WAVELET PACKETS BASED AUDIO WATERMARKING APPROACH
FOR DIGITAL APPLICATIONS

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Abstract

Audio watermarking is a technique that embeds information with specific meaning into the host media without perceptible interference to the original quality. As a complement to conventional encryption techniques, watermarking provides powerful tools for protecting the copyright of audio works and has become an active research area in recent years. In our proposed framework we present a novel watermarking procedure to embed for copyright protection and authentication into digital audio by directly changing the audio samples then after modifying the audio signals. The modified audio signals are divided into no. of samples each sample is decomposed adaptively by the method of novel Wavelet Packets decomposition. Finally in our proposed algorithm we show the robustness efficiency of the hidden watermark for additive noise re-quantization, MP3 compression, filtering, cropping and re-sampling. A comparison analysis shows that our proposed framework has high end performance than the other watermarking schemes proposed recently in the literature.

KEYWORDS: Audio watermarking, Wavelets packets additive noise re-quantization, MP3 compression, filtering, cropping and re-sampling.

1. INTRODUCTION

Over the last decade much work has been done in applying time frequency transforms to the problem of signal representation and classification. Signals possessing non-stationary character are not well suited for detection and classification by traditional Fourier methods. It has been shown that wavelets can approximate time varying non-stationary signals in a better way than the Fourier transform representing the signal on both time and frequency domains. Hence they can easily detect local features in a signal. Furthermore, wavelet decomposition allows analyzing a signal at different resolution levels. The discrete wavelet transform (DWT) provides a very efficient representation for a broad range of real-world signals. This property has been exploited to develop powerful signal de-noising and estimation methods and extremely low-bit-rate compression algorithms.

The discrete wavelet transform (DWT) is usually implemented using an octave-band tree structure. This is accomplished by dividing each sequence into a component containing its approximated version (low-frequency part) and a component with the residual details (high-frequency part) and then iterating this procedure at each stage only on the low-pass branch of the tree. The main drawback of the octave-band tree structure is that it does not provide a good approximation of the critical
sub band decomposition. An alternate means of analysis is sought, so that valuable time-frequency information is not lost. The Wavelet Packet Transform (WPT) is one such time frequency analysis tools. It is a transform that brings the signal into a domain that contains both time and frequency information (Wickerhauser, 1991). Thus, analysis of the signal can be done simultaneously in frequency and time. The most basic way to do time frequency analysis is by making FFT analysis in short windows. That has the drawback that the window needs to be short to find out fast changes in the signal and long to determine low frequency components.

The wavelet packet transform (WPT) offers a great deal of freedom in dealing with different types of transient signals. Indeed the development of the wavelet transform (WT) and wavelet packets has sparked considerable activity in signal representation and in transient and non stationary signal analysis. Wavelet packet decomposition (WPD) (sometimes known as just wavelet packets) is a wavelet transform where the signal is passed through more filters than the DWT. Wavelet packets are the particular linear combination of wavelets. They form bases which retain many of the orthogonality, smoothness, and localization properties of their parent wavelets. The coefficients in the linear combinations are computed by a recursive algorithm making each newly computed wavelet packet coefficient with the result that expansions in wavelet packet bases have low computational complexity. In the DWT, each level is calculated by passing the previous approximation coefficients through high and low pass filters. However, in the WPD, both the detail and approximation coefficients are decomposed. For n levels of decomposition the WPD produces different sets of coefficients (or nodes) as opposed to (n+1) sets for the DWT. However, due to the down sampling process the overall number of coefficients is still the same and there is no redundancy.

2. BACKGROUND

2.1 Discrete Wavelet Transform

The DWT, which is based on sub band coding, is found to yield a fast computation of Wavelet Transform. It is easy to implement and reduces the computation time and resources required. In continuous wavelet transform (CWT), the signals are analyzed using a set of basis functions which relate to each other by simple scaling and translation. In the case of DWT, a time scale representation of the digital signal is obtained using digital filtering techniques. The signal to be analyzed is passed through filters with different cutoff frequencies at different scales. In the discrete wavelet transform, a signal can be analyzed by passing it through an analysis filter bank followed by a decimation operation. When a signal passes through these filters, it is split into two bands. The low pass filter, which corresponds to an averaging operation, extracts the coarse information of the signal. The high pass filter, which corresponds to a differencing operation, extracts the detail information of the signal. The output of the filtering operations is then decimated by two. Filters are one of the most widely used signal processing functions. Wavelets can be realized by iteration of filters with rescaling. The DWT is computed by successive low pass and high pass filtering of the discrete time-domain signal as shown in Figure 1. This is called the Mallat algorithm or Mallat-tree decomposition.

At each decomposition level, the half band filters produce signals spanning only half the
frequency band. This doubles the frequency resolution as the uncertainty in frequency is reduced by half. In accordance with Nyquist’s rule if the original signal has a highest frequency of \( \omega \), which requires a sampling frequency of \( 2\omega \) radians, then it now has a highest frequency of \( \omega/2 \) radians. It can now be sampled at a frequency of \( \omega \) radians thus discarding half the samples with no loss of information. This decimation by 2 halves the time resolution as the entire signal is now represented by only half the number of samples. Thus, while the half band low pass filtering removes half of the frequencies and thus halves the resolution, the decimation by 2 doubles the scale. The filtering and decimation process is continued until the desired level is reached. The maximum number of levels depends on the length of the signal. The DWT of the original signal is then obtained by concatenating all the coefficients, approximation and details, starting from the last level of decomposition.

![Figure 1: Level 3 decomposition using wavelet transform](image)

2.2 Wavelet Packet Decomposition

The wavelet packet method is a generalization of wavelet decomposition that offers a richer range of possibilities for signal analysis and which allows the best matched analysis to a signal. It provides level by level transformation of a signal from the time domain into the frequency domain. It is calculated using a recursion of filter-decimation operations leading to the decrease in time resolution and increase in frequency resolution. The frequency bins, unlike in wavelet transform, are of equal width, since the WPT divides not only the low, but also the high frequency sub band. In wavelet analysis, a signal is split into an approximation and a detail coefficient. The approximation coefficient is then itself split into a second-level approximation coefficients and detail coefficients, and the process is repeated.

In wavelet packet analysis, the details as well as the approximations can be split. This yields more than different ways to encode the signal. When the WT is generalized to the WPT, not only can the low pass filter output be iterated through further filtering, but the high pass filter can be iterated as well. This ability to iterate the high pass filter outputs means that the WPT allows for more than one basis function (or wavelet packet) at a given scale, versus the WT which has one basis function at each scale other than the deepest level, where it has two. The set of wavelet packets collectively make up the complete family of possible bases, and many potential bases can be constructed from them. If only the low pass filter is iterated, the result is the wavelet basis. If all low pass and high pass filters are iterated, the complete tree basis results. The top level of the WPD tree is the time representation of the signal. As each level of the tree is traversed there is an increase in the tradeoff between time and frequency resolution. The bottom level of a fully decomposed tree is the frequency representation of the signal. Figure 2
shows the level 3 decomposition using wavelet packet transform.

Based on the above analysis, Figure 1 and Figure 2 give the comparison of a three-level wavelet decomposition and wavelet packet decomposition. It can be seen in Figure 1 that in wavelet analysis only the approximations (represented by capital A in the figure) at each resolution level are decomposed to yield approximation and detail information (represented by capital D in the figure) at a higher level. However, in the wavelet packet analysis [Figure 2], both the approximation and details at a certain level are further decomposed into the next level, which means the wavelet packet analysis can provide a more precise frequency resolution than the wavelet analysis.

**Figure 2: Level 3 decomposition using wavelet packet transform**

3. **PROPOSED METHOD**

3.1 **Audio Signals Embedded**

In the watermark embedding process Synchronization of code are efficiently combined with watermark bits from a obtained binary sequence Before embedding, then after it is denoted by \( n_j \in \{0, 1\} \) bit of watermark (Fig. 2). Basics of our watermark embedding are shown in Fig. 3 and detailed as follows:

**Steps to be embedded**

1. In step Divide the original host signals to the no. of samples
2. Each sample is decomposed into wavelet packet coefficients
3. Embed T times the binary sequence \( \{n_i\} \) into extrema

\[
 e'_j = \begin{cases} 
 \left\lfloor \frac{e_j}{H} \right\rfloor * H + \text{sgn} \left( \frac{3H}{4} \right) & \text{if } n_j = 1 \\
 \left\lfloor \frac{e_j}{H} \right\rfloor * H + \text{sgn} \left( \frac{H}{4} \right) & \text{if } n_j = 0 
\end{cases}
\]

Where \( e_j \) and \( e'_j \) are the extrema of \( IMF_z \) of the original host audio signal and the watermarked signal respectively. If “Sgn” function is equal to “[+]” then it is a maxima, and equal to “[-]” then it is a minimal. Denotes the floor function, and \( S \) denotes the embedding strength chosen to maintain the inaudibility constraint
4. Reconstruct the samples by using inverse EMD modified \( IMF_z \) and concatenate the watermarked frames to retrieve the watermarked signal
3.2 STEPS TO WATER MARK EXTRACTION

Host signal is splitted into samples and wavelet packet coefficients is performed on each one as in embedding for the watermark extraction. Extract binary data using rule given by (3). Then find out the SCs in the extracted data. This procedure is continuously repeated by shifting the selected segment one sample at a time until a SC is found. With the position of SC determined, after that we can extract the hidden data i.e. information bits, which given as

\[
e_j^* = \begin{cases} 
1 & \text{if } e_j^* - \left[ e_j^*/H \right] H \geq \text{sgn}(H/2) \\
0 & \text{if } e_j^* - \left[ e_j^*/H \right] H \leq \text{sgn}(H/2) 
\end{cases} 
\]

5. Set the start index of the extracted data, \( Y \), to \( I = 1 \) and select samples \( L = N1 \) (sliding window size).
6. Evaluate the similarity between the extracted segment \( V = Y(I:L) \) and \( U \) bit by bit. If the similarity value is \( \geq \tau \), then is taken as the SC and go to Step 8. Otherwise proceed to the next step.
7. Increase by 1 and slide the window to the next samples and repeat Step 6.
8. Evaluate the similarity between the second extracted segments \( V' = Y(I + N_1 + N_2 ; I + 2N_1 + N_2) \) and bit by bit.
9. \( I \leftarrow I + N_1 + N_2 \) of the new value is equal to sequence length of bits, go to Step 10 else repeat Step 7.

10. Extract the P watermarks and make comparison bit by bit between these marks, for correction, and finally extract the desired watermark. Watermarking embedding and extraction processes are summarized.

5. SIMULATION RESULTS

![Figure 5: Original Signal](image1)

![Figure 6: Watermarked Signal](image2)

![Figure 7: Watermarked image and Extracted image](image3)

6. CONCLUSION

In our proposed framework we present a novel watermarking procedure to embed for copyright protection and authentication into digital audio by directly changing the audio samples then after modifying the audio signals. The modified audio signals are divided into no. of samples each sample is decomposed adaptively by the method of novel Wavelet Packets
decomposition. Finally in our proposed algorithm we show the robustness efficiency of the hidden watermark for additive noise re-quantization, MP3 compression, filtering, cropping and re-sampling.

REFERENCES


