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Localized & self adaptive audio watermarking algorithm in the wavelet domain

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ABSTRACT

This paper presents an adaptive audio watermarking algorithm in the wavelet domain to optimize the payload under the perceptual transparency constraints of audio signal by strategically using some of its local features. Unlike existing algorithms, the watermark payload in this approach is made adaptive based on the nature of the audio signal. This localized feature based approach to determine the payload addresses the issue of over-loading and under-loading the audio signals with watermark data making the payload optimized for each individual audio host signal. Some audio features are strategically extracted and the most discriminatory features are selected using Principal Component analysis (PCA) approach. A mathematical model is designed using selected audio features like energy, zero cross mean and short time energy to evaluate the degree of embedding under perceptual transparency. It is used to estimate the number of watermarking bits to be inserted for a particular audio signal which makes the approach adaptive in nature optimizing the watermarking payload. At the embedding stage, watermark is embedded in the host audio signal in the third level detailed coefficient of wavelet domain which strikes a balance between the contradicting design parameters of perceptual transparency, robustness and optimized payload. Watermark extraction in this paper is blind with good robustness to signal processing attacks. Experimental results validate that the proposed adaptive algorithm provide good imperceptibility with good robustness against signal processing attacks at adjustable payload for different types of audio signals. Comparative analysis indicates that this proposed adaptive algorithm has better performance in terms of imperceptibility and robustness in comparison to uniform watermarking algorithm.

1. Introduction

Watermarking of audio signals have been in existence since last few years for secure transmission of audio signals over the network addressing copyright issues, piracy issues, ownership issues, fingerprinting and military applications etc. There are wide range of audio watermarking algorithms that are available in the existing literature for varied applications. Audio watermarking can be done in time domain, frequency domain, wavelet domain, cepstrum domain etc. Audio watermarking algorithms are basically divided into two categories: blind and non-blind. If original signal or watermark is required at the receiver side, then the watermarking algorithm is said to be non-blind, else it is blind algorithm. It is very challenging to design a blind audio watermarking algorithm because detection of efficient watermark at the receiver end, after application of various signal processing attacks is a crucial task. The embedding algorithm should be effective enough such that the original signal and the watermarked signal are perceptually similar. The watermarked signal must withstand serious signal processing attacks in order to extract the exact watermark with good accuracy. Imperceptibility, robustness and payload are the three contradictory design requirements of an audio watermarking algorithm [1]. In order to design an efficient audio watermarking algorithm, the three contradictory design parameters of the watermarking algorithm should be optimized. There are a large number of existing algorithms which provide a good watermarking solution at a high payload with good robustness against signal processing attacks maintaining perceptual constraints [2–6]. However, these algorithms may be suitable for some particular set of audio signals because property of audio signals is not taken into consideration while designing the embedding and extraction algorithm. In order to provide a solution to this problem, many researchers have proposed adaptive audio watermarking algorithm in which embedding strength of the algorithm is decided by using some of the audio features. The embedding strength here represents quantization pa-

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rameter or some alpha factor etc. Li et al. has presented an adaptive audio watermarking algorithm in wavelet domain based on SNR to determine watermark embedding intensity by using scaling parameter,α [7]. Pooyan et al. [8] proposed a robust method of audio watermarking in wavelet domain which determines the quantization and embedding strength adaptively according to the characteristics of human auditory system (HAS). Wang et al. presented an adaptive audio watermarking algorithm in discrete multi wavelet transformation based on the energy relationship of two consecutive frames in multi wavelet domain [9]. Dahui Li presented an adaptive audio watermarking algorithm in DCT and DWT domain. This algorithm also chooses the quantization step according to the masking properties of HAS [10]. Dutta et al. proposed an efficient watermarking algorithm which embeds watermark data adaptively in SVD and DWT domain. In [11] high energy peaks are used to evaluate degree of embedding for each audio frame. Youssef presented a novel hybrid fuzzy self-adaptive digital audio watermarking scheme (HFSA-A W) in DWT which is based on local audio features [12]. Fuzzy c-means clustering is used in [12] for segmentation of audio features related to rhythm, timbre and harmony to estimate the strength of a frame for each sub-band and to ensure that the embedded watermark in the original audio is self-adaptive. But the method is non-blind because it requires an original signal at the receiver end for watermark extraction. Peng et al. presented an adaptive blind audio watermarking algorithm in wavelet domain based on local audio feature and support vector regression (SVR) [13]. In their paper, frame energy and maximal peaks of its all sub-bands are extracted as local features and SVR is used to model the relationship between the local features and the embedding strength of the audio frame in order to adaptively control the embedding strength of the audio frame.

It can be seen from the state-of-art that there exists various adaptive algorithms addressing the issue of robustness and imperceptibility in audio watermarking. In these algorithms, watermark is made inaudible by using different embedding intensities in different audio frames that depends on some features of that particular audio frame. Robustness is another design parameter of audio watermarking which is optimized with good perceptual transparency. Watermark payload capacity of the signal is not being considered in these methods. Different audio signals may have different capacity of carrying watermark data. For example, if an audio signal has low energy or less noise but carrying more watermarking bits than its capacity for fixed payload watermarking algorithm, it will lead to overloading of audio signal with watermark data. The changes due to insertion of watermark may be audible to human ear or it may also affect the robustness of the algorithm. In other case, if a high energy audio signal carry less number of bits than its capacity, this will lead to under loading of audio signal with watermark data. In this case robustness or imperceptibility will not be disturbed or even may be improved. Thus, there is an unused room available which should be utilized for hiding watermarking bits. In order to overcome the issue of overloading and under loading of audio signal with watermark data, an algorithm is proposed in this paper by considering the features of audio signal such that payload capacity directly depends on the signal leading to an adaptive nature of watermarking algorithm. This paper presents an adaptive audio watermarking algorithm in wavelet domain with varied payload for different audio signals. This proposed algorithm can estimate the payload for the audio signal by using local audio features like energy, zero cross mean etc.

The main contribution of this paper is an adaptive audio watermarking algorithm that can estimate the watermark payload by using local audio features from the host audio signal. This will optimize the payload that to an acceptable limit for an audio signal under perceptual transparency constraints. Unlike various existing algorithms that involves uniform payload, this adaptive algorithm will help in resolving the issue of over-loading and under-loading the audio signal with the watermark data. This will ensure that the most optimized payload is employed for each host audio signal depending on its local properties which makes the algorithm adaptive in nature. To address the design requirements like imperceptibility, robustness and adaptive payload which are mutually contradictory, the watermarking method is strategically chosen based on a wavelet decomposition method.

The rest of the paper is organized as follows: Section 2 describes the challenges faced while designing an adaptive audio watermarking algorithm, detailed methodology of the proposed algorithm is given in Section 3, experimental results are discussed in Section 4 and finally Section 5 concludes the paper and proposes a future prospect.

2. Adaptive audio watermarking: challenges

In any watermarking method, payload may affect both imperceptibility and robustness of the algorithm. Generally, more payload has less robustness and lower imperceptibility, whereas watermarking algorithms with less payload rates can retain watermark even under signal processing attacks exhibiting good imperceptibility and robustness of the algorithm. In general, existing audio watermarking techniques have fixed watermarking payload for any type of signal without considering the nature of the signal. This can cause the problem of under-loading and over-loading the audio signal. So, due to different natures of the audio signals, the payload capacity should differ from signal to signal. Adjusting the payload or embedding intensity on the basis of audio signal for an algorithm is considered as an adaptive audio watermarking. In this paper, an adaptive audio watermarking algorithm is proposed in which payload is estimated from local features of the audio signal. It can avoid the issue of over-loading or under-loading of audio signal keeping the payload optimized under perceptual transparency conditions and good robustness toward signal processing attacks. In an adaptive audio watermarking algorithm wherein payload of an algorithm varies from signal to signal, many challenges have been faced. Some of the challenges faced while designing adaptive algorithm are listed as below:

1. The major challenge while designing an adaptive audio watermarking is the selection of audio features which may impact the payload of the signal such that number of bits to be embedded may depend on the localized audio features.
2. The design of mathematical model to evaluate the payload of the signal from its inherent features is considered as this will serve as root for the adaptive algorithm to decide the payload based on the local property of the audio signal.
3. The effective design of embedding algorithm is one mandatory requirement for adaptive audio watermarking such that variable payload can be embedded in different types of audio signals without affecting the perceptual transparency and the robustness.
4. Evaluating the payload of the watermarked signal at the receiver side is also a challenging task needing attention. The embedding is done in such a way that the features used for evaluating the payload should not be disturbed by hiding the watermark.

Therefore, there is a need to design an adaptive audio watermarking algorithm maintaining good robustness under perceptual constraints with varying payload for different types of audio signals.