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Audio Water marking for Digital Image Protection using Source-Channel Coding

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Abstract: Digital watermarking is new procedure of delivering protection to multimedia knowledge and digital content material from unauthorized copying. Potent and imperceptible audio watermarking scheme utilizing Discrete Wavelet Transformation (DWT) is existed. So It is mighty information hiding manner for audio signal. Therefore, an appropriate design of channel code can protect the reference bits against tampering. In the present proposed method, the total watermark bit-budget is dedicated to three groups: 1) source encoder output bits; 2) channel code parity bits; and 3) check bits. In watermark embedding phase, the original image is source coded and the output bit stream is protected using appropriate channel encoder. By using this source coding and channel coding we will get the output information without loss and will get better PSNR values, bit error rate values in proposed method comparing to the existing method. By using this source coding we can calculate the maximum and minimum pixel values and using channel coding for compression of the image.

Keywords: watermarking, data embedding, self recovery, source, channel coding audio signal.

I. INTRODUCTION

Digital watermarking was once introduced to provide security of understanding in opposition to intellectual piracy. Now a day's protection is essential challenge as Computer efficiency, use of internet is increasing day by day. Photograph, audio, video watermarking are the forms of digital watermarking centred on multimedia information. Audio watermarking process is extra complicated than snapshot and video watermark manner [2][3]. Human auditory procedure is extra complex and sensitive than human visual procedure as human ear is ready of detecting frequency exchange in audio sign. Embedded understanding is significant as measurement and period of audio sign is shorter than snapshot and video file which degrade the first-class of audio signal.

Audio watermarking procedure grouped into two types: time domain methods and frequency domain procedures [1]. In DWT (discrete wavelet develop into) audio watermarking manner, watermark is embedded into decomposed audio sign. Embedded watermark in audio signal bear many signal processing operations like linear filtering, compression, quantization. It does no longer have an effect on the fine of sign but corrupt the watermark know-how. Embedded watermark need to be survived and mighty in contrast various malicious assaults and sign processing operations. Watermark resistance against removing and degradation method robustness. In line with IFPI (international Federation of the Phonographic enterprise), audio watermarking procedure should meet the specifications like robustness, potential, perceptual quality, security. Robustness is the capacity of watermark to survive towards special signal processing operation and malicious assaults. Potential of audio watermarking algorithm approach without Degrading the nice of audio signal, algorithm must be capable to carry more expertise. Embedded watermark will have to no longer produce audible distortion to common audio is referred to as imperceptibility. Security approach most effective licensed person should realize the watermark or able to make some alterations in it. Time domain systems of audio watermarking incorporate LSB (Least large Bit), unfold spectrum, segment coding and echo hiding. Time area tactics are effortless to put into effect than frequency domain procedures however it is much less mighty towards malicious attacks. Time domain strategies of audio watermarking are unable to present detail expertise about all frequencies of audio signal. It requires less quantity of resources. LSB (Least tremendous Bit), spread spectrum, phase coding and echo hiding are the time domain tactics of audio watermarking. In LSB know-how embedded in least massive bit without problems via overwriting the usual bits. Echo hiding watermark technique embeds understanding into discrete audio signal.

Human perceptual homes are employed in frequency domain procedures of audio watermarking. Segment and amplitude of the develop into domain coefficients are modified to embed understanding in these strategies. DCT (Discrete Cosine become), DFT (Discrete Fourier grow to be), DWT (Discrete wavelet become), FFT (quick Fourier turn into) these are the standard become systems. DWT preferred over other develop into for audio watermarking considering the fact that it gives time and frequency illustration of sign This paper deals with implementation of effective audio watermarking system to prevent copyright infringement of long-established audio file.

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Algorithm is based on Discrete Wavelet grow to be. Proposed work goals at powerful implementation of audio watermarking that possess capability of watermark to live on towards different attacks scaling, re-sampling, low move filtering, re-quantization, random cropping.

II. EXISTING METHOD

DISCRETE WAVELET TRANSFORM

Discrete wavelet grow to be is used in DSP purposes. It symbolizes sign in each time and frequency domain.DWT decomposition of signal. It separate high move and low go component of sign. When sign cross by means of the high go filter it gives detail coefficient and low cross filter offers approximate coefficient [6]. Approximate coefficients are much less prone to noise as they are low pass coefficient. Thus understanding embedded into approximate coefficients. More stage of decomposition of DWT approach more correct illustration of sign in time and frequency. Each and every stage is referred to as an octave. High pass component offers details of signal at the same time low pass produces mean. The Haar Wavelet turn out to be is the simplest of all wavelet become. It is the normal orthogonal wavelet filter. In inverse DWT procedure both approximates coefficient and element coefficient up-sampled then passes through the low pass and high move filter. Through convolving the samples of low go and excessive cross filter, reconstructed sign is obtained.

III. PROPOSED METHOD

To overcome the drawbacks of existing technique, a digital audio watermarking with source-channel coding is proposed.

It consists of '2' modules

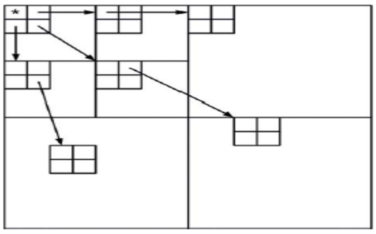
1. Embedding
2. Extraction

In embedding process information is added into decomposed audio signal. While extraction is used to retrieve that information without distortion decompose audio signal.

Source Coding-Self Recovery

Self recovery encoding is utilized as supply encoder in the proposed procedure. SPIHT is an embedded compression algorithm, that is, it is easy to truncate its output bit stream on the favoured fee and come to a targeted reconstruction of the common snapshot. The more output cost exploited, the better pleasant of reconstruction is workable. To meet this goal, the algorithm sorts the rounded multi-resolution wavelet grow to be coefficients

Examples of root-leaves dependencies within the spatial-orientations of Image pyramid decomposition. According to their magnitudes and transmits them established on giant bit order. The sorting order ought to be on hand to the decoder as well.



A cosmopolitan sorting approach is required to scale down the "bit-price range" used for sorting move. To do so, self healing exploits the self-similarities across different sub bands of wavelet turn out to be.

Beside the low computational complexity, the truth that Self recuperation is an embedded compression algorithm with adaptive output fee makes it suitable for our utility where we could must take advantage of distinct compression rates to satisfy specific purposes. Our algorithm truncates the Self recovery output on the price of ns bits per pixel. Channel coding is applied to source encoder output bit flow to preserve it against tampering. As a consequence, the highest conceivable Height signal to noise ratio (PSNR) of our reconstruction algorithm happens when channel code has labored flawlessly and retrieved all source encoded bits, and equals the PSNR of Self restoration for fashioned picture at the cost of ns. For illustration, Self recovery offers the PSNR of 83.2435 dB for the Cameraman image when compressed at the expense of 1 bpp. As a final result, if we set ns = 1 in our algorithm, no PSNR recovery of extra than eighty three.2435 dB is doable.

Channel Coding- Self Recovery

Some source encoder output bits possibly misplaced if now not Safeguarded considering of snapshot tampering; accordingly, the supply encoder end result ought to be blanketed by way of some channel codes. Besides, tampered blocks will be famous utilizing determine bits. It is noteworthy that their knowledge is on hand to channel decoder. Due to the fact this source-channel code design and having error areas to be had, tampering can also be modeled and handled as an erasure error, the place the locations of error are recognized to decoder.

Accordingly, an erasure decoder which uses the channel coded information in preserved blocks and the locations of erasure have got to be applied for the motive of image healing. However, when a block is well-known as tampered, all its bc embedded channel code bits are assumed to be erased. SR (Self recovery) codes with tremendous code words over huge fields tackle this task conveniently. In case of the use of SR (self restoration) codes with tremendous code words, several bits are congregated to one image, leading to restricted quantity of code words affected by image tampering.

$$m = \min_{\hat{m}}(\hat{m}|2^t - 1 \quad \text{and} \quad \hat{m} > n).$$

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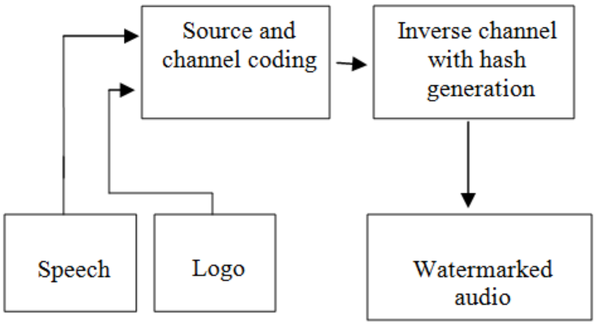
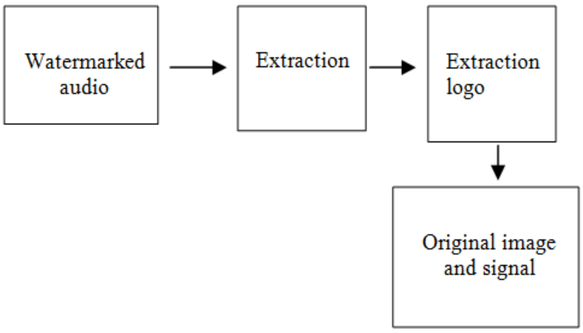
Embedded process:



In source and channel coding process contain inputs audio and image. In source coding SPIHT algorithm is used for image compression. In channel coding DCT techniques are used for maximum to minimum pixel after encoding process this data is given to input of inverse channel coding with has generation Hash information which was derived from the image blocks in are now extracted from the speech frames finally watermarked audio is generated

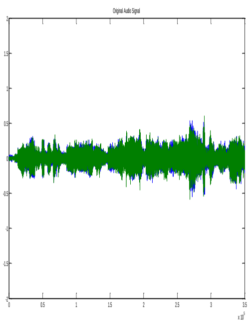
Extraction process:



In extraction process water marked audio is given to input of extraction process module. In inverse source and channel coding techniques are adopted. Finally speech and image is extracted from water marked audio. To get back original image

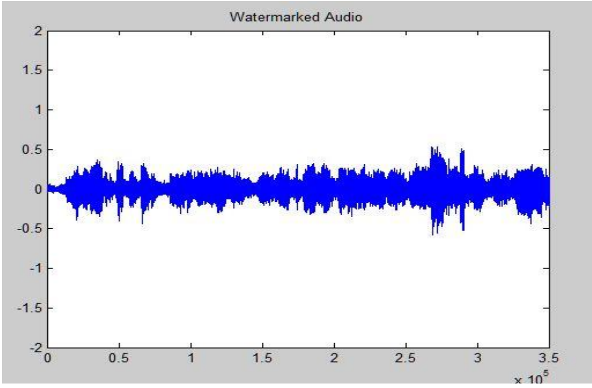
Experimental Results:

Embedded process



(a) **Original audio**

(b)) watermark logo



(c) Watermarked audio

Fig a shows the audio signal contains frequency (0-0.35MHz). Here pixel rate is decreases using DCT techniques. Fig b shows input image having lower pixel rate. Fig c shows embedded audio and image data

Extraction process:

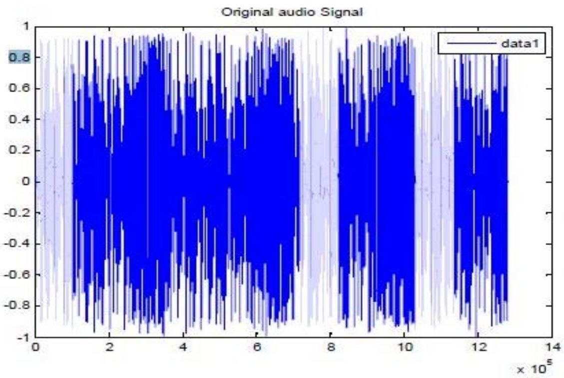
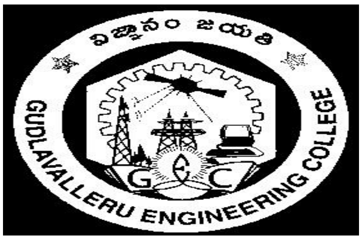


Fig: 2. Original audio signal

(d) Watermarked audio



(e) Original logo

Fig d shows extraction process of water marked audio data. This data is input to extraction block finally we get original image from output of block of extraction process. Fig e shows final original image.

Results:

Existing method (using DWT):

Embedded process	Recover process
Image size:12.00KB	Image size:1.42KB
Audio size: 1.68MB	Audio size: 1.28MB

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Proposed method (using source channel coding):

Embedded process	Recover process
Image size:12.00KB	Image size:1.85KB
Audio size: 1.68MB	Audio size: 1.40MB

Evaluation Parameters:
Existing method:
PSNR:-48.1308
MSE:-1
Proposed method:
PSNR:-83.2435
MSE:-1.3722e+13

CONCLUSION

This paper proposes a efficient data hiding procedure for audio watermarking situated on Discrete Wavelet turn into is proposed. As in comparison with time area technique grow to be area system for audio watermarking shows more robustness. Arnold converted is used to furnish more security for watermark in transmission. DWT is effective algorithm. Using DWT audio signal without distortion is obtained after extraction of watermark. SNR, PSNR are used as evaluation parameter which verify robustness of algorithm. SNR have got to be more than 20 dB which fulfil IFPI necessities. DWT audio watermarking technique is priceless in copyright defence and authenticity verification.

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