = \{f_j | p_j^w = -1\}$ $M \leq N$ of Fourier coefficients are those, whose magnitudes will be altered $\mathcal{A}_w \cup \mathcal{B}_w \subseteq \mathcal{F}$.
2. Change the Fourier coefficients $f_j, j = 1, \dots, 2N$ according to

$$f'_j = f_j + p_j^w \Delta f_j \quad (1)$$

which is equivalent to changes of the coefficients in the subsets $\mathcal{A}_w, \mathcal{B}_w$

$$a'_i = a_i + \Delta a_i, \quad b'_i = b_i - \Delta b_i, \quad i = 1, \dots, M \quad (2)$$

The strength $\Delta a_i, \Delta b_i, i = 1, \dots, M$ are controlled by the masking threshold to achieve inaudibility.

Detection of the watermark

1. Map the secret key onto the seed of the random number generator in order to generate the pattern $p^w, w \in \{0, 1\}$ for the '1' and '0' bit.
2. Define a threshold T for detection.
3. The model for a testable hypotheses in the reading process will be derived by the definition of a appropriate test statistic z . Furthermore the test and alternative hypothesis must imply different *expected values* for z , that is

$$E(z | H_0 \text{ correct}) \neq E(z | H_1 \text{ correct}) \quad (3)$$

*The algorithm used for embedding and detecting the watermark is presented in detail in.¹

z will be a function $g(\vec{a}', \vec{b}')$ of the elements of subsets \mathcal{A}_w and \mathcal{B}_w †

$$z = g(\vec{a}', \vec{b}') = \frac{\bar{a}' - \bar{b}'}{\sigma_{\bar{a}' - \bar{b}'}} \quad (4)$$

having two distinct distributions in the unmarked $\phi(z)$ and marked case $\phi_m(z)$. Calculate the sample mean $E(z) = E(g(\vec{a}', \vec{b}'))$ for each pattern. Use the maximum value of the '1' or '0' bits to decide between the two mutually exclusive propositions:

$$\begin{aligned} H_0 : E(z) \leq T & \quad \text{bit } w \text{ is **not** embedded} \\ H_1 : E(z) > T & \quad \text{bit } w \text{ is **is** embedded} \end{aligned} \quad (5)$$

4. WATERMARKING SYSTEM

4.1. Watermark encoder

The task of a watermark encoder is to shape the watermark in order to ensure indaibility on the one hand and simultaneously embed the watermark with the maximum power according to the carrier signal to provide maximum robustness. A typical audio watermark encoder consists of several components (Fig. 1). The noise generator produces noise by means of a secret key *Key*. The psychoacoustic model (PAM) block analyzes the original signal $s(t)$ in order to calculate the minimum masking threshold LT_{Min} . This threshold defines the frequency response of the LT_{Min} -Filter, which shapes the watermark. The filtered watermark signal is scaled in order to shift the watermark noise below the masking threshold. The result is the actual watermark signal added to the delayed original signal to produce the watermarked track $s_w(t)$. The delay component takes into account the time delay introduced by the filtering process.

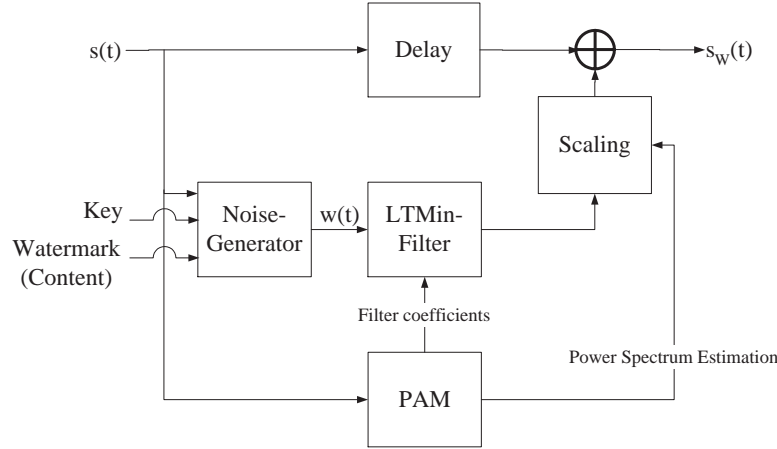


Figure 1: Watermark encoder and its components

4.1.1. Noise Generator

The task of a noise generator is to produce an additional signal $w(t)$ to the carrier signal $s(t)$, that may not be distinguished from random noise. This watermark is the input of the LT_{Min} -Filter block. The watermark noise produced has an amplitude normalized to 1. The phases of the fourier coefficients are taken from the original signal. The alteration for each '1' or '0' bit of the magnitude is determined by the two different pseudo-randomly selected pattern according to section 3.

In general the signal can undergo a transformation T in an other domain (Wavelet, Cepstrum etc.) than the fourier domain used in this implementation. Nevertheless the statistical considerations are independent of the embedding domain used.

†The altered elements are \vec{a}' and \vec{b}' .

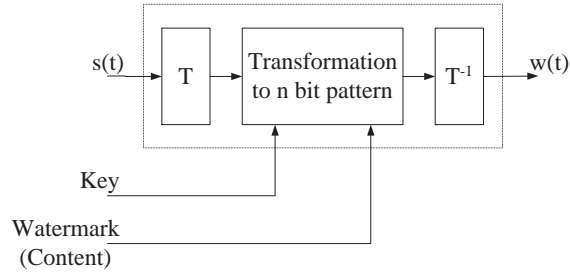


Figure 2: Watermark noise generator

4.1.2. Psychoacoustic model

Psychoacoustic models used in current audio compression coders apply the frequency and temporal masking effects in order to ensure inaudibility by shaping the quantization noise according to the masking threshold. In turn a natural approach is to use already existing models for shaping the watermark noise. The different psychoacoustic models differ in complexity and the implementation of the different masking effects. In this paper we used the psychoacoustic model 1 layer I of ISO-MPEG with $f_s = 44.1kHz$ sampling rate.⁹ In order to iteratively allocate the necessary bits the MPEG standard calculates the signal-to-mask ratios (SMR) of all the sub-bands. This is not necessary in our case since only the masking threshold for each block is of interest. Therefore we perform the following steps:

1. Calculation of the power spectrum.
2. Identification of the tonal (sinusoid-like) and non-tonal (noise-like) components.
3. Decimation of the maskers to eliminate all irrelevant maskers.
4. Computation of the individual masking thresholds.
5. Computation of the global masking threshold.
6. Determination of the minimum masking threshold in each sub-band.

4.1.3. Spectral weighting block

The LT_{Min} filter block is based on a piecewise linear approximation of the masking threshold which represents the frequency response. We used a m -tap FIR filter (with m even). Therefore the original has to be delayed by $m/2$ samples before added to the watermark noise. The masking threshold $LT_{Min}(n)$ is calculated for each sub-band n of the frequency spectrum:

$$LT_{Min}(n) = \min_{f^{(i)} \in n} LT_g(i) [dB], \quad n = 1, \dots, 32, \quad i = 1, \dots, 106 \quad (6)$$

It is the minimum of the local masking threshold and the threshold in quiet $LT_q(i)^\ddagger$ in the corresponding sub-band n . The 32 sub-bands are the result of a reduction of the 106 frequency indexes (for Layer I and sampling rate $f_s = 44.1kHz$) with a nonlinear frequency resolution. In order to maximize the power of the embedded watermark the minimum relation in equation (6) should be lowered. To investigate the effect we changed the implementation of the PAM to allow different number of sub-bands in the frequency domain. As an evaluation criteria one can measure the relative factor of embedding strength defined by

$$k := \frac{\Delta \bar{a} + \Delta \bar{b}}{\bar{a} + \bar{b}}, \quad \Delta \bar{a} = \frac{1}{M} \sum_{i=1}^M \Delta a_i, \quad \Delta \bar{b} = \frac{1}{M} \sum_{i=1}^M \Delta b_i \quad (7)$$

[‡] i is the index of the sub-sampled frequency domain and tabulated in the ISO standard.

for varying number of sub-bands and different audio files (see Fig. 3). k describes the relative variation of all fourier coefficients used for embedding.

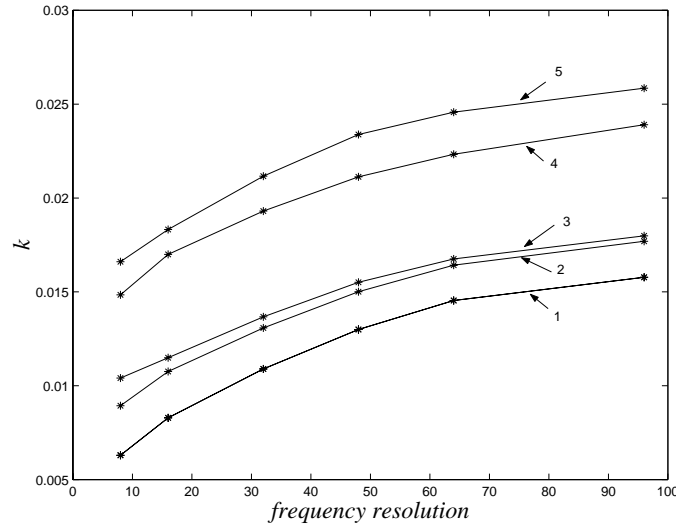


Figure 3: Effective factor as function of filter resolution for different tracks (1 – 5).

According to the results presented above a higher frequency resolution corresponds to a higher effective embedding strength. Despite the fact that individual audio tracks have different embedding factors the run of the curve is the same.

4.1.4. Scaling block

The masking threshold LT_{Min} is based on the FFT analysis of the psychoacoustic model which normalizes the maximum sound pressure level to $+96dB$. After filtering the watermark noise the sound pressure level has to be adjusted. To calculate the correct attenuation of the watermark noise W_{Noise} the power spectrum is estimated and adjusted (see Fig. 4) to a Noise-to-Mask-Ratio (NMR) defined by

$$NMR = W_{Noise} - LT_{Min} \text{ [dB]} \quad (8)$$

One would expect an $NMR = W_{Noise} - LT_{Min} = 0$ dB to be inaudible. NMR is the quality parameter we use in order to adjust the power of the watermark noise in relation to the masking threshold LT_{Min} .

4.2. Decoding

The decoding procedure is pretty straightforward (Fig. 5). One has to transform with transformation $T = DFT$ in the domain used during the embedding procedure. The hypothesis testing has to be performed as described in section 3.

5. QUALITY EVALUATION

To perform a reliable testing of the quality of watermarked audio tracks the whole test has to be specified in detail including:

- What is the aim of the test?
- What are the parameters used during testing?
- Explanation of the selection of test items used during testing.
- Description of the test equipment, setup and implementation.
- Statistical evaluation of the received test results in order to perform a quantitative analysis.

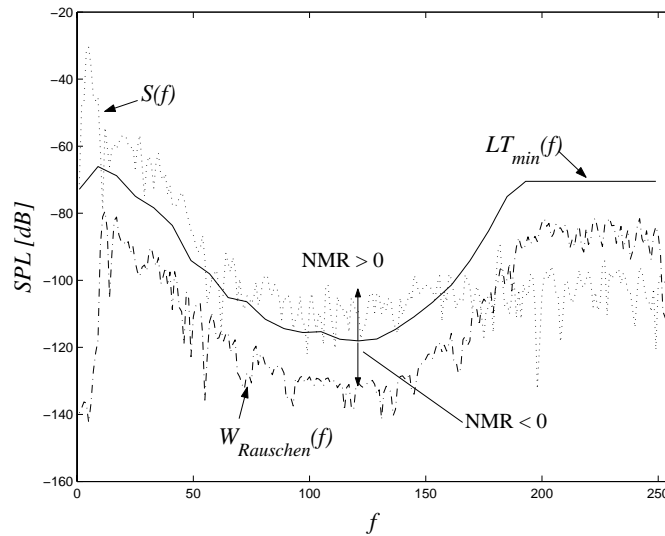


Figure 4: Sound pressure levels

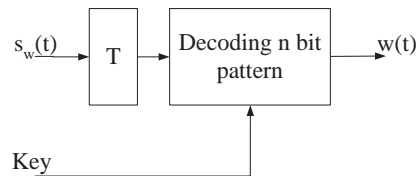


Figure 5: Watermark decoder

Aim of this quality evaluation is to measure the transparency of the watermarked audio tracks with regard to different NMR settings.

Parameters The NMR serves as the quality parameter¹⁰ throughout the quality evaluation. The determination of suitable parameter values will be done by performing a subjective listening test as a function of the NMR (for three different settings):

- -5dB (W_{Noise} 5 dB below the masking threshold)
- 0dB (W_{Noise} exactly on the masking threshold)
- 2dB (W_{Noise} 2 dB above the masking threshold)

Since the optimal NMR value may be dependent on the group of listeners and the test item, the test is performed for different items and evaluated with regard to different types of listeners. Therefore the audio tracks used in the evaluation of the quality are the second parameter varied during the test.

Test items used for this test are:

Harpichord Testing hard attacks (origin SQAM¹¹ material).

Marissa Testing pop music.

Speech Testing German language (origin SQAM¹¹ material).

Vivaldi Testing classical orchestral music.

All audio tracks are excerpts from the original tracks with an amplitude resolution of 16 bit and a sampling rate of 44.1 kHz. The duration of each excerpt is about 10s.

Test setup A test is performed on each combination of the items and *NMR* setting. A training phase precedes the actual test phase. During the training the test person listens to the original and the watermarked item until he or she believes to hear a difference. The whole listening test was performed under the following conditions:

- The test person is alone in the listening room.
- The test is performed by using a high quality equipment:
 - Headphone amplifier STAX Vacuum Tube Output Driver Unit
 - Headphone STAX Signature Series II
 - Sound Card, SEK'D RME-Audio Digi 96
 - DA/AD Converter SEK'D ADDA 2496 S
- No listening time limit during the training and test phase.
- Full knowledge which item is presented during the training phase.
- The listener is guided through the test by a web application, which is also used to record the results.
- The results are stored in a database for the statistical evaluation.

For the actual test of one setting (item, *NMR*), ten pairs are randomly chosen from the possible combinations $\{(O, O), (O, W), (W, O), (W, W)\}$, where "O" denotes the original and "W" the watermarked item. For each of these pairs, the subject is asked whether both items were equal or not. A correct decision about items being equal or different is called a "hit", so a subject produces a result of the form "*k* hits of a number of trials". During the test phase no limit is imposed on the number of repetitions to hear each of the individual items in the pair for comparison.

Three types of listeners were differentiated throughout the test:

Professional listeners E.g. sound engineers belong to this group.

Musicians Persons playing any music instrument belong to this group.

Normal listeners Persons without special knowledge in music.

Although the subjects are divided into these groups, the evaluation can still be done for all subjects (without consideration of the group). Since the evaluation is performed for a group of persons, the hits within this group are summed and taken as the test variable. The number of tries is the product of the number of persons within a group and the number of pairs for a combination of a specific item and *NMR* setting.

Test implementation To simplify the test, a web-based application was created. The application runs on a server and generates HTML documents that are sent to the browser of the test person and make up the graphical user environment. The main reason for the decision of a web-based application is the great number of potential test persons reachable by the world wide web. Since the volume of data being transmitted for such a test is fairly large, we decided to present two alternatives for the test:

- Online evaluation
- Offline evaluation

The online evaluation is intended to be used if the test person has access via a high bandwidth connection to the server. This is especially the case when performing a local test. During the online evaluation, the subject is interactively led through the test by the application, presenting the different items and collecting the results from the user.

If the subject does not have a fast connection to the server, it still can participate via the offline evaluation. Therefore all pairs are randomly generated and put on an audio CD that can be listened to by the subject. After all items are compared, the subject fills in a HTML form also contained on the CD and sends the results back to the server to be stored in the database. This reduces the data being transmitted between server and browser because no audio files need to be loaded from the server. Furthermore the same code is used for the pseudorandomly selection of the pairs which are used in the offline or online evaluation. The selection is stored in the database for the comparison with the test results.

5.1. Statistical considerations

In the following, an item is called *transparent* if no differences between the original and the watermarked version are audible. Otherwise it is called *non-transparent*. To prove the hypothesis of transparency statistical tests have to be implemented.

5.1.1. Test for non-transparency

A test for non-transparency is performed by trying to reject the transparency hypothesis. Therefore the following null and corresponding alternative hypotheses are formulated:

H_0 : Distortions are not perceivable.

H_1 : A subject can perceive distortions in a watermarked item.

What is tested is the ability to detect differences between the original and the watermarked audio track. Therefore the test variable is the number of hits k out of the number of pairs n . Under the hypothesis of H_0 the probability to get k hits out of n with detection probability $p = 0.5$ - because the subject is guessing - is:

$$P_1(k, n, p) = \binom{n}{k} p^k (1-p)^{n-k} = \binom{n}{k} 0.5^n \quad (9)$$

Therefore the distribution function of k is a binomial distribution $B(n, p)$. The critical region can be determined according to the following equation by choosing a level of significance α :

$$P(T \in \mathbf{B} \mid H_0) \leq \alpha \quad (10)$$

Choosing the level of significance $\alpha = 0.05$ and applying equation (10) for $n = 40$ pairs leads to a critical region of $B = \{26, \dots, 40\}$. I.e. if a subject has more then 25 hits, the error probability of **wrongly rejecting** the null hypothesis is 5%.

To ensure independent experiments the best approach would be to use as many subjects as possible each one testing only one pair. If the subjects are able to distinguish between the original and the watermarked version they will do it with a certain probability of detection $p > 0.5$. In general we have no knowledge about this detection probability. The parameter $p = 0.5$ corresponds to the null hypothesis H_0 whereas all values of $0.5 < p \leq 1$ correspond to the alternative hypothesis H_1 of non-transparency.

What can be calculated is the so-called operation characteristic (OC) function $\beta(p)$. $\beta(p)$ is a function of the detection probability p and reports the error of **wrongly accepting** H_0 if the alternative hypothesis H_1 is correct:

$$\beta(p) = \begin{cases} 0.95, & p = 0.5 \\ P_p(T \notin \mathbf{B} \mid H_1), & 0.5 < p \leq 1 \end{cases} \quad (11)$$

The adjustment of the OC-function $\beta(p) = P_p(T \notin \mathbf{B} \mid H_1)$ is done according to equation (11). By using the OC-function we have the full knowledge about the errors and therefore the quality of the hypothesis test.

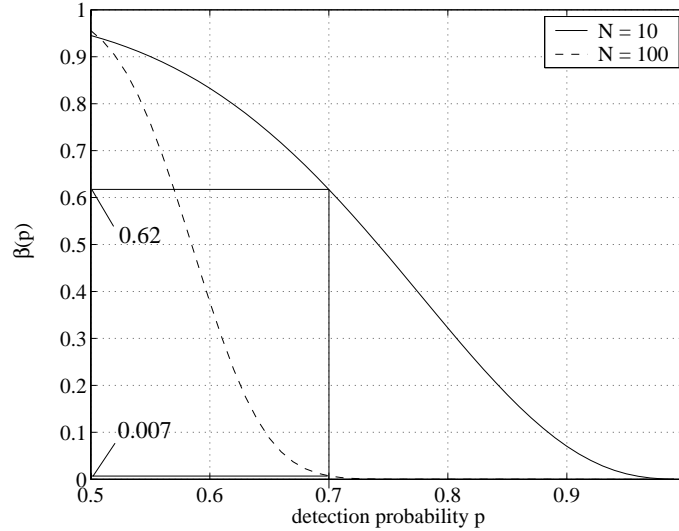


Figure 6: Quality function for hypothesis test

The quality of the test strongly depends on the number of experiments performed. For example if the subjects are able to distinguish with a detection probability of $p = 0.7$ the error probability is

$$\beta(p) \approx \begin{cases} 0.617, N = 10 \\ 0.007, N = 100 \end{cases} \quad (12)$$

The quality of the test increases with a higher number of tested pairs (see fig. 6). This can be used in order to design the test by calculating the number of pairs needed to ensure the two kind of errors incorporated in hypothesis testing if the detection probability of the subjects are known.

5.2. Test results

We performed an online evaluation with a total number of 20 persons. 16 persons belong to the group of *normal listeners*, four were *musicians*. No *professional listeners* participated in this test.

The critical regions for each group are shown in the following table:

Listener group	Critical region B
<i>Normal listeners</i>	$\{91, \dots, 160\}$
<i>Musicians</i>	$\{26, \dots, 40\}$
<i>Global</i>	$\{112, \dots, 200\}$

Two watermarks were simultaneously embedded to all audio tracks. The data rate for each watermark was $8bit/s$.

The results of both tests are included in the following table:

Item	<i>NMR</i>	Normal listeners		Musicians		Global	
<i>Harpsichord</i>	-5 dB	105	no	26	no	131	no
	0 dB	142	no	37	no	179	no
	2 dB	147	no	39	no	186	no
<i>Marissa</i>	-5 dB	84	yes	19	yes	103	yes
	0 dB	87	yes	19	yes	106	yes
	2 dB	81	yes	23	yes	104	yes
<i>Speech</i>	-5 dB	82	yes	16	yes	98	yes
	0 dB	92	no	29	no	121	no
	2 dB	123	no	30	no	153	no
<i>Vivaldi</i>	-5 dB	83	yes	18	yes	101	yes
	0 dB	75	yes	18	yes	93	yes
	2 dB	76	yes	22	yes	98	yes

Combinations of items and *NMR* settings are marked with either "yes" or "no" depending on whether they are considered to be transparent or not.

It is remarkable that the evaluation shows the same result for each group. The item *Harpsichord* cannot be considered transparent for any *NMR* setting. Also *Speech* is not transparent for the two worst quality settings. These two items are the critical ones. Both tracks have a simpler structure than the *Vivaldi* and *Marissa* tracks, which of course makes it more difficult to embed the watermark without introducing audible distortions. Both tracks are from the SQAM CD¹¹ and are considered critical in data reduction applications, which introduces distortions in form of quantization noise. For future tests and to enable the comparison of different audio watermarking algorithms we propose to use items from this CD in testing quality and robustness.

6. POSSIBLE ENHANCEMENTS

Improvements can be made by taking into account temporal masking effects to raise the power of the embedded watermarks and to use analyze windows with varying block length. To provide a better robustness one can use a different transformation T (Cepstrum, Wavelet etc.). Furthermore the pre-selection of the Fourier coefficients with regard to minimization of the variation coefficient¹ can improve the detection values. This adaptive approach can lead to lower embedding strength improving the quality of the watermarked audio tracks. In addition of the improvements of the algorithm the reliability of the subjective listening tests can be improved by the determination of the ability (detection probability p) of the subjects to distinguish between original and watermarked audio tracks.

7. CONCLUSION

This paper presents an intensive quality evaluation of watermarked audio tracks. The underlying statistical algorithm working in the Fourier domain was presented and the performance improvements with integrated PAM and the decoding procedure discussed. The test results showed transparency of the watermarked items in classical and pop pieces of music in all quality settings. Problems remain in pieces with a relativ simple structure like solo instruments and speech. However even in the best quality setting the robustness of the critical watermarked pieces is sufficient, offering the possibility to decrease the *NMR* in order to ensure the quality. This will require further investigations. Some possible enhancements to improve quality were discussed and are subject of current research.

8. ACKNOWLEDGMENTS

This work was supported by WEDELMUSIC project (<http://www.wedelmusic.org/>), a Research and Development project in the IST programme (IST-1999-10165) of the European Commission.

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