Combined Compression/Watermarking for Audio Signals

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ABSTRACT

Perceptual audio coding has become a customary technology for storage and transmission of audio signals. Audio watermarking enables the robust and imperceptible transmission of data within audio signals, thus allowing to attach valuable information to the content, such as song title, name of the composer and artist or property rights related data.
This paper describes a new concept for simultaneous low bitrate encoding and watermark embedding for audio signals. In particular, the advantages of this combined technique over separate steps of encoding and watermark embedding are discussed (i.e. encoding of watermarked PCM audio signals or watermarking of existing bitstreams). Experimental results obtained from a first implementation of an extended MPEG-2/4 AAC encoder are shown.

1 Introduction

Today, watermarking is a well-known technology for attaching property rights information or additional valuable data for the customer to audio material [1, 2]. Currently, there are a number of systems available to transmit a watermark in a hidden channel within uncompressed audio signals. However, music distribution over the Internet becomes a more and more important business and, therefore, most of the content is compressed using a perceptual audio encoder in order to save disk space while storing the data and to minimize transmission time. Embedding a watermark in the PCM domain and encoding afterwards is a common way. However, due to the quantization in the audio encoder the watermark information can be partly disturbed. On the other hand, there are methods to embed a watermark directly into the compressed audio material [3, 4]. The disadvantage of this solution is that in this case the watermark has to be treated as an additional disturbance to the readily included quantization noise.

This paper presents a method for embedding watermarks during the encoding process of the audio content that is favorable for both reasons of watermark robustness and signal quality. Firstly, a short overview over the state of the art in audio coding and some aspects regarding to watermarking are given. It is followed by the discussion about some more details on PCM watermarking, bitstream watermarking and - in contrast to those - combined audio encoding/watermarking. Finally, results gathered from a first implementation of an extended MPEG-2/4 AAC encoder are given.

2 State of the Art

A description of perceptual audio encoding can be found in many publications. Nevertheless, it is necessary to become familiar with the basic principles of audio encoding technology to understand the concepts of the proposed way of watermark embedding. Therefore, a short overview over audio encoding will be provided in this section. The requirements for a watermarking system are also explained.

2.1 Audio Encoding

Uncompressed high quality audio material demands high bandwidth for transmission or high storage capacity. However, above all, bandwidth is very expensive in most cases and, therefore, different data reduction algorithms for audio signals were developed. These algorithms can be classified into two groups: lossless and lossy algorithms.

Lossless algorithms make use of the redundancy of the signal source, i.e. only the information which is not readily known to the recipient is transmitted. The signal can be perfectly reconstructed by the recipient. In contrast, lossy algorithms rely on the fact that the recipient does not notice slight deviations of the signal. These algorithms attempt to remove both redundancy and irrelevancy of the signal before transmission.

Perceptual encoders are lossy encoders. They exploit knowledge of the human perception in order to shape the noise distribution introduced by the irrelevancy reduction in a way that the best possible audio quality is achieved. The basic structure of a perceptual encoder and the corresponding decoder as used in the family of MPEG audio encoding schemes is shown in Fig. 1. Subsequently, a short description of the components is given [5, 6].

- **Filterbank:** A time-frequency mapping (analysis filterbank) is used to decompose the input signal into subsampled spectral components. These spectral components are called subband values or frequency lines, dependent on the filterbank that is applied. The corresponding filterbank in the decoder forms the synthesis part of the system.
- **Perceptual Model:** By using either the output of the analysis filterbank or the input signal and an additional transformation, an estimate of the current masking threshold is computed. The masking threshold determines the maximum time and frequency dependent energy that can be added to the signal without any influence on the audio quality and is calculated using rules known from psychoacoustics [7]. If this estimate is more accurate better subjective audio quality can be achieved.
- **Quantization and Coding:** The spectral components are quantized and coded with the aim of keeping the quantization noise energy below the masking threshold while meeting the requirements on the bit consumption. Depending on the audio encoding scheme, this step is done in very different ways.
- **Bitstream Multiplexer:** The quantized and coded spectral components as well as some side information are assembled frame by frame, resulting in a bitstream which can be transmitted or saved.

The fundamental task of a perceptual audio encoding system is to compress the digital audio data in such a way that the highest possible compression rate is achieved while the sound quality of the reconstructed signal is

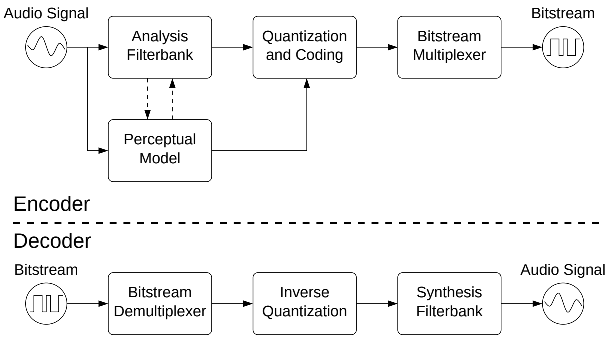

Figure 1: Basic structure of a perceptual encoding/decoding system.

exactly or as close as possible to the sound quality of the original audio signal. Other requirements include low complexity and flexibility for a wide application area. Among highly developed perceptual models and efficient redundancy reduction schemes, modern audio encoding systems offer a multitude of coding tools like Joint Stereo Coding [8] or TNS (Temporal Noise Shaping) [9] to fulfill these requirements.

2.2 Watermarking

The basic idea of watermarking is to provide a hidden channel that can be used in existing distribution channels. This channel offers the possibility to transmit user specific data.

Various schemes are available for embedding a watermark into some audio material, such as echo hiding [10, 11], direct modification of the time domain signal [12], narrow band systems [13] and the group of systems based on the spread spectrum technology [14, 15, 16]. The following terms are commonly used to classify and/or to describe the properties of the different watermarking schemes:

- **Inaudibility:** In most cases inaudibility, i.e. perceptual transparency of the watermark signal, is considered to be the most important issue in audio watermarking. In other words, the noise introduced by the watermark should not alter the audio quality noticeably. However, the maximum allowed disturbance should always relate to the target sound quality. If the target sound quality is very low an adequate watermark does not need to fulfill absolute inaudibility, while in application areas with very high target sound quality inaudibility must be ensured.
- **Robustness:** Often, the robustness of the watermark signal is also a very demanding aspect. It refers to the idea that the unintentional or intentional attempt to remove the watermark should only lead to success by accepting a clear degradation of the audio quality.
- **Data Rate:** The data rate specifies the number of bits per second that can be transmitted by the watermarking system. It depends on the underlying technology and the choice of parameters of the watermark scheme. Watermark systems using the spread spectrum technology typically offer bitrates between a few to a few hundred bits/s.
- **Operation Domain:** Both the input signal and the output signal of the watermark embedder can either be uncompressed or compressed. Accordingly, the term *PCM watermarking* is used to describe a system expecting and producing an uncompressed audio signal. In contrast, *bitstream watermarking* denotes a watermarking system working completely in the compressed domain. Finally, *combined compression/watermarking* characterizes systems with an uncompressed input signal and a compressed output signal. Depending on the application, either one of the system types may be required.
- **Interoperability:** In this context, interoperability is synonymous with the fact that the same watermark extractor may be used regardless whether the watermark was embedded using a PCM watermarking, a bitstream watermarking or a combined compression/watermarking system. In order to achieve this, the same watermark signal representation must be employed in each system.
- **Complexity:** Both, the maximum tolerated complexity of the watermark embedding process and the complexity of the extracting process depend on the application. This is due to the fact that various trade-offs exist between the watermark system complexity and other system properties, e.g. “complexity vs. audibility” in the watermark embedder or “complexity vs. reliability” in the watermark extractor.
- **Blind vs. Non-Blind Watermark Detection:** Most of the watermarking systems are currently capable of extracting the watermark without knowledge of the unwatermarked original signal. This is called blind, public or oblivious watermarking. However, the extraction performance can be greatly enhanced if the original audio signal is available since in this case the disturbing cover signal for the extraction can be eliminated. This is possible

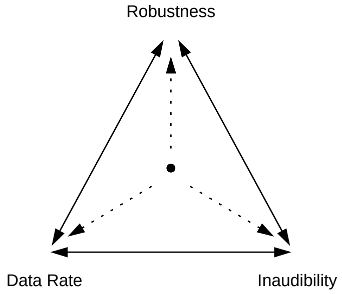

Figure 2: Trade-off between inaudibility, robustness and data rate.

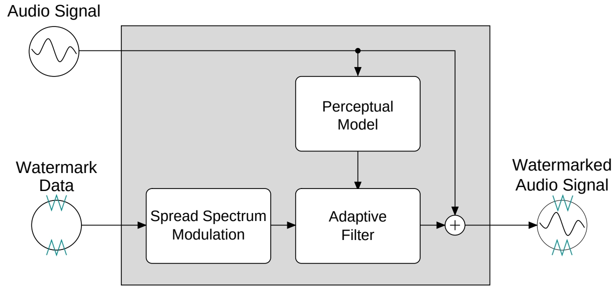

Figure 3: Block diagram of a PCM watermarking embedder.

in some watermarking systems, allowing non-blind watermarking extraction.

Most of the above mentioned characteristics are already determined by the system design itself or system inherent properties. However, inaudibility, robustness and data rate cannot be adjusted independently. For example, the audio quality of a watermarking system can be enhanced by lowering the robustness at the same time. The same applies for a high data rate system that may increase its rate by degrading the robustness. These trade-offs form a triangle shown in Fig. 2. For every different application an appropriate operation point within the limits of the triangle has to be chosen.

3 Spread Spectrum Watermarking

The spread spectrum modulation is a very popular way to embed a watermark into a carrier signal. Thus, some details of this technology are given [17]:

The basic concept of spread spectrum modulation systems is to add a pseudo noise signal with high bandwidth and low energy density to the carrier data. The pseudo noise sequence results from a multiplication of the data signal with the so-called spreading sequence. In the watermark extractor, the watermark can be detected using a matched filter with the filter coefficients matching the reversed spreading sequence applied in the embedder. In this context, the spreading sequence can be treated as a key. The extractor is not able to perform the fitting correlation without knowledge of the correct sequence and therefore the detection of the watermark will fail.

Due to the low energy density of the modulated watermark signal the distortion of the carrier signal can be kept at a low level. Additionally, this technology provides a high measure of robustness. The data rates provided by such watermarking systems are usually only moderate.

Earlier work described watermarking schemes for PCM and bitstream watermarking which are based on the spread spectrum modulation in detail [2, 3, 4]. A new way of embedding watermark information into audio material is then introduced: The combined audio encoding and watermarking. The scheme of this technology is presented and the differences to the two commonly used methods are characterized.

3.1 PCM Watermarking

The term PCM watermarking refers to the embedding of a watermark into uncompressed audio signals. A block diagram of such a scheme is shown in Fig. 3. The watermark data is expanded in bandwidth using the spread spectrum modulation. Typical bandwidths after spreading are 12 kHz to 18 kHz. At the same time, a perceptual model is applied to the audio signal resulting in an estimate of the masking threshold, i.e. the maximum amount of time and frequency-dependent noise energy that can be introduced into the original signal without decreasing the audio quality. The masking threshold is used by the adaptive filter in order to shape the spectral energy distribution of the spread spectrum data signal. After performing this operation, inaudibility of the watermark is ensured if the masking model is accurate enough. Otherwise, the audio quality can be improved by decreasing the overall watermark energy which, of course, leads to a degradation of the robustness as well.

3.2 Bitstream Watermarking

In contrast to PCM watermarking, bitstream watermarking operates in the compressed domain, i.e. the input signal as well as the output signal are encoded

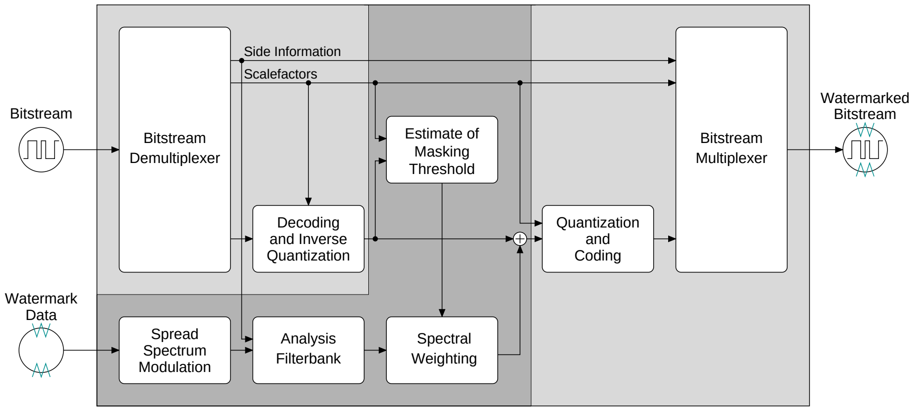

Figure 4: Block diagram of a bitstream watermarking embedder.

audio signals (bitstreams). In principle, embedding of watermarks into existing bitstreams can be achieved by subsequent audio decoding, PCM watermarking and re-encoding of the signal. However, doing so would cause both a decrease in sound quality of the audio material and robustness of the watermark. Furthermore, the computational complexity for processing the chain of the single applications would be rather high. Therefore, a reasonable bitstream watermarking scheme will operate as a "shortcut" and combine only the relevant operations of this sequence of steps.

The basic structure of such a scheme is shown in Fig. 4. In the decoder part of the scheme, the bitstream is parsed by the bitstream demultiplexer and divided into side information and the quantized and coded spectral values. In order to retrieve the spectral representation of the signal, a Huffman decoding and an inverse quantization process is applied by taking the side information into consideration.

The watermark data consists of the binary representation of the information to be embedded. Each single bit is spread in bandwidth by using the spread spectrum modulation. The watermark data signal is then converted into the spectral representation of the encoder by means of the same analysis filterbank as used for audio encoding alone. After spreading, the frequency spectrum of the watermark data signal is shaped by applying a time variant filter. The filter coefficients are chosen dependent on the masking threshold that is estimated with the information gathered in the decoder part.

The watermark data signal and the audio spectrum are added, quantized and coded in the encoder part. For the quantization, the retrieved scalefactors are used in order to avoid tandem coding effects. Finally, the output bitstream is generated by the bitstream multiplexer.

3.3 Combined Compression/Watermarking

The basic idea of the new watermarking system is the combination of audio encoding and watermark embedding in one step. Such a system can provide several benefits for both audio quality and robustness of the watermark compared to the conventional way of watermarking and encoding in separate steps. Afterwards, the basic structure of the combined compression/watermarking system is explained and its benefits are discussed.

Basic Structure

The combined compression/watermarking system consists of all relevant parts of a perceptual encoder (as described in Sec. 2.1) and the modules performing the necessary operations on the watermark. The basic structure for the system is shown in Fig. 5.

The uncompressed audio input signal is processed by the analysis filterbank with parameters gathered from both the signal itself and the perceptual model. The same analysis filterbank type and parameters are used to transform the watermark data signal that is obtained by spreading the watermark data using the spread spectrum modulation.

A time variant filter is applied in order to shape the spectrum of the watermark data signal. For that purpose, information from the perceptual model is evaluated and frequency dependent scaling factors are calculated. The algorithms for this calculation are chosen in such a way that both a minimum distortion of the audio quality and

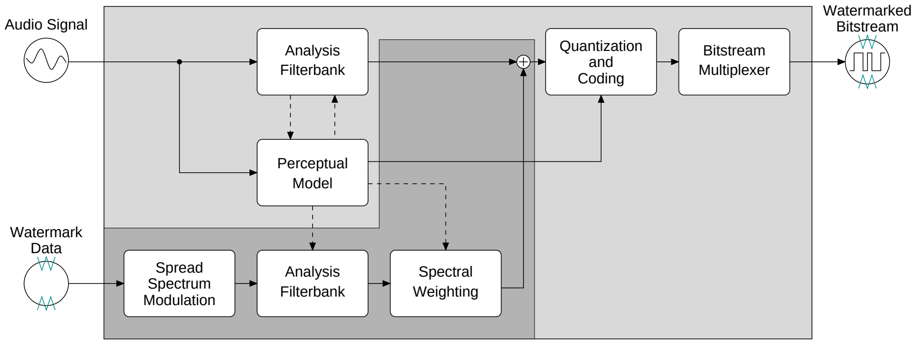

Figure 5: Block diagram of a system for combined compression/watermarking.

sufficient robustness of the watermark is achieved.

Since the audio signal and the watermark data signal have a compatible spectral representation, the spectral values of the two signals can be added on a line-by-line basis. The resulting spectrum is quantized and coded subsequently. Finally, the bitstream multiplexer produces a valid bitstream consisting of the quantized and coded spectral values and the side information.

Benefits

In general, there are three options to obtain encoded and watermarked audio material from an uncompressed source:

- Embedding a watermark into the uncompressed material using a PCM watermarking system and audio encoding afterwards.
- Encoding and embedding a watermark into the compressed material using a bitstream watermarking system.
- The above introduced combined compression/watermarking system.

For some kind of applications, the latter system can provide some benefits which are described here:

Combined compression/watermarking enables an optimal coordination between the quantization strategy of the encoder and the watermark embedding process. All information about the chosen quantization parameters can be taken into account by the watermark embedding process. Note that this is also true for the case of bitstream watermarking.

In contrast to bitstream watermark embedding there is only one quantization step which is a direct mapping of the combined "signal plus watermark" onto a quantized representation. By comparison, bitstream watermark embedding is constrained to work on previously quantized spectral coefficients. Thus, an already coarsely quantized value might be altered further by the embedding process. This can be avoided by the combined compression/watermarking which has the potential for delivering a very high audio quality at a given level of robustness.

Finally, combined compression/watermarking provides an efficient way of carrying out both processes in one step.

4 Results

In this section results of a first implementation of an extended MPEG-2/4 AAC [18, 19] encoder are presented that implements the capability of simultaneous watermark embedding. The audio quality of a set of watermarked critical MPEG test signals were assessed by subjective listening tests. The watermark bit error rate occurred while watermark extraction is shown for some average music material. Finally, the results of a test concerning the computational complexity are given.

4.1 Audio Quality

Fig. 6 shows the quality of both AAC watermarking schemes, i.e. the AAC bitstream watermarking system and the combined compression/watermarking system. Furthermore, the quality for MPEG-2 AAC encoded items as well as the quality for items processed by a chain of PCM watermarking and subsequent MPEG-2 AAC encoding are shown for comparison purposes. For all encoding steps a consumer grade MPEG-2 AAC en-

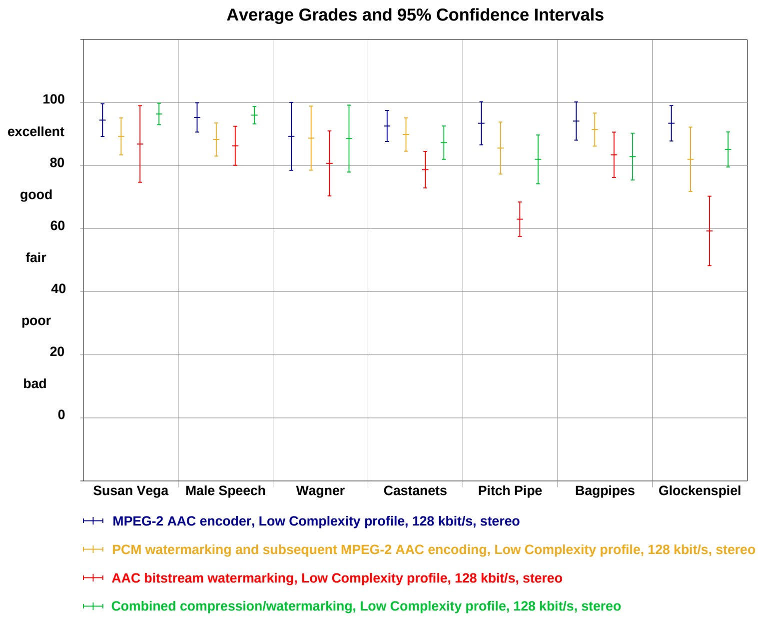

Figure 6: Listening test results of an MPEG-2 AAC encoder, a chain of a PCM watermarking embedder and a subsequent MPEG-2 AAC encoder, an AAC bitstream watermark embedder and a system for combined compression/watermarking.

coder was used. The test was carried out as a MUSHRA listening test [20].

The test items represent a variety of typical audio material and were used extensively for assessment of the subjective audio quality in the MPEG-4 audio development process before. The quality was evaluated by seven listeners, both experienced and familiar with the set of test items.

The figure shows mean values and 95% confidence intervals for these critical test items. The test results indicate that there is no statistically significant degradation between MPEG-2 AAC encoded items and the combined compression/watermarking scheme. The overall quality of the new scheme is comparable to the subjective sound quality of the MPEG-2 AAC encoded PCM watermarked items.

For some of the test items the audio quality of the AAC bitstream watermarking scheme is statistically significantly degraded. Since both of the AAC watermarking schemes are rather new implementations [21], an enhancement in the near future is expected. Please note that this implementation of the AAC bitstream watermarking scheme is not identical to the one presented in [3, 22], but is able to operate without transmission of helper information.

4.2 Watermark Bit Error Rate Measurement

This section presents the results of the watermark bit error rate (WBER) measurements. The watermark bit error rate is defined as the ratio of erroneous extracted watermark bits and the overall number of watermark bits. For this purpose a fixed watermark sequence is embedded into the test items. During the extraction process the retrieved watermark bits are compared with the known sequence on a bit-by-bit basis and the number of unequal bits is measured.

The audio material in this test represents a wide range of music genres. Altogether, approximately 1.4 million watermark bits were tested. The results of the watermark bit error rate measurements for the three watermarking options (PCM watermarking and subsequent MPEG-2 AAC encoding, AAC bitstream watermarking and combined compression/watermarking) are shown in Tab. 1.

The parameters for each watermarking option were adjusted to keep a good balance between audio quality and robustness. The resulting watermark bit error rates ($1.54 \cdot 10^{-2} \ldots 2.52 \cdot 10^{-2}$) are low enough for a multitude of applications. Moreover, it should be noted that no error correction algorithms were employed. Thus, the

Watermarking Option	WBER
PCM watermarking + MPEG 2 AAC encoding	$1.54 \cdot 10^{-2}$
AAC bitstream watermarking	$2.01 \cdot 10^{-2}$
Combined compression/watermarking	$2.52 \cdot 10^{-2}$

Table 1: Watermark bit error rates (WBER) for a chain of a PCM watermarking embedder and a subsequent MPEG-2 AAC encoder, an AAC bitstream watermark embedder and a system for combined compression/watermarking.

watermark bit error rate displays the raw channel bit error rate. Improvements can be achieved by the use of channel coding techniques.

4.3 Computational Complexity

To draw a statement about the computational complexity, the execution time of three different watermarking options was measured. Starting point for the test was an uncompressed test item (Antonín Dvořák, Largo, Symphonie No. 9 in E minor "From the new world": Two channels, 44.1 kHz sampling frequency) with a playing time of 709 seconds. The result after applying the watermarking options was an MPEG-2 AAC encoded and watermarked audio signal in all cases.

Fig. 7 illustrates and compares the implementation complexity of the different watermarking options. The y-axis is normalized to the duration of the test item, i.e. 1.0 corresponds to execution in real time. The results were obtained on a PC equipped with an Intel Pentium II with 400 MHz and 128 MByte of main memory. All file input and output operations were carried out using the local hard disk. The implementation of the single applications is based on source code containing both C and C++ modules.

Firstly, the execution time is given for processing by a chain of a PCM watermarking system and MPEG-2 AAC encoding afterwards. This operation is done in 77.8% of real time (PCM watermarking: 54.0% of real time; MPEG-2 AAC encoding: 23.8% of real time). The second entry in the figure gives the equivalent execution time needed by a system consisting of an MPEG-2 AAC encoder and subsequent AAC bitstream watermarking which is 54.3% of real time (AAC bitstream watermarking: 30.5% of real time). In contrast, the combined compression/watermarking performs in 44.6% of real time. Tab. 2 shows the execution time of each used application in seconds.

From these results it can be seen that all tested watermarking options perform faster than playing time of the item on the test equipment. The combined compression/watermarking system is even more than 2 times faster than real time and is, therefore, the method with the lowest computational complexity to encode and watermark uncompressed audio material.

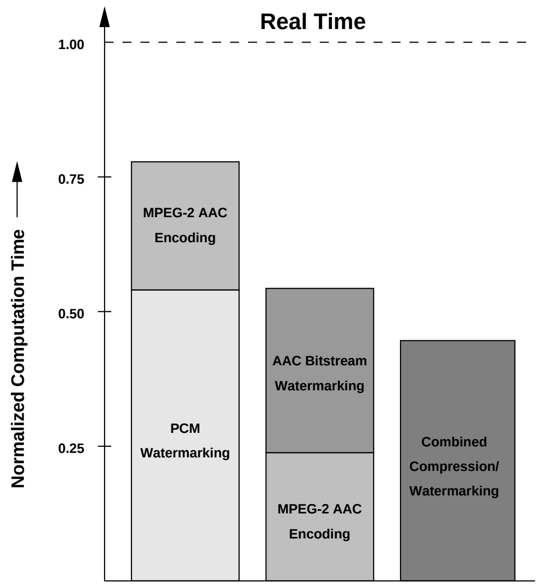

Figure 7: Comparison of the computational complexity of the three different watermarking options PCM watermarking and subsequent MPEG-2 AAC encoding, MPEG-2 AAC encoding and subsequent AAC bitstream watermarking and combined compression/watermarking.

However, as mentioned before, the system for combined compression/watermarking as well as the new AAC bitstream watermarking system are rather new implementations and not fully optimized yet. A further reduction of the computational complexity can be expected in future.

Application	Execution Time
MPEG-2 AAC encoder	168.68 s
PCM watermarking	382.80 s
AAC bitstream watermarking	216.43 s
Combined compression/watermarking	316.25 s

Table 2: Execution time of a MPEG-2 AAC encoder, PCM watermarking system, AAC bitstream watermarking system and a system for combined compression/watermarking processing a test item with a playing time of 709 seconds.

5 Conclusion

This paper presented a novel way of embedding imperceptible, robust watermarks into high quality audio signals. Combined compression/watermarking unifies the formerly separate steps of low bitrate encoding and watermarking into one efficient scheme. Beyond computational efficiency the combined approach has a number of distinct advantages over the traditional, two-step procedures by enabling optimum coordination between the two component processes. Initial results on subjective sound quality and robustness seem to confirm the potential of this new technology. The new scheme can provide an attractive solution in application scenarios where embedding of watermarks is used to convey the origin of the compressed content.

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